ANSI/CTA Standard

**Standard Method of Measurement
for In-Home Loudspeakers**

ANSI/CTA-2034-A

(Formerly ANSI/CEA-2034-A)

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ANSI

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Consumer
Technology
Association

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(Formulated under the cognizance of the CTA **R3 Audio Systems Committee**.)

FOREWORD

This standard describes an improved method for measuring and reporting the performance of a loudspeaker in a manner that should help consumers better understand the performance of the loudspeaker and convey a reasonably good representation of how it may sound in a room based on its off-axis response and how this response affects the consumer's experience.

Unlike previously published standards, this standard describes how to measure and report the directivity of a loudspeaker, whether it stands by itself or is mounted in or on a wall or ceiling. It also describes how to use this directivity data to estimate the in-room frequency response that more recent research has shown correlates well to subjective listening preferences of consumers.

It describes how to measure and report the maximum on-axis usable sound pressure level of a loudspeaker, and how to measure and report the impedance of a loudspeaker. It also describes how to calculate and report the size of the power amplifier needed for the consumer to get the desired SPL from the loudspeaker.

Finally, it includes a number of informational annexes to help readers gain a more thorough understanding of techniques for acquiring loudspeaker data in both anechoic and non-anechoic environments, as well as methods for using this acquired data to predict loudspeaker performance.

When used properly this standard should assist manufacturers in accurately measuring the capabilities of a full-range loudspeaker system and specifying them to consumers. Using this in conjunction with ANSI/CEA-2010-B – Standard Method of Measurement for Subwoofers, the ANSI/CEA-2034-A full-range loudspeaker ratings should make it easier for the consumer to select, purchase, and enjoy a subwoofer that will complement their main full-range loudspeaker system.

This standard was developed by the Consumer Electronics Association's Audio Systems Committee (CEA R3 Committee).

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Standard Method of Measurement for In-Home Loudspeakers

1 Scope

This standard describes how to determine the frequency response, directivity and maximum output capability of a residential loudspeaker. It is intended to determine the audio performance of a loudspeaker, not the loudspeaker's ability to survive a given input signal.

This standard applies only to loudspeaker systems, and not to raw transducers.

2 Revision History
Added Table of Figures and Tables
Added items to Symbols & Abbreviations
Added section for Compliance Notation
Updated General Test Setup & Conditions
Added section for Frequency Response and Crossover Categories
Added section for Powered Loudspeaker Rating
Added column for voltage to the Required Amplifier Power tables
Added section for Matching a Subwoofer to a Full-Range Loudspeaker System
Incorporated new additions to Consolidated Reporting Requirements section

3 References

3.1 Normative References

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

3.1.1 Normative Reference List

IEC Publication 225, *Octave, Half-Octave, and Third-Octave Band Filters Intended for the Analysis of Sound and Vibration* (1966)

3.1.2 Normative Reference Acquisition

IEC Standards:
International Electrotechnical Commission, IEC Central Office, 3, rue de Varembé
P.O. Box 131, CH - 1211 Geneva 20 – Switzerland; Phone +41 22 919 02 11;
Fax +41 22 919 03 00; Internet http://www.iec.ch

3.2 Informative References

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

3.2.1 Informative Reference List

AES Convention Paper 5638, *Characterizing the Amplitude Response of Loudspeaker Systems,* presented at 113th Audio Engineering Society Convention October 2002, Allan Devantier

CEA-490-A R-2008, *Test Methods of Measurement for Audio Amplifiers*

IEC 60263 ed3.0 (1982-01), *Scales and Sizes for Plotting Frequency Characteristics and Polar Diagrams*

IEC 60268-1 ed2.0 (1985-01), *Sound system equipment. Part 1: General*

IEC 60268-1-am1 ed2.0 (1988-01), *Amendment 1 - Sound system equipment. Part 1: General*

IEC 60268-1-am2 ed2.0 (1988-01), *Amendment 2 - Sound system equipment. Part 1: General*

IEC 60268-2 ed2.0 (1987-06), *Sound system equipment. Part 2: Explanation of general terms and calculation methods*

IEC 60268-2-am1 ed2.0 (1991-05), *Amendment 1 - Sound system equipment. Part 2: Explanation of general terms and calculation methods*

IEC 60268-5 ed3.1 Consol. with am1 (2007-09), *Sound system equipment - Part 5: Loudspeakers*

NIST Special Publications 811 2008 Edition, Guide for the use of the International System of Units universally abbreviated SI (from the French *Le Systeme Intgernational d'Unites*)

3.2.2 Informative Reference Acquisition

Audio Engineering Society (AES): http://www.aes.org/e-lib/browse.cfm?elib=11234

CEA Standards:
Techstreet, 3916 Ranchero Drive, Ann Arbor, MI, USA 48108; Phone 800-699-9277; Fax 734-780-2046; Internet http://www.techstreet.com; Email techstreet.service@thomsonreuters.com

IEC Standards:
International Electrotechnical Commission, IEC Central Office, 3, rue de Varembé
P.O. Box 131, CH - 1211 Geneva 20 – Switzerland; Phone +41 22 919 02 11;
Fax +41 22 919 03 00; Internet http://www.iec.ch

NIST Standards:
National Institute of Standards and Technology, 100 Bureau Drive, Gaithersburg, MD 20899-8930; Phone 301 975-6478; Internet http://www.nist.gov/publication-portal.cfm Email
inquiries@nist.gov

3.3 Definitions

Continuous	The root mean squared value of a signal over a one second interval
Normalized Transfer Function	A transfer function measurement that has been referenced to (divided by) another transfer function measurement
Peak	The instantaneous maximum value, either positive or negative, of a signal
Point of Rotation (POR)	The point on the front baffle of the loudspeaker that intersects the reference axis

3.4 Symbols and Abbreviations

CF	Crest Factor
dB	decibel
DUT	Device Under Test
ERDI	Early Reflections Directivity Index
FFT	Fast Fourier Transform
HD+N	Harmonic Distortion + Noise
Hz	hertz
kHz	kilohertz
LFCS	Low Frequency Calibration Source
m	meter
MIV	Maximum Input Voltage
Pa	pascal
PIR	Predicted In-Room Amplitude Response
POR	Point of Rotation
RCF	Room Correction Factor
rms	Root Mean Square
SPDI	Sound Power Directivity Index
SPL	Sound Pressure Level

Note: Throughout this document all references to "dB SPL" refer to sound pressure level referenced to 20 µPa.

SPL_{MUCO}	Sound Pressure Level Maximum Usable Continuous Output
THD+N	Total Harmonic Distortion + Noise
V	volt

3.5 Compliance Notation

As used in this standard "shall" and "must" denote mandatory provisions of the standard. "Should" denotes a provision that is recommended but not mandatory. "May" denotes a feature whose presence does not preclude compliance, and implementation of which is optional. "Optional" denotes items that may or may not be present in a compliant device.

4 Test Setup and Conditions

> **CAUTION — WEAR EAR PROTECTION!** High sound levels can cause hearing damage with prolonged exposure. The operator is advised to wear appropriate protection such as ear plugs, preferably balanced attenuation devices which will still permit the operator to listen critically for distortion or other signs of distress in the device under test.

4.1 General Test Setup and Conditions

All testing shall be conducted in an anechoic chamber or equivalent environment.

During testing the DUT shall be operated as per manufacturer's instructions. Complete systems shall be tested intact.

Testing shall be conducted at an ambient temperature of 22 °C ± 5 °C (71.6 °F ± 9 °F) and relative humidity of 30 % to 80 %. If the conditions are different, they shall be noted at the time of the test. Both of these items shall be recorded at the time of the tests. The ambient atmospheric pressure (in Pascals) shall also be recorded.

For powered loudspeakers the mains voltage used to power the DUT shall be within 1 % of the rated value specified for use by the DUT (i.e. for a DUT that requires 120 V, the mains voltage used shall be between 118.8 V and 121.2 V).

All test equipment used during the tests shall operate within its linear operating range. For example, amplifiers shall be sufficiently powerful that they never clip or distort during any of the tests.

All test equipment used during the tests shall operate within its linear operating range. For example, amplifiers shall be sufficiently powerful that they rarely clip or distort during any of the tests.

If a crossover or filter network is specified by the manufacturer, it shall be installed according to the manufacturer's instructions during all tests, including frequency response and maximum SPL tests.

If a small loudspeaker (satellite, bookshelf, etc.) without a manufacturer-specified filter network is being tested then a two pole Butterworth 80 Hz high pass filter network shall be used.

All test equipment shall be properly calibrated, and its calibration shall be documented.

The frequency response measurements shall be a transfer function type measurement employing a dual channel FFT or similar measurement apparatus.

The measurement system shall have an amplitude resolution of at least 0.1 dB.

See Appendix E for a list of all equipment needed for perform the tests in this document.

4.2 Verification of Frequency Response at DUT Input Terminals

Measure the voltage at the DUT input terminals over the range 20 Hz to 20 kHz and confirm that it is within ± 0.5 dB from the specified voltage. If it deviates more than ± 0.5 dB the appropriate corrective action shall be taken, such as the addition of equalization prior to the input terminals of the DUT.

4.3 Measurement Resolution and Calibration

The measurement resolution shall be no wider than 1/20 octave. The specific bandwidth, resolution and test stimulus used should be documented.

Calibration of the microphone and data acquisition system, and of the post processing of the resulting data, shall be such that the system yields a measured sound level identical to that of a stable pure tone stepped or swept with an rms input voltage of 2.83 V.

5 Directivity Response – Standalone Loudspeakers

The following procedure applies to conventional floor standing loudspeakers, or to small loudspeakers intended to be used on stands, both of which would normally be placed away from the walls. Loudspeakers intended to be installed in walls or ceilings or mounted on walls or ceilings are not covered by this section.

The procedure is described in two parts, one dealing with the acquisition of the raw data and one dealing with the post-processing of this data and its presentation. The description assumes that an anechoic measurement space (chamber or outdoors) is available. This is a requirement for data that are not significantly compromised in terms of frequency resolution or signal-to-noise ratio. It is recognized that such facilities are not available to all users of this document. Appendix A references some methods for acquiring data that may be useful approximations to anechoic data.

The objective of these measurements is to be able to anticipate how a loudspeaker might sound in a normally reflective room. No single curve can sufficiently describe how a loudspeaker might sound in a normally reflective room. To better predict its sound, many measurements shall be made at multiple positions surrounding the loudspeaker. These data shall then be processed to estimate the direct, early-reflected and late-reflected sound in a typical small listening room.

5.1 Data Acquisition

The specific bandwidth, resolution and test stimulus used shall be documented.

Measurement Environment

Measurements should be made in a reflection-free environment – a qualified anechoic chamber or outdoors.

Acknowledging that not everyone has access to such facilities, Appendix A references methods for acquiring data that approximates data acquired in a true anechoic space.

Measurement Apparatus

Amplitude response measurements shall be made at 10° intervals on one horizontal and one vertical orbit around the DUT. The orbits intersect at the reference axis, described below, which defines the zero degree azimuth and elevation angles (*i.e.,* the "on axis" measurement). Make 72 measurements (36 horizontal and 36 vertical). Measurements at 0° and 180° are duplicated, so the data set consists of 70 distinctive amplitude response measurements as illustrated in Figure 1.

The measurements shall be made using a single, stationary microphone. The loudspeaker shall be placed on a rotating table, the front edge of which does not project beyond the front of the loudspeaker baffle. The base of the loudspeaker should be horizontal, as it would be in normal use. However, if this results in the reference axis of the DUT being tilted up or down, the turntable shall be mounted at an angle so that the reference axis of the DUT is horizontal for the purposes of making the measurements. For convenience and measurement consistency, it is useful if the rotating platform can be vertically adjusted to compensate for loudspeakers of different heights. A possible setup is illustrated in Figure 2.

Vertical Axis

Horizontal Axis

Reference Axis
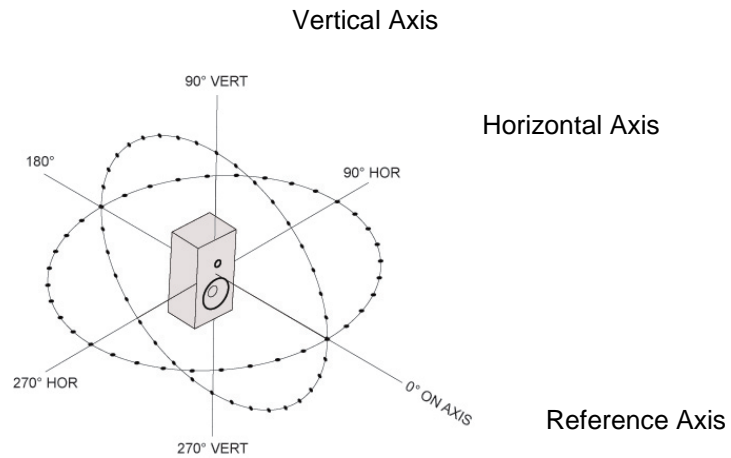
Figure 1: The 70 microphone locations used to acquire the amplitude response data (from Toole, *Sound Reproduction,* Focal Press, 2008)
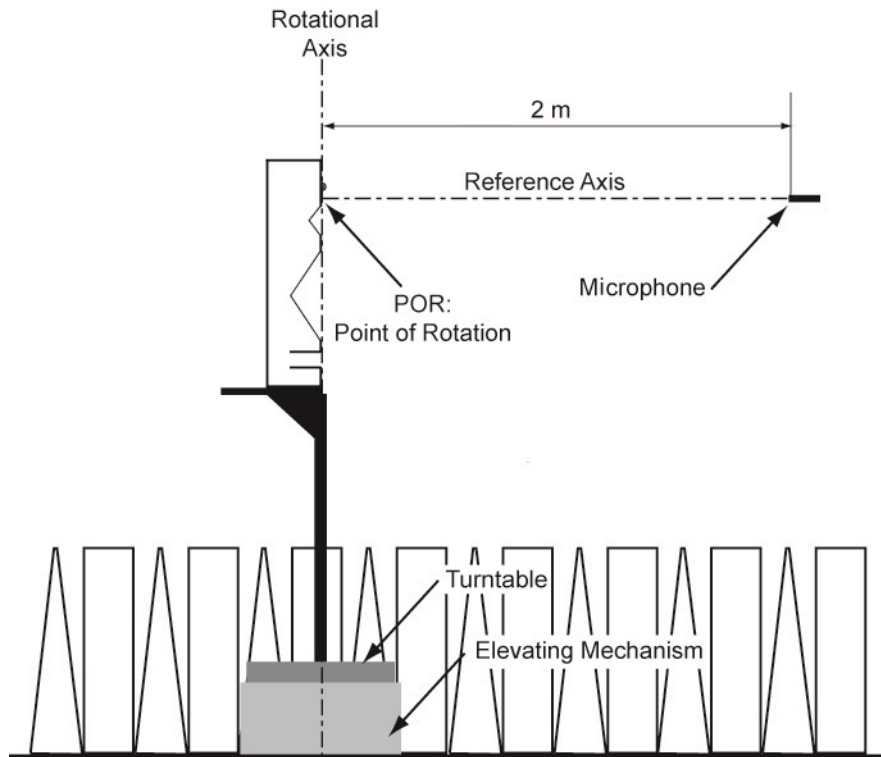
Figure 2: One of several possible physical arrangements within an anechoic chamber that allows rotation and vertical adjustment of the loudspeaker under test. The objective is to minimize acoustical clutter around the loudspeaker and the microphone

The reference axis shall be aimed at the microphone and the DUT rotated in 10° increments to complete measurements on the horizontal orbit. The DUT shall then be placed on its side, and the height of the DUT adjusted so that the reference axis is again aimed at the microphone, and the process shall be repeated. When the DUT is placed on its side to conduct the vertical directivity measurements it shall be rotated about the DUT reference axis. Care must be taken to

ensure that the height of the reference axis never deviates more than 5 cm (≈2 inches) between the horizontal and vertical measurements. This is illustrated in Figure 3. With large floor-standing loudspeakers placed on their sides it may be necessary to provide special outriggers to support the off-center mass.

