

INTERNATIONAL STANDARD

NORME INTERNATIONALE



**Sound system equipment –
Part 21: Acoustical (output-based) measurements**

**Équipements pour systèmes électroacoustiques –
Partie 21: Mesures acoustiques (basées sur la sortie)**



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SOUND SYSTEM EQUIPMENT –**Part 21: Acoustical (output-based) measurements****FOREWORD**

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Full information on the voting for the approval of this International Standard can be found in the report on voting indicated in the above table.

The French version of this standard has not been voted upon.

This document has been drafted in accordance with the ISO/IEC Directives, Part 2.

A list of all parts in the IEC 60628, published under the general title *Sound system equipment*, can be found on the IEC website.

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INTRODUCTION

Loudspeakers, headphones and other actuators have become more versatile and, as a result, new measurement techniques are required to evaluate these systems. The following is a list of examples where new measurement techniques are required:

- Limited access to the electrical terminals of the transducer
The higher integration of electrical, acoustical and mechanical elements limit the access to the electrical terminals of the transducer.
- Analogue or digital audio input signals
Audio inputs can accept analogue or digital signals in various formats.
- Latency and other kinds of distortion associated with digital signal processing
Digital signal processing is used to correct the transfer behaviour of the passive system and to generate a desired sound output and as a result, latency and other kinds of distortion not found in analogue equipment can be generated.
- Excessive equalization
Excessive equalization can force the transducer to operate in the large signal domain causing thermal and nonlinear effects.
- Active protection
Active protection attenuates the input signal to prevent a mechanical and thermal overload of the transducer and other components.
- Other transducer principles
Although most loudspeaker systems use a moving coil in an electro-dynamical transducer, there is a need to expand the application to electro-static, electro-magnetic or any other transduction principles.
- Other mechanical and acoustical elements
To improve sound radiation, vented enclosures, sealed enclosures, passive radiators, horns, wave guides, flat panels, and other mechanical and acoustical elements are implemented.
- Impulsive distortions
Defects in manufacturing (e.g. voice coil rubbing) or operating under overload conditions can create impulsive distortions, which have a high impact on perceived sound quality but cannot be detected by conventional measurements (e.g. total harmonic distortion).
- Directional characteristics and complex near field properties
The comprehensive evaluation of professional equipment, including directional characteristics, can be realized by considering the complex near-field properties as a supplement to the existing far-field measurement techniques. In addition, devices intended for use in the near field, such as hand-held personal audio devices (e.g. laptops, tablets, smart phones) and other portable sound systems, need to be evaluated in a manner appropriate to their intended use.

SOUND SYSTEM EQUIPMENT –

Part 21: Acoustical (output-based) measurements

1 Scope

This part of IEC 60268 specifies an acoustical measurement method that applies to electro-acoustical transducers and passive and active sound systems, such as loudspeakers, TV-sets, multi-media devices, personal portable audio devices, automotive sound systems and professional equipment. The device under test (DUT) can be comprised of electrical components performing analogue and digital signal processing prior to the passive actuators performing a transduction of the electrical input into an acoustical output signal. This document describes only physical measurements that assess the transfer behaviour of the DUT between an arbitrary analogue or digital input signal and the acoustical output at any point in the near and far field of the system. This includes operating the DUT in both the small and large signal domains. The influence of the acoustical boundary conditions of the target application (e.g. car interior) can also be considered in the physical evaluation of the sound system. This document does not assess the perception and cognitive evaluation of the reproduced sound and the impact of perceived sound quality.

NOTE Some measurement methods defined in this document can be applied to headphones, headsets, earphones and earsets in accordance with [1]¹. This document does not apply to microphones and other sensors. This document does not require access to the state variables (voltage, current) at the electrical terminals of the transducer. Sensitivity, electric input power and other characteristics based on the electrical impedance will be described in a separate future standard document, IEC 60268-22, dedicated to electrical and mechanical measurements.

2 Normative references

The following documents are referred to in the text in such a way that some or all of their content constitutes requirements 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 60263, *Scales and sizes for plotting frequency characteristics and polar diagrams*

IEC 60268-1, *Sound system equipment – Part 1: General*

IEC 60268-2:1987, *Sound system equipment – Part 2: Explanation of general terms and calculation methods*

IEC 61094-4, *Measurement microphones – Part 4: Specifications for working standard microphones*

IEC 61260-1, *Electroacoustics – Octave-band and fractional-octave-band filters – Part 1: Specifications*

ISO 3, *Preferred numbers – Series of preferred numbers*

ISO 3741:2010, *Acoustics – Determination of sound power levels and sound energy levels of noise sources using sound pressure – Precision methods for reverberation test rooms*

¹ Numbers in square brackets refer to the Bibliography.

ISO 3744, *Acoustics – Determination of sound power levels and sound energy levels of noise sources using sound pressure – Engineering methods for an essentially free field over a reflecting plane*

ISO 3745, *Acoustics – Determination of sound power levels and sound energy levels of noise sources using sound pressure – Precision methods for anechoic rooms and hemi-anechoic rooms*

CTA 2034-A, *Standard Method of Measurement for In-Home Loudspeakers*, Consumer Technology Association (Formerly CEA), 02/01/2015

CTA 2010-B, *Standard Method of Measurement for Powered Subwoofers*, standard by Consumer Technology Association (Formerly CEA), 11/28/2014

3 Terms, definitions and abbreviated terms

3.1 Terms and definitions

No terms and definitions are listed in this document.

ISO and IEC maintain terminological databases for use in standardization at the following addresses:

- IEC Electropedia: available at <http://www.electropedia.org/>
- ISO Online browsing platform: available at <http://www.iso.org/obp>

3.2 Abbreviated terms

DUT device under test

4 Type description

The type description shall be provided by the manufacturer, including the following information:

- type, principles and number of the transducers used in the loudspeaker system;
- acoustical loading (e.g. enclosure, horn, bass reflex, column, line array);
- power amplification;
- DSP processing (e.g. equalizer, active protection).

5 Physical characteristics

5.1 Marking of terminals and controls

The terminals and controls shall be marked in accordance with IEC 60268-1 and IEC 60268-2.

5.2 Dimensions

The outer dimensions of the DUT shall be specified.

5.3 Mass

The total mass of the DUT when ready for use shall be specified.

5.4 Connectors and cable assemblies

Cable assemblies and connectors shall be in accordance with IEC 60268-11 and IEC 60268-12.

NOTE In some circumstances, the connectors that are currently standardized are unsuitable and the use of other types is unavoidable.

6 Design data

Further design data shall be specified as additional information such as:

- type of transducer principle;
- number of transducers (drive units);
- digital processing of the audio signal (equalization, linearization, active protection).

7 Conditions

7.1 Rated conditions

For convenience, this document specifies how sound system equipment shall be set up for measurement. Normal measuring conditions are defined in this document. To obtain the actual conditions for measurement, some values (known as "rated conditions") shall be taken from the manufacturer's specification.

These rated conditions are not subject to measurement, but they constitute the basis for performing the measurements to determine the other characteristics.

The following rated conditions are of this type, and shall be stated by the manufacturer:

- rated maximum sound pressure output or maximum input value;
- rated frequency range;
- geometrical conditions as per 10.1;
- ambient conditions as per Clause 15.

7.2 Climatic conditions

IEC 60268-1 states that tests should be carried out in the following environment in order to prevent the influence of temperature and humidity that can affect the properties of the drive unit suspensions [2]:

- ambient temperature T_{norm} : 15 °C to 35 °C;
- relative humidity: 25 % to 75 %;
- air pressure: 86 kPa to 106 kPa.

7.3 Normal measuring conditions

The DUT shall be understood to be under normal measuring conditions if all the following conditions are defined:

- a) The DUT to be measured is mounted in accordance with Clause 13;
- b) The acoustical environment is specified and selected from those given in Clause 9;
- c) Unwanted acoustical signals, electrical signals, and noise generated by other sources shall be kept at the lowest levels possible because their presence may obscure low-level signals. Data related to signals, which are less than 20 dB above the noise level in the frequency band being considered, shall be discarded or marked as corrupted by noise;

- d) The DUT is positioned with respect to the measuring microphone and the walls in accordance with Clause 10;
- e) The DUT is acclimatized to the ambient condition in accordance with IEC 60268-1 or the DUT has a defined working temperature that is typical for the target application and determined by measuring the compression in accordance with 19.4;

NOTE The influence of the ambient atmospheric condition on acoustic measurements is discussed in greater detail in [3].

- f) Additional cooling periods are required between successive tests if compression $C > 0,5$ dB in accordance with 19.4;
- g) The DUT is supplied with a test signal with specified properties (spectrum, duration, etc.) in accordance with Clause 8 at a specified RMS input value $\tilde{u}$ for the rated frequency range in accordance with Clause 16;
- h) Attenuators, equalizers, dynamics or any other active control elements, shall be set to their "normal" position as stated by the manufacturer. If other positions are chosen, for example, those providing a maximally flat frequency response or minimum attenuation, they shall be specified;
- i) Measuring equipment suitable for determining the wanted characteristics is connected in accordance with Clause 11.

8 Test signals

8.1 General

Some measurements can be performed by using any audio signal $x(t)$ as an input signal (stimulus) applied to the electro-acoustical device under test (DUT) while other measurement techniques use the test signals outlined in 8.2 to 8.9.

8.2 Sinusoidal chirp

The transient sinusoidal chirp (gliding tone, sweep) is defined as

$$x_c(t) = \sqrt{2}A(f(t))\cos(2\pi f(t)t)$$

with the amplitude $A(f(t))$ depending on the instantaneous frequency

$$f(t) = f_{\text{start}} 2^{\beta t} \quad 0 \leq t \leq T_s$$

varying logarithmically with time from the starting frequency f_{start} to the end frequency f_{end} during sweep length T_s , which is $0,2 \text{ s} < T_s < 10 \text{ s}$ typically. The sweep rate parameter

$$\beta = \frac{1}{T_s} \log_2 \left(\frac{f_{\text{end}}}{f_{\text{start}}} \right)$$

describes the time required for doubling the instantaneous frequency.

A constant value can be applied to the amplitude $A(f(t))$ but an amplitude varying with frequency can be used to protect the DUT in particular frequency bands. Ultra-fast measurements, as sometimes required in end-of-line testing, require other chirp signals where also the sweep rate $\beta(f)$ of the instantaneous sinusoidal signal varies with frequency f . The varying rate profile can be used to realize a desired power density spectrum of the stimulus and provide sufficient accuracy for the frequency band of interest and a desired peak value of the signal $x_c(t)$.

8.3 Steady-state single-tone signal

The electro-acoustical system is excited by a single tone

$$x_s(t) = \sqrt{2} \cos(2\pi f_1 t) \quad 0 \leq t \leq T_P + T_M = \frac{N_P + N_M}{f_1}$$

of defined frequency f_1 . The pre-excitation time $T_P = N_P/f_1$ corresponding with a number of periods $N_P > 5$ is required to generate the steady-state condition of the electro-acoustical system. The measurement time

$$T_M = \frac{N_M}{f_1}$$

with a number of periods $N_M \geq 1$ is recommended for temporal averaging and time-discrete signal processing.

8.4 Steady-state two-tone signal

The electro-acoustical system is excited by two tones

$$x_t(t) = \sqrt{\frac{2}{A_1^2 + A_2^2}} (A_1 \cos(2\pi f_1 t) + A_2 \cos(2\pi f_2 t)) \quad 0 \leq t \leq (T_P + T_M)$$

of defined frequencies $f_1 < f_2$ and amplitude scaling A_1 and A_2 . The pre-excitation time $T_P = N_P/f_1$ corresponding with a minimal number of repetitions $N_P > 5$ is required to generate the steady-state condition of the DUT. The measurement time

$$T_M = \frac{N_M}{f_1 f_2} 1\text{Hz}$$

corresponding with a given number of repetitions $N_M \geq 1$ is comprised of an integer number of periods of both tones. This is recommended for temporal averaging and time-discrete signal processing.

8.5 Sparse multi-tone complex

The multi-tone complex defined by

$$x_m(t) = \sqrt{\frac{2}{\sum_{i=0}^N A(f_i)^2}} \sum_{i=0}^N A(f_i) \cos(2\pi f_i t + \varphi_i) \quad 0 < t \leq T_P + T_M$$

comprises a multitude of $(N + 1)$ sinusoidal tones with amplitude $A(f_i)$ at logarithmically spaced frequencies

$$f_i = f_b \text{int}\left(T \cdot f_0 \cdot 2^{i/R}\right) \quad \text{with } i = 0, \dots, N$$

between the lowest tone f_0 and the highest tone f_N at multiples of the base frequency f_b . The base frequency f_b defined by

$$f_b = \frac{1}{T} = \frac{f_s}{N_{FT}}$$

corresponds to the minimal length T of the multi-tone complex, sampling frequency f_s and the size N_{FT} of the Fourier transform (DFT or FFT) to avoid a smearing of the fundamental tones over multiple frequency bins.

The resolution parameter R describes the number of tones per octave. The tones have a pseudo-random phase

$$\phi_i = \frac{2\pi}{m} n_i \quad \text{for } i = 0, \dots, N$$

with

$$n_i = (an_{i-1} + c) \bmod_m \quad \text{for } i = 1, \dots, N$$

generating a deterministic waveform of the stimulus [4].

NOTE The phases of the low-frequency tones determine the waveform of the voice coil displacement, which has a high impact on the nonlinear distortion generated by motor and suspension nonlinearities.

The modulus parameter m is usually defined by the digital word size used. The other free parameters a , c and the start value n_0 (seed) shall be positive and smaller than m . Modifying these parameters affects the randomness of the generated phases and thus the crest factor of the resulting time signal. The crest factor may be optimized for a specific frequency range, signal length and resolution. Reasonable default values are $a = 843\,314\,861$, $c = 453\,816\,693$, $m = 2^{31}$ and $n_0 = 0$. If other parameters are used, they shall be stated.

