

DSZOB

Digitálne spracovanie zvuku, obrazu a biosignálov

Introduction

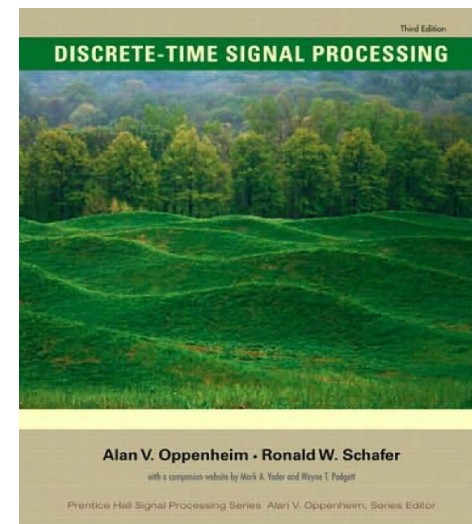
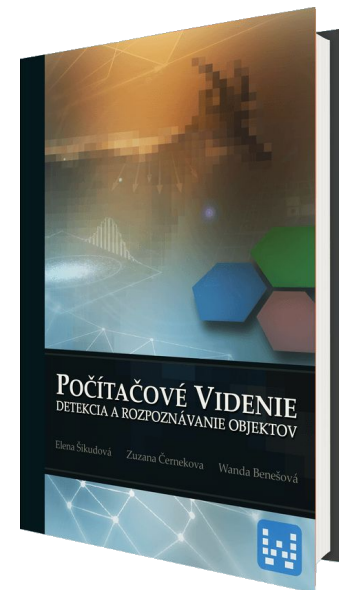
Prof. Ing. Vanda Benešová, PhD.
(Wanda Benesova)

Literature

<https://vgg.fiit.stuba.sk/kniha/L>

<http://cs.haifa.ac.il/~nimrod/Compression/Papers/Rabiner-Ch123.pdf>

third edition discrete-time
signal processing Alan V.
Oppenheim



Rules

40% exercises

10% test

50% final exam

Zápočet

- Min 18 bodov z cvičení (z maxim. 40)
- Účast' na cvičeniach (okrem ospravedlnených)
- V prípade ospravedlnenej neprítomnosti študent dopracuje po dohode s cvičiacim protokol
- Cvičenia študent venuje práci na zadaní
- Odovzdané protokoly

Cvičenia

Pre jednotlivé cvičenia je vždy samostatné zadanie

Úloha bude vypracovaná v Matlabe a študent vypracuje protokol

Odobzdanie protokolu po každom cvičení do začiatku ďalšieho cvičenia do AISu

Prezentácie protokolu :

- Náhodne! vybratí študenti (jeden až dvaja) sú na začiatku cvika vyzvaní aby prezentovali svoj odobzdaný protokol za predchádzajúce cviko. (max 5 minút)
- Cvičiaci im dáva doplnujúce otázky aby zistil či študent porozumel tomu čo robil.
- Prezentácie začnú od 3. cvika a prezentujú vždy dvaja na jednom cviku.
- Študent prezentuje 1x za semester - náhodne sa vyberá iba zo študentov čo ešte neprezentovali

Hodnotenie:

- | | |
|---------------------------|---|
| •1. protokol : | max 2 body |
| •Ostatné protokoly (2-10) | max 4 body za protokol (spolu max 9x4b) |
| •Prezentacia protokolu | max 2 body |

Spolu max 40 bodov

Signal , Signal processing

Signal , Signal processing

In signal processing, a **signal** is a function that conveys information about a phenomenon.

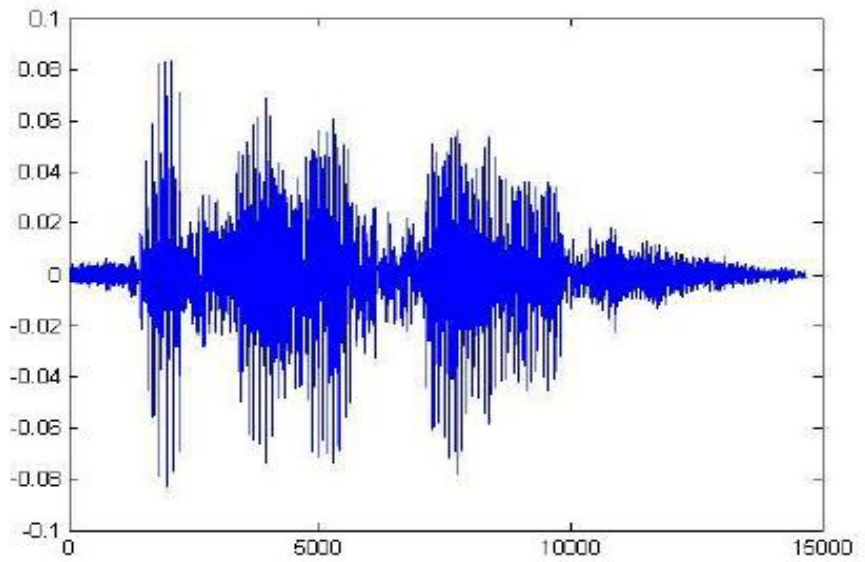
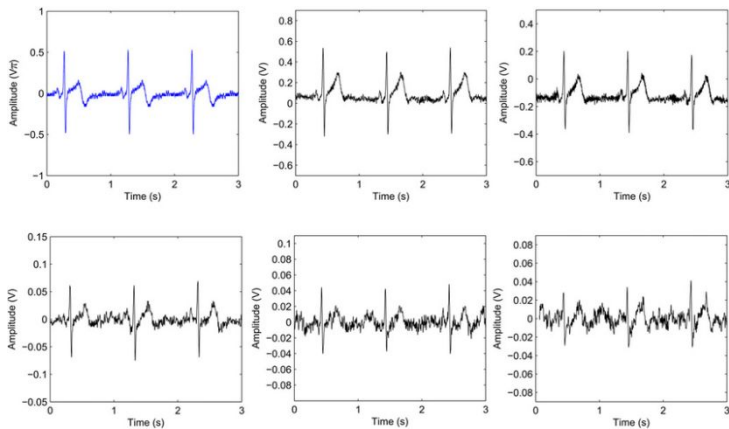
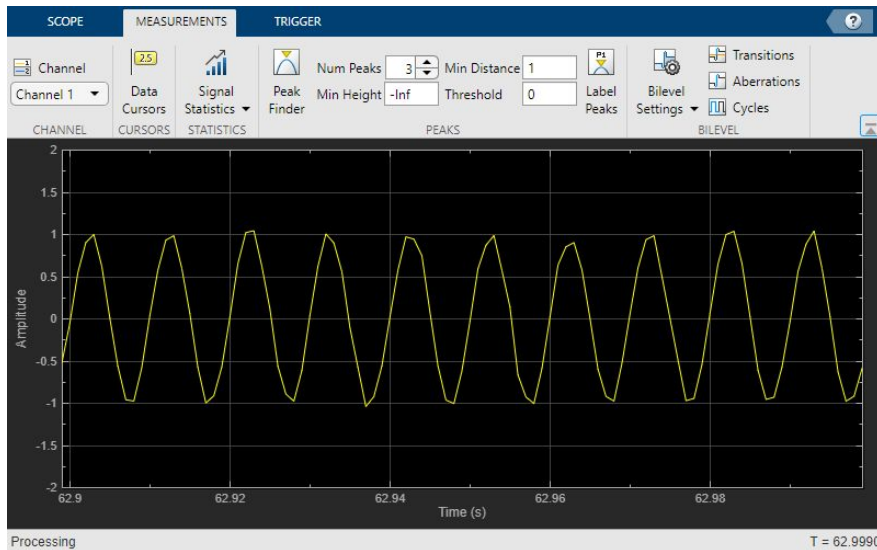
In electronics and telecommunications, it refers to **any time varying** voltage, current, or electromagnetic wave that carries information.

$$y = x(t) \quad \text{where } t \text{ is time}$$

Any quality, such as physical quantity that exhibits variation in space or time can be used as a signal to share messages between observers.

According to the IEEE Transactions on Signal Processing, a signal can be audio, video, speech, image, sonar, and radar-related and so on...

Signal visualization - examples



Analog signal vs. Digital signal

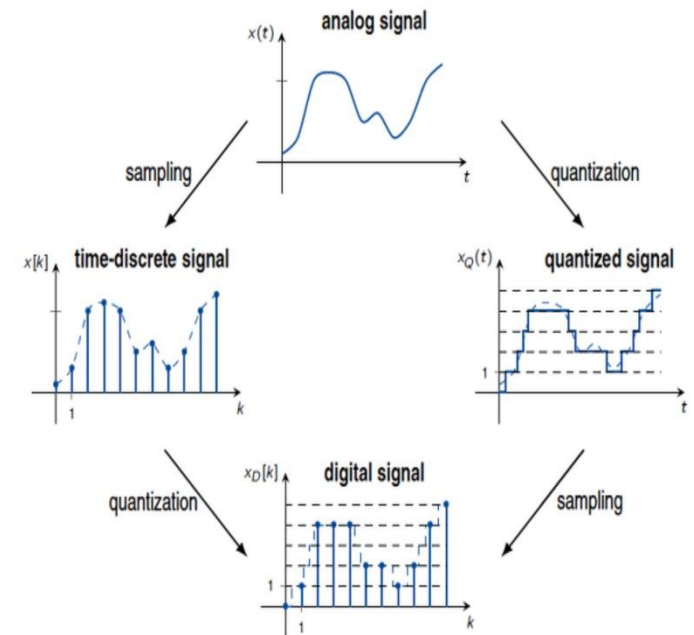
Analog signal is **any continuous signal**

$y = x(t)$ where t is time and $x, y, t \in \mathbb{R}$

...any information may be conveyed by an analog signal; often such a signal is a measured response to changes in physical phenomena, such as sound, light, temperature, position, or pressure...

