MEK 4600

Labview

Acquiring acoustic data from turbulence

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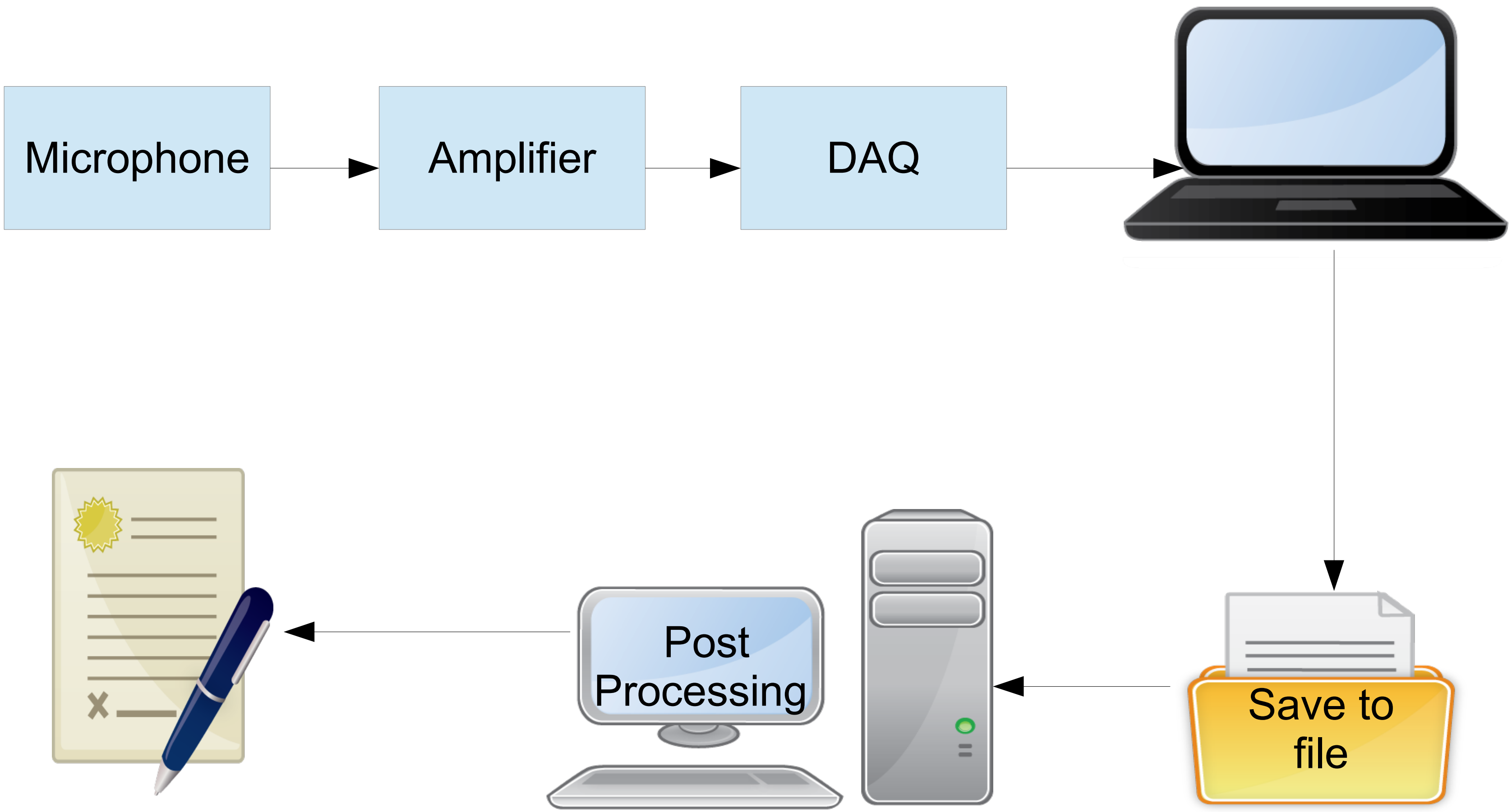
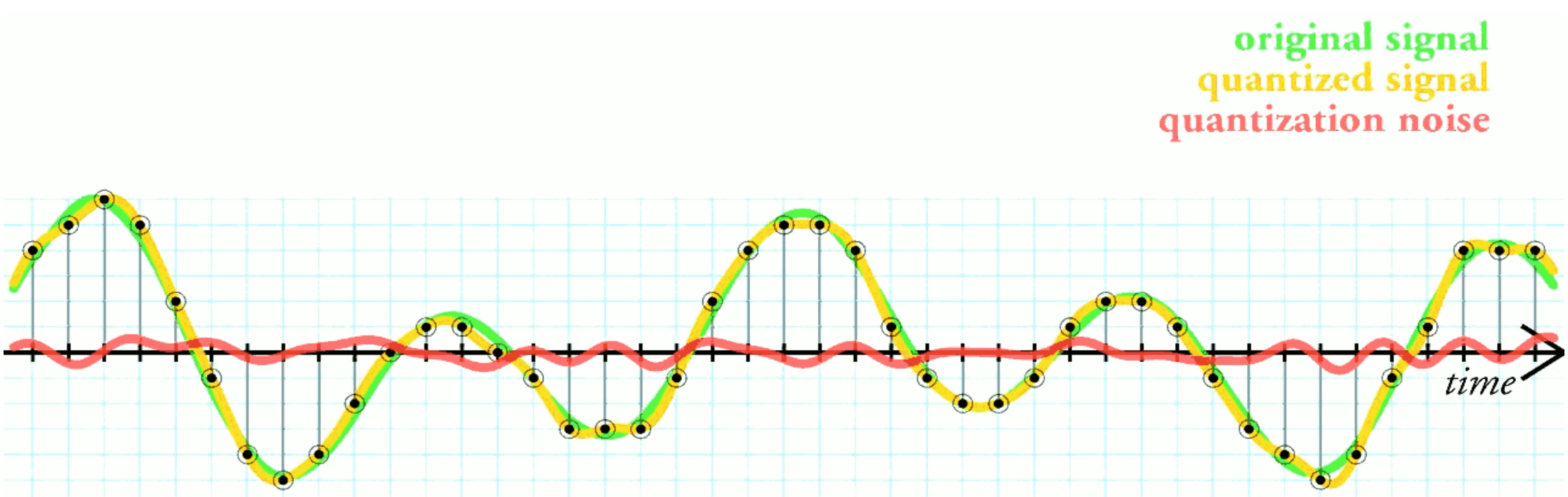
System setup



