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AVIXA A102.01:202X
Audio Coverage Uniformity in Listener Areas

SUPERCEDES AVIXA A102.01:2017

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Abstract

This Standard provides a procedure to measure and classify the uniformity of early arriving energy from a sound system across a listener area.

Keywords

ACU; audio coverage uniformity; audio system; early arriving sound; early arriving energy; listener area; sound pressure level; sound system; spatial coverage; uniformity

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Foreword

The performance of a sound system can be characterized by several factors including uniformity of coverage, tonal balance and consistency, gain before feedback, and maximum sound pressure level. This Standard focuses on the uniformity of coverage of a sound system's early arriving energy to the listener area(s). An ideal sound system design allows all listeners to hear reproduced content at approximately the same sound pressure level independent of the listener's position in a designated listener area. This performance Standard provides a procedure to measure and a means to classify the uniformity of coverage.

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A special thanks to Ann Brigida, CStd, CTS for her invaluable contributions towards the development of this Standard.

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1.1 Scope

This Standard defines parameters for characterizing a sound system's coverage of defined listener areas. It provides measurement procedures and performance classifications to assess the uniformity of coverage of a sound system's early arriving energy, with the goal of achieving consistent sound pressure levels throughout defined listener areas.

The procedure associated with this Standard is one of many verifications of the deployment and performance of a sound system. This Standard specifically excludes testing or measuring for spectral balance, gain before feedback, maximum sound pressure level, and other parameters required to assess the total performance of a sound system.

1.2 Purpose

The purpose of this performance Standard is to establish a method by which a sound system's coverage can be assessed and classified. This is accomplished by measuring and evaluating the uniformity of coverage of the early arriving energy from the loudspeaker system(s) throughout the designated listener area(s).

1.3 Application

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

1.4 Exceptions

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

This Standard is not intended for use in the following applications:

a) Cinema (refer to SMPTE: Society of Motion Picture and Television Engineers)

b) Home theater (refer to CEDIA: Custom Electronic Design & Installation Association)

c) Sound masking/speech privacy

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.

a) 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.

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

c) International Electrotechnical Commission. *Electroacoustics – Sound Level Meters – Part 2: Pattern Evaluation Tests*. IEC 61672-2:2013. Geneva: IEC, approved September 30, 2013.

d) Acoustical Society of America (ASA). *Procedure for Measuring the Ambient Noise Level in a Room*. ANSI/ASA S12.72. Melville, NY: ASA, approved 2015.

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) Acoustical Society of America. *Measurement of Sound Pressure Levels in Air*. ANSI/ASA S1.13-2020. Melville, NY: ASA, approved July 28, 2020.

c) Audio Engineering Society (AES). *Standard on Acoustics-Sound Source Modeling – Loudspeaker Polar Radiation Measurements*. AES56-2008 (r2019). New York, NY: AES, reaffirmed 2019.

d) Haas, Helmut. “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.

e) Sinclair, Rex. “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.

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 ACU: Audio Coverage Uniformity

3.1.2 ANL: Ambient Noise Level

3.1.3 DFT: Discrete Fourier Transform

3.1.4 FFT: Fast Fourier Transform

3.1.5 IR: Impulse Response

3.1.6 SPL: Sound Pressure Level

3.2 Definitions

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

3.2.1 Coverage envelope

The difference (in decibels) between the highest and lowest wideband measurement values recorded at the defined measurement points within the listener area(s).

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 70 Hz to 17.7 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 70 Hz to 11.2 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

Contiguous 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 areas. The system may be single or multi-channel in nature.

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 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.

3.2.13 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 175 Hz to 5.6 kHz.

3.2.14 Single-channel loudspeaker system

A loudspeaker system designed so that a single source feed is distributed to all designated coverage areas. An example is a system that contains a central loudspeaker cluster with delayed loudspeakers, which might be found in an auditorium or lecture hall.

3.3 Units

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

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.

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

   2) capturing a transfer function measurement (or an equivalent) which can be windowed with a 50 ms (or an equivalent) impulse response window. See Annex F for further details.

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 than the highest measured 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.

4.4 Audio Coverage Uniformity Process Map

For visual reference, Annex A contains a process map that shows the ACU 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 175 Hz to 5.6 kHz.

b) Limited Bandwidth System: systems used primarily for speech or other limited-bandwidth content shall be evaluated from 70 Hz to 11.2 kHz.

c) Full Bandwidth System: systems used primarily for music or other full-bandwidth content shall be evaluated from 70 Hz to 17.7 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 locations as required by site conditions.

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).

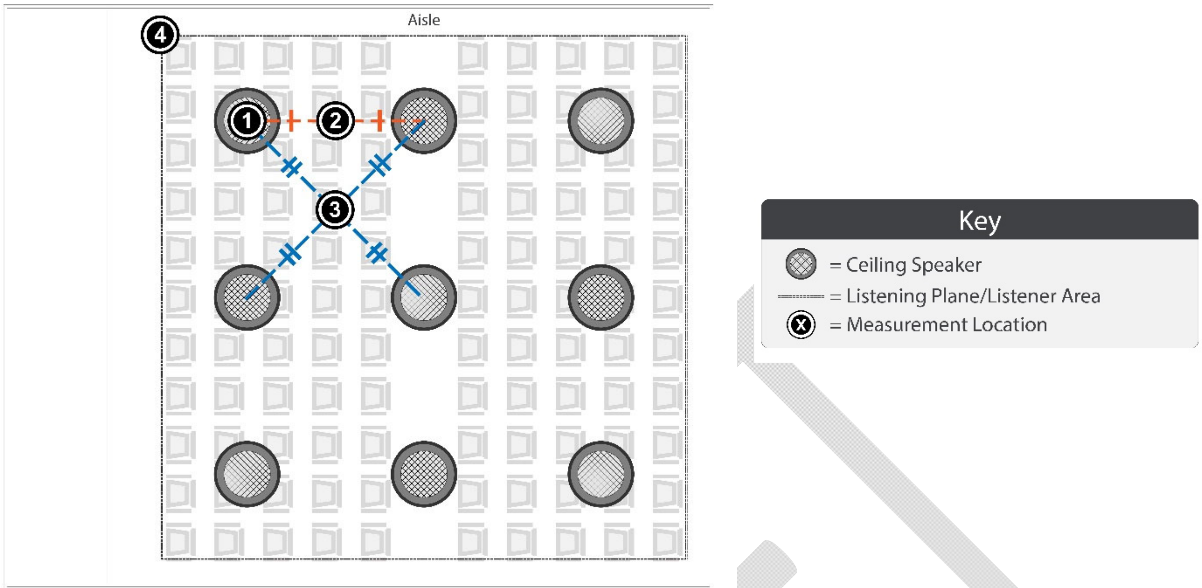


Figure 1 – Distributed loudspeaker measurement locations (plan view)

