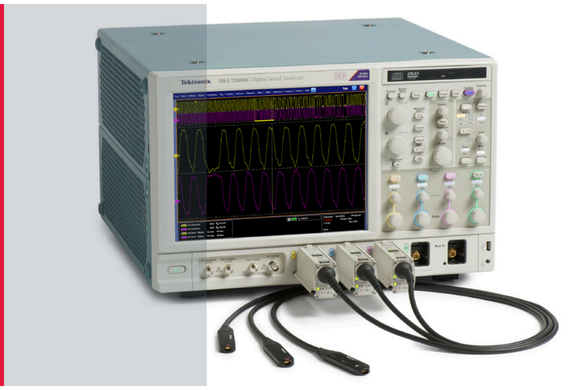

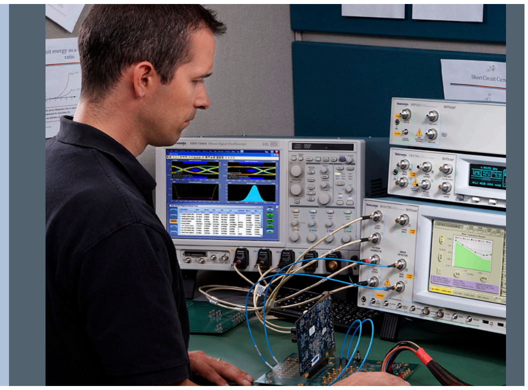

Measurement System Signal Integrity: Important Factors to Consider

Technical Brief

Introduction

When oscilloscope users choose an oscilloscope for making critical measurements, banner specifications are often the only available measures used to make this choice. The top three oscilloscope banner specification categories are:

- Bandwidth
- Sample Rate
- Record Length

And of these banner specification categories, the number one asked for capability in an oscilloscope is bandwidth. After all, more bandwidth means higher performance, right? Well, not necessarily. This article will point out the pitfalls in this very simple assumption, and discuss other important factors that weigh in the decision on measurement system effectiveness. Depending upon your expectation to see and analyze signals as they really are, more knowledge about the true performance of your oscilloscope will be needed.

Bandwidth – What Does This Specification Tell Us?

Analog bandwidth is a measurement specification that simply defines the frequency at which the measured amplitude of a sinewave is 3 dB lower than the actual sinewave amplitude (see IEEE-1057).

If you want to make an amplitude measurement on a sinewave with only 3% error, you would want to measure sinewaves lower in frequency than the rated bandwidth of the oscilloscope. Because most signals are more complex than sinewaves, it has been a general rule of thumb to use a measurement device, like an oscilloscope, that has 5 times the bandwidth of the signal you intend to measure. This general rule cannot be applied in all cases as rise times vary and harmonic content vary for the same frequency signal.

What Does Bandwidth Not Tell Us?

Most typical users choose an oscilloscope to display and measure complex electrical and optical signals, seen on the instrument display as a graph of signal amplitude versus time. Analog Bandwidth, a key oscilloscope specification, is defined by necessity in the frequency domain, not the time domain. Complex signals, according to sampling theory, will contain many spectral components (e.g., multi-tones that contain several discrete but harmonically related sinewave components).

Using spectral analysis, we can learn what these components are for a sampled signal. However, to fully characterize these components, we must know both the accurate amplitude and phase of each of these components making up the complex signal. In this case, the bandwidth specification tells us almost nothing about how the instrument will capture these details. From a bandwidth perspective, all we know is that for a sine wave input, the amplitude error approaches 30% at the specified bandwidth.

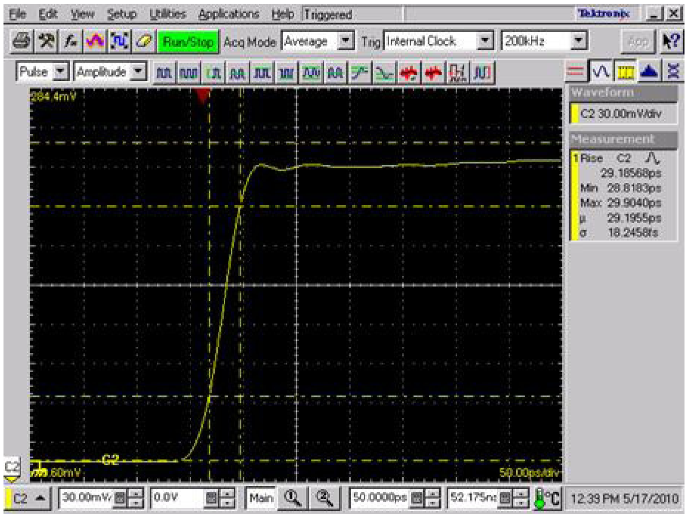

Figure 1. 30ps risetime pulse captured on a Tektronix 50GHz Equivalent-Time Sampling Oscilloscope.

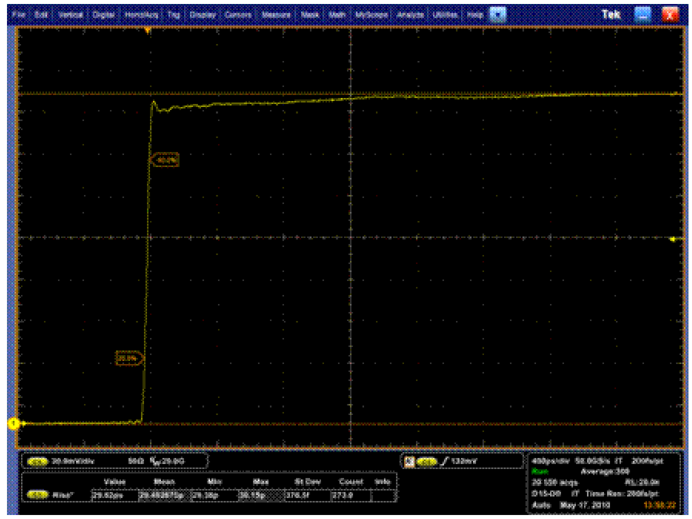

Figure 2. 30ps rise time pulse captured on a Tektronix 20GHz Real-Time Oscilloscope.

When is Rise Time More Important than Bandwidth?

Beyond general purpose signal analysis, most engineers are also interested in time measurements such as rise time and fall time. Therefore, to estimate the oscilloscope system rise time from its specified bandwidth we might use an approximation such as:

$$T_r = \frac{0.35}{BW}$$

This 0.35 factor between bandwidth and rise time is based on a simple one-pole model for 10-90% rise time, and assumes the response of the system is Gaussian. In reality, especially with today's digital oscilloscopes, this assumption is far from correct and the bandwidth times rise time constant of an oscilloscope can approach 0.45.

