

AES17-1998

Revision of AES17-1991

**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 digital audio equipment. Many tests are substantially identical to those used when testing analog equipment. However, because of the unique requirements of digital audio equipment and the effects of its imperfections, additional tests are necessary.

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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 comment record is posted at <http://www.aes.org/standards/comments>.

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.

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Working Group SC-02-01 on Digital Audio Measurement Techniques
1998-03

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

1 Scope

This standard provides methods for specifying and verifying the performance of digital audio equipment. It describes the testing of medium- and high-performance digital audio equipment. The characteristics of voice-grade digital audio devices 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 such applications. Low-bit-rate coders are an example of devices that will require additional test techniques to be developed. The nature of such coders dictates that the test methods be based on psychoacoustic models which can predict audible performance. However, the techniques described here should still be informative for such systems. Another caveat concerns digital devices which purposely modify the time-domain characteristics of the audio signal, such as pitch shifters and reverberators. Many of the tests in this standard were prepared assuming that the frequency spectrum of the output signal is substantially the same as that of the input signal. Also, large-amplitude interfering signals (as would be encountered with reverberators) have not been considered.

Many tests are substantially identical to those used when testing analog equipment. However, because of the unique requirements of digital audio equipment and the effects of its imperfections, additional tests are necessary. Some tests have been omitted while the appropriate modifications required for use in testing digital audio systems are being developed.

2 Normative references

The following standards contain provisions that, through reference in this text, constitute provisions of this document. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this document are encouraged to investigate the possibility of applying the most recent editions of the indicated standards.

AES3-1992, *AES recommended practice for digital audio engineering — Serial transmission format for two-channel linearly represented digital audio data*. New York, NY, USA: Audio Engineering Society, 1992.

ITU-R BS.468-4, *Measurement of audio-frequency noise voltage in sound broadcasting*. Geneva, Switzerland: International Telecommunication Union, 1986.

IEC 61260 (1995-08), *Electroacoustics — Octave-band and fractional-octave-band filters*. Geneva, Switzerland: International Electrotechnical Commission, 1995.

IEC 60268-3 (1988-09), *Sound system equipment Part 3: Amplifiers*. Geneva, Switzerland: International Electrotechnical Commission, 1988.

3 Definitions and abbreviations

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

3.1

folding frequency

one-half the sampling frequency of the digital system

NOTE Signals applied to the input which are greater than this frequency are subject to aliasing. In systems that sample at a frequency higher than that ultimately used, stored, or produced by the system, the sampling frequency to be considered is the lowest sampling frequency that occurs in the signal path.

3.2**aliasing components**

components that fold down from the folding frequency due to input signals above the folding frequency

3.3**full-scale amplitude**

amplitude of a 997-Hz sine wave whose positive peak value reaches the positive digital full scale, leaving the negative maximum code unused.

NOTE In 2's-complement representation, the negative peak is 1 LSB away from the negative maximum code.

3.3.1**decibels, full scale**

dB FS

amplitude expressed as a level in decibels relative to full-scale amplitude (20 times the common logarithm of the amplitude over the full-scale amplitude)

NOTE The rules of the International System of Units (SI) require that a space appear after the standard symbol dB.

3.3.2**percent, full scale**

% FS

amplitude expressed as a percentage of full-scale amplitude (100 times the amplitude over the full-scale amplitude)

3.4**full-scale signal level**

FS

signal amplitude relative to the full-scale amplitude expressed in decibels, full scale or percent, full scale

NOTE Because the definition of full scale is based on a sine wave, it will be possible with square-wave test signals to read as much as + 3,01 dB FS. Square-wave signals at this level are not recommended because tilt or overshoot introduced by any filtering operations will cause clipping of the signal.

3.5**upper band-edge frequency**

highest signal frequency to be measured

NOTE See 4.2.2

3.6**digital zero**

signal that has all zeros for all samples

NOTE Digital zero is the only test signal in this standard that is not dithered.

3.7**jitter susceptibility**

effect on equipment-under-test (EUT) performance of jitter present on the signal or reference inputs.

4 Procedures

4.1 Measurement conditions

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

4.1.2 Power-line (mains) voltage shall be set within 2 % of the nominal value listed on the panel of the device being tested. If a range of values is given, the specifications are assumed to be valid over the entire range and may be so verified.

4.1.3 Power-line (mains) frequency shall be set within 1 % of the nominal value listed on the panel of the device being tested. If a range of values is given, the specifications are assumed to be valid over the entire range and shall be so verified.

4.1.4 For dc-powered devices the dc supply voltage shall have a peak-to-peak ripple content of less than 0,5 % of the nominal supply voltage.

4.2 Instrumentation

4.2.1 Standard low-pass filter

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

- a) passband response deviation: $\leq \pm 0,1$ dB, $10 \text{ Hz} \leq f \leq 20 \text{ kHz}$;
- b) stop-band attenuation: > 60 dB, $f > 24 \text{ kHz}$.

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

- a) pass-band response deviation: $\leq \pm 0,1$ dB, $10 \text{ Hz} \leq f \leq$ upper band-edge frequency;
- b) stop-band attenuation: ≥ 60 dB, $f >$ (sampling frequency – upper band-edge frequency).

