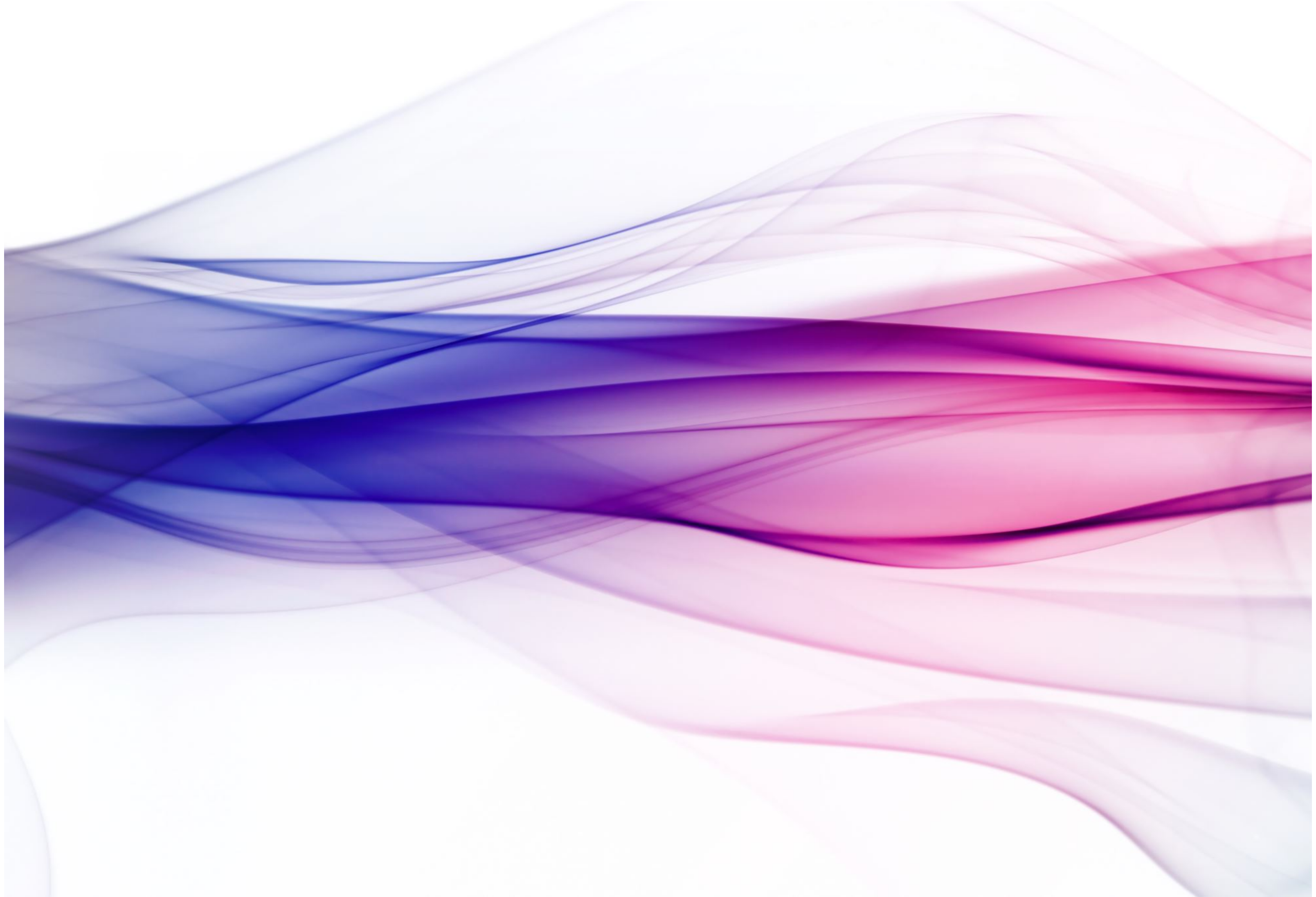

Digital Filter

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Contents

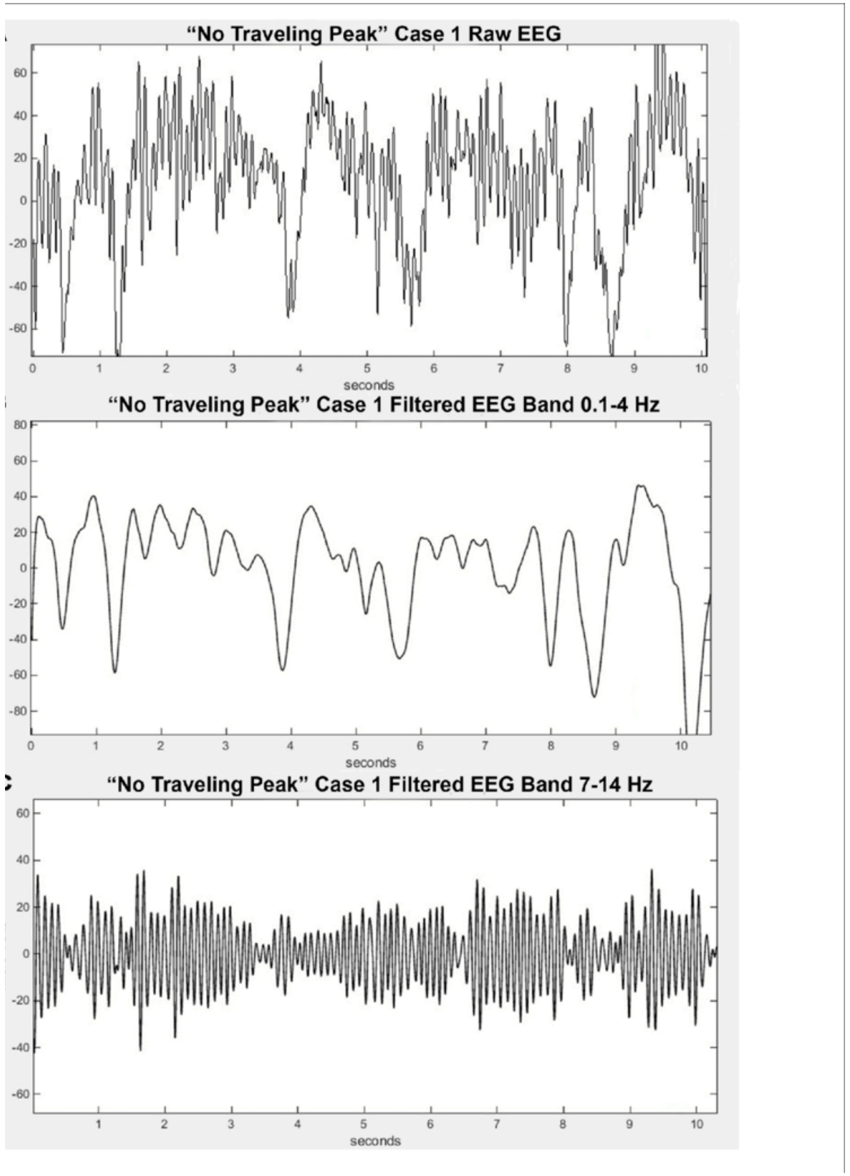
Information contained in raw signal waveforms

A signal usually consists of useful information and undesired noise. Instead of the time-domain value only, information in different frequency bands give different useful information and noise should be eliminated.

Example:

- A – Raw data
- B – Filtered signal (0.1 – 4 Hz)
- C – Filtered signal (7 – 14Hz)

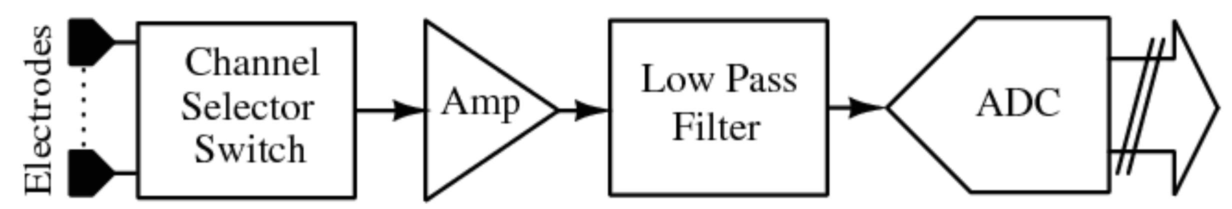
How to extract necessary information?



Preprocessing of signals

A filter is placed between the output of the amplifier and the input of the ADC.

The filter is used to extract the necessary information between the defined frequency band(s).



