STANDARDS AND INFORMATION DOCUMENTS

AES17-2015
(revision of AES17-1998)

STANDARDS

AES standard method for digital audio engineering - Measurement of digital audio equipment

AUDIO ENGINEERING SOCIETY, INC.
551 Fifth Avenue, Suite 1225, New York, NY 10176, US.

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AES17-2015
revision of AES17-1998

AES standard method for digital audio engineering — Measurement of digital audio equipment

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Abstract

This standard provides methods for specifying and verifying the performance of medium-to-high performance digital audio equipment. It comprises an exhaustive list of measurements applicable to digital audio equipment.

This edition substantially revises and updates AES17-1998.

Audio Engineering Society Inc. 551 Fifth Avenue, New York, NY 10176, US.

www.aes.org/standards   standards@aes.org

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Foreword

[This foreword is not a part of *AES standard method for digital audio engineering — Measurement of digital audio equipment*, AES17-1998]

This document has been prepared by the SC-02-01 Working Group on Digital Audio Measurement Techniques of the SC-02 Subcommittee on Digital Audio of the Audio Engineering Society Standards Committee. It is a revision of AES17-1991. With the permission of AESSC, it also had been independently released by ANSI Accredited Standards Committee S4 as ANSI S4.51-1991.

Discussions on the revision project, AES17-R, began in the autumn of 1995. Proposals for revision have been discussed at five subsequent open working group meetings and over the working group reflector, SC_02_01@aessc.aes.org. The call for comment on its draft was published 1997-10-09 on http://www.aes.org/standards and was distributed with the *Journal of the Audio Engineering Society*, vol. 45, no. 11.

The following individuals contributed to the preparation of the 1991 edition of this document: Robert Adams, Richard Cabot, Louis Fielder, David Haynes, and Tomlinson Holman. The revision was prepared by R. Cabot based on the working group discussions.

Richard Cabot, Chairman

Working Group SC-02-01 on Digital Audio Measurement Techniques

1998-03

Foreword to 2015 revision

[This foreword is not part of AES17-2015, *AES standard method for digital audio engineering - Measurement of digital audio equipment*.]

This document substantially revises and updates AES17-1998, *AES standard method for digital audio equipment - Measurement of digital audio equipment*. It includes measurements using more up-to-date measurement tools, and includes annexes describing advanced measurement methods for production testing, and frequency-domain and window-width filters.

The members of the writing group that developed this document in draft are: R.C. Cabot, I. Dennis, T. Duffy, J. Emmett, B. Katz, T. Kite, C. Lacinak, T. Lund, J.G. McKnight, H. Morfett-Jones, J. Novick, M. Poimboeuf, B. Putzeys, P. Soper, J. Woodgate, M. Yonge.

Tom Kite, Chairman

Working group SC-02-01 on Digital Audio Measurement Techniques

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Note on normative language

In AES standards documents, sentences containing the word “shall” are requirements for compliance with the document. Sentences containing the verb “should” are strong suggestions (recommendations). Sentences giving permission use the verb “may”. Sentences expressing a possibility use the verb “can”.

AES standard method for digital audio engineering - Measurement of digital audio equipment

0 Introduction

This standard provides methods for specifying and verifying the performance of medium- to high-performance digital audio equipment.

Many tests are very similar to those used for testing analogue equipment. However, because of the unique requirements of digital audio equipment and the effects of its imperfections, additional tests are needed.

The characteristics of voice-grade digital audio EUTs are sufficiently different from those of high-performance equipment that some of the test levels and frequencies specified in this document may need to be revised for these applications. Low bit-rate coders are an example of EUTs that require additional test techniques. The nature of such coders dictates that the test methods be based on psychoacoustic models which can predict subjective performance. However, the techniques described here should still be informative for such systems.

Another caveat concerns digital EUTs which purposely modify the time-domain characteristics of the audio signal, such as pitch shifters and reverberators. Many of the tests in this standard assume that the frequency spectrum of the output signal is substantially the same as that of the input signal. Also, high-level interfering signals (as would be encountered with reverberators) have not been considered.

1 Scope

This standard specifies basic measurement methods of medium- to high-performance digital audio equipment.

It includes definitions, and measuring conditions and methods applicable to professional equipment.

This standard does not consider:

− measurement of low-quality audio devices,

− measurement of low-bit-rate audio devices (‘sub-band’ or ‘perceptual’ coding devices),

− measurement of devices which significantly modify time or frequency characteristics of the signal, such as pitch shifters or reverberators,

− measurement of signals from analogue input to analogue output, beyond the most general,

− EMC and safety related testing.

2 Normative references

The following documents, in whole or in part, are normatively referenced in this document and are indispensable for its application. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

ITU-R BS.468-4, *Measurement of audio-frequency noise voltage in sound broadcasting*. International Telecommunication Union, Geneva, Switzerland.

IEC 61260-1, *Electroacoustics - Octave-band and fractional-octave-band filters.* International Electrotechnical Commission, Geneva, Switzerland.

3 Terms and definitions

For the purposes of this standard, the following definitions apply.

3.1
Equipment under test
EUT

3.2
Folding frequency
one-half the sampling frequency of the digital system

NOTE The folding frequency defines the maximum upper band-edge frequency. A signal applied to the input whose frequency is above the folding frequency is subject to aliasing. For EUTs which operate at multiple simultaneous sampling frequencies, the sampling frequency to be considered is the lowest sampling frequency in the signal path.

3.3
Upper band-edge frequency
highest signal frequency to be measured (see 4.3)

NOTE 1 The non-ideal behaviour of anti-aliasing filters and reconstruction filters in the EUT leads to the upper band-edge frequency being lower than the folding frequency.

NOTE 2 A higher upper band-edge frequency may be used for higher sampling rates.

3.4
Passband
frequency band from 20 Hz to the upper band-edge frequency

3.5
Alias products
signal components that appear in the passband, caused by input signals above the folding frequency

3.6
Image products
signal components that appear above the passband, caused by input signals below the folding frequency

3.7
Input word length
maximum number of bits in a digital word which can be applied to the digital input of an EUT for which none of the bits is ignored

NOTE The input word length of an EUT determines the correct dither level in the test signal.

3.8
Output word length
maximum number of bits in a digital word transmitted by the digital output of an EUT for which all bits have values that vary over time

3.9
Positive digital full scale
highest value code in the digital representation

NOTE In 24-bit two's complement, this code is $\mathtt{7FFFFF}_{16}$.

3.10
Negative digital full scale
lowest value code in the digital representation

NOTE In 24-bit two's complement, this code is 800000_{16}.

3.11
Least significant bit
LSB
smallest change that can be represented in a digital word

3.12
Levels

3.12.1
Full-scale level
level of a dc-free 997-Hz sine wave whose undithered positive peak value is positive digital full scale, leaving the code corresponding to negative digital full scale unused

NOTE 1 Full-scale level may be reported as “1,0 FS” or “0 dBFS”.

NOTE 2 All test signals defined in this document are dithered as described in section 5.1.3. For dithered signals at full-scale level, the level should be reduced so that the dithered positive peak value just reaches positive digital full scale.

3.12.2
Signal level, FS
rms signal level expressed as a fraction of full-scale level (see 3.12.1)

NOTE Levels reported in FS are always rms. It is invalid to use FS for non-rms levels.

3.12.3
Decibels full scale
dBFS
rms level expressed in decibels relative to full-scale level. dBFS is computed from FS using the relation

$$\textbf{Level in dBFS} = \mathbf{20\,lg(level\ in\ FS)}$$

NOTE 1 Levels reported in dBFS are always rms. It is invalid to use dBFS for non-rms levels.

NOTE 2 Because the definition of full scale is based on a sine wave, the level of signals with a crest factor lower than that of a sine wave may exceed 0 dBFS. In particular, square-wave signals may reach +3,01 dBFS. Signals above 0 dBFS are not recommended, because tilt or overshoot introduced by signal processing may cause clipping.

3.13
Dither signal
noise signal added to a signal before quantization to linearize the quantizer

NOTE The correct dither has a triangular probability density function, with a maximum level of ±1 LSB.

3.14
Digital zero
signal consisting only of dither at the appropriate level for the word length

3.15
Jitter susceptibility
effect on EUT performance of jitter present on the signal or clock reference inputs

3.16
Crest factor
ratio of the absolute peak level of a signal to its time-averaged rms level

3.17
Code
value of a digital audio sample. A complete set of codes describes the entire dynamic range of the coded audio

4 Measurement conditions

4.1 Environmental

All tests in this standard shall be performed at an ambient temperature of (23 ± 5) °C unless otherwise specified by the manufacturer. When specifications are listed for an EUT with a temperature range of operation, these specifications are assumed to be valid over the entire range, and shall be verified at the rated limits.

4.2 Power supply

4.2.1 EUTs powered directly from the mains

Power-line (mains) voltage shall be set within 2 % of the nominal value listed on the EUT. If a range of values is given, the specifications are assumed to be valid over the range, and shall be verified at the rated limits.

Power-line frequency shall be set within 1 % of the nominal value listed on the EUT. If a range of values is given, the specifications are assumed to be valid over the range, and shall be verified at the rated limits.

4.2.2 EUTs powered from the mains via an external power supply

The external power supply shipped with the EUT shall be used. The power line conditions listed in 4.2.1 shall apply.

4.2.3 EUTs using battery power

One of the following power sources shall be used:

- A battery of the type required or recommended for the EUT; or
- A dc power supply with a peak-to-peak ripple content of less than 0,5 % of the nominal dc supply voltage.

4.3 Bandwidth

An EUT using a 44,1 kHz or 48 kHz sampling frequency shall have an upper band-edge frequency of 20 kHz.

An EUT which uses a sampling frequency other than 44,1 kHz or 48 kHz may have an upper band-edge frequency defined by the manufacturer. This frequency, if different from 20 kHz, shall be stated in the manufacturer's specifications.

If an upper band-edge frequency specified by the EUT manufacturer is used, it shall be the same for all tests.

4.4 EUT settings

The equipment controls shall be set to their normal operating positions, except where noted. The settings of the switches and controls of the EUT shall be consistent for all measurements in this standard.

The normal source impedance for terminating an analogue input shall be 50 Ω, unless otherwise specified by the manufacturer. Measurements which use an input termination different from 50 Ω shall be so noted.

If any emphasis is provided, it shall be set to the manufacturer's recommended position. This setting shall be clearly indicated in the specifications. If a recommended position is not stated by the manufacturer, emphasis shall not be used. If desired, some measurements may be repeated with other settings. Such measurements shall be clearly indicated as supplementary, and shall be reported in addition to the results of the same tests performed using the recommended position.

If the EUT includes limiters, compressors, or other means to affect signal dynamics, they shall be disabled. The results of any additional tests on these features shall be reported separately.

4.5 EUT preconditioning

The EUT shall be connected under normal operating conditions for the preconditioning period specified by the EUT manufacturer before any measurements are performed. This condition is intended to allow the EUT to stabilize. If no preconditioning period is specified by the manufacturer, a 5 minute period shall be assumed. The EUT manufacturer should state if operational requirements preclude preconditioning.

If power to the EUT is interrupted during the measurements, enough time shall be allowed for the EUT to stabilize again before resuming.