Digital signal is a **sequence of discrete values**

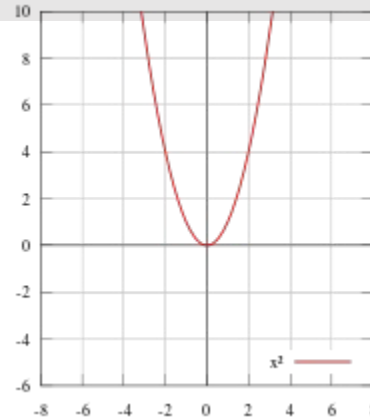
$Y_n = X(n \cdot \Delta T)$
where ΔT is sampling time period
and $X, Y \in \mathbb{Z}$



Even and odd signals

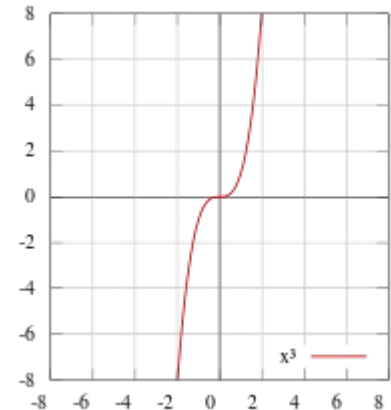
$$f(x) = x^2 f(x) = x^2$$

is an example of an even signal.



$$f(x) = x^3 f(x) = x^3$$

is an example of an odd signal.



An even signal satisfies the condition $x(t) = x(-t)$

An odd signal satisfies the condition $x(t) = -x(-t)$

Deterministic vs. random signal

Deterministic signals are those whose values at any time are **predictable** and **can be calculated by a mathematical equation**.

Random signals are signals that take on random values at any given time instant and **must be modeled stochastically**.

Periodic signals

A signal is said to be periodic if it satisfies the condition:

$$x(t) = x(t + T)$$

Analog:

$$x(n) = x(n + N)$$

Digital:

Where:

T [s] is fundamental time period,

$1/T=f$ f [Hz] is fundamental frequency.

A periodic signal will repeat for every period.

Signals

- 1D/2D Signal processing
 - Signal acquisition
 - Signal representation
 - Conversion analog to digital
- Sound processing
 - Audio signal
 - Speech signal processing
- Image processing
 - Image
 - Video
- Biosignals
 - 1D signals
 - Medical imaging

Methods of signal processing

Spectral analysis

- Fourier transform

- Cosine transform

- Wavelet transform

Statistical methods

Signal processing in the time/space domain

Signal 1D, 2D

1-Dim signal

Audio signal

Frequency range 20 Hz – 20 kHz

Speech - Frequency range 20 Hz – cca. 4KHz

Next type of signals :

seismic, EKG, EEG, ultrasound, ...

2-Dim signal

Image

1D Signal Processing Introduction

Audio signal

- Sampling
- Processing
- Coding
- Data compression
- Processing in frequency domain
- Speech representation

Multimedia data compression standards

MP3 (MPEG1 layer 3 MP3)

Advanced Audio Coding (AAC)

JPEG

JPEG 2000

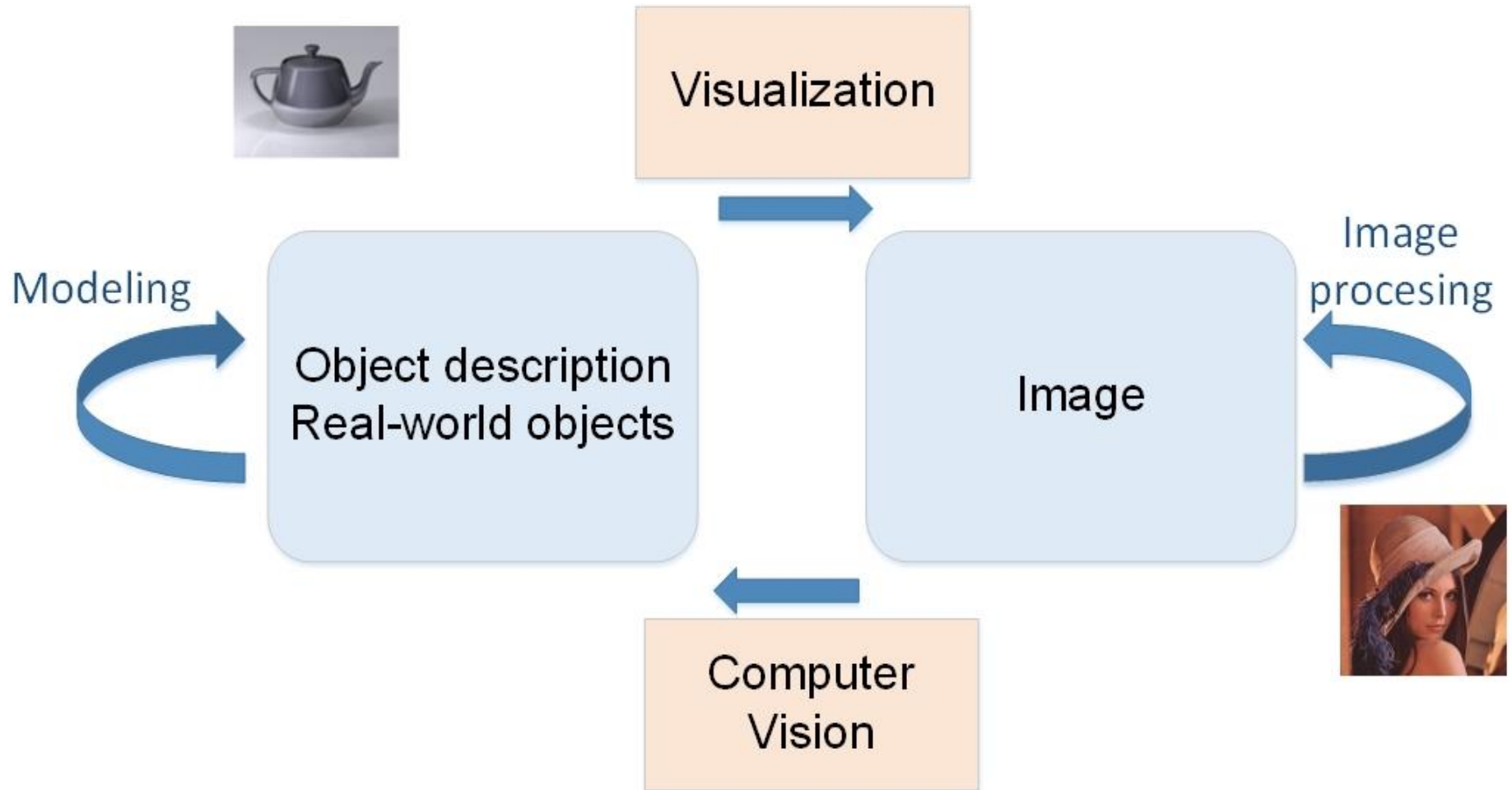
MPEG-4

MPEG 7

....

Computer graphics
Computer vision

Modeling, Computer graphics , Computer vision, Visualization



Raster Images vs. Vector Images

Raster Images

- pixel-based images
- camera photo, bitmap and any pixel-based image.

Vector Images

- vector images or objects which are drawn by a mathematical algorithm that defines points and curves rather than pixels.
- scalable
- not resolution dependent

What is an (Raster) Image ?

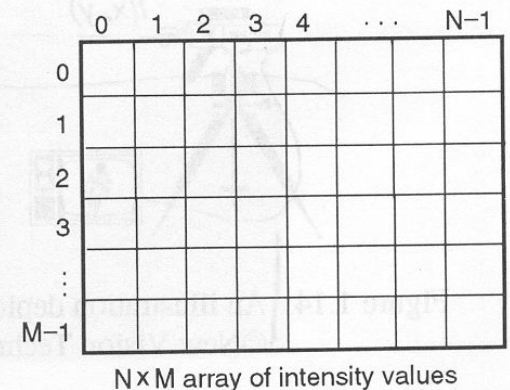
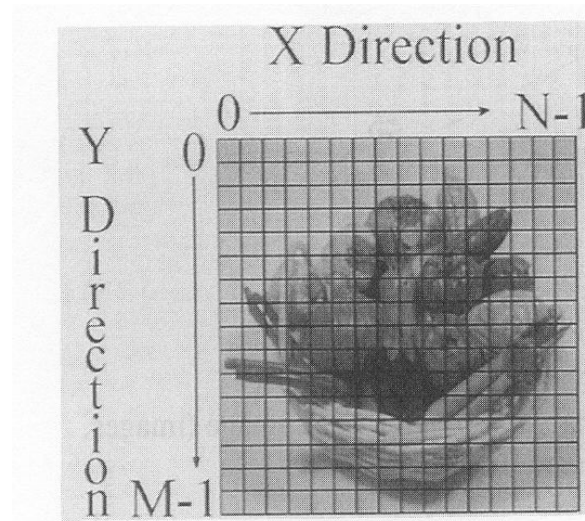
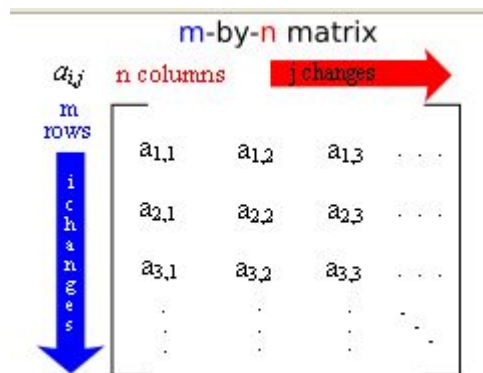
- A gray-scale image may be defined as a two-dimensional function $f(x, y)$
- x and y are:
 - *spatial (plane) coordinates*
- the amplitude of f at any pair of coordinates (x, y)
 - *intensity or gray level* (..jas)

Matrix Representing of Digital Image

The result of sampling and quantization is a matrix of numbers.

