Electrical signals

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What is a signal?

We have seen ”electricity” in terms of voltages and currents.

What do we mean when we say (electrical) signal?

From the Merriam-Webster dictionary:

> **signal … 4 c:** *A detectable physical quantity or impulse (as a voltage, current, or magnetic field strength) by which messages or information can be transmitted.*

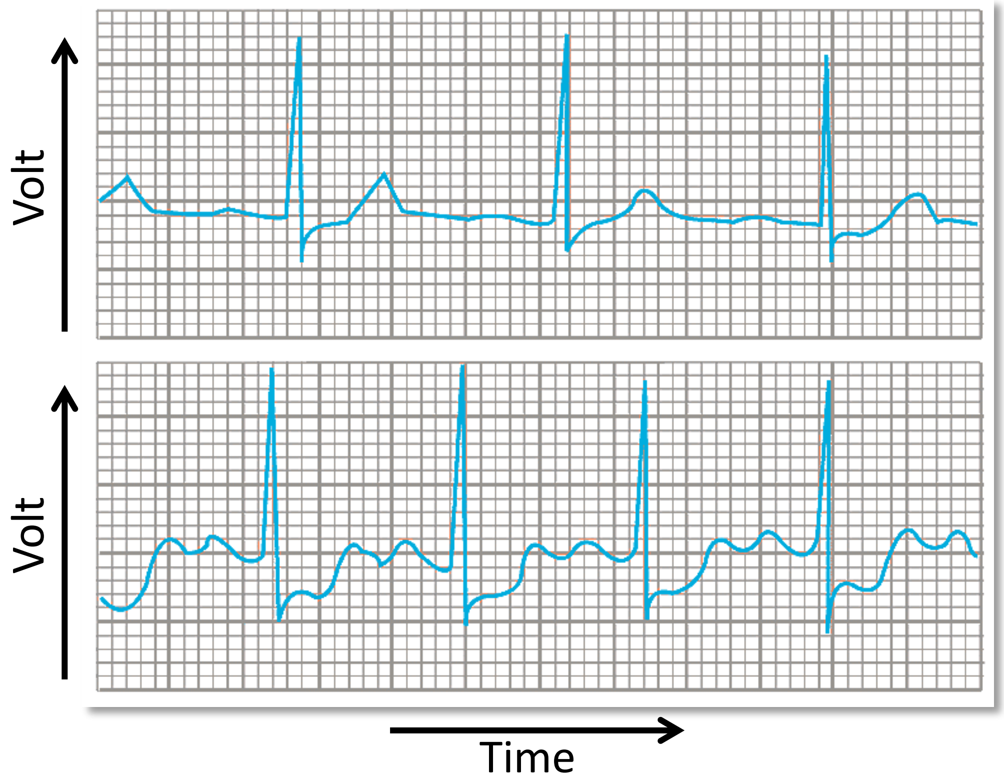
The news here, as far as we are concerned, is that a signal is something that **can be used to transmit information**.

What is this?

Signal examples [1]

Electrocardiogram (ECG)

A measurement of the electrical activity of the heart.



Patient at rest.

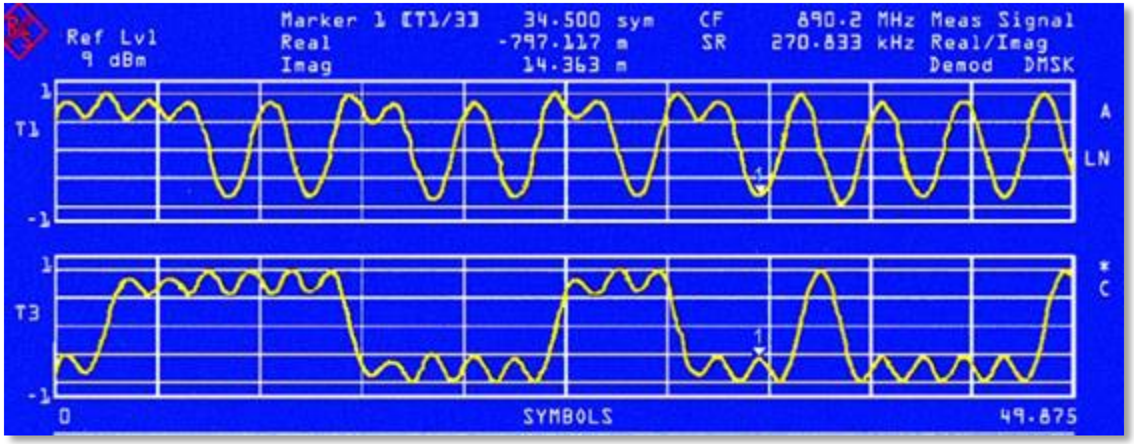
Does it contain any information and about what?

After 4 ½ minutes of exercise.

Signal examples [2]

GSM radio signal

The electrical signal received at a GSM antenna.

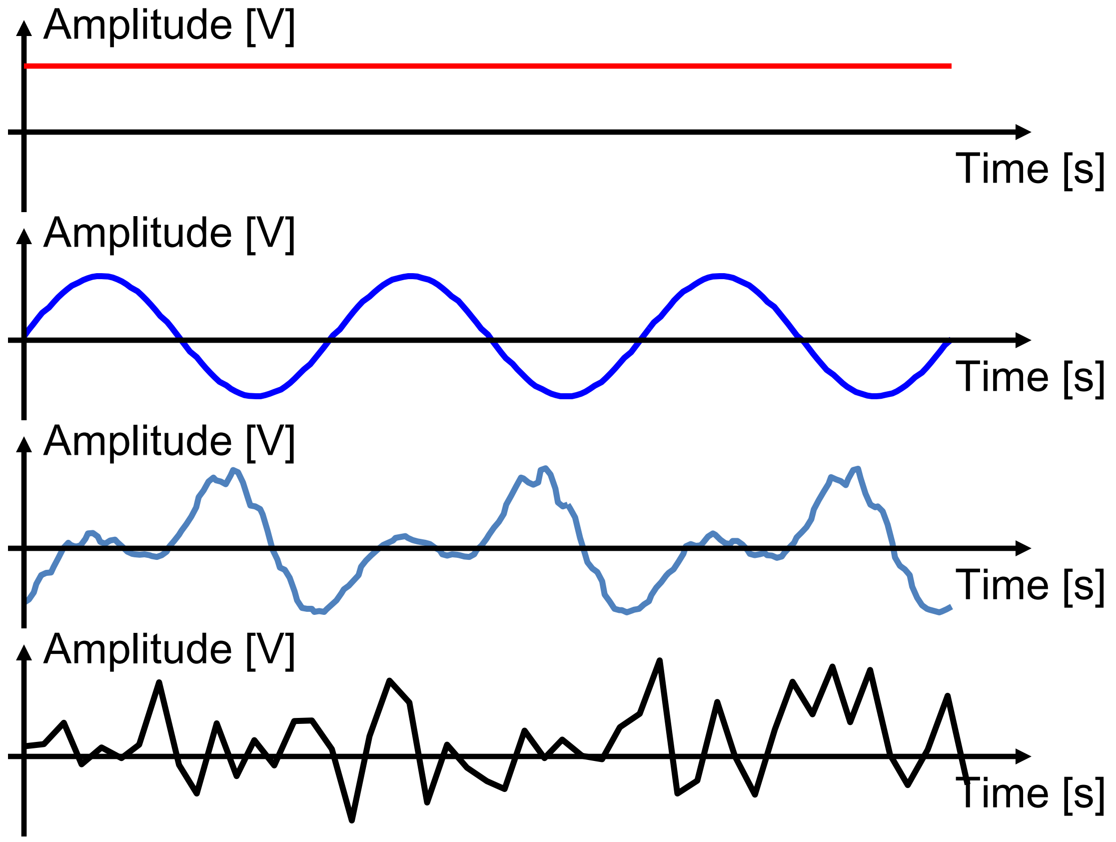


Does it contain any information and about what?

Some ”amplitude” properties

Time varying amplitude

An electrical signal typically has a time-varying amplitude.



No amplitude changes.

A smooth and **periodic** amplitude change (sinusoid).

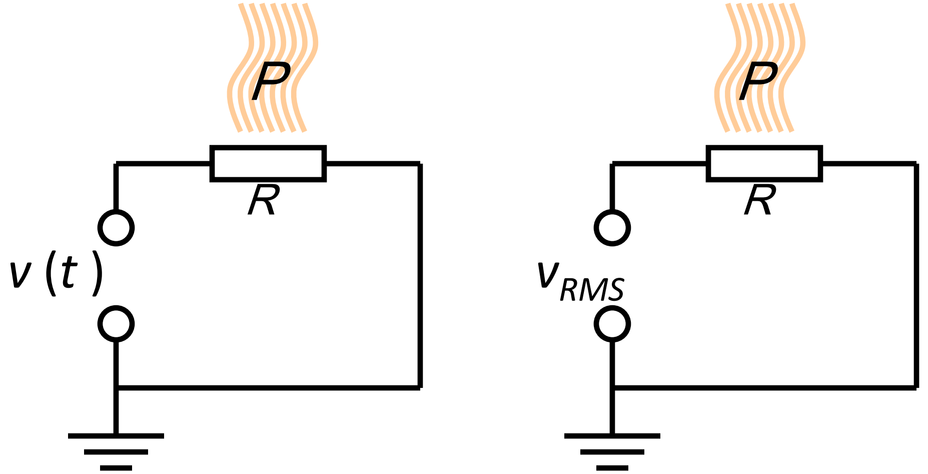
A somewhat erratic change of amplitude, but with a certain **periodic** structure.

No apparent structure at all.

”Average” amplitude

In many applications it is useful to know some sort of ”average” amplitude. A normal average is not very useful though.

A good measure on the average is a constant amplitude that would generate the same average power as the original signal if they were both connected to equal resistors.

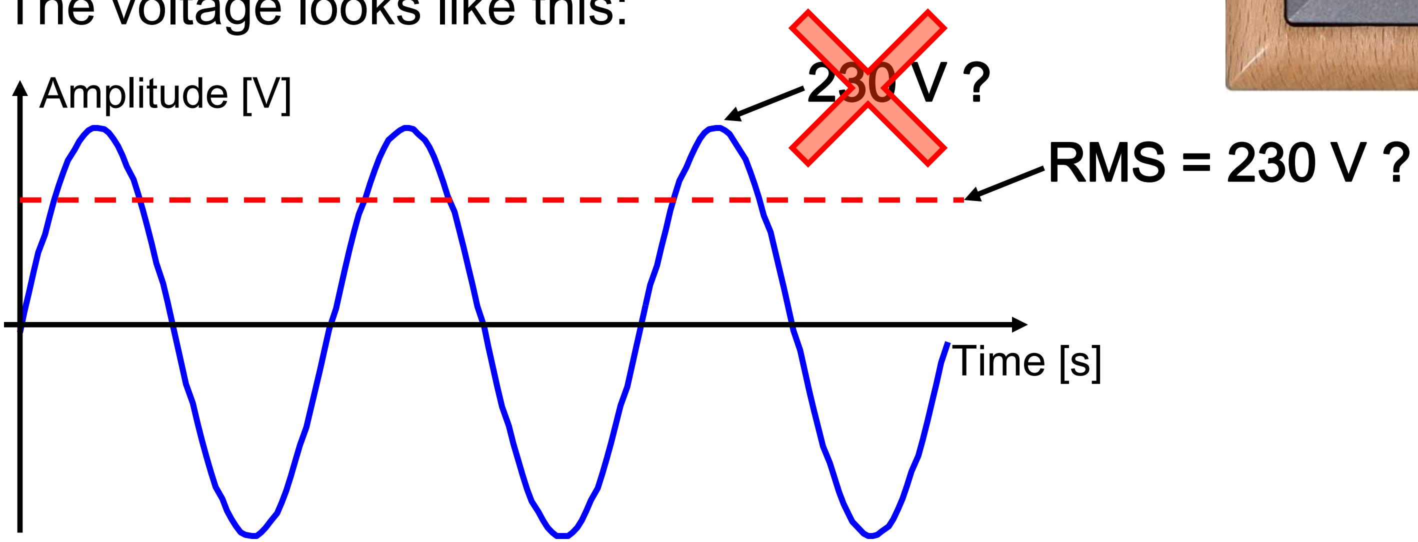


The **root mean-squared** (RMS) amplitude: $v_{RMS} = \sqrt{\frac{1}{T}\int_T v^2(t)\,dt}$

An RMS amplitude example

In the wall socket at home, you have 230 V.
What does that mean?

