Digital oscilloscopes: When things go wrong

Arthur Pini - October 27, 2016

DSOs (digital oscilloscopes) offer a great many advantages over their analog equivalents but as they say, "There’s no such thing as a free lunch." Digital scopes sample, digitize, and store waveforms and let you for measure, analyze, and archive signals. But, that sampling process brings a few issues along as "baggage."

Aliasing (this page), synchronous sampling (page 2), and interpolator (page 3) errors can cause you to misinterpret the measurement results unless you understand these issues. As you might expect, most DSO manufacturers don’t spend a lot of time talking about negative issues so learning about them is a discovery experience. Let's examine these problems and discuss how to detect and, hopefully, work around them.

Aliasing
The sampling theorem, which rules over all digital instruments and systems, requires that a signal be sampled at a rate that is greater than twice the maximum frequency contained in the signal. If the signal is properly sampled, then an oscilloscope can reconstruct it from the samples with no loss of information. Under sampling, or sampling at less than twice the highest frequency component, results in a recovered signal with lower component frequencies than the original signal, this unwanted signal is called an alias. Half of the sample rate is called the Nyquist Frequency, which marks the highest frequency that can be digitized at that sample rate.

Figure 1 provides an example of aliasing. The waveform in the upper left grid is a 400 MHz sine wave sampled at 1 (GSamples/s. There are 2½ samples per cycle as seen in the horizontally expanded zoom trace shown in the second grid from the top on the left side. Note that this is the raw sampled data with no interpolation. In the trace third from the top on the left side sin(x)/x interpolation has been applied. This is what most DSO's will display as this is their default display interpolator.