Analogue vs Digital signal

- Analog signals are continuous signals
- Digital signals are non-continuous
- Most «real» signals are analogue
- Time, distance, temperature, strain....



Analogue vs Digital signal (II)

- Digital audio involves taking an analog waveform (i.e. sound waves) and converting it to a series of individual samples, each of which has an amplitude value

- The range of possible amplitude levels are defined by the bit depth, e.g. 8-bit quantization = 256 possible values; 16 bit quantization = 65,536 possible values, etc.

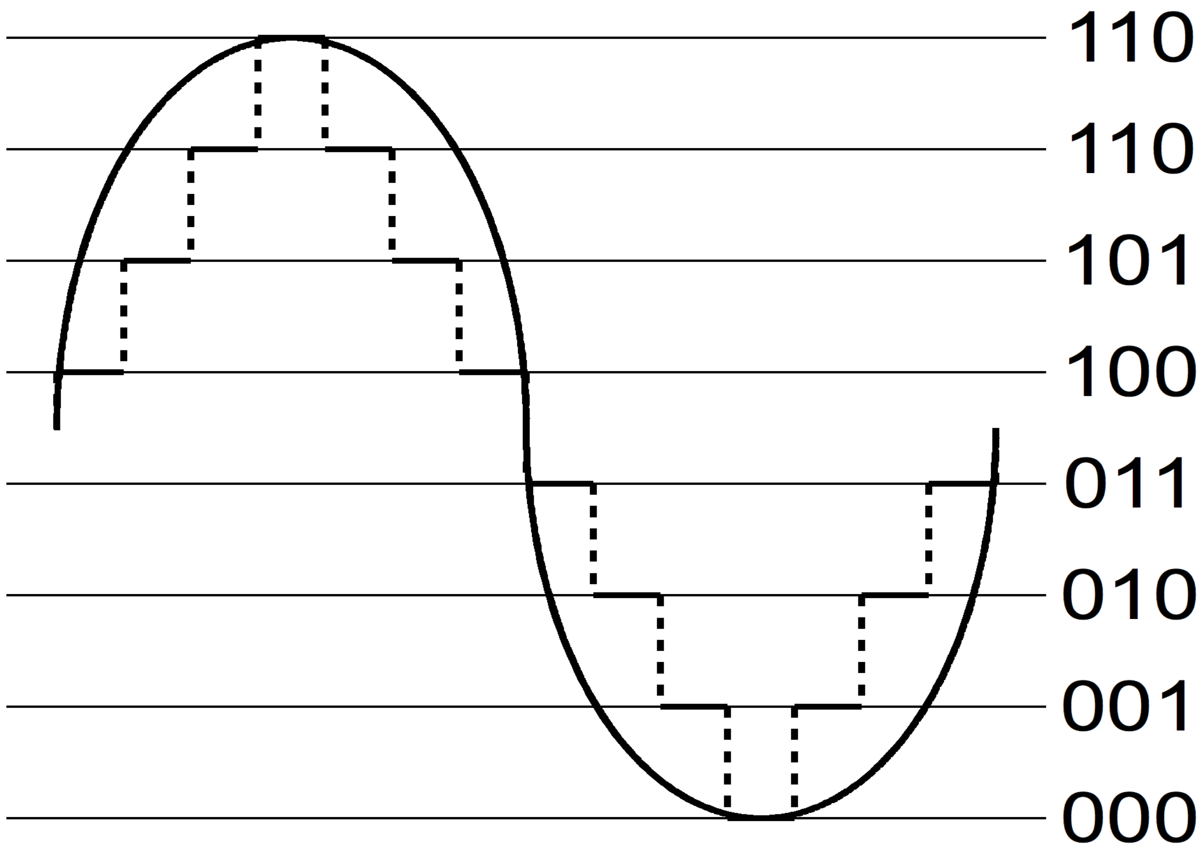
Quantization

The computers are not able to compute an analogue signals in its original form, thus it needs to be *quantized* and *converted* to digital numbers

In order to use the analogue signal on the computer you will have to make it «digital»

Quantization (II)

