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100/2531/NP

NEW WORK ITEM PROPOSAL

	Proposer Germany	Date of proposal 2015-05-22
	TC/SC 100	Secretariat Japan
	Date of circulation 2015-06-26	Closing date for voting 2015-10-02

A proposal for a new work item within the scope of an existing technical committee or subcommittee shall be submitted to the Central Office. The proposal will be distributed to the P-members of the technical committee or subcommittee for voting on the introduction of it into the work programme, and to the O-members for information. The proposer may be a National Committee of the IEC, the secretariat itself, another technical committee or subcommittee, an organization in liaison, the Standardization Management Board or one of the advisory committees, or the General Secretary. Guidelines for proposing and justifying a new work item are given in ISO/IEC Directives, Part 1, Annex C (see extract overleaf). **This form is not to be used for amendments or revisions to existing publications.**

The proposal (to be completed by the proposer)

Title of proposal

IEC 60268-X - SOUND SYSTEM EQUIPMENT – LOUDSPEAKERS – ACOUSTICAL (OUTPUT BASED) MEASUREMENTS

☒ Standard ☐ Technical Specification

Scope (as defined in ISO/IEC Directives, Part 2, 6.2.1)

This International Standard applies to passive and active sound systems such as loudspeakers, headphones, TV-sets, multi-media devices, personal portable audio devices, automotive sound systems and professional equipment. The device under test (DUT) may 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. The standard describes only physical measurements which 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. The standard does not assess the perception and cognitive evaluation of the reproduced sound and the impact of perceived sound quality.

Purpose and justification, including the market relevance, whether it is a proposed horizontal standard (Guide 108)[1)] and relationship to Safety (Guide 104), EMC (Guide 107), Environmental aspects (Guide 109) and Quality assurance (Guide 102) . (attach a separate page as annex, if necessary)

This standard satisfies the requirement of modern active audio systems where existing measurement standards (e.g. IEC 60268-5) developed for passive systems are not applicable. Furthermore, portable and personal audio devices require a more comprehensive assessment of the acoustical output in the 3D space considering a physical assessment of the signal distortion which have a high impact on the perceptual sound quality.
This International Standard is not proposed as a horizontal standard and has no relationship to Safety, EMC, Environmental aspects and Quality assurance.

Target date	for first CD 2015-11	for IS/ TS 2017-11
Estimated number of meetings	Frequency of meetings: per year	Date and place of first meeting:
Proposed working methods	☐ E-mail	☒ Collaboration tools

Relevant documents to be considered

IEC 60268-5

Relationship of project to activities of other international bodies

Liaison organizations	**Need for coordination within ISO or IEC**

1) Other TC/SCs are requested to indicate their interest, if any, in this NP to the TC/SC secretary.

Preparatory work

Ensure that all copyright issues are identified. Check one of the two following boxes

☒ A draft is attached for comment* ☐ An outline is attached

* Recipients of this document are invited to submit, with their comments, notification of any relevant patent rights of which they are aware and to provide supporting documentation.

We nominate a project leader as follows in accordance with ISO/IEC Directives, Part 1, 2.3.4 (name, address, fax and e-mail): Wolfgang Klippel, Email: wklippel@klippel.de

Concerns known patented items (see ISO/IEC Directives, Part 2)	**Name and/or signature of the proposer**
☐ Yes. If yes, provide full information as an annex ☒ no	Wolfgang Klippel

Comments and recommendations from the TC/SC officers

1) Work allocation

☒ Project team ☐ New working group ☐ Existing working group no:

2) Draft suitable for direct submission as

☒ CD ☐ CDV/ DTS

3) General quality of the draft (conformity to ISO/IEC Directives, Part 2)

☒ Little redrafting needed ☐ Substantial redrafting needed ☐ no draft (outline only)

4) Relationship with other activities

In IEC

In other organizations

5) Proposed horizontal standard

☐ [1)]

Remarks from the TC/SC officers

This proposal has been discussed in GMT meetings with the other three proposals from DE, CN and US. It was agreed to circulate those NPs at the GMT meeting which was held on 2014-10-10 in Los Angeles. This project will be allocated directly under TC 100.

[1)] Other TC/SCs are requested to indicate their interest, if any, in this NP to the TC/SC secretary.

Approval criteria:

- Approval of the work item by a simple majority of the P-members voting;
- At least 4 P-members in the case of a committee with 16 or fewer P-members, or at least 5 P-members in the case of committees with more than 17 P-members, have nominated or confirmed the name of an expert and approved the new work item proposal.

Elements to be clarified when proposing a new work item

Title

Indicate the subject matter of the proposed new standard or technical specification.

Indicate whether it is intended to prepare a standardor a technical specification.

Scope

Give a clear indication of the coverage of the proposed new work item and, if necessary for clarity, exclusions.

Indicate whether the subject proposed relates to one or more of the fields of safety, EMC, the environment or quality assurance.

Purpose and justification

Give details based on a critical study of the following elements wherever practicable.

a) The specific aims and reason for the standardization activity, with particular emphasis on the aspects of standardization to be covered, the problems it is expected to solve or the difficulties it is intended to overcome.

b) The main interests that might benefit from or be affected by the activity, such as industry, consumers, trade, governments, distributors.

c) Feasibility of the activity: Are there factors that could hinder the successful establishment or general application of the standard?

d) Timeliness of the standard to be produced: Is the technology reasonably stabilized? If not, how much time is likely to be available before advances in technology may render the proposed standard outdated? Is the proposed standard required as a basis for the future development of the technology in question?

e) Urgency of the activity, considering the needs of the market (industry, consumers, trade, governments etc.) as well as other fields or organizations. Indicate target date and, when a series of standards is proposed, suggest priorities.

f) The benefits to be gained by the implementation of the proposed standard; alternatively, the loss or disadvantage(s) if no standard is established within a reasonable time. Data such as product volume of value of trade should be included and quantified.

g) If the standardization activity is, or is likely to be, the subject of regulations or to require the harmonization of existing regulations, this should be indicated.

If a series of new work items is proposed, the purpose and justification of which is common, a common proposal may be drafted including all elements to be clarified and enumerating the titles and scopes of each individual item.

Relevant documents

List any known relevant documents (such as standards and regulations), regardless of their source. When the proposer considers that an existing well-established document may be acceptable as a standard (with or without amendments), indicate this with appropriate justification and attach a copy to the proposal.

Cooperation and liaison

List relevant organizations or bodies with which cooperation and liaison should exist.

Preparatory work

Indicate the name of the project leader nominated by the proposer.

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INTERNATIONAL ELECTROTECHNICAL COMMISSION

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SOUND SYSTEM EQUIPMENT – ACOUSTICAL (OUTPUT BASED) MEASUREMENTS

FOREWORD

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SOUND SYSTEM EQUIPMENT – ACOUSTICAL (OUTPUT BASED) MEASUREMENTS

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 may 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 may 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.

1 Scope

This International Standard applies to passive and active sound systems such as loudspeakers, headphones, TV-sets, multi-media devices, personal portable audio devices, automotive sound systems and professional equipment. The device under test (DUT) may 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. The standard describes only physical measurements which 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. The standard does not assess the perception and cognitive evaluation of the reproduced sound and the impact of perceived sound quality.

NOTE This standard does not apply to microphones and other sensors. This standard 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 standard document IEC 60268-Xb dedicated to electrical and mechanical measurements.

2 Normative references

The following documents, in whole or in part, are normatively referenced in this document and are indispensable for its application. For dated references, only the cited edition 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, *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 61260-1:2014, *Electroacoustics – Octave-band and fractional-octave-band filters – Part 1: Specifications*

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

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

ISO/IEC Guide 98-3, *Uncertainty of measurement – Part 3: Guide to the expression of uncertainty in measurement (GUM:1995)*

NOTE Informative references are listed in the appendix **Fehler! Verweisquelle konnte nicht gefunden werden.**.

3 Abbreviations

The abbreviation DUT represents the 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 Marking of terminals and controls

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

6 Physical characteristics

6.1 Dimensions

The outer dimensions of the DUT in accordance with IEC 60268-14 shall be specified.

6.2 Mass

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

6.3 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 which are currently standardized are unsuitable and the use of other types is unavoidable.

7 Design data

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

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

8 Conditions

8.1 Rated conditions

For convenience, this standard specifies how sound system equipment shall be set up for measurement. Normal measuring conditions are defined in this standard. 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;
- evaluation point;
- rated frequency range;
- reference plane;
- reference point;
- reference axis;
- orientation vector.

8.2 Normal measuring conditions

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

a) The DUT to be measured is mounted in accordance with clause 14;

b) The acoustical environment is specified and selected from those given in clause 10;

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 11;

e) The DUT is acclimatised 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 clause 20.3.

NOTE The influence of the ambient atmospheric condition on acoustic measurements is discussed in greater detail in [19].

f) Additional cooling periods are required between successive tests if compression C > 0.5 dB in accordance with clause 20.3.

g) The DUT is supplied with a test signal with specified properties (spectrum, duration, etc.) in accordance with clause 9 at a specified r.m.s. input value $\tilde{u}$ for the rated frequency range in accordance with clause 17;

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 12.

9 Test signals

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 following test signals.

9.1 Sinusoidal chirp

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

$$x_c(t) = \sqrt{2}A(f(t))\cos\left(2\pi f(t)t\right) \tag{1}$$

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

$$f(t) = f_{\text{start}}2^{\beta t} \qquad 0 \le t \le T_s \tag{2}$$

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 s < T_s < 10 s typically. The sweep rate parameter

$$\beta = \frac{1}{T_S} \log_2 \left(\frac{f_{\text{end}}}{f_{\text{start}}} \right) \tag{3}$$

describes the time required for doubling the instantaneous frequency.

NOTE 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.

9.2 Steady-state single-tone signal

The electro-acoustical system is excited by a single tone

$$x_s(t) = \sqrt{2}\cos(2\pi f_1 t) \qquad 0 \le t \le T_P + T_M = \frac{N_P + N_M}{f_1} \tag{4}$$

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

$$T_M = \frac{N_M}{f_1} \tag{5}$$

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

9.3 Steady-state two-tone signal

The electro-acoustical system is excited by two tones

$$x_t(t) = \sqrt{\frac{2}{A_1^2 + A_2^2}}\left(A_1 \cos(2\pi f_1 t) + A_2 \cos(2\pi f_2 t)\right) \qquad 0 \le t \le (T_P + T_M) \tag{6}$$

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

$$T_M = \frac{N_M}{f_1 f_2} 1\,\text{Hz} \tag{7}$$

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.

9.4 Sparse multi-tone complex

The multi-tone complex defined by

$$x_m(t) = \sqrt{\frac{2}{\sum_{i=0}^{N} A(f_i)^2}} \sum_{i=0}^{N} A(f_i)\cos(2\pi f_i t + \varphi_i) \qquad 0 < t \le T_P + T_M \tag{8}$$

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

$$f_i = f_b \operatorname{int}\left(T \cdot f_0 \cdot 2^{i/R}\right) \quad with\ i = 0,...,N \tag{9}$$

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

$$f_\mathrm{b} = \frac{1}{T} = \frac{f_s}{N_\mathrm{FT}} \tag{10}$$

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

$$\varphi_{i+1} = \frac{2\pi}{m}\left[\left(\frac{a\varphi_i m}{2\pi}\right)\mathrm{mod}_m\right] \qquad with\ i = 0,...,N-1 \tag{11}$$

generating a low crest factor of the stimulus and a deterministic waveform of the stimulus.

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

The parameters a, m and the starting phase φ_0 determine the crest factor of the generated time signal. If the recommended values a=48271, m=2^{31}–1 and φ_0=1 rad are not used, other values of parameters shall be stated by the manufacturer. The pre-excitation time T_P with 0.5 $T < T_P < T$ is required to generate the steady-state condition of the electro-acoustical system under test. The measurement time $T_M = N_M T$ with an integer number of repetitions $N_M \ge 1$ is recommended for temporal averaging and Fourier analysis.

9.5 Broadband noise signal

As defined in IEC 60268-2, clause 1.4.

9.6 Narrow-band noise signal

As defined in IEC 60268-2, clause 1.4.

9.7 Hann-burst signal

The Hann-burst signal,

$$x_b(t) = \begin{cases} \left(1-\cos\left(\dfrac{2\pi f_0 t}{6.5}\right)\dfrac{\sin(2\pi f_0 t)}{2}\right) & for\ 0 \le t \le 6.5/f_0 \\ 0 & ,\text{elsewhere} \end{cases} \tag{12}$$

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.

9.8 Impulsive signal

As defined in IEC 60268-2.

10 Acoustical environment

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

10.1 Free-field conditions

Acoustical conditions which 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 ±0.5 dB and ±10 degrees 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 entire frequency range of the measurement, the manufacturer shall state the valid frequency range.

NOTE The amplitude response may be corrected in accordance with 20.5 if the free-field conditions are not fulfilled. The correction applied, if any, shall be stated.

10.2 Half-space, free-field conditions

Acoustical conditions in which the free field, according to clause 10.1, exists in a half-space may be used. 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 ±0.5 dB and ±10 degrees in the measured sound pressure amplitude and phase values at the specified frequency. If the environment (e.g. anechoic room at low frequencies) does not fulfil these free field conditions over the entire frequency range of the measurement, the manufacturer shall state the valid frequency range.

10.3 Simulated free-field conditions

Acoustical conditions of any environment (e.g. non-anechoic room) shall be satisfactory if the direct sound radiated by the DUT, including any secondary reflections from the DUT's surface, are separated from the room reflections by using measurement techniques such as gating of the impulse response, wave separation, or acoustical holography. These measurement techniques shall ensure an accuracy of ±0.5 dB and 10 degrees in the measured sound pressure amplitude and phase values at the specified frequency. The frequency resolution Δf = 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 which 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 will increase resolution virtually but provide no additional information.

10.4 Half-space simulated free-field conditions

Acoustical conditions, according to clause 10.3, where the device under test is mounted in a reflecting plane of sufficient size.

10.5 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, Appendix A.