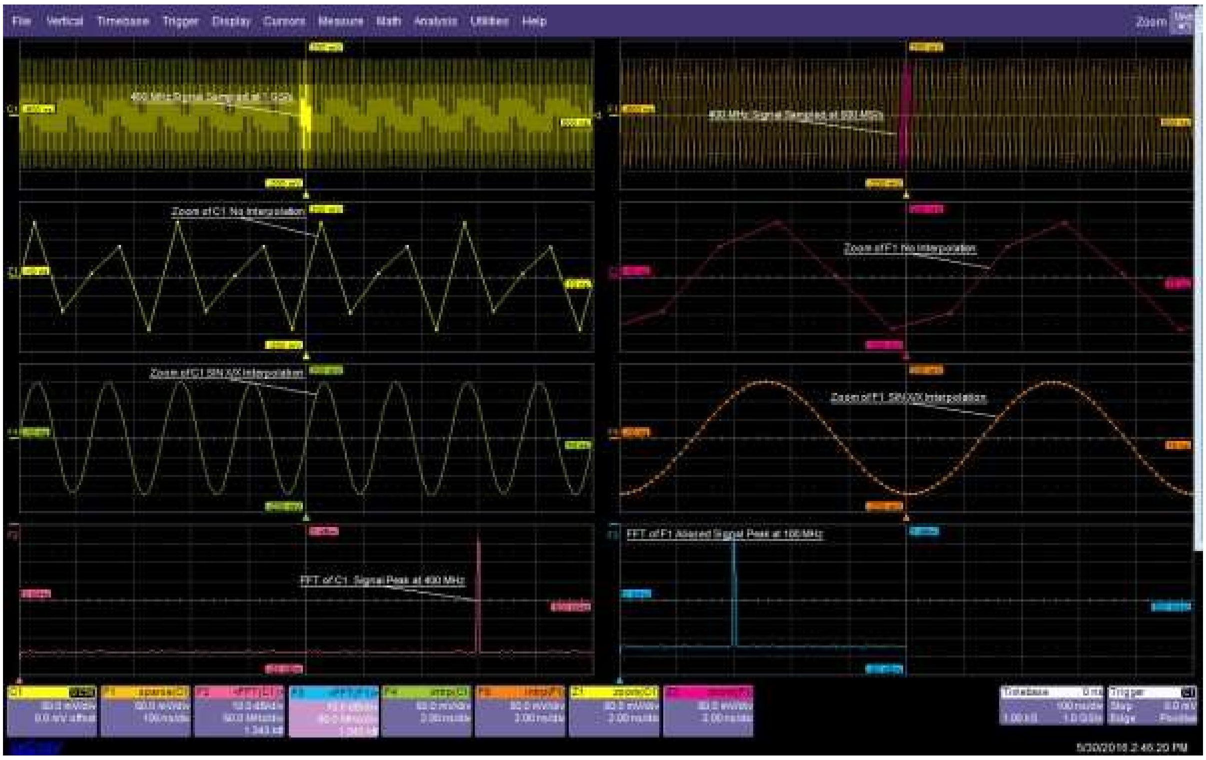

Figure 1. When a 400 MHz signal is undersampled, it loses signal fidelity and aliasing will occur.

The bottom trace on the left side is the FFT (Fast Fourier Transform) of the input signal showing the spectral or frequency domain view of the signal. It shows a spectral peak at 400 MHz as expected for this signal.

The waveform in the top right grid is the same 400 MHz sine sampled at 500 Msamples/s. The sample rate is below twice the signal frequency and the signal is aliased. The second grid from the top on the right side is the zoom view of the aliased trace. Note that the signal frequency is lower. In this case it is 100 MHz. The next lower trace is the aliased signal with interpolation applied. The FFT of the aliased trace has a frequency peak at 100 MHz. Note that the FFT trace is truncated at 250 MHz, the Nyquist frequency for the 500 MS/s sample rate.

Because Fig. 1 is an unanimated graphic, the aliased waveform appears to have a stable trigger, but but it doesn't. The trigger level is set for zero volts, and a positive slope and the non-aliased waveform shows the correct trigger level. The aliased waveform only has every other sample point of the non-aliased waveform and will hop between samples adjacent to the trigger point. This results in a trace with horizontal "jitter."

Probably the best way to investigate aliasing is to view it in the frequency domain. Sampling is similar to an analog mixing process. It essentially multiplies the sampled waveform by the sample clock, which is usually a very narrow pulse. The sampling clock is rich in harmonics. The sampling/mixing process produces frequency components that include the original baseband signal being sampled, the sample clock and all its harmonics, and lower and upper sideband images of the sampled signal about each sample clock harmonic as shown in the upper view in **Figure 2**.