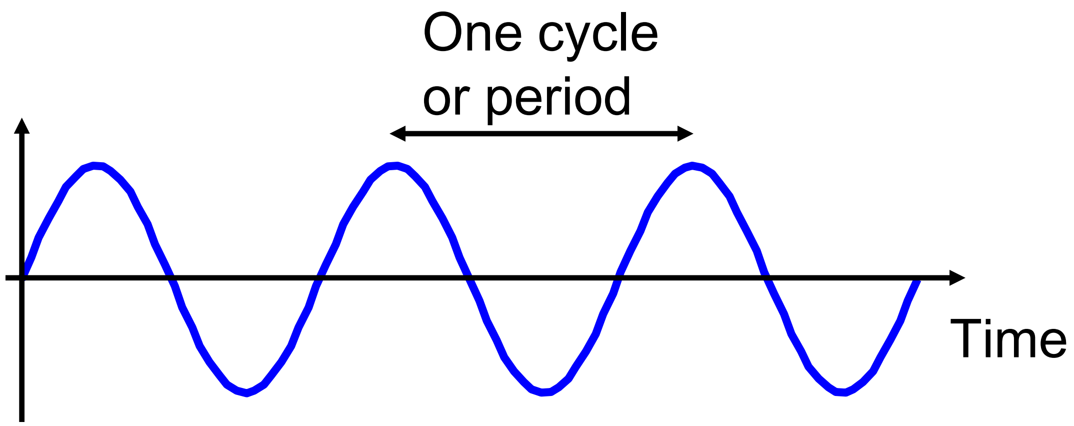
The voltage looks like this:



Frequency and bandwidth

Frequency

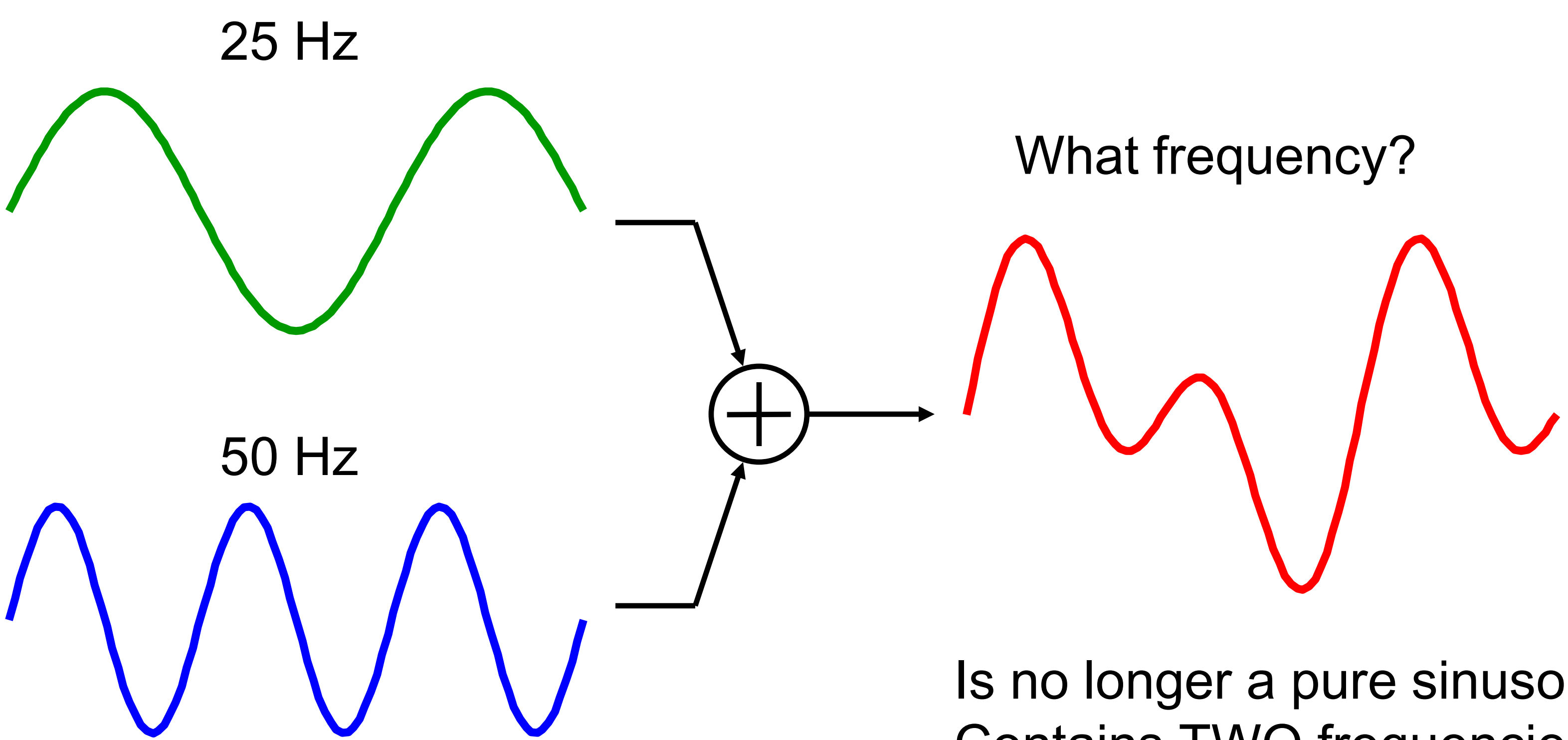
Sinusoidal signals:



Frequency = Number of cycles per second [Herz]

Example: The AC power in your home has a frequency of 50 Hertz.
This also means that the cycle time is 20 ms.

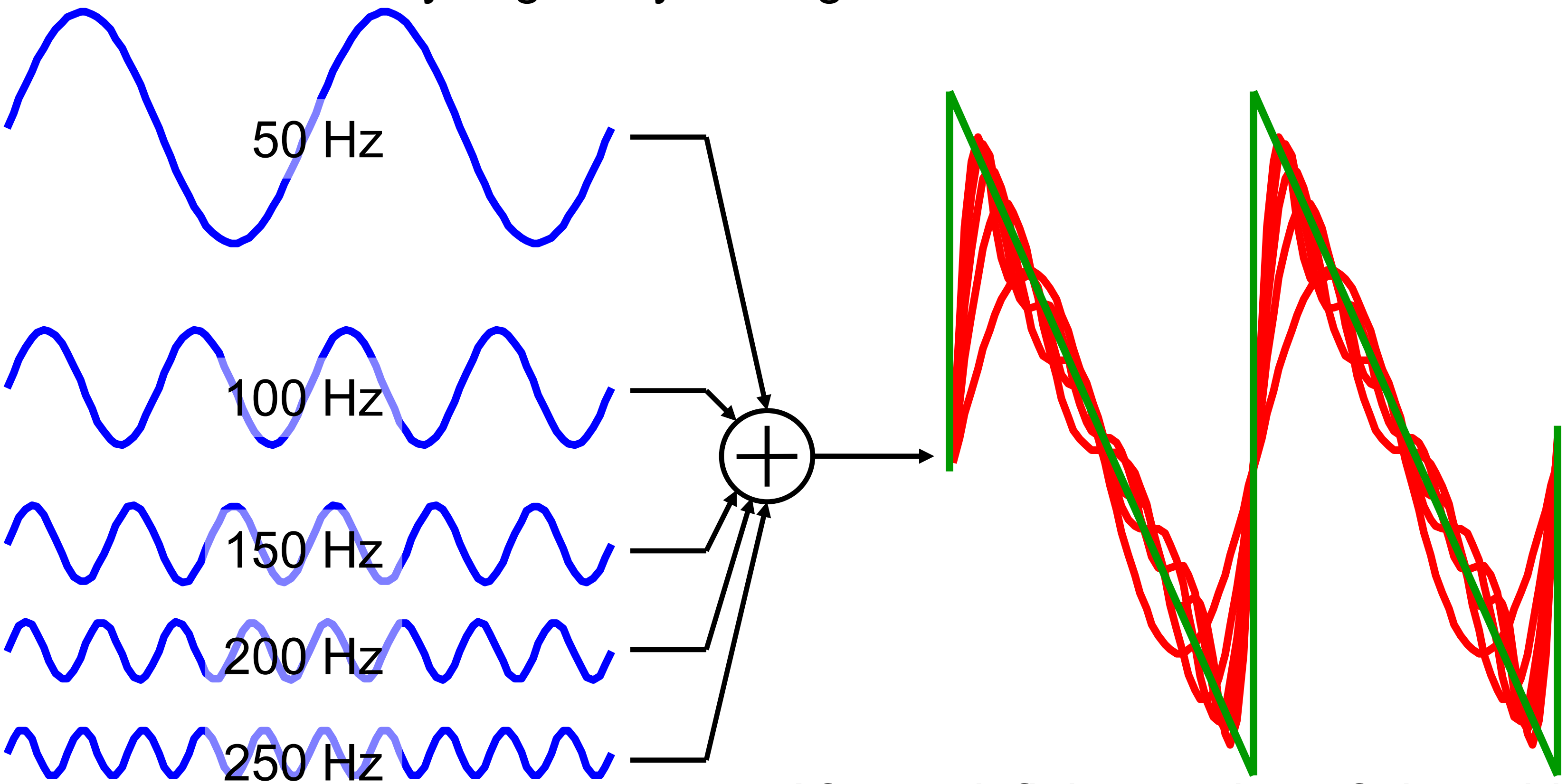
Adding sinusoids [1]



Is no longer a pure sinusoid.
Contains TWO frequencies.

Adding sinusoids [2]

Can we build "any" signal by adding sinusoids? Yes!

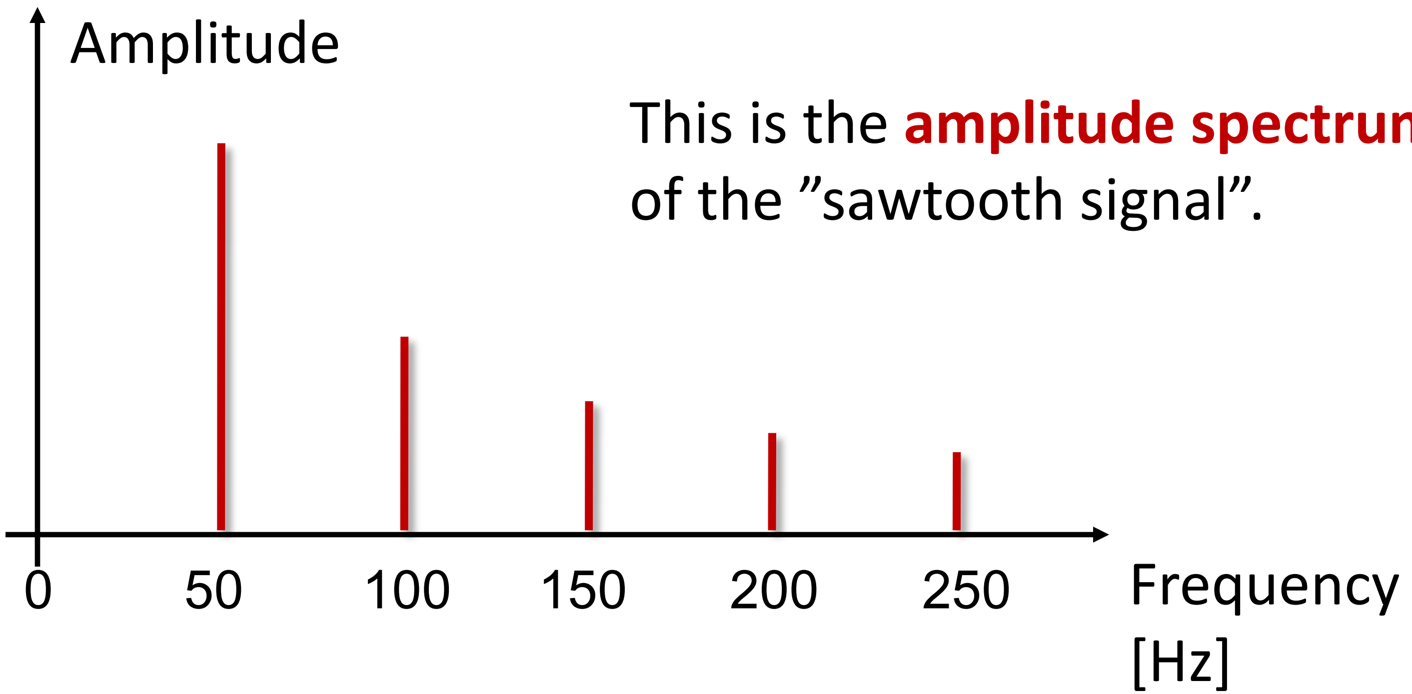


After an infinite number of sinusoids we get a sawtooth signal!

Spectrum

Spectrum [1]

If we can build any signal by adding sinusoids ... can we view the frequency content of a signal in some way?

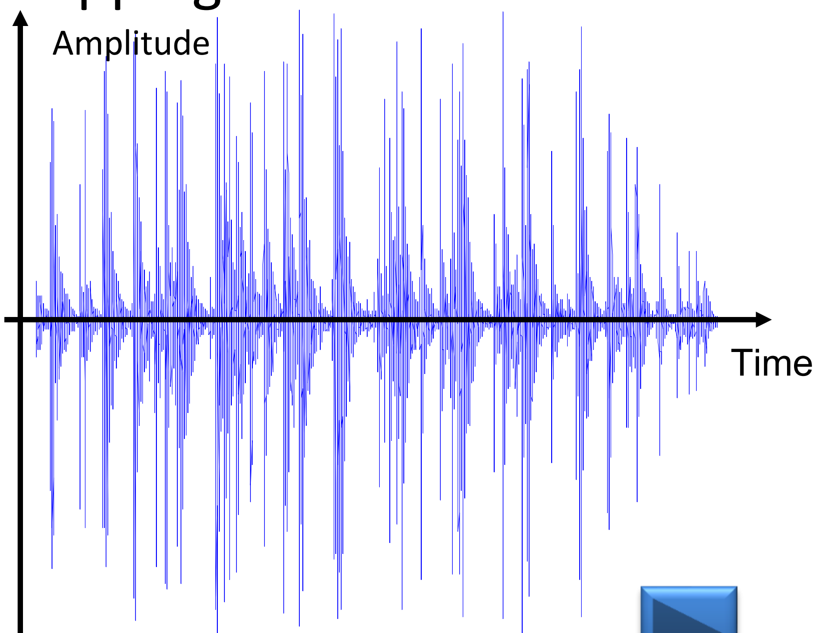


This is the **amplitude spectrum** of the ”sawtooth signal”.

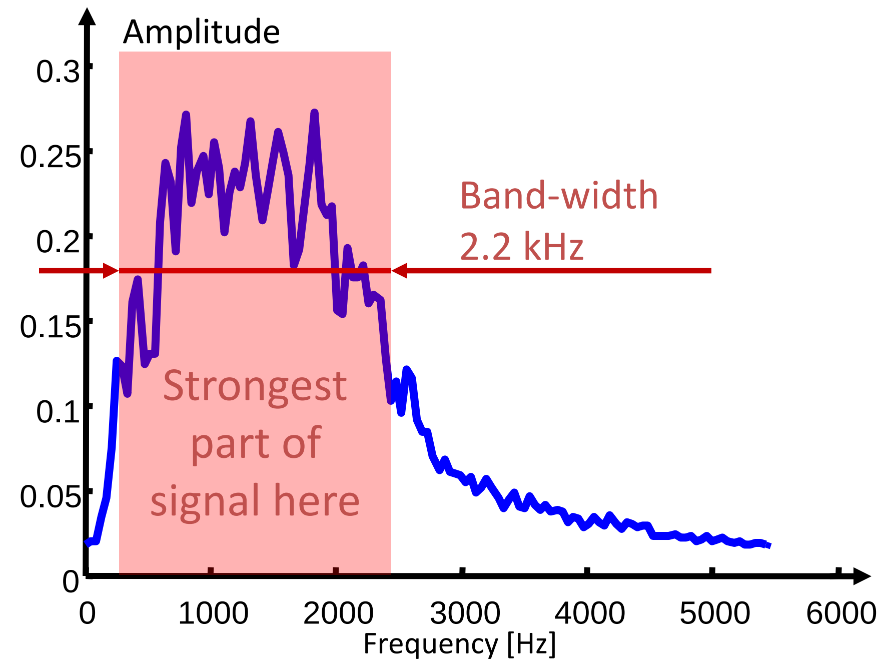
Spectrum [2]

How about the spectrum of ”any” signal?

The sound of an audience clapping its hands:



... and the corresponding amplitude spectrum.



