STANDARDS AND INFORMATION DOCUMENTS

AES2-2012
revision of AES2-1984

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AES standard for acoustics - Methods of measuring and specifying the performance of loudspeakers for professional applications - Drive units

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AES2-2012

Revision of AES2-1984

AES standard for acoustics - Methods of measuring and specifying the performance of loudspeakers for professional applications - Drive units

Abstract

This document is a recommended practice for describing and specifying loudspeaker components used in professional audio and sound-reinforcement systems. These components include high-frequency drivers and low-frequency drivers. Specifications are given for describing frequency response, impedance, distortion, and power handling.

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Contents

This foreword is not part of the AES2-2012 *AES standard for acoustics - Methods of measuring and specifying the performance of loudspeakers for professional applications - Drive units*

Foreword to 1984 edition

The purpose of this document is to recommend methods of specifying the performance of loudspeaker components used in music, speech, and fixed-signal (such as siren alert) systems. It is needed so that these components may be compared on an equal basis, by methods which directly relate to their specific real use. Previously, no such practice or standard existed for this class of acoustical product. Tests and nomenclature used in this document are compatible with IEC Standard, Publication 268-5 (1972) and Supplement 268-5A (1980).

The document presented here is a complete recommendation.

This committee was suggested and formed by John Eargle in 1975 November, and the following members have contributed to the processing and approval of this Recommended Practice:

Clifford Henricksen, *Chairman*

J. Robert Ashely, George Augspurger, George Brettell, Bob Davis, Howard Durbin, David Klepper, Bart Locanthi, Manny Mohageri, Harold Mosier, Richard Negus, Daniel Queen, Ludwig Sepmeyer, and Melvin Sprinkle.

Foreword to 2012 edition

This document substantially revises and updates AES2-1984.

The members of the writing group that developed this document in draft were: D. Blore , J. Brusi, M. Buck, R. Campbell, D. Carlstrom, D. Cinanni, P. Conejos, E. Geddes, D. Gunness, C. Hughes, R. Hagenbach, S. Hutt, D. Keele, W. Klippel, H. van Maanen, B.C. Olson, S. Orth, R. Sauro, E. Simon, R. Small, J.S. Stewart, J. Stratman, C.J. Struck, A. Voishvillo, J. Woodgate.

Steve Hutt
Chair, Working Group SC-04-03 on Loudspeaker Modeling and Measurement

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 for acoustics - Methods of measuring and specifying the performance of loudspeakers for professional applications - Drive units

1 Scope

This document defines a minimum set of characteristics of loudspeaker drivers for inclusion in manufacturers' specification documents, and identifies the relevant methods of measurement.

The document considers drivers and passive loudspeaker systems for professional applications. It does not consider sub-components such as spiders or cones. It is intended for loudspeaker system designers, and drive-unit manufacturers.

2 Normative references

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

IEC 60268-1: *Sound system equipment, Part 1: General*. International Electrotechnical Commission, Geneva, Switzerland.

IEC 60268-2: *Sound system equipment, Part 2: Explanation of general terms & calculation methods*. International Electrotechnical Commission, Geneva, Switzerland.

IEC 60268-5 Ed.3.1 2007: *Sound system equipment, Part 5: Loudspeakers*. International Electrotechnical Commission, Geneva, Switzerland.

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

ISO 3741: *Acoustics - Determination of sound power levels of noise sources using sound pressure - Precision methods for reverberation rooms*. International Organization for Standardization, Geneva, Switzerland.

3 Definitions

3.1
loudspeaker
an electroacoustic transducer system intended to produce sound to be heard at a distance from the transducer.

3.2
drive unit, driver
a basic electroacoustic transducer forming an article of commerce, used in conjunction with other parts, such as baffles, enclosure or horns, to construct a loudspeaker or loudspeaker system.

3.3
broadband noise signal
See IEC 60268-2.

3.4
narrow-band noise signal
See IEC 60268-2.

3.5
impulsive signal
A short duration pulse having constant spectral power per unit bandwidth over at least the bandwidth of interest of the measurement.

3.6
simulated program signal
See IEC 60268-1.

3.7
free-field conditions
See IEC 60268-5.

3.8
half-space free-field conditions
See IEC 60268-5.

3.9
simulated free-field conditions
Conditions which are equivalent to those of free space for the time required to carry out a measurement.

> NOTE These conditions, which may apply in free space or half space, are used for gated or impulse measurements.

3.10
diffuse sound field conditions
See ISO 3741.

3.11
power compression
Reduction in sensitivity of a loudspeaker with increasing input voltage or power.

