Preliminary Considerations on ADC Standard Harmonization

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***Abstract*—In this paper, an analytic comparison of the dynamic parameters that are used for qualifying analog-to-digital converters (ADCs) in the frequency domain reported in the most diffused standards is provided. This could be the first step toward their harmonization.**

***Index Terms*—Analog-to-digital converter (ADC), effective number of bits (ENOB), IEC 60748, IEC 62008, IEEE Standard 1057, IEEE Standard 1241, "Methods and draft standards for the DYNamic characterization and testing of Analog to Digital converters" (DYNAD), SIgnal-to-Noise And Distortion ratio (SINAD), signal-to-noise ratio (SNR), spurious-free dynamic range (SFDR), total harmonic distortion (THD).**

I. Introduction

NOWADAYS, analog-to-digital converters (ADCs) are used in a wide range of applications, comprising Data AcQuisition (DAQ), precision industrial measurement, and voice-band and audio applications [1]. Considering the thousands of currently available converters, selecting the proper ADC for a particular application appears to be a difficult task. Different manufacturers, in fact, specify parameters in different ways, often using different specifications to describe their products. This has become more relevant still, considering that ADCs are produced and used around the world. The standardization plays a determinant role [2] in clarifying this scenario by introducing common terminology and test methods that guide manufacturers in describing their products and customers in understanding converter characteristics.

At the level of official international standardization bodies, the current standard International Electrotechnical Commission (IEC) Standard 60748-97 [3] is devoted to stand-alone ADC components and only covers quasi-static operations [4]. Within the Fourth Framework Program "Standard, Measurement and Testing SMT" [5] of the European Union, the research project "Methods and draft standards for the DYNamic characterization and testing of Analog to Digital converters" (DYNAD) was proposed and successfully financed [6]. The project aimed at integrating and complementing IEC standards [3] for the part that concerns dynamic testing by proposing a list of parameters that specify the dynamic behavior of the converter or sample and hold device and indicate in detail the measurement conditions and the data processing algorithms to be adopted. Furthermore, this project seems to be mainly devoted to characterization of the ADC as a stand-alone component, rather than to a complex digital measuring system that includes relevant hardware and software [2].

At the level of category standardization, a remarkable effort has been done by the Technical Committee 10 of the IEEE Instrumentation and Measurement Society through IEEE Standards 1057-94 [7] and 1241-00 [8]. The former has the great value of making a punctual state of the art by focusing metrological performance specifications and testing procedures in an unambiguous way. However, it is specifically devoted to waveform recorders rather than to general digital measuring systems [2]. The latter provides both standard terminology for specifying the performance of ADCs and test methods for measuring it [2]. It has many similarities to IEEE Standard 1057, many of the terms and tests are nearly the same, since ADCs are a necessary part of digitizing waveform recorders [2].

Very recently, IEC Standard 62008 [9], which is otherwise known as the "Performance characteristics and calibration methods for digital data acquisition systems and relevant software" standard, has been released. This standard is aimed to ensure that all measurement systems that rely on DAQ devices meet a common standard. This standard covers the following: 1) the minimum specifications that the DAQ device manufacturer must provide to describe the performance of the analog-to-digital module (ADM) of the device; 2) standard test strategies to verify the minimum set of specifications; 3) the minimum calibration information, required by the ADM, which is stored on the DAQ device; and 4) the minimum calibration software requirements for external and self-calibration of the ADM [9].

The current situation of the standardization of measuring systems based on ADCs is, therefore, characterized by the coexistence of more standards, highlighting the lack of a unified approach, which is an essential requirement for standard harmonization. International trade, in fact, makes products produced in one country and sold in others to be developed or redesigned to meet different standards in different countries. Consequently, standard harmonization is needed so that standards of different countries can be substantially the same, providing benefits for manufacturers, in terms of reduced products variations, inventory and cost, product quality, and for focusing on new products; for customers, giving more choices among cheaper and better products; and for governments to promote international trade and cooperation.

To achieve an effective harmonization of the existing standards, a comparison of the different definitions from the theoretical point of view is necessary to highlight similarities and/or possible ambiguities in the parameter definitions and

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descriptions. Successively, an extensive experimental comparison, starting from the indications achieved in the former phase, should be carried out. The experiments should involve laboratories from different countries after the identification of common test benches and procedures. The results of such a phase would constitute a quantitative basis to find and remove possible metrological incompatibilities.

This paper deals with the first phase, providing an analytic comparison of parameters used for qualifying ADCs reported in the most diffused standards that can be used at an international level. This paper describes the first results of this work, considering the most widely used ADC dynamic parameters in the frequency domain, which are listed as follows: 1) spurious-free dynamic range (SFDR); 2) total harmonic distortion (THD); 3) SIgnal-to-Noise And Distortion ratio (SINAD); 4) signal-to-noise ratio (SNR); and 5) effective number of bits (ENOB).