5 Measurement equipment

5.1 Signal generator

5.1.1 Overview

Depending on the nature of the EUT, either an analogue or digital generator may be required. This clause defines their characteristics.

5.1.2 Output impedance

Unless otherwise specified, the analogue signal generator shall have an output impedance of 50 Ω or less. The digital signal generator shall have an output impedance appropriate for the physical layer to which it connects.

5.1.3 Dither signal

For measurements where the stimulus originates in the digital domain, the test signals shall be dithered. When the stimulus is applied in the analogue domain, the inclusion of dither shall be the responsibility of the EUT manufacturer, and shall not be supplemented during testing.

The dither signal shall be a random or pseudorandom sequence having a triangular probability density function and a peak level of ±1 LSB of the digital audio input word length (see 3.7). The dither power shall be constant per unit bandwidth to the upper band-edge frequency.

> NOTE Such a dither signal may be generated by subtracting two random or pseudorandom numbers whose values are uniformly distributed between 0 LSB and 1 LSB.

If a pseudorandom signal is used, its repetition interval shall be at least as long as the measurement interval of the meters used. It shall be generated from a maximum-length polynomial of greater than two terms. The samples should be spaced by several shifts of the polynomial to reduce sample-to-sample correlation. Care should be taken to avoid polynomials that create large short-term energy variation, or that produce substantial asymmetry in the probability distribution when low-pass filtered.

5.1.4 Accuracy

The analogue and digital signal generators shall provide the ability to set frequency with an accuracy of at least ±0,05 %.

The analogue signal generator shall have a level accuracy of at least ±(0,2 dB + 3 μV) at 1 kHz, and at least ±(0,3 dB + 3 μV) from 20 Hz to the upper band-edge frequency.

The digital signal generator shall have a level accuracy of at least at least ±(0,01 dB + 0,5 LSB) from 20 Hz to the upper band-edge frequency.

5.2 Signal analyzer

5.2.1 Overview

Depending on the nature of the EUT, either an analogue or digital analyzer may be required. This clause defines their characteristics.

Some measurements require one or more standard filters (the filter requirements are noted in each measurement). The standard filters are described in the following clauses.

5.2.2 Input impedance

Unless otherwise specified, the analogue signal analyzer shall have an input impedance of 100 kΩ or greater. The input capacitance shall be 500 pF or less. The digital signal analyzer shall have an input impedance appropriate for the physical layer to which it connects.

5.2.3 Level meter

Level meters shall be true root-mean-square (rms) responding devices with an accuracy of at least ±0,25 dB at 1 kHz. This accuracy shall be maintained for signals having a crest factor of 5 or less. RMS-calibrated average or peak-responding devices are not acceptable. If the accuracy specification in 5.2.11 requires better performance than this, then 5.2.11 shall take precedence.

Level meters shall integrate the signal for a minimum of 25 ms to ensure that enough codes are exercised in the EUT. At low frequencies the integration time shall be increased to ensure that at least one full cycle of the test signal is measured.

5.2.4 Phase meter

Phase meters shall have a minimum accuracy of ±1° at 1 kHz, and ±5° at 20 kHz. This accuracy shall be maintained for any signal whose rms sum of harmonic, non-harmonic, and spurious components does not exceed 1 % of the total signal level.

5.2.5 Standard low-pass filter

For an upper band-edge frequency of 20 kHz, the standard low-pass filter shall have the following characteristics:

- Passband response deviation: ≤ ±0,1 dB, 20 Hz ≤ f ≤ 20 kHz
- Stopband attenuation: ≥ 60 dB, f > 24 kHz

For an upper band-edge frequency other than 20 kHz, the standard low-pass filter shall have the following characteristics:

- Passband response deviation: ≤ ±0,1 dB, 20 Hz ≤ f ≤ upper band-edge frequency
- Stopband attenuation: ≥ 60 dB, f > (sampling frequency – upper band-edge frequency)

5.2.6 Standard high-pass filter

For an upper band-edge frequency of 20 kHz, the high-pass filter shall have the following characteristics:

- Passband response deviation: ≤ ±0,5 dB, 26 kHz ≤ f ≤ 200 kHz
- Stopband attenuation: ≥ 40 dB, 20 Hz ≤ f < 20 kHz

For an upper band-edge frequency other than 20 kHz, the high-pass filter shall have the following characteristics:

- Passband response deviation: ≤ ±0,5 dB, 1.3 times upper band-edge frequency ≤ f ≤ 200 kHz
- Stopband attenuation: ≥ 40 dB, 20 Hz ≤ f ≤ upper band-edge frequency

5.2.7 Standard weighting filter

The standard weighting filter shall conform to ITU-R BS.468-4, with an additional gain of -5,63 dB. (This puts the unity gain frequency at 2 kHz.) This gain-shifted response is known as the “CCIR-RMS” filter. Table 1 shows the required response and tolerance of the filter.

Measurements made with this filter shall be denoted “[unit] CCIR-RMS”, where [unit] is the appropriate unit for the measurement.

See also: 5.5.2 for more information on units.

Table 1 - CCIR-RMS weighting filter response

Frequency (Hz)	Gain (dB)	Tolerance (dB)
31,5	-35,5	±2,0
63	-29,5	±1,4
100	-25,4	±1,0
200	-19,4	±0,85
400	-13,4	±0,70
800	-7,5	±0,55
1 000	-5,6	±0,50
2 000	0,0	±0,50
3 150	3,4	±0,50
4 000	4,9	±0,50
5 000	6,1	±0,50
6 300	6,6	±0,01
7 100	6,4	±0,20
8 000	5,8	±0,40
9 000	4,5	±0,60
10 000	2,5	±0,80
12 500	-5,6	±1,2
14 000	-10,9	±1,4
16 000	-17,3	±1,6
20 000	-27,8	±2,0
31 500	-48,3	+2,8; -∞

5.2.8 Standard notch filter

The standard notch filter shall have a quality factor Q of at least 1,2 and not more than 3, where Q is defined as the ratio of the center frequency to the difference between the -3 dB frequencies. Multistage notch filters are acceptable if their combined Q measures within these limits using this definition.

5.2.9 Standard band-pass filter

The standard band-pass filter shall conform to the class 1 or class 2 response limits described in IEC 61260-1. The attenuation shall be at least 30 dB one octave away from the filter center frequency, and at least 60 dB three octaves away.

NOTE A filter complying with ANSI S1.11-2004 Class 2 requirements with a bandwidth designator *b* of 2 (that is, a half-octave filter) easily meets this requirement.

If the EUT is very noisy, certain measurements may benefit from the use of a band-pass filter centered on the test frequency to achieve accurate results. Where such measurements are made using a band-pass filter, this shall be noted.

See also: 5.2.11 for more information on the required gain accuracy of the standard band-pass filters.

5.2.10 Frequency-domain band-pass filters

NOTE 1 Frequency-domain band-pass filters are able to isolate tones better than conventional time-domain filters. They are also known as *window-width filters*, because it is possible to create band-pass filters with them whose bandwidth is equal to the number of bins in the main lobe of the transform window.

NOTE 2 Typically, frequency-domain filters are implemented by Fast Fourier Transform (FFT), an analysis method also used by FFT analyzers. A measurement that specifies a frequency-domain filter therefore implies the use of an FFT analyzer.

Frequency-domain band-pass filters are specified for certain measurements for which a band-pass filter with a very narrow bandwidth is needed to obtain accurate results. They may also be used to provide better noise rejection than the standard band-pass filter for measurements which do not depend on the specific shape of the band-pass filter used. The filter must still meet the gain accuracy requirements of 5.2.11.

The combination of sample rate, acquisition length, and window type shall be sufficient to achieve the bandwidth needed for the measurement. The bandwidth shall be specified in each measurement.

See also: Annex B for more information.

5.2.11 Accuracy

For all measurements in this standard, unless otherwise specified, the measurement equipment shall have an accuracy in the parameter being measured of at least three times better than the specification being verified. Table 2 contains some examples.

Table 2 - Measurement accuracy - Examples

Parameter	EUT figure	Measurement equipment figure	Worst-case error
Noise, 20 Hz to 20 kHz, rms	10 μV	<3,3 μV	0,53 μV
THD+N ratio, 20 kHz BW	-95 dB	<-104,5 dB[1]	0,45 dB
Gain flatness, 20 Hz to 20 kHz	±0,03 dB	<±0,01 dB	±0,01 dB

NOTE ([1]) This figure is computed either by converting to a linear ratio, dividing by 3, and converting back to dB; or by noting that 20 lg(1/3) = -9,54 dB.

5.3 Scope of measurements

Some of the methods in the standard specify a range and density of settings over which to make the measurement. For example, the frequency response measurement in 6.2.3 states:

"The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of no more than one octave."

This should be taken as the minimum requirement to achieve acceptable results. It is permissible to use a wider range, or a higher point density, if desired.

See also: 5.4 for a table of standard octave-spaced and third-octave-spaced test frequencies.

Measurements indicated "**Optional**" are not required. If they are made, they shall be performed in addition to the recommended measurements, not instead of them.

5.4 Standard frequencies

The following table lists standard frequencies used in some measurements (see IEC 61260-1). Frequencies shown in bold shall be used for octave-spaced measurements. All the frequencies shall be used for third-octave-spaced measurements.

Table 3 - Standard Frequencies

Octave Number	Nominal Frequencies (Hz)		
1	20	25	**31,5**
2	40	50	**63**
3	80	100	**125**
4	160	200	**250**
5	315	400	**500**
6	630	800	**1 000**
7	1 250	1 600	**2 000**
8	2 500	3 150	**4 000**
9	5 000	6 300	**8 000**
10	10 000	12 500	**16 000**
11	20 000	-	-

5.5 Documentation of results

5.5.1 General

Measurement results shall be presented in the form outlined in each measurement method in this standard. The intent is to create a uniform presentation of data, allowing meaningful comparisons among competing equipment.

The settings of all controls which can affect the measured performance of the equipment shall be stated in the reporting of the test results.

When reporting results measured in accordance with the methods presented in this standard, there should be a statement to that effect in the text or footnotes of the data. For example, this might read: “All results measured in accordance with AES17-2015”.

5.5.2 Recommended units

Some values can be stated in more than one unit. The units listed in table 4 should be used.

Table 4 - Recommended units

Parameter	EUT type			
	Analogue-to-Analogue	Analogue-to-Digital	Digital-to-Analogue	Digital-to-Digital
Input level	dBu	dBu	dBFS	dBFS
Output level	dBu	dBFS	dBu	dBFS
Gain	dB	-	-	dB
Phase	degrees	degrees	degrees	degrees
Group delay	seconds	seconds	seconds	seconds
Frequency	Hz	Hz	Hz	Hz
Ratio (THD+N, IMD, etc.)	dB	dB	dB	dB

NOTE Gain is not measured for cross-domain (analogue-to-digital and digital-to-analogue) EUTs. For analogue-to-digital devices, the analogue input to achieve a full-scale digital output is measured. For digital-to-analogue EUTs, the analogue output for a full-scale digital input is measured. See 6.2.1 for more information.

5.5.3 Graph results and equivalent text

Some of the methods in the standard require the results to be presented as a graph. However, sometimes it may not be possible to generate or render graphs. The results may be presented in the following form:

"±[Range][Y-axis unit] from [Low][X-axis unit] to [High][X-axis unit]".