The modulo computation shall be done exactly with no round-off error. For the given parameters, this can be achieved by using data type 32-bit unsigned integer.

The pre-excitation time T_p with $0,5 T < T_p < T$ is required to ensure steady-state condition of the electro-acoustical system under test. The measurement time $T_M = N_M T$ with an integer number of repetitions $N_M \geq 1$ is recommended for temporal averaging and Fourier analysis.

8.6 Broadband noise signal

As defined in IEC 60268-2:1987, 1.4.

8.7 Narrow-band noise signal

As defined in IEC 60268-2:1987, 1.4.

8.8 Hann-burst signal

The Hann-burst signal,

$$x_b(t) = \begin{cases} \left(1 - \cos\left(\frac{2\pi f_0 t}{6,5}\right) \frac{\sin(2\pi f_0 t)}{2}\right) & \text{for } 0 \leq t \leq 6,5 / f_0 \\ 0 & \text{, elsewhere} \end{cases}$$

is defined as the time product between a continuous sine wave of frequency f_0 and an Hann window of 6,5 cycles in length, generating a bandwidth of one third of an octave [5].

8.9 Impulsive signal

As defined in IEC 60268-2.

9 Acoustical environment

9.1 General

Acoustical measurements shall be made under one of the conditions in the following subclauses. The acoustical environment used for testing shall be stated.

9.2 Free-field conditions

Acoustical conditions that approach those of free space may be used. Any environment shall be considered satisfactory when the reflected sound components are sufficiently suppressed to ensure an accuracy of $\pm 0,5$ dB and $\pm 10^\circ$ in the measured sound pressure amplitude and phase values at the specified frequency, respectively. If the environment (e.g. anechoic room at low frequencies) does not fulfil these free-field conditions over the rated frequency range of the measurement, the manufacturer shall state the valid frequency range.

The amplitude response may be corrected in accordance with 19.5 if the free-field conditions are not fulfilled. The correction applied, if any, shall be stated.

9.3 Half-space, free-field conditions

Acoustical conditions in which the free field, according to 9.2, exists in a half-space may be used [6]. These conditions shall be satisfactorily met with a reflecting plane of sufficient size. Reflections of the radiated sound on other reflecting surfaces (e.g. walls) shall be sufficiently suppressed to ensure an accuracy of $\pm 0,5$ dB and $\pm 10^\circ$ in the measured sound pressure amplitude and phase values at the specified frequency. If the environment (for example, anechoic room at low frequencies) does not fulfil these free-field conditions over the entire rated frequency range of the measurement, the manufacturer shall state the valid frequency range.

9.4 Simulated free-field conditions

The acoustical conditions of any environment (e.g. non-anechoic room) shall be satisfactory if the direct sound radiated by the DUT is separated from the room reflections by using measurement techniques such as gating of the impulse response [7], wave separation [8], or acoustical holography [9]. These measurement techniques shall ensure an accuracy of $\pm 0,5$ dB and $\pm 10^\circ$ in the measured sound pressure amplitude and phase values at the specified frequency. The frequency resolution $\Delta f = 1/T$, where T is the effective or truncated length of the impulse response where the simulated free-field conditions are fulfilled, and any other limitations that impair the measurement results, shall be stated.

NOTE The gating technique truncates the impulse response, which reduces frequency resolution. Extending the length of the truncated impulse response by zero padding increases resolution virtually but provides no additional information.

9.5 Half-space simulated free-field conditions

Acoustical conditions, in accordance with 9.4, where the device under test is mounted in a reflecting plane of sufficient size.

9.6 Diffuse sound field conditions

Diffuse sound field conditions for measurements using pink noise and 1/3 octave band analysis, as defined and specified in ISO 3741, may be used. The lower limiting frequency shall be determined as specified in ISO 3741:2010, Appendix A.

9.7 Target application conditions

Acoustical conditions that correspond with the final target application of the device under test (for example, sound system mounted in a car).

10 Positioning of the DUT

10.1 Rated geometrical conditions

10.1.1 General

As illustrated in Figure 1, the position and the orientation of the DUT shall be stated using the reference plane with normal vector $\mathbf{n}_{\text{ref}}$, the reference point r_{ref} , the reference axis and the orientation vector $\mathbf{o}_{\text{ref}}$. In addition, the evaluation point and the evaluation distance shall be stated as illustrated in Figure 2.

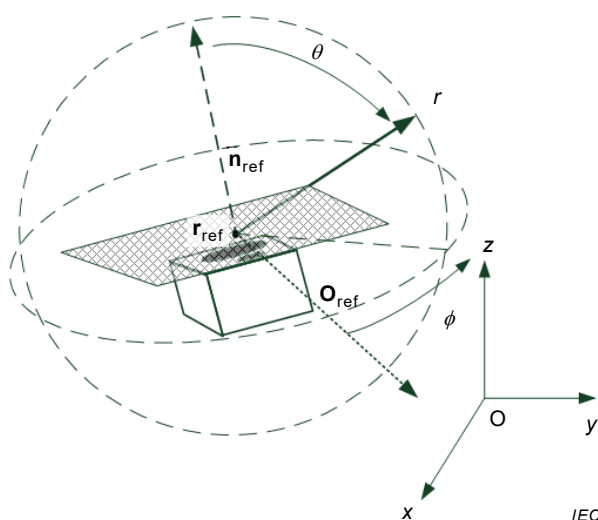


Figure 1 – Rated conditions used to describe the position of the DUT in the coordinate system

10.1.2 Reference plane and normal vector

The reference plane with the normal vector $\mathbf{n}_{\text{ref}}$ shall be used to define the reference axis and the reference point r_{ref} . The normal vector $\mathbf{n}_{\text{ref}}$ also defines the polar angle $\theta = 0$ in spherical coordinates.

NOTE For symmetrical structures, the reference plane is usually parallel to the radiating surface or to a plane defining the front of the DUT. For asymmetrical structures, the reference plane can be better indicated by means of a diagram.

10.1.3 Reference point

A point on the reference plane. The position of the reference point r_{ref} shall be specified by the manufacturer.

NOTE For symmetrical structures, reference point r_{ref} is usually the point of axial symmetry of the radiator within the reference plane; for asymmetrical structures, the reference point can be better indicated by means of a diagram.

For directional measurements performing a rotation of the DUT, the reference point is identical to the point of rotation (POR) and usually chosen close to or identical to the centre of gravity (COG) of the DUT.

10.1.4 Reference axis

The line that passes through the reference plane at the reference point; its direction shall be specified by the manufacturer. The reference axis shall be used as the zero reference axis for directional and frequency response measurements.

10.1.5 Orientation vector

The orientation vector $\mathbf{o}_{\text{ref}}$ defines the orientation of the sound system within the reference plane and the direction of azimuthal angle $\phi = 0$ in spherical coordinates. The preferred orientation vector places the reference point $\mathbf{r}_{\text{ref}}$ at the origin O of the coordinate system, pointing the normal vector $\mathbf{n}_{\text{ref}}$ of the reference plane into the z -direction where the polar angle $\theta = 0$ and turning the audio system in such a way that the orientation vector $\mathbf{o}_{\text{ref}}$ (e.g. top of the enclosure) points into the x -direction as illustrated in Figure 2. By doing this, the relationship between spherical and Cartesian coordinates can be expressed as

$$\mathbf{r} = x\vec{e}_x + y\vec{e}_y + z\vec{e}_z = r \cos\phi \sin\theta \vec{e}_x + r \sin\phi \sin\theta \vec{e}_y + r \cos\theta \vec{e}_z$$

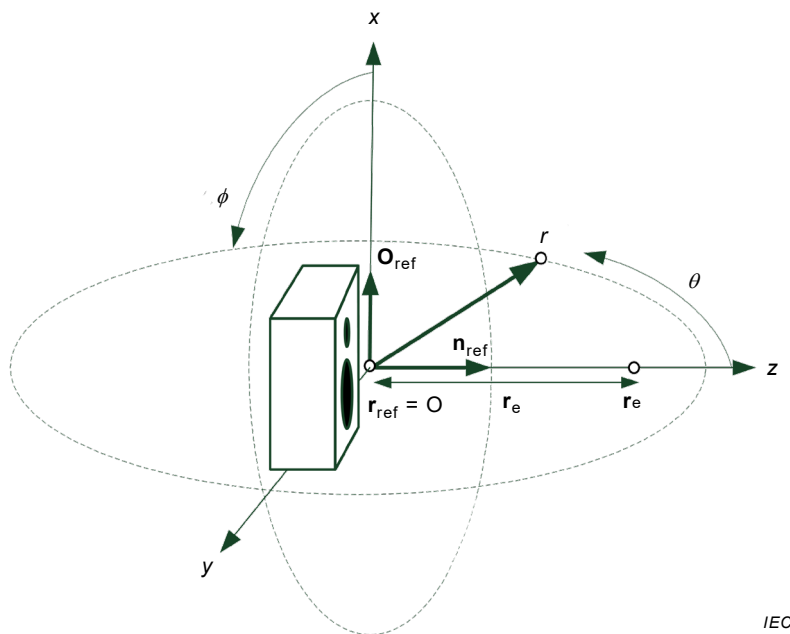


Figure 2 – Recommended position and orientation of the DUT

10.1.6 Evaluation point

The evaluation point $\mathbf{r}_e$ specified by the manufacturer is the position where the rated maximum sound pressure level SPL_{max} is determined. The evaluation point $\mathbf{r}_e$ shall be on the reference axis.

The evaluation point is a particular measurement point, which is required for the calibration of the maximum input value u_{max} and the maximum output level SPL_{max} in accordance with Clause 17 and Clause 18.

10.1.7 Evaluation distance

The evaluation distance $r_e = |\mathbf{r}_e - \mathbf{r}_{\text{ref}}|$ is the distance between the evaluation point $\mathbf{r}_e$ and the reference point $\mathbf{r}_{\text{ref}}$.

NOTE The evaluation distance r_e depends on the size of the DUT, typically 1 m but shorter for micro-speakers and larger for line arrays.

Ideally, the evaluation distance places the evaluation point in the far field of the DUT (see 10.2.1).

10.2 Measuring distance between DUT and microphone

10.2.1 Far-field conditions

Measurements under free-field and half-space conditions shall ideally be carried out in the far field of the DUT at a distance $r > r_{\text{far}}$ where the sound pressure decreases with the distance r according to the $1/r$ law. The location in the far field is determined when the following three conditions [10] are fulfilled:

- 1) the distance r is much larger than the geometrical dimension d of the DUT,
- 2) the distance r is much larger than the wavelength λ ,
- 3) the ratio between distance r and dimension d is much larger than the ratio between dimension d and wavelength λ ($r/d \gg d/\lambda$).

However, in practice, limitations of the measuring environment (e.g. room size [9]) and the effects of the background noise set an upper limit to the distance r that can be used.

Measurements under simulated free-field conditions shall be made with the microphone and DUT positioned within the measuring environment so that the time required for the unwanted first reflection to reach the microphone is maximized.

If the measurement space is an anechoic chamber, attention shall be paid to reflections from wedge tips, flooring, DUT supports and microphone supports.

Errors from these sources shall not exceed $\pm 0,5$ dB and $\pm 10^\circ$ in the measured sound pressure amplitude and phase at any frequency within the frequency range of measurement.

10.2.2 Near-field conditions

10.2.2.1 General

Except under special conditions as per 10.2.2.2, the sound pressure in the near field can only be measured by using holographic techniques [9]. In addition, valid far-field information can only be calculated from near-field data by using holographic methods.

The advantages of measurements in the near field [11] are as follows:

- maximize time available before first reflections;
- maximize the direct sound pressure.

The main advantage of holographic techniques is to allow large DUTs to be measured in small rooms [4].

10.2.2.2 Extrapolation technique for monopoles

The extrapolation technique developed by Keele [12] provides reliable far field information at low frequencies based on a single near-field measurement if the woofers are mounted in a sealed enclosure and the sound field is omnidirectional.

10.2.3 Diffuse field conditions

The position and orientation of the DUT and microphone with respect to the walls shall be described by means of a diagram appended to the measurement results.

An arrangement for the simultaneous movement of the DUT and the microphone is permitted for the evaluation of the power delivered by the loudspeaker in accordance with the method described in 20.4. The microphone system and nearest microphone position shall meet the requirements of ISO 3741.

10.2.4 Target application condition

The position and orientation of the DUT and microphone with respect to the target environment (e.g. car interior, artificial ear) shall be described by means of a diagram appended to the measurement results.

The acoustical properties of the boundaries shall be described by using area-averaged random-incidence energy absorption coefficients [13].

11 Measurement equipment and test results

Measurements in free-field conditions shall be made using a WS2F or WS3F free-field microphone as per IEC 61094-4. For measurements under diffuse-field conditions, a WS3P pressure microphone or a WS2P pressure microphone with diffuse field correction shall be used.

Graphical results shall conform to one of the preferred aspect ratios in IEC 60263.

Frequency data, if provided in addition to the graphs, shall be at the R40 preferred frequencies specified in ISO 3, unless otherwise stated.

12 Accuracy of the acoustical measurement

12.1 General

Probable error sources in both the instrumentation and measuring environment shall be identified, quantified and their contribution specified. Uncertainties in the position and calibration of the microphone shall be stated (see 12.2). This information shall be included with the test report.

NOTE Techniques for quantifying acoustic measurement tolerances are discussed in [14]. Working with two microphones located at different positions (or distances) can be used to check the possible errors in the measurement setup.

12.2 Measurement uncertainty

The measurement uncertainty is composed of several factors:

- uncertainty of the test equipment used, such as sound generator, level meters, measuring microphones, etc.;
- tolerances of the mechanical coupling or mounting;
- the sound field or test environment;
- positioning the DUT in the test space.