By definition the POR is the point of intersection of the rotational axis of the turntable and the DUT reference axis. The POR should be located on the front baffle of the DUT, and shall not be any farther than 5 cm (≈2 inches) from the front baffle.
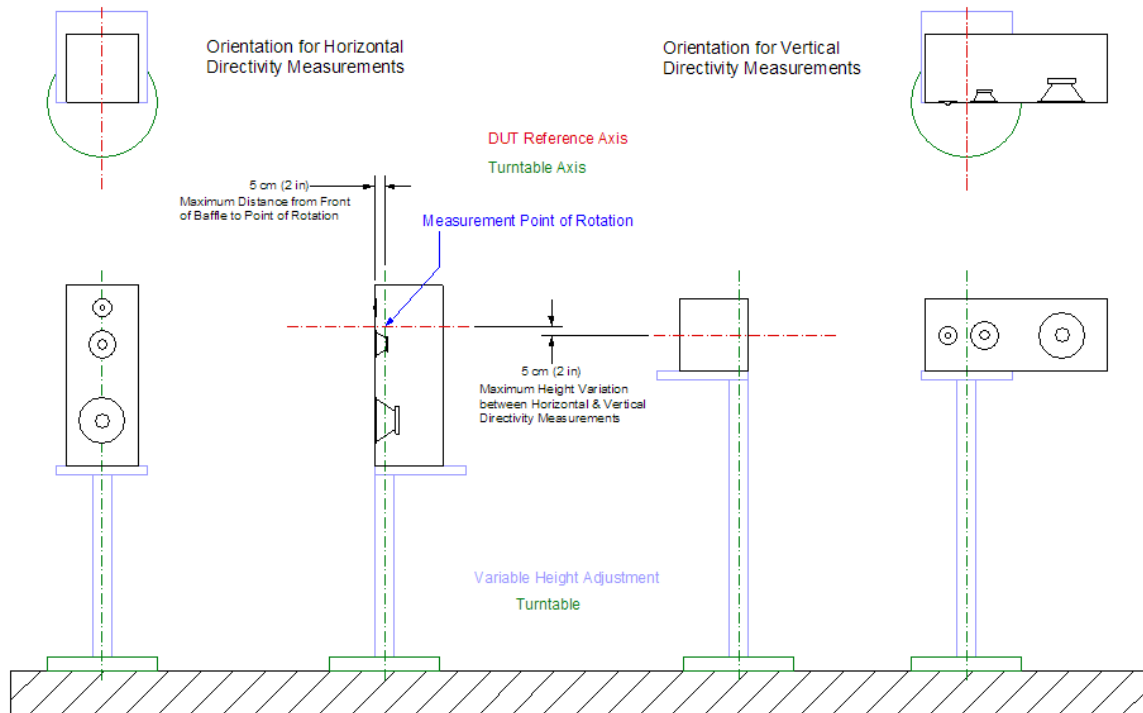
Figure 3: DUT orientation for horizontal & vertical directivity measurements showing the DUT Reference Axis and the Measurement Point of Rotation

The Reference Axis

The reference axis should be specified by the manufacturer. It approximates the direct-sound listening axis for a seated listener in the prime listening location, at a normal listening distance.

Defining the reference axis requires that assumptions be made, and perhaps for this reason manufacturers rarely specify it. Because the reference axis is needed to perform the tests described in this standard the following method shall be used to define it whenever the manufacturer has not specified it.

The starting point of the reference axis shall be either the center of the high-frequency driver (normally a tweeter) or a point midway between the high-frequency and midrange drivers. The specific point that is used shall be specified in the final report. The orientation of the reference axis depends on the design of the loudspeaker. For most conventional loudspeakers with a front baffle, the reference axis shall be perpendicular to the front baffle. If the loudspeaker design is such that there is not a conventional front baffle, the reference axis shall be in the direction of the intended forward radiation of sound from the loudspeaker.

<u>Measurement Distance</u>

Ideally measurements should be made in the far field of the DUT and, for reasons of standardization the sensitivity should be referenced to a distance of 1 m. The far field for large diaphragm loudspeakers can be several meters away and listeners may sit in the near field of these loudspeakers.

For typical loudspeakers the far field begins about 2 m from the DUT, and the typical listening distance is closer to 3 m from the DUT. By taking many measurements and displaying the results as spatial averages useful data can be gathered within the near field. Therefore measurements shall be made at 2 m from the DUT and the data shall be reported as the equivalent sound pressure level (SPL) at 1 m, which is a convenient 6 dB higher than the SPL at 2 m.

<u>Input Signal</u>

When conducting measurements on passive loudspeakers using the procedure described in this standard an rms input voltage of 2.83 V shall be used.

Self-powered loudspeakers or those with a line level active electronic input to the DUT shall be tested with an input voltage that produces 79 dB SPL averaged over the frequency range 500 Hz to 2 kHz at 2 m from the DUT in an anechoic environment.

The spectrum of the input signal shall span a bandwidth of at least 20 Hz to 20 kHz.

The level of the input signal that is fed into the loudspeaker under test shall be documented and shall be within the linear operating range of the loudspeaker. During the frequency response test the signal-to-noise ratio shall be measured and shall always be at least 40 dB.

5.2 Post Processing of Data

The following composite response curves shall be calculated. In each instance a power average of the specified magnitude responses shall be calculated.

Listening Window
The listening window curve is a spatial average of the nine magnitude responses in the ±10⁰ vertical and ±30⁰ horizontal angular range.
- 0°
- ± 10⁰ vertical
- ± 10⁰, ± 20⁰, ± 30⁰ horizontal

Early Reflections
The early reflections curve is an estimate of all single-bounce, first-reflections, in a typical listening room.
- Floor Bounce: 20⁰, 30⁰, 40⁰ down
- Ceiling Bounce: 40⁰, 50⁰, 60⁰ up
- Front Wall Bounce: 0⁰, ± 10⁰, ± 20⁰, ± 30⁰ horizontal
- Side Wall Bounces: ± 40⁰, ± 50⁰, ± 60⁰, ± 70⁰, ± 80⁰ horizontal
- Rear Wall Bounces: 180⁰, ± 90⁰ horizontal

Vertical Reflections
The "floor reflection" is defined as the spatial average of three measurements at 30 degrees below the main-axis ± 10°. The "ceiling reflection" is defined as the spatial average of three measurements at 50° above the main-axis ± 10°.
- Floor Reflection: - 20°, - 30°, - 40° vertical

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- Ceiling Reflection: + 40°, + 50°, + 60° vertical

Horizontal Reflections
- Front: 0°, ± 10⁰, ± 20⁰, ± 30⁰ horizontal
- Side: ± 40°, ± 50°, ± 60°, ± 70°, ± 80° horizontal
- Rear: ± 90°, ± 100°, ± 110°, ± 120°, ± 130°, ± 140°, ± 150°, ± 160°, ± 170°, 180° horizontal, (i.e.: the horizontal part of the rear hemisphere).

Sound Power
The sound power is the weighted rms average of all 70 measurements, with individual measurements weighted according to the portion of the spherical surface that they represent. Calculation of the sound power curve begins with a conversion from SPL to pressure, a scalar magnitude. The individual measures of sound pressure are then weighted according to the values shown in Appendix C and an energy average (rms) is calculated using the weighted values. The final average is converted to SPL.

Sound Power Directivity Index (SPDI)
For the purposes of this standard the Sound Power Directivity Index is defined as the difference between the listening window curve and the sound power curve. An SPDI of 0 dB indicates omnidirectional radiation. The larger the SPDI, the more directional the loudspeaker is in the direction of the reference axis.

Early Reflections Directivity Index (ERDI)
The Early Reflections Directivity Index is defined as the difference between the listening window curve and the early reflections curve.

5.3 Data Presentation

Data shall be referenced to an rms input voltage of 2.83 V at a distance of 1 m.

Data shall be reported in graphical form as illustrated in Figure 4. This chart follows the guidance of IEC 60263, which stipulates that the horizontal length for a 10:1 frequency ratio be equal to 50 dB on the vertical scale. For the reported data to be useful it is important that the X-Y proportions and the horizontal and vertical scaling of the graphs be standardized in this manner.
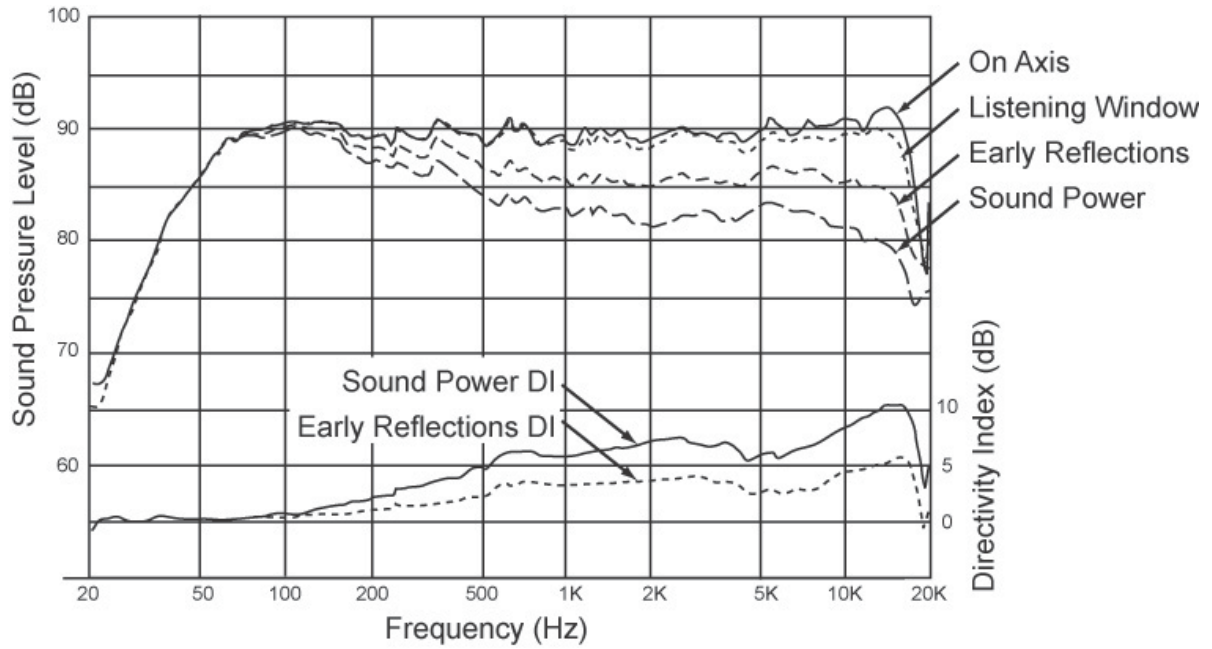
Figure 4: Standard format for reporting data, and sample data

The curves on the graph may be identified by color coding and/or by various dot-dash combinations for black and white displays. If color coding of the curves is used then the colors must be available in any printed or electronic versions made available for public consumption.

6 Directivity Response – In-Wall, In-Ceiling, On-Wall and On-Ceiling Loudspeakers

The device under test shall be tested in an anechoic, or hemi-anechoic, chamber. It shall also be mounted in (for in-wall/in-ceiling DUTs) or on (for on-wall/on-ceiling DUTs), a rigid surface that represents a wall, ceiling or floor to yield a 2 π measurement space, or in a baffle. It may be easier to extend the size of the rigid surface if it is laid horizontally on the floor of the test chamber. If the manufacturer specifies a particular rear box design then this rear box shall be installed during testing, otherwise the cavity behind the device under test shall be no larger than 40 liters, which approximates the cavity between the studs in a typical wall. If a baffle is used it must be at least 1.98 m high by 1.98 m wide, and the device under test shall be mounted in the center of the baffle.

The microphone used for taking measurements shall be 2 m from the device under test.

All of the requirements regarding the measurement environment, measurement apparatus, measurement distance and input signal described in Section 5.1 also apply here.

With the device under test mounted as described above, one of the following measurement methods shall be used for collecting data.

6.1 Data Acquisition – Stationary Microphone

Record complete (180°) horizontal and vertical polar measurements at 10° increments. These measurements shall be taken by rotating the turntable and DUT in a 180° rotation to acquire the horizontal data. The DUT is then rotated 90° about its reference axis. Another 180° rotation of the turntable and DUT is made to acquire the vertical data. The measurement microphone remains stationary during this test.

6.2 Data Acquisition – Rotating Microphone

Record the front hemispherical (180°) horizontal and vertical polar measurements at ten degree increments. These measurements shall be taken by moving the measurement microphone from one angular position to the next in sequence around the vertical axis as shown in Figure 1 until the front 180° have been covered. Then by moving the measurement microphone from one angular position to the next in sequence around the horizontal axis as shown in Figure 1 until the front 180° have been covered.

Instead of moving a single microphone it may be preferable to use 37 separate stationary microphones (note that the microphone directly in front of the DUT would be used twice, once for each axis). The DUT remains stationary during this test.

Also, instead of moving the microphone around the horizontal axis it may be preferable to rotate the DUT 90° about its reference axis and repeat the original rotation of the microphone about the vertical axis to obtain the required measurements.

6.3 Data Presentation

Data shall be presented using the method described in Section 5.3.

7 Frequency Response and Crossover Category

The frequency response of a loudspeaker shall be measured in order to determine how low in frequency it may be used. This should be helpful in determining if a particular loudspeaker can be recommended for use with a particular subwoofer system, or vice-versa. This assumes that a similar high frequency limit is known for the subwoofer. Four *Crossover Categories* are introduced to simplify and assist with this matching process.

7.1 Data Acquisition

The loudspeaker (DUT) shall be loaded in a manner similar to that of normal operating conditions.

If there are any adjustable high pass or low pass filters on the DUT the high pass filters shall be set to the highest frequency possible. The low pass filters shall be set to the lowest frequency possible. If the roll-off rate of these filters is adjustable it shall be set to the slowest roll-off rate possible.

A transfer function measurement with a frequency resolution of at least 1/12 octave within the pass band limits of the DUT shall be made. The measurement shall cover the entire operating bandwidth of the DUT.

7.2 Upper Frequency Limit

The level of the transfer function measurement shall be averaged over the frequency range from 100 Hz to 1 kHz. This level shall be the Average Level of the DUT.

The frequency at which the transfer function has a level of no less than the Average Level minus 3 dB shall be the Lower Frequency Limit of the DUT.

7.3 Crossover Category Qualification

The DUT shall be qualified for each of four crossover categories W, X, Y, and Z. The frequency limit for each of these categories is shown in Table 1.

Table 1: Lower frequency limit of Crossover Categories

Crossover Category			
W	**X**	**Y**	**Z**
80 Hz	100 Hz	120 Hz	150 Hz

If the Lower Frequency Limit of the DUT is lower than the frequency of a crossover category it qualifies for that crossover category.

As an example, a DUT with a Lower Frequency Limit of 90 Hz would qualify for crossover categories X, Y, and Z. It would not qualify for crossover category W.

7.4 Crossover Category Data Presentation

The crossover categories for a loudspeaker shall be reported in the following format:

Crossover Categories: W, X, Y, and/or Z

Where W, X, Y, and Z represents each of the categories for which the DUT qualifies.

The following statement shall also accompany the report.
"Consult the manufacturers recommended settings for optimizing the response of their products"

For the example given in Section 7.3 the required reporting format would state the following.

Crossover Categories: X Y Z
Consult the manufacturers recommended settings for optimizing the response of their products

8 On-Axis Maximum Usable Sound Pressure Level

8.1 On-Axis Maximum Usable Sound Pressure Level – Continuous

8.1.1 Input Signal

The input signal shall be weighted pink noise with a 12 dB crest factor. Its power spectrum shall be as described in Table 2 when measured with one-third octave filters in accordance with IEC Publication 225. This signal may be generated by feeding a pink noise source through the filter processing shown in Figure 6.

