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PAK

ANTI-ALIASING

Aliasing occurs when frequency components which are higher than the Nyquist frequency, are visible in the sampled signal level gradient. Caused by interferences on the measurement hardware, it can have serious consequences, as in the engineering misinterpretations of the signal might be used. Aliasing is an effect which occurs when sampling an analog signal. It is important to recognize that there is no way to prevent this effect from happening. The only thing we can do is to minimize its effect. This white paper is intended to explain the use of appropriate anti-aliasing filter.

Basics

A common misunderstanding is that aliasing can be prevented using a **"high enough" sampling rate**, making sure that the frequencies of interest lie below the Nyquist Frequency. For acoustics one might, for example, argue that 50-100 kHz should be more than sufficient to capture all relevant (NVH) frequency content. There is however no guarantee that sensors / cabling won't pick up (EMI) noise with operating frequencies in Mega or Giga Hertz region. Picked-up, these very high frequencies Mirror / Alias onto your FFT spectrum falsifying your analysis.

So what should front-ends do to minimize the effect of aliasing? The answer lies not in digital signal processing but in the analog world: Application of a proper **analog low-pass filter**. Indeed, once transformed into the digital domain, there is nothing one can do anymore to distinguish **aliasing effects**. As a consequence we can only tackle the problem in the analog world.

A low-pass filter (in this context commonly referred to as **anti-aliasing filter**) can be a relative simple circuit of resistors and capacitors. The circuit makes sure that frequencies above the maximum frequency of interest (**Nyquist Frequency**) are severely reduced in amplitude (reduction in the order of a million for example). Although therefor still present after sampling, the **aliasing effect** on the FFT spectrum is minimal.

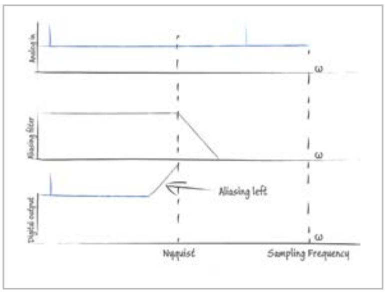

Figure 1:
Anti-Anliasing filter

A typical anti-aliasing filter is shown in figure 1. It has a flat response below the Nyquist Frequency with a gain equal to 1. Above the Nyquist frequency, the filter kicks in and tries to diminish any frequency content. Please notice that there will always be a certain "grey area" approaching the Nyquist Frequency where the signal is not yet fully attenuated. The low-pass filter simply can't create a complete sudden drop at the Nyquist Frequency in practice. For this reason, very often dynamic analysers use a factor less than the Nyquist Frequency (½ sampling frequency). In PAK, a "safety factor" of the sampling frequency divided by 2.56 is applied making sure that our module specifications are met in the entire bandwidth.

In the past, high order low-pass filter had been developed – built out of several electrical components. In order to attenuate the frequency range above the Nyquist frequency sufficient enough, the slope after the cut-off frequency should be steep. This can only be realized with electrical components having very tight specifications, and are therefore very expensive. Moreover, as we are talking about electrical components with very tight specifications, the properties of the filter become temperature dependent due to slight variations in, for example, the resistance of the resistor. So in the past, users had to wait before their data acquisition systems had come to operating temperature to be able to use them.

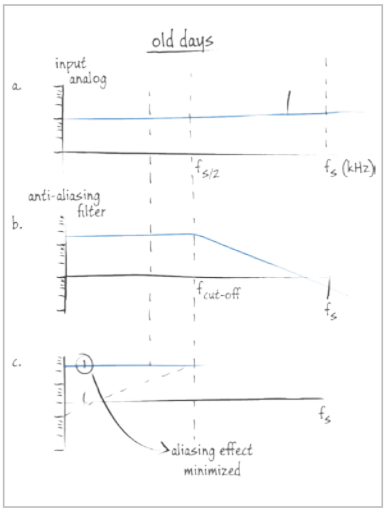

Figure 2a:
Filter characteristics of the past

Figure 2a shows the filter characteristics as used in the old days. The filter kicks in at about half of the sampling frequency. Aliasing effects caused by frequency content close to the sampling rate are minimized, but only to a certain extent.

