Overshoot in data acquisition systems

Effects of filters on sudden signal changes in time domain representation

Summary
A “discontinuity” in a signal, for example a step or transient, is a big challenge when it comes to accurate digital representation of such a signal in the time domain. Understanding physical phenomena, signal filtering and overshoot is important when interpreting amplitudes of a measured signal.

by Aleix Riera, Siemens Industry Software

Contents

Background

A digital representation of a signal starts with an understanding of the Fourier Transform and Fourier series. According to the Fourier series, every periodic function can be **approached** by the summation of a unique set of sinusoidal waves. Some examples are shown in Figure 1.

Notice that some functions in Figure 1, e.g. the square wave and impulse, have infinite frequency content. Both functions have discontinuities, or a sudden step, contained in them.

This sudden transient in an actual signal, and the associated infinite frequency content that comes with it, cannot be correctly replicated with data acquisition systems, with intrinsically finite frequency capabilities. To overcome problems related to that limitation (i.e. aliasing), filters are used in measurement systems.

Filter may then produce overshoot **depending on filter design**.

"Sudden transients in actual signals and associated infinite frequency content cannot be correctly represented on data acquisition systems with finite frequency capabilities"

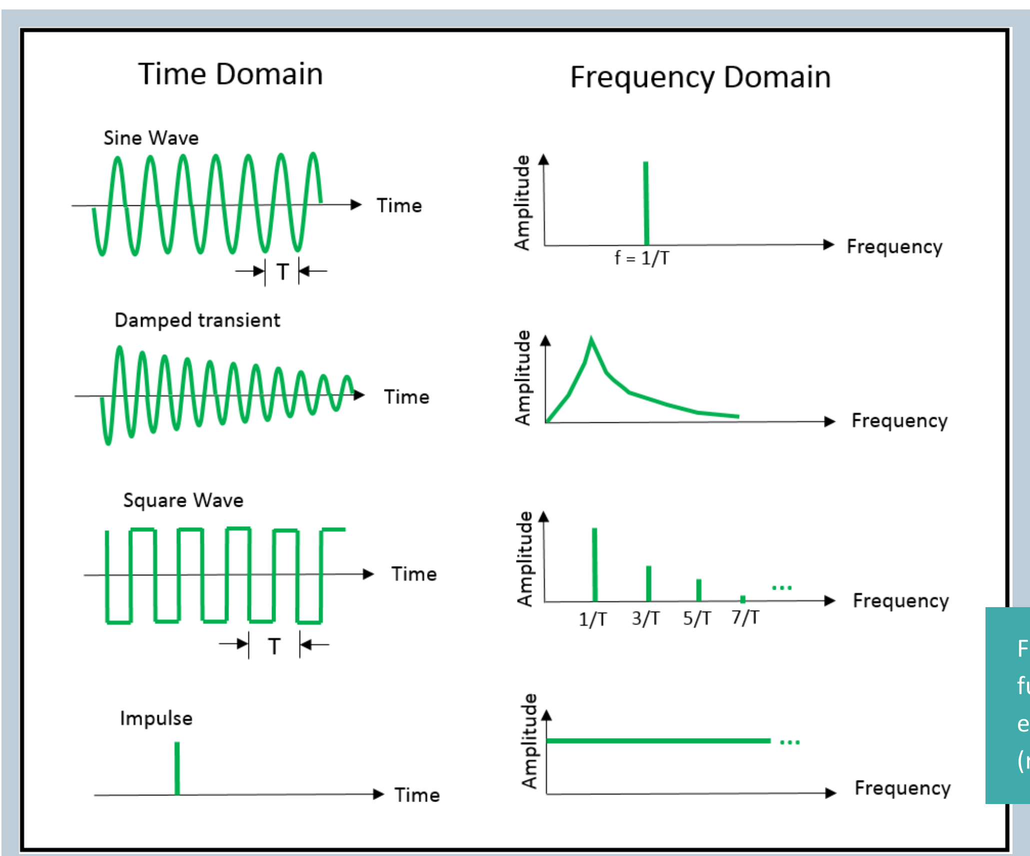


Figure 1: Various time functions (left) and their equivalent frequency content (right).

Theory

In digital data acquisition systems, representation of a continuous, non-periodic time signal by a digital, discrete time signal involves two important steps: sampling and alias protection.

Sampling

First, a time signal is represented by a series of digital samples. Expressed in terms of Fourier transformation, sampling can be modelled as a multiplication of our signal with a **periodic series of Dirac impulses**:

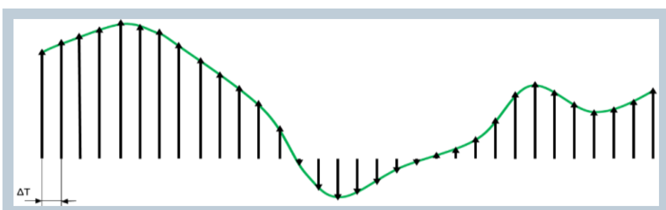


Figure 2: Sampling an analog, continuous non-periodic time signal

The Fourier transform of a continuous function is a complex function (i.e. including a real and imaginary part, or more practical in signal processing, magnitude and phase) in the frequency domain. The Fourier transform of a series of Dirac impulses is **another series of Dirac impulses**, this time in the frequency domain (Figure 3).

“Sampling can be modelled as a multiplication of our signal with a periodic series of Dirac impulses”

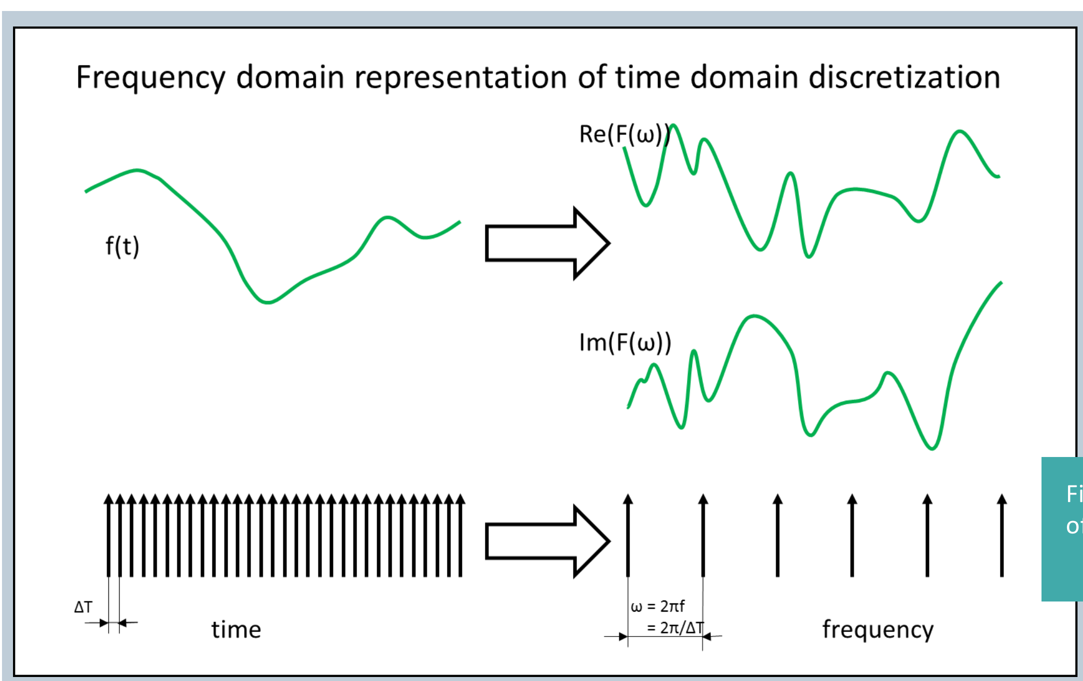


Figure 3: Fourier transform of time functions

A multiplication in time domain corresponds to a convolution in frequency domain, and vice-versa. This convolution of our signal's Fourier transformation with a Dirac train results in the following:

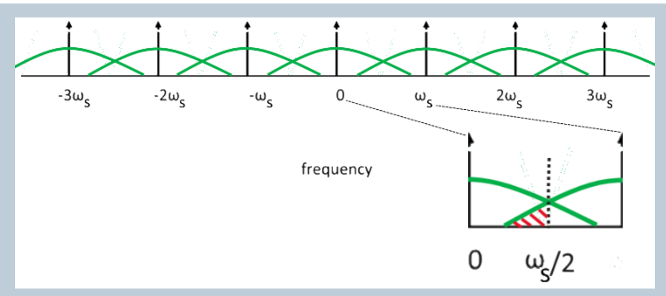


Figure 4: Fourier transform of sampled signal.

