Audio

As with most signals, we can distinguish between analogue and digital audio.

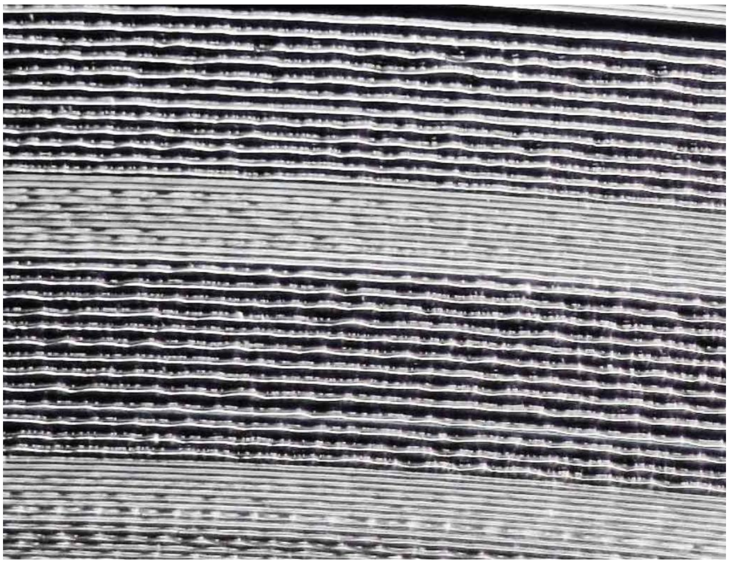
A signal is called analog when it is continuous-time and continuous-values. Conversely, digital signal is a discrete-time signal for which not only the time but also the amplitude has discrete values; in other words, its samples take on only values from a discrete set.

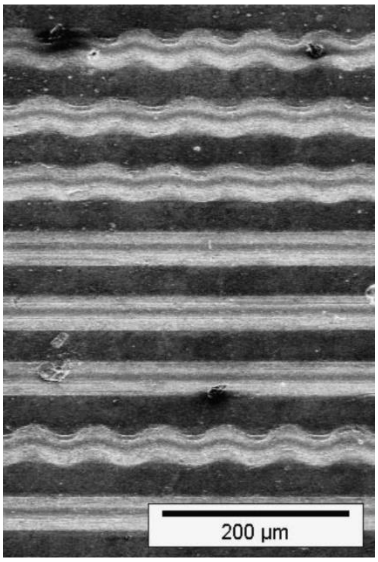
Analog representation

It is a representation that works analogously. The curve continues in the time of amplitude changes and is represented by a continuous curve in the time of voltage changes. The recorded track follows the curve of the amplitude curve after being converted into an electrical signal.

To get analogue audio from a sound:

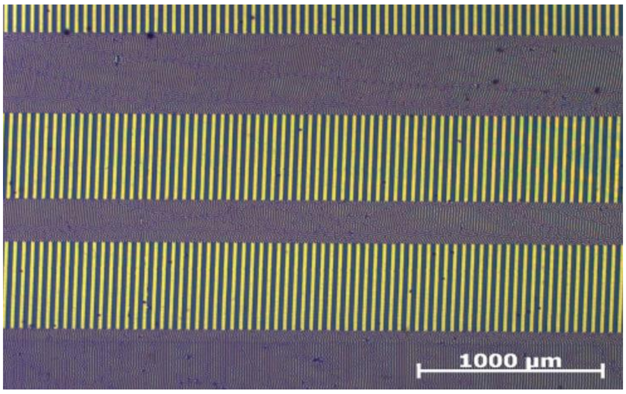
The sound is detected by a transducer, which converts the pressure waves into electrical waves (voltage fluctuations). The signal that was obtained in this way is pretreated and recorded on an analogue carrier (e.g. record, audiocassette).