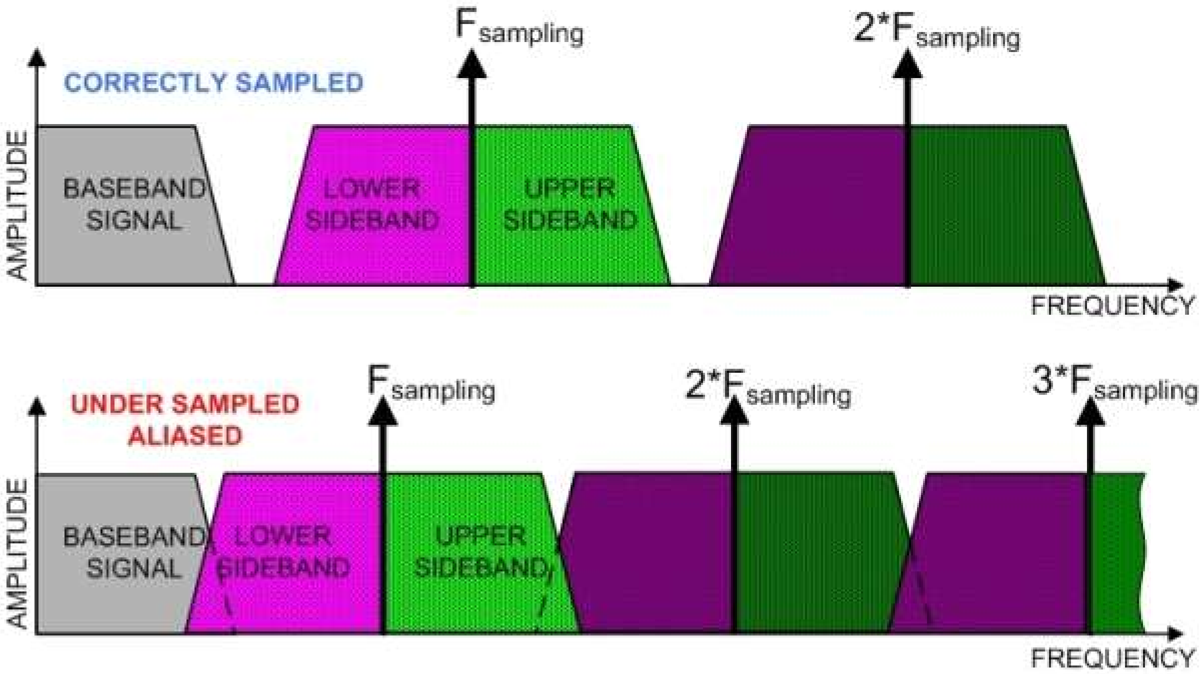

Figure 2. The sampling process viewed in the frequency domain shows both correct and aliased sampling.

The baseband signal component approximates the frequency response of a typical DSO. The bandwidth is generally specified at the "knee" of the response with a rapidly attenuated "roll off" response above the bandwidth limit. Because there can be spectral components above the oscilloscope's bandwidth, most manufacturers sample at 2.5 times or greater than the bandwidth to prevent aliased components from this region.

Lowering the sample rate moves the sampling frequency component of the spectrum and all its harmonics to the left in the frequency domain display. Aliasing occurs when the lower sideband component about the sample frequency intersects the baseband signal as shown in the lower diagram. Once spectral components overlap, it's no longer possible to filter the resulting waveform to recover the original baseband signal.

Oscilloscope designers generally try to limit aliasing in several ways. First, they chose a maximum sampling frequency that is much greater than the minimum required over sampling. Rates of 3 to 20 times the Nyquist frequency are not uncommon. Next, they lengthen the acquisition memory. This keeps the sampling rate high even when long acquisitions are used. When choosing a DSO, you should know the maximum duration acquisitions you need to make and then choose an instrument with enough memory to support the required sampling rate for the bandwidth your signal requires.

Figure 3 illustrates how acquisition memory length affects the sample rate of an oscilloscope. This chart plots sample rate as a function of the oscilloscope's time/division setting with acquisition memory length as a parameter.