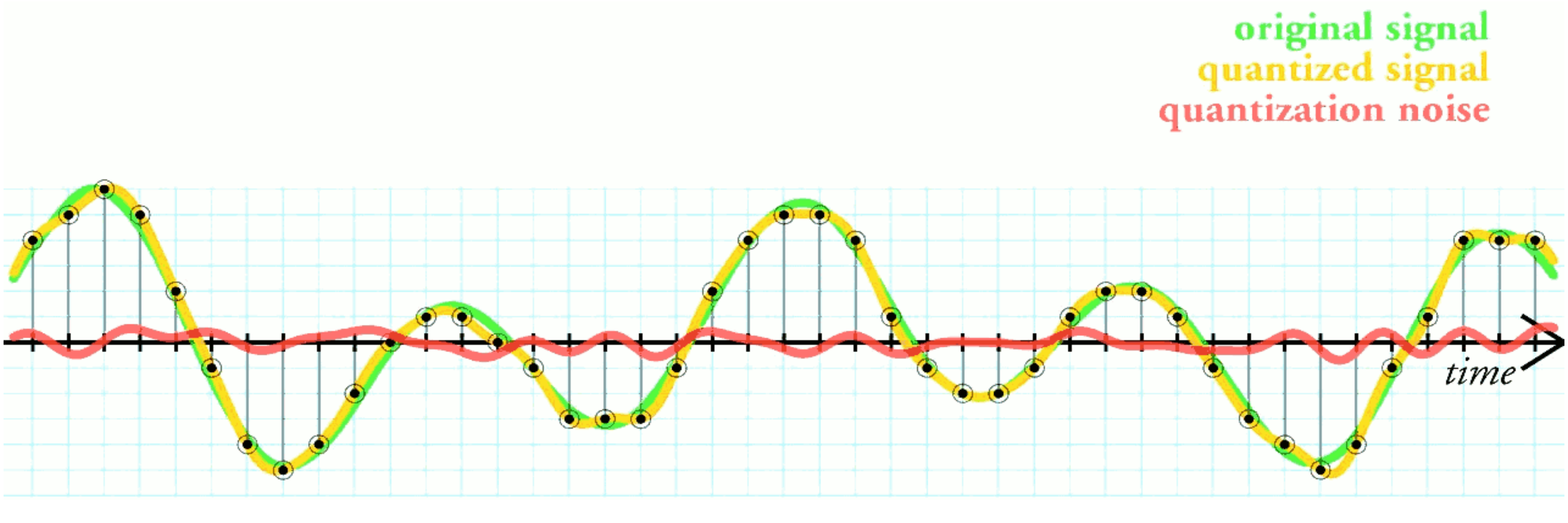
Analog signals can be converted to digital signal by an Analogue to Digital Converter (ADC)



Sampling

In signal processing, sampling is the reduction of a continuous signal to a discrete signal.

A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal)



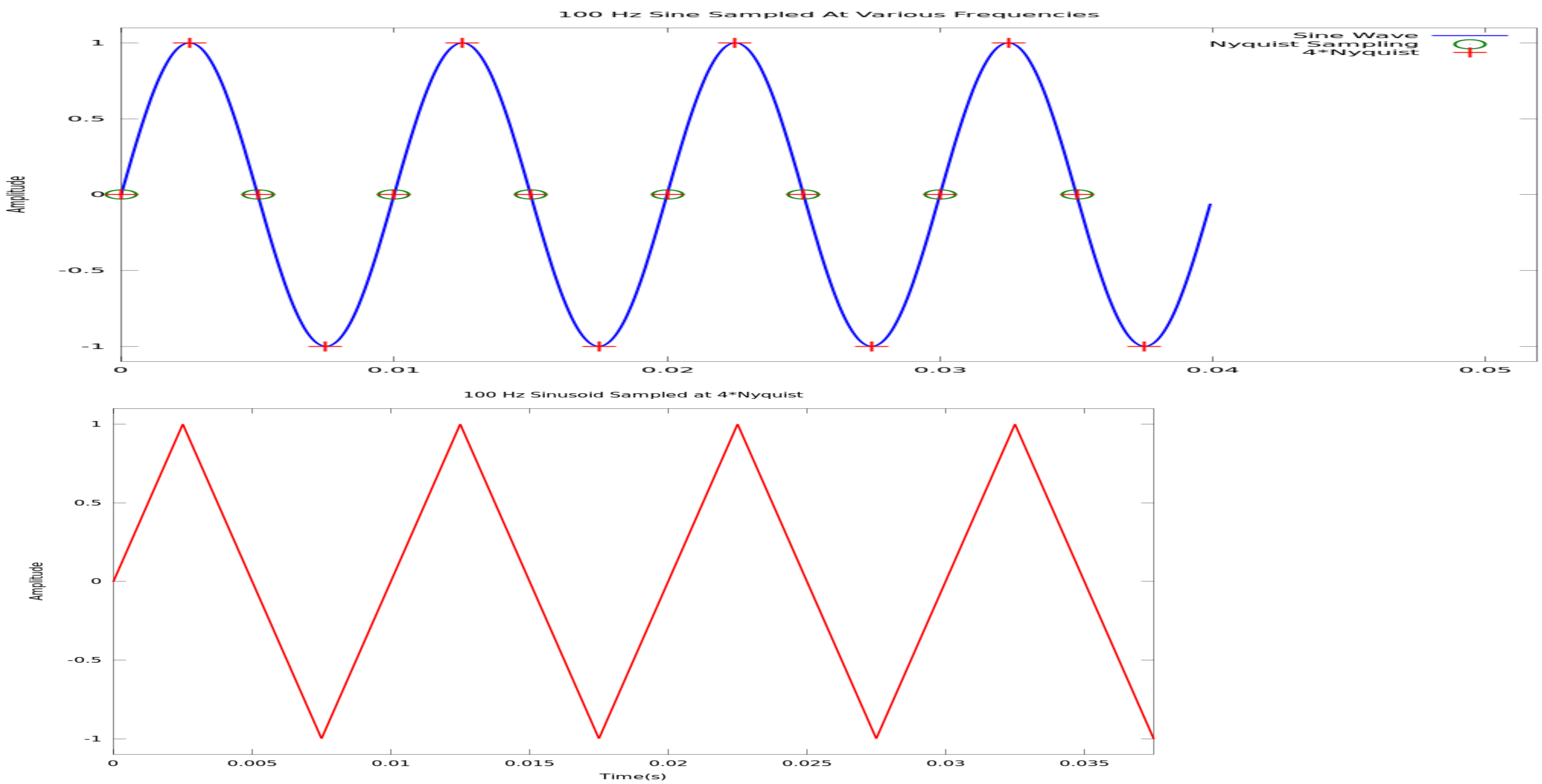
Sampling (II)