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)

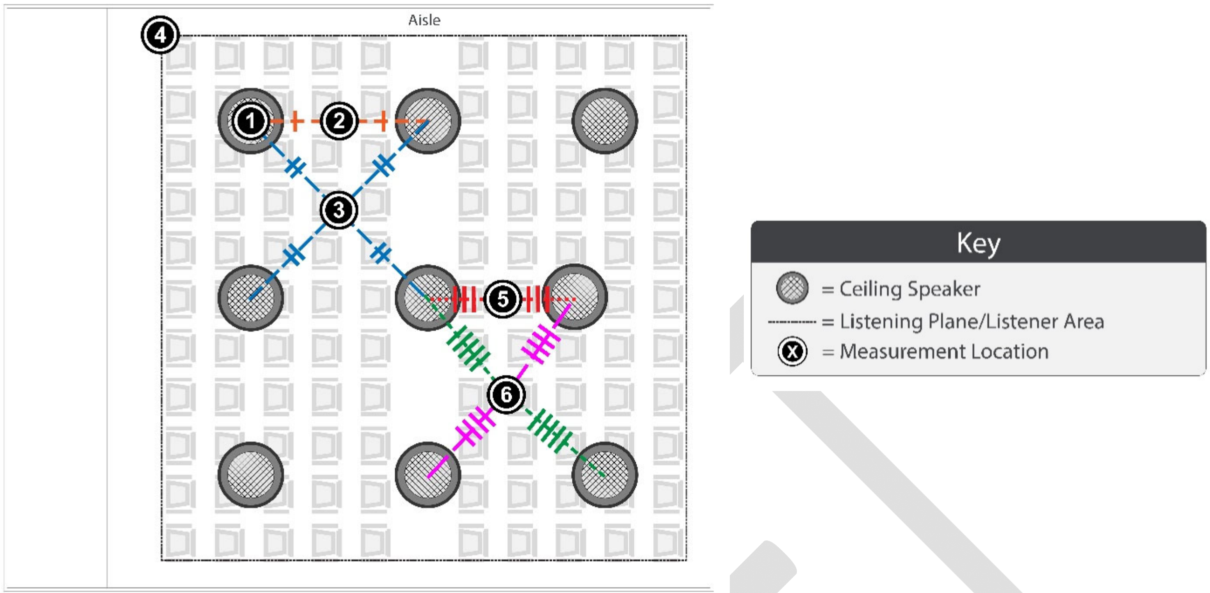
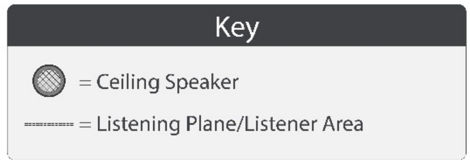
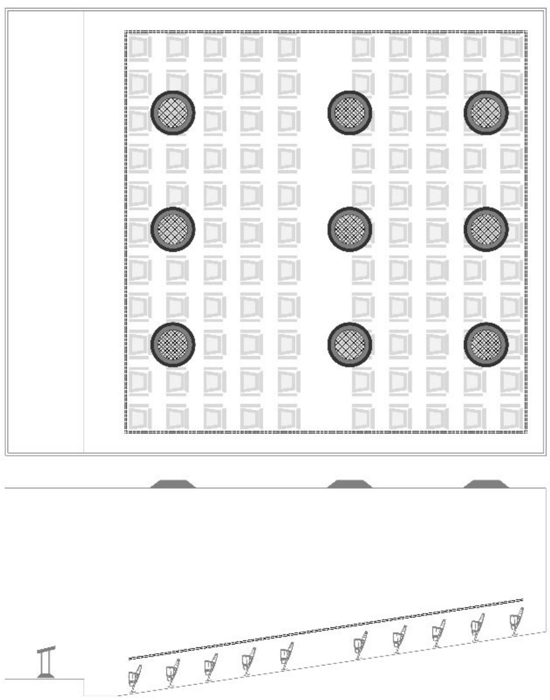
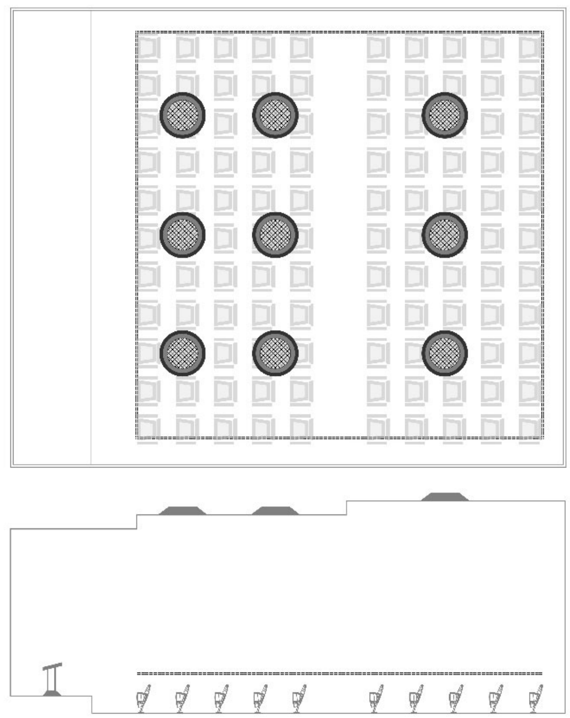


Figure 2 – Distributed loudspeaker measurement locations with minor anomalies (plan view)

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

<table>
<tr><td colspan="2">
</td></tr>
<tr><td>Figure 3 –Flat ceiling and sloped listener plane (plan and section view)</td><td>Figure 4 – Stepped ceiling and flat listener plane (plan and section view)</td></tr>
<tr><td></td><td></td></tr>
</table>

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 will 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.