In frequency domain, this convolution results in a continuous, **periodic** function, of which only the first period in the frequency domain is of interest and represented by digital data acquisition systems. **Discretization of a time domain signal makes the signal periodic in the frequency domain**[1]. Because of reciprocity, discretization in the frequency domain results in turn in a periodic signal in the time domain. In other words, discretization in frequency domain imposes periodicity of the time domain signal sampled during interval [0, T): Note that in order to represent the frequency domain with a digital device (for example, PC with software), the frequency domain is inherently also discretized.

Next, the Nyquist-Shannon sampling theorem states that for a given sample rate F_S, perfect reconstruction of the signal is possible for a bandwidth $BW < F_S/2$.[2]

Alias protection

Digital data acquisition systems use **anti-aliasing filters** to ensure enough attenuation of the signal, at least, at the Nyquist frequency (being half of the sampling rate $F_S/2$). This way, potential frequency alias folding back from F_S (see Figure 4) will be properly attenuated, and thus not harm our original signal frequency content.

Again, this can be represented by means of the Fourier transform. If we are interested in the frequency domain of a function, a conceptual anti-aliasing filter model is a rectangle (or brick-wall) function in frequency domain:

"If we are interested in the frequency domain, an ideal, conceptual anti-aliasing filter model is a rectangle function in frequency domain"

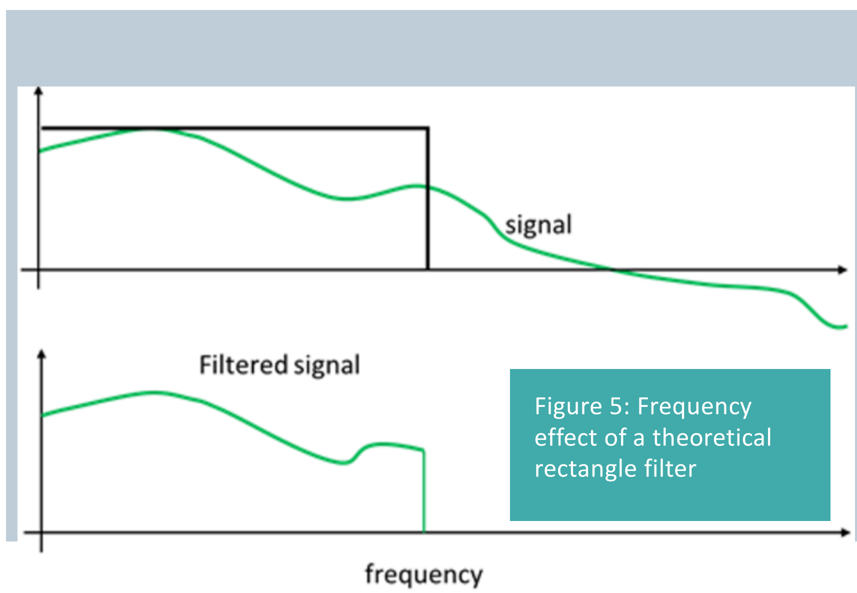


Figure 5: Frequency effect of a theoretical rectangle filter

[1] Note: to force periodicity in time domain, a technique called "windowing" is typically used. This is not further elaborated in this White Paper.

[2] Inversely stated: a signal that contains information with a maximum frequency content of BW, can be accurately represented in frequency domain if sampled with a sample rate of $F_S > 2 \cdot BW$

Multiplying functions in frequency domain corresponds to convolution in time domain.

This means that correct representation of filtering in time domain, corresponds to convoluting the original time signal with the inverse Fourier transform of this filter model. This is how, by the way, actual LTI systems (linear, time-invariant and causal systems) process physical signals. And filters are LTI systems.

The inverse Fourier transform of a rectangle function in frequency domain is a sinc function ($\sin x/x$) in time domain. Figure 7 shows the effect of this convolution on a square wave function. As a result of this convolution, a ringing effect occurs at the transitions.

The Gibbs effect
Adding harmonic after harmonic to build up a square function will result, only at infinity, in a correct representation of the discontinuities. Inversely stated: practically, we cannot add an infinite number of harmonics, so our result will not converge to a correct representation of a discontinuity. This non-convergence results in ringing at the discontinuity. This observation is known as the Gibbs phenomenon, and should not be confused with the topic of this document. Overshoot is a product of a filter.

Further, a square wave can be approached by a summation of an infinite amount of harmonic sine waves:

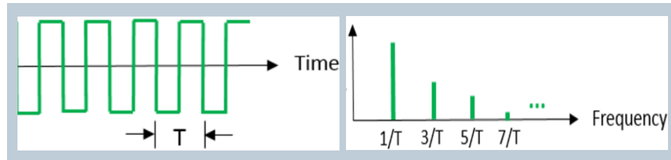


Figure 6: Square is infinite number of oddly spaced harmonics.

Using the square wave function period (T), the first harmonic in a sine wave is at frequency 1/T. An infinite number of harmonics are then added with frequencies of 3/T, 5/T, 7/T, etc. to infinity.

The amplitude of each harmonic is reduced as frequency is increased. All the harmonics, **up to infinity**, are needed to reconstruct the square properly in the time domain.

If some harmonics amplitude and/or phase are modified, then the time domain representation of the function is not exactly square. Digital data acquisition systems cannot represent a signal including all its harmonics up to infinity; an anti-aliasing filter is needed to always meet Nyquist independently of the input signal, and so they will not be able to represent such a signal without some distortion.

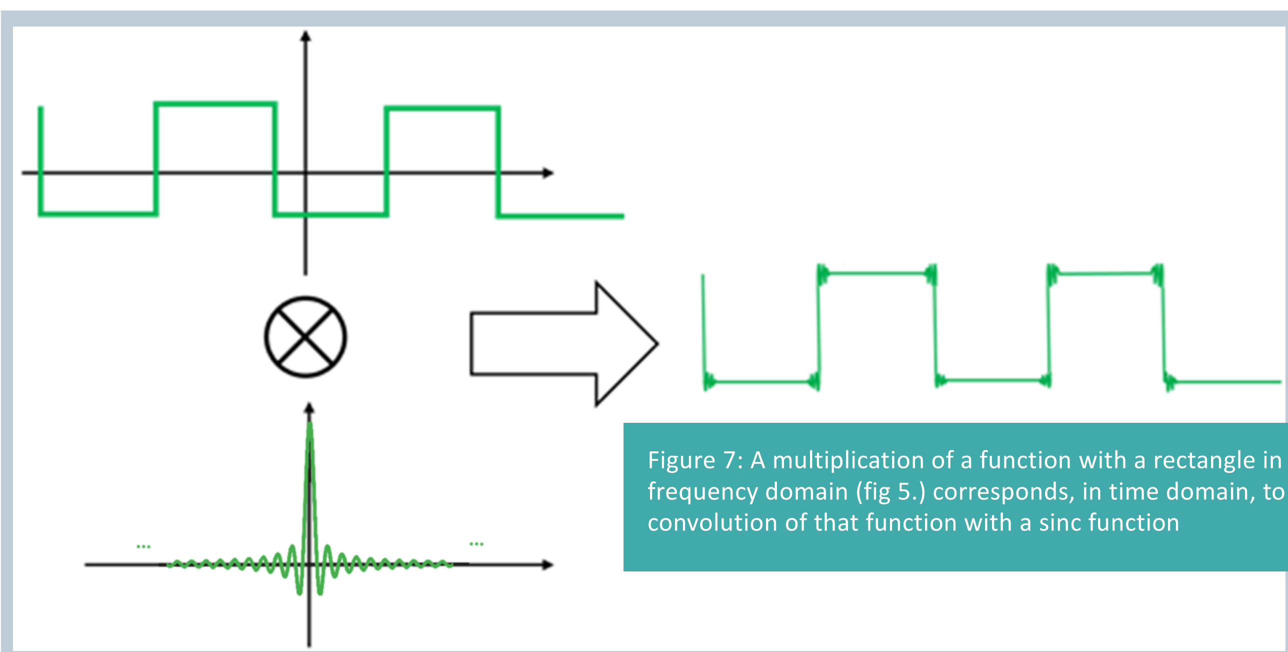


Figure 7: A multiplication of a function with a rectangle in frequency domain (fig 5.) corresponds, in time domain, to a convolution of that function with a sinc function

Overshoot

In the physical world, overshoot can be present in filter outputs as a ringing at the step/transition location in the signal as shown in Figure 8.

In **digital** data acquisition systems, this ringing is not consistent at each step function in the signal. It can vary in amplitude or not occur at all. This also depends on timing of the transient relative to the data samples.

Causes

This ringing is a product of a sharp filter on a signal with transients, which frequency content falls in filter's stopband.

Anti-aliasing filters, pre-decimation filters, shaping filters, etc. are needed and often used in measurement devices, and introduce overshoot depending on their design: **type/shape, sharpness, and phase response** are important contributing parameters.

