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AVIXA A104.01:202X
Dynamic Range in Listener Areas
TECHNICAL REVIEW DRAFT

Abstract

Sound systems must be capable of accurately reproducing an audio source to listeners. A key to achieving this goal is the dynamic range of a given system. Systems must be capable of delivering sufficient loudness to the listener at low distortion and with optimum headroom in the signal chain. Proper attention to dynamic range during design and commissioning will also help ensure that systems will provide adequate speech intelligibility throughout the listener area along with accurate reproduction of live or pre-recorded musical sources.

The challenges of maximizing dynamic range have shifted with broader incorporation of digital processing in system(s). This performance standard contains a straightforward method that defines criteria for pass/fail of system(s). It establishes a method that will assess the dynamic range of the acoustic and electronic components of a system. This is accomplished by measuring the noise level with the system on and off and measuring the maximum sound pressure level of the early arriving sound from the loudspeaker system(s) throughout the designated listener area(s) conforming to the design requirement.

Keywords

audio; AVIXA; direct sound; dynamic range; frequency response; full bandwidth; gain structure; impulse response; limited bandwidth; loudspeaker; measurement; microphone; noise floor; sound pressure level; sound system; sound system uniformity; spectral balance; time window; voice communication

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ISBN:

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1 Scope, Purpose, and Application

1.1 Scope

This Standard will:

a) Characterize the required sound pressure level from an audio system, recommend a signal-to-noise ratio (based on the anticipated environmental noise level in a venue) for different venue types, and provide measurement techniques using commonly available test equipment.

b) Define the sound system noise level contribution.

c) Define performance requirements for audio system gain structure for a variety of venues and provide methods for verifying and reporting the levels of the audio system at various stages of the signal path.

d) Determine an appropriate testing procedure for a system to sustain the specified qualities and characteristics over a defined time scale.

e) Define the performance metrics to measure the temporal output characteristics of an amplified audio system that will provide high quality and fidelity—a key factor in producing high intelligibility, acoustics notwithstanding.

The procedure associated with this Standard is one of many verifications of the deployment and performance of a sound system.

1.2 Application

This Standard's procedures apply to sound reinforcement and audiovisual (AV) presentation systems. A variety of applications implement these systems, including conference rooms, training rooms, classrooms, auditoria, theatres, houses of worship and other venues that employ sound reinforcement. Additionally, one may use this Standard to establish design criteria for new systems used for the applications listed above.

1.3 Exceptions

This Standard may be used in conjunction with, but does not supersede, regulatory authority requirements and/or other applicable standards.

This Standard specifically excludes testing or measuring for spectral balance, gain before feedback, coverage uniformity, and other parameters required to assess the total performance of a sound system.

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) ANSI/ASA S1.13:2010, *Measurement of Sound Pressure Levels in Air*.

b) ANSI/AVIXA A102.01:2022, *Measurement and Classification of Audio Coverage Uniformity in Listener Areas* (ACU).

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) AES75 – 2023: *AES standard for Acoustics – Measuring loudspeaker maximum linear sound levels using noise.*

b) ANSI/AVIXA A103.01:2022, *Measurement and Classification of Spectral Balance of Sound Systems for Listener Areas.*

c) ANSI/ASA 3.5-1997 (R2017), *Methods for Calculation of The Speech Intelligibility Index*, Acoustical Society of America.

d) ITU-T G.100.1-2001 (R2015), *The use of the decibel and of relative levels in speechband telecommunications*. International Telecommunication Union.

e) M. van Veen, and R. Schwenke, "Coherence as an Indicator of Distortion for Wide-Band Audio Signals such as M-Noise and Music," Engineering Brief 559, (2019 October). (https://www.aes.org/e-lib/browse.cfm?elib=20582)

f) Architectural Acoustics. Chapter 4, Sound Isolation. M. David Egan, 1988 McGraw Hill

g) Yamaha Sound Reinforcement Handbook, Section 4, “Dynamic Range and Headroom.”

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:

ACU: ANSI/AVIXA A102.01:2022, *Measurement and Classification of Audio Coverage Uniformity in Listener Areas*.

dBFS: Decibels Full Scale

RMS: Root Mean Square

SPL: Sound Pressure Level

3.2 Definitions

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

3.2.1
document
To record information in a way that a person could replicate the procedure and obtain the same results, assuming they can perform this Standard and have access to the test report. For instance, "documenting" the microphone position could mean making a drawing of the room showing the microphone position, or it simply could mean including the seat number in the name of a measurement in your measurement software.