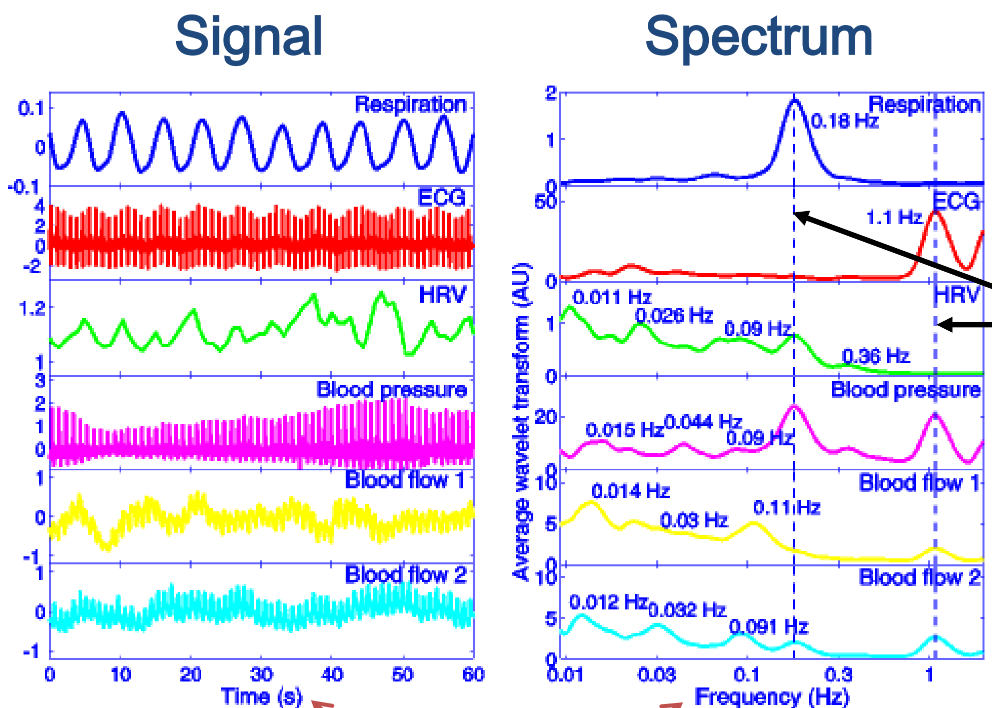
Spectrum [3]

Some typical frequencies and band-widths:

	Center frequency	Bandwidth
Speech:	0 (base band)	3 kHz (telephone)
Music:	0 (base band)	20 kHz
2G GSM radio signal:	900 or 1800 MHz	200 kHz
4G LTE radio signal:	450 MHz – 3.8 GHz	1.4-100 MHz

Spectrum examples [1]

Below is an example of multiple signals from a patient, with their respective spectra to the right.



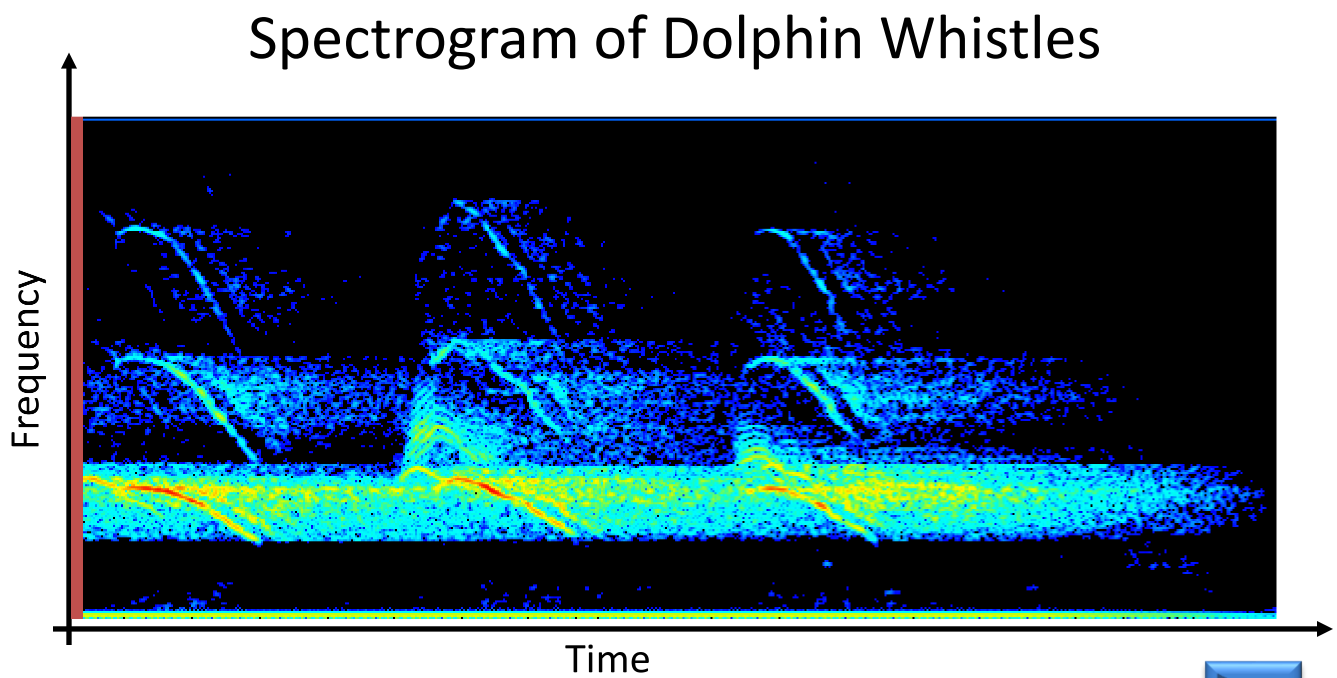
Observation:
There are strong 0.18 Hz and 1.1 Hz signal components present in several of the signals.

Any idea why?

Note!

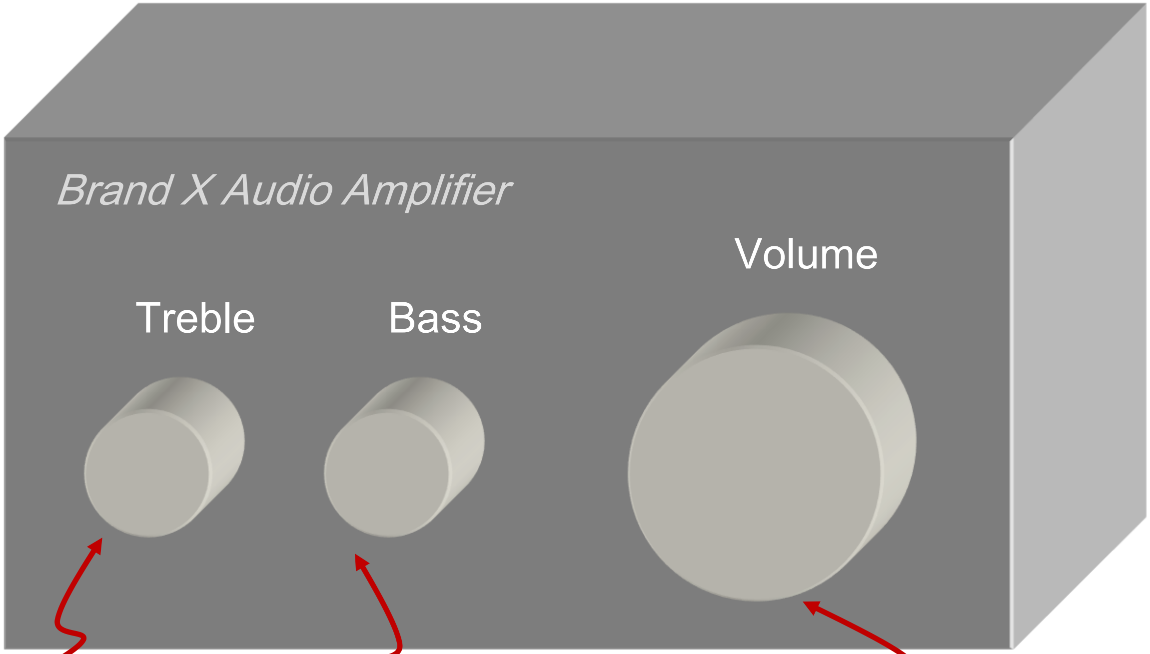
Spectrum examples [2]

If the signal changes in time, the spectrum also changes. Showing this change over time is called a **spectrogram**



Audio amplifiers [1]

In traditional audio amplifiers you have three ”volume buttons”.



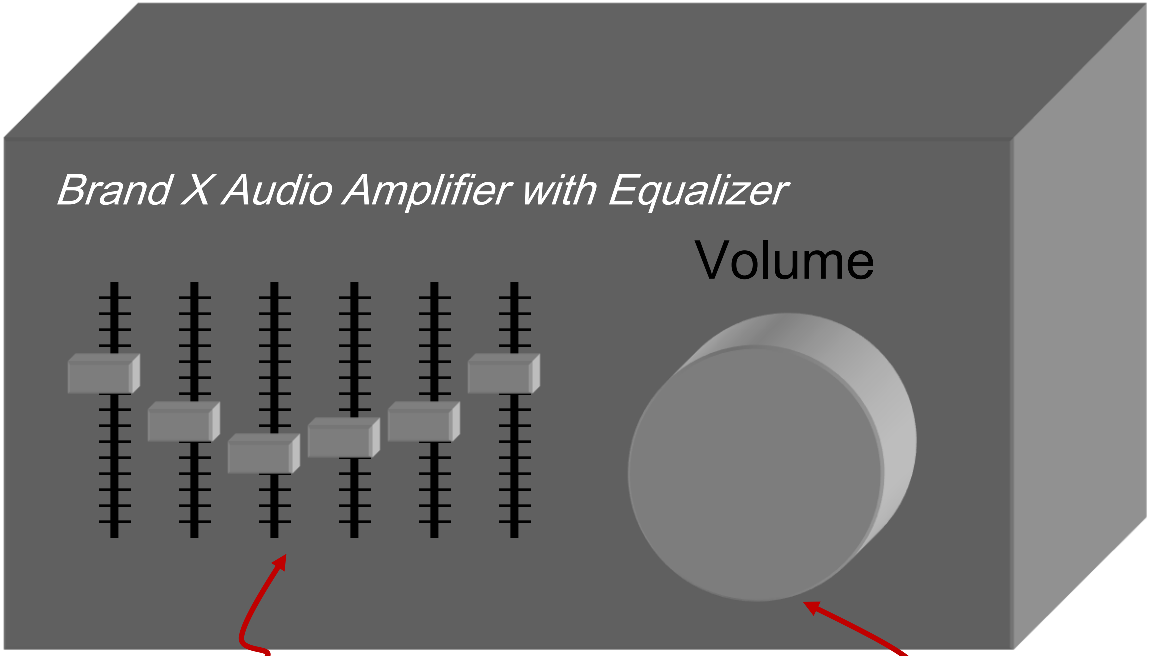
Controls volume of high-frequency part of sound.

Controls volume of low-frequency part of sound.

Controls overall volume.

Audio amplifiers [2]

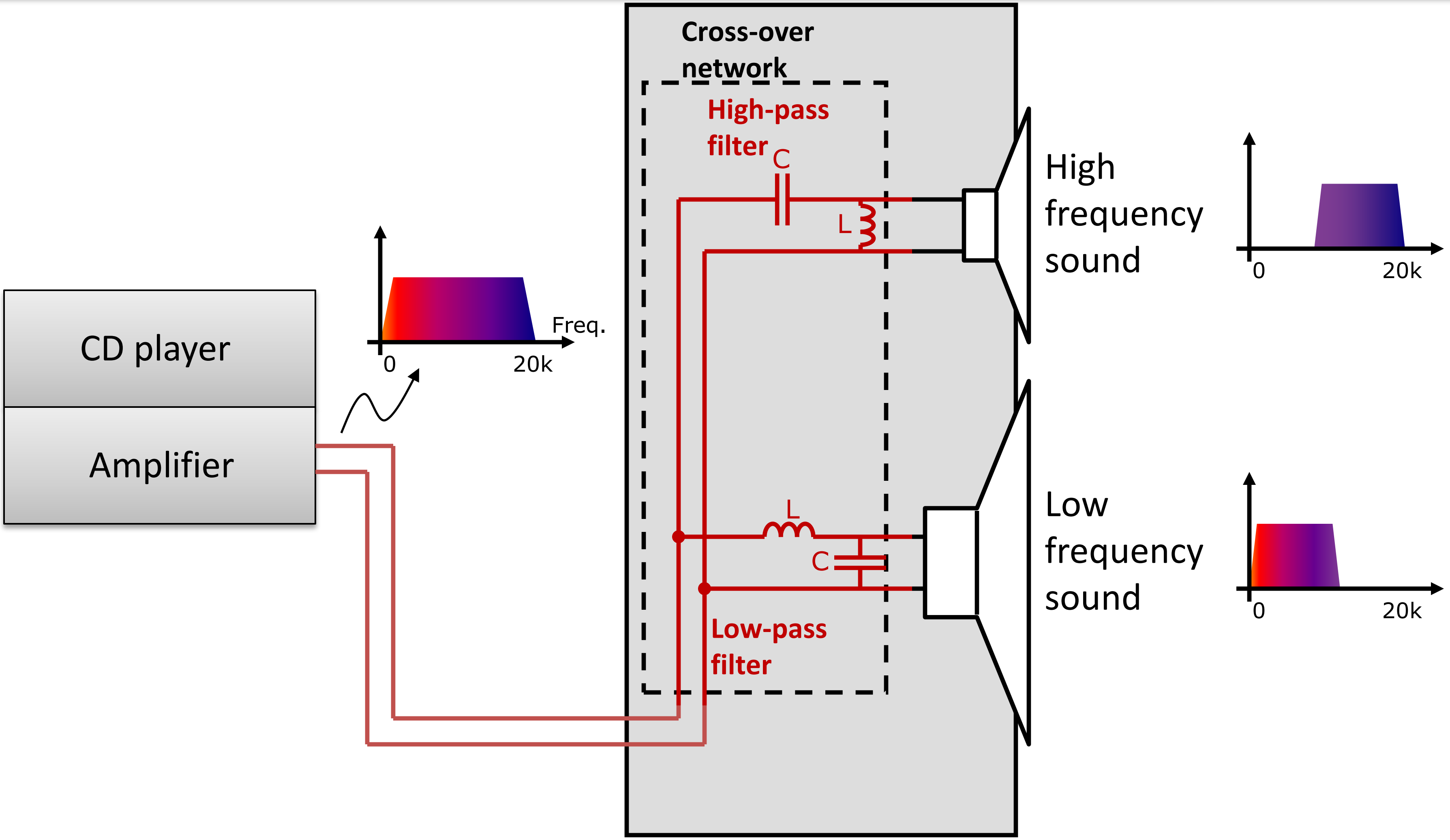
You may also have a more advanced ”equalizer”:



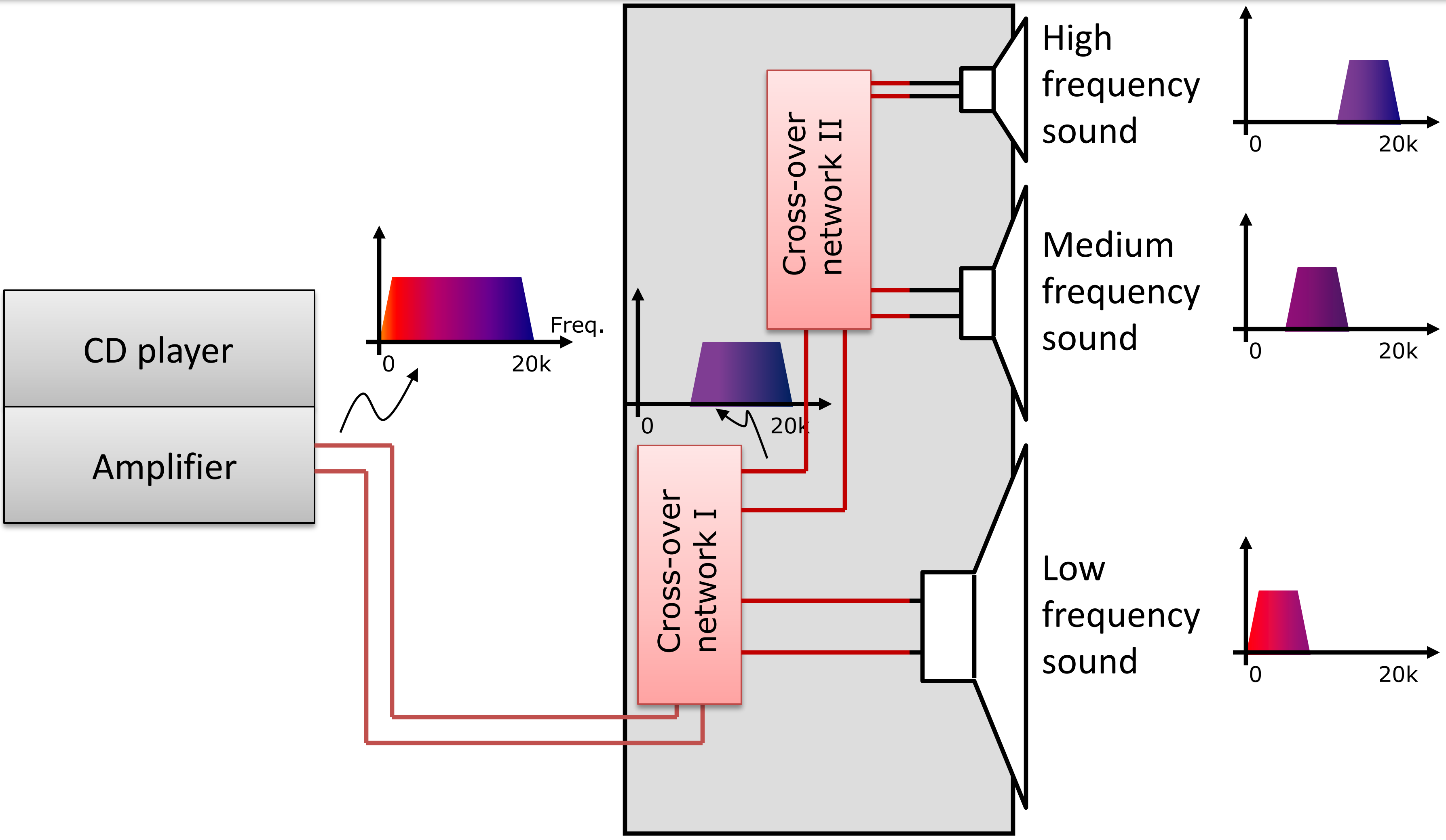
Several volume controls for low to high frequencies.

Controls overall volume.

Two-way loudspeakers



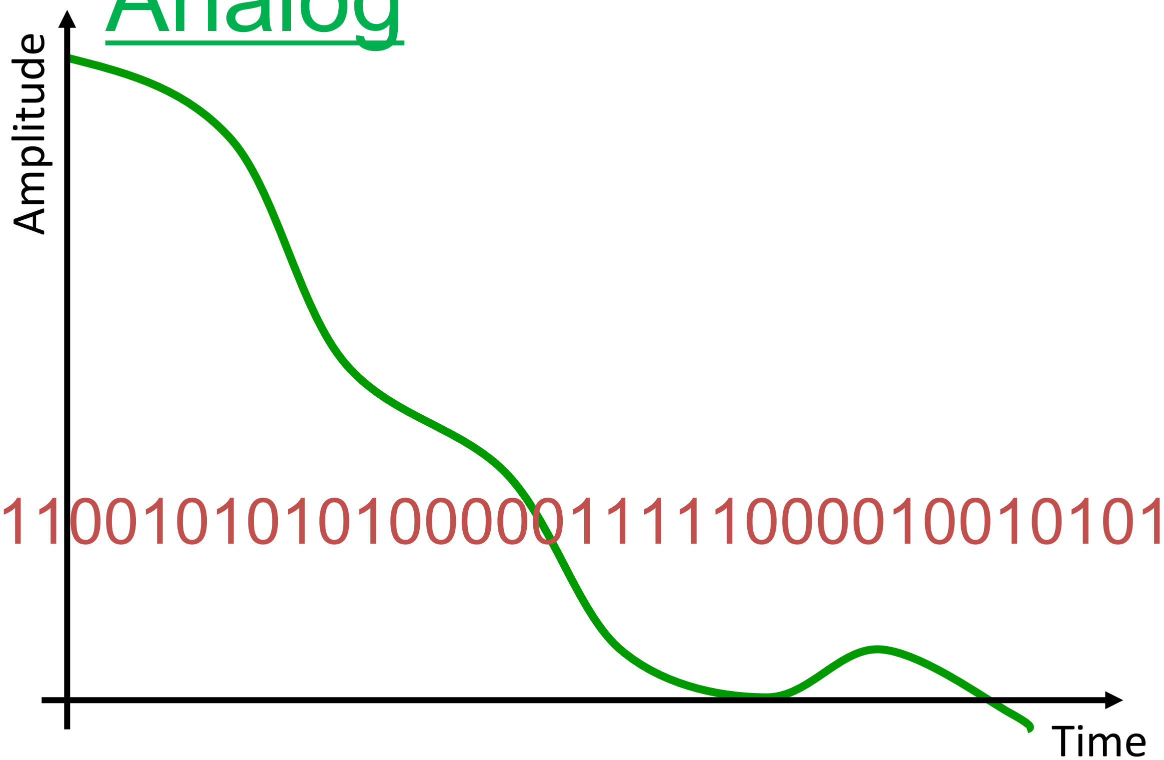
Three-way loudspeakers



Analog vs Digital

Analog and Digital Signals

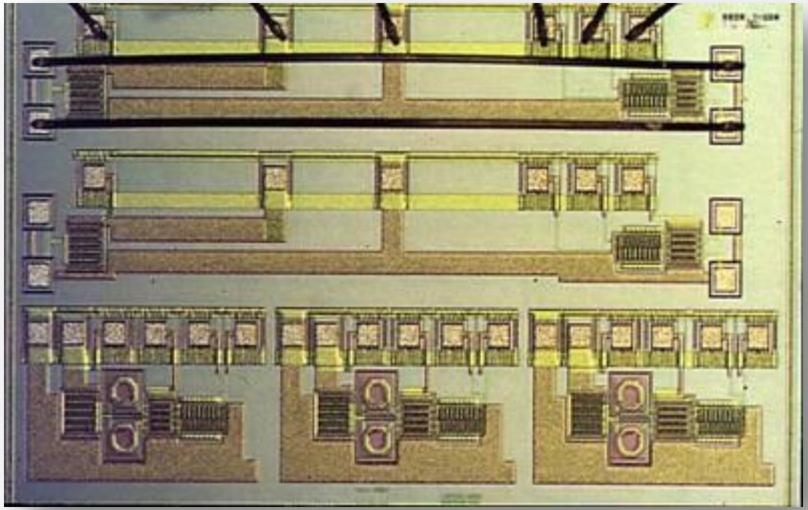
Analog



Digital

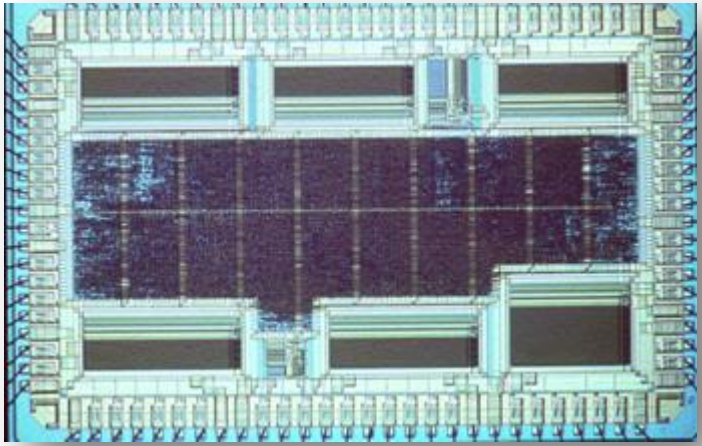
0011001010101000001111100001001010100010111000100010000

Analog or Digital



Analog

- ”real” signals
- few components
- low power consumption?



Digital

- Complex algorithms
- High precision
- Better Storage Capabilities

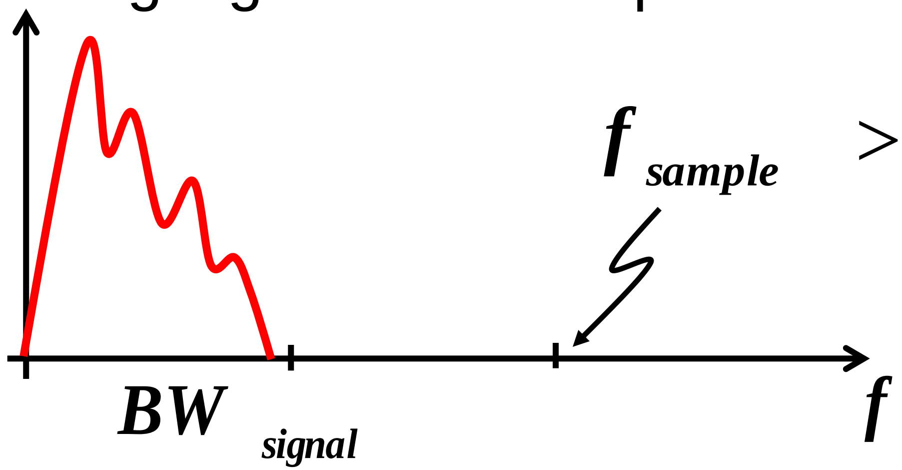
CD/DVD, MP3, Digital Camera, GSM, 3G, Computers, etc, etc

Nyquist's Sampling theorem

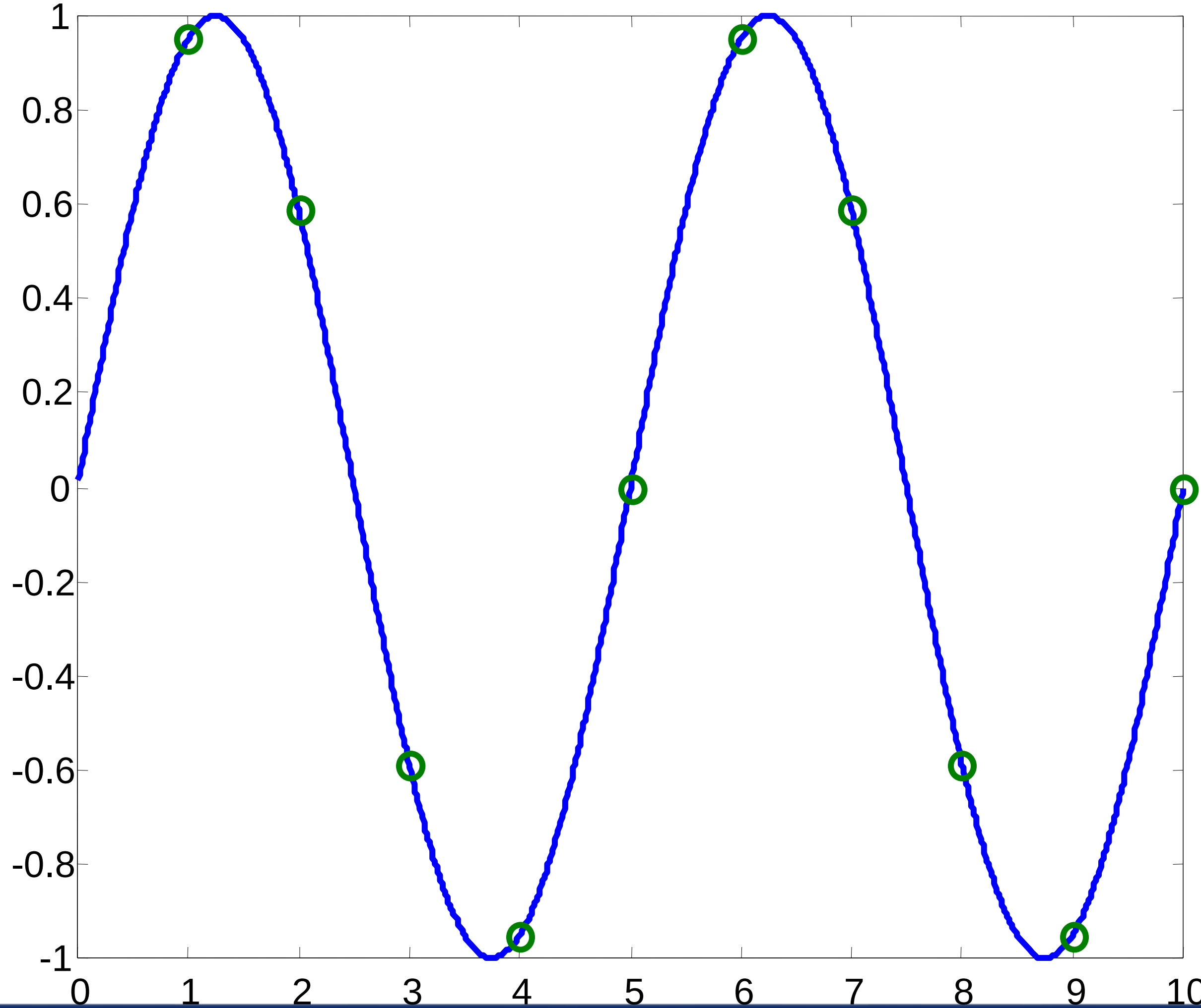
If an analog signal with a bandwitdth of BW_{signal} , is sampled with a sampling frequency of