4.2.2 Standard high-pass filter

The standard high-pass filter shall have a passband ripple of less than 1 dB peak to peak and a stop-band attenuation of greater than 40 dB. The lower edge of the filter transition band shall equal the upper band-edge frequency defined in 3.5. The upper edge of the transition band shall be no greater than 1,3 times the upper band-edge frequency defined in 3.5.

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

- a) passband response deviation: $\leq \pm 0,5$ dB, $1,3$ times upper band-edge frequency $\leq f \leq 200$ kHz;
- b) stop-band attenuation: ≥ 40 dB, $10 \text{ Hz} \leq f \leq$ upper band-edge frequency.

4.2.2.2 If the standard upper band-edge frequency of 20 kHz is used, the standard high-pass filter shall have the following characteristics:

- a) passband response deviation: $\leq \pm 0,5$ dB, $26 \text{ kHz} < f < 200 \text{ kHz}$;
- b) stop-band attenuation: ≥ 40 dB, $10 \text{ Hz} < f < 20 \text{ kHz}$.

NOTE It is possible to implement this characteristic with a 5-pole elliptic or 7-pole Chebyshev filter.

4.2.3 Standard weighting filter

The standard weighting filter for all weighted noise measurements should conform to ITU-R BS 468-4 except for overall gain. The filter unity-gain frequency should be taken to be 2 kHz, equivalent to inserting an attenuation of 5,629 dB at all frequencies. Relative amplitude measurements, such as signal-to-noise ratio, performed using this recommended standard weighting filter, shall be abbreviated dB CCIR-R.M.S. Absolute amplitude measurements performed using this recommended filter shall be denoted by the appropriate quantity abbreviation followed by CCIR-R.M.S.; for example, dB FS shall be dB FS CCIR-R.M.S. If a standard weighting filter differing from this recommendation is used for a measurement according to this standard, the filter network — and gain if appropriate — shall be specified.

4.2.4 Standard notch filter

The standard notch filter shall be a separate notch filter when used instead of a THD + N distortion analyzer. It shall have an electrical Q of at least 1 and not more than 5. This value shall be verified by measuring the -3 -dB frequencies and computing the ratio of the center frequency to the difference between the -3 -dB frequencies. Multistage notch filters shall be acceptable if their combined Q measures within these limits using this definition.

4.2.5 Dither signal

The dither signal shall be a random or pseudorandom sequence having a triangular probability density and a peak-to-peak amplitude of 2 least significant bits (which is ± 1 LSB) of the digital audio input word length quoted for 5.7. The amplitude of the noise shall be constant per unit bandwidth (white) to at least the upper band-edge frequency.

4.2.5.1 The dither signal may be generated by summing two uniformly distributed random or pseudorandom numbers with amplitudes of 1 LSB. If a pseudorandom signal is used, its repetition rate shall be at least as long as the measurement interval of the level 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 any polynomials that create significant short-term energy variation or that produce significant asymmetry in probability distribution when low-pass filtered.

4.2.5.2 For measurements where the stimulus is generated in the digital domain, such as when testing Compact-Disc (CD) players, the reproduce sections of record/replay devices, and digital-to-analog converters, the test signals shall be dithered. When the stimulus is applied in the analog domain, the inclusion of dither shall be the responsibility of the manufacturer and shall not to be supplemented during test except for investigative purposes.

4.2.5.3 The dither may be omitted in special cases for investigative purposes. One example of when this is desirable is when viewing bit weights on an oscilloscope with ramp signals. In these circumstances the dither signal can obscure the bit variations being viewed.

4.2.5.4 The standard dither signal may be replaced by an alternate dither signal having a spectral density that increases with increasing frequency. It shall have a triangular probability density and a peak-to-peak amplitude of 2 LSB (which is ± 1 LSB) of the digital signal driving the EUT.

When measured with an unweighted level meter, the amplitude of the alternate dither signal is identical to the standard dither signal. Weighted measurements are improved due to the lower energy in the middle of the audio band and the weighting-filter rolloff at high frequencies.

This signal may be easily generated by subtracting two uniformly distributed random or pseudorandom numbers with amplitudes of 1 LSB. One number should be new for each new output value and one should be the new number from the previous dither sample calculation. If a pseudorandom signal is used, its repetition rate shall be at least as long as the measurement interval of the level meters used for subsequent measurement. If it is generated from a maximum-length polynomial, the polynomial should be of greater than two terms and the samples should be spaced by several shifts of the polynomial to reduce sample-to-sample correlation. Care should be taken to avoid any technique that creates significant short-term energy variation or that produces significant asymmetry in probability distribution when low-pass filtered.

4.3 Equipment accuracy

4.3.1 Unless otherwise specified, the analog signal generators used for measurements in this standard shall have an output impedance of 50 Ω or less.

4.3.2 Unless otherwise specified, the measurement devices used for measurements in this standard shall have an input impedance of 100 000 Ω or greater. The input capacitance shall not exceed 500 pF.

4.3.3 Unless otherwise specified, the equipment used for measurements in this standard shall have an accuracy in the parameter being measured of at least 3 times better than the specification being verified.

4.3.4 All voltmeters and level meters used for measurements in this standard shall be true root-mean-square (r.m.s.) responding devices with a minimum required accuracy of 0,25 dB at 997 Hz. This accuracy shall be maintained for a signal having a crest factor of 5 or less. R.M.S. calibrated average or peak-responding devices are not acceptable. If the accuracy specification in 4.3.3 requires better performance than this, 4.3.3 shall take precedence.