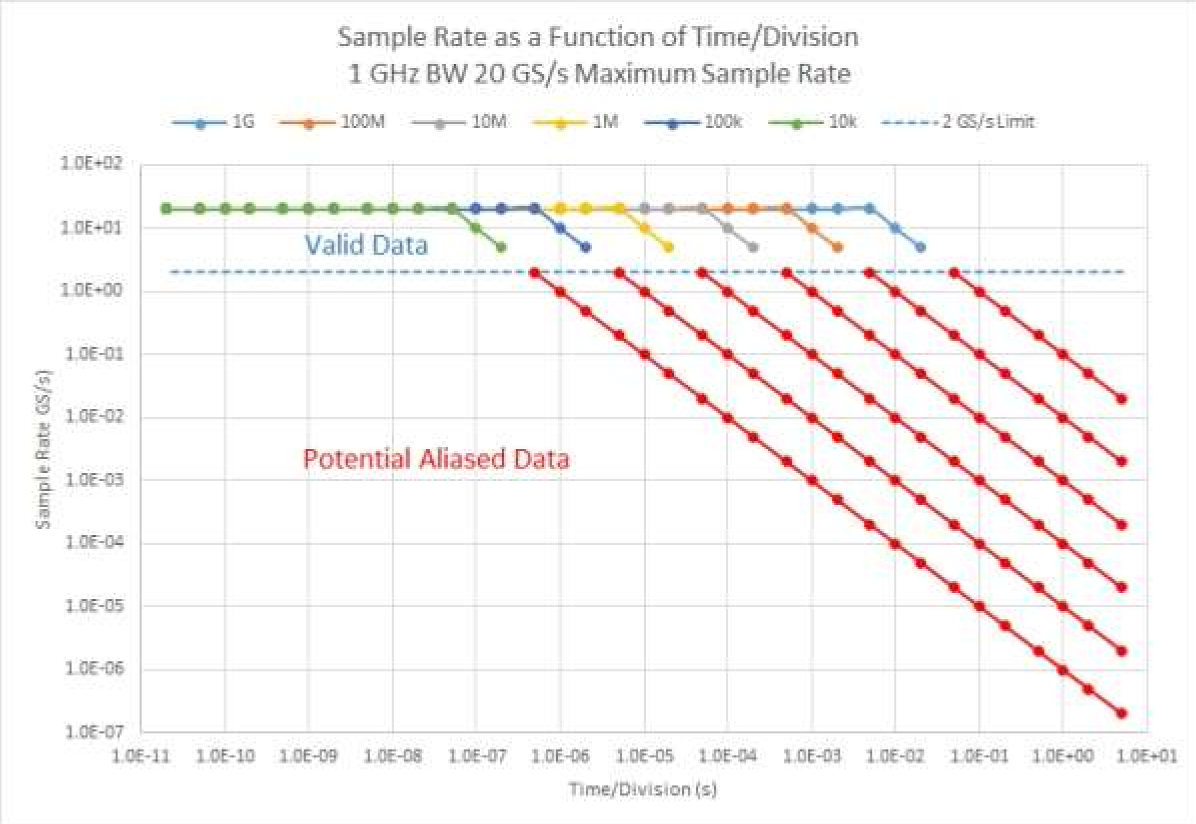

Figure 3. Chart of sample rate vs. time/division setting for a 1 GHz bandwidth scope with a maximum sample rate of 20 Gsamples/s. Note that once the sample rate falls to 2 Gsamples/s or lower, the oscilloscope will alias signals at 1 GHz.

The oscilloscope in this example has a maximum sample rate of 20 Gsamples/s and a bandwidth of 1 GHz. As long as the sample rate is above 2 Gsamples/s, the acquired data is valid. If sampling drops to exactly 2 Gsamples/s or less, the data may be aliased. The sampling rate stays at the maximum 20 Gsamples/s as the time/division setting is increased until all the acquisition memory is engaged. Beyond that point, the sample rate drops. So for an acquisition memory length of 10 ksamples, the sample rate falls to 2 Gsamples/s at 50 ns/division. With a memory length of 100 ksamples, the oscilloscope can reach 5 µs/division before the sampling rate falls to 2 Gsamples/s. As the acquisition memory increases the sample rate remains above the critical 2 Gsamples/s over more time/division settings. So the longer the acquisition memory, the less chance of aliasing.

When it comes to operating a digital oscilloscope, you should start with the fastest sweep speed available—the lowest time per division setting to detect and avoid aliasing. Doing so will result in the highest sampling rate. As you increase the time/division setting, keep your eyes on the waveform. If aliasing occurs, the waveform's frequency will drop suddenly; it's quite dramatic when it happens. If you do run into aliasing, see if you can *increase the depth of the acquisition memory to increase the sampling rate*.

Synchronous sampling

If the sampling clock is synchronous or near synchronous with the signal, then samples are taken at or near the same phase every time. As the signal repeats, the same parts of the signal are sampled. This is most obvious when there are only a few samples per cycle. There's nothing wrong with this as long as the sampling rate is greater than the Nyquist limit, but the oscilloscope display will start to look strange, the signal will appear to be modulated as shown in **Figure 4**.

Figure 4. If the sampling rate is a multiple of the signal frequency, then samples are taken at or near the same phase points in each cycle. This can result in a display that appears modulated.

This 399.9 MHz sine wave is sampled at 1 GS/s, the signal frequency was incremented until false modulation occurred. The upper left trace, C1, is the full acquisition and it appears to be modulated. The "modulation" frequency of approximately 500 kHz (period of 2 µs). The effect is not, however, true amplitude modulation. Trace Z1, second from the top on the left, is a horizontally expanded zoom trace overlaid with a persistence or history display. Linear interpolation is being used for this acquisition. The yellow zoom trace shows individual cycles of the acquired waveform. Note that the sample locations are marked by dots. There are 2½ samples per cycle (five samples over two input signal cycles). The zoom trace location is shown as the highlighted area on the acquired trace.