The [y-axis unit] is the unit of the measured value. The [x-axis unit] is the unit of the parameter that was varied. For example, the frequency response result (see 6.2.3) may be presented as:

±0,03 dB from 20 Hz to 20 kHz.

See also: 5.5.2 for more information on units.

6 Measurement methods

6.1 Overview

The measurement methods in this clause shall apply to all EUTs, regardless of their input and output domains. Before making measurements, the measurement conditions described in clause 4 should be established.

For the preferred units to use for each measurement, see the table in 5.5.2.

6.2 Linear characteristics

6.2.1 Maximum input level

This test measures the maximum level that can be applied to the input of the EUT before excessive distortion occurs.

NOTE 1 This measurement is not intended for EUTs that mute the input.

NOTE 2 This measurement is typically used for EUTs with analogue inputs, but it may also be useful for measuring digital signal paths with non-flat frequency response or imperfect gain structure.

Any gain controls in the EUT shall be set to their minimum operational value. If the EUT contains pre-emphasis or de-emphasis filters, the measurement results shall be reported separately for each setting of the filter, including off.

The test signal shall be a 997 Hz sine wave. The analyzer shall include the standard low-pass filter (see 5.2.5). One of the following methods shall be used to determine the maximum input level:

a. **THD+N ratio method.** The THD+N ratio at the EUT output is measured (see 6.3.1). The level of the test signal is increased until the THD+N ratio reaches -40 dB.

b. **Compression method.** The level at the EUT output is measured. The level of the test signal is increased until the corresponding increase in EUT output level is 0,3 dB lower than the increase in the test level.

The maximum input level shall be the level of the test signal at this point.

NOTE The THD+N ratio method is recommended for EUTs that hard clip, that is, EUTs that have an essentially linear transfer function to the point where the maximum input code is reached. The compression method is recommended for EUTs that soft clip, that is, EUTs whose distortion increases in a controlled manner as the input level increases.

Optional: The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of no more than one octave. The maximum input level is measured at each frequency, according to the chosen method. The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and maximum input level on the y-axis.

See also: 5.5.3 for presenting graphical data as text.

See also: 6.6.8 for determining EUT overload behaviour.

6.2.2 Gain

This test measures the ratio of EUT output level to EUT input level at 997 Hz.

NOTE This measurement is only applicable for analogue-to-analogue or digital-to-digital EUTs.

The test signal shall be a 997 Hz sine wave at -20 dB relative to the maximum input level (see 6.2.1). The analyzer shall include the standard low-pass filter (see 5.2.5). The gain shall be the ratio of the output level of the EUT to the level of the test signal. It shall be reported in dB.

See also: 5.2.9 if the EUT is very noisy.

6.2.3 Frequency response

This test measures the variation of EUT gain with frequency.

The test signal shall be a sine wave at -20 dB relative to the maximum input level (see 6.2.1). The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of no more than one octave. One of the test frequencies shall be 997 Hz. The analyzer shall include the standard low-pass filter (see 5.2.5). The output level of the EUT shall be measured at each frequency.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and the level relative to the level at 997 Hz on the y-axis.

See also: 5.5.3 for presenting graphical data as text.

See also: 5.2.9 if the EUT is very noisy.

See also: Annex A for alternative methods that offer quicker measurements more suitable for production test.

6.2.4 Gain matching between channels

This test measures the matching of gain between channels in a multichannel EUT.

The test signal shall be a 997 Hz sine wave at -20 dB relative to the maximum input level (see 6.2.1). The test signal shall be preferably applied to all EUT channels simultaneously. The analyzer shall include the standard low-pass filter (see 5.2.5). The output level of each EUT channel shall be measured. The gain matching shall be the greatest difference in levels, expressed in dB.

If the EUT includes a ganged gain control, the output level of each EUT channel shall be measured at multiple positions over the full range of the gain control. The gain-matching shall be the greatest difference in levels measured according to the method above.

> NOTE It may be necessary to reduce the level of the test signal if the gain of the EUT is high enough to cause clipping at certain settings of the gain control.
>
> **See also:** 5.2.9 if the EUT is very noisy.

6.2.5 Gain stability

This test measures the variation of EUT gain over time.

The test signal shall be a 997 Hz sine wave at -6 dB relative to the maximum input level (see 6.2.1). The analyzer shall include the standard low-pass filter (see 5.2.5). The output level of the EUT is measured for a period of at least one hour immediately following the preconditioning period (see 4.5) at intervals of no more than one minute. The gain stability is the difference between the maximum measured level and the minimum measured level, expressed in dB.

> **See also**: 5.2.9 if the EUT is very noisy.

6.2.6 Maximum output level

This test measures the maximum level attainable at the output of the EUT before excessive distortion occurs.

> NOTE This measurement is typically used for EUTs with analogue outputs, but may also be useful for measuring digital signal paths with non-flat frequency response or imperfect gain structure.

Any gain controls in the EUT shall be set to maximize output level without causing input saturation. If the EUT contains pre-emphasis or de-emphasis filters, the measurement results shall be reported separately for each setting of the filter, including off.

The test signal shall be a 997 Hz sine wave. The analyzer shall include the standard low-pass filter (see 5.2.5). One of the following methods shall be used to determine the maximum output level:

a. **THD+N ratio method.** The THD+N ratio at the EUT output is measured (see 6.3.1). The level of the test signal is increased until the THD+N ratio reaches -40 dB.

b. **Compression method.** The level at the EUT output is measured. The level of the test signal is increased until the corresponding increase in EUT output level is 0,3 dB lower than the increase in the test level.

The maximum output level shall be the EUT output level at this point.

> NOTE The THD+N ratio method is recommended for EUTs that hard clip, that is, EUTs that have an essentially linear transfer function to the point where the maximum output code is reached. The compression method is recommended for EUTs that soft clip, that is, EUTs whose distortion increases in a controlled manner as the output level increases.

Optional: The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of no more than one octave. The maximum output level shall be measured at each frequency, according to the method used above. The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and maximum output level on the y-axis.

See also: 5.5.3 for presenting graphical data as text.

6.2.7 Inter-channel phase response

This test measures the variation of phase between channels in a multichannel EUT.

NOTE Because EUT inter-channel phase differences are typically quite small, the inter-channel phase response can usually be measured using sine waves at discrete frequencies. It is also possible to derive the inter-channel phase response from the phase response (see 6.8.3) by subtraction of one channel response from another.

If the EUT contains pre-emphasis or de-emphasis filters, the measurement results shall be reported separately for each setting of the filter, including off.

The test signal shall be a sine wave at -20 dB relative to the maximum input level (see 6.2.1). The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of no more than one octave. One channel of the EUT shall be designated the reference, and the phase at the output of each channel of the EUT relative to the phase on the reference channel shall be measured.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and the phase difference on the y-axis. There shall be one graph trace for each channel, apart from the reference channel, which is omitted.

See also: 5.5.3 for presenting graphical data as text.

See also: 5.2.9 if the EUT is very noisy.

See also: annex A for alternative methods that offer quicker measurements more suitable for production test.

6.2.8 Polarity

This test measures whether the EUT is non-inverting or inverting.

Several methods can be used to measure polarity. For all the following methods, the signal level shall be -20 dB relative to the maximum input level (see 6.2.1).

a. **Time domain, burst signal**. A burst signal is applied to the EUT. Typically, a 997 Hz sine wave is used, toggled at positive-going zero crossings, on for 5 cycles, and off for 20 cycles. If the output of the EUT is positive-going at the onset of the burst, the EUT is non-inverting.

b. **Time domain, continuous signal**. An asymmetrical signal is applied to the EUT. Typically, a 997 Hz sine wave is used, summed with a 1 994 Hz sine wave whose phase is shifted by minus a quarter cycle relative to the main tone. If the output of the EUT has the single peak positive, the EUT is non-inverting.

c. **Impulse response**. The impulse response of the EUT is measured. If the peak of the impulse response is positive, the EUT is non-inverting.

The polarity shall be reported as either “non-inverting” or “inverting”.

See also: annex A for measuring the impulse response.

6.3 Non-linear characteristics

6.3.1 Overview: Performing a THD+N ratio measurement

The following clauses describe total harmonic distortion plus noise (THD+N) ratio measurements. In this clause, the general measurement method is described.

The test signal shall be a sine wave. The analyzer shall include the standard low-pass filter (see 5.2.5). The total signal level at the EUT output is measured and recorded. The fundamental component of the test signal present in the output shall be removed with the standard notch filter (see 5.2.8), and the unweighted level of the residual shall be measured. The THD+N ratio shall be the difference between the residual level and the total signal level, expressed in dB.

6.3.2 THD+N ratio vs frequency

This test measures the total harmonic distortion and noise ratio of the EUT as the test signal frequency is varied.

The test signal level shall be a sine wave at -1 dB relative to the maximum input level (see 6.2.1). The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of not more than one octave. The THD+N ratio shall be measured at each step (see 6.3.1). The test shall be then repeated with a test signal level of -20 dB relative to the maximum input level, according to the method above.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and THD+N ratio on the y-axis.

See also: 5.5.3 for presenting graphical data as text.

6.3.3 THD+N ratio vs input level

This test measures the total harmonic distortion and noise ratio of the EUT as the test signal level is varied.

The test signal shall be a 997 Hz sine wave. The level of the test signal shall be varied from the maximum input level (see 6.2.1) to -80 dB relative to the maximum input level, in steps no larger than 10 dB. The THD+N ratio shall be measured at each step (see 6.3.1). The results shall be presented as a graph with test signal level relative to the maximum input level on the x-axis, and THD+N ratio on the y-axis.

See also: 5.5.3 for presenting graphical data as text.

6.3.4 THD+N ratio vs output level

This test measures the total harmonic distortion and noise ratio of the EUT as the output signal level is varied.

The test signal shall be a 997 Hz sine wave. The level of the test signal shall be varied from the level needed to produce the maximum output level (see 6.2.6) to -80 dB relative to this level, in steps no larger than 10 dB. The total output signal level and THD+N ratio shall be measured at each step (see 6.3.1). The results shall be presented as a graph with total output signal level relative to the maximum output level on the x-axis, and THD+N ratio on the y-axis.

See also: 5.5.3 for presenting graphical data as text.

6.3.5 Difference-frequency distortion

This test measures the intermodulation distortion (IMD) ratio of the EUT when driven by two tones close in frequency near the upper band-edge.

NOTE used to be called Intermodulation distortion ratio (close-tone).

The test signal consists of a sum of two tones of equal level. The tones shall be 2 kHz apart in frequency. For EUTs whose upper band-edge frequency is at least 20 kHz, the upper tone frequency shall be 20 kHz. For EUTs whose upper band-edge frequency is below 20 kHz, the upper tone frequency shall be the upper band-edge frequency. The peak level of the test signal shall equal the peak level of a sine wave at the maximum input level (see 6.2.1).

NOTE Typically, this means that the level of each tone is 0,5× the maximum input level.

The output of the EUT shall be filtered with a frequency-domain band-pass filter (see 5.2.10) whose bandwidth shall be 500 Hz.