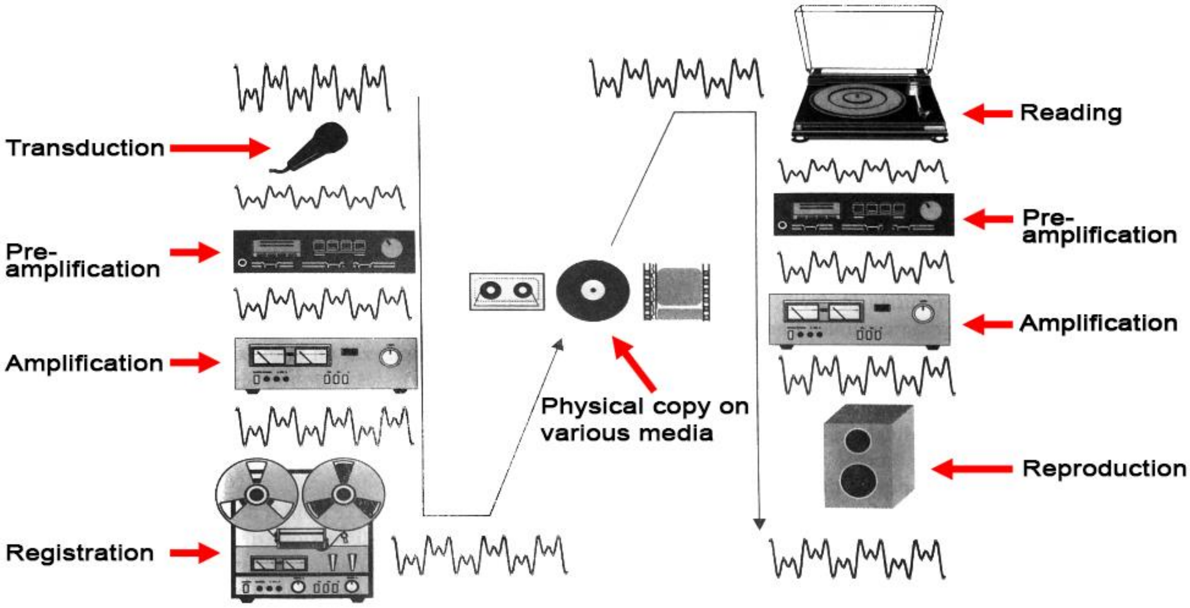
Turns of the groove of a record under the microscope. The course of the groove corresponds to the acoustic vibration of the stored signal.

Playback



Magnetic domains lined up on an audio cassette.
Visualization of the magnetically stored information (a 1 kHz test tone) on a stereo audio cassette; Image taken with a Faraday magnetometer.

The original sound can be reproduced by interpreting the variations of the physical quantities present on the support. In the case of the vinyl record, the variation in the depth or irregularity of the grooves, while for the audio cassettes the variation in the intensity of the magnetic fields.



Every physical transformation that affects the original sound always has a certain error. The error propagates until the analogue audio signal that differs with respect to the start signal is reproduced. The introduced distortion is also called noise.

Analog audio - Distortion

In the transduction there is usually an alteration of the original parameters. This alteration results in distortion. This distortion will propagate until the sound is reproduced. It is measurable through a system quality index called SNR (Signal Noise Ratio), which is a numerical quantity that relates the power of the useful signal with respect to that of noise in any acquisition system.

Signal-to-noise ratio is defined as the ratio of the power of a signal (meaningful information) to the power of background noise (unwanted signal): $SNR = \frac{S}{N}$

The SNR value can also be measured in decibels.

$$SNR_{dB} = 10\log_{10}\frac{S^2}{N^2} = 20\log_{10}\frac{S}{N}$$

A human being needs at least an SNR of around 6 dB in a noisy signal in order to be able to hear the speech contained therein.

Worn vinyl discs introduce tics and other imperfections. Some of these are considered "pleasant" by audiophiles. In the audio cassettes, instead, a background noise is typical.

Analog audio - pro and contra

The advantage is that the simple representation of very extensive frequency ranges is possible, and the reproduction is possible without sophisticated equipment. The physical distortions and the continuous degradation of the media are disadvantageous.