=> *matrix operations*

Coordinate convention :



Dimension of Image Matrix

- Gray-scale image $m \times n$
- Colour image $m \times n \times 3$
- ...
- Multi-spectrale image $m \times n \times x$

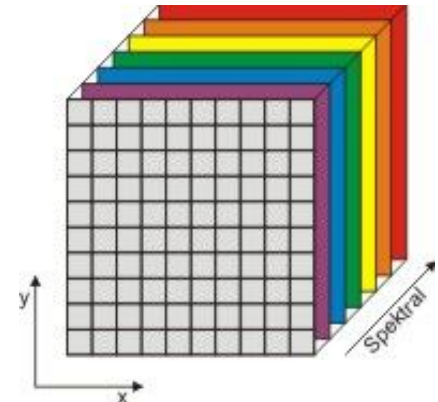


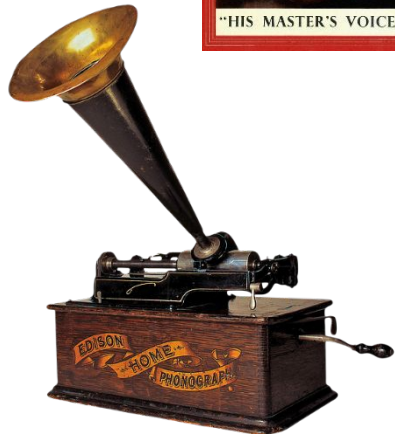
Image sequence (video)

- + time (...next dimension...)
- Frequency of images [frame/sec]
 - Human sensing ~ aprox. from 10 frames/sec.

Analog audio

History of the Record Player

- Thomas Edison invented the **phonograph** in 1877
- Emile Berliner took record players to the next level. He dubbed his creation the "**gramophone**" and secured a patent for the device in 1887. The gramophone was made of hard rubber and **shellac** before being constructed with vinyl.
- Record players became extremely popular in the 60s and 70s when Dual released the first turntables to provide stereo playback. High-fidelity sound reproduction hit the scene and motivated countless people to add a record player to their home.



45 MINUTES OF MUSIC FROM A SINGLE RECORD
... ANOTHER "FIRST" BY COLUMBIA RECORDS

COLUMBIA
LONG PLAYING
MICROGROOVE
RECORD

THIS COLUMBIA (LP) PLAYER ATTACHMENT
plays LP records through your present radio or phonograph

Finer tone quality! So listen you'll hardly believe you're listening to a record. Tone, tone, tone. High notes are heard without distortion. And practically no surface noise!

Uninterrupted music! Many records are recorded either on 2 sides or 3 sides of a single LP record. As for the more interesting records.

More than twice as much music for your money! Columbia LP Records save you up to 50% per selection over conventional 78s. You'll have much better your money goes to... how much better you'll have a fine record collection.

Non-splittable Vinyl! Made from a new plastic material, a thing of the past, another source of savings. And super-records! No 78s means less noise.

Saves storage space! Every inch of shelf space holds 3 hours of music.

Over 400 selections already in catalog! Gramophones, recorders, record collections, play, store, children's records... 100 different models! A beautiful collection of convenience by your favorite store who record exclusively for Columbia. Many new releases every month.

You only need to add a down speed player attachment to your present set to enjoy the play of LP Records. The handsome Columbia Player attachment is quickly installed, extremely practical, and provides a perfect reproduction of Columbia LP Records. It insures your present set to play both LP and your regular records. This amazing development! One set playing only 1/2 of an hour! Your savings is a lot! LP Records pay for it. See your dealer today!

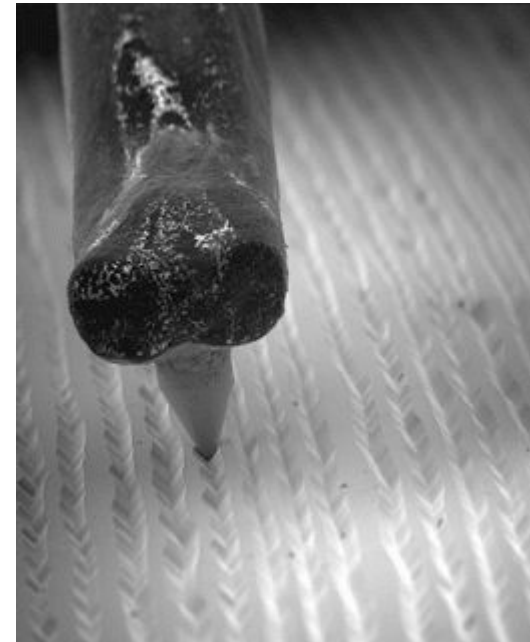
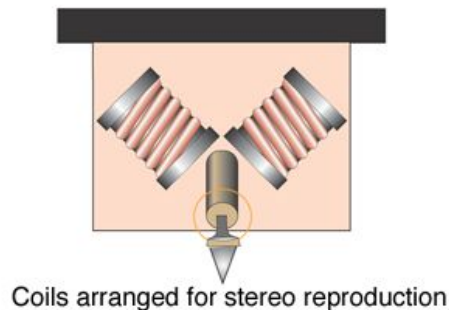
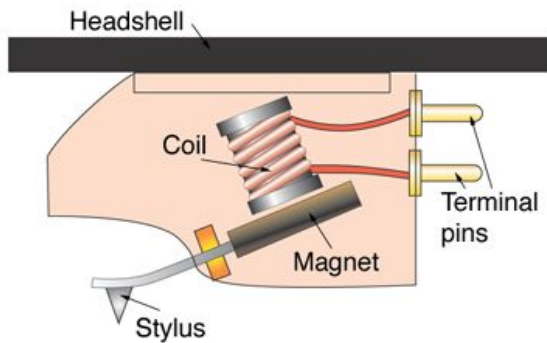
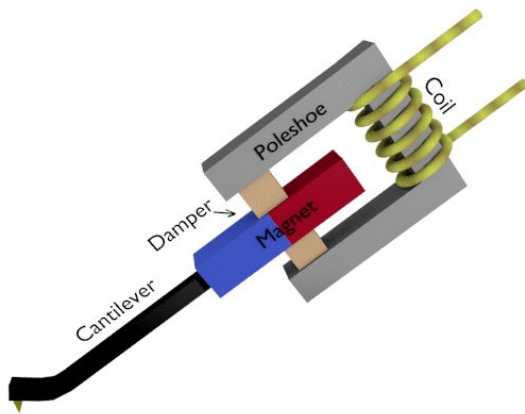
Columbia Records and the LP logo are trademarks of Columbia Records Inc. © 1955 Columbia Records Inc. New York, N.Y.

Electromagnetic cartridge

A pickup system consists of a stylus and a cartridge. These are the components of the turntable that **convert the sound waves etched on a record into an electrical signal**.

Two types of pickup systems are:

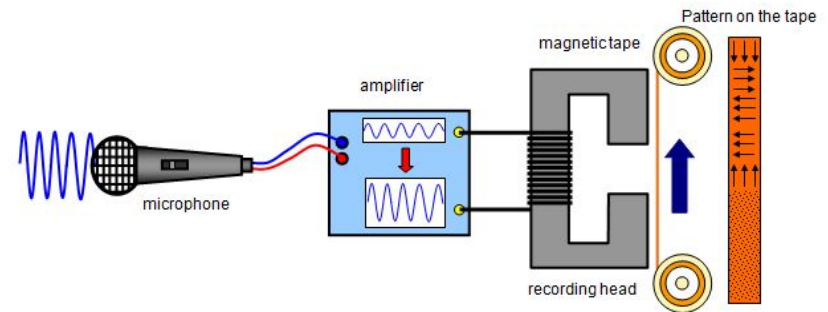
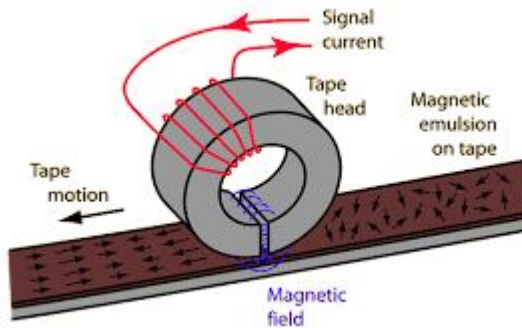
- Piezo-electric pickup systems
- Magnetic pickup systems



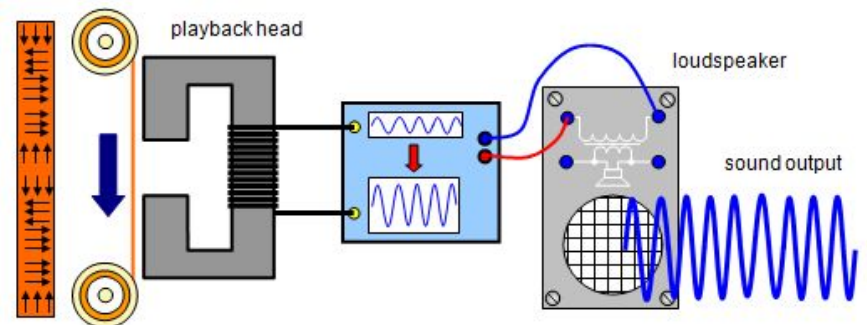
Magnetic tape recording



- the microphone picks up the sound wave and converts to voltage
- the amplifier amplifies this voltage
- the output from the amplifier is fed to the recording head where a changing magnetic field is produced
- this changing magnetic field arranges the grains of iron oxide on the tape into a pattern that "mirrors" the changing sound received by the microphone.

