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SOME CRITICAL NOTES ON DAC FREQUENCY DOMAIN SPECIFICATIONS

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Abstract: In this paper currently used DAC frequency domain parameters definitions are evaluated in order to achieve a consistent terminology that can be used in a DAC standard. New definitions for DAC THD, SFDR, SNR, SINAD and ENOB are proposed taking into account the most used definitions, the need of avoiding any ambiguities and the measurability of them.

Keywords: DAC, THD, SFDR, SNR, SINAD, ENOB.

1. Introduction

Purpose of a DAC is to provide a link between digital and analog sections of a mixed signal system.

Interest in these devices is growing both from scientific and industrial worlds increasing also the number of applications they are required for. Industrial process control, optical networking, automatic test equipment, high-end medical imaging are only a few examples of them.

However, the very large number of converter products available in the marketplace, even from the same manufacturer, can overwhelm the user when faced with the problem of selecting a DAC for a given application.

The correct interpretation of the specifications is made more complex from the lack of standardized definitions of specifications that all manufacturers can agree upon.

Aware of these problems the Waveform Measurement and Analysis Technical Committee (TC-10) of the IEEE Instrumentation and Measurement Society is working to produce a new standard defining terminology and test methods for DACs [1]. A DAC metrology standard, in fact, can provide manufacturing or performance requirements that the device has to meet, and evaluation methods for this purpose leading to a better product quality.

This paper arises from a research project aimed to give a contribution and ideas for discussion concerning the new DAC standard. The main objective of this project is to provide a unified approach to DAC standardization concerning terminology. A standardized terminology in fact is essential for three main reasons: i) to understand, as it ensures that when a given term is used, it means the same thing to everyone, ii) to evaluate and compare, because testing something that has not been first rigorously defined is impossible, and iii) to ensure consistency, since terms need to be organized so that they can be used not only individually, but together with related terms.

The method followed to achieve a set of DAC standardized parameter definitions starts from carrying out a comprehensive overview on DAC terminology taken from existing standards, scientific literature and manufacturers' documentation for highlighting similarities, ambiguities and lacks in the parameter definitions. After that, DAC specifications have to be chosen establishing for each parameter of being unambiguously defined and practically measurable. The final set of DAC specifications has to take into account the most used definitions coming from the collected references and the existing definitions have to be entirely adopted, modified or completely rewritten in order to satisfy the above reported requirements.

Previous work [2,3] provided some critical notes on DAC static and some dynamic parameter definitions, suggesting definitions for resolution, full scale range, LSB (Least Significant Bit), offset and gain, INL (Integral NonLinearity) and DNL (Differential NonLinearity), settling time, rise time, fall time and glitch.

This paper continues the previous work considering DAC dynamic parameter definitions in the frequency domain.

To satisfy the requirement for each parameter to be measurable, each proposed definition is joined with its measurement unit taking as reference the International System of Units (SI). Since some DAC specifications could be derived from the ADC ones with the appropriate changes, the IEEE Standard for Terminology and Test Methods for Analog-to-Digital Converters (IEEE Std. 1241) [4] has been used as guideline.

A set of DAC frequency domain specifications including DAC THD (Total Harmonic Distortion), SFDR (Spurious Free Dynamic Range), SINAD (SIgnal to Noise And Distortion ratio), SNR (Signal to Noise Ratio), ENOB (Effective Number Of Bit) has been considered. The paper analyzes the quoted above parameters, providing some useful comments to highlight ambiguity in the mostly used definitions for further discussions and suggesting new definitions.

2. Total Harmonic Distortion

THD gives an indication of a circuit's linearity in terms of its effect on the harmonic content of an input signal. In the DAC case this parameter represents the difference between an ideal sinewave and its reconstructed version using the converter [5].

THD figure of merit is almost the same for different sources, defining this parameter as the ratio between the Root Mean Square (RMS) level of the harmonic content of the output signal to the RMS level of the fundamental. Alternatively it is the ratio of the power in the harmonics of the fundamental frequency and the power in the fundamental itself.

The number of considered harmonics is essential as it can greatly influence the outcome of any THD test if any relevant harmonic is excluded [6]. Moreover, finding an agreement on this number among the different references is very hard. For someone "*usually ten harmonics are sufficient*" [6], other states that "*THD is usually calculated using the first ten-twenty harmonics or until the harmonics can not be distinguished from the noise floor*" [7], the first five [8,9] or six [8,10] are considered as well.

