Waveform Recorder Testing: IEEE Standard 1057 and You

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Waveform recorders and digital oscilloscopes have become indispensable tools for electronics measurement and diagnosis. However, because they are complex, sophisticated instruments, it is often difficult to know how to specify and test their performance. To address these problems, the Waveform Measurements and Analysis Committee developed an IEEE Trial-Use Standard that was first issued in 1989. Since then, The Standard for Digitizing Waveform Recorders, IEEE Standard 1057, has been substantially revised; the new edition was recently approved for publication as a full-use standard and is now in print. This paper will present an overview of the new standard, with an emphasis on how it can help the bewildered engineer understand and test the latest digitizers.

INTRODUCTION

When contemplating testing a waveform recorder, it is important to understand how the waveform recorder will be used. That is, what is the nature of the data to be acquired? Further, what characteristics of the data are important to the experiment? Even if the answers to these questions are approximate (e.g., the pulse will probably rise from 10% to 90% of its final level in 1 - 10 ns), they will still serve to focus the waveform testing.

The next task is to translate the measurement requirements into a set of parameters which commonly specify waveform recorders. For the sake of discussion, assume we are to acquire a pulse. It may be necessary to estimate the highest frequency components of the anticipated data in order to determine the minimum bandwidth required of the recorder. The bandwidth alone may determine the minimum sampling rate, or other factors, such as a minimum number of points on the rising edge, may also be considered. If the shape of the pulse is critical, then the recorders' harmonic-distortion and phase linearity can be specified. If there is a possibility of a greater-than-full-scale signal, then overvoltage recovery should be specified.

With an initial set of characterizing parameters in hand, the next step is to compare these parameters to the manufacturer's published specification. Does the published specification appear to cover your requirement? Does the manufacturer specify all the parameters of interest to you? Usually, some form of recorder testing will be indicated either to verify specified performance or to establish characteristics not specified by the manufacturer.

At this point, it may be helpful to consult the IEEE Standard for Digitizing Waveform Recorders (IEEE Std. 1057-94). The standard was prepared to assist both users and manufacturers in specifying and testing waveform recorders. This standard was first released as a trial-use standard in 1989. Since then it has been substantially revised and reissued as a full-use standard. The Standard's table of contents, which has been reprinted as Fig. 1, indicates the parameters included and suitable test methods.

CLAUSE PAGE

Fig. 1. Standard 1057-94 Table of Contents.

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It is important to note that Std. 1057 deals with recorders (and digital oscilloscopes) which have digital outputs. (Accordingly, much of Std. 1057 is appropriate for specifying and testing an analog-to-digital converter (ADC). The recorders are assumed to incorporate both signal sampling and quantization. The nature of these processes substantially govern the operation of the recorder, the output data available, and the nature of its errors. For example, Fig. 2 shows the nature of quantization error in the vicinity of a sharp peak. A cusp of a sine wave is shown both ideally and as quantized by a recorder. Note that the recorder only approximates the true waveform.

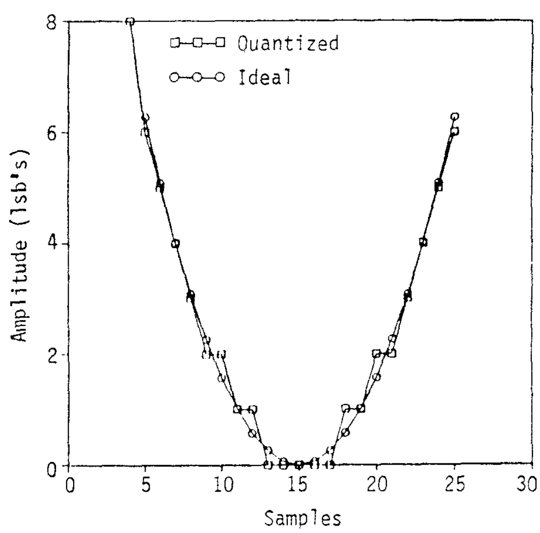

Fig. 2. Quantization Error at a Data Peak.

Again, for the sake of discussion, let us assume that the parameters of interest include:

- Signal-to-Noise Ratio (SNR) and Effective Bits
- Transition Duration
- Gain Flatness
- Nonlinear Phase Error
- Total Harmonic Distortion
- Overvoltage Recovery

Upon reading about each parameter, it will become apparent that all of this data can be obtained from two general tests (sine wave response and step response) plus a special test for overvoltage recovery.