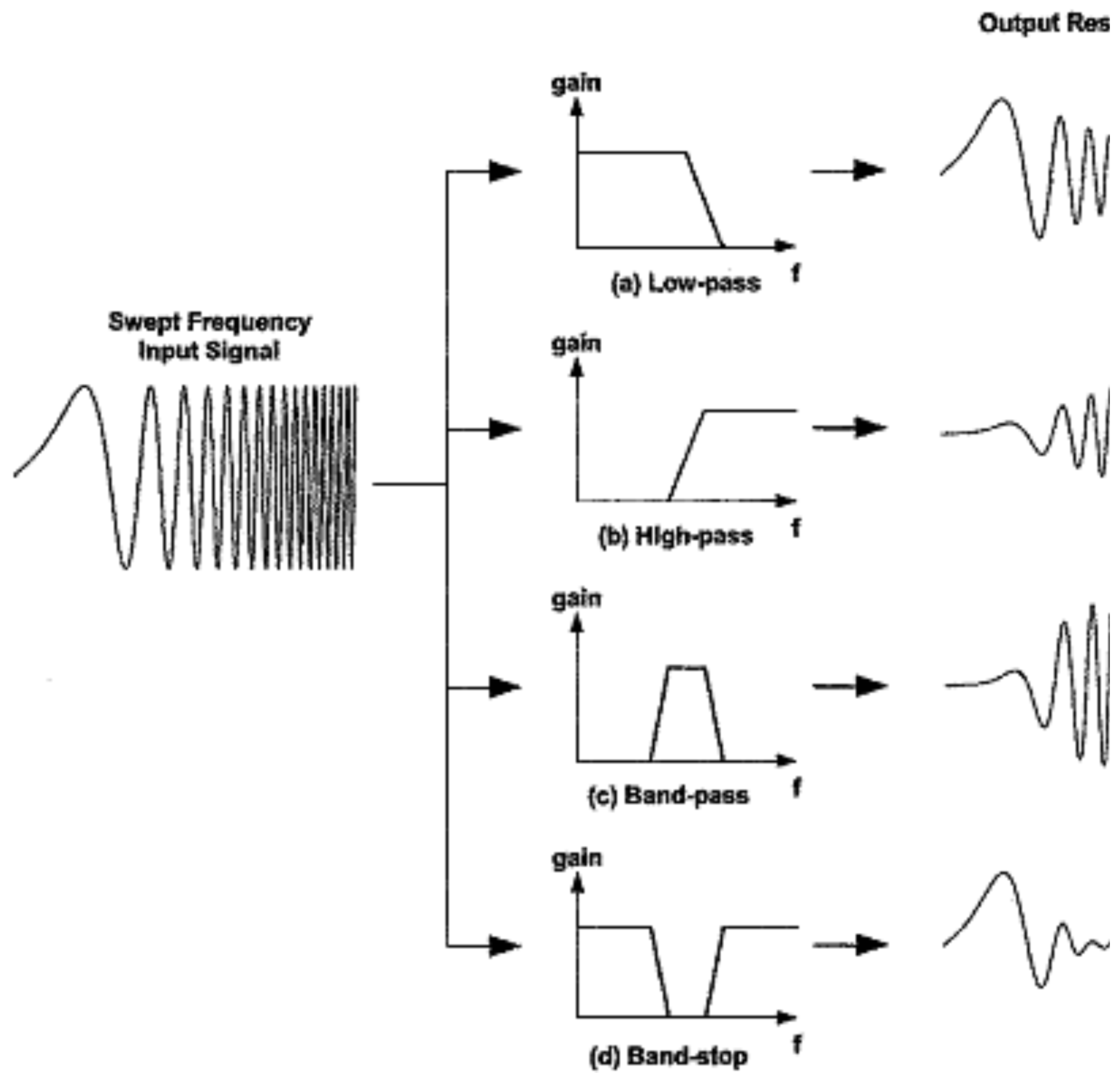
Different types of filters

Passband - the range of frequencies or wavelengths that can pass through a filter.

Stopband – the range of frequencies, between specified limits, through which a circuit does not allow signals to pass.