This paper analyzes the ADC dynamic parameters that are included in the standards produced by official international standardization bodies, namely, IEC Standard 60748-4 and IEC Standard 62008, as well as those that belong to category standardization, namely, IEEE Standards 1057 and 1241, as they have a very diffused usage around the world. As quoted above, IEC Standard 60748-4 only covers static parameters; therefore, while waiting for the release of the new IEC Standard 60748-4-3, including dynamic criteria for ADC characterization, the DYNAD definitions have been considered since they have been designed to fill the part of the IEC standard that concerns dynamic parameters. IEEE TC-10 is currently engaged in the revision of IEEE Standards 1057-1994 and 1241-2000; the analysis reported in this paper is referred to the released versions.

This paper has been divided in seven sections. After a brief introduction on frequency domain testing, each of the succeeding sections analyzes a single parameter including the following: 1) its definitions with the formulas; 2) its description; 3) connected formulas, including relations with other parameters or alternate forms of the definition formula and comments; and 4) comments and preliminary proposals for standard harmonization. The last section sketches conclusions.

II. Frequency Domain Test

The frequency domain tests extracts SFDR, THD, SINAD, SNR, and ENOB from the frequency spectrum of the ADC output response. In particular, the evaluation of the ADC performance is carried out by processing a DFT of a record of data. Before the test execution, some considerations have to be made on the choice of input waveform, clock frequency, and record size, as well as on the accuracy of the input frequency. It is also important to take into account that the DFT of a data record can contain spectral components that does not correspond to the exact center of a DFT frequency bin, making some portion of the signal leak to other bins. These components, which are termed spectral leakage, are undesirable because they often mask spurious signals produced by the ADC. Coherent sampling, in which the bin number of the applied signal is an exact integer, is usually the best approach when dealing with spectral leakage. The leakage problem, in fact, can be completely eliminated if exact coherence is obtained. When the input frequencies do not satisfy the condition for coherent sampling with sufficient accuracy or are unknown, noncoherent sampling is considered, and the windowed DFT is used to reduce the problems caused by spectral leakage.

In this paper, an analytical comparison of the formulas provided by the quoted above standards is presented, not dealing with the frequency domain test setup configuration issues. More information about them can be found in IEEE and DYNAD standards, since IEC Standard 62008 addresses the IEC Standard 60748-4-3, which has not been published yet for the description of the dynamic test methods.

III. SFDR

An ideal ADC, receiving as input a pure sine wave, provides at the output a sampled version of the sampled signal. Actual ADCs are, on the contrary, characterized by outputs, including unwanted signals produced within the device. These signals, generally, are a combination of the harmonics of the fundamental and intermodulation products, but they can also be caused by nonharmonic persistent frequency components, which are called spurii. ADC standards make a distinction between the spurious tones [6] or components [7], [8] and the harmonic distortion [6]–[8], which means that the first ones refer to "*persistent sine wave at frequency other than the harmonic frequencies*" [7], [8], with [6] also adding "*or intermodulation frequencies.*" The harmonic distortion, instead, is defined as "*output components at frequencies that are an integer multiple of the applied sine wave frequency which are induced by the input sine wave*" for a pure sine wave input [6]–[8]. Both spurious components and harmonic distortion degrade the range of ADC input signal levels that can be reliably measured simultaneously, in particular, the ability to accurately measure small signals in the presence of large ones. SFDR is used as a measure of this degradation.

IEEE Standard 1057 and IEC Standard 60748 do not include any SFDR parameter definition in characterizing the ADC dynamic range, despite its wide use. On the contrary, IEEE Standard 1241 defines SFDR as "*the ratio of the amplitude of the ADC output averaged spectral component at the input frequency, f_i, to the amplitude of the largest harmonic* or *spurious spectral component observed over the full Nyquist band*" for a pure sine wave input of specified amplitude and frequency. The formula is reported in Table I.

According to DYNAD, SFDR "*expresses the range, in dB, of input signals lying between the averaged amplitude of the ADC's output fundamental tone, f_i, to the averaged amplitude of the highest frequency harmonic or spurious spectral component observed over the full Nyquist band.*" The formula of SFDR for a pure sine wave input of specified amplitude and frequency is also reported in Table I.

The IEEE Standard 1241 definition uses the word "*ratio*," whereas DYNAD adopts the word "*range*," making immediately understandable what is represented by SFDR on