So what does this really mean when choosing the best oscilloscope to use for making rise time measurements? Two oscilloscopes that have the same bandwidth performance can have very different rise times, amplitude and phase response. So, knowing only the bandwidth of an oscilloscope will not reliably tell us its measurement capability nor its ability to accurately capture complex signals like high speed serial data streams. Furthermore, specifying rise times from the Bandwidth*Rise Time approximation can be inaccurate. The most reliable way to know the rise and fall time response of an oscilloscope is to measure it with an ideal step signal that is much faster than the oscilloscope.

Figure 1 shows an approximate 30ps risetime step being measured by a 50GHz equivalent-time sampling oscilloscope. This same signal is applied in Figure 2 to a 20GHz real-time oscilloscope measuring the same signal. The 50GHz sampling oscilloscope measures a risetime of 29.2ps, and the 20GHz real-time oscilloscope measures a risetime of 29.6ps. Characterizing the fastest 3rd Generation high-speed serial rise times of ~30ps, the 20GHz RT scope has a rise time error of 0.3ps or ~1%.

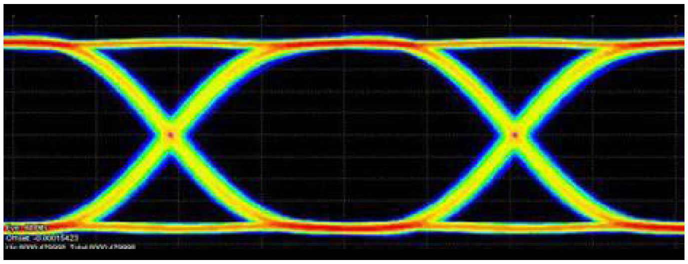

Figure 3: Eye-Diagram of 12.5Gb/s PRBS7 data captured with a Tekronix 20GHz Real-Time Oscilloscope.

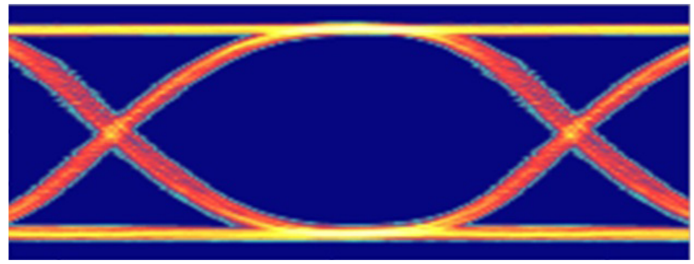

Figure 4. Eye-Diagram of 12.5Gb/s PRBS7 data captured with a Tektronix 50GHz Equivalent-Time Sampling Oscilloscope.

The experiment above points to the crux of this discussion; the bandwidth of a signal cannot by itself tell us its spectral content. The ability of your measurement system to accurately measure rise times will have a more significant effect than bandwidth alone; faster rise times create higher harmonics of the fundamental frequency. Depending on how fast the transitions are in a signal, there may or may not be enough spectral content out at higher harmonics to need a higher bandwidth instrument. Most high speed serial signals, have reduced rise times as they traverse a channel. In Figures 3 and 4, you see two eye diagrams both capturing 12.5Gb/s PRBS7 data; one (Figure 3) taken on a 20GHz real time oscilloscope, and the other (Figure 4) taken on a 50GHz equivalent time sampling oscilloscope. You can see from the comparison of the two pictures that there is essentially no degradation seen on the real time scope. This is because the rise times of the signal are not significantly fast enough to push energy out into higher harmonics, which would necessitate higher bandwidth equipment.

Rise times have also been limited due to the materials used on many high volume, reduced cost circuit boards in production today. There are more exotic circuit board materials available that extend these limitations, but they are often cost prohibitive in many high volume manufacturing environments. Most FR-4 materials in production today have a risetime limitation around ~30ps.

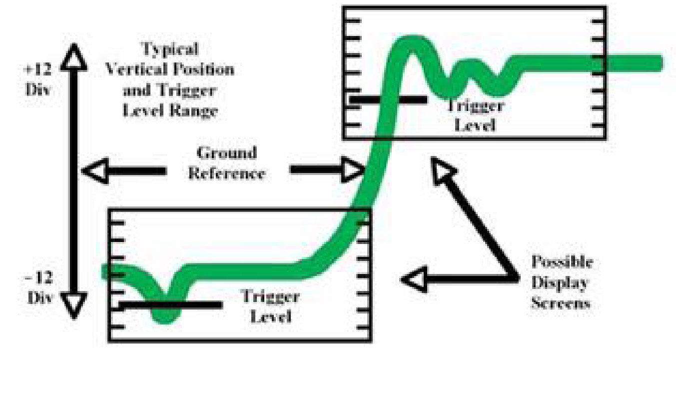

Figure 5. Step Response Abberations.

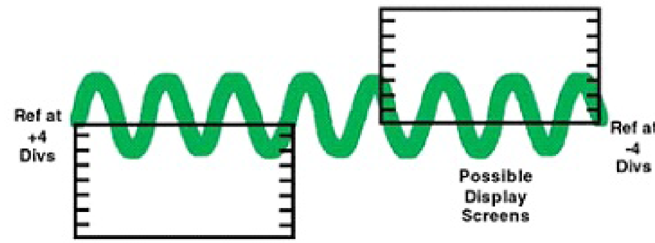

Figure 6. Dynamic range for vertical position and trigger level.

What is Step Response?

In reality, most oscilloscope users want an oscilloscope that has excellent overall step response. Bandwidth, as a specification, tells us next to nothing about how well an oscilloscope can reproduce a complex waveform shape. To verify step response performance, a very clean step generator is needed. When an oscilloscope reproduces this clean (close to ideal) step signal on its screen, the displayed deviations are known as aberrations.

Figure 5 shows what step response aberrations and rise time might look like on an oscilloscope screen.

So how much deviation from an ideal step response are you willing to tolerate when using your oscilloscope? There can be four key contributing categories to these step response deviations:

- Base Oscilloscope Analog Performance
- Probing Effects
- Under Sample Alias Effects
- Digital Signal Processing Effects

What Defines the Base Analog Performance of an Oscilloscope?

True analog performance is defined by the analog oscilloscope circuitry leading up to the analog-to-digital (A/D) converter. This includes the vertical input attenuators, as well as the vertical amplifiers, position controls and trigger pick off circuitry in each channel of the oscilloscope. The following discussion will explain what you will need to consider in this circuitry when using your oscilloscope to look for subtle signal integrity details.