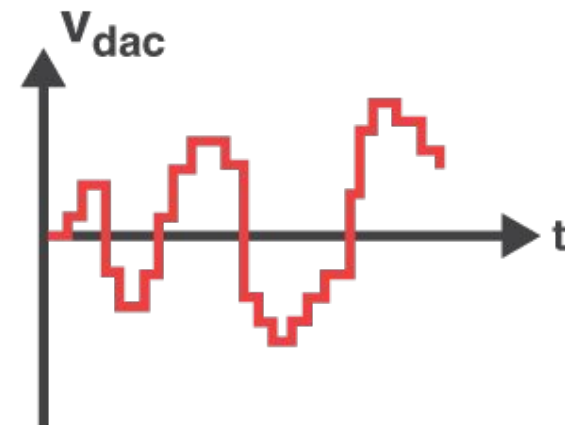
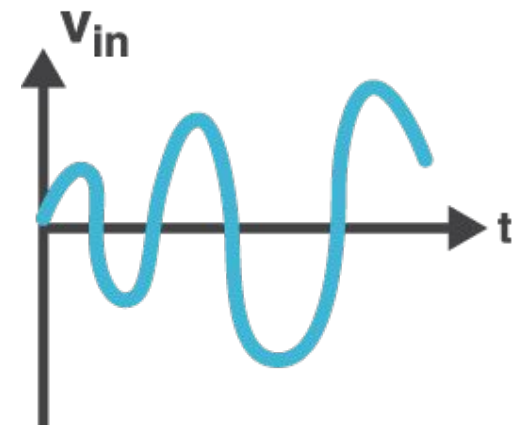
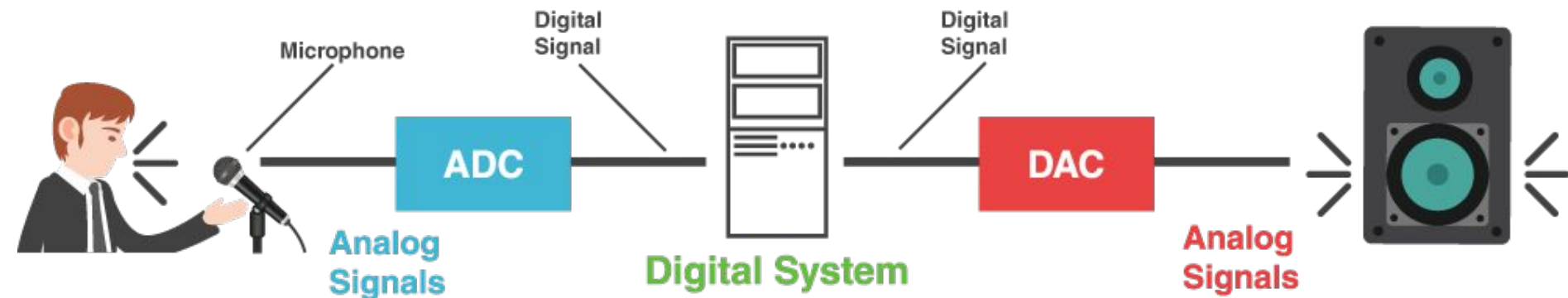


- The reverse happens at the playback stage: A changing magnetic field on the tape is converted to a voltage by the playback head, this is amplified by the amplifier and then fed to a loudspeaker.



Conversion: Analog =>
digital

Analog signal, digital signal



Analog amplifier, digital amplifier

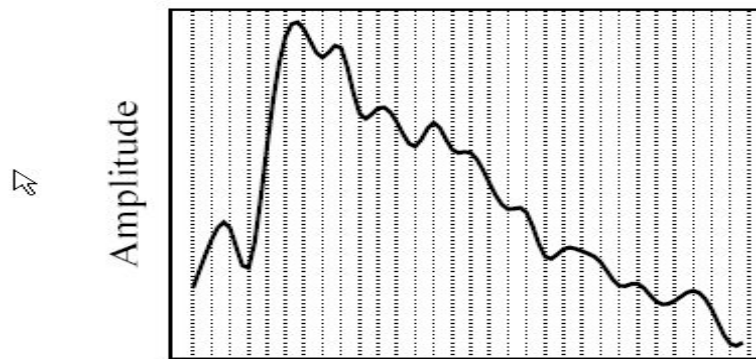
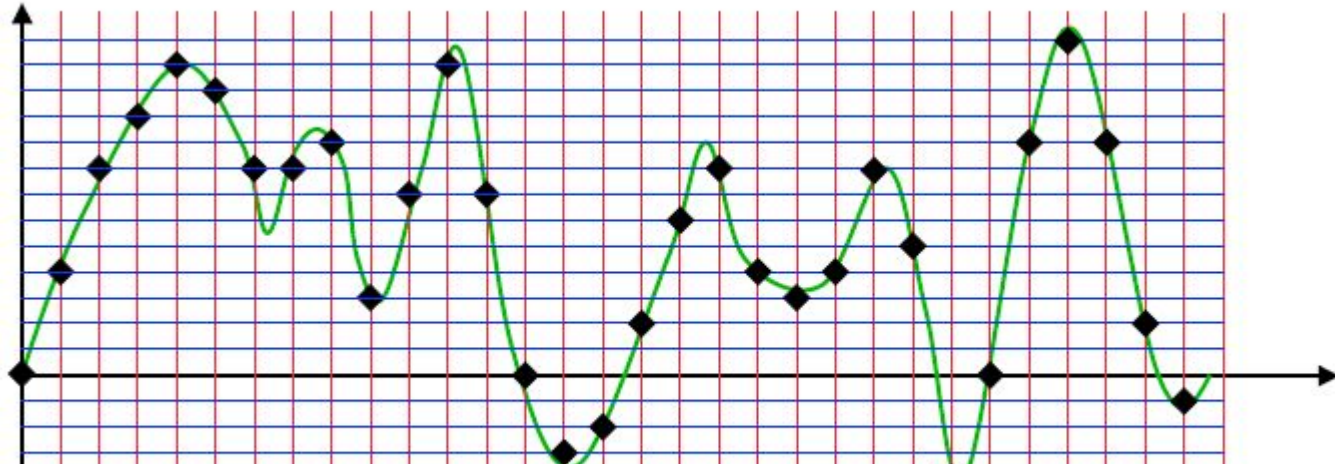
Signals in analog amplifier



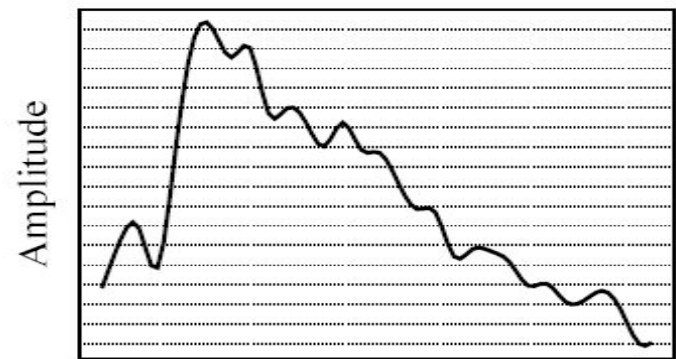
Signals in digital amplifier



Digitalisation: sampling + quantisation



(a)



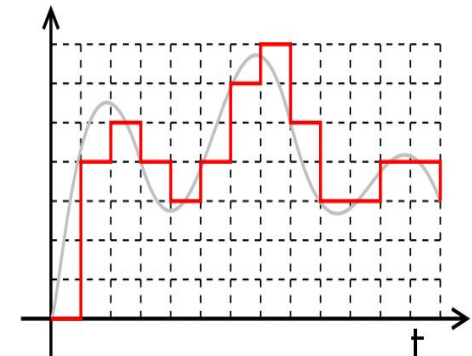
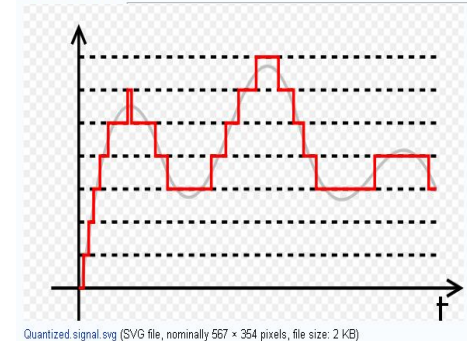
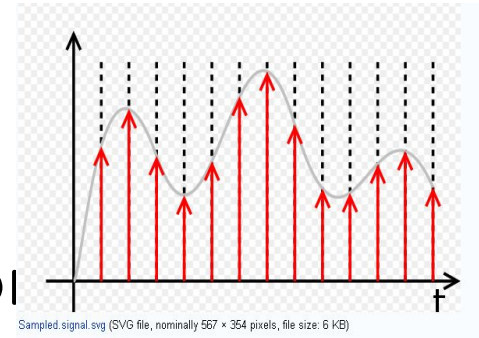
(b)