NOTE The standard band-pass filter (see 5.2.9) has inadequate selectivity for this measurement. However, a time-domain filter may be used provided that its attenuation at 2 kHz from the center frequency is at least 100 dB.

The band-pass filtered rms level at the output of the EUT shall be measured at the following center frequencies:

- For EUTs with an upper band-edge frequency of 20 kHz: 2 kHz (the second-order product), 16 kHz (the lower third-order product), 18 kHz (the lower fundamental), and 22 kHz (the upper third-order product).
- For EUTs with a different upper band-edge frequency: 2 kHz (the second-order product), 4 kHz below the upper band-edge frequency (the lower third-order product), 2 kHz below the upper band-edge frequency (the lower fundamental), and 2 kHz above the band-edge frequency (the upper third-order product).

The close-tone difference-frequency distortion ratio shall be the difference between the total rms level and the second-order and third-order products to the rms level of the lower fundamental, expressed in dB.

NOTE This measurement may be accomplished in one step by rms summing the output of multiple frequency-domain band-pass filters on the FFT of a single acquisition.

6.3.6 Modulation distortion

This test measures the intermodulation distortion (IMD) ratio of the EUT when driven by two tones far apart in frequency.

NOTE Used to be called Intermodulation distortion ratio (spread-tone).

The test signal consists of a sum of two tones. The upper tone shall be at 7 993 Hz. The lower tone shall be at 41 Hz, with a level four times that of the upper tone. The peak level of the test signal shall equal the peak level of a sine wave at the maximum input level (see 6.2.1).

NOTE Typically, this means that the level of the lower tone is 0,8× the maximum input level, and the level of the upper tone is 0,2× the maximum input level.

The output of the EUT shall be filtered with a frequency-domain band-pass filter (see 5.2.10) whose bandwidth shall be no more than 40 Hz. The band-pass filtered rms level at the output of the EUT shall be measured at the following center frequencies: 7 952 Hz (the lower modulation sideband), 7 993 Hz (the upper fundamental), and 8 034 Hz (the upper modulation sideband).

NOTE The standard band-pass filter (see 5.2.9) has inadequate selectivity for this measurement. However, it is still possible to perform this measurement in the time domain by demodulating the sidebands around the upper tone and using appropriate filtering.

The spread-tone IMD ratio shall be the ratio of the total rms level of the two modulation sidebands to the RMS level of the upper fundamental.

NOTE This measurement may be accomplished in one step by rms summing the output of multiple frequency-domain band-pass filters on the FFT of a single acquisition.

6.3.7 Gain non-linearity

This test measures the variation of the gain of the EUT as the input signal level is varied.

NOTE This characteristic is also known as linearity.

The test signal shall be a sine wave at 997 Hz. The output of the EUT shall be filtered with a band-pass filter with center frequency 997 Hz. The band-pass filter shall be either the standard band-pass filter (see 5.2.9) or a frequency-domain band-pass filter (see 5.2.10). If a frequency-domain band-pass filter is used, its bandwidth shall be no more than 500 Hz.

The idle channel noise level (see 6.4.2) shall first be measured and recorded.

Next, the level of the test signal shall be set to -5 dB relative to the maximum input level (see 6.2.1). This is the *reference input level*. The rms level at the output of the band-pass filter shall be measured. This is the *reference output level*.

The level of the test tone shall then be reduced in steps no larger than 5 dB. At each step, the rms level at the output of the band-pass filter shall be measured. This iteration shall continue until the rms level at the output of the band-pass filter is within 5 dB of the idle channel noise level.

For each step, the deviation of the output level of the band-pass filter from the ideal level shall be computed as:

$$\textit{Deviation from ideal} = \text{Output level} - \text{Ideal output level.}$$

The ideal output level is the output level of a perfectly linear EUT. This shall be computed as:

$$\textit{Ideal output level} = \textit{Reference output level} \times \text{Test level} - \text{Reference test level.}$$

The results shall be plotted as a graph with the test level in a dB-based unit on the x-axis, and the deviation of the output level of the band-pass filter from the ideal in a dB-based unit on the y-axis.

> **See also:** 5.5.3 for presenting graphical data as text.

6.4 Noise

6.4.1 Dynamic range

This test measures the ratio of the full-scale level at the output of the EUT to the weighted noise and distortion level in the presence of a low-level signal. It includes all harmonic, inharmonic, and noise components.

> NOTE This characteristic is also known as *signal-to-noise ratio*.

The test signal shall be a 997 Hz sine wave with a level of -60 dB relative to the maximum input level (see 6.2.1). The output of the EUT shall be filtered with the standard low-pass filter (see 5.2.5) and the standard notch filter (see 5.2.8), whose center frequency is set to 997 Hz. The output of the standard notch filter shall be filtered with the standard weighting filter (see 5.2.7). The rms level of the final filter output shall be measured. The dynamic range shall be the ratio of the maximum output level (see 6.2.6) to this measured level. It is reported as dB CCIR-RMS.

6.4.2 Idle channel noise level

This test measures the weighted output of the EUT when driven with no signal.

For EUTs with an analogue input, the input shall be short-circuited. For EUTs with a digital input, the test signal shall be digital zero (see 3.14).

The output of the EUT shall be filtered with the standard low-pass filter (see 5.2.5) and the standard weighting filter (see 5.2.7). The idle channel noise level shall be the rms level of the final filter output. For EUTs with an analogue output, the idle channel noise level shall be measured in dB(V), and reported as dB(V) CCIR-RMS. For EUTs with a digital output, the idle channel noise level shall be measured in dBFS, and reported as dBFS CCIR-RMS.

6.4.3 Out-of-band noise and spurious level

This test measures the total output of the EUT above the folding frequency when driven with no signal. This measurement only applies to EUTs with analogue outputs, or EUTs with digital outputs whose internal sampling frequency is lower than its output sampling frequency.

> NOTE The internal folding frequency of the EUT must be known for this test. If necessary, it can be determined by observing image products at the EUT output. The internal folding frequency is the frequency of the lowest frequency test tone which coincides with its own image. The internal sampling frequency is twice the internal folding frequency.

For EUTs with an analogue input, the input shall be short-circuited. For EUTs with a digital input, the test signal shall be digital zero (see 3.14).

The output of the EUT shall be filtered with the standard high-pass filter (see 5.2.6). The out-of-band noise level shall be the level at the output of the high-pass filter.

Optional: The spectrum of the EUT output under the test conditions may be computed and presented as a graph with frequency on the x-axis, shown logarithmically, and the level on the y-axis.

6.4.4 Signal modulation noise

NOTE The characteristic to be specified consists of amplitude modulation sidebands.

The test signal shall be a sine wave at 0,4999 times the upper band-edge frequency. The amplitude shall be 5 dB below the full-scale input or output level of the system, as appropriate. The output signal shall be full-wave rectified. The resulting signal shall be measured in third-octave bands from 50 Hz to 500 Hz with a level meter. Its level shall be expressed in dB relative to the level of the original sine wave. The results shall be displayed graphically.

NOTE This measurement may be performed with most commercial IM distortion analyzers plus a third-octave band analyzer.

6.4.5 Low-level noise modulation

This test measures the variation in the noise floor of the EUT at low signal levels.

NOTE This test can reveal problems such as inadequate dithering and poor linearity.

The idle channel noise level (see 6.4.2) shall first be measured and recorded.

The test signal shall be a sine wave at 41 Hz. The output of the EUT shall be filtered in one of the following ways:

- With the standard notch filter whose center frequency is set to 41 Hz, cascaded with the standard band-pass filter (see 5.2.9).
- With a frequency-domain band-pass filter (see 5.2.10) whose bandwidth shall be no more than 100 Hz.

The center frequency of the band-pass filter shall be each of the standard third-octave frequencies (see 5.4) from 200 Hz to the upper band-edge frequency in turn.

The test signal level shall be set to -40 dB relative to the EUT maximum input level (see 6.2.1). The level at the output of the band-pass filter at each center frequency shall be measured and recorded. The test signal level shall then be lowered by 10 dB, and the measurement repeated. This iteration shall continue until the test signal level is below the idle channel noise level.

The low-level noise modulation ratio at each frequency shall be the highest level recorded at that frequency divided by the lowest level recorded at that frequency.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and the low-level noise modulation ratio on the y-axis. Alternatively, the largest ratio, irrespective of frequency, may be quoted as a scalar result.

See also: 5.5.3 for presenting graphical data as text.

6.5 Interference and crosstalk

6.5.1 Power line (mains) related products

This test measures the level of products in the EUT output due to the mains power supply.

NOTE The local mains frequency (typically 50 Hz or 60 Hz) must be known for this test. If it is not known, it should be determined in a safe manner.

For EUTs with an analogue input, the input shall be short-circuited. For EUTs with a digital input, the test signal shall be digital zero (see 3.14). The output of the EUT shall be filtered with a band-pass filter. The band-pass filter shall be either the standard band-pass filter (see 5.2.9) or a frequency-domain band-pass filter (see 5.2.10). If a frequency-domain band-pass filter is used, its bandwidth shall be no more than half the mains frequency.

The band-pass center frequency shall be an integer multiple M of the mains frequency. The output of the band-pass filter shall be measured for values of M from 1 to 5 in turn. The power line level shall be the rms sum of these levels.

NOTE This measurement may be accomplished in one step by rms summing the output of multiple frequency-domain band-pass filters on the FFT of a single acquisition.

6.5.2 Inter-channel crosstalk ratio

This test measures the degree to which signal leaks from one channel of a multi-channel EUT into another channel.

NOTE This characteristic is also known as *separation*.

One channel of the EUT shall be designated the *driven channel*. The remaining channels shall be designated *undriven channels*. For EUTs with analogue inputs, the undriven channels shall be terminated with the normal source impedance (see 4.4). For EUTs with a digital input, the test signal on the undriven channels shall be digital zero (see 3.14).

The test signal on the driven channel shall be a sine wave at -20 dB relative to the maximum input level (see 6.2.1). The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of no more than one octave. The analyzer shall use the standard low-pass filter (see 5.2.5). The output of each undriven EUT output channel shall be measured. The inter-channel crosstalk ratio for each undriven channel shall be the ratio of the level on the undriven channel to the level on the driven channel, expressed in dB.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and the crosstalk ratio on the y-axis.

See also: 5.5.3 for presenting graphical data as text.

6.5.3 Non-linear crosstalk: high frequency

This test measures the non-linear crosstalk of high frequency signals in the channels of a multichannel EUT. Since the method involves overdriving the analogue circuits of the EUT, it can only be applied to EUTs with analogue inputs. Even-order and odd-order non-linear crosstalk, or cross intermodulation, are measured separately.

One channel of the EUT shall be designated the *channel under test*. The remaining channels shall be designated *interfering channels*. For EUTs with analogue inputs, the interfering channels shall be terminated with the normal source impedance (see 4.4). For EUTs with a digital input, the test signal on the interfering channels shall be digital zero (see 3.14).

The test signal in the channel under test shall be a -20 dBFS sine wave at the upper band-edge frequency. The test signal in the interfering channels shall be a +3 dBFS sine wave at a frequency 3 kHz below the upper band-edge frequency, applied simultaneously to all channels. The analyzer shall include the standard band-pass filter (see 5.2.9). Two separate analyses are made:

- **Even-order non-linear crosstalk**. The band-pass filter has a center frequency of 3 kHz. The even-order non-linear crosstalk shall be the ratio of the rms band-pass filtered level in the channel under test to the total rms level in the channel under test, expressed in dB.

