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AVIXA A103.01:202X
Measurement and Classification of Spectral Balance of Sound Systems in Listener Areas

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ICS: 33.160.30

Abstract

This Standard defines the parameters for characterizing the spectral balance of sound systems by evaluating its transfer function to identify variations in frequency response averaged across the audience listening area. The Standard defines a process to measure, document, and classify a sound system's ability to reproduce a relatively uniform spectral balance, also known as a uniform frequency response.

Keywords

audio; audiovisual; AV; AVIXA; direct sound; frequency response; full bandwidth; impulse response; limited bandwidth; loudspeaker; measurement; microphone; sound system; sound system uniformity; spectral balance; time window; voice communication

Disclaimer

The application of this Standard is strictly voluntary. AVIXA recommends its use but does not assume responsibility for misinterpretation or misapplication. AVIXA does not assume liability for disputes resulting from non-conformance to this Standard. Conformance does not imply certification of a system. Any reference to a specific product or service is not an endorsement by AVIXA. Inclusion is for informational purposes only.

Spectral balance is the focus of this Standard, which specifically does not include testing or measurements for spatial uniformity or other parameters required to assess the total performance of a sound system. The test procedure associated with this Standard is one of many verification procedures used to determine the performance quality of a sound system in a facility.

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ISBN:

Foreword

Well-executed sound system design and implementation will reproduce program material with an approximately uniform spectral balance. This contributes to a positive user experience. This performance Standard provides a procedure to measure the spectral balance and classify the spectral performance of the system.

There are numerous factors that define the performance of a sound system. These factors include but are not limited to speech intelligibility; spectral balance; adequate sound pressure level; spatial sound pressure level uniformity; phase response; and distortion of the sound produced by the sound system. This Standard covers one of the many factors that make a sound system suitable for its intended use. Other standards cover other factors required to document the performance of a sound system. The use of those standards is highly recommended.

AVIXA Standards Developers

This Standard is dedicated to the memory of Ray Rayburn. Ray moderated this Standard's task group from 2011 to 2021, and this document would not exist without his faithful leadership, insight, and guidance. He is greatly missed.

At the time of this Standard's development, contributors' names and affiliations are as shown:

Sound System Spectral Balance in Listener Areas Task Group

Charlie Hughes, Excelsior Audio (Moderator)
Frederick Ampel, Technology Visions Analytics
Jason Antinori, CTS-D, TELUS Collaboration Services
Calvert Dayton, Rational Acoustics
Tim Habedank, CTS-D, Parsons Technologies
Ryan Knox, CTS-D, Idibri
John Monitto, CTS, Meyer Sound
John Murray, Optimum System Solutions
Pete Swanson, CTS, Lendlease
Read Wineland, CTS, Biamp Systems (ret.)

AVIXA Standards Steering Committee

Greg Jeffreys, Visual Displays Ltd., Committee Chair
Ben Boeshans, CTS-D, Idibri
James Colquhoun, CTS-D, CTS-I, Avidex Industries LLC.
Lance Feldenkreiss, CTS-D, JKL Technologies, Inc.
Kenneth Ng, Mojoworx Asia Limited
Lisa Perrine, CTS, Ed.D., Cibolla Systems
Rodrigo Sanchez-Pizani, CTS, King's College London
Timothy Troast, Legrand AV
Pomona Valero, CTS, PMP, PITM Consulting

AVIXA Staff

Bob Higginbotham (Director of On-Demand Content & Standards)
Loanna Overcash, AStd (Standards Developer)
Michelle Bollen (Standards Developer)

A special thanks to Ann Brigida, CStd, CTS for her invaluable contributions towards the development of this Standard and to Ben Boeshans, CTS-D for voluntarily acting as Standards Developer for this project during COVID-19 staff shortages.

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1.1 Scope

This Standard defines the parameters and procedures for assessing spectral balance as one of the means to characterize sound systems. Spectral balance is the amplitude uniformity of the system’s direct sound output and early reflections at selected locations relative to its signal input over a specified bandwidth.

For the purposes of this Standard, a sound system includes all the connected components that perform the functions of amplifying and processing a non-acoustical audio signal source/input (analog or digital) through to the electro-acoustic output transducers (loudspeakers).

1.2 Purpose

The purpose of this Standard is to define the requirements that indicate sound system spectral balance uniformity and the methods to verify whether a sound system conforms to these requirements.

1.3 Application

The procedures described in this Standard are to be applied to sound reinforcement, playback systems, and audiovisual (AV) presentation systems. These systems are implemented in a variety of applications, including conference rooms, training rooms, classrooms, auditoria, theatres, houses of worship, and other venues that employ sound reinforcement. Additionally, the metrics and classifications in this Standard may be used to establish design criteria for new systems.

The requirements of this Standard are aligned with the AVIXA A102.01:202x, *Audio Coverage Uniformity in Listener Areas* standard (Audio Coverage Uniformity). This maximizes efficiency when using both Standards.

1.4 Exceptions

This Standard may be used in conjunction with, but does not supersede, regulatory authority requirements.

This Standard does NOT:

a) Apply to electronic architecture systems;

b) Pertain to adjusting equalization or any similar parameter that might be changed during operation of the system, such as input channel specific EQ or tonal balance controls, dynamics, or effects on a mixing console;

c) Suggest specific equipment to be used in system measurement, adjustment, and operation;

d) Apply to theatrical effects systems or other specialty loudspeaker channels/systems;

e) Apply to cinema sound systems when used for cinema playback. (This Standard can be applied to those same cinema sound systems when used for non-cinema events);

f) Apply to system input sources, such as microphones and playback devices.

2 Referenced Publications

2.1 Normative References

The following documents contain provisions that, through reference in this text, constitute provisions of this Standard. At the time of approval, the editions indicated were valid. Because standards are periodically revised, users should consult the latest revision approved by the sponsoring Standards Developing Organizations.

2.2 Informative References

The following publications contain information that supports the design and application of this Standard but are not required provisions of the Standard. Use the latest edition unless otherwise specified.

a) Acoustical Society of America. *Acoustical Terminology*. ANSI/ASA S1.1-2013. Melville, NY: ASA, approved October 14, 2013.

b) ———. *Preferred Frequencies and Filter Band Center Frequencies for Acoustical Measurements*. ANSI/ASA S1.6-2016. Melville, NY: ASA, approved August 15, 2016.

c) Audiovisual and Integrated Experience Association. *Audio Coverage Uniformity in Listener Areas.* ANSI/AVIXA A102.01:202X. Fairfax, VA: AVIXA, approved X, 202X.

d) International Electrotechnical Commission (IEC). *Electroacoustics - Octave-Band and Fractional-Octave-Band Filters - Part 1: Specifications*. IEC 61260-1:2014. Geneva: IEC, approved February 14, 2014.

e) International Electrotechnical Commission. *Electroacoustics – Sound Level Meters – Part 1: Specifications*. IEC 61672-1:2013. Geneva: IEC, approved September 30, 2013.

3 Definitions

As used in this document, “shall” and “must” denote mandatory provisions of the Standard. “Should” denotes a provision that is recommended, but not mandatory.

3.1 Acronyms

For the purposes of this Standard, the following acronyms apply:

3.1.1 ANL: Ambient Noise Level

3.1.2 DFT: Discrete Fourier Transform

3.1.3 FFT: Fast Fourier Transform

3.1.4 IR: Impulse Response

3.1.5 IRW: Impulse Response Window

3.1.6 rms: Root Mean Square (square root of the average of the individual squared values)

3.1.7 SPL: Sound Pressure Level

3.2 Definitions

For the purposes of this Standard, the following definitions apply:

3.2.1 Complex smoothing

The process of performing spectral smoothing on a complex spectral data set. For more information, see Annex F.2.

3.2.2 Early arriving energy

Energy, both direct and reflected, which arrives at a measurement location within 50 ms of the direct sound’s arrival.

3.2.3 Fill loudspeakers

The portion(s) of a sound system designed to supplement the main loudspeakers’ coverage of listener area(s). These loudspeakers are not adjacent to the main loudspeakers. Examples include stage lip, over-balcony, and under-balcony loudspeakers.

3.2.4 Full-bandwidth sound system

A sound system whose frequency reproduction limits are driven by the need to reproduce material with spectral content at the lower and upper limits of the audible spectrum. These systems are often employed for concerts, contemporary worship, and musical theatre. For the purposes of this Standard, these systems are evaluated from 40 Hz to 12.5 kHz.

3.2.5 Limited-bandwidth sound system

A sound system whose frequency reproduction limits are driven by the need to reproduce speech and background music. This type of system is often found in ballrooms, conference rooms, and lecture halls. For the purposes of this Standard, these systems are evaluated from 100 Hz to 10 kHz.

3.2.6 Listener plane

A stated distance above the floor determined to be the average audience member’s ear height across a listener area. This distance is dictated by the intended use of the system and establishes the height of the measurement microphone.

3.2.7 Listener area

Continuous space(s) intended to be covered by a sound system.

3.2.8 Loudspeaker system

An implementation of loudspeaker(s) designed to provide audio coverage to specific listener area(s). The system may be single or multi-channel in nature.

3.2.8.1 Multi-channel loudspeaker system

A loudspeaker system designed so that multiple loudspeaker locations provide coverage of unique content to the same listening area(s). An example would be a Left/Center/Right system where each feed is discretely provided to all listeners.

3.2.8.2 Single-channel loudspeaker system

A loudspeaker system designed so that all loudspeaker locations provide the same monophonic content to the listening area(s). An example is a distributed mono system.

3.2.9 Main loudspeakers

The portion of a sound system designed to serve as the primary (or “front-of-house”) loudspeaker system for a venue. For the purposes of this Standard, this includes adjacent loudspeaker(s) which serve as a direct extension of the main loudspeakers. Examples include center or side coverage loudspeakers.

3.2.10 Measurement grid origin point

The physical point in space from which measurement grid locations for a listener area are determined.

3.2.11 Measurement grid reference line

A line drawn between the two outermost points of the loudspeaker(s) that make up the main loudspeaker system.

3.2.12 Paging sound system

A sound system whose frequency reproduction limits are driven by the need to reproduce voice messages. The systems are often found in schools, convention centers, and transportation hubs. The systems place priority on a message being communicated rather than the faithful reproduction of the source content. For the purposes of this Standard, these systems are evaluated from 250 Hz to 4 kHz.

3.2.13 Power averaging

The process of calculating the arithmetic mean of squared linear magnitudes at each frequency for two or more level-adjusted transfer function magnitude spectra. The result is expressed in decibels.

3.2.14 Spectral balance

The magnitude uniformity of the frequency response of the direct sound and first 50 ms of reflections of a sound system’s transfer function over a specified bandwidth.

3.2.15 Transfer function

A comparison of the magnitude and phase of the acoustic output of a sound system divided by the electrical/digital input.

330 ## 3.3 Units

Measurement Quantity	Unit
ANL	dB(LZ_{eq})
Stimulus Level	dB(Z)
Transfer Function Magnitude	dB

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4 Requirements

4.1 Sound System Prerequisites

These conditions shall be met prior to testing:

a) The system shall be in its intended operating state with confirmation of loudspeaker functionality and polarity as well as adjustments for gain structure, system equalization, and time offset corrections having already been performed.

a) The venue shall be in its intended operating configuration. This means that all construction activity has ceased, room finishes are in place, the room is in its typical seating configuration, and extraneous noise from people or equipment is minimized.

4.2 Measurement System Requirements

All measurement instrumentation shall meet the following requirements:

a) Be calibrated as required by the manufacturer’s instructions to ensure measurement accuracy and consistency.

b) Express sound pressure level in Z-weighted decibels.

c) Microphones shall be free-field, omni-directional, with a capsule diameter no greater than 15 mm (0.59 in) and conform to frequency response requirements of Class 1 sound level meter systems. For additional information on measurement microphones, see Annex B.

d) The measurement system(s) shall be capable of:

   1) performing a LZ_{eq} measurement of ANL per *ANSI/ASA S12.72 Procedure for Measuring the Ambient Noise Level in a Room*.

   2) performing a transfer function measurement

   3) applying a 50 ms impulse response window or an equivalent function (e.g., time delay spectrometry or frequency domain complex smoothing [magnitude and phase]). This is distinct from the data window function used for signal acquisition and the record length (e.g., FFT block size) used for the measurement. See Annex F for further details.

   4) a frequency spacing no greater than 1/12 octave (for log spaced) or 15 Hz (for linear spaced).

4.3 Test Signal Requirements

a) The test signal shall be injected into the system under test electronically, not acoustically.

b) The test signal shall be supplied to the system under test before the main system processing functions, such as equalization or time delay. Moreover, it should pass through the system under test free from the effects of compressors, limiters, and other non-linear processing. If the test signal does not pass through the system under test free from non-linear processing, then a broadband noise-like test signal shall be used.

c) The sound pressure level (L_P) of the test signal produced as an acoustic output from the system under test shall be at least 15 dB greater in each one-octave band than the corresponding octave band LZ_{eq} of the ANL measurement taken across the space. If the system is incapable of meeting this requirement, it shall not be evaluated under this Standard. An exception for this requirement is noted in Section 4.5 System Purpose.