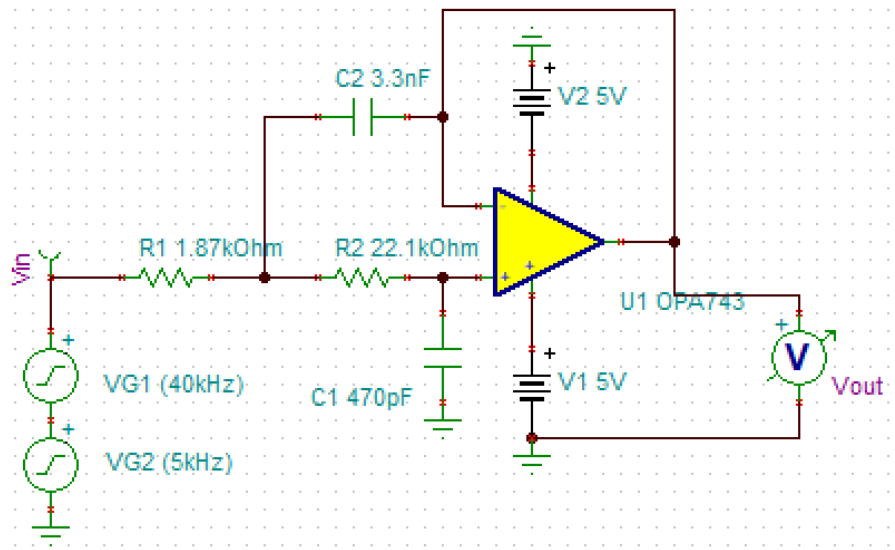
There are four main types of filters:

- Low-pass filter
- High-pass filter
- Band-pass filter
- Band-stop filter



Simulation on TINA

20kHz 2nd- Order Sallen- Key Low- Pass Filter



Achieving high stopband attenuation at high frequency in a low- pass filter requires an op amp with a high gain- bandwidth (GBW) product. In cases where supply current is an important factor, the OPA743 offers a unique combination of GBW (7MHz), slew rate (10V/us), quiescent current (1.1mA) and the ability to operate on +/-5V or a single +12V supply.

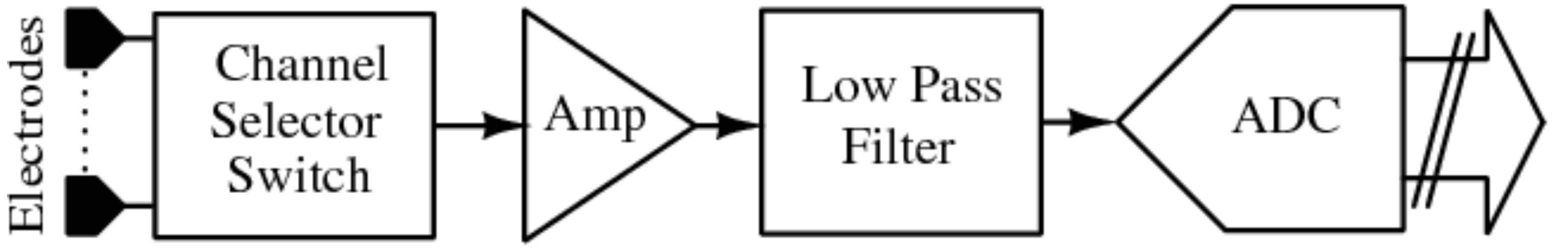
Resistors should be 1% metal- film types and capacitors should be accurate, high- stability types such as ceramic NPO, silver mica, or PPS film types.

Download the simulation file and Try!

What is frequency response?

Frequency	Input voltage (Vin)	Output voltage (Vout)	Gain (Vout / Vin)	Gain (in dB)
1kHz	0.5	0.5	1	0
5kHz	0.5	0.5	1	0
10kHz	0.5	0.482	0.964	-0.318
15kHz	0.5	0.4272	0.8544	-1.367
20kHz	0.5	0.34474	0.68948	-3.23
30kHz	0.5	0.1961	0.3922	-8.13