Given knowledge of the aforementioned factors, the measurement uncertainty can, in general, be determined [15]. For more information, see Annex A.

NOTE It is good practice to validate the measurement uncertainty by comparing measurement results with those from an accredited test laboratory. For the evaluation of the sound pressure output generated by loudspeakers, Annex A gives an example of an uncertainty budget calculation. For additional information, see ISO/IEC Guide 98-3 [15].

Theoretically, measurement error tolerances only apply to the performance of a device under test when the measurement equipment is 100 % accurate. To ensure that DUT performance is within a specified tolerance when using imperfect real world measurement equipment, the inside acceptance interval on the measured value shall be less than the tolerance specified by the maximum permitted uncertainty for that measurement. This may apply for outgoing inspection applications. To confirm that DUT performance is outside the specified tolerance under real-world conditions, the outside acceptance interval on the measured value shall be greater than the tolerance specified by the maximum permitted uncertainty for that measurement. This may apply for incoming QC/QA applications. The maximum permitted uncertainty is applied to each tolerance limit to obtain the acceptance interval (see Figure A.1).

13 Mounting of the DUT

13.1 Mounting and acoustic loading of drive units

The performance of the drive unit (transducer) is determined by the properties of the unit itself and its acoustic loading. The acoustic loading depends upon the mounting arrangement, which shall be clearly described in the presentation of the results. Annex B describes ways of mounting the transducer.

One of the following types of mounting shall be used:

- a) Half-space free-field condition generated by mounting the transducer flush with a plane reflecting surface having a diameter d ten times larger than the wavelength λ of the lowest working frequency or lowest frequency specified or rated by the manufacturer. Half-space loading for transducers is preferred.
- b) If this condition cannot be fulfilled for low-frequency transducers (e.g. woofer) in the available anechoic room, the manufacturer shall state the lowest frequency limit where this condition becomes valid. Alternatively, the baffle measurement or a room correction curve shall be used.
- c) A standard baffle with a plane front surface that is acoustically reflecting as specified in Clause B.1.
- d) A standard measuring enclosure specified in Clause B.2 (type-A or type-B) generating a defined rear air volume and radiation condition at the front of the transducer.
- e) A test cabinet not in accordance with Clause B.2 (type-A or type-B), generating a defined air load and radiation condition at the front of the transducer may be used for end-of-line testing and relative measurements (see [16]). An additional chamber at the rear side of the transducer can be used to provide additional noise isolation or to consider the influence of the air volume of a sealed box in the target application.
- f) A defined horn, coupler or other kind of wave guide that couples the radiating surface of the transducer to the surrounding sound field.

NOTE Application to headphones, headsets, earphones and earsets is specified in IEC 60268-7.

- g) A plane wave tube.
- h) The transducer may be operated in free air without baffle, enclosure, etc.

13.2 Mounting and acoustic loading of an electro-acoustic system

Electro-acoustic systems are usually measured without any additional baffle. If the manufacturer specifies a special type of mounting for the loudspeaker system, this shall be used for the measurement. The mounting method used shall be specified with the results.

14 Preconditioning

Temporary or permanent changes may take place when a signal generating high displacement is applied to the new DUT for the first time. Sufficient preconditioning shall be performed according to the intended use of DUT in the field (e.g. bandwidth).

NOTE For example, the suspension of a driver is subject to significant changes in its mechanical properties. Likewise, the effects of heating and cooling on the DUT.

The DUT shall be preconditioned before measurements are performed by applying a broadband signal at half the maximum input value $\tilde{u}_{\max}$ in accordance with Clause 17 for at least one hour (3 h recommended for woofers).

Before proceeding with the measurement, the period of preconditioning shall be followed by a recovery period of at least one hour while ensuring that the temperature of the internal parts is re-acclimatized to the ambient conditions.

15 Rated ambient conditions

15.1 Temperature ranges

15.1.1 Performance limited temperature range

The temperature range over which the variation of the characteristics of the DUT shall not exceed the specified tolerances.

15.1.2 Damage limited temperature range

The temperature range that, if exceeded during operation or storage, may result in permanent changes in the operating characteristics of the DUT.

15.2 Humidity ranges

15.2.1 Relative humidity range

The relative humidity range over which the variation of the characteristics of the DUT shall not exceed the specified tolerances.

15.2.2 Damage limited humidity range

The relative humidity range that, if exceeded during operation or storage, may result in permanent changes in the operating characteristics of the DUT.

16 Rated frequency range

The frequencies f_l and f_u describe the lower and upper limits of the audio band for which the maximum sound pressure of the DUT is rated.

NOTE The rated frequency range corresponds to the intended use of the DUT in the final application and can differ from the DUT's effective frequency range as defined in 19.6.

17 Input signal

17.1 Rated maximum input value

17.1.1 Condition to be specified

The maximum input value $\tilde{u}_{\max}$ is the RMS value

$$\tilde{u} = \sqrt{\frac{1}{T_s} \int_0^{T_s} u^2(t) dt}$$

of a broadband stimulus $u(t) = a[i]x(t)$ scaled by $a[i]$, which can be applied to the DUT for more than 100 h while keeping measured characteristics within defined tolerances.

The properties of the broadband stimulus $x(t)$, such as the spectral properties, the crest factor and the lower and upper limits f_l and f_u , respectively, of the rated frequency band in accordance with Clause 16, shall be stated by the manufacturer.

The maximum input value $\tilde{u}_{\max}$ can be stated by the manufacturer for transducers, passive loudspeaker systems and active systems having a single input and a fixed transfer function between the input and evaluation point r_e . The maximum input value $\tilde{u}_{\max}$ applied to the terminal voltage can be used for end-line testing of the passive transducer if the measuring conditions (e.g. geometry of the test box) are not defined or different. The maximum input value $\tilde{u}_{\max}$ can be used to evaluate the influence of various measurement conditions, such as acoustic loading, mounting of the driver, etc., on the transfer behaviour of the DUT. In active systems where the maximum input value $\tilde{u}_{\max}$ depends on the selected input channel, gain control, equalizer and other settings of the signal processing prior to the signal reaching the passive transducer the maximum input value can be determined based on the rated maximum sound pressure level $SPL_{\max}$ in accordance with 18.2.

17.1.2 Direct measurement

The following equipment or equivalent shall be included in the chain of measurement:

- a pink noise generator and or multi-tone generator in accordance with 8.5;

NOTE 1 A DUT intended for full-band audio signals can be tested using a broadband stimulus such as a multi-tone complex with sufficient resolution $R = 10$, a pink-noise signal or any noise like signal such as the simulated programme signal in accordance with Annex C. For a DUT with a small usable bandwidth in the target application (e.g. subwoofer), a sufficiently good approximation of the "broadband" signal can be a sinusoidal signal, a Hann burst signal or a narrow band noise in accordance with Clause 8.

- an optional weighting network to obtain the noise signal in accordance with Annex C;
- an optional clipping network to realize a defined crest factor;
- an optional power amplifier for passive electro-acoustical DUT.

The clipped noise at the terminals of the loudspeaker under test shall have a frequency distribution as specified in Annex C and a peak-to-RMS ratio (crest factor) between 1,8 and 2,2. The amplitude of the frequency response between the input of the optional power amplifier and the voltage at the input terminals of a passive DUT shall be constant to within $\pm 0,5$ dB in the frequency range 20 Hz to 20 kHz. The high-pass filter in the power amplifier may increase the crest factor of the transferred noise signal. The optional power amplifier shall have an output impedance not greater than 1/3 of the rated impedance of the passive DUT. The amplifier shall be capable of supplying the passive DUT with a peak sinusoidal voltage of at least twice the RMS voltage of the test noise without clipping.

- a) The manufacturer defines a test value $\tilde{u}_{\text{test}}$ as a candidate for the rated maximum input value $\tilde{u}_{\max}$. The chosen test value considers the intended application and behaviour of the DUT (e.g. permissible heating and maximum mechanical excursion of the voice coil) so that damage to the DUT during the following test can be avoided.

NOTE 2 Annex D discusses the considerations and measurements that are required to specify test value $\tilde{u}_{\text{test}}$.

- b) The DUT shall be placed in a room of sufficient size (typically $> 8 \text{ m}^3$) with climatic conditions as specified in 7.2. The size of the room depends on the space required around the DUT to avoid any additional acoustical loading of the DUT and to ensure stable climatic conditions during the test.

- c) The DUT shall be tested under a specified climatic condition for a continuous period of 100 h at the test value $\tilde{u}_{\text{test}}$ of the input stimulus, which is a broadband stimulus as stated by the manufacturer.

NOTE 3 Since the test value $\tilde{u}_{\text{test}}$ is constant during the 100 h test, the heating of the DUT will cause thermal compression of the acoustical output. This effect can be evaluated by measuring the compression in accordance with 19.4.

- d) Immediately after the test, the DUT shall be stored under climatic conditions that would normally exist in ordinary rooms or laboratories. Unless otherwise specified, the recovery period shall be 24 h.
- e) At the end of the recovery period, the acoustical characteristics of the DUT are measured. If the measured values are within the tolerances of the values stated in the data sheet, the DUT may be considered to have fulfilled the requirement of this test and the test value $\tilde{u}_{\text{test}}$ is assigned to the rated maximum input value $\tilde{u}_{\text{max}}$.

If the DUT has been damaged during this test, or the measured characteristics are outside the defined tolerances, the test shall be repeated using a new DUT at a reduced input test value $\tilde{u}_{\text{test}}$ until the rated maximum input value $\tilde{u}_{\text{max}}$ is determined.

17.1.3 Indirect measurement based on SPL_{max}

- a) To determine the maximum input value $\tilde{u}_{\text{max}}$ based on rated maximum sound-pressure level SPL_{max} , the DUT is placed under normal measurement conditions in accordance with 7.3.
- b) The DUT is excited by a scaled stimulus $u(t) = \alpha[i]x(t)$ as stated by the manufacturer in accordance with 17.1. In the first step of the measurement $I = 0$, a sufficiently low scaling factor $\alpha[0]$ is used to prevent damage to the device under test.
- c) The sound pressure signal $p(t, r_e)$ is measured at the evaluation point r_e and the resulting sound pressure level $SPL[i]$ for the stimulus with RMS value $\tilde{u}[i]$ is calculated for the excitation time $T_s = 1$ s.
- d) After waiting for at least 10 s the scaling factor is adjusted by

$$\alpha[i+1] = \alpha[i] 10^{(SPL_{\text{max}} - SPL[i])/20}$$

and the measurement steps 17.1.2 b) to 17.1.2 d) are repeated until $SPL[i]$ deviates from SPL_{max} by less than 0,1 dB.

The final RMS value $\tilde{u}[i]$ is assigned to the maximum input value $\tilde{u}_{\text{max}}$.

NOTE In cases where a filter having a bandwidth equal to the rated frequency range is not available, an approximation can be made by dividing this frequency range into n sets of 1/3 octave bands in accordance with IEC 61260-1. The 1/3 octave filters are fed with the broadband stimulus. As a result, the voltage fed to the DUT in each 1/3 octave frequency band is equal to $\tilde{u}/\sqrt{n}$. The sound-pressure level is given by the formula:

$$SPL[i] = 10 \log_{10} \left(\frac{1}{p_0^2} \sum_{k=1}^n \tilde{p}_k^2 \right)$$

where $\tilde{p}_k$ is the RMS sound pressure value in a given 1/3 octave band and p_0 is the standard reference sound pressure (20 μPa).

17.2 Maximum input level

Expressed in decibels, the maximum input level shall be specified as twenty times the logarithm of the ratio between the maximum input value $\tilde{u}_{\text{max}}$ and a stated reference value.

The RMS value 1 V shall be used as a reference value if the maximum input is expressed in dB(V). The digital full-scale value is used as a reference value if maximum input level is expressed in dB(FS).

18 Sound-pressure output

18.1 Rated maximum sound pressure

18.1.1 Conditions to be specified

The manufacturer specifies the rated maximum sound pressure $\tilde{p}_{\max}(\mathbf{r}_e)$ defined as the RMS value

$$\tilde{p}_{\max}(\mathbf{r}_e) = \left(\frac{1}{T_s} \int_0^{T_s} p^2(t, \mathbf{r}_e) dt \right)^{1/2}$$

of the measured sound pressure signal $p(t, \mathbf{r}_e)$ at the specified evaluation point $\mathbf{r}_e$ under normal measurement conditions, in accordance with 7.3, using a specified broadband stimulus limited by the frequencies f_l and f_u of the rated frequency band within the excitation time $T_s = 1$ s.

The properties of the broadband stimulus $x(t)$, such as the spectral properties, the crest factor and the lower and upper limits of the rated frequency band, shall be stated by the manufacturer.

NOTE The rated maximum sound pressure $\tilde{p}_{\max}(\mathbf{r}_e)$ can be determined either by a direct measurement or indirectly derived from the rated maximum input value $\tilde{u}_{\max}$ in accordance with 17.1.

18.1.2 Direct measurement

The following equipment or equivalent shall be included in the chain of measurement:

- a pink-noise generator and or multi-tone generator in accordance with 8.5;

NOTE 1 A DUT intended for full-band audio signals can be tested using a broad-band stimulus such as a multi-tone complex with sufficient resolution $R = 10$, a pink noise signal or any noise like signal such as the simulated programme signal in accordance with Annex C. For a DUT with a small usable bandwidth in the target application (e.g. subwoofer), a sufficiently good approximation of the "broadband" signal can be a sinusoidal signal, a Hann burst signal or a narrow band noise in accordance with Clause 8.

- an optional weighting network to obtain the noise signal in accordance with Annex C;
- an optional clipping network to realize a defined crest factor;
- an optional power amplifier for passive electro-acoustical DUT.