When you decide to explore a detail of your waveform, you will likely discover the limits of the traditional vertical position control quite quickly. Each time you change the volts/div control to expand the trace, you must reposition the waveform. And when you wish to expand a portion of the waveform that is not close to the ground reference, the typical –12 divisions of vertical position range quickly limits the expansion or zoom range you can use. Also, when you reposition a vertically expanded trace, you may also need to retain a trigger point or timing reference on that detail so that it remains on-screen. This requires that you also consider the range of your trigger level control. Figure 6 illustrates these analog limits.

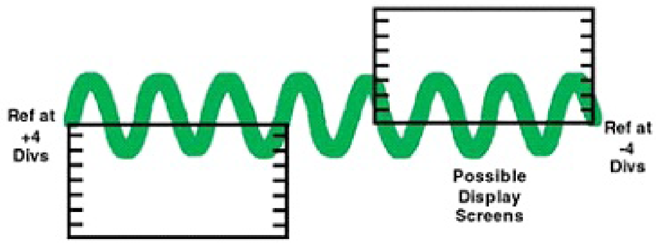

Figure 7: Vertical position moves the volts/div zero reference point.

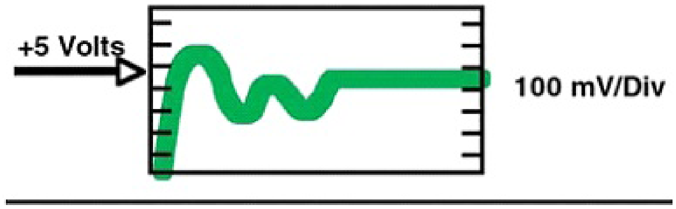

Figure 8. Vertical offset changes the volts/div reference point from zero, to some other voltage.

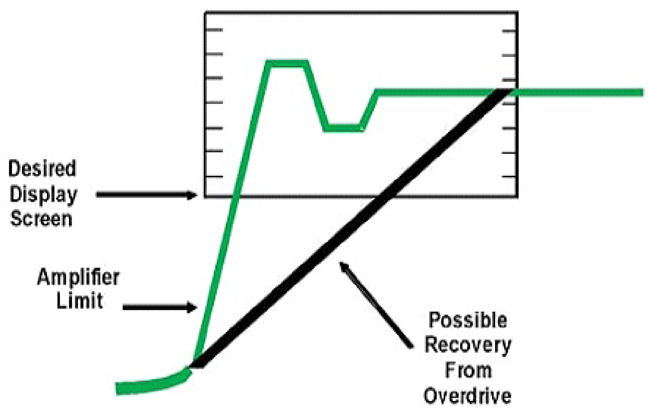

Figure 9. Overdrive recovery characteristics can cause high-speed details to disappear.

When you vertically reposition the trace, you also reposition the expansion reference, which is fixed at ground, as shown in Figure 7.

If you want to “zoom” in on a waveform detail that is not at ground, consider specifying an offset control, illustrated in Figure 8.

Vertical offset allows you to redefine the displayed reference for expansion. For example, if you want to zoom in on a waveform detail at the top of a pulse that is at +5 volts, simply set the offset control to +5 volts. Then change the volts/div to the needed sensitivity without the need to reposition the trace. Offset results in a tremendous increase in the vertical analog dynamic range of your oscilloscope.

One drawback to “zooming” in on vertical waveform details will be the overdrive recovery limit of your oscilloscope’s vertical amplifiers and acquisition system. When you drive a portion of your waveform off screen in order to expand on a particular detail, the vertical system will need to recover from that overdriven condition. A typical overdrive recovery specification could be “90% recovery in 1 nsec.” This would imply “99% recovery in 2 nsec.” On current models of Tektronix DPO/DSA/MSO Series performance oscilloscopes, overdrive recovery can be as low as 100ps for a 15 division overdrive.

True analog performance goes well beyond bandwidth. Bandwidth, step response aberrations which affect in-band flatness and phase-response, rise time and fall time must be understood. Having the ability to zoom in on details requires both vertical position and offset controls that have adequate range for your applications. This implies that the oscilloscope vertical system also has adequate overdrive recovery characteristics for your measurement requirements.

And any time a part of the signal is taken out of the range of the analog-to-digital converter (as just described), digital signal processing of the signal can become more of a problem than a solution, and having the ability to disable DSP is crucial to accurately evaluating regions of a waveform using the techniques mentioned above. This is true because the entire signal is no longer available to the processing system and DSP artifacts can swamp subtle signal features. We will cover more on this topic later in this technical brief.

How Can a Probe Affect Bandwidth and Rise Time?

If you solder in a resistor, a capacitor, or even a piece of wire to a circuit, would you expect these components to affect the signals in that part of the circuit? Of course, you would. These components affect signal amplitude, slow down a signal, and influence signal shape. Likewise, every oscilloscope probe has some capacitance. Every oscilloscope probe has a resistance value. Every oscilloscope probe is going to affect the signals at the measurement point. It's not a matter of if the probe will change the signal; it s a matter of how much.

An ideal probe would capture any signal with perfect fidelity and would be non-invasive to the circuit under test. The requirements for the probe designer seem clear: extremely high bandwidth, broad dynamic range, and don't affect the signal under test. We'll consider the following topics and their impact on bandwidth, rise times, and the step response:

- Different Requirements. Different Probes.
- Probe Bandwidth and Rise Time
- Short Leads & Selecting the Right Accessories

Different Requirements. Different Probes.

Probes are used in many different environments and are used in testing to numerous industry standards. For example, measuring high voltage in power applications requires adherence to safety certification standards. This type of measurement requires mechanically rugged probes with very large dynamic range, but these probes do not require high bandwidth. On the other hand, applications such as modern serial standards require measurement tools that use precision parts that are high bandwidth and have a low dynamic range.

It is important to realize that oscilloscope probes are designed for target markets that influence the probe's design requirements. For varying measurement environments, probes have varying bandwidth capabilities. When determining the right probe for your measurement, make sure the probe has enough bandwidth. How much is enough? Let's take a look.

Probe Bandwidth and Rise Time

Bandwidth

Bandwidth is the range of frequencies that an oscilloscope probe is designed for. For example, a 100 MHz oscilloscope probe is specified for measurements on all frequencies up to 100 MHz. However, the probes ability to capture signals changes across the specified frequency range. In fact, every probe manufacturer assumes that at the maximum specified bandwidth, the probe's frequency response is down 3 dB. At frequencies beyond the 3 dB point, signal amplitudes are overly attenuated and measurement results may be unpredictable.

Rise Time

Bandwidth describes frequency domain characterization but does not provide the complete picture of how the probe and scope will reproduce a complex waveform shape over time. To get the full story, the step response is essential in obtaining the time domain characterization. This characterization is provided through the probe's rise time value where rise time is obtained by evaluating the response of a system to a step input that is faster than the capabilities of the test system.