$$f_{sample} > 2BW_{signal} ,$$

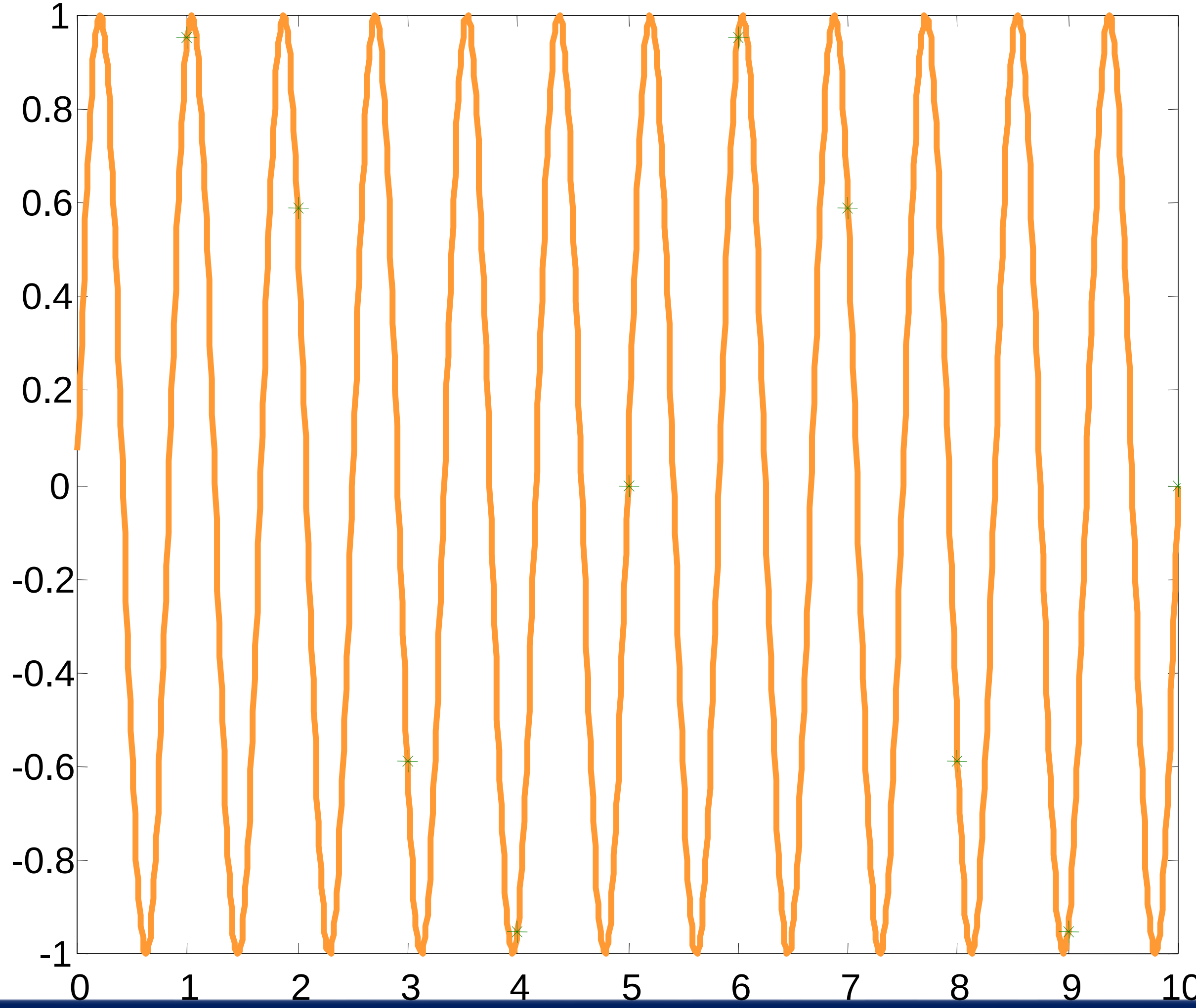
the analog signal can be reproduced.