TABLE I
SFDR FORMULAS

IEEE Std 1241	$SFDR(dB)=20\log_{10}\left(\frac{\left\|X_{avg}(f_i)\right\|}{\max_{f_S,f_h}\left\{\left\|X_{avg}(f_h)\right\| or \left\|X_{avg}(f_S)\right\|\right\}}\right)$	X_{avg}: averaged spectrum of the ADC output; f_i: input signal frequency; f_h and f_s: frequencies of the set of harmonic and spurious spectral components.
DYNAD coherent sampling	$SFDR(dB)=20\log\left(\frac{\left\|Y_{avm}(f_i)\right\|}{\max_{f_{sp},f_h}\left\{\left\|Y_{avm}(f_h)\right\|,\left\|Y_{avm}(f_{sp})\right\|\right\}}\right)$	Y_{avm}: average of an adequate number of amplitude spectra, corresponding to different data records collected at the ADC output; f_i: input signal frequency; f_h and f_{sp}: frequencies of the set of harmonic and spurious spectral components.
DYNAD non-coherent sampling	$SFDR_{dB}=10\log\left(\frac{\left\|Y_{avm}[J]\right\|^2}{\max_{f_h,f_{sp}}\left\{\left\|Y_{avm}[f_h]\right\|^2,\left\|Y_{avm}[f_{sp}]\right\|^2\right\}}\right)+$ $+10\log\left(\frac{\left\|W[0]\right\|^2}{\left\|W_C\left(\frac{\varepsilon_{jr}f_S}{M}\right)\right\|^2}\right)$	ε_{jr}: number of inaccuracy cycles rounded to one decimal; J: integer; $W[0]=\sum_{n=0}^{M-1}w[n]$ w[n]: window function coefficient for a DFT; $W_C\left(\frac{\varepsilon_j f_S}{M}\right)=\int_{-\infty}^{+\infty}e^{-i2\pi\frac{\varepsilon_j f_S}{M}}w(t)dt$ M: number of samples.
IEC Std 62008	$SFDR=20\log_{10}\left(\frac{\text{rms value of output signal}}{\text{rms value of largest single other component}}\right)$	

an amplitude spectrum. In the test section of IEEE Standard 1241, SFDR is specified as the ratio between amplitudes of averaged DFT values. Moreover, IEEE Standard 1241 highlights that SFDR is generally a function of the amplitude and the frequency of the input sine wave, as well as, eventually, of the ADC sampling frequency and input noise or dither.

In the section devoted to the calculation of the dynamic parameters in the frequency domain, DYNAD reports the SFDR formula in terms of averaged power spectrum. Information about the number of records used to calculate the average spectrum is also given, stating that, in practice, to smooth the noise and make the spurii emerge, the acquired records should be "*generally lower than or equal to 10*" because "*taking a number of records greater than 10 will not lead to a great improvement of the noise smoothness.*" The above-quoted SFDR formulas reported in [6] and [8] apply when an integer number of periods of the sampled waveform are acquired, which is the coherent sampling condition. DYNAD takes into account, also for SFDR computation, the case of noncoherent sampling conditions when the ratio between the input frequency f_i and the sampling frequency f_s is not an integer value, considering the relationship between f_i and f_s as

$$f_i=(J\pm\varepsilon_j)\frac{f_s}{M} \tag{1}$$

where J is an integer number of cycles of the input waveform, so that the periodic extension of the sample set is continuous, and ε_j is the number of inaccuracy cycles ($\varepsilon_j\leq 0.5$; $\varepsilon_j=0$ in the case of coherent sampling). DFT is performed on an M-sample-long record. To take into account the effect of the windowing, the FFT amplitude at the frequency bin J is multiplied by the factor

$$\frac{|W[0]|}{\left|W_c\left(\frac{\varepsilon_j f_s}{M}\right)\right|} \tag{2}$$

where $W[0]$ and $W_c(\varepsilon_j f_s/M)$ are reported in Table I. As can be seen in Table I, the two SFDR formulas provided in [6] in the cases of coherent and noncoherent sampling are equal, except for the additive term in the case of noncoherent sampling, which corrects only the signal amplitude, as the exact frequency of the spurious tone is generally unknown. Moreover, except for a notation difference, the DYNAD SFDR formula in the case of coherent sampling is equal to the one found in [8].

The IEC Standard 62008 SFDR definition follows the same approach as those reported in DYNAD and IEEE Standard 1241 and states, "*For a pure sine-wave input, ratio, expressed in dB, of the r.m.s. value of the output signal at the input frequency to the largest persistent r.m.s. value of the output at any other single frequency.*" In the formula reported to calculate this parameter (Table I), the rms value of the output signal is "*determined from the amplitude of the ADM output at the input signal frequency*," and the rms value of the largest other component is "*the r.m.s. value of the largest component excluding the fundamental of the input signal.*" To improve test accuracy, the IEC 62008 standard suggests to "*acquire multiple test records, compute the corresponding Fourier Transforms and average the values corresponding to each component of the Fourier Transform to obtain an averaged Fourier Transform.*"

From the analysis of the definitions reported in the existing standards, it can be observed that the basic principle is the same. All of them are theoretically correct, each in its definition domain. The definitions are clear and easy to comprehend. The DYNAD definition is clearly more complete, taking in consideration also the noncoherent sampling operation. This approach simplifies the test setup by removing a strict synchronization between the ADC clock generator and the test signal generator. The result, however, is a higher mathematical complexity, leading to a processing increase.

Concerning the SFDR formulas, it is possible to assert that the standards are *almost* harmonized. The main effort to making them more similar is a matter of notation.

The mathematical complexity of the formulas, actually, gives the user a degree of freedom; therefore, once the existence of a simpler formula is provided, one should be free to choose the best tradeoff between the test bench cost and the processing weight.

IV. THD

THD is another parameter to measure the harmonic distortion caused by ADC nonlinearity, which takes into account the amplitude of the harmonics of the fundamental signal.