- **Odd-order non-linear crosstalk**. The band-pass filter has a center frequency 6 kHz below the upper band-edge frequency. The odd-order non-linear crosstalk shall be the ratio of the rms band-pass filtered level in the channel under test to the total rms level in the channel under test, expressed in dB.

The measurements shall be performed for each channel of the EUT. Even-order and odd-order non-linear crosstalk shall be reported separately for each channel. Alternatively, the worst even-order and odd-order results across the channels shall be reported.

6.5.4 Non-linear crosstalk: low frequency

This test measures the non-linear crosstalk, or cross intermodulation, of low frequency signals in the channels of a multichannel EUT. Since the method involves overdriving the analogue circuits of the EUT, it can only be applied to EUTs with analogue inputs.

One channel of the EUT shall be designated the *channel under test*. The remaining channels shall be designated *interfering channels*. The test signal in the channel under test shall be a -20 dBFS sine wave at half the upper band-edge frequency. The test signal in the interfering channels shall be a +3 dBFS sine wave at 40 Hz.

The output of the EUT shall be filtered with a frequency-domain band-pass filter (see 5.2.10) whose bandwidth shall be no more than 40 Hz.

> NOTE The standard band-pass filter (see 5.2.9) has inadequate selectivity for this measurement. However, it is still possible to perform this measurement in the time domain by demodulating the sidebands around the upper tone and using appropriate filtering.

The band-pass filtered rms level at the output of the EUT shall be measured at the following center frequencies: half the upper band-edge frequency - 40 Hz (the lower modulation sideband), and half the upper band-edge frequency + 40 Hz (the upper modulation sideband). The non-linear crosstalk ratio shall be the ratio of the total rms level of the two modulation sidebands and the total rms level, expressed in dB.

The measurements shall be performed for each channel of the EUT. Alternatively, the worst measured value of all the channels shall be reported.

> NOTE This measurement may be accomplished in one step by rms summing the output of two frequency-domain band-pass filters on the FFT of a single acquisition.

6.5.5 Input-to-output leakage

This test measures the degree to which signal leaks from inactive EUT inputs to the output. This measurement only applies to EUTs capable of outputting a signal uncorrelated to that on its input. An example is a digital recorder operating in playback mode.

> NOTE This characteristic is also known as *feed-through*.

The EUT shall be placed in a mode that sends digital zero to the outputs (see 3.14). The test signal shall be a sine wave at the maximum input level (see 6.2.1), applied to all EUT inputs simultaneously. The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of no more than one octave.

The analyzer shall include the standard band-pass filter (see 5.2.9), with center frequency equal to the frequency of the test signal. The output level of the band-pass filter shall be measured at each frequency.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and the measured level on the y-axis.

> **See also:** 5.5.3 for presenting graphical data as text.

6.6 Digitization artifacts

6.6.1 Overview: Jitter susceptibility measurements

Analogue-to-digital converters, digital-to-analogue converters, and sample-rate converters render or interpolate samples at fixed times which are determined by a clock. The clock may be recovered from the EUT's digital audio input, may be applied directly to the EUT's reference input, or may be internal to the EUT.

The distortion and noise performance of the EUT are potentially affected by jitter present on the sampling clock. The degree to which performance is affected is known as the jitter susceptibility.

Jitter susceptibility measurements require two simultaneous test signals: an audio test signal, and a clock test signal. The audio test signal is sinusoidal. The clock test signal is first jitter-free to establish baseline performance, and then jittered with sinusoidal phase jitter. For EUTs with a digital audio input, the two test signals may be combined on one interface.

The methods in this clause measure the susceptibility of the EUT to jitter present on an external clock. Because the external clock must be jittered by a known amount, these methods do not apply to EUTs with internal clocks that cannot be accessed outside the EUT.

6.6.2 Overview: Performing a jitter susceptibility measurement

The audio test signal is a sine wave at -3 dB relative to the maximum input level (see 6.2.1). A minimum of three audio test frequencies shall be used:

- One quarter of the sampling frequency.
- 1/192 of the sampling frequency.
- 997 Hz.

  NOTE These audio test frequencies are designed to expose common jitter-related problems exhibited by typical EUTs.

Optional: The measurements shall be repeated for other audio test frequencies.

The clock test signal shall first be substantially free of jitter. The THD+N ratio of the output shall be measured (see 6.3.1) and recorded in dB. This is known as the *jitter-free THD+N ratio*.

Next, the clock test signal shall have its phase jittered with a sinusoidal jitter signal. The frequency of this jitter test signal shall be varied from 80 Hz to 20 kHz in steps of not more than one octave. The jitter level shall be equal to the high frequency jitter tolerance limit of the interface used. If this level is not known, a peak level value of 40 ns time interval error (TIE) jitter shall be used.

The THD+N ratio of the output shall be measured at each step and recorded in dB. The difference between the measured THD+N ratio and the jitter-free THD+N ratio shall be computed, and the results presented as a graph with frequency on the x-axis, shown logarithmically.

6.6.3 Analogue-to-digital jitter susceptibility

This test measures the degree to which jitter on the sampling clock affects the distortion at the output of an analogue-to-digital converter. It can only be used for EUTs with an external clock input.

  See also: 6.6.2 for details on making a jitter susceptibility measurement.

The analogue input shall be driven by the audio test signal. The clock input shall be driven by the clock test signal. The measurement shall be repeated for all clock inputs.

6.6.4 Digital-to-analogue jitter susceptibility

This test measures the degree to which jitter on the digital word clock affects the distortion at the output of a digital-to-analogue converter.

  See also: 6.6.2 for details on making a jitter susceptibility measurement.

The digital audio input shall be driven by the audio test signal. The clock input shall be driven by the clock test signal. The measurement shall be repeated for all clock inputs.

NOTE The digital audio input and the clock input may use the same connector.

6.6.5 Digital-to-digital jitter susceptibility

This test measures the degree to which jitter on the digital word clock affects the distortion at the output of a sample rate converter.

See also: 6.6.2 for details on making a jitter susceptibility measurement.

The digital audio input shall be driven by the audio test signal. The clock input shall be driven by the clock test signal. The measurement shall be repeated for all clock inputs.

NOTE The digital audio input and the clock input may be the same connector.

6.6.6 Attenuation of alias products

This test measures the ability of the anti-aliasing filter in the EUT to attenuate signals above the internal folding frequency. This measurement is primarily intended for EUTs with analogue inputs. However, it can be applied to EUTs with digital inputs whose internal rate is lower than the interface rate at the input (that is, EUTs that decimate). The EUT output may be analogue or digital.

NOTE The internal folding frequency of the EUT must be known for this test (see 3.2). If necessary, it can be determined by observing alias products at the EUT output. The internal folding frequency is the frequency of the lowest frequency test tone which coincides with its own alias.

The test signal shall first be a 997 Hz sine wave at -20 dB relative to the maximum input level (see 6.2.1). The analyzer shall include the standard low-pass filter (see 5.2.5). The level at the output of the EUT shall be measured and recorded as the reference level.

The frequency of the test signal shall then be varied, in steps of not more than one-third of an octave, as follows:

- For EUTs with analogue inputs, from four times the internal sampling frequency or 192 kHz, whichever is lower, down to the internal folding frequency.
- For EUTs with digital inputs, from the input folding frequency down to the internal folding frequency.

The level at the output of the EUT at each step shall be measured and expressed as a ratio to the reference level. The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and the relative level of the alias products on the y-axis in dB. Alternatively, the graph may be replaced by a specification indicating the worst-case relative level.

6.6.7 Attenuation of image products

This test measures the ability of the reconstruction filter in the EUT to attenuate signals above the internal folding frequency. This measurement is primarily intended for EUTs with analogue outputs. However, it can be applied to EUTs with digital outputs whose internal rate is higher than the interface rate at the input (this is, EUTs that interpolate). The EUT input may be analogue or digital.

NOTE The internal folding frequency of the EUT must be known for this test. If necessary, it can be determined by observing image products at the EUT output. The internal folding frequency is the frequency of the lowest frequency test tone which coincides with its own image. The internal sampling frequency is twice the internal folding frequency.

The test signal shall first be a 997 Hz sine wave at -20 dB relative to the maximum input level (see 6.2.1). The analyzer shall include the standard low-pass filter (see 5.2.5). The level at the output of the EUT shall be measured and recorded as the reference level.

The frequency of the test signal shall then be varied from 20 Hz to the upper band-edge frequency in steps of not more than one-third of an octave. The output of the EUT shall be filtered with the standard notch filter (see 5.2.8) whose center frequency is set to the frequency of the test signal, and by the standard high-pass filter (see 5.2.6). The level at the final filter output at each step shall be measured and expressed as a ratio with the reference level, expressed in dB.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and the relative level of the image products on the y-axis in dB. Alternatively, the graph may be replaced by a specification indicating the worst-case relative level.

Optional: This measurement is the sum of the image products and any spurious products. The image products may be measured independently of the spurious products by using a frequency-domain band-pass filter centered on the alias frequency, rather than the notch and high-pass filters.

> **See also**: annex A for more information on how this measurement may be performed using FFTs and frequency-domain band-pass filters.

> **See also:** 6.4.3 for measurement of the spurious output above the folding frequency.

6.6.8 Overload behavior

This test measures the behavior of the EUT when driven with a signal beyond the maximum input level. It is intended to identify unstable behavior, a condition commonly called rollover. This measurement is typically used for EUTs with analogue inputs, but it may also be useful for measuring digital signal paths with non-flat frequency response or imperfect gain structure.

> NOTE This test is intended to be used for EUT diagnosis, not for published specifications. Rollover should always be avoided because it results in very high distortion levels.

The test signal shall be a 997 Hz sine wave at +3 dB relative to the maximum input level (see 6.2.1). The unweighted THD+N ratio of the output signal shall be measured (see 6.3.1). A THD+N ratio greater than 20 % (-14 dB) is a strong indication that rollover has occurred.

Optional: The measurement may be repeated at other frequencies to examine the frequency dependence of the overload behavior. This is important for EUTs with non-flat frequency responses. Rollover is most likely to occur at frequencies where the gain is maximum.

> **See also:** 6.3.3 for measurement of THD+N ratio vs input signal level.

6.7 Digital interface characteristics

6.7.1 Overview

The nature of any digital interface in the EUT shall be clearly stated in its specifications. This includes any dedicated reference synchronization interface. This clause lists the items that shall be stated.

6.7.2 Input interface

The interface standard to which all digital inputs of the EUT conform shall be stated, including any applicable grade or level of conformance. All dedicated reference synchronization inputs should be included.

> NOTE Methods for testing the conformance of the EUT are beyond the scope of this document, and should be established with reference to the relevant standards. In general, methods should be applied to establish the handling of both audio and non-audio data, and the susceptibility to relevant carrier quality parameters including sampling frequency accuracy and jitter, at each digital input.

6.7.3 Input word length

The number of active audio bits accepted by the EUT's digital inputs shall be stated. See 3.7 for the definition of input word length.

