Filtering on Dewesoft HS amplifiers

Sirius HS has a fast 1 MS/sec SAR type AD converter. That means that the ADC by itself doesn't filter the incoming signal, but acquires it »as is« at the time of sampling.

This is very useful in some cases where we want to have bandwidth of the signal as high as possible, for example when capturing transients, non sine-wave (rectangular, triangular) signals, when acquiring voltage and current from frequency inverters and others.

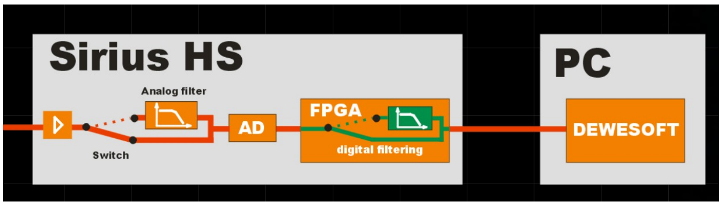
Then the bandwidth must be as high as possible. With Dewesoft HS HV and LV the bandwidth of the amplifier is as high as 2 MHz, ACC module has 500 kHz ...

Of course this kind of amplifier startegy passes through also higher harmonics, which in other cases (when we are interesting in real frequency contents of the signal) doesn't work well. In such cases we need the strong anti-alias fitlering of the signals. What makes it even more difficult is that each application requires different cutoff frequencies.

We have solved this issue in HS amplifiers by adding optional hardware (analog) sharp 100 kHz filter, which can be switched in measurement chain.

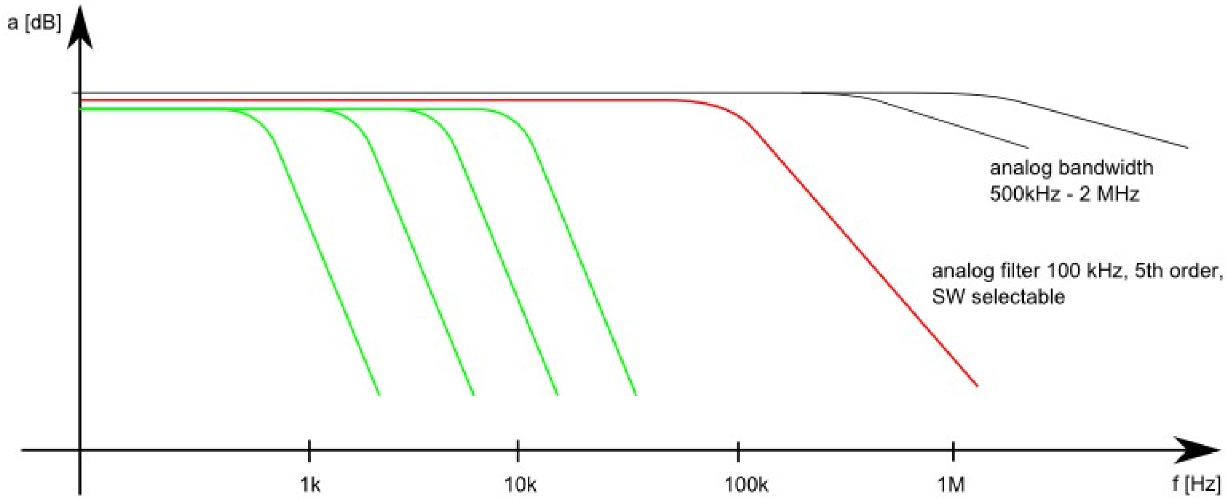
This 5th order filter cuts all unwanted frequencies above 400 kHz. On top of that Dewesoft can set internal FPGA to calculate additional shar 8 th filter with variable cutoff frequencies and shapes. In this case we can be sure that we have a brick-wall alias free data acquisition with just a simple choice in the software.

In addition to that the FPGA will downsample the data before sending it to software. Therefore Sirius is sampling data always with or close to the full speed, filter the data and downsample it, providing alias free acquisition with any sample rate.