"Sharp filters release previously stored stopband energy more abruptly in the passband, causing overshoot around each transient or step in the signal"

"While things seem fine in the frequency domain, in the time domain this effect is not desirable, especially from an amplitude accuracy point of view"

For example, a sharp filter will release previously stored stopband energy more abruptly in the passband, giving overshoot around each transient in the signal. Note that stored energy must be released anyway, and most of the time it isn't noticed because it's shadowed by passband signal.

While things seem fine in the frequency domain, in the time domain this effect is not desirable, especially from an amplitude accuracy point of view. The measured and observed amplitude after sampling and filtering can be very different from the actual signal itself.

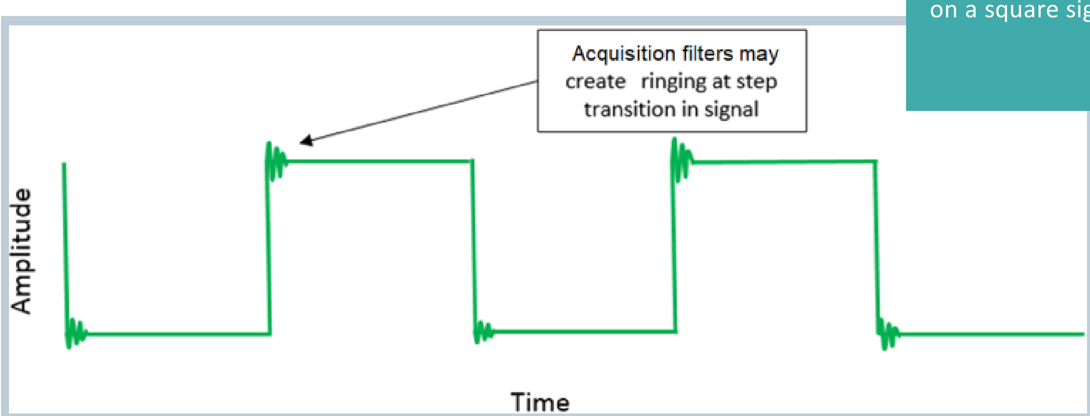


Figure 8: Overshoot on a square signal

Filter selection

Filters are characterized by their passband, transition and stopband characteristics. Figure 8a gives an overview of typical filter parameters.

Keeping in mind that overshoot is consequently only possible with filters of order 2 or higher (where energy can effectively be stored), some guidelines for selection of a suitable filter to minimize overshoot at the output are:

Phase as linear as possible in bandwidth: That is, group delay as constant as possible. This ensures minimal shape distortion and helps distributing overshoot, if any, before and after transients. It's necessary but not sufficient for an overshoot free signal.

"Analog and digital IIR filters have non-linear phase responses because they are intrinsically causal, i.e. they can't foresee a transient before it actually happens at their inputs"

Analog and digital IIR filters have non-linear phase responses because they are intrinsically causal, i.e. they can't foresee a transient before it actually happens at their inputs. A non-linear phase response (group delay not constant) means that the signal won't propagate through the filter with the same delay for all its frequency components.

In causal filters, the sharper their transition band, the more non-linear their phase, resulting in a more distorted signal time shape: overshoot energy can't be spread, and is released only after the transient, therefore magnifying its effects. Note that most phase dispersion generally occurs in the vicinity of the cut-off frequency (f_{co}).

Smoothness (low sharpness) transition band filters. This implies smoother energy dosing for a given filter order, thus minimizing overshoot. The compromise is that the stopband is larger than a sharp filter, and high attenuation figures (i.e. alias attenuation in anti-aliasing filters) are reached farther above the f_{co} of the filter (that should normally be set to our BW of interest). In order to have high alias attenuation using a smooth filter, we should use oversampling and higher sample rates, so that the Nyquist frequency is pushed safely above f_{co}. At Nyquist, the smooth filter will then have enough attenuation to remove aliasing.

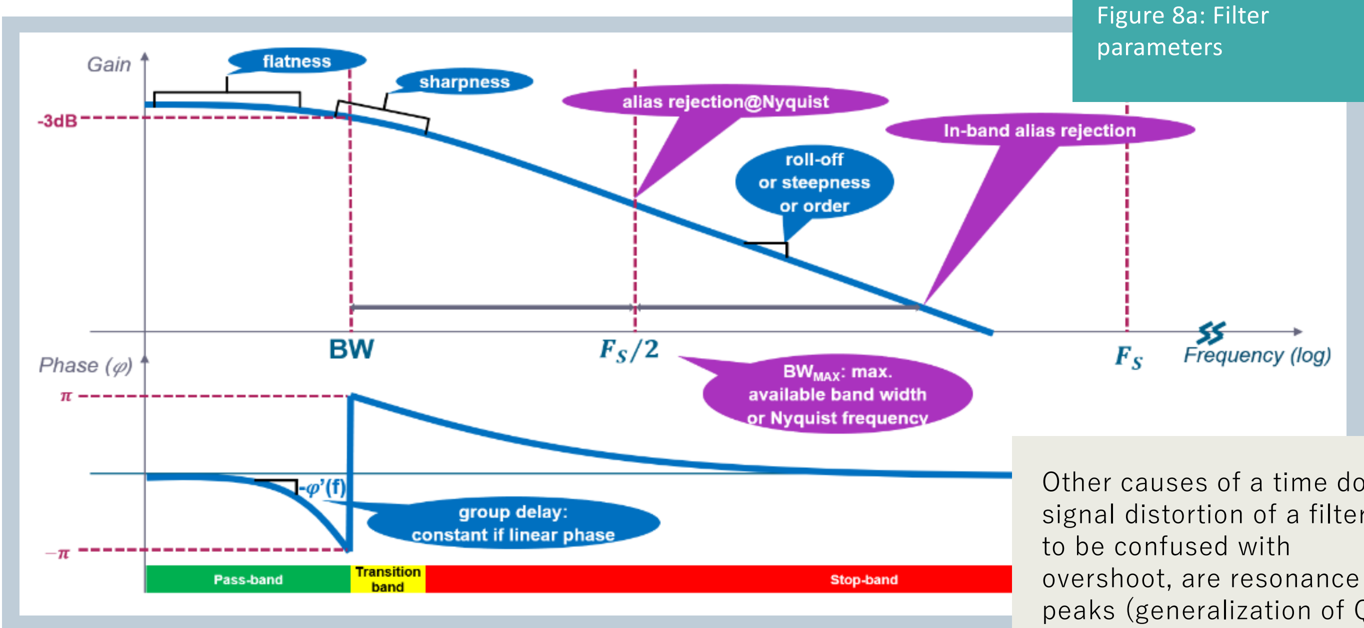


Figure 8a: Filter parameters

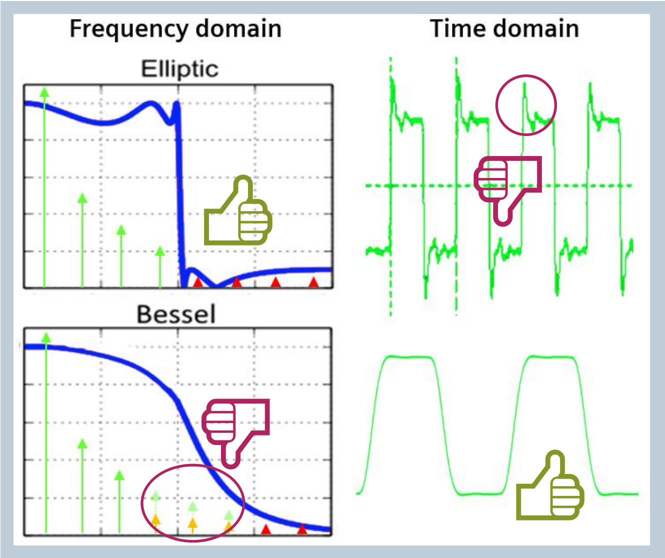
Other causes of a time domain signal distortion of a filter, not to be confused with overshoot, are resonance peaks (generalization of Q factor), and potential passband ripples, like in Chebyshev design.

Choosing a proper filter type: Figure 9 shows a comparison between 2 filter types. In general, increasing filter order reduces phase response linearity. But there are exceptions where phase linearity improves with filter order, e.g. maximally flat delay with Chebyshev stopband filter, Bessel filter, linear phase active filter, transitional filters (Gaussian), etc.

Note that there is always a compromise between sharpness and overshoot: the sharper the filter, the more overshoot. This is not the case for the order of a filter, that gives its steepness or roll-off behavior. For example, a 2nd order Butterworth overshoots, and a 12th order Bessel does not.