As the digital signal is acquired at discrete time intervals we understand that the time between each acquisition (sampling) is critical for how much information in the signal you will obtain;

If the signal varies fast (wind speed, speech, radio waves) you will need to obtain (sample) faster than if the signal varies more slowly (temperature in a pool)

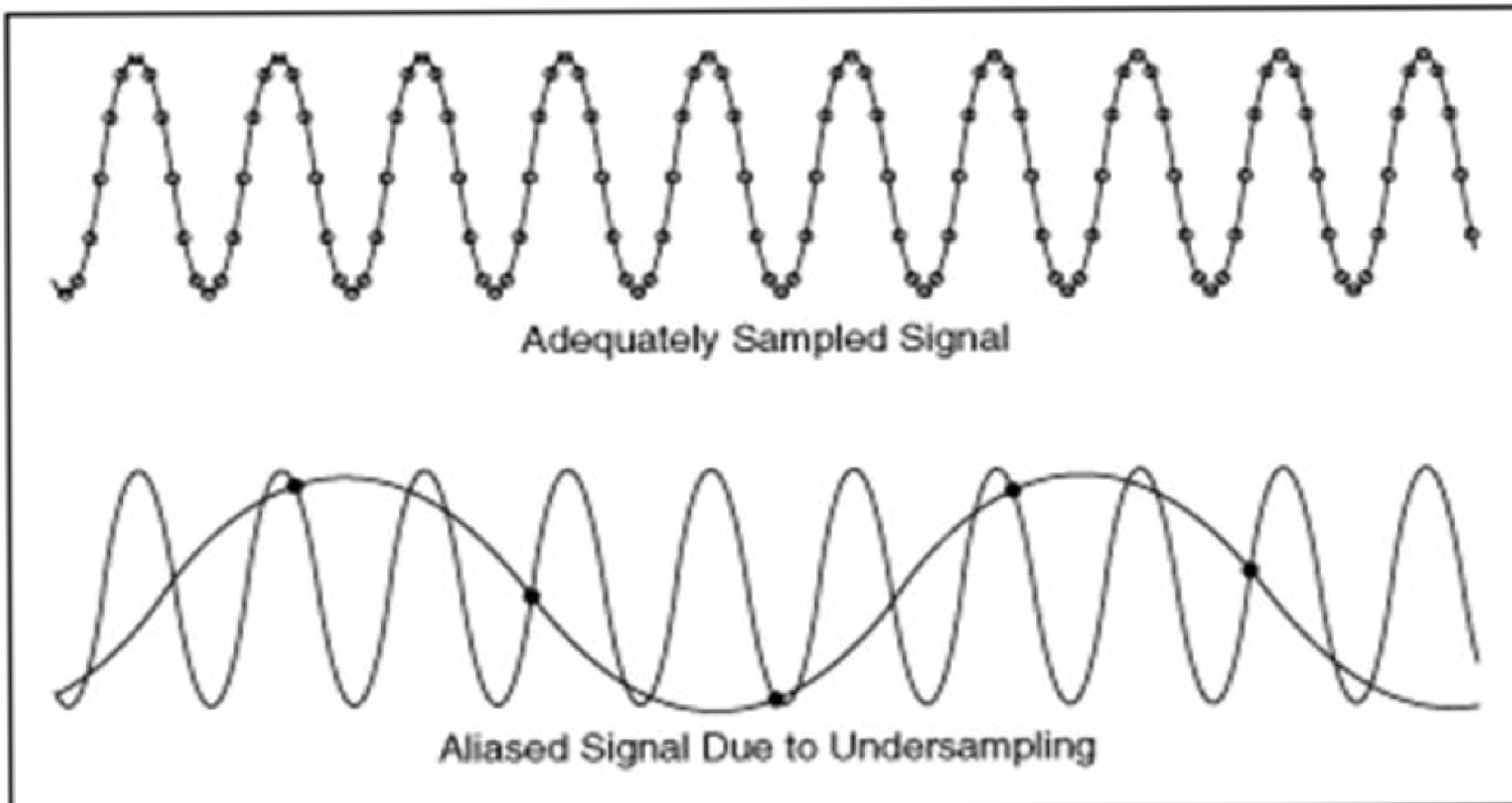
Nyquist–Shannon sampling theorem

“If a function x(t) contains no frequencies higher than B hertz, it is completely determined by giving its ordinates at a series of points spaced 1/(2B) seconds apart.”



Nyquist (II)

What happens if you sample slower than 1/2B?