NOTE The input word length is important because it defines the generator dither level needed for other measurements using that EUT input.

The input word length may be specified by the manufacturer. If not specified, it may be inferred by one of the following methods:

a. **Dynamic range method.** This method applies if a unity-gain path is available from the digital input of the EUT to a digital output with greater or equal word length. The generator word length is first set to 12 bits and the dynamic range measured (see 6.4.1). The generator word length is then increased one bit at a time, and the change in dynamic range from the previous word length is noted. If the change is less than 3 dB, the word length is reduced by one bit. The resulting word length is the input word length.

b. **Toggling LSB method.** The test signal shall be a sequence of samples with all bits set to zero except the LSB, which follows the repeating sequence `1, 1, 0, 0`. The analyzer shall use the standard band-pass filter (see 5.2.9) with center frequency set to half the folding frequency. The generator word length is first set to 12 bits and the level at the output of the band-pass filter is measured. The generator word length is then increased one bit at a time, and the change in the filtered level from the previous word length is noted. If the change is less than 3 dB, the word length is reduced by one bit. The resulting word length is the input word length.

6.7.4 Output interface

The interface standard to which all digital outputs of the EUT conform shall be stated, including any applicable grade or level of conformance. Any dedicated reference synchronization output should be included.

NOTE Methods for testing the conformance of the EUT are beyond the scope of this document, and should be established with reference to the relevant standards. In general, methods should be applied to establish the generation of both audio and non-audio data, and the relevant carrier quality parameters, at each digital output. For dedicated reference synchronization outputs, intrinsic jitter and jitter transfer characteristics should also be included.

6.7.5 Output word length

The number of active audio bits which are transmitted by the EUT's digital outputs shall be stated. See 3.8 for the definition of output word length.

The output word length may be specified by the manufacturer. If not specified, it may be determined by observing the bit activity on the digital output using equipment conforming to the output's interface standard. The test signal shall be a sine wave at 997 Hz at a level between -60 dB and -20 dB relative to the maximum input level (see 6.2.1). The output word length shall be the number of bits whose values vary over time.

NOTE Neither the test frequency nor level is critical for this measurement. The values given here are recommendations only.

6.8 Audio measurements requiring advanced equipment

6.8.1 Overview

The measurements in this clause require access to advanced test equipment with the following capabilities:

- Ability to play back non-sinusoidal waveforms.
- Ability to acquire a signal into memory.
- Ability to trigger a signal acquisition at the start of waveform playback.

All measurements in this clause are optional.

6.8.2 Delay through the EUT

This test measures the delay imposed upon a signal passing through the EUT.

NOTE For most converter-based EUTs, the delay of the EUT consists of a relatively large component due to the converter that is constant with frequency, and a relatively small component due to analogue filtering, ac coupling, etc. that varies with frequency. The first component is measured by this test.

The following methods may be used to measure delay. For both methods, the signal level shall be -20 dB relative to the maximum input level (see 6.2.1). Each channel of the EUT shall be measured separately.

a. **Impulse response method.** The test signal shall be a wideband signal, such as a swept sine wave. The impulse response of the EUT is derived from the inverse Fourier transform of the complex division of the EUT output spectrum by the test signal spectrum. The delay through the EUT is the time value corresponding to the absolute peak in the impulse response.

b. **Cross-correlation method.** The test signal shall be a wideband signal, such as pseudorandom noise, or a swept sine wave. The input and output signals shall be cross-correlated to obtain a measurement of delay. The delay through the EUT is the time value corresponding to the absolute peak in the correlation function.

The results shall be presented as a single figure stating the EUT delay.

See also: Annex A for measuring the impulse response.

6.8.3 Input-to-output phase response

This test measures the deviation of EUT phase from linear with frequency.

NOTE For most converter-based EUTs, the delay of the EUT consists of a relatively large component due to the converter that is constant with frequency, and a relatively small component due to analogue filtering, ac coupling, etc. that varies with frequency. The second component is measured by this test.

The following methods may be used to measure phase response. For both methods, the signal level shall be -20 dB relative to the maximum input level (see 6.2.1). Each channel of the EUT shall be measured separately.

a. **Transfer function method.** The test signal shall be a wideband signal, such as a swept sine wave. The transfer function of the EUT shall be derived from the complex division of the EUT output spectrum by the test signal spectrum. The input-to-output phase response of the EUT shall be the difference between the phase of the transfer function and a linear fit to this phase.

b. **Sine wave method.** First, the delay through the EUT shall be measured (see 6.8.2). The test signal shall be a sine wave. The frequency of the test signal shall be varied from 20 Hz to the upper band-edge frequency in steps of no more than one octave. The phase difference between the output of the EUT and its input shall be measured at each frequency. A phase corresponding to the EUT delay shall be computed as phase (degrees) = EUT delay (s) × frequency (Hz) × 360, and subtracted from the measured phase difference.

NOTE For both methods, the phase curve may need to be unwrapped to obtain a meaningful result. The low frequency resolution of the sine wave method may make this difficult or impossible. It may therefore be necessary to increase the frequency resolution, or to use the transfer function method.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and the phase difference on the y-axis. The graph may be replaced by a specification indicating the worst-case variation in phase deviation from linear over the measured frequency range, for example, "+1,0/-2,5 degrees from 20 Hz to 20 kHz".

See also: 5.2.9 if the EUT is very noisy.

6.8.4 Group delay vs frequency

This test measures the variation of EUT group delay with frequency.

Group delay can be computed from the input-to-output phase response (see 6.8.3) according to the formula $g = -\mathrm{d}\varphi/\mathrm{d}f$, where g is the group delay and φ is the measured phase at frequency f. The change in phase $\mathrm{d}\varphi$ can be approximated by the difference of neighboring phase measurements.

The results shall be presented as a graph with frequency on the x-axis, shown logarithmically, and group delay on the y-axis.

Annex A (informative): Advanced measurement methods for production testing

A.1 Characterization vs production testing

Characterization of an EUT involves establishing its performance, ideally using the methods described in clause 6. Typically, the results of these measurements are used to establish the published specifications. Characterization of the EUT is done in the research and development phase.

The goal of *production testing* is to establish with high confidence that a particular EUT sample meets the published specifications. It may take too long to measure all the specifications of an EUT in production using the methods of clause 6. Typically, therefore, test time is reduced by:

- Measuring a subset of the EUT characteristics; and
- Making use of fast measurement methods whose results are correlated with, rather than identical to, the results obtained by the sine-based methods used to characterize the EUT.

This annex describes fast measurement methods for production testing. These methods provide results that are correlated with equivalent measurements in clause 6 It is therefore possible to use them for production testing to establish with high confidence that the EUT sample meets specifications. This is done as follows:

1. In the research and development phase, the specifications are established by characterizing the EUT using the methods in clause 6.
2. Also in the research and development phase, a sample EUT which meets the specifications is measured using one or more of the methods in this annex. The results are used as production test limits, with an appropriate tolerance applied to account for measurement noise, and for expected sample-to-sample variation.
3. In the production phase, each sample EUT is measured using the method or methods chosen in step 2, and compared against the test limits.

Both of the methods described in this annex measure multiple characteristics of the EUT in a short time. They are therefore well suited to production testing.

A.2 Overview of FFT-based analysis methods

Modern test equipment can perform fast Fourier transform (FFT) analysis quickly and at low cost. FFT analysis enables measurement methods that are difficult or impossible to achieve with traditional sine-based equipment.

The FFT is a fast algorithm for computing the discrete Fourier transform (DFT). The DFT transforms a uniformly sampled time-domain signal into its spectral equivalent. The resulting spectrum has the same number of points as the input signal. Each point in the spectrum (also known as a *bin*) is a complex number whose magnitude represents the level of the bin and whose angle represents the phase of the bin.

The DFT is inherently circular. The result of performing a DFT is the spectrum of the input sequence is copied and repeated infinitely in both directions. (The spectrum of a periodic sequence is itself periodic. The DFT computes one period of the periodic spectrum.)

The circularity of the DFT means that the last point of the input signal is effectively the neighbor of the first point when the transform is performed. If the signal does not consist of a whole number of cycles, this will lead to a discontinuity. In the transform, this will cause energy to leak into the neighboring bins of prominent tones, potentially obscuring low-level information.

To reduce this problem, an acquired signal is usually windowed before transformation. The acquisition is multiplied point-for-point by a window function whose value is unity at the center of the acquisition and tapers to zero at either end. There are many examples of windows in the published literature, but all have the same purpose: to control the degree to which energy leaks into neighboring bins.

If the signal is known to have a whole number of cycles in the acquisition, no window is needed. This is the basis for synchronous multitone analysis (see annex A.3).

Because of Parseval's theorem of the equivalence of power across the Fourier transform, rms measurements can be made in either the time or the frequency domain. Since all the measurements in this standard use rms detection, any of them can in principle be made in the frequency domain using FFTs.

A.3 Synchronous multitone analysis

A.3.1 General

In multitone analysis, the EUT is stimulated with many frequencies at once. The EUT output is acquired and analyzed with FFT-based methods. This allows many measurements to be derived from a single acquisition.

Synchronous multitone analysis requires the sampling frequencies during generation and analysis to be identical within a very fine tolerance. The FFT analysis may then be performed without windowing. Each stimulus tone will occupy only one bin of the resulting FFT.

Synchronous multitone analysis can be used on EUTs with analogue or digital inputs and outputs, as long as the sampling rates of the test equipment's signal generator and signal analyzer are equal. This is inherent in most digital-to-digital EUTs, and can usually be arranged in cross-domain EUTs by synchronizing the analogue generator or analyzer to the sampling frequency of the EUT. For digital-to-digital EUTs where the input and output have different or unlocked sampling frequencies (such as sample-rate converters), synchronous multitone analysis can be used if the signal analyzer is capable of resampling its input signal to match the sampling frequency of the signal generator.

This clause describes a set of methods based on synchronous multitone analysis. For all the methods described below, unless specifically stated, the EUT is configured with the standard settings (see 4.4). Wherever different settings are employed, these shall be clearly stated.

A.3.2 Stimulus signal

The multitone stimulus is typically a pre-computed waveform. The sample rate SR and the length L of the stimulus waveform determine the available tone frequencies in the stimulus. They are given by

$$\mathrm{f} = \mathrm{N} \times \frac{\mathrm{SR}}{\mathrm{L}} \qquad (A.1)$$

where N is an integer and $1 \leq N < L/2$. A waveform is constructed with the desired properties by summing sine waves of appropriate frequency and level. For example, tones might be spaced on approximate octave boundaries between 20 Hz and 20 kHz, and of equal level.

Additional properties that are commonly exploited are:

- Randomized or optimized tone phases to reduce the crest factor of the waveform, improving the signal-to-noise ratio of the measurement.
- Double-length analysis to measure EUT noise in the odd-numbered (signal-free) frequency bins.
- For stereo or multichannel EUTs, offset upper tone frequencies in each channel to measure crosstalk.

To avoid transients in the EUT, the stimulus is generated periodically, with no gap between the end of one cycle and the beginning of the next. The EUT is allowed to settle before an acquisition is made.