The amplitude of the frequency response between the input of the optional power amplifier and the voltage at the input terminals of a passive DUT shall be constant to within $\pm 0,5$ dB in the frequency range 20 Hz to 20 kHz. The clipped noise at the terminals of the loudspeaker under test shall have a frequency distribution as specified in Annex C and a peak-to-RMS ratio between 1,8 and 2,2. The optional power amplifier shall have an output impedance not greater than 1/3 of the rated impedance of the passive DUT. The amplifier shall be capable of supplying the passive DUT with a peak sinusoidal voltage of at least twice the RMS voltage of the test noise without clipping.

- a) The manufacturer defines a test value $\tilde{p}_{\text{test}}(\mathbf{r}_e)$ as a candidate for the rated maximum sound pressure $\tilde{p}_{\max}(\mathbf{r}_e)$, in the rated frequency band. The chosen test value considers the intended application and behaviour of the DUT (e.g. permissible heating and maximum mechanical excursion of the voice coil) so that damage to the DUT during the following test can be avoided.

NOTE 2 Annex D discusses the considerations and measurements which are required to specify test value $\tilde{p}_{\text{test}}(\mathbf{r}_e)$.

- b) The DUT shall be placed in an acoustical environment, in accordance with Clause 9, and the test value $\tilde{p}_{\text{test}}(\mathbf{r}_e)$ shall be generated at the specified evaluation point $\mathbf{r}_e$ by adjusting the amplitude of the stimulus. This measurement shall use the broadband stimulus as stated by the manufacturer's test value in 18.1.1 and an excitation time of $T_s = 1$ s with a sufficient break of at least 10 s between successive measurements to avoid heating of the DUT and thermal compression of the output amplitude. The RMS value of the input stimulus generating the test value $\tilde{p}_{\text{test}}(\mathbf{r}_e)$ is used as a test value $\tilde{u}_{\text{test}}$ for further testing.
- c) The DUT shall be placed in a room of sufficient size (typically $> 8 \text{ m}^3$) with climatic conditions as specified in 7.2. The size of the room depends on the space required around the DUT to avoid any additional acoustical loading of the DUT and to ensure stable climatic conditions during the test.
- d) The DUT shall be tested under a specified climatic condition for a continuous period of 100 h at the test value $\tilde{u}_{\text{test}}$ of the input stimulus.

NOTE 3 Since the test value $\tilde{u}_{\text{test}}$ is constant during the 100 h test, the heating of the DUT causes thermal compression of the acoustical output and the sound pressure level falls below the rated maximum sound pressure $\tilde{p}_{\text{test}}(\mathbf{r}_e)$. This effect can be evaluated by measuring the compression in accordance with 19.4.

- e) Immediately after the test, the DUT shall be stored under climatic conditions that would normally exist in ordinary rooms or laboratories. Unless otherwise specified, the recovery period shall be 24 h.
- f) At the end of the recovery period, the electrical, mechanical and acoustical characteristics of the DUT are measured. If the measured values are within the tolerances of the values stated in the data sheet, the DUT may be considered to have fulfilled the requirement of this test and the test value $\tilde{p}_{\text{test}}(\mathbf{r}_e)$ is assigned to the rated maximum sound pressure $\tilde{p}_{\text{max}}(\mathbf{r}_e)$. The permissible shift of the resonant frequency shall be defined in the data sheet.

If the DUT has been damaged during this test or if the measured characteristics are outside the defined tolerances, the test shall be repeated using a new DUT at a reduced test value $\tilde{p}_{\text{test}}(\mathbf{r}_e)$ until the rated maximum sound pressure $\tilde{p}_{\text{max}}(\mathbf{r}_e)$ is determined.

18.1.3 Indirect measurement based on maximum input value

- a) The DUT shall be placed in an acoustical environment, in accordance with Clause 9, and the RMS value of the input stimulus is adjusted to the rated maximum input value $\tilde{u}_{\text{max}}$ of the stimulus in accordance with 17.1.
- b) The RMS value of the sound pressure $\tilde{p}(\mathbf{r}_e)$ shall be measured at the specified evaluation point $\mathbf{r}_e$ and assigned to the rated maximum sound pressure $\tilde{p}_{\text{max}}(\mathbf{r}_e)$.

18.2 Rated maximum sound-pressure level

Expressed in decibels, the rated maximum sound pressure level SPL_{max} in a rated frequency band shall be specified as twenty times the logarithm of the ratio between the RMS sound pressure value $\tilde{p}_{\text{max}}(\mathbf{r}_e)$, measured in accordance with 18.1, and the standard reference sound pressure (20 μPa).

18.3 Short term maximum sound pressure level

18.3.1 Conditions to be specified

The short term maximum sound pressure level SPL_{short} is defined as the level

$$SPL_{\text{short}}(\mathbf{r}_e) = 10 \log_{10} \left(\frac{1}{T_s p_0^2} \int_0^{T_s} p^2(t, \mathbf{r}_e) dt \right)$$

of the RMS sound pressure signal $p(t)$ at the specified evaluation point $\mathbf{r}_e$ referred to the reference sound pressure ($p_0 = 20 \mu\text{Pa}$) using a sequence of gated broadband stimulus comprised of an excitation period of $T_s = 1 \text{ s}$ followed by a cooling interval of $T_{\text{off}} = 1 \text{ min}$ repeated 60 times. A multi-tone complex signal with sufficient resolution $R = 10$ or a pink-noise signal or any noise-like signal, such as the simulated programme signal in accordance with Annex C, may be used as the stimulus. The properties of the broadband stimulus $x(t)$, such as the spectral properties, the crest factor and the lower and upper limits of the rated frequency band, shall be stated.

NOTE The short term maximum sound pressure level SPL_{short} can be larger than the rated value SPL_{max} because the short excitation period $T_s = 1 \text{ s}$ and the cooling interval ($T_{\text{off}} = 1 \text{ min}$) keeps thermal heating and power compression negligible.

18.3.2 Method of measurement

- The manufacturer defines a test value $SPL_{\text{test}}(\mathbf{r}_e)$ as a candidate for the short-term maximum sound pressure level $SPL_{\text{short}}(\mathbf{r}_e)$ in the rated frequency band. The test value $SPL_{\text{test}}(\mathbf{r}_e)$ considers the intended application and the behaviour of the DUT (e.g. permissible heating and maximum mechanical excursion of the voice coil) so that damage to the DUT during the following test can be avoided.
- The DUT shall be placed in an acoustical environment, in accordance with Clause 9. While measuring the sound pressure at the specified evaluation point $\mathbf{r}_e$, the amplitude of the stimulus is adjusted to obtain a sound pressure equal to the test value specified by the manufacturer in point a) by using an excitation period of $T_s = 1 \text{ s}$ followed by a cooling interval of $T_{\text{off}} = 1 \text{ min}$.
- After the adjustment, the sequence of the defined excitation period and cooling interval shall be repeated 60 times.
- After the test, no signal is supplied to the DUT for a recovery period of 5 min.
- At the end of the recovery period, the electrical, mechanical and acoustical characteristics of the DUT are measured. If the measured values are within the tolerances of the stated values in the data sheet, the DUT may be considered to have fulfilled the requirement of this test, and the test value $SPL_{\text{test}}(\mathbf{r}_e)$ is assigned to the rated sound pressure $SPL_{\text{short}}(\mathbf{r}_e)$. A change in the resonance frequency may occur but the acceptability of this change is subject to negotiation, therefore, it shall be stated.

If the DUT has been damaged during this test or the measured characteristics are outside the defined tolerances, the test shall be repeated using a new DUT at a reduced test value $SPL_{\text{test}}(\mathbf{r}_e)$ until the short term maximum sound pressure level $SPL_{\text{short}}(\mathbf{r}_e)$ is determined.

18.4 Long term maximum sound pressure level

18.4.1 Conditions to be specified

The long term maximum sound pressure level SPL_{long} is the level

$$SPL_{\text{long}}(\mathbf{r}_e) = 10 \log_{10} \left(\frac{1}{T_s p_0^2} \int_0^{T_s} p^2(t, \mathbf{r}_e) dt \right)$$

of the RMS value of the sound pressure signal $p(t)$ at the specified evaluation point $\mathbf{r}_e$ referred to the reference sound pressure ($p_0 = 20 \mu\text{Pa}$) using a broadband stimulus with an excitation period of $T_s = 1 \text{ min}$ and a cooling interval of $T_{\text{off}} = 2 \text{ min}$.

A multi-tone complex signal with sufficient resolution $R = 10$ or a pink-noise signal or any noise like signal, such as the simulated programme signal in accordance with Annex C, may be used as the stimulus. The properties of the broadband stimulus $x(t)$, such as the spectral properties, the crest factor, the durations of T_s and T_{off} and the lower and upper limits of the rated frequency band, shall be stated by the manufacturer.

NOTE The long term maximum sound pressure level SPL_{long} could be larger than the rated value SPL_{max} because the cooling interval $T_{off} = 2$ min reduces the thermal heating and power compression found in the 100 h test in accordance with 18.1.2.

18.4.2 Method of measurement

- a) The manufacturer defines a test value $SPL_{test}(r_e)$ as a candidate for the long term maximum sound pressure level $SPL_{long}(r_e)$ in the rated frequency band. The test value $SPL_{test}(r_e)$ considers the intended application and the particularities of the DUT (e.g. permissible heating and maximum mechanical excursion of the voice coil) so that damage to the DUT during the following test can be avoided.
- b) The DUT shall be placed in an acoustical environment, in accordance with Clause 9. While measuring the sound pressure at the specified evaluation point r_e , the amplitude of the stimulus is adjusted to obtain a sound pressure equal to the test value specified by the manufacturer in 18.4.2 a) by using a excitation period of $T_s = 1$ min followed by a cooling interval of $T_{off} = 2$ min.
- c) After the adjustment, the sequence of the defined excitation period and cooling interval shall be repeated 10 times.
- d) After the test, no signal is supplied to the DUT for a recovery period of 1 h.
- e) At the end of the recovery period, the electrical, mechanical and acoustical characteristics of the DUT are measured. If the measured values are within the tolerances of values stated in the data sheet, the DUT may be considered to have fulfilled the requirement of this test and the test value $SPL_{test}(r_e)$ is assigned to the long-term sound pressure level $SPL_{long}(r_e)$. A change in the resonance frequency may occur but the acceptability of this change is subject to negotiation, therefore, it shall be stated.

If the DUT has been damaged during this test or the measured characteristics are outside the defined tolerances, the test shall be repeated using a new DUT at a reduced test value $SPL_{test}(r_e)$ until the long-term maximum sound pressure level $SPL_{long}(r_e)$ is determined.

18.5 Sound pressure in a stated frequency band

18.5.1 Condition to be specified

The sound pressure $\tilde{p}(r)$ produced by a DUT at a stated measurement point r excited with a band-limited pink-noise signal with a stated RMS value $\tilde{u} = \alpha \tilde{u}_{max}$.

18.5.2 Method of measurement

- a) The loudspeaker shall be operated under normal measuring conditions.
If the maximum input value $\tilde{u}_{max}$ is not stated by the manufacturer, this value can be determined by using the rated maximum sound pressure level SPL_{max} in accordance with 18.2.
The bandwidth of a pink-noise signal is limited to the stated frequency limits by using a band-pass filter having slopes of at least 24 dB/octave. The RMS value of the stimulus is adjusted to the stated value $\tilde{u} = \alpha \tilde{u}_{max}$ and applied to the input of the DUT.
- b) The sound-pressure $\tilde{p}(r)$ shall be measured at the stated measurement point r .
- c) Any deviations from the conditions stated in 18.5.2 a) and b) shall be stated with the results.

18.6 Sound-pressure level in a stated frequency band

Expressed in decibels, the sound pressure level in a stated frequency band shall be specified as twenty times the logarithm of the ratio between the sound pressure $\tilde{p}(\mathbf{r})$ measured in accordance with 18.5 and the standard reference sound pressure (20 µPa).

18.7 Mean sound-pressure in a stated frequency range

18.7.1 Condition to be specified

The square root of the arithmetic mean of the squares of the sound pressure $\tilde{p}_k$ of all $1/n^{\text{th}}$ octave sub-bands with $k = 1, \dots, K$ and $n \geq 3$ filling the stated frequency range.

$1/3^{\text{rd}}$ octave sub-bands with $n = 3$ as defined by IEC 61260-1 shall be used if the upper and lower limits of the stated frequency band correspond to the limits of the $1/3^{\text{rd}}$ octave bands.

18.7.2 Method of measurement

- The measurement shall be made in accordance with 18.5.
- The sound pressure signal $\tilde{p}(\mathbf{r})$ measured at the measurement point $\mathbf{r}$ is analyzed by using a set of filters with $1/n^{\text{th}}$ octave bandwidth generating the RMS value of the sound pressure $\tilde{p}_k$ of each sub-band with $k = 1, \dots, K$.
- The mean sound-pressure $\tilde{p}_m$ in the stated frequency range is determined by the formula:

$$\tilde{p}_m = \left[\frac{1}{K} \sum_{k=1}^K (\tilde{p}_k)^2 \right]^{1/2}$$

18.8 Mean sound-pressure level in a stated frequency range

Expressed in decibels, twenty times the logarithm of the ratio between $\tilde{p}_m$ in accordance with 18.7 and the standard reference sound-pressure (20 µPa).

19 Frequency response of the fundamental component

19.1 Transfer function

19.1.1 Conditions to be specified

The transfer function $\underline{H}(f, \mathbf{r})$ between the input signal $u(t)$ and the sound pressure output $p(t, \mathbf{r})$ at the measurement point $\mathbf{r}$ measured using a broadband stimulus $\tilde{u}(t) = \alpha \tilde{u}_{\text{max}} x(t)$ scaled by the maximum input value $\tilde{u}_{\text{max}}$ and scaling factor α . The properties of the stimulus $u(t)$, the measurement time T_s and either the RMS value $\tilde{u} = \alpha \tilde{u}_{\text{max}}$ or the scaling factor α shall be stated.