4.4 Spectral Balance Process Map

For visual reference, Annex A contains a process map that shows the Spectral Balance measurement procedure and necessary documentation for this Standard.

4.5 System Purpose

Systems shall be evaluated based on their intended purpose:

a) Paging System: systems used primarily for spoken word or similar content shall be evaluated from 250 Hz to 4 kHz.

b) Limited Bandwidth System: systems used primarily for speech or other limited bandwidth content shall be evaluated from 100 Hz to 10 kHz.

c) Full Bandwidth System: systems used primarily for music or other full bandwidth content requiring low frequency reproduction or reinforcement shall be evaluated from 40 Hz to 12.5 kHz.

A system unable to meet the 15 dB above ANL requirement of Section 4.3 at the upper and/or lower boundaries of the frequency range may be evaluated as a system with a more restrictive frequency range, provided it meets the 15 dB above ANL requirement across the more restrictive frequency range.

For information on determining system purpose, see Annex C.

4.6 Establishing Measurement Locations

Prior to establishing measurement locations, identify the listener area(s) and plane(s).

This Standard provides two procedures for determining measurement locations based upon loudspeaker system topology: distributed topology (Section 4.6.1) and point-source or line-source topology (Section 4.6.2).

The Standard outlines the minimum number of measurement locations required to characterize coverage uniformity; the user may add measurement locations as they deem necessary due to site conditions. All additional measurement locations and their rationale shall be documented.

4.6.1 Distributed System Measurement Locations

Measure a distributed system using the following scenarios:

4.6.1.1 Consistent Distribution

In spaces where the distribution of loudspeakers and the distance from the loudspeakers to the listening plane are consistent, as in Figure 1, measurements shall be taken:

1) Directly on-axis of a loudspeaker (Figure 1, location 1)
2) Equidistant between two adjacent loudspeakers (Figure 1, location 2)
3) At the point of greatest overlap created by three or more adjacent loudspeakers that is equidistant from each of those loudspeakers (Figure 1, location 3)
4) At the edge of the listener area furthest from any loudspeaker (Figure 1, location 4).

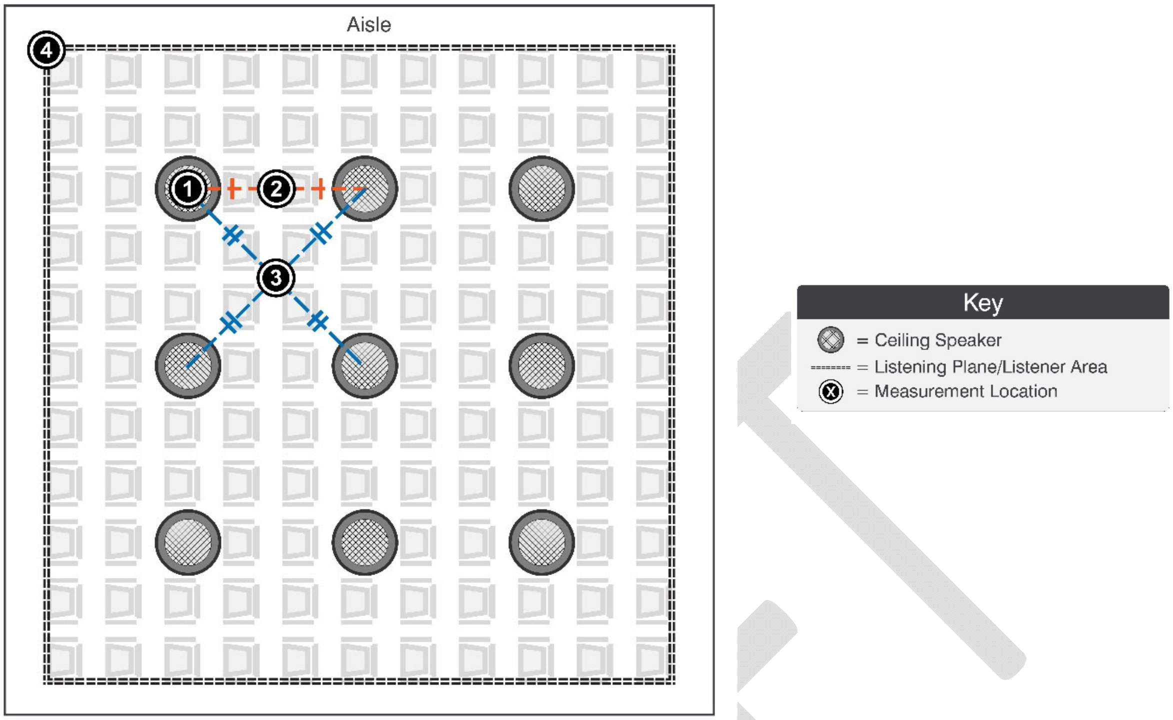


Figure 1 – Distributed loudspeaker measurement locations (plan view)

4.6.1.2 Consistent Distribution with Minor Anomalies

Sometimes, within a consistently distributed loudspeaker system, loudspeaker locations have been shifted to accommodate other ceiling devices, such as lighting fixtures or heating, ventilation, and air conditioning (HVAC) systems. In such scenarios, include these additional measurement locations:

5) The coverage overlap zone halfway between the shifted loudspeaker and a consistently spaced loudspeaker (Figure 2, location 5)
6) The position of greatest overlap of three or more loudspeakers as a result of shifting one of those loudspeakers (Figure 2, location 6)

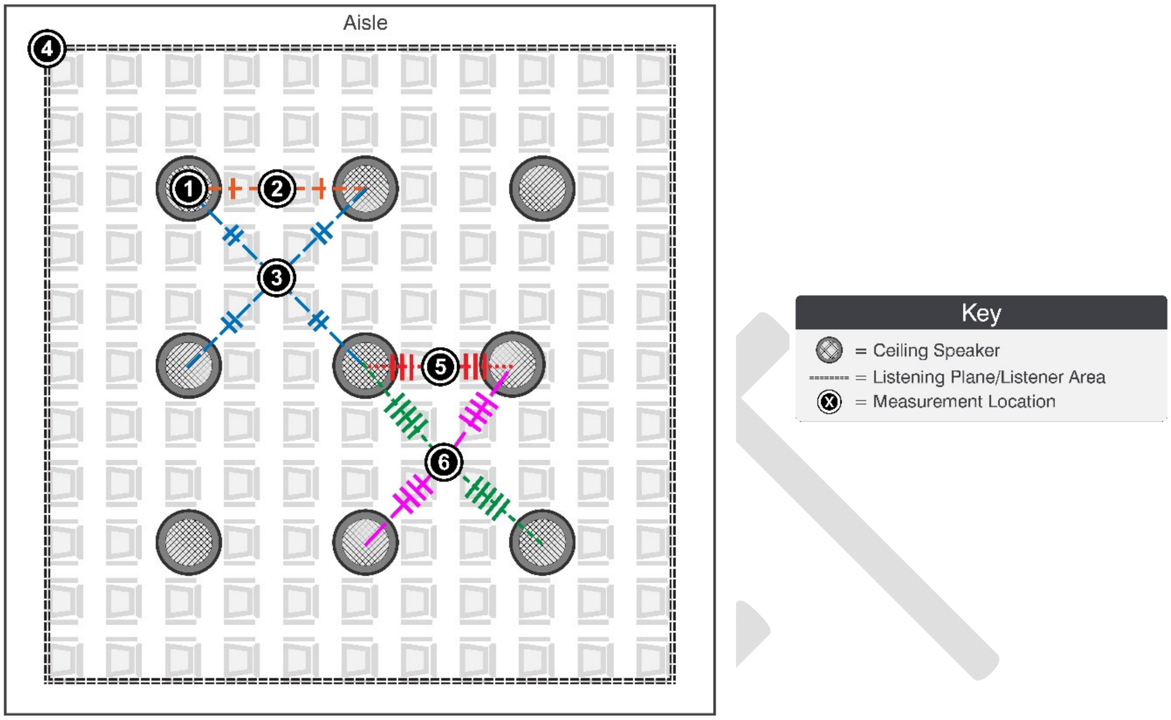
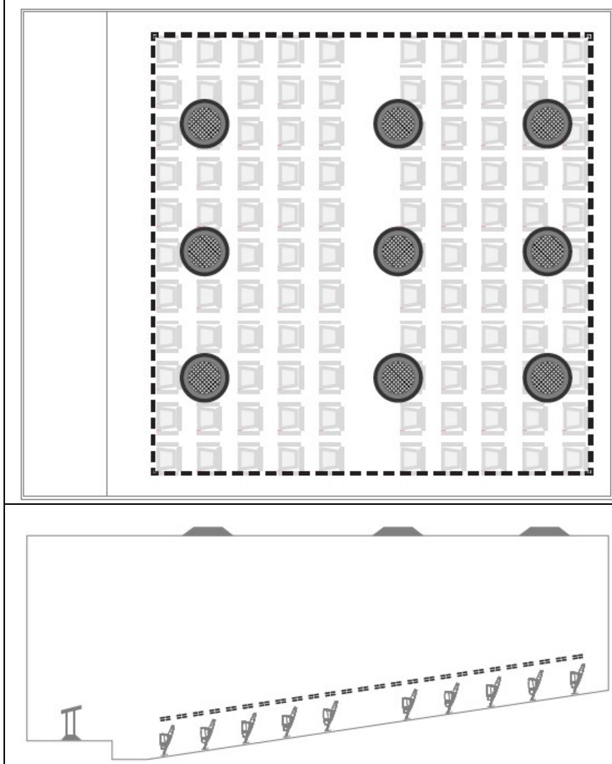
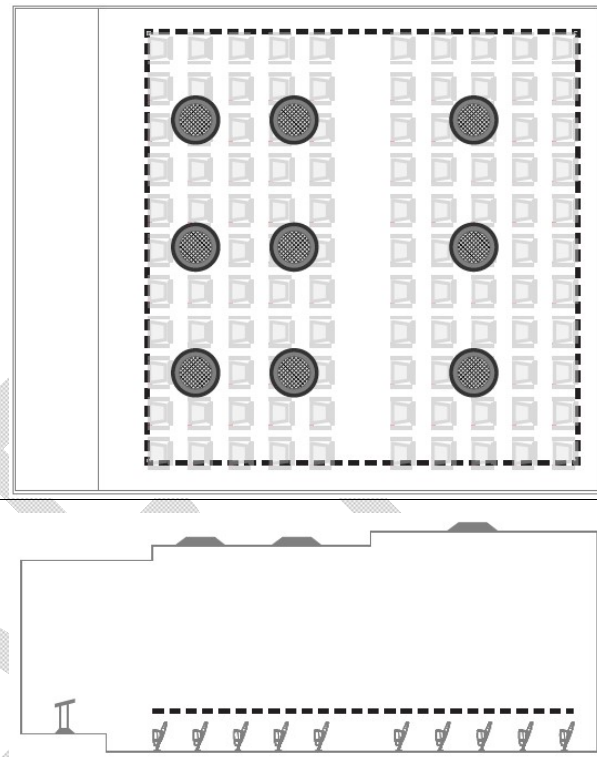


Figure 2 – Distributed loudspeaker measurement locations with minor anomalies (plan view)

4.6.1.3 Inconsistent Distribution

In a scenario where either the loudspeaker spacing or the distance between the loudspeaker plane and the listener plane are not consistent, each unique overlap zone (Figure 1, locations 2 and 3) shall be measured in addition to the on-axis (Figure 1, location 1) and listener area edge (Figure 1, location 4) measurement locations.

Table 1 – Distributed loudspeaker systems with inconsistent distribution

Figure 3 –Flat ceiling and sloped listener plane (plan and section view)	Figure 4 – Stepped ceiling and flat listener plane (plan and section view)
	

4.6.2 Point-Source and Line-Source System Measurement Locations

Measurement of point-source or line-source systems shall include all loudspeakers, based on the following rules:

a) Single-channel loudspeaker systems shall be measured with all loudspeakers operating.

b) Multi-channel loudspeaker systems shall have each channel measured independently.

4.6.2.1 Measurement Grid Origin Point

The measurement grid origin point of a point-source or line-source system is the physical point in space from which the grid measurement locations for a listening area are determined. It shall be established based upon the system's topology.

4.6.2.1.1 Horizontal Location of the Measurement Grid Origin Point

Horizontally, the measurement grid origin point shall be the midpoint of a line drawn between the two outermost points of the loudspeaker(s) that make up the main loudspeaker system. This line is referred to as the measurement grid reference line.