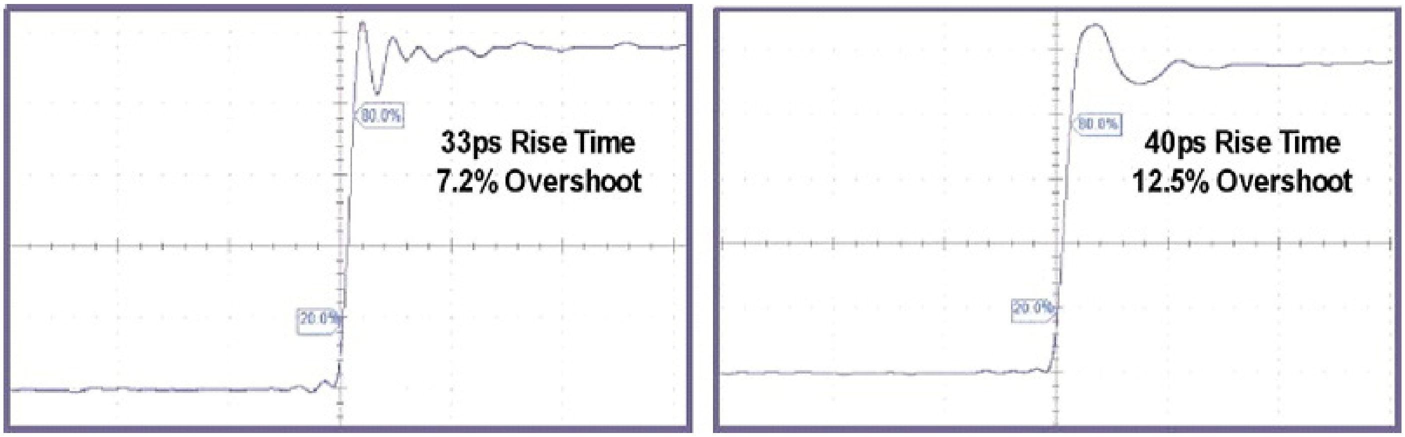

Figure 10. Long leads can cause very different measurement results.

Short Leads & Selecting the Right Accessories

When a probe is attached to a measurement point, it will induce a load on the circuit. The capacitive and inductive elements typically come from the probe tip geometry and wire lengths. The inductive element is likely to have variation due to the addition of different probe tip accessories and varying wire lengths for signal and ground leads.

Keep Signal Leads and Ground Leads Short

At times, connecting wire leads to the test point may be challenging. Users may compensate for this connectivity problem by making the signal or ground wires too long. As shown in Figure 10, lengthening the signal leads on a P7500 probe influences the measurement. The plot on the left has shorter leads whereas the plot on the right has much longer signal wires; the rise time and overshoot values change due to wire length. The P7500 probe series is a TriMode™ probe that offers the ability to view the signal in single-ended, differential and common mode configurations without the need to make a new connection every time. The probe allows all of these combinations to be made with the simple press of a button on the probe.

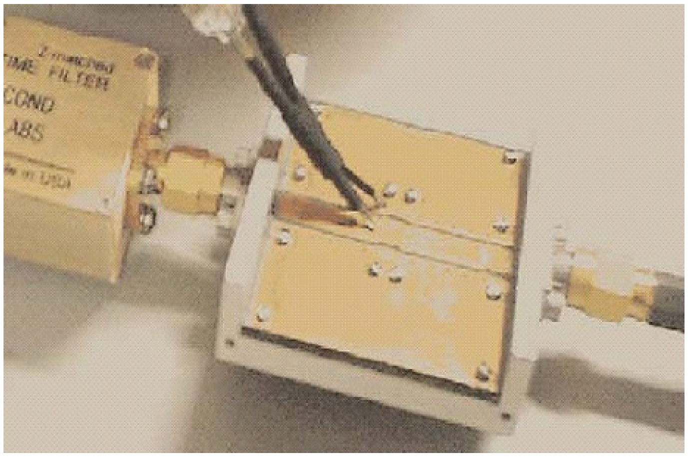

Figure 11. Make sure you understand how the probe tip accessory will affect your measurement.

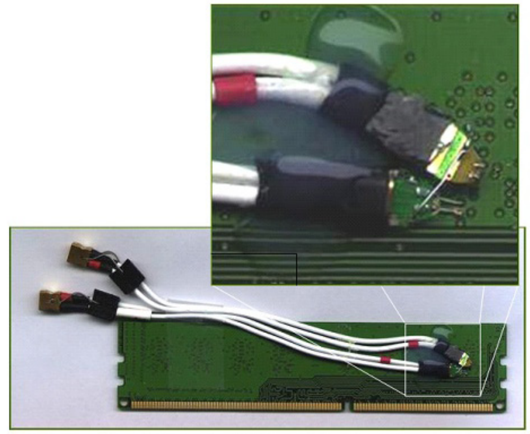

Figure 12. P7500 series Long-Reach solder probe tips get to the source of the measurement.

Selecting the Right Accessories

Oscilloscope probes are typically equipped with a number of different probe tip accessories. The different accessories are included with the probe to support the various design activities such as validation, debug, compliance testing, etc. Additionally, some probes are equipped with solder down solutions to enable a solid, hands-free connection.

Users should be aware that different probe accessory tips may have different measurement results. In an effort to make the connectivity to the device under test more convenient, some test leads have clips, some leads are longer, and some probe connectors have square pin sockets. Consider the connection in Figure 11. The probe in this picture has 1" signal and ground leads. Obviously, these longer leads will be more inductive and will cause effects such as ringing, aberrations or overshoot. Figure 12 shows the connection of a Long-Reach Solder Tip that is available for use with the P7500 TriMode™ probe series. The Long-Reach Solder Tip allows you to connect as close to the reference signal as possible, minimizing effects of inductance by bringing terminations to the head of the tip; ensuring you have the shortest leads to the DUT. The point is that there are a wide variety of connection methods and these varying methods may cause varying measurement results. Probe manufacturers will typically specify how the probe tip accessory will influence the measurement.

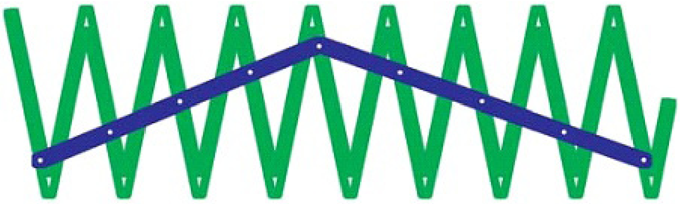

Figure 13. Undersample Aliasing.

What are Under Sample Alias Effects?