MATRIX SINE FITTING ALGORITHMS

Many of the tests prescribed in Std. 1057 call for least-squares sine wave curve fitting of recorded data from sine wave tests. The curve fitting process presented in Standard 1057 as Clause 4.1.3 is variously used to estimate recorder offset and gain errors, time base errors (e.g., fixed error in sample time), signal-to-noise ratio, and effective bits, among other parameters. Because sine functions are nonlinear in one of the four defining parameters (i.e., frequency), closed form least-squares solutions don't exist, so iterative search techniques are often used. For this reason, it is especially important that "standardized" algorithms be defined. Two new sine fitting algorithms using matrix operations have been included in the full-use standard, to augment the 3- and 4-parameter fitting algorithms proposed in the earlier trial-use version. The 3-parameter, fixed frequency methods used in each version are based on the closed form, classical least-squares approach, and consequently, the solutions given by both are identical. The advantage of the new matrix method is the relative ease with which it is coded, compared to the earlier method.

The new 4-parameter matrix method also gives substantially identical solutions, provided that the initial starting conditions are the same. In the past, the 4-parameter fit did not always converge if the initial estimates were not very close to the actual values. The full-use standard recommends using the 3-parameter fit to derive the initial estimates for the 4-parameter fit. This procedure virtually eliminates the convergence problem if the signal frequency is reasonably well known. In addition to ease of coding, the 4-parameter matrix method has a further advantage in the speed of convergence for records containing a small number of sine wave cycles. With each iteration, this algorithm estimates the three linear parameters (amplitude, offset, and phase), as well as the deviation of the frequency from the assumed value. The assumed frequency is updated with this Δf information in each subsequent iteration, until convergence is reached.

SIGNAL-TO-NOISE RATIO AND EFFECTIVE BITS

Figs. 3 and 4 are the results of signal-to-noise ratio (§ 4.5.1) and effective bits (§ 4.5.2) tests of a digitizer that use equivalent-time sampling. The full-scale range of the instrument is ± 2 V, so the sine wave signal shown in Fig. 3 (± 1.9 V or 95% FS) is "a large signal." The test frequency in this case was 10 MHz, and the equivalent sample rate was 2 GHz. The rms of the residuals was 1.107 mV, giving an SNR of 61.7 dB, or an effective bits of 10.03. The 4-parameter matrix sine fit algorithm was used which returned the following parameter estimates: amplitude = 1.89971 V, offset = 0.285 mV, frequency = 9.9966 MHz, and phase = 31.269°.

Several features are evident from examining the residuals: harmonic distortion (predominantly, 2nd) is present; aperture uncertainty is evidenced by the noise bursts that coincide with the regions of highest slew rate (near the zero

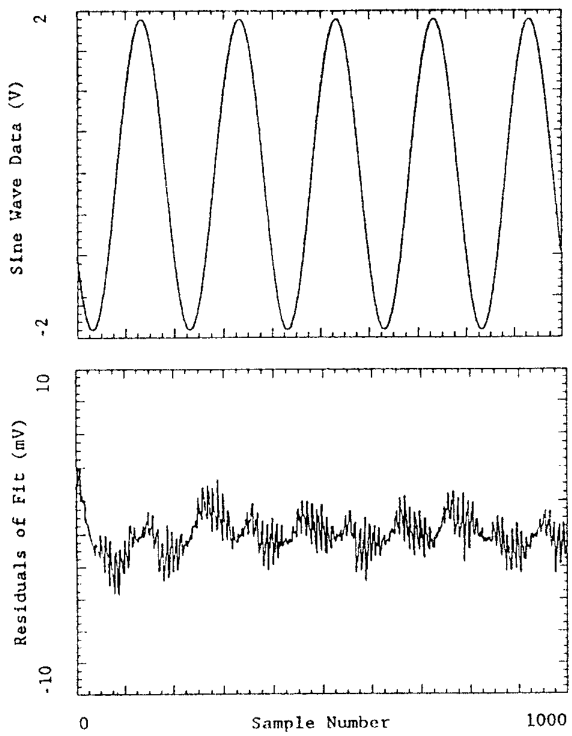

Fig. 3. Sine Wave Data and Residuals of Fit.