"Sharpness is always a compromise; steepness/ roll-off is not"

Figure 9: The Elliptic filter example shows high energy release from stop and transition bands, while triggering maximum phase scattering after the discontinuity from transition band. On the contrary, Bessel filter has a wide transition band, implying smooth energy release. Furthermore, the near-linear phase response tends to distribute energy release before and after the discontinuity.



Time domain or frequency domain filters

There is a fundamental difference in filter selection based on whether analysis is done in time or in frequency domain:

- Time domain: use oversampling filters. Oversampling filters have a f_{co} set to the BW of interest, way below Nyquist frequency, so we can reach sufficient alias suppression at Nyquist.
- Frequency domain: a sharp filter with f_{co} as close as possible to the Nyquist frequency can be used. Overshoot can be large but is not important: it is a time domain effect and does not influence amplitude at each spectral line in the frequency domain.

Filter Type	Frequency domain	Time domain
Elliptic	-Sharp: BW of interest can be close to $F_S/2$, and still be good in alias suppression	-Large overshoot -Phase scattering (asymmetry)
Bessel	-Smooth: BW of interest needs to be a factor below $F_S/2$, for good alias suppression	-Almost overshoot-free -Almost linear-phase (keeps symmetry)

Measuring overshoot

The overshoot can be quantified in three ways. These quantities are illustrated in Figure 10:

- **Frequency** – The frequency of the ringing overshoot
- **Amplitude** – How much the signal under- and overshoots the original signal
- **Duration** – Ringing decays with time

In the following examples, a square signal is used to illustrate the overshoot. Note that sharp filters cause overshoot for signals with fast transients or steps.

Examples of these types of signals include forces from driving over a pothole, the sound of an explosion, or the vibration from a golf club while hitting a ball.

Frequency

The frequency of the ringing is related to the **frequency cut-off of the filter**. Normally, and considering a generic filter type, we encounter next phenomena in the vicinity of f_{co} (see Figure 10):

> *"Sharp filters cause overshoot for signals with fast transients or steps"*

- Maximum phase dispersion: maximum delay difference between frequency components. Frequencies near the f_{co} get delayed and superimposed with less delayed lower frequency components. This contributes to an asymmetric, damped, f_{co} ringing overshoot
- The highest frequency component before the f_{co} gets still thru just before the transition band. So, for the same reason, it will be at same time the first frequency where the filter will overshoot: the stored stopband energy will encounter a first opportunity to be released exactly at this frequency

Theoretical background

The inverse Fourier Transform of the frequency domain filter shape, is the impulse response of the filter (a filter can be considered itself a signal (or a function), as a property of LTI systems). The filter has a wide frequency domain characteristic, resulting in an impulse response function of short duration in the time domain (time and frequency domain are reciprocal). For a given filter design, overshoot energy is less when the duration of impulse response of the filter is shorter. In other words, the filter buffer is smaller and so is the energy for an overshoot.

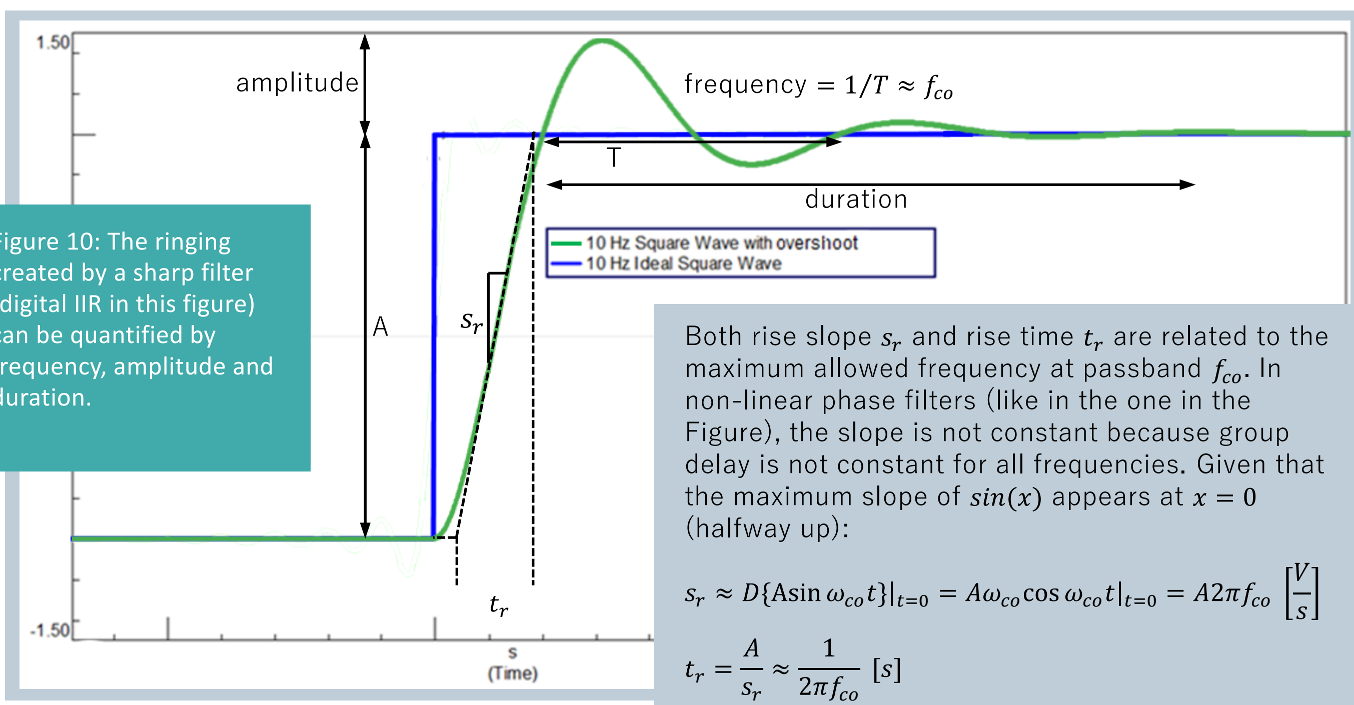


Figure 10: The ringing created by a sharp filter (digital IIR in this figure) can be quantified by frequency, amplitude and duration.

Both rise slope s_r and rise time t_r are related to the maximum allowed frequency at passband f_{co}. In non-linear phase filters (like in the one in the Figure), the slope is not constant because group delay is not constant for all frequencies. Given that the maximum slope of $sin(x)$ appears at $x = 0$ (halfway up):

$$s_r \approx D\{A\sin\omega_{co}t\}|_{t=0} = A\omega_{co}\cos\omega_{co}t|_{t=0} = A2\pi f_{co}\left[\frac{V}{s}\right]$$

$$t_r = \frac{A}{s_r} \approx \frac{1}{2\pi f_{co}}\,[s]$$

Duration and amplitude

The duration of the overshoot is expressed by the damping time of the ringing and is determined by its frequency and initial (highest) amplitude.

Both amplitude and damping time **depend on the type of filter** used, i.e. shape, sharpness and phase response.

The graph in the lower half of Figure 11 shows that, with the same filter type, ringing lasts longer if more harmonics are filtered. In addition, the transient slope in the square is not as pronounced counting only on the lower harmonic frequencies.

Applying filters to remove frequency content of a sine signal creates no overshoot, because a sine signal has only one frequency component: itself. In Figure 12, a sine signal of the same frequency of the square signal in Figure 11 is not affected by differing filter settings. There is no frequency content being stopped by a filter, so there is no overshoot:

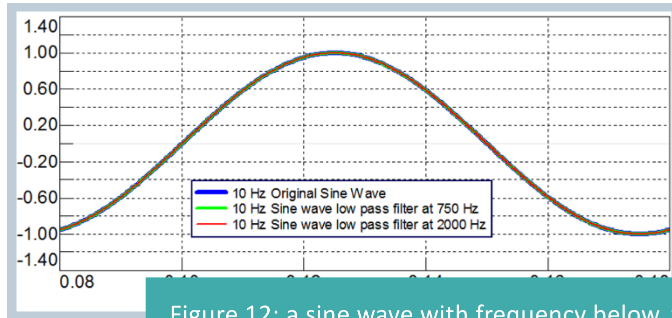


Figure 12: a sine wave with frequency below f_{co} will not experience any distortion. Overshoot only occurs on signals with sudden changes, i.e. containing high frequency content in the filter stopband.