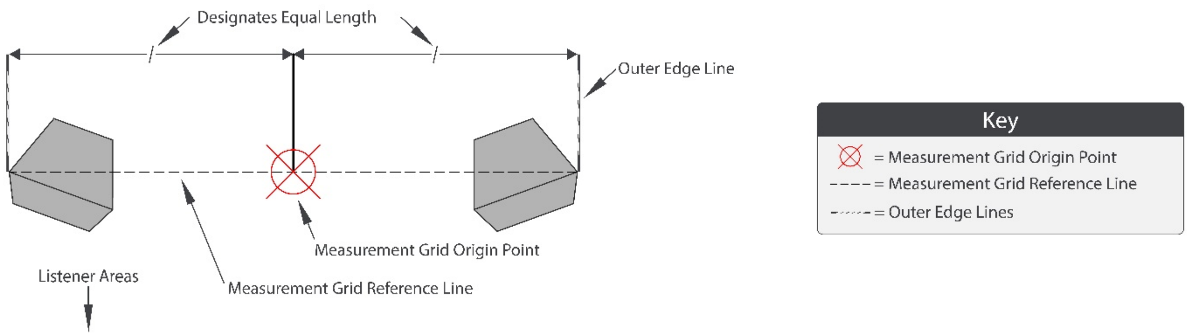


Figure 5 – Measurement grid origin point

For examples of finding the measurement grid origin point, see Table 1 and Table 2.

Table 2 – Measurement Grid Origin Points for Single-Channel Systems with a Single Main Loudspeaker Location

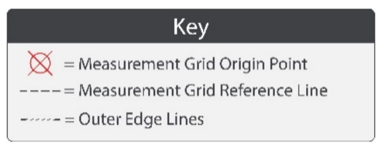
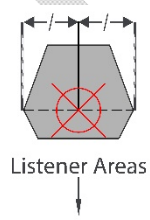
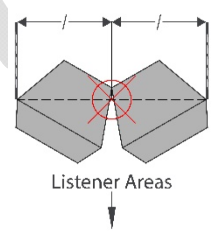
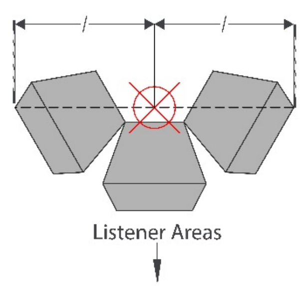
	
Figure 6 – Measurement grid origin point for a single loudspeaker	
Figure 7 – Measurement grid origin point for a two loudspeaker cluster	
Figure 8 – Measurement grid origin point for a three loudspeaker cluster	

Table 3 – Measurement Grid Origin Points for Single-Channel Systems with Multiple Main Loudspeaker Locations

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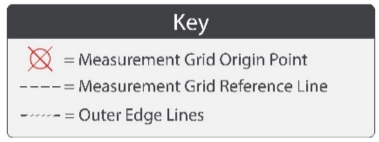
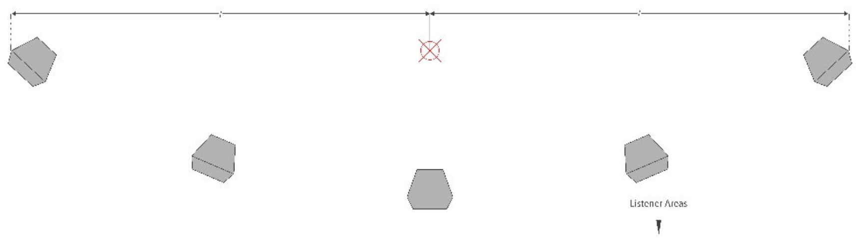
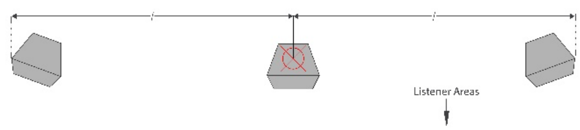
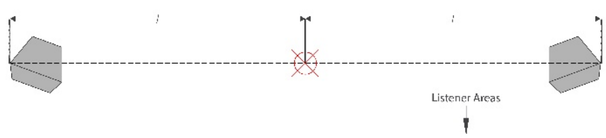
<tr><td colspan="2">
</td></tr>
<tr><td>Figure 9 – Measurement grid origin point for an exploded single-channel cluster</td><td>
</td></tr>
<tr><td>Figure 10 – Measurement grid origin point for a single-channel system with three loudspeaker locations</td><td>
</td></tr>
<tr><td>Figure 11 – Measurement grid origin point for a single-channel system with two loudspeaker locations</td><td>
</td></tr>
</table>

4.6.2.1.2 Vertical Location of the Measurement Grid Origin Point

The height of the measurement grid origin point will be at the top, front of the uppermost box in the main loudspeaker system.

Table 4 – Examples of vertical locations of measurement grid origin points

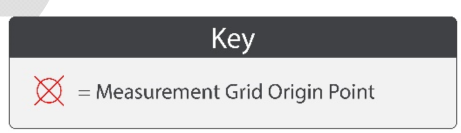
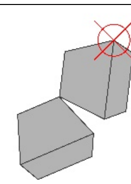
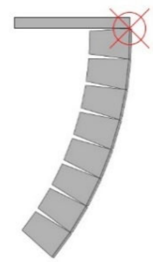
<table>
<tr><td colspan="2">
</td></tr>
<tr><td>Figure 12 – Example of the vertical location of the measurement grid origin point of a two-loudspeaker cluster</td><td></td></tr>
</table>

Figure 13 – Example of the vertical location of the measurement grid origin point of a multi-box line array	

4.6.2.1.3 Measurement Grid Origin Points for Multi-Channel Systems

Systems with multiple program contents whose output channels individually cover the same listener areas shall have each output channel measured independently. Such systems will have multiple measurement grid origin points. Repeat Section 4.6.2.1.1 and Section 4.6.2.1.2 for the measurement grid origin point of each loudspeaker system channel.

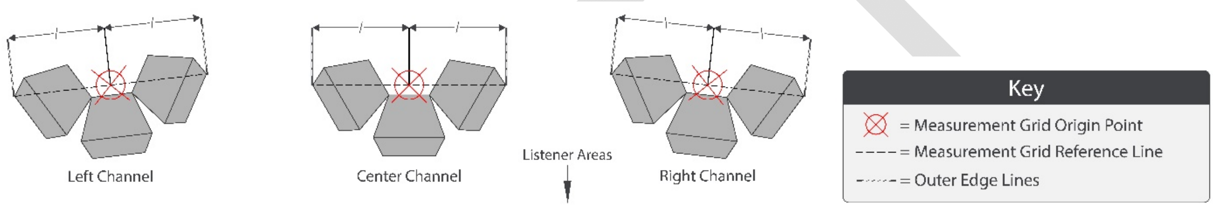


Figure 14 – Measurement grid origin points for a multi-channel left/center/right loudspeaker system

4.6.2.2 Establishing Measurement Locations for Point or Line Source Systems

Establish measurement locations using the following procedure:

a) Establish the 0-degree horizontal radial line by drawing a line perpendicular to the measurement grid reference line from the measurement grid origin point.

b) Additional horizontal radial lines shall be established by rotating the 0-degree horizontal radial line about the measurement grid origin point in 20-degree increments.

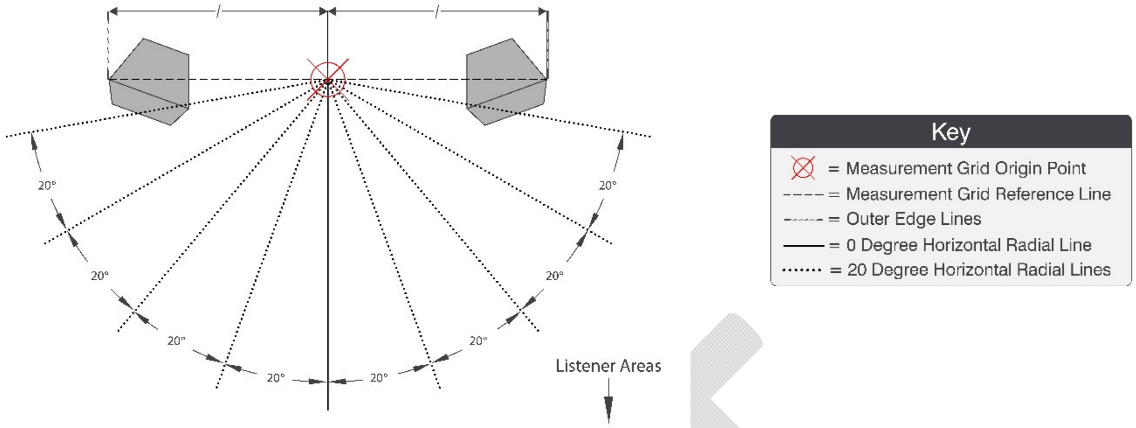


Figure 15 – Establishing radial lines (plan view)

c) The first measurement location on each horizontal radial line shall be the location in the listener area on the horizontal radial furthest from, but within sight of, the measurement grid origin point.

   1) If the first point along each horizontal radial line is not at the back of the listener area, an additional point shall be added at the back of the listener area."

d) Using the measurement grid origin point as reference, establish a second measurement location along the horizontal radial that is 5 degrees closer to the measurement grid origin point.

e) Continue to establish additional measurement locations in 5-degree increments along the horizontal radial line up to the front of the listener area.

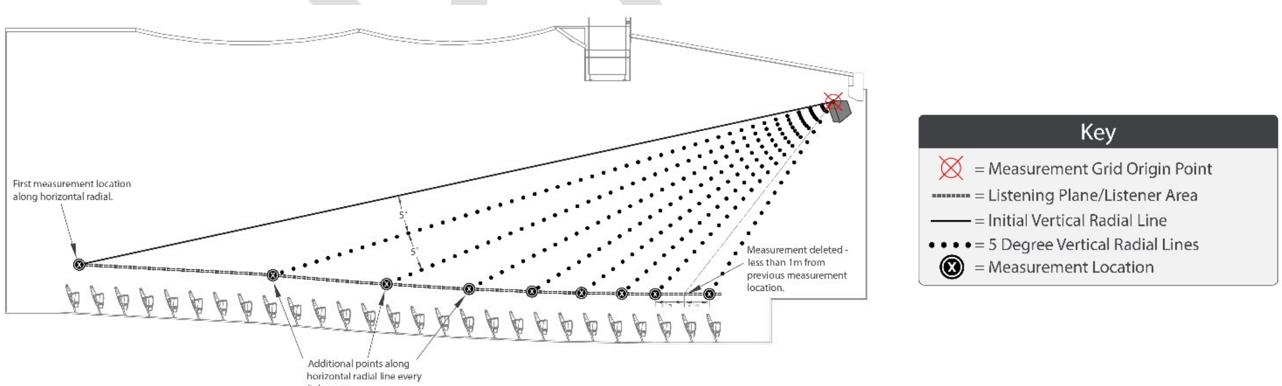


Figure 16 – 5-degree vertical radials along a horizontal radial (section view)

f) Repeat this procedure for each horizontal radial line within the listener area.

If multiple measurement locations are located within a 1 m (3.3 ft) radius, only one of the measurement locations shall be used. The intent is to have maximum spacing between adjacent measurement locations.

4.6.2.2.1 Tiered Venues

In tiered venues, repeat steps 4.6.2.2.c through 4.6.2.2.e within each tier.

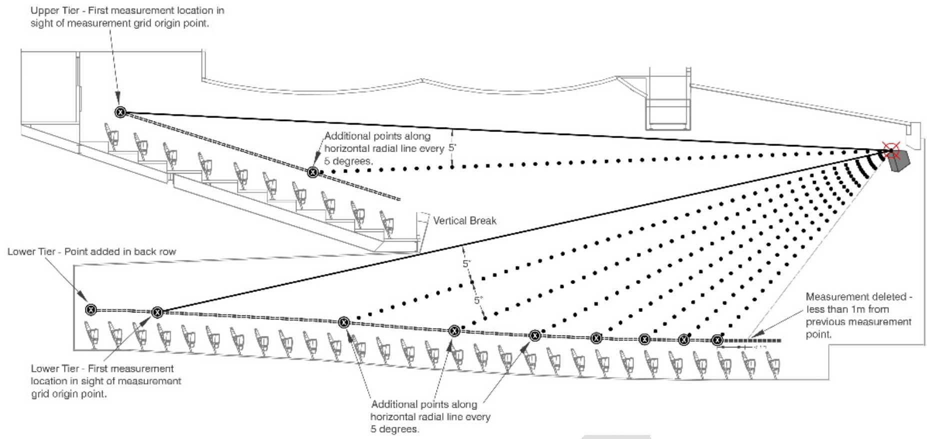


Figure 17 – Establishing measurement locations in a tiered venue (section view)

4.6.2.2.2 Venues with a Center Aisle

If the 0-degree horizontal radial falls in the center aisle of a venue, the measurement locations shall be shifted to the edge of the closest listener area. Only the 0-degree horizontal radial shall be shifted; none of the other radials shall be shifted within the grid.