Table 2: Power spectrum of input signal

Frequency (Hz)	Relative level (dB)	Tolerance (dB)		Frequency (Hz)	Relative level (dB)	Tolerance (dB)
5	- 24.1	± 2.0		400	0	± 0.5
6.3	- 20.1	± 2.0		500	0	± 0.5
8	- 16.0	± 2.0		630	- 0.1	± 0.5
10	- 12.2	± 2.0		800	- 0.1	± 0.5
12.5	- 8.8	± 1.0		1,000	- 0.2	± 0.6
16	- 5.4	± 1.0		1,250	- 0.4	± 0.7
20	- 3.0	± 1.0		1,600	- 0.6	± 0.8
25	- 1.5	± 0.8		2,000	- 1.0	± 1.0
32	- 0.6	± 0.6		2,500	- 1.5	± 1.0
40	- 0.3	± 0.5		3,150	- 2.3	± 1.0
50	- 0.1	± 0.5		4,000	- 3.5	± 1.0
63	0	± 0.5		5,000	- 4.9	± 1.0
80	0	± 0.5		6,300	- 6.8	± 1.0
100	0	± 0.5		8,000	- 9.1	± 1.0
125	0	± 0.5		10,000	- 11.8	± 1.0
160	0	± 0.5		12,500	- 14.9	± 1.5
200	0	± 0.5		16,000	- 18.2	± 2.0
250	0	± 0.5		20,000	- 21.8	± 3.0
315	0	± 0.5				
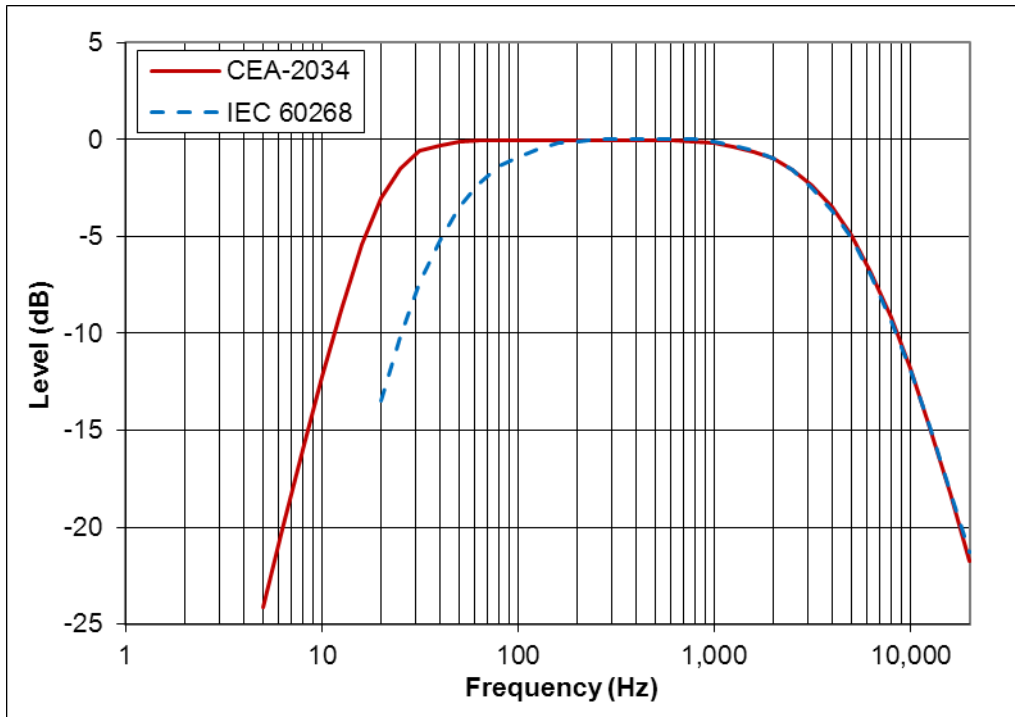
Figure 5: Spectral content of CEA-2034 noise compared to IEC 60268 shaped noise
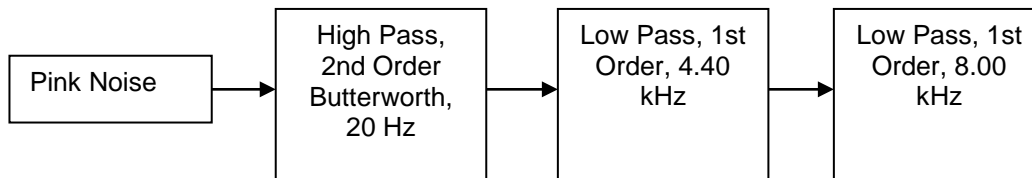
Figure 6: Filter processing blocks for generating input signal

Measure the crest factor (CF) of the input signal to the power amplifier and note the value. The crest factor shall be within the range 11 dB to 13 dB. If the rms voltage output of the power amplifier is operating close to or in excess of 1/8 of its maximum rms output voltage it may clip the output signal, resulting in a lower crest factor. Therefore it is important to measure the crest factor at the input to the DUT, and note its value, at each power level used. It is important to ensure that the amplifier used for testing is capable of cleanly passing the input signal without misbehavior. Misbehavior can best be observed with an oscilloscope - look for unusual spiking, seemingly extended periods of clipping, or mutes. These can often be heard, if the sound of the DUT is audible. Typical crest factors at the input DUT should be between 9 dB and 13 dB. A loudspeaker is a high pass filter and its acceleration, and therefore its SPL, may generally have a crest factor a bit higher than the crest factor of the input signal.

8.1.2 Data Acquisition

8.1.2.1 Measurement Set-up
The loudspeaker (DUT) shall be loaded in a manner similar to that of normal operating conditions. The test signal to be used shall be the input signal specified in Section 8.1.1.

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

The signal flow for the measurement shall be that shown in Figure 7. The output of the signal source shall be connected to the input of a level control device as a means to increase or decrease the amplitude of the signal during the test. The output of the level control shall be connected to the input of a power amplifier. The power amplifier will drive the DUT.

A true rms responding voltmeter shall be connected across the output of the power amplifier to measure the signal amplitude while testing the DUT.

The measurement microphone does not have to be in the far-field of the DUT. However, the distance from the DUT to the measurement microphone should be sufficient so that the magnitude of the radiation from one area of the DUT is not overly emphasized compared to the other areas. In no case shall the measurement microphone be less than 2 m from the DUT.
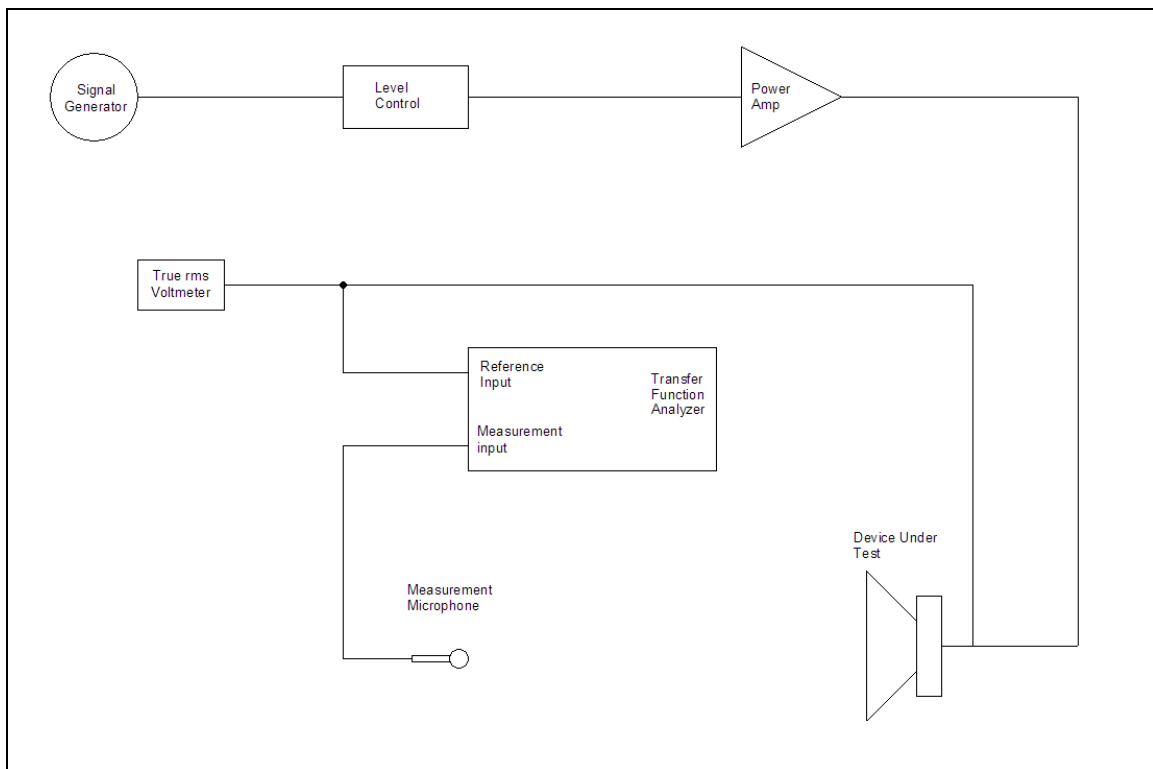
Figure 7: Signal Flow Diagram

8.1.2.2 Measurement Procedure

The level control shall be adjusted so that the broadband rms voltage of the test signal at the input of the DUT is 1.0 V. A transfer function measurement shall be recorded and stored. This measurement shall be used as the Normalization Reference for all subsequent measurements.

By normalizing this to the 1.0 V reference all other measurements will show a flat Normalized Transfer Function at 0 dB as long as there is no deviation in the amplitude response of the DUT.

The level control should be increased 3 dB and the Normalized Transfer Function examined for at least 1 minute to determine the amount of amplitude change in the output of the DUT. If the Normalized Transfer Function has not changed by more than 1 dB, from 0 dB flat, at any point within the pass band limits of the DUT, the level control should be increased by another 3 dB and the Normalized Transfer Function examined for at least 1 minute. This procedure shall be repeated until the Normalized Transfer Function has changed by at least 1 dB from 0 dB flat at any point within the pass band limits of the DUT. The broadband rms voltage at the input of the DUT shall be recorded.

The level control should be increased 1 dB and the Normalized Transfer Function examined for at least 1 minute to determine the amount of change in the output of the DUT. If the Normalized Transfer Function has not changed by more than 3 dB, from 0 dB flat, at any point within the pass band limits of the DUT, the level control should be increased by another 1 dB and the Normalized Transfer Function examined for at least 1 minute. This procedure shall be repeated until the Normalized Transfer Function has changed by approximately 3 dB (but no more than 3.5 dB) from 0 dB flat at any point within the pass band limits of the DUT. The broadband rms voltage at the input of the DUT shall be recorded. This shall be the Maximum Input Voltage, Continuous (MIV continuous) for the DUT.

If the test is being conducted in an anechoic environment the SPL shall be recorded. This SPL shall be referenced to a distance of 1 meter for free-field conditions and shall be the Maximum Usable Continuous Output SPL (SPLmuco) of the DUT.

The transfer function and the Normalized Transfer Function at this input voltage shall also be recorded. This concludes the test and the test signal may be terminated.

As an alternate method to determine the Maximum Usable Continuous Output SPL (SPLmuco) of the DUT if the test is not being conducted in an anechoic environment one of the following procedures can be used.

Method A
1. Apply an appropriate window to the impulse response recorded at the conclusion of the test. This window shall attenuate the level of sound arrivals from any reflecting surfaces to greater than 40 dB below the level of the direct sound from the DUT.

2. Perform an FFT on the windowed impulse response to obtain the anechoic transfer function.

3. Measure the spectral content and level at the input of the DUT at the conclusion of the test.

4. Multiply the transfer function from step 2 by the spectral content and level from step 3.

5. Compute the broadband SPL of the result of step 4 and reference it to 1 meter to yield a suitably anechoic representation of the DUT as if it was driven at its MIV continuous.

Method B
1. Use gating or windowing to measure the transfer function of the DUT at a relatively low input voltage (such that no compression effects are occurring) and reference it to an rms voltage of 1.0 V.

2. Multiply (in the frequency domain) this transfer function and the Normalized Transfer Function recorded at the conclusion of the test.

3. Multiply the result of step 2 with the normalized (to 1.0 V broadband) spectral content of the Simulated Program test signal.

4. Add the gain level represented by the increased signal level from the MIV continuous to the result of step 3 [Gain Level = 20 * log(MIV)].

5. Compute the broadband SPL of the result of step 4 and reference it to 1 meter to yield a suitably anechoic representation of the DUT as if it was driven at its MIV continuous.

See Appendix D for an example of this procedure.

8.2 On-Axis Maximum Sound Pressure Level – Peak

8.2.1 Input Signal

The input signal shall be 1/3 octave band-limited tone bursts. This signal can be generated by applying a cosine weighting function to allow only 6.5 cycles of a sine wave with a frequency of the center frequency of the 1/3 octave band to be tested.

If the power amplifier is operating near its maximum level it will likely clip the signal, adding distortion to the signal, therefore it is important to ensure that the amplifier is not clipping. If clipping occurs it may detrimentally affect the results of this test, therefore it is wise to use a sufficiently large power amplifier. The signal at the input to the DUT should not have distortion in excess of 0.5 % THD+N.

8.2.2 Data Acquisition

This test may be conducted in a calibrated anechoic chamber, in a suitable ground plane environment or in a large calibrated room. (See Appendix B for information about room correction for smaller than required measurement spaces.)

An external power amplifier used for this test should exhibit no individual harmonic distortion components greater than 10 dB lower than those specified in Table 2, at the voltage levels required for the DUT. This will help to assure that the power amplifier does not adversely affect the results of this test. The test procedure in this section can be conducted on the power amplifier alone to validate it as a suitable amplifier to use for this test.

The DUT shall be tested at ISO 1/3 octave center frequencies within the range 20 Hz to 5 kHz. See Table 2 in Section 8.1.1 for a list of frequencies within this range.

With the input of the DUT connected to an amplifier, or with its input shorted to ground, and without any input signal presented to the DUT, its volume control shall be set to the maximum position and the sound pressure level from the DUT and the background noise of the environment and test equipment shall be measured. If the smoothed root mean square (rms) slow sound pressure level (SPL) (i.e., the rms SPL determined using a slow integration time and then 1/12 octave smoothed) exceeds 50 dB (flat/unweighted) at any test frequency (even though no input signal is present) the DUT, test environment and equipment does not comply with CEA-2034, and the reporting method described in Section 8.3 shall not be used for the DUT. Also, no reporting method that might be confused with the reporting method described in Section 8.3 shall be used for the DUT.

For each test frequency, a 1/3 octave band limited tone burst centered at the test frequency shall be fed to the DUT. The SPL at the tone-burst test frequency f_1 (fundamental) and over the frequency range bounded by 1.26 * f_1 and 20.0 kHz, where harmonic (and non-harmonic) distortion and noise (HD+N) occurs, shall be monitored. The signal level presented to the DUT shall be increased until the peak SPL of a 1/12 octave smoothed individual component of HD+N exceeds the level specified in Table 3 and illustrated in Figure 8. The signal level presented to the DUT shall then be decreased to just below the occurrence of the above set of conditions. The peak SPL of the fundamental shall then be recorded. That is, the peak SPL is the level of the highest peak sound pressure within the 1/3 octave of the tone burst stimulus fundamental frequency range, f_1.

Time windowing the output signal of the DUT is allowed. However, the window applied to the output signal from the DUT shall not be shorter than twice the length of the test signal itself. Additionally, the window shall not attenuate any data in the time domain between the initial output of the stimulus and twice the period of the stimulus.

This SPL shall be referenced to a one meter free-field level. If the SPL of the fundamental cannot be recorded due to low output signal level or low signal-to-noise ratio using a single burst tone, averaging techniques shall be used to increase the signal-to-noise ratio of the measurement. Averaging shall be done in the time domain, or vector averaging shall be done in the frequency domain, prior to spectral smoothing. Inability to measure the fundamental output is most likely to occur at the lowest frequencies. If it is not possible to measure the peak SPL at a particular frequency band a peak SPL of 0 dB shall be recorded for that frequency band.