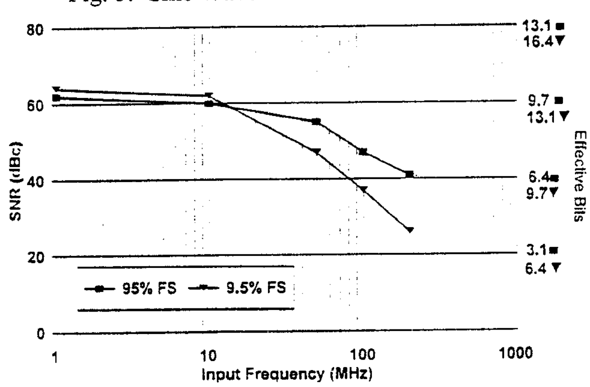

Fig. 4. SNR and Effective Bits vs. Frequency.

crossings); some error due to fixed error in sample time (time base nonlinearity) is evident, especially in the beginning of the record where a peak occurs in the residuals that is not periodic.

The results of several such signal-to-noise ratio/effective bits tests are plotted in Fig. 4 as a function of frequency, for two different levels of input signal amplitude (95% and 9.5% of full scale). As often happens, the SNR improves (at higher frequencies) as the signal level decreases. This reflects the fact that harmonic distortion is often lower for small signals. Note that, for a given signal-to-noise ratio, the effective bits increases by 3.3 bits as the applied signal level is reduced by a factor of 10, as indicated by the different right-hand scales. For these last two reasons, Std. 1057 recommends that a large signal, i.e., one at least 90% of full scale, be used for effective bits tests, to more accurately reflect the worst-case conditions.

EQUIVALENT TIME SAMPLING

For some tests, it is difficult to obtain accurate estimates of performance because of contributions made by aliased components of the test signals. A case in point is the measurement of step response, and parameters that can be derived from it, such as transition duration, impulse response and frequency response. For these cases, the test signal (i.e., input step) contains frequency components that are often much higher than the Nyquist frequency, resulting in low frequency aliases.

The revised standard introduces a simple, effective procedure (§ 4.7.1) that can be used to reduce these aliasing errors. The method, based on the principle of equivalent-time sampling, increases the effective sampling rate of the waveform recorder by an integer, D, by carefully selecting the repetition frequency of the input signal. A record of data obtained in this way is illustrated in the lower plot of Fig. 5. Simply rearranging the samples using a simple algorithm gives the waveform shown in the upper plot. The rearranged waveform is one cycle of the test waveform, sampled at D (4, in this case) times the actual sampling rate of the recorder.

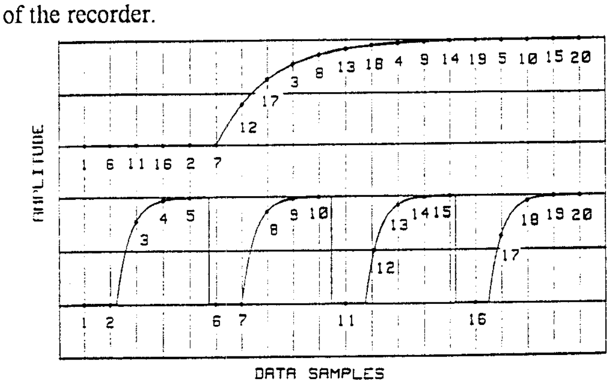

Fig. 5. Equivalent Time Sampling.

Figs. 6 - 10 illustrate the application of this method to the measurement of the step, impulse, and frequency responses of a 10-bit, 60 MHz waveform recorder. (For this test, a reference step generator was used for which the positive transition and top line region are well defined. Although recorded, the ill-defined negative transition was not used in the impulse and frequency response computations.) The raw, repetitive step response data is recorded (Fig. 6). Next, the equivalent, oversampled step response is computed

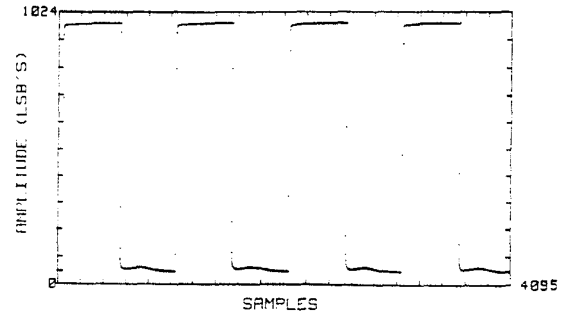

Fig. 6. Repetitive Step Data.