4.3.5 All distortion meters used for measurements in this standard shall be true r.m.s. responding devices with a minimum required accuracy of 1,0 dB. This accuracy shall be maintained up to a crest factor of 5. R.M.S. calibrated average or peak-responding devices are not acceptable.

4.3.6 All total harmonic distortion plus noise (THD + N) type distortion analyzers used for measurements in this standard shall utilize a notch filter having an electrical Q of at least 1 and not more than 5. This value shall be verified by measuring the -3 -dB frequencies and computing the ratio of the center frequency to the difference between the -3 -dB frequencies. Multistage notch filters shall be acceptable if their combined Q measures within these limits using this definition. High-pass or band-pass filters should not be part of the measurement path unless specifically required for the test being performed. While such filters may not respond to harmonics only, to be acceptable they must respond to noise, since distortion products which alias in frequency will appear at inharmonic frequencies.

4.3.7 Third-octave band-pass filters used for measurements in this standard shall conform to class II or class III response limits as outlined in IEC 61260. This will provide at least 30 dB of attenuation of signals one octave away from the filter center frequency and 60 dB, three octaves away.

4.3.8 Level meters used for measurements in this standard shall integrate the signal for a minimum of 25 ms to ensure an adequate number of codes to be exercised in the converters under test. At low frequencies the required time shall be increased to ensure that at least one full cycle of the test signal shall be measured.

4.3.9 Signal generators used for measurements in this standard shall provide control over frequency with an accuracy of at least 0,05 %. Alternatively, the frequency may be measured with a frequency counter and adjusted to be within the required accuracy. The frequency adjustment resolution shall be adequate to produce the frequencies specified in the appropriate test.

4.3.10 If a manufacturer-specified upper band-edge frequency is used, it shall be the same for all tests in this standard.

NOTE Systems using a 44,1-kHz or 48-kHz sampling frequency shall, for the purposes of this definition, have an upper band-edge frequency of 20 kHz. Broadcast systems or others which use lower sampling frequencies often have upper band-edge frequencies defined by the manufacturer. This frequency, if different from 20 kHz, shall be stated in the manufacturer's specifications.

4.3.11 The measurement frequencies specified in this standard increase the number of codes exercised in the EUT at the preferred sample frequencies. For example, a 1-kHz signal exercises only 48 codes at 48 kHz and, depending on initial phase, might never produce the peak code. Annex A, Finger 1987, contains an explanation.

4.4 Equipment-under-test (EUT) settings

4.4.1 The equipment controls shall be set to their normal operating positions except where noted. The switches and controls of the equipment under test (EUT) shall be consistent for all measurements in this standard.

4.4.2 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, but measurements so obtained shall be clearly indicated as supplementary and shall be reported in addition to the results of the same tests performed using the recommended position.

4.4.3 If a dither is provided, it shall be turned on, and this fact shall be clearly indicated in the specifications.

If desired, some measurements may be repeated without dither. Measurements so obtained shall be clearly indicated as supplementary and shall be reported in addition to the results of the same tests performed with dither.

4.4.4 If selectable limiter or compression circuits are included in the EUT, they shall be disabled. If their effect may be measured with additional tests, the results shall be reported separately.

4.5 Device preconditioning

4.5.1 The device shall be connected under normal operating conditions for the manufacturer-specified preconditioning period prior to any measurements being performed. This condition is intended to allow the device to stabilize. If no preconditioning period is specified by the manufacturer, a 5-min period shall be assumed. Should operational requirement preclude preconditioning, the manufacturer shall so state.

4.5.2 Should power to the device be interrupted during the measurements, sufficient time shall be allowed for restabilization to occur.

4.6 Documentation

4.6.1 Measurement results shall be presented in the form outlined in each test procedure in this standard. Where optional measurements are suggested, these are not to be performed in lieu of the recommended measurements, but in addition to them. The intent is to create a uniform presentation of data, allowing meaningful comparisons among competing equipment.

4.6.2 The settings of all controls which can affect the measured performance of the equipment shall be specified in the reporting of the test data.

4.6.3 When reporting results measured in accordance with the techniques 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."

4.6.4 Amplitude measurements relative to full-scale amplitude (see 3.3) shall be reported in dB FS or % FS.

NOTE To prevent clipping of the signal introduced by the addition of dither, the full-scale amplitude shall be reduced by the peak amplitude of the dither signal added. In digital systems of 16 bits or more, an error introduced by this adjustment on triangular dither of less than 0,003 % or 0,0003 dB is acceptable.

5 Input characteristics

5.1 Measurement of the suppression of aliasing components

The suppression of aliasing components shall be measured using a sinusoidal signal at -20 dB FS. The signal frequency shall be set to 997 Hz to establish an output reference level. The signal shall be swept from at least 4 times the sampling frequency or 192 kHz (whichever is lower) to the folding frequency. For systems in which the sampling frequency is not known, this frequency may be determined by monitoring the frequency of the output alias component.

When making measurements at the analog output of the EUT, the signal shall be filtered with the standard low-pass filter. This may be cascaded with a notch filter at the frequency of the applied sine wave if necessary to suppress input to output leakage. The remaining components shall be measured with an r.m.s. level meter. Their amplitude shall be expressed relative to the amplitude resulting from the reference measurement. The data shall be graphed in decibels as a function of the input frequency.

