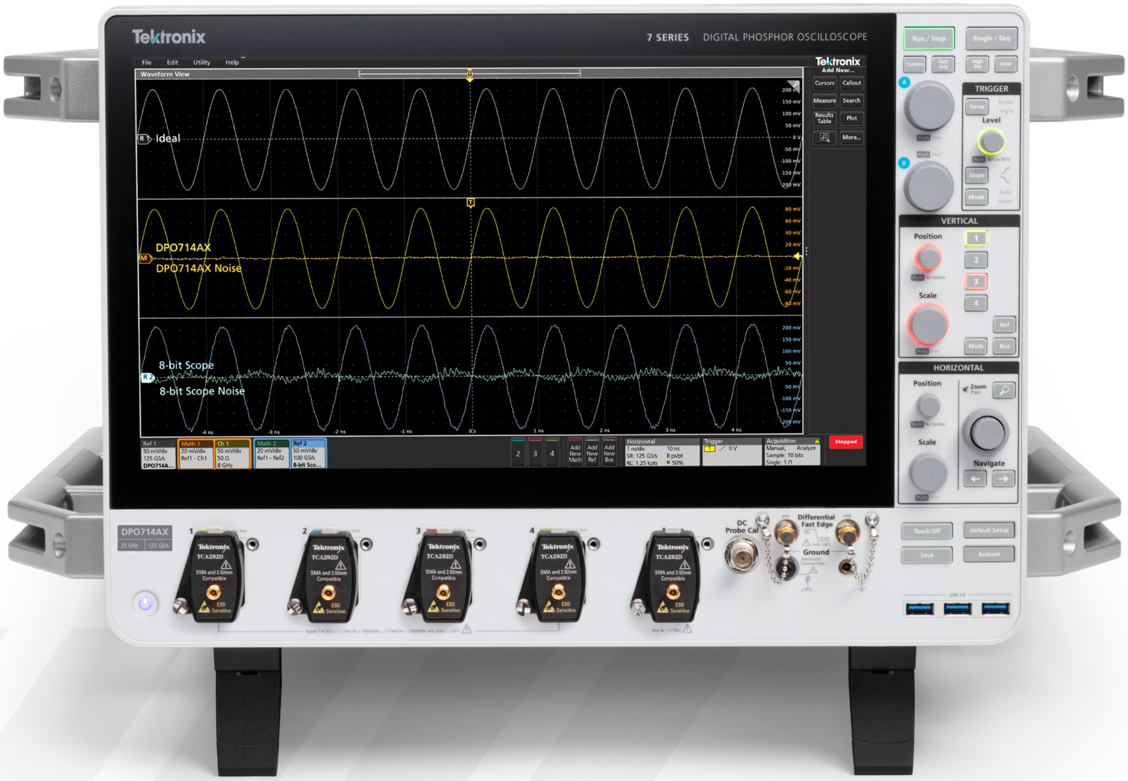
Tektronix

Understanding ENOB (Effective Number of Bits) in ADCs: Key Metrics for Dynamic Performance of Digitizing Instruments

APPLICATION NOTE



Effective Bits versus Resolution

Increasing measurement accuracy demands have made it extremely difficult to understand how an oscilloscope or digitizer impacts the results. Modern instruments have datasheets with dozens of pages of specifications that could be relevant to a measurement, and these specifications interact with each other in complex ways. For example, noise creates jitter and jitter can increase noise. The effective number of bits (ENOB) is a figure of merit for ADCs, digitizers and oscilloscopes that captures most of the errors caused by acquiring a signal. ENOB was developed by the IEEE to capture impairments like random noise, sample jitter and non-linearity in a single figure of merit to describe the real performance of an ADC or instrument.

Some vendors of data converters and instruments will feature the resolution of their products. The resolution of a product creates a limit to the noise floor due to the inability to measure within ±½ of the least significant bit.

This error is called quantization noise and limits the signal-to-noise ratio to 6.02N + 1.76dB where N is the resolution. This quantization noise is the basis for ENOB. The effective number of bits is the measured performance of an ADC or instrument that is equivalent to an ADC with quantization noise as the only impairment or performance limit. A 12-bit oscilloscope that specifies an ENOB of 8 is the equivalent of an ideal 8-bit ADC.