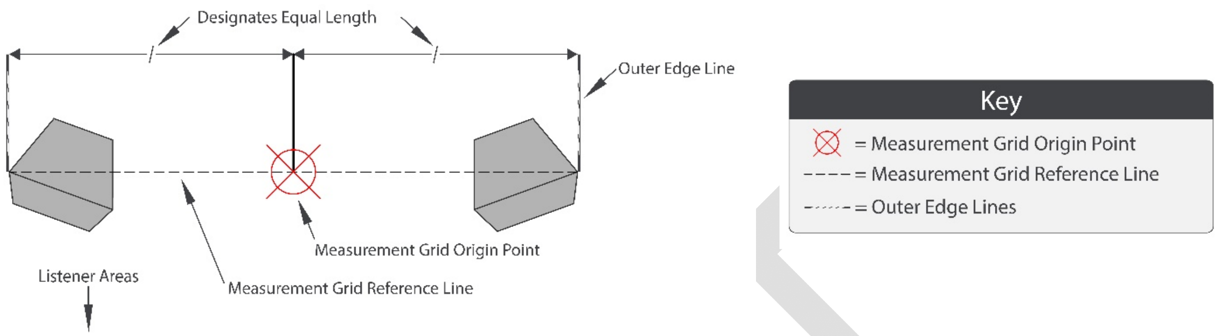


Figure 5 – Measurement grid origin point

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

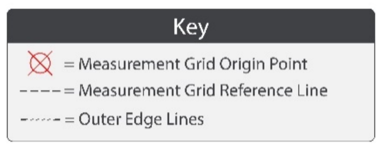


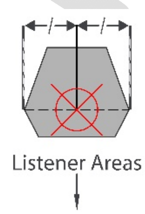
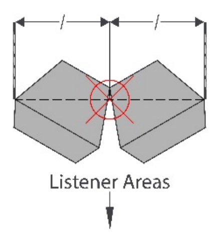
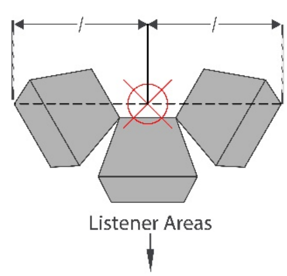
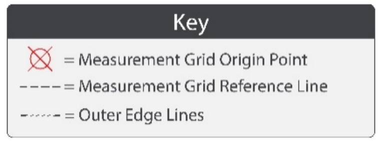
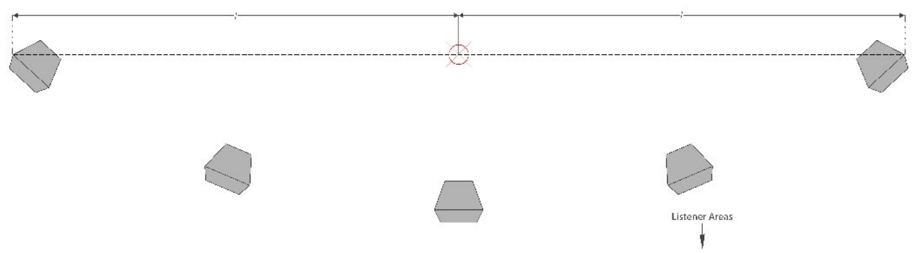
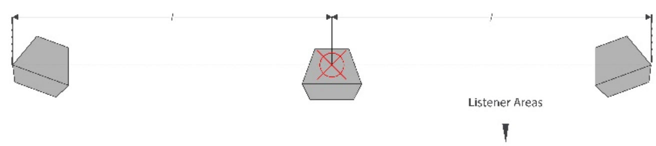
Figure 6 – Measurement grid origin point for a single loudspeaker	
Figure 7 – Measurement grid origin point for a two loudspeaker cluster	
Figure 8 – Measurement grid origin point for a three loudspeaker cluster	

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

<table>
<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>
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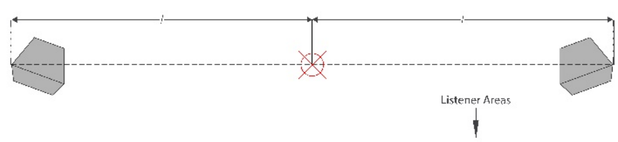
<tr><td>Figure 11 – Measurement grid origin point for a single-channel system with two loudspeaker locations</td><td>
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</table>

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

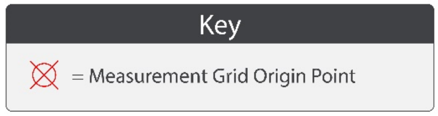
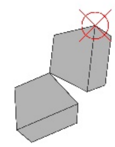
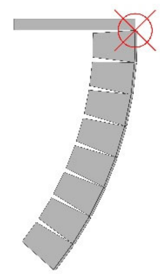


Figure 12 – Example of the vertical location of the measurement grid origin point of a two-loudspeaker cluster	
Figure 13 – Example of the vertical location of the measurement grid origin point of a multi-box line array	

4.6.2.1.3 Measurement Grid Origin Points for Multi-Channel Systems

Systems with multiple inputs 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.

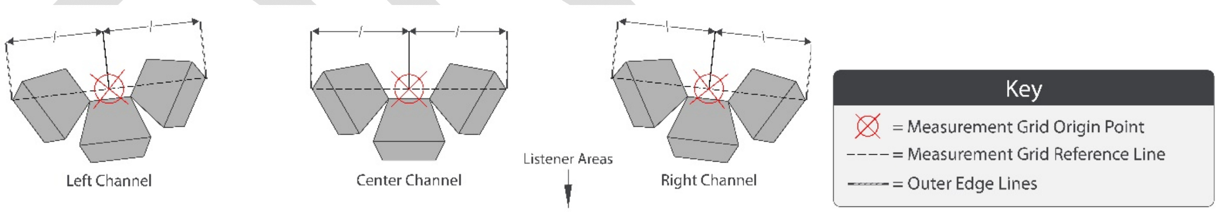


Figure 14 – Measurement grid origin points for a multi-channel left/center/right loudspeaker system

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.

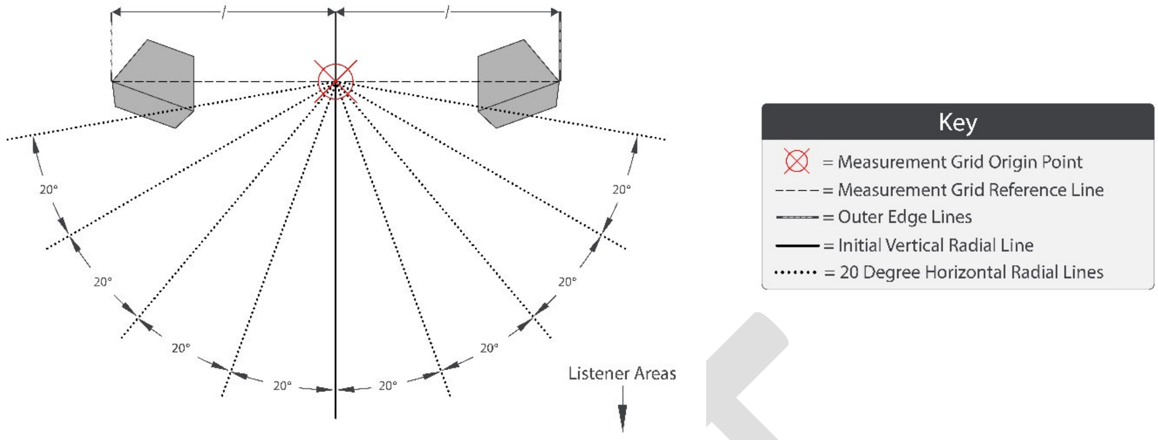


Figure 15 – Establishing radial lines (plan view)

c) The first measurement location on each horizontal radial line shall be the location on the horizontal radial furthest from the measurement grid origin point.