When making measurements at the digital output of the EUT, the signal shall be filtered with a notch filter at the frequency of the applied sine wave. When the input signal exceeds the folding frequency, the notch filter should no longer be used. The r.m.s. amplitude of the remaining components shall be computed from the samples. Their level shall be expressed relative to the amplitude resulting from the reference measurement. The data shall be graphed in decibels as a function of the input frequency.

5.2 Susceptibility to radio-frequency interference

Under consideration.

5.3 Overload behavior

The overload characteristics of an analog input to a digital device shall be measured by applying a sinusoidal signal 3 dB above the full-scale input level ($+3$ dB FS). The unweighted THD + N of the output signal shall be measured with a notch-type distortion meter and recorded in decibels. The measurement shall then be repeated at -3 dB FS. The overload distortion shall be the difference in decibels between the two distortion measurements. This measurement shall be performed at 997 Hz. If desired, the measurement may be repeated at other frequencies to examine the frequency dependence of the overload behavior.

NOTE This test is intended to identify unstable behavior in converters, a condition commonly called rollover.

5.4 Input for full-scale amplitude

NOTE The characteristic to be specified is the analog signal voltage required to reach digital clipping under normal device settings.

In systems where the output is accessible in the digital domain, the input for full-scale amplitude shall be the r.m.s. voltage of a 997-Hz sine wave that shall be applied to the input to obtain a digital signal whose positive peak value reaches the positive digital full scale.

When the digital signal is not accessible or when the digital full-scale amplitude cannot be reached (as occurs when input limiters are present), the input for full-scale voltage shall be 0,5 dB below the voltage of the lowest 997-Hz sine wave that may be applied to the input of the EUT with its gain controls set to their normal operating positions before introducing 1 % THD + N or 0,3-dB compression at the EUT output, whichever comes first.

NOTE Input gain controls shall be set to the reference position specified by the manufacturer or to their normal operating position if none is specified. Other gain controls in the EUT shall be adjusted to minimize the potential for overload in the EUT output circuitry.

5.5 Maximum input amplitude

NOTE The characteristic to be specified is the maximum analog signal that may be applied to the device for correct operation.

The maximum input amplitude shall be the maximum voltage of a 997-Hz sine wave that may be applied to the EUT input, regardless of gain settings, before introducing 1 % THD + N or 0,3-dB compression at the EUT output, whichever occurs first.

In systems where the output is accessible in the digital domain, the maximum input amplitude shall be the r.m.s. voltage of the sine wave that may be applied to the input, regardless of gain settings, before introducing 1 % THD + N or 0,3-dB compression at the digital output, whichever occurs first.

NOTE Gain controls in the EUT shall be adjusted to minimize the potential for overload in the EUT output circuitry.

5.6 Input logarithmic-gain stability

The EUT shall be stimulated with a sinusoidal signal source at 6 dB below the full-scale input amplitude. The level at the digital output of the EUT shall be measured for a period of at least 1,0 h immediately following the warm-up interval. The worst-case variation in level above and below the first measurement value shall be the input logarithmic-gain stability. The result shall be expressed as a level in decibels.

If the device cannot be measured in the digital domain, the test may be performed with an analog r.m.s. level meter. Because the input logarithmic-gain stability cannot be separated from the output logarithmic-gain stability under this condition, the results shall be reported as the system logarithmic-gain stability.

5.7 Digital audio input format

If a digital audio input is provided, the format of the input shall be stated. For example, if the input conforms to the AES3 serial interface format, this conformity shall be so stated.

The maximum digital audio word length that the equipment will accept without truncation should be stated. This word length shall define the level of dither used in the measurements.

5.8 Jitter susceptibility

The performance of analog inputs and analog outputs is potentially affected by jitter present on the reference input, the digital input, or both. If the sampling clock for the analog-to-digital converter inside the equipment under test is derived from or locked to either the reference input or a digital input, jitter present on that input can impact performance.

5.8.1 Analog-to-digital jitter susceptibility

The analog input shall be driven with a –3-dB-FS sine wave at one-fourth the sampling frequency. The digital input shall be driven with a signal whose phase is jittered with a sine-wave jitter signal. The frequency of this jitter signal is varied from 80 Hz to 20 kHz in octave steps. The jitter amplitude shall be set at the high frequency jitter tolerance limit of the interface used. If this frequency is not known, a value of 40 ns shall be used. The THD + N of the output shall be measured at each step and the results shall be graphed. The measurement may be repeated for other input signal frequencies. It should be repeated at $1/192$ the sampling frequency (f_s) and 997 Hz. The $1/192 f_s$ signal identifies any anomalous low-frequency behavior while the 997 Hz signal maximizes any interaction with converter codes. If the EUT has a reference input, the measurements shall be repeated with the jitter applied to the reference input.

5.8.2 Digital-to-analog jitter susceptibility

The digital input shall be driven with a –3-dB-FS sine wave at $0,25 f_s$. This signal shall be phase jittered with a sine-wave jitter signal. The frequency of the jitter shall be varied from 80 Hz to 20 kHz in octave

steps. The jitter amplitude shall be set at the high-frequency jitter tolerance limit of the interface used. If this limit is not known, a value of 40 ns shall be used. The THD + N of the output shall be measured at each step and the results shall be graphed. The measurement may be repeated for other input signal frequencies. It should be repeated at $1/192 f_s$ and 997 Hz. The $1/192 f_s$ signal identifies any anomalous low-frequency behavior while the 997-Hz signal maximizes any interaction with data codes. If the EUT has a reference input, the measurements shall be repeated with the jitter applied to the reference input.

