

Human Body as a signal source

Signal processing

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Human Body as signal source

Signals in medicine

Information content of signals

Signal detection - transducers

Explained through examples
there are endless possibilities

Signals in medicine

$$H = \sum p * \log_2 \frac{1}{p}$$

Signal is something which carries Information

Information content in Bits

Human body as signal source: everything which is a signal, and comes from the body

Here in the cartoon:

Information : Head or Tail?

Signal:

- Optical: we simply look at the coin, and see the image
- Digital: after encoding: 1/0



Transmitting information – information coding

in general

Information source

encoding

Transmission channel

decoding

Information receiver
destination

an example

Which side is up?

Sides : Head or Tail
into Numbers: 1,0

Speech, waves in the air, sms

1,0 → head, tail

Decide who wins

$$H = p_{tail} * \log_2 \frac{1}{p_{tail}} + p_{head} * \log_2 \frac{1}{p_{head}} = \frac{1}{2} * \log_2 \frac{1}{\frac{1}{2}} + \frac{1}{2} * \log_2 \frac{1}{\frac{1}{2}} = 1 \text{ [Bit]}$$

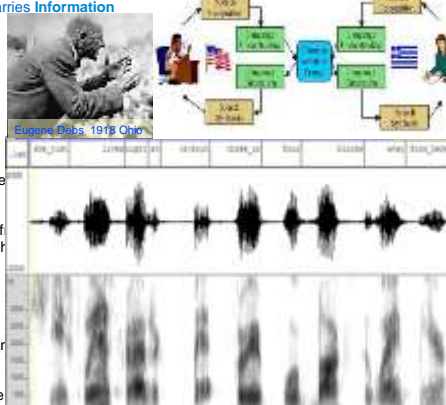
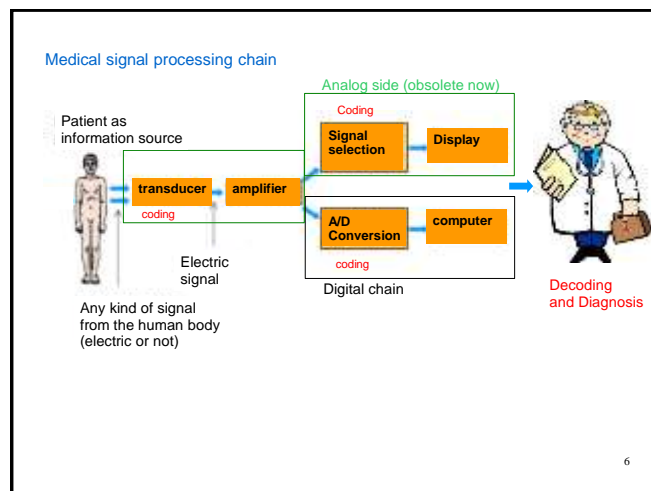
Signals in medicine
Signal is something which carries Information

Here in speech:

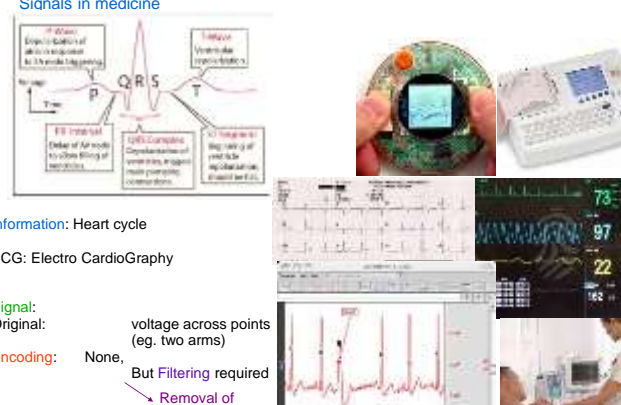
Information : „what we say“

Signal:

- Audio: pressure wave the air
- encoding: electrical signal from Microph
- encoding: formal grammar
- decoding: electrical to Mechar (loudspeaker)
- decoding: natural language underst