For a complex waveform, the spectral sine waves that exist in the waveform can be determined with a spectrum analyzer, or a Fast Fourier Transform (FFT) of the waveform. Nyquist Theory states that, for a signal to be properly digitized, it needs to be sampled more than twice for each and every spectral sine wave cycle that is in the waveform. If the fastest sine wave in your signal is not sampled faster than this two times rate, then Nyquist Theory is violated and the signal will be reconstructed in a false way (aliased) that cannot be corrected. Figure 13 shows how under sampling can cause false waveform reconstruction.

If the signal present on the display seems to have an appearance of being not triggered, even though the trigger light is solidly lit, then under sampling is very likely to be the problem. If this is suspected, turn the sec/div control to a faster speed and you should eventually see a stable triggered display. This can be true for a repeating waveform.

For single-shot conditions, it is not possible to have a hint about this type of aliasing (false waveform reconstruction) without some real initial knowledge about the waveform. The shape of a repeating waveform can appear to be correct, but can have the wrong timing. Or, the shape of fast moving waveform details can be incorrect, due to under sampling.

Perceptual aliasing is where your eyes can really be fooled when looking at a displayed waveform, even though you have satisfied Nyquist Theory. This means that you have more than two samples for every spectral sine wave component in your waveform, as previously described. Figure 14 shows this type of dots display.

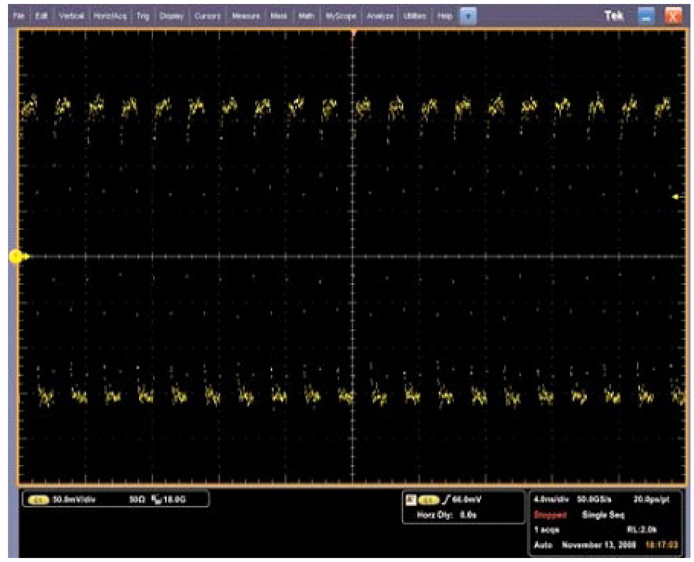

Figure 14. Dot mode does not show true waveform shape.

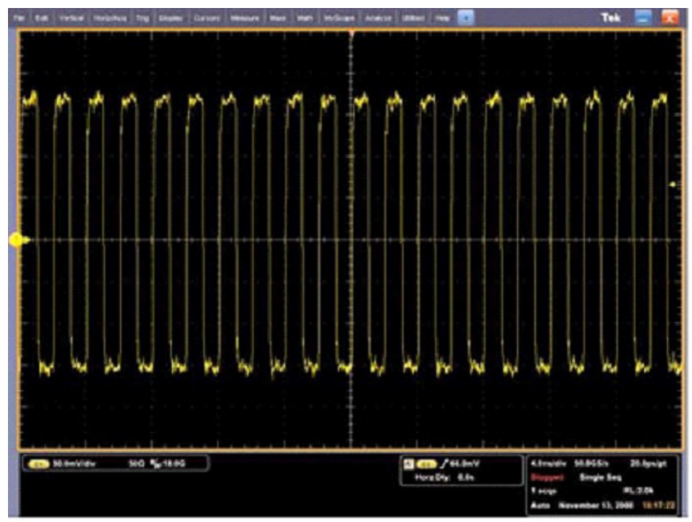

Figure 15. Vector mode improves perceptual aliasing.

Perceptual aliasing can appear as patterns of dots. The real waveform, of course, is still not visible. This type of displayed aliasing, or false waveform reproduction, can be significantly improved by joining the dots with various types of lines. This process of joining the dots is called interpolation, as shown in Figure 15, using the same dots used in Figure 14.

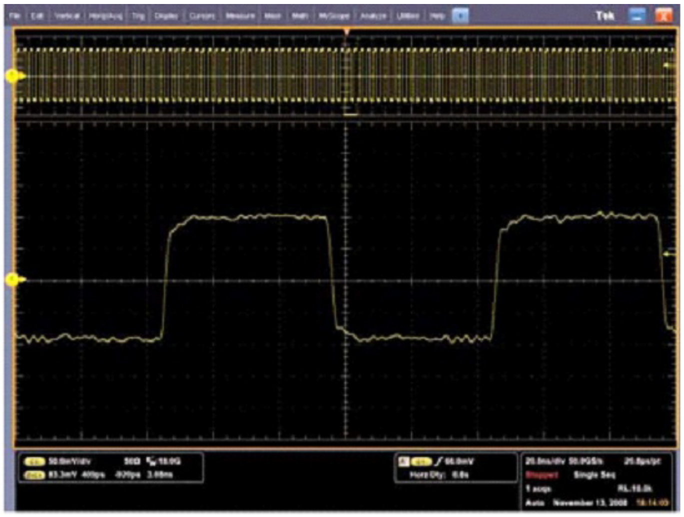

Figure 16a. Oversampled waveform using sin(x)/x interpolation.

In order to truly remove the effects of perceptual aliasing, we must use the digital filter from Nyquist Theory, called sine(x)/x. This mathematical filter allows truly correct intermediate points to be calculated between real samples on a waveform, provided that no “actual aliasing” is present. This means that more than 2 samples exist for each and every spectral (sine wave component) cycle in the signal that reaches the oscilloscope analog-to-digital converter.

So what will sine(x)/x do to a step response that is under sampled? In Figure 16a the waveform is over sampled, and is displayed correctly with sine(x)/x interpolation. In Figure 16b the waveform is under sampled, with sine(x)/x interpolation used for the display, and results in ringing that isn’t present on the original signal. In Figure 16c, the waveform is also under sampled, with linear interpolation (straight lines between acquisition samples) used for the display, and results in a better representation of the original signal.

As you can see, under sampling combined with interpolation can give you very misleading information about a waveform. So be careful about your choice of sample rate and display interpolation to ensure the best measurement signal fidelity for your signals.