IEC Standard 60748 does not include THD parameter definition. IEC Standard 62008 does not provide any THD formula giving only its definition: THD is "*for a sine wave signal, the sum of power of all harmonics.*"

IEEE Standard 1057 defines THD as "*the root sum square of all harmonic distortion components including their aliases.*"

IEEE Standard 1241, in the terminology section, states that THD is "*for a pure sine wave input of specified amplitude and frequency, the root-sum-of-squares (rss) of all the harmonic distortion components including their aliases in the spectral output of the ADC. Unless otherwise specified, THD is estimated by the rss of the second through the tenth harmonics, inclusive. THD is often expressed as a decibel ratio with respect to the root-mean-square amplitude of the output component at the input frequency.*"

In DYNAD, THD is "*the ratio of the rss (root-sum-of-squares) of all the harmonic distortion components, including their aliases in the spectral output of the ADC, to the rms amplitude of the output fundamental component, expressed in dB. The input stimulus is assumed to be a high purity sine wave. Unless otherwise specified, THD is estimated considering the second through the tenth harmonics, inclusive*." DYNAD uses the term *ratio* in the definition, whereas [8] states that THD is often expressed as a ratio. Except for this, the two definitions are the same.

IEEE Standard 1057 provides a procedure for calculating the THD both in the cases of coherent and noncoherent sampling. In the former case, THD is the square root of the sum of squares of the S_{rms} values that arise from the following formula:

$$S_{\mathrm{rms}} = \sqrt{\frac{1}{M^2}\left(|X_{fa}|^2 + |X_{-fa}|^2\right)} \tag{3}$$

where M is the acquired record length, and X_{f_a} and X_{-f_a} are the DFT values at the positive and negative frequencies f_a and $-f_a$ of each harmonic or spurious component. THD in the case of noncoherent sampling is calculated as the square root of the sum of squares of S_{rms} (the rms value of the DFT values in the positive and negative frequency bands), which is determined by using the following formula:

$$S_{\mathrm{rms}} = \sqrt{\frac{1}{M^2 \cdot NNPG}\left(\sum_{f=f_1}^{f_2} |X_f|^2 + \sum_{f=-f_2}^{-f_1} |X_f|^2\right)} \tag{4}$$

with

$$NNPG = \frac{1}{M}\sum_{i=0}^{M-1} w_i^2 \tag{5}$$

where $NNPG$ is the noise power gain normalized by M, which is the noise power gain of the rectangular window, and f_1 to f_2 and $-f_2$ to $-f_1$ are the selected positive and negative frequency bands "*that the harmonic or spurious component occupies.*"

The IEEE Standard 1241 THD formula (Table II) is the rss of a specified set of harmonic distortion components (h), including their aliases. The averaged spectral magnitude X_{avg} is used in this formula, as done for SFDR, because it has a variance that is smaller than that of the nonaveraged spectral magnitude. Alternate THD% and $\mathrm{THD}_{\mathrm{dB}}$ formulas are included in the standard as well.

DYNAD, on the contrary, considers the nonaveraged spectral magnitude in the THD computation (Table II). It also reports THD expressed in terms of power in the cases of coherent and noncoherent sampling, including terms that correct both the amplitudes of the fundamental signal and of the harmonics to take into account the windowing effect in the second case (Table II). To minimize errors in the measurements of THD, SINAD, and SFDR, DYNAD states that "*the harmonic distortion of the input sinewave must be less than the THD of the ADC under test.*" Therefore, DYNAD adds a guideline for the THD measurement in the worst case, when "*the THD of the ADC under test and the input sinewave distortion are dominated by the same harmonic component (same frequency* and *same phase)*," and in all the other cases, when "*the THD of the ADC as well as the distortion of the input sinewave result from a distortion over many harmonic components*," having not necessarily the same frequency and/or the same phase.

Concerning THD, there seems to be an agreement in principle among the different standards, as well as if the formulas are different. The IEEE definitions for THD are simpler from the mathematical point of view, whereas the DYNAD ones are more complete, including the detailed models of the windowing effect on the FFT of the output signal. In this case, the applicability of the THD formulas provided in [7] for noncoherent sampling seems higher than those provided in [6].

V. SINAD AND SNR

Any deviation between the ADC output signal (converted to input units) and the input signal, not including 1) deviations