3.12
Thiele-Small parameters of a transducer

parameter	symbol
d.c. resistance of the voice coil	R_E

effective piston area of the radiating element	S_D
resonance frequency of the mass and compliance of the radiating element	f_S
mechanical Q of the resonance at f_S	Q_{MS}
electrical Q of the resonance at f_S	Q_{ES}
equivalent air volume of the compliance of the radiating element	V_{AS}
maximum useful displacement of the voice-coil or equivalent element in either direction from its rest position	X_{MAX}

3.13
directivity index
D_i
specified in IEC 60268-5, 23.3

3.14
crest factor
the ratio of the peak amplitude to the time-averaged rms amplitude of an alternating waveform

3.15 Principle of the transducer
3.15.1
electrodynamic (moving-coil) loudspeaker
loudspeaker, the diaphragm of which is driven by a mechanical force that occurs when current flows through an electric conductor placed in a magnetic field

3.15.2
electrostatic (condenser) loudspeaker
loudspeaker, the diaphragm of which is driven by an electrostatic force

3.15.3
piezoelectric (crystal) loudspeaker
loudspeaker, the diaphragm of which is driven by a force of piezoelectric effect

3.15.4
electromagnetic (moving-iron) loudspeaker
loudspeaker, the diaphragm of which is driven by a magnetic force applied to a movable part of a ferromagnetic substance

3.16 Type
3.16.1
direct radiator loudspeaker
loudspeaker that directly radiates an acoustic sound from the diaphragm

3.16.2
horn loudspeaker
loudspeaker to which an end of a horn, the cross-sectional area of which changes continuously, is attached at the front of the diaphragm, so that the other end of the horn radiates an acoustic sound

3.16.3
compression driver
loudspeaker drive unit, of which the opening area to be connected to the horn is made smaller than the diaphragm area

4 Conditions for measurements

4.1 Rated conditions

For effective and realistic testing, certain data which the manufacturer provides serves to establish a basis for measurements. These data are:

- rated impedance
- rated sinusoidal voltage or power
- rated noise voltage or power
- rated frequency range
- reference plane
- reference point
- reference axis

A full explanation of the term 'rated' is given in IEC 60268-2.

4.2 Normal measuring conditions

To establish normal measuring conditions, all of the following conditions shall be fulfilled:

a) the loudspeaker is mounted as specified in 4.7;

b) the acoustic environment is as specified by the manufacturer;

c) the distance of the microphone from the loudspeaker is as specified in 4.5;

d) the loudspeaker is supplied with a specified test signal (see 4.3), at the relevant voltage specified in this standard;

e) The interconnections between the amplifier and loudspeaker shall have a maximum impedance less than 5% of the loudspeaker's rated impedance.

f) all controls are set to the positions specified by the manufacturer

g) relevant measuring equipment is in place.

4.3 Requirements for test signals

4.3.1 Sinusoidal signal

A sinusoidal signal shall have a total harmonic distortion of less than 0,1% (-60 dB) if it is used for measuring distortion. In addition, no relative harmonic component in the loudspeaker input voltage shall be greater than one tenth of that measured in the sound pressure output of the loudspeaker.

Unless otherwise stated, the voltage across the terminals of the loudspeaker shall be kept constant during a measurement.

4.3.2 Broadband noise signal

A broadband signal should have a crest factor of between 3 and 4. The voltage shall be measured with a true rms voltmeter having a time-constant sufficiently long and a bandwidth sufficiently wide to enable the voltage to be measured with the required accuracy.

4.3.3 Narrow-band noise signal

A narrow-band signal should be derived by applying pink noise to an appropriate band-pass filter. Unless otherwise specified, 1/3-octave filters conforming to IEC 61260 shall be used.

4.4 Unwanted acoustical and electrical noises

The levels of noise shall be sufficiently low that the accuracy of measurement of the smallest wanted signals is adequately preserved. Normally, it is sufficient that the noise measured with a wanted signal is at least 10 dB lower in level than the signal.

4.5 Distance of the measuring microphone from the loudspeaker

4.5.1 Free-field and half-space measurements

For free-field and half-space measurements, the distance shall, unless otherwise stated, be sufficient to ensure that measurements are made in the far field of the loudspeaker.

> NOTE 1 For large loudspeakers, this may be very difficult to achieve, but there is at present no practical way of using near-field results to reliably derive far-field results, except at low frequencies.
>
> NOTE 2 This does not prohibit the use of other, fully described methods of measurement, where the microphone is not in the far field, which can be shown to produce the same or more accurate results.

4.5.2 Diffuse-field measurements

For diffuse-field measurements, the measurement method shall be stated.

4.6 Measuring equipment

For measurements other than in a diffuse field, a microphone having a known free-field calibration shall be used.