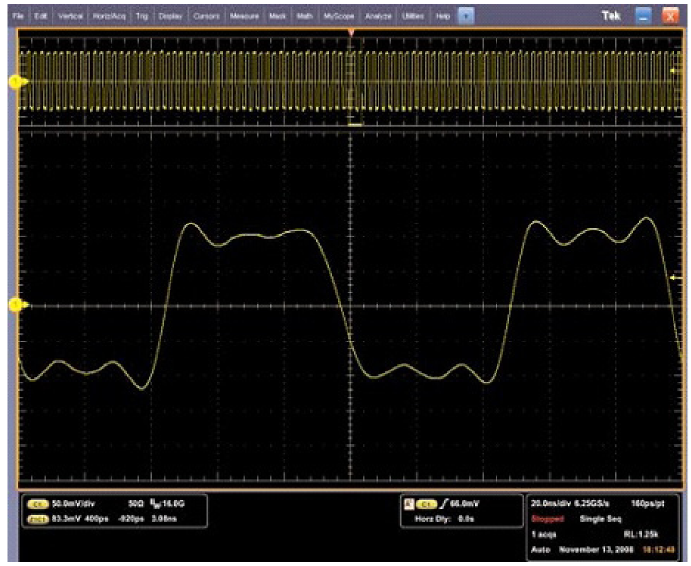

Figure 16b. Undersampled waveform using sin(x)/x interpolation.

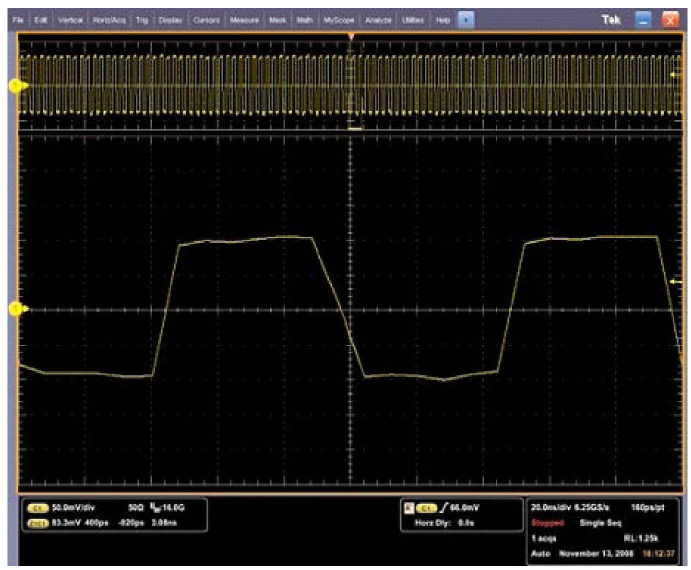

Figure 16c. Undersampled waveform using linear interpolation.

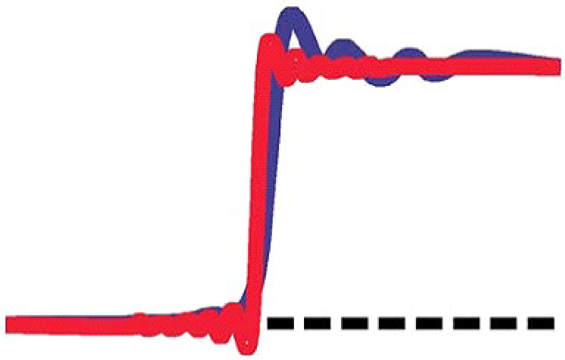

Figure 17. DSP enhanced shape, rise time and bandwidth.

How can Oversampling Improve Data Acquisition?

Oversampling can reduce quantization noise that is produced as part of the Analog to Digital conversion in an oscilloscope. The amount of noise reduction will depend on the passband, or frequency response of the scope in question, and how much oversampling is applied.

For example, consider an oscilloscope with a 20GHz bandwidth, and a sample rate of 50GS/s. The Nyquist frequency of this instrument would be 25GHz. With a passband of 20GHz and a maximally flat response, you might see some of the higher frequency noise that exists around the fundamental of the sampling frequency. If you were to extend that sample rate of the instrument out to 100GS/s, your Nyquist frequency would now be at 50GHz and noise that may have existed as a result of the digitizing process would now be spread over twice the frequency span. At the same time, your passband for your oscilloscope has not changed, so less noise is included into the bandwidth of the instrument.

Additionally, this concept also points to the benefit of using only the bandwidth of the instrument that is needed for measuring the signal of interest; if you are using an instrument at full bandwidth, and measuring signals that are at half of the rated bandwidth of the instrument, then you are seeing any noise that may exist at that higher bandwidth included into your measurement.

What can Digital Signal Processing do to Risetime, Bandwidth and Signal Fidelity?

In reality, interpolation between real samples is a form of Digital Signal Processing (DSP). Processing waveforms can serve many purposes, including the following:

- Bandwidth Enhancement
- Rise Time Improvement
- Gain and Wave Shape Calibration
- Spectral Magnitude and Phase Correction
- Optical Reference Receiver Normalization
- Jitter Analysis of Waveform Deviations and Anomalies

In Figure 17, the blue trace is the uncorrected waveform through a less than perfect vertical amplifier system on an oscilloscope. The red trace shows DSP correction of shape, as well as enhanced bandwidth and improved rise time.

A DSP filter can be used to improve the pass band magnitude and phase response of an oscilloscope acquisition channel. This filter can extend the bandwidth, flatten the oscilloscope channel frequency response, improve phase linearity, and provide a better match between channels. Fourier Series DSP filtering is most commonly used for bandwidth and rise time improvement.

When enhancing the rise time of a fast rise step, the Fourier Series DSP converges to the mid point of the step. On both sides of the step, the series will oscillate. The height of the peaks of the oscillation decreases away from the step, but the heights of peak1, peak2, etc. remain the same as the number of terms summed increases, making the amplitude and shape of the ring the same but at a higher frequency. The peak overshoot of each ring has a constant height (=18% of the step) and moves towards the step edge as the number of terms increases. This effect is referred to as the Gibbs Phenomenon.

So, according to Gibbs Phenomenon, pre and post ringing will occur on the step edge, when enhancing bandwidth to the limit with Fourier math. This is shown in Figure 17, where the oscilloscope channel response is low pass and linear in phase. The amount of Gibbs Phenomenon ringing will depend on the amount of rise time and bandwidth improvement being implemented with the DSP, as well as the speed of the signal being measured.

In order for DSP bandwidth enhancement to work consistently as described here, two conditions must be met. First, the sample rate must be kept high enough to ensure that no spectral frequency component at or above the Nyquist rate (half the sample rate) gets through to the oscilloscope's analog-to-digital converter. If this condition is not met, under sampling will occur, and DSP will very likely destroy the integrity of the displayed waveform. Second, the total waveform must be kept within the range of the analog-to digital converter. If you choose to zoom in on a detail of the waveform, and subsequently drive another part of the signal vertically off screen, digital signal processing is very likely to cause unwanted distortions.

What is "Effective Number of Bits", and how will it effect my measurements?