10.6 Target application conditions

Acoustical conditions which correspond with the final target application of the device under test (e.g. sound system mounted in a car).

11 Positioning of the DUT

11.1 Rated geometrical conditions

The position and orientation of the DUT shall be stated using the reference point $\mathbf{r}_{ref}$, the normal vector $\mathbf{n}_{ref}$ and the orientation vector $\mathbf{o}_{ref}$ as illustrated in Figure 1.

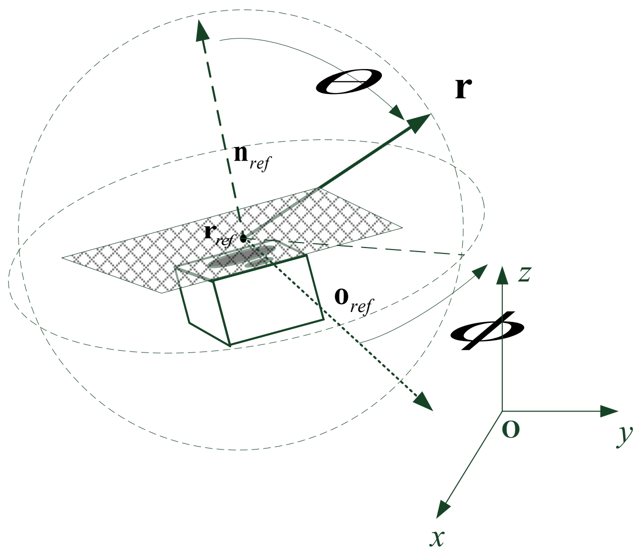


Figure 1 – Rated conditions used to describe the position of DUT in the coordinate system

11.1.1 Reference plane and normal vector

11.1.1.1 Condition to be specified

The reference plane with the normal vector $\mathbf{n}_{ref}$ shall be used to define the reference axis and the reference point $\mathbf{r}_{ref}$. The normal vector $\mathbf{n}_{ref}$ also defines the polar angle θ=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 is better indicated by means of a diagram.

11.1.2 Reference point

11.1.2.1 Condition to be specified

A point on the reference plane. The position of the reference point $\mathbf{r}_{ref}$ shall be specified by the manufacturer.

NOTE For symmetrical structures, reference point $\mathbf{r}_{ref}$ is usually the point of axial symmetry of the radiator within the reference plane; for asymmetrical structures, the reference point is better indicated by means of a diagram.

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

11.1.3 Reference axis

11.1.3.1 Condition to be specified

The line which passes through the reference plane at the reference point and 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.

NOTE For symmetrical structures, the reference axis is usually perpendicular to the radiating surface or to the reference plane.

11.1.4 Orientation vector

11.1.4.1 Condition to be specified

The orientation vector $\mathbf{o}_{ref}$ defines the orientation of the sound system within the reference plane and the direction of azimuthal angle ϕ=0 in spherical coordinates.

NOTE The preferred orientation vector places the reference point $\mathbf{r}_{ref}$ at the origin O of the coordinate system, pointing the normal vector $\mathbf{n}_{ref}$ of the reference plane into the z-direction where the polar angle θ=0 and turning the audio system in such a way that the orientation vector $\mathbf{o}_{ref}$ (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

$$\mathbf{r} = x\vec{e}_x + y\vec{e}_y + z\vec{e}_z = r\ \cos\phi\sin\theta\vec{e}_x + r\ \sin\phi\sin\theta\vec{e}_y + r\ \cos\theta\vec{e}_z \qquad (13)$$

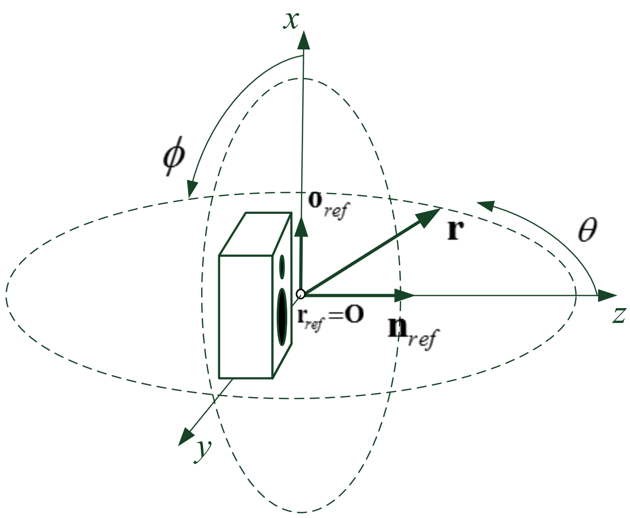


Figure 2 – Recommended position and orientation of the DUT

11.1.5 Evaluation point

Condition to be specified

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.

NOTE 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 18 and clause 19. Ideally, the evaluation distance places the evaluation point in the far field of the DUT (see 11.2.1).

11.1.6 Evaluation distance

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

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 11.2.1).

11.2 Measuring distance between DUT and microphone

11.2.1 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 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 ($r/d >> d/\lambda$).

However, in practice, limitations of the measuring environment (e.g. room size) 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 of ±0.5 dB and ±10 degrees in the measured sound pressure amplitude and phase at any frequency within the frequency range of measurement.

11.2.2 Near-field conditions

Measurements of the sound pressure in the near field of the sound source provide additional information about the DUT, which is required for assessing studio monitors, personal audio devices, laptops, automotive and other applications where the listener is placed at a short distance to the sound source.

It is also beneficial to perform measurements under simulated free field conditions close to the source in order to maximize the time available before the first unwanted reflections reach the microphone. In addition, measurements made close to the source maximize the sound pressure of the direct sound, thereby dominating the room reflections and ambient noise.

The measurements of DUTs of large size (e.g. line array loudspeakers) in small anechoic rooms cannot be performed under far-field conditions because the measured sound pressure data is affected by the influence of the near field.

Valid far-field information can be calculated from the near-field data by using holographic methods.

Note: The extrapolation technique developed by Keele [8] provides reliable far field information at low frequencies based on a single near field measurement if the woofers is mounted in a sealed enclosures and the sound field is omnidirectional.

11.2.3 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 clause 21.3. The microphone system and nearest microphone position shall meet the requirement of ISO 3741.

11.2.4 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.

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

12 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. 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 the preferred aspect ratios in IEC 60263.

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

13 Accuracy of the acoustical measurement

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 (13.1). This information shall be included with the test report.

NOTE Techniques for quantifying acoustic measurement tolerances are discussed in [19]. Working with two microphones located at different positions (or distances) can be used to check the possible errors in the measurement setup.

13.1 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.

NOTE It is good practice to validate the measurement uncertainty by comparing measurement results with an accredited test laboratory.

The interpretation of the measurement uncertainty is different for the manufacturer, who has to guarantee the nominal performance as specified in the data sheet, and the purchaser:

- Manufacturer production test limits: Tolerance *minus* measurement uncertainty.
- Purchaser measurement acceptance limits: Nominal data *plus* measurement uncertainty.

14 Mounting of the DUT

14.1 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.

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) A standard baffle with a plane front surface that is acoustically reflecting as specified in clause A.1.1.

c) A standard measuring enclosure specified in annex A.1.2 (type-A or type-B) generating a defined rear air volume and radiation condition at the front of the transducer.

d) A test cabinet not in accordance with annex A.1.2 (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 [15]). 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.

e) A defined horn, coupler or other kind of wave guide that couples the radiating surface of the transducer to the surrounding sound field.

f) A plane wave tube.

g) The transducer may be operated in free air without baffle, enclosure, etc.

14.2 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.

15 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. Therefore, the DUT shall be preconditioned before measurements are performed by applying a broadband signal at half the maximum input value $\tilde{u}_{ref}$ in accordance with clause 18 for at least one hour.

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.

16 Rated ambient conditions

16.1 Temperature ranges

16.1.1 Performance limited temperature range

16.1.1.1 Conditions to be specified

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

16.1.2 Damage limited temperature range

16.1.2.1 Conditions to be specified

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

16.2 Humidity ranges

16.2.1 Relative humidity range

16.2.1.1 Conditions to be specified

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

16.2.2 Damage limited humidity range

16.2.2.1 Conditions to be specified

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

17 Rated frequency range

17.1 Conditions to be specified

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 with the intended use of the DUT in the final application and may differ from the DUT’s effective frequency range as defined in clause 20.6.

18 Input signal

18.1 Maximum input value

18.1.1 Condition to be specified

The maximum input value $\tilde{u}_{max}$ is the r.m.s. value

$$\tilde{u} = \sqrt{\frac{1}{T_s}\int_0^{T_s} u^2(t)dt} \qquad (14)$$

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 17, shall be stated by the manufacturer.

NOTE 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 $\mathbf{r}_e$. The maximum input value $\tilde{u}_{max}$ applied to the terminal voltage shall 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 behavior 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 shall be determined based on the rated maximum sound pressure level SPL_{max} in accordance with clause 19.2.

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 9.4;

NOTE A DUT intended for full-band audio signals shall 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 program signal in accordance with A.2. 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 9.

– an optional weighting network to obtain the noise signal in accordance with A.2;

– an optional clipping network to realize a defined crest factor;

– an optional power amplifier for passive electro-acoustical DUT;

NOTE The clipped noise at the terminals of the loudspeaker under test shall have a frequency distribution as specified in A.2 and a peak-to-r.m.s. 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 ±0.5 dB in the frequency range 20 Hz to 20 kHz. The high pass filter in the power amplifier may increase crest factor of the tranferred 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 without clipping. The peak voltage capability shall be at least twice the r.m.s. voltage of the test noise.

a) The manufacturer shall define an input test value $\tilde{u}_{\text{test}}$ to achieve the rated maximum input value $\tilde{u}_{\text{max}}$. This 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.

b) The DUT shall be placed in a room of sufficient size (typically > 8 m^3) with climate conditions as specified in A.2.

NOTE The size of the room depends of the space required around the DUT to avoid any additional acoustical loading of the DUT and to ensure stable climate 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 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 clause 20.3.

d) Immediately after the test, the DUT shall be stored under climate 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 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 $\tilde{u}_{\text{test}}$ is assigned to the rated maximum input value $\tilde{u}_{\text{max}}$. The permissible shift of the resonant frequency shall be defined in the data sheet.

NOTE 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.

18.1.3 Indirect measurement based on $SPL_{\max}$

a) To determine the maximum input value $\tilde{u}_{\max}$ based on specified maximum sound-pressure level $SPL_{\max}$, the DUT is placed under normal measurement conditions in accordance with clause 8.2.

b) The DUT is excited by a scaled stimulus $u(t)= \alpha[i]x(t)$ as stated by the manufacturer in accordance with clause 18.1. 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 $p(t, \mathbf{r}_e)$ is measured at the evaluation point $\mathbf{r}_e$ and the resulting sound pressure level $SPL[i]$ for the stimulus with r.m.s. 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

$$\alpha[i+1] = \alpha[i]10^{(SPL_{\max}-SPL[i])/20} \tag{15}$$

and the measurement steps 18.1.2b) to 18.1.2c) are repeated until $SPL[i]$ deviates from $SPL_{\max}$ less than 0.1 dB.

e) The final r.m.s. value $\tilde{u}[i]$ is assigned to the maximum input value $\tilde{u}_{\max}$.

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 broad-band stimulus. As a result, the voltage fed to the DUT in each 1/3 octave frequency band shall be equal to $\tilde{u}/\sqrt{n}$. The. sound-pressure level is given by the formula:

$$SPL[n] = 10\log_{10}\left(\frac{1}{p_0^2}\sum_{k=1}^{n}\tilde{p}_k^2\right) \tag{16}$$

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

18.2 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 $\tilde{u}_{\max}$ and a stated reference value.

NOTE The r.m.s. value 1 Volt shall be used as a reference value if the maximum input will be expressed in dBV. The digital full scale value is used as a reference value if maximum input level will be expressed in dBFS.

19 Sound-pressure output

19.1 Rated maximum sound pressure

19.1.1 Conditions to be specified

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

$$\tilde{p}_{\max}(\mathbf{r}_e) = \left(\frac{1}{T_s}\int_0^{T_s} p^2(t,\mathbf{r}_e)dt\right)^{1/2} \tag{17}$$

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 clause 8.2, 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 clause 18.1.

19.1.2 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 9.4;

NOTE A DUT intended for full-band audio signals shall 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 program signal in accordance with A.2. 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 9.

– an optional weighting network to obtain the noise signal in accordance with A.2;

– an optional clipping network to realize a defined crest factor;

– an optional power amplifier for passive electro-acoustical DUT;

NOTE 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 ±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 A.2 and a peak-to-r.m.s. 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 without clipping. The peak voltage shall be at least twice the voltage of the test noise.

a) The manufacturer shall define a test value $\tilde{p}_{\text{test}}(\mathbf{r}_e)$ to achieve the rated maximum sound pressure $\tilde{p}_{\max}(\mathbf{r}_e)$, in the rated frequency band. This value considers the intended application and behavior 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 10, 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 19.1.1 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 r.m.s. 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 m^3) with climate conditions as specified in A.2.

NOTE The size of the room depends of the space required around the DUT to avoid any additional acoustical loading of the DUT and to ensure stable climate 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 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 and the sound pressure level will fall 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 clause 19.3.

d) Immediately after the test, the DUT shall be stored under climate 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 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 $\tilde{p}_{\max}(\mathbf{r}_e)$. The permissible shift of the resonant frequency shall be defined in the data sheet.

NOTE 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 $\tilde{p}_{\text{test}}(\mathbf{r}_e)$ until the rated maximum sound pressure $\tilde{p}_{\max}(\mathbf{r}_e)$ is determined.

19.1.3 Indirect measurement based on maximum input value

a) The DUT shall be placed in an acoustical environment, in accordance with clause 10, and the r.m.s. value of the input stimulus is adjusted to the rated maximum input value $\tilde{u}_{\max}$ of the stimulus in accordance with clause 18.1.

b) The r.m.s. 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 $\tilde{p}_{\max}(\mathbf{r}_e)$.