4.7 Mounting

Details of the mounting of the drive unit shall be recorded with the results of the measurements. The drive unit should be mounted in one of the following ways:

a) on the standard baffle specified in IEC 60268-5;
b) on a similar baffle, modified to minimize diffraction at the front surface when the loudspeaker is mounted;
c) in a plane-wave tube or duct, on a baffle similar to, but much smaller than, type a) or b);
d) in one of the standard enclosures specified in IEC 60268-5;
e) suspended in free air
f) on a large round baffle of at least 2,44 m in diameter

> NOTE a), b) or f) is appropriate for loudspeakers intended by the manufacturer to radiate into half-space.

5 Characteristics to be specified, methods of measurements and presentation of results

5.1 Characteristics and methods of measurement

5.1.1 General

Methods of measurement for various drive-unit characteristics shall be as set out in table 1. The entries in table 1 refer to clauses within IEC 60268-5, except where indicated otherwise.

5.1.2 Impedance curves

Curves of impedance against frequency should be plotted with a logarithmic impedance axis. The test signal should be within the linear range of the device. The units shall be listed as ohms. The minimum impedance, and the rated impedance, shall be indicated on the graph. Complex impedance vs frequency should be plotted.

5.1.3 Acoustic power

When reporting acoustic power, the field condition - free-field, half-space or diffuse - shall be stated.

5.1.4 Frequency response

The frequency response shall be measured at a stated voltage.

5.1.5 Harmonic distortion

5.1.5.1 Measurement signal

Harmonic Distortion should be measured with either a sine sweep or a stepped sine with adequate frequency resolution (1/24 octave suggested) over the rated frequency range. The voltage input should be set to that which produces 10 % of the full rated power into the rated impedance. Second and third harmonics shall be measured.

5.1.5.1 Measurement report

Distortion measurements shall present values of both second- and third-harmonic components compared with the fundamental frequency. The levels of distortion shall be expressed in decibels referenced to the level of the fundamental.

Table 1 - Characteristics and methods of measurement

Type / Characteristic	HF driver	HF horn	LF driver
Principle	13.2.1		
Type	13.2.2		
Reference plane, point and axis	15		
Rated and minimum impedances	16.1 The rated impedance and the minimum impedance within the effective frequency range shall be stated The value of a pure resistance which is to be substituted for the loudspeaker when defining the available electric power of the source shall be specified by the manufacturer. The lowest value of the modulus of the impedance in the rated frequency range shall be not less than 80 % of the rated impedance. If the impedance at any frequency outside this range (including d.c.) is less than this value, this shall be stated in the specifications.		
Impedance/ frequency curve	16.2 See also 5.1.2 of this standard		
Acoustic power	22.1.2 See also 5.1.3 of this standard		
Frequency response	See 5.2 of this standard	21.1 See also 5.1.4 of this standard	
Effective frequency range	Not applicable	21.2	
Sensitivity	Not applicable	Calculated from the frequency response and effective frequency range, as the sound pressure level produced at 1 m on the reference axis by an applied voltage of 2,83 V. Narrow-band sensitivity: the test signal is 1/3-octave filtered noise centered at 1 kHz, or at the geometric mean of the limit frequencies of the effective frequency range if different from 1 kHz. The frequency shall be stated. Broad-band sensitivity: the test signal is 2-octave filtered noise centered at 1 kHz, or at the geometric mean of the limit frequencies of the effective frequency range if different from 1 kHz. The frequency shall be stated.	
Directional response	Not applicable	See 5.6 of this standard	
Directivity index	Not applicable	23.3	
Harmonic distortion	24.3. See also 5.1.5 of this standard		
Modulation distortion	Not applicable	24.4, 24.5	
Rated noise voltage or power	See 6 of this standard		
Power compression	See 5.3 of this standard		
Thiele-Small parameters	19.2. See also 5.5 of this standard. Only f_S and X_{MAX} apply.		16.3, 16.4,19.2 See also 5.4 and 5.5 of this standard
Dimensions and mounting data	27.1		
Mass	27.2		

NOTE unless otherwise stated, references are to IEC 60268-5 Ed.3 2003, incl. Am.1 2007.

5.2 Method of measurement of frequency response using a plane-wave tube

The frequency response of high-frequency drivers should be measured with a plane-wave tube.

5.3 Method of measurement for maximum usable continuous output sound-pressure level

5.3.1 Measurement Set-up

The loudspeaker driver, or device under test (DUT), shall be loaded in a manner similar to that of normal operating conditions. The test signal to be used shall be the Simulated Program Signal defined in clause 7 of IEC 60268-1, edition 2 (1985).

A transfer function measurement with a frequency resolution of at least 1/3 octave within the pass band limits of the DUT shall be used to measure the amplitude linearity of the output of the DUT. The reference for the transfer function measurement shall be either the output of the power amplifier directly driving the DUT or the output of the level control device used to adjust the amplitude of the test signal.

The signal flow for the measurement shall be that shown in figure 1. The output of the signal source shall be connected to the input of a level control device as a means to increase or decrease the amplitude of the signal during the test. The output of the level control shall be connected to the input of a power amplifier. The power amplifier shall drive the DUT.

