# Signal Filters for Measuring Electrical Parameters in Power Electronic Converters

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## I. Introduction to Signal Filtering

Filtering is a basic operation to signal acquisition. Is a key part of the signal conditioning process (along with signal adapting, protection, etc.). Filter can be done by means of analog or digital filters.

Analog, hardware filters are essential in most applications. We assume a digital, periodical sampling of a given electrical signal. Analog filtering is the only manner of removing noise components above Nyquist frequency, that cannot be removed by software filtering. If the behavior of the filter is required to vary, then the hardware itself needs to be changed.

## II. Analog Filters

The electrical parameters enter a specific circuit that provides one desired transfer function with a predefined behavior in frequency domain. This behavior is characterized by poles and zeros, with specific frequencies that define the filter.

These filters can be either passive or active filters.

## III. Passive vs. Active Filters

### *a) Passive Filters.*

Passive filters, are formed only by passive elements (resistors, capacitors, inductors). They are simple, cheap filters. These kind of filters are used in power circuits, but not really in signal conditioning. The most important drawbacks are:

- They are very sensitive to the component value tolerances.
- For low frequencies, the values of R and C can be quite large, leading to physically large components.
- A first or second-order filter may not give adequate roll-off (6dB at the corner frequency…)
- If gain is required in the circuit, it cannot be added to the filter itself.
- The filter can potentially have a high output impedance. Since the resistor value is typically large, to keep the capacitors a reasonable value, the next stage device can see a significant source impedance. An op-amp could be added at the output, but then it is actually an active filter. Moreover, why adding the op amp just at the output when it can be used to improve the filter's performance in addition to lowering the output impedance? This give rise to active filters.

### *b) Active Filters*

These signal filters are implemented by means of operational amplifiers. They provide better selectivity, tight gain and , generally speaking, a perfectly matched frequency response and almost-ideal impedance behavior to the measured system. The main drawback is the more complex design, and the need for a power supply for the Operational Amplifiers (that might introduce some noise as well).

In any case, they are the most used option for signal conditioning applications.

## IV. Active Filter Topologies

There exist some schemes, known as ***electronic filter topologies***, used to implement 2nd order active filters (any higher order filter can be implemented by cascading these topologies). The most used topologies are:

### *a) Sallen-Key*

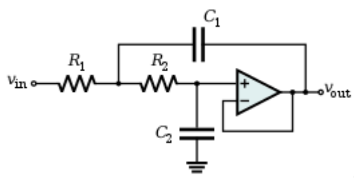
The unity-gain Sallen-Key inherently has the best gain accuracy because its gain is not dependent on component values. Provides a 2nd order filter with one opamp, 2 capacitors and 2 resistors. It is simple, yet not recommended for applications that require a very high Q. the GBW of the selected opamp must be relatively high to have a good filter performance.

### *b) Multiple Feedback*

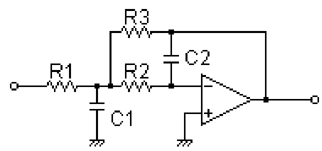
This topology is generally preferred because it has better sensitivity to component variations and better high-frequency behavior. Is slightly more complicated than the Sallen-key, but ensures better Q. However, it is an inverting circuit.

### *c) Other topologies*

There are more complicated topologies (e.g. State-Variable and Biquad architectures), but are beyond this document.




a)




b)

Figure 1.a) Sallen-key filter. b) Multiple feedback filter.

## V. Filter Responses

There are also different **Filter Response**s, e.g. types of filters, understood as transfer functions that can correspond to filter circuits. Provided that these transfer fucnitos are $2^{nd}$ order expressions, then they can be implemented by the aforementioned topologies.

If an ideal low-pass filter existed, it would completely eliminate signals above the cutoff frequency, and perfectly pass signals below the cutoff frequency. In real filters, various trade-offs are made to get optimum performance for a given application.

The real responses that can be implemented are:

*a) Butterworth Filters*

Butterworth filters are termed maximally-flat-magnitude-response filters, optimized for gain flatness in the pass-band. the attenuation is –3 dB at the cutoff frequency. Above the cut-off frequency the attenuation is –20 dB/decade/order. The transient response of a Butterworth filter to a pulse input shows moderate overshoot and ringing.

*b) Bessel Filters*

Bessel filters are optimized for maximally-flat time delay (or constant-group delay). This means that they have linear phase response and excellent transient response to a pulse input. This comes at the expense of flatness in the pass-band and rate of roll-off. The cutoff frequency is defined as the –3 dB point.

*c) Chebyshev Filters*

Chebyshev filters are designed to have ripple in the pass-band, but steeper roll-off after the cutoff frequency. Cutoff frequency is defined as the frequency at which the response falls below the ripple band. For a given filter order, a steeper cutoff can be achieved by allowing more pass-band ripple. The transient response of a Chebyshev filter to a pulse input shows more overshoot and ringing than a Butterworth filter.