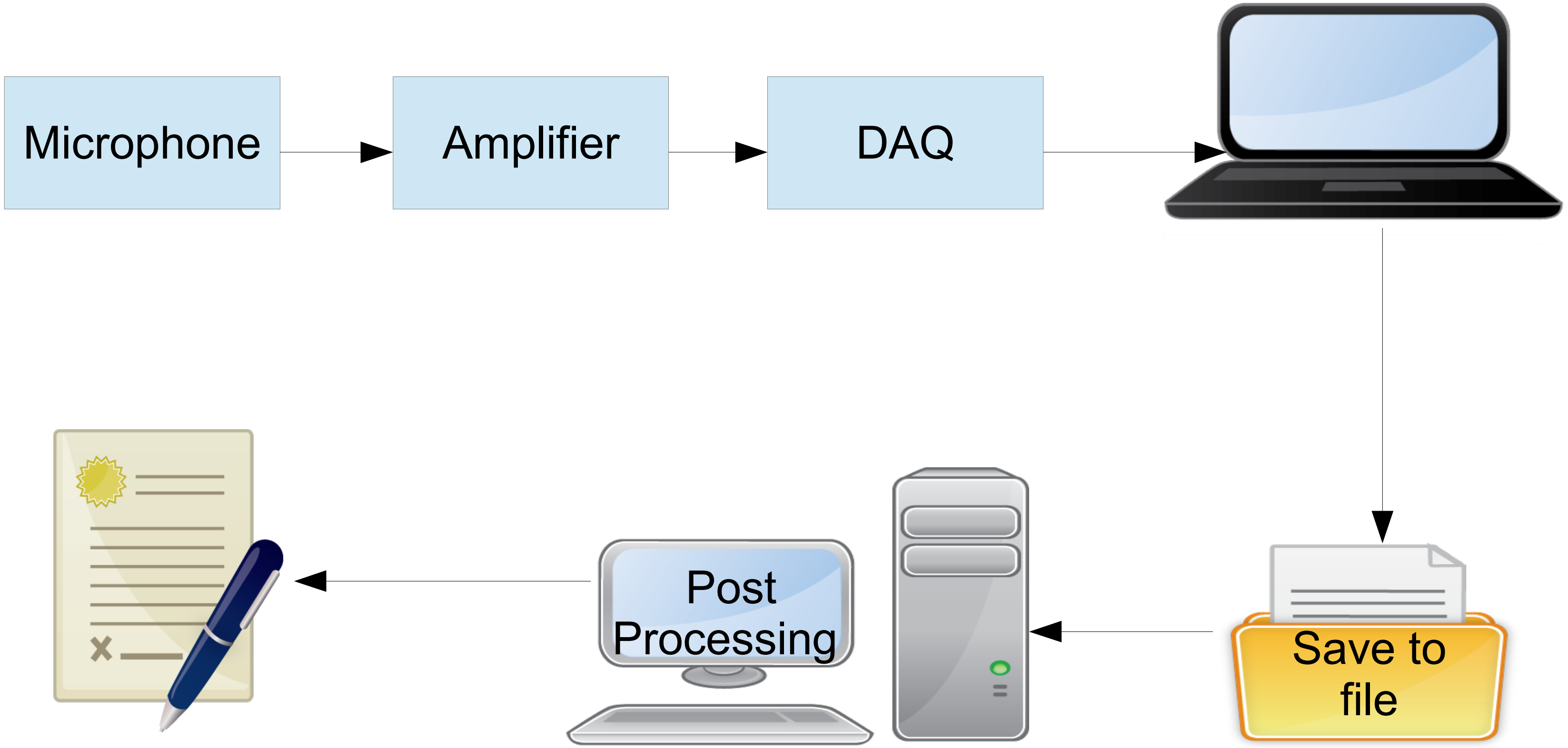


Nyquist (III)

A band-limited function can be perfectly reconstructed from a countable sequence of samples if the band-limit, B, is no greater than half the sampling rate (samples per second) [Wiki]

Noise

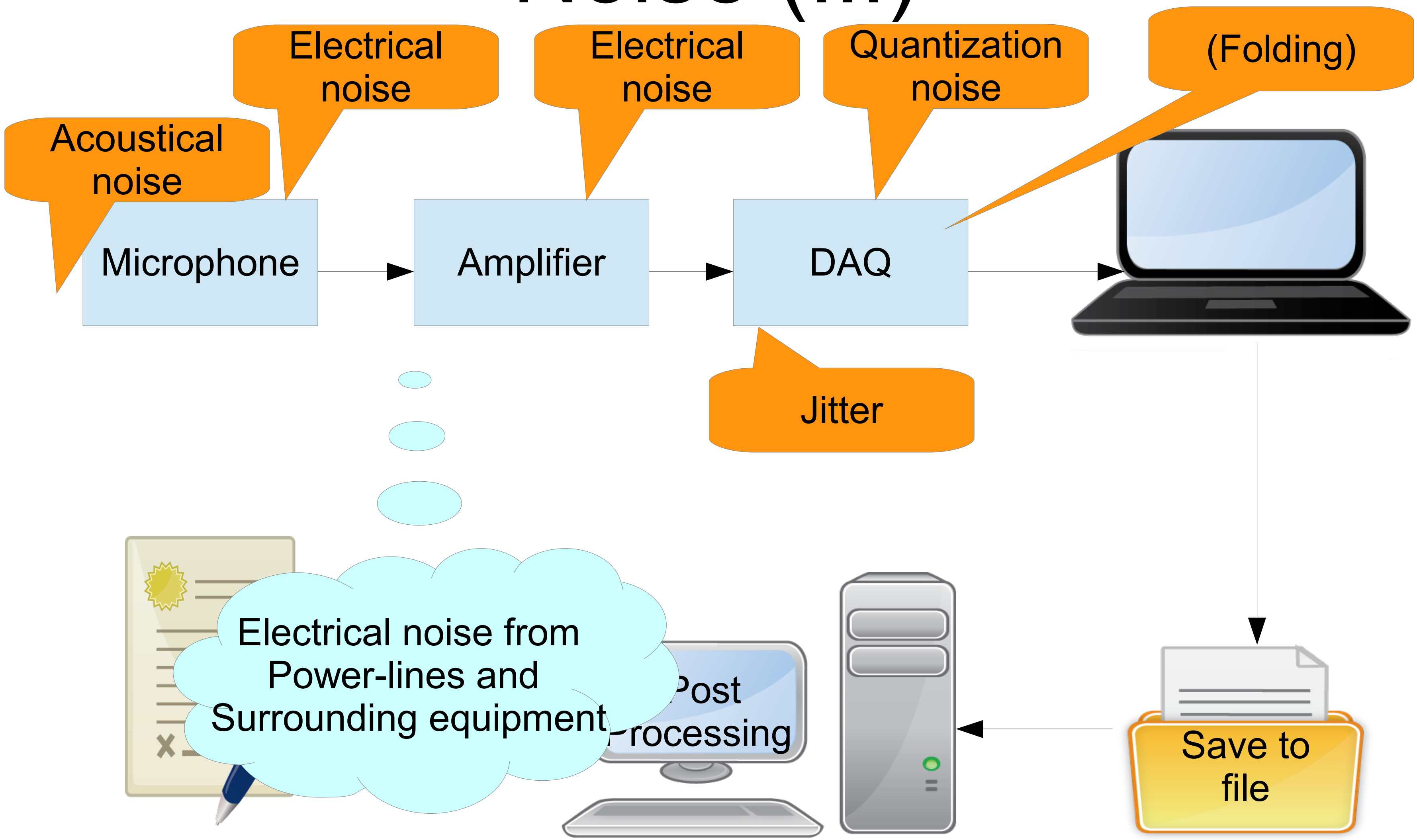
All real life systems do have noise. In our measuring setup we will see noise in several forms and from different sources.