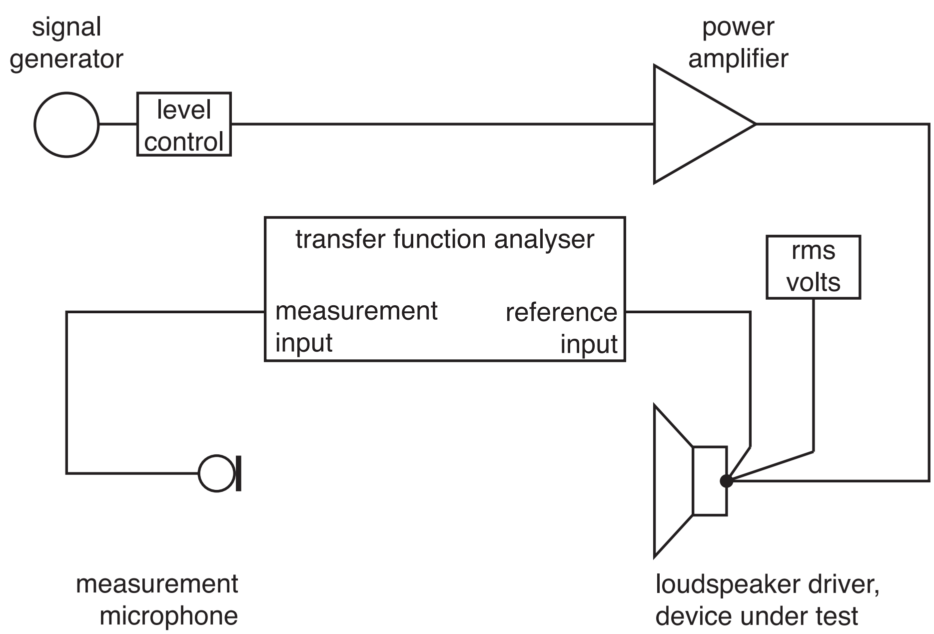

Figure 1 - Test setup for maximum usable continuous output sound-pressure level

An rms-reading voltmeter shall be connected across the loudspeaker terminals to measure the signal amplitude while testing the DUT.

The measurement microphone shall be used to measure sound pressure level (SPL). The distance from the DUT to the measurement microphone should be sufficient so that the magnitude of the radiation from one area of the DUT is not overly emphasized compared to the other areas.

5.3.2 Measurement Procedure

The level control should be adjusted so that the broadband rms amplitude of the test signal at the input to the loudspeaker is 1,0 V, or lower if necessary to ensure linear operation. The frequency response shall be recorded and stored. This result shall be used as the reference for the normalization of all subsequent measurements, such that each subsequent measurement is divided by the reference.

> NOTE By normalizing to this result, all other measurement results are converted into a flat response at 0 dB so long as there is no change in the amplitude response of the DUT.

The level control should be increased in 3-dB increments. At each increment, the frequency response shall be normalized to the reference response. The normalized result shall be examined after at least 1 minute to determine the amount of change in the amplitude response of the DUT. When the normalized transfer function has changed by at least 1 dB at any point within the pass-band limits of the DUT, the broad-band rms voltage at the input to the DUT and the amount of deviation should be recorded.

The level control shall then continue to be increased in 1 dB increments. At each increment the normalized frequency response shall be examined after at least 1 minute to determine the amount of change in the amplitude response of the DUT. When the response has changed by approximately 3 dB (but no more than 3,5 dB) compared with the reference response, at any point within the pass-band limits of the DUT, the broad-band rms voltage at the input to the DUT shall be recorded as the Maximum Input Voltage, Continuous (MIV continuous) for the DUT.

5.3.3 Anechoic measurement

If the test is being conducted in an anechoic environment the SPL shall be recorded. This SPL shall be referenced to a distance of 1 meter and shall be the Maximum Usable Continuous Output SPL (*SPL muco*) of the DUT.

The transfer function and the normalized transfer function at this input voltage shall also be recorded.

5.3.4 Non-anechoic measurement

After the procedure in 5.3.2, multiply (in the frequency domain) the normalization reference and the normalized transfer function recorded at the conclusion of the test.

Then multiply this result with the normalized spectral content of the simulated program test signal (see 5.3.1). To this result add the gain level represented by the increased signal level from MIV continuous [Gain = 20log(*MIVc*)]. Finally, take the broad-band SPL of this result and reference it to 1 metre to yield a suitably anechoic representation of the DUT if was driven at its *MIVc*.