TABLE II
THD FORMULAS

IEEE Std 1241	$THD = \frac{1}{M}\sqrt{\sum_{h}(X_{avm}(f_h))^2}$	$X_{avm}(f_h)$: averaged magnitude of the component at the hth harmonic of the DFT of the ADC output data record M: number of samples in the data record.
DYNAD coherent sampling	$THD = 20\,log\frac{\sqrt{\sum_{h=2}^{m} X(f_h)^2}}{X(f_1)}$ $THD_{dB} = 10\,\log\left(\frac{\sum_{h=2}^{h_{\max}} \lvert Y[hJ]\rvert^2}{\lvert Y[J]\rvert^2}\right)$	f_h: set of output components at frequencies that are an integer multiple of the applied sine wave frequency f_1 up to the frequency $fm = m \cdot f_1$, where m is application dependent. Y[hJ]: h^{th} harmonic component.
DYNAD non-coherent sampling	$THD_{dB} = 10\,\log\left(\frac{\sum_{h=2}^{h_{\max}} \lvert Y[hJ]\rvert^2 \frac{\lvert W[0]\rvert^2}{\left\lvert W_C\left(\frac{frac_r\lvert h(J\pm\varepsilon_j)\rvert f_S}{M}\right)\right\rvert^2}}{\lvert Y[J]\rvert^2 \frac{\lvert W[0]\rvert^2}{\left\lvert W_C\left(\frac{\varepsilon_{jr} f_S}{M}\right)\right\rvert^2}}\right)$ $= 10\,\log\left(\frac{\sum_{h=2}^{h_{\max}} \frac{\lvert Y[hJ]\rvert^2}{\left\lvert W_C\left(\frac{frac_r\lvert h(J\pm\varepsilon_j)\rvert f_S}{M}\right)\right\rvert^2}}{\lvert Y[J]\rvert^2}\left\lvert W_C\left(\frac{\varepsilon_{jr} f_S}{M}\right)\right\rvert^2\right)$	$frac_r[x]$: fractional part of x rounded to the first decimal.

caused by linear time invariant response of the system, 2) harmonics of the fundamental up to a prefixed order, or 3) a DC level shift, is commonly attributed to noise [6]–[9]. Noise is caused by phenomena that act on either the phase or the amplitude of the input signal, like, e.g., the effects of random errors (random noise), fixed-pattern errors, high-order harmonics or intermodulation distortion, and aperture uncertainty [6]–[8].

IEC Standard 62008 adds that "*for DC or very low frequency input signals it is usual to describe system noise which does not include the effect of non-linearity and time base errors*."

ADC noise performances are dealt with in different ways in [6]–[9] by using different terms, depending on the considered output noise that can include harmonic distortion or not. IEEE Standard 1057 defines the amount of noise present in the output using SNR as "*the ratio of root mean square (rms) signal to rms noise for sine wave input signals. The SNR depends on the amplitude and frequency of the applied sine wave. The amplitude and frequency at which the measurement was made shall be specified.*"

IEEE Standard 1241, instead, uses SINAD, which is defined in the same manner (and with the same formula for the calculation in the time domain) as SNR in [7]. The term SNR is not used in [8] because, in the context of ADC testing, it is used in different ways, where it is very ambiguous. SINAD, as stated in [8], is defined as "*for a pure sine wave input of specified amplitude and frequency, the ratio of the rms amplitude of the ADC output signal to the rms amplitude of the output noise, where noise is defined to include not only random errors but also nonlinear distortion and the effects of sampling time errors.*" The formulas are reported in Table III. IEEE Standard 1241 also reports the formula for calculating this parameter in the frequency domain (Table III). Moreover, it introduces another parameter, the signal-to-nonharmonic ratio (SNHR), which is defined for a pure sine wave input of specified amplitude and frequency as "*the ratio of the rms amplitude of the ADC output signal to the rms amplitude of the output noise which is not harmonic distortion.*" The SNHR formula provided in [8] is reported in Table IV. As can be seen, the use of the SINAD and SNHR couple instead of SNR clarifies whether the test result includes the harmonic distortion or not.

However, because of its utility in a variety of ADC applications and comparative purposes, IEEE Standard 1241 reports a *normalized SNR* measure. It is usually obtained from the ratio of the rms signal to the portion of rms noise that is not harmonic distortion, as for the SNHR, using a sine wave test signal.

DYNAD defines both SNR and SINAD. The former is "*a measure of the broadband noise and spurious that are introduced into the ADC output signal by the sampling and AD conversion processes. It is given by the ratio, expressed in dB, of the signal power to noise (including spurious) power, i.e.,*