Signals in medicine



Information: Heart cycle

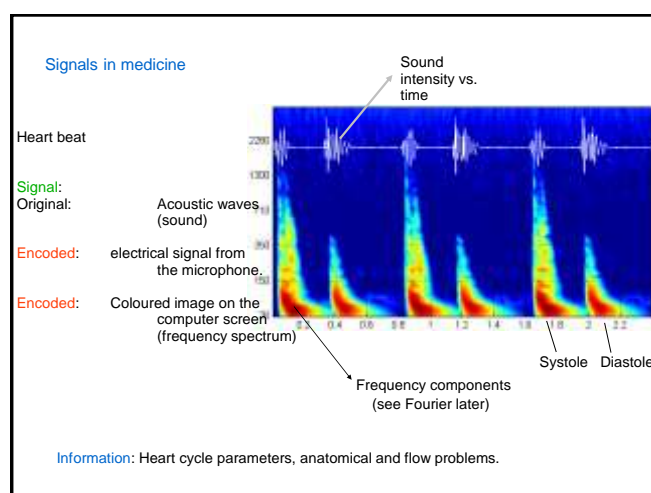
ECG: Electro CardioGraphy

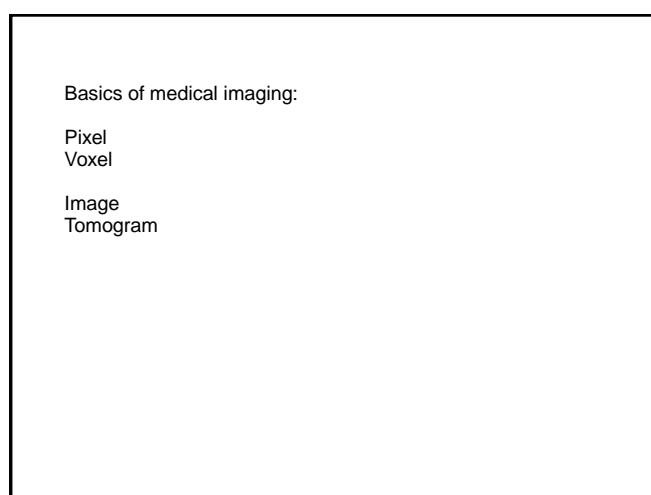
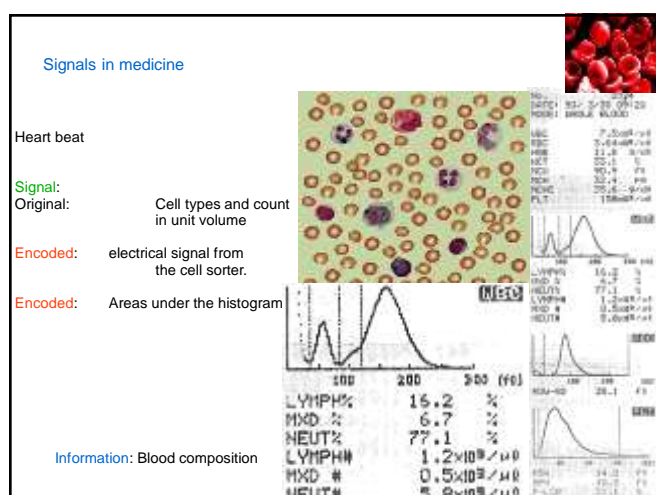
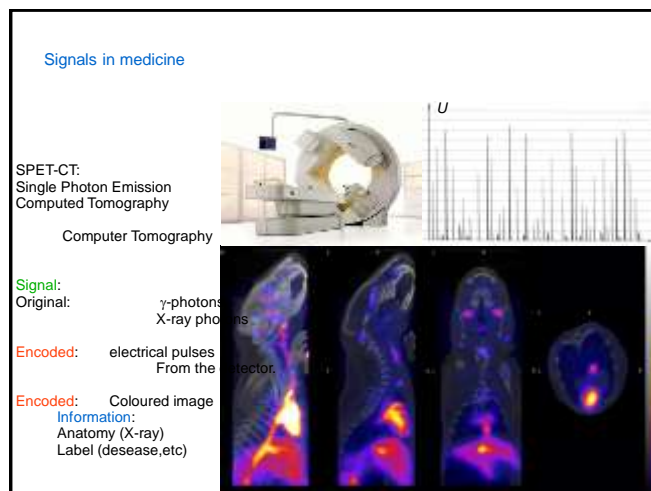
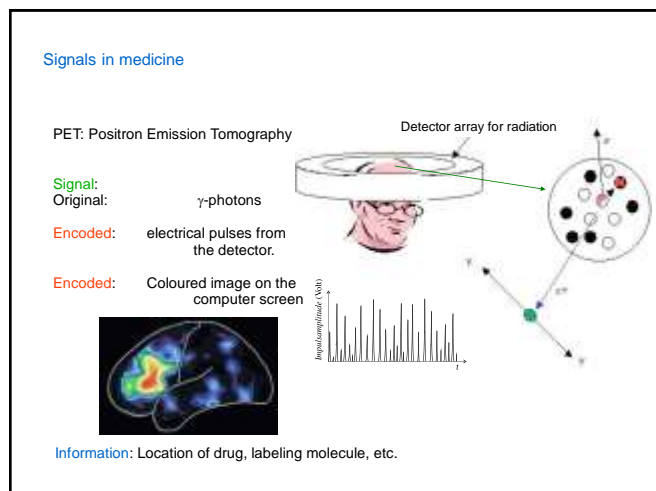
Signal:

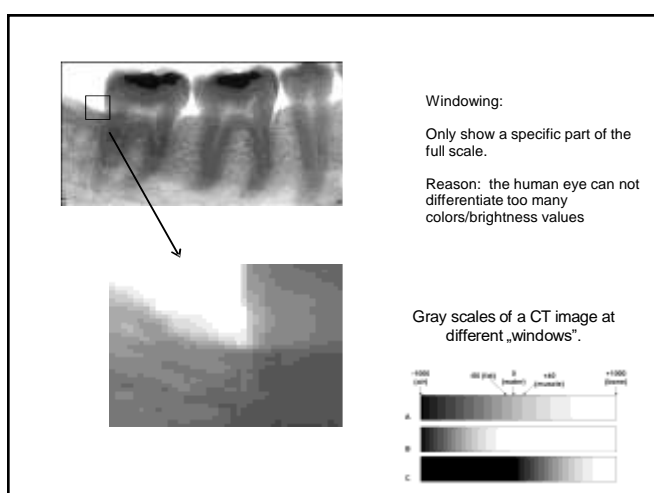
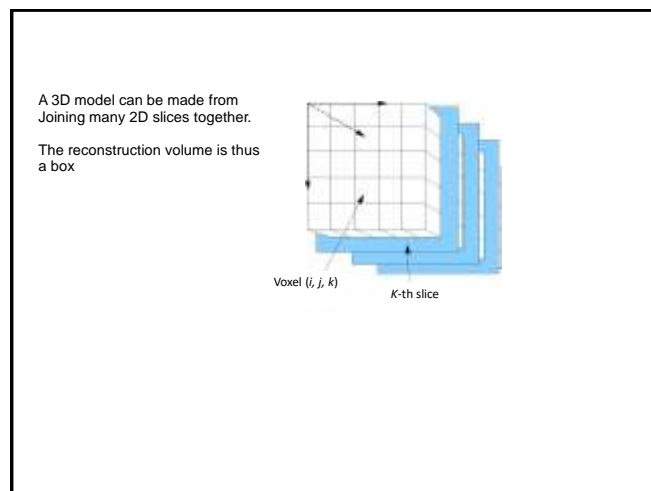
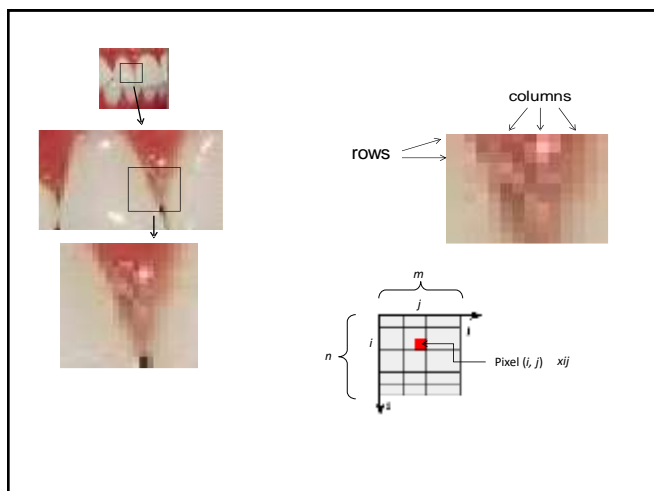
Original: voltage across points (eg. two arms)

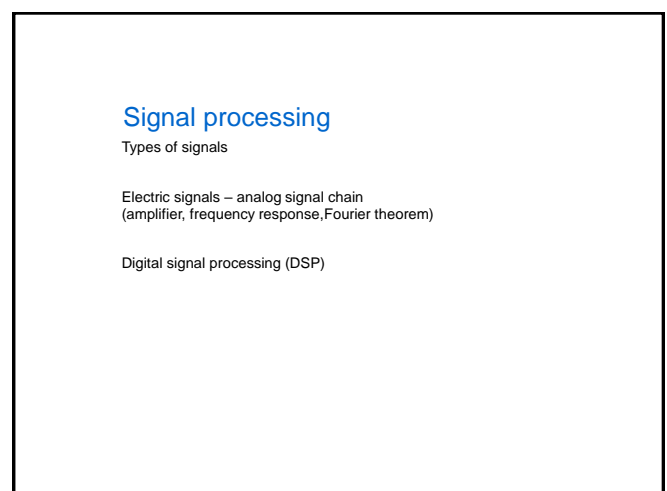
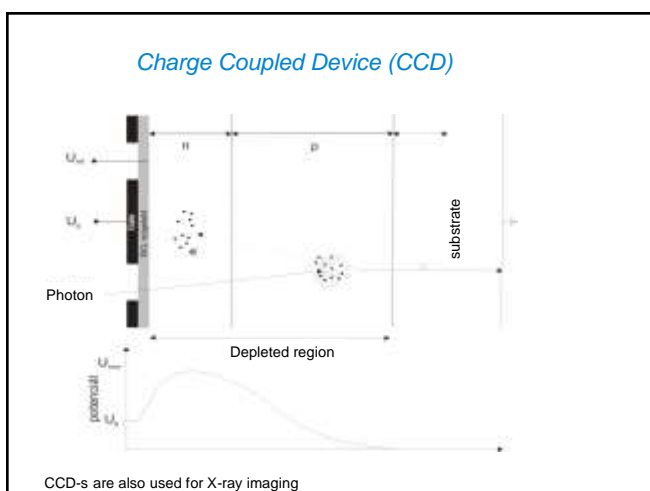
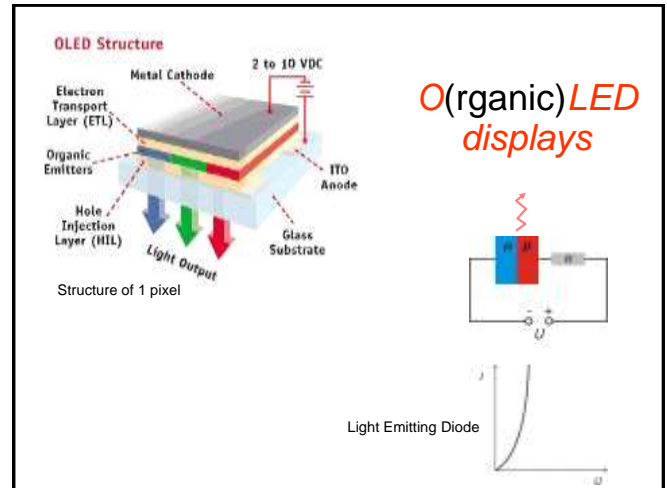
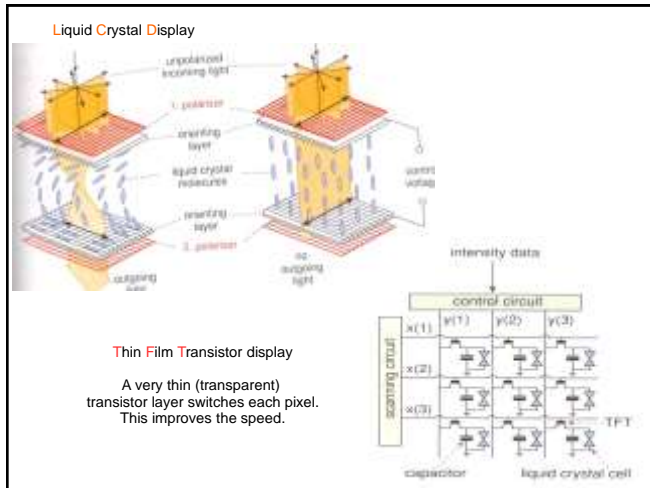
Encoding: None, But Filtering required

