



Enhanced Sample Rate Mode Measurement Precision

Summary

Enhanced Sample Rate, combined with the low-noise system architecture and the tailored brick-wall frequency response in the HDO4000A, HDO6000A, HDO8000A and MDA800A series oscilloscopes dramatically improves the measurement precision when the input frequency signals are very high frequency, such as a high-frequency sinewave or fast edge.

Enhanced Sample Rate Mode Overview

The low-noise system architecture and the tailored frequency response employed in the HDO4000A, HDO6000A, HDO8000A and MDA800A series provides the foundation for enhancing ADC sample rates through additional techniques. In this case, carefully constructed filters combined with a pristine front-end amplifier and a frequency response carefully limited to 1 GHz provide the opportunity to provide more measurement precision than would otherwise be possible. The technique utilized to achieve higher measurement precision is *interpolation*, and this technique is used by default as an *Enhanced Sample Rate* up to 10 GS/s. By integrating the Enhanced Sample Rate functionality with the normal Timebase controls for Sample Rate, Time and Acquisition Memory adjustment, the oscilloscopes are optimized for best waveform signal fidelity in all situations.

Interpolation Overview

Interpolation is an important feature in today's digital oscilloscopes. The main purpose of a digital oscilloscope is to analyze and view analog waveforms. To do this, the digital oscilloscope samples a waveform at some finite sample rate, generating a vector of voltages with respect to time. Since this vector represents a set of points (not the actual smooth analog waveform), it is often desirable to modify the acquired waveform by generating samples that are predicted between the actual acquired points. The generation of samples that occur in between actual waveform samples is called interpolation. When done properly, this results in a waveform with a higher resultant sample rate that is a closer approximation of the analog waveform under analysis.

Interpolation Filter Construction

There are two widely used interpolation methods – linear and SinX/x (also referred to as simply SinX interpolation). Linear interpolation simply joins digitally sample points with a straight line whereas SinX/x interpolation provides a more reasonable result if some key criteria for oversampling and frequency response can be met.

Linear Interpolation

The simplest form of interpolation is linear interpolation. Linear interpolation is performed by assuming that a straight line joins each waveform sample. This is a very simple, but naïve, method and provides limited improvements. It is rarely used for upsampling, but studying how it works provides a solid basis for understanding more rigorous techniques.

Linear interpolation can be viewed as a convolution of an upsampled waveform with a triangular window. The triangular window is formed by generating a triangle that has a height of 1.0 and a width that is twice the sample period. As the window slides to the right, interpolated points are calculated by calculating the sum of the values of the window multiplied by the value of actual samples at the times where the window and the sample points intersect. The interpolated sample is placed at the time of the window apex (reference image below):



The width of the window defines its memory - the time over which actual samples affect the interpolated samples. Since the width of the window is twice the sample period, only samples that bracket the time of the interpolated sample affect the interpolated value. This convolution can also be realized with a digital filter in an upsampling arrangement, as shown in the image below:





The arrangement shown above is a four point upsampler. In this arrangement, every new sample produces four new ones. The filter output is a function only of the new input and the last one. The filter coefficients are generated by sampling the window.



By examining the diagrams, it is easy to understand the pattern for developing the filter coefficients.

SinX/x Interpolation

A popular and more complicated form of interpolation is called SinX/x (also referred to as Sync or simply SinX interpolation). SinX interpolation gets its name from the well known shape of the window function used for the convolution. Unlike the narrow pointed triangle of linear interpolation, the window for SinX interpolation is a theoretically never ending damped sinewave, as shown in the image below:



This window shape derives from an important assumption that Nyquist's criterion has been obeyed in the sampling of the original waveform. In other words, it assumes that all of the frequency content in the analog waveform sampled lies below one-half the sample rate that the waveform was sampled at - a reasonable assumption. When this assumption is made, and the inverse Fourier transform of this assumed spectrum is calculated, the result is this well-known function.



In the general case, this assumption is the best assumption that can be made - but it is not always correct, as we shall see. As a result, SinX interpolation is truly the most valid interpolation method. This is understood by examining the meaning of Nyquist's criterion. Nyquist said that when all of the frequency content of a signal lies below one-half the sample rate, then the continuous analog signal can be completely determined from the sampled points. SinX interpolation is merely the mechanics for obtaining the continuous analog signal.

SinX interpolation suffers from some mathematical and practical technicalities that make it impossible for this method to be perfect. First of all, the Sync function goes forever and must be truncated at a point where the truncation error dips acceptably low. This is due to the fact that a truly bandwidth limited signal must have infinite length implying that all sample points must be known for all time. As it turns out, the influence of points further and further from the point being interpolated diminishes rapidly and the truncation provides highly acceptable results. Another drawback is that in a sampled system, noise and artifacts due to DSO architecture like channel interleaving creep in, causing noise and distortion above the Nyquist limit. Again, the errors caused by this can be kept acceptably low.

When is Interpolation Valid?