19.2 Rated maximum sound-pressure level

19.2.1 Condition to be specified

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 r.m.s. sound pressure value $\tilde{p}_{\max}(\mathbf{r}_e)$, measured in accordance with clause 19.1, and the standard reference sound pressure (20 μPa).

19.3 Short term maximum sound pressure level

19.3.1 Conditions to be specified

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

$$SPL_{\text{short}}(\mathbf{r}_e) = 10\log_{10}\left(\frac{1}{T_{\text{s}}p_0^2}\int_0^{T_s} p^2(t,\mathbf{r}_e)dt\right) \tag{18}$$

of the r.m.s. sound pressure signal $p(t)$ at the specified evaluation point $\mathbf{r}_e$ referred to the reference sound pressure (p_0=20 μPa) using a sequence of gated broadband stimulus comprised of an excitation period of T_{s}=1 s followed by a cooling interval of T_{off}=1 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 program signal in accordance with A.2, 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 s and the cooling interval (T_{off}=1 min) keeps thermal heating and power compression negligible.

19.3.2 Method of measurement

a) 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.

b) The DUT shall be placed in an acoustical environment, in accordance with clause 10. 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 0 by using an excitation period of T_s = 1 s followed by a cooling interval of T_off=1 min.

c) After the adjustment, the sequence of the defined excitation period and cooling interval shall be repeated 60 times.

d) After the test, no signal is supplied to the DUT for a recovery period of 5 min.

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 the stated values in the data sheet, the DUT may be considered to have fulfilled the requirement of this test and the test value $SPL_\mathrm{test}(\mathbf{r}_e)$ is assigned to the rated sound pressure $SPL_\mathrm{short}(\mathbf{r}_e)$. A change in the resonance frequency may occur but the acceptability of this change is subject to negotiation, therefore, it shall be stated.

NOTE 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 $SPL_\mathrm{test}(\mathbf{r}_e)$ until the short term maximum sound pressure level $SPL_\mathrm{short}(\mathbf{r}_e)$ is determined.

19.4 Long term maximum sound pressure level

19.4.1 Conditions to be specified

The long term maximum sound pressure level SPL_long is the level

$$SPL_\mathrm{long}(\mathbf{r}_e) = 10\log_{10}\left(\frac{1}{T_\mathrm{s}p_0^2}\int_0^{T_s} p^2(t,\mathbf{r}_e)dt\right) \tag{19}$$

of the r.m.s. 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 μPa) using a broadband stimulus with an excitation period of T_s=1 min and a cooling interval of T_off=2 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 program signal in accordance with A.2, 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 19.1.2.

19.4.2 Method of measurement

a) The manufacturer defines a test value $SPL_\mathrm{test}(\mathbf{r}_e)$ as a candidate for the long term maximum sound pressure level $SPL_\mathrm{long}(\mathbf{r}_e)$ in the rated frequency band. The test value $SPL_\mathrm{test}(\mathbf{r}_e)$ 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 10. 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 19.4.2 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 $SPL_{\mathrm{test}}(\mathbf{r}_e)$ is assigned to the long term sound pressure level $SPL_{\mathrm{long}}(\mathbf{r}_e)$. A change in the resonance frequency may occur but the acceptability of this change is subject to negotiation, therefore, it shall be stated.

NOTE 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 $SPL_{\mathrm{test}}(\mathbf{r}_e)$ until the long term maximum sound pressure level $SPL_{\mathrm{long}}(\mathbf{r}_e)$ is determined.

19.5 Sound pressure in a stated frequency band

19.5.1 Condition to be specified

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

19.5.2 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 clause 19.2.

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 r.m.s. 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 $\tilde{p}(\mathbf{r})$ shall be measured at the stated measurement point $\mathbf{r}$.

c) Any deviations from the conditions stated in 19.5.2 a) to 19.5.2 d) shall be stated with the results.

19.6 Sound-pressure level in a stated frequency band

19.6.1 Condition to be specified

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 clause 19.5 and the standard reference sound pressure (20 µPa).

19.7 Mean sound-pressure in a stated frequency range

19.7.1 Condition to be specified

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

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

19.7.2 Method of Measurement

a) The measurement shall be made in accordance with clause 19.5.

b) The sound pressure signal $\tilde{p}(\mathbf{r})$ measured at the measurement point $\mathbf{r}$ is analysed by using a set of filters with $1/n^{\mathrm{th}}$ octave bandwidth generating the r.m.s. value of the sound pressure $\tilde{p}_k$ of each sub-band with k=1,...,K.

c) The mean sound-pressure $\tilde{p}_m$ in the stated frequency range is determined by the formula:

$$\tilde{p}_m = \left[\frac{1}{K}\sum_{k=1}^{K}\left(\tilde{p}_k\right)^2\right]^{1/2} \qquad (20)$$

19.8 Mean sound-pressure level in a stated frequency range

19.8.1 Condition to be specified

Expressed in decibels, twenty times the logarithm of the ratio between p_m in accordance with clause 19.7 and the standard reference sound-pressure (20 μPa).

20 Frequency response of the fundamental component

20.1 Transfer function

20.1.1 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 $u(t) = \alpha\tilde{u}_{\max}x(t)$ scaled by the maximum input value $\tilde{u}_{\max}$ and scaling factor α. The properties of the stimulus $u(t)$, the measurement time T_s and either the r.m.s. value $\tilde{u} = \alpha\tilde{u}_{\max}$ or the scaling factor α shall be stated.

NOTE The transfer function measured with a high amplitude stimulus may 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.

20.1.2 Method of measurements

a) To assess the nonlinear behaviour of the DUT, the electro-acoustical system is excited by the stimulus $u(t) = \alpha\tilde{u}_{\max}x(t)$ by using the maximum input value $\tilde{u}_{\max}$ and a stated scaling factor α. It is recommended to use a sinusoidal chirp signal or multi-tone complex signal in accordance with clause 9.1 and 9.4.

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 The logarithmic frequency-time mapping of the chirp signal can be used to separate the fundamental and harmonic components, see Farina [10].

d) The complex transfer function is calculated by

$$\underline{H}(f,\mathbf{r}) = \frac{\underline{P}_{\text{fund}}(f,\mathbf{r})}{\underline{U}(f)} \qquad (21)$$

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

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 $F^{-1}\{\ \}$.

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 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

$$\varphi_{\mathrm{res}}(f,\mathbf{r}) = \varphi(f,\mathbf{r}) - 360° f \tau_{\mathrm{mean}}(\mathbf{r}) \tag{22}$$

represents the minimum-phase + all pass behavior of the DUT separated from the phase corresponding with the mean group delay [22].

h) The unwrapped total phase response

$$\varphi_{\mathrm{un}}(f,\mathbf{r}) = \varphi_{\mathrm{res}}(f,\mathbf{r}) + 360° f \tau_{\mathrm{mean}}(\mathbf{r}) \tag{23}$$

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

NOTE Unwrapping is a necessary but ambiguous, noise-sensitive and error-prone process for frequency-discrete phase data (at least in acoustics). The frequency resolution must be high enough so that the phase difference between two discrete frequencies does not exceed ±90 degree.

i) The group delay response

$$\tau(f) = -\frac{1}{360°}\frac{d\varphi_{un}(f)}{df} \tag{24}$$

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

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 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 Spectral averaging generates blurred 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 Pascal per Volt).

20.2 SPL frequency response

20.2.1 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 **r** using a narrow-band signal with bandwidth *B* and center frequency f. The input signal $u(t) = \alpha \tilde{u}_{\max} x(t)$ has a constant r.m.s. value $\tilde{u} = \alpha \tilde{u}_{\max}$ 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 r.m.s. value $\tilde{u} = \alpha \tilde{u}_{\max}$ 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 to appear better than they are.

20.2.2 Method of measurement

a) The loudspeaker shall be operated under normal measuring conditions.

b) If the maximum input value $\tilde{u}_{\mathrm{max}}$ is not stated by the manufacturer, this value can be determined by using the rated maximum sound pressure level SPL_{max} in accordance with clause 19.6.

c) For each frequency f in the rated frequency range $f_{\mathrm{l}} \leq f \leq f_{\mathrm{u}}$ 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 r.m.s. value $\alpha\tilde{u}_{\mathrm{max}}$ within the measurement time $T_{\mathrm{s.}}$ The sound pressure $p(\mathrm{t,r})$ is measured at the measurement point **r** and filtered by band pass filter of defined bandwidth *B* and center 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 clause 20.1. The $SPL(f)$ in dB is calculated by

$$SPL(f,\mathbf{r}) = 20\log\left(\frac{|H(f,\mathbf{r})|\alpha\tilde{u}_{\mathrm{max}}}{p_0}\right) \qquad (25)$$

using the reference sound pressure (p_0 = 20 μPa).

d) 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.

20.3 Short-term amplitude compression of the fundamental component

20.3.1 Conditions to be specified

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

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

20.3.2 Method of measurement

a) The DUT is adjusted to ambient temperature.

b) The transfer function $\underline{H}(f,\mathbf{r},u_{\mathrm{max}})$ 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 clause 20.1.

c) The linear transfer function $\underline{H}_{\mathrm{lin}}(f,\mathbf{r},\alpha u_{\mathrm{max}})$r is measured by using a specified broad-band stimulus with an r.m.s. value αu_{max} within the measurement time T_{s}=1 s in accordance with clause 20.1.

d) The response of the long term amplitude compression shall be calculated using:

$$C_{\mathrm{short}}(f) = 20\log\left(\left|\underline{H}_{\mathrm{lin}}(f,\mathbf{r},\alpha u_{\mathrm{max}})\right|\right) - 20\log\left(\left|\underline{H}(f,\mathbf{r},u_{\mathrm{max}})\right|\right) \qquad (26)$$

20.4 Long-term amplitude compression of the fundamental component

20.4.1 Conditions to be specified

The long-term amplitude compression $C_{\mathrm{long}}(f)$ is the level difference between the magnitude frequency response of the transfer function $\underline{H}(f,\mathbf{r},\tilde{u}_{\max})$ 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_{pre}=1 min and the linear transfer function $\underline{H}_{\mathrm{lin}}(f,\mathbf{r},\alpha\tilde{u}_{\max})$ measured with the same stimulus at attenuated maximum input value $\alpha\tilde{u}_{\max}$.

NOTE The long-term amplitude compression $C_{\mathrm{long}}(f)$ 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.

20.4.2 Method of measurement

a) The DUT is adjusted to ambient temperature.

b) After exciting the DUT by a broad-band stimulus $u(t)$ with maximum input value $\widetilde{u}_{\max}$ for a pre-excitation time T_{pre}=1 min, the transfer function $\underline{H}(f,\mathbf{r},\tilde{u}_{\max})$ is measured by using the same stimulus for the measurement time T_{s}=1 s in accordance with clause 20.1.

c) The linear transfer function $\underline{H}_{\mathrm{lin}}(f,\mathbf{r},u_{\max})$ r is measured by using a specified broad-band stimulus with an r.m.s. value $\alpha u_{\max}$ within the measurement time T_{s}=1 s in accordance with clause 20.1.

d) The response of the long term amplitude compression shall be calculated by:

$$C_{\mathrm{long}}(f) = 20\log\left(\left|\underline{H}_{\mathrm{lin}}(f,\mathbf{r},\alpha\tilde{u}_{\max})\right|\right) - 20\log\left(\left|\underline{H}(f,\mathbf{r},\tilde{u}_{\max})\right|\right) \qquad (27)$$

20.5 Measurement corrections at low frequencies

Whenever the low-frequency absorption characteristic of an anechoic room causes a deviation from free-field conditions, such that an accurate measurement of free-field response is not possible down to the lower limit of the rated frequency range in accordance with 2.2, the low-frequency measurement results may be corrected as follows:

The loudspeaker under test shall be removed from the room and replaced by a calibrated reference loudspeaker located such that its reference point $\mathbf{r}_{ref}$ and reference axis take the positions previously occupied by those of the loudspeaker under test.

The reference loudspeaker shall have a directivity index *DI* in accordance with clause 21.2.4 which deviates from the directivity index *DI* of the 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.

It is necessary to determine the frequency response of the reference loudspeaker accurately. 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.

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.

The frequency response of the reference loudspeaker shall be measured using the same equipment and technique as for the loudspeaker under test.

Over the low-frequency range where the frequency response measured for the reference loudspeaker deviates from its known calibrated free-field response, the difference between the calibrated and the measured responses shall be used to correct the measured response of the loudspeaker under test and this shall be stated with the reported results.

20.6 Effective frequency range

20.6.1 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 20.2 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.

20.6.2 Method of Measurement

a) The frequency response $SPL(f,\mathrm{r})$ shall be measured in the rated frequency range according to clause 17 at resolution corresponding to 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 clause 19.7.

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}.

20.7 Internal latency

20.7.1 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 $\mathbf{r}_{ref}$ to the measurement point $\mathbf{r}$.

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 $\mathbf{r}_{ref}$ which may be not identical with the acoustical center depending on the frequency. Thus, the internal latency is a rough estimate of the time delay generated by the electronic components in the loudspeaker.

20.7.2 Methods of measurement

a) The mean group delay $\tau_{\mathrm{mean}}(\mathbf{r})$ between input signal $u(t)$ and sound pressure signal $p(t,\mathbf{r})$ at stated measurement point $\mathbf{r}$ shall be measured in accordance with 20.1.2 clause 0.

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

$$\tau_{\mathrm{lat}} = \tau_{\mathrm{mean}} - \frac{\left|\mathbf{r} - \mathbf{r}_{ref}\right|}{c} \tag{28}$$

where c is the speed of sound.

21 Directional characteristics

The direct sound radiated from the device under test is described by directional characteristics 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.

21.1 Direct sound field in 3D space

21.1.1 Directional transfer function

21.1.1.1 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 10.