	Advantages	**Disadvantages**
Analogue	• Analog tape recording is better than digital, because the sound wave, in the form of alternating current, is continuously recorded on the tape • Representation of many frequencies • Unsophisticated equipment like the vinyl record and the audio cassette • Tried and tested format - 2”24 track tape is a worldwide standard	• The metal particles that cover the surface of the tape are rather lazy and do not move from their stillness unless the magnetic field of the recording head is strong enough to polarize them. → the weakest sounds of intensity are not properly engraved, as are the high frequencies • Transients are very rapid events, dynamic and rich in high frequencies, qualities that the tape cannot fully satisfy • Deterioration of the material that occurs at each step and the risk of self-elimination of the material. • Dependence of physical support • copying deteriorates the sound

Digital audio

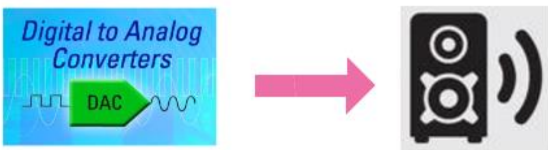
A signal is called digital if it is discrete time and discrete values. A **digital signal** is a discrete-time signal for which not only the time but also the amplitude has discrete values; in other words, its samples take on only values from a discrete set.

It is a representation that does NOT try to imitate the continuous amplitude curve with a curve analogous to it, it shows the numbers that represent the value of the amplitude in successive instants of time in each case. The sequence of numbers will represent the trend of the amplitude curve.

Acquisition of digital audio from a sound:

- The sound is detected by a transducer that transforms the pressure waves into electric waves (voltage variations) → Ex.: Piezoelectric materials (such as quartz)
- The signal thus obtained is pretreated and sent to an Analog-Digital converter (ADC).
- In the output from the ADC, a discrete-time signal with discrete values is obtained, which is digital.
- Finally, the digital signal is presented in a specific format and stored in mass storage device.

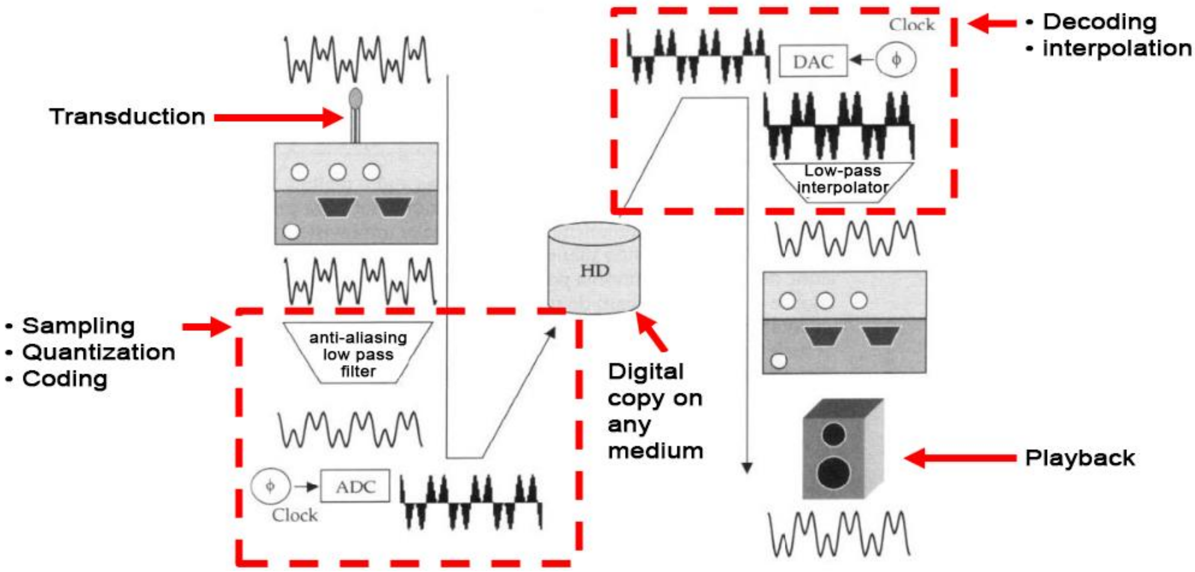
Play digital audio:



- The format in which the audio data is stored is interpreted and the output is sent to a digital-to-analogue converter (DAC)
- The DAC (in our case the sound card) generates an electrical signal that describes the sound, i.e. analogue audio
- The variations in the electrical quantity are interpreted by a device (for example audio boxes) in order to generate a vibration, i.e. the sound wave source

Data acquisition devices (DAQ) are complex devices that both collect and reproduce data.

Digital Audio Chain:



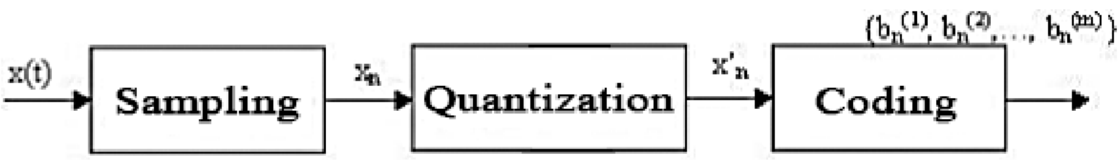
Analog to digital conversion

Most devices used today process digital signals. For this reason, it is necessary to convert the naturally analogue signals into digital signals.

This requires the transition from continuous domains to discrete domains by:

- Converting a continuous-time signal into a discrete-time signal -> **sampling**
- Transforming a signal into continuous values with discrete values -> **quantization**

The coding phase follows.

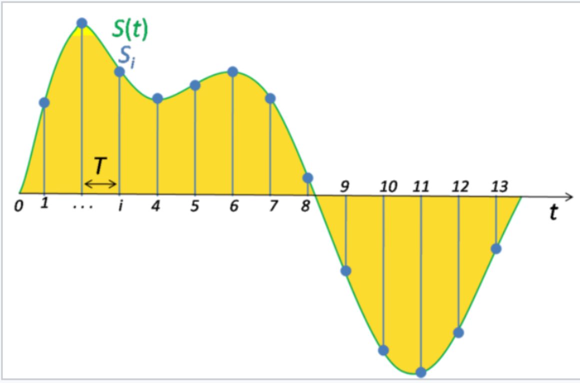


Signal filtering → Signal sampling → Quantization of samples → Coding of quantized samples

Sampling

The transformation of a continuous-time signal into a discrete-time signal is called sampling.

In practice, only some equidistant signal values are considered. This is described as the signal, using only a limited number of samples. The frequency with which the samples are taken is called the sampling rate.



Signal sampling representation. The continuous signal is represented with a green colored line while the discrete samples are indicated by the blue vertical lines.

Nyquist frequency

The Nyquist frequency, the highest frequency available in the signal spectrum, is defined in a periodic and limited band signal.

- To define the Nyquist frequency, the Fourier series is used to get the spectrum.

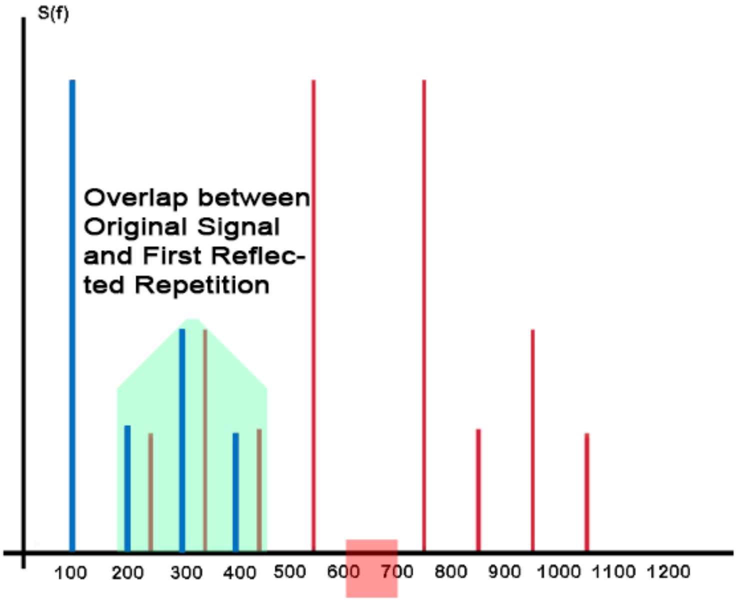
- The Fourier transform can be used for non-periodic signals. For unlimited band signals (unlimited spectrum) it is imperative to use a low pass filter or to return to a limited band signal.