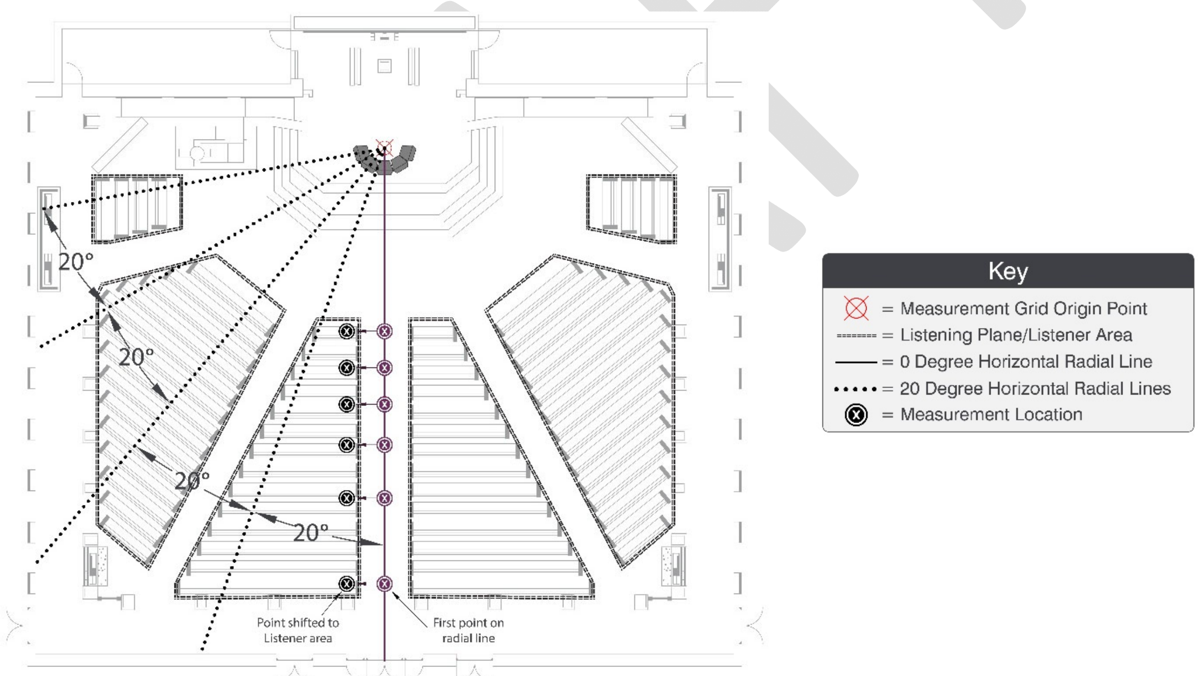


Figure 18 – Measurement locations along the 0-degree horizontal radial shifted to accommodate a center aisle (plan view)

4.6.2.2.3 Symmetrical Loudspeaker Systems in Symmetrical Venues

If a symmetrical loudspeaker system is deployed in a symmetrical venue, measurements are only required on, and to one side of the 0-degree horizontal radial line.

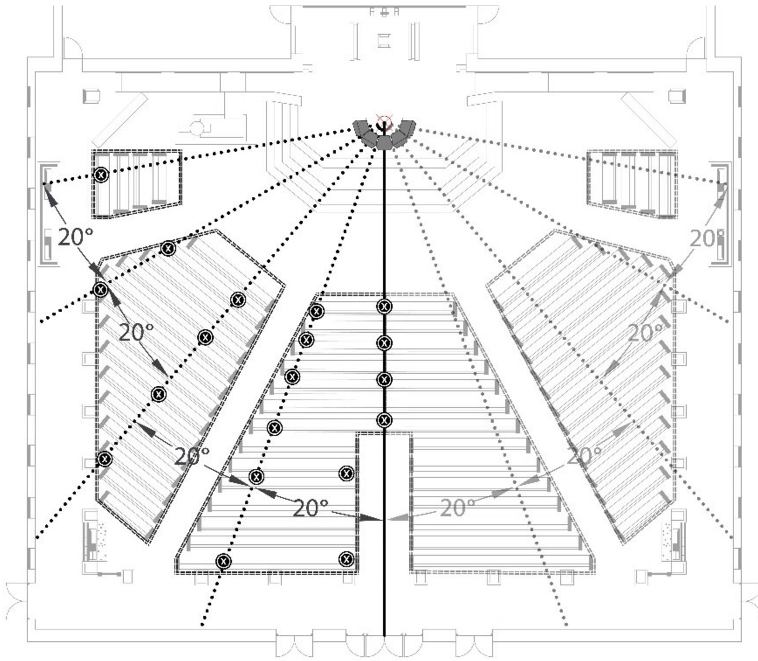


Figure 19 – Symmetrical loudspeaker systems in symmetrical venues (plan view)

4.6.3 Fill Loudspeaker System Measurement Locations

In venues that utilize fill loudspeakers, establish measurement locations in the following places:

1) Directly on axis of each fill loudspeaker
2) At each midpoint between adjacent measurement locations taken in step one
3) At the edge of the listener area covered by the outermost fill loudspeaker(s)
4) In the transition between each fill loudspeaker and the main loudspeaker system

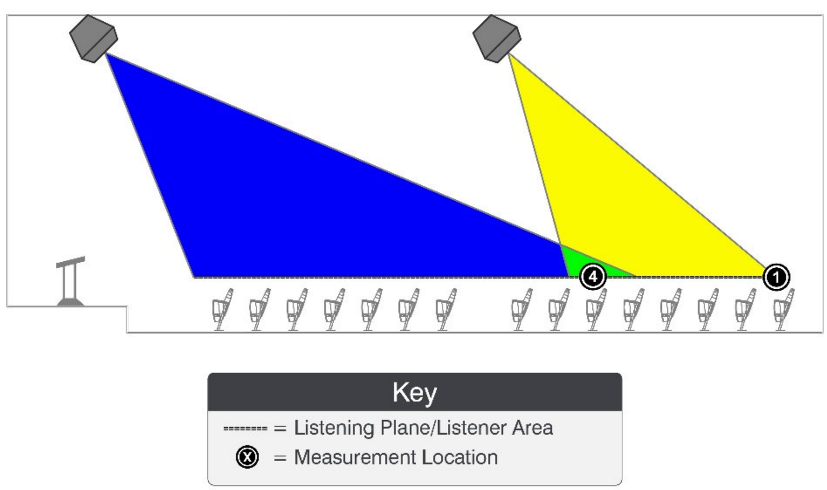


Figure 20 – Point-source/line-source loudspeaker system with fill system (over or under balcony)

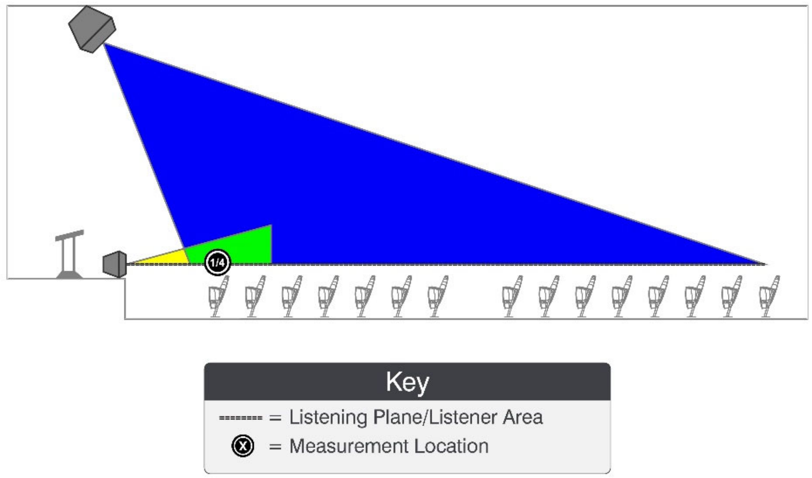


Figure 21 – Point-source/line-source loudspeaker system with front fills

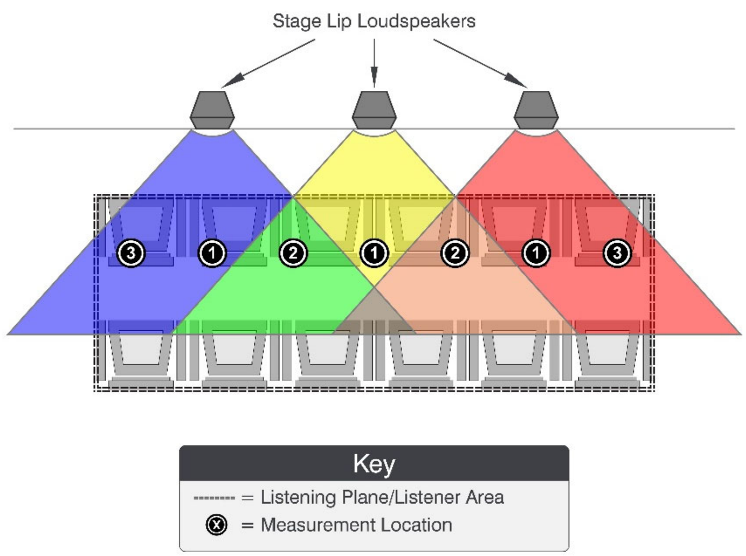


Figure 22 – Stage lip loudspeaker measurement locations

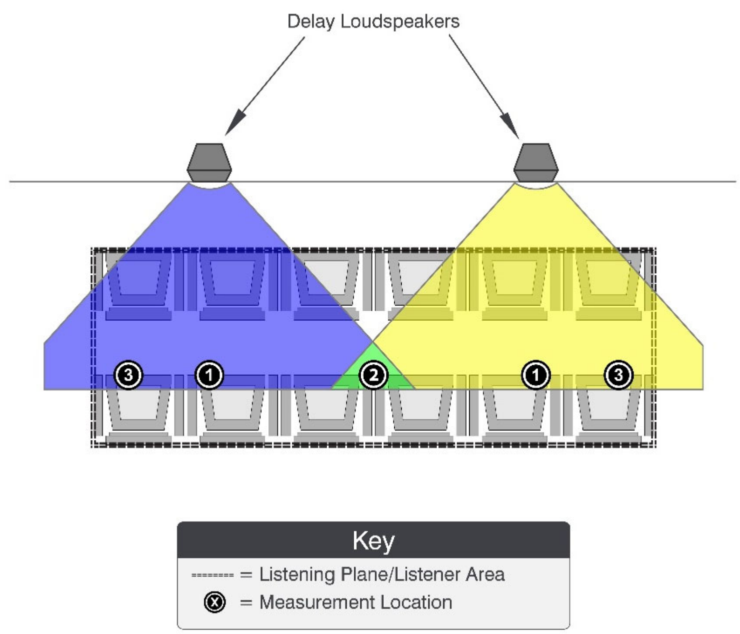


Figure 23 – Delay loudspeaker measurement locations

4.7 Procedure

The following procedure defines the data collection and computational steps required by this Standard. While each step shall be completed, the method for completing these steps may vary among software platforms. For example, some users may prefer to apply an impulse response window as each measurement is performed rather than during post-processing.

4.7.1 Measurements

a) Prepare a drawing (similar in nature to a ceiling, furniture, or facilities plan) which includes the following:

   1) Location of all loudspeakers

   2) Location of all listener areas marked with the listener plane height.

      i) For listener areas with varying physical configurations (such as operable partitions), measurements shall be taken and reported separately for each configuration.

b) Record the following:

   1) The system purpose: paging, limited, or full bandwidth

   2) The type of test signal to be used (broadband noise, sweep, etc.)

   3) The measurement tools to be used (make, model, calibration status, software version)

c) Determine the spatially averaged ANL in each one-octave band contained within the evaluation range by taking a LZ_{eq} measurement for a minimum of 15 s across all listener areas, as per the survey method in ANSI/ASA S12.72 *Procedure for Measuring the Ambient Noise Level in a Room*.

   1) Measurement duration shall be adequate to survey the entire listener area(s).

   2) If a listener area has a noticeably louder ANL than that of other listener areas, an additional LZ_{eq} measurement(s) shall be taken in that listener area. This measurement shall determine the ANL for the test.

d) Connect the test signal generator to the system and route the signal to all loudspeaker elements within a given output channel.

   1) Fill loudspeakers shall be operating during measurements.

e) Ensure that the test signal meets the requirements of Section 4.3.

f) Record any changes to system settings so that they may be reset at the conclusion of the test.

g) Perform a transfer function measurement at each measurement location identified in Section 4.6. Save each transfer function measurement as a unique data set.

   1) Microphones shall be placed in the listener plane to a height tolerance of ±25 mm (1 in).

   2) The position of these locations within the space shall be located to a tolerance of ±300 mm (12 in). Note any measurement locations that are outside of that tolerance and the reason for the deviation.

   3) The frequency spacing shall be no greater than 1/12 octave (for log spaced) or 15 Hz (for linear spaced).

h) For multi-channel sound systems or rooms with varying physical configurations, repeat steps d through g.

i) Return any system parameters changed for this measurement procedure to their pre-existing operating conditions.

4.7.2 Data Processing

After all measurements have been taken, the data shall be processed as detailed below.

4.7.2.1 Apply the Impulse Response Window

An impulse-response window (in the time domain) or equivalent function shall be applied to the data from each individual measurement location.