5.8.3 Digital-to-digital jitter susceptibility

The digital input shall be driven with a 3-dB-FS sinewave at $0,25 f_s$. This signal shall be phase jittered with a sinewave jitter signal. The frequency of the jitter shall be varied from 80 Hz to 20 kHz in octave steps. The jitter amplitude shall be set at the high frequency jitter tolerance limit of the interface used. If this limit is not known, a value of 40 ns shall be used. The THD + N of the digital output shall be measured at each step and the results are graphed. The measurement may be repeated for other input signal frequencies. It should be repeated at $1/192 f_s$ and 997 Hz. The $1/192 f_s$ signal identifies any anomalous low-frequency behavior while the 997-Hz signal maximizes any interaction with data codes. If the equipment under test has a reference input, the measurements shall be repeated with the jitter applied to the reference input.

6 Output characteristics

6.1 Out-of-band spurious components

Out-of-band spurious components shall be measured at the analog output of the EUT with no test-signal stimulus. The level of all components above the upper band-edge frequency of the EUT shall be expressed as a level in decibels relative to the full-scale signal level (that is, in dB FS). The out-of-band signals shall be measured with the standard high-pass filter. The output of the filter shall be measured with a true r.m.s. level meter. The level of the components shall be expressed relative to the full-scale output level in dB FS.

The spectrum of the spurious components should be measured and reported. This analysis provides useful diagnostic information. The spectrum also may be used with the susceptibility to alias-product measurements of the succeeding device to estimate audio-frequency alias-product levels.

6.2 Suppression of imaging components

NOTE The characteristic to be specified is the suppression of all out-of-band components measured in the presence of signal. The measurement is identical to that for spurious components with the addition of a test signal and a notch filter to remove the signal at the device output.

The signal shall be output from the EUT at -20 dB FS. The frequency range of the test signal shall be from 10 Hz to one-half the upper band-edge frequency or 10 kHz, whichever is lower. The amplitude of all components above the upper band-edge frequency of the EUT shall be measured relative to the full-scale output level and expressed in dB FS.

The test signal shall be removed from the output of the EUT with a standard notch filter at the signal frequency. The out-of-band signals shall be separated with a standard high-pass filter as described in 4.2.4. The output of the filter shall be measured with a true r.m.s. voltmeter having a scale of level.

The display, in graphic form, should show the total r.m.s. level as a function of frequency. If a graph is not provided, the worst case shall be specified.

NOTE This measurement, strictly speaking, is the sum of the spurious signal and the image products. The only practical method of separating these signals would be with a spectrum analysis of both tests whereby one eliminates the components present in both from the image-products value.

As in the case of spurious components (6.1), the spectrum of the imaging components should be measured and reported.

6.3 Output amplitude at full scale

In systems where the input is accessible in the digital domain, the output amplitude at full scale shall be the r.m.s. voltage that results from a sine wave whose positive peak value reaches the positive digital full scale under normal settings of gain controls.

In systems where the input is not accessible in the digital domain, the output amplitude at full scale shall be 0,5 dB below the maximum level of a 997-Hz sine wave that may be achieved at the EUT output, under normal settings of gain controls, before introducing 1 % THD + N or 0,3-dB compression, whichever occurs first.

NOTE While performing this test, gain controls in the EUT other than the output gain controls should be adjusted to minimize the potential for overload in the EUT input circuitry.

6.4 Maximum output amplitude

In systems where the input is accessible in the digital domain, the maximum output amplitude shall be the r.m.s. voltage that results from a sine wave whose positive peak value reaches the positive digital full scale when the gain controls are set to maximize the undistorted output signal. The maximum undistorted output amplitude is that obtained before introducing 1 % THD + N or 0,3-dB compression, whichever occurs first.

In systems where the input is not accessible in the digital domain, the maximum output amplitude shall be the maximum level of a 997-Hz sine wave that may be achieved at the EUT output before introducing 1 % THD + N or 0,3-dB compression, whichever occurs first.

NOTE While performing this test, gain controls in the EUT other than the output gain controls should be adjusted to minimize the potential for overload in the EUT input circuitry.

6.5 Output-level stability

The EUT shall be stimulated with a digital sine-wave source at a level -6 dB FS. The output level of the EUT shall be measured for a period of at least 1,0 h immediately following the warm-up interval. The worst-case variation in level above and below the first measurement value shall be the output level stability. The result shall be expressed as a level in decibels.

If the device cannot be stimulated in the digital domain, the test may be performed with an analog signal generator at the analog input. Because the input logarithmic-gain stability cannot be separated from the output-level stability under this condition, the results shall be reported as the "system level stability."

6.6 Digital audio output format

If a digital audio output is provided, the format and active word length of the output shall be stated. For example, if the output conforms to the AES3 serial interface format with a word length of 20 bits, this conformity shall be so stated. If the output word length is adjustable, the word length used for the measurements should be stated.

7 Linear response

7.1 Amplitude versus frequency

NOTE When emphasis filters are incorporated in the EUT, measurement results should be reported separately with and without emphasis.