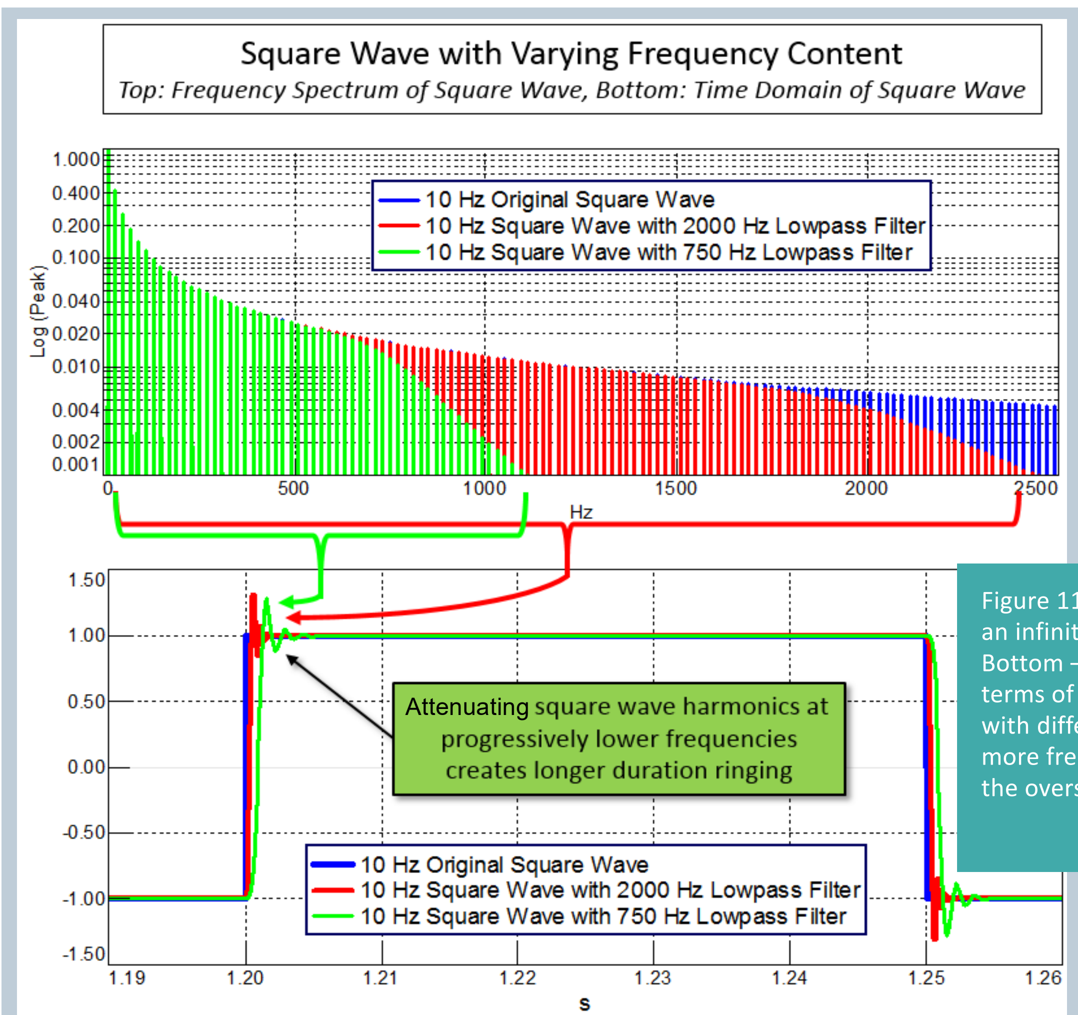


Figure 11: Top – The square signal has an infinite number of harmonics. Bottom – Using same filter type (IIR) in terms of sharpness and steepness, but with different cut-off frequencies. As more frequency harmonics are filtered, the overshoot has a longer duration.

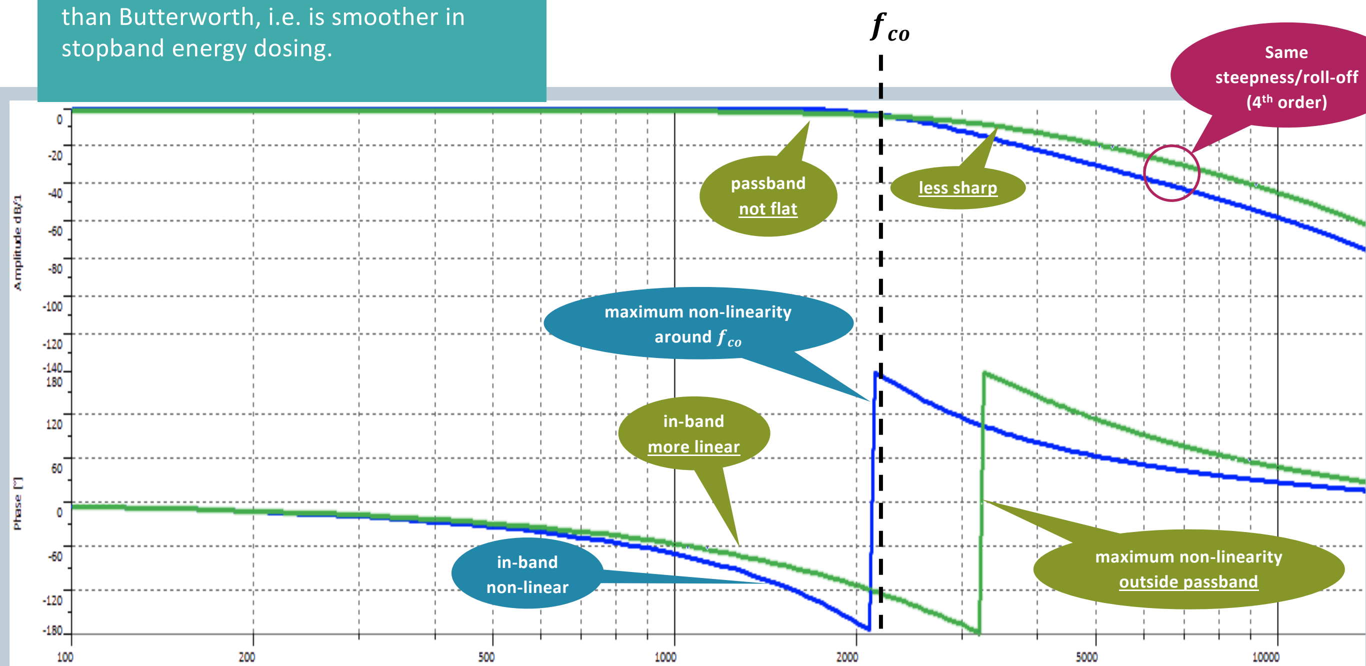
We have seen that filters can overshoot only if the signal falls into their stopband. So next to any filters used in a data acquisition system, the presence of overshoot also depends on the signal itself.

As a rule of thumb, **the sharper the filter, the larger the amplitude of the ringing**.

In Figure 13, the shape of two different filters is overlaid: a Bessel and a Butterworth filter. The Bessel filter is less sharp compared to the Butterworth filter, but it has almost linear phase at passband (its phase even improves with higher filter orders).

In Figure 14, the amplitude of the overshoot of a Bessel filtered square signal is less than the amplitude overshoot of the same square signal using a Butterworth filter. In fact, the Butterworth filter is designed to have controlled (fixed) overshoot.

Figure 13: Bessel (green) and Butterworth (blue), both 4th order, have same 3dB f_{co} (where they cross), but given same filter order, the Bessel asymptotic roll off is achieved later than Butterworth, that is, Bessel is less sharp than Butterworth, i.e. is smoother in stopband energy dosing.



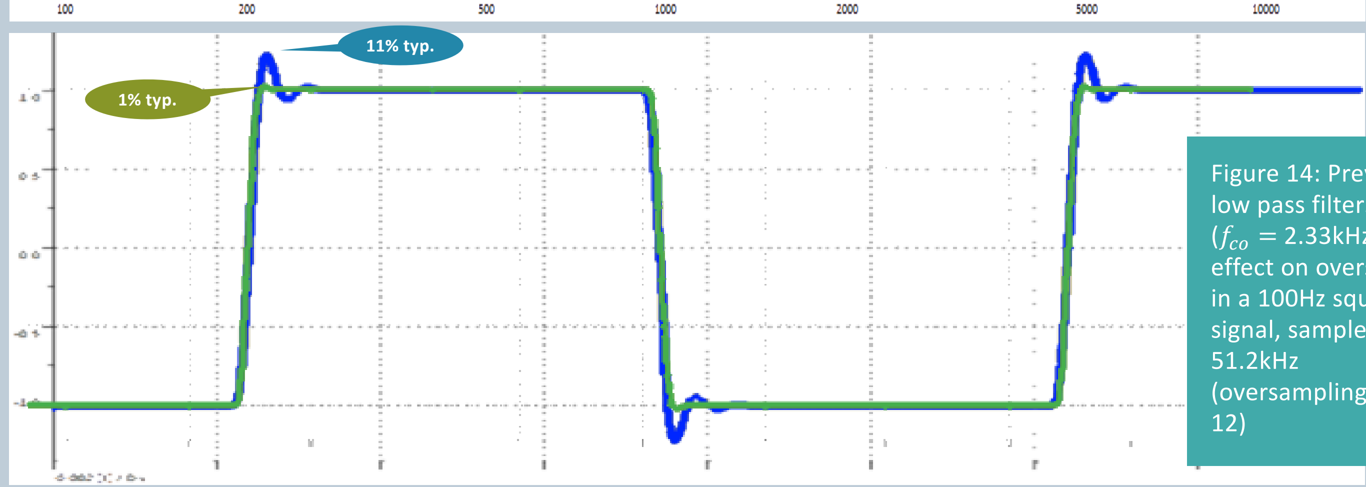


Figure 14: Previous low pass filters ($f_{co} = 2.33$kHz) effect on overshoot in a 100Hz square signal, sampled at 51.2kHz (oversampling factor 12)

Pre-ringing

When viewing signals with overshoot, sometimes this ringing can be observed before the transient or step in the signal, other times the ringing might only be seen after the transient as shown in Figure 15.