Noise (II)

- Quantization noise
- Electrical noise from the microphone
- Noise from the amplifier
- Electrical noise from other electrical equipment and from power-lines (50Hz) around us
- Acoustical noise picked up by the microphone
- Jitter

Noise (III)



Electrical noise
Electrical noise
Quantization noise
(Folding)
Acoustical noise
Microphone
Amplifier
DAQ
Jitter
Electrical noise from Power-lines and Surrounding equipment
Post Processing
Save to file

Signal to Noise Ratio (SNR or S/N)

Signal-to-noise ratio is defined as the power ratio between a signal (meaningful information) and the background noise (unwanted signal):

$$SNR = 10 \cdot \log(A_{signal}/A_{noise})^2 = 20 \cdot \log(A_{signal}/A_{noise})$$

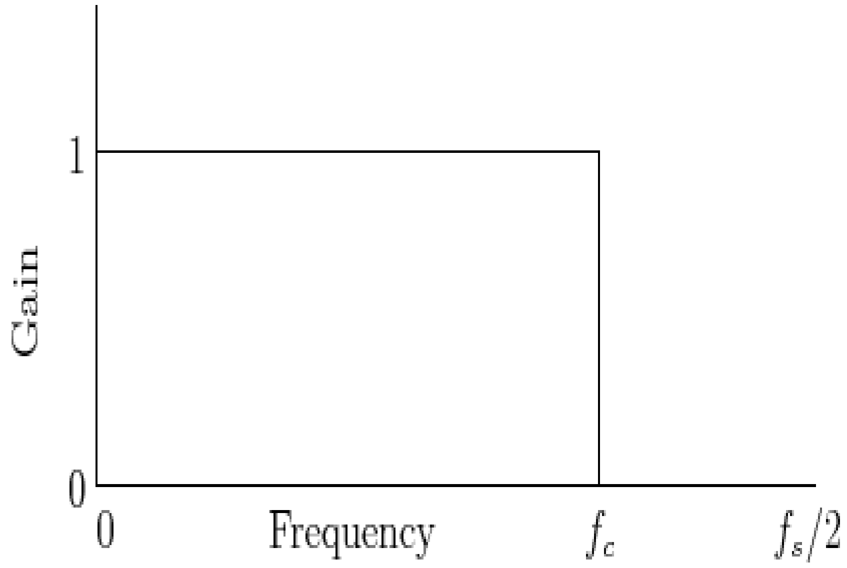
Teoretical max for Digital signals:

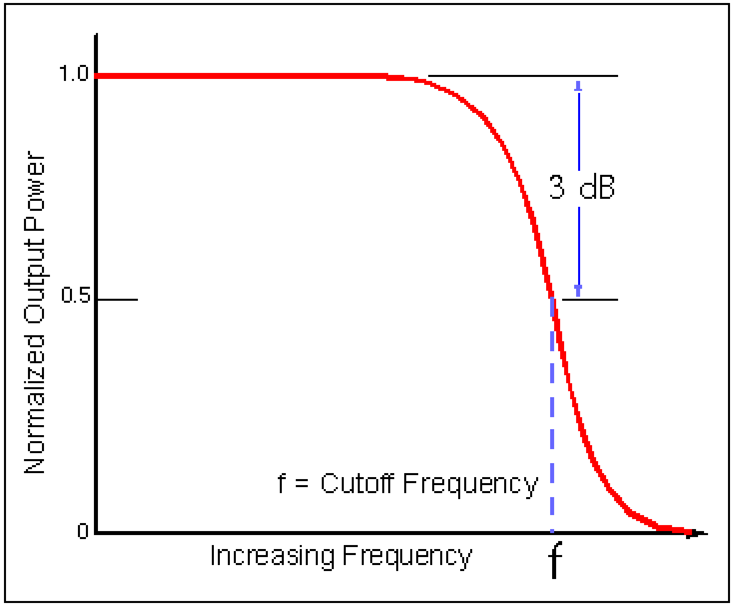
$$SNR = 20 \cdot \log(2^n) \approx 6 \cdot n \text{ dB}$$

(where n = number of bits)

Filters

- To avoid aliasing we need to limit the highest frequency of the signal -*before sampling*
- Low-pass (LP) filter