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. If the measurement grid origin point is not visible at this location, then move forward along the horizontal radial until it is visible and use this point as the second measurement location.

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

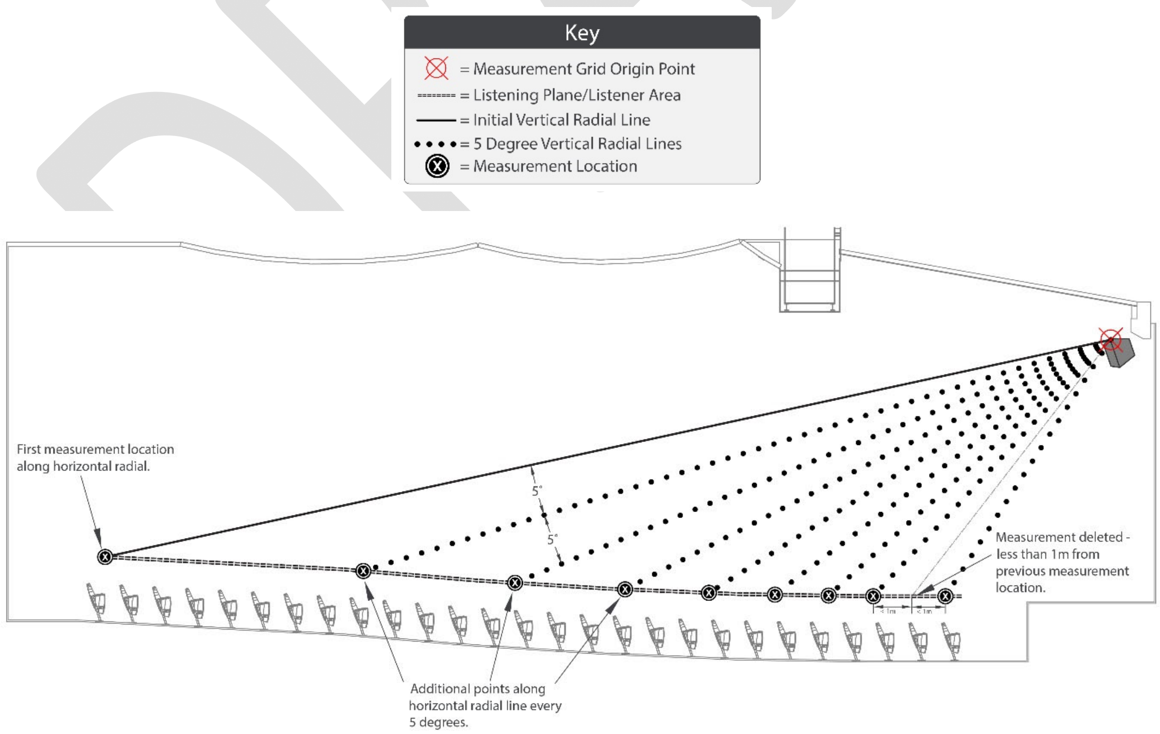


Figure 16 – 5-degree vertical radials along a horizontal radial (section view)

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.

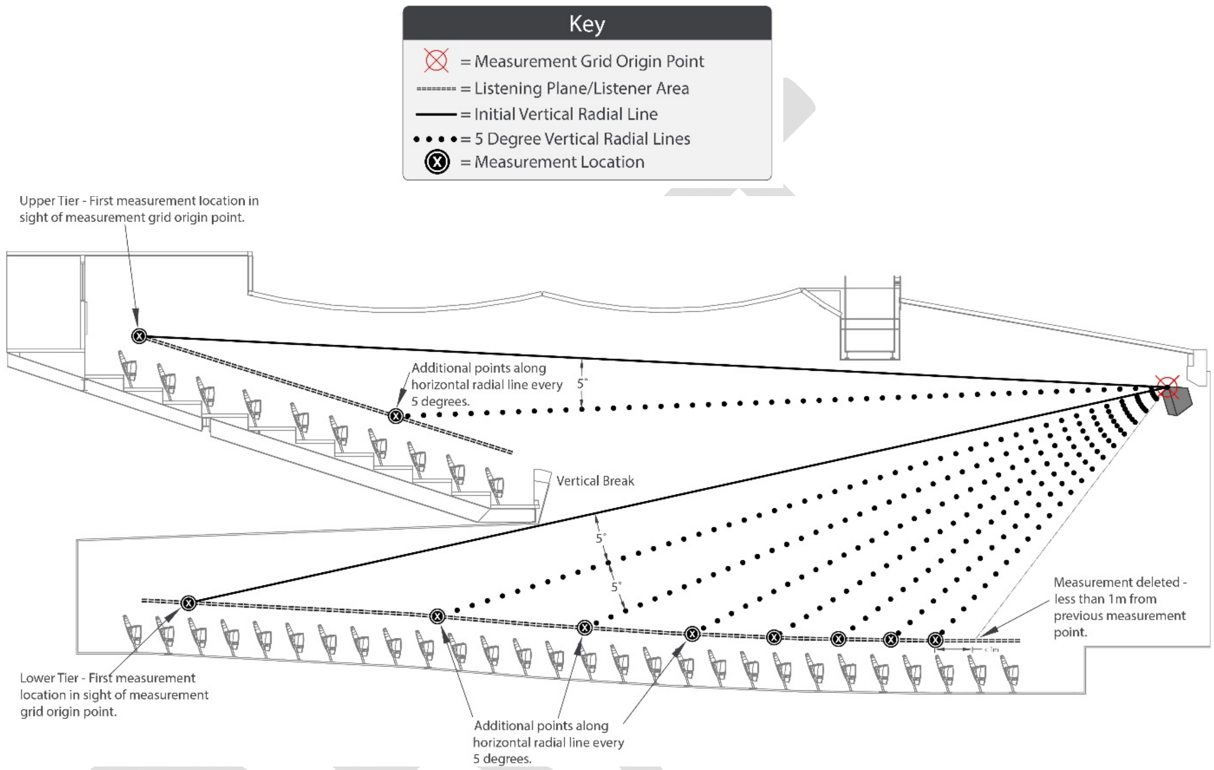


Figure 17 – Establishing measurement locations in a tiered venue (section view)

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.