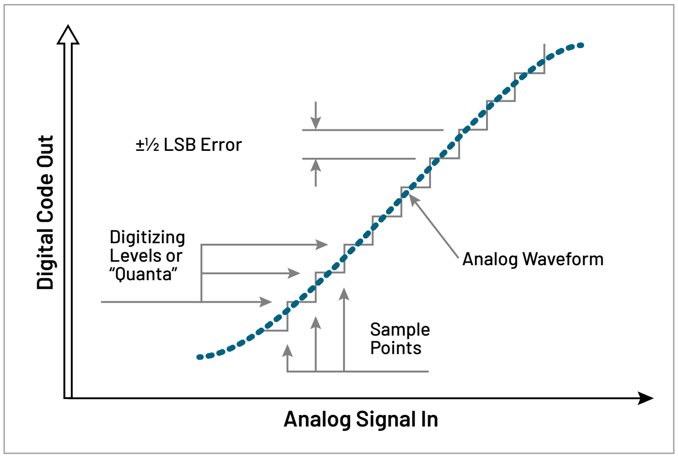


Figure 1 - Quantizing error.

Resolution or Effective Bits (N)	Quantizing Levels	Signal-to-Noise Ratio in dB (6.02N+1.76dB)
4	16	25.84
6	64	37.88
8	256	49.92
10	1,024	61.96
12	4,096	74.00
14	16,384	86.04
16	65,536	98.08

Table 1 - Digitizer is ±½ LSB of error.

Figure 2 shows the ENOB for four oscilloscopes currently on the market. There are several important takeaways from this plot. While the differences between these specifications may seem small, an improvement of 1 effective bit is twice as accurate when acquiring. Also, the 12-bit oscilloscope and the 8-bit oscilloscope are close in performance at high frequency, and the 8-bit instrument is significantly better than the 12-bit instrument at lower frequencies.

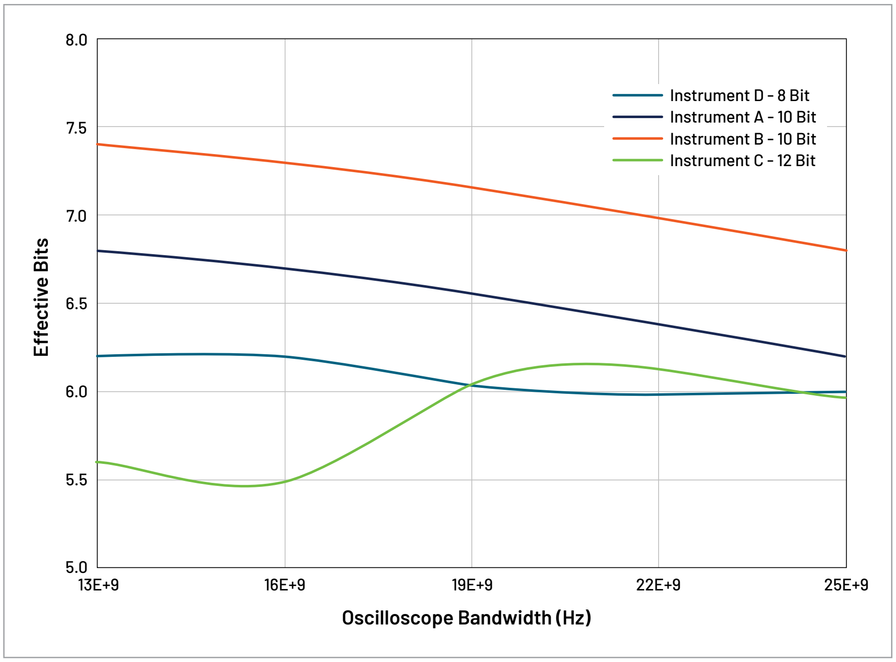


Figure 2 – ENOB comparison of high bandwidth oscilloscopes.