Digitalisation: sampling + quantization

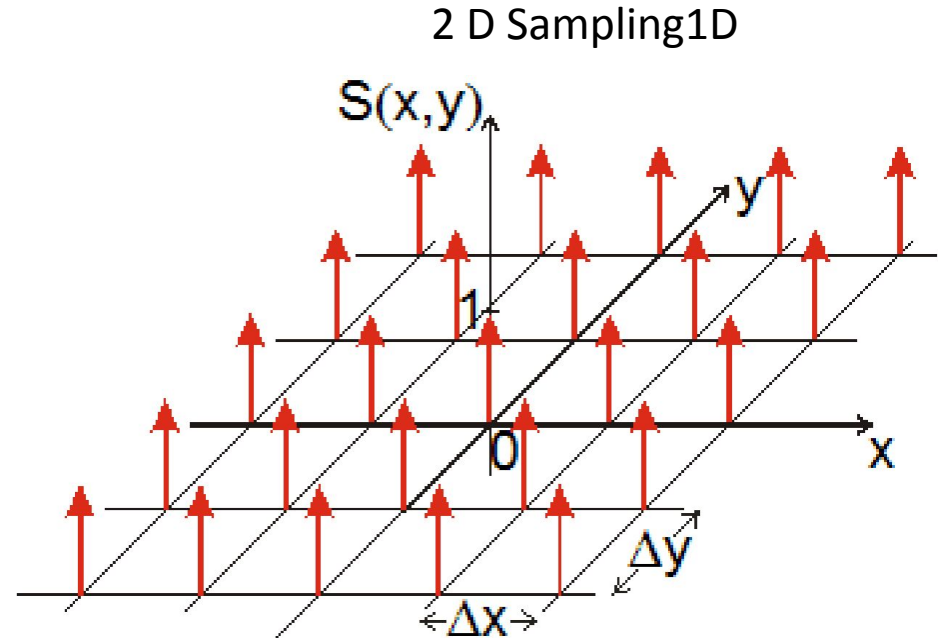
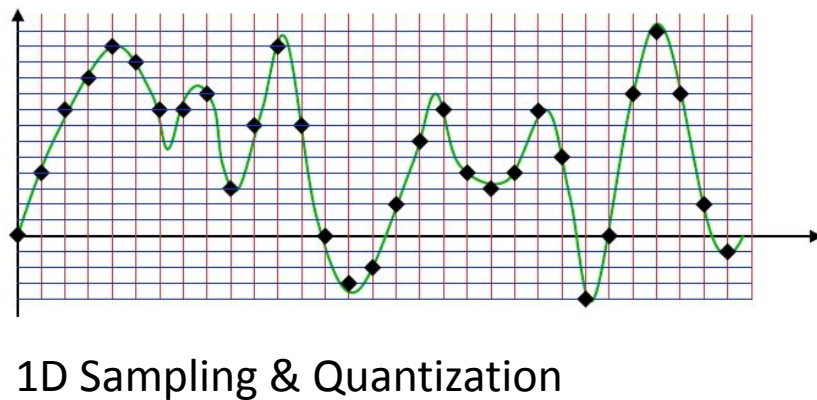
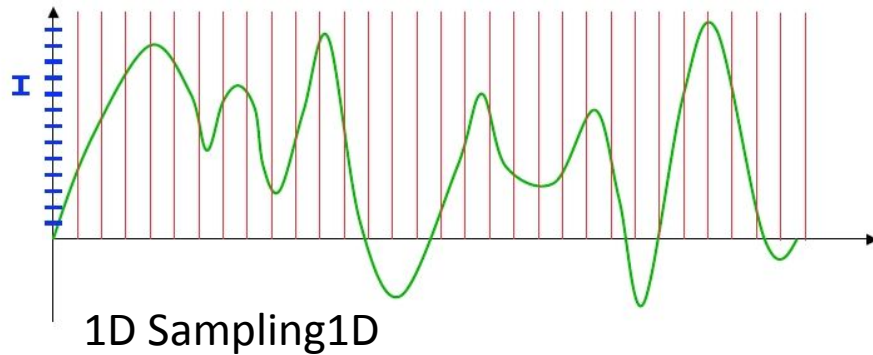
- Sampling
 - Spatial resolution
 - (pixel resolution...)
 - Nyquist–Shannon sampling theorem

+

- Quantization
 - Uniform /non uniform
 - Dynamic range
 - 8 bit -> 256 levels



Sampling + Quantization



Quantization

Dynamic range

ratio between the largest and smallest possible values of a changeable quantity (such as in sound and light)

Dynamic range in Decibel dB:

$L_A \text{ [dB]} = 20 \log_{10} (A_{\max}/A_{\min})$ where A is amplitude

$L_P \text{ [dB]} = 10 \log_{10} (P_{\max}/P_{\min})$ where P is power

Unit used for dynamic range:

Decibel dB - analog , digital

No of Bits - digital

Examples :

Dynamic range in dB	49,8 dB	73,7 dB	85,7 dB	97,6 dB
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No of Bits	8 bit	12 bit	14 bit	16 bit
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(Every bit ~ 6 decibels)

Decibel dB

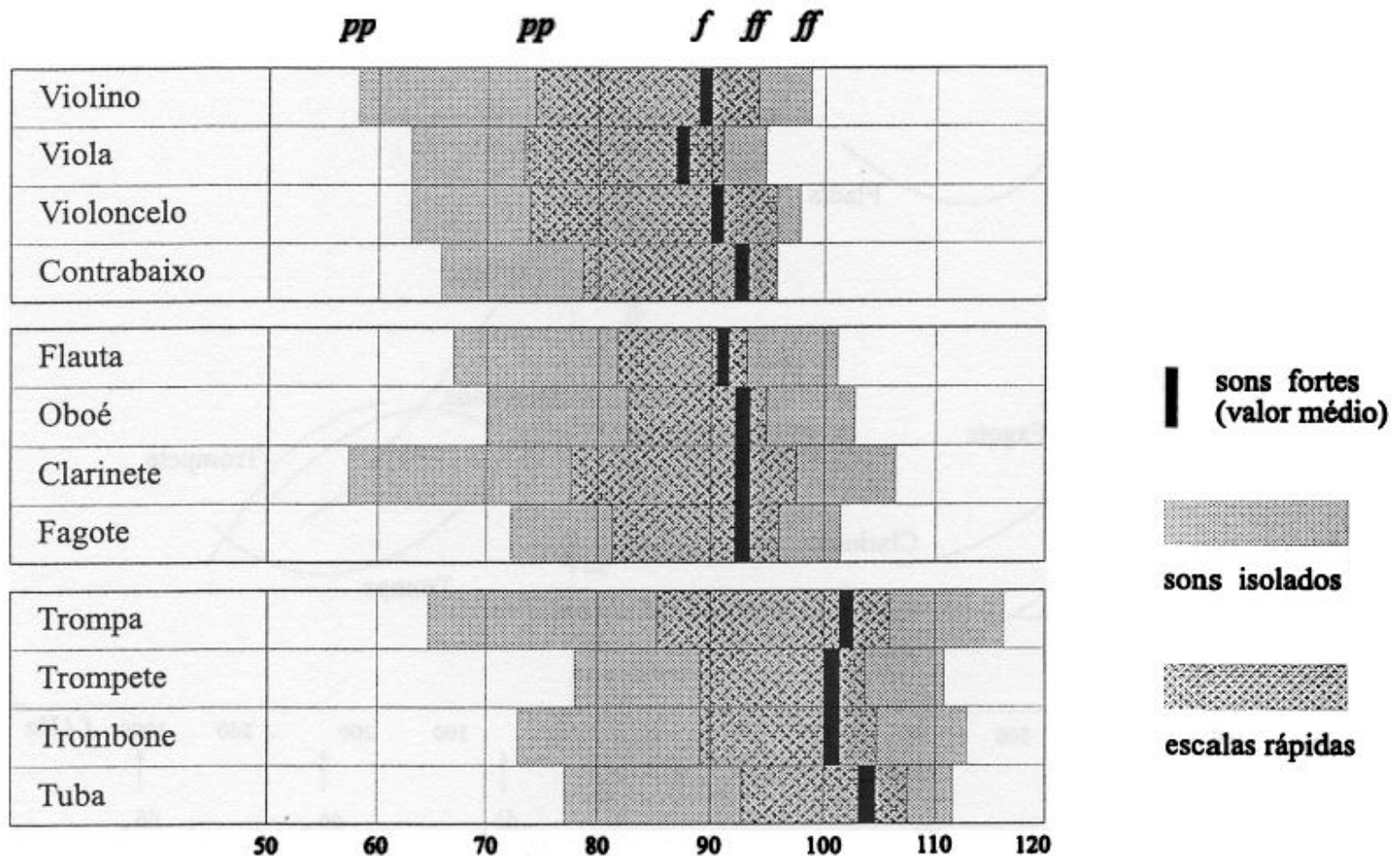
$$L_A \text{ [dB]} = 20 \log_{10} (A_{\max}/A_{\min}) \quad \text{where } A \text{ is amplitude}$$

$$L_P \text{ [dB]} = 10 \log_{10} (P_{\max}/P_{\min}) \quad \text{where } P \text{ is power}$$

Since the decibel is a logarithmic quantity, it is especially good at representing values that range from very small to very large numbers.

The logarithmic scale approximately matches the human perception of sound.