In many cases the number of harmonics is not provided [11-15] but someone recommends to consider only harmonics within the first Nyquist band [11,12].

Sometimes different harmonic orders are considered by the same manufacturer depending on the particular product. To give an example, Analog Device AD5424/AD5433/AD5445 datasheet [16], states that "*usually only the lower order harmonics*" are included in the THD formula "*such as second to fifth*". AD9734/AD9735/AD9736 datasheet [17] considers the first six harmonics, and AD5643R/AD5663R datasheet [18] takes into account harmonics present on the DAC output without specifying any number.

IEEE Std. 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*".

This definition is open to different possibilities concerning the number of harmonics to be considered, however it must be specified if different from ten.

The IEEE Std. 1241 definition states that the aliases in the spectral output have to be included in the THD calculation. In the case of DACs this is subject to debate.

Burns and Roberts [19] state that "*since the distortion appears at the expected frequencies, rather than appearing at aliased frequencies*", calculating the alias is not necessary. However, a check has to be done to control that no overlap between distortion components of one tone and images of other test tones caused by reconstruction is present.

On the contrary Hendriks, in [8], states that THD "*includes most of the harmonically related spurs as well as aliased, folded back, harmonics within the DAC Nyquist bandwidth*" adding that "*the dominant harmonic at higher output frequencies typically is a second or third harmonic, which could also be aliased back into the Nyquist bandwidth*".

For some DAC datasheets produced by Intersil the problem of alias is solved including in the THD definition the use of a filter, more precisely "*an output filter of 1/2 the clock frequency is used to eliminate alias products*" [20]. In fact, only the harmonics that exceed the Nyquist frequency, alias back into the first Nyquist zone (Fig. 1).

The THD definition has to be valid apart from the presence of a filter and should include information about the harmonics to be considered to avoid confusion and ambiguity. Considering only harmonics in band (within the Nyquist limit) that can be distinguished from the noise floor could be a good solution. Taking into account the previous considerations, the proposed definition for THD is the following:

"*For a pure sine wave input of specified amplitude and frequency, THD is the ratio between the root-sum-of-squares of all the harmonic distortion components that can be distinguished from the noise floor including their aliases, in the output spectrum of the DAC and the RMS value of the output component at the input frequency*".

3. Spurious Free Dynamic Range

Spurious Free Dynamic Range (SFDR) is the usable DAC dynamic range before spurious noise interferes or distorts the fundamental signal [21]. It is the difference between the fundamental and the highest spur power over a frequency band of interest.

There are different definitions for SFDR, mainly depending on the exclusion of harmonics and on the definition of a frequency window around the fundamental in the computation of this parameter (Fig.2) [12].

Considering or not the harmonically related distortion components as spurious is subject to debate.

A spur is defined in [19] as "*any non signal component that*

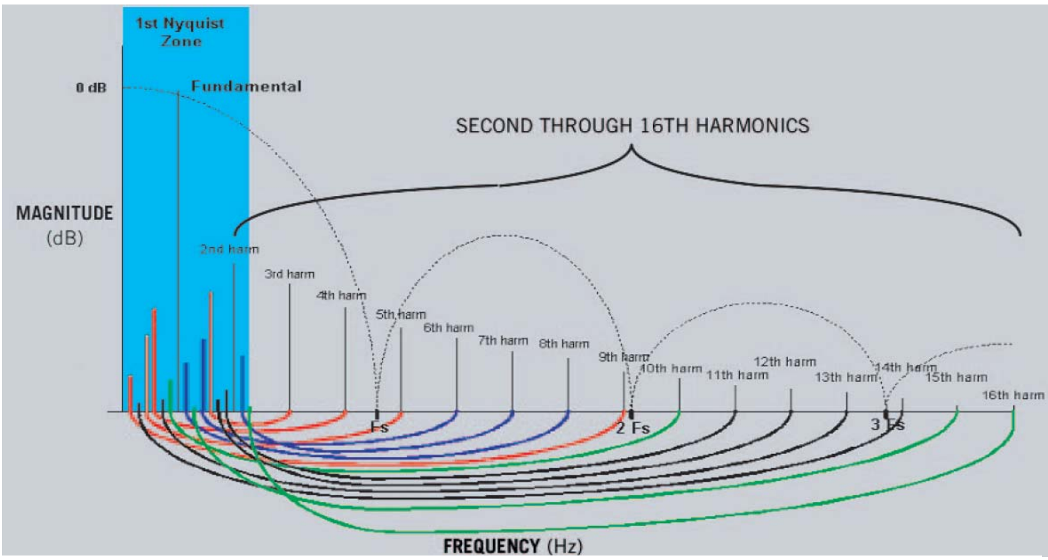

Fig. 1. Harmonics beyond the first Nyquist zone mapped back to the first Nyquist zone.