5.4 Methods of measurement of Thiele-Small parameters

5.4.1 Method using added mass

Suspend the driver in free air and supply a variable-frequency sinusoidal signal at a low level (typically 0,1 V). Observe the supply voltage and current with an oscilloscope having an X-Y display and determine the resonance frequency, using a counter, by detecting the zero-phase frequency. Alternatively, a frequency-dependent measurement of the complex impedance may be used to determine the zero-angle frequency corresponding to resonance. It is likely to be necessary for the driver to be operated for many minutes before measurements are made, so as to allow the resonance frequency to stop drifting.

Repeat the measurement with mass added to the cone so as to reduce the resonance frequency by 10 % to 20 %. This amount of added mass is experimentally determined and shall be made from a material that has no inductive or magnetic effect on the loudspeaker performance. The means of attachment shall be such that no audible independent vibration of the added mass occurs. The DUT should be oriented so that the added mass does not affect voice-coil rest position in the magnetic gap.

The mechanical moving mass M is then given by:

$$M = mf^2/(F^2 - f^2), \text{ where:}$$

m = added mass
F = original resonance frequency
f = resonance frequency with added mass.

This value of M includes the air mass load on both sides of the cone (see below). From the cone mass, the compliance and the equivalent volume can be calculated conventionally. Sources of error include: vibration of the added mass; non-linearity of the suspension; and rounding errors affecting the divisor in the equation (- the difference of the squares of two nearly equal quantities).

5.4.2 Method using a change in enclosure volume

Mount the driver in a carefully sealed rigid enclosure whose volume is such as to raise the resonance frequency by 10 % to 15 %. The enclosure shall be provided with a very small 'anti-aneroid' leakage hole (usually about 1 mm diameter) to prevent build-up of static air pressure inside it. The required enclosure volume is determined experimentally and may be achieved by inserting rigid blocks of wood or other material into an enclosure of greater volume than required. Non-rigid or low-density materials, such as expanded plastics, are unsuitable. The integrity of the sealing may be checked by increasing the internal pressure of the enclosure slightly by means of an air pump and noting the time for the driver cone to return to its stable position. Apply to the driver a variable-frequency sinusoidal signal at a low level (typically 0,1 V). Observe the supply voltage and current with an oscilloscope having an X-Y display and determine the resonance frequency, using a counter, by detecting the zero-phase frequency. Alternatively, a frequency dependent measurement of the complex impedance may be utilized to determine the zero angle frequency corresponding to resonance. It is likely to be necessary for the driver to be operated for many minutes before measurements are made, so as to allow the resonance frequency to stabilize.

Repeat the measurement with the volume of the enclosure decreased, by adding rigid blocks, so that the resonance frequency is raised by a further 10 % to 15 %.

The mechanical suspension compliance C is then given approximately by:

$$C = (F^2V_1 - f^2V_2)/\{V_1V_2(f^2 - F^2)\}, \text{ where:}$$

V_1 = original enclosure volume;
F = original resonance frequency;
f = resonance frequency with reduced volume V_2.

Sources of error include: inadequate sealing of the enclosure, vibration of the walls and/or contents of the enclosure, rounding errors affecting the denominator in the equation (- the difference of the squares of two nearly equal quantities).

5.4.3 Method using an enclosure with a sealable port

See IEC 60268-5, 16.4.

5.5 Measurement of maximum useful displacement of the voice-coil or equivalent element

5.5.1 Characteristic to be specified

The peak displacement of the voice-coil or equivalent element at which neither the total harmonic distortion d_t or the *n*th-order modulation distortion (where n=2 or 3) exceeds 10 % in the radiated sound pressure.

5.5.2 Method of measurement

The driver is operated in free air and is excited by the linear superposition of a first tone at the resonance frequency $f_1 = f_s$ and a second tone $f_2 = 8.5 f_s$ with an amplitude ratio of 4:1. The sound pressure is measured with the measurement microphone in the near field. The microphone signal is passed to a distortion analyzer. The total harmonic distortion d_t assesses the harmonics of f_1 and the modulation distortions are measured by the modulation components $f_2 \pm nf_1$.

Displacement is usefully measured with a laser displacement sensor.

This method is described more fully in [1] and [6].

NOTE The difference in definitions between IEC 60268-5 and this standard is not expected to cause any difficulty since the limiting percentage distortion is only 10 %. See also 5.1.5.

5.6 Directional responses

This standard on measurement of loudspeaker drive units does not consider directional measurements. Where appropriate, see AES56 [4].

6 Power-Handling

6.1 Test Conditions and Equipment

A compression driver shall be mounted on an appropriate constant- or expanding-area acoustical load whose initial area is no smaller than that of the driver exit. The manufacturer shall specify the method of loading, including mechanical dimensions. A direct radiator shall be mounted in free air at sufficient distance from surrounding surfaces such that there is no substantial acoustic loading.

The driver shall be excited with noise extending one decade upward from the manufacturer's stated low-frequency (lf) limit of the driver. The noise shall be pink noise, bandpass filtered at 24 dB per octave, with Butterworth filter response characteristics, and the peak-to-rms voltage ratio of the noise signal supplied to the driver shall be 4:1 (12 dB).