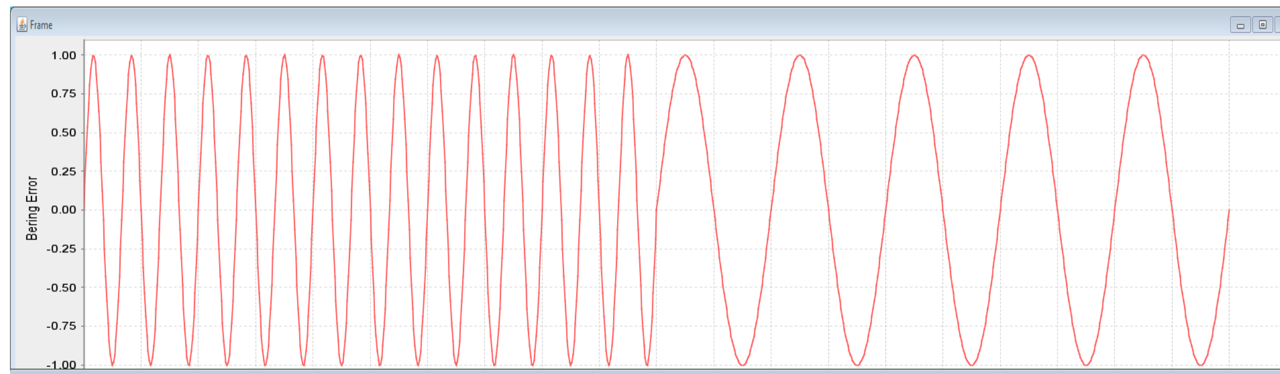




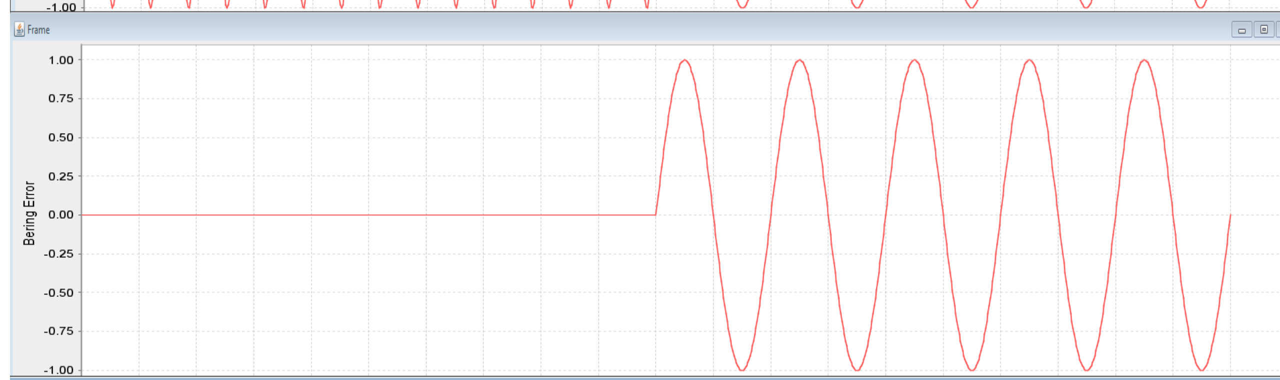
low pass filter

Low pass filter output

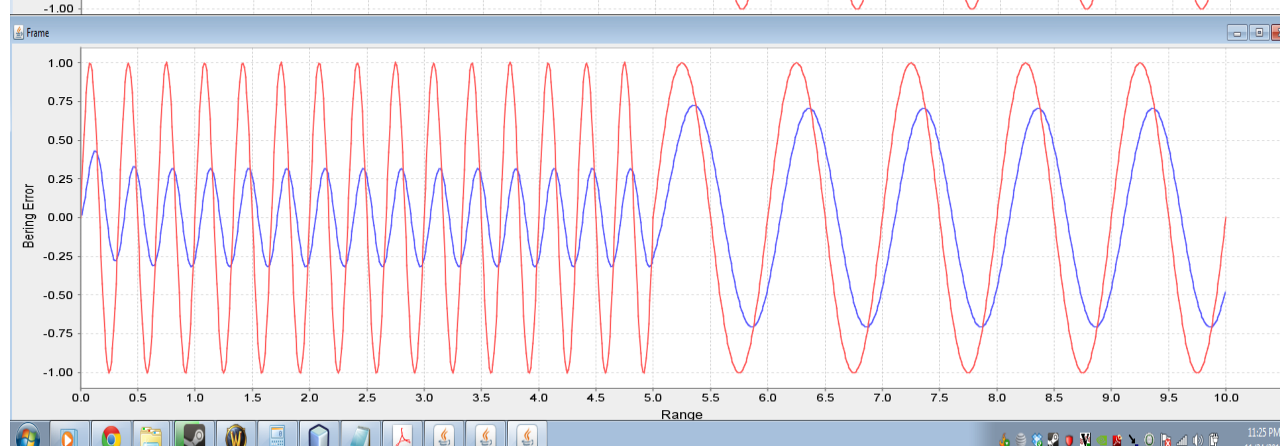
Input:



Ideal output:



Real output:



Filters (II)