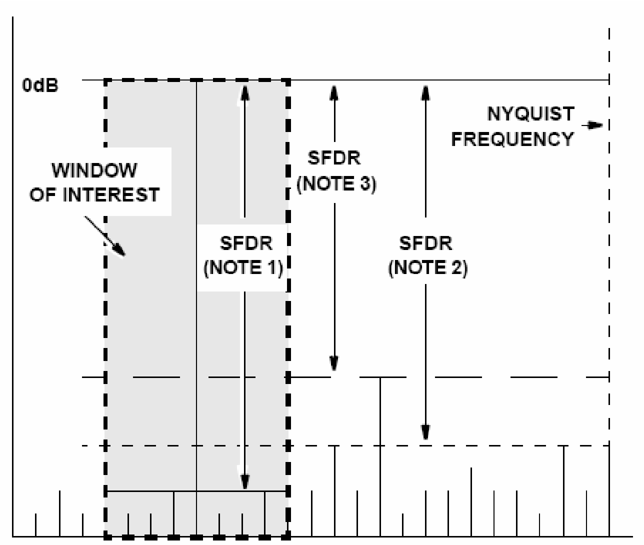

Fig. 2. SFDR computation methods [12]. NOTE 1: SFDR defined in a narrow band; NOTE 2: SFDR to Nyquist without harmonics; NOTE 3: SFDR to Nyquist including harmonics.

is confined to a single frequency", and can be caused by "*harmonic and intermodulation distortion, clock feedthrough, sigma-delta converter self-tones, stray oscillations, or any of dozens of other undesiderable processes*".

The worst spur, considered in the SFDR computation is "*the largest spectral component excluding the input signal and DC*" [22], "*the highest peak of any of the harmonic or intermodulation distortion products*" [23], "*possibly harmonic component*" [9] or "*usually but not necessary always a harmonic of the fundamental*" [24].

On the contrary, Hendriks [8] states that "*the spur does not have to be harmonically related to the fundamental*". Reference [6] addresses the problem of considering or not harmonic distortion components as spurii by introducing another parameter in addition to the SFDR, the Distortion Free Dynamic Range (DFDR). Therefore, SFDR "*specifies the available signal range as the spectral distance between the amplitude of the fundamental and the amplitude of the largest non-harmonic, spurious component in the frequency band of interest*", while the DFDR is referred to the "*largest harmonic component in the frequency band of interest*".

IEEE Std. 1241 defines SFDR for a pure sine wave input, as "*For a pure sinewave input of specified amplitude and frequency, the ratio of the amplitude of the analog-to-digital converter's 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*", so "*both harmonic distortion and spurious signals are considered to be undesirable spurs in the spectrum of a sampled pure sinewave*". This definition considers the averaged spectral magnitude because it has a smaller variance than the non-averaged one.

Reference [12] explains the exclusion of harmonics in the SFDR computation stating that "*since harmonic distortion typically exceeds noise in the D/A converter's spectrum, little information about the characteristics of the noise floor are obtained*". But the inclusion of harmonics gives important information for example in high frequency telecommunications DACs, because "*an offending spur at an odd harmonic is likely related to amplitude distortion and at even harmonics is likely related to phase distortion*" [25]. Moreover, both spectral spurious and harmonics restrict the dynamic range.

The frequency band of interest over which the SFDR is specified is not always the full Nyquist band. Many manufacturers instead consider a more lower frequency interval. Many definitions don't set the frequency band to be considered for the SFDR only mentioning "*a specified bandwidth*", [5,7,8,9].