The amplifier for the test should be capable of passing voltage peaks at least 9 dB, but preferably 12 dB higher than the rms voltage. This requires an amplifier rating at least 4 times, but preferably 8 times the power level of the test.

The spectrum, rms voltage and crest factor shall be verified at the input terminals of the driver under test. The voltage shall be measured with a true rms voltmeter having a bandwidth greater than or equal to the noise signal upper cutoff frequency, a time constant or block-averaging time equal to, or greater than, twenty times the period of the lowest frequency of interest, and having the capability to measure crest factors of at least 4 (that is, 12 dB).

The manufacturer shall state the upper and lower cutoff frequencies (-3 dB) of the noise signal.

6.2 Test procedure

The device under test shall be subjected to successively higher powers and allowed to reach thermal equilibrium at each increment (approximately 2 h). Power shall be determined as the square of applied rms voltage, as measured with a true rms voltmeter, divided by the rated impedance. The rated power of the device shall be that power the device can withstand for 2 h without significant permanent change in acoustical, mechanical, or electrical characteristics.

6.3 Displacement limit

The manufacturer shall specify the maximum excursion of the device which, when exceeded, results in permanent mechanical damage to the device. The manufacturer shall state the cause of damage (for example: elastic limit of suspension, or the diaphragm striking a mechanical stop).

6.4 Thermal test information

The temperature rise of the voice coil and magnet assembly at the end of the 2-h power-handling test shall be stated. The manufacturer shall state the method of temperature measurement.

6.5 Low Resonance

The frequency of the low resonance shall be stated as measured before and after the power test.

Annex A: (Informative) - Informative references

[1] **Klippel, W.**, *Assessment of Voice Coil Peak Displacement X_{MAX}*, JAES Volume 51 Issue 5 pp. 307-324; May 2003. www.aes.org/e-lib/ Audio Engineering Society, New York, NY., US.

[2] **AES-1id**: *AES Information Document - Plane wave tubes: design and practice*. Audio Engineering Society, New York, NY., US.

[3] **AES-5id**: *AES information document for Room acoustics and sound reinforcement systems - Loudspeaker modeling and measurement - Frequency and angular resolution for measuring, presenting and predicting loudspeaker polar data*. Audio Engineering Society, New York, NY., US.

[4] **AES56**: *AES standard on acoustics - Sound source modeling - Loudspeaker polar radiation measurements*. Audio Engineering Society, New York, NY., US.

[5] **ANSI CEA-426-B**: *Loudspeaker, Optimum Amplifier Power*. Consumer Electronics Association, Arlington, VA., US.

[6] **IEC 62458,** *Sound System Equipment - electroacoustic transducers - Measurement of large signal parameters*. International Electrotechnical Commission, Geneva, Switzerland.

Annex B (Informative) - Crest Factor

B.1 Crest Factor of Random Noise

Crest factor is defined as, "the ratio of the peak amplitude to the time-averaged rms amplitude of an alternating waveform".

The crest factor of gaussian random noise is theoretically infinite, because a gaussian distribution is unbounded. However, extreme values are rare. A truly gaussian white-noise source sampled at 48 kHz can be expected to produce a sample 12 dB or more above the rms value about once every 200 milliseconds; a sample at +15 dB can be expected about once every 13 minutes; a sample at +18 dB can be expected about once every 200 years. So, in human terms, the crest factor of random noise is not actually "very infinite": but neither is it characterized by a precise value. A source which happens to produce a rare outlying value would be measured as having a high crest factor, but it is not significantly different in effect from a source that hasn't produced such a value yet.

Pink noise also has a Gaussian distribution, but the rate of occurrence of extreme values is lower. This can be attributed to the averaging effect of the pinking filter or, if preferred, to the increased weighting of low frequencies; which has an effect similar to lowering the sample rate. For a pink noise source covering the spectrum from 20 Hz to 20 kHz, extreme values can be expected to occur about one tenth as often as they do in a white-noise signal with a bandwidth up to 24 kHz: a +12 dB sample can be expected to occur about once every 2 seconds; a +15 dB sample can be expected about once every two hours. Here is a chart of the average occurrence rate with which samples exceed levels between 9 dB and 15 dB.