NOTE The transfer function measured with a high amplitude stimulus can be affected by nonlinear and thermal properties of the DUT.

The linear transfer response shall be measured in the small signal domain by using an attenuated stimulus (scaling factor $\alpha \leq 0,1$) or by performing response measurements at reduced levels until the observed response is unchanged, verifying linear conditions.

19.1.2 Method of measurements

- To assess the nonlinear behaviour of the DUT, the electro-acoustical system is excited by the stimulus $\tilde{u}(t) = \alpha \tilde{u}_{\text{max}} x(t)$ by using the maximum input value $\tilde{u}_{\text{max}}$ and a stated scaling

factor α . It is recommended to use a sinusoidal chirp signal or multi-tone complex signal in accordance with 8.2 and 8.5.

- b) The sound pressure signal $p(t, \mathbf{r})$ is measured at stated measurement point $\mathbf{r}$ and the complex amplitude of the fundamental component $\underline{p}_{\text{fund}}(f, \mathbf{r}) = F\{p(t, \mathbf{r})\}$ with respect to instantaneous frequency f is calculated by using a time-frequency transformation such as the Fourier transform $F\{\}$.
- c) The complex amplitudes of the fundamental component $\underline{u}(f)$ with respect to instantaneous frequency f are determined from stimulus $u(t)$ by using the frequency-time mapping of the chirp $x(t)$.

NOTE 1 The logarithmic frequency-time mapping of the chirp signal can be used to separate the fundamental and harmonic components, see [17].

- d) The complex transfer function is calculated by

$$\underline{H}(f, \mathbf{r}) = \frac{\underline{p}_{\text{fund}}(f, \mathbf{r})}{\underline{u}(f)}$$

providing the magnitude response $|\underline{H}(f, \mathbf{r})|$ and the phase response $\varphi(f, \mathbf{r}) = \arg(\underline{H}(f, \mathbf{r}))$.

- e) The impulse response $h(t, \mathbf{r}) = F^{-1}\{\underline{H}(f, \mathbf{r})\}$ may be calculated by using a frequency-time transformation such as the inverse Fourier transform $F^{-1}\{\}$.

NOTE 2 Time windowing can be applied to the impulse response to suppress reflections from acoustical boundaries (e.g. room walls) and to generate simulated free-field conditions in accordance with 9.4. A corrected complex transfer can be generated by applying a Fourier transform to the windowed impulse response.

- f) The mean group delay $\tau_{\text{mean}}(\mathbf{r})$ may be determined by searching for the maximum of the energy-time curve in the impulse response $h(t, \mathbf{r})$.

NOTE 3 The energy-time curve is the envelope of the impulse response, which can be calculated by using the Hilbert transform. A useful approximation is the log-squared impulse response.

- g) The residual phase

$$\varphi_{\text{res}}(f, \mathbf{r}) = \varphi(f, \mathbf{r}) - 360^\circ f \tau_{\text{mean}}(\mathbf{r})$$

represents the minimum-phase + all pass behaviour of the DUT separated from the phase corresponding to the mean group delay [18].

- h) The unwrapped total phase response

$$\varphi_{\text{un}}(f, \mathbf{r}) = \varphi_{\text{res}}(f, \mathbf{r}) + 360^\circ f \tau_{\text{mean}}(\mathbf{r})$$

is calculated by unwrapping the residual phase response and adding the phase corresponding with the mean group delay.

NOTE 4 Unwrapping is a necessary but ambiguous, noise-sensitive and error-prone process for frequency-discrete phase data (at least in acoustics).

The frequency resolution shall be high enough so that the phase difference between two discrete frequencies does not exceed $\pm 90^\circ$.

- i) The group delay response

$$\tau(f) = -\frac{1}{360^\circ} \frac{d\varphi_{\text{un}}(f)}{df}$$

may be calculated by differentiating the unwrapped total phase response $\varphi_{\text{un}}(f, \mathbf{r})$.

- j) Smoothed curves of the magnitude and phase response are generated by applying spectral averaging to the frequency response $\underline{H}(f, \mathbf{r})$ by stating the smoothing technique applied (complex averaging or separate averaging of the level and phase response) and the used smoothing bandwidth B , which is typically between 1/3rd octave and 1/12th octave.

NOTE 5 Spectral averaging generates smoothed curves, which gives the following benefits:

- improved visual separation of multiple curves plotted on the same axis, e.g., directional off-axis responses;
- conversion of linear averaging to equivalent exponential averaging;
- as a surrogate for averaging the results from multiple devices, e.g. a 'virtual golden sample' for generating tolerance limits;
- simplified interpretation of peaks and dips with respect to spectral audibility;
- reduced errors caused by poor SNR and room reflections in the original measurement.

The magnitude response of the transfer function may be presented as twenty times the logarithm of the ratio between the absolute value of the complex transfer function $|\underline{H}(f, \mathbf{r})|$ and c_{ref} which has a stated reference value (e.g. 1 Pa/V).

19.2 SPL frequency response

19.2.1 Conditions to be specified

The sound pressure level $SPL(f, \mathbf{r})$ as a function of frequency, measured under normal conditions at the measurement point $\mathbf{r}$ using a narrow-band signal with bandwidth B and centre frequency f . The input signal $\tilde{u}(t) = \alpha \tilde{u}_{\text{max}} x(t)$ has a constant RMS value $\tilde{u} = \alpha \tilde{u}_{\text{max}}$ for all frequencies f varied in the rated frequency range. The properties of the stimulus $u(t)$, the measurement time T_s , and either the RMS value $\tilde{u} = \alpha \tilde{u}_{\text{max}}$ or the scaling factor α , shall be stated.

NOTE The SPL frequency response measured at high amplitudes depends on the interaction between the stimulus and the linear, nonlinear and thermal properties of the DUT.

The linear properties shall be measured in the small signal domain by using a small scaling factor ($\alpha \leq 0,1$) or by performing response measurements at reduced levels until the observed response is unchanged, verifying linear conditions.

The graphical representation of the SPL frequency response shall correspond to IEC 60263 to ensure that the results are not stretched or compressed.

19.2.2 Method of measurement

- a) The loudspeaker shall be operated under normal measuring conditions.
- b) If the maximum input value $\tilde{u}_{\text{max}}$ is not stated by the manufacturer, this value can be determined by using the rated maximum sound pressure level SPL_{max} in accordance with 18.6.
- c) For each frequency f in the rated frequency range $f_l \leq f \leq f_u$, the $SPL(f)$ may be determined either by:
 - 1) Exciting the DUT with a steady-state tone or with a narrow band-filtered noise signal with stated bandwidth B having an RMS value $\alpha \tilde{u}_{\text{max}}$ within the measurement time T_s . The sound pressure $p(t, \mathbf{r})$ is measured at the measurement point $\mathbf{r}$ and filtered by band pass filter of defined bandwidth B and centre frequency f to reduce noise and nonlinear distortion generated by the DUT before the sound pressure level is calculated.
 - 2) Measuring the transfer function $\underline{H}(f, \mathbf{r})$ by using a broad-band stimulus within the measurement time T_s in accordance with 19.1. The $SPL(f)$ in dB is calculated by

$$SPL(f, \mathbf{r}) = 20 \log \left(\frac{|H(f, \mathbf{r})| \alpha \tilde{u}_{\max}}{p_0} \right)$$

d) Using the reference sound pressure ($p_0 = 20 \mu\text{Pa}$).

- 1) The measurements are performed at a sufficient spectral resolution over the rated frequency band. Additional smoothing of the frequency response can be applied if the bandwidth B of the spectral averaging is stated by the manufacturer. The results shall be presented in a graph as a function of frequency and the measurement condition shall be stated.

19.3 Time-varying amplitude compression of the fundamental component

19.3.1 General

The amplitude compression $C(f, t_m)$ is the level difference between the magnitude frequency response of a time-varying transfer function $\underline{H}(f, \mathbf{r}, t_m)$ measured in interval $t_m < t < t_m + T_s$ and the linear transfer function $\underline{H}_{\text{lin}}(f, \mathbf{r}, \alpha u_{\max})$ measured using a broad-band stimulus at an attenuated maximum input value $\alpha u_{\max}$.

NOTE The amplitude compression $C(f, t_m)$ is a time-varying characteristic (i.e. reduction or increase in the fundamental component), which depends on the voice coil temperature and nonlinearities inherent in the device under test. The amplitude compression corresponds to the negative value of the normalized transfer function as defined in CTA 2010-B.

19.3.2 Method of measurement

- a) The DUT is adjusted to ambient temperature.
- b) The linear transfer function $\underline{H}_{\text{lin}}(f, \mathbf{r}, \alpha u_{\max})$ is measured by using a broad-band stimulus with an rms value $\alpha u_{\max}$ within the measurement time $T_s = 1 \text{ s}$ in accordance with 19.1.
- c) The transfer function $\underline{H}(f, \mathbf{r}, t_m)$ is measured by using a broad-band stimulus within the measurement interval $t_m < t < t_m + T_s$ in accordance with 19.1.
- d) The response of the time varying amplitude compression shall be calculated using:

$$C(f, t_m) = 20 \log \left(\left| \underline{H}_{\text{lin}}(f, \mathbf{r}, \alpha u_{\max}) \right| \right) - 20 \log \left(\left| \underline{H}(f, \mathbf{r}, t_m) \right| \right)$$

19.4 Amplitude compression at maximum input

19.4.1 Short term amplitude compression

The short-term amplitude compression $C_{\text{short}}(f)$ is the level difference between the magnitude frequency response of the transfer function $\underline{H}(f, \mathbf{r}, u_{\max})$ measured by using a broad-band stimulus at maximum input value $\tilde{u}_{\max}$ within the measurement time $T_s = 1 \text{ s}$ and the linear transfer function $\underline{H}_{\text{lin}}(f, \mathbf{r}, \alpha u_{\max})$ measured using the same stimulus at an attenuated maximum input value $\alpha u_{\max}$.

NOTE The short-term amplitude compression $C_{\text{short}}(f)$ reveals the nonlinear mechanisms of the transducer, the effect of the protection system and the limiting effects from other electronics (e.g. amplifier).

19.4.2 Method of measurement

- a) The DUT is adjusted to ambient temperature.
- b) The linear transfer function $\underline{H}_{\text{lin}}(f, \mathbf{r}, \alpha u_{\max})$ is measured by using a specified broad-band stimulus with an RMS value $\alpha u_{\max}$ within the measurement time $T_s = 1 \text{ s}$ in accordance with 19.1.

- c) The transfer function $\underline{H}(f, r, \alpha u_{\max})$ is measured by using a specified broad-band stimulus with the maximum input value $u_{\max}$ within the measurement time $T_s = 1$ s in accordance with 19.1.
- d) The response of the long-term amplitude compression shall be calculated using:

$$C_{\text{short}}(f) = 20 \log(|\underline{H}_{\text{lin}}(f, r, \alpha u_{\max})|) - 20 \log(|\underline{H}(f, r, u_{\max})|)$$

19.4.3 Long-term amplitude compression

The long-term amplitude compression $C_{\text{long}}(f)$ is the level difference between the magnitude frequency response of the transfer function $\underline{H}(f, r, \tilde{u}_{\max})$ measured by using a broad-band stimulus at maximum input value $\tilde{u}_{\max}$ within the measurement time $T_s = 1$ s after applying a pre-excitation of the DUT with the same stimulus for $T_{\text{pre}} = 1$ min and the linear transfer function $\underline{H}_{\text{lin}}(f, r, \alpha \tilde{u}_{\max})$ measured with the same stimulus at attenuated maximum input value $\alpha \tilde{u}_{\max}$.

NOTE The long-term amplitude compression $C_{\text{long}}(f)$ reveals the thermal and nonlinear mechanisms of the transducer and the effect of the active protection system limiting the maximum output of the device under test.

19.4.4 Method of measurement

- a) The DUT is adjusted to ambient temperature.
- b) The linear transfer function $\underline{H}_{\text{lin}}(f, r, u_{\max})$ is measured by using a specified broad-band stimulus with an RMS value $\alpha u_{\max}$ within the measurement time $T_s = 1$ s in accordance with 19.1.
- c) After exciting the DUT by a broad-band stimulus $u(t)$ with maximum input value $\tilde{u}_{\max}$ for a pre-excitation time $T_{\text{pre}} = 1$ min, the transfer function $\underline{H}(f, r, \tilde{u}_{\max})$ is measured by using the same stimulus for the measurement time $T_s = 1$ s in accordance with 19.1.
- d) The response of the long-term amplitude compression shall be calculated by:

$$C_{\text{long}}(f) = 20 \log(|\underline{H}_{\text{lin}}(f, r, \alpha \tilde{u}_{\max})|) - 20 \log(|\underline{H}(f, r, \tilde{u}_{\max})|)$$

19.5 Corrections based on a free-field reference measurement

19.5.1 General

Whenever the low-frequency absorption characteristic of the acoustical test environment (room walls, equipment, etc.) causes a deviation from free-field conditions, such that an accurate measurement of the free-field response under far field conditions is not possible in the rated frequency range in accordance with Clause 16, the low-frequency measurement may be corrected by either filtering the measured sound pressure signal with the complex room correction function or by correcting the resulting amplitude frequency response by a real-value based room correction function.

A reference loudspeaker shall be selected having a directivity index DI , in accordance with 20.3.4, which deviates from the directivity index DI of the device under test (DUT) by not more than 0,5 dB over the frequency range where correction is required and its calibrated free-field response shall extend to the lowest frequency of interest.