21.1.1.2 Methods of measurement

a) The transfer function $\underline{H}(f,r,\theta,\phi)$ shall be measured under free field or simulated free field conditions in accordance with clause 10 unless the influence of the acoustical environment (e.g. limited size of the baffle) cannot be separated from the DUT.

b) 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.

c) The transfer function $\underline{H}(f,r,\theta,\phi)$ is measured in accordance with clause 20.1 by using an attenuated stimulus (attenuation factor α=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 clause 11.2.1.

21.1.2 Extrapolated far-field data

21.1.2.1 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 ($\phi_2,\ \theta_2$) 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 ($\phi_1=\theta_2,\ \theta_1=\theta_2$).

21.1.2.2 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 10 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

$$\underline{H}(f,r_2,\theta_2,\phi_2) = \underline{H}(f,r_1,\theta_1,\phi_1)\frac{r_1}{r_2}e^{-jk(r_2-r_1)} \tag{29}$$

21.1.3 Parameters of the holographic sound field expansion

21.1.3.1 Conditions to be specified

The coefficients $\mathbf{C}(f)$, 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

$$\underline{H}(f,\mathbf{r}) = \mathbf{C}(f)\mathbf{B}(f,\mathbf{r}) \tag{30}$$

between the input signal $u(t)$ and the sound pressure output $p(t,\mathbf{r})$ at measurement point $\mathbf{r}$ at a distance $r=|\ \mathbf{r} - \mathbf{r}_{ref}\ |$ from the reference point $\mathbf{r}_{ref}$ 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 A.3.

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.

21.1.3.2 Method of measurements

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

NOTE A simulated free field condition, in accordance with clauses 10.3 and 10.4 can be generated by performing an additional scan on a second scanning surface S_2 placed between the surface S_D of the DUT and the first scanning surface S_1 as illustrated in Figure 3. Wave separation techniques [6] 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 $i=1,...,M$ are measured on the scanning surfaces S_k with $k=1,2,...$ in accordance with clause 20.1.

c) The expansion coefficients $\mathbf{C}(f)$ 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 $i=1,...,M$. 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 $\mathbf{r}_{ref}$ surrounding the scanning surface S_1.

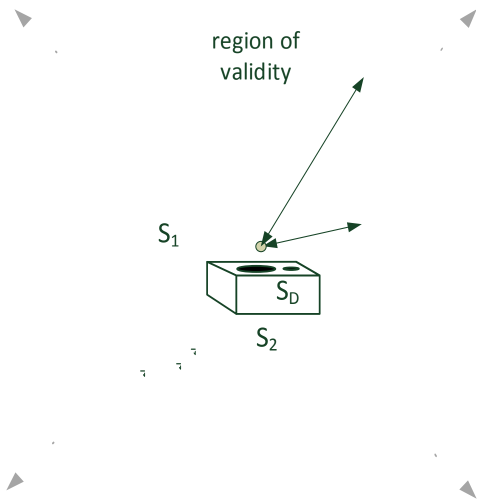


Figure 3 – Valid region of expansion of the sound pressure $p(\mathbf{r})$ at the observation point $\mathbf{r}$ at the distance $\mathbf{r}>a$

21.1.4 Extrapolated near-field data

21.1.4.1 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 $r=|\ \mathbf{r} - \mathbf{r}_{ref}\ |$ from the reference point $\mathbf{r}_{ref}$ which is larger than the validity radius a can be calculated based on the parameters of the holographic sound field expansion.

21.1.4.2 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 clause 20.1. The coefficients $\mathbf{C}(f)$, the order $N(f)$ of the expansion and the validity radius a shall be stated.

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

21.2 Directional far field characteristics

21.2.1 Directional Factor

21.2.1.1 Conditions to be specified

The directional factor

$$\underline{\Gamma}(f,\theta,\phi) == \frac{\underline{P}(f,r,\theta,\phi)}{\underline{P}(f,r,\theta_r,\phi_r)} = \frac{\underline{H}(f,r,\theta,\phi)}{\underline{H}(f,r,\theta_r,\phi_r)} \tag{31}$$

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 $\phi_{\mathrm{r}} = 0$ and $\theta_{\mathrm{r}} = 0$ corresponding to the normal vector $\mathbf{n}_{ref}$ and the orientation vector $\mathbf{o}_{ref}$ in accordance with clause 11.

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

$$|\ \underline{H}(f,r,\theta_{\max},\phi_{\max})| = \max_{\phi,\theta}(|\ \underline{H}(f,r,\theta,\phi)|) \tag{32}$$

In this case the modulus of the directional factor $|\underline{\Gamma}(f,r,\theta,\phi)| \le 1$ would stay below one but the reference angles $\phi_{\mathrm{r}}(f)$ and $\theta_{\mathrm{r}}(f)$ would vary in general with frequency f.

21.2.1.2 Methods of measurement

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

1) In accordance with clause 21.1.1, 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,..., M in the far field of the source at a fixed distance r with defined angular resolution.

2) The coefficients $\mathbf{C}(f)$ of the wave expansion are determined in accordance with clause 21.1.3. 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 in accordance with A.3.2.

c) The following directional response pattern displays shall be chosen:

d) One 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.

NOTE 1 In accordance with AES-5id-1997 (1998) Room acoustics and sound reinforcement systems –Loudspeaker modelling and measurement – Frequency and angular resolution for measuring, presenting and predicting loudspeaker polar data.

NOTE 2 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. 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.

NOTE 3 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 A.2.

NOTE 4 Care shall be taken that the level in the direction of the reference angles corresponds to the zero level of the polar diagram.

21.2.2 Beam pattern

21.2.2.1 Conditions to be specified

Expressed in decibels, the beam pattern

$$b(f,\theta,\phi) = 20\log\left|\underline{\Gamma}(f,\theta,\phi)\right| \tag{33}$$

shall be specified as twenty times the logarithm of the ratio between the directional factor $\underline{\Gamma}(f,\theta,\phi)$ measured in accordance with clause 21.2.1 and the standard reference sound-pressure (20 μPa).

21.2.3 Directivity function

21.2.3.1 Conditions to be specified

The directivity function is defined as

$$Q(f) = \frac{4\pi}{\int\limits_{\Omega} \left|\underline{\Gamma}(f,\theta,\phi)\right|^2 d\Omega} \tag{34}$$

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

21.2.3.2 Methods of measurement

a) The directivity factor $\underline{\Gamma}(f,\theta,\phi)$ is measured in accordance with clause 21.2.1 with defined angular resolution. Alternatively, the coefficients $\mathbf{C}(f)$ of the wave expansion can be determined by scanning the sound pressure field on a scanning surface S_k in accordance with clause 21.1.3.

b) The directivity may be calculated based on the directional factor $\underline{\Gamma}(f,\theta,\phi)$ or the coefficients $\mathbf{C}(f)$ of the wave expansion in accordance with A.3.3.

NOTE 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 function can be calculated by

$$Q(f) = \frac{360°}{\pi \sum_{n=1}^{180°/\Delta\theta} \left|\underline{\Gamma}(f,\theta_n)\right|^2 \sin\theta_n \ \Delta\theta} \tag{35}$$

where $\Delta\theta$ is separation in degrees of the successive points around the sound source, $180°/\Delta\theta$ 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 Beranek [2] , p. 165).

21.2.4 Directivity index

21.2.4.1 Conditions to be specified

Expressed in decibels, the directivity index

$$DI(f) = 10\log_{10} Q(f) \tag{36}$$

shall be specified as ten times the logarithm of the ratio between the directivity function $Q(f)$ measured in accordance with clause 21.2.3.

21.2.4.2 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 $\underline{\Gamma}(f,\theta,\phi)$ shall be measured in accordance with clause 21.2.1 at sufficient angular resolution. The directivity index $DI(f)$ is calculated based on measured directivity $Q(f)$ in accordance with clause 21.2.3.

b) Calculation based on coefficients $\mathbf{C}(f)$ of the wave expansion:

The coefficients $\mathbf{C}(f)$ of the wave expansion are determined in accordance with 21.1.3. and the directivity index $DI(f)$ is calculated based on measured directivity $Q(f)$ in accordance with clause 21.2.3.

c) Measurements under free field and diffuse conditions:

1) In accordance with 20.2, the sound pressure level response $SPL_{\mathrm{ax}}(f)$ shall be measured under free-field conditions on the reference axis at a distance of 1 m using an attenuated broadband stimulus (attenuation factor α=0.1).

2) The sound pressure level $SPL_{\mathrm{dif}}(f)$ 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 [21]:

$$DI(f) = SPL_{\mathrm{ax}}(f) - SPL_{\mathrm{dif}}(f) + 10\lg\left(T(f)\right) - 10\lg\left(V(f)\right) + 25 \tag{37}$$

where

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

$SPL_{\mathrm{dif}}(f)$ is the sound pressure level in dB measured under diffuse field conditions,
$T(f)$ is the reverberation time of the reverberation room in seconds,
$V(f)$ is the reverberation room volume in cubic meters.

21.3 Acoustic Output Power

21.3.1 Conditions to be specified

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

21.3.2 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

$$\Pi(f) = \frac{r^2}{\rho c}\int_{\Omega} \tilde{p}^2(f,r,\theta,\phi)d\Omega \qquad (38)$$

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

$$\Pi(f) = \frac{4\pi r^2}{\rho c}\overline{\tilde{p}^2(f)} = 0{,}031 r^2 \overline{\tilde{p}^2(f)} \qquad (39)$$

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

r is the sphere radius in meters,

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

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:

$$\Pi(f) = \frac{2\pi r^2}{\rho c}\overline{\tilde{p}^2(f)} = 0{,}016 r^2 \overline{\tilde{p}^2(f)} \qquad (40)$$

NOTE In the case of full-space free-field conditions, the square of the r.m.s. 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

$$\Pi(f) = \tilde{u}^2(f)\frac{r^2}{\rho c}\int_{\Omega} \left|\underline{H}(f,r,\theta,\phi)\right|^2 d\Omega \qquad (41)$$

using the r.m.s. value $\tilde{u}(f)$ with a given frequency band and a center frequency f.

b) Calculation based on far-field characteristics:

The acoustic output power may be calculated by the following equation

$$\Pi(f)=\frac{\tilde{p}^2(f,r,\theta_r,\phi_r)}{\rho c}\int_{\Omega}\left|\underline{\Gamma}(f,\theta,\phi)\right|^2 d\Omega=\frac{\tilde{p}^2(f,r,\theta_r,\phi_r)4\pi r^2}{Q(f)\rho c} \tag{42}$$

using the r.m.s. 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 function $Q(f)$ measured in accordance with clause 21.2.1 or 21.2.3, 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 A.3.4.

d) Measurement of acoustic power under diffuse field conditions:

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

$$\Pi(f)=\frac{V}{T_{60}(f)}\tilde{p}(f)^2 10^{-4} \tag{43}$$

where

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

V is the reverberation room volume in cubic meters,

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

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

NOTE 1 The filtering may take place either in the loudspeaker chain or in both the loudspeaker and the microphone chains.

NOTE 2 An alternative method for measuring the sound power of loudspeakers, using a sound power source, is described in ISO 3743.

21.4 Sound power level

21.4.1 Conditions to be specified

Expressed in decibels, the sound power level

$$L_{\Pi}(f)=10\log_{10}\left(\frac{\Pi(f)}{\Pi_0}\right) \tag{44}$$

shall be specified as ten times the logarithm of the ratio between the acoustic output power $\Pi(f)$ measured in accordance with clause 21.3 and the reference power $\Pi_0 = 10^{-12}$ W.

21.5 Mean acoustic output power in a frequency band

21.5.1 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.

21.5.2 Method of measurement

a) The measurement shall be made in accordance with 21.3.2.

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.

21.6 Radiation Angle

21.6.1 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.

21.6.2 Method of measurement

a) The radiation angle shall be deduced from the beam pattern measured in accordance with 21.2.2 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.

21.7 Coverage angle or angles

21.7.1 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 beam pattern, in accordance with 21.2.2 centered 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.

21.7.2 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 centered 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 center 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 center 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.

22 Harmonic distortion

22.1 Introduction

A general explanation of harmonic distortion can be found in IEC 60268-2.

22.2 N^{th}-order harmonic component

22.2.1 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

$u(t) = \alpha\tilde{u}_{\max}x(t)$ with frequency f. The properties of the stimulus $u(t)$, the measurement time T_s and either the r.m.s. value $\tilde{u} = \alpha\tilde{u}_{\max}$ or the scaling factor α shall be stated.

22.2.2 Method of measurement

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

NOTE A sinusoidal chirp is a preferred stimulus, because the step-by-step method using steady-state tones may cause important information to be missed.

b) The sound pressure signal $p(t)$ is measured at the measurement point $\mathbf{r}$ under normal measurement conditions in accordance with 8.2. If not stated otherwise, the measurement point $\mathbf{r}$ is the evaluation point $\mathbf{r}_e$.

c) The r.m.s. value of the fundamental component $\tilde{p}_f(f)$ and the r.m.s. 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 $p(t)$.

d) 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 r.m.s. value $\tilde{p}_f(f)$ and the standard reference sound pressure (20 μPa).

e) 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 r.m.s. 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 r.m.s. 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 r.m.s. value or the scaling factor α and the sound-pressure level at the measurement point $\mathbf{r}$;
- 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 $\mathbf{r}_e$ and the conditions of the measurement (free-field or half-space free-field);
- Bandwidth of spectral smoothing if applied to the frequency response.

22.3 Total harmonic components

22.3.1 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 r.m.s. value $\tilde{u} = \alpha\tilde{u}_{\max}$ or the scaling factor α shall be stated.

22.3.2 Method of measurement

a) The r.m.s. value $\tilde{p}_{nf}(f)$ of the nth-order harmonic components in the measured sound-pressure signal $p(t)$ shall be measured in accordance with 22.2.2 clause a) and b).

b) The r.m.s. value of the total harmonic component fundamental component $\tilde{p}_{TH}(f)$ and the shall be determined by the formula:

$$\widetilde{p}_{TH}(f) = \sqrt{\sum_{n=2}^{N} \widetilde{p}^{2}{}_{nf}(f)} \tag{45}$$

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 r.m.s. value $\widetilde{p}_{TH}(f)$ 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 level individual nth-order harmonic components $L_{nH}(f)$.