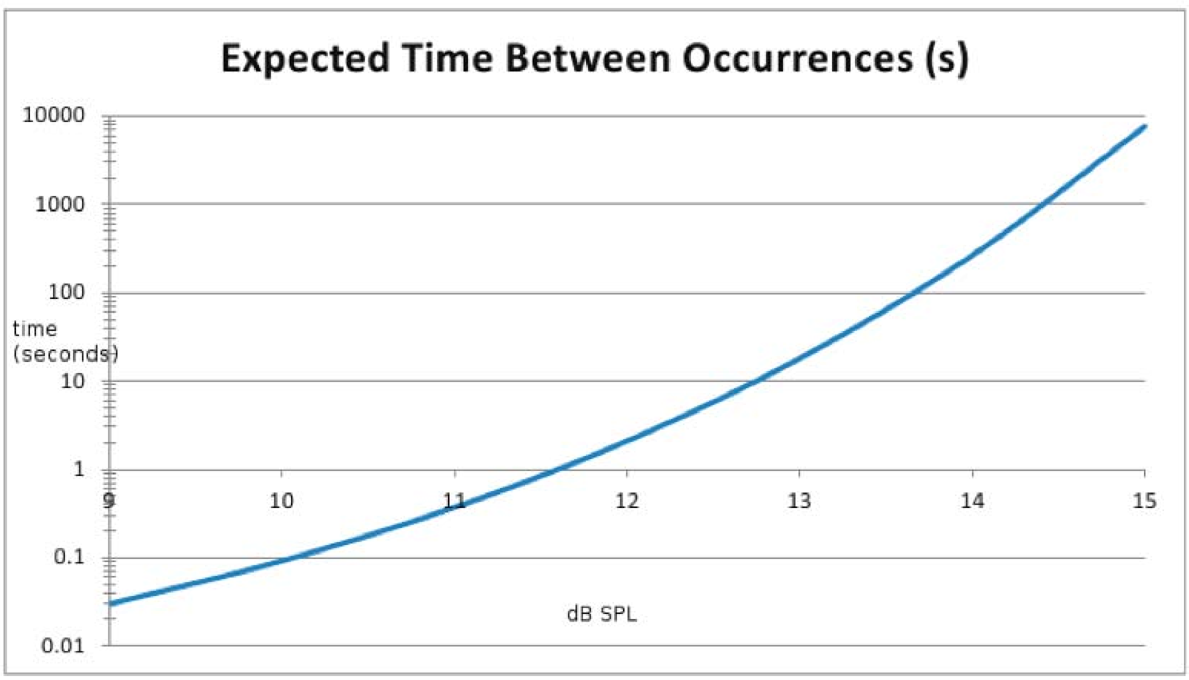

Figure B.1 - Repetition rate between 9 and 15 dB

B.2 Signal Organization

There are several applicable definitions of "random", with regard to noise signals. White noise is completely random: each sample is completely independent of all other samples. If the samples are randomly re-ordered, the spectrum will still be white. In the frequency domain the magnitudes and/or phases of the frequency components can be randomly exchanged and the noise will still be white.

Pink noise is somewhat less random. Each sample is more similar to its nearby neighbors than it is to distant samples. If the samples are randomly re-ordered, the result will not have a pink spectrum - it will be white. Obviously, the magnitudes of the frequency components cannot be exchanged, but the phases of the frequency components can be randomly exchanged - and the result will still be pink noise.

Importantly, the characteristic that defines “random”, as it applies to pink noise, is that there is no organization to the phase response. Not only can the phase components be shuffled, but pink noise can be passed through an arbitrary all-pass filter, and it will still have all of the characteristics of pink noise, including the spectral content and the approximately normal distribution of the sample values.

A sampled or modeled noise source of typical length (0,5 s to 5 s) will have a crest factor of approximately 12 dB. This is the expected crest factor for a pink noise source of that length with no phase organization. A lower crest factor can only be obtained by organizing the phase response in a very specific way, and some loudspeaker measurement methods require this (see [5]). This can be achieved by applying compression and/or soft clipping and by adjusting the magnitudes of the frequency components, and by iterating that process until both the magnitude response and the crest factor are within acceptable tolerance. Whether by this algorithm or any other, a reduction of the crest factor of pink noise can only be achieved by organizing the phase components. That organization is very fragile, by which is meant that any filter that affects the phase and/or magnitude response even slightly will affect the way the frequency components sum and the crest factor will revert toward 12 dB.

To demonstrate this effect, a CEA-426-B signal from a Wave file was measured in a signal analysis program called Sigview, the crest factor was measured as a precise 6,01 dB. After applying a second order Butterworth high pass filter at 20 Hz, the crest factor of the signal increased to 8,3 dB.

A fourth-order 40 Hz Butterworth filter was selected to represent a typical protective high pass filter. With this filter applied to the 6 dB crest factor signal, the crest factor increased to 11,6 dB. The rms level only dropped by 0,1 dB because CEA-426-B does not contain very much energy below 40 Hz, but the peak levels increased by 5,7 dB. The radical change in summation can be presumed to be caused primarily by the phase shift of the high pass filter.