19.5.2 Correction of the measured sound pressure signal

This method filters the measured sound pressure signal with the room correction function, which compensates for any phase and amplitude distortion before the signal is analyzed. This is a requirement for accurate measurement of nonlinear distortion and to apply windowing of the impulse response to the device under test.

- a) The complex transfer response $\underline{H}_{\text{free}}(f, \mathbf{r})$ of the reference loudspeaker operated under free-field conditions shall be measured at a defined measurement point $\mathbf{r}$ (e.g. distance r on the reference axis).

For a reference loudspeaker dedicated to reproduce only low frequencies, such as a subwoofer, a half-space anechoic condition shall be used provided a ground floor measurement is applied to the DUT in the non-anechoic test environment.

For a reference loudspeaker with limited low-frequency response (main resonance above 150 Hz), measurements in a very large anechoic room (8 m × 10 m × 12 m, for example) can be sufficiently accurate.

For reference loudspeakers with extended low-frequency response, measurements on a tower (typically 10 m or more above ground level) in the open air can become necessary.

Windowing may be applied to the impulse response to suppress noise, provided there is no change in the time delay nor a reduction in the spectral resolution required to capture the properties of the reference loudspeaker.

- b) The reference loudspeaker shall be moved to the position in the room or non-anechoic test environment located such that its reference point $\mathbf{r}_{\text{ref}}$ and reference axis take the positions occupied by the device under test.

The radiated sound pressure generated by the reference loudspeaker operated under test conditions shall be determined at the same measurement point (e.g. distance r on the reference axis) as used in step a).

The complex transfer response $\underline{H}_{\text{test}}(f, \mathbf{r})$ shall be determined in accordance with 19.1. Windowing may be applied to the impulse response to suppress noise provided there is no change in the time delay nor a reduction in the spectral resolution required to capture the properties of the reference loudspeaker and the interaction with the boundaries (e.g. room walls).

NOTE For measurement of the low-frequency response of a multi-unit loudspeaker system, the reference point is ideally the reference point of the bass unit.

- c) The complex room correction function $\underline{C}_{\text{room}}(f) = \underline{H}_{\text{free}}(f) / \underline{H}_{\text{test}}(f)$ shall be calculated as the ratio between the transfer function measured under free-field and test conditions.
- d) The reference loudspeaker shall be replaced by the DUT and the sound pressure signal shall be measured using the same equipment. The sound pressure signal $p(t)$ shall be filtered with the complex room correction function $\underline{C}_{\text{room}}(f)$ to generate the corrected sound pressure signal $p'(t)$ with the spectrum $\underline{P}'(f) = \underline{P}(f)\underline{C}_{\text{room}}(f)$. The corrected sound pressure signal $p'(t)$ is the basis for spectral and transient signal processing to calculate the complex transfer response $\underline{H}_{\text{DUT}}(f)$ of the DUT and distortion components.

19.5.3 Correction of the amplitude response

This method applies a correction curve to the measured level of the amplitude response of the fundamental component. This correction curve shall only be used for correcting the frequency response at low frequencies (< 500 Hz) while using defined windowing properties. This technique cannot be used to correct the results of the nonlinear distortion measurement.

- a) The amplitude response $L_{\text{free}}(f) = 20\log|\underline{H}_{\text{free}}(f)|$ of the reference loudspeaker operated under free-field conditions shall be measured at a defined measurement point $\mathbf{r}$ (e.g. distance r on the reference axis).

For a reference loudspeaker dedicated to reproduce only low frequencies, such as a subwoofer, a half-space anechoic condition shall be used provided a ground floor measurement is applied to the DUT in the non-anechoic test environment.

For a reference loudspeaker with limited low-frequency response (main resonance above 150 Hz), measurements in a very large anechoic room (8 m × 10 m × 12 m, for example) can be sufficiently accurate.

For reference loudspeakers with extended low-frequency response, measurements on a tower (typically 10 m or more above the ground level) in the open air can become necessary. A window large enough to provide sufficient spectral resolution for the device under test at low frequencies shall be applied to the impulse response.

- b) The reference loudspeaker shall be moved to the position in the room or non-anechoic test environment located such that its reference point r_{ref} and reference axis take the positions occupied by the device under test.

The radiated sound pressure generated by the reference loudspeaker operated under test conditions shall be determined at the same measurement point (e.g. distance r on the reference axis) as used in step a).

The amplitude response $L_{test}(f) = 20\log|H_{test}(f)|$ of the reference loudspeaker operated under test conditions shall be determined in accordance with 19.1. The same windowing as used under free-field conditions shall be applied.

NOTE For measurement of the low-frequency response of a multi-unit loudspeaker system, the reference point is ideally the reference point of the bass unit.

- c) The real-value based correction curve $C_{room}(f) = L_{free}(f) - L_{test}(f)$ shall be calculated as the difference between the amplitude response measured under free-field and test conditions.
- d) The reference loudspeaker shall be replaced by the DUT and the uncorrected amplitude response $L_{DUT,test}(f)$ of the sound pressure shall be measured using the same equipment. The same windowing as used under free-field conditions shall be applied.
- e) The corrected amplitude response $L_{DUT}(f) = L_{DUT,test}(f) + C_{room}(f)$ of the DUT shall be calculated by adding the room correction curve to the uncorrected amplitude response.

19.6 Effective frequency range

19.6.1 Conditions to be specified

The range of frequencies, bounded by stated upper and lower limits, for which the measured frequency response $SPL(f)$ of the loudspeaker, in accordance with 19.2 measured on the reference axis and smoothed by a bandwidth B , is not more than 10 dB below the mean sound pressure level SPL_{mean} averaged over a stated frequency range (typically the rated frequency range). The preferred smoothing bandwidth is 1/12th octave.

19.6.2 Method of measurement

- a) The frequency response $SPL(f,r)$ shall be measured in the rated frequency range in accordance with Clause 16 at a resolution corresponding to a narrow-band filter with a relative bandwidth B . The relative bandwidth B shall be stated if the default value 1/12th octave is not used.
- b) The mean sound pressure level SPL_{mean} is calculated in the stated frequency range in accordance with 18.7.
- c) The effective frequency range shall be determined where the smoothed frequency response is not more than 10 dB below the mean sound pressure level SPL_{mean} .

19.7 Internal latency

19.7.1 Conditions to be specified

The internal latency of the DUT is the difference between the mean group delay of the DUT and the time required for the sound wave to propagate from the reference point r_{ref} to the measurement point r .

NOTE For many electronic devices, latency is understood as the signal delay between the input and output terminals. For a loudspeaker the output signal is evaluated at the reference point r_{ref} , which may be not identical with the acoustical centre depending on the frequency. Thus, the internal latency is a rough estimate of the time delay generated by the electronic components in the loudspeaker.

19.7.2 Methods of measurement

- a) The mean group delay $\tau_{mean}(r)$ between input signal $u(t)$ and sound pressure signal $p(t,r)$ at stated measurement point r shall be measured in accordance with 19.1.2 f).

- b) The internal latency caused by digital signal processing and other electronic components shall be determined by the formula:

$$\tau_{\text{lat}} = \tau_{\text{mean}} - \frac{|\mathbf{r} - \mathbf{r}_{\text{ref}}|}{c}$$

- c) where c is the speed of sound.

20 Directional characteristics

20.1 General

The direct sound radiated from the device under test is described by directional characteristics [19] used to predict the sound pressure at points relative to the source. If only far-field data is measured for the device under test, the sound pressure can only be modelled for points within the approximate far field of the device. A more complete 3D-based measurement performed in the near field, in combination with wave expansion techniques, can be used to model both the near field and the far field of the device [20].

20.2 Direct sound field in 3D space

20.2.1 Directional transfer function

20.2.1.1 Conditions to be specified

The directional transfer function $\underline{H}(f, r, \theta, \phi)$ between the input signal $u(t)$ and the sound pressure $p(t, \mathbf{r})$ of the direct sound measured at the measurement point $\mathbf{r}$ shall be specified. The measurement point $\mathbf{r}$ is described by the spherical coordinates distance r , azimuthal angle ϕ and angle θ in the stated acoustical environment in accordance with Clause 9.

20.2.1.2 Methods of measurement

- The transfer function $\underline{H}(f, r, \theta, \phi)$ shall be measured under free-field or simulated free-field conditions in accordance with Clause 9 unless the influence of the acoustical environment (e.g. limited size of the baffle) cannot be separated from the DUT.
- The position of the measurement points are defined in the region of interest in spherical coordinates with a spatial resolution required for the particular application.
- The transfer function $\underline{H}(f, r, \theta, \phi)$ is measured in accordance with 19.1 by using an attenuated stimulus (attenuation factor $\alpha = 0,1$) or by performing response measurements at reduced levels until the observed response is unchanged, verifying linear conditions.

NOTE 1 For systems where only far-field data is required, it is common practice to measure the directional transfer function at a fixed distance r that is approximately in the far field of the DUT ($r > r_{\text{far}}$).

NOTE 2 The measurements performed at two measurement points at the same directions ($\theta_1 = \theta_2$, $\phi_1 = \phi_2$) but different distances ($r_1 \neq r_2$) can be used to check the far field condition and to determine the critical distance r_{far} in accordance with 10.2.1.

20.2.2 Extrapolated far-field data

20.2.2.1 Conditions to be specified

A directional transfer function $\underline{H}(f, r_2, \phi_2, \theta_2)$ between the input signal $u(t)$ and the sound pressure $p(t, \mathbf{r}_2)$ of the direct sound in the far field at distance $r_2 > r_{\text{far}}$ and angles (ϕ_2, θ_2) is extrapolated from a transfer function $\underline{H}(f, r_1, \phi_1, \theta_1)$ measured in the far field at the distance $r_1 > r_{\text{far}}$ at the same angles ($\phi_1 = \phi_2$, $\theta_1 = \theta_2$).

20.2.2.2 Methods of measurement

- a) The transfer function $\underline{H}(f, r_1, \theta_1, \phi_1)$ shall be measured in far field at distance $r_1 > r_{\text{far}}$ under free-field or simulated free-field conditions in accordance with Clause 9 unless the influence of the acoustical environment (e.g. limited size of the baffle) cannot be separated from the DUT.
- b) The transfer function $\underline{H}(f, r_2, \theta_2, \phi_2)$ to a point at distance $r_2 > r_{\text{far}}$ in the in the same direction is calculated by

$$\underline{H}(f, r_2, \theta_2, \phi_2) = \underline{H}(f, r_1, \theta_1, \phi_1) \frac{r_1}{r_2} e^{-jk(r_2 - r_1)}$$

20.2.3 Parameters of the holographic sound field expansion

20.2.3.1 Conditions to be specified

The coefficients $C(f)$, the approximation order $N(f)$ depending on frequency f , the validity radius a and the general basic functions $\mathbf{B}(f, \mathbf{r})$ of the wave expansion describe the directional transfer function

$$\underline{H}(f, \mathbf{r}) = C(f) \mathbf{B}(f, \mathbf{r})$$

between the input signal $u(t)$ and the sound pressure output $p(t, \mathbf{r})$ at measurement point $\mathbf{r}$ at a distance $r = |\mathbf{r} - \mathbf{r}_{\text{ref}}|$ from the reference point $\mathbf{r}_{\text{ref}}$ which is larger than the validity radius a as illustrated in Figure 3.

NOTE The basic functions $\mathbf{B}(f, \mathbf{r})$ are general solutions of the wave equation and are independent of the properties of the particular DUT. Spherical harmonics and Hankel functions are suitable basis functions of the sound waves generated by compact sound system as shown in Annex B.

The region of validity shall be defined by specifying the space (2π half-space or 4π full-space) of expansion and the radius of the sphere, which is outside of the scanning surface S_s .

20.2.3.2 Method of measurements

- a) By ensuring sufficient spatial resolution, the multiple measurement points $\mathbf{r}_i$ with $i = 1, \dots, M$ are defined on at least one scanning surface S_k with $k = 1, \dots$ in the near or far field of the DUT.

NOTE A simulated free-field condition, in accordance with 9.4 and 9.5, can be generated by performing an additional scan on a second scanning surface S_2 placed between the surface S_{DUT} of the DUT and the first scanning surface S_1 as illustrated in Figure 3. Wave separation techniques [20] can be used to separate the outgoing direct sound from the incoming reflected sound generated by the acoustical environment.

- b) The transfer functions $\underline{H}(f, \mathbf{r}_i)$ between input signal $u(t)$ and the sound pressure $p(t, \mathbf{r}_i)$ at the measurement points $\mathbf{r}_i$ with $i = 1, \dots, M$ are measured on the scanning surfaces S_k with $k = 1, 2, \dots$ in accordance with 19.1.
- c) The expansion coefficients $C(f)$ are estimated by ensuring a minimum of the fitting error, which is the squared difference between measured and predicted sound pressure at all measurement points $\mathbf{r}_i$ with $i = 1, \dots, M$. The order $N(f)$ is determined to ensure that the fitting error of the wave expansion stays below a defined threshold. The radius a describing the validity of the expansion is determined by searching for a sphere with the smallest radius centred at the reference point $\mathbf{r}_{\text{ref}}$ surrounding the scanning surface S_1 .

