FORMULAS

Nyquist frequency: $f_{Nq} = \frac{f_{sampling}}{2}$

Frequency resolution: $\Delta f = \frac{1}{T}$

Acquisition time: $T = \frac{1}{\Delta f} = \frac{N}{f_{Sampling}}$

Check to aliasing: $f_{sampling} > 2 \cdot f_{max\_signal}$ or $f_{Nq} > f_{max\_signal}$

Check to leakage: $\frac{f_{signal}}{\Delta f} = integer\ number$

Electrical resolution: $Res_v = \frac{range}{2^n}$

Resolution in E.U.: $Res_{E.U.} = s \cdot$

Least significant beat: $LSB = \frac{range}{2^n}$

Cut off frequency: $f_{cut\ off} = f_{max\_signal} \cdot 1{,}1$

this value can be chosen as we want, the most important thing is the fact that it must be higher than zero

Sampling frequency: $f_{sampling} = 2{,}56 \cdot f_{cut\ off}$

Frequency of the signal: $f_{signal} = \frac{1}{T_{signal}}$

Number of samples: $N = T \cdot f_{sampling}$

Conversion: $1\frac{rad}{s} = \frac{1}{2\pi}Hz$ $1rpm = \frac{1}{60}Hz$

Angles relationships: $\sin\alpha \cdot \sin\beta = \frac{1}{2}[\cos(\alpha - \beta) - \cos(\alpha + \beta)]$

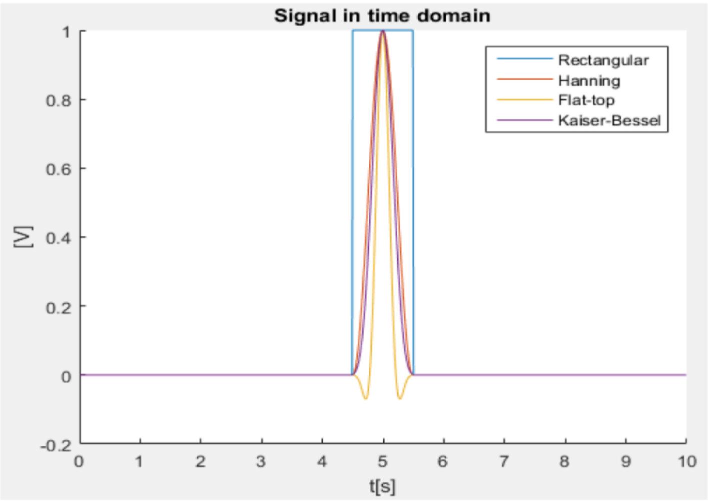
Inside the sine or the cosine: $number \cdot \pi \cdot t = 2 \cdot f \cdot \pi \cdot t \rightarrow f = \frac{number}{2}$

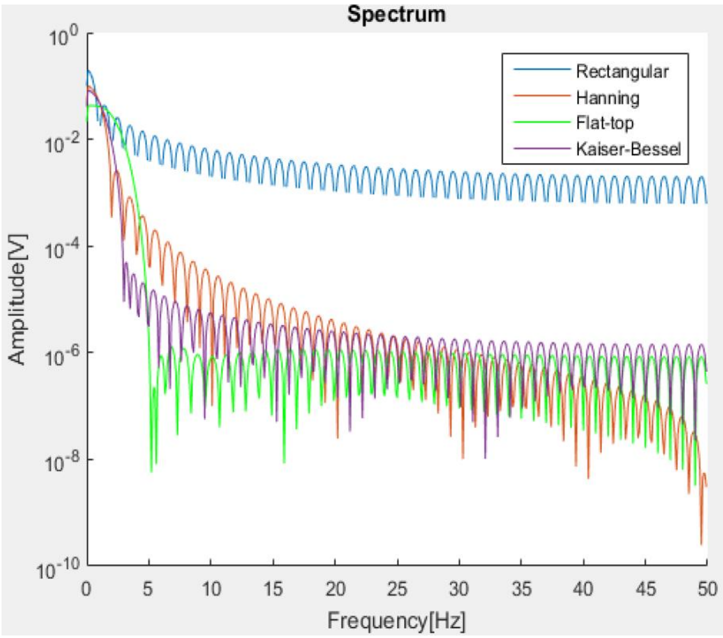
Range: must be chosen when we know the output of the transducer as a range in which it fits. This range can go form 0 to a positive number of form a negative value to a positive one and this depend on the type of measure that we are doing. If we are measuring a pressure or a mass, for example, we cannot choose a negative range because this wouldn't have any meaning.

We have to put an anti-aliasing filter if they don't' tell us how many components we have or if we have some noise. If we are sure to have only that component, we can decide to not put the aliasing.

DEFINITIONS

- Sampling = is the process of extracting meaningful data from a population.
- Aliasing = it's a problem that occurs during the acquisition process according to which we are not able anymore to reconstruct correctly the original signal. To check if we have aliasing, we should the Shannon's theory. In general, to avoid this problem we have to put an anti-aliasing filter between the transducer and the ADC.
- Clipping = it's a problem during the acquisition phase that can occur when the choice of the range is wrong and especially it occurs when the amplitude of the signal that we are acquire exceed our threshold and so when it is higher than the range that we have chosen.
- Leakage = it's a problem that can occur when we do a frequency analysis. It consists in the smearing of the energy all over the spectrum. It occurs when the frequency resolution is not a integer submultiple than each frequency of the signal. We can solve it with two different methods:
  - Changing the frequency resolution in order to increase the acquisition time. But this could lead us to have an acquisition time too much long and so we cannot be anymore sure to have a stable signal
  - Apply a window
- Masking = it's a phenomenon that occurs when we have leakage according to which all the real values of amplitude are "masked" from the fake ones.
- Windowing = it's a technique used to reduce leakage. It consists in multiplying the original signal per a function in the time domain or, in the frequency domain, in consist in making a convolution between the spectrum of the original signal and the spectrum of the window. By applying a window, we are reducing the leakage, but we are underestimating the overall energy of the signal. We could have different type of window, more selective one if we are more interested in the frequency than the energy or the less selective one if we are more interested in preserving the energy. We could have a rectangular, a hanning, a flat top window and so on and so forth.





- RMS average = it's a method that can be applied to reduce noise. Of course, in order to apply it we need a lot of measures of the same phenomenon. In this type of average, we take into account only the real part of the value that is the output of the FFT. By doing this we will not be able to eliminate completely the noise, since every value has a positive sign, and so we'll always have a carpet of noise. This carpet of noise could also mask some main frequencies of our initial signal. Anyway, it's a quite good method to reduce the noise but not to completely eliminate it and to have a more accurate measurement with a quite low computational cost

- Complex average = this type of average is done in order to decrease the noise. In this case we use the entire output of the FFT and so both the real and the imaginary part. For this reason, it's also called vectorial averaging, since we are dealing with vectors. This method is able to completely reduce the noise. This approach requires a quite important computational costs and it also require a phase reference because otherwise all the signal tends to zero and not only the non-deterministic components.
- Synchronous acquisition =
- Power spectral density = it is the process of normalising the spectrum with respect to the frequency resolution. It can be computed as the ratio between the autospectrum and the frequency resolution. The autospectrum is the product between the complex vector output of the FFT and its conjugate one.