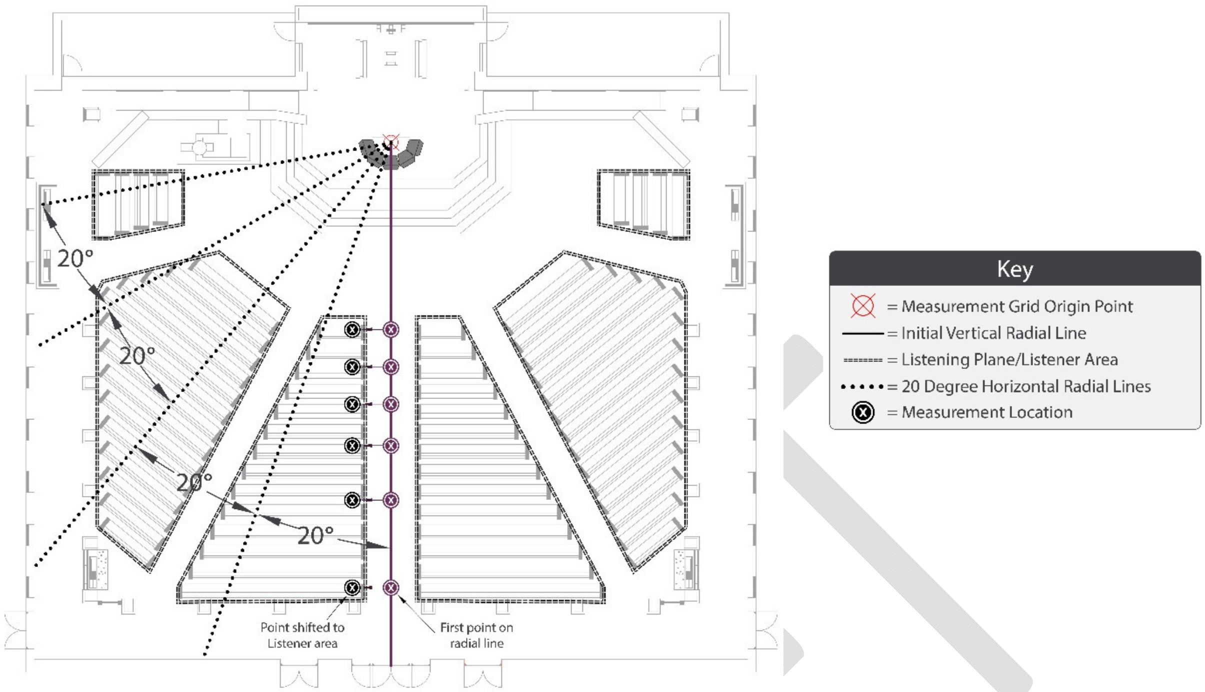


Figure 18 – Measurement locations along the 0-degree horizontal radial shifted to accommodate a center aisle (plan view)

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 one side of the 0-degree horizontal radial line.

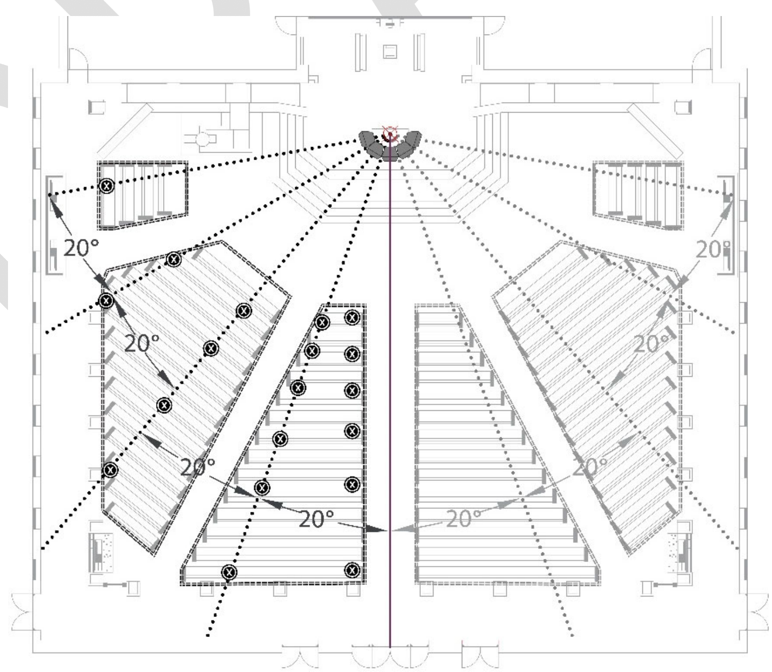


Figure 19 – Symmetrical loudspeaker systems in symmetrical venues (plan view)

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

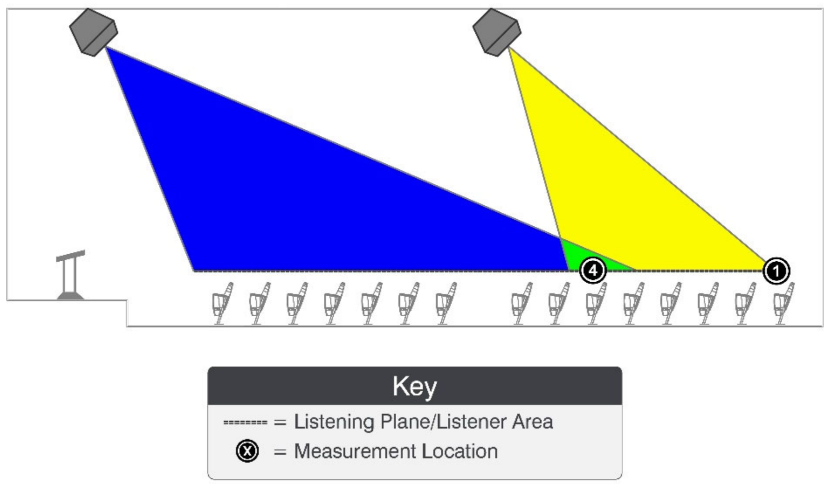


Figure 20 – Point-source/line-source loudspeaker system with fill system (over or under balcony)