22.4 Total harmonic distortion

22.4.1 Conditions to be specified

The total harmonic distortion in terms of total sound pressure $\widetilde{p}$ 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 r.m.s. value $\tilde{u} = \alpha \tilde{u}_{\max}$ or the scaling factor α shall be stated.

22.4.2 Method of measurement

a) The r.m.s. value $\tilde{p}_{nf}(f)$ of the nth-order harmonic components in the measured sound pressure signal $p(t)$ shall be measured in accordance with clause 22.2.2.

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

NOTE The r.m.s. value $\widetilde{p}_f(f)$ of the fundamental component or the mean sound pressure $\tilde{p}_m$ in a stated frequency range are useful alternatives for the reference value $\widetilde{p}_{ref}(f)$.

c) The total harmonic distortion can be determined by the formula:

in percentage:

$$THD(f) = \frac{\sqrt{\sum_{n=2}^{N} \widetilde{p}^{2}{}_{nf}(f)}}{\widetilde{p}_{ref}(f)} 100\% \tag{46}$$

in decibels:

$$L_{THD}(f) = 20\lg\left(\frac{THD}{100\%}\right) \tag{47}$$

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

d) 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 r.m.s. 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 $\mathbf{r}_e$ and the conditions of the measurement (free-field or half-space free-field);
- Bandwidth of spectral smoothing if applied to the frequency response.

22.5 Maximum sound pressure level limited by total harmonic distortion

22.5.1 Conditions to be specified

The maximum sound pressure level $SPL_{THD}(f,\mathbf{r})$ at the measurement point $\mathbf{r}$ for a sinusoidal stimulus at frequency f generating a defined value of THD_{lim} in the sound pressure output.

22.5.2 Method of measurement

a) A series of sinusoidal input signals $u(t) = \alpha\tilde{u}_{\max}x(t)$ with increasing frequency and sufficient spectral resolution, up to 10 kHz, and increasing r.m.s. input $\tilde{u} = \alpha\tilde{u}_{\max}$ in 1 dB steps by using an appropriate scaling factor $\alpha_{\min} \le \alpha \le \alpha_{\max}$ shall be supplied to the DUT.

NOTE The minimum and maximum scaling factors $\alpha_{\min}$ and $\alpha_{\max}$, respectively, depend on the value THD_{lim} (typical values are $\alpha_{\min}$=0.1 and $\alpha_{\max}$=1).

b) The sound pressure signal $p(t)$ is measured at the stated measurement point $\mathbf{r}$ under normal measurement conditions in accordance with 8.2.

c) Based on measured sound pressure signal $p(t)$, the total harmonic distortion $THD(f,\alpha)$ and sound pressure level $SPL(f,\alpha)$ are measured as a function of frequency f and scaling factor α.

d) The maximum sound pressure level $SPL_{THD}(f, \mathbf{r})$ at a defined threshold value THD_{lim} is determined by linear interpolation between the measured data points in $THD(f,\alpha)$ and $SPL(f,\alpha)$.

NOTE The linear interpolation requires sufficient resolution of the amplitude variation (1 dB steps).

22.6 Nth-order equivalent input harmonic distortion component

22.6.1 Conditions to be specified

The equivalent n^{th}-order harmonic input distortion $EIHD_n(f)$ in the input signal $u(t)$ corresponds to the harmonic distortion measured in the sound pressure output $p(t,\mathbf{r}_i)$ at the measurement point $\mathbf{r}$ while exciting the DUT under normal measurement conditions with a sinusoidal stimulus $u(t) = \alpha\tilde{u}_{\max}x(t)$ at frequency f. The properties of the stimulus $u(t)$, the measurement time T_{s} and either the r.m.s. value $\tilde{u} = \alpha\tilde{u}_{\max}$ or the scaling factor α shall be stated.

NOTE If the source of the harmonic distortion (e.g. force factor nonlinearity) is located close to the input of the transducer, the ETHD(f, $\mathbf{r}_i$)=const. will be independent of the position of the measurement point $\mathbf{r}_i$ with i=1,…,N. This simplifies the interpretation of distortion generated by motor and suspension nonlinearities as discussed in A.4 and Klippel [14].

22.6.2 Method of measurement

a) A series of sinusoidal input stimuli with increasing frequencies, up to 5 000 Hz, shall be supplied to the loudspeaker. The chosen r.m.s. value $\tilde{u}$ of the input signal $u(t) = \alpha\tilde{u}_{\max}x(t)$ shall be those which are the most relevant for the intended use but shall not exceed the maximum input value $\tilde{u}_{\max}$.

NOTE A sinusoidal chirp is a preferred stimulus, because the step-by-step method using steady-state tones may cause important information to be missed.

b) The sound pressure signal $p(t)$ is measured at the measurement point $\mathbf{r}$ under normal measurement conditions in accordance with 8.2. If not stated otherwise, the measurement point $\mathbf{r}$ is the evaluation point $\mathbf{r}_e$.

c) The linear transfer function $H(f,\mathbf{r})$ between input $u(t)$ and measurement point $\mathbf{r}$ shall be measured in the small signal domain accordance with clause 20.1.

d) The sound pressure signal $p(t,\mathbf{r})$ measured in the large signal domain shall be filtered with the inverse transfer function $H(f,\mathbf{r})^{-1}$ to transfer the measured sound pressure into a virtual input signal $u'(t,\mathbf{r})$.

NOTE The virtual input signal $u'(t,\mathbf{r})$ is not identical with the input signal $u(t)$ at high amplitudes but reveals harmonic components and other nonlinear and thermal effects in the fundamental component.

e) The r.m.s. value of the fundamental component $\tilde{u}'_f(f)$ and the r.m.s. values $\tilde{u}'_{nf}(f)$ of n^{th}-order equivalent input harmonic components shall with n > 1 be determined by applying spectral analysis to virtual input signal $u'(t,\mathbf{r})$.

f) Expressed in decibels the amplitude compression of the fundamental component shall be determined by the formula

$$C_f(f) = -20\lg\left(\frac{\tilde{u}'_f(f)}{\tilde{u}}\right) \tag{48}$$

using the r.m.s. value of the input stimulus.

NOTE The amplitude variation $C_f(f)$ corresponds with short-term amplitude compression $C_{\text{short}}(f)$ and long-term amplitude compression $C_{\text{long}}(f)$ amplitude compression depending on the measurement time in accordance with clauses 20.3 and 20.4, respectively.

g) The n^{th}-order equivalent input harmonic distortion shall be determined by the formula:

in percentage:

$$EIHD_n(f) = \frac{\tilde{u}'_{nf}(f)}{\tilde{u}}100\% \tag{49}$$

in decibels:

$$L_{nEIHD}(f) = 20\lg\left(\frac{EIHD_n(f)}{100\%}\right) \tag{50}$$

h) 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 r.m.s. value or the scaling factor α and the sound-pressure level at the measurement point $\mathbf{r}$;
- 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 $\mathbf{r}_e$ and the conditions of the measurement (free-field or half-space free-field);
- Bandwidth of spectral smoothing if applied to the frequency response.

22.7 Equivalent input total harmonic distortion

22.7.1 Conditions to be specified

The equivalent total harmonic input distortion (EITHD) in the input signal $u(t)$ corresponds to the harmonic distortion in the sound pressure output $p(t,\mathbf{r})$ at the measurement point $\mathbf{r}$ while exciting the DUT under normal measurement conditions 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 r.m.s. value $\tilde{u}=\alpha\tilde{u}_{\max}$ or the scaling factor α shall be stated.

22.7.2 Method of measurement

a) The r.m.s. values $\tilde{u}'_{nf}(f)$ of n^{th}-order equivalent harmonic components with n > 1 in the virtual input signal $u'(t,\mathbf{r})$ shall be measured in accordance with 22.6.2 clauses a) to e).

b) The equivalent total harmonic input distortion shall be determined by the formula:

in percentage:

$$EITHD(f)=\frac{\sqrt{\sum_{n=2}^{N}\tilde{u}'^{2}_{nf}(f)}}{\tilde{u}'(f)}100\% \qquad (51)$$

in decibels:

$$L_{EITHD}(f)=20\lg\left(\frac{EITHD(f)}{100\%}\right) \qquad (52)$$

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

c) The results of the measurement shall be presented in accordance with 22.6.2 clause h).

23 Two-tone intermodulation distortion

23.1 Variation of excitation frequencies

The device under test shall be excited by a two-tone stimulus $u(t)=\alpha\tilde{u}_{\max}x_t(t)$ according to clause 9.3 and the intermodulation distortion components generated in the sound pressure output shall be measured as a function of the frequencies f_1 or f_2, where frequency f_1 being lower than frequency f_2.

NOTE It is recommend to keep the frequency f_1 significantly lower than the frequency f_2 that the intermodulation components provide unique information which cannot be found in the harmonics. (See comments to Figure 11 in annex A.5).

One of the following techniques shall be used (as illustrated in Figure 11 in Annex A.5):

a) Variation of the high frequency tone f_2 while keeping the low frequency tone f_1 at a constant frequency (e.g. to generate high voice coil displacement);

b) Variation of the low frequency tone f_1 while keeping the high frequency tone f_2 at a constant frequency (to investigate the influence of displacement and current);

c) Variation of both frequencies f_1 and f_2 while keeping the frequency ratio $f_2/f_1=\mathrm{const.}$;

d) Variation of both frequencies f_1 and f_2 while keeping the frequency difference $f_2-f_1=\mathrm{const.}$

The method used shall be stated with the results.

23.2 Intermodulation distortion

23.2.1 Conditions to be specified

The intermodulation distortion measured in the sound pressure output $p(t,\mathbf{r})$ at the measurement point $\mathbf{r}$ while exciting the DUT under normal measurement conditions shall be specified as the ratio between the arithmetic sum of the r.m.s. values of the nth-order intermodulation component and the r.m.s. value of the fundamental component at excitation frequency f_2.

The properties of the stimulus $u(t)$ including the frequencies f_1 and f_2, the total measurement time T_s and either the r.m.s. value $\tilde{u}=\alpha\tilde{u}_{\max}$ or the scaling factor α shall be stated.

23.2.2 Method of measurement

a) The DUT shall be excited by the two-tone stimulus $u(t)=\alpha\tilde{u}_{\max}x_t(t)$ in accordance with clause 9.3 and clause 18.2. The measurement shall be performed after exciting the DUT during a pre-excitation time T_pre ensuring a steady-state condition of the DUT.

b) The sound pressure signal $p(t,\mathbf{r})$ is measured at the measurement point $\mathbf{r}$ under normal measurement conditions in accordance with 8.2. If not stated otherwise, the measurement point $\mathbf{r}$ is the evaluation point $\mathbf{r}_e$.

c) A spectral analysis of the sound pressure signal $p(t,\mathbf{r})$ shall be performed to determine the r.m.s. value of the n^{th}-order intermodulation distortion components $\tilde{p}(f_2 \pm (n-1)f_1)$ at sum and difference frequencies and the r.m.s. value of the fundamental component $\tilde{p}(f_2)$ at the excitation frequency f_2.

d) The 2nd-order intermodulation distortion shall be calculated by the following formula:

in percentage:

$$IMD_2(f_1,f_2) = \frac{\tilde{p}(f_2 - f_1) + \tilde{p}(f_2 + f_1)}{\tilde{p}(f_2)} 100\% \tag{53}$$

in decibels:

$$L_{2IMD}(f_1,f_2) = 20\lg\left(\frac{IMD_2(f_1,f_2)}{100\%}\right) \tag{54}$$

The 3rd-order intermodulation distortion shall be calculated by the following formula:

in percentage:

$$IMD_3(f_1,f_2) = \frac{\tilde{p}(f_2 - 2f_1) + \tilde{p}(f_2 + 2f_1)}{\tilde{p}(f_2)} 100\% \tag{55}$$

in decibels:

$$L_{3IMD}(f_1,f_2) = 20\lg\left(\frac{IMD_3(f_1,f_2)}{100\%}\right) \tag{56}$$

e) The total intermodulation distortion shall be calculated by the following formula considering only intermodulation components up to 3rd-order:

in percentage:

$$IMD_T(f_1,f_2) = \frac{\sum_{k=1}^{2} \tilde{p}(f_2 - kf_1) + \tilde{p}(f_2 + kf_1)}{\tilde{p}(f_2)} 100\% \tag{57}$$

in decibels:

$$L_{TIMD}(f_1,f_2) = 20\lg\left(\frac{IMD_T(f_1,f_2)}{100\%}\right) \tag{58}$$

NOTE The normalization of the intermodulation components to the r.m.s. value $\tilde{p}(f_2)$ of the fundamental at frequency f_2 (consistent with standard IEC 60268-2) reveals the modulation of the high frequency carrier while suppressing the influence the linear transfer response from the loudspeaker nonlinearity (e.g. force factor) to the measurement point.

23.3 Amplitude modulation distortion

23.3.1 Conditions to be specified

The amplitude modulation distortion measured in the sound pressure output $p(t,r)$ at the measurement point $\mathbf{r}$ while exciting the DUT under normal measurement conditions shall be specified as the ratio between the AC signal of the signal envelope and the DC signal of the envelope at excitation frequency f_2.

The properties of the stimulus $u(t)$ including the frequencies f_1 and f_2, the total measurement time T_s and either the r.m.s. value $\tilde{u} = \alpha \tilde{u}_{max}$ or the scaling factor α shall be stated.

NOTE Some loudspeaker nonlinearities (e.g. force factor Bl(x)) cause amplitude modulation distortions which are more audible than phase or frequency modulation distortion caused by other nonlinearities (e.g. Doppler effect).