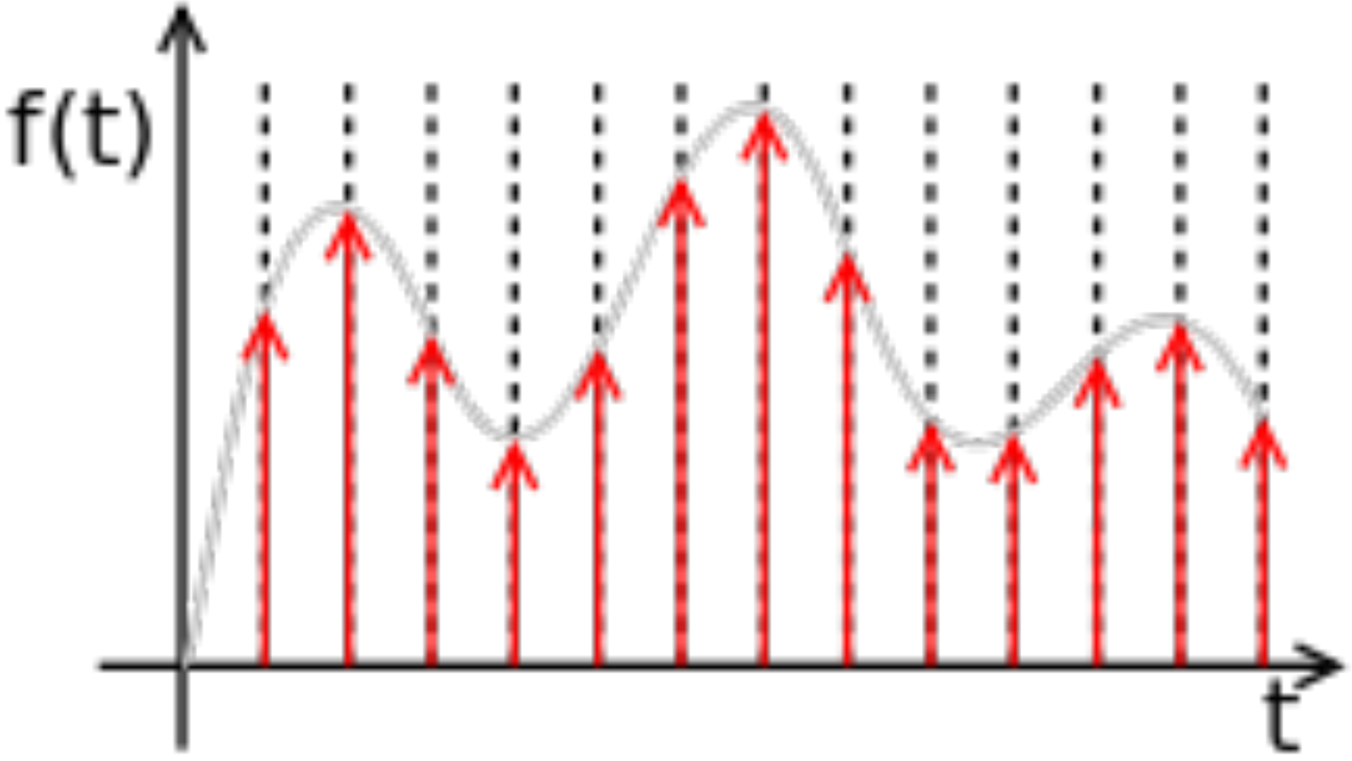
Try your simulation!! Do you remember how to convert the gain into in dB?



Digital filter

Instead of using a separate an analog filter, the signal is filtered by the microcomputer. That is, the low-pass filtering function is implemented by a software algorithm.

Continuous-time and discrete-time signals



- A **continuous signal** or a **continuous-time signal** is a varying quantity (a signal) whose domain, which is often time, is a continuum (e.g., a connected interval of the reals). [Grey line]
- A **discrete signal** or **discrete-time signal** is a time series consisting of a sequence of quantities. **[Red arrows]**
- The function's domain of continuous signal is an uncountable set. Conversely, a discrete time signal has a countable domain, like the natural numbers.

Basic idea

In the following discussion, we will focus on low-pass filtering only.

Let’s start from something simple. If the time series is

[10, 10, 10, 10, 10, 10, 10, 10, 10, 10], what is it?

However, if there is noise introduced, the time series is

[10, 11, 10, 9, 12, 10, 9, 9, 10, 11]

How can we extract the original signal and get rid of noise signal?