Only two of these instruments get close enough to 8 effective bits to make use of the lower quantization error of a > 8-bit converter. If the highest performance 10-bit instrument had been designed with an 8-bit converter with everything else the same in the system, the ENOB at 13 GHz would only be 7.1 effective bits instead of the specified 7.4 bits. The other 10-bit oscilloscope would drop from 6.8 effective bits at 13 GHz to 6.7 effective bits. At higher instrument bandwidths and on the 12-bit instrument, the higher resolution doesn't provide anything beyond the 8-bit converter. The number of ADC bits doesn't always translate to effective bits or better measurements.

ENOB performance limitations are also important to understand for automation. If the instrument and settings used don't provide an ENOB of 7 or above, then there is little or no benefit in saving or querying 16-bit data. Saving processing time and disk space is more advantageous in these cases.

ADC versus Instrument Resolution

ENOB is a specification used for Analog to Digital Converters as well as instruments that use ADCs. For instruments, the resolution specification also needs to consider the width of the data all of the way into the acquisition memory as well as to the display and measurement system.

The DPO714AX is a good example of this difference between ADC and instrument resolution. The Tek079 ADC developed by Tektronix is a 12-bit 62.5 GSa/s ADC. At the maximum product bandwidth of 25 GHz, this is an ENOB of 6.5 which equates to an SNR of 40.9 dB. Limiting this instrument all the way to 1 GHz provides an ENOB of 7.6 or an SNR equivalent of 47.5 dB. The DPO714AX stores data as 10 bits since any additional bits would just bloat logic and fill 1/6th of the memory with unnecessary bits. The calculation in **Table 1** shows the SNR would need to be close to 62 dB for this not to be a waste.

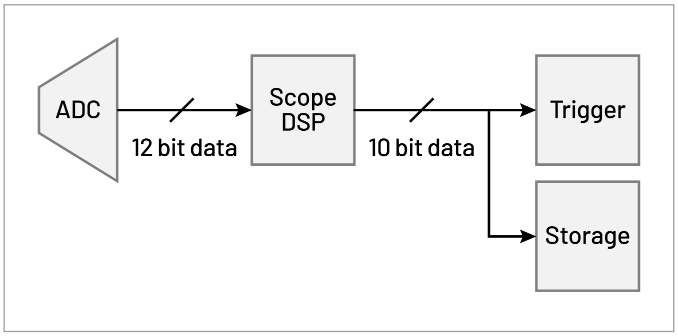


Figure 3 - DPO714AX dataflow.

Effective Number of Bits Derivation

Decline in digitizer performance is specified as an increasing level of noise on the digitized signal. "Noise", here, refers to any random or pseudorandom 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),

$$SNR = \frac{rms\_signal}{rms\_error} \qquad \text{[Equation 1.]}$$

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,

$$EB = \log_2(SNR) - \tfrac{1}{2}\log_2(1.5) - \log_2\left(\frac{A}{FS}\right) \qquad \text{[Equation 2.]}$$

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,

$$EB = N - \log_2\left(\frac{rms\_error}{ideal\_quantization\_error}\right) \qquad \text{[Equation 3.]}$$

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

$$EB = \log_2(rms\_error * \frac{\sqrt{12}}{FS}) \qquad \text{[Equation 4.]}$$

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 minimum rms error for an N-bit digitizer is the ideal quantization error. 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 $FS/(2^n \sqrt{12})$ 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.

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 ± ½ LSB (least significant bit). As illustrated in **Figure 1** 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.)
- And other error sources such as trigger jitter

Aside from DC offset and gain error, ENOB captures all these errors into a single figure of merit.

Figure 4 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.

On the other hand, aperture uncertainty and time base inaccuracies are phenomena associated with the sampling process that accompany waveform digitizing. The basic concept of aperture uncertainty is illustrated in **Figure 5**.