A.3.3 Acquisition

The acquisition of the EUT output is usually triggered at the start of the stimulus. The acquisition is typically either the same length as the stimulus, or, if noise is also being measured, twice its length (see A.3.7). Since the analysis is by FFT, and efficient algorithms exist for power-of-two length FFTs, the stimulus and acquisition lengths are often a power of two. However, this is not a requirement.

In some setups, the stimulus and analysis sample rates cannot be made equal. Examples are measurements on playback-only EUTs (such as portable music players) and measurements over a channel which does not share the sample clock between transmitter and receiver (such as Bluetooth). The acquisition must then be resampled at the stimulus rate before performing the FFT. Typically this is done with asynchronous sample rate conversion.

A.3.4 Result: Frequency response

This result is correlated with the sine-based frequency response measurement (6.2.3).

The EUT output level is computed by measuring the rms level of each bin in the analysis spectrum that corresponds to a tone in the stimulus. The rms output level is the magnitude of the complex value in the bin.

The EUT frequency response is computed by dividing each measured level by the rms level of the corresponding stimulus tone. The level of each stimulus tone is derived from knowledge of the stimulus waveform and the generator output level setting. The frequency response is shown by plotting the gain at each tone bin on an x-y graph.

A.3.5 Result: Inter-channel phase response

This result is correlated with the sine-based inter-channel phase response measurement (6.2.7).

Typically the low density of tones in a multitone waveform precludes measuring the input-to-output phase response of converter-based EUTs. However, the inter-channel phase response of stereo or multichannel EUTs can be measured with multitones.

The EUT output phase is computed by measuring the phase of each bin in the analysis spectrum that corresponds to a tone in the stimulus. The phase is the angle of the complex value in the bin. This is done for all EUT channels.

The tones in a multitone stimulus may have random phases, or phases that are optimized to minimize the crest factor of the waveform. The phase of each stimulus tone may therefore be different in the two channels. The EUT phase is the difference between each measured phase and the phase of the corresponding stimulus tone.

The EUT inter-channel phase response is the difference in phase between corresponding tones on two channels. The inter-channel phase response is shown by plotting the phase difference at each tone bin on an x-y graph.

A.3.6 Result: Total distortion plus noise (TD+N) ratio

This result is correlated with the sine-based THD+N ratio vs frequency measurement (6.3.2) and intermodulation distortion ratio measurements (see 6.3.5 and 6.3.6).

Consider a multitone stimulus consisting of a single tone. The EUT output spectrum contains the stimulus tone and its harmonic distortion products. Each harmonic, being an integer multiple of the stimulus frequency, occupies one bin in the spectrum. The total harmonic distortion plus noise (THD+N) is the rms sum of the levels of all bins except the bin containing the stimulus tone. The THD+N ratio is the ratio of this sum to the rms sum of all bins.

A typical multitone, however, has more than one tone. Therefore the EUT distortion spectrum consists not just of harmonics of the stimulus tones, but their intermodulation products as well. For even a moderate number of tones, a large proportion of the bins contain distortion energy. As with harmonic products, each intermodulation product occupies a single bin in the spectrum.

The EUT total distortion plus noise (TD+N) level is the rms sum of the levels of all bins except those containing the stimulus tones. The TD+N ratio is the ratio of this sum to the rms sum of all bins. The TD+N ratio is shown as a single value, usually in dB. The entire distortion plus noise spectrum can also be shown on an x-y graph. Typically the stimulus tones are deleted from this graph.

A.3.7 Result: Signal-to-noise ratio (SNR)

This result is correlated with the sine-based dynamic range measurement (6.4.1).

If the analysis length is twice that of the multitone stimulus (that is, if two full cycles of the stimulus waveform are acquired), then the stimulus tones and all distortion products occupy the even-numbered bins in the spectrum. The odd-numbered bins contain only noise generated in the EUT. It is therefore possible to measure the EUT noise level in the presence of the multitone stimulus.

The EUT noise level is computed as $\sqrt{2}$ times the rms sum of the levels of all the odd-numbered bins. (The factor of $\sqrt{2}$ compensates for the fact that the even-numbered bins are not included in the sum.) The signal-to-noise ratio (SNR) is the ratio of the rms sum of all bins to the noise level. The SNR is shown as a single value, usually in dB. A noise density spectrum can also be shown on an x-y graph. Typically the stimulus tones are deleted from this graph.

A.3.8 Result: Inter-channel crosstalk ratio

This result is correlated with the sine-based inter-channel crosstalk ratio measurement (6.5.2).

In the sine-based crosstalk measurement on stereo or multichannel EUTs, one channel is driven with a tone, while the remaining channels are undriven. The crosstalk ratio is defined as the ratio of the measured level of the tone in an undriven channel to the measured level of the tone in the driven channel. The measurement is repeated on each channel in turn.

In the multitone crosstalk measurement, all channels are driven simultaneously, but each channel contains at least one stimulus tone that is not present in the other channels. Typically a stimulus tone in one channel is paired with a stimulus tone very close in frequency on the other channels. This allows crosstalk to be measured in both directions simultaneously in the same part of the spectrum.

The variation of crosstalk with frequency can be measured by using multiple tone clusters. Because crosstalk tends to be capacitive in origin, these tone cluster are typically placed in the higher frequency regions of the audio band. However, crosstalk can be measured over the full band if desired.

The EUT inter-channel crosstalk ratio is computed by dividing the level of the tone in the undriven channel by the level of the same tone in the driven channel. It is stated as a single value for each channel, usually in dB. The inter-channel crosstalk ratio response can also be shown on an x-y graph.

A.3.9 Comparability of results to sine-based methods

The frequency response, inter-channel phase vs frequency, and inter-channel crosstalk ratio results derived from the multitone are all comparable to sine-based measurements made at the multitone stimulus frequencies. Because a multitone consists of a sum of tones, however, the level of each tone is lower than the total signal level. For a dense multitone, each tone may be 20 dB or more lower than the total signal level. For low-noise EUTs, this reduction in the signal-to-noise ratio is of little consequence in a production test.

SNR is measured differently from the equivalent sine-based measurement. One difference is that the multitone version measures the noise in the presence of a full-scale signal. For EUTs with an analogue output, this may lead to dynamic range limitations in the analysis equipment that limit the maximum SNR that can be measured. Nevertheless, the results of the two measurement methods correlate well, particularly for medium-performance line-level EUTs.

The TD+N ratio result is fundamentally different from sine-based THD+N ratio and IMD ratio measurements, being in some ways a combination of both. The sine-based THD+N ratio measurement sums all spectral content other than the fundamental, as does the multitone TD+N ratio measurement. The sine-based IMD ratio measurement measures the levels of intermodulation products and noise in the region of those products, as does the multitone TD+N ratio measurement. The main differences between the two approaches lie in the larger number of tones in the multitone (leading to many more intermodulation products) and the difference in the denominator component of the ratio.

Again, the TD+N ratio result correlates well with the sine-based equivalents. Because of the higher number of tones in the multitone, the distortion spectrum is much richer than that produced by sine-based measurements. As such, the multitone can be considered to exercise the EUT more thoroughly than sine-based methods.

A.4 Exponential sine sweep (chirp) analysis

A.4.1 General

In exponential sine sweep analysis, the EUT is stimulated with a swept tone whose frequency starts at the low end of the passband and moves at a constant rate in octaves per unit time to the upper band-edge frequency. The output of the EUT is analyzed using FFT-based methods to recover the impulse response of the EUT. This analysis is known as deconvolution.

In principle, any wideband stimulus can be used to determine the impulse response of the EUT. However, the exponential sine sweep is unique in largely separating the harmonic distortion responses of the EUT from its linear response (and from each other) during the deconvolution. It is therefore possible to measure the linear and non-linear characteristics of the EUT in a single acquisition, with minimal corruption of the linear response by distortion. This makes exponential sine sweep analysis particularly suitable for production testing.

Exponential sine sweep analysis can be applied to EUTs with analogue or digital inputs and outputs, so long as the sampling rates of the test equipment's signal generator and signal analyzer are equal. This is inherent in most digital-to-digital EUTs, and can usually be easily arranged in cross-domain EUTs by synchronizing the analogue generator or analyzer to the sampling frequency of the EUT. For digital-to-digital EUTs where the input and output have different or unlocked sampling frequencies (such as sample-rate converters), exponential sine sweep analysis can be applied if the signal analyzer is capable of resampling its input signal to match the sampling frequency of the signal generator.

The following clause describes a set of methods based on exponential sine sweep analysis. For all the methods described below, unless specifically stated, the EUT is configured with the standard settings (see 0). Wherever different settings are employed, these shall be clearly stated.

A.4.2 Stimulus signal

The basic stimulus signal is a sine wave whose frequency increases continuously and exponentially with time. That is, the stimulus tone starts at a predefined lower frequency, increases by a constant rate in octaves per unit time, and stops at a predefined upper frequency. The table below gives an example.

Table A.1 - Stimulus signal - Example

Time (s)	Stimulus Frequency (Hz)
0,0	20
0,1	200
0,2	2 000
0,3	20 000

A.4.3 Acquisition

Acquisition of the EUT output is triggered at the start of the stimulus. The acquisition is typically slightly longer than the stimulus, to allow for EUT delay that would otherwise cause the end of the sweep to be truncated. If the stimulus is presented once (rather than periodically), the acquisition should be long enough to capture all the transient behavior of the EUT. Failure to do so will result in inaccurate low frequency measurements.

A.4.4 Analysis

The impulse response of the EUT is derived by deconvolution. The acquired EUT response and the stimulus are Fourier transformed, the transforms are divided point-for-point using complex division, and the result is inverse Fourier transformed to recover the EUT impulse response. Careful use of windowing is required to obtain good results.

The deconvolved impulse response consists of two main regions:

- The right-hand side ($t \geq 0$). This region contains the linear response of the EUT, from which the frequency and phase responses are derived.
- The left-hand side ($t < 0$). This region contains the non-linear response of the EUT, from which the harmonic distortion and THD responses are derived.

The exponential nature of the sweep confers the unique property that harmonic distortion products generated in the EUT are effectively advanced in time relative to the fundamental, while following the same trajectory of increasing frequency. The higher the harmonic number, the more advanced the response is in time. The left-hand side of the deconvolved impulse response therefore consists of a series of harmonic distortion responses, starting with the second harmonic close to $t = 0$, and with higher harmonics appearing at increasingly negative values of t.

A.4.5 Result: Frequency response

This result is correlated with the sine-based frequency response measurement (6.2.3).

The EUT frequency response is computed by forward transforming the linear impulse response and computing the magnitude of each spectral bin. The linear impulse response can be optionally truncated (gated) before transformation to prevent reflections in an acoustical setup from corrupting the frequency response. The frequency response is shown by plotting on an x-y graph, typically in dB.

A.4.6 Result: Inter-channel phase response

This result is correlated with the sine-based inter-channel phase response measurement (6.2.7).

The EUT phase response is computed by forward transforming the linear impulse response and computing the angle of each spectral bin. Typically this is done at the same time as the computation of the frequency response. Because the phase response has high resolution, it is possible to correctly unwrap the input-to-output phase response, if desired.

The EUT inter-channel phase response is the difference in phase between the EUT channels. The inter-channel phase response is shown by plotting the phase difference on an x-y graph in degrees.