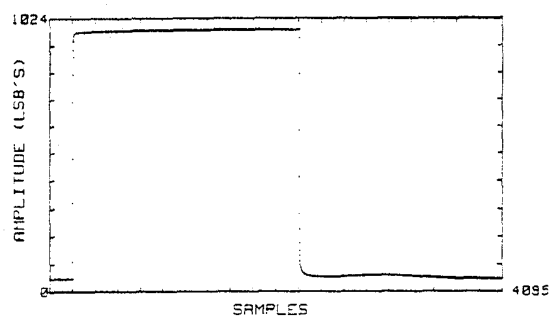

Fig. 7. Oversampled Step Response (from Step response).

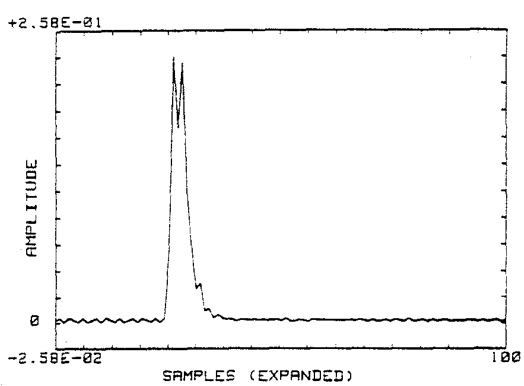

Fig. 8. Impulse Response (from Step Response).

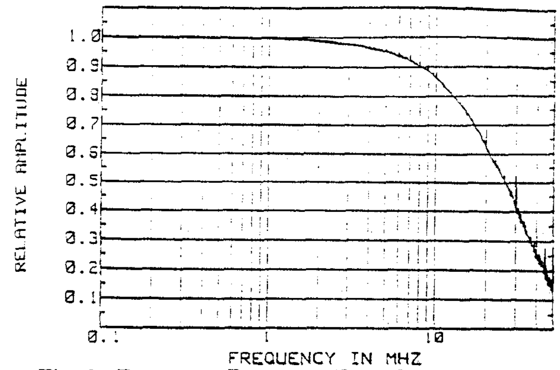

Fig. 9. Frequency Response (from Step Response).

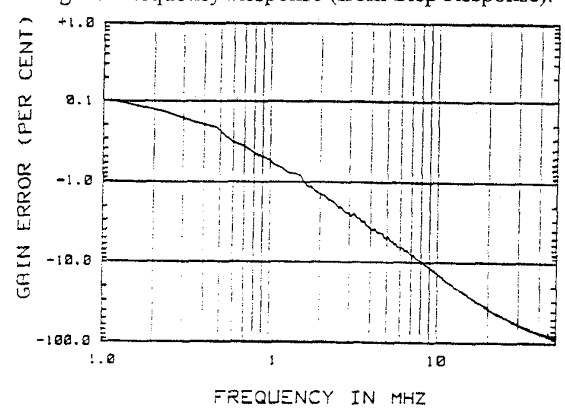

Fig. 10. Relative Gain Error (Flatness) (from Step Response).

(Fig. 7). From this, the impulse response is derived by differentiation (Fig. 8). Finally, the Fourier transform of the impulse response gives the frequency response (Fig. 9). The oversampling minimizes the aliasing errors, producing a frequency response estimate from which accurate gain flatness errors can be determined (Fig. 10).

The phase response is derived from the linear (delay) term in the phase portion of the transform. Since the equivalent delay of the recorded signal with respect to the input waveform is indeterminate, the true phase response can only be approximated. Fig. 11 shows the approximate phase response assuming the phase is 0° at dc and -45° at the -3 dB point at 17 MHz.

When the data to be acquired is a pulse, phase linearity may be more relevant than the complete phase response. Nonlinear phase error can be obtained by forcing the dc phase value and the last phase value to be equal. The resulting nonlinear phase error as shown in Fig. 12 is obtained by subtracting the linear phase term from the Fig. 11 curve.

HARMONIC DISTORTION

For a pure sine wave, the harmonic distortion (§ 4.4.4) is the set of output components at frequencies that are an integer multiple of the applied sine wave. While total harmonic distortion is actually the square root of the sum of the squares of all harmonic distortion components, it is commonly expressed as a subset of all harmonics (e.g., the 2nd through the 10th harmonic) to avoid including a lot of extraneous noise. Spurious components, or spurs, are persistent sine wave components which are not integer multiples of the input sine wave.