When using a symmetric impulse response FIR filter, its linear phase **allows overshoot energy to spread before the transient**. Overshoot amplitude will then be lower and pre-ringing is created. Pre-ringing is per definition non-causal, so it's not present in signals filtered by analog or digital IIR filters. In other words, exact linear phase is not possible with these filters.

"Pre-ringing is - by definition - non-causal, so it's not present in signals filtered by analog or digital IIR filters. In other words, exact linear phase is not possible with these filters"

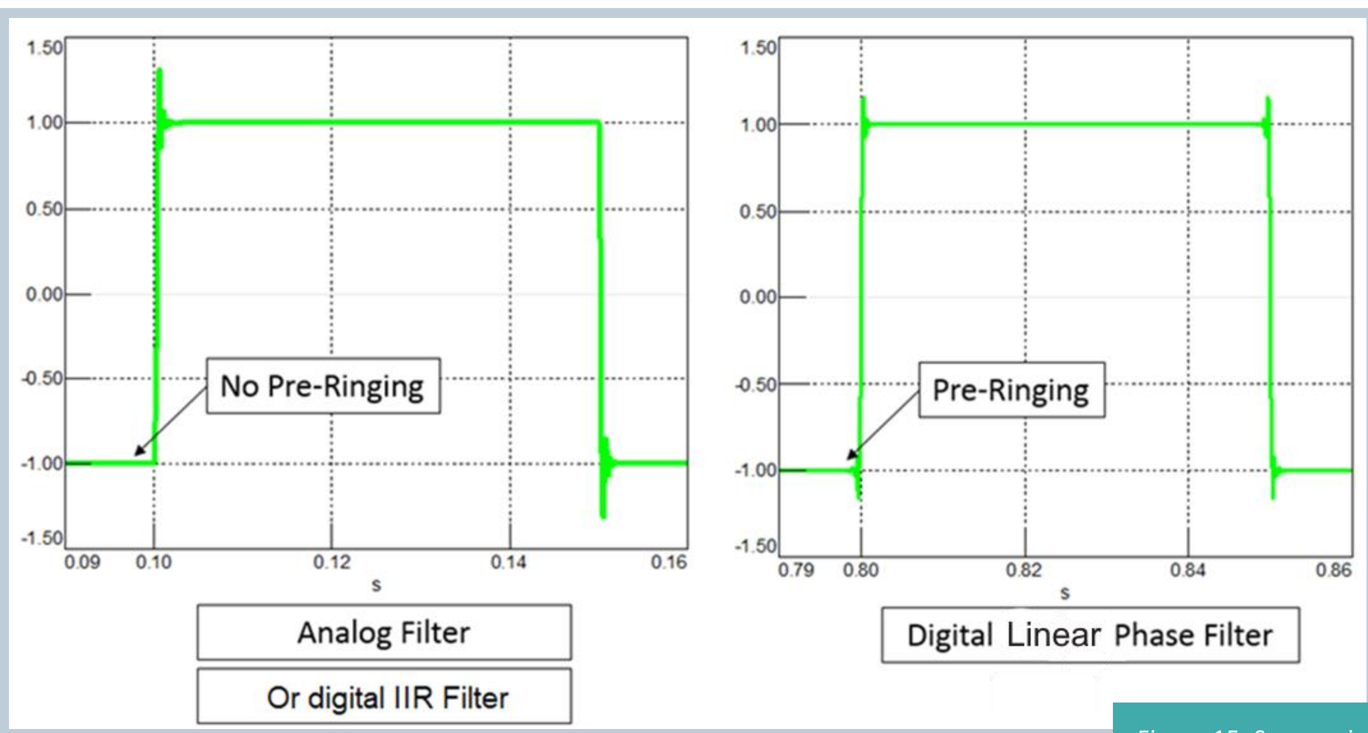


Figure 15: Square signal on left exhibits no pre-ringing, square signal on right has pre-ringing.

Causes in time domain

This document analyses signals in the frequency domain, which is normally more intuitive to understand. Filtering is easily understood by multiplying the Fourier transform of the input signal with the transfer function of the filter.

In time domain, a multiplication is mathematically equivalent to convolution. We use this approach to understand the cause of overshoot in time domain. The convolution operation calculates for each x value, the area of the multiplication of 2 functions, one of them flipped in the x axis (a 'flip & slide operation', see Figure 15a).

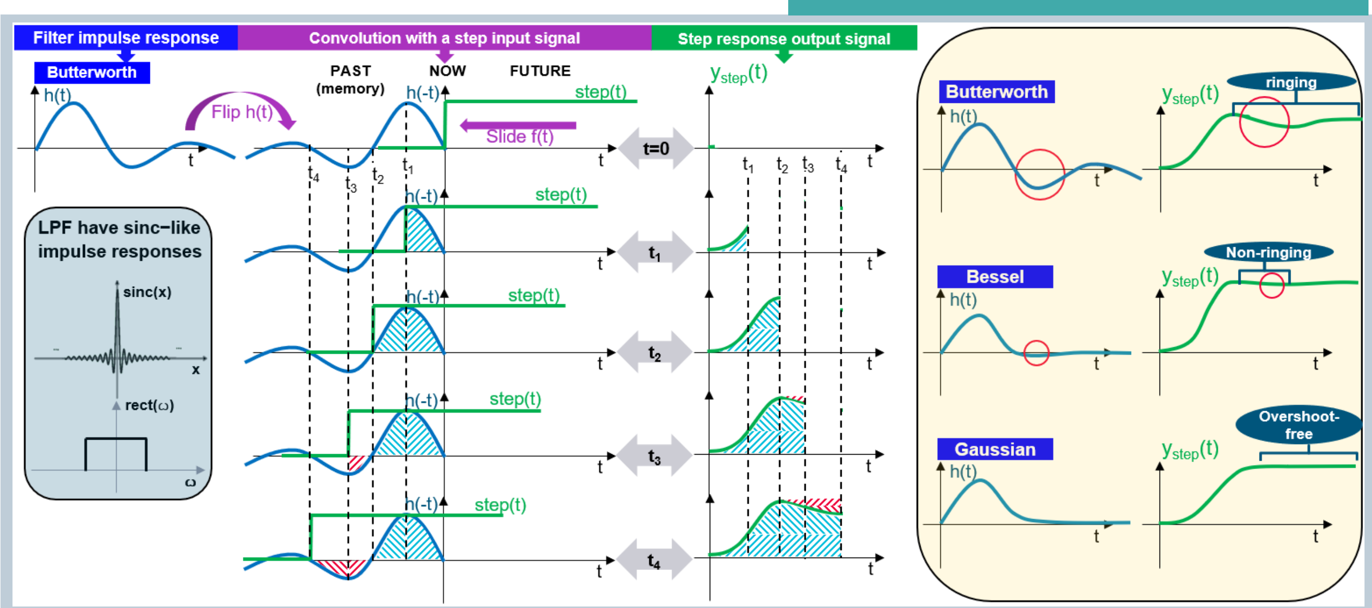
In the physical world, those 2 functions are our signals (input signal and filter impulse response "signal"). The same idea holds for the digital version of the convolution, that would apply for a digital filter. In case of a FIR filter, each output sample in time is calculated by multiply-and-accumulation (MAC) operations, that are carried out for all filter impulse response samples, otherwise called coefficients.

The Fourier transform of a $sinc(x)$ function is a rectangle function. LPFs have normally rectangle-like shaped transfer functions, so they have sinc-like impulse responses. Remember that Fourier transform enables us to go from one domain to the other. The $sinc(x)$ function is defined as $\sin x/x$, so for example, the frequency of the overshoot will be defined by $\sin x$, and its decay behavior will follow the multiplying term of $\sin x$. In this case, for a perfect $sinc(x)$, the overshoot will decay with $1/x$.