A.4.7 Result: Total harmonic distortion (THD) ratio vs frequency

This result is correlated with the sine-based THD+N ratio vs frequency measurement (6.3.2).

The EUT harmonic distortion responses are found on the left-hand side of the impulse response ($t < 0$). Their locations are a function of the sweep rate and length. The N-th order response, where N is the harmonic number, represents the component of the EUT response whose frequency is N times the stimulus frequency.

Time-domain harmonic distortion responses are not easy to interpret. A more common way of showing a harmonic product is as a response vs stimulus frequency. This can be obtained by isolating the desired harmonic response, forward transforming it, and plotting it on an x-y graph against generator frequency (rather than analysis frequency). This requires the transform of the harmonic response to be frequency compressed by a factor of N, where N is the harmonic number. The result is the N-th harmonic distortion response vs frequency.

The EUT total harmonic distortion (THD) ratio is computed as the rms sum of the N-th harmonic distortion response results divided by the frequency response. The number of harmonics included in the sum should be large enough to include most of the harmonic output of the EUT. The THD ratio vs frequency is shown by plotting the THD ratio on an x-y graph, usually in dB.

A.4.8 Result: Group delay response

This result is correlated with the sine-based group delay vs frequency measurement (6.8.4).

The EUT group delay vs frequency is computed by:

- Multiplying the linear impulse response point-for-point by the corresponding time axis values;
- Forward transforming this vector;
- Forward transforming the linear impulse response;
- Finding the point-for-point ratio of these two vectors; and
- Taking the real part of each point.

Typically this is done at the same time as the computation of frequency response. Group delay vs frequency is shown by plotting on an x-y graph in seconds.

A.4.9 Result: Linear impulse response

The EUT impulse response is not a measurement required in the standard. However, it can provide useful information in both development and production. Furthermore, it has to be computed to derive the results given above.

The result of deconvolution is the combined linear and harmonic impulse responses of the EUT. The linear portion is recovered by discarding the deconvolution result before time zero. The linear impulse response is shown by plotting on an x-y graph, where the x-axis unit is seconds. The y-axis is unitless for analogue-analogue and digital-digital EUTs. For cross-domain EUTs, it is necessary to express levels on the output side of the EUT in the units of the input side. This requires measuring the maximum input level (for analogue-digital EUTs) or the maximum output level (for digital-analogue EUTs).

> **See also:** 6.2.1 and 6.2.6 for measuring maximum input level and maximum output level, respectively.

A.4.10 Comparability of results to sine-based methods

Because the exponential sine sweep stimulus is at its heart a sine wave (albeit one whose frequency increases continuously over time), the results it returns are very similar to those returned by sine-based measurements. This includes distortion measurements.

The frequency response, inter-channel phase response, and group delay vs frequency results derived from the exponential sweep are all comparable to sine-based measurements. Any differences are likely to stem from EUT transients, and from the lower signal-to-noise ratio that is a result of fast sweeping. Longer sweeps will reduce the contribution of both of these effects.

The THD ratio vs frequency result is also comparable to the sine-based THD+N ratio vs frequency measurement. This is because the stimulus is a constant-level sine wave that exercises memoryless EUT distortion mechanisms identically to a static sine wave. There are two main differences in the results:

- The sine-based measurement includes noise (the “+N” part), while the exponential sweep does not.
- The sine-based measurement uses the entire EUT output signal as the denominator; the exponential sweep uses the fundamental only.

When driven to near maximum input level, most EUTs are distortion dominated, yet still have low distortion. In this region, the sine-based THD+N ratio and exponential sweep THD ratio are very similar.

Annex B (informative): Frequency-domain and window-width filters

B.1 Overview

Frequency-domain filters can exhibit the following characteristics:

- Extremely narrow band of interest (the passband, for band-pass filters, or stopband, for notch filters).
- Extremely high attenuation outside the band of interest.
- Extremely sharp transition bands.

The band of interest is defined as the minimum number of bins required to effectively isolate the energy in a single tone. Its width is determined by the sampling frequency, the frequency transform record length, and the window type.

Typically, frequency-domain filters are applied to spectra derived from a Fast Fourier Transform (FFT). FFT-based analysis is within the capabilities of most modern test instruments, bringing with it the option of frequency-domain filters for certain measurements. This annex discusses some theory and implementation details of these filters. It also presents reasons both for and against using them for certain measurements in the standard.

The term *window-width filter* is used to describe a frequency-domain band-pass filter whose passband is the main lobe of the window. Such filters are useful for isolating individual tones in a signal. A general frequency-domain band-pass filter differs from a true window-width filter in that more bins may be included in the filter passband to achieve a fixed measurement bandwidth. See B.4 for more information.

B.2 Windowing

It is standard practice to apply a window to a signal before it is transformed into the frequency domain. Without windowing, tones which are not synchronous with the record length (that is, tones which do not have an exactly integer number of cycles in the record) will spread out over many bins in the transformed result. This can make tone identification difficult, and can obscure low-level details in the transform.

The discrete Fourier transform (DFT) is a cyclical operation. The record is effectively periodically extended before transformation. The result is a spectrum consisting of discrete bins (sometimes known in earlier literature as 'lines'). The energy spreading seen in a windowless transform of a non-synchronous tone can be thought of as being due to the discontinuity that results when the record is periodically extended, causing the end of the record to be effectively joined to the start. (For synchronous tones, no discontinuity exists, and there is no energy spread. This is a special case that rarely occurs unless the system is designed that way. Synchronous multitone analysis, described in A.3, is a notable example.)

A window smoothly tapers the signal at the ends of the record, reducing the size of the discontinuity. This mitigates the effect of energy spread. The trade-off is that what may have formerly been narrow lines in the spectrum are broadened. Typically, this broadening is of little consequence, but under certain conditions such as very closely spaced tones, it may make it difficult to distinguish certain features in the spectrum unless the acquisition is increased in length to improve the frequency resolution.

The degree to which features are broadened in the spectrum is determined by the window function. In general, a window that tapers faster broadens features more. (There is an extensive literature on window functions. See annex C for more information.)

B.3 Method of operation of frequency-domain and window-width filters

Frequency-domain filters in general, and window-width filters in particular, rely on the use of a window that confines the energy spread of a tone to a few bins in the transform. No window can confine 100 % of the tone energy to a small number of bins; some energy will always fall into every bin in the transform. However, for all practical windows, the vast majority of the tone energy is confined to a few bins around the tone frequency (a region known as the *main lobe*). Some windows are able to do this to the degree that outside the main lobe, the tone energy is below the noise floor of the measurement system. Such windows are ideal for constructing frequency-domain filters. The Dolph-Chebyshev family of windows is a good example.

For practical purposes, the main lobe contains all the energy in the tone. It follows then that the rms level of the tone can be determined by summing the energy in the main lobe. (Parseval's theorem shows that an rms measurement made in the transform domain is identical to one made in the time domain.)

The advantage of summing the main lobe over performing the rms sum in the time domain is that all frequencies outside the main lobe are excluded from the sum. It is therefore possible to make extremely accurate measurements of tone levels in the presence of distortion, hum, noise, and other spurious components. As an example, consider the measurement of EUT linearity (described in 6.3.7). In this measurement, the test signal level is progressively reduced until it reaches the EUT noise floor. With conventional filtering, the noise that falls into the passband of the filter is substantial. With window-width filtering, only the noise in the main lobe is included. That makes it possible to measure linearity down to lower test signal levels, and with higher accuracy.

B.4 Bandwidth of window-width filters

The bandwidth of a window-width filter is given by

$$\text{Bandwidth (Hz)} = \frac{\text{Main lobe width (bins)} \times \text{Sample rate (Hz)}}{\text{Acquisition length (samples)}} \ .$$

Typically the window type (and therefore the main lobe width) is fixed. The sample rate, which determines the system bandwidth, is often also fixed (48 kHz being a common value for audio test applications). This leaves the acquisition length as the usual way to set the filter bandwidth.

As an example, the Dolph-Chebyshev 150 dB window has a main lobe width of 11 bins. The following table shows the window-width filter bandwidth for a 48 kHz sample rate, for various power-of-two acquisition lengths. Note that the bandwidth decreases with increasing acquisition length.

Table B.1 - Window-width filter bandwidth

Acquisition length (samples)	Filter bandwidth (Hz)
8 192	64,5
16 384	32,2
32 768	16,1
65 536	8,06

B.5 The noise problem

B.5.1 General

When using FFTs and window-width filters to make rms measurements of tones, the FFT must be scaled to give correct results. As long as the tone can be effectively isolated from other components in the spectrum, the measured rms level of the tone will be constant, even if the bin width changes. This is because the tone energy is spread over a fixed number of bins, regardless of their width in Hz.

An extremely linear EUT, such as a processor with digital inputs and outputs, may have no discernible distortion components. The energy falling into the passband of a true window-width filter will then be entirely due to the noise floor of the EUT. The number of bins summed in the filter is constant, but the width of each bin in Hz varies with the sample rate and the FFT length. Therefore the effective bandwidth of the filter in Hz also varies, as shown in the table above. If the signal within the filter passband is noise-like, the measured rms level inside the filter will vary with the bin width.

Consider, for example, an EUT driven with a pure tone. The level of the second harmonic distortion component can be measured by centering a window-width filter on the FFT bin nearest to the frequency of the second harmonic. If the second harmonic is well above the noise floor, the measured level will not vary with the bin width. However, if the second harmonic is below the noise floor, the measured level inside the filter will be a function of the filter bandwidth, which varies with the bin width.

It is highly undesirable that an EUT's apparent performance be dependent on the FFT length and sample rate of the analyzer. Ideally, these parameters should be hidden from the user.

The solution to this problem is to use frequency-domain band-pass filters whose bandwidths are chosen to be a fixed size in Hz. As the bin width decreases, the number of bins included in the measurement is increased to maintain a constant bandwidth in Hz. The FFT length must not fall below a minimum size dictated by this bandwidth. The shortest permissible FFT length is that which results in a true window-width filter.

B.5.2 Example: IMD measurements (close tone)

The close-tone IMD measurement is described in 6.3.5. The stimulus consists of two tones spaced by 2 kHz. The intermodulation products are separated from other products by 2 kHz, and must be measured individually. This is a good application for frequency-domain filters.

However, using a true window-width filter without regard for its actual bandwidth in Hz will lead to the noise problem described in the previous clause if the EUT distortion is low enough that the energy inside the filter passband consists mainly of noise. If a larger FFT is used for analysis, the filter bandwidth will decrease, and the measured IMD level will decrease. The EUT will measure better, even though the only change is the analysis FFT length.

The filter bandwidth therefore needs to be fixed. A reasonable choice for the bandwidth in this application is ±250 Hz from the tone center. Thus the filter centered on the 2 kHz intermodulation product would measure the band from 1 750 Hz to 2 250 Hz, for instance. This choice is a trade-off between noise rejection (which improves with longer acquisitions) and measurement speed (which improves with shorter acquisitions).

An advantage of using frequency-domain band-pass filters in this application is that, unlike constant-Q filters, the filter bandwidth in hertz does not vary with the filter center frequency. Thus the contribution to the total distortion figure by the noise floor is equal for each potential distortion component. If constant-Q filters were used, the noise contribution would increase with the center frequency of the filter.

Annex C (informative): Informative references

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