The reason reported in [8] is that "*SFDR is sometimes specified over a narrow bandwidth which typically excludes the worst spur falling within the Nyquist zone*", assuming the user "*will operate over a narrow frequency band and will filter out any larger out of band spurs*", as for example in the case of frequency synthesizers employing direct digital synthesis, for maintaining low phase noise [8].

Reference [21] takes into account the designers' point of view stating that "*by picking an arbitrary window size, the 2nd or 3rd harmonic are often not included in the measurement. Because many systems designers intend to use a narrow band pass filter around the fundamental signal, they are more interested in the spectral performance within a band that the filter will pass. However, having full knowledge of a DAC's spectral performance is essential to the selection of an appropriate band pass filter to remove the harmonics*". Therefore, in the same way the spurs and spectral performance up to Nyquist frequency are important for designers to know DAC actual performance.

Although SFDR over a narrow bandwidth takes into account the high frequency spurs generated by glitch impulses that fold back in band, reside close to the fundamental, cannot be filtered and dominate the noise within that range of frequencies, "*unless the user filters the output signal in a similar fashion to that being used to test the converter, the effect of the remainder of the noise floor on the application is unknown*" [12].

In any case, different SFDR values can arise depending on the method used for its measurement. SFDR is generally a function of the amplitude and the frequency of the input sine wave and the sample frequency.

For signal amplitudes near full scale, one of the first input

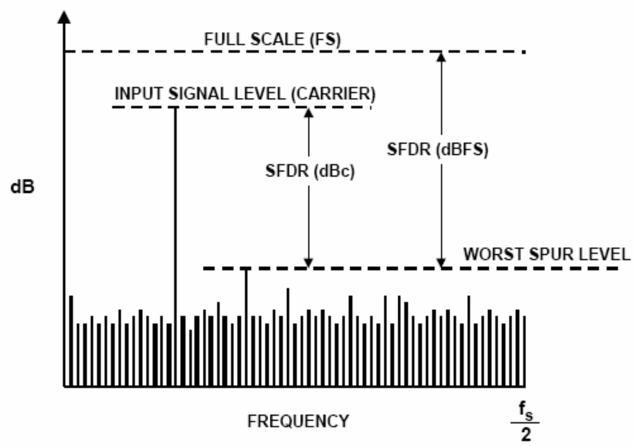

Fig. 3. SFDR units of measurement.

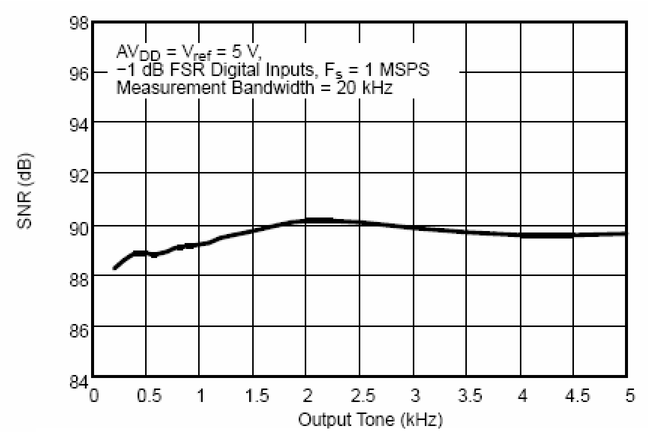

Fig. 4. SNR vs output frequency of the Burr-Brown 8554 DAC [26].

harmonics usually determines the highest spectral spur. As the signal falls several dB below full scale, other spurs which are not harmonics of the input signal can become dominant [24]. SFDR is usually measured with respect to the carrier frequency amplitude in dBc, (degrades as signal level is lowered) or with respect to the DAC full-scale range, in dBFS, (always relates back to converter full scale), (Fig.3).

Taking into account all these considerations, the proposed definition for SFDR is the following:

"*For a pure sinewave input of specified amplitude and frequency, SFDR is the ratio of the amplitude of the DAC output averaged spectral component at the input frequency to the amplitude of the largest unwanted spectral component observed over a specified frequency band. SFDR is expressed in dBc or in dBFS*".

This definition requires the specification of amplitude and frequency of the input sinewave as well as of the frequency band considered.

Reporting in the definition *unwanted spectral component* allows the choice of considering the largest harmonic spectral component or not in the SFDR without having to define what the term "spurious" is intended to mean. Harmonics can be excluded by a suitable choice of frequency band over which the SFDR is computed.

