2011 International Workshop on ADC Modelling, Testing and Data Converter Analysis and Design and IEEE 2011 ADC Forum
June 30 - July 1, 2011. Orvieto, Italy

New Features of IEEE Std 1057-2007 Digitizing Waveform Recorders

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Abstract IEEE Standard 1057-2007 Digitizing Waveform Recorders defines specifications and describes test methods for measuring the performance of electronic digitizing waveform recorders. The latest standard contains many new features not present in the original version. These include new test methods, discussions of requirements for test signals, and new ways of displaying results.

I. Introduction and History

IEEE Standard 1057-2007 Digitizing Waveform Recorders [1] defines specifications and describes test methods for measuring the performance of electronic digitizing waveform recorders, waveform analyzers, and digitizing oscilloscopes with digital outputs. The main purpose of this standard is to ensure that manufactures and users of waveform recorders have a well-defined set of specifications and test methods so they can understand, describe, and compare the performance of these recorders using a common language.

IEEE Std 1057-2007 was created by the Waveform Recorder Subcommittee of Technical Committee 10 (TC-10) Waveform Generation, Measurement, and Analysis. TC-10 is part of the IEEE Instrumentation and Measurement Society. The first full version of IEEE Std 1057-1994 was published in 1994 and was approved for a 5 year extension in 2001. Subsequent to the publishing of 1057-1994 the Waveform Recorder Subcommittee started developing a User Guide which provided in-depth information on how to use the waveform recorder standard.

After the 5 year extension was granted, the subcommittee decided to produce a revised version of 1057 that would include the material developed for the User Guide and would be compatible with IEEE Std 1241 on Analog to Digital Converters [2] and with IEEE Std 181 on Pulse Terms and Test Methods [3]. As a result an updated version of the Digitizing Waveform Recorder Standard, IEEE Std 1057-2007 was created. This document was approved in December 2007 and printed and released in summer 2008.

The new waveform recorder standard incorporates several new test methods. Some of these had been developed since the original standard was published. The new test methods include:

- Random equivalent time sampling
- Locating code transitions using a feedback loop network
- Locating code transitions using triangle waves
- Using sine fitting to compute Total Harmonic Distortion (*THD*) for noncoherently sampled data
- Out of range input impedance
- Noise Power Ratio
- Spurious noise and Spurious free dynamic range
- Signal to Noise Ratio (*SNR*) (for new noise definition)
- Low noise Effective Number of Bits (*ENOB*)

The material transferred from the User Guide and other sources include:

- Selecting frequencies used in computing *ENOB*
- Setting up the recorder
- Selecting and connecting signal sources
- Grounding and shielding

- Discrete Fourier Transform (DFT), spectral leakage, windowing, coherent sampling, aliasing
- Sine fit frequency selection
- Sine fit signal source discussion
- Cable losses
- Step response aliasing error bounds
- Step response jitter error bounds
- Phase noise
- ENOB presentation.

II. New Test Methods

A few of the new test methods are described below.

A. Random equivalent time sampling

Equivalent time sampling allows the user to reduce aliasing errors in step response measurements by producing a more finely sampled waveform by combining samples from several instances of an input test pulse from either within a single recorded pulse waveform or from multiple recordings of the pulse.

The random equivalent-time sampling test method has the advantage of not requiring accurate control of the pulse repetition frequency. Instead, it requires taking many, typically 100 or more, records of data. Application of the method requires that the input signal, as recorded on the waveform recorder, have a level, v_0, with the property that the signal is increasing during a time interval T after crossing v_0, where T is the reciprocal of the sampling frequency. The method also requires that the input pulse occur at times that are random relative to the clock of the waveform recorder. With these conditions met, do the following:

a) Collect K records of data $x_k[i]$, for $k = 1…K$ and $i = 1…M$.
b) Let i_0 be the first index in each record for which the recorded signal value is equal to or greater than v_0. It is assumed that i_0 is the same for each record. If it is not, the beginnings of some records shall be truncated to make it the same.
c) Reorder the numbering of the records so that $x_{k+1}[i_0] \geq x_k[i_0]$. Because the signal is increasing in this interval, the records are now ordered in time.
d) Construct a single record of length $K \times M$ by taking the first point from each record (in the order of the records), followed by the second point of each record, then the third, and so on.
e) Treat this larger record as if it had a sampling frequency of $K \times f_s$, where f_s is the sampling frequency of the individual records.