TABLE III
SINAD FORMULAS

IEEE Std 1241	$SINAD = \dfrac{\text{rms signal}}{\text{rms noise}}$ Equivalent to SNR in IEEE Std 1057 $\text{rms noise} = \dfrac{1}{M}\sqrt{\sum_{m=0}^{M-1} E_{avm}(f_m)^2}$ Frequency domain calculation	E_{avm}: residual spectrum of X_{avm} $X_{avm}(f_m)$ the averaged magnitude spectral component at discrete frequency f_m after a DFT
DYNAD coherent sampling	$SINAD_{dB} = 10\log\left(\dfrac{\lvert Y[J]\rvert^2 - \lvert NFl\rvert^2}{\sum_{k=1,k\neq J}^{M/2-1} \lvert Y[k]\rvert^2 + 2\lvert NFl\rvert^2 + \frac{1}{2}\left\lvert Y\left[\frac{M}{2}\right]\right\rvert^2}\right)$	$\lvert NFl\rvert^2 = \dfrac{\sum_{k=1,k\neq J,k\neq hJ}^{M/2-1} \lvert Y[k]\rvert^2 + \frac{1}{2}\left\lvert Y\left[\frac{M}{2}\right]\right\rvert^2}{\frac{M}{2} - h_{\max}}$ h_{max}: the highest harmonic to remove. $h = 2 \ldots h_{\max}$
DYNAD non-coherent sampling	$SINAD_{dB} = 10\log\left(\dfrac{\lvert Y[J]\rvert^2 - \lvert NFl\rvert^2}{A+B}\right) + 10\log(ENBW) +$ $+ 10\log\left(\dfrac{\lvert W[0]\rvert^2}{\left\lvert W_c\left(\frac{\varepsilon_{jr} f_s}{M}\right)\right\rvert^2}\right)$ $A = \sum_{k=1,k\neq J\pm l,k\neq rnd[h(J\pm\varepsilon_j)]\pm l}^{M/2-1} \lvert Y[k]\rvert^2 + (2l_{max}+2)\lvert NFl\rvert^2 + \frac{1}{2}\left\lvert Y\left[\frac{M}{2}\right]\right\rvert^2$ $B = ENBW\sum_{h=2}^{h_{max}} \left\lvert Y\left[rnd\left[h\left(J\pm\varepsilon_j\right)\right]\right]\right\rvert^2 \cdot \dfrac{\lvert W[0]\rvert^2}{\left\lvert W_c\left(\frac{frac_r\left[h\left(J\pm\varepsilon_j\right)\right] f_s}{M}\right)\right\rvert^2}$	$\lvert NFl\rvert^2 = \dfrac{\sum_{k=1,k\neq J\pm 1,k\neq rnd[h(J\pm\varepsilon_j)]\pm l}^{M/2-1} \lvert Y[k]\rvert^2 + \frac{1}{2}\left\lvert Y\left[\frac{M}{2}\right]\right\rvert^2}{\frac{M}{2} - h_{\max}(2l_{\max}+1)}$ rnd[x]: round to the nearest integer of x, Y[hJ]: h[th] harmonic component, $2l_{max}$: number of bins to remove around the signal and its harmonics. $frac_r[x]$: is the fractional part of x rounded to the first decimal. $l = 0 \ldots l_{\max}$ $ENBW = \dfrac{M\sum_{n=0}^{M-1} w^2[n]}{\left[\sum_{n=0}^{M-1} w[n]\right]^2}$
IEC Std 62008	$SINAD = 20\log_{10}\left(\dfrac{\text{rms value of output signal}}{\text{rms value of noise}}\right)$	

TABLE IV
SNHR AND SNR FORMULAS

IEEE Std 1241	$SNHR = \dfrac{rms\ signal}{rms\ noise}$ $rms\ noise = \dfrac{1}{M}\sqrt{\sum_{m=0}^{M-1} N_{avm}(f_m)^2}$ Frequency domain calculation	N_{avm}: averaged DFT spectrum, X_{avm}, with the components (bins) at dc, the test frequencies f_i and $(f_s\text{-}f_i)$ and the specified harmonic frequencies all set to zero.
DYNAD coherent sampling	$SNR_{dB} = 10\log\left(\dfrac{\lvert Y[J]\rvert^2 - \lvert NFl\rvert^2}{\sum_{k=1,k\neq J,k\neq hJ}^{M/2-1} \lvert Y[k]\rvert^2 + (h_{\max}+1)\lvert NFl\rvert^2 + \frac{1}{2}\left\lvert Y\left[\frac{M}{2}\right]\right\rvert^2}\right)$	$\lvert NFl\rvert^2 = \dfrac{\sum_{k=1,k\neq J,k\neq hJ}^{M/2-1} \lvert Y[k]\rvert^2 + \frac{1}{2}\left\lvert Y\left[\frac{M}{2}\right]\right\rvert^2}{\frac{M}{2} - h_{\max}}$ h_{max}: the highest harmonic to remove. $h = 2 \ldots h_{\max}$
DYNAD non-coherent sampling	$SNR_{dB} = 10\log\left(\dfrac{\lvert Y[J]\rvert^2 - \lvert NFl\rvert^2}{\sum_{k=1,k\neq J\pm 1,k\neq rnd[h(J\pm\varepsilon_j)]\pm l}^{M/2-1} \lvert Y[k]\rvert^2 + \left(h_{max}\left(2l_{max}+1\right)+1\right)\lvert NFl\rvert^2 + \frac{1}{2}\left\lvert Y\left[\frac{M}{2}\right]\right\rvert^2}\right) +$ $+ 10\log(ENBW) + 10\log\left(\dfrac{\lvert W[0]\rvert^2}{\left\lvert W_c\left(\frac{\varepsilon_{jr} f_s}{M}\right)\right\rvert^2}\right)$	$\lvert NFl\rvert^2 = \dfrac{\sum_{k=1,k\neq J\pm 1,k\neq rnd[h(J\pm\varepsilon_j)]\pm l}^{M/2-1} \lvert Y[k]\rvert^2 + \frac{1}{2}\left\lvert Y\left[\frac{M}{2}\right]\right\rvert^2}{\frac{M}{2} - h_{\max}(2l_{\max}+1)}$ $ENBW = \dfrac{M\sum_{n=0}^{M-1} w^2[n]}{\left[\sum_{n=0}^{M-1} w[n]\right]^2}$ rnd[x]: round to the nearest integer of x, Y[hJ]: h[th] harmonic component, $2l_{max}$: number of bins to remove around the signal and its harmonics. $l = 0 \ldots l_{\max}$