A fourth-order Linkwitz-Riley high-pass filter at 80 Hz, combined with a Linkwitz-Riley low-pass filter at 1500 Hz were selected to represent a filter set that might be applied to the woofer in a multi-way system: When applied to the 6-dB crest-factor noise signal, the filtered signal displayed a 12,4-dB crest factor.

For a more complete example, the 6 dB crest factor signal was passed through an FIR-based one-way loudspeaker processor (Fulcrum Acoustic CX1295), which produced a 12,3-dB crest factor, representative of the amplifier output voltage. Convolving this signal with the measured response of the processed loudspeaker produced a 13,2 dB crest factor, representing the crest factor of the sound pressure.

Repeating the analysis with a similarly-filtered noise source but without the crest-factor restriction provided a comparison for the results shown in table B.1.

Table B.1: Crest factor of noise

	CEA-426-B shaped noise, 6 dB restricted crest factor	**Shaped noise, unrestricted crest factor**
Source-signal crest factor	6,0 dB	11,6 dB
With HPF, 20 Hz BW2	8,3 dB	11,7 dB
With HPF, 40 Hz BW4	11,6 dB	11,7 dB
With BPF, 80 Hz to 1500 Hz	12,4 dB	12,6 dB

Complex Processor Output	12,3 dB	10,8 dB
Loudspeaker Output	13,1 dB	11,6 dB

The crest factor of recorded music is also affected by linear filters. Table B.2 presents the change in crest factor for several genres of music, when passed through varying degrees of filtering. The recordings were selected to demonstrate the effects of different amounts of dynamic-range compression: an over-compressed album by Metallica, a lightly compressed pop song by Alana Davis, and an uncompressed song by Diana Krall. The Metallica recording is squeezed into 8,4 dB in the source file, but it “expands” to a crest factor over 13 dB once it is passed through a loudspeaker processor. The Diana Krall song’s crest factor is relatively unaffected by any of the filters.

Table B.2: Crest-factor comparison of noise and music

	“All Nightmare Long” Metallica	**“Love and Pride” Alana Davis**	**“Popsicle Toes” Diana Krall**
Source-signal crest factor	8,4 dB	15,8 dB	18,7 dB
With HPF, 20 Hz BW2	9,8 dB	17,1 dB	19,0 dB
With HPF, 40 Hz BW4	12,0 dB	18,0 dB	18,6 dB
With BPF, 80 Hz to 1500 Hz	13,6 dB	17,9 dB	18,3 dB
Complex Processor Output	13,2 dB	17,1 dB	18,3 dB
Loudspeaker Output	13,8 dB	18,7 dB	19,3 dB

B.3 Implications for Loudspeaker Testing

Random noise has a natural crest factor of approximately 12 dB. If the phase components of a noise loop are carefully massaged, a signal can be created that has a smaller crest factor, like 6 dB; but the phase relationships that constrain the crest factor are very delicate, and even very gentle filtering will cause the crest factor to revert towards that of random noise.

The examples above make it very clear that though often assumed, it is not the case that a noise source restricted to a 6-dB crest factor will produce an amplifier voltage with a crest factor of approximately 6 dB; nor is it true that a loudspeaker driven by that amplifier will produce a sound pressure with a crest factor of approximately 6 dB. In fact it would require an inordinate amount of effort to constrain an amplifier voltage to 6 dB crest factor.

It might be presumed that the origin of the 6-dB crest factor was a simple expedient: “It would be convenient to be able to test a loudspeaker with an amplifier as small as two times the power to be tested. To make this possible and ensure consistency, we’ll also constrain the crest factor when ample power is available.”

Considering the size of amplifiers that are now commonly available, there appears to be no benefit to a reduced crest factor. In fact, a test signal that is more similar to music should be preferable – keeping in mind that even hyper-compressed recorded music expands to 12 dB crest factor, or greater, once it encounters a filter.

One producer of test signals with a restricted crest factor has observed, “Loudspeaker power and life-test standards require a specific crest factor for the noise signals that are used to energize the loudspeaker. In most situations, when the test signal is measured at the speaker terminals, it is found that the crest factor is significantly higher than the generated test signal. This increase in crest factor is directly due to phase non-linearities of the high- and low-pass filters in the transmission chain between the noise generator output and the speaker’s terminals. For a 20 to 100-Hz band-limited pink noise signal with a crest factor of 6 dB, a simple first-order high pass signal at 2 Hz will raise the crest factor of the test signal by more than 0.5 dB. To ensure that the crest factors will not change, a DC-coupled path is required between the noise generator and the speaker’s terminal. A CD player cannot be used to properly reproduce the test signal because it is not

DC-coupled. Only a computer data acquisition card with a DC-capable D/A converter will generate the proper noise signal."

Because of the difficulty of preserving a 6 dB crest factor, it is believed that most companies have in fact been testing with a higher crest factor for many years. It appears that the standard noise-source specification could be changed with minimal effect on the power-handling ratings of existing products.