In pure mathematical terms, neither interpolation method is valid. For example, linear interpolation implies that waveform points are joined by straight lines - a technical impossibility due to bandwidth limitations alone. SinX interpolation, as mentioned, is valid only when Nyquist's criteria is met - which is never fully the case - and when the waveform is infinitely long.

Suffice it to say, interpolation can be valid to a large extent. If you are uncomfortable with this concept, consider the fact that a digital oscilloscope is used to view, analyze, measure and otherwise make judgements on an analog signal. The validity of interpolation is philosophically related to the concept that statements can be made about an analog waveform utilizing only an imperfect digital representation. Since we know that depending on bandwidth, sample rate, signal fidelity etc. that we can make good assumptions about an analog waveform with the digital scope, we can also say that interpolation is a generally good method.

Enhanced Sample Rate with SinX/x Interpolation

Enhancing the sample rate using SinX/x interpolation is a very valid approach in the HDO4000A, HDO6000A, HDO8000A and MDA800A oscilloscopes given their following characteristics of:

• The frequency response of a 1 GHz oscilloscope is carefully tailored to roll off quickly to avoid any Nyquist criteria violations and aliasing of the signal by the analog-to-digital converter (ADC).





- The sample rate to bandwidth ratio is at least 2.5:1 (worst case for a 1 GHz oscilloscope, in this case) and more realistically at least 3:1 given the maximum frequency most engineers would typically use a 1 GHz scope at (700-800 MHz sinewave, or a rise time equivalent to one-third to one-half of the oscilloscope's capability).
- HD4096 low-noise architecture with high-performance input amplifiers and 12-bit ADCs provides much better vertical gain precision and accuracy compared to a conventional 8-bit oscilloscope, and this improves the interpolation result.

In the worst-case scenario (SR:BW = 2.5:1), the Nyquist criteria is satisfied, and with the carefully tailored fast bandwidth rolloff beyond 1 GHz, the likelihood of aliasing is essentially eliminated. In this scenario, SinX/x is still very effective. In the more realistic scenarios (SR:BW = 3:1), the SinX/x interpolation works extremely well in adding to measurement precision, as we will show in some examples to follow.

Enhanced Sample Rate Practical Results

Overview

The 10 GS/s Enhanced Sample Rate improved precision and accuracy is demonstrated with two different input signals:

- Sinewave of frequencies ~250 MHz, ~500 MHz and ~800 MHz with measurements and statistical analysis made of amplitude and frequency measurements with and without Enhanced Sample Rate.
- Fast rise time pulse input with measurements and statistical analysis of overshoot and rise time with and without Enhanced Sample Rate.

Input Sinewave Examples

Input sinewaves of 246.2, 506.2, and 816.2 MHz were acquired with Channel 1 (yellow) of an HDO6104A-MS (1 GHz, 12-bit resolution, mixed signal high definition oscilloscope). A Hewlett-Packard 8648B Signal Generator was used to generate the sinewaves. Five thousand 5 µs long acquisitions were made and the amplitude and frequency were measured on each acquisition. Amplitude was measured once per acquisition, and frequency was measured for each full period in the acquisition (approximately 6 to 20 million measurements, depending on input frequency). Statistics were accumulated for each measurement, and Histograms were used to show the distribution for each measurement. A Track was used to show the variation in the measurement over time, time-correlated to the original acquisition. The Measure Table statistical sdev (standard deviation) value describes the one sigma standard deviation of the measurement set comprised of a given population (num) of measurements. A Histogram with a smaller range (i.e., narrower width) and a more normal distribution equates to more measurement precision and accuracy. A Track that is vertically smaller also represents less measurement variation and more precision and accuracy.

The basic acquisition screen image is explained in detail below:





- Histograms visually describe the statistical variation of a numerical data set. The vertical scale of the
 Histogram is the number of measurement instances, and the horizontal scale is the range of measurements.
 In all comparisons, the vertical and horizontal scaling the same to make comparisons visually intuitive. A
 more normal (Gaussian) distribution with less horizontal range is an indication of a more precise
 measurement. In the lower right Histogram (of frequency), the horizontal center is fixed to the generator
 output frequency.
- **Tracks** visually describe the variation of a measurement over time, time-correlated to the originally acquired waveform upon which the measurement was based. The vertical scale of the Track is in the measurement unit value (in this case, MHz) and the horizontal scale is in time (time-correlated to the original acquisition). In the lower left Track (of frequency), the vertical center is fixed to the generator output frequency.