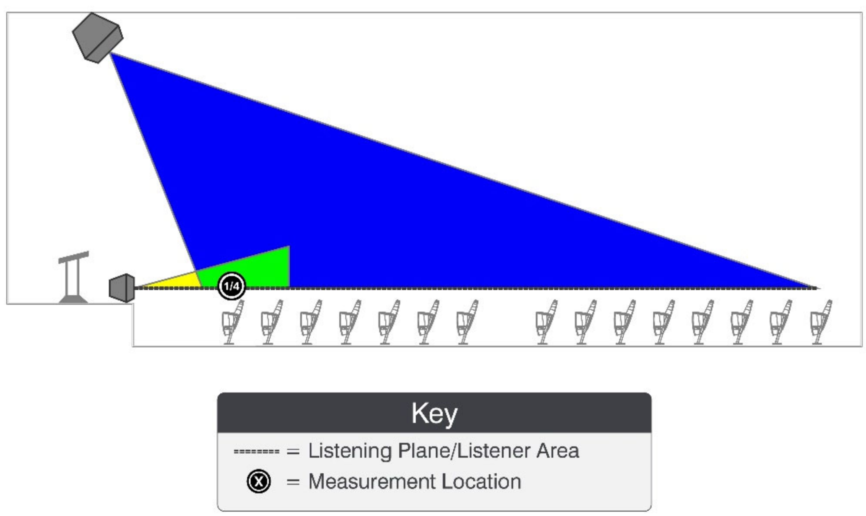


Figure 21 – Point-source/line-source loudspeaker system with front fills

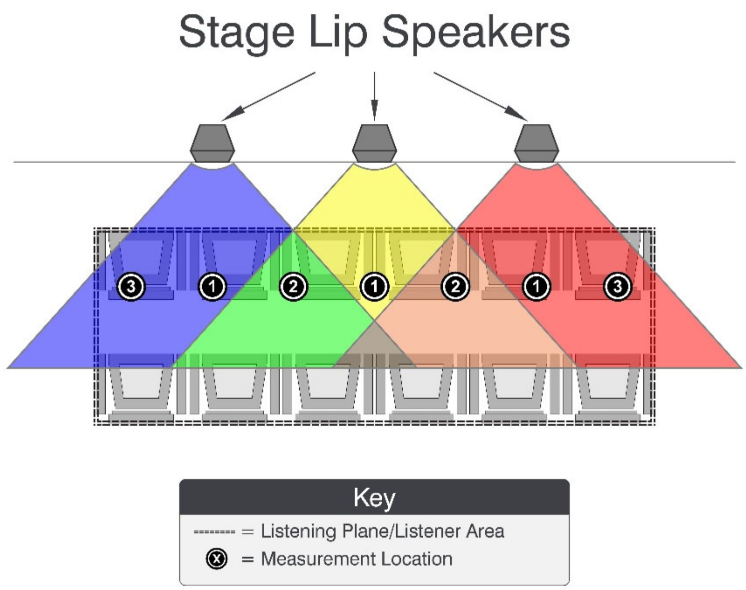


Figure 22 – Stage lip speaker measurement locations

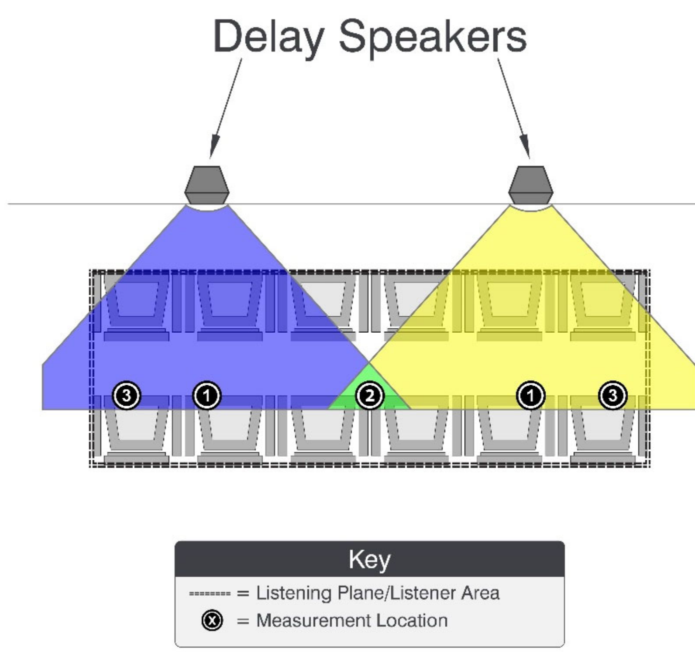


Figure 23 – Delay speaker measurement locations

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.

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 the listener area, 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) Capture a transfer function measurement at each measurement location identified in Section 4.6. Save each 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.

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

j) Within the time domain, apply a 50 ms impulse response window to each measurement. For additional information on impulse response windowing, see Annex F.

k) Within the frequency domain, apply the bandwidth limits as defined by the system purpose to each time windowed measurement.

l) Sum the frequency data points within each measurement data set to a single number, as follows:

1) Convert the values from decibel to linear

2) Sum them

3) Convert this sum into decibels
4) Record the result

m) Determine and record the coverage envelope for each loudspeaker system channel or room configuration.

5 Performance Classification

5.1 System Classification

The system shall be classified according to the table shown below.

Coverage Envelope	System Classification
0-3 dB	3 dB
3-6 dB	6 dB
6-9 dB	9 dB
9-12 dB	12 dB
12+	> 12 dB

The system purpose (full, limited, or paging) shall also be reported.

The following format shall be used to designate the performance classification:

ACU-[System Classification]-[System Purpose]

For example, a system could be described as **ACU-3-Full** or **ACU-6-Paging**.

Annex G contains additional guidance about coverage envelopes and the user experience.

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
b) A plan that designates measurement locations and their unique, assigned numbers.
c) The measured ANL.
d) The value in decibels for each measurement location.
e) The performance classification.
   1) A performance classification shall be generated for each loudspeaker system channel or room configuration.

5.3 Test Report 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:
Other:	

System Purpose (Circle One)		
Paging (175 Hz to 5.6 kHz)	Limited (70 Hz to 11.2 kHz)	Full (70 Hz to 17.7 kHz)