An impulse-response window of 50 ms shall be used.

The impulse-response window shall satisfy the following three requirements:

a) The highest peak of the IR shall be aligned with the portion of the impulse-response window where the least attenuation occurs.

b) The right half, or trailing edge of the window, shall taper smoothly to be closed (terminal attenuation) at 50 ms after the highest peak in the IR.

c) The left half or leading edge of the window shall not truncate the IR too early or too abruptly so that it smoothly encompasses the full arrival of direct sound from the loudspeaker system under test.

For additional information on satisfying these requirements, see Annex F.

Some measurement/analysis systems do not directly use an impulse-response window, but instead employ equivalent functionality. It is up to the operator to ensure that the measurement/analysis system is configured so that the processing performed by the measurement/analysis system is equivalent to the duration requirement for the impulse-response window given in this Standard.

4.7.2.2 Level Adjust

The impulse-windowed transfer function from each measurement location shall be level adjusted so that the broadband levels of all transfer functions are approximately the same.

The log-frequency based average level of each transfer function shall be calculated over the bandwidth from 225 Hz to 8.8 kHz.[1] For each transfer function, the decibel level values at each frequency data point over this bandwidth shall be averaged to yield the broadband average level. For instructions on how to calculate a log-frequency based average, see Annex H

A gain value (either positive or negative) shall be calculated for each broadband level, so that each average broadband level has the same value. The gain value calculated for each broadband level shall be applied to its respective transfer function. This will result in a matched broadband level across all transfer functions.

4.7.2.3 Symmetrical Loudspeaker Systems in a Symmetrical Venue

In a symmetrical venue (as detailed in clause 4.8.3), the measurement positions located along the center line of the room shall be included once in calculation of standard deviation (4.11.4) and power average magnitude (4.11.5) at each frequency. All other measurement positions shall be included twice.

4.7.2.4 Calculate Standard Deviation

A standard deviation figure for each frequency shall be calculated based on level-adjusted decibel magnitude values for all measurement positions. The resulting decibel standard deviation spectrum shall be plotted on a semi-log chart with logarithmic frequency on the *x* axis and linear standard deviation on the *y* axis, to be included in the test report.

Additionally, a log-frequency based average of all standard deviation values within the evaluation frequency range, based on the intended System Purpose (see clause **Error! Reference source not found.** and Annex C), shall be calculated and stated in the report. For instructions on how to calculate a log-frequency based average, see Annex H.

4.7.2.5 Calculate the Power Averaged Magnitude Response

A power averaged magnitude response shall be calculated from the impulse-response windowed, level-adjusted transfer functions for all measurement positions. For instructions on how to calculate the power average for each frequency, see Annex G.

4.7.2.6 Smooth the Results

Apply one-octave smoothing to the power-averaged magnitude response.

4.7.2.7 Normalize the Results to the Mid-Band Average Level

Calculate the log-frequency based average level of the one-octave smoothed, power-averaged magnitude response over the frequency range of 355 Hz to 2.82 kHz, inclusive. This result will be the normalization offset. For instructions on how to calculate a level average, see Annex G.

Subtract the normalization offset from the level of the one-octave smoothed, power-averaged magnitude response. This will normalize the one-octave smoothed, power-averaged magnitude response so that the average level in the mid band is at 0 dB.

If the normalization offset is a positive value, the one-octave smoothed, power-averaged magnitude response will be decreased. If the normalization offset is a negative value, the one-octave smoothed, power-averaged magnitude response will be increased.

[1] These are the lower and upper frequency limits, respectively, for the 250 Hz and 8 kHz one-third octave bands.

4.7.3 Identify the System Tolerance Limits

Once the one-octave smoothed, power-averaged response has been prepared as described in Section 4.7.2.7, and the frequency limits for the system purpose have been determined in Section 4.5, the tolerance limits of the system shall be identified.

The spectral balance performance classifications are SB-A, SB-B, SB-C, SB-D, or SB-F, where SB-A is the tightest tolerance limit, SB-D is the loosest tolerance limit, and SB-F falls outside of all limits. The system purpose indicates the frequency range over which the performance classification shall be evaluated.

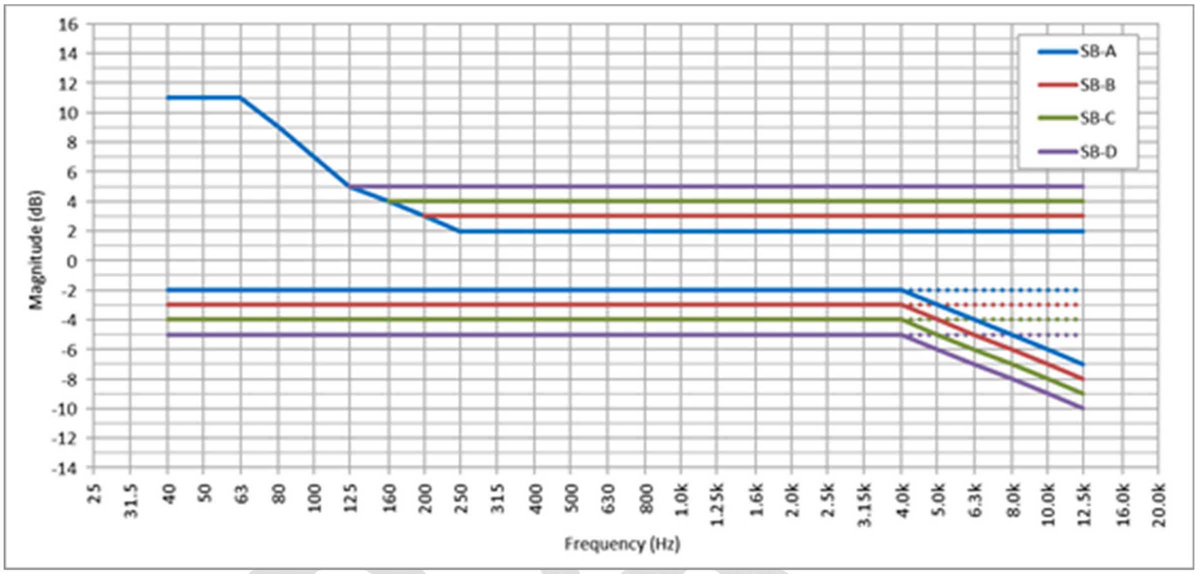


Figure 24 – Tolerance limits for spectral balance (SB) performance classifications

In Figure 24, the dotted lines represent the preferred frequency response. However, a roll-off above 4 kHz is allowed.

To see the tolerance limits represented in tabular form, see Annex I.

4.7.3.1 SB-A

If the one-octave smoothed, power-averaged response as plotted on the tolerance graph does not exceed the upper or lower limits of Tolerance Limit A in Figure 25, the performance classification shall be **SB-A.**

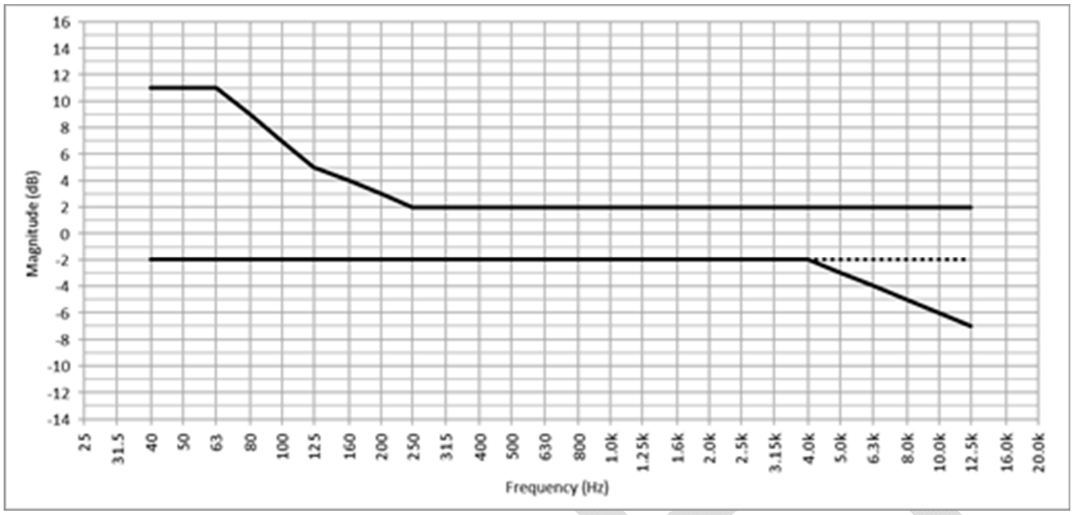


Figure 25 – Tolerance limits for SB-A

The dotted lines represent the preferred frequency response. However, a roll-off above 4 kHz is allowed.

4.7.3.2 SB-B

If the one-octave smoothed, power-averaged response as plotted on the tolerance graph exceeds the limits of Tolerance Limit A but does not exceed the upper or lower limits of Tolerance Limit B in Figure 26, the performance classification shall be **SB-B.**

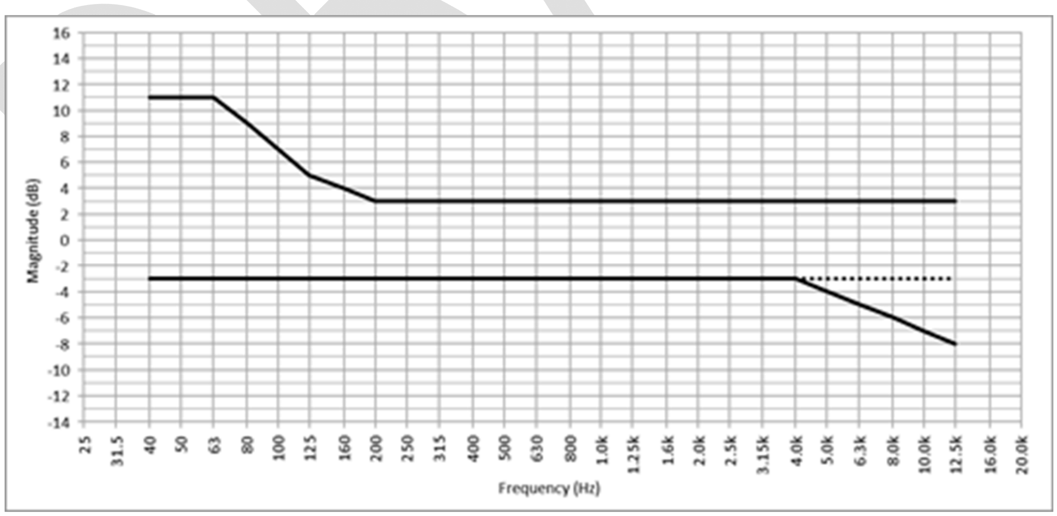


Figure 26 – Tolerance limits for SB-B

The dotted lines represent the preferred frequency response. However, a roll-off above 4 kHz is allowed.

4.7.3.3 SB-C

If the one-octave smoothed, power-averaged response as plotted on the tolerance graph exceeds the limits of Tolerance Limit B but does not exceed the upper or lower limits of Tolerance Limit C in Figure 27, the performance classification shall be **SB-C.**

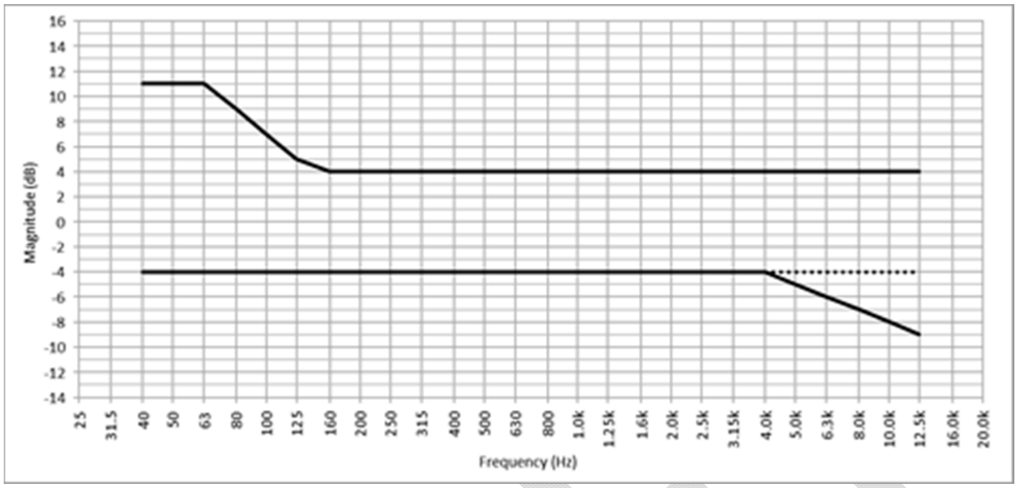


Figure 27 – Tolerance limits for SB-C

The dotted lines represent the preferred frequency response. However, a roll-off above 4 kHz is allowed.