This approach eliminates the aliasing error and replaces it with a different error—the error due to the fact that the records are not truly uniformly distributed in time, but differ randomly from a uniform distribution.[4]

B. Using sine fitting to compute THD for noncoherently sampled data

This method uses sine fitting rather than the DFT to determine the input signal and harmonic amplitudes. It is somewhat more computationally intensive than using fast algorithms for the DFT, but that is usually not a problem with today's computers. Its advantage over other methods is that it is less sensitive to noise and more thoroughly eliminates spectral leakage.

Apply a sinewave input signal to the recorder. To maximize accuracy, each data record shall be truncated so that it has approximately an integer number of cycles of the input signal. Perform either a three-parameter or four-parameter sine fit to the data to determine the input amplitude A_1 and, if using a four-parameter fit, the input frequency f_i. Calculate the residuals.

For each harmonic number, h, between 2 and N_H, perform a three-parameter sine fit to the residuals with a frequency of hf_i to determine the harmonic amplitude A_h.

Use the values of THE and A_{rms} computed in Equation (3) to calculate THD using Equation (4).

$$THE = \frac{1}{2}\sum_{h=2}^{N_H} A_h^2, \quad \text{and} \quad A_{rms} = \frac{A_1}{\sqrt{2}} \tag{3}$$

$$THD = \frac{\sqrt{THE}}{A_{rms}} \tag{4}$$

The reason for truncating the records to an approximate integer number of cycles is to allow each harmonic amplitude to be accurately determined separately. If multiple records are used, the value for the A_i used in Equation (3) shall be the average of the values from the individual records.

C. Out of range input impedance

Sometimes waveform recorders are used in applications where the input signal can be greater than the Full Scale Range (FSR) of the recorder but within the maximum safe operating level. The input impedance for signal levels greater than the FSR can be different from the input impedance for signals within the FSR. There can be a problem with reflected signals when the impedance for out-of-range signals no longer matches the input transmission line impedance.

Time Domain Reflectometers normally do not have sufficient output levels to make this measurement. Thus to measure this impedance, arrange a network similar to that shown in Figure 2. The key components are a pulse generator, an isolating coupler, well-characterized transmission lines, and a second waveform recorder.

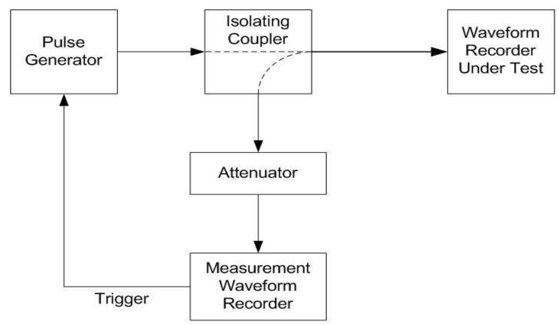

Figure 2—Test setup for measuring input impedance for out-of-range signals by time-domain reflectometry

The pulse source shall have a fast leading transition duration and a flat top. The duration of the pulse shall be less than the two-way transit time between the recorder under test and the recorder being used to make the measurement.