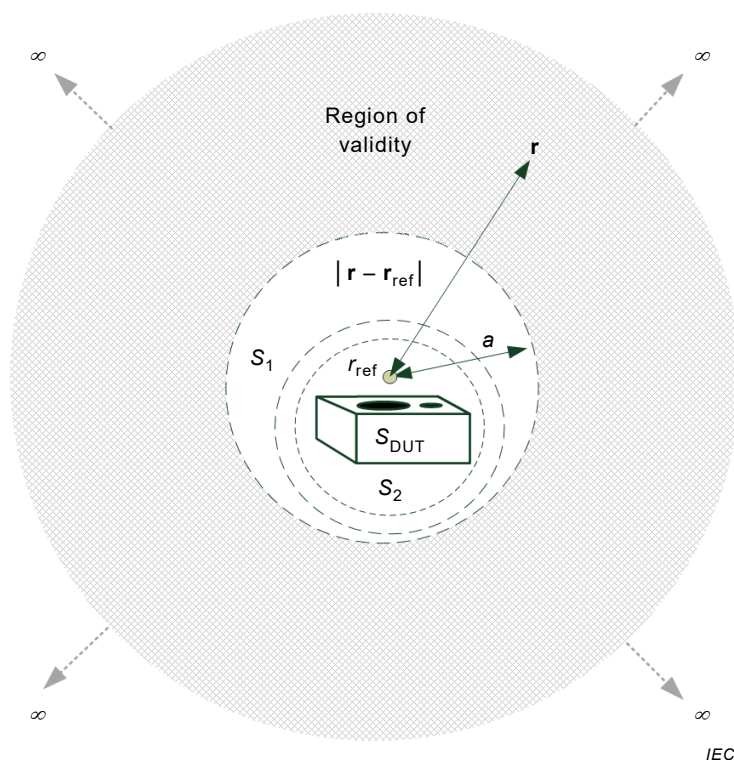


Figure 3 – Valid region of expansion of the sound pressure $p(r)$ at the observation point r at the distance $r > a$

20.2.4 Extrapolated near-field data

20.2.4.1 Conditions to be specified

The directional transfer function $\underline{H}(f, \mathbf{r})$ between the input signal $u(t)$ and the sound pressure output $p(t, \mathbf{r})$ at an evaluation point $\mathbf{r}$ at a distance $r = |\mathbf{r} - \mathbf{r}_{\text{ref}}|$ from the reference point $\mathbf{r}_{\text{ref}}$, which is larger than the validity radius a , can be calculated based on the parameters of the holographic sound field expansion.

20.2.4.2 Methods of measurement

- a) The parameters of the holographic sound field expansion shall be determined by using the scanned sound pressure in the near field of the DUT in accordance with 19.1. The coefficients $C(f)$, the order $N(f)$ of the expansion and the validity radius a shall be stated.

The sound pressure shall be scanned in the near field of the DUT to generate a minimum value of the validity radius a and to extrapolate the sound pressure at any point $\mathbf{r}$ in the near or far field of the DUT with a distance $r > a$.

- b) The directional transfer function $\underline{H}(f, \mathbf{r}) = C(f)B(f, \mathbf{r})$ shall be calculated at an evaluation point $\mathbf{r}$ at a distance $r = |\mathbf{r} - \mathbf{r}_{\text{ref}}| > a$ based on given coefficients $C(f)$ and defined basic solutions $B(f, \mathbf{r})$ of the wave equation.

20.3 Directional far field characteristics

20.3.1 Directional factor

20.3.1.1 Conditions to be specified

The directional factor

$$\underline{\Gamma}(f, \theta, \phi) = \frac{\underline{P}(f, r, \theta, \phi)}{\underline{P}(f, r, \theta_r, \phi_r)} = \frac{\underline{H}(f, r, \theta, \phi)}{\underline{H}(f, r, \theta_r, \phi_r)}$$

is the ratio between the complex sound pressure value $\underline{P}(f, r, \theta, \phi)$ at any azimuthal angle ϕ and polar angle θ to the sound pressure value $\underline{P}(f, r, \theta_r, \phi_r)$ defined by reference angles ϕ_r and θ_r measured in the far-field of the DUT. This ratio also applies using the transfer functions $\underline{H}(f, r, \theta, \phi)$ and $\underline{H}(f, r, \theta_r, \phi_r)$. If not otherwise defined by the manufacturer, the reference angles $\phi_r = 0$ and $\theta_r = 0$ correspond to the normal vector $\mathbf{n}_{\text{ref}}$ and the orientation vector $\mathbf{o}_{\text{ref}}$ in accordance with Clause 10.

NOTE It is not recommended to derive the reference angles $\phi_r = \phi_{\text{max}}(f)$ and $\theta_r = \theta_{\text{max}}(f)$ from the main radiation direction defined by

$$|\underline{H}(f, r, \theta_{\text{max}}, \phi_{\text{max}})| = \max_{\phi, \theta} (|\underline{H}(f, r, \theta, \phi)|)$$

In this case, the modulus of the directional factor $|\underline{\Gamma}(f, r, \theta, \phi)| \leq 1$ would stay below one but the reference angles $\phi_r(f)$ and $\theta_r(f)$ would vary in general with frequency f [21].

20.3.1.2 Methods of measurement

a) The measurements are performed by selecting one of the procedures described below:

- 1) In accordance with 20.2.1, the directional transfer functions $\underline{H}(f, \mathbf{r}_i)$ between the input voltage and the output sound pressure are measured at multiple points $\mathbf{r}_i$ with $i = 1, \dots, M$ in the far field of the source at a fixed distance r with defined angular resolution.
- 2) The coefficients $C(f)$ of the wave expansion are determined in accordance with 20.2.3. The wave expansion is used to extrapolate the sound propagation and to determine the sound pressure $\underline{P}(f, r, \theta, \phi)$ or the transfer functions $\underline{H}(f, r, \theta, \phi)$ at the distance r in the far field.

b) The directional factor is determined from the transfer function $\underline{H}(f, r, \theta, \phi)$ measured at a distance r on a spherical surface in the far field, or it is determined by using the coefficients of the wave expansion determined as shown in Annex E.

c) One or more of the following directional response pattern displays shall be chosen:

- 1) A family of two-dimensional polar response curves in the vertical and horizontal planes at a minimum of the preferred frequencies of 500 Hz, 1 000 Hz, 2 000 Hz, 4 000 Hz and 8 000 Hz.
- 2) Three-dimensional polar response surfaces (balloon plots) at a minimum of the preferred frequencies of 500 Hz, 1 000 Hz, 2 000 Hz, 4 000 Hz and 8 000 Hz. The minimum azimuthal angle interval shall be 10°.
- 3) A family of frequency response curves of sound pressure level or phase at various angles from the reference axis. The minimum azimuthal angle interval shall be 15°.
- 4) Directional maps displaying isocontours of sound pressure level or phase with respect to vertical/horizontal angle and frequency.
- 5) A three-dimensional display of the sound pressure level or phase at a stated frequency.

Great care shall be used to ensure that significant lobes and nulls are adequately explored. This may require an azimuthal angular interval of 2° or less in some cases [22]. In presenting the results, the orientation of the measuring axis with respect to the reference axis shall be stated. If a point-by-point method is used, the graph shall clearly show the angles used.

For very small loudspeakers such as tweeters, it may be necessary to use higher frequencies outside those mentioned above. These frequencies shall comply with those given in Annex C.

Care shall be taken that the level in the direction of the reference angles corresponds to the zero level of the polar diagram.

20.3.2 Directional gain

Expressed in decibels, the directional gain

$$D(f, \theta, \phi) = 20 \log |\underline{\Gamma}(f, \theta, \phi)|$$

shall be specified as twenty times the logarithm of the ratio between the directional factor $\underline{\Gamma}(f, \theta, \phi)$ measured in accordance with 20.3.1.

20.3.3 Directivity factor

20.3.3.1 Conditions to be specified

The directivity factor is defined as

$$Q(f) = \frac{4\pi}{\int_{\Omega} |\underline{\Gamma}(f, \theta, \phi)|^2 d\Omega}$$

with $d\Omega = \sin(\theta)d\phi d\theta$ by taking the integral of the squared directional factor over the unit sphere.

20.3.3.2 Methods of measurement

- The directional factor $\underline{\Gamma}(f, \theta, \phi)$ is measured in accordance with 20.3.1 with defined angular resolution. Alternatively, the coefficients $C(f)$ of the wave expansion can be determined by scanning the sound pressure field on a scanning surface S_k in accordance with 20.2.3.
- The directivity factor may be calculated based on the directional factor $\underline{\Gamma}(f, \theta, \phi)$ or the coefficients $C(f)$ of the wave expansion in accordance with E.3.
- For the special case of axial symmetry where, for any particular value of θ , the directional factor $\underline{\Gamma}(f, \theta, \phi)$ produced by the sound source is independent of the value of ϕ , the directivity factor can be calculated by

$$Q(f) = \frac{360^\circ}{\pi \sum_{n=1}^{180^\circ/\Delta\theta} |\underline{\Gamma}(f, \theta_n)|^2 \sin \theta_n \Delta\theta}$$

where $\Delta\theta$ is the separation in degrees of the successive points around the sound source, $180^\circ/\Delta\theta$ is the number of measurements that were made in passing from a point directly in front of the source to one directly behind the source (see [17], p. 165).

20.3.4 Directivity index

20.3.4.1 Conditions to be specified

Expressed in decibels, the directivity index

$$DI(f) = 10 \log_{10} Q(f)$$

shall be specified as ten times the logarithm of the ratio between the directivity factor $Q(f)$ measured in accordance with 20.3.3.

20.3.4.2 Methods of measurement

The directivity index $DI(f)$ may be determined by one of the following methods:

a) Calculation based on directional far-field characteristics:

The directional factor $\Gamma(f, \theta, \phi)$ shall be measured in accordance with 20.3.1 at sufficient angular resolution. The directivity index $DI(f)$ is calculated based on measured directivity factor $Q(f)$ in accordance with 20.3.3.

b) Calculation based on coefficients $C(f)$ of the wave expansion:

The coefficients $C(f)$ of the wave expansion are determined in accordance with 20.2.3 and the directivity factor index $DI(f)$ is calculated based on measured directivity $Q(f)$ in accordance with 20.3.3.

c) Measurements under free-field and diffuse conditions:

- 1) In accordance with 19.2, the sound pressure level response $SPL_{ax}(f)$ shall be measured under free-field conditions on the reference axis at a distance of 1 m using an attenuated broadband stimulus (attenuation factor $\alpha = 0,1$).
- 2) The sound pressure level $SPL_{dif}(f)$ shall be measured under diffuse field conditions using the same attenuated broadband stimulus.
- 3) The directivity index $DI(f)$, expressed in dB, is then determined from the formula:

$$DI(f) = SPL_{ax}(f) - SPL_{dif}(f) + 10 \lg(T(f)) - 10 \lg(V(f)) + 25$$

where

$SPL_{ax}(f)$ is the sound pressure level in dB under free-field conditions measured on the reference axis and referred to a distance of 1 m,

$SPL_{dif}(f)$ is the sound pressure level in dB measured under diffuse field conditions,

$T(f)$ is the reverberation time of the reverberation room in seconds (s),

$V(f)$ is the reverberation room volume in cubic metres (m³).

20.4 Acoustic output power

20.4.1 Conditions to be specified

The total sound power radiated by a DUT shall be specified in a given frequency band with a centre frequency f and a defined input signal $u(t)$ generating an RMS value $\tilde{u}(f)$ in this frequency band.

20.4.2 Methods of measurement

The acoustic output power is determined by one of the following methods:

a) Integration of the far-field sound pressure on a sphere:

The acoustic output power may be calculated by integrating the sound pressure on the sphere or by the following equation

$$\Pi(f) = \frac{r^2}{\rho c} \int_{\Omega} \tilde{p}^2(f, r, \theta, \phi) d\Omega$$

of the RMS sound pressure $\tilde{p}(f, r)$ on a sphere in the far field. The acoustic power under free-field full space conditions may be determined by the formula:

$$\Pi(f) = \frac{4\pi r^2}{\rho c} \overline{\tilde{p}^2(f)} = 0,031r^2 \overline{\tilde{p}^2(f)}$$

where

$\Pi(f)$ is the acoustic power in W,

r is the sphere radius in m,

$\overline{\tilde{p}^2(f)}$ is the squared sound-pressure averaged over a large sphere in squared Pa,

ρ and c are the density and the sound speed of air under standard conditions.

The acoustic power under half-space free-field conditions shall be determined by the formula:

$$\Pi(f) = \frac{2\pi r^2}{\rho c} \overline{\tilde{p}^2(f)} = 0,016r^2 \overline{\tilde{p}^2(f)}$$

In the case of full-space free-field conditions, the square of the RMS sound pressure shall be averaged over a large sphere, and, in the case of half-space free-field conditions, over a large hemisphere in accordance with ISO 3744 and ISO 3745. In both cases, there is a requirement for a large number of points evenly distributed around the system under measurement.

If the system has axial symmetry of revolution, measurements in a plane containing this axis might be considered sufficient provided the measurements are suitably weighted in the averaging process.

Alternatively, the acoustic output power may be calculated by integrating the transfer function on the sphere or by the following equation

$$\Pi(f) = \tilde{u}^2(f) \frac{r^2}{\rho c} \int_{\Omega} |H(f, r, \theta, \phi)|^2 d\Omega$$

using the RMS value $\tilde{u}(f)$ with a given frequency band and a centre frequency f .

b) Calculation based on far-field characteristics:

The acoustic output power may be calculated by the following equation

$$\Pi(f) = \frac{\tilde{p}^2(f, r, \theta_r, \phi_r)}{\rho c} \int_{\Omega} |\underline{\Gamma}(f, \theta, \phi)|^2 d\Omega = \frac{\tilde{p}^2(f, r, \theta_r, \phi_r) 4\pi r^2}{Q(f) \rho c}$$

using the RMS sound pressure value $\tilde{p}(f, r, \theta_r, \phi_r)$ at distance r on the reference axis defined by reference angles ϕ_r , θ_r and the directional factor $\underline{\Gamma}(\theta, \phi)$ or directivity factor $Q(f)$ measured in accordance with 20.3.1 or 20.3.3, respectively.

c) Calculation based on coefficients of the wave expansion:

The acoustic output power is calculated by using the coefficients $C(f)$ in accordance with E.4.

d) Measurement of acoustic power under diffuse field conditions in accordance with ISO 3741:

The acoustic power of the loudspeaker $\Pi(f)$ is approximately given by the relation:

$$\Pi(f) = \frac{V}{T_{60}(f)} \tilde{p}(f)^2 10^{-4}$$

where

$\Pi(f)$ is the acoustic power in W,

V is the reverberation room volume in m³,

$T_{60}(f)$ is the reverberation time in seconds of the room in the frequency band considered,

$\tilde{p}(f)$ is the sound-pressure in Pa.