4. SIgnal to Noise And Distortion ratio and Signal to Noise Ratio

Specifications and characterization curves that reveal the signal-to-noise ratio performance including or not distortion (SNR and SINAD) are beginning to appear on DAC datasheets [26,27], (Fig.4, Fig.5), however, the higher the resolution the more challenging the accurate characterization and testing of these important performance parameters.

SNR and SINAD depend on the converter resolution of the converter and include specifications of linearity, distortion, sampling time uncertainty, glitches, noise and settling time [28].

In general, all the spectral components different from the signal applied at the DAC input are considered noise. Consequently, both distortion components and other signal degradation factors are contribute of the noise. However, for characterization purposes these two contributes should be considered separately [19].

For this reason, the SNR is commonly measured by excluding harmonic distortion components, that are instead included in the SINAD. SINAD "*encompasses all noise in band (both harmonically and non-harmonically related over the full Nyquist bandwidth), defining the overall effective resolution of the converter being tested*" [12]. SNR, instead, not including any harmonic contributions, provides "*insight to the overall characteristic of its noise floor*", that in combination with THD gives more information about the nature of both harmonically and non-harmonically generated distortion than the one provided by SINAD alone [12].

The IEEE Std. 1241 considers the term SNR ambiguous, since it has been used to represent both signal-to-noise including and excluding harmonic distortion. Therefore, this standard introduces the Signal to Non-Harmonic Ratio (SNHR) defined as "*for a pure sine-wave input of specified amplitude and frequency, the ratio of the root-mean-square (rms) amplitude of the analog-to-digital converter output signal to the rms amplitude of the output noise which is not harmonic distortion*", together with the SINAD that is "*for a pure sine wave input of specified amplitude and frequency, the ratio of the root-mean-square (rms) amplitude of the analog-to-digital converter 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*" "*except deviations caused by linear time-invariant system response (gain and phase shift), or a dc level shift*".

However, the term SNHR is rarely used by manufacturers, being the SNR the preferred one for indicating the signal to noise ratio without harmonic distortion. Aware of this practice, the IEEE TC10 is changing SNHR with SNR in the next version of the IEEE 1241, that is currently under revision [29]. Since SINAD and SNR can be measured in the time domain as well as in the frequency domain the IEEE standard definitions do not refer to the DAC output spectrum.

SNR is also calculated excluding not all but selected harmonic distortion components. In practice, the first four

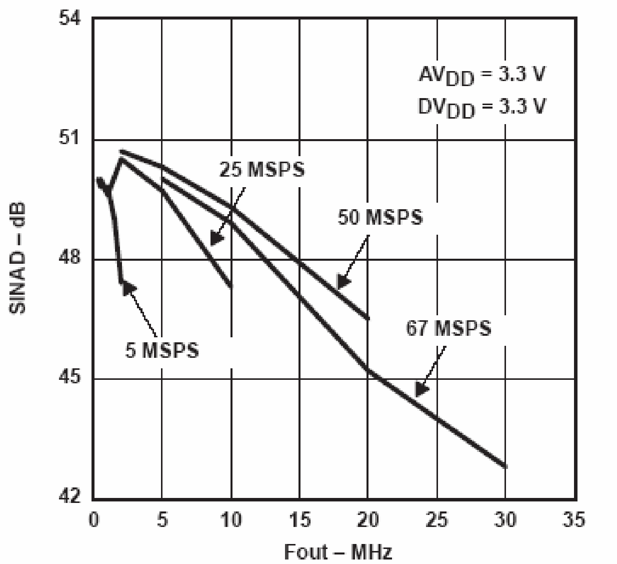

Fig. 5. SINAD vs output frequency of the Texas Instruments THS5641 DAC [27] for different output update rates.

[30,31], five [24], six [10], as well as nineteen [26] harmonics can be considered significant. SNR and SINAD are commonly calculated over the Nyquist band, although in some cases they can be specified in a "*given bandwidth*" [6].

For audio DAC, SNR is measured with input data set to zero, as in this case the output consists of any noise sources in the DAC [32]. A more proper term to describe such noise floor for an audio DAC is idle channel noise (Fig.6) [32].