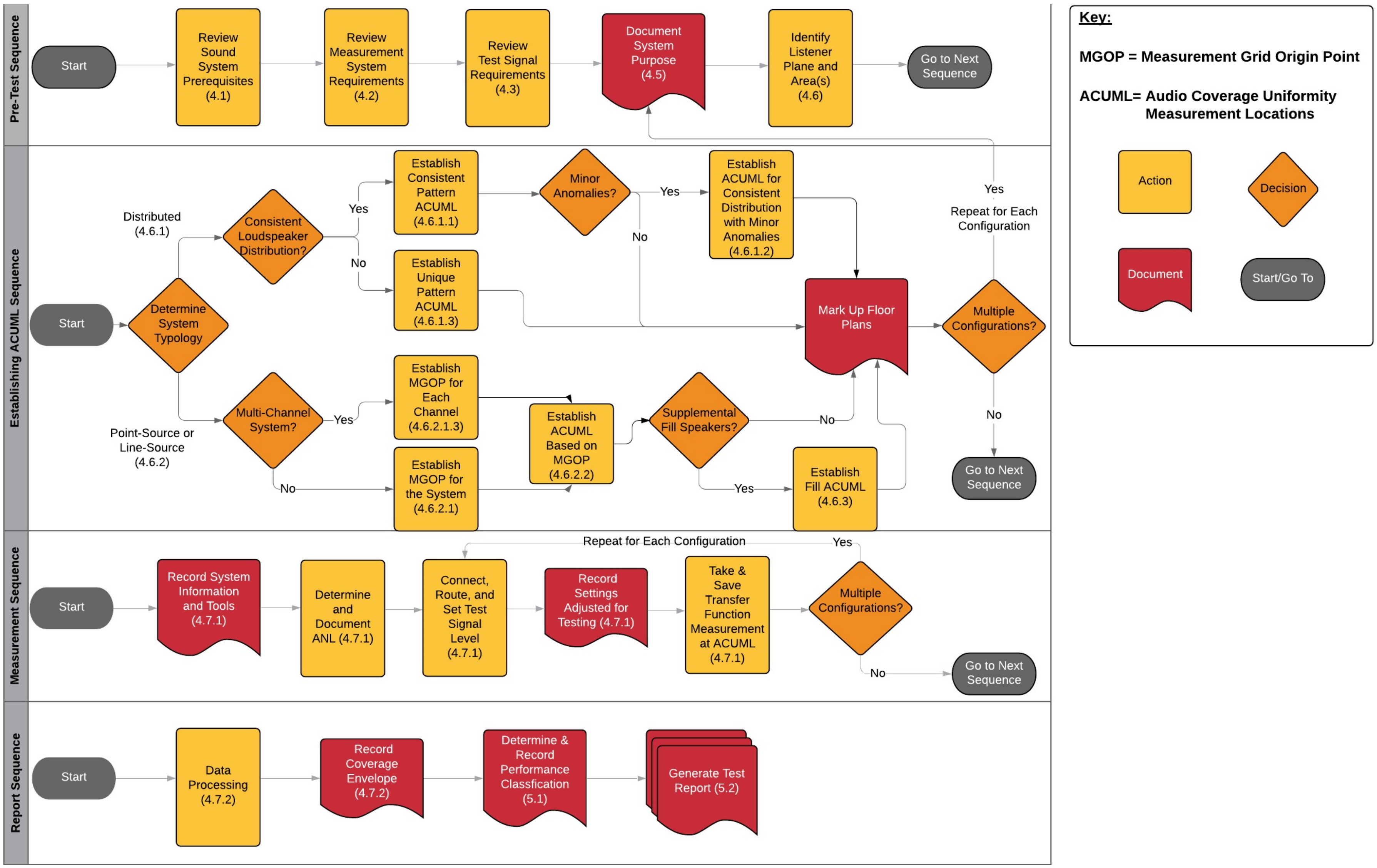
Envelope (Circle One)				
3 dB	6 dB	9 dB	12 dB	>12 dB
Notes & Explanation:				

Measurements	
	Value
Ambient Noise Level (LZ_{eq})	
Ambient Noise Level + 15 dB	
Transfer Function Values	
Measurement Location	**Value**
1	
2	
3	
4	
5	
6	
7	
8	
9	
10	
11	
12	
Minimum Level	
Maximum Level	
Coverage Envelope	

Included with report:

- Plan and/or elevation drawings showing measurement locations.

Annex A: Process Map (Normative Annex)



Annex B: Measurement Microphones (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.

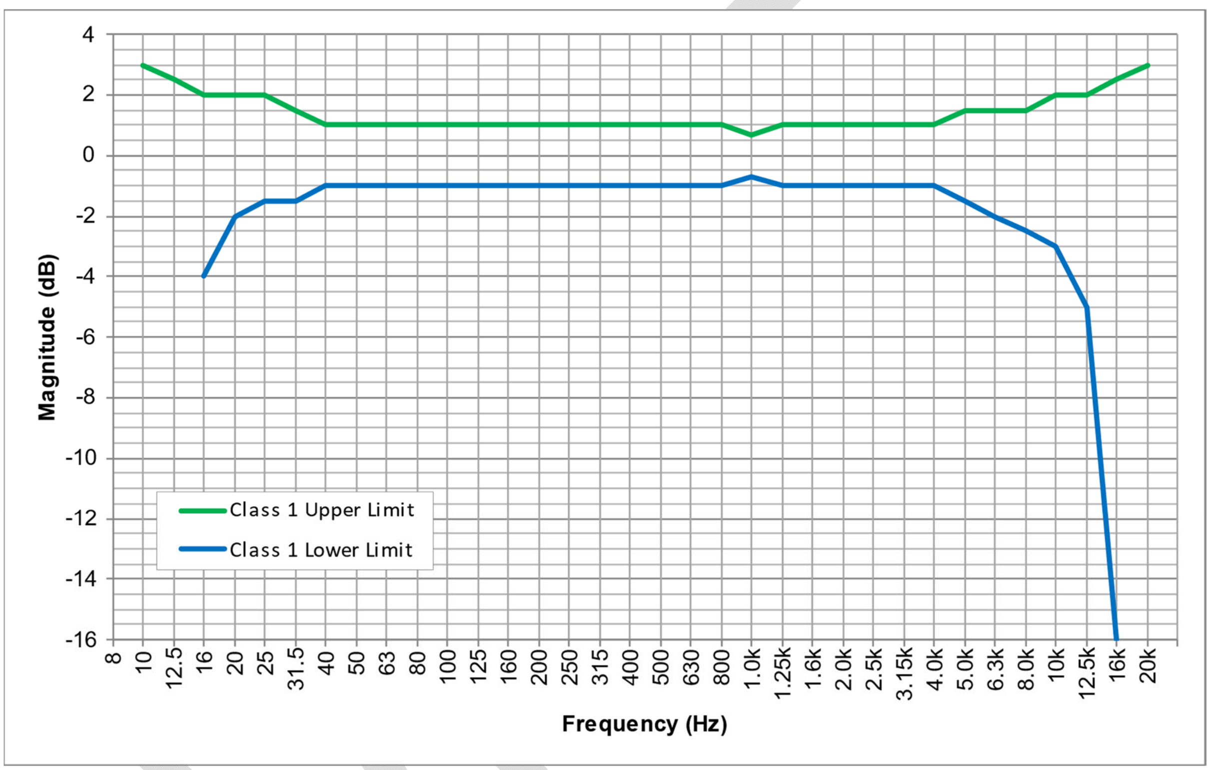


Figure B.1 – Class 1 sound level meter response (per IEC 61672-1:2013)

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 175 Hz (the lower limit of the 250 Hz one octave band) to 5.6 kHz (the upper limit of the 4 kHz one octave band).