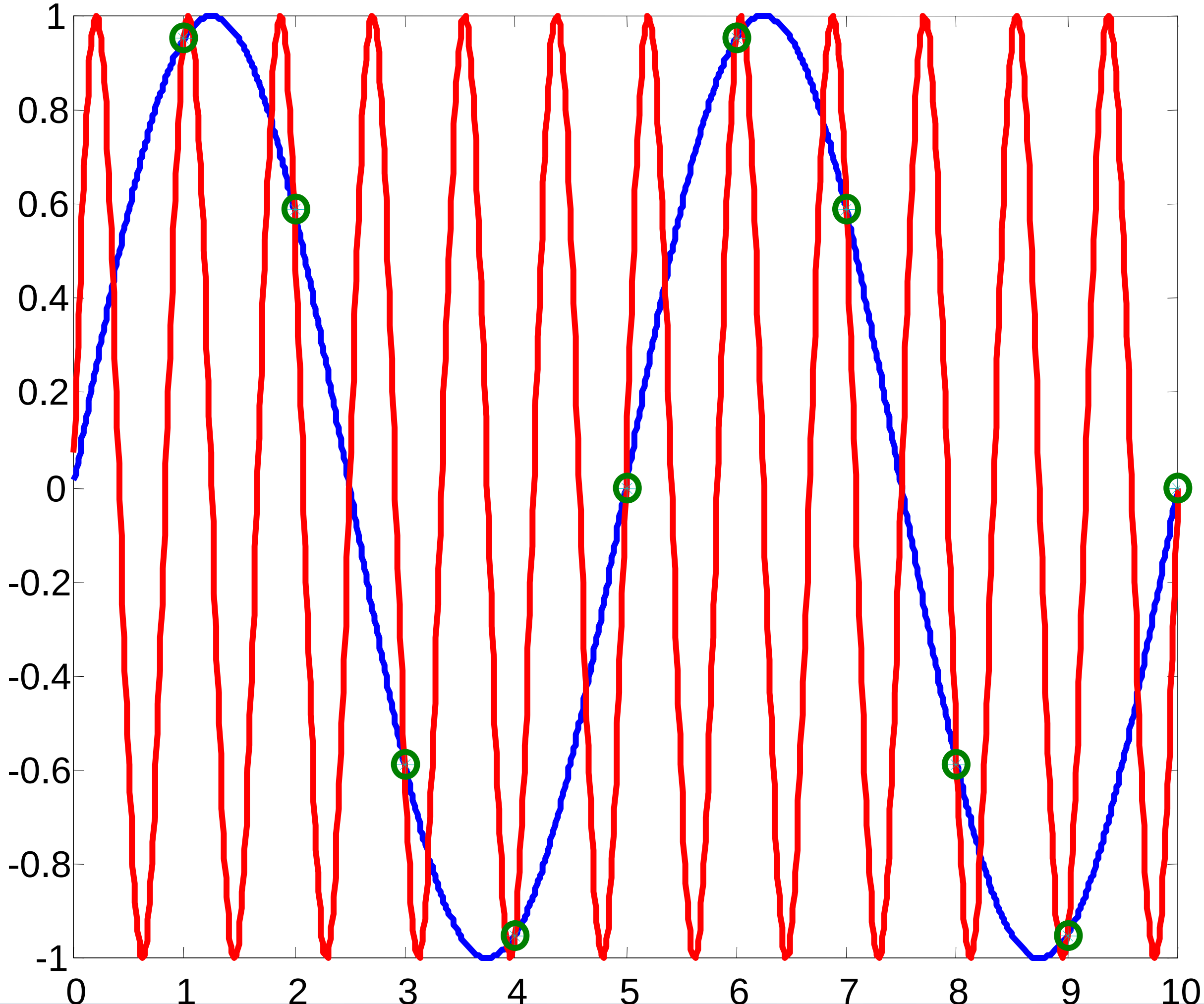
Sampling – $f_s > 2f_{signal}$



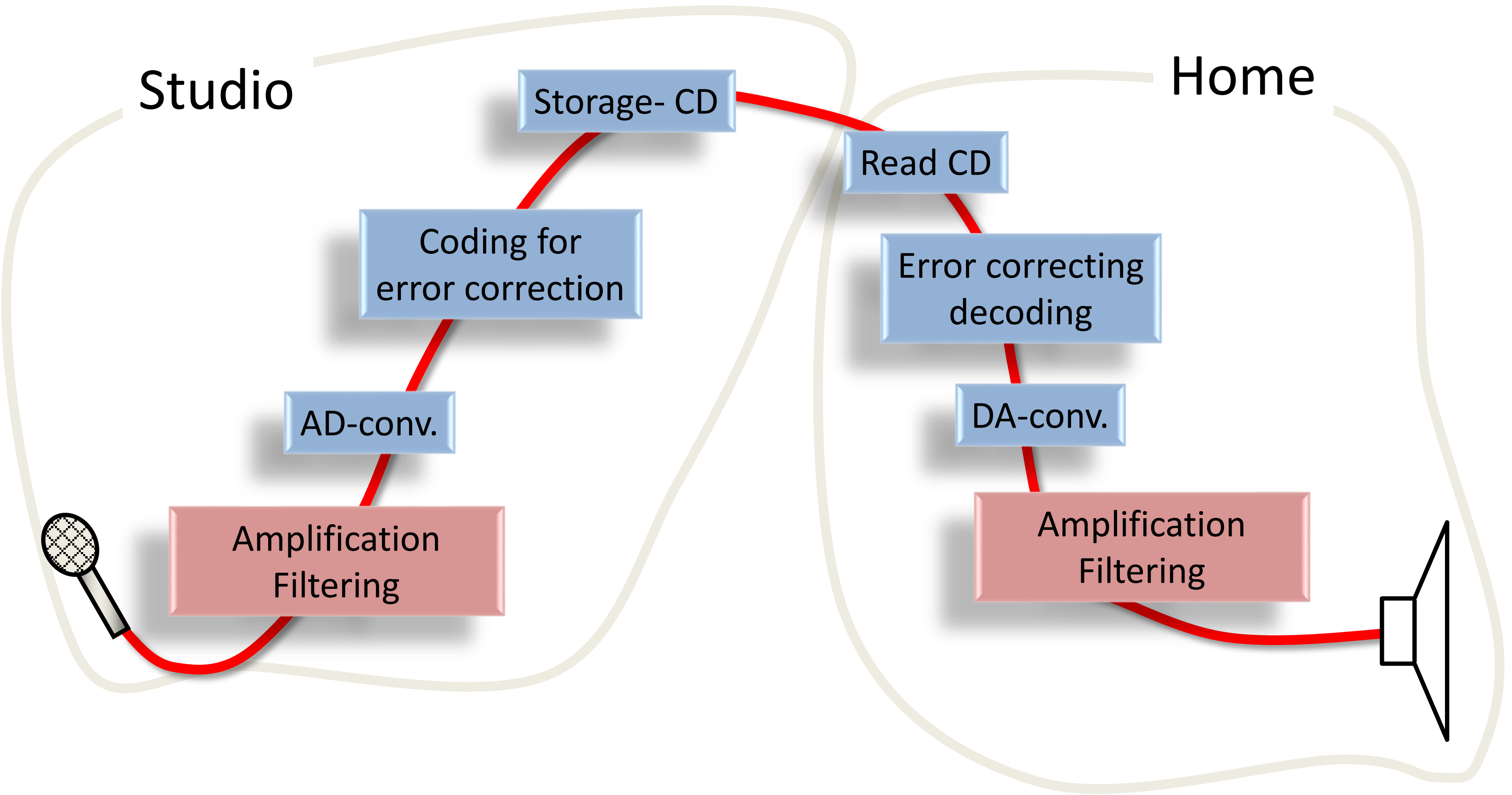
Sampling – $f_s < 2f_{signal}$



Sampling points are the same - aliasing



Example: Digital Audio

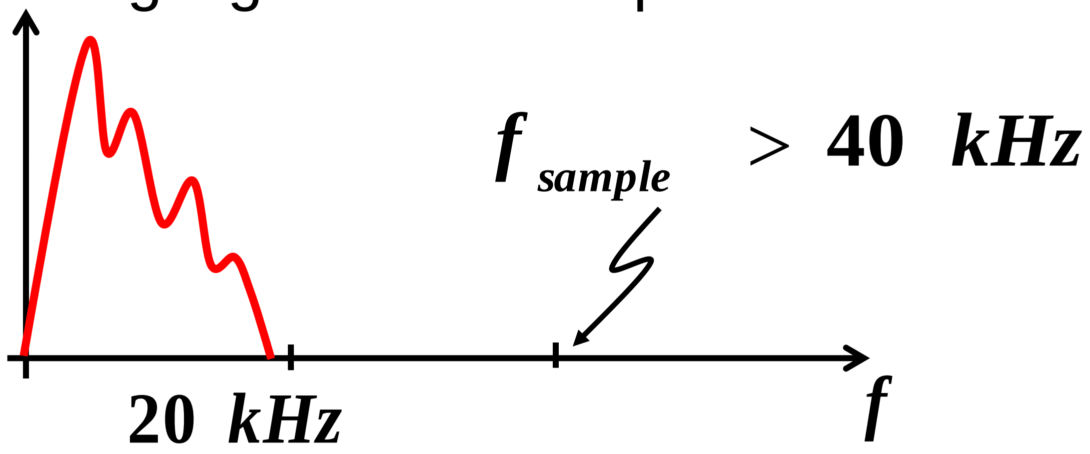


Nyquist's Sampling theorem - Audio

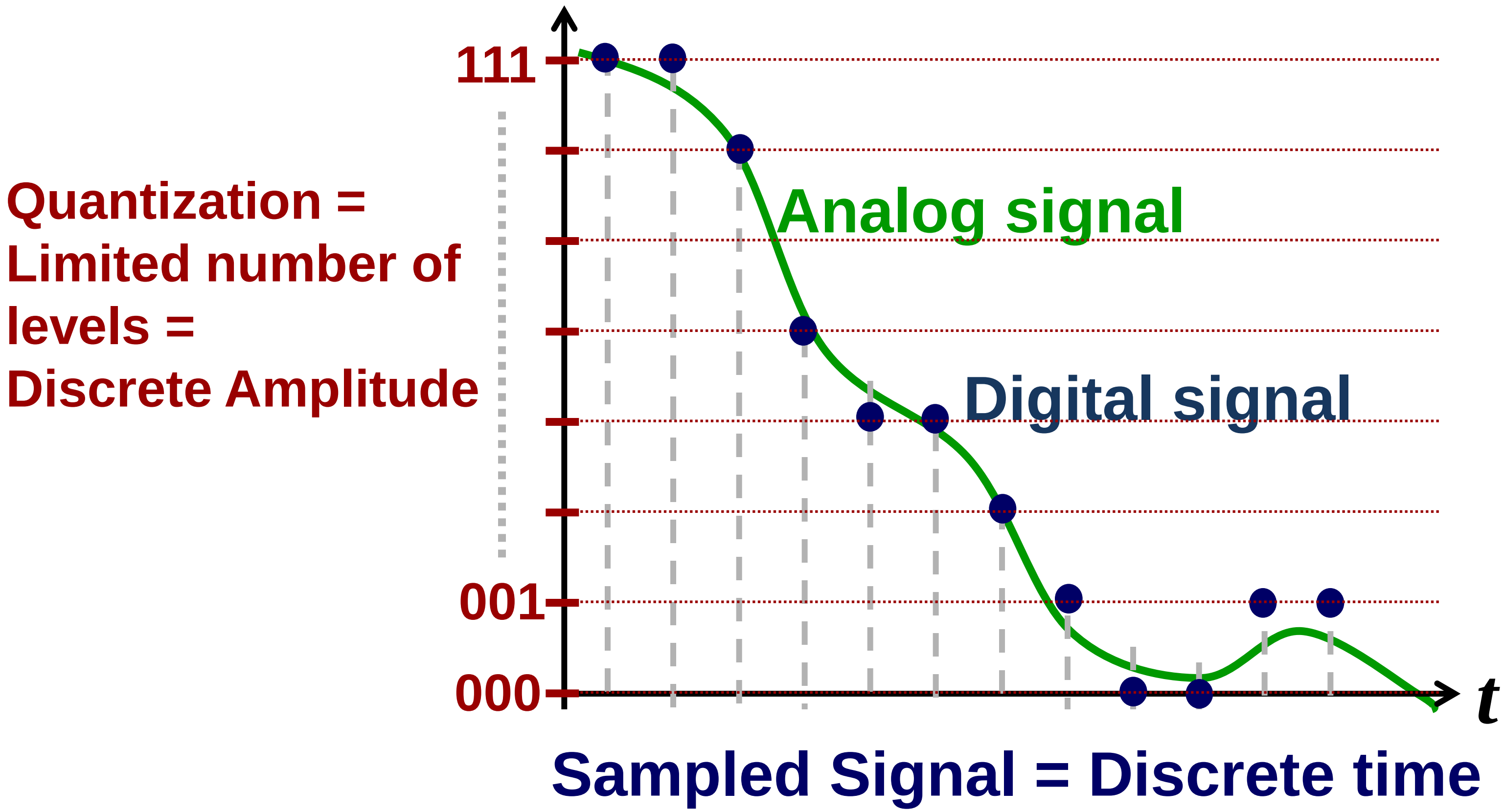
The human ear has a range 20Hz-20kHz, if

$$f_{sample} > 2 \times 20\ kHz = 40\ kHz$$

the analog signal can be reproduced.



Digital Signals

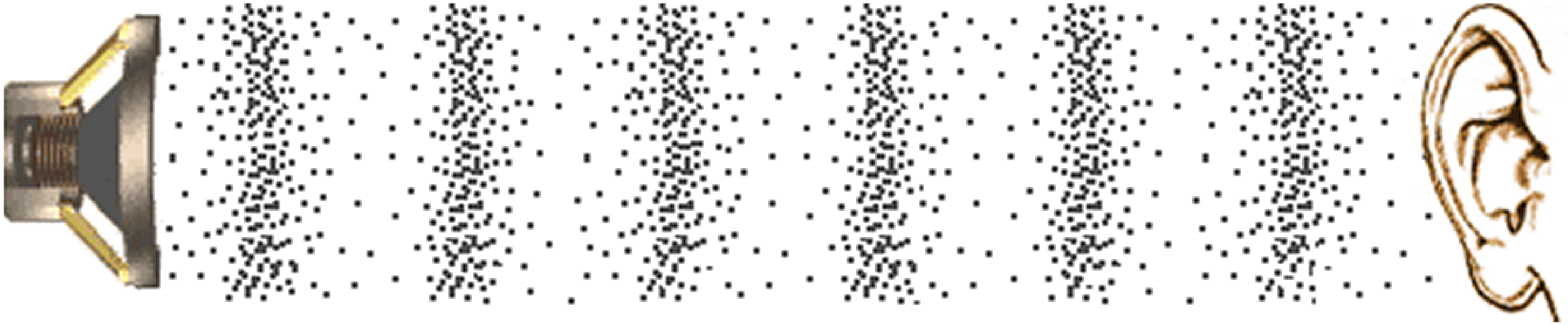


Digital Signal = Discrete time and amplitude

Sound

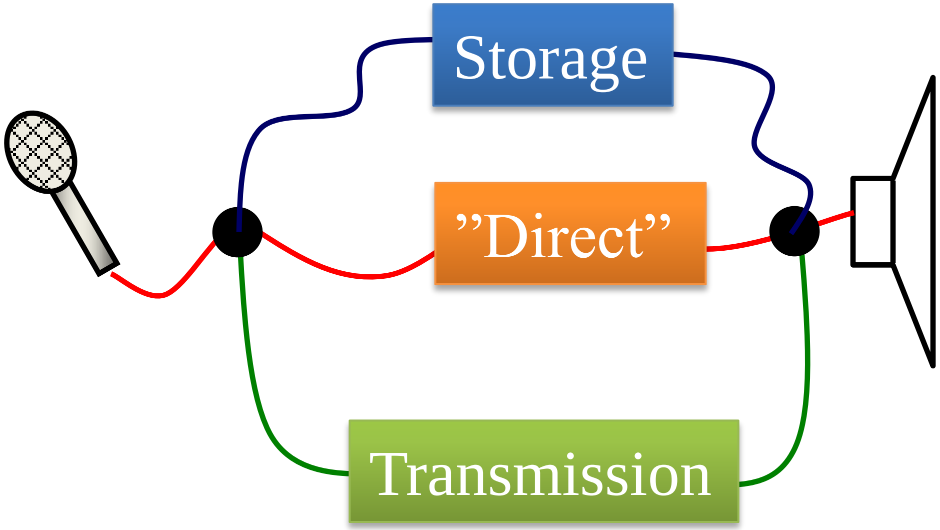
What is sound?

Velocity in air: 340m/s
Audible to the human ear: 20-20.000Hz



Audio systems/signals

- We can use the sound in different ways
  - direct amplification from mic to speaker
  - store it on some media: CD, harddrive, tape, etc
  - tansmit it to another location: radio, mobile phones, et
- and of course combinations of the above



Audio

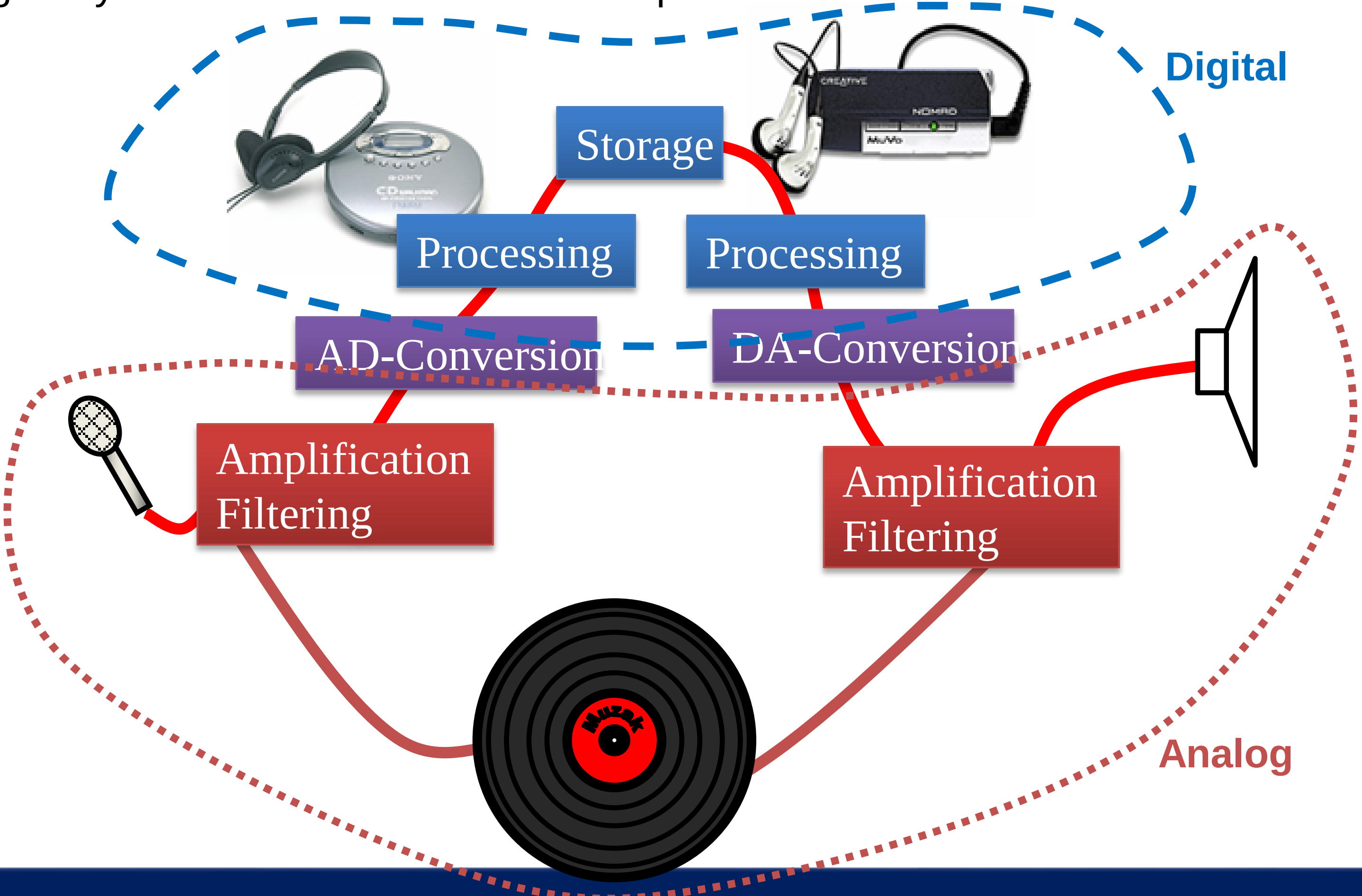
Recording and Storage

When we deal with audio (and images) it is the **perceived** sound quality that is important. This is very hard to measure!

perception = physical sensation interpreted in the light of experience

Audio systems/signals

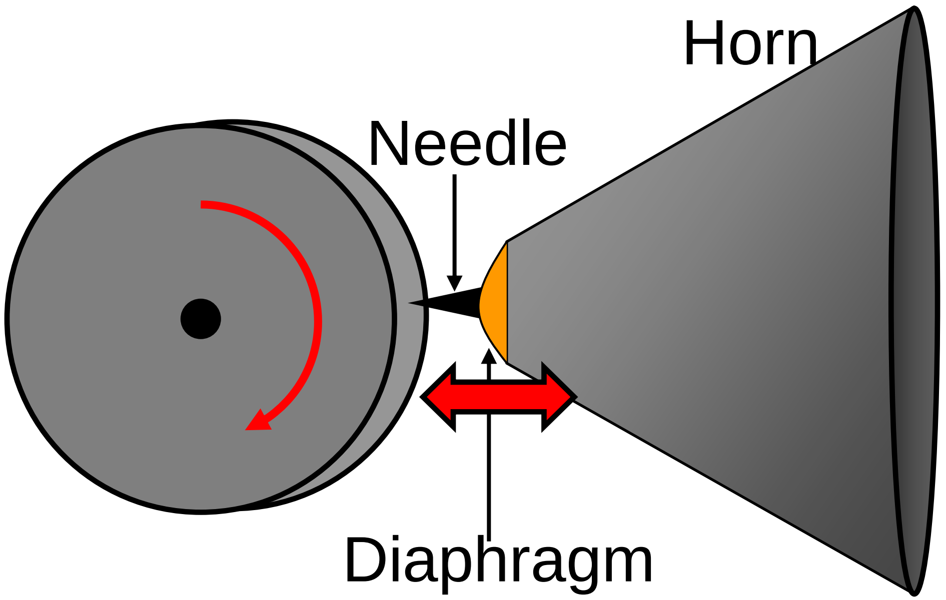
- A audio system consists of many different parts of which not all are necessary for all systems.
- A digital system contains more data manipulations



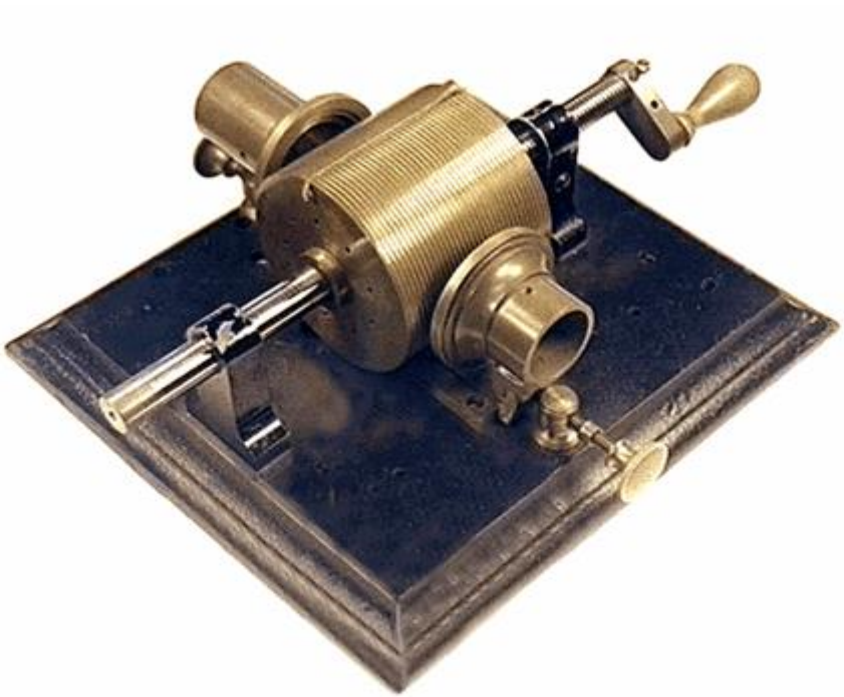
First Audio Recording - 1877

Thomas Edison is credited with creating the first device for recording and playing back sounds. A diaphragm directly controlled a needle, and the needle scratched an analog signal onto a tin foil cylinder. First recorcing: "Mary had a little lamb"

Analog wave stored mechanically on tin foil cylinder.