Simple moving average filter

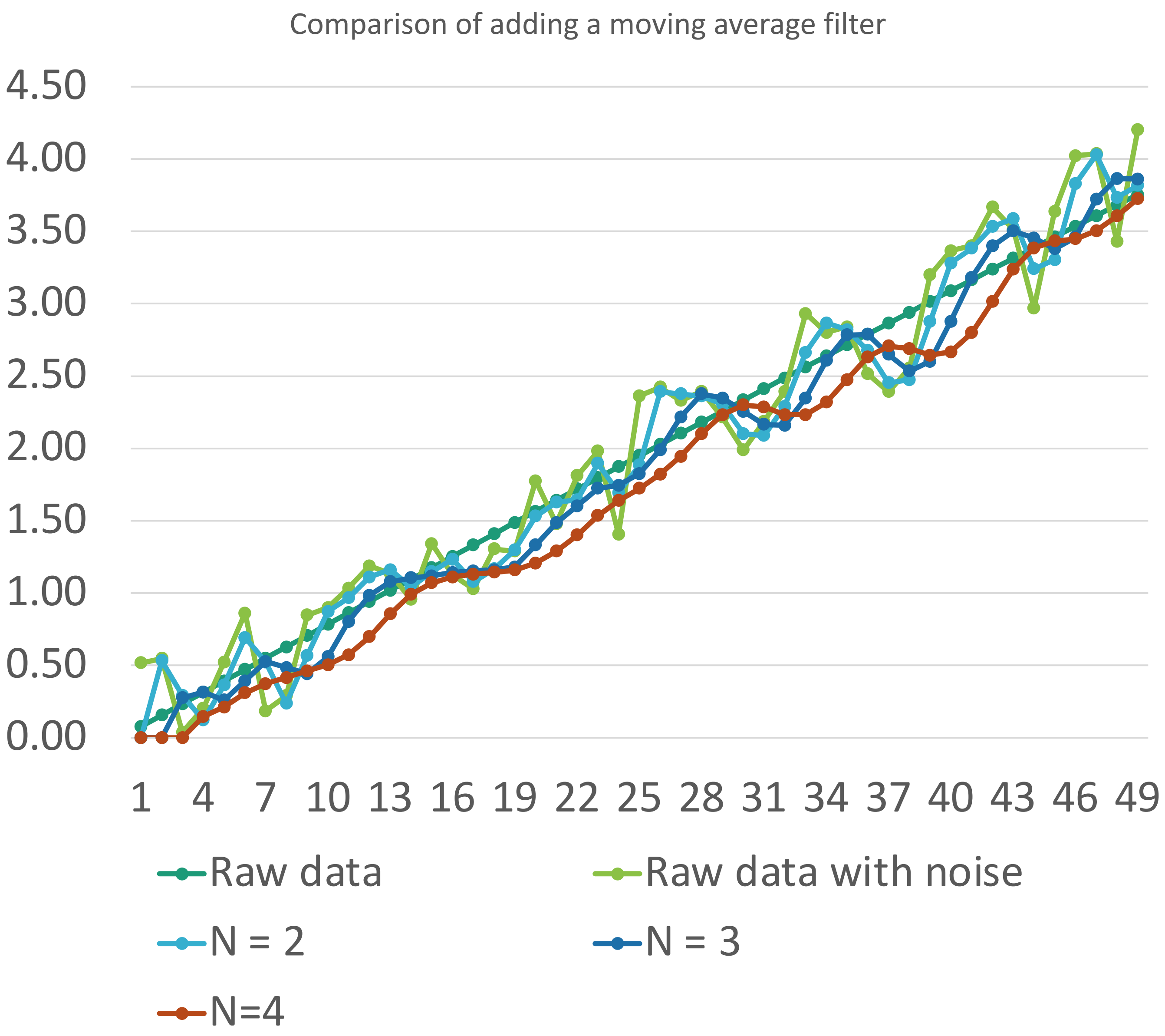
The output of the filter is the average of consecutive data.

For example, if the no. of data in averaging is 2, then

$$y_n = \frac{x_n + x_{n-1}}{2}$$

Actual	With noise	% error	Average	Average	%error
10	10	0%	10	10	0%
10	11	10%	(10+11)/2	10.5	5%
10	10	0%	(11+10)/2	10.5	5%
10	9	-10%	(10+9)/2	9.5	-5%
10	12	20%	(9+12)/2	10.5	5%
10	10	0%	(12+10)/2	11	10%
10	9	-10%	(9+10)/2	9.5	-5%
10	9	-10%	(9+9)/2	9	-10%
10	10	0%	(10+9)/2	9.5	-5%
10	11	10%	(11+10)/2	10.5	5%

Example - Comparison with different values of *N*



Difference Equation

The moving average filter is an example of a difference equation.

The difference equation is to discrete signal processing what the differential equation is to analogue signal processing.

In this particular filter, the values that multiply the input values are all the same. That is the coefficients of the filter are all the same. b_k is the same.

$$y_n = \sum_{k=0}^{2} b_k \, x_{n-k}$$

There are filters with different b_k .