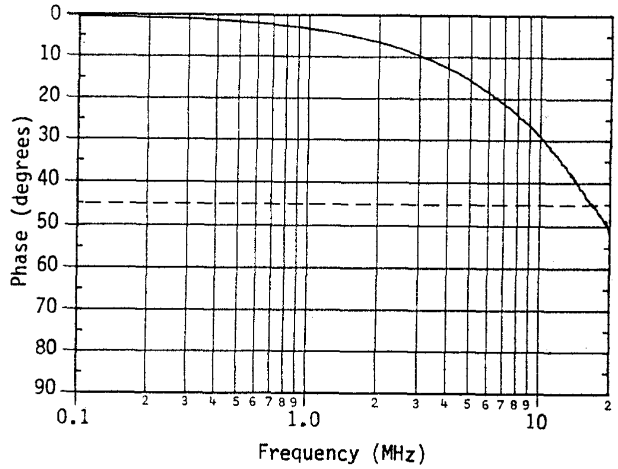

Fig. 11. Phase Respone (from Step Response).

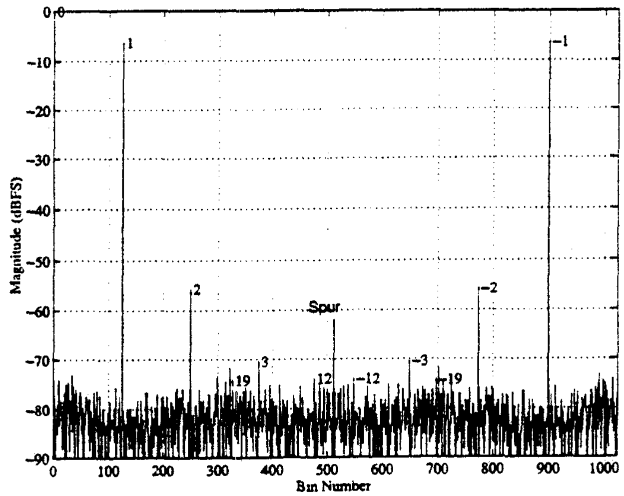

Fig. 13. Harmonic Distortion.

Some insight into the distortion imposed on a pulse as it is recorded may be gained by measuring the recorder's total harmonic distortion. Recall that a repetitive pulse train may be represented as a fundamental sine wave plus a weighted series of its harmonics. Introducing harmonics with erroneous amplitude will distort the pulse.

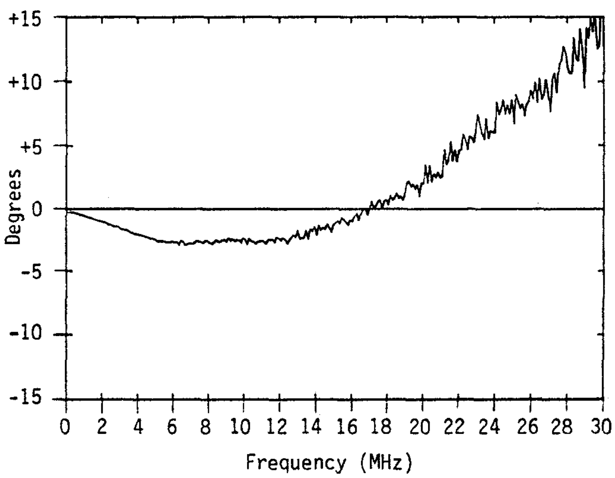

Fig. 12. Nonlinear Phase Error (from Step Response).

The harmonic distortion data displayed in Fig. 13 was obtained with coherent sampling as described in Std. 1057 (§ 4.4.4.1.1). A discrete Fourier transform (DFT) was performed on a carefully constructed record of sine wave data. The frequency of the sine wave is carefully chosen to be one of the DFT frequencies. No window is required other than the rectangular window implicit in taking a record of data. A 25.000 MHz sine wave was recorded at 204.8 Ms/s to obtain the data displayed in Fig. 13. Both the positive and negative components of the input sine wave are labeled (+1 and -1) together with its positive and negative harmonic components which show above the noise. A spur is also shown. Distortion components are then calculated relative to the carrier (i.e., dBc) yielding a -49.40 dBc second harmonic, -49.12 dBc total harmonic distortion (2nd through 10th harmonic), and a -58.66 dBc spur.

OVERVOLTAGE RECOVERY

An overvoltage is any voltage whose magnitude is less than the maximum safe input voltage of the recorder, but is greater than the full scale value for the selected range. Fig. 14 shows a record of sine wave data which contains a positive overvoltage pulse near mid record. In accordance with Std. 1057 (§ 4.1.4.2), a sine wave was fit to a portion of the data prior to the occurrence of the overvoltage pulse. For this data, the sine wave was fit over the first one-third of the record. The fitted sine wave is extrapolated across the entire record and subtracted from the original data yielding the residuals also plotted in Fig. 14. Overvoltage recovery time was measured from the last full-scale point associated with the pulse to the first point which deviated less than, and stays within, a prescribed tolerance (in this case, 0.95 lsb) of the fitted sine wave. A histogram of the overvoltage recovery time for multiple records of data will indicate the stability of the recorder's overvoltage recovery characteristic.