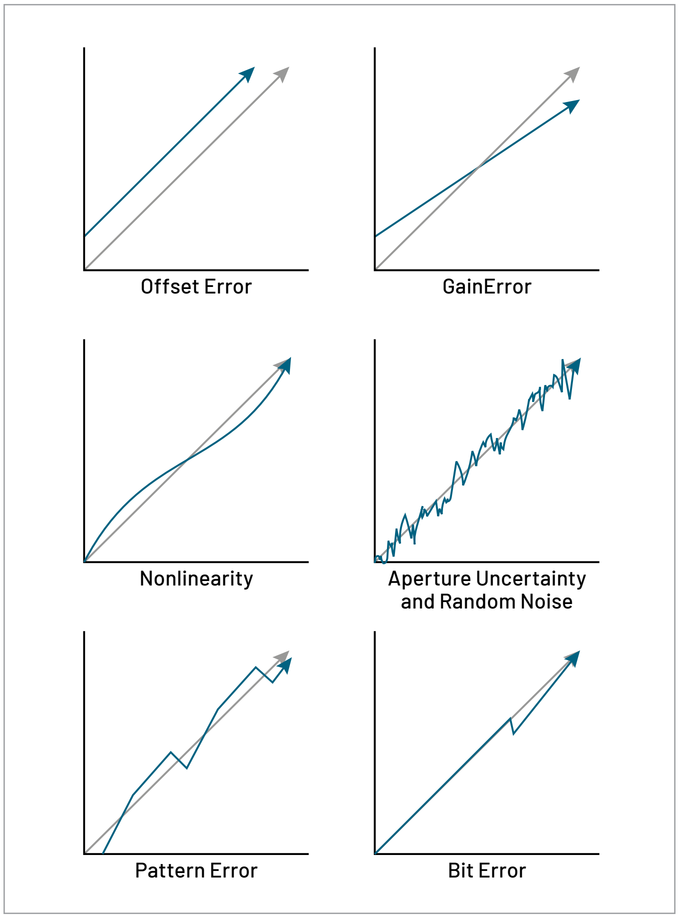


Figure 4 - Errors associated with non-ideal digitizing.

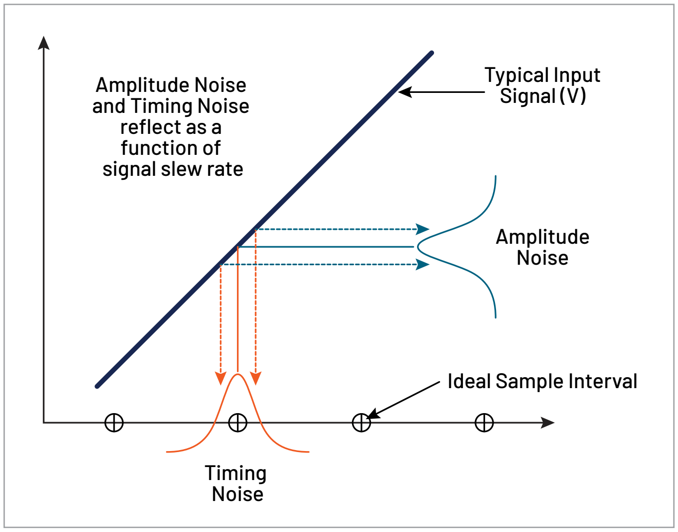


Figure 5 - Aperture uncertainty, or sample jitter.

The important thing to note from **Figure 5** is that aperture uncertainty results in an amplitude error and the error magnitude is slope dependent. 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.

$$f = \frac{1}{\sqrt{6} \cdot \pi \cdot \Delta t \cdot 2^N} \qquad \text{[Equation 5.]}$$

In Equation 5, 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 5.

There is, however, a way around the necessary decrease in bits, n, for increasing frequency. This relates back to the concepts illustrated in **Figure 5**. 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 depend 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 acquisition system.

The Effective Bits Measurement Process

Beyond the error sources and considerations mentioned thus far, there are still other possible sources of digitizing error. For example, in high-speed real-time digitizing without a sample-and-hold or track-and-hold, the least significant bits must change at extremely high rates in order to follow a fast-changing signal. This puts high bandwidth requirements on the data lines and buffer inputs for these lesser bits. If these bandwidth requirements are not met, fast changing lesser bits will be "dropped", leaving the digitizer with a lower effective bits performance. This is, of course, in addition to the many other possible error sources prior to and after the digitizing device.