7.1.1 Frequency response

To prevent overload effects in emphasized systems, frequency response shall be measured at -20 dB FS. When significant aliasing or noise products can be present, the output of the EUT shall be passed through a band-pass filter at the frequency of the sine-wave stimulus. The output of the band-pass filter shall be measured with a level meter. The band-pass filter used shall have a bandwidth equal to or narrower than that of the standard band-pass filter.

NOTE Care should be taken to ensure that the filter gain is sufficiently constant with frequency to be neglected in the measurement.

The response should be expressed as a graph in decibels relative to the amplitude at 997 Hz. If a graph is not provided, the specification shall indicate the worst-case variation in amplitude over a specified frequency range. This range shall include the upper band-edge frequency; for example, $+x - y$ dB from 10 Hz to z kHz, where x and y are the deviations and z kHz is greater than or equal to the upper band-edge frequency.

7.1.2 Maximum signal level versus frequency

Maximum signal level versus frequency shall be measured using a sinusoidal stimulus of adjustable frequency and amplitude. The output signal shall be monitored with a level meter and a notch-filter-based THD + N analyzer. The analyzer shall include a standard low-pass filter at the band-edge frequency or at 20 kHz, whichever is lower. The frequency of the stimulus shall be swept from 10 Hz to the upper band-edge frequency. The test-signal level shall be readjusted at each new frequency to maintain the specified output conditions. The resulting data shall be presented graphically. If a stepwise sweep is used, it shall contain data points at least every octave.

7.2 Phase versus frequency

7.2.1 Phase response

Except where otherwise stated in this section, the phase response of a EUT input and output shall be compared by direct means, such as fast Fourier transforms (FFT) of pseudorandom sequences or impulses, and the deviation from linear phase recorded in degrees.

NOTE When using impulses, it may be necessary to average the results of several measurements to obtain the required measurement accuracy.

However, the phase response of a EUT that processes signals in real time and allows simultaneous access to the input and output terminals may be measured using comparative techniques such as sine-wave display. The phase shift produced by any time delay through the EUT shall be subtracted before recording the results.

7.2.2 Group delay

Group delay may also be computed from the phase response of the EUT measured in 7.2.1.

7.2.3 Interchannel phase response

Interchannel phase response shall be measured by applying a sine wave to all channels of the multichannel device. One channel shall be selected as the reference and so specified. The phase differences between every other channel and the reference channel shall be reported in degrees as a function of frequency. If the r.m.s. sum of the harmonic, inharmonic, and spurious components in each output signal does not exceed 1 % of the test signal amplitude, the phase may be measured based on the zero crossings of the two output sine waves.

A graph of the phase differences among the channels shall be plotted for each channel except the reference. If desired, the graph may be replaced by specification of the maximum phase difference over the frequency range from 10 Hz to the upper band-edge frequency; for example, $+x - y$ degrees from 10

Hz to z kHz, where x and y are the differences and z kHz is greater than or equal to the upper band-edge frequency.

NOTE When emphasis filters are incorporated in the EUT, measurement results should be reported separately with and without emphasis.

7.3 Delay through a device

Two general methods should be used to measure delay. The first should be to apply an impulse test signal to the EUT. The input and output signals should be displayed on a time-calibrated oscilloscope and the delay time read directly from the display. Alternatively, a time-interval counter may be used in place of the oscilloscope.

The second technique uses a random or pseudorandom noise signal as a stimulus. The input and output signals should be cross-correlated to obtain a measurement of delay. The time value corresponding to the peak in the correlation function shall be reported as the delay through the EUT.

The signal amplitude for delay measurements shall be -20 dB FS. When measuring a dual-channel device, each channel should be measured separately because some equipment performs alternate sampling of signals. However, this characteristic is also shown in interchannel phase measurements.

When delay measurements are made between digital and analog signals the timing reference point corresponding to the timing of the digital audio data shall be specified. For AES3 format signals, the timing reference point shall be the first transition of the frame containing each sample. This specification gives both samples in the frame the same time reference.

If a separate synchronization reference can be used, then a second delay measurement should be made with the timing reference point defined with respect to a point in the reference with a defined timing relationship to the digital audio signal. For a reference specified to AES11, the reference is approximately co-timed with the digital audio and measurements shall be made with respect to the timing reference point in the reference signal closest to the timing reference point in the digital audio data.

7.4 Polarity

A 997-Hz or 1-kHz sine-wave tone burst shall be used as a stimulus. It shall be gated on and off at the positive-going zero crossings. The signal shall be on for 5 cycles and off for 20 cycles. The amplitude shall be -20 dB FS.

Polarity may also be measured with any asymmetrical signal. It may be ascertained by observing the output signal from the EUT on an oscilloscope or by an automated device that can sense the direction of the asymmetry.

NOTE When multiple inputs and outputs are present in the EUT, one input shall be chosen as a reference, and the polarity of the remaining inputs and outputs shall be expressed relative to the chosen input.

The result shall be reported as either

- a) polarity: noninverting; or
- b) polarity: inverting.

If the EUT contains both analog-to-digital and digital-to-analog sections and the signal is accessible in the digital domain, results shall be stated separately for each. Positive input and output voltages should be assumed to correspond in the noninverting case to positive digital codes. In the 2's-complement number system, a positive code is one carrying a logic 0 sign bit.