The filtering may take place either in the loudspeaker chain or in both the loudspeaker and the microphone chains.

20.5 Sound power level

Expressed in decibels, the sound power level

$$L_{\Pi}(f) = 10 \log_{10} \left(\frac{\Pi(f)}{\Pi_0} \right)$$

shall be specified as ten times the logarithm of the ratio between the acoustic output power $\Pi(f)$ measured in accordance with 20.4 and the reference power $\Pi_0 = 10^{-12}$ W.

20.6 Mean acoustic output power in a frequency band

20.6.1 Conditions to be specified

The arithmetic mean of the acoustic output power in all 1/3 octave frequency bands located within the frequency band being considered.

20.6.2 Method of measurement

- a) The measurement shall be made in accordance with 20.4.2.
- b) The mean acoustic output power shall be calculated as the arithmetic mean of the individually measured acoustic output powers for each of the 1/3 octave frequency bands contained within the frequency range being considered.

20.7 Radiation angle

20.7.1 Conditions to be specified

The angle measured with respect to the reference axis (in a plane containing this axis) where the sound-pressure level at the measuring distance has decreased with off-axis angle for the first time by 10 dB with respect to the sound-pressure level on the reference axis. In addition, the frequency range over which this specification is met shall be stated.

NOTE This definition is less practical at high frequencies where many, narrow lobes require a high angular resolution.

20.7.2 Method of measurement

- a) The radiation angle shall be deduced from the beam pattern measured in accordance with 20.3.2 in the rated frequency range.
- b) The radiation angle may be plotted as a graph with frequency on the abscissa and the angles on the ordinate, symmetrical with respect to 0°.

NOTE A logarithmic ordinate (vertical) axis is a useful presentation, as the asymptotes of radiation angle at low frequencies can be more clearly seen.

- c) If the directional response pattern of the loudspeaker has no cylindrical symmetry, the value shall be given in two perpendicular planes.

20.8 Coverage angle or angles

20.8.1 Conditions to be specified

The angle, measured in a plane containing the reference axis, between the two directions on either side of the main lobe of the directional gain, in accordance with 20.3.2 centred on a specific frequency, where the sound-pressure level is 6 dB less than that in the direction of the reference axis.

For loudspeakers which are designed to have different coverage angles in different planes through the reference axis, coverage angles shall be specified in at least two orthogonal planes.

20.8.2 Method of measurement

- a) If the effective frequency range of the loudspeaker includes both 2 800 Hz and 5 700 Hz (1/2 octave above and below 4 000 Hz), the coverage angle or angles shall be deduced from the directional response pattern or patterns measured with an octave band centred on 4 000 Hz.

If the effective frequency range does not include the octave band centered on 4 000 Hz, the coverage angle or angles shall be deduced from measurements in an octave band of specified centre frequency near the upper limit of the effective frequency range.

In addition, the coverage angle or angles may be specified for other center frequencies of octave bands.

The octave band centre frequency or frequencies used for the measurements shall be presented with the measured data.

- b) The values shall be presented in a table or as a diagram.

20.9 Mean sound pressure level in an acoustical zone

20.9.1 General

The mean sound pressure level shall be calculated based on the mean value of the squared sound pressure $p(t, \mathbf{r}_n)$ at multiple measurement points $\mathbf{r}_n$ with $n = 1, \dots, N$ representing an acoustical zone in the direct sound field generated by the device under test. The manufacturer shall state the coordinates of acoustical zone, the number of points and a weighting function $w(f, \mathbf{r}_n)$, if applied.

20.9.2 Method of measurement

- a) Define a grid of points $\mathbf{r}_n$ with $n = 1, \dots, N$ in the stated acoustical zone to sample the sound field with sufficient resolution.

For the evaluation of the far field properties, it is recommended to define the acoustical zone as a segment or line on a sphere of radius r (e.g. $r = 2$ m as defined for in-home loudspeakers for CTA 2034-A). For the evaluation of other applications, it is recommended to define the acoustical zone as a cuboid as defined in IEC 62777.

- b) Measure the directional transfer function $\underline{H}(f, \mathbf{r}_n)$ between the input signal $u(t)$ and sound pressure $p(t, \mathbf{r}_n)$ at points $\mathbf{r}_n$ with $n = 1, \dots, N$ in accordance with 20.2.1.
- c) Calculate the weighted mean sound pressure level in an acoustical zone as

$$L_{\text{zone}}(f) = \frac{1}{\sum_{n=1}^N w(f, \mathbf{r}_n)} \sum_{n=1}^N w(f, \mathbf{r}_n) L(f, \mathbf{r}_n)$$

$$= \frac{1}{\sum_{n=1}^N w(f, \mathbf{r}_n)} \sum_{n=1}^N w(f, \mathbf{r}_n) 20 \log \left(\frac{\tilde{u}(f)}{p_0} |H(f, \mathbf{r}_n)| \right)$$

using the weighting function $w(f, \mathbf{r}_n)$ defined for the particular application, the RMS value of the input signal and reference sound pressure $p_0 = 20 \mu\text{Pa}$.

21 Harmonic distortion

21.1 General

A general explanation of harmonic distortion can be found in IEC 60268-2, and there is further information in Annex F.

21.2 N^{th} -order harmonic component

21.2.1 Conditions to be specified

The n^{th} -order harmonic component in terms of total sound pressure $\tilde{p}$ are measured under normal measurement conditions by exciting the DUT with a sinusoidal stimulus $u(t) = \alpha \tilde{u}_{\text{max}} x(t)$ with frequency f . The properties of the stimulus $u(t)$, the measurement time T_s and either the RMS value $\tilde{u} = \alpha \tilde{u}_{\text{max}}$ or the scaling factor α shall be stated.

21.2.2 Method of measurement

- A series of sinusoidal input stimuli with increasing frequencies, up to 10 kHz, shall be supplied to the loudspeaker. The chosen RMS value $\tilde{u}$ of the input signal $u(t) = \alpha \tilde{u}_{\text{max}} x(t)$ shall be those that are the most relevant for the intended use, but shall not exceed the maximum input value $\tilde{u}_{\text{max}}$.

A sinusoidal chirp is a recommended stimulus, because the step-by-step method using steady-state tones may cause important information to be missed. A Hann-Burst signal as defined in 8.8 can be used as a short excitation signal distributing the spectral power over one third of an octave. This measurement technique developed by Keele [5] and shall be performed in accordance with CTA 2034-A and CTA 2010-B.

- The sound pressure signal $p(t)$ is measured at the measurement point $\mathbf{r}$ under normal measurement conditions in accordance with 7.3. If not stated otherwise, the measurement point $\mathbf{r}$ is the evaluation point $\mathbf{r}_e$.
- The RMS value of the fundamental component $\tilde{p}_f(f)$ and the RMS values $\tilde{p}_{nf}(f)$ of n^{th} -order harmonic components shall with $n > 1$ be determined by applying spectral analysis to the measured sound pressure signal $p(t)$.
- Expressed in decibels, the sound pressure level $L_f(f)$ of the fundamental component shall be specified as twenty times the logarithm of the ratio between the RMS value $\tilde{p}_f(f)$ and the standard reference sound pressure (20 μPa).
- Expressed in decibels, the sound pressure level $L_{nH}(f)$ of the n^{th} -order harmonic components shall be specified as twenty times the logarithm of the ratio between the RMS value $\tilde{p}_{nf}(f)$ of the n^{th} -order harmonic components ($n > 1$) and the standard reference sound pressure (20 μPa).

NOTE The level of the harmonic distortion components (not referred to the total RMS value) can directly be compared with the level of the fundamental component.

- f) The results of the measurement shall be presented graphically as a function of the fundamental frequency. With the results, the following information shall be given:
- the input RMS value or the scaling factor α and the sound-pressure level at the measurement point r ;
 - the method used, a sinusoidal chirp signal or a steady-state signal varied step-by-step or Hann-Burst stimulus;
 - any discrete frequencies used;
 - the measurement position of the microphone if it differs from the evaluation point r_e and the conditions of the measurement (free-field or half-space free-field);
 - bandwidth of spectral smoothing if applied to the frequency response.

21.3 Total harmonic components

21.3.1 Conditions to be specified

The total harmonic components in terms of total sound pressure $\tilde{p}$ are measured under normal measurement conditions by exciting the DUT with a sinusoidal stimulus $u(t) = \alpha \tilde{u}_{\max} x(t)$ with frequency f . The properties of the stimulus $u(t)$, the measurement time T_s and either the RMS value $\tilde{u} = \alpha \tilde{u}_{\max}$ or the scaling factor α shall be stated.

21.3.2 Method of measurement

- a) The RMS value $\tilde{p}_{nf}(f)$ of the n th-order harmonic components in the measured sound-pressure signal $p(t)$ shall be measured in accordance with 21.2.2 a) and b).
- b) The RMS value of the total harmonic component fundamental component $\tilde{p}_{TH}(f)$ and the shall be determined by the formula:

$$\tilde{p}_{TH}(f) = \sqrt{\sum_{n=2}^N \tilde{p}_{nf}^2(f)}$$

where the highest harmonic order N considered in the $THD(f)$ shall be stated.

- c) Expressed in decibels, the sound pressure level $L_{TH}(f)$ of the total harmonic components shall be specified as twenty times the logarithm of the ratio between the RMS value $\tilde{p}_{TH}(f)$ of the total harmonic components and the standard reference sound pressure (20 μPa).

NOTE The level of the total harmonic components $L_{TH}(f)$ can directly be compared with the level $L_f(f)$ of the fundamental component and with the levels of individual n th-order harmonic components $L_{nf}(f)$.

21.4 Total harmonic distortion

21.4.1 Conditions to be specified

The total harmonic distortion in terms of total sound pressure $\tilde{p}$ under normal measurement conditions by exciting the DUT with a sinusoidal stimulus $u(t) = \alpha \tilde{u}_{\max} x(t)$ with frequency f . The properties of the stimulus $u(t)$, the measurement time T_s and either the RMS value $\tilde{u} = \alpha \tilde{u}_{\max}$ or the scaling factor α shall be stated.

21.4.2 Method of measurement

- a) The RMS value $\tilde{p}_{nf}(f)$ of the n th-order harmonic components in the measured sound pressure signal $p(t)$ shall be measured in accordance with 21.2.2.

- b) The RMS value $\tilde{p}_{\text{ref}}(f)$ of a reference signal shall be determined. The RMS value $\tilde{p}(f)$ of the total sound pressure signal including the fundamental, all harmonics and noise shall be used as the reference value $\tilde{p}_{\text{ref}}(f)$ if not stated otherwise.

The total harmonic distortion can be determined by the formula:

in percentage:

$$THD(f) = \frac{\sqrt{\sum_{n=2}^N \tilde{p}_{nf}^2(f)}}{\tilde{p}_{\text{ref}}(f)} 100\%$$

in decibels:

$$L_{THD}(f) = 20 \lg \left(\frac{THD}{100\%} \right)$$

where the highest harmonic order N considered in the $THD(f)$ shall be stated.

- c) The results of the measurement shall be presented graphically as a function of the fundamental frequency on a log abscissa (y -axis), if in percent. With the results, the following information shall be given:
- the input RMS value or the scaling factor α and the sound pressure level at the measurement point r ;
 - the method used, a sinusoidal chirp signal or a steady-state signal varied step-by-step;
 - any discrete frequencies used;
 - the measurement position of the microphone if it differs from the evaluation point r_e and the conditions of the measurement (free-field or half-space free-field);
 - bandwidth of spectral smoothing if applied to the frequency response.

21.5 Higher-order harmonic distortion

21.5.1 Conditions to be specified

The higher-order harmonic distortion in terms of total sound pressure $\tilde{p}$ under normal measurement conditions by exciting the DUT with a sinusoidal stimulus $u(t) = \alpha \tilde{u}_{\text{max}} x(t)$ with frequency f . The properties of the stimulus $u(t)$, the measurement time T_s and either the RMS value $\tilde{u} = \alpha \tilde{u}_{\text{max}}$ or the scaling factor α shall be stated.

NOTE The higher-order harmonic distortion HOHD evaluates the energy of spectral components at frequencies nf , which are multiples n of the excitation frequency f with $N_1 \leq n \leq N$. This characteristic reveals the deterministic, nonlinear properties of the device under test such as hard limiting of the mechanical suspension at high excursion. However, the higher-order harmonic distortion cannot reliably detect loose particles, and other loudspeaker defects with random properties (e.g. voice coil rubbing, buzzing, air leakage noise) generating impulsive distortion comprising non-harmonic components. The measurement of impulsive distortion based on time-domain analysis according to Clause 24 provides maximum sensitivity for both random and deterministic defects as described in Clause F.3.

21.5.2 Method of measurement

- a) The RMS value $\tilde{p}_{nf}(f)$ of the n th-order harmonic components in the measured sound pressure signal $p(t)$ shall be measured in accordance with 21.2.2.
- b) The RMS value $\tilde{p}_{\text{ref}}(f)$ of a reference signal shall be determined. The RMS value $\tilde{p}(f)$ of the total sound pressure signal including the fundamental, all harmonics and noise shall be used as the reference value $\tilde{p}_{\text{ref}}(f)$ if not stated otherwise.