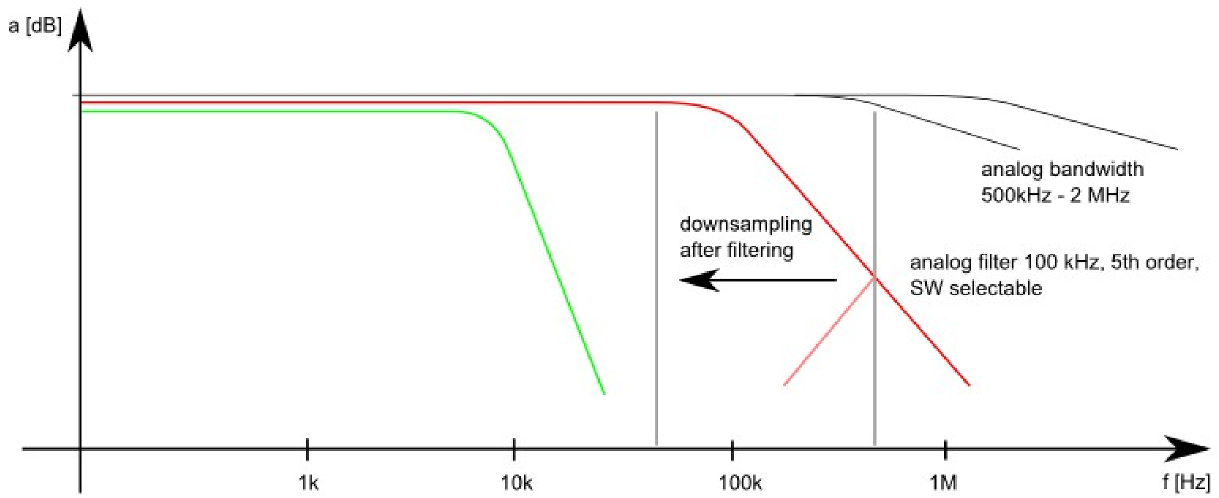


Let's look how that looks in frequency domain. The black curve is the analog bandwidth of the amplifier. When filters are switched to OFF, then we can acquire also high frequency content.

After that we can switch in the 100 kHz analog filter (red curve), cutting off the high frequency content in analog world before ADC conversion. To be fully flexible, Sirius has programmable digital filters which can be programmed to virtually any frequency, any sharpness and different filter charactersistics.



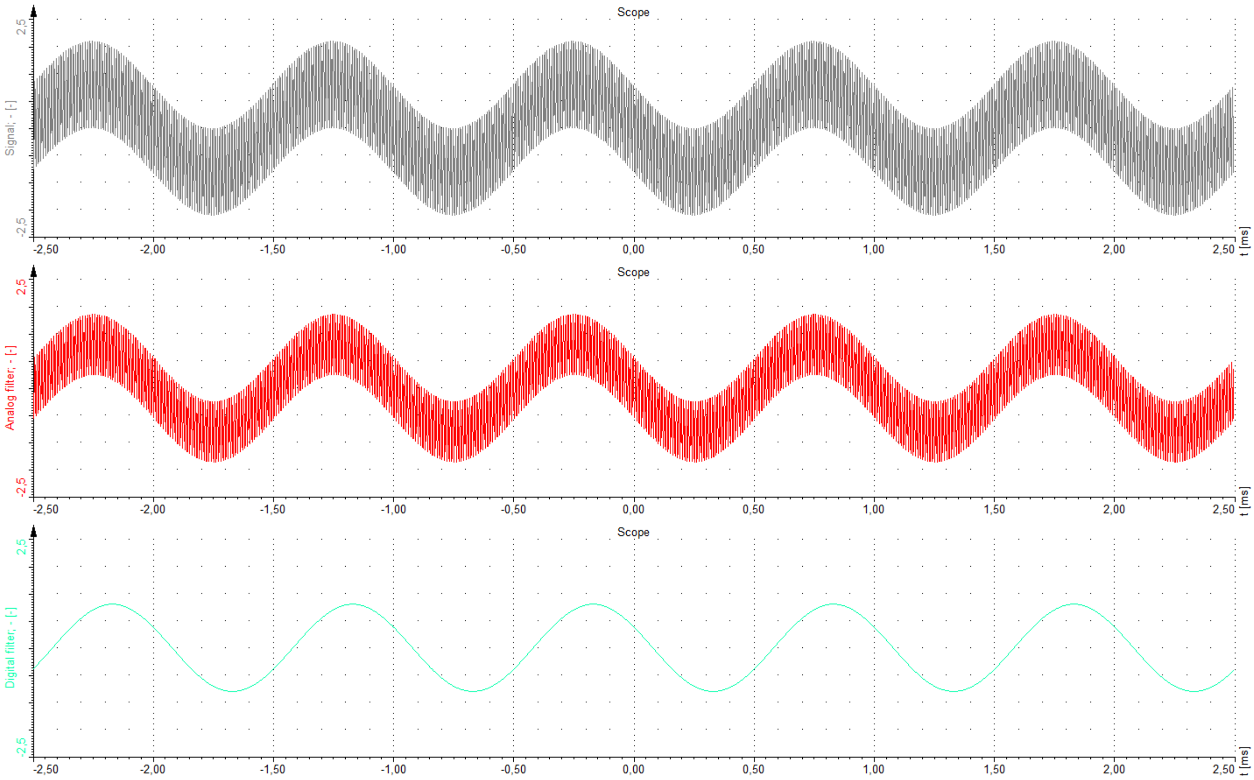
So let's look at one example. Let's say we want to sample with 100 kHz and have a signal bandwidth of 10 Khz. Sirius will still sample the data with 1 Mhz, analog filter will cut off all frequencies above let's say 400 Khz, making sure that below 100k the data will be alias free. After that the digital filter will give the wanted bandwidth and in the last stage the data, already alias free, will be downsampled to 100 kHz.



So let's look to this example in time domain. We have mixed signal of 1 V with 1 Khz and 1V of 100 kHz.

The analog filter will already cut the 100 Khz (and of course all above signals).

What is left will be cut with the digital signal, giving a perfect alias free result.



With this technology you have two amplifiers in one – a very high bandwidth amplifier to capture transients and alias free amplifier for standard data acquisition and frequency analysis.