3.2.2
initial linear frequency response level
A low-level reference signal used to establish the linear frequency response of the system which produces an acoustic level at least 3 dB louder than the background noise at the measurement location in the frequency range of the loudspeaker system.

3.2.3
interruption point
The system under test requires an effective partitioning between the input and the output at a convenient, user-accessible connection point; at which point the reference signal may be inserted and/or measured. This connection point will be designated "the interruption point" and appears in the signal flow after the primary gain and processing stages, but before the speaker/power-amplifier gain and processing stages.

See **Annex H** for example diagrams.

3.2.4
linear frequency response
The frequency response when the change in the frequency response between the initial and confirmation levels is less than the tolerance.

3.2.5
linear frequency response confirmation level
A low-level reference signal used to confirm the linear frequency response of the system which is at least 3 dB louder than the initial linear frequency response level.

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
nominal level
The nominal level at the audience is 15dB above the noise floor or 65dBA, whichever is louder. For a microphone input, the nominal level of the anticipated source is a person speaking at 65dBA at 1 meter. For line level sources, the nominal level of the anticipated source is 18dB below maximum.

3.2.8
maximum linear SPL
Slow weighted SPL at the reference microphone position using the reference signal when the acceptable compression criteria has been reached. The standard's default frequency weighting for Speech is A, the default frequency weighting for wide-band signals is C.

3.2.9
primary loudspeaker system
The set of speakers that cover the largest portion of the listener plane and that all come to limit at the same drive level. For example, the primary system may be the single largest speaker, a coupled array of speakers, or a distributed overhead array of speakers (all the same model and ceiling height). All other speakers are to be considered secondary speakers.

3.2.10
reference signal
The continuous, broadband signal injected into the system for testing.

3.2.11
secondary loudspeaker system
A subset of speakers, designed to supplement the primary system, all of which come to limit at the same drive level, and which cover a contiguous segment of the listener plane. For example, a secondary system might include effects speakers, single speaker acting as a side fill, one or more speakers acting as a center fill, the set of stage lip speakers, a subsystem for a secondary listening space, the set of all under balcony speakers, or a delay subsystem to the primary.

3.2.12
total ambient noise level
The sum of the sound system and environmental noise levels.

3.2.13
environmental noise level
The contribution to the total noise floor from sources such as, but not limited to, mechanical systems, lighting systems, external noises, listeners, etc.

3.2.14
sound system noise level
The contribution to the total noise floor from the sound system.

4 Requirements

4.1 Process map

The following sections define and detail the processes that comprise the process map in **Annex A**. The map shows the dynamic range measurement procedure, necessary documentation, and a decision tree for this Standard.

4.2 Sound system prerequisites

The sound system shall meet these conditions for this Standard to apply:

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) In its intended operating state, the system shall be capable of an acoustic output of at least 15 dB above the environmental noise level in each one-octave band to be tested.
c) 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.

4.3 Measurement system prerequisites

The instrumentation used for measurement of this Standard shall meet the following requirements:

a) 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.

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

c) Express sound pressure levels in appropriately weighted decibels (dB). The standard's default frequency weighting for Speech is A, the default frequency weighting for wide-band signals is C. If the default weighting is not used, it shall be noted in the test report.

d) Be capable of a source independent transfer function that calculates magnitude, phase, and coherence.

e) Use transfer function measurements or an equivalent function to capture a 50-millisecond impulse-response-windowed or multi-resolution frequency response.

4.4 Reference signal requirements

The reference signal - the signal injected into the system and used for testing - shall be a continuous, broadband stimulus, such as pink noise or other industry recognized signals like:

- Noise with standard speech spectrum as defined by ANSI/ASA S3.5, *Methods for Calculation of the Speech Intelligibility Index*, or
- Music-noise as defined by AES 75-2023, *AES Standard for Acoustics - Measuring loudspeaker maximum linear sound levels using noise* (https://www.aes.org/publications/standards/search.cfm?docID=116).

The default reference signal for speech is noise with standard speech spectrum as defined by ANSI/ASA 3.5-1997 (R2017), *Methods for Calculation of The Speech Intelligibility Index*.

The default reference signal for wide band systems is ST 2095-1:2015 - SMPTE Standard - *Calibration Reference Wideband Digital Pink Noise Signal*.

If the default signal is not used, it shall be noted in the test report.