Adjust the output level of the pulse generator to the desired level. Again the level shall be less than the maximum safe operating level of the recorder under test. Measure the test pulse level at the recorder input using a calibrated attenuator. Then record the pulse reflected from the input of the recorder under test. Compute the reflection coefficient, ρ, from Equation (5).

$$\rho = \frac{V_{ref}}{V_{in}} \tag{5}$$

where V_{in} is the amplitude of the incident pulse at the recorder under test and V_{ref} is the amplitude of the reflected pulse corrected for losses in the coupler and attenuation. Use Equation (6) to compute the desired input impedance.

$$Z_i = Z_0 \frac{1+\rho}{1-\rho} \tag{6}$$

where Z_i is the recorder impedance and Z_0 is the input transmission line impedance

D. Low noise *ENOB*

When the noise level of the waveform recorder is low enough so that the quantization error is significant compared to the random noise, the *ENOB* calculation can be affected by the amplitude and offset of the applied signal in an undesirable way. This effect can be greatly reduced by changing the calculation as follows:

a) Apply the test sine wave and collect a record of data of M samples $x(i)$.
b) Let x_0 be the average of the maximum and minimum data values collected.
c) Form a histogram of the values of the data samples collected.
d) Let $x_{\max}$ be the data value greater than x_0 with the highest histogram count.
e) Let $x_{\min}$ be the data value less than x_0 with the highest histogram count.
f) Remove from the data record any values with $x \geq x_{\max}$ or with $x \leq x_{\min}$.
g) Fit a sinewave to the reduced data set (which will not have uniform time spacing).

h) Calculate noise and distortion (*NAD*)

$$NAD = \left[\frac{1}{M} \sum_{n=1}^{M} (x[n] - x'[n])^2 \right]^{1/2}$$

i) Calculate

$$ENOB = \log_2 \left(\frac{\text{FullScaleRange}}{NAD\sqrt{12}} \right)$$

This procedure eliminates the data near the peaks of the sinewave and is illustrated on Figure 3**Figure** , where the omitted code bins are cross-hatched.

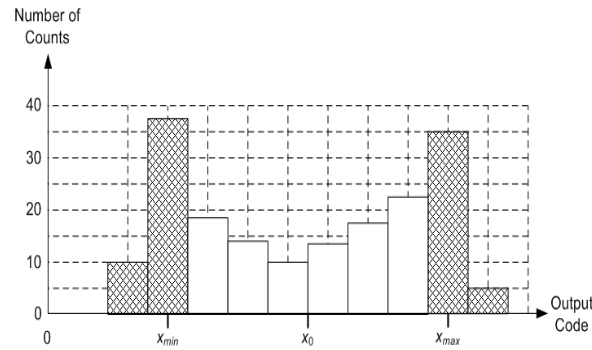

Figure 3—Illustration of the use of the histogram to remove the sinewave peaks

E. Presentation of *ENOB* residuals

Much can be learned by viewing the residuals of a particular sinewave test. In particular, a user can determine the sources of error that are responsible for an *ENOB* that is less than expected. The residuals can be viewed in either the time domain or the frequency domain, and it is most informative to look at both.. The time-domain presentation is called the *modulo time plot* [5]. The frequency domain presentation is called the *power spectral distribution* (PSD) [6]. Typical displays of each are shown in Figure 4.