of the rms amplitude of the ADC output fundamental tone to the rms amplitude of the spectral content defined by the sum of all frequencies in the Nyquist band ($f_S/2$) *excluding DC, fundamental, and harmonics.*" This corresponds to the SNHR definition in [8]. The latter is "*for a pure sinewave input of specified amplitude and frequency, the ratio of the rms amplitude of the ADC output fundamental tone to the rms amplitude of the output noise, where noise is defined as to include not only random errors but also nonlinear distortion and the effects of sampling time errors, i.e., the sum of all non-fundamental spectral components in the range from DC (excluded) up to half the sampling frequency* ($f_S/2$)." This definition corresponds to the SINAD definition reported in IEEE Standard 1241. The formulas reported in [6] and Tables III and IV comprise the SINAD and SNR in the frequency domain in the cases of coherent and noncoherent sampling.

To compute SINAD and SNR by using the formulas provided in [6], the noise floor parameter NFl is required. The ADC output noise is assumed to be white and to evaluate NFl, neither the DC bin nor the signal bin nor the harmonic bins have to be considered. The number of harmonics to remove for the calculation of NFl depends on the ADC under test and the accuracy required. In practice, removing the second through the tenth harmonics is often sufficient. In the case of noncoherent sampling, the NFl formula reported in DYNAD takes into account the effect of the main lobe broadening due to the windowing by removing a few bins before and after the signal and harmonic bins. Concerning the SINAD and SNR formulas in the case of noncoherent sampling, the equivalent noise bandwidth (ENBW) of the window applied must be used (Tables III and IV).

Two chapters are also devoted to their time-domain measurement. Another chapter in [6] deals with SINAD calculation in the dual tone test. Concerning the SNR and/or SINAD relations with other parameters, [8] reports a formula that relates SINAD with ENOB, whereas [7] reports the same relation for SNR.

IEC Standard 62008 defines SINAD as "*For a pure sine-wave input, ratio of the r.m.s. amplitude of the ADM output signal at the input frequency to the r.m.s. amplitude of all other signal in the ADM output*," and in a note, it is also suggested that "*SINAD information should be supplied at a range of gains over a range of input and sampling frequencies.*" In the SINAD formula reported in Table III, the rms value of the output signal is the same as in the SFDR formula, whereas the rms value of noise that includes harmonics is "*determined by the root of the sum of squares of all of the terms of the output, excluding the DC term and input frequency.*" This definition is the same as those reported by IEEE Standard 1241 and DYNAD. Concerning the SNHR, IEC Standard 62008 reports only its definition, stating that it is the "*ratio, expressed in dB of the power of the signal with all possible harmonics, to the overall noise.*" Considering that IEC Standard 62008 does not include a THD formula, information about the harmonic distortion content is taken into account only in the SINAD parameter. In fact, by simply inverting the SINAD formula, a figure of merit, which is referred to as THD + N, can be obtained [10].

Currently, there are open questions on whether to use the term SNR or not and on how noise is measured. Once again, the models included in DYNAD are mathematically more complete but difficult to implement for a laboratory technician, whereas the IEEE definitions are more practice oriented. However, in this case, there is no agreement on terminology. In this case, different terms are used from different standards to refer to the same parameter, leading to a possible confusion in the end users.

A proposal for harmonization could be done, recalling the main principles for quantifying the effect of noise only and using the SNR definition in [6] instead of SNHR. SNR, in fact, is the standard figure of merit in the industry to refer to the ratio between the signal power and the noise power.

To define the ratio between the signal power and the power of noise and distortion, there is an agreement among [6], [8], and [9] on the SINAD definition. Therefore, this could be used in the form provided in [8].

Concerning the definition implementations, the best applicability can be found in the IEEE and IEC formulas, limiting the implementation of formulas reported in [6] to scientific fields.

VI. ENOB

Excessive noise in an ADC can make it appear to have fewer bits of resolution than it actually has. The apparent resolution of an ADC is specified by the ENOB.

IEEE Standard 1057 and IEC Standard 69748 do not include the ENOB definition.

However, IEEE Standard 1057 reports formulas for calculating the effective bits E for an input sine wave of specified frequency and amplitude, after gain and offset correction, and for expressing the relationship between SNR and E (Table V).

IEEE Standard 1241 defines ENOB as "*a measure of the signal-to-noise and distortion ratio used to compare actual ADC performance to an ideal ADC.*" The same ENOB formulas reported in [7] can be found in IEEE Standard 1241 (Table V), with the exception that, in this standard, the relationship between SINAD and ENOB is reported since SINAD is the same as the term SNR in IEEE Standard 1057. To take into account the effect of jitter on ENOB, [8] provides another formula (Table V).

DYNAD defines ENOB (N_{ef}) as the number that "*in practice identifies the actual resolution of the converter taking into account the signal to noise and distortion ratio,*" adding that it can be interpreted as follows: "*if the actual noise is attributed only to the quantisation process, the ADC under test can be considered as equivalent to an ideal* N_{ef}*-bit ADC insofar as they produce the same rms noise level.*" The ENOB formula and the relation with SINAD according to [6] are reported in Table V.