Other industry recognized signals include:

- ITU-T Recommendation P.50 (09/99), *Artificial Voices*. Telecommunication Standardization Sector of ITU; and
- IEC 60268-1 Ed. 2,0 b: 1985 *Sound system equipment, Part 1: General. Clause 7, Simulated programme signal.*

4.5 Establish measurement locations within listener areas

The AVIXA ACU standard provides the mathematical means to determine conformance at other locations in the venue.

One microphone location is necessary for the primary system, and one additional microphone location is needed for each secondary system (if any). For the purposes of this conformance verification, the following single test locations shall be used.

4.5.1 Measurement location(s) for distributed systems

For an overhead distributed system, the measurement location shall be directly on axis of a loudspeaker, to correspond with location 1 in ACU (Section 4.6.1).

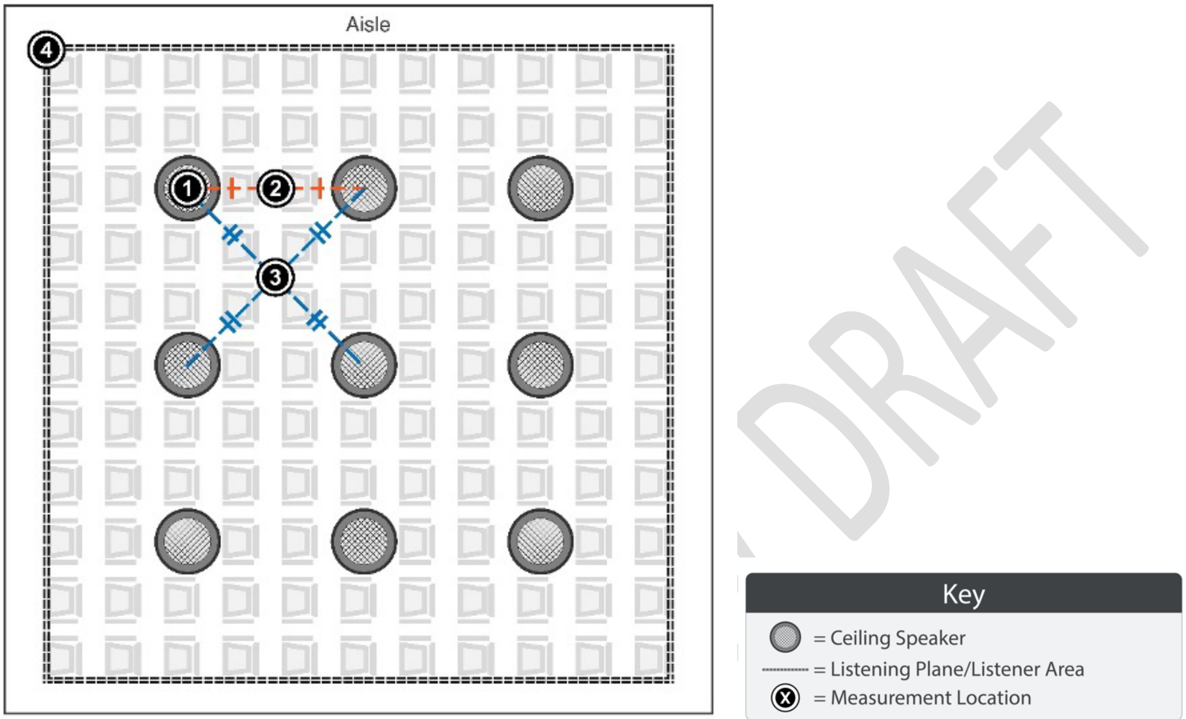


Figure 1 – Distributed loudspeaker measurement locations (plan view)

4.5.2 Measurement location(s) for point-source and line-source systems

The midpoint between the closest and furthest listener positions along the 0-degree horizontal radial line, rounded to the nearest 5-degree increment that aligns with a measurement location within ACU (Section 4.6.2).