Thus seen from a time domain approach, if the input signal contains slope(s) allowing transient(s) shorter than the duration of the filter's impulse response lobes (i.e. has frequency content that will fall in the filter stopband), the cause of overshoot is the presence of negative lobes in the (sinc-like) impulse response of filters. In other words, those filters having negative lobes in their impulse response will overshoot at their output in these conditions.

"If the input signal contains transient(s), those filters having negative lobes in their impulse response will overshoot at their output"

Figure 15a: For simplicity, we can take the analog example of a filter. The analog filter will process the signal like a continuous, mathematical convolution. For each output moment in time, the result is the area of the multiplication of both signals along all past and present times (in practical terms, it is enough to do it only along the filter impulse response, given that it is 0 elsewhere).



Simcenter SCADAS and overshoot

SCADAS Lab and Mobile

Overshoot can be greatly reduced in Simcenter Testlab by applying a low pass Bessel filter to the incoming signal. In the Channels worksheet, right-click the channel header and select "Show columns..." (Figure 16) to activate the low pass filter settings.

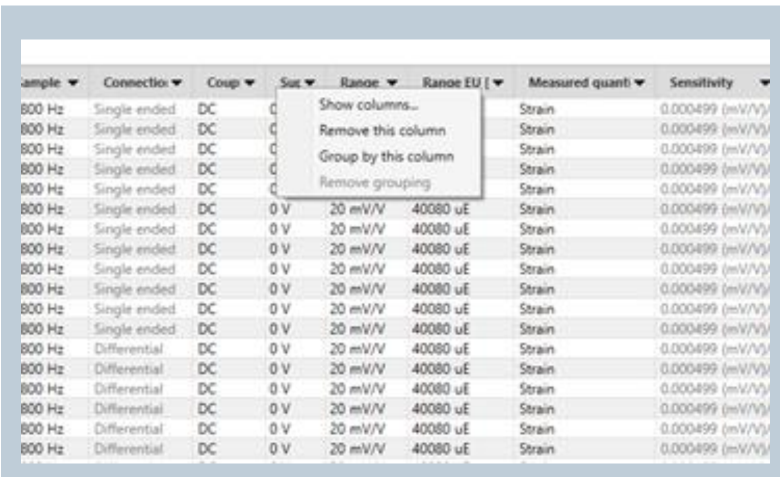


Figure 16: To add the low pass filter setting fields to channel setup, right-click the channel header and select "Show columns...".

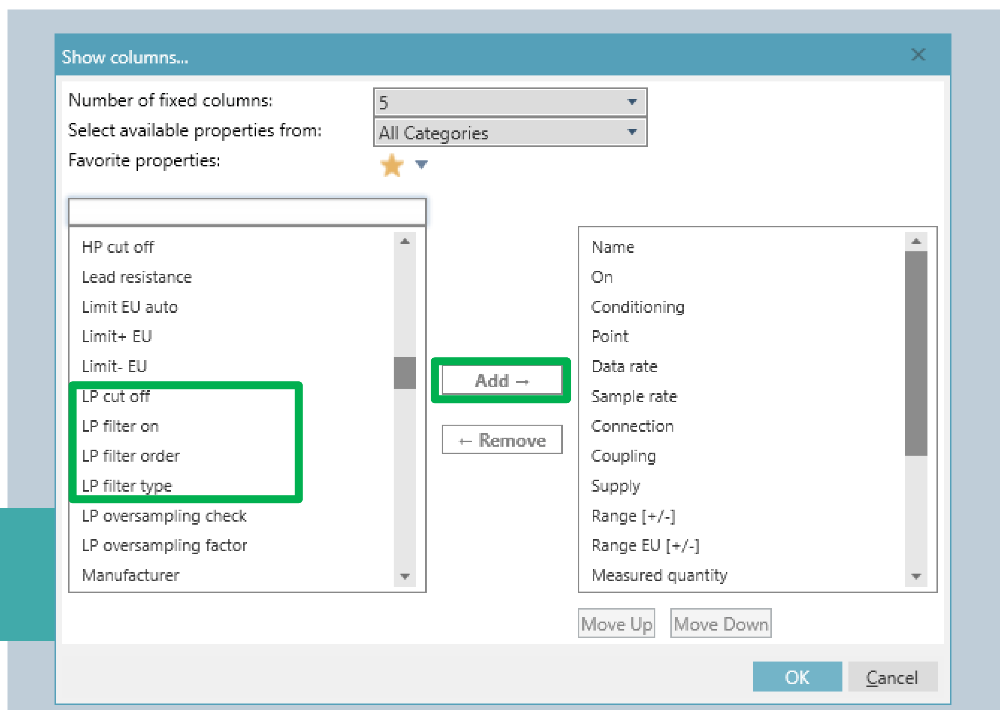


Figure 17: "Show columns..." menu.

Four new columns are added to the channel information in the Channel Setup worksheet as shown in Figure 18.

LP cut off	LP filter on	LP filter order	LP filter type
1000 Hz	☑	6	Butterworth
1200 Hz	☑	6	Butterworth
100 Hz	☑	6	Bessel
500 Hz	☑	6	Bessel

Figure 18: Lowpass (LP) filter selections in Channel Setup.

The following parameters can be set:

- **LP filter type** – Either Bessel or Butterworth can be selected. Bessel has almost linear phase
- **LP cut off** –Bessel filters are less sharp. A good “rule of thumb” is to set the lowpass filter cutoff five times less than the bandwidth of the measurement, for a given filter order
- **LP filter on** – Turns the filter ON or OFF
- **LP filter order** – The lower the order, the more gradual the filter roll off

A 2nd order Bessel overshoot will largely reduce overshoot of a measured square wave signal, as shown in Figure 19.

Not every SCADAS data acquisition module has extra lowpass Bessel and Butterworth filters. Check the product information sheet or your local support if you have questions about filter characteristics of your SCADAS hardware.

These low pass filter settings are available in the Simcenter SCADAS Lab and Mobile modules. The signal is oversampled at high frequency and filtered with steep filters in the decimation stages. This way, the frequency content of the signal is still completely valid near Nyquist frequency, which means that applying inline or post processing filters, overshoot of the time domain signal can be largely reduced.

“ These low pass filter settings are available in the Simcenter SCADAS Lab and Mobile”

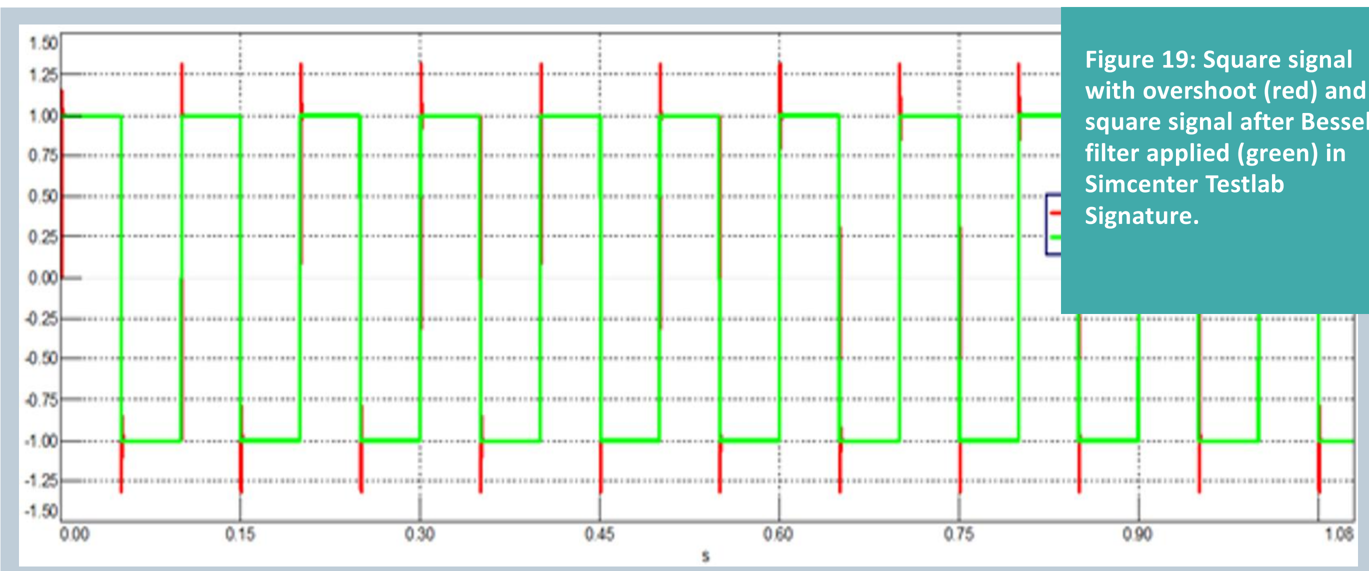


Figure 19: Square signal with overshoot (red) and square signal after Bessel filter applied (green) in Simcenter Testlab Signature.