Dynamic Range in Sound



Dynamic range of the sound power of orchestral musical instruments in dB

Dynamic Range in a Grayscale Image

The number of grey levels typically is an integer power of 2:

$$L = 2^k \quad \text{Where } k - \text{No of Bits}$$

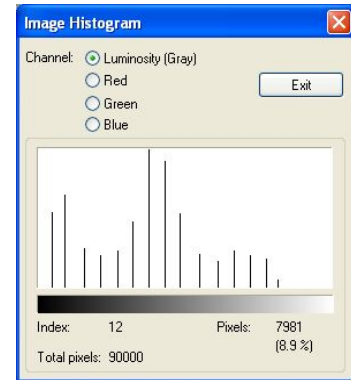
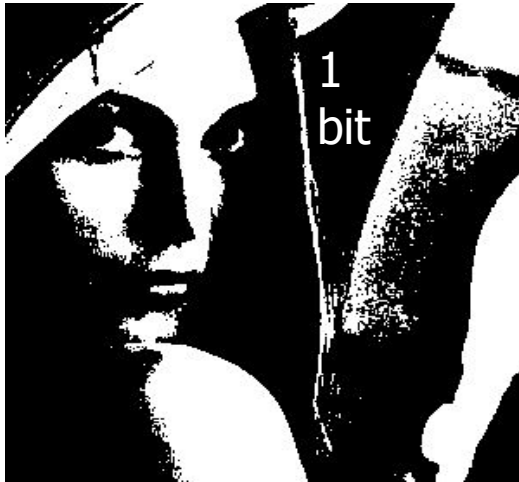
We assume that the discrete levels are equally spaced and that they are integers in the interval $[0, L-1]$.

Grey-level resolution refers to the smallest discernible change in grey level

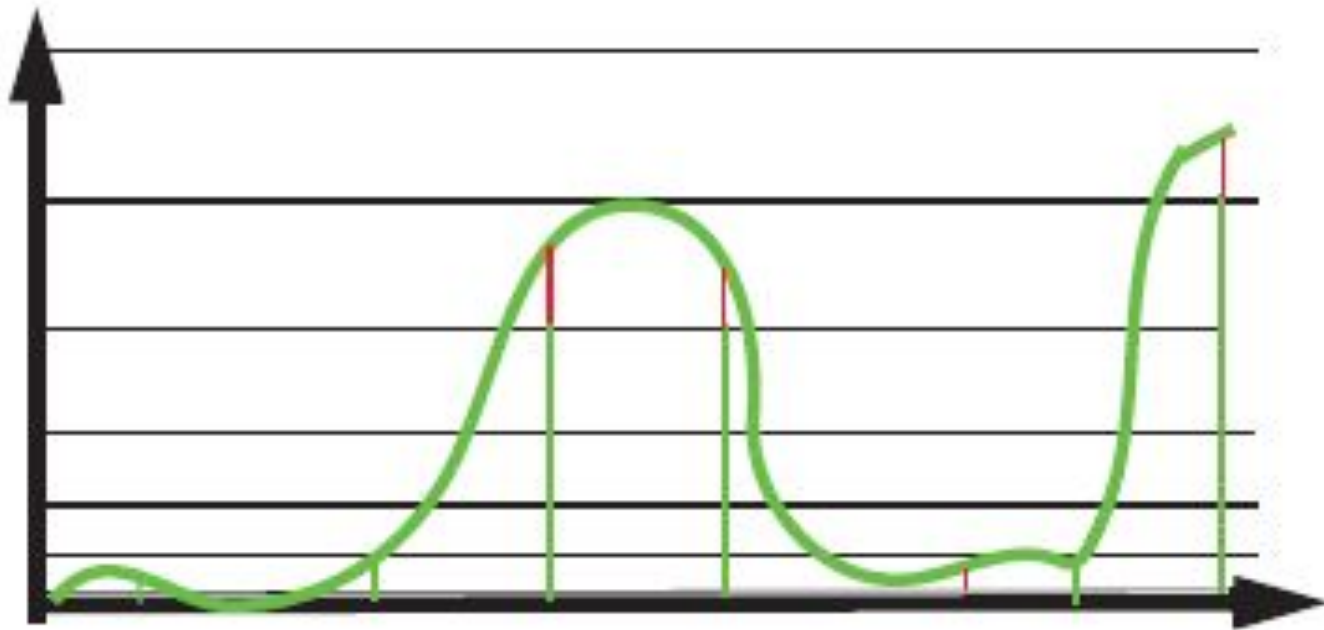
LSB (Least Significant Bit)

The smallest measurable signal is determined by the **Least Significant Bit (LSB)** value of the analog-to-digital converter (ADC). Signal changes smaller than the LSB value will not be able to force the ADC to the next digital level.

Dynamic range - Lena example

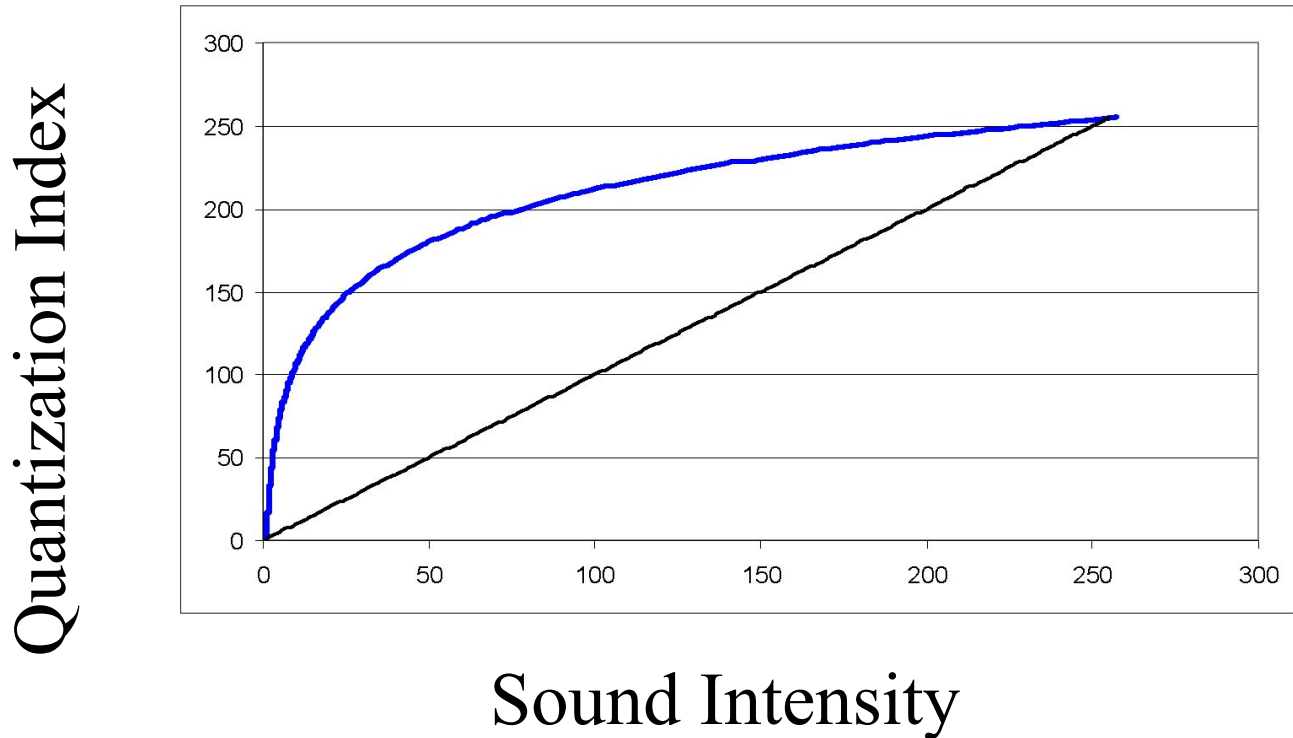


Linear / Non-uniform Quantization



Perceptual Quantization (μ -Law)

Intensity values logarithmically mapped over N quantization units

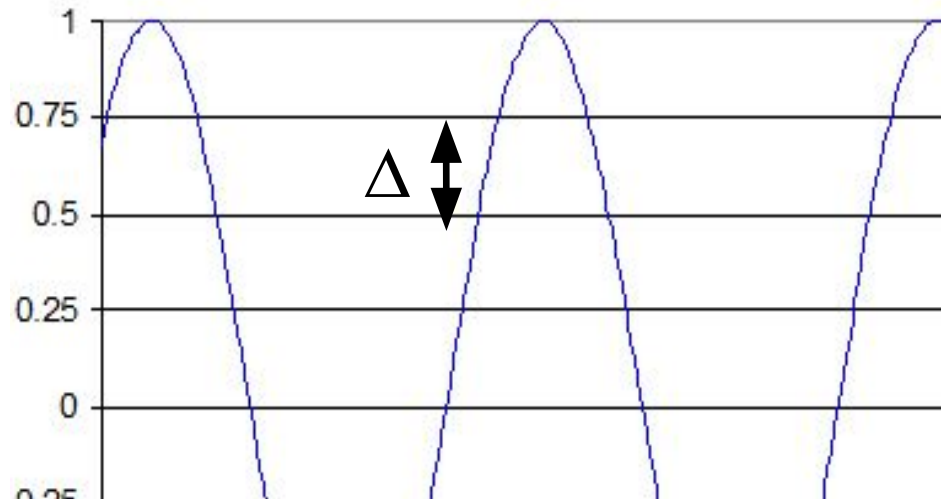


Quantization Error

Difference between actual and sampled value
amplitude between $[-A, A]$
quantization levels = N

$$\Delta = 2A/N$$

e.g., if $A = 1$,
 $N = 8$, $\Delta = 1/4$



Noise level in signal

Noise level in signal : L_{noise}

amplifier noise, electrode noise...

Noise level in signal should be $<$ Least Significant Bit (LSB)

Signal to Noise Ratio

The signal to noise ratio is defined as the ratio of signal power to noise power: $SNR = P_{\text{signal}}/P_{\text{noise}}$
or $SNR = (A_{\text{signal}}/A_{\text{noise}})^2$ where A is the root-mean-square amplitude.

It can be expressed on a logarithmic scale in decibels dB

$$SNR = 10 \log_{10}(P_{\text{signal}}/P_{\text{noise}}) \quad \text{or} \quad SNR = 20 \log_{10}(A_{\text{signal}}/A_{\text{noise}})$$

e.g. a SNR ratio of 0 dB means that the amplitude of the signal and the noise fluctuations are similar;

60 dB means that the rms amplitude of the signal is 1000 times that of the noise.