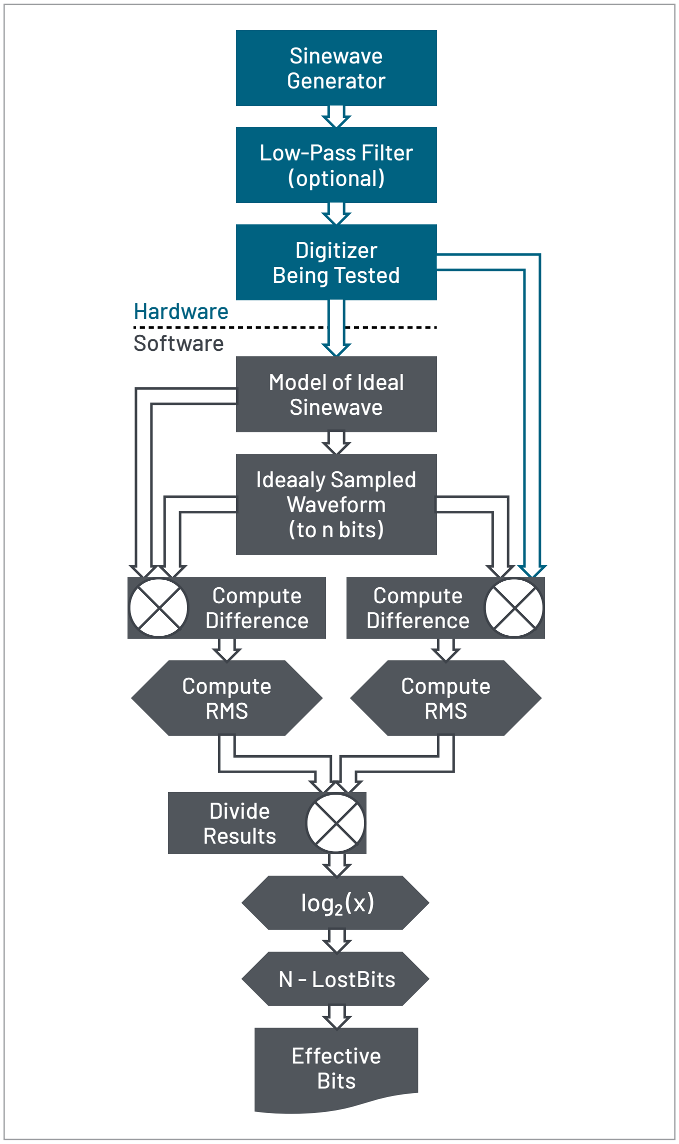


Figure 6 - The process of effective bits measurements.

Rather than trying to distinguish and measure each individual error source within a digitizing system, it is easier to measure overall performance. In other words, given an ideal input signal, what are the overall error contributions of the digitizing system in the output signal? A good place to start is determining the digitizing system's SNR and the resulting effective bits as defined by Equations 2, 3 or 4. This provides an easily understood and universal figure of merit for comparisons.

The basic test process is illustrated in **Figure 6**. It involves applying a known, high-quality signal to the digitizer and then computer analyzing the digitized waveform. A sine wave is used as the test signal because high-quality sine waves are relatively easy to generate and characterize. The general test requires that the sine wave generator's performance must significantly exceed that of the digitizer under test, preferably by 10 dB or more.

Otherwise, the test will not be able to distinguish digitizing errors from signal source errors. It may be necessary to add filters to the source in order to reduce source harmonics to levels significantly below what might be expected from the digitizer under test. To obtain an effective bits number, a perfect (idealized) sine wave is computed and fitted to the digitized sine wave. This perfect sine wave is described by,

$$A \cdot \sin(2\pi f t + \Theta) + C \qquad \text{[Equation 6.]}$$

where A is the sine wave's amplitude, f is its frequency, Θ is phase, t is time and C is DC offset. The actual process of fitting this sine wave can use any of several software algorithm variations designed to converge quickly on an optimum result. This result is considered to be a description of the analog input to the digitizer. It should be noted, however, that because the analog signal parameters are computed from the digitizer's output, DC offset, gain, phase and frequency errors are not included. These excluded errors need to be measured by separate tests, such as a histogram test or other test appropriate to the specific error of interest. After computing a model of the ideal input sine wave, further computations are done to determine ideal sampling and digitizing of the sine wave. This simulates what the N-bit digitizer under test would produce if it were an ideal N-bit digitizer. The difference between the computed ideal sine wave and the perfectly sampled and digitized version is then computed. The rms value of this provides the ideal quantization error used in Equation 3. The rms error value used in the effective bits equations (3 and 4) is obtained by subtracting the ideal sine wave from the actual digitized sine wave and finding the rms value of the result. Or, as an alternative, the rms value of the signal and the rms error can be found and used to compute SNR for use in Equation 2.