23.3.2 Method of measurement

a) The DUT shall be excited by the two-tone stimulus $u(t) = \alpha \tilde{u}_{max} x_t(t)$ in accordance with clause 9.3 and clause 18.2. The measurement shall be performed after exciting the DUT during a pre-excitation time T_{pre} ensuring a steady-state condition of the DUT.

b) The sound pressure signal $p(t,\mathbf{r})$ is measured at the measurement point $\mathbf{r}$ under normal measurement conditions in accordance with 8.2. If not stated otherwise, the measurement point $\mathbf{r}$ is the evaluation point $\mathbf{r}_e$.

c) A spectral analysis of the sound pressure signal $p(t,\mathbf{r})$ shall be performed to separate difference and sum intermodulation distortion components $\underline{P}(f_2 \pm (n-1) f_1)$ and the fundamental component $\underline{P}(f_2)$.

d) The envelope $E(t)$ of the high-frequency fundamental component $\underline{P}(f_2)$ and the intermodulation components $\underline{P}(f_2 \pm (n-1) f_1)$ is calculated by using an analytic signal generated by the Hilbert Transform.

e) The mean value of the envelope

$$\overline{E} = \frac{1}{T_1} \int_0^{T_1} E(t) dt \qquad (59)$$

is calculated by averaging over one period $T_1 = 1/f_1$ of the low frequency tone f_1.

f) Amplitude modulation distortion is calculated by the relative variation of the envelope of the high-frequency signal f_2:

in percentage:

$$AMD(f_1, f_2) = \frac{\sqrt{\frac{2}{T_1} \int_0^{T_1} \left(E(t) - \overline{E} \right) dt}}{\overline{E}} * 100\% \qquad (60)$$

in decibels:

$$L_{AMD}(f_1, f_2) = 20 \lg \left(\frac{AMD(f_1, f_2)}{100} \right) \qquad (61)$$

NOTE The amplitude modulation distortion is comparable to the total intermodulation distortion as defined in section e). If the distortion values are on the same order of magnitude the amplitude modulation (e.g. caused by force factor Bl(x)) is dominant and frequency modulation (e.g. caused by Doppler Effect) is negligible.

24 Multi-tone distortion

24.1 Conditions to be specified

The level of the multi-tone distortion spectrum $MDS(f)$ measured in the sound pressure output $p(t,\mathbf{r})$ at the measurement point $\mathbf{r}$ comprising of components at frequencies $f \neq f_i$ which are not excited by the multi-tone complex $u(t)=\alpha\tilde{u}_{ref}x_m(t)$ in accordance with clause 18.2.

24.2 Method of measurement

a) The device under test shall be excited by the multi-tone stimulus $u(t)=\alpha\tilde{u}_{ref}x_m(t)$ according to clause 9.4. The following properties of the stimulus shall be stated:

- r.m.s. value $\tilde{u}=\alpha\tilde{u}_{ref}$ or the scaling factor α;
- pre-excitation time T_p (generating steady-state conditions);
- measurement time T_M (including repetition of the stimulus for averaging);
- Total number N of excited frequency lines;
- Starting frequency f_{start};
- Resolution R (number of excited lines per octave);
- Length T of the multi-tone complex;
- Spectral shaping function $U(f)$ applied to the stimulus (if used).

b) The sound pressure signal $p(t)$ is measured at the measurement point $\mathbf{r}$ under normal measurement condition in accordance with 8.2. If not stated otherwise, the measurement point $\mathbf{r}$ is the evaluation point $\mathbf{r}_e$.

c) The multi-tone distortion spectrum $MDS(f)$ at frequencies $f \neq f_i$ which are not excited by the fundamentals shall be determined and expressed in dB.

NOTE The multi-tone distortion spectrum $MDS(f)$ can be compared with the spectrum of the fundamental components. To simplify the interpretation it is recommended to smooth the multi-tone distortion spectrum $MDS(f)$ and to calculate a relative distortion metric as proposed by Voishvillo [20].

25 Impulsive Distortion

25.1 Impulsive distortion level

25.1.1 Conditions to be specified

The impulsive distortion level $ID(f(t))$ measured in the sound pressure signal $p(t,\mathbf{r})$ at the measurement point $\mathbf{r}$, the properties of the sinusoidal stimulus $u(t)=\alpha\tilde{u}_{\max}x(t)$ and the properties of the high-pass filter shall be stated.

NOTE Voice coil rubbing, buzzing, air leakage noise and other irregular loudspeaker defects generate distortions covering a wide frequency range with low spectral energy close to the noise level. By exploiting amplitude and phase information of higher-order harmonics and other random distortion components, the impulsive distortion level $ID(f(t))$ provides maximum sensitivity for random defects such as loose particles as described in Annex A.7.

25.1.2 Method of measurement

a) The DUT shall be excited by the stimulus $u(t)=\alpha\tilde{u}_{\max}x_c(t)$ using a chirp signal $x(t)$ in accordance with clause 9.1 The measurement time T_s and either the r.m.s. value $\tilde{u}=\alpha\tilde{u}_{\max}$ or the scaling factor α shall be stated.

b) The sound pressure signal $p(t)$ is measured at the measurement point $\mathbf{r}$ under normal measurement conditions in accordance with 8.2. If not stated otherwise, the measurement point $\mathbf{r}$ is the evaluation point $\mathbf{r}_e$.

c) The impulsive distortion time signal $d_h(t)$ shall be generated by high-pass filtering of the measured sound pressure signal $p(t)$. The cut-off frequency $f_c(t)$ of the high-pass filter or alternative analysis methods shall be stated by the manufacturer. It is recommended to use a cut-off frequency $f_c(t)$ which is significantly higher than the instantaneous frequency *f(t)* of the chirp. Alternative techniques used for separating the impulsive distortion signal in the time domain shall be stated by the manufacturer.

d) The peak level of the impulsive time signal

$$ID(f(t)) = 20\log_{10}\left(\frac{\max\limits_{T}\left(\left|d_h(t)\right|\right)}{p_0}\right) \tag{62}$$

shall be determined within one period $T=1/f(t)$ of the instantaneous frequency and expressed in decibels by using the standard reference sound-pressure (p_0 = 20 μPa).

25.2 Maximum impulsive distortion ratio

25.2.1 Conditions to be specified

The maximum impulsive distortion ratio IDR measured in the sound pressure signal $p(t,\mathbf{r})$ at the measurement point $\mathbf{r}$, the properties of the sinusoidal stimulus $u(t) = \alpha\tilde{u}_{\max}x(t)$ and the properties of the high-pass filter shall be stated.

25.2.2 Method of measurement

a) The impulsive distortion level *ID(f(t))* of the electro-acoustical system is measured within the frequency range in accordance with clause 25.1. The frequency range corresponds with the rated frequency range in accordance with clause 17 unless stated otherwise.

b) The mean value of the sound pressure level SPL_{mean} in the stated frequency range is measured in accordance with clause 19.8.

c) The maximum level of the impulsive distortion is determined within the stated frequency range $f_{\mathrm{l}}<f<f_{\mathrm{u}}$ and the impulsive distortion ratio is calculated by the following equation:

$$IDR = \max_{f_{\mathrm{l}}<f<f_{\mathrm{u}}}\left(ID(f)\right) - SPL_{\mathrm{mean}} \tag{63}$$

25.3 Mean impulsive distortion level

25.3.1 Conditions to be specified

The mean impulsive distortion level *MID(f(t))* measured in the sound pressure signal $p(t,\mathbf{r})$ at the measurement point $\mathbf{r}$, the properties of the sinusoidal stimulus $u(t) = \alpha\tilde{u}_{ref}x(t)$ and the properties of the high-pass filter shall be stated.

NOTE The mean impulsive distortion level *MID(f)* neglects the phase information of higher-order harmonics and other random distortion components and does not assess the waveform of impulsive distortion as described in Annex A.7.

25.3.2 Method of measurement

a) The DUT shall be excited by the stimulus $u(t) = \alpha\tilde{u}_{\max}x_c(t)$ using a chirp signal *x(t)* in accordance with clause 9.1 The measurement time T_{s} and either the r.m.s. value $\tilde{u} = \alpha\tilde{u}_{\max}$ or the scaling factor α shall be stated.

b) The sound pressure signal *p(t)* is measured at the measurement point $\mathbf{r}$ under normal measurement conditions in accordance with 8.2. If not stated otherwise, the measurement point $\mathbf{r}$ is the evaluation point $\mathbf{r}_e$.

c) The impulsive distortion time signal $d_h(t)$ shall be generated by high-pass filtering of the measured sound pressure signal *p(t)*. The cut-off frequency $f_{\mathrm{c}}(t)$ of the high-pass filter or alternative analysis methods shall be stated by the manufacturer. It is recommended to use a cut-off frequency $f_{\mathrm{c}}(t)$ which is significantly higher than the instantaneous frequency *f(t)* of the chirp.

d) The r.m.s. value of the impulsive distortion time signal $d_h(t)$ is calculated within one period $T=1/f(t)$ of the instantaneous frequency. The mean impulsive distortion level

$$MID(f) = 10\log_{10}\left(\frac{f\int\limits_0^{1/f} d_h^{\,2}(t)dt}{p_0^2}\right) \tag{64}$$

shall be expressed in decibels relative to the standard reference sound-pressure (p_0 = 20 µPa).

25.4 Crest factor of impulsive distortion

25.4.1 Conditions to be specified

The crest factor of impulsive distortion CID(f) measured in the sound pressure signal $p(t,\mathbf{r})$ at the measurement point $\mathbf{r}$, the properties of the sinusoidal stimulus $u(t) = \alpha\tilde{u}_{\max}x(t)$ and the properties of the high-pass filter shall be stated.

NOTE The crest factor of the impulsive $CID(f)$ is a relative measure exploiting the phase information of the distortion components. The amplitude information is suppressed. Most loudspeaker defects generate impulsive distortion with a high crest factor (CID > 12 dB). Noise and motor and suspension nonlinearities (e.g. force factor Bl(x), stiffness Kms(x)) generate a CID which is less than 12 dB.

25.4.2 Method of measurement

a) The impulsive distortion level $ID(f(t))$ of the electro-acoustical system is measured within the stated frequency range in accordance with clause 25.1. The frequency range corresponds with the rated frequency range in accordance with clause 17 unless stated otherwise.

b) The mean impulsive distortion level $MID(f(t))$ of the DUT is measured within the stated frequency range in accordance with clause 25.3.

c) The crest factor of the impulsive distortion is calculated as follows:

$$CID(f) = ID(f) - MID(f) \tag{65}$$

26 Stray magnetic fields

NOTE It is sometimes necessary to know the value of the magnetic field generated by the DUT in order to prevent interference with other nearby components such as television and video components, computer devices, aircraft on-board instrumentation, medical devices such as pacemakers, etc.

26.1 Static component

26.1.1 Characteristic to be specified

The maximum value of the static magnetic field strength in amperes per metre, produced by the magnet system of the loudspeaker at 30 mm from any part (or associated parts) of its rear side, or from any part of its enclosure, when no audio signal is applied shall be specified. The static component (H) can also be measured as the magnetic induction; in that case the measured value shall be converted into ampere per metre using the following relations:

$$H = \frac{B}{\mu_0} \tag{66}$$

where

$\mu_0 (= 4\pi 10^{-7}\,\mathrm{H/m}$ is the magnetic permeability of vacuum/air.

B is the magnetic flux density in Tesla.

26.1.2 Method of Measurement

a) The static magnetic flux shall be measured using a Hall (or other suitable type) probe flux meter. A non-magnetic holder (wood or plastic, for example) shall be fitted onto the probe to control the measuring distance from the loudspeaker under test, as shown in Figure 14.

b) Before starting the measurement, it shall be necessary to set the instrument to zero as instructed by the manufacturer, on order to remove the influence of the Earth's magnetic field. To do this, the Hall probe fixture shown in Figure 14 shall be properly oriented and fixed to null the indication due to the Earth's magnetic field. Care shall be taken to exclude any magnetic material from the measuring area which exhibit a low and uniform ambient magnetic field.

c) When the Hall probe is properly oriented, the loudspeaker under test shall be moved about the Hall probe holder to obtain the highest measured value. Measurement may also be made by changing the position and orientation of the Hall probe instead of changing that of the loudspeaker. If the positioning of the Hall probe is used, the measuring space shall not exhibit any external magnetic influence that exceeds 1/10 of the magnetic field strength to be measured.

d) The highest measured value of magnetic field strength expressed in ampere per metre shall be recorded as the result.

NOTE The report shall include the position and orientation for the maximum values with respect to the reference plane and the reference point of the loudspeaker. This information can be shown in a diagram.

26.2 Dynamic components

26.2.1 Characteristics to be specified

Maximum values of both static and alternating components of the magnetic field strength in A/m created by the loudspeaker and associated parts at a measuring distance of 30 mm, when the loudspeaker is driven at its rated noise voltage of simulated program signal in accordance with IEC 60268-1 shall be specified.

Both static and alternating components shall be specified. The rated noise voltage shall be stated with the results.

26.2.2 Method of measurement

a) During the measurement, the loudspeaker to be tested shall be electrically driven by the simulated programme signal in accordance with IEC 60268-1 generating the rated maximum sound pressure level $SPL_{\max}$ in accordance with 19.2.

b) The static and alternating components shall be measured using a Hall effect probe flux meter (or other suitable type with measuring range up to 10 000 Hz) while alternating component measurement is possible with the standard search coil in accordance with IEC 60268-1. A non-magnetic holder (wood or plastic, for example) shall be fitted onto the probe to control the measuring distance from the loudspeaker under test as in 26.1.2.

c) For a static component the measuring procedure shall be the same as described in 26.1.2b).

d) For an alternating component, before starting the measurement, the Hall effect probe (or search coil) fixture shall be oriented so that no external influence can reach 1/10 of the alternating component to be measured, within the measuring frequency band. Care shall be taken to remove any electromagnetic influence from the measuring area that can degrade the required measuring accuracy.

e) When the magnetic probe is properly oriented, the loudspeaker under test shall be applied against the magnetic probe holder, in any possible position, until the highest measured value shall be found. Measurement can also be made by changing the position and orientation of the Hall effect probe instead of changing that of the loudspeaker. In this case, the measuring area shall not exhibit any external magnetic influence which exceeds a 1/10 of the magnetic field strength to be measured.

f) The highest measured values of the static component and of the alternating component of the magnetic field strength expressed in ampere per metre shall be recorded as the results.