4.7.3.4 SB-D

If the one-octave smoothed, power-averaged response as plotted on the tolerance graph exceeds the limits of Tolerance Limit C but does not exceed the upper or lower limits of Tolerance Limit D in Figure 28, the performance classification shall be **SB-D.**

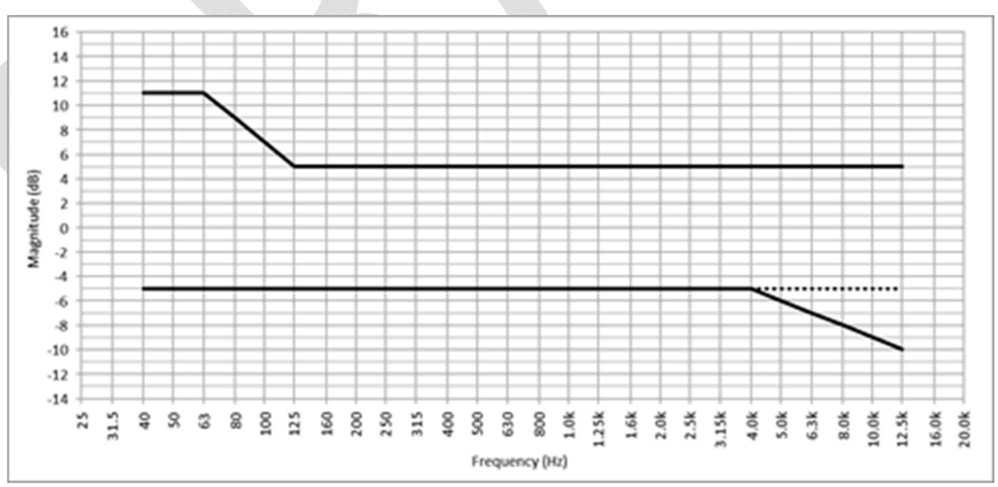


Figure 28 – Tolerance limits for SB-D

The dotted lines represent the preferred frequency response. However, a roll-off above 4 kHz is allowed.

4.7.3.5 SB-F

If the one-octave smoothed, power-averaged response as plotted on the tolerance graph exceeds the limits of Tolerance Limit D, the performance classification shall be **SB-F.**

5 Performance Classification

5.1 System Classification

The performance classification of the system is determined by the tolerance limit into which the one-octave smoothed, power-averaged response fits. Upon completion of the measurements, data processing, and analysis, the system shall be classified as SB-A, SB-B, SB-C, SB-D, or SB-F, as described in Section 4.7.3 using the specified tolerance limits. The system purpose (full, limited, or paging) and the average standard deviation value shall also be reported.

This shall be reported in the following format:

SB-[Performance Classification]-[System Purpose]-[Average Standard Deviation Value]

For example, a system could be described as SB-D-Full-4.5 or SB-C-Paging-3.2.

Each channel of a multi-channel loudspeaker system shall be measured, processed, evaluated, and reported independently.

5.2 Reporting

A test report shall be generated which at a minimum contains:

a) A tabular form (such as in Section 5.3) indicating:
   1) The system purpose
   2) The test signal used
   3) The measurement equipment used
   4) The frequency spacing of the measurement data

b) A plan that designates measurement locations and their unique, assigned numbers

c) The measured ANL

d) Graph of the average frequency response with tolerance limits as specified in Section 4.7.3

e) Graph of the standard deviation as specified in Section 4.7.2.4

f) The performance classification
   1) A performance classification shall be generated for each loudspeaker system channel and/or room configuration.

5.3 Test Report Format Example

Venue, Date, and Evaluator		
Venue	Name:	
	Location:	
Date:		
Evaluator:		

Measurement Equipment	
Calibrator:	Calibration Date:
Computer/Measurement Device:	Measurement Software:
Measurement Tools:	Microphone:
Test Signal:	Pre-Amplifier:
Frequency Spacing:	Other

System Purpose (Circle One)		
Paging (250 Hz to 4 kHz)	Limited (100 Hz to 10 kHz)	Full (40 Hz to 12.5 kHz)

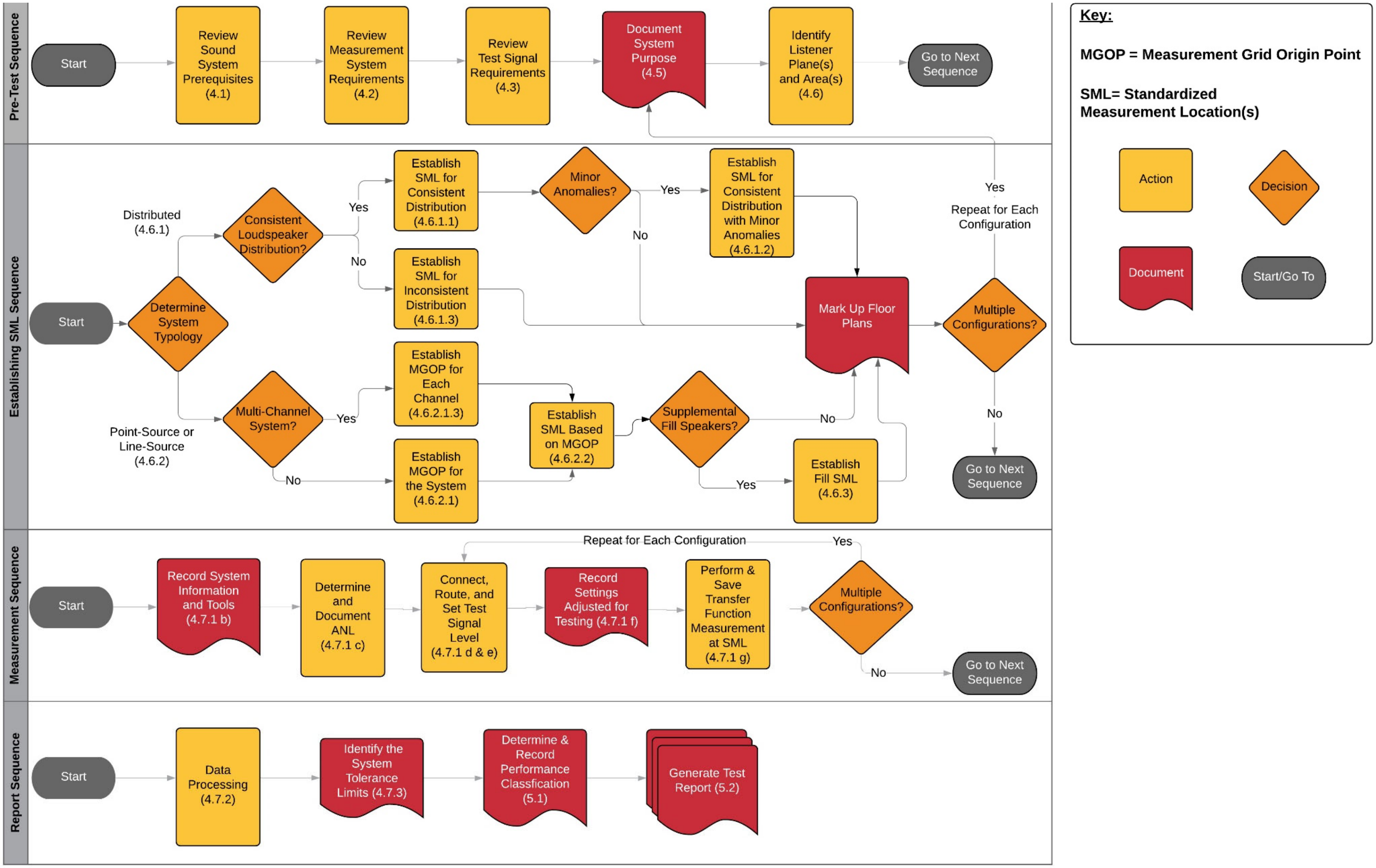
Performance Classification
Notes & Explanation:

Measurements		
One-Octave Band	**Ambient Noise Level (LZ_{eq})**	**Ambient Noise Level + 15 dB**
63 Hz		
125 Hz		
250 Hz		
500 Hz		
1 kHz		
2 kHz		
4 kHz		
8 kHz		

Included with report:

- Documentation that shows each channel plotted on a graph with tolerance limits
- Documentation that shows standard deviation graph for each channel
- Plan and/or elevation drawings showing measurement locations

695 # Annex A: Process Map (Normative Annex)



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Annex B: Measurement Microphones and Transmission (Informative Annex)

B.1 Measurement Microphone Frequency Response

The frequency response tolerances for sound level meters are often used to define the grade of a microphone's frequency response. To measure the frequency response of a sound system accurately, microphones need to have as flat a response as is practical. Therefore, as stated in Section 4.2, this Standard requires that instrumentation used for measurements specified within this Standard shall conform to the frequency response requirements of Class 1 sound level meter systems per IEC 61672-1:2013.

Figure B.1 shows the frequency response requirements for Class 1 sound level meters. This figure is based on a table provided in IEC 61672-1:2013 that shows sound level meter response limits.

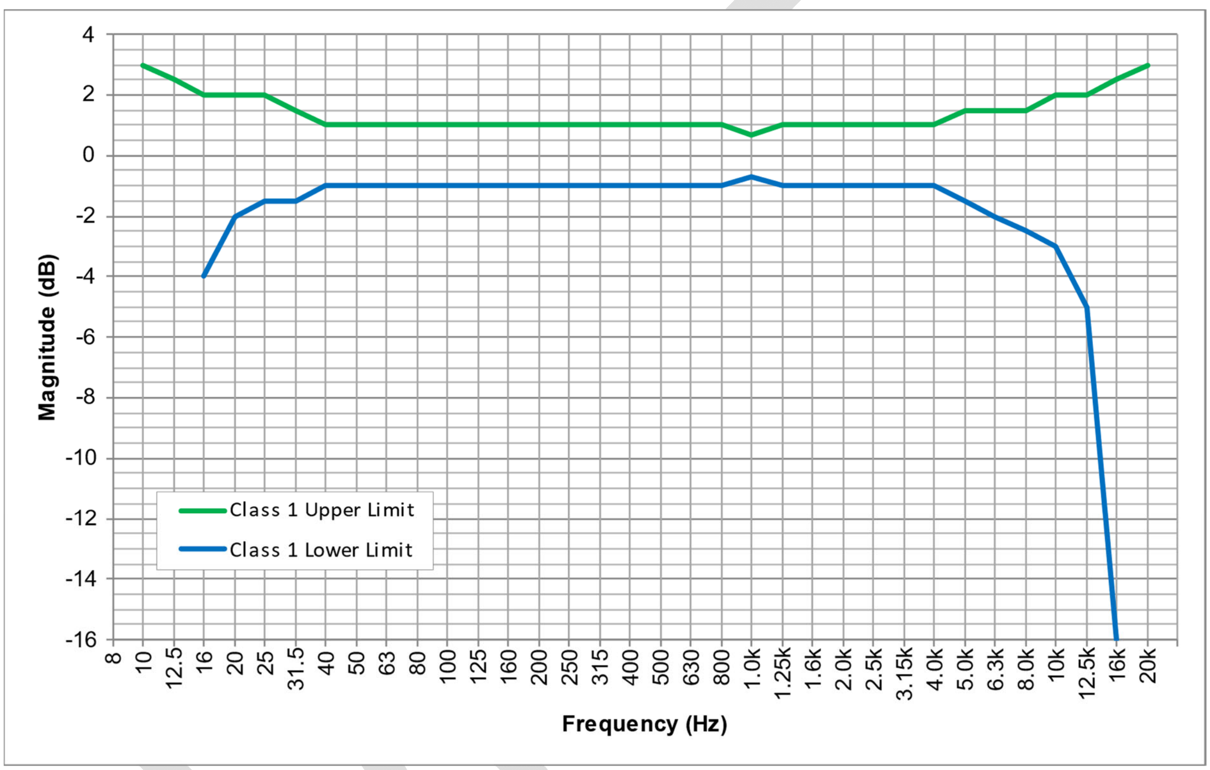

Figure B.1 – Class 1 sound level meter response (per IEC 61672-1:2013)

B.2 Measurement Microphone Diaphragms

There are many types of measurement microphones on the market. Many use polymer diaphragms that are inherently temperature sensitive. If such a microphone is exposed, even briefly, to excessive temperatures, then the response of the microphone will often change. Therefore, it is recommended that microphones with metal diaphragms be used, because they have better stability of their frequency response with time and temperature.

All microphones, especially those with polymer diaphragms, should be checked regularly for conformance with the Class 1 frequency response curve.

B.3 Microphone Correction Curves

Inexpensive measurement microphones are often provided with correction curves that must be applied to the microphone response in order to achieve a stated accuracy. The use of a microphone that requires correction to achieve the frequency response of a Class 1 sound level meter is not recommended.