In the formula that describes the relationship between N_{ef} and SINAD, the latter is measured when applying a full-scale sine wave at the input of the ADC ($\mathrm{SINAD}_{\mathrm{dBfs}}$). DYNAD, however, states that "*in practice, it is impossible to use a full-scale sine wave to measure the dynamic parameters of an ADC.*" Therefore, $\mathrm{SINAD}_{\mathrm{dBfs}}$ is calculated as the difference

TABLE V
ENOB FORMULAS

IEEE Std 1057	$E = N - \log_2\left(\dfrac{\text{rms noise}}{\text{ideal rms quantization error}}\right) = \log_2\left(\dfrac{\text{full scale range}}{\text{rms noise}\cdot\sqrt{12}}\right)$	N: number of digitized bits
E vs SNR	$E = \log_2(SNR) - \dfrac{1}{2}\log_2(1.5) - \log_2\left(\dfrac{A}{V}\right)$ $\quad SNR = \sqrt{1.5}\left(\dfrac{A}{V}\right)2^E$	A: amplitude of the sine wave fitted to the output V: one half of the full-scale range of the waveform recorder input.
IEEE Std 1241	$ENOB = N - \log_2\left(\dfrac{\text{rms noise}}{\text{ideal rms quantization error}}\right) = \log_2\left(\dfrac{\text{full scale range}}{\text{rms noise}\cdot\sqrt{12}}\right)$	N: number of digitized bits
ENOB vs SINAD	$ENOB = \log_2(SINAD) - \dfrac{1}{2}\log_2(1.5) - \log_2\left(\dfrac{A}{(V/2)}\right)$ $\quad SINAD = \sqrt{1.5}\left(\dfrac{A}{V/2}\right)2^{ENOB}$	A: amplitude of the sine wave fitted to the output V: full-scale range of the ADC under test.
ENOB (jitter effects)	$ENOB = \log_2\left[\dfrac{FSR}{V_{noise}\cdot\sqrt{12}}\right]$ $\quad V_{noise} \approx V_P\dfrac{\omega\sigma_t}{\sqrt{2}}$	V_p: the sine wave peak amplitude. ω: equal to $2\pi f$, with f the signal frequency fundamental component. σ_t: the standard deviation of the jitter
DYNAD	$N_{ef} = N - \log_2\dfrac{\eta_{rms}}{\sigma_{eq}}$	η_{rms}: *rms* total noise including jitter and harmonic distortion σ_{eq}: ideal *rms* quantisation noise for a sinusoidal input.
ENOB vs SINAD	$N_{ef} = \dfrac{SINAD_{dBFS} - 1.76dB}{6.02}$ $\quad SINAD_{dBFS} = SINAD_{dB} - 20\log(SFSR)$	SFSR: for a pure sinewave input of specified amplitude and frequency, the ratio of the ADC's output fundamental tone to the amplitude of a full scale sinewave at the same frequency.
IEC Std 62008	$ENOB = (\text{signal to noise and distortion ratio} - 1.76)/6.02$	

between the SINAD obtained by using a sine wave of amplitude A different from the full scale and a signal-to-full-scale ratio (SFSR) computed as the ratio (in decibels) between A and the ADC full scale (Table V). DYNAD specifies that, in practice, N_{ef} is often given without specifying the value of SFSR, which can lead to a misunderstanding of the performances of the ADC. However, the ENOB is advised to be avoided, since the same information is contained in the SINAD measured when applying a full-scale sine wave at the ADC input. In [6], there is a chapter devoted to the time-domain measurement of N_{ef} and another one that concerns its calculation in the dual-tone test.

IEC Standard 62008 defines ENOB as a "*practical limit of the resolution of an ADM due to inherent noise and linearity error. Effective number of bits represents that the ADM performs as an ideal ADM with this number of bits.*" In a note, it is also suggested that "*effective number of bits information should be supplied at a range of gains over a range of input and sampling frequencies.*" ENOB is directly calculated from SINAD, as reported in Table V.

Even if there is still a discussion on the utility of a figure of merit that reports all the ADC nonidealities in terms of equivalent quantization noise, all the references consider ENOB. There is an agreement on what the ENOB represents, with the DYNAD definition seemingly the clearest one.

VII. CONCLUSION

The analytic comparison of the most used ADC parameters according to the most diffused standards is the necessary first step toward the realization of a unique ADC standard. The results of such a study, in fact, are the basis for the start of the design of a measurement campaign for assuring the metrological compatibility of the achieved results, following the different standards. This paper starts the analysis by considering SFDR, THD, SINAD, SNR, and ENOB to put in evidence similarities and/or possible ambiguities in the parameter definitions and descriptions taken from IEC Standards 62008, 1057, and 1241, as well as DYNAD. Future developments of this work will concern the analysis of other ADC parameters, as well as those included in this paper, taking into account the new IEC Standard 60748-4-3 as soon as it is released. Next, an experimental comparison will be carried out on a set of currently available ADCs, with the test setups described in the references to assess the metrological compatibility of the achieved results.

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