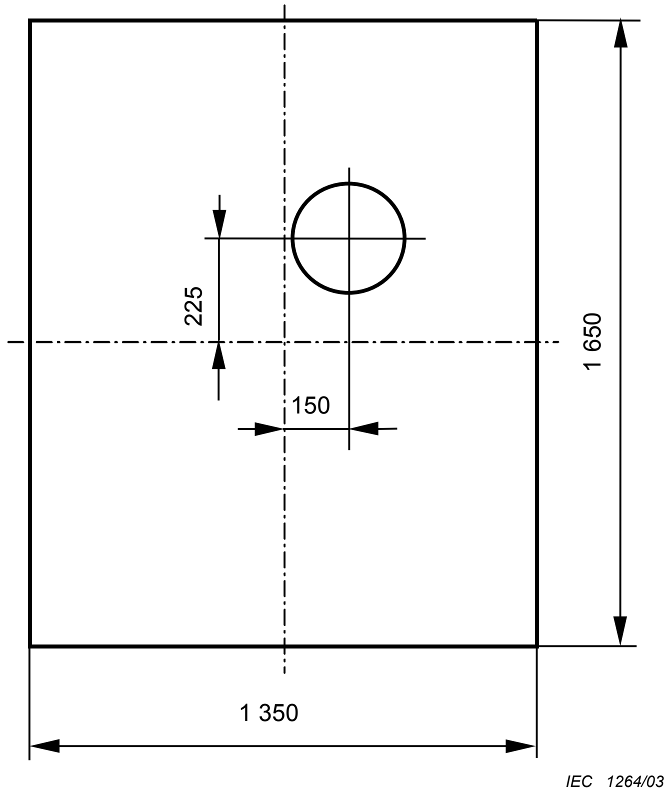
NOTE The report shall include the position and orientation of the maximum values with respect to the reference plane and the reference point of the loudspeaker. This information can be shown in a diagram.

Annex A

A.1 Mounting of transducer

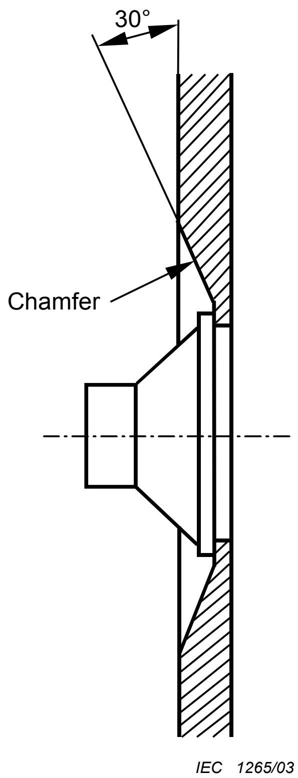
A.1.1 Standard baffle

The standard baffle shall be made with a plane front surface that is acoustically reflecting. It shall be of a material with adequate thickness to ensure negligible vibration. The baffle shall have the dimension of Figure 4. The edge of the radiating element shall be substantially flush with the front surface of the baffle. This may be achieved by means of a chamfer as shown in Figure 5 or by the use of a thin rigid sub-baffle, with or without a chamfer, as shown in Figure 6.



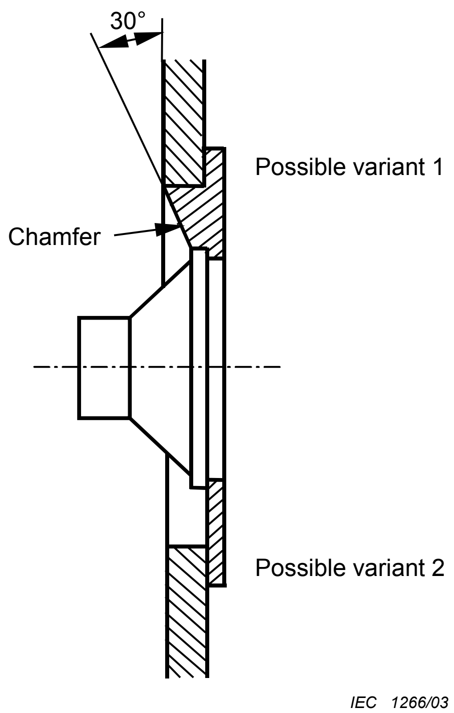
Dimensions are in millimeters

Figure 4 – Standard baffle, dimensions



1967

1968 **Figure 5 – Standard baffle with chamfer**



1969

1970 **Figure 6 – Standard baffle with sub-baffle**

1971

A.1.2 Standard measuring enclosures

One of the two types of standard measuring enclosures specified in Figure 7 and Figure 8 shall be used. The type of choose for testing shall be stated by the manufacturer. A standard measuring enclosure shall meet the following conditions:

a) The enclosure shall have plane or curved surfaces that have an acoustically reflective characteristic. The material shall be appropriately thick so that the effect of vibrations can be disregarded for measurement. If necessary, braces shall be used for reinforcement between facing surfaces at or around their centers so as to avoid panel vibrations.

b) The enclosure shall be airtight.

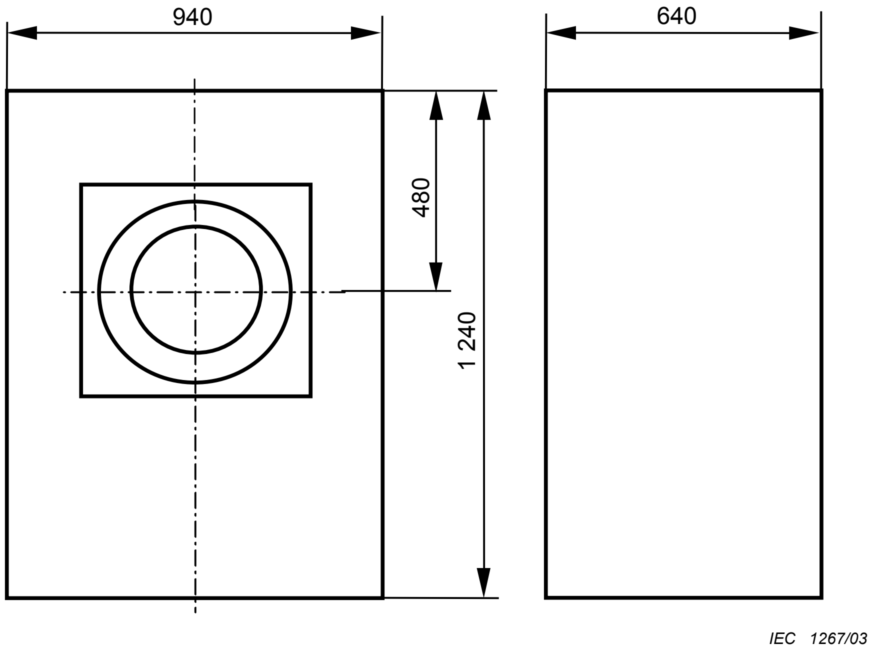
c) The edge of the loudspeaker shall be, in principle, set on the same plane as that of the front part of the baffle.

d) To remove standing waves which may otherwise occur in the enclosure, an appropriate sound absorbing material shall be used. Handles or joints may be installed if their effect on acoustical reflections and undesired vibrations can be ignored.

A.1.2.1 Type-A

The dimensions of standard measuring enclosure type-A shall be as shown in Figure 7.

NOTE All the surfaces of this type enclosures are plane and the joints of the surfaces are made at right angles. No change in size is allowed. This causes the diffraction characteristic to be repeatable. Correction curves for the standard measuring enclosure shall be determined at a measuring distance of 1 m on the reference axis from free-field to half-space free-field condition. The type A is useful when analysing, studying or comparing the characteristics of transducers in detail.



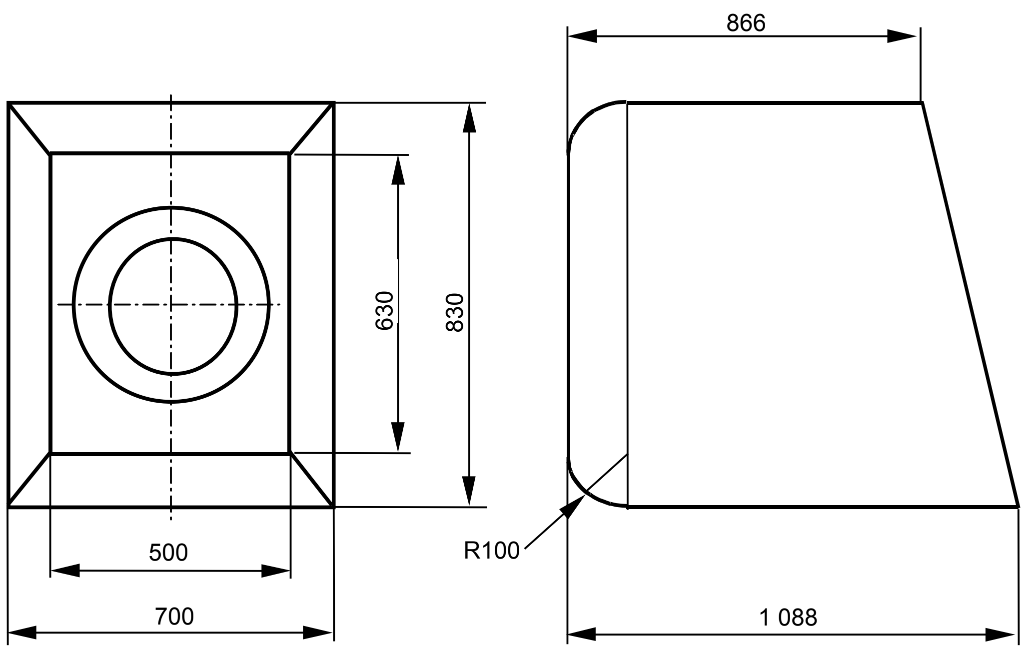
Dimensions are in millimeters

Figure 7 – Standard measuring enclosure type-A (net volume is about 600 l)

A.1.2.2 Type-B

The dimensions of standard measuring enclosure type-A shall be as shown in Figure 8.

NOTE If a smaller or larger measuring enclosure of type-B is required, proportional scaling shall be applied and the outer dimension shall be stated. Correction curves for the standard measuring enclosure shall be determined at a measuring distance of 1 m on the reference axis from free-field to half-space free-field condition.



Dimensions are in millimeters

Figure 8 – Standard measuring enclosure type-B (net volume is about 450 l)

A.2 Simulated programme signal

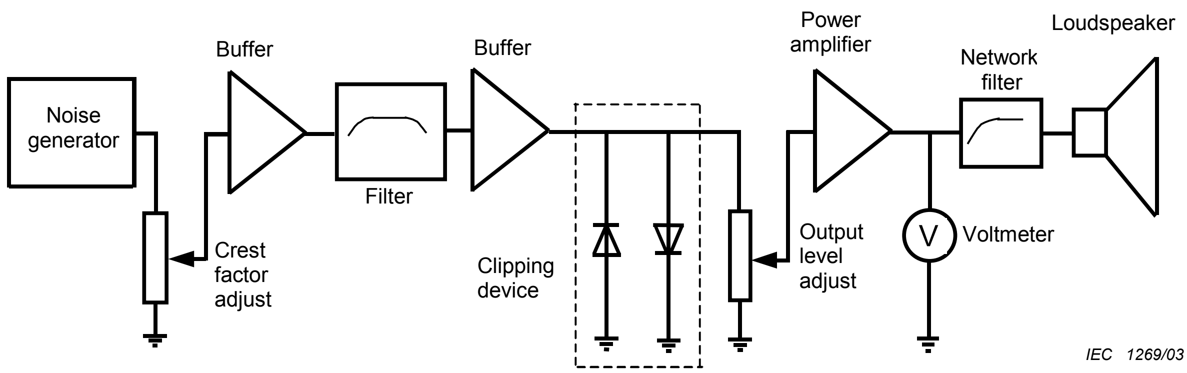


Figure 9 – Block diagram of test setup for generating the simulated noise signal used for testing passive loudspeaker systems consisting of a network filter

NOTE In passive loudspeaker systems consisting of a passive network (crossover), the maximum input value $\tilde{u}_{max}$ is applied to the voltage at the input of the network. The maximum input value $\tilde{u}_{max}$ shall be applied to the transducer terminals if they are accessible in active systems.