Data is summarized in the table below, and screen images for each case are in the table that follows:

	Amplitude		Frequency	
	2.5 GS/s	10 GS/s ESR	2.5 GS/s	10 GS/s ESR
816.2 MHz				
mean value	639.1 mV	637.8 mV	816.2 MHz	816.2 MHz
1σ (sdev)	811.2 μV	501.4 μV	3.857 MHz	0.972 MHz
506.2 MHz				
mean value	623.7 mV	622.4 mV	506.2 MHz	506.2 MHz
1σ (sdev)	939.2 μV	475.2 μV	0.645 MHz	0.637 MHz
246.2 MHz				
mean value	622.6 mV	623.5 mV	246.2 MHz	246.2 MHz
1σ (sdev)	1587 μV	729 μV	0.587 MHz	0.592 MHz





The numerical values and the Histograms demonstrate that the Amplitude measurement precision is improved by approximately a factor of two (1 σ standard deviation of the measurement at 10 GS/s is approximately half that at 2.5 GS/s). The Frequency measurement precision is improved by a factor of four in the case of the 816.2 MHz input sinewave, but not improved in the case of the lower frequency input sinewaves. Note that at the 816.2 MHz input frequency, the 10 GS/s ESR acquisition has a notably more Gaussian Histogram shape for the frequency measurement. The frequency measurement data is consistent with our predictions of significant improvement with ESR when the signal is only slightly oversampled (3x) and moderate or very little improvement once oversampling of 5x is achieved.

10 GS/s ESR (and to a lesser extent, 2.5 GS/s) also demonstrates the amplitude resolution of the HD4096 12-bit technology in the HDO6104A-MS. The Histogram is set to bin values in 1000 distinct bins and the horizontal scale of the Histogram is shown as either 1 mV/div (816.2 MHz and 506.2 MHz) or 2 mV/div (246.2 MHz). The amplitude peaks are shown ~0.33 mV apart, or roughly the real-world measurement resolution of the HD4096 technology, in this case. This could also be related to the output precision of the generator.

Input Fast Edge Example

A fast edge was generated with a Picosecond Pulse Labs TD-1110C Tunnel Diode Pulse Generator and TD1107B Tunnel Diode Pulse Head. The output rise time of this combination is 25 to 30 ps – much faster than the HDO6104A-MS can measure. The rising edge is measured on Channel 1 of the HDO (yellow trace) and the HDO is triggered on the trigger out signal from the TD-1110C (this achieves the least trigger jitter in the measurement). The measurement is relatively straightforward – capture of the rising edge with 2.5 GS/s and 10 GS/s ESR in real-time single-shot mode (1000 accumulated measurement values, but only the last acquisition displayed) and real-time sequence mode (1000 accumulated measurement values and 1000 acquisitions overlaid into one display). The overlaid sequence mode acquisitions are a good method to view a persistence-like display of 1000 waveforms and provides a simple way to view the average behavior of the acquisition.



Data is summarized in the table below, and screen images for each case are in the table that follows:

	2.5 GS/s	10 GS/s ESR
Overshoot- (Preshoot)		
mean value	0.27%	1.66%
1σ (sdev)	0.056%	.67%
Rise Time (10-90%)		
mean value	567 ps*	447 ps
1σ (sdev)	32.6 ps*	22.2 ps
Overshoot+		
mean value	7.04%	9.78%
1σ (sdev)	2.69%	1.31%

* The oscilloscope measurement algorithm detects fewer points on the edge than necessary for a precise measurement result.





The tailored frequency response of the 1 GHz HDO6104A-MS provides a near brick-wall like filter for the frequency response. This minimizes the Gibbs ears phenomenon associated with applying SinX/x on a very fast edge signal with an infinite bandwidth response. As shown in the data above, with 10 GS/s ESR:

1. The rising edge is appropriately sampled to ensure an accurate measurement result



- 2. The pre-shoot ringing is kept to a minimum
- 3. The rise time performance is improved by 120 ps with less standard deviation of the measurement value
- 4. The mean value of the pre-shoot and overshoot increase by ~1.5%, but the overall edge measurement precision is improved, especially for the overshoot measurement, and the oversampling also improves the overall signal fidelity.

Enhanced Sample Rate and Long Acquisitions

Teledyne LeCroy's X-Stream architecture permits math and measurement calculations on full acquired waveforms, regardless of length. Additionally, the X-Stream architecture permits the 10 GS/s ESR application to the full acquired waveform, regardless of length, and not just a zoomed view of a long acquisition. This is unique in the industry, and a significant advantage over the simple SinX/x implementations on short records or only in zooms that is common with lower-performance oscilloscopes. The benefit to the user is that the measurement precision enhancements of ESR are available with the full measurement set or math waveform calculated on the full record length, as shown in the input sinewave examples above.

Summary

Enhanced Sample Rate, combined with the low-noise system architecture and the tailored brick-wall frequency response in the HDO4000A, HDO6000A, HDO8000A and MDA800A oscilloscopes, dramatically improves the measurement precision when the input frequency signals are very high frequency, such as a high-frequency sinewave or fast edge. For an input sinewave, the improvement is very dramatic for both amplitude and frequency measurements when the input frequencies are very high, but even with lower input frequencies there is improvement in amplitude measurement precision. Fast edge measurement precision and displayed signal fidelity also improve dramatically with Enhanced Sample Rate, with a 120 ps reduction in the rise time measurement during real-time acquisition mode and more precision in the overshoot measurement with a reasonably minor addition to pre-shoot.