The persistence trace shows a history of multiple acquisitions and we can see over time the sample points trace a smooth sine wave. There are not enough samples to "paint" the entire wave shape and the existing samples are nearly phase locked so that the same phase points repeat in adjacent cycles. The samples slowly move through the acquired waveform, eventually filling in the display as witnessed in the persistence history. Thus, the acquired waveform is correct but the display appears modulated due to the limited number of samples per cycle and a nearly phase-locked condition between the input signal and the sampling clock.

The third trace from the top on the left side is the FFT of the input signal, centered on 399.9 MHz with a scale factor of 1 MHz/division. Note the absence of 500 kHz modulation sidebands on either side of the carrier. This confirms that the effect is not amplitude modulation.

The display can be improved by increasing the number of samples per cycle. One way to do that is to change the display interpolator. The waveform in Fig. 4 is using a linear interpolator. Sine x/x and linear interpolation are two ways to join the sample points acquired on a waveform. If the signal is a band-limited waveform (that is, if there is little frequency content in the waveform above the Nyquist frequency, half the sample rate) then sine x/x interpolation applied with a high-quality algorithm can properly and accurately reconstruct the waveform shape and amplitude up to a frequency of 0.25 to

0.4 times the sample rate. In our case, the input frequency is 0.399 of the 1 GS/s sampling rate. The upper right trace, C2, in Fig. 4 is the same signal acquired using sin(x)/x interpolation. It shows that the sin(x)/x interpolator improves the display a bit does not correct it.

The second trace from the top on the right is a zoom expansion of the same input signal using the sin(x)/x interpolator. The waveform shows alternate cycles having different peak-to-peak amplitudes. The interpolator is having trouble due to the small number of samples per cycle. This oscilloscope offers a user configurable interpolation function as part of its math functions. The dialog box shows the setup of this interpolation function operating on the trace C1, which was acquired with linear interpolation. The output of the interpolation function is displayed in the third grid from the top on the right side. Below that is a zoom of this trace. Note that a "stronger" interpolator function has eliminated the issue.

The display can also be improved by raising the sampling rate and getting enough samples on the acquired waveforms to fill in the whole waveform. As we saw before, for a given time per division selection the sampling rate can be raised by increasing the size of the acquisition memory.

Again, I would like to point out that this "modulation" effect is not an error. All the oscilloscope's measurement functions will reflect the correct amplitudes because like the persistence display they are based on statistical methods. But is can still cause confusion.

Gibbs Ears: How I learned not to trust interpolators

Sin(x)/x interpolation works great with sinusoidal signals. Unfortunately, many of our signals are digital in nature and look like rectangular pulses. If the signals have "fast" edges, with few sample on an edge, then the sin(x)/x interpolator can cause problems, as shown in **Figure 5**, which compares the response of the oscilloscope’s interpolators to a rectangular pulse with a fast edge. The top trace is the response to a linear interpolator, below that we see a horizontal zoom expansion of the same signal. The third trace from the top is the response of the sin(x)/x interpolator with the zoom expansion of that signal in the bottom trace.