8 Amplitude non-linearity

8.1 Level-dependent logarithmic gain

NOTE The characteristic to be specified is a change in logarithmic gain of the EUT with signal level, frequently called deviation from level linearity.

Level-dependent logarithmic gain shall be measured by applying a sine wave at 997 Hz to the EUT. The output signal from the EUT shall be passed through a third-octave 1-kHz band-pass filter. The output of the filter shall be measured with a level meter. The test tone shall be applied at -5 dB FS and the output amplitude level noted. The ratio of output amplitude to input amplitude, expressed in decibels, shall be the logarithmic gain of the EUT. The amplitude level of the tone shall be reduced in increments no larger than 5 dB, noting the output level at each step and calculating the logarithmic gain. This procedure shall be continued until an output level within 5 dB of the idle channel noise measured in the 1-kHz third-octave band is reached. The logarithmic gain at each step shall be graphed as a function of the applied level. If a single-number measurement is reported, it shall be the worst-case deviation of all steps from the logarithmic gain at the first measurement.

When accesses to the analog-to-digital and the digital-to-analog portions are available separately, the measurement shall be reported for each portion independently. The sinusoidal stimulus may be applied in the digital domain where appropriate, using a digital waveform generator. When the stimulus is provided in the digital domain, it shall be dithered as described in 4.2.5. As the signal level is reduced during this test, the dither amplitude shall remain unchanged.

When testing systems in the presence of large amounts of interfering noise, it may be necessary to restrict the bandwidth of the measurement further than provided by a third-octave filter. This will not be a problem if the EUT has noise in the presence of signal performance commensurate with the performance levels being measured in this section.

8.2 Intermodulation (IM) measurements

IM measurements shall be performed with a twin tone signal, one tone at the upper band-edge frequency and one tone 2 kHz below this frequency. The amplitudes shall be in a 1:1 amplitude ratio, the peak amplitude being adjusted to equal the peak amplitude of an equivalent sine wave at the full-scale level. The output signal shall be measured with a spectrum analyzer or narrow band-pass filter to obtain the level of the second- and third-order difference frequency components. Their r.m.s. sum shall be reported in decibels relative to the output level.

Additional IM measurements may be performed with a pair of tones at 41 Hz and 7993 Hz. The amplitude of the high-frequency tone shall be one-fourth the amplitude of the low-frequency tone. The peak amplitude of the signal shall be adjusted to equal the peak amplitude of an equivalent sine wave at the full-scale level. The level of the modulation sidebands on the 7993-Hz tone shall be expressed as a percentage or decibel value of the amplitude of the 7993-Hz tone.

8.3 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 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.

8.4 Low-level noise modulation

The EUT shall be excited with a 41-Hz sine wave at a level that produces an output signal 40 dB below the full-scale level. The output signal from the EUT shall be passed through a 41-Hz notch filter. The output signal from the filter shall be measured with a third-octave filter set. The output from the band-pass filters shall be measured with an r.m.s.-responding meter. The levels in each of the third-octave bands from 200 Hz to 20 kHz shall be recorded. A single band-pass filter may be used if its frequency is stepped to each standard third-octave center frequency from 200 Hz to 20 kHz. The signal level shall then be reduced in 10-dB steps until the sine-wave portion of the output signal from the EUT is below the idle channel noise level. The third-octave levels shall be recorded at each of these steps, and the level difference at each frequency shall be calculated. The maximum difference in spectral level, regardless of frequency, shall be reported as the low-level noise modulation. If desired, a graph of the worst-case difference versus frequency may be supplied.

8.5 Total harmonic distortion and noise (THD + N)

NOTE The characteristic to be specified is the transfer characteristic and dynamic non-linearities in the EUT. The results are indicative of anomalies in device behavior but may not be indicative of audible performance.

Harmonic distortion and noise is the ratio of the output noise and distortion level to the output signal level. Both levels shall include all harmonic, inharmonic and noise components. All components shall be included because harmonics often alias above the folding frequency and often appear anywhere in the audio band.

8.5.1 Total harmonic distortion and noise versus frequency

The measurement should be conducted with a sine wave at $-1,0$ dB FS and repeated with a sine wave at -20 dB FS. The test signal present in the output should be removed by means of a standard notch filter and the remaining signal bandwidth limited to the upper band-edge frequency or 20 kHz, whichever is lower. The level of the filtered signal should be measured and reported as a ratio to the unfiltered signal level. The measurement should be repeated at each octave frequency from 20 Hz to one-half the upper band-edge frequency.

8.5.2 Total harmonic distortion and noise versus level

The measurement should be conducted with a sine wave at 997 Hz. The test signal present in the output should be removed by means of a standard notch filter and the remaining signal bandwidth limited to the upper band-edge frequency or 20 kHz, whichever is lower. The level of the filtered signal should be measured and reported as a ratio to the unfiltered signal level. The measurement should be repeated at each level from 0 dB FS to -80 dB FS in steps no larger than 10 dB.

9 Signal-to-noise measurement

9.1 Idle channel noise

NOTE The characteristic to be specified is a weighted noise measurement with no signal applied to the EUT input, that is, an electrical back-termination for analog signals and digital zero for digital signals.

Idle channel noise shall be measured with a level meter preceded by the standard weighting filter and a low-pass filter at the specified band-edge frequency. The level shall be read in dB FS and reported in dB FS CCIR-R.M.S.

NOTE The low-pass filter may not be necessary when aliasing components are not present.