For this example, the selected filter is a second order low-pass filter, using an analog, active stage), with Sallen-Key topology configuration, and Butterworth response. This filter is located after the adaptation stage (to filter also possible switching noise coming from the DC sources at the OpAmp of the adapting stage).

## VI. Switching Frequency vs. Sampling frequency

Prior to implement the filter, the selection of the corner frequency must be carried out. An analysis of the system frequencies is initially required. Figure 2 shows the most important frequencies for a given application.

In our example, the application is the measurement of the current through an inductance, that actually is controlled in an inner cascaded loop scheme, of the converter. The converter is tied to the grid to deliver the power from a PV panel. The converter is switching a t a fixed switching frequency.

*a) Switching Frequency*

The power converter is switching at a given switching frequency, $f_{SW}$ (e.g. 50 kHz). This provides noise to the system. If synchronous sampling is used, theoretically it can be not considered to affect. (but switching noise might extend up to the sampling instant…).

*b) Sampling Frequency*

The sampling frequency, $f_S$, is the rate at which the measurements are taken and processed by the digital system. The $f_S$ selection largely depends on the control strategy that is intended for the converter, and on the waveform shape of the magnitude to measure.

Assuming PWM triangle generation, then the inductor current follows the following triangular waveform shape.

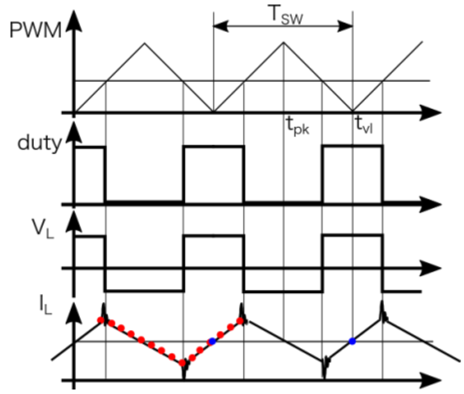



Figure 2.

For **peak (current) control**; usually the current waveform is analogically measured and compared to a peak value by means of an analog comparator. Therefore, it is not a digital sampling process.

For **average control** (the most usual one, and the one targeted in this discussion), there are two approaches:

- The triangular waveform is analogically averaged, to remove the ripple, switching noise, etc. For instance, it can be attenuated to a 1% of the ripple at the switching frequency. Then it can be sampled. The sampling frequency in this case can be lowered down to the corner frequency of the averaging filter, to ensure:

$$f_{NYQ} < f_{SAMPLING} < f_{AnalogFilter} << f_{SW} \qquad (1)$$

- Alternatively, by using **synchronous sampling**, several benefits are ensured. First, the switching noise is automatically removed if the phase instant is chosen adequately at the siwtching period. At the example (blue dot), by sampling at instant $t_{pk}$ (peak time) of the triangular waveform, the switching instants are as far as possible from the sampling instant. In addition, the measured value is automatically the average one. It is required that the microcontroller has the ability of providing synchronous sampling features. Then the sampling frequency equals the switching frequency:

$$f_{SAMPLING} = f_{SW} \qquad (2)$$

The sampling frequency is pushed to the highest possible value for average control of the converter.

In a general case (forgetting the converter application, but focusing on sampling a triangular waveform), may be this triangular waveform must be exactly represented. (i.e. triangular shape storage and representation required). Then, the highest harmonic targeted must be defined. The sampling frequency required to acquire this harmonic is given by the Nyquist frequency. That is to say:

$$f_{Triang} << f_{NthHarmonic} < f_{NYQ} < f_{SAMPLING} \quad (3)$$

Thus, to store a true representation of the triangular waveform, the current must be sampled many times in a switching period. This is, usually, not required at all. (Red plots in figure, 14 samples per period, for reaching the 7th harmonic.).

*c) Nyquist Frequency*

This Nyquist frequency, $f_{NYQ}$, is, by definition, equal to $f_S/2$. The information entering the sampling process has components only below $f_S/2$. In order to avoid aliasing, it must be ensured that all noise components above $f_S/2$ are removed. A way to achieve this is by means of a hardware filter. The corner frequency of the filter must be related to this Nyquist frequency.

*d) Control Frequency*

In this case, the current loop control bandwidth, BWi denotes the highest frequency at which the control must be performing adequately. The desired bandwidth of the current controller. For a proper cascaded control, the BWu bandwidth must be around 10 times smaller, and the grid frequency must also be 10 times smaller than BWu.

Finally, for Synchronous Sampling, the following frequencies are given by figure 3.