SCADAS RS

A (digital) sharp anti-aliasing filter is typically used on top of any other filter chain when measuring a signal, especially after intermediate decimation stages. This sharp anti-aliasing filter influences the behavior of the low-pass filters that may follow: the closer the low-pass f_{co} the anti-aliasing filter, the more distortion the former will experience from the latter.

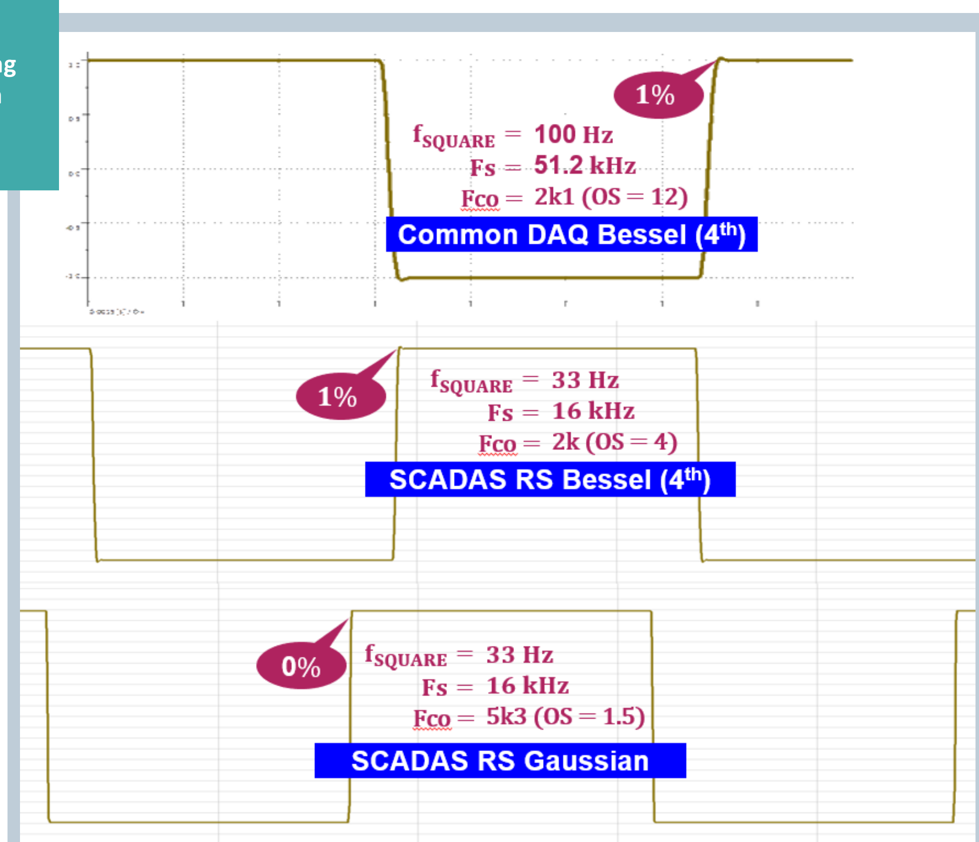
A dedicated approach for time data acquisition and recording is implemented in Simcenter SCADAS RS. This hardware includes high performance dedicated extra shaping and anti-aliasing filters, including Bessel, Butterworth, Gaussian and Steep FIR types.

As a result, using traditional filter techniques for frequency domain, measurements show a 1% overshoot with a Bessel filter using an oversampling factor of 12 (being the ratio between the Nyquist frequency and f_{co} of the filter).

With SCADAS RS, an oversampling of 4 is enough to provide 1% overshoot using the same Bessel filter. Even more, overshoot free (0%) is guaranteed when using a Gaussian filter at 1/3 of the sampling rate (oversampling factor of 1.5).

In other words, outstanding overshoot values are achieved already for low oversampling factors.

Figure 20: Overshoot comparison. SCADAS RS uses an improved filtering strategy for time domain analysis



Conclusion

Using the appropriate sampling rate and filters, overshoot on measured time signals can largely be reduced:

- **Careful selection of low-pass filter type** can reduce and even eliminate overshoot
- Overshoot is best observed when measuring **transient signals** that contain a step or fast transient. It occurs when an arbitrarily sharp filter, used in the conversion from analog to digital domain, stops higher frequency content of a signal in its stop band
- **Filter design** including filter shape, sharpness, phase response and passband width are the contributing factors to the presence of overshoot at the output of the filter
- Overshoot effects are visible in **the time domain**, not frequency domain

Using a proper filter strategy with any analog to digital converter reduces or even eliminates overshoot. The effect of the filter should not be confused with the type of analog to digital converter.

Sometimes there is confusion about a Successive Approximation Register (SAR) versus Sigma-Delta analog to digital converters and overshoot. Many Sigma-Delta converters have sharp decimation anti-aliasing filters which prevent alias errors. But these sharp filters are not inherent to Sigma-Delta converters; any type of filter can be used.

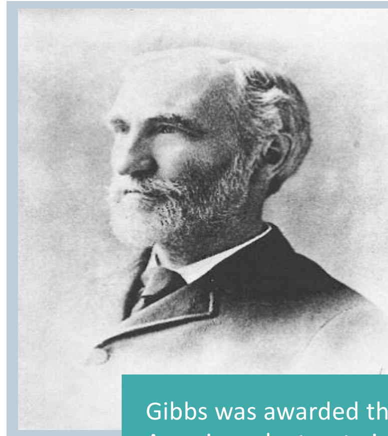
Data acquisition systems targeting frequency domain applications are optimized for a maximum frequency range coverage including amplitude accuracy in frequency domain up to high bandwidths.

When studying phenomena in time domain, data acquisition systems need to be optimized for time domain filtering, avoiding inaccuracies in time domain amplitude because of filter effects. This is especially important in areas such as durability tests, road load data acquisition, image processing, medical EKG's, etc.

"Studying phenomena in time domain is especially important in areas such as durability tests, road load data acquisition, image processing, medical ECG's, etc.."

Addendum: The Gibbs Effect

Josiah Willard Gibbs (1839-1903) was an American scientist from Yale University. In 1899, he published observations about the Fourier series approximation of a **function with discontinuities**, and how it presents undershoot and overshoot. That observation later became known as the “Gibbs Phenomenon” or “Gibbs Effect”.



Gibbs was awarded the first American doctorate in Engineering. He specialized in mathematical physics, and his work affected diverse fields like chemical thermodynamics and physical optics.

A **Fourier series approximation** of a periodic function:

- is an **approximation** of the function by a summation of a (finite set) of sines with different phases
- **does not converge** at discontinuities when present in the function

The Gibbs Phenomenon is an observation of the Fourier series’ asymptotic behavior. It highlights its limitations when approximating periodic functions with discontinuities using a finite set of Fourier terms.

In other words, even when representing a signal with a large (N) number of terms $N \rightarrow \infty$ (approaching, but not reaching infinity) of the Fourier series **a ringing overshoot at the original function’s discontinuities will always be present in the result**. Only when $N = \infty$, the sum converges to the original function:

- Ringing amplitude is independent of N; it doesn’t fade when $N \rightarrow \infty$
- Ringing amplitude is about 9% of the original amplitude discontinuity (“step”)
- Ringing duration is inversely proportional to N, so to its energy

Later it was discovered that this had already been described by an English mathematician, Henry Wilbraham, in 1848. Despite this revelation, the phenomenon continued to be named after Gibbs.

Questions? Call us!

”The Gibbs Phenomenon is an observation of the Fourier series’ asymptotic behavior and highlights its limitations when approximating periodic functions with discontinuities using a finite set of Fourier terms.”

Siemens Digital Industries Software

Headquarters
Granite Park One
5800 Granite Parkway
Suite 600
Plano, TX 75024
USA
+1 972 987 3000

Americas
Granite Park One
5800 Granite Parkway
Suite 600
Plano, TX 75024
USA
+1 314 264 8499

Europe
Stephenson House
Sir William Siemens Square
Frimley, Camberley
Surrey, GU16 8QD
+44 (0) 1276 413200

Asia-Pacific
Unit 901-902, 9/F
Tower B, Manulife Financial Centre
223-231 Wai Yip Street, Kwun Tong
Kowloon, Hong Kong
+852 2230 3333

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About the author

Aleix Riera – Product Manager Hardware

Further references (digital signal processing)

- Digital Signal Processing: Sampling Rates, Bandwidth, Spectral Lines, and more...
- Gain, Range, Quantization
- Aliasing
- Overloads
- Averaging Types: What's the difference?
- Spectrum versus Autopower
- Autopower Function...Demystified!
- Power Spectral Density
- Windows and Leakage
- Window Types
- Window correction factors
- RMS Calculations

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