9.2 Idle channel noise spectrum

The idle channel noise spectrum shall be measured with a third-octave filter set. The levels in each of the third-octave bands from 20 Hz to 20 kHz shall be recorded. If the upper band-edge frequency is lower than 20 kHz, the measurements may be truncated at the highest center frequency less than the upper band-edge frequency. A single band-pass filter may be used if its frequency is stepped to each standard third-octave center frequency from 20 Hz to the maximum frequency required. The levels shall be reported in dB FS.

9.3 Signal-to-noise ratio or noise in the presence of signal

NOTE The characteristic to be specified is the ratio of the full-scale amplitude to the weighted r.m.s. noise and distortion, expressed in decibels, in the presence of signal. It includes all harmonic, inharmonic, and noise components. It is identical to a measurement of noise in the presence of signal.

The test signal for the measurement shall be a 997-Hz sine wave producing – 60 dB FS at the output of the EUT. Any 997-Hz test signal present in the output shall be removed by means of a standard notch filter. The remaining noise shall be filtered with the standard weighting filter. The measurement shall be limited in bandwidth to the upper band-edge frequency or 20 kHz, whichever is lower. The resulting measurement shall be read as dB FS and reported as dB FS CCIR-R.M.S.

9.4 Power-line (mains) related products

NOTE The characteristic to be specified consists of the components of the EUT noise caused by the power supplied to the EUT.

The low-frequency component of the EUT noise shall be assumed to be due to power-line interference.

Any artifacts due to a high-frequency switching type power supply in the EUT shall be classified as spurious components. Only components related to the power-source input shall be included as power-line interference.

Power-line (mains) related products separate from other noise and spurious components should be reported. If reported, the r.m.s. summation of the signal component at the power line (mains) frequency and its second through fifth harmonic shall be called power line (mains) related products.

10 Cross-talk and separation

10.1 Interchannel cross-talk and separation

NOTE The characteristic to be specified is the linear leakage of information from one channel of a multichannel EUT into another channel. The measurement methods for cross-talk and separation are identical, so only cross-talk will be described.

Cross-talk shall be measured by applying a signal to one input. Other inputs shall be terminated with 50- Ω resistors (for analog inputs).

The signal applied to the driven channel shall be a sine wave at – 20 dB FS. The output of the driven channel shall be measured with a level meter. The output signal from each undriven channel shall be filtered with a third-octave filter at the frequency of the signal source. The signal out of the filter shall be measured with a level meter. The ratio of the undriven-channel output amplitude to the driven-channel output amplitude shall be expressed in decibels. The measurement shall be made at each frequency of interest from 10 Hz to the upper band-edge frequency. Measurements shall be made at least as often as one per octave. The results shall be displayed as a graph.

10.2 Non-linear interchannel cross-talk

NOTE The characteristic to be specified is the non-linear interaction of information in the channels of a multichannel EUT. The term, cross-talk, is used whether or not the information among the channels is related.

Non-linear cross-talk at high frequencies shall be measured by applying a signal to all inputs of the EUT. The channel being measured shall be driven to a level of -20 dB FS with an upper band-edge frequency sine wave. The other channels, called the driven channels, shall be connected together and driven from the output of a second sine-wave generator. The signal applied to the driven channels shall be a sine wave at 3 dB above the full-scale level whose frequency is 3 kHz below that of the measured channel. The ratio of the amplitude of the second-order difference frequency component at 3 kHz in the measured channel to the signal amplitude in the measured channel shall be expressed in decibels. The ratio of the amplitude of the third-order IM component at 6 kHz below the upper band-edge frequency to the signal amplitude in the measured channel shall be expressed in decibels. This measurement shall be repeated for each of the channels in the EUT. The measured values shall be reported separately as even- and odd-order non-linear cross-talk for each channel, or the worst value of the even- and odd-order products shall be reported.

Non-linear cross-talk at low frequencies shall be measured by applying a signal to all inputs of the EUT. The channel being measured shall be driven to a level of -20 dB FS with a sine wave having one-half the upper band-edge frequency. The other channels, called the driven channels, shall be connected together and driven from the output of a second sine-wave generator. The signal applied to the driven channels shall be a 40-Hz sine wave at 3 dB above the full-scale level. The ratio of the r.m.s. amplitude of the modulation sidebands introduced onto the signal in the measured channel to the signal amplitude in the measured channel shall be expressed in decibels. The measurement shall be repeated for each of the channels in the EUT. The worst measured value of all the channels shall be reported.

10.3 Input-to-output leakage

NOTE The characteristic to be specified is the interference in the EUT output caused by an undesired input signal. It is often called feed-through. The measurement is only relevant for a EUT capable of outputting a signal uncorrelated to that on its input, for example, when a tape recorder operates in its playback mode.

Input to output leakage shall be measured by applying a sine wave at the full-scale input level simultaneously to all channels of the EUT. The EUT shall be placed in a mode that sends a digital-zero signal to the outputs. The amplitude of the output signal at the frequency of the input shall be measured using a third-octave band-pass filter and a level meter. The result shall be expressed in dB FS. This procedure shall be repeated for each channel of the EUT. The frequency of the input signal and the filter shall be swept from 10 Hz to the upper band-edge frequency in steps no further than one octave apart. The worst channel measurement at each frequency shall be reported on a graph.

Annex A (informative)**Informative references**

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