The AES "Standard method for digital audio engineering-measurement of digital audio equipment" [33], defines the SNR as the noise in presence of the signal, that includes "*all harmonic, inharmonic and noise components*" and the idle channel noise when no signal is applied at the input. Although in that way mixing up the noise in presence of the input signal and the one without is avoided, SNR definition, including the harmonic distortion, is the same of SINAD, parameter not considered in this standard.

The EIAJ Standard CP-307 uses the term SNR to refer to the idle channel noise [34].

Most of the audio DAC datasheets [34,35,36] comply with the EIAJ Standard CP-307 [37] that uses the term SNR to refer to the idle channel noise [34].

In other cases, understanding if the SNR is computed considering the presence or not of the input signal is not possible, since information about the compliance with the EIAJ is missing in datasheets [38,39].

Taking into account the previous considerations, the proposed definition for SINAD and SNR are the following:

"*For a pure sine wave input of specified amplitude and frequency, the ratio of the root-mean-square (rms) amplitude of the DAC output signal to the rms amplitude of the output noise over a specified window of interest. Noise includes all the output components that are not the input signal applied except deviations caused by linear time-invariant system response (gain and phase shift) or a dc level shift. SINAD is expressed in dB*".

"*For a pure sine wave input of specified amplitude and frequency, SNR is the ratio of the root-mean-square (rms) amplitude of the DAC output signal to the rms amplitude of the output noise over a specified window of interest. Noise includes all the output components that are not the input signal applied except harmonic distortion and deviations caused by linear time-invariant system response (gain and phase shift) or a dc level shift. SNR is expressed in dB*".

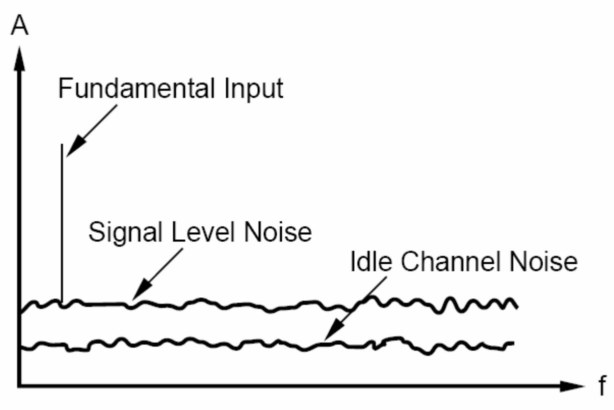

Fig. 6. Comparison of the dynamic range and idle channel noise floors [32].

These definitions can be used both in the time and in the frequency domain.

Mentioning a "window of interest" takes into account the common practice of excluding not all but selected harmonic distortion components for the SNR computation as well as considering a bandwidth that can be different from the Nyquist one.

Since noise has historically been an ambiguous term, particular attention is required by the meaning that is intended to give to this term.

The IEEE Std. 1241 states that the 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), or a dc level shift. For example, noise includes the effects of random errors (random noise), fixed pattern errors, nonlinearities (e.g., harmonic or intermodulation distortion), and aperture uncertainty*".

Reference [24] explains that the greater noise contribute in DACs is interference in the form of high amplitude, low energy (hence low rms) spikes appearing at the output, caused by coupling of digital signals in different ways including coupling via stray capacitance, power supplies, inadequate ground systems, feedthrough and glitch generation.

However, DAC noise has different sources and its contributions depend on several different factors, (e.g. the particular device architecture, the environmental conditions, grounding and shielding etc.), making listing them in a standard definition quite useless.

For this reason it has been chosen of not specifying the noise components unlike the IEEE Std. 1241, addressing to references [40-45] for a deeper analysis.

5. Equivalent Number Of Bit

Excessive noise can make DACs appear to have fewer bits of resolution than they actually have [19]. ENOB specifies how many bits an ideal DAC would require in order to obtain the same SINAD as that measured on the real DAC. Therefore, comparing this number to the nominal number of bits indicates how well the DAC approaches the ideal case [6].

IEEE Std.1241 defines ENOB as "a *measure of the signal-to-noise and distortion ratio used to compare actual analog-to-digital converter (ADC) performance to an ideal ADC*". Moreover, "*For an input sine wave of specified frequency and amplitude, after correction for gain and offset*".