Edison and Phonograph

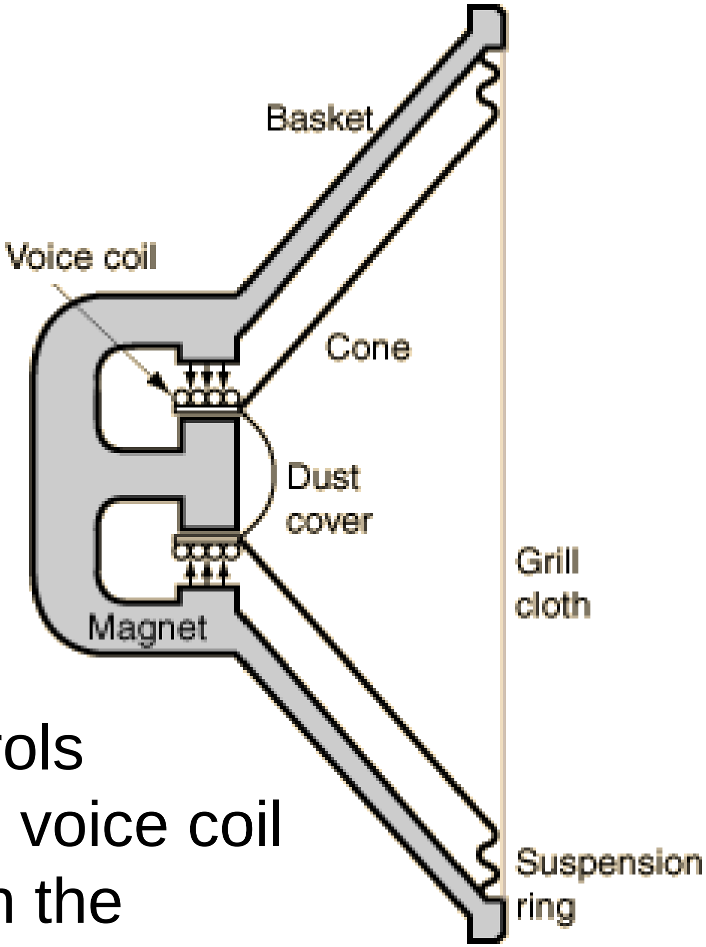


Replica of first phonograph

Loudspeaker

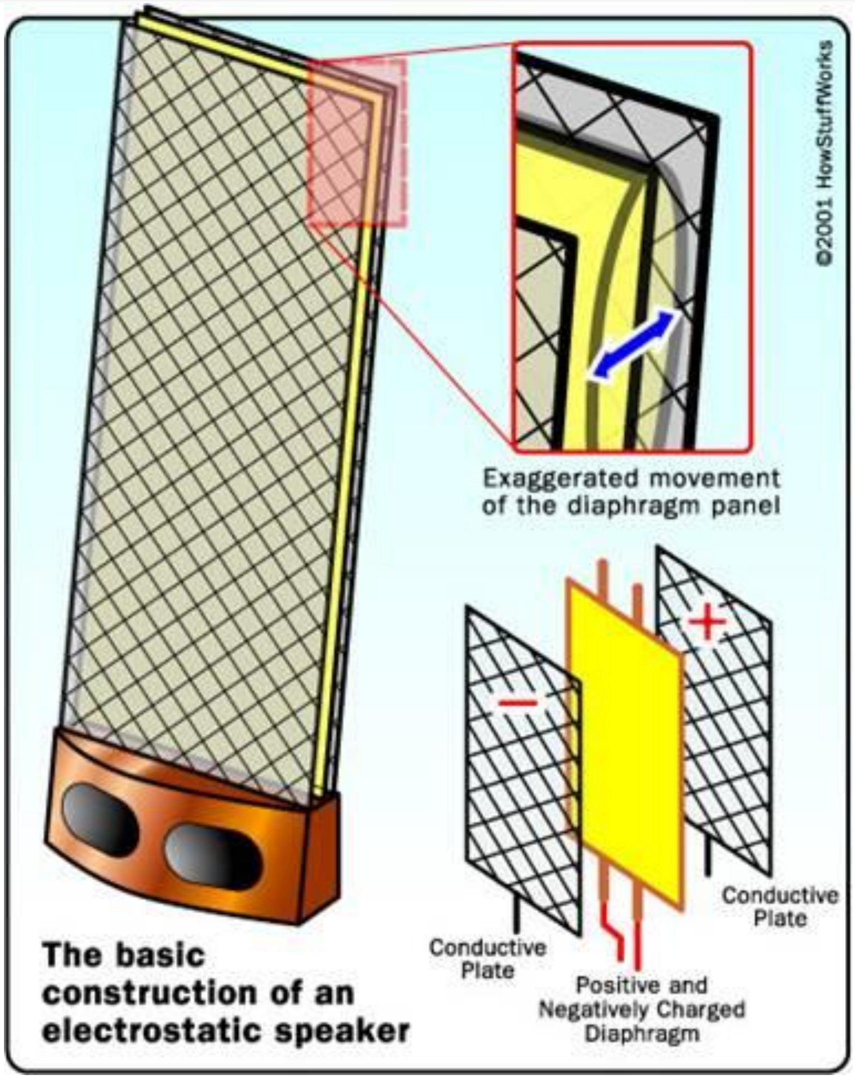
Woofer

Tweeter



Audio signals controls current through the voice coil which interacts with the permanent magnet

Electrostatic Speaker



planar magnetic speaker – magnetc instead of electrical field

The panel has a low mass ⇨
moves quickly and precisely to signal changes changes in the audio signal ⇨
clear, extremely accurate sound reproduction.

The panel doesn't move a great distance ⇨
not very effective at producing lower frequency sounds. Therefore, often used together with woofer that boosts the low frequency

Electrostatic speakers need power from socket ⇨
extra wires and more difficult to place

Microphones

Carbon microphones - The oldest and simplest microphone uses carbon dust. The carbon dust has a thin metal or plastic diaphragm on one side. As sound waves hit the diaphragm, they compress the carbon dust, which changes its resistance.

Dynamic microphones - the diaphragm moves either a magnet or a coil when sound waves hit the diaphragm, and the movement creates a small current.

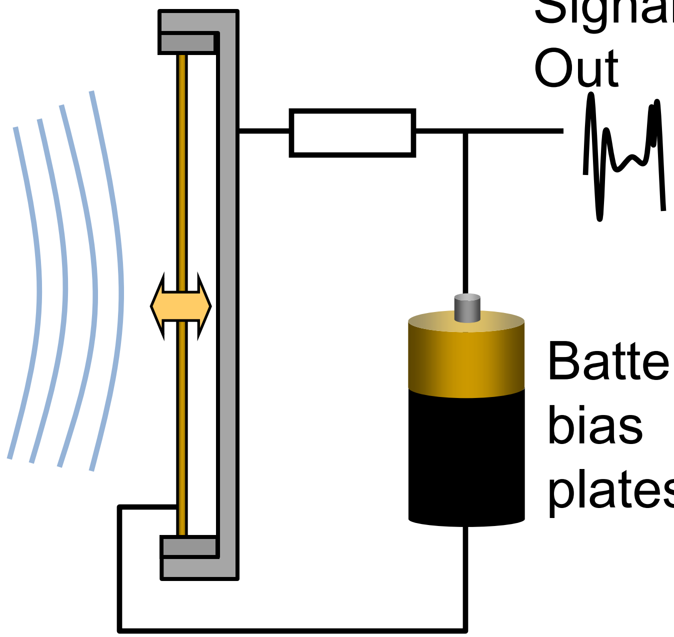
Ribbon microphones - a thin ribbon is suspended in a magnetic field. Sound waves move the ribbon which changes the current flowing through it.

Condenser microphones - A condenser microphone is essentially a capacitor, with one plate of the capacitor moving in response to sound waves. The movement changes the capacitance of the capacitor, and these changes are amplified to create a measurable signal.

Crystal microphones - Certain crystals change their electrical properties as they change shape. Attaching a diaphragm to a crystal, the crystal will create a signal when sound waves hit the diaphragm.

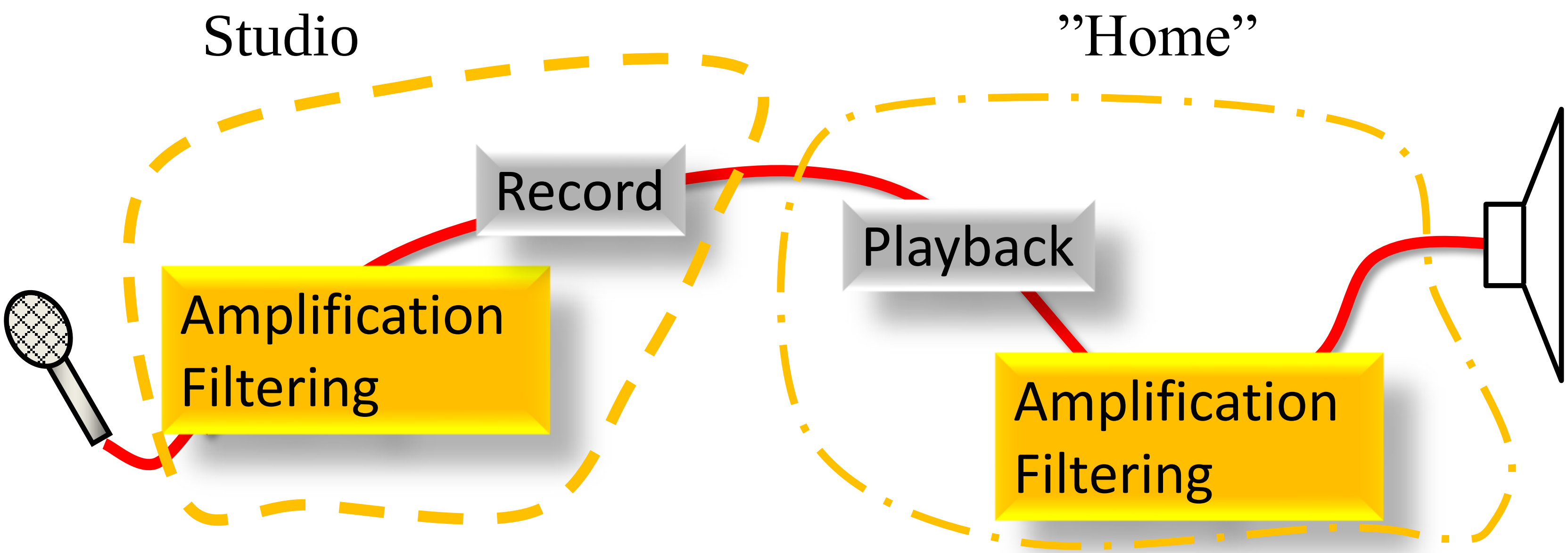
Condenser Microphone

Already seen in a previous lecture.



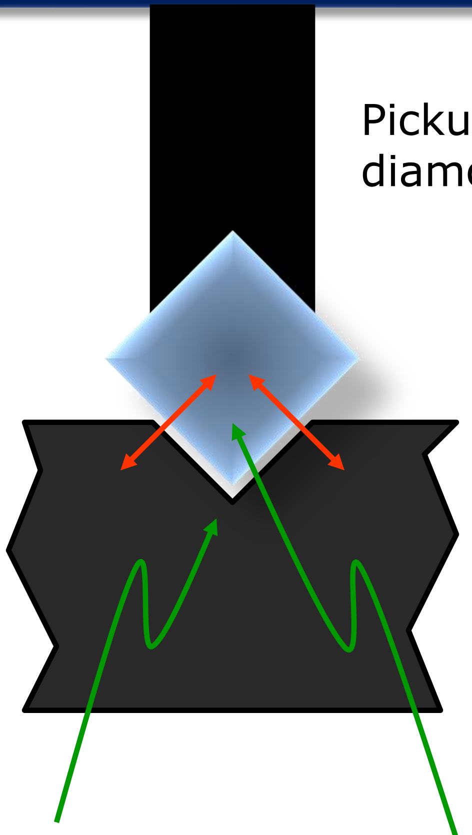
Vinyl records

Analog recording/playback



Vinyl Records (1)

Pickup with diamond tip.



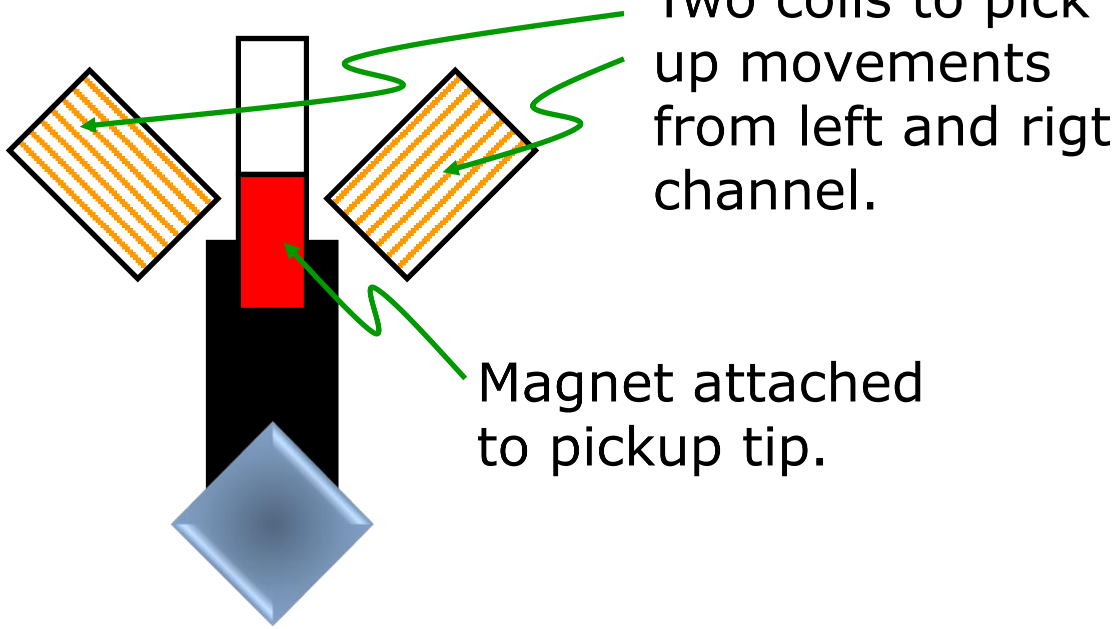
Music is engraved as a V-shaped track on the vinyl surface.

Movement in two directions gives stereo sound.