B.4 Wired Audio Links

The use of long cables with relatively high capacitance can have negative effects on the frequency response of a measurement system. Excess capacitance in the wiring can cause a level reduction in the high-frequency region of the signal. Wired audio links that are longer than 30 m (100 ft) such as cables for measurement microphones should be compared with a 15 m (50 ft) or shorter cable using the source (typically a microphone) and load (typically a pre-amplifier) the longer cable will be used with, to make sure functionally equivalent results are obtained. In general, low capacitance cables such as those designed for digital audio applications (AES3) will allow longer cable lengths to be used without performance loss.

B.5 Wireless Audio Links

If wireless audio links are used, they should first be compared to a relatively short cable, no greater than 15 m (50 ft), to make sure functionally equivalent results are obtained. Wireless, analog links utilizing companding are not recommended. Digital, wireless audio links are generally preferred since the transmission tends to be closer to that obtained with a cable.

Annex C: System Purposes Guidance (Informative Annex)

This Standard provides three system purposes based upon a system's intended usage. The source material being reproduced and the purpose a system is serving determines the minimum frequency ranges over which each system purpose should be evaluated.

Paging Systems: The primary function of a paging system is to communicate short voice messages. A higher value is placed on communicating the message rather than faithful reproduction of the source material. Examples of similar systems include traditional telephone systems (POTS) with a nominal frequency range of 300 Hz to 3 kHz and an alarm system's voice announcement, with a nominal frequency range of 400 Hz to 4 kHz. To encompass the above examples, this Standard defines paging systems as operating from 250 Hz to 4 kHz.

Limited-bandwidth Systems: The primary function of a limited-bandwidth system is the reproduction of speech. These systems are often found in ballrooms, lecture halls, and conference rooms. The system needs to accurately reproduce spoken word. The lower limit of the evaluation range is established based upon the male voice's fundamental frequencies. The upper limit of the evaluation range captures the upper frequencies of speech consonant sounds. Therefore, this Standard evaluates limited-bandwidth systems from 100 Hz to 10 kHz. A limited-bandwidth system will also function for background music because of its similar frequency content.

Full-bandwidth System: The primary function of a full-bandwidth system is the reproduction of full range music. The system must reproduce the fundamental frequencies of the lowest instruments such as bass guitar (41 Hz) and piano (27 Hz), as well as the upper (typically third order) harmonics of instruments such as piano (12.3 kHz), piccolo (11.9 kHz,) and cymbals (16 kHz+). This Standard, however, excludes evaluation of the highest frequencies due to the potential for repeatability issues measuring very high frequencies. Therefore, this Standard evaluates full-bandwidth systems from 40 Hz to 12.5 kHz.

Annex D: Justifications for Measurement Locations (Informative Annex)

D.1 Distributed Loudspeaker Systems

In a system where the spacing of distributed loudspeakers and the distance from the loudspeakers to the listening plane are consistent, the repeatability of the layout and the predictability of the loudspeakers' interactions permit the use of the simplified measurement technique detailed in Section 4.6.1. The work of Rex Sinclair establishes that:[2]

- The loudest measurement from a single loudspeaker will occur directly on-axis of the loudspeaker, in that it is the shortest distance from the loudspeaker to the measurement microphone.
- The greatest summation from any two loudspeakers will occur at the midpoint of a line directly between the two loudspeakers.
- The greatest contribution from multiple loudspeakers will occur at the point equidistant from all adjacent loudspeakers, as typically found in a hexagonal or square grid.
- An edge of a loudspeaker's coverage pattern that does not overlap with that of another loudspeaker will have the lowest measurement value, as that location is the greatest distance from the loudspeaker that sound will travel to reach the measurement microphone. This occurs at the edge of the listening area, off-axis of a loudspeaker.

These four assumptions form the basis for the required measurement locations for distributed loudspeaker systems in this Standard.

When a consistent distribution pattern is not present, the Standard requires a similar set of measurements for each unique loudspeaker layout pattern.

D.2 Point-Source and Line-Source Loudspeaker Systems

This Standard applies a measurement-point distribution scheme that measures the coverage of a loudspeaker system at a consistent angular resolution, regardless of the distance from the loudspeaker(s) to the listener area(s). The partial-sphere wavefront from a loudspeaker or a group of loudspeakers expands radially, so it is fitting to measure it radially, giving equal weight to every portion of the coverage pattern. This is accomplished by distributing points throughout the listening plane on a radial grid, which originates from the measurement grid origin point(s) of the loudspeaker system(s) under test.

Spacing of the radial measurement locations was determined through a series of onsite measurements performed by the *ANSI/AVIXA A102.01:2017, Audio Coverage Uniformity in Listener Areas* Task Group. Members laid out a seating grid and collected measurement data at each seat. Analyzing the collected data, the task group members determined that variances in data occurred at about 3.7 to 4.6 m (about 12 to 15 ft). Using a 4.3 m (14 ft) spacing, the members trigonometrically calculated and verified angles by comparing the resulting radial measurement grids to the initial series of onsite measurements. From this exercise, it was determined the angles for measurement location spacing would be 5 degrees vertically and 20 degrees horizontally.

Every site is unique. This Standard identifies the minimum number of points necessary to characterize the spectral balance of a system. During the course of testing a user may discover

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[2] Rex Sinclair, "The Design of Distributed Sound Systems from Uniformity of Coverage and Other Sound-Field Considerations," *Journal of the Audio Engineering Society* 30, no. 12 (December 1982): 871-881, http://www.aes.org/e-lib/browse.cfm?elib=3805.

the grid does not adequately characterize a particular condition in the listener area(s). This Standard allows the user to add and document measurement locations as deemed necessary.

Annex E: Early Arriving Energy and the 50 millisecond Window (Informative Annex)

This Standard strives to quantify sound systems in a way that is meaningful in terms of how human listeners perceive sound. Psychoacoustic research dating back to 1948 demonstrates that for speech, the human ear and brain (the "hearing system") can integrate the first arrival of sound from a source with other acoustical energy arriving within 25-35 ms of the first arrival.[3] This neuro-physical integration allows listeners to perceive the signals as a single source without seriously affecting intelligibility.

Subsequent research has shown that this integration of differences in arrival times extends out to 50 milliseconds for speech signals.[4] This is often referred to as “precedence,” the “Haas effect,” or the “law of the first wavefront.” Late-arriving energy (after 50 ms for speech or about 100 ms for music) results in a decrease in clarity due to the distinguishability of multiple arrivals at the listener’s ears. In keeping with this research, this Standard limits the time window to the first 50 ms after arrival of the direct sound.

The wide availability of measurement tools that can capture impulse response windowed measurements, combined with the inability of listeners to separate direct from early arriving sounds, allows the output of this Standard to reflect a listener’s experience more accurately than by solely measuring the direct sound or by using time-blind measurement tools.

A common method for limiting the integration period of a frequency response measurement is to capture the response of a system under evaluation over some period greater than the desired integration period and apply an appropriately sized window function to the impulse response (IR) in the time domain. The windowed IR is then transformed into the frequency domain by a discrete Fourier transform (DFT) or fast Fourier transform (FFT) for evaluation of its magnitude response spectrum.

The impulse response of a system and its complex transfer function, from which the magnitude response spectrum is calculated, are related by the Fourier transform. The forward Fourier transform of the IR yields the transfer function (magnitude and phase response) of the system in the frequency domain; the inverse Fourier transform of the transfer function in the frequency domain produces the IR in the time domain.

[3] Helmut Haas, “The Influence of Single Echo on Audibility of Speech,” *Journal of the Audio Engineering Society* 20, no. 2 (March 1972): 146-159, http://www.aes.org/e-lib/browse.cfm?elib=18873.

[4] Ruth Y. Litovsky and H. Steven Colburn, “The Precedence Effect,” *The Journal of the Acoustical Society of America* 106, no. 4 (August 1999): 1633, https://doi.org/10.1121/1.427914.

Annex F: Impulse Response Window Selection and Application (Normative Annex)

The requirements for IR windowing are:

a) The highest peak of the IR shall be aligned with the portion of the IR window where the least attenuation occurs.

b) The right half, or trailing edge of the IR window shall taper smoothly to be closed (terminal attenuation) at a point 50 ms after the highest peak in the IR.

c) The left half or leading edge of the IR window shall not truncate the IR too early or too abruptly so as to smoothly encompass the full arrival of direct sound from the loudspeaker system under evaluation.

The precise shape of the right half window is not critical. The shape of the left half window is less critical. Different time-window functions produce different smoothing functions in the frequency domain; however, given the 50 ms length of the right half window, small differences in window shape are not expected to produce significantly different results after one-octave smoothing is applied to the power-average of multiple measurements or after broadband summation.

Since the IR of an electro-acoustical system is a causal, one-sided function, the left half of the IR window may be symmetrical or asymmetrical relative to the right side of the window. The right side may be fully or partially tapered.

In all cases, the window shall be positioned relative to the measured IR such that the highest peak in the IR aligns with the portion of the time window where the IR being windowed is least attenuated. In the case of a symmetrical, fully tapered window function such as a raised cosine window function (popularly called a Hann window), the point of minimum attenuation will occur exactly in the center of the window, and the required full window length would be 100 ms (50 ms before the center of the window and 50 ms after the center of the window). When using a hybrid window such as a “right half” or Tukey window consisting of both rectangular (no attenuation) and tapered window segments, the peak of the IR may be positioned anywhere in the rectangular portion of the window such that the peak of the IR is not attenuated.

Figure F.1 shows three examples of IR window functions that satisfy the requirements of this standard and demonstrate proper alignment with the peak in the IR. The first example is a symmetric, fully tapered raised cosine window with the peak in the IR positioned exactly in the center of the window. The center graph shows an asymmetrical hybrid window with a rectangular middle section. The third example is a “right half” window with a rectangular left side and a tapered right side.

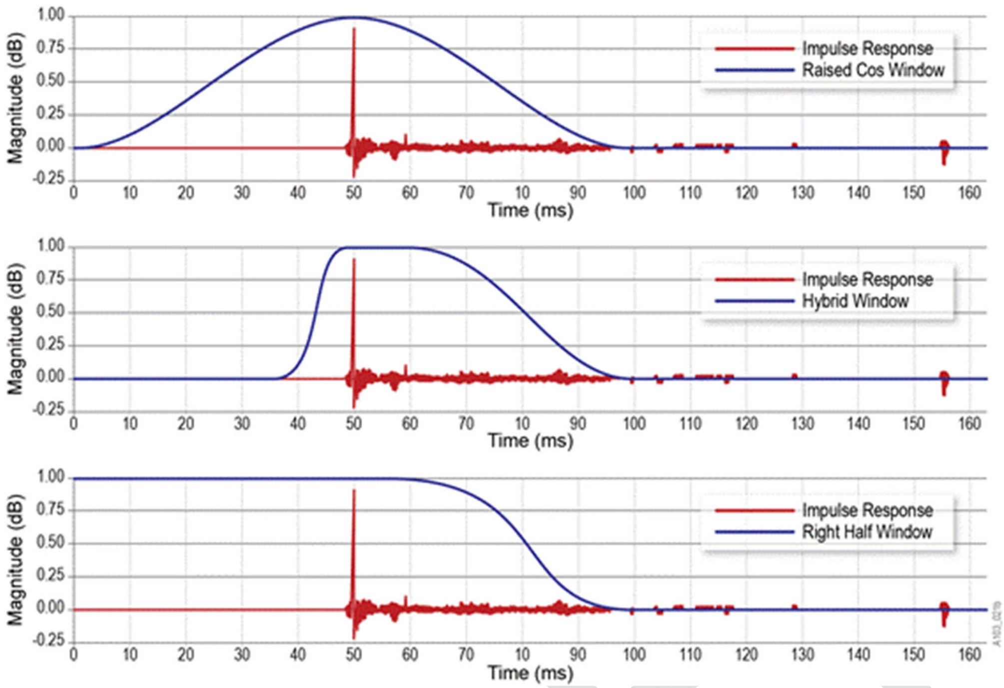

Figure F.1 – Three examples of conforming IR time windows

Notice that in the latter two examples, the rectangular portion of the window extends later in time, past the peak in the IR, giving low frequencies more time to "ring out" before the window function begins to attenuate the IR. This can result in less attenuation of the measured low frequency response relative to measurements made using fully tapered window functions, particularly when measuring full bandwidth systems.

A related concern when measuring a multi-driver loudspeaker system and/or a system comprising multiple enclosures is that the highest peak in the IR will typically coincide with the arrival of energy from the high-frequency element(s), which may not be the earliest arriving sound from the system under test. Care must therefore be exercised when working with asymmetrical windows, to ensure that the leading edge of the window does not truncate the IR too early or too abruptly before the peak to smoothly encompass the full arrival of direct sound from the system under test.

Applying the time window after selection and alignment with the IR is a matter of multiplying the amplitude of each sample in the IR by the amplitude in the corresponding sample of the time window function. Samples outside the non-zero portion of the time window function may be set to zero or discarded depending on the DFT/FFT size to be used to transform the windowed IR to the frequency domain.