The following formula is provided:

$$ENOB = N - log_2\left(\frac{rms\ noise}{ideal\ rms\ quantization\ error}\right) = log_2\left(\frac{full\ scale\ range}{rms\ noise \cdot \sqrt{12}}\right), \quad (1)$$

where N is the number of digitized bits and the second equality comes from taking the ideal rms quantization error equal to $Q/\sqrt{12}$.

The quantization error is defined in [4] as "*the error caused by conversion of a variable having a continuous range of values to a quantized form having only discrete values, as in analog-to-digital conversion. The error is the difference between the original (analog) value and its quantized (digital) representation*".

While in the case of ADCs *the original analog value* is the input and *its quantized digital representation* is the output, for DACs both the input and the output are quantized, therefore the DAC quantization error defined as the difference between an analog and a digital value does not make sense, since information about the analog signal to be considered are not available.

It is possible instead to define the DAC quantization noise. The quantization noise is modeled as a random variable assumed i) to be independent of the analog signal, ii) uniformly distributed over (-q/2,+q/2), where q is the quantization step, and iii) white. Although the IEEE Std. defines quantization error and noise as the same thing, the quantization error is a deterministic value, the quantization noise is statistically modeled. Therefore, the ENOB formula in the case of DACs can be rewritten changing *ideal rms quantization error* with *ideal rms quantization noise*.

Most of references define ENOB through the SINAD, by using the formula

$$ENOB = \frac{SINAD - 1.76\,dB}{6.02}, \qquad (2)$$

someone [10,14,19] considering instead of SINAD the SNR. However, since ENOB provides an overall measure of the DAC performance and the harmonic distortion affects its value, when the SNR is defined excluding harmonic distortion components, relating ENOB with SINAD is the more correct choice. In the other cases the problem of the ambiguity of the term SNR arises.

Defining ENOB in terms of another figure of merit, as *a measure of the signal-to-noise and distortion ratio* makes this parameter useless as it is redundant.

Taking into account the previous considerations, the proposed definition for ENOB is:

"*For an input sinewave of specified frequency and amplitude, after correction for gain and offset, effective number of bits is the difference between the DAC digital resolution and the number of bits affected by DAC nonideality. This number has to be obtained as the binary logarithm of the ratio between the measured rms noise and the ideal rms quantization noise*".

In this way ENOB definition is not directly related to the SINAD.

6. CONCLUSIONS

In the paper the need for a unique set of DAC specifications including both the end-user and the manufacturers point of view has been highlighted. DAC THD, SFDR, SINAD, SNR and ENOB definitions coming from different sources are collected, compared and discussed in order to achieve a unique set of definitions. The final set of DAC specifications has been achieved considering that each parameter has to be unambiguously defined and practically measurable.

REFERENCES

[1] IEEE Std. 1658 Draft, "IEEE Standard for terminology and test methods for digital-to-analog converters", 2005.

[2] E.Balestrieri, S.Moisa, S.Rapuano, "DAC static parameter specifications - some critical notes", Proc. of 10th IMEKO TC4 Workshop on ADC Modelling and Testing, Gdynia and Jurata, Poland, 12-15 September 2005, vol. I, pp.81-86.

[3] E.Balestrieri, "Some critical notes on DAC time domain specifications", Proc. of IEEE IMTC'06, Sorrento, Italy 24-27 April 2006, pp.930-935.

[4] IEEE Std. 1241, "IEEE Standard for terminology an test methods for analog-to-digital converters", 2000.

[5] Analog Device, "Analog-digital conversion handbook", Prentice Hall, 1986.

[6] Philips, "Mixed signal testing cookbook", Nat.Lab. Report 7103, ver. 2.1 - October 1999.

[7] M.Gustavsson, J.J.Wikner, N.N.Tan, "CMOS data converters for communications", Kluwer Academic Publishers, 2000.

[8] P.Hendriks, "Specifying communication DACs", IEE Spectrum, July 1997.

[9] Atmel, "Data converter terminology", Application Note, available at www.atmel.com

[10] Analog Device, "Data converters: high speed digital-to-analog converters glossary", available at www.analog.com.

[11] Maxim, "ADC and DAC glossary", Application Note 641, December 2000, available at www.maxim-ic.com

[12] Intersil, "Understanding the HI5721 D/A converter spectral specifications", Application Note, available at www.intersil.com

[13] Soft Test Inc. "The fundamental of mixed signal testing", version 3.2, 2003.