Multi-Sampling Rate Methods

Nowadays, these problems are circumvented by using a so called '**Multi-Sampling Rate**' technique. Instead of sampling at slightly more than double the frequency range of interest (order of magnitude kHz), the analog signal is sampled much higher (order of magnitude MHz or GHz).

As a result, only frequency content close to the high sampling rate will mirror back into frequency content of interest. This allows the slope of the low-pass filter to be less steep (lower order filter) and therefore less electrical components with lower specifications are required. This principle is shown in figure 2b. Already a simple filter will reduce the contribution of the signal mirrored back into the signal.

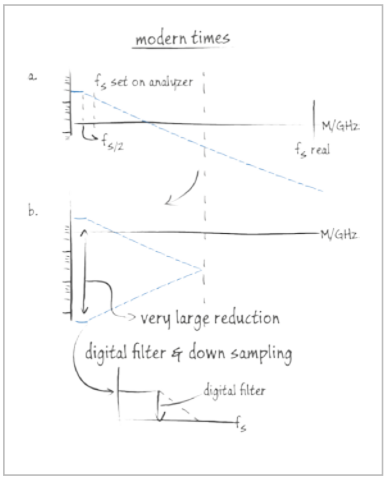

Figure 2b: Downsampling

A consequence is however that we are left with a high-sampled signal of which only a small frequency range is of interest. So obviously, the signal needs to be down sampled to an order of magnitude of kHz.

Down sampling consists of two steps:

1. Low-pass filtering
2. Sample decimation

The latter one is simply "throwing" away a large part of the samples. Each of the decimated samples would however still contain all the frequency content of the originally MHz / GHz sampling. It is therefore less simple as such. Before decimation, it is essential to filter out the unwanted higher frequencies. This is again achieved by applying a low-pass filter. Indeed, for the same reason (aliasing) one needs to filter the analog signal before sampling, the digitized signal needs to be filtered before down sampling.

In other words, downsampling can be understood by realising that every time sample is in fact a summation of the contributions of every frequency component. By simple decimation (throwing samples away), these contributions are not lost and will cause aliasing effects in the down sampled signal. In order to reduce the higher frequency contributions, again a (digital) low-pass filter is required.

As the MHz / GHz signal is already in the digital domain, the low-pass filter is also a digital filter. The accuracy of the digital filter now only depends on the processors **floating point accuracy**. It is not temperature dependent and also (much) less expensive than the analog solution of the past.

Summary

To avoid aliasing, several methods are applicable. In the past, the focus was only on low-pass filter that had steep slope at the Nyquist frequency. As these filter are extremely expensive and sensitive, nowadays another method will be applied that is based on a simpler low-pass filter in the analog **and** digital field:

- Analog low-pass filter to minimize the aliasing in the analog signal
- Oversampling in the Mega- or Gigahertz range
- Digital low-pass filter
- Downsampling back into the original frequency range

This method benefits users – with regard to the standard oversampling – by:

- Expensive low-pass filter with steep slope are not necessary
- Simple downsampling through previous digital transformation possible
- Accurate results

The PAK software offers a multitude of tools for data acquisition and data analysis, especially for the fields of acoustic, vibration, structural and rotational analyses. PAK provides a flexible, effective and compact set of tools for all applications and is most effective in the context of highly standardized tasks and procedures, quality control or troubleshooting.

If you would like to gain a deeper insight into the field of signal analysis and other topics related to the analysis tools provided by Müller-BBM VibroAkustik Systeme, you are welcome to attend one of our practical training sessions. More information can be found at **http://www.muellerbbm-vas.com/services/training/.**

Müller-BBM VibroAkustik Systeme is one of the world's leading suppliers of vibroacoustic measurement technology. We focus on the interpretation of dynamic or physical data, especially in the fields of NVH, strength and comfort. For more than 30 years, our system competence results in innovative solutions that seamlessly integrate into existing environments.

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