The final computation (using Equations 2, 3 or 4) results in an effective bits number for the digitizer. By keeping input signal amplitude constant for various frequencies, further effective bits numbers can be computed for the subject digitizer or digitizing system. These numbers can then be plotted against frequency to obtain a digitizer performance curve such as illustrated in **Figure 2**.

Effective bits lumps many of the key digitizer system errors into a figure of merit that is easy to understand and use in comparisons. As noted previously, however, effective bits does depend on the input signal's percent of full-scale digitizer amplitude. A digitizer should be tested as near to full scale as possible without noise or frequency response causing clipping on the pre-amplifier. A good value to use is typically 90% of full scale. Testing at lower percentages of signal amplitude will reduce the impact of sample jitter and harmonic distortion on ENOB.

Caution also needs to be exercised in selecting frequencies for developing an effective-bits plot. If the test signal frequency is harmonically related to the digitizer's sampling rate, there is a possibility of beat frequencies interfering with the test results. Consequently, it is best to ensure that the test signal is asynchronous with the digitizer's sampling.

Digitizer triggering is still another area that needs caution. In general, capture of the test signal should be done in a single-shot mode. This eliminates the harmful effects of trigger jitter from the effective bits measurement and concentrates the evaluation on the digitizer itself.

However, to operate at higher bandwidths, many digitizers must use repetitive triggering and equivalent time sampling to build a full sample complement over many repetitions of the input waveform. Trigger jitter and long-term drift effects can increase the noise level associated with equivalent-time digitizers. This is most often dealt with by using signal averaging to decrease noise, thus raising the effective bits of the digitizer. If signal averaging is going to be used in conjunction with effective bits testing, the amount of signal averaging used should be stated with the effective bits results. Also, you should be aware that the built-in signal averaging used in some digitizing instruments may employ computing at a higher resolution than the digitizer itself.

For example, an 8-bit digitizing instrument may use internal 16-bit computations for signal averaging. This high-resolution averaging can make an 8-bit digitizer appear to be a 10- or 11-bit digitizer. This, of course, can result in higher effective bits testing results when averaged input signals are used. In general, it's rarely valid to compare digitizers that use signal averaging to digitizers that capture signals on a single-shot basis, unless both digitizers can be set to a common operating mode for the purposes of "fair and equal" comparisons.

To see a quick method for approximating ENOB without having to perform the complex sine fit, watch this video: "Maximize Test Margins with Lowest Noise and Highest Effective Number of Bits (ENOB)".

Other Figures of Merit

Many communications standards and applications have recognized the need for a figure of merit that encompasses multiple system errors, including the digitizer, and their complex interactions. Channel Operating Margin (COM) and Signal-to-noise and Distortion Ratio (SNDR) are used by wired communication standards as a single number that reflects the performance of the system. Transmitter Dispersion and Eye Closure (TDEC) is a similar figure of merit used for optical standards. These industry figures of merit use a pulse to characterize the system instead of the swept sinewave used to characterize ENOB. Even though the signal input is different, these figures of merit capture a lot of the same errors and in most cases will correlate well with ENOB.

In some optical and wireless standards that use Quadrature Amplitude Modulation (QAM), Error Vector Magnitude (EVM) is a measure of the vector to the ideal constellation point placement compared to the vector at the actual point. EVM does capture digitizing errors like gain and frequency response flatness that aren't accounted for in ENOB. This makes ENOB less predictive of EVM performance.

