Filtering on DEWESoft Dual Core and HD amplifiers

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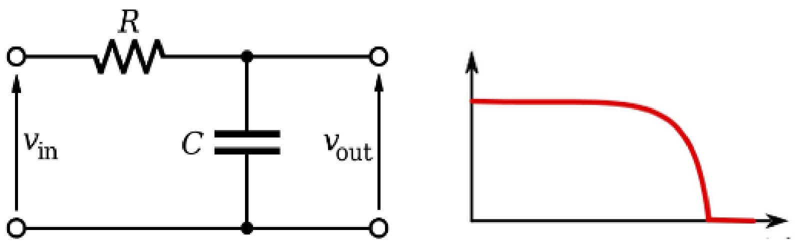

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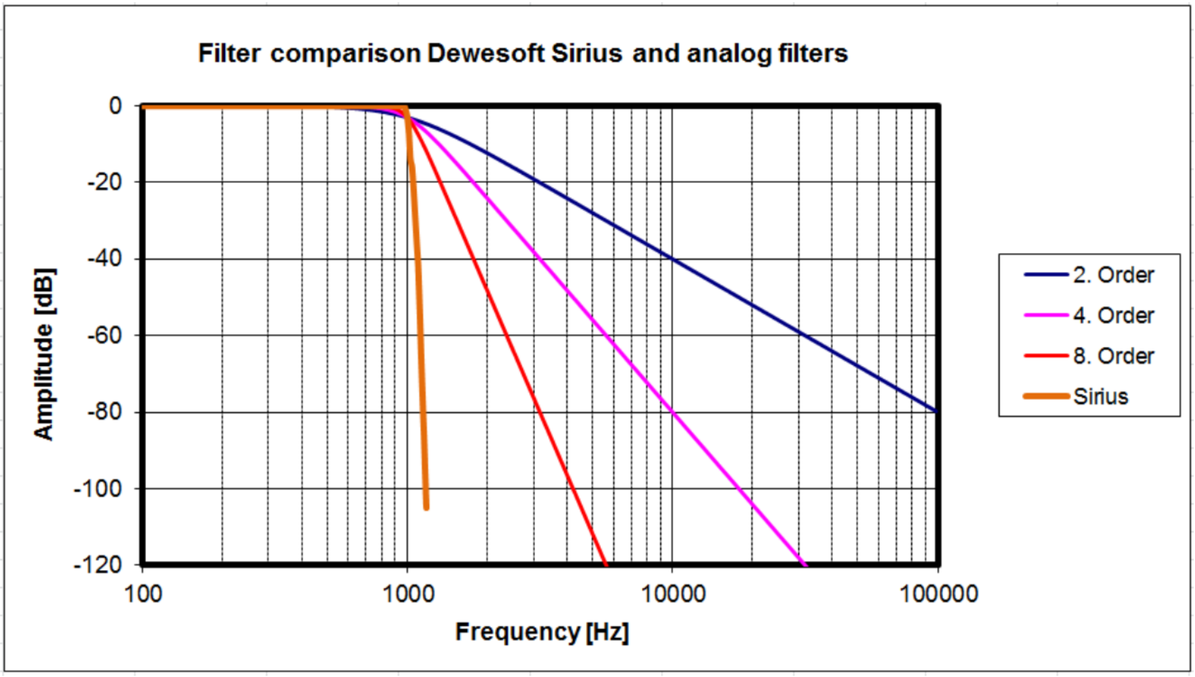
In many applications, it is absolutely necessary to guarantee aliasing free acquisition to avoid misinterpretation in the frequency domain analysis of the measured data. Aliasing effects are seen if the input frequency of the measured signal is higher than half of the acquisition rate.

Therefore, on one side we need to cut all frequencies in the signal higher than half the sample rate, but on the other side, we should not damp the amplitude of the signal with frequencies lower than half of the sample rate. High filter damping rates can be reached with higher filter orders.

Analog filters can be designed up to the 8th order with selecting all components very carefully, but even this high-quality analog filter is by far not as sharp as the filter technology used in a Sirius. The diagram below compares the filter of a Sirius with analog filters. You see nearly no damping on up to 1 kHz input frequencies and very sharp damping above 1 kHz.



How is this filter behavior possible at all if it is not analog? To get a filter characteristic like the one shown in the above picture we need a different sort of technology.

First, we need digital FIR (Finite Impulse Response) filters to create sharp filters. But the digital filter also requires aliasing free data, which can be guaranteed using oversampling technology. That means that the internal ADC sampling rate is higher than the output data rate.

The Sirius amplifiers from DEWESoft combine both of these technologies together for getting guaranteed aliasing free acquisition. The digital filter's cut-off frequency is automatically set to half of the sample rate. The internal sampling frequency is up to 256 times higher than the output rate and depends on the sample rate like shown below.

Digital Filter (vs. Sample Rate)	1 Ks/S ... 50 Ks/s	50 Ks/s ... 100 Ks/s	100 Ks/S ... 200 Ks/s
Bandwidth (-3 dB)	0.494 fs	0.49 fs	0.38 fs
Alias-free Bandwidth	DC to 0.42 fs	DC to 0.32 fs	DC to 0.22 fs
Alias Rejection	-95 dB	-92 dB	-97 dB
Delay through ADC	12/fs	9/fs	5/fs
Oversampling	256	128	64

Depending on the sample rate, the analog filter at the input of the ADC is selected automatically. Due to the above-mentioned oversampling technology, only two filters with 5 kHz and 100 kHz are needed. When the sampling rate is equal or below 2 Ks/S the filter frequency will be set to 5 kHz, otherwise, the filter will have a frequency of 100 kHz.

The diagram below gives an overview of the different filter and sample rate settings in the frequency domain:

- the sample rate of Sirius is set to 2 kS/sec,
- the filter of the Sirius is automatically set to 1kHz for the measured data,
- the internal sampling frequency of the Sigma-Delta ADC is 512 kS/sec,
- the internal the analog aliasing filter (5kHz) guarantees aliasing free acquisition.