Whether you are designing or buying a digitizing system, you need some means of determining actual, real-life digitizing performance. How closely does the output of any given analog-to-digital converter (ADC), waveform digitizer or digital storage oscilloscope actually follow any given analog input signal?

At the most basic level, digitizing performance would seem to be a simple matter of resolution. For the desired amplitude resolution, pick a digitizer with the requisite number of "bits" (quantizing levels). For the desired time resolution, run the digitizer at the requisite sampling rate. Those are simple enough answers. Unfortunately, they can be quite misleading, too.

While an "8-bit digitizer" might provide close to eight bits of accuracy and resolution on DC or slowly changing signals, that will not be the case for higher speed signals. Depending on the digitizing technology used and other system factors, dynamic digitizing performance can drop markedly as signal speeds increase. An 8-bit digitizer can drop to 6-bit, 4-bit, or even fewer effective bits of performance well before reaching its specified bandwidth. If you are designing an ADC device, a digitizing instrument, or a test system, it is important to understand the various factors affecting digitizing performance and to have some means of overall performance evaluation. Effective bits testing provides a means of establishing a figure of merit for dynamic digitizing performance. Not only can effective bits be used as an evaluation tool at various design stages, beginning with ADC device design or selection, but it can also be used to provide an overall system dynamic performance specification.

For those making digitizing system purchase decisions, effective bits is an equally important evaluation tool. In some instances, effective bits may already be stated as part of the system or instrument specification. This is becoming increasingly common for waveform digitizing instruments. However, effective bits may not always be specified for individual instruments or system components. Thus, it may be necessary to do effective bits evaluation for purposes of comparison. If equipment is to be combined into a system, an effective bits evaluation can provide an overall system figure-of-merit for dynamic digitizing system performance.

Essentially, effective bits provides a means of specifying the ability of a digitizing device or instrument to represent signals of various frequencies.

Like gain-bandwidth or Bode plots, ENOB generally – but not always – decreases with frequency. The major difference is that the ENOB plot compares digitization precision or digital bits of accuracy rather than analog gain (or attenuation) accuracy.

Fundamentally, an 8-bit digitizer provides eight effective bits of accuracy only at DC and low frequencies or slow signal slopes. As the signal being digitized increases in frequency or speed, digitizing performance drops to lower and lower values of effective bits.

This decline in digitizer performance is manifested as an increasing level of noise on the digitized signal. “Noise”, here, refers to any random or pseudo-random error between the input signal and the digitized output. This noise on a digitized signal can be expressed in terms of a signal-to-noise ratio (SNR),

$$SNR = \left(\frac{rms\_signal}{rms\_error} \right)$$

where rms (signal) is the root-mean-square value of the digitized signal and rms (error) is the root-mean-square value of the noise error. The relationship to effective bits (EB) is given by,

$$EB = \log_2 (SNR) - \frac{1}{2} \log_2 (1.5) - \log_2 \left(\frac{A}{FS} \right)$$

where A is the peak-to-peak input amplitude of the digitized signal and FS is the peak-to-peak full-scale range of the digitizer's input. Other commonly used formulations include,

$$EB = N - \log_2 \left(\frac{rms\_error}{ideal\_quantization\_error} \right)$$

where N is the nominal (static) resolution of the digitizer, and,

$$EB = -\log_2 \left(rms\_error * \sqrt{\frac{12}{FS}} \right)$$

Notice that all these formulations are based on a noise, or error level, generated by the digitizing process. In the case of Equation 3, the “ideal quantization error” term is the rms error in ideal, N-bit digitizing of the input signal. Both Equations 2 and 3 are defined by the IEEE Standard for Digitizing Waveform Recorders (IEEE std. 1057). Equation 4 is an alternate form for Equation 3. It is derived by assuming that the ideal quantization error is uniformly distributed over one least significant bit (LSB) peak-to-peak. This assumption allows the ideal quantization error term to be replaced with

$$\frac{FS}{2^n * \sqrt{12}}$$

where FS is the digitizer's full-scale input range.

Another important thing to notice about these equations is that they are based on full-scale signals (FS). In actual testing, test signals at less than full scale (e.g., 50% or 90%) may be used. This can result in improved effective bits results. Consequently, any comparisons of effective bits specifications or testing must take into account test signal amplitudes as well as frequency.

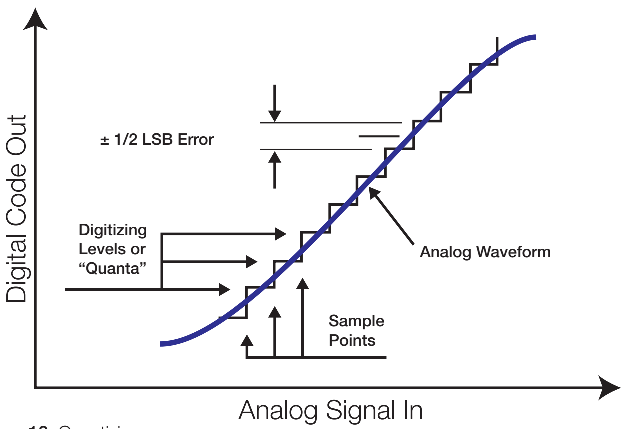

Figure 18. Quantizing error.

Resolution or Effective Bits (N)	Quantizing Levels	Signal-to-Noise Ratio in dB (6.08N+1.8dB)
4	16	26.12
6	64	38.28
8	256	50.44
10	1,024	62.60
12	4,096	74.76
14	16,384	86.92
16	65,536	99.08

Table 1. Digitizer is ±1/2 LSB of error.

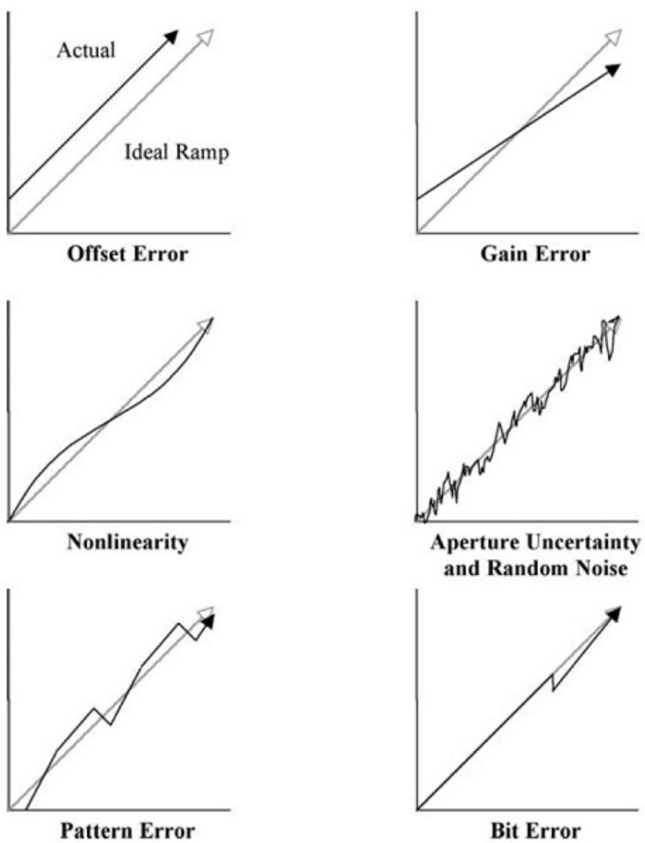

Figure 19. Typical A/D errors.