This procedure shall be repeated for each test frequency. The recorded SPL shall be repeatable to within ± 0.5 dB.

Some loudspeakers never surpass the HD+N threshold but begin to limit and/or compress the output at some frequencies. If this is the case the peak SPL of the fundamental at the test frequency shall be entered.

Note: In developing the above test procedure a goal was to take into account the fact that higher frequency noise that is not harmonically related to the fundamental can have a negative impact on perceived loudspeaker performance. It is assumed that the higher in frequency the noise, the more annoying it will be to the listener. It is also assumed that noise that is closer in frequency to the fundamental will be more easily masked within the human hearing system, and can therefore be somewhat higher in level and still be acceptable.

Table 3: Peak SPL Limits

Attenuation of harmonics for fundamentals of 20 Hz to 50 Hz	Attenuation of harmonics for fundamentals of 63 Hz to 200 Hz	Attenuation of harmonics for fundamentals of 250 Hz to 5,000 Hz	Lower Frequency Limit (Hz)	Upper Frequency Limit (Hz) (not to exceed 20 kHz)[1]	Notes
0 dB	0 dB	0 dB	16	1.26 x f_1	Bandwidth from 16 Hz to 1/3 octave above fundamental
- 12 dB	- 24 dB	- 33 dB	0.8909 x f_2	1.1225 x f_2	Bandwidth from 1/6 octave below to 1/6 octave above 2nd harmonic

- 18 dB	- 30 dB	- 42 dB	$0.8909 \times f_3$	$1.1225 \times f_3$	Bandwidth from 1/6 octave below to 1/6 octave above 3rd harmonic
- 21 dB	- 35 dB	- 48 dB	$0.8909 \times f_4$	$1.1225 \times f_4$	Bandwidth from 1/6 octave below to 1/6 octave above 4th harmonic
- 23 dB	- 40 dB	- 53 dB	$0.9438 \times f_5$	$1.0595 \times f_5$	Bandwidth from 1/12 octave below to 1/12 octave above 5th harmonic
- 25 dB	- 44 dB	- 57 dB	$0.9438 \times f_6$	$1.0595 \times f_6$	Bandwidth from 1/12 octave below to 1/12 octave above 6th harmonic
- 27 dB	- 48 dB	- 60 dB	$0.9438 \times f_7$	$1.0595 \times f_7$	Bandwidth from 1/12 octave below to 1/12 octave above 7th harmonic
- 29 dB	- 52 dB	- 63 dB	$0.9438 \times f_8$	$1.0595 \times f_8$	Bandwidth from 1/12 octave below to 1/12 octave above 8th harmonic
- 31 dB	- 55 dB	- 65 dB	$0.9438 \times f_9$	$1.0595 \times f_9$	Bandwidth from 1/12 octave below to 1/12 octave above 9th harmonic
- 33 dB	- 58 dB	- 67 dB	$0.9438 \times f_{10}$	$1.0595 \times f_{10}$	Bandwidth from 1/12 octave below to 1/12 octave above 10th harmonic
- 35 dB	- 60 dB	- 69 dB	$0.9715 \times f_{11}$	$1.0293 \times f_{11}$	Bandwidth from 1/24 octave below to 1/24 octave above 11th harmonic
- 37 dB	- 62 dB	- 70 dB	$0.9715 \times f_{12}$	$1.0293 \times f_{12}$	Bandwidth from 1/24 octave below to 1/24 octave above 12th harmonic

- 38 dB	- 63 dB	- 71 dB	0.9715 x f_{13}	1.0293 x f_{13}	Bandwidth from 1/24 octave below to 1/24 octave above 13th harmonic
- 39 dB	- 64 dB	- 72 dB	0.9715 x f_{14}	1.0293 x f_{14}	Bandwidth from 1/24 octave below to 1/24 octave above 14th harmonic
- 40 dB	- 65 dB	- 73 dB	0.9715 x f_{15}	1.0293 x f_{15}	Bandwidth from 1/24 octave below to 1/24 octave above 15th harmonic

[1]The upper frequency for analysis for all fundamental frequencies and harmonics shall be that specified in Table 3 or 20 kHz, whichever is lower.
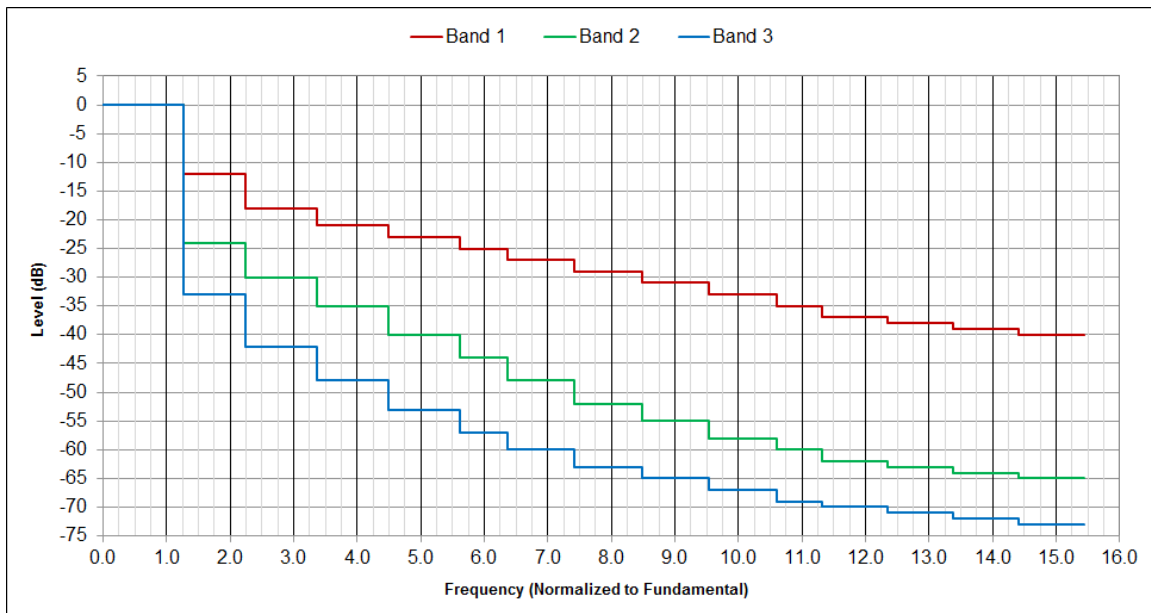
Figure 8: HD+Noise Threshold vs. Frequency

8.2.3 Max Peak SPL Data Post-Processing

1. The recorded peak SPL at each 1/3 octave band shall be weighted by the power spectrum given in Table 2 (8.1.1).
2. The Average Weighted SPL shall be calculated by averaging the weighted SPL values from 100 Hz to 5 kHz, inclusive.
3. The weighted SPL for any 1/3 octave band exceeding the Average Weighted SPL by 2 dB shall be reduced to a value of the Average Weighted SPL plus 2 dB.
4. The weighted & limited SPL in each 1/3 octave band shall be converted to its squared-pressure value ($p^2 = 10^{(SPL/10)}$)
5. The sum of all of the squared-pressure values for each 1/3 octave band shall be calculated.
6. The SPL of the summed squared-pressure shall be calculated ($SPL = 10 * \log(sum(p^2))$).
7. The Broadband Peak SPL shall be the SPL of the summed squared-pressure minus 14 dB.

8.3 Maximum SPL Data Presentation

Subtract the Maximum Usable Continuous Output SPL (Section 8.1.2.2) from the Broadband Peak SPL (Section 8.2.3). If the result is 9 dB or greater, the Maximum Usable Continuous Output SPL shall be reported as the Maximum SPL of the DUT. If the result is less than 9 dB the reported Maximum SPL shall be the Broadband Peak SPL minus 9 dB.

The computed broadband Maximum SPL for the DUT shall be reported using the following format:

Maximum Sound Pressure Level: XX dB SPL referenced to 1 m.

The information presented to the consumer shall include the following statement in a prominent location:

CAUTION: High sound levels can cause hearing damage with prolonged exposure. Wear appropriate protection such as ear plugs when listening to loud audio.

9 Powered Loudspeaker Rating

The loudspeaker shall be rated in four different listening levels. This calculation shall use the maximum usable continuous SPL and the voltage required to achieve this SPL in Section 8.1 as its basis. The SPL shall be computed using the following assumptions.

1. Two loudspeakers in a listening room each reproducing uncorrelated signals of approximately the same level.
2. The listening position shall be at 4 meters from the loudspeakers.
3. The listening room is partially reflective so there is a "room gain" of 6 dB compared to the direct sound only.

The four listening levels and their corresponding average SPL shall be:

Listening Level[1]	Required Target Loudspeaker System SPL
Quiet	75 dB
Moderate	85 dB
Loud	95 dB
Very Loud	105 dB

It is possible that some of these listening levels may not be attainable with some loudspeakers.

9.1 Calculation

Calculate the Maximum In-Room SPL at a distance of 4 meters from two loudspeakers in a room. For this calculation use the Maximum SPL at 1 m from Section 8.3.

$$SPL_{In-Room} = MaxSPL + 3 \text{ dB} + 6 \text{ dB} - 12 \text{ dB}$$

9.2 Data Presentation

The results shall be reported in the following format.

Listening Levels: Quiet, Moderate, Loud, and/or Very Loud

Where Quiet, Moderate, Loud, and Very Loud represents each of the categories for which the DUT qualifies.

In addition to the above required reporting presentation, an optional expanded reporting may be published. This should include the format detailed in the table below.

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[1] This standard provides a method for measuring loudspeaker output signals at different signal levels at a distance of 4 m. The provision of a method for measuring loudspeaker outputs at high levels shall not be interpreted as a recommendation that people listen to audio at these high levels. Some loudspeaker applications naturally prevent people from listening at a distance of 4 m from the loudspeaker, such as the case when a loudspeaker is mounted at the top of a tall mast at an outdoor venue. This standard provides a method for testing such loudspeakers, but does not recommend that people listen to audio at dangerously high levels. People conducting tests in accordance with this standard shall take appropriate precautions, including the use of appropriate ear protection, to avoid exposure to high audio levels.

Table 4: Subwoofer In-Room Listening Level

Listening Level	Subwoofer Required SPL at 4 m	Full-Range Required SPL at 4 m	Full-Range Loudspeaker Recommended for this SPL
Quiet	85 dB	75 dB	Yes
Moderate	95 dB	85 dB	Yes
Loud	105 dB	95 dB	Yes
Very Loud	115 dB	105 dB	No

* The "No" recommendation indicates a subwoofer system with higher SPL capability should be selected to achieve this Listening Level.

10 Impedance (Passive Loudspeakers Only)

10.1 Data Acquisition

The impedance of the DUT shall be determined as follows:
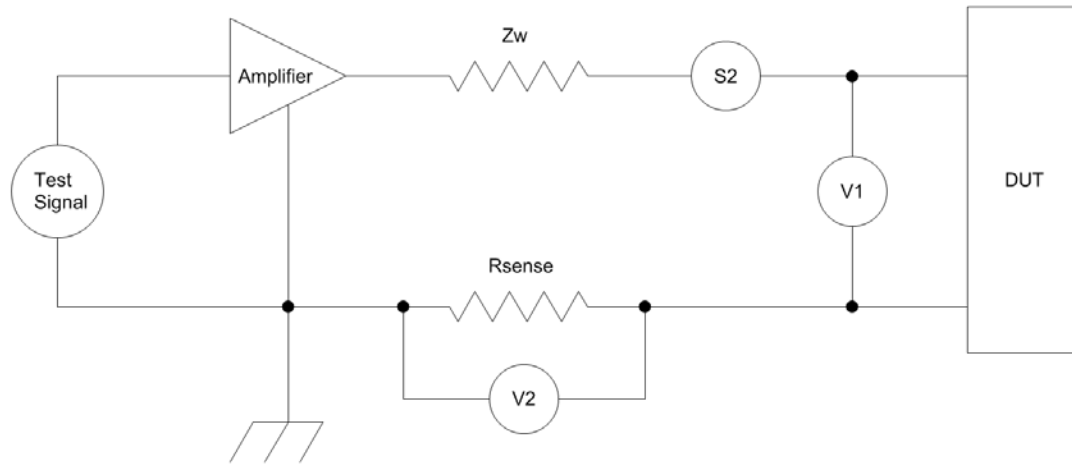
Figure 9: Impedance measurement schematic, R_{Sense} method

OR
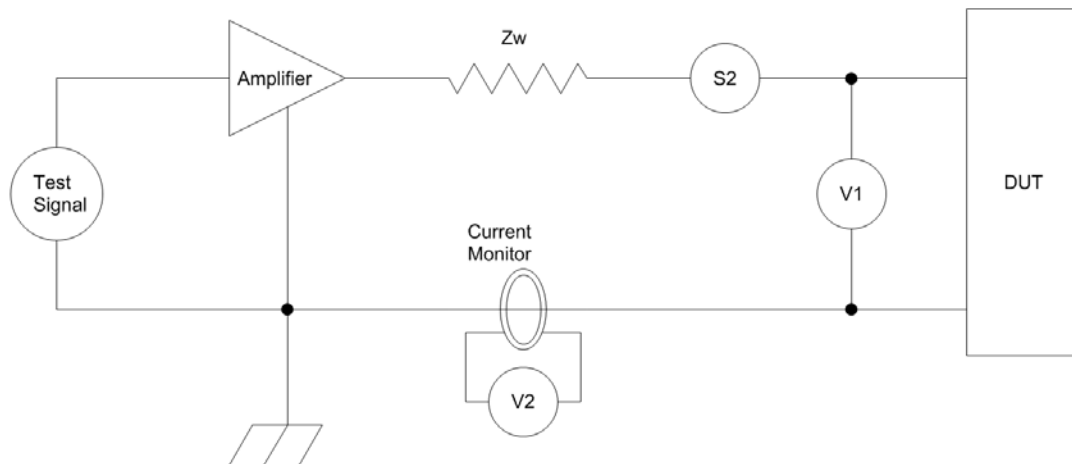
Figure 10: Impedance measurement schematic, current monitor method

The voltmeters, especially V2 across the R_{Sense} resistor or the current monitor, shall have high resolution and high sensitivity, such as 5.5 digits or greater with sensitivity in the microvolt range. Both voltmeters must have flat voltage as a function of frequency response in the 20 Hz to 20 kHz range. Additionally, both voltmeters shall be able to derive complex data with respect to each other, such as a dual-channel FFT measurement system. Use a low inductance sense resistor, tested to have insignificant inductance in the 20 Hz to 20 kHz range. Resistors are available with a 4-wire Kelvin configuration for better accuracy. Zw is any impedance of any wire(s) within the measurement loop. It should not be a part of the measured speaker impedance and shall be kept to a value no greater than 5 % of the Rdc of the DUT. The measurement error as a result of Zw

can be minimized by measuring V1 directly at the speaker cabinet terminals. S2 is any voltage due to back emf, thermal noise and the speaker acting as a microphone.

Before applying any desired signal, record the voltmeter values.

Apply the desired signal (discrete spot frequencies, pink noise, swept sine, etc.) such that the signal-to-noise ratio is at least 40 dB. Verify the speaker is not driven into a non-linear range. This can be verified by testing at several drive levels above and below the minimum drive level determined to provide the 40 dB signal-to-noise ratio, and comparing the impedance curves to see if they change. If the impedance curves are different then measurements at lower drive levels need to be made to determine the highest drive level that can be used in the linear range of the DUT. If this drive level does not allow at least 40 dB of signal-to-noise ratio, appropriate measures need to be taken (such as averaging) to ensure at least 40 dB of signal-to-noise ratio without increasing the drive level.

Measure the voltages, V1 and V2
Calculate the current as $I = V2 / R_{Sense}$ or determine I from current monitor
Calculate complex impedance as V1 / I

A single frequency shall be reported as the minimum impedance frequency. The impedance measurement shall not be smoothed to any greater than 1/12 of an octave. The minimum impedance frequency reported shall be within the range 10 Hz to 20 kHz.