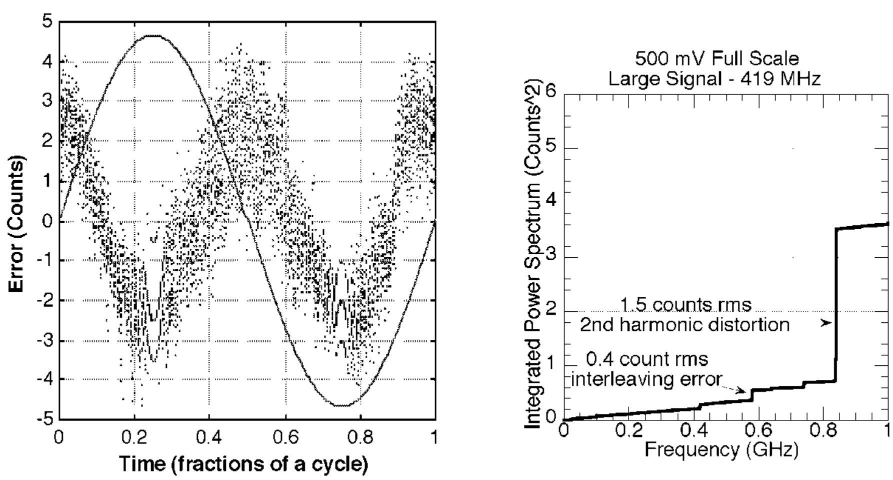

Figure 4—Example plots of sine fit residuals
(The left side shows a modulo time plot, and the right side shows a PSD. Both represent the same data.)

To construct the modulo time plot, from the time coordinate of each data point in the record is calculated the phase value, between 0 and 2π, relative to the input sinewave, as determined by the fit. This phase angle, divided by 2π, is shown on the horizontal axis of the plot. The residual value is shown on the vertical axis. On this particular plot, the units are Least Significant Bits (LSB), but any other units, such as volts, might be used. A scaled replica of the fitted input signal is displayed as the continuous curve. On this plot, a user can clearly see 2^{nd} harmonic distortion of approximately ±2.5 LSB superimposed on random noise of about ±2 LSB. The plot clearly displays the phase relationship of the harmonic to the input signal, having its negative peaks at the peaks of the input signal and having its positive peaks at the zero crossings.

The PSD is the integral of the power spectral density. Its units are the square of input signal amplitude units, or, as in the case of Figure 4, LSBs. A jump in the PSD represents energy concentrated at a single frequency. A straight line represents white noise with density equal to the slope of the line. [7]

The interpretation of a PSD will be illustrated using Figure 4. At 840 MHz, the 2^{nd} harmonic of the input signal, the PSD jumps from 0.8 LSB^2 to 3.6 LSB^2, an increase of 2.8 LSB^2, or (taking the square root) 1.7 LSB. This value is the rms value of the sinewave component at 840 MHz. The peak value of 2.4 is obtained by multiplying this value by $\sqrt{2}$. This result agrees with the 2nd harmonic observed in the modulo time plot.

These data were taken from a waveform recorder that consists of two interleaved recorders with sampling rates of 1 GSa/s, giving a combined sampling rate of 2 GSa/s. An interleaving error would occur at 581 MHz, which is the difference between the applied frequency of 419 MHz and the sampling frequency of 1 GHz. A jump of 0.2 LSB^2 is seen at this frequency, which corresponds to an rms error of $\sqrt{0.2} \cong 0.4\text{LSB}$. This jump is masked by other errors in the modulo time plot. The primary sources of interleaving error are differences in gain, offset, and delay between the interleaved channels.

II. Requirements for Signal Sources

This section presents selected topics defining requirements for selecting and evaluating test signal sources.

A. Sine fit frequency selection

The original 1057 included guidance for fine-scale frequency selection for Effective Number of Bits (*ENOB*) testing. The requirement is that a frequency is chosen to optimize the spacing between the samples recorded.

The revised 1057-2007 also includes guidance for selecting frequencies on medium-scales and coarse-scales. On the medium scale, a user selects frequencies to cause errors from different sources to occur at different frequencies. For example, for a recorder with a sampling rate of 2 GSa/s, if a frequency of 400 MHz were selected, 3rd harmonic distortion would be at a frequency of 1200 MHz. Since this is above the Nyquist frequency of 1000 MHz, it would be aliased down to 800 MHz. This is the same frequency as 2nd harmonic distortion; therefore, the two would be indistinguishable. With a frequency of 420 MHz, the 2nd harmonic is at 840 MHz while the 3rd harmonic aliases down to 760 MHz, allowing the user to distinguish between the two.