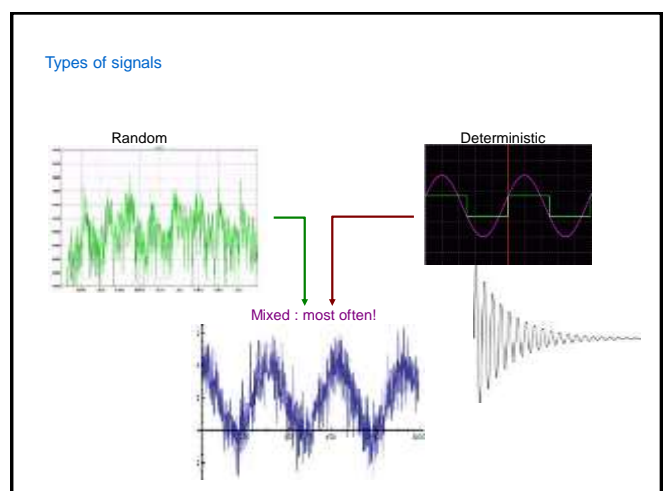
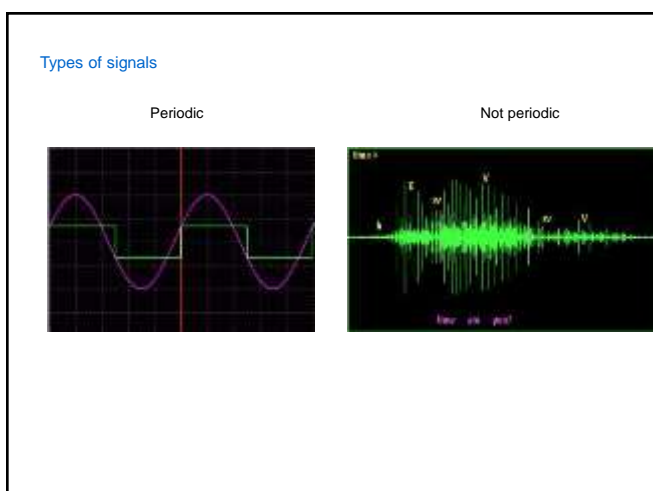
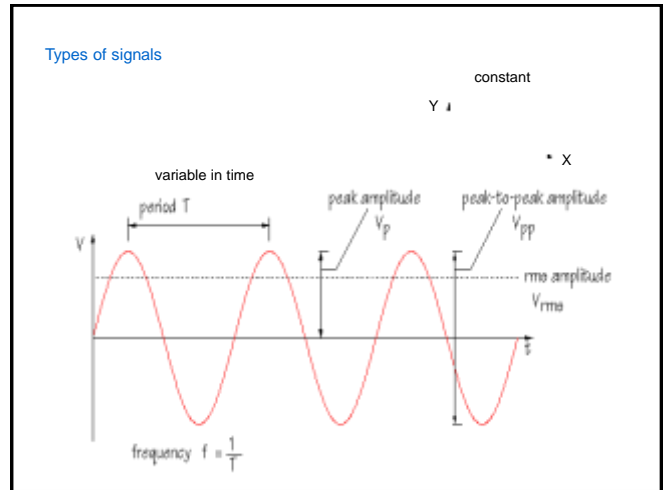
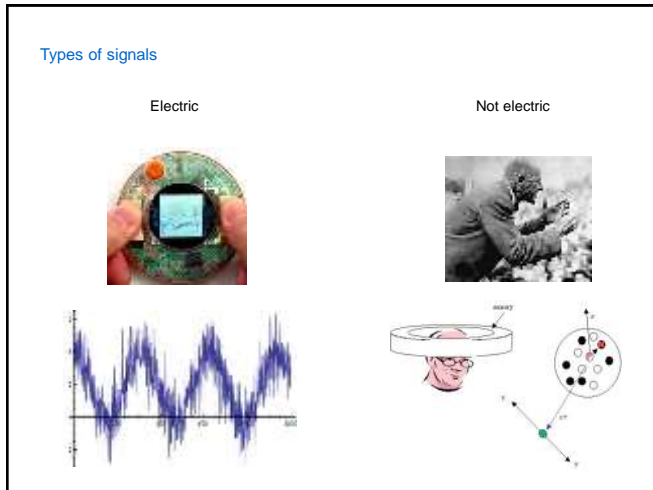
Removal of unwanted portion of the signal