Sampling

Sampling: mathematical point of view

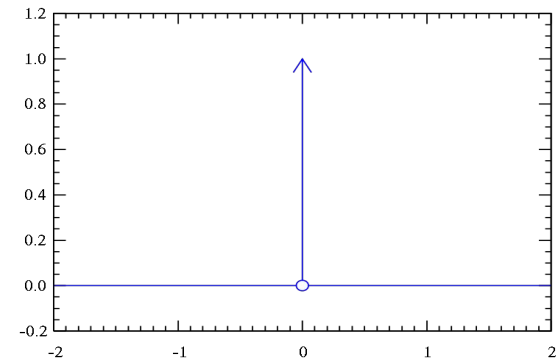
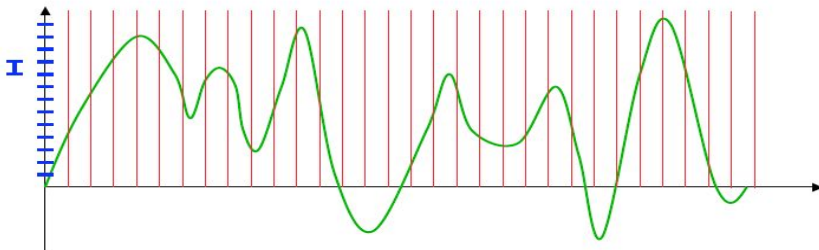
The mathematical operation of sampling multiplicative operation:

$f(t)$ is multiplied by a Dirac sampling function $s(t; \Delta T)$, consisting of a set of delayed Dirac delta functions

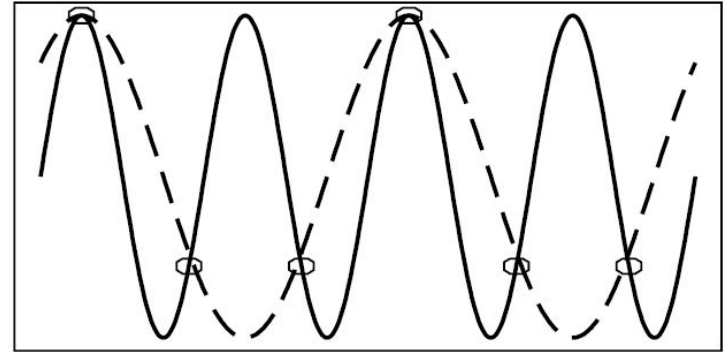
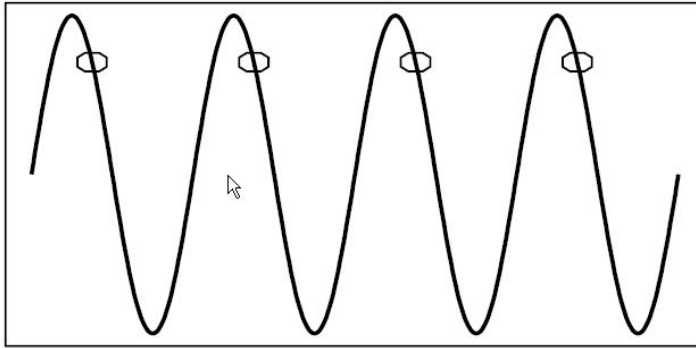
$$s(t; \Delta T) = \sum_{n=-\infty}^{\infty} \delta(t - n\Delta T)$$

Dirac delta function: 1) $\int_{-\infty}^{\infty} \delta(x) dx = 1$.

$$2) \delta(x) = \begin{cases} +\infty, & x = 0 \\ 0, & x \neq 0 \end{cases}$$



What is the correct sampling rate?



Sampling theorem

...discovered in various forms by Shannon, Nyquist, Whittaker and Kotelnikov:

If a system uniformly samples an analog signal at a rate that exceeds the signal's highest frequency by at least a factor of two, the original analog signal can be recovered exactly from the discrete values produced by sampling.

Nyquist sampling frequency

The sampling frequency must be at least twice the highest frequency contained in the signal.

In mathematical terms:

$$f_s \geq 2 * f_m$$

where f_s is sampling frequency and f_m is the maximum frequency in the signal

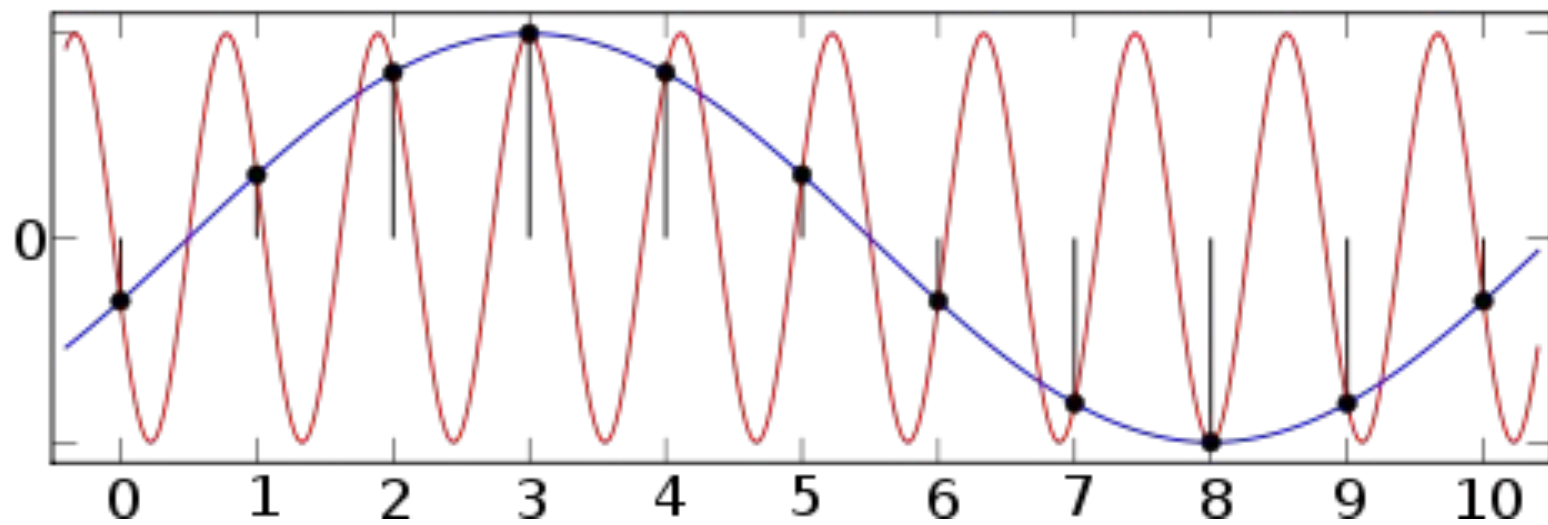
Aliasing

When the Nyquist criterion is violated,

frequency components above half the sampling frequency appear as frequency components below half the sampling frequency, resulting in an **erroneous representation of the signal**.

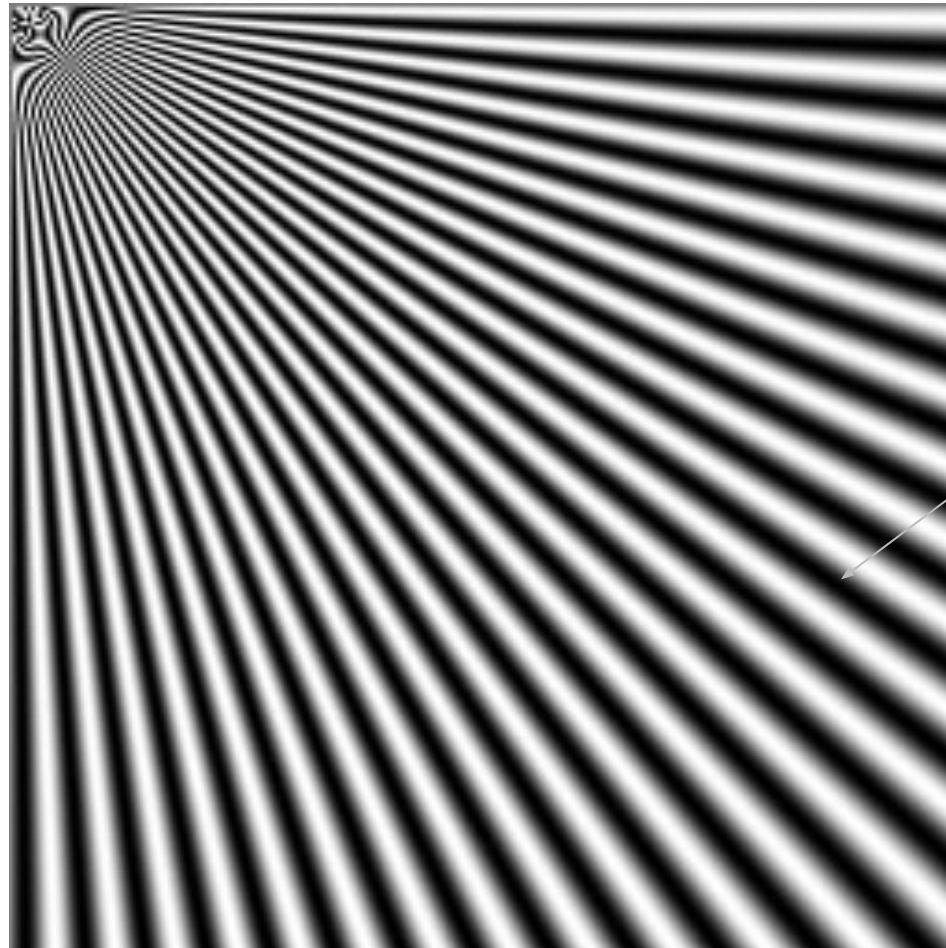
Aliasing

Example:



Aliasing in 2D signal

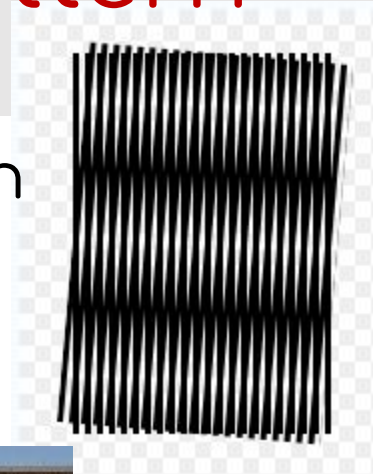
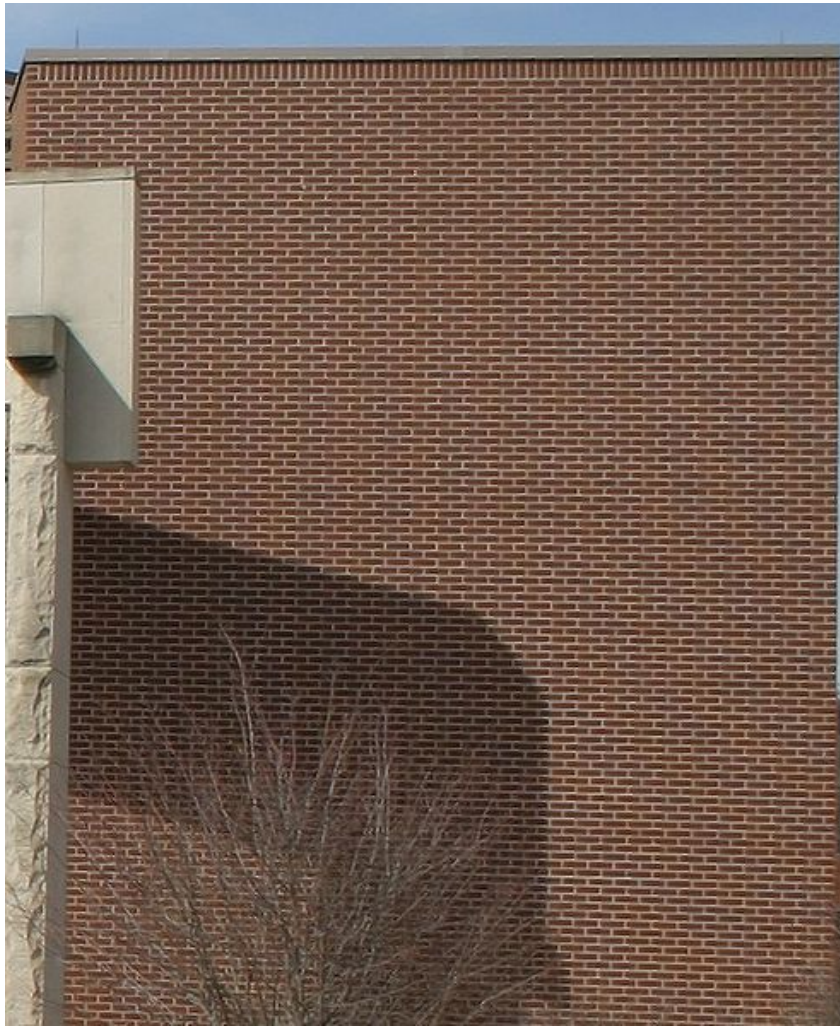
High
frequencies



Low
frequencies

Subsampled image – Moiré pattern

- Subsampled image showing a Moiré pattern



Anti-aliasing filter (analog filter)

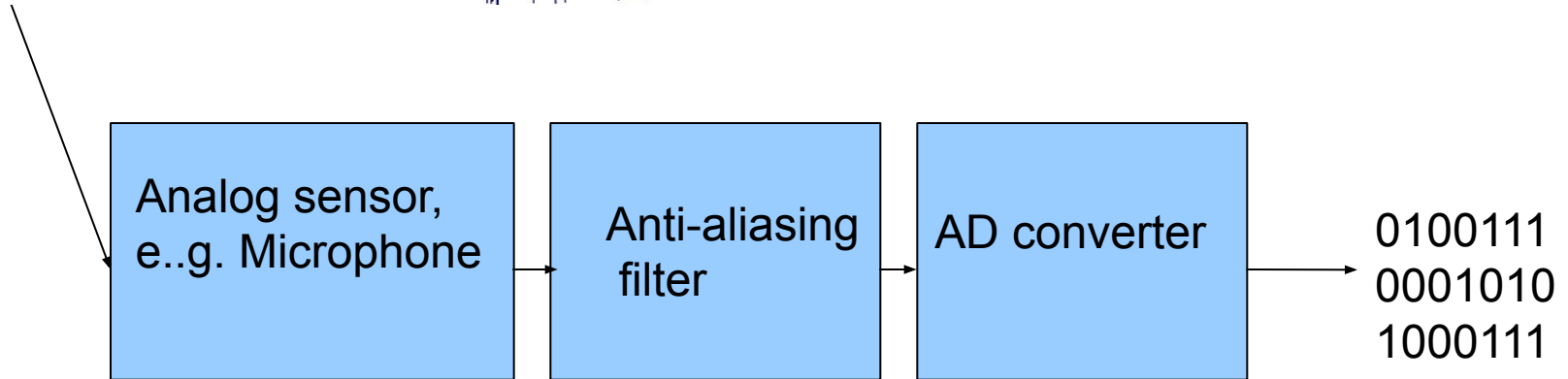
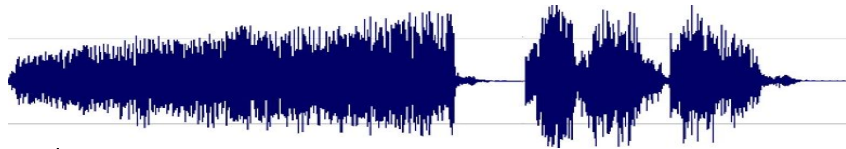
The goal – to reduce signal frequency to half of maximum sampling frequency:

low-pass filter removes higher-frequencies

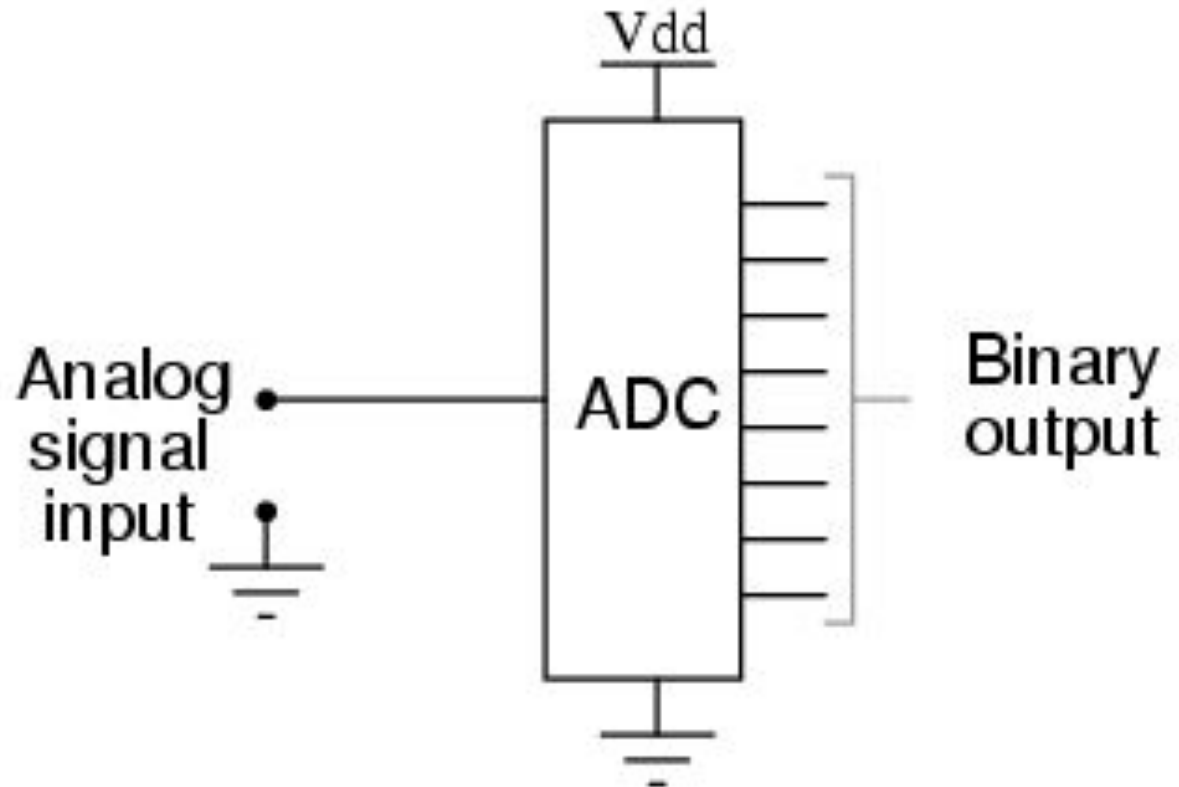
e.g., if max. sampling frequency is 22kHz, must low-pass filter a signal down to 11kHz

Analog to digital conversion – technical point of view

Input – analogue signal



AD converter – example (recap.)



analog devices a-d

audio a-d