If the same minimum impedance occurs at multiple frequencies within the range of 30 Hz to 2 kHz, the frequency of minimum impedance with the highest absolute value of the angle of impedance shall be reported as the minimum impedance frequency.

If the minimum impedance occurs at multiple frequencies outside of the range of 30 Hz to 2 kHz, and there are no occurrences of the minimum impedance within the range of 30 Hz to 2 kHz, the frequency of minimum impedance closest to the range of 30 Hz to 2 kHz shall be reported as the minimum impedance frequency.

Practical measurements of complex impedance will typically be performed with computer controlled analog or digital 2 channel analyzers, including some that use high quality 2 channel sound boards with appropriate commercially available software.

Any measurement system setup must be checked for accuracy by measuring a known value non-inductive resistor, which should have a flat resistance magnitude in Ohms and a flat phase response of zero degrees. Special auxiliary software such as Macros are provided by some manufacturers for use with their instruments to perform the complex mathematical calculations to obtain phase as well as magnitude, and to scale the impedance to Ohms using the value chosen for the current sense resistor, typically 0.100 ohms.

10.2 Data Presentation

The rated and minimum impedance for the DUT shall be reported using the following format:

Rated Impedance: x ohms
Minimum Impedance: y.y ohms at zzz Hz (or z.z kHz)
The rated impedance shall be given as an integer number.

The minimum impedance shall be reported to the nearest one-tenth ohm.

The frequency for the minimum impedance shall be reported as an integer value when the units are given in Hz or to the nearest one-tenth kilohertz when the units are given in kHz.

11 Required Power Amplifier Size (Passive Loudspeakers Only)

The required power amplifier size shall be calculated for an average SPL in four different listening levels. This calculation shall use the maximum usable continuous SPL and the voltage required to achieve this SPL in Section 8.1 as its basis. The SPL shall be computed using the following assumptions.

4. Two loudspeakers in a listening room each reproducing uncorrelated signals of approximately the same level.
5. The listening position shall be at 4 meters from the loudspeakers.
6. The listening room is partially reflective so there is a "room gain" of 6 dB compared to the direct sound only.

The four listening levels and their corresponding average SPL shall be:

Listening Level[2] Target SPL
Quiet   75 dB
Moderate   85 dB
Loud   95 dB
Very Loud   105 dB

It is possible that some of these listening levels may not be attainable with some loudspeakers.

11.1 Calculation

Prior to calculating the required amplifier power, the Adjusted Maximum Input Voltage should be confirmed. If the Maximum SPL is the same as the SPL_{MUCO} (Maximum Usable Continuous Output) the Adjusted MIV shall be the same as the measured MIV. If the Maximum SPL is different than the SPL_{MUCO}, the Adjusted MIV level shall be the MIV level minus the difference between the SPL_{MUCO} and the Maximum SPL

MIV Level = 20 * log(MIV)

Adjusted MIV Level = MIV Level – (SPL_{MUCO} – Maximum SPL)

Adjusted MIV = 10^(Adjusted MIV Level/20)

The following steps shall be used to calculate the required amplifier power.

1. Calculate the Maximum In-Room SPL at a distance of 4 meters from two loudspeakers in a room. For this calculation use the Maximum SPL at 1 m from section 8.3
$SPL_{In-Room}$ = MaxSPL + 3 dB + 6 dB – 12 dB

2. Subtract the $SPL_{In-Room}$ from the Target SPL to get the Required Gain

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[2] This standard provides a method for measuring loudspeaker output signals at different signal levels at a distance of 4 m. The provision of a method for measuring loudspeaker outputs at high levels shall not be interpreted as a recommendation that people listen to audio at these high levels. Some loudspeaker applications naturally prevent people from listening at a distance of 4 m from the loudspeaker, such as the case when a loudspeaker is mounted at the top of a tall mast at an outdoor venue. This standard provides a method for testing such loudspeakers, but does not recommend that people listen to audio at dangerously high levels. People conducting tests in accordance with this standard shall take appropriate precautions, including the use of appropriate ear protection, to avoid exposure to high audio levels.

Required Gain = Target SPL – SPL$_{In-Room}$

3. Calculate the Required Voltage (rms) using the Adjusted MIV and the Required Gain.
Required E$_{rms}$ = AdjustedMIV * 10^(Required Gain / 20)

4. Calculate the Required Voltage (peak) by multiplying the Required Voltage by 4 to account for a 12 dB crest factor in the program material.
Required E$_{peak}$ = 4 * Required E$_{rms}$

5. Calculate the Equivalent Sine Voltage (rms) for the Required Voltage (peak)
Equivalent Sine E$_{rms}$ = Required E$_{peak}$ / 10^(3/20)

6. Calculate the Required Power for each amplifier channel by dividing the square of the Equivalent Sine Voltage (rms) by the rated impedance of the loudspeaker.
Required Power = (Equivalent Sine E$_{rms}$)^2 / Z$_{Rated}$

11.2 Data Presentation

The results shall be reported in the following format.

The value of the Required Power shall be reported as an integer number.

Table 5: Required Amplifier Power

Listening Level	Subwoofer Required SPL at 4 m	Full-Range Required SPL at 4 m	Full-Range Loudspeaker Recommended for this SPL	Power Amplifier Required at 8 Ω	Amplifier Voltage (rms) Required	Amplifier Voltage (peak) Required
Quiet	85 dB	75 dB	Yes	2 W	4 V	5 V
Moderate	95 dB	85 dB	Yes	16 W	11 V	16 V
Loud	105 dB	95 dB	Yes	158 W	36 V	50 V
Very Loud	115 dB	105 dB	No	N/A	N/A	N/A

* The "No" recommendation indicates a loudspeaker system with higher SPL capability should be selected to achieve this Listening Level.

The following table may be used to modify the desired listening level based on the size and the acoustics of the listening room. Smaller and more reflective rooms may require less SPL from the loudspeaker to achieve the desired listening level. The numbers in the table indicate to move up or down 0, 1, or 2 listening levels from the initial desired listening level. This modified listening level should then show the required power amplifier size for use with a given loudspeaker system.

Table 6: Listening Level Adjustment Based on Room Acoustics and Size

	Room Acoustics	Hard - Very Reflective (marble, tile, glass, etc.)	Normal - Average Décor & Furnishings	Soft - Very Absorptive (lots of carpet, heavy drapes, large plush furniture, pillows)
Room Size	**Large** (≥600 ft²)	0	1	2
	Medium	-1	0	1
	Small (≤200 ft²)	-2	-1	0

Note: This assumes an 8-9 ft average ceiling height

Here is an example of how Table 5 and Table 6 may be used together:

If a loudspeaker requires 158 W to achieve a Loud listening level, but the loudspeaker will be used in a Small, Hard-Very Reflective room, then the recommended amplifier power to achieve the Loud listening level may be adjusted down by two listening levels (-2), which results in a recommended amplifier power of 2 W according to Table 5.

12 Matching a Subwoofer to a Full-Range Loudspeaker System

A subwoofer would be a good candidate for matching with a full-range loudspeaker system when the following occur.

- Both the subwoofer and the full-range loudspeaker system have at least one overlapping Crossover Category in common. That is, they share one or more of the same Crossover Categories (see Section 7).
- Both the subwoofer and the full-range loudspeaker system have the desired Listening Level in common (see Section 9 or Section 11).

As an example, a subwoofer and full-range system with the following ratings for use in the Moderate listening level would be good candidates to be matched with each other.

Subwoofer
Crossover Categories: W X Y
Listening Levels: Quiet and Moderate

Full-Range
Crossover Categories: X Y Z
Listening Levels: Quiet, Moderate, and Loud

Alternatively, a subwoofer and full-range system with the following ratings for use in the Loud listening level would *not* be good candidates to be matched with each other. Note also that they have no crossover categories in common.

Subwoofer
Crossover Categories: W X
Listening Levels: Quiet and Moderate

Full-Range
Crossover Categories: Y Z
Listening Levels: Quiet, Moderate, and Loud

13 Estimating In-Room Response from Anechoic Data

The Estimated In-Room Response shall be calculated using the directivity data acquired in Section 5 or Section 6. It shall be comprised of a weighted average of 12 % Listening Window, 44 %, Early Reflections, and 44 % Sound Power. The sound pressure levels shall be converted to squared pressure values prior to the weighting and summation. After the weightings have been applied and the squared pressure values summed they shall be converted back to sound pressure levels.

The Estimated In-Room Response shall be presented using the method described in Section 5.3.
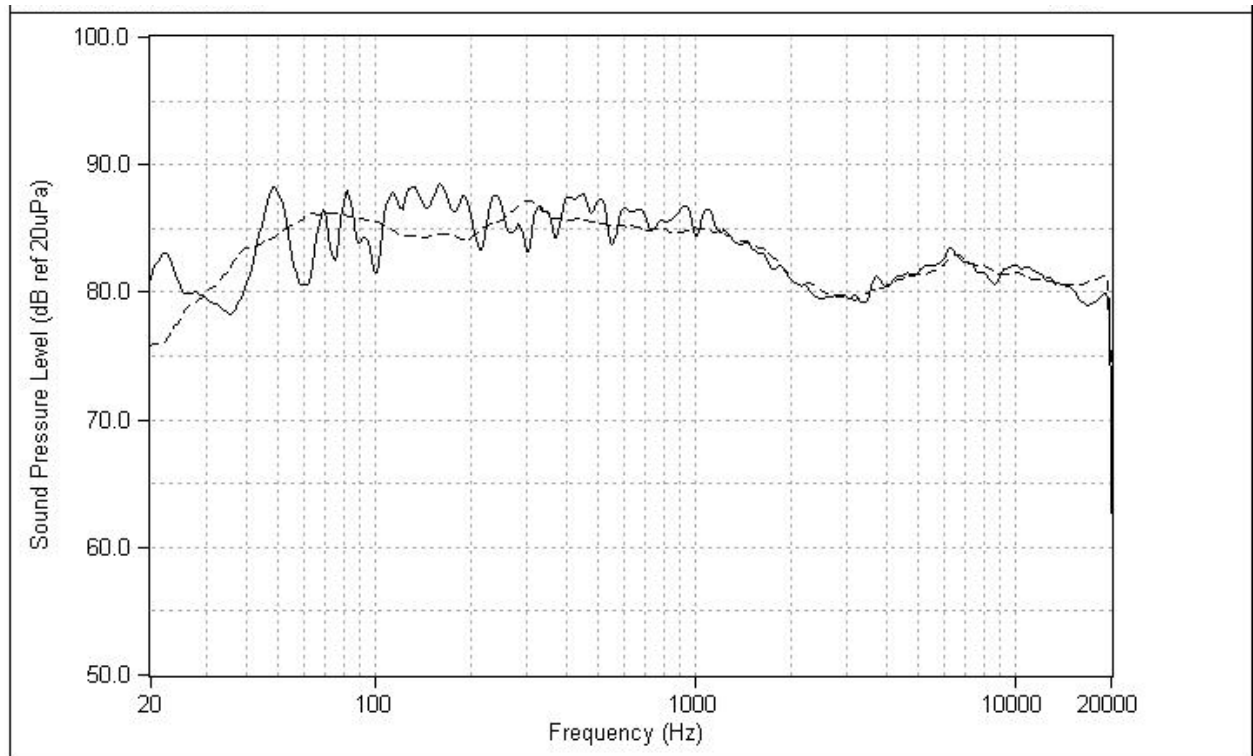
Figure 11: In-room response of loudspeaker B. Solid: predicted from anechoic data. Dotted: average response in a real room, with a ¼-inch microphone aimed at the ceiling, random-incidence amplitude response corrected to be flat. Note that the predicted curve cannot anticipate the effects of room modes at frequencies below the transition/Schroeder frequency around 300 Hz to 400 Hz.

14 Consolidated Reporting Requirements – Example Data Sheet

This section includes all of the reporting requirements from all of the other sections, and explains how data should be reported to consumers in a combined format.

An example of the consolidated reporting required is shown below. This includes the following.

1.   A multi-curve graph for the Directivity Response
2.   A single-curve graph for the Estimated In-Room Response
3.   The Rated Maximum SPL referenced to 1 meter
4.   *For passive loudspeakers only* – The Rated Impedance and the Minimum Impedance, along with the frequency at which the Minimum Impedance occurs
5.   *For passive loudspeakers only* – A table detailing the Recommended Amplifier Size to achieve the four Listening Levels at 4 meters
6.   The Listening Levels for which the loudspeaker qualifies. Optionally, the expanded reporting of the table (Section 9 or Section 11) may be included.
7.   The Crossover Categories for the loudspeaker

The Rated Maximum SPL shall be an integer number.

The Rated Impedance shall be reported as an integer number.
The Minimum Impedance shall be reported to the nearest one-tenth ohm.
The frequency at which the Minimum Impedance occurs shall be reported as the nearest integer value in hertz for values less than 1,000 Hz or to the nearest one-tenth kilohertz for values of 1,000 Hz or greater.

The values of the Required Power shall be reported as integer numbers.

Example Reporting for a Passive Loudspeaker System
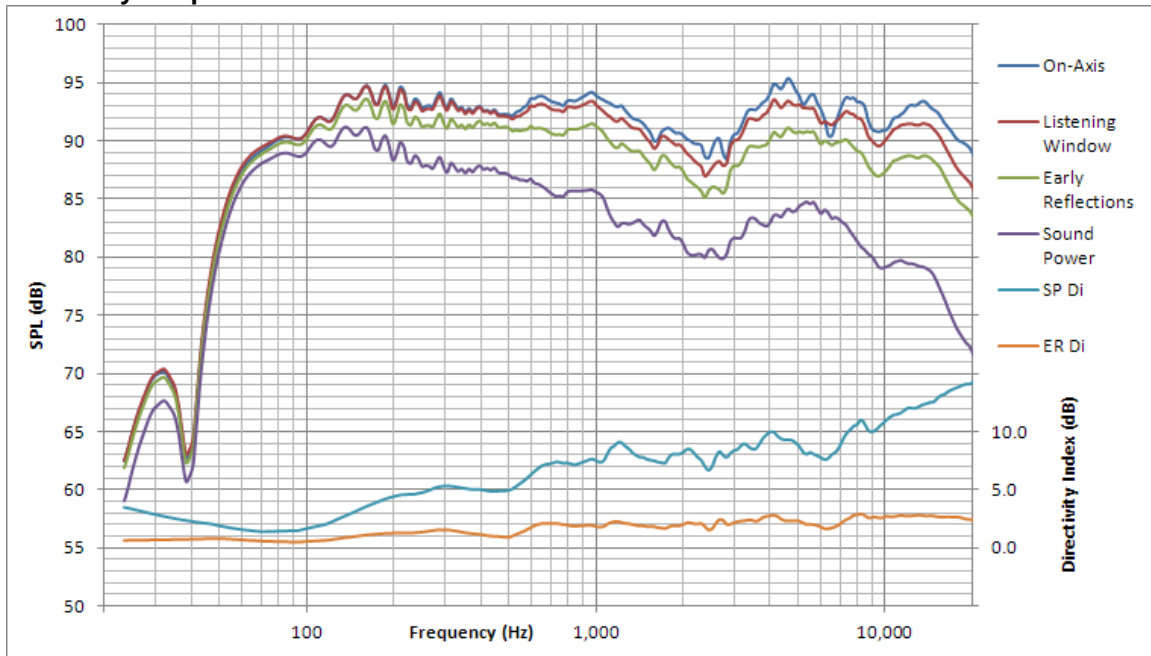
Directivity Response