Nyquist-Shannon sampling theorem

The theorem states that in order to faithfully reconstruct a sampled signal, the sampling rate must be at least twice the Nyquist frequency. It establishes a sufficient condition for a sample rate that permits a discrete sequence of *samples* to capture all the information from a continuous-time signal of finite bandwidth.

$$f_C > 2 f_N$$

<u>The original signal gets lost if the sampling rate is too low</u>. We introduce information not originally present, that is a distortion: **Aliasing**. The aliasing results in a superposition of the highest frequencies of the original signal spectrum, with the frequencies introduced in the sampled signal.



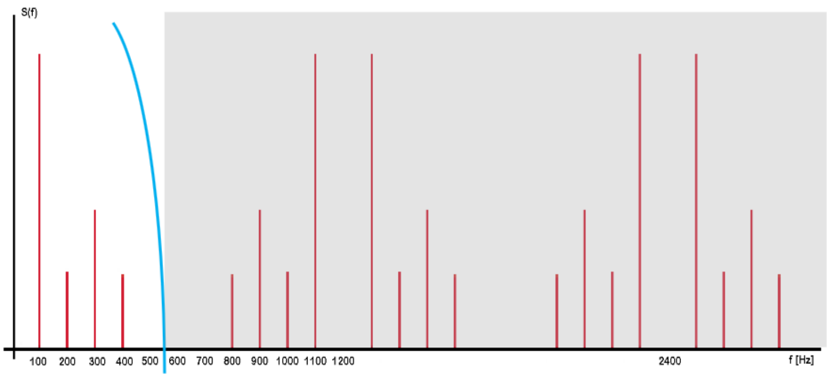
In the example the signal was sampled at a frequency of 650Hz, smaller than that suggested by the sampling theorem. Some spectral lines around the sampling frequency were superimposed on the original frequencies.

The higher the sampling rate, the better the quality? This has disadvantages because it creates costs and more waste.

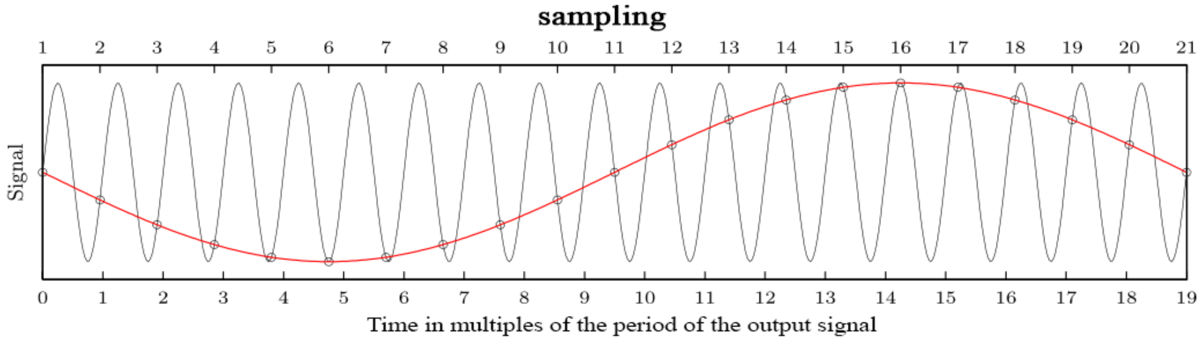
The right sampling rate: The right frequency corresponds to more than twice the maximum human frequency of 20 kHz. (Sample at 44.1 kHz)

Reconstruction

To reconstruct the original signal, all unwanted frequencies (not originally present) must be eliminated. For this a low pass filter is used which eliminates all frequencies above that of the original Nyquist.