CONCLUSION AND CLOSING REMARKS

A wealth of information can be gathered on a waveform recorder by conducting two general tests: the sine-fit test and the step-response test. The sine-fit test includes the effects of noise, nonlinearities, and aperture uncertainty, but does not measure amplitude flatness or phase linearity. Sine

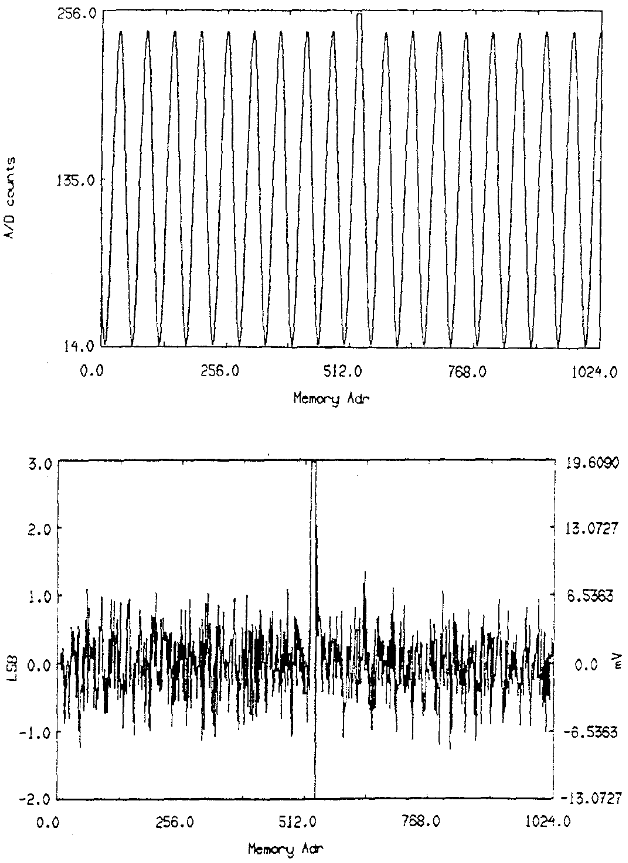

Fig. 14. Sine Wave Data and Residuals Containing Overvoltage Pulse.

wave data may also be used to obtain a code-bin histogram, differential nonlinearity (DNL) and amplitude as a function of frequency. The step response yields pulse parameters (e.g., transition duration, overshoot, and settling time). In addition, the step response can be used to generate amplitude and phase versus frequency, amplitude flatness, and phase linearity. The frequency response of these parameters can be extended by acquiring the step response in equivalent time mode. It may be appropriate to supplement these tests with specialized tests (e.g., overvoltage recovery) which may be important to a particular measurement.

As the trial-use standard (Std. 1057-89) evolved into a full-use standard (Std. 1057-94), a number of changes were incorporated. The section describing histogram testing was expanded in a number of ways. The analysis was extended to give code transition levels as well as integral and differential nonlinearity. The guidance on the choice of measurement parameters is more specific, and the error analysis is more thorough. Many additions were made to the section on testing for harmonic distortion and spurious components. The tests in this section are based on Fourier analysis of the results of a sine wave test. The new standard covers both a coherent sampling method and an incoherent sampling method. The coherent sampling method gives the most accurate results but restricts the frequency of the input signal. The data analysis for the incoherent sampling method was modified to be more accurate than that in the trial-use standard. Finally, a number of comments and cautionary notes were added to guide the user.

IEEE Std. 1057-94, *IEEE Standard for Digitizing Waveform Recorders,* has been issued by the IEEE effective December 30, 1994. It can be ordered from the IEEE by phone (1/800/678-IEEE) or fax (1/908/981-9667). Now that Std. 1057 has been issued, its authors, the Waveform Measurement and Analysis Committee (TC-10) of the IEEE Instrumentation and Measurement Society has focused its efforts on developing a *User's Guide* and on drafting another standard (Std. 1241) for analog-to-digital converters. Those interested in participating in these projects, or balloting on draft documents, are welcome to apply.

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