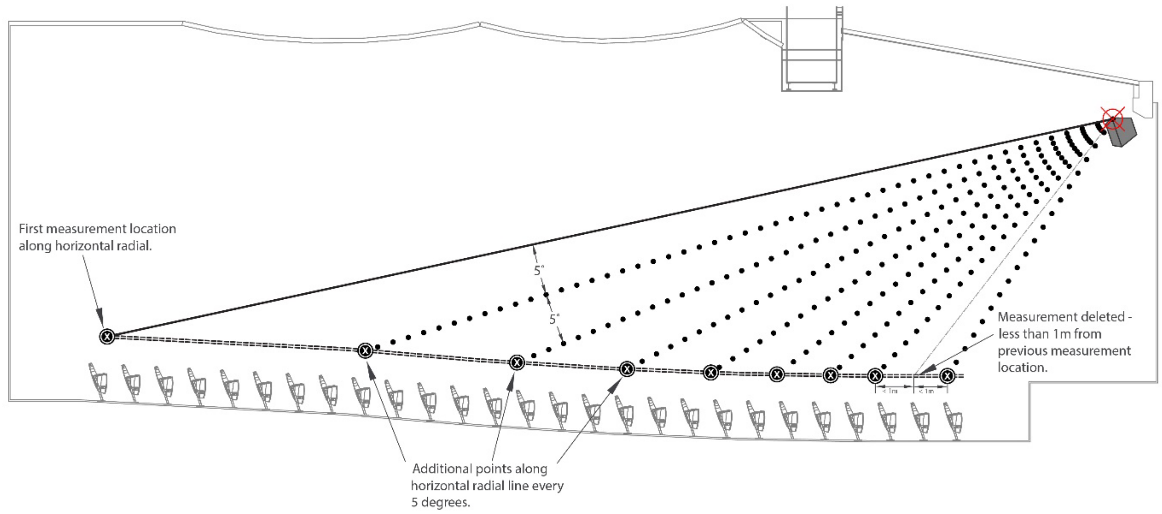
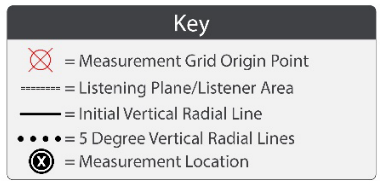


Figure 2 – 5-degree vertical radials along a horizontal radial (section view)



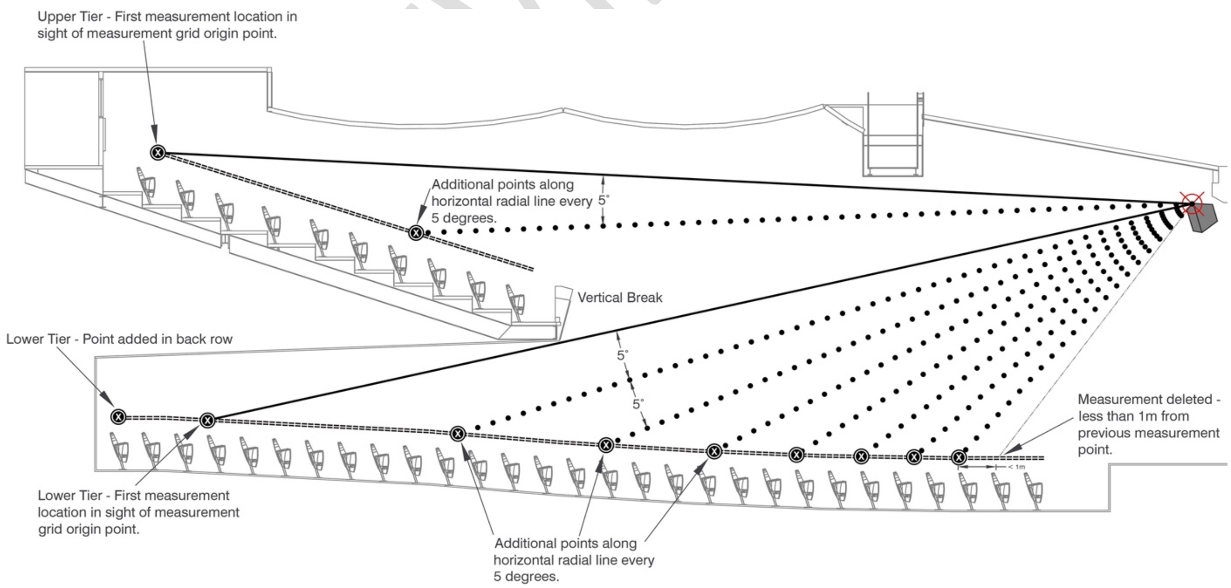


Figure 3 – Establishing measurement locations in a tiered venue (section view)

4.5.3 Measurement location(s) for secondary (fill) system

Once the Max Interruption Point Level of the primary system has been determined, each type of secondary system shall be individually measured to confirm that it can linearly reproduce the Max Interruption Point Level. Each secondary system which uses different speaker models or has a different throw distance shall be measured.