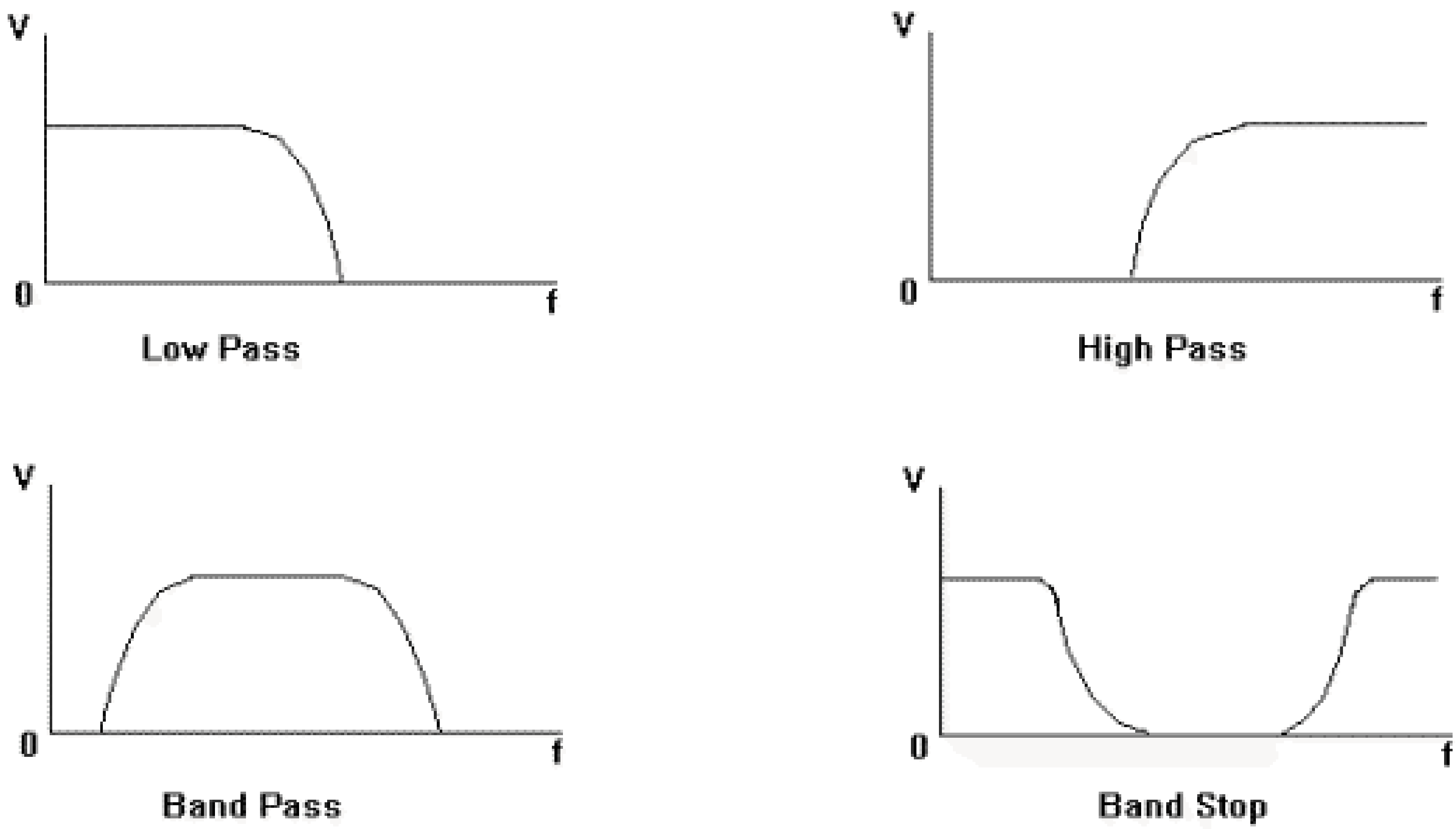
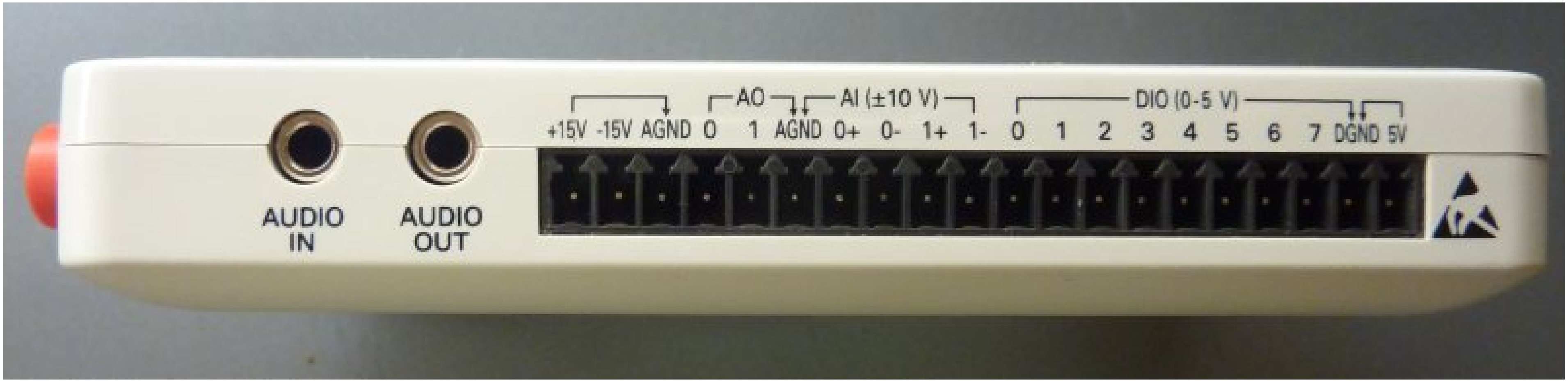


Fig. 3

DAQ

- myDAQ

myDAQ


AO
AI (±10 V)
DIO (0-5 V)
+15V -15V AGND 0 1 AGND 0+ 0- 1+ 1- 0 1 2 3 4 5 6 7 DGND 5V
AUDIO IN
AUDIO OUT

MyDAQ schematic

Labview overview

- Explorer
- Tasks
- Labview VI's
- Block diagram
- Front panel