Figure of Merit	Gain	Offset	Flatness	Random Noise	Spurious Noise	Harmonics	Interleave	Jitter	Phase
ENOB	✗	✗	✗	✓	✓	✓	✓	✓	✗
EVM	✓	✗	✓	✓	✓	✓	✓	✓	✓
COM/SNDR	✗	✗	✓	✓	✓	✓	✓	✓	✓
TDEC/TDECQ	✗	✗	✓	✓	✓	✓	✓	✓	✓

Table 2 -Figures of Merit Errors Included.

These industry tests have a lot of overlap in what digitizing errors are included. **Table 2** shows which errors are accounted for in each figure of merit. Depending on the application, ENOB can be a good stand in for these other figures of merit as test instruments and ADCs will specify ENOB and may not have a relevant specification for other figures of merit.

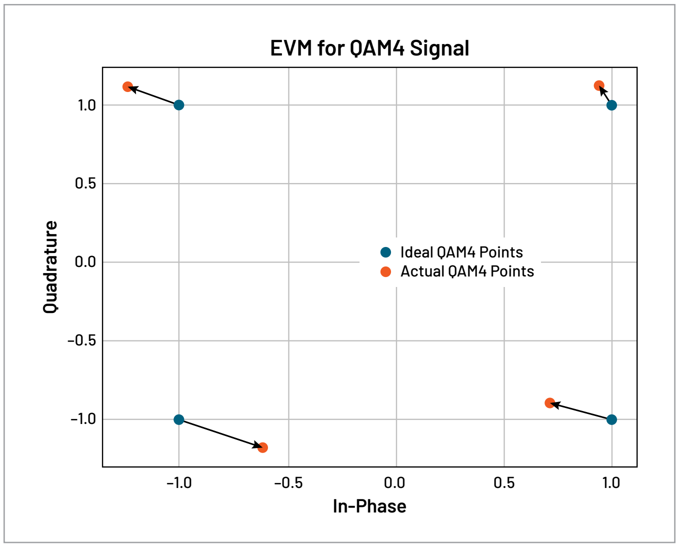


Figure 7 - QAM4 Error Vectors.

Bibliography

Bednarek, C., "Dynamic Characterization of A/D Converters," *Handshake*, Tektronix, Inc., Summer 1988 pp. 9-11.

Bird, S.C., and J.A. Folchi, "Timebase Requirements for a Waveform Recorder," *Hewlett-Packard Journal*, Nov. 1982, pp. 29-34.

DeWitt, L., "Dynamic Testing Reveals overall Digitizer Performance," *Handshake*, Tektronix, Inc. Spring/Summer 1980, pp. 9-12.

Dornberg, J., H.S. Lee, and D. Hodges, "Full Speed Testing of A/D Converters," *IEEE Journal of Solid State Circuits*, Vol. SC-19, No. 6, Dec. 1984, pp. 820-827.

Harris, F.J., "On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform," *Proceedings of the IEEE*, Vol. 66, No. 1, Jan 1978, pp. 51-83.

IEEE Standard 1057, Standard for Digitizing Waveform Instruments.

Jenq, Y.C., "Digital Spectra of Non-Uniformly Sampled Signals, Theory and Applications, Part I Fundamentals of Waveform Digitizers," *IEEE Transactions on Instrumentation and Measurement*, Vol IM-37, No. 2, June 1988.

Jenq, Y.C., and P. Crosby, "Sinewave Parameter Estimation Algorithm with Application to Waveform Digitizer Effective Bits Measurement," *IEEE IMTC/88*, San Diego, CA., April 19-22, 1988.

Jenq, Y.C., "Measuring Harmonic Distortion and Noise Floor of an A/D Converter Using Spectral Averaging," *IEEE IMTC/88*, San Diego, CA., April 19-22, 1988.

Jenq, Y.C., "Asynchronous Dynamic Testing of A/D Converters," *Handshake*, Tektronix, Inc., Summer 1988, pp. 4-7.

Peets, B.E., A.S. Muto, and J.M. Neil, "Measuring Waveform Recorder Performance," *Hewlett-Packard Journal*, Nov. 1982, pp. 21-29.

Ramreriz, R.W., "Digitizer Specifications and Their Applications to Waveforms," *Electronics Test*, Sept. 1981.

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