[14] A.Van den Bosch, M.Steyaert, W.Sansen, "Static and dynamic performance limitations for high speed D/A converters", Kluwer Academic Publishers, 2004.

[15] R.C.Cabot, "Fundamental of modern audio measurement", Audio Precision Inc., www.audioprecision.com

[16] Analog Device, AD5424/AD5433/AD5445 datasheet, available at www.analog.com

[17] Analog Device, AD9734/AD9735/AD9736datasheet, available at www.analog.com

[18] Analog Device, AD5643R/AD5663R datasheet, available at www.analog.com

[19] M.Burns, G.W. Roberts, "An introduction to mixed-signal IC test and measurement", Oxford University Press, 2001.

[20] Intersil, ISL5857 datasheet, available at www.intersil.com.

[21] Intersil, Measuring Spurious Free Dynamic Range in a D/A Converter, Technical Brief, available at www.intersil.com.

[22] M.Baker, "Demystifying mixed signal test methods", Newnes, 2003.

[23] D.Tweed, "Digital processing in an analog world", Circuit Cellar INK, Issue 99, October 1998.

[24] W.Kester Editor, “The data conversion handbook”, Analog Device Inc., Elsevier, 2005.

[25] B.Jasper, “Practical telecom DAC testing”, www.testedgeinc.com

[26] Burr-Brown products from Texas Instruments, DAC 8554 datasheet, available at www.ti.com

[27] Texas Instruments, THS5641 datasheet, available at www.ti.com

[28] R.J.Van de Plassche, ”CMOS integrated analog-to-digital and digital-to-analog converters”, Kluwer Academic Publishers, Boston, 2003.

[29] E.Balestrieri, J.Blair, P.Daponte, L. De Vito, S. Max, S.Rapuano, S.J.Tilden, “ADC parameters and characteristics”, IEEE Instrumentation & Measurement Magazine, Vol.8, No.5, December 2005, pp.44-54.

[30] Maxim, MAX5894 datasheet, available at www.maxim-ic.com

[31] Maxim, MAX5876 datasheet, available at www.maxim-ic.com

[32] L.Gaddy, H.Kawai, “Dynamic performance testing of digital audio D/A converters”, Burr-Brown Application Bulletin, May 1997.

[33] AES17-1998 (r2004), “Standard method for digital audio engineering-measurement of digital audio equipment”, Audio Engineering Society Inc..

[34] Crystal, CS4390 datasheet available at www.cirrus.com

[35] Texas Instruments, PCM1748 datasheet, available at www.ti.com

[36] Analog Device, AD1851/AD1861 datasheets, available at www.analog.com

[37] EIAJ CP-307, “Methods of measurement for CD players”, Electronics Industries Association of Japan, June 1985.

[38] Philips Semiconductor, UDA1334BT datasheet, available at www.semiconductors.philips.com

[39] Maxim, MAX9850 datasheet, available at www.maxim-ic.com

[40] J. A. Connelly, K. P. Taylor, “An analysis methodology to identify dominant noise sources in D/A and A/D converters”, IEEE Trans. on Circuits and Systems, Vol. 38, No. 10, October 1991, pp.1133 – 1144.

[41] V. N. Kuleshov, H. Y. Liu, “Fundamental noise in direct digital frequency synthesizers”, Proc. of the 49th IEEE International Frequency Control Symposium, 31 May-2 June 1995, pp. 288 – 293.

[42] I.Galton, “Digital cancellation of D/A converter noise in pipelined A/D converters”, IEEE Trans. on Circuits and Systems, Vol. 47, No. 3, March 2000, pp.185-196.

[43] P.M.Lavrador, N.Borges de Carvalho, J.C.Pedro, “Evaluation of Signal-to-Noise and Distortion Ratio degradation in nonlinear systems”, IEEE Trans. on Microwave Theory and Techniques, Vol.52, No.3, March 2004, pp.813-822.

[44] N.Kurosawa, H.Kobayashi, H.Kogure, T.Komuro, H.Sakayori, “Sampling clock jitter effects in digital-to-analog converters”, Measurement, Vol.31, No.3, April 2002, pp.187-199.

[45] J.J.Wikner, N.N.Tan, “Influence of circuit imperfections on the performance of DACs”, Analog Integrated Circuits and Signal Processing, Vol.18, No.1, January 1999, pp.7-20.