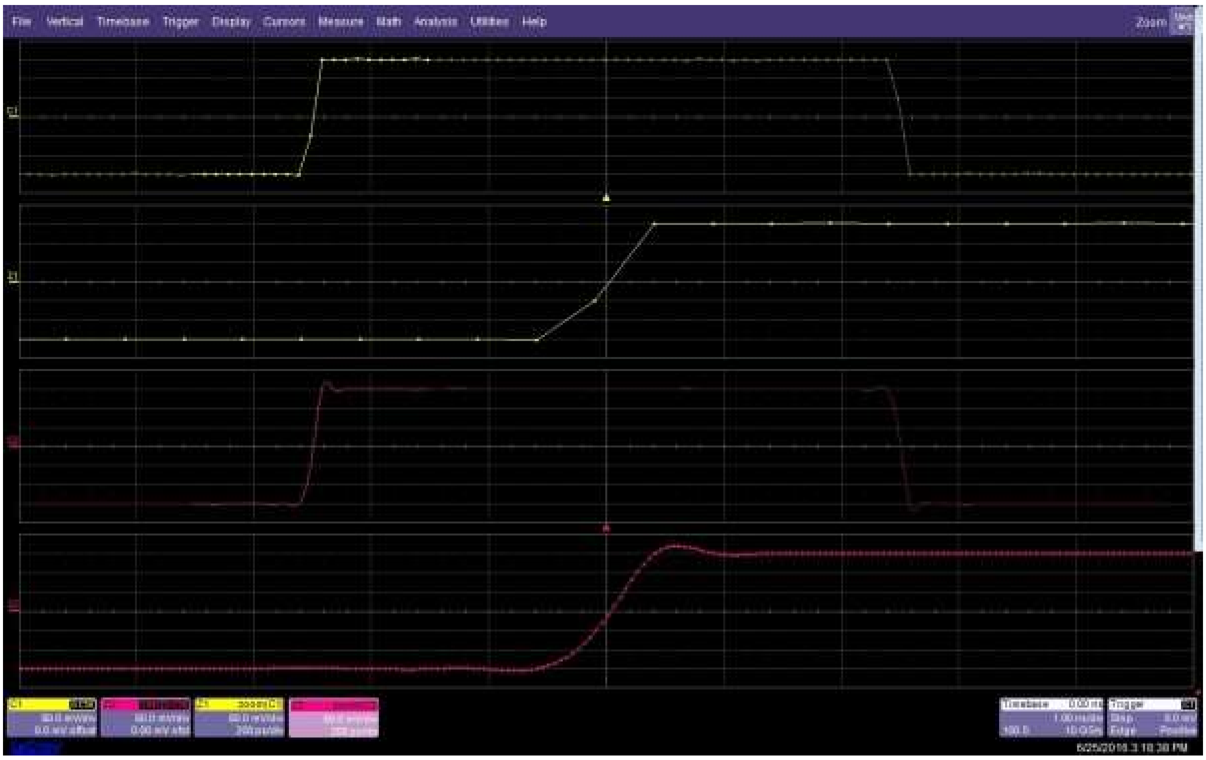

Figure 5. Comparing the response of linear and sin(x)/x interpolators to a fast edge on a

rectangular pulse reveals problems of measuring signals with fast edges.

The linear interpolator connects the samples with a straight line. Even with only a single sample on the edge, there is no evidence of pre-shoot or overshoot on the waveform. The sin(x)/x interpolator has a problem fitting the samples on the edge and blesses the waveform with an obvious overshoot and a less pronounced pre-shoot. These artifacts are called *Gibb's Ears* and can lead to a signal integrity fire drill looking for the source of a nonexistent overshoot. If you observe pre-shoot or overshoot on a pulse waveform, you should change the display interpolator to linear interpolation and see if these artifacts disappear.

In general, it is best to use the linear interpolation on pulse-like waveforms, which prevents this condition. Reserve the sin(x)/x interpolator for sinusoidal signals.

You can minimize problems like this if there are more samples on the waveform edge. Maintaining a high sample rate can help prevent Gibbs Ears.

Conclusion

Hopefully, being mindful of these potential sources of error will make you less likely to fall prey to them. Develop the habit of following these operating guidelines:

- Keep the sampling rate as high as practical.
- When analyzing an unfamiliar signal start with the minimum time/division control setting guaranteeing the highest available sample rate and increase the time/division selection while observing the signal for the onset of aliasing.
- If a waveform appears to be unexpectedly modulated view a horizontally expanded display of the waveform showing sample locations. Set the display to view the trace with persistence on and overlaying the last trace (as in Figure 4). If the displayed samples do not overlap, the peaks and valleys shown in the persistence display and they don't change locations from cycle to cycle you may be sampling synchronously to the signal frequency.
- If you observe pre-shoot and overshoot on a pulse-like waveform using sin(x)/x interpolation view the signal using linear interpolation and see if they disappear.

Remember that the rewards of using a digital oscilloscope greatly outweigh these annoyances, go boldly forth enjoying the benefits with an awareness of what can go wrong.

See EDN collection: Oscilloscope articles by Arthur Pini

Also see

- Oscilloscope rise time and noise explained
- Intuitive sampling theory, part 1
- Intuitive sampling theory, part 2
- Intuitive sampling theory, part 3
- Intuitive sampling theory, part 4