F.1 Transforming the Windowed IR

When transforming the windowed IR into the frequency domain for evaluation of its magnitude response spectrum, the DFT/FFT size in samples shall be at least equal to the full length of the time window used, inclusive of both the left and right sides of the window. If a DFT/FFT size greater than the time window size is used, the value of all samples exceeding the length of the time windowed IR shall be set to zero. This is commonly referred to as "zero padding."

F.2 Complex Smoothing in the Frequency Domain as an Alternative to Impulse-Response Windowing

When the impulse-response windowed measurement is transformed into the frequency domain, the DFT of the window function becomes convolved with the DFT of the original, un-windowed measurement and the practical result is a complex smoothing function in the frequency domain.[5] Because the resulting smoothed transfer function and the windowed IR are related by their Fourier transforms, it is also possible to obtain a functionally equivalent result by applying a complex smoothing function in the frequency domain. This is sometimes referred to as frequency-domain windowing and it may be a desirable alternative to IR time windowing for measurements originating in the frequency domain.[6] For example, complex smoothing in the frequency domain using a smoothing function with an effective bandwidth of 20 Hz corresponds to a 50 ms half window length in the time domain.

Because timing relationships encoded in the complex transfer function are discarded when magnitude is calculated, magnitude or power smoothing in the frequency domain is not an alternative to impulse-response time windowing.

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[5] Richard C. Heyser, "Determination of Loudspeaker Signal Arrival Times, Part 1," *Journal of the Audio Engineering Society* 19, no. 9 (October 1971): 734-743, http://www.aes.org/e-lib/browse.cfm?elib=2136.

[6] Richard G. Lyons, *Understanding Digital Signal Processing*, 3rd ed. (Upper Saddle River, NJ: Pearson, 2011).

Annex G: Power Averaging (Normative Annex)

For averaging transfer function measurements at multiple microphone locations, power, or energy, averaging shall be used. This differs from other methods of averaging complex signals or transfer functions (i.e., those having both magnitude and phase information).

Field quantities (e.g., volts or pascals) are always used in the calculations. Decibel (logarithmic) values are never used to calculate a power average. The level values must be converted to field quantities (linear) prior to use in the calculations.

Power averaging will bias the result towards the measurement with the highest level. Therefore, level adjustment of each individual measurement is required prior to calculating the power average.

A power average employs the root mean square (rms) method. This technique uses the square of the amplitude values at each frequency data point, calculates the average (mean) of the squared values, and determines the square root of the mean. A decibel value is then calculated from this result. The mathematics for this method are as follows.

$$L_p = 10\log\left(\frac{P^2}{{p_0}^2}\right) = 20\log\left(\frac{P}{P_0}\right)$$

This can then be expressed as:

$$\left(\frac{P_i}{P_0}\right)^2 = 10^{\frac{L_i}{10}}$$

where:

i is the index number for a particular measurement (1, 2, 3, …n), and

n is total number of measurements.

The average is simply written as the sum of the antilogarithmic divided by the number of the measurements:

$$Mag_{Avg} = \frac{1}{n}\left(\sum_{(i=1)}^{n} 10^{(L_i/10)}\right)$$

or

$$Mag_{Avg} = \frac{(Mag_1 + Mag_2 + \ldots + Mag_n)}{total\ number\ of\ measurements}$$

This needs to be converted back to sound pressure level:

$$L_p = 10\log\left(\frac{1}{n}\left(\sum_{(i=1)}^{n} 10^{(L_i/10)}\right)\right)$$

or

$$L_p = 10 * \log_{10}(Mag_{Avg})$$

Where p and L_p are the sound pressure and sound pressure level respectively and n represents the number of measurements in the average.

This is the power-averaged level at a given frequency for all the measurements included in the calculations. Figure G.1 shows the results for a calculation of the power average for the 500 Hz data point of three different measurements.

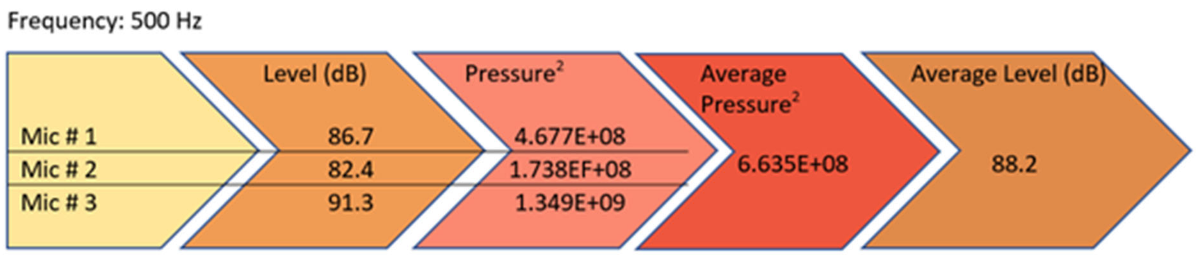

Figure G.1 – Power Averaging Example

It would not be appropriate to use a complex average (also called a vector average) for averaging multiple microphone locations to determine the subjectively perceived average for the different locations. Any differences in the phase of the transfer functions among the measurements could lead to large differences in the average that would not correspond to any meaningful or perceived differences in the sound level among the various microphone locations used for the measurements. An example of a vector average would be to combine the outputs of multiple microphones using a mixing console and use the output of the mixer for the averaged measurement result. This method shall not be used for this Standard.

Microphone multiplexers shall not be used for this Standard.

Annex H: Averaging Over a Specified Frequency Range (Normative Annex)

H.1 Log-spaced Data Points

For calculating the average level of the magnitude response over a specified frequency range, the levels (decibel value) at each frequency data point within the specified range are added together. The total value is then divided by the number of frequency data points.

$$Level_{Avg} = \frac{1}{n}\left(\sum_{(i=0)}^{n} Level_{Fi}\right)$$

or

$$Level_{Avg} = \frac{(Level_{F0} + Level_{F1} + \dots + Level_{Fn})}{total\ number\ of\ frequency\ data\ points}$$

Where n = number of frequency data points

H.2 Linear-Spaced Data Points

When calculating a single-figure average for a specified range of decibel magnitude values at linearly spaced frequency intervals, a log-weighted average shall be used for consistency with averaged values calculated from log-spaced frequency data. A weighted average is the sum of the products of the values being averaged and their weighting factors, divided by the sum of the weighting factors. The weighting factor for each linear-spaced value in the log-weighted average is the ratio of its frequency (F_i) to the start frequency (F_0) of the range to be averaged (F_0/F_i).

$$Level_{Avg} = \frac{\left(\sum_{(i=0)}^{n} \frac{F_0}{F_i} Level_{Fi}\right)}{\left(\sum_{(i=0)}^{n} \frac{F_0}{F_i}\right)}$$

or

$$Level_{Avg} = \frac{(\frac{F_0}{F_0} Level_{F0} + \frac{F_0}{F_1} Level_{F1} + \dots + \frac{F_0}{F_n} Level_{Fn})}{(\frac{F_0}{F_0} + \frac{F_0}{F_1} + \dots + \frac{F_0}{F_n})}$$

H.3 Standard Deviation

The average standard deviation for a frequency range shall be calculated using a simple, arithmetic average of log-spaced frequency values (see H.1) or a log-weighted average of linear-spaced frequency values (see H.2)

Annex I: Performance Classification Tables (Normative Annex)

This Annex contains tables of the tolerance limits for performance classifications SB-A, SB-B, SB-C, and SB-D. The related requirements can be found in Section 4.7.3.

Table I.1 – Tolerance Limits for SB-A

Frequency (Hz)	Magnitude (dB): Lower Limit	Magnitude (dB): Upper Limit
40	-2	11
50	-2	11
63	-2	11
80	-2	9
100	-2	7
125	-2	5
160	-2	4
200	-2	3
250	-2	2
315	-2	2
400	-2	2
500	-2	2
630	-2	2
800	-2	2
1.0 k	-2	2
1.25 k	-2	2
1.6 k	-2	2
2.0 k	-2	2
2.5 k	-2	2
3.15 k	-2	2
4.0 k	-2	2
5.0 k	-3	2
6.3 k	-4	2
8.0 k	-5	2
10.0 k	-6	2
12.5 k	-7	2

976 **Table I.2 – Tolerance Limits for SB-B**

Frequency (Hz)	Magnitude (dB): Lower Limit	Magnitude (dB): Upper Limit
40	-3	11
50	-3	11
63	-3	11
80	-3	9
100	-3	7
125	-3	5
160	-3	4
200	-3	3
250	-3	3
315	-3	3
400	-3	3
500	-3	3
630	-3	3
800	-3	3
1.0 k	-3	3
1.25 k	-3	3
1.6 k	-3	3
2.0 k	-3	3
2.5 k	-3	3
3.15 k	-3	3
4.0 k	-3	3
5.0 k	-4	3
6.3 k	-5	3
8.0 k	-6	3
10.0 k	-7	3
12.5 k	-8	3

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978 **Table I.3 – Tolerance Limits for SB-C**

Frequency (Hz)	Magnitude (dB): Lower Limit	Magnitude (dB): Upper Limit
40	-4	11
50	-4	11
63	-4	11
80	-4	9
100	-4	7
125	-4	5
160	-4	4
200	-4	4
250	-4	4
315	-4	4
400	-4	4
500	-4	4
630	-4	4
800	-4	4
1.0 k	-4	4
1.25 k	-4	4
1.6 k	-4	4
2.0 k	-4	4
2.5 k	-4	4
3.15 k	-4	4
4.0 k	-4	4
5.0 k	-5	4
6.3 k	-6	4
8.0 k	-7	4
10.0 k	-8	4
12.5 k	-9	4

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Table I.4 – Tolerance Limits for SB-D

Frequency (Hz)	Magnitude (dB): Lower Limit	Magnitude (dB): Upper Limit
40	-5	11
50	-5	11
63	-5	11
80	-5	9
100	-5	7
125	-5	5
160	-5	5
200	-5	5
250	-5	5
315	-5	5
400	-5	5
500	-5	5
630	-5	5
800	-5	5
1.0 k	-5	5
1.25 k	-5	5
1.6 k	-5	5
2.0 k	-5	5
2.5 k	-5	5
3.15 k	-5	5
4.0 k	-5	5
5.0 k	-6	5
6.3 k	-7	5
8.0 k	-8	5
10.0 k	-9	5
12.5 k	-10	5

Annex J: Spectral Balance Ratings and User Experience (Informative Annex)

This Standard provides ratings for sound systems based on how balanced the average frequency response of the sound system is. A relatively uniform frequency response is typically perceived as neutral and “good sounding” to most listeners. This translates perceptually to a more accurate tonal balance.

The resulting response curve of this Standard is an average. This Standard does not quantify the response differences between different measurement locations or how they relate to the average response. To investigate the specific amount of deviation from seat to seat, the AVIXA A103.01:2017, *Audio Coverage Uniformity in Listener Areas* Standard may be utilized. That said, the experiences detailed below are a guideline, not a hard rule.

The following chart relates spectral balance ratings to typical user experiences:

Performance Classification	**Listener Experience**
SB-A	• Listeners may find it difficult to find differences in system spectral balance. • Everyone in the listener area has a similar audio experience.
SB-B	• Listeners may notice differences in the spectral balance if they move around the space. • Most listeners in the area have similar audio experiences, although some locations might have a different tonality than others.
SB-C	• Most listeners will notice differences in the spectral balance when they move around the space. Some listeners may find the sound quality less than desirable. •
SB-D	• Listeners are likely to notice significant differences in the spectral balance and will need to work to hear the content. • Listeners may feel the need to move to areas of better spectral balance to listen to content.
SB-F	• Listeners will notice extreme differences in the spectral balance and would likely describe sound quality as objectionable.

This Standard provides an objective measure of spectral balance; it does not provide a subjective qualification of a system. An acceptable spectral balance depends on the system's application or use. For instance, a speech reinforcement system might need a better rating than a background music system, just as a system in a performance hall needs a better rating than a bar band's system.

The user experience, as defined by the owner, determines the acceptability of the system’s spectral balance. It is for this reason the Standard does not attempt a pass/fail system of verification of conformance. While the following chart is somewhat subjective, it provides examples of system uses and possible acceptable spectral balance.

Standard deviation is a metric of confidence in consistency of the average spectral balance in the listening area. A system with a low standard deviation will deliver similar content across the listener areas. Users can have confidence that the average spectral balance curve is representative of the user's experience. Conversely, systems with higher standard deviation will have more spectral variations from seat to seat.

Listening Purpose	**System Uses**	**Performance Classification**		
		Might Be Suitable	**Good**	**Excellent**
Audio content is critical to the experience.	Concert Halls Conference Spaces Classrooms Houses of Worship	SB-B	SB-A	SB-A
Audio content is secondary to experience.	Sporting Venues Gymnasia Paging Systems	SB-C	SB-B	SB-A
Audio content is one of many contributors to the experience.	Environmental Experience Background Music Systems	SB-D	SB-C	SB-B
The audio content needs to be shared, but spectral balance is a low priority.	Parking Lot Paging Systems Transit Facility Systems	SB-F	SB-D	SB-C

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