Table 1 – Power spectrum of simulated programme signal

	CEA 2034 [16]			IEC 60268-1	
Frequency (Hz)	**Relative Level (dB)**	**Tolerance (dB)**		**Relative Level (dB)**	**Tolerance (dB)**
5	–24.1	±2.0			
6.3	–20.1	±2.0			
8	–16.0	±2.0			
10	–12.2	±2.0			
12.5	–8.8	±1.0			
16	–5.4	±1.0			
20	–3.0	±1.0		–13.5	±3.0
25	–1.5	±0.8		–10.2	±2.0
32	–0.6	±0.6		–7.4	±1.0
40	–0.3	±0.5		–5.2	±1.0
50	–0.1	±0.5		–3.5	±1.0
63	0	±0.5		–2.3	±1.0
80	0	±0.5		–1.4	±1.0
100	0	±0.5		–0.9	±0.8
125	0	±0.5		–0.5	±0.6
160	0	±0.5		–0.2	±0.5
200	0	±0.5		–0.1	±0.5
250	0	±0.5		0	±0.5
315	0	±0.5		0	±0.5
400	0	±0.5		0	±0.5
500	0	±0.5		0	±0.5
630	–0.1	±0.5		0	±0.5
800	–0.1	±0.5		0	±0.5
1 000	–0.2	±0.6		–0.1	±0.6
1 250	–0.4	±0.7		–0.3	±0.7
1 600	–0.6	±0.8		–0.6	±0.8
2 000	–0.1	±1.0		–1.0	±1.0
2 500	–1.5	±1.0		–1.6	±1.0
3 150	–2.3	±1.0		–2.5	±1.0
4 000	–3.5	±1.0		–3.7	±1.0
5 000	–4.9	±1.0		–5.1	±1.0
6 300	–6.8	±1.0		–7.0	±1.0
8 000	–9.1	±1.0		–9.4	±1.0
10 000	–11.8	±1.0		–11.9	±1.0
12 500	–14.9	±1.5		–14.8	±1.5
16 000	–18.2	±2.0		–18.2	±2.0
20 000	–21.8	±3.0		–21.6	±3.0

A.3 Spherical Wave Expansion

A.3.1 Coefficients of spherical wave expansion

A DUT having the largest geometrical dimensions d generates a far field at the distance $|\mathbf{r}\text{-}\mathbf{r}_{\mathrm{ref}}| >> d$ which can be described by a superposition of spherical waves. The coefficients

$$\mathbf{C}(f) = \left\lfloor \underline{c}_{0,0}(f) \quad \dots \quad \underline{c}_{n,m}(f) \quad \dots \quad \underline{c}_{N(f),N(f)}(f) \right\rfloor \tag{67}$$

of the spherical wave expansion of order $N(f)$ depending on frequency f describe the transfer function

$$\begin{aligned} \underline{H}(f,\mathbf{r}) &= \mathbf{C}(f)\mathbf{B}(f,\mathbf{r}) \\ &= \sum_{n=0}^{N(f)} \sum_{m=-n}^{n} \underline{c}_{n,m}(f) h_n^{(2)}\left(\frac{2\pi fr}{c}\right) Y_n^m(\theta,\phi) \end{aligned} \tag{68}$$

between the input signal $u(t)$ and the sound pressure output $p(t,\mathbf{r})$ at point $\mathbf{r}$ at a distance $r=|\ \mathbf{r} - \mathbf{r}_{ref}\ |$ from the reference point $\mathbf{r}_{ref}$ which is larger than the validity radius a in accordance with clause 21.1.3.

The spherical harmonics

$$Y_n^m(\theta,\phi) \tag{69}$$

show the dependency versus azimuthal angle ϕ and polar angle θ and the Hankel functions of the second kind

$$h_n^{(2)}\left(\frac{2\pi fr}{c}\right) \tag{70}$$

describes the dependency in radial direction for an outgoing wave radiated from the reference point $\mathbf{r}_{ref}$.

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

A.3.2 Directional factor

$$\underline{\Gamma}(f,\theta,\phi) = \lim_{r\to\infty} \frac{\underline{H}(f,r,\theta,\phi)}{\underline{H}(f,r,\theta_r,\phi_r)} = \frac{\sum_{n=0}^{N(f)} \sum_{m=-n}^{n} \underline{c}_{n,m}(f) i^{n+1} Y_n^m(\theta,\phi)}{\sum_{n=0}^{N(f)} \sum_{m=-n}^{n} \underline{c}_{n,m}(f) i^{n+1} Y_n^m(\theta_r,\phi_r)} \tag{71}$$

is the ratio between the transfer function $H(f,r,\phi,\theta)$ at any azimuthal angle ϕ and polar angle θ to the transfer function $H(f,r,\phi_r,\theta_r)$ on the reference axis defined by reference angles ϕ_r, θ_r with the complex operator $i = \sqrt{-1}$.

A.3.3 Directivity

A.3.3.1 Characteristics to be specified

The directivity is calculated by

$$Q(f) = 8\pi \frac{\left| \sum_{n=0}^{N(f)} \sum_{m=-n}^{n} \underline{c}_{n,m}(f)\, j^{n+1} Y_n^m(\theta = \pi/2, \phi = 0) \right|^2}{\sum_{n=0}^{N(f)} \sum_{m=-n}^{n} |\underline{c}_{n,m}(f)|^2} \tag{72}$$

A.3.4 Acoustic output power

The acoustic output power in a defined frequency band at center frequency f is calculated by

$$\Pi(f) = \frac{\tilde{u}(f)^2}{\rho c} \sum_{n=0}^{N} \sum_{m=-n}^{n} \left| \underline{c}_{n,m}(f) \right|^2 \tag{73}$$

using the coefficients $C(f)$ of the wave expansion and the r.m.s. value $\tilde{u}(f)$ of the input signal $u(t)$ in this band.

A.4 Equivalent harmonic input distortion

The equivalent harmonic input distortion (EHD) in the input signal $u(t)$ can be measured in the harmonic distortion in the sound pressure output $p(t,\mathbf{r}_i)$ at any measurement point $\mathbf{r}_\mathrm{i}$ with i=1,...,N. If the source of the harmonic distortion in the sound pressure output is located in the one-dimensional signal path close to the DUTs input (see nonlinear system N_1 in Figure 10), the same EHD will be found in every measurement point at all locations $\mathbf{r}_i$ in the sound field. The dominant transducer nonlinearities in the motor and suspension system can be represented by the single nonlinear system N_1 generating EHD in the one-dimensional signal path. If there are nonlinear sources located in the multi-dimensional signal path (see $N_{2,1}$, $N_{2,2}$ in Figure 10), the EHD generated from these sources will be superimposed onto the respective EHD generated in the one-dimensional signal path. Therefore, if nonlinearities exist in the multi-dimensional signal path, the total EHD at each point $\mathbf{r}_i$ will depend on the points position in the sound field [14].

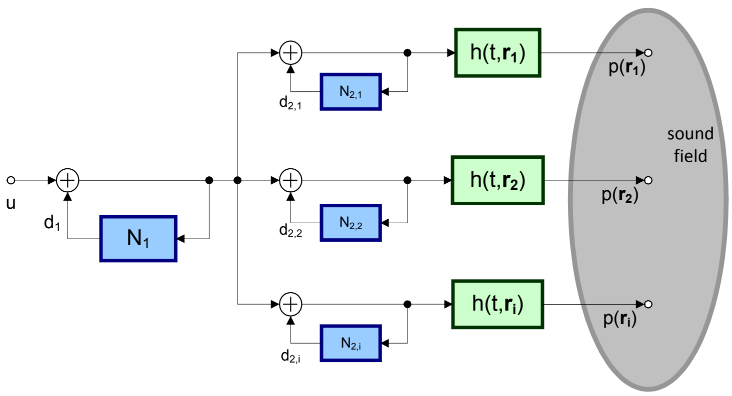


Figure 10 – Signal flow chart of the electro-acoustical system

A.5 Two-tone intermodulation

Intermodulation distortion can easily be interpreted and it is a valuable diagnostic tool. The frequencies f_1 and f_2 of the two-tone stimulus shall be carefully selected to measure intermodulation distortion. The possible combinations of the frequency settings are represented as a two-dimensional frequency plane in

Figure 11. The condition $f_2 >> f_1$ ensures that the intermodulation distortion components of f_2 will not interfere with the fundamental and harmonic components of f_1. Shown as blue shaded areas in Figure 11, the intermodulation distortion generated with a large ratio or difference between the two fundamentals reveals the nonlinear properties of the DUT. These nonlinear properties cannot be assessed by harmonics generated when $f_1=f_2$ which is the black diagonal line shown in

Figure 11. A difference tone measurement using a small distance $f_2–f_1$ provides similar information as the harmonic components which are shown as white shaded areas along the black diagonal line in Figure 11.

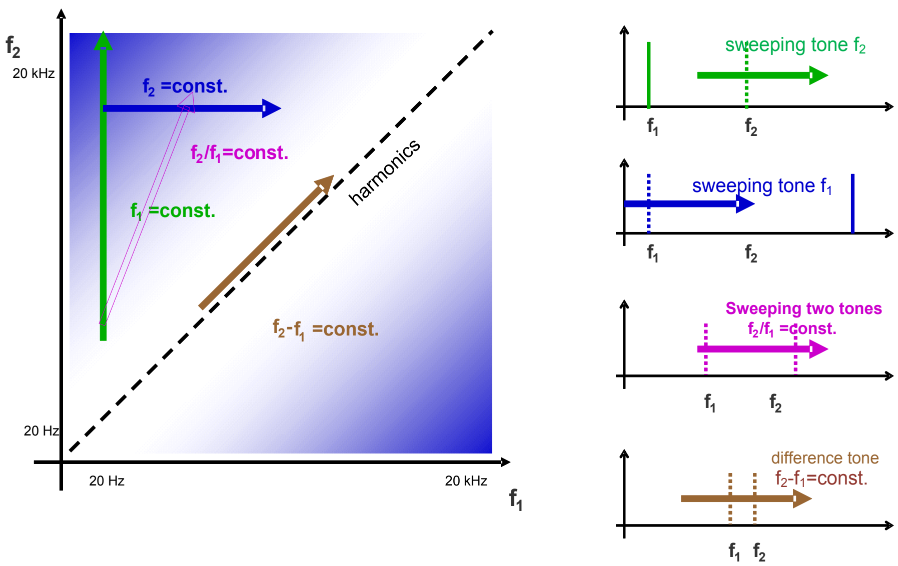


Figure 11 – Variation of the frequencies of the two-tone stimulus in the intermodulation measurement

A.6 Multi-tone distortion

The device under test shall be excited by a sparse multi-tone stimulus comprised of a multitude of tones at frequencies f_i as defined in clause 9.4. The distortion components generated at frequencies $f \neq f_i$ between the excited fundamentals are evaluated. The multi-tone stimulus can be used to simulate the steady-state properties of music and other audio signals allowing the assessment of both harmonic and intermodulation distortion components [20].

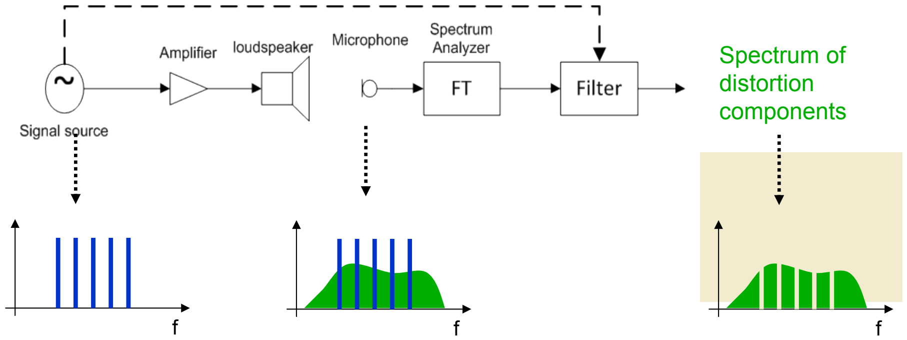


Figure 12 – Measurement of the distortion generated by a multi-tone stimulus

A.7 Impulsive distortion

A loudspeaker overload or defect, a limiting amplifier or artefacts generated by electric protection systems generate nonlinear distortion with transient properties perceived as a rattling, squeezing or buzzing sound. Contrary to regular nonlinear distortion as found in well-made loudspeakers where the 2nd or 3rd-order component is dominant and the amplitude of the higher-order distortion decays rapidly, the spectrum of impulsive distortion contains high-frequency components corresponding to the generation process. Bottoming of the voice coil former at the lower pole-plate is an example of a deterministic defect generating repetitive clicks at negative peak displacements corresponding with higher-order harmonics having approximately the same amplitude. However, defects based on a random generation process, such as turbulent air leak noise and loose particles, generate a dense broad-band signal similar to white noise.

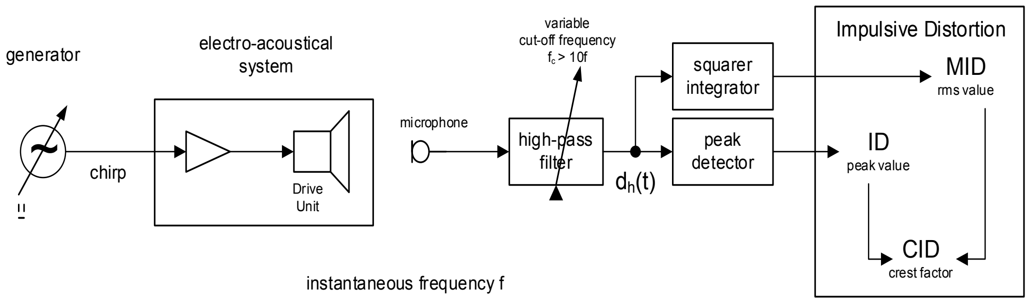
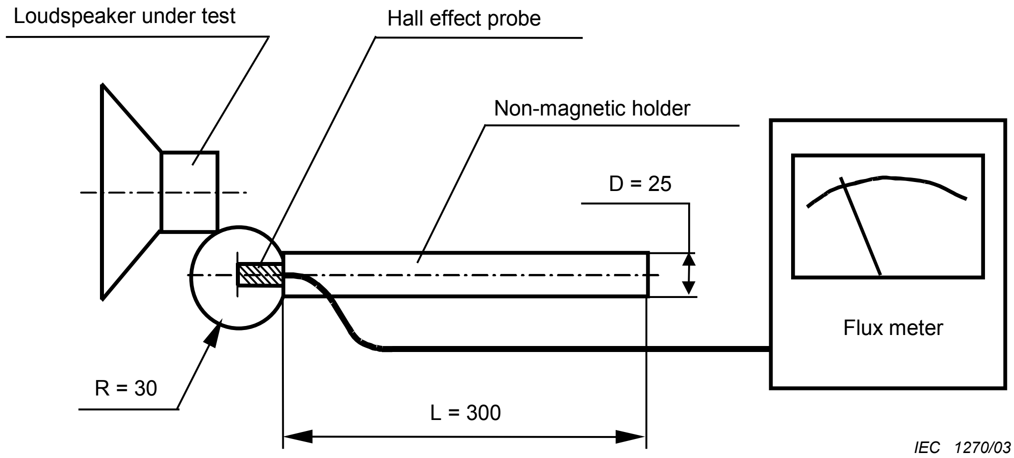


Figure 13 – Measurement of impulsive distortion

2103 ## A.8 Stray magnetic field



2104

2105 *Dimensions are in millimeters*

2106 **Figure 14 – Measuring apparatus for stray magnetic field**

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