Estimated In-Room Response
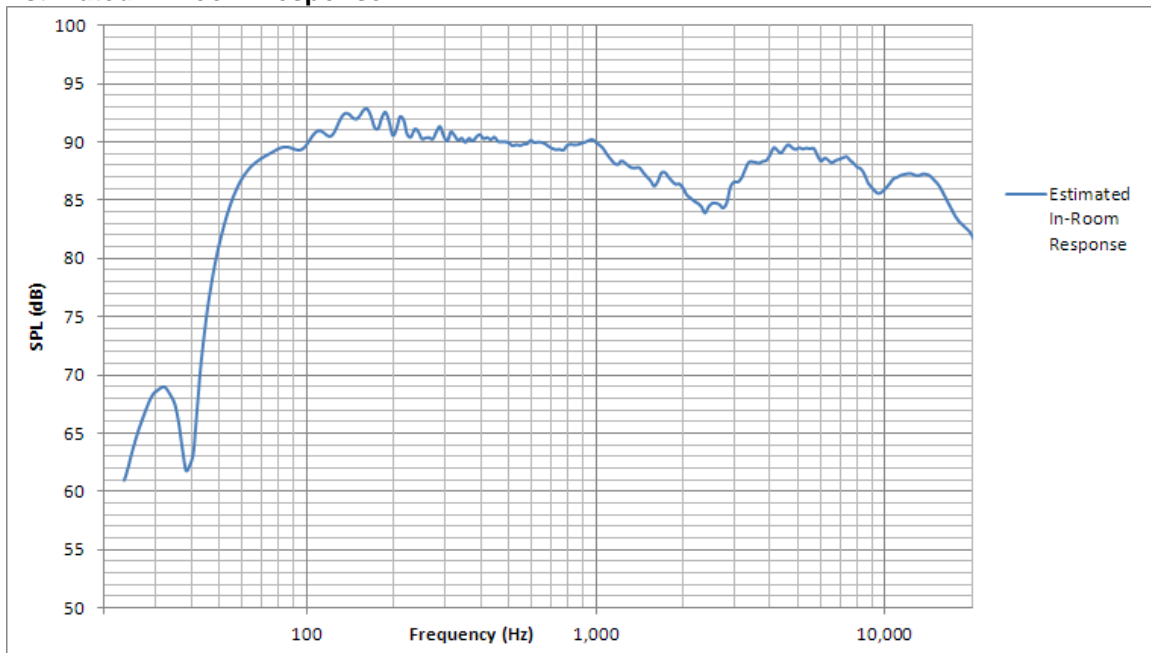
Maximum SPL
Rated Maximum Sound Pressure Level: 102 dB SPL referenced to 1 m

Impedance
Rated Impedance: 8 ohms
Minimum Impedance: 6.7 ohms at 1.8 kHz

Required Amplifier Size

Listening Level	Subwoofer Required SPL at 4 m	Full-Range Required SPL at 4 m	Full-Range Loudspeaker Recommended for this SPL	Power Amplifier Required at 8 Ω	Amplifier Voltage (rms) Required	Amplifier Voltage (peak) Required
Quiet	85 dB	75 dB	Yes	2 W	4 V	5 V
Moderate	95 dB	85 dB	Yes	16 W	11 V	16 V
Loud	105 dB	95 dB	Yes	158 W	36 V	50 V
Very Loud	115 dB	105 dB	No	N/A	N/A	N/A

* The "No" recommendation indicates a loudspeaker system with higher SPL capability should be selected to achieve this Listening Level.

CAUTION: High sound levels can cause hearing damage with prolonged exposure. Wear appropriate protection such as ear plugs when listening to loud audio.

Listening Levels
Quiet, Moderate, and Loud

Crossover Category
Crossover Categories: X Y Z

Consult the manufacturers recommended settings for optimizing the response of their products

Example Reporting for a Powered Loudspeaker System

Directivity Response

Estimated In-Room Response
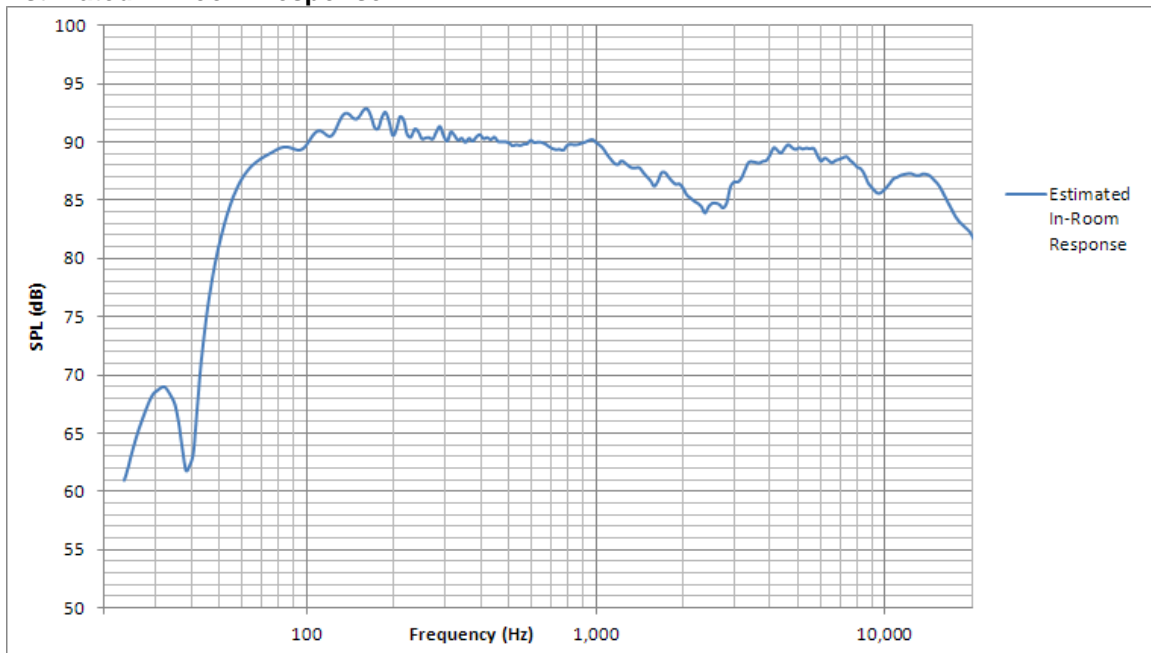
Maximum SPL
Rated Maximum Sound Pressure Level: 102 dB SPL referenced to 1 m

In-Room Listening Level

Listening Level	Subwoofer Required SPL at 4 m	Full-Range Required SPL at 4 m	Full-Range Loudspeaker Recommended for this SPL
Quiet	85 dB	75 dB	Yes
Moderate	95 dB	85 dB	Yes
Loud	105 dB	95 dB	Yes
Very Loud	115 dB	105 dB	No

* The "No" recommendation indicates a loudspeaker system with higher SPL capability should be selected to achieve this Listening Level.

CAUTION: High sound levels can cause hearing damage with prolonged exposure. Wear appropriate protection such as ear plugs when listening to loud audio.

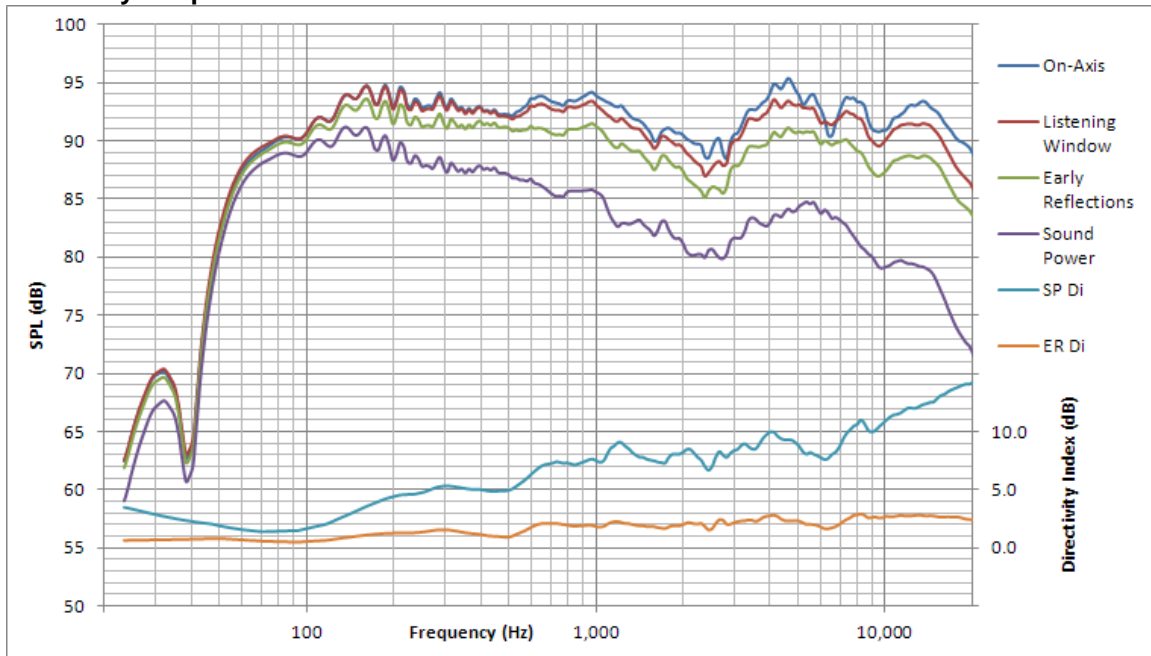**Listening Levels**
Quiet, Moderate, and Loud

Crossover Category
Crossover Categories: X Y Z

Consult the manufacturers recommended settings for optimizing the response of their products

Appendix A.   Data Acquisition in Non-Anechoic and Non-Open Field Spaces

The following techniques are useful in assessing the frequency response of a loudspeaker in the absence of an anechoic chamber. Many of them work well. All measurements have errors, including measurements in anechoic chambers and outdoors, so it is important to validate your own techniques by comparing the results to a measurement made in a free field environment. A good way to do this is to submit one of your speakers to a recognized testing laboratory. An agreement of ± 1.5 dB is considered to be good.

All of these papers are available from the AES Electronic Library.

Quasi-Anechoic Loudspeaker Measurement Using Notch Equalization for Impulse Shortening
        Author: Stroud, Richard
        Affiliation: Stroud Audio Inc., Kokomo, IN, USA
        AES Convention: 129 (November 2010) Paper Number: 8170
        http://www.aes.org/e-lib/browse.cfm?elib=15593

A Study of Low-Frequency Near- and Far-Field Loudspeaker Behavior
        Authors: Vanderkooy, John; Rousseau, Martial
        Affiliations: University of Waterloo, Waterloo, Ontario, Canada; B&W Group Ltd.,
        Steyning, West Sussex, UK(See document for exact affiliation information.)
        AES Convention: 126 (May 2009) Paper Number: 7747
        http://www.aes.org/e-lib/browse.cfm?elib=14943

Can One Perform Quasi-Anechoic Measurements in Normal Rooms?
        Authors: Vanderkooy, John; Lipshitz, Stanley
        Affiliation: University of Waterloo
        AES Convention: 125 (October 2008) Paper Number: 7525
        http://www.aes.org/e-lib/browse.cfm?elib=14677

Comparison of Four Subwoofer Measurement Techniques
        Authors: Melon, Manuel; Langrenne, Christophe; Rousseau, David; Herzog, Philippe
        Affiliations: CNAM Laboratoire d'Acoustique, Paris Cedex, France ; BC Acoustique,
        Alfortville, France ; LMA—CNRS, Marseille Cedex, France*(See document for exact affiliation information.)*
        JAES Volume 55 Issue 12 pp. 1077-1091; December 2007
        http://www.aes.org/e-lib/browse.cfm?elib=14183

Extending Quasi-Anechoic Measurements to Low Frequencies
        Author: Benjamin, Eric
        Affiliation: Dolby Laboratories, San Francisco, CA
        AES Convention: 117 (October 2004) Paper Number: 6218
        http://www.aes.org/e-lib/browse.cfm?elib=12875

Simultaneous Measurement of Impulse Response and Distortion with a Swept-Sine Technique
        Author: Farina, Angelo
        Affiliation: Dipartimento di Ingegneria Industrial, Universita' di Parma, Italy
        AES Convention: 108 (February 2000) Paper Number: 5093
        http://www.aes.org/e-lib/browse.cfm?elib=10211

Reduction of the Room Effects in Loudspeaker Measurements
        Authors: Larsson, Lars; Dalbjörn, Klas; Kleiner, Mendel
        Affiliations: dba audioab, Kungsbacka, Sweden; Chalmers University of Technology,
        Gothenburg, Sweden *(See document for exact affiliation information.)*
        AES Convention: 106 (May 1999) Paper Number: 4967

http://www.aes.org/e-lib/browse.cfm?elib=8213

Simulated Free Field Measurements
      Authors: Struck, Christopher J.; Temme, Steve F.
      Affiliation: Bruel & Kjaer, Naerum, Denmark
      JAES Volume 42 Issue 6 pp. 467-482; June 1994
      http://www.aes.org/e-lib/browse.cfm?elib=6937

Maximum Entropy, Auto Regression, Pole-Zero Modeling-On the Use of Modern Spectral Estimation in Audio Testing
      Author: Geddes, Earl R.
      Affiliation: Ford Motor Co., Dearborn, MI
      AES Convention: 87 (October 1989) Paper Number: 2883
      http://www.aes.org/e-lib/browse.cfm?elib=5811

Predicting Farfield Pressures from Nearfield Loudspeaker Measurements
      Author: Kessel, Ron T.
      Affiliation: Audio Research Group, University of Waterloo, Waterloo, Ontario, Canada
      AES Convention: 85 (November 1988) Paper Number: 2729
      http://www.aes.org/e-lib/browse.cfm?elib=10235

Low Frequency Measurement of Loudspeakers by the Reciprocity Method
      Author: Merhaut, Josef
      Affiliation: Technical University, Prague, Czechoslovakia
      JAES Volume 30 Issue 12 pp. 882-888; December 1982
      http://www.aes.org/e-lib/browse.cfm?elib=3804

Low-Frequency Loudspeaker Assessment by Nearfield Sound-Pressure Measurement
      Author: Keele, Jr., D. B. (Don)
      Affiliation: Electro-Voice, Inc., Buchanan, MI
      JAES Volume 22 Issue 3 pp. 154-162; April 1974
      http://www.aes.org/e-lib/browse.cfm?elib=2774

Appendix B.   Calibrating Amplitude Response of Anechoic Chambers

Use only for on-axis frequency response (sensitivity) measurements. This is not valid for directivity measurements.

Because few acoustic spaces are sufficiently large to permit measurements down to a high-performance DUT's low-frequency cutoff and because room effects are generally important well above that frequency, measurement accuracy and low-frequency extension may be greatly enhanced by applying a known room correction factor (RCF) to data acquired indoors. The acoustic space in which measurements are taken will affect the low frequency response of a loudspeaker. Therefore, the purpose of the RCF is to "remove the room" from the measurement so that only the "true" magnitude response of the device under test (DUT) remains.

The means by which one may derive the RCF for a particular acoustic space and measurement setup are as follows:

1. In accordance with good engineering practices, measure the free field ground plane response of a sealed box subwoofer whose low-frequency extension at least matches or exceeds that of the DUT. If no such low frequency calibration source (LFCS) exists, it may be necessary to design and build one.
    a. For an anechoic measurement space
        Place the measurement microphone on the ground plane in a free field to make the reference measurement for the calibration. The LFCS shall be placed in the free field, above the ground plane at least one-quarter (1/4) wavelength of the lowest frequency to which the calibration will be valid.

    b. For a hemi-anechoic measurement space
        Place the measurement microphone on the ground plane in a free field to make the reference measurement for the calibration. The LFCS shall also be placed on the ground plane in the free field.

2. Place the LFCS and measurement microphone in the measurement room at the intended measurement locations of the DUT and microphone, taking into account proximity of major acoustic reflectors (*e.g.*, walls) to maximize reflection times.

3. Measure the in-room ground plane magnitude response of the LFCS in an accepted manner.

4. Compute the RCF from the dB magnitude difference between results of steps (c) and (a) above.

The RCF need not be derived more than once for a particular measurement space so long as it's major features (*e.g.* placement of reflectors, room absorption) and the measurement set-up (*e.g.* source and receiver location in the room, temperature and placement of major acoustic reflectors) are constant. One simply applies the once- derived RCF to the DUT's "in-room" ground plane magnitude response in order to derive its true magnitude response.