Error Sources in the Digitizing Process

Noise, or error, related to digitizing can come from a variety of sources. Even in an ideal digitizer, there is a minimum noise or error level resulting from quantizing. This "quantizing error" amounts to ± 1 LSB (least 2 significant bit). As illustrated in Figure 18 and Table 1, this error is an inherent part of digitizing. It is the resolution limit, or uncertainty, associated with ideal digitizing. To this basic ideal error floor, a real-life digitizer adds further errors. These additional real-life errors can be lumped into various general categories –

- DC offset (also AC offset or "pattern" errors, sometimes called "fixed pattern distortion," associated with interleaved sampling methods)
- Gain error (DC and AC)
- Nonlinearity (analog) and Nonmonotonicity (digital)
- Phase error
- Random noise
- Frequency (time base) inaccuracy
- Aperture uncertainty (sample time jitter)
- Digital errors (e.g. data loss due to metastability, missing codes, etc.)
- Other error sources such as trigger jitter

Figure 19 illustrates some of the more basic error categories to give you a visual idea of their effects. Many of the errors encountered in digitizers are the classical error types specified or associated with any amplifier or analog network. For example, DC offset, gain error, phase error, nonlinearity and random noise can occur anywhere in the waveform capture process, from input of the analog waveform to output of digitized waveform values.

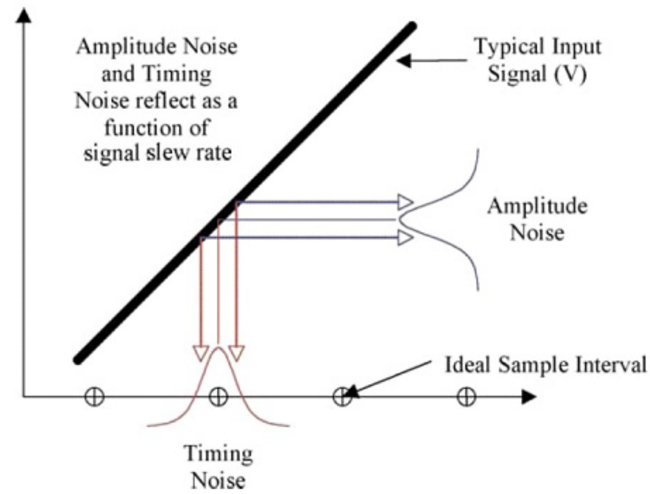

Figure 20. Aperture uncertainty, or sample jitter, makes an amplitude error contribution that is a function of slew rate and timing jitter. Similarly, amplitude noise can impact timing measurements.

On the other hand, aperture uncertainty and time base inaccuracies are phenomena associated with the sampling process that accompanies waveform digitizing. The basic concept of aperture uncertainty is illustrated in Figure 20.

The important thing to note from Figure 20 is that aperture uncertainty results in an amplitude error and the error magnitude is slope dependant. The steeper the slope of the signal, the greater the error magnitude resulting from a time jittered sample. Aperture uncertainty is only one of many reasons for decreases in effective bits at higher signal frequencies or slopes. However, aperture uncertainty serves as a useful and graphical example for exploring input signal frequency and amplitude related issues.

To gain further insight into the effects of aperture uncertainty, consider sampling the amplitude of a sine wave at its zero crossing. For a low-frequency sine wave, the slope at the zero crossing is low, resulting in minimal error from aperture uncertainty. However, as the sine wave's frequency increases, the slope at the zero-crossing increases. The result is a greater amplitude error for the same amount of aperture uncertainty or jitter. Greater error means lower SNR and a decrease in effective bits. In other words, the digitizer's performance falls off with increasing frequency. This is expressed further by the following equation.

$$f = \frac{1}{\sqrt{6} * \pi * \Delta t * 2^N}$$

In equation 7, f is the frequency of a full-scale sine wave that can be digitized to n bits with a given rms aperture uncertainty, Δt. If aperture uncertainty remains constant and frequency is increased, then the number of bits, n, must decrease in order to maintain the equality in Equation 7.

There is, however, a way around the necessary decrease in bits, n, for increasing frequency. This relates back to the concepts illustrated in Figure 20. If the amplitude of the sine wave is decreased from full scale, the zero-crossing slope decreases. Thus, the amplitude error decreases, resulting in a better effective bits number. This points out an important fact when comparing effective-bit numbers from various digitizers. Effective bits depends not only on frequency, but on the amplitude of the test waveform. Any one-to-one testing or comparison of digitizers must include specifications of the input waveform's amplitude (typically 50% or 90% of full scale) as well as frequency.

Also, it should be noted that input amplifier roll-off, post-acquisition filtering and other processing can reduce signal amplitude internal to the digitizing instrument. This can result in effective bit specifications that overstate the actual, real-life dynamic performance of the instrument.

Will Effective Bits matter to you?

It will greatly depend on what you are trying to measure, as to whether or not the Effective Number Of Bits will affect your measurement outcome. High speed serial data has harmonics at very specific frequencies, which may pass through the measurement system mostly unaffected by a decrease in Effective Bits. It is important that you know what the effective Bits performance of the instrument you choose to measure with looks like across the full rated bandwidth of the instrument. Only then can you know whether the performance is sufficient for your needs.

Summary

Bandwidth, as a banner specification, tells you something about how well your oscilloscope will reproduce the true nature of your waveform, but not the entire story. Step response rise time, fall time, aberrations, and in-band flatness and phase response will tell you much more about the true fidelity of your measurement system. And when you want to explore waveform details, remember that vertical offset and trigger level range, along with good overdrive recovery capability, will allow you to see these details as you expect. And don’t forget probe loading effects, especially from signal tip and ground lead adapters.

Sample rate is another banner specification. When you have the right amount of sample rate, combined with the correct interpolation between acquired samples, with proper trigger-to-sample time correction, you can be less concerned about under sample aliasing effects. And make sure that digital signal processing enhancements of bandwidth and rise time, when used, delivers to you the signal fidelity that you expect. Lastly, ensure that the effective bits of the measurement system you have will meet the signal integrity requirements of your test signals.

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