Vinyl Records (2)

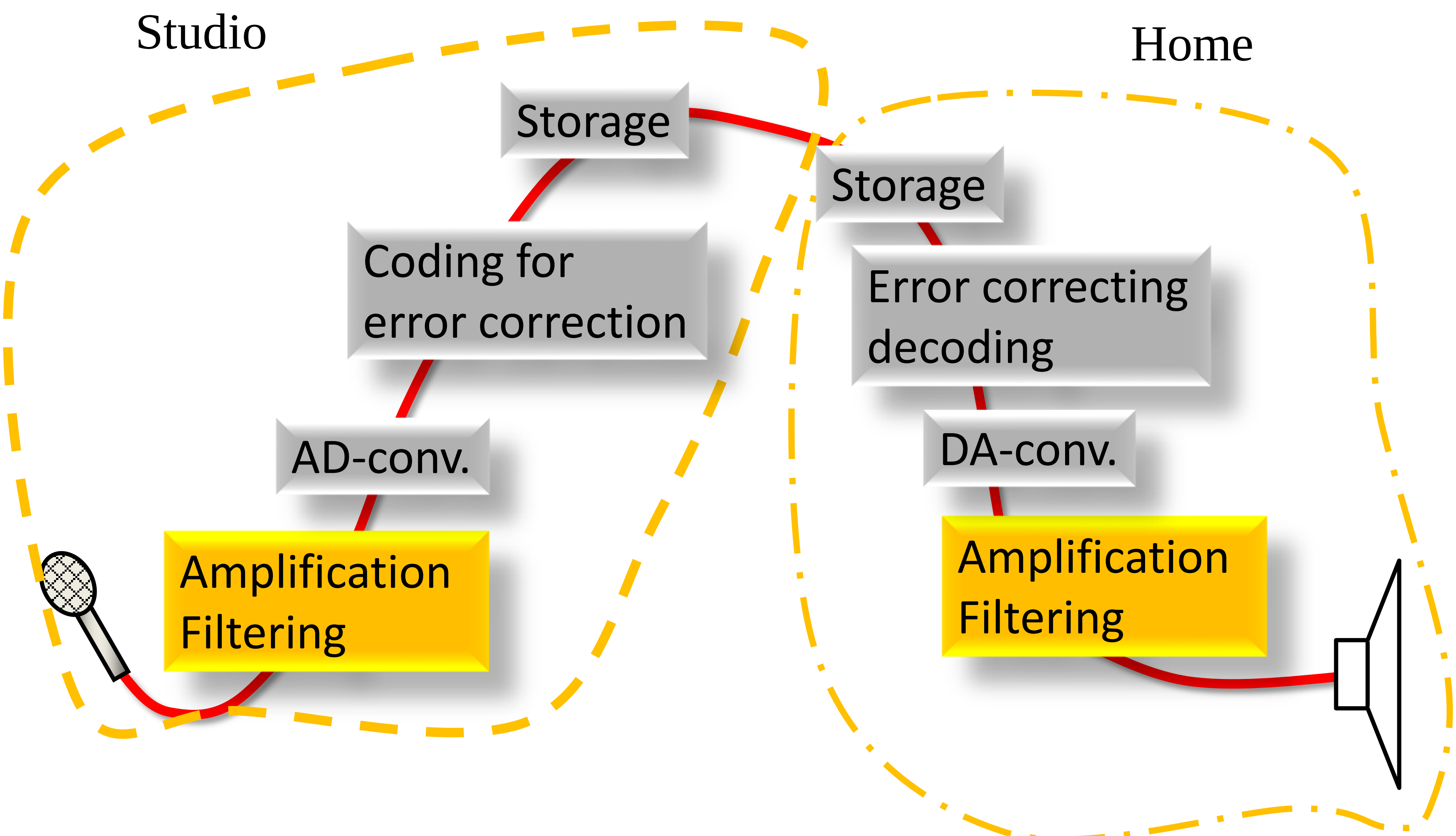
Magnetic fields are used for ”measuring” movements of the pickup tip and converting it to stereo sound:

Simple illustration:

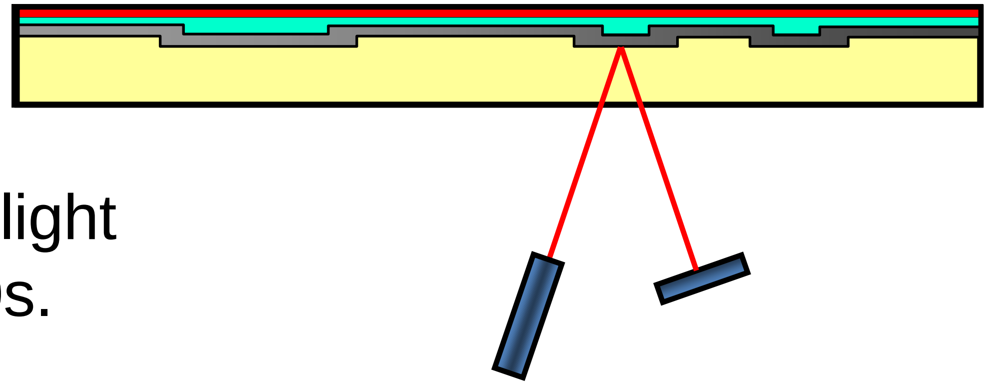


Pick-up can be either ”moving coil” or ”moving magnet”

Digital recording and playback



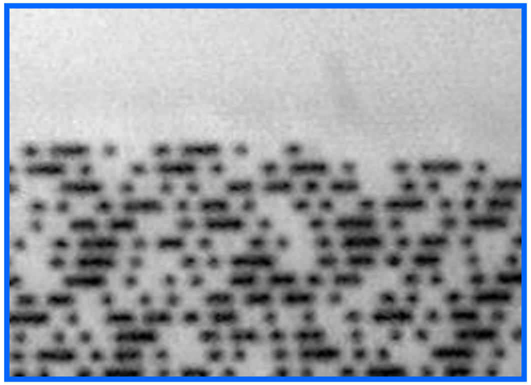
Reading CDs/DVDs/BlueRays



- Bumps and pits reflects light differently, thus 1s and 0s.

- Tracking makes sure that the laser hits the center of the ”digit” by adjusting the speed, e.g. three detectors.

- Data is encoded to avoid long stretches without bumps which would make the laser loose track.

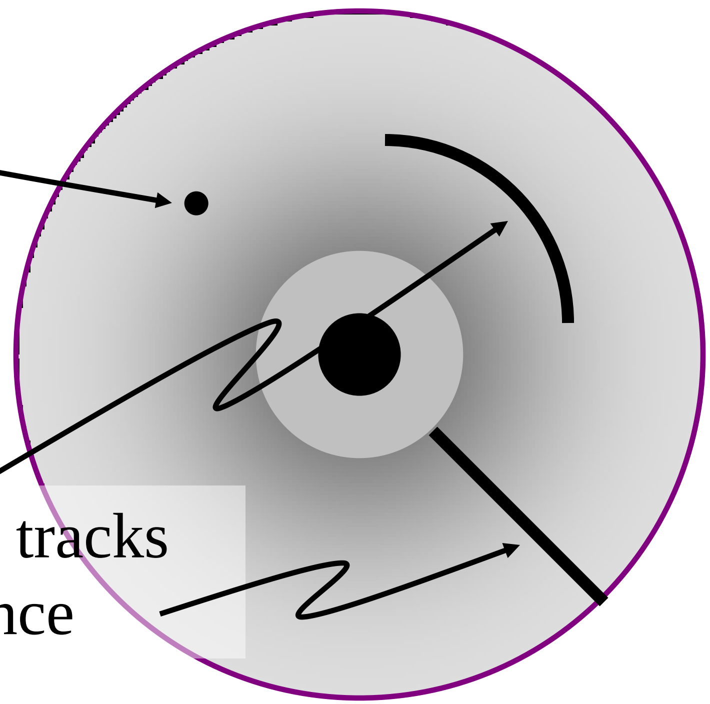


Error correcting codes

The data is coded so that if there is a ”limited” error, they error can be corrected

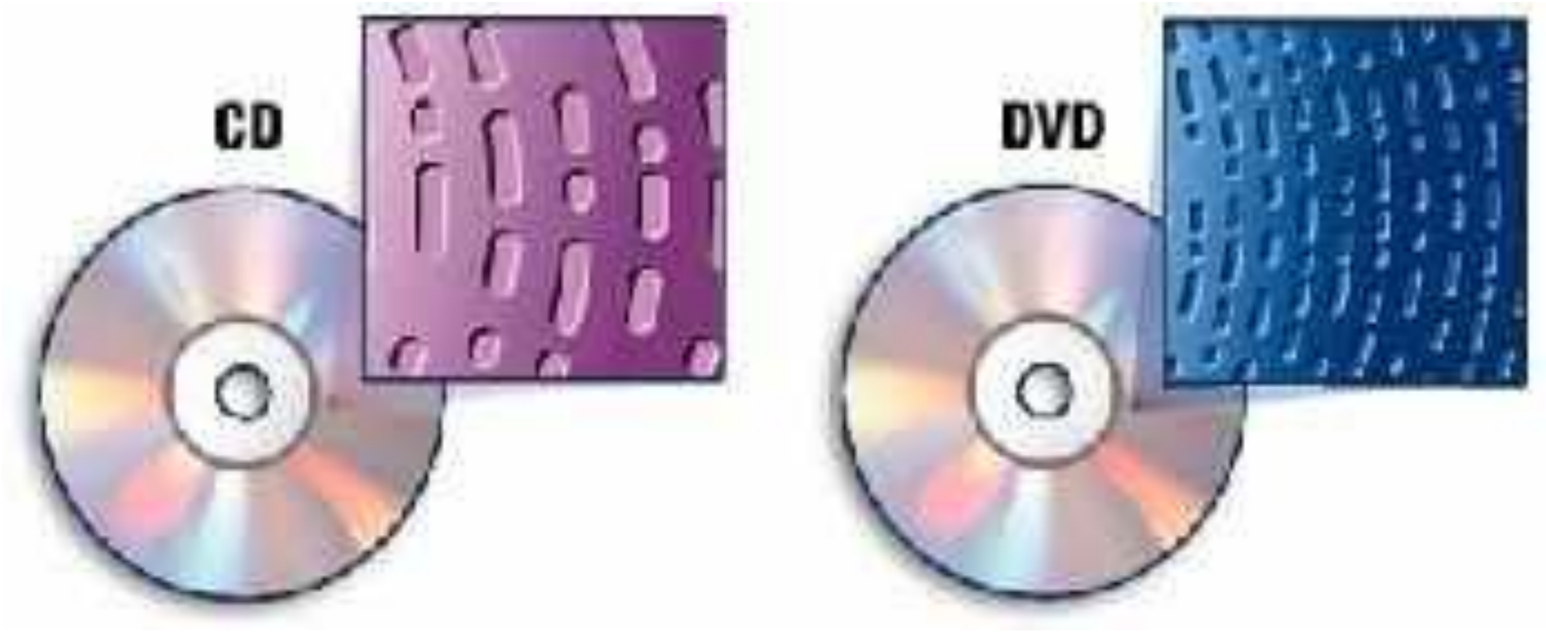
Error correcting coding will increase the number of bits.

You can drill a small hole in the disc and still playback the CD



Scratches or dirt along the tracks are worse than diagonal once

CD vs. DVD (Digital Versatile Disc) first Digital Video Disc



	CD	DVD
Track Pitch:	1.6µm	0.74µm
Bump width:	0.5µm	0.32µm

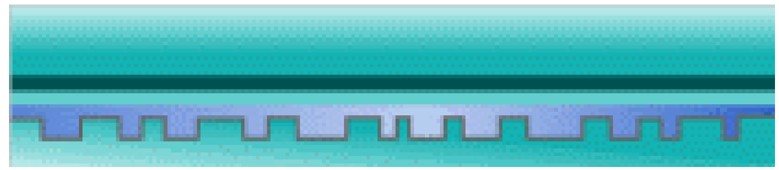
CD vs. DVD audio format

Specification	CD Audio	DVD Audio*
Sampling Rate	44.1 kHz	96/192 kHz
Sampling Accuracy	16-bit	24-bit
Number of Possible Output Levels	65,536	16,777,216

*Lots of DVD audio standards

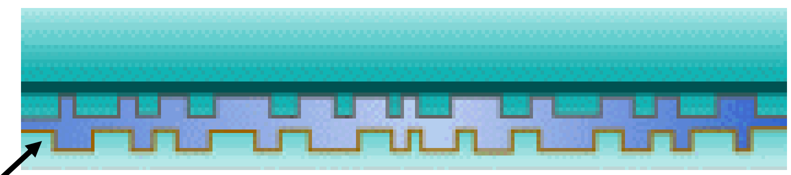
DVD

Single-sided, single layer (4.38GB)



Movie
2 hours

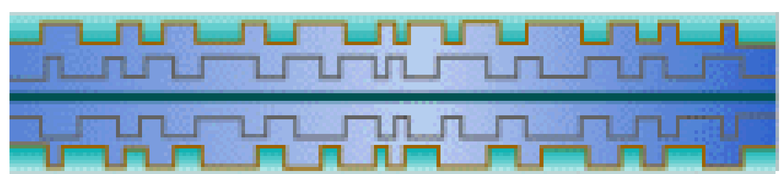
Single-sided, double layer (7.95GB)



Movie
4 hours

semi-reflective gold layer, laser can focus through the outer layers.

Double-sided, double layer (15.9GB)



Movie
8 hours

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MP3

CD: 44,100 samples/second * 16 bits/sample * 2 channels
⇨ **1,411,200 bits per second**

If a song is 3 minutes and 1.4 Mbits per second ⇨
252 Mbits

⇩

This is why you want to use compression

MP3 - compression

Compression can be

- Loss-less - exact data can be retrieved
- Lossy (MP3) - information is lost

MP3 has a constant coding rate independent of input.

Led Zeppelin:	Full CD -	45MBytes	
	MP3 56kbits/s -	1.8MBytes	⇨25times
	MP3 16kbits/s -	0.5MBytes	⇨90times
	MP3 8kbits/s -	0.25MBytes	⇨180times
Schubert:	Full CD -	47MBytes	
	MP3 56kbits/s -	1.9MBytes	
	MP3 16kbits/s -	0.5MBytes	
	MP3 8kbits/s -	0.27MBytes	

MP3 coding

A technique called **perceptual noise shaping** is used. It is "perceptual" part because the MP3 format uses characteristics of the human ear

- There are certain sounds that the human ear cannot hear.
- There are certain sounds that the human ear hears much better than others.
- If there are two sounds playing simultaneously, we hear the louder one but cannot hear the softer one.