Appendix C.   Sound Pressure Weighting Values

In order to approximate the total sound power radiated from a source by using the measurements at equal angular increments, the measurement at each angular increment must be weighted by the appropriate value. The weighting value corresponds to the area of the spherical quadrangle centered at the microphone position for a particular angular position.

The table below gives the weighting values applicable for horizontal and vertical polar measurements at 10° increments.

Table 7: Weighting Values for Horizontal & Vertical Polars at 10° Increments

Angle	Spherical Quadrangle Weighting	Angle	Spherical Quadrangle Weighting
0°	0.000604486	180°	0.000604486
10°	0.004730189	190°	0.004730189
20°	0.008955027	200°	0.008955027
30°	0.012387354	210°	0.012387354
40°	0.014989611	220°	0.014989611
50°	0.016868154	230°	0.016868154
60°	0.018165962	240°	0.018165962
70°	0.019006744	250°	0.019006744
80°	0.019477787	260°	0.019477787
90°	0.019629373	270°	0.019629373
100°	0.019477787	280°	0.019477787
110°	0.019006744	290°	0.019006744
120°	0.018165962	300°	0.018165962
130°	0.016868154	310°	0.016868154
140°	0.014989611	320°	0.014989611
150°	0.012387354	330°	0.012387354
160°	0.008955027	340°	0.008955027
170°	0.004730189	350°	0.004730189

The table below gives the weighting values applicable for horizontal and vertical polar measurements at 5° increments.

Table 8: Weighting Values for Horizontal & Vertical Polars at 5° Increments

Angle	Spherical Quadrangle Weighting	Angle	Spherical Quadrangle Weighting
0°	0.000151456	180°	0.000151456
5°	0.001208191	185°	0.001208191
10°	0.002397988	190°	0.002397988
15°	0.003550813	195°	0.003550813
20°	0.004650038	200°	0.004650038
25°	0.005677375	205°	0.005677375
30°	0.006616875	210°	0.006616875
35°	0.007453213	215°	0.007453213
40°	0.008172263	220°	0.008172263
45°	0.008761238	225°	0.008761238
50°	0.009208863	230°	0.009208863
55°	0.009505500	235°	0.009505500
60°	0.009643288	240°	0.009643288
65°	0.009616238	245°	0.009616238
70°	0.009420288	250°	0.009420288
75°	0.009053425	255°	0.009053425
80°	0.008515638	260°	0.008515638
85°	0.007809050	265°	0.007809050
90°	0.007175775	270°	0.007175775
95°	0.007809050	275°	0.007809050
100°	0.008515638	280°	0.008515638
105°	0.009053425	285°	0.009053425
110°	0.009420288	290°	0.009420288
115°	0.009616238	295°	0.009616238
120°	0.009643288	300°	0.009643288
125°	0.009505500	305°	0.009505500
130°	0.009208863	310°	0.009208863
135°	0.008761238	315°	0.008761238
140°	0.008172263	320°	0.008172263
145°	0.007453213	325°	0.007453213
150°	0.006616875	330°	0.006616875
155°	0.005677375	335°	0.005677375
160°	0.004650038	340°	0.004650038
165°	0.003550813	345°	0.003550813
170°	0.002397988	350°	0.002397988
175°	0.001208191	355°	0.001208191

Appendix D.   Calculating Maximum Usable Continuous SPL in a Non-Anechoic Environment

An example of the alternate procedure for calculating the SPL using Method A from Section 8.1.2.2 is shown below. The steps for Method B would be similar.

Step 1
*Apply an appropriate window to the impulse response recorded at the conclusion of the test. This window shall attenuate the level of sound arrivals from any reflecting surfaces to greater than 40 dB below the level of the direct sound from the **DUT**.*

Any reflections from walls, the ceiling, or any other objects must be eliminated from the measurement results. Figure 12 below shows the windowed and unwindowed impulse response (IR) of the DUT. In this case the IR window (Half Blackman) began at 10 ms and ended at 20 ms. During this time span the magnitude of the IR was gradually attenuated until it was zero at 20 ms and thereafter.

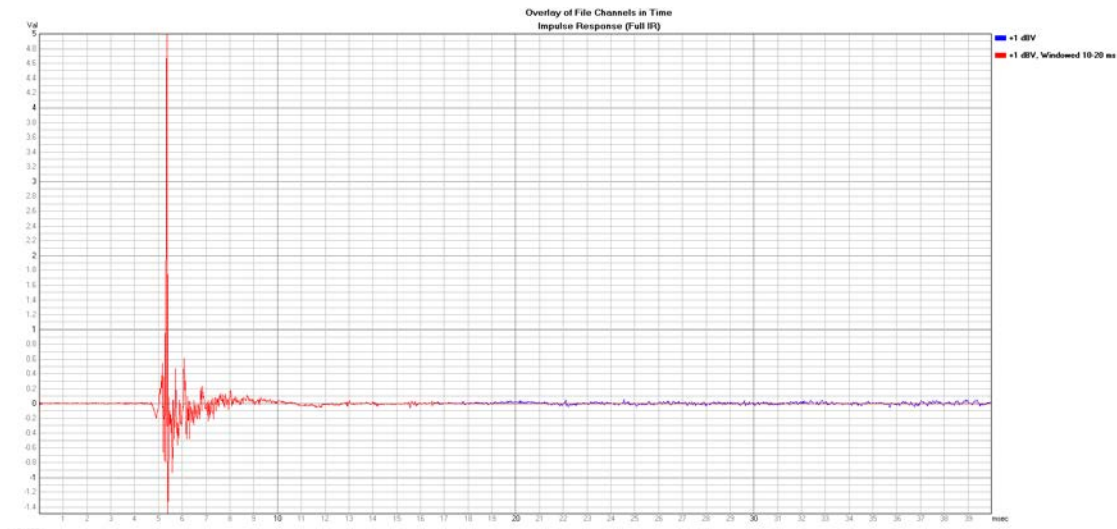**Figure 12: Windowed & Unwindowed IR of the DUT**

Step 2
Perform an FFT on the windowed impulse response to obtain the anechoic transfer function.

This will yield the frequency response of the DUT at the termination of the test (Figure 13).
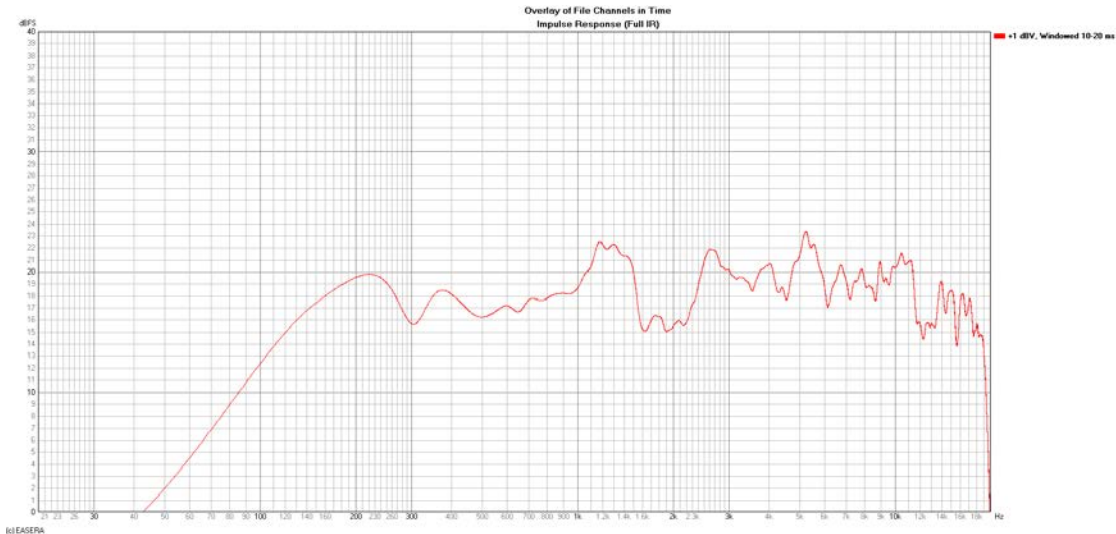
Figure 13: FFT of the windowed IR from Step 1

Step 3
Measure the spectral content and level at the input of the DUT at the conclusion of the test.

In the Figure 14 below the spectral content of the test signal is shown. This does not resemble the typical spectrum of the test signal which is usually displayed as the level on a unit bandwidth basis. Here it is displayed as the level on a unit frequency basis.

Figure 14: Spectral content of the test signal

Step 4
Multiply the transfer function from step 2 by the spectral content and level from step 3.

The measurements from steps 2 and 3 are show to scale in Figure 15. These are to be multiplied together in the frequency domain. (Note that this is identical to them being convolved in the time domain.) The result of this multiplication is shown in Figure 16.
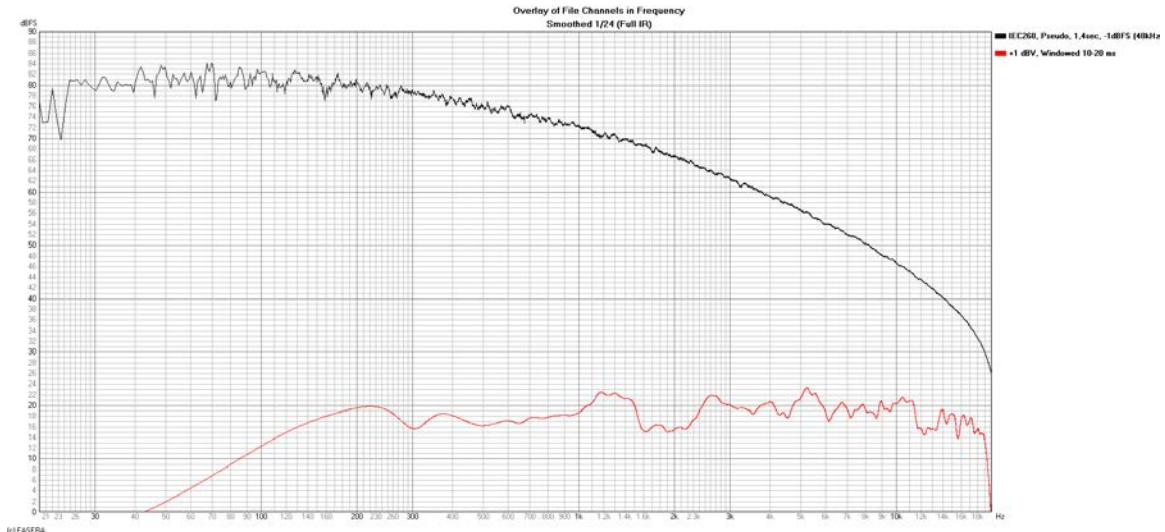
Figure 15: Spectral content of the test signal and the frequency response of the DUT
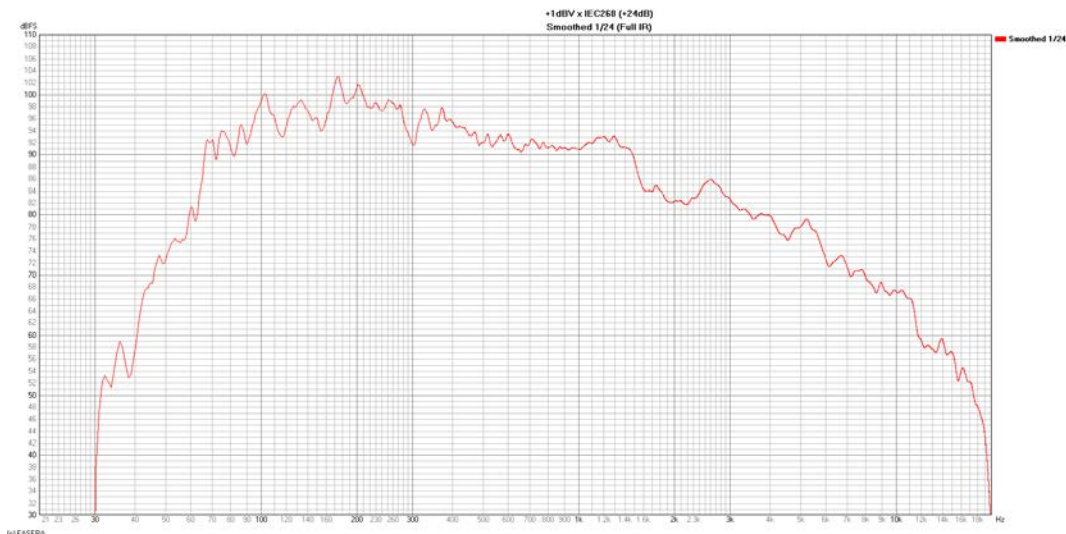
Figure 16: Calculated spectral content of the output from the DUT at the termination of the test

Step 5
*Compute the broadband SPL of the result of step 4 and reference it to 1 meter to yield a suitably anechoic representation of the **DUT** as if it was driven at its MIV continuous.*

The data from Figure 16 is exported in text format at 1/24 octave frequency spacing and copied into a spreadsheet application where further calculations can be made using the data. The following calculations are made.

   A. At each frequency the squared pressure is calculated from the SPL.

   B. The pressure squared for all the frequency data points are summed together to get the total pressure squared.

   C. The total SPL is calculated from the total pressure squared.

D.   The total SPL is then referenced to a 1 m, free-field level.

For the example given here the following are the calculated values.

Total Pressure^2 = 1.85 x 10^{11} Pa^2
Total SPL = 112.7 dB

The measurement was made at 3 meters with the microphone placed using a ground plane measurement technique.

Level adjustment for 3 m to 1 m = +9.5 dB
Level adjustment for Ground Plane to Free-Field = -6 dB
Total Level Adjustment = +3.5 dB

SPL_{MUCO} = 112.7 dB + 3.5 dB
SPL_{MUCO} = 116.2 dB

Appendix E.   List of Equipment Needed to Perform Tests Described in This Document

Section 5 - Directivity Response – Standalone Loudspeakers
- Sufficiently large reflection-free environment – a qualified anechoic chamber or outdoors
- Rotation mechanism capable of holding the DUT
- Tape measure, protractor or similar tools to measure distance and angle
- Type 2 or better measurement microphone
- Constant voltage source (power amplifier) capable of producing sufficient voltage at frequencies from 20 Hz to 20 kHz

Section 6 - Directivity Response – In-Wall, In-Ceiling, On-Wall and On-Ceiling Loudspeakers
- Sufficiently large reflection-free environment – a qualified anechoic chamber or outdoors
- Rotation mechanism capable of holding the DUT
- Tape measure, protractor or similar tools to measure distance and angle
- Type 2 or better measurement microphone
- Constant voltage source (power amplifier) capable of producing sufficient voltage at frequencies from 20 Hz to 20 kHz
- Rigid surface that represents a wall, ceiling or floor

Section 7 - Frequency Response and Crossover Category
- Sufficiently large reflection-free environment – a qualified anechoic chamber or outdoors
- Tape measure, or similar tool, to measure distance
- Type 2 or better measurement microphone
- Constant voltage source (power amplifier) capable of producing sufficient voltage at frequencies from 20 Hz to 20 kHz (for passive DUT only)

Section 8.1 - On-Axis Maximum Usable Sound Pressure Level – Continuous
- CEA-2034 noise source
- Level control
- Constant voltage source (power amplifier) capable of producing sufficient voltage at frequencies from 20 Hz to 20 kHz
- True rms voltmeter with adequate crest factor compliance and bandwidth
- Peak reading voltmeter
- Transfer function analyzer
- Type 2 or better measurement microphone

Section 8.2 - On-Axis Maximum Sound Pressure Level – Peak
- 1/3 octave tone bursts
- Level control
- Constant voltage source (power amplifier) capable of producing sufficient voltage at frequencies from 20 Hz to 20 kHz
- Type 2 or better measurement microphone
- FFT analyzer with usable bandwidth to 20 kHz and a frequency resolution of 5 Hz or finer

Section 10 - Impedance
- Transfer function analyzer
- Constant voltage source (power amplifier) capable of producing sufficient voltage at frequencies from 20 Hz to 20 kHz
- True rms voltmeter with adequate crest factor compliance and bandwidth

Appendix F.   Informational Appendix

Gated and Windowed Measurements

Gated or windowed measurement data are approximations, not equivalents, of anechoic data. The most common compromise is in frequency resolution at lower frequencies, because of a requirement for time windowing in FFT and related types of measurements. The requirement for 1/20-octave resolution cannot be achieved with windowed measurements except in the geometric center of very large, auditorium-sized, rooms. The lack of such resolution means that the nature of the low-frequency roll-off in loudspeakers is not accurately revealed and the audibility of medium- and high-Q resonances cannot be accurately estimated at lower frequencies in the amplitude response data. These limitations are discussed in the references in Appendix A.