In order for the original signal to be correctly restored, only frequency components that are smaller than the Nyquist frequency may appear in the signal to be sampled. However, if there are frequency components that are higher than the Nyquist frequency, these are interpreted as lower frequencies. The higher frequencies pretend to be another (lower) so to speak (see graphic), hence the name alias.

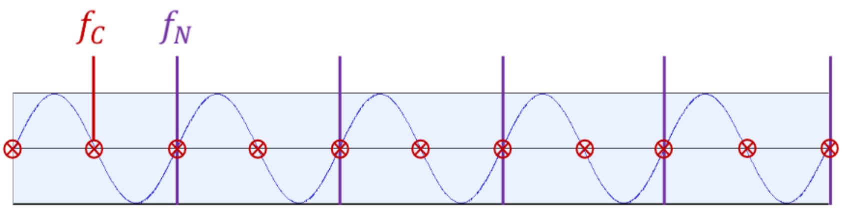


To avoid such aliasing effects, the input signal is filtered by a low-pass filter (anti-aliasing filter). The filter effect of this cutting off of the high frequencies can also be described by the terms treble cut, treble filter, high cut, and treble cut. This filtering must be done before digitization - a subsequent correction of alias effects is no longer possible.

<u>Note: Critical Sampling is achieved with close equality</u>

100 Hz pure tone

- $f_N = 100Hz \rightarrow f_C = 200Hz$



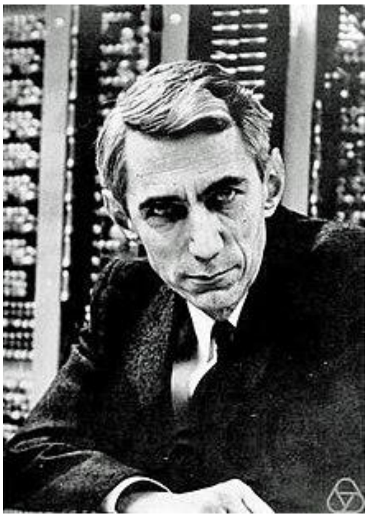
How will the signal be reconstructed with this sampling?

... null!

Sampling exactly twice the Nyquist frequency can cause critical sampling.

That is not necessarily the case. For example, if we had a 90 ° phase shift it would be possible to reconstruct the signal without any problem.

Claude Shannon (1916 –2001)



Claude Elwood Shannon was an American mathematician, electrical engineer, and cryptographer known as "the father of information theory". Amongst other things Shannon developed information entropy as a measure of the information content in a message, which is a measure of uncertainty reduced by the message, while essentially inventing the field of information theory. Shannon was interested in everything and creative. Side products of his professional activity include a juggling machine, rocket-driven frisbees, motorized pogo sticks, a machine for mind reading, a mechanical mouse (1950) that could be oriented in labyrinths using a simple memory consisting of relay circuits, and an early chess computer as early as the 1960s. He died of Alzheimer's disease.

Harry Theodor Nyquist (1889 –1976)

Swedish-born American Electronic engineer who made important contributions to communication theory. He worked in Bell's Industries and after retirement he worked for the army. As an engineer at Bell Laboratories, Nyquist did important work on thermal noise ("Johnson–Nyquist noise"), the stability of feedback amplifiers, telegraphy, fax, television, and other important communications problems. His early theoretical work on determining the bandwidth requirements for transmitting information laid the foundations for later advances by Claude Shannon, which led to the development of information theory.