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 70 Hz to 11.2 kHz. A limited-bandwidth system will also function for light 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 lowest frequencies due to the challenges presented to measurement techniques by room modes and measurement microphone boundary conditions. Therefore, this Standard evaluates full-bandwidth systems from 70 Hz to 17.7 kHz.

Annex D: Justifications for Measurement Locations (Informative Annex)

D.1 Distributed Loudspeakers

In a condition where the spacing of distributed loudspeakers and the distance from the loudspeakers to the listening plane are consistent, the repeatability of the loudspeaker's layout and the predictability of the loudspeaker's behavior can be leveraged to create the simplified measurement technique utilized in the Standard. The work of Rex Sinclair establishes that:[1]

- 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 contribution from any two loudspeakers will occur at the point 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 pattern that does not overlap with the coverage 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 repeatable 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 3700 to 4600 mm (about 12 to 15 ft). Using a 4300 mm (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 coverage uniformity of a system. During the course of testing a user may discover

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[1] 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 capture a particular spot in the listener area(s). This Standard allows the user to add measurement locations as deemed necessary.

Annex E: Early Arriving Energy and the 50 Millisecond Window (Normative 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.[2] This neuro-physical integration allows listeners to perceive the signals as a single source without seriously affecting intelligibility.

Subsequent research has shown that this window extends out to 50 ms for speech signals.[3] 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.

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 magnitude response of the system in the frequency domain; the inverse Fourier transform of the magnitude response in the frequency domain produces the IR in the time domain. Advancements in technology have made capture of these measurements widely available.

[2] 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.

[3] 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.

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. 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.

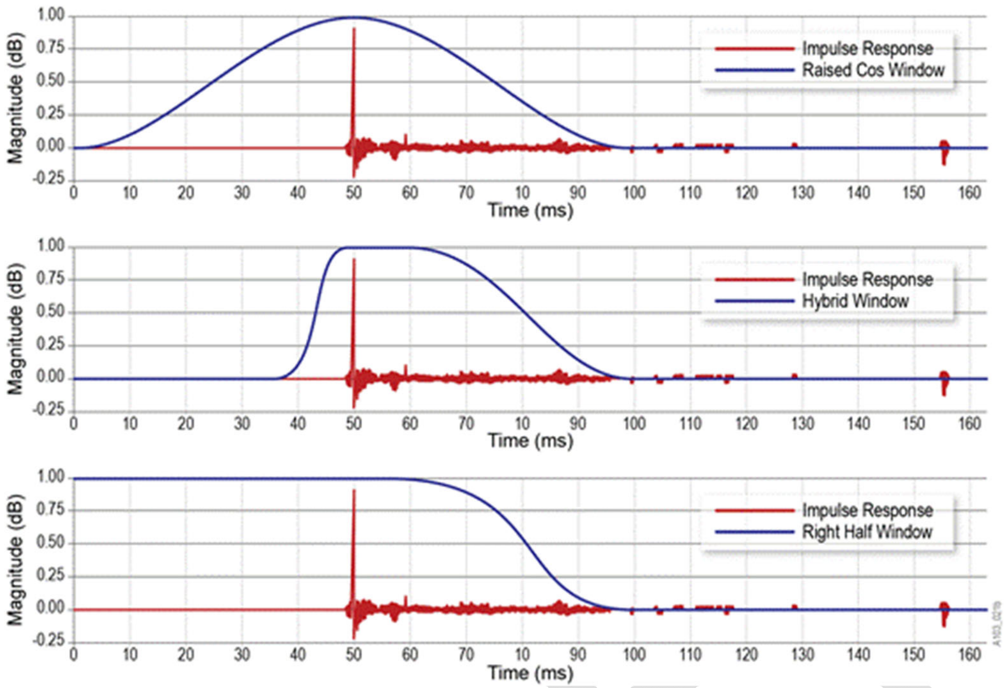

Figure F.1 – Three examples of conforming IR time windows

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 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 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 Time Windowing

When a time-windowed IR is transformed into the frequency domain, the DFT of the time window becomes convolved with the DFT of the original, un-windowed measurement and the practical result is a complex smoothing function in the frequency domain.[4] Because the resulting smoothed transfer function and the time-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.[5] 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.

[4] 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.

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

Annex G: Coverage Envelopes and User Experience (Informative Annex)

Variations in the coverage envelope illustrate how uniform the coverage is across any listener area. This is described in terms of five different coverage envelopes: 3 dB, 6 dB, 9 dB, 12 dB, and >12 dB. Research has shown that the minimum variation in level that the average listener can perceive is a 3 dB envelope, which forms the basis for the coverage envelopes in this Standard.[6]

The following chart relates coverage envelopes to typical user experiences:

Coverage Envelope	Listener Experience
3 dB	• Listeners may find it difficult to find differences in system coverage. • Everyone in the listener area has a similar audio experience.
6 dB	• Listeners may notice differences in coverage if they move around the space. • Most listeners in the area have similar audio experiences, although some locations are louder than others.
9 dB	• Most listeners will notice differences in coverage when they move around the space. Some listeners may find it difficult to hear the content. • It becomes difficult to strike a balance where everyone is satisfied with the level. Owner may receive complaints.
12 dB	• Listeners are likely to notice significant differences in coverage and will need to work to hear the content. • Listeners may feel the need to move to areas of better coverage to listen to content.
> 12 dB	• Listeners will notice extreme differences in coverage and would likely describe coverage as sporadic. • Some locations in the listener area are hot spots, and some may not be covered.

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

The user experience, as defined by the owner, determines the acceptability of the system's uniformity of coverage. 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 coverage envelopes.

6 John Eargle and Chris Foreman, *JBL Audio Engineering for Sound Reinforcement* (Milwaukee, WI: Hal Leonard Corporation, 2002).

Listening Purpose	System Uses	Coverage Envelopes		
		May Be Suitable	Typical	Exceptional
Audio content is critical to the experience.	Concert Halls Conference Spaces Classrooms Houses of Worship	9 dB	6 dB	3 dB
Audio content is secondary to experience.	Sporting Venues Gymnasia Paging Systems	12 dB	9 dB	6 dB
Audio content is one of many contributors to the experience.	Environmental Experience Background Music Systems	>12 dB	12 dB	9 dB
The audio content needs to be shared, but coverage uniformity is a low priority.	Parking Lot Paging Systems Transit Facility Systems	>12 dB	>12 dB	12 dB

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