Anechoic chambers cease to be perfectly anechoic at low frequencies, depending on the depth of the absorptive treatment on the room boundaries. A thickness of approximately one quarter wavelength is required for effective absorption, and a "cutoff frequency" is often defined that is related to this measure. At and below this frequency one can expect reflected sound to increase, and errors to be evident. Measurements can still be made within this low frequency range but without calibration of the chamber, it cannot be trusted. Appendix B describes methods of calibrating the amplitude response of anechoic chambers at frequencies below cutoff.

The anechoic chamber should be free from unneeded apparatus and reflecting objects in the vicinity of the microphone(s) and the DUT. Tensioned wire "trampoline" floors are usually acoustically transparent at the frequencies for which amplitude measurements are taken during loudspeaker testing. Some structural floors can exhibit significant reflections. Fortunately such floors are usually removable and this is recommended, at least in the sound propagation path between the loudspeaker and the microphone(s). A test with and without the floor can be done to observe the level of effect.

Input Signal Level

In the traditional method of measuring loudspeaker sensitivity (SPL measured one meter from the DUT with the DUT receiving a one watt input signal) the input voltage has to vary with frequency because loudspeaker impedance varies with frequency. To maintain the input signal power at one watt while the loudspeaker impedance varies it is necessary to vary the voltage of the input signal. Making accurate measurements while varying the input voltage each time the frequency changes is time consuming, and the results it produces are not really related to real world circumstances because program material is not spectrally adapted to loudspeaker impedance.

Conventional power amplifiers are essentially constant voltage sources. They have very low output impedances and substantial current capability and thus can maintain a constant voltage across a loudspeaker load over the normal audio frequency range. As a result it has become normal procedure to measure loudspeakers with a constant voltage input of 2.83 volts – which is the voltage required to deliver one watt into eight ohms. Eight ohms is used because it is the standard resistive load used to measure the power output capabilities of power amplifiers in CEA-490-A R-2008, *Test Methods of Measurement for Audio Amplifiers.*
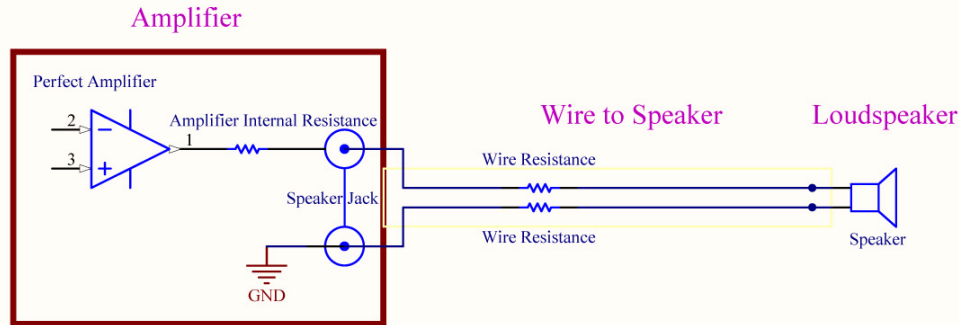
Figure 17: Possible Sources of Error in Testing a Loudspeaker System

How to Measure Crest Factor

The crest factor of the signal is defined as its peak voltage divided by its rms voltage. When working with logarithmic levels (dB) this is equivalent to the peak level minus the rms level.

Peak voltage can be measured by a calibrated oscilloscope. The rms voltage can be measured by a suitable AC voltmeter, which has an adequate bandwidth (approximately 20 Hz to 20 kHz) with good accuracy, and a crest factor capability suitable for the signal being measured. A true rms meter should be used, not one that reads rms only on a sine wave. The device used to measure the intended signal shall be AC coupled and have a flat passband with rolloff filters at the extremes to reject out of band energy.

An easy way to measure the crest factor of a signal is with a two-channel audio analyzer. Connect the signal to both Channel A and Channel B inputs, set Channel A to read peak, and Channel B to read rms, and set the reading to ratio. This will provide a continuous readout of the crest factor.

Crest Factor Considerations

The following is why a 12 dB crest factor (CF) signal source is appropriate for testing loudspeakers when in the past 6 dB has been recommended.

If one starts with a 6 dB CF noise (whether this is accomplished by clipping the peaks or digitally generating this noise) once the signal passes through a high pass filter, as is usually found in a power amplifier, the CF tends to increase to approximately 9-12 dB. This is a result of the effect of the high pass filter's phase shift on the signal. This might as well be recognized by starting with 12 dB CF random noise in its natural state. Uncompressed music and speech also tend to have a CF of 12 dB or greater.

Relationship of Maximum Continuous and Maximum Peak SPL

The test signal used in this standard to determine the maximum usable continuous output SPL (SPL_{MUCO}) has a crest factor of 12 dB. Different program material (speech, movie soundtracks, different types of music) all have different crest factors. The use of 12 dB crest factor for the test signal is a choice as an average for these as well as so other practical and engineering reasons.

It is desirable that a DUT with a certain SPL_{MUCO} can also cleanly reproduce a maximum peak SPL that is at least 12 dB greater than the SPL_{MUCO}. This ensures that the DUT can cleanly reproduce the signal presented to it at those levels. A small reduction in the maximum peak SPL

can be permitted with only minor, if any, audible effects from a loudspeaker system. For this reason this standard requires that the maximum peak SPL be at least 9 dB greater than the rated maximum SPL as given in Section 8.3.

Complex Impedance

Impedance is a measure of the "opposition" to the flow of electrons in a circuit in response to an applied voltage. For a resistor (or set of resistors), the impedance is just the resistance, which is the real value and frequency independent. In circuits with reactive elements (capacitors and inductors), impedance takes on real and imaginary values and becomes frequency dependent. A circuit with a complex impedance, $Z(\omega)$, has a phase shift between the applied voltage to the circuit and the current through the circuit generated by the applied voltage. The complex impedance (Z) of a circuit is the ratio of the complex voltage $V(\omega)$ across and complex current $i(\omega)$ through a circuit: $Z(\omega) = V(\omega)/i(\omega)$.

Purpose of the Measured Directivity Data for Post-Processing

The purpose of acquiring data on loudspeaker performance is to be able to understand how the loudspeaker might sound in a normal, semi-reflective, room. The set of anechoic measurements is a means to this end, but only after these data have been manipulated will it become clear how the loudspeaker may be heard by a listener in a normal room.

It is well known that the human hearing system responds to direct sounds as distinct from later arriving reflected sounds, and that, among the later arriving sounds there appears to be greater weight given to early reflections than to later reflections. The earliest reflections encounter only one room surface en route to the listener, and thus are potentially among the loudest reflections. Later reflections, commonly lumped into a descriptor such as reverberation, are much less loud because of the distance travelled, and because they have encountered several different surfaces in the room during their transit. The inverse-square law attenuates sound with distance, and reflective surfaces are rarely perfectly reflective, and energy is lost.

Research has led to an understanding of how important the direct, early and late reflected sounds are to listeners in small rooms. Consequently, these are the factors to examine in post processing of the raw data. From a diagnostic perspective, it is found that spatial averages are useful as a means of separating resonances (which tend to be similar in many measurements) from acoustical interferences (which tend to be different in different measurements, and are therefore attenuated by averaging). In all of the following post processing, the averages are energy (rms) averages.

- The **on-axis frequency response** is the universal starting point and in many situations it is a fair representation of the first sound to arrive at a listener's ears. However, as shown in the Devantier (2002) survey, over half of those investigated had the prime listening position 10º to 20º off axis. A further reason to question the value of this single on-axis curve is that, in some (mostly horizontally symmetrical) loudspeakers, it is contaminated by acoustical interference irregularities that do not exist at other angles. This has special consequences when this curve is used as a reference for normalization or for the calculation of Directivity Index – these single-axis irregularities which are very likely inaudible in a room, cause derived metrics to be distorted in an unrealistic way. For all of these reasons it was decided to consider the following measure as a more realistic indicator of "direct sound" in a normal listening situation.

- The **listening window** is a spatial average of the nine amplitude responses in the ±10⁰ vertical and ±30⁰ horizontal angular range. This encompasses those listeners who sit within a typical home theater audience, as well as those who disregard the normal rules when listening alone. Because it is a spatial average, this curve attenuates small fluctuations caused by acoustical interference, something far more offensive to the eye than to the ear, and reveals evidence of resonances, something to which the ear is very sensitive. Interference effects change with microphone position and are attenuated by the spatial averaging, while resonances tend to radiate similarly over large angular ranges, and remain after averaging. Bumps in spatially-averaged curves tend to be caused by resonances.

- The **early reflections** curve is an estimate of all single-bounce, first-reflections, in a typical listening room. Measurements were made of early reflection "rays" in 15 domestic listening rooms. From these data a formula was developed for combining selected data from the 70 measurements in order to develop an estimate of the first-reflections arriving at the listening location in an "average" room (Devantier, 2002). It is the average of the following:
  - Floor bounce: average of 20⁰, 30⁰, 40⁰ down
  - Ceiling bounce: average of 40⁰, 50⁰, 60⁰ up
  - Front wall bounce: average of 0⁰, ± 10⁰, ± 20⁰, ± 30⁰ horizontal
  - Side wall bounces: average of ± 40⁰, ± 50⁰, ± 60⁰, ± 70⁰, ± 80⁰ horizontal
  - Rear wall bounces: average of 180⁰, ± 90⁰ horizontal

  The number of "averages" mentioned in that description may make it seem as though anything useful would be lost in statistics. However, this turns out to be a very useful metric. Being a substantial spatial average, a bump that appears in this curve, and in other curves is clear evidence of a resonance. It is also, as will be seen, the basis for a good prediction of what is measured in rooms. In the detailed evaluation of a loudspeaker it is often useful to examine horizontal and vertical behavior separately, and so the following measures may be derived.

- **Vertical reflections.** The "floor reflection" is defined as the spatial average of three measurements at 30 degrees below the main-axis ± 10 degrees. The "ceiling reflection" is defined as the spatial average of three measurements at 50 degrees above the main-axis ± 10 degrees.

- **Horizontal reflections.** The following three spatial averages are defined:

  1. "Front" is the average of seven measurements at 0 degrees ±30 degrees.
  2. "Side" is defined as the average of ten measurements at 60 degrees ± 20 degrees to either side.
  3. "Rear" is defined as the average of 19 measurements at 180 degrees off the main-axis ±90 degrees (i.e.: the horizontal part of the rear hemisphere).

- **Sound power** represents all of the sounds arriving at the listening position after any number of reflections from any direction. It is the weighted rms average of all 70 measurements, with individual measurements weighted according to the portion of the spherical surface that they represent. Sound power is a measure of the total acoustical energy radiating through an imaginary spherical surface with the radius equal to the measurement distance. Thus the on-axis curve has very low weighting because it is in the middle of other closely adjacent measurement points (see the perspective sketch at the top of Figure 1) and measurements further off axis have higher weighting because of the larger surface area that is represented by each of those measurements. Ideally, such a measurement would be made at equally-spaced points on the entire surface of the sphere, but this simplified spatial-sampling process turns out to be a very good approximation. The result could be expressed in acoustic watts, the true measure of sound power, but for the purposes of this standard it is expressed as sound level, a

frequency response curve having the same shape. Any bump in the curve that shows up in the other curves (on-axis, early reflections, etc.) and persists through this ultimate spatial average is a notable resonance. Calculation of the sound power curve begins with a conversion from dB to a scalar magnitude. The individual measures of sound pressure are then weighted according to the values shown in Appendix C and an energy average (rms) is calculated using the weighted values. The final average is converted to dB.

- **Sound Power Directivity Index (SPDI)** is normally defined as the difference between the on-axis curve and the sound-power curve, expressed in dB. It is a measure of the degree of forward bias (directivity) in the sound radiated by the loudspeaker. The SPDI is defined slightly differently in this standard because, due to the symmetry in the layout of transducers on baffles, the on-axis frequency response often contains acoustical interference artifacts, caused by diffraction, that do not appear in any other (off-axis) measurement. In such cases the directivity index exhibits irregularities that can have no significant effects on real listening. In this standard the SPDI is defined as the difference between the listening window curve and the sound power curve. For the majority of conventional cone and dome loudspeakers this difference is negligible. However, in highly-directional (*e.g.,* large panel and horn) systems the listening window curve can be significantly different from the on-axis curve so some interpretation may be required. An SPDI of 0 dB indicates omnidirectional radiation. The larger the SPDI, the more directional the loudspeaker in the direction of the reference axis.

- **Early Reflections Directivity index (ERDI)** is defined as the difference between the listening window curve and the early reflections curve. In small rooms, early reflections figure prominently in what is measured and heard in the room so this curve may provide insights into potential sound quality.

Basis for Estimated In-Room Response Calculation

In theory, with complete 360-degree anechoic data on a loudspeaker and sufficient acoustical and geometrical data on the listening room and its layout it would be possible to estimate with good precision what would be measured by an omnidirectional microphone located in the listening area of that room. By making some simplifying assumptions about the listening space, the data set described above permits a usefully accurate preview of how a given loudspeaker might perform in a typical domestic listening room. Obviously, there are no guarantees, because individual rooms can be acoustically aberrant. Sometimes rooms are excessively reflective ("live") as happens in certain hot, humid climates, with certain styles of interior décor and in under-furnished rooms. Sometimes rooms are excessively "dead" as in other styles of décor and in some custom home theaters where acoustical treatment has been used excessively. This form of post processing is offered only as an estimate of what might happen in a domestic living space with carpet on the floor and a "normal" amount of seating, drapes and cabinetry. Reverberation time would typically be in the vicinity of 0.4 s and relatively constant with frequency over most of the frequency range.

For these limited circumstances it has been found that a usefully accurate Predicted In-Room (PIR) amplitude response, also known as a "room curve" is obtained by a weighted average consisting of 12 % listening window, 44 % early reflections and 44 % sound power. At very high frequencies errors can creep in because of excessive absorption, microphone directivity, and room geometry. These discrepancies are not considered to be of great importance. Free-field data of the kind shown in Figure 4 are generally trustworthy indicators of performance at very high frequencies.

Figure 11 shows a comparison in which the agreement is impressive above the transition/Schroeder frequency for the room (300 Hz to 400 Hz). At lower frequencies it is clear that the PIR cannot anticipate the effects of room modes and standing waves, although an

underlying trend seems to be evident. This particular loudspeaker exhibits a dip in the frequency range 2-5 kHz, which from a display of the kind shown in Figure 4 can be seen to be caused by a combination of an irregular amplitude response and frequency-dependent directivity. Individual users may experiment with other proportions of direct and spatially-averaged curves to find a good PIR fit for rooms they find themselves working in.

It is important to keep in mind that scientific research has shown that human listeners are far more analytical than a microphone in a room. While it is tempting to think that a good looking room curve or PIR is a guarantee of good sound, it is not. The first requirement for good sound in a room is a good loudspeaker, something that will be evident from inspection of a data set of the kind shown in Figure 4. A room curve is not a diagnostic tool, in that it reflects a combination of the loudspeaker and the room, however by inspecting the comparison of a PIR and a measured room curve, it may be possible to learn something about the effects of the room, and from that what acoustical modifications might be required.

Consumer Technology Association Document Improvement Proposal

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