In the coarse-scale frequency selection, a user normally selects nice round numbers (e.g., 250 MHz, 500 MHz). These round numbers then have to be modified to separate the errors from different harmonics and from interleaving. It is important to take aliasing into account in this step. The resulting frequencies must then be modified a second time to meet the criteria for fine-scale frequency selection.

Select several test frequencies that span the range of major expected frequency components in the final-use input signal. It is important that the highest frequency signal have at least as large of a maximum slew rate (derivative with respect to time) as the maximum slew rate of final-use input signal.

The test frequencies can be categorized as low, medium, and high. Low frequencies are low enough to not cause significant dynamic errors (e.g., frequency dependant distortion and time jitter) in the waveform recorder. Frequencies less than a few percent of the analog bandwidth are generally safe to consider low. Medium frequencies are those high enough to cause some dynamic effects, but still well below the analog bandwidth. These will generally be in the range of 10% to 30% of the analog bandwidth. High frequencies are near enough to the analog bandwidth that the amplitude roll-off is a significant factor. The test frequencies shall include at least one frequency in each category.

B. Sine-fitting signal source discussion

A number of tests in the 1057standard use sinewave sources, and the analyses of the test results assume that the signal is a pure sinewave. The new version of 1057 includes a description of how the impurities of a sinewave are quantified, how they are measured, and how users can control them.

The impurities in a sinewave are described as follows:

- Harmonic distortion - the presence of sinusoidal signals at frequencies that are integer multiples of the signal frequency.
- Spurious components - sinusoidal signals at frequencies that are not integer multiples of the signal frequency
- Wide-band noise - random signal that is spread over a large frequency range
- Amplitude modulation - amplitude variations with time when the signal is a pure sinusoid
- Phase modulation – nonrandom variation of phase with time
- Phase noise – random variation of phase with time

One approach to dealing with the problem of potential sinewave impurities is to assume that they are negligible and proceed with sine-fitting tests. The results of the sine-fitting tests can then be used to determine potential problems with the signal source. This approach requires performing the sine-fit test with two different amplitudes of the same frequency. Observe the harmonic distortion, spurious components, and wideband noise in the residuals. If each is negligible, it is reasonable to assume that the corresponding impurity in the signal source is negligible. If the impurity remains the same for the two different amplitudes relative to the signal, the signal source should be tested for that impurity. Exceptions can be spurious signals caused by internal clocks and interleaving.

The spectrum of the residuals will have a peak at the signal frequency if either amplitude or phase modulation is significant. The two types of modulation can be distinguished by looking at the modulo time plot of the residuals described above. Amplitude modulation will appear as random noise multiplied by a sinusoidal envelope at the frequency of, and in phase with, the input signal. Phase modulation is the same except that the envelope is 90° out of phase with the signal. The observed phase modulation will be the difference between that of the signal and that of the clock of the waveform recorder. One way to discriminate between the two is to simultaneously test two waveform recorders that have independent clocks with the same signal. By correlating the residuals from the two waveform recorders, a user can determine how much of the phase modulation is due to the signal and how much is due to the waveform recorder clocks.

III. New Definition of Noise

The definition of noise was changed to exclude *THD*. This was done to support the measurement of the parameter Signal to Noise and Distortion (*SINAD*) which is commonly used in digitizer specifications. Noise is now defined as follows:

> Noise is any deviation between the output signal (converted to input units) and the input signal except deviations caused by linear time invariant system response (gain and phase shift), a dc level shift, *THD*, or an error in the sample rate.

IV. Conclusion

The TC-10 Waveform Recorder subcommitte feels that the newly revised version of 1057 is a significant improvement over the original version.

References

[1] IEEE Std 1057-2007, Digitizing Waveform Recorders.

[2] IEEE Std 1241-2000, Standard for Terminology and Test Methods for Analog-to-Digital Converters.

[3] IEEE Std 181-2003, IEEE Standard on Transitions, Pulses, and Related Waveforms.

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