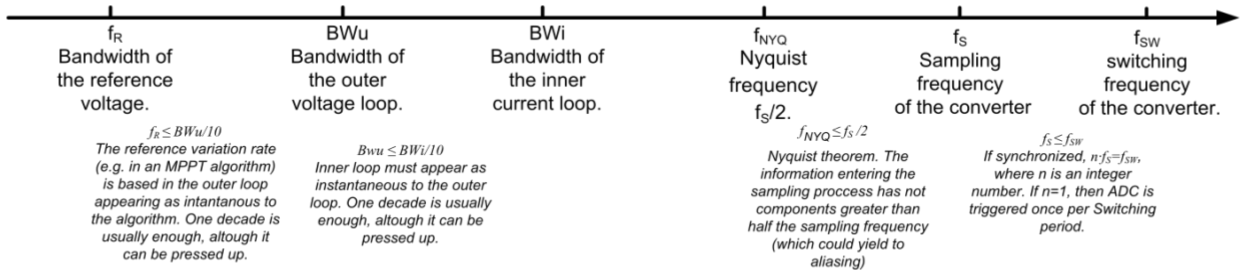



Figure 3.

## VII. Antialiasing Filter Characterization and Corner Frequency Selection

So the question is how to settle the gains/frequencies of the Antialiasing Filter, AAF, for adequate operation of the system. In order not to affect the measurement, the gain is assumed to be 0dB at low frequencies.

A 2nd order filter is targeted, as mentioned before. This filter can be characterized as shown in picture 4, considering that:

$$f_C = f_{AAF} \quad (4)$$

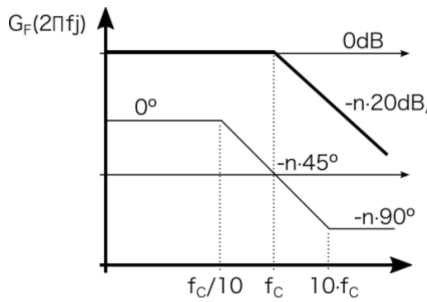



Figure 4.

The absolute maximum/minimum limits for the values of this frequency are given by:

*a) Lower Limit for $f_{AAF}$*

At $f=f_C$, for a second order filter, there is a phase shift of 90º. In order to ensure that the phase shift of the AAF filter does not affect the control performance (limited at BWi), then at this BWi frequency, the phase shift must be small. Thus,

$$BW_i \leq f_C/10 = f_{AAF} / 10 \Rightarrow BW_i << f_C \quad (5)$$

If this is not fulfilled, then the phase margin of the control loop might be significantly reduced, and this implies a higher overshoot and, eventually, instability.

*b) Higher Limit for $f_{AAF}$*

Assuming an ideal filter (all frequencies filtered above $f_C$), then the theoretical highest limit for $f_C$ is $f_{NYQ}$. The 2nd order is obviously non-ideal, but the synchronous sampling provides enough safety margin as to guarantee to some extent that no aliasing is present. This is generally accurate for not too noisy systems, using synchronous sampling far from the switching instants. Therefore, the theoretical limit is given by $f_C = f_{AAF} \leq f_{NYQ}$.

Thus, the $f_C$=$f_{AAF}$ must be located fulfilling:

$$10 \cdot BW_i \leq f_C = f_{AAF} \leq f_{NYQ} \quad (6)$$

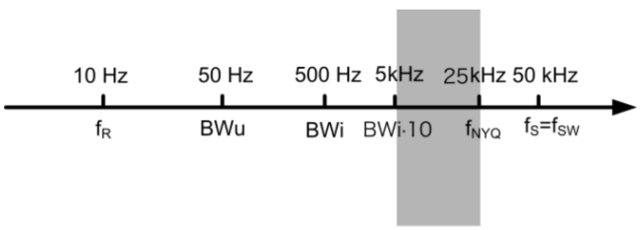



Figure 5.

As a conclusion, the frequency of the AAF filter must be placed at the grey area in figure 5, fulfilling:

$$10 \cdot BW_i \leq f_C = f_{AAF} \leq f_{NYQ} \quad (7)$$

Initially, a rule of thumb is to locate $f_{AAF}$ as close as possible to $f_{NYQ}$. It is suggested to start with $f_{AAF}$=$f_{NYQ}$, and if noise issues appear in the experimental setup, then push the $f_{AAF}$ towards smaller frequencies. In any case, always ensure $10 \cdot BWi \leq f_C$; if still there are noise issues, increase the order of the filter.

## VIII. Digital Filters

The exact digital filter implemented depends on the application, however it must be remarked that these filters cannot avoid the aliasing effect, and therefore they are generally used with an analog filter prior to the ADC block.

After the signal is converted to the digital domain, and provided that aliasing has been avoided (by using an analog, hardware antialiasing filter), some magnitudes require a further digital processing. These digital filters are interesting particularly in the following cases:

- In notch filters, where a specific band of frequencies must be ignored. In this case, the control scheme must be "blind" to these frequencies, i.e. the control is designed not to react to them. An example is mechanical resonances in machines, avoi9ding the drive to react to this. Another example might be grid harmonics, etc.
- Another example is an noisy encoder, avoiding the noisy frequencies and focusing on the ones interesting for the information of the position of the shaft.