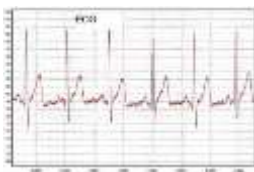




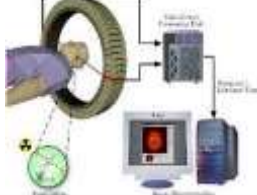
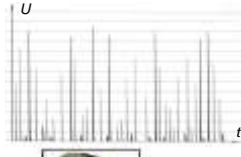


Types of signals

Continuous

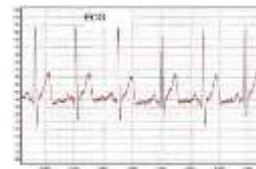


Pulses



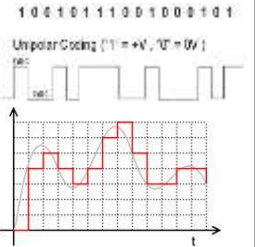
Types of signals

Analog



Theoretically unlimited resolution
in time and magnitude
(measurement system limit only)

Digital



Digital: represented with numbers

Finite resolution

Digital signals are a form of
encoding: digital to electrical
electrical to digital

Information content of signals

Analog signals – infinite information content?

Do we really need **unlimited** resolution?

Do we even **have** unlimited resolution in
real-life analog signals?

Theoretically unlimited resolution
in time and magnitude
(measurement system limit only)

No!

We always have a real signal as:

$$S = \text{Information} + \text{Noise}$$

Information

+

Noise

Information content of signals

Analog signals – infinite information content?

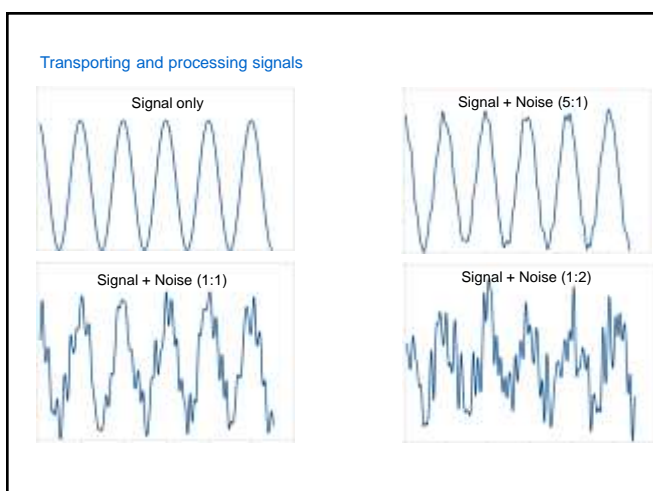
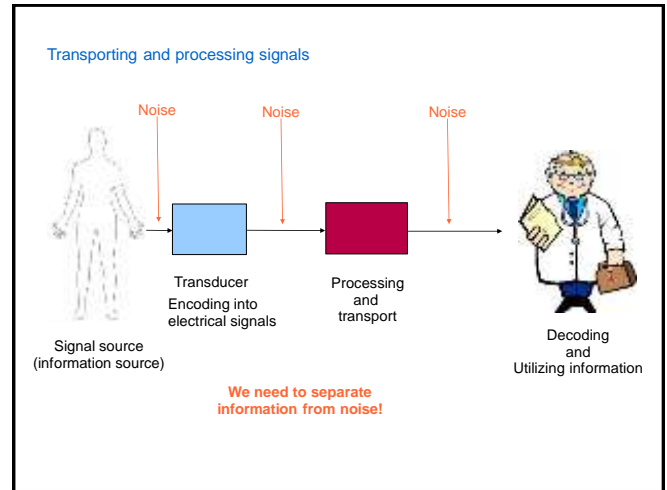
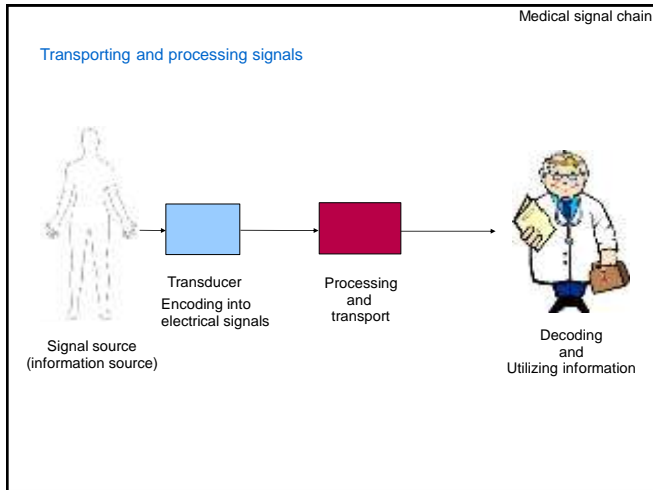
We have Information + Noise

Goal: **Preserve and transport information**
without increasing the noise content.

Information $U = A_{\text{inf}} \cdot \cos(\omega t + \phi)$

+

Noise $\text{Noise}(t) = A_{\text{noise}} \cdot \text{Random}(t)$



Transporting and processing signals

Amplifiers

Task: amplify signal, without addition of noise
(only transport information)

Combat noise in the chain: Amplify the signal at the beginning!

In real-life no amplifier is ideal, they always distort the signal

We need to characterize amplifiers, and other signal-transporting / processing elements of the signal chain.

Analysis of amplifiers

Basic analysis: amplifier gain

The technique is applicable to any transport/coding

$P = U \cdot I = U^2 / R$

$n = 10 \log \frac{P_{\text{output}}}{P_{\text{input}}} \quad [\text{dB}]$

U_2/U_1	P_2/P_1	dB
1,414	2	3
2	4	6
3,16	10	10
10	100	20
100	10000	40
1000	1000000	60

$\frac{P_2}{P_1} = 10^{0,3 \text{ dB}}$

$\frac{P_2}{P_1} = 2 \Rightarrow 10^{0,3 \text{ dB}}$

$\frac{P_2}{P_1} = 10 \Rightarrow 10^{1,0 \text{ dB}}$

$\frac{P_2}{P_1} = 100 \Rightarrow 10^{2,0 \text{ dB}}$

$\frac{P_2}{P_1} = 1000 \Rightarrow 10^{3,0 \text{ dB}}$

Analysis of amplifiers - complex signals

Fourier theorem: Any arbitrary (periodic) signal can be split into sine/cosine functions with varying frequency and amplitude OR from a set of such functions it can be recovered

$$\text{Signal}(t) \leftrightarrow \sum_i A_i \sin(\omega_i t) + B_i \cos(\omega_i t)$$

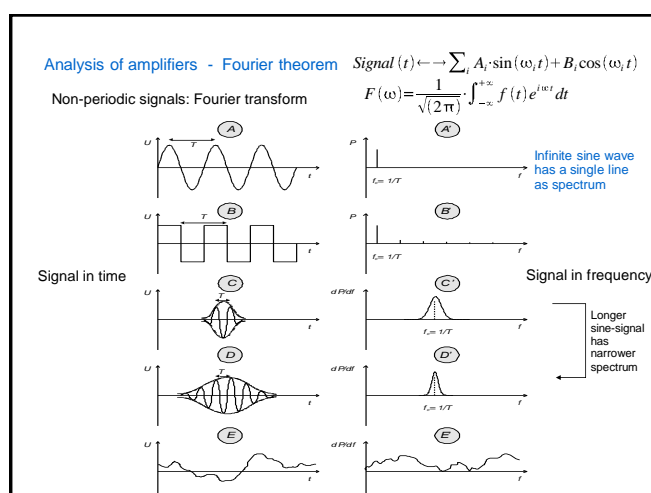
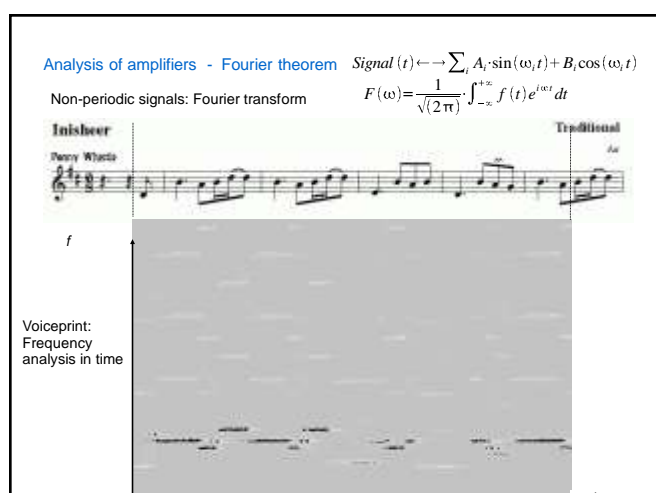
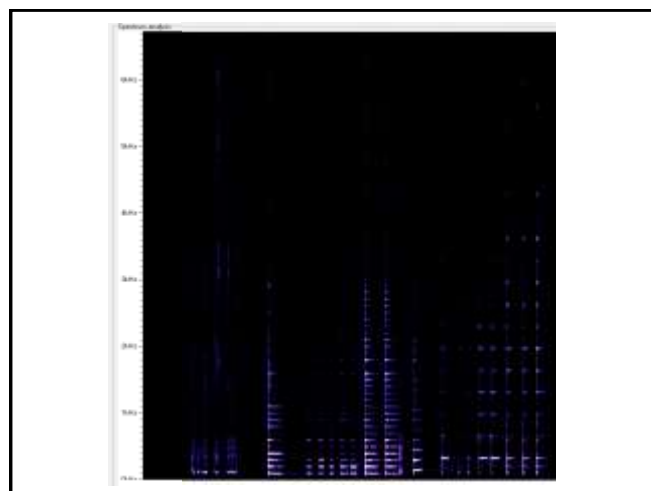
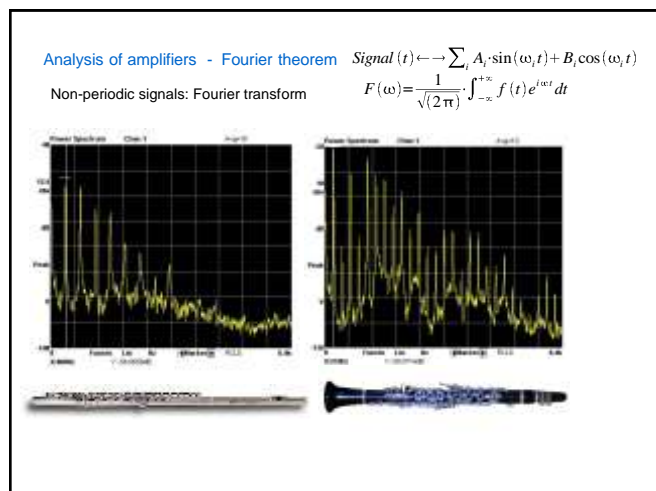
Where in the case of periodic signals $\omega_k = k \cdot f$, $f = 1/T$ and $k = 1, 2, 3, 4, 5, \dots$

Base frequency

overtones

Analysis of amplifiers - Fourier theorem $\text{Signal}(t) \leftrightarrow \sum_i A_i \sin(\omega_i t) + B_i \cos(\omega_i t)$

Analysis of amplifiers - Fourier theorem $\text{Signal}(t) \leftrightarrow \sum_i A_i \sin(\omega_i t) + B_i \cos(\omega_i t)$



Analysis of amplifiers - Fourier theorem $Signal(t) \leftarrow \sum_i A_i \sin(\omega_i t) + B_i \cos(\omega_i t)$

Non-periodic signals: Fourier transform $F(\omega) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{+\infty} f(t) e^{i\omega t} dt$

Any signal is just a representation of information


We can have many pictures of the same

Time-based (more conventional)

or

Frequency-based (useful, but a bit abstract)

Fourier-transform is the „art of engineering“ (Picasso: La Crucifixion)



Analysis of amplifiers - Transfer function of filters

Low-pass filter

High-pass filter

$n(\text{dB})$

$\log f$

U_{in} R_1 R_2 U_{out}

$U_{output} = U_{input} \cdot \frac{R_2}{R_1 + R_2}$

Substitute one R with C

R U_{in} C U_{out}

C U_{in} R U_{out}

Analysis of amplifiers - Transfer function of filters

Low-pass filter

High-pass filter

$n(\text{dB})$

$\log f$

U_{in} R_1 R_2 U_{out}

$U_{output} = U_{input} \cdot \frac{R_2}{R_1 + R_2}$

Substitute one R with C

$R_C = \frac{1}{C\omega}$

R U_{in} C U_{out}

C U_{in} R U_{out}

$U_{out} = \frac{1}{\sqrt{1 + R^2 C^2 \omega^2}} U_{input}$

$U_{out} = \frac{RC\omega}{\sqrt{1 + R^2 C^2 \omega^2}} U_{input}$

Analysis of amplifiers - Transfer function of amplifiers

input U_1 U_2 output

$n(\text{dB})$

n_{max}

$n_{max} - 3$

Transfer band

f_l f_u

$\log f (\text{Hz})$

Ideal amplifier

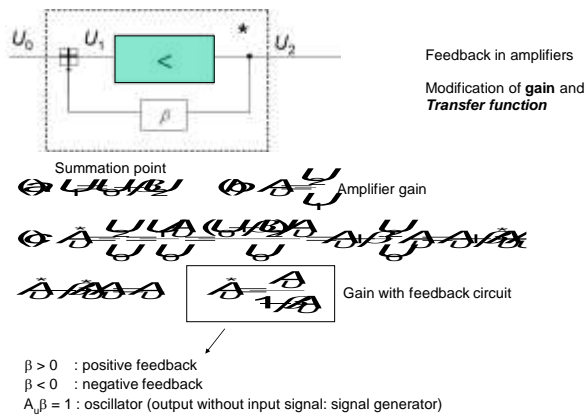
Real-world amplifier

Amplifiers are not ideal, they have input and output capacitance, etc.

The output signal may *not* contain all frequency components!

Distortion, information loss / alteration

Analysis of amplifiers - Transfer function of amplifiers



Analysis of amplifiers - Transfer function of amplifiers

Gain Bandwidth Product

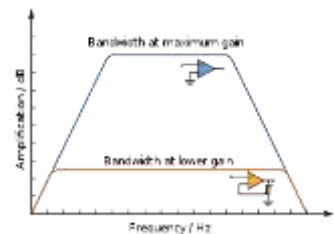
$$\text{Gain} \cdot \text{Bandwidth} = \text{constant}$$

The available power to the amplifier can either be put to use as:

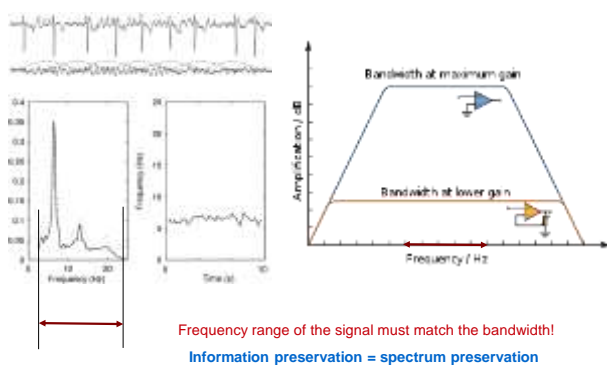
high signal gain over a limited bandwidth

or

limited gain over a wide bandwidth.



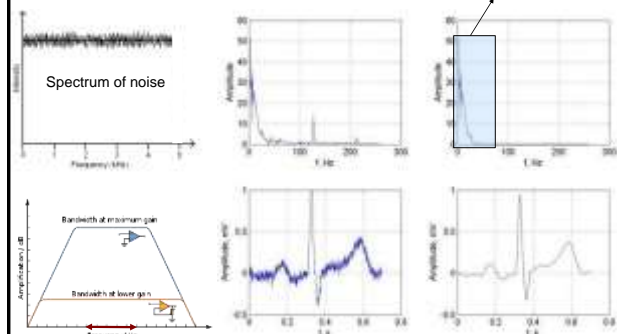
Analysis of amplifiers - Transfer function of amplifiers



Analysis of amplifiers - Transfer function of amplifiers

During analog signal transport at every stage noise will be added! → degradation

Just transport that part of the spectrum which contains the information!



Digital signals – A/D conversion (ADC)

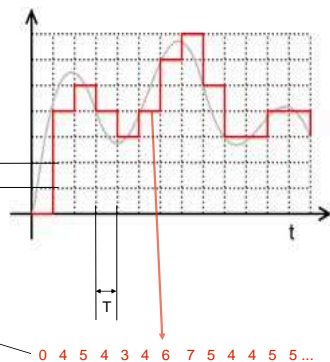
The analog signal can be represented by numbers:

We measure the signal every T seconds, and transmit the result only.

Measurement accuracy
(how many bits)

Digital signals are discrete
in time and in value

Numbers can be transported / stored
or processed losslessly!

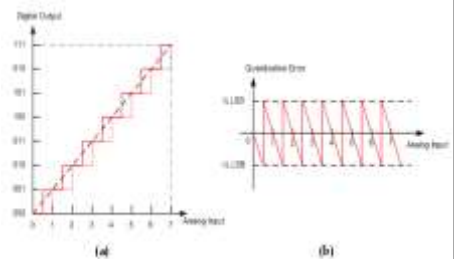


Digital signals - Quantization

Digital signals are discrete
in time and in value

What happens to the original parts between?

They get lost!

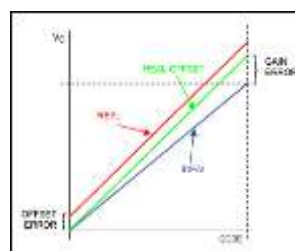


Digital signals – Restoration (DAC)

Recovery of analog signals:

Digital to analog converter

This is easily realized to be near-ideal
Many-bits, fast DAC-s are cheap



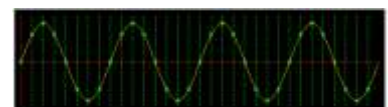
Pitfalls to avoid

Digital signals – Sampling of sine waves

For non-sine signals: „first apply Fourier, then sample each sine”

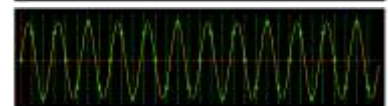
$f = 1000$ Hz
 $f_s = 8000$ Hz

No problem



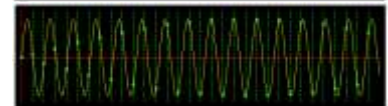
$f = 3000$ Hz
 $f_s = 8000$ Hz

Still no problem



$f = 3900$ Hz
 $f_s = 8000$ Hz

Still no problem

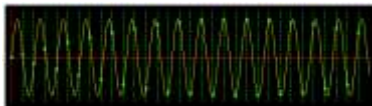


Digital signals – Sampling of sine waves

For non-sine signals: „first apply Fourier, then sample each sine“

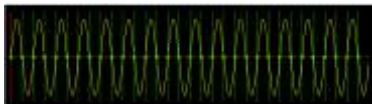
$f = 3900 \text{ Hz}$
 $f_s = 8000 \text{ Hz}$

Still no problem



$f = 4000 \text{ Hz}$
 $f_s = 8000 \text{ Hz}$

Signal lost!



Nyquist theorem: sampling frequency must be at least 2x the frequency of the sine

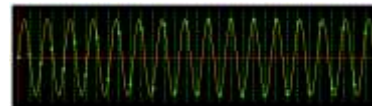
Digital signals – Nyquist

Nyquist theorem: sampling frequency must be at least 2x the frequency of the sine

good

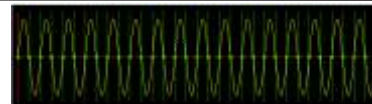
$f = 3900 \text{ Hz}$
 $f_s = 8000 \text{ Hz}$

Still no problem



$f = 4000 \text{ Hz}$
 $f_s = 8000 \text{ Hz}$

Signal lost!

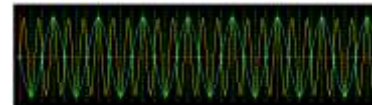


bad

$f = 6000 \text{ Hz}$
 $f_s = 8000 \text{ Hz}$

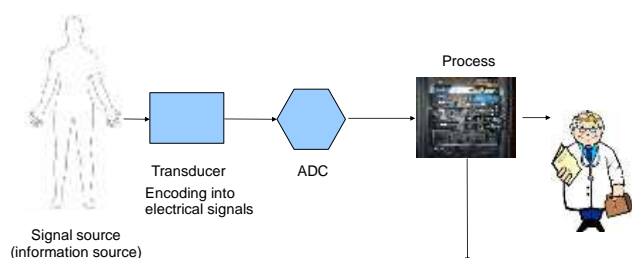
Signal lost!

Aliasing



Aliased sine appears instead of the real input

Digital signals – Digital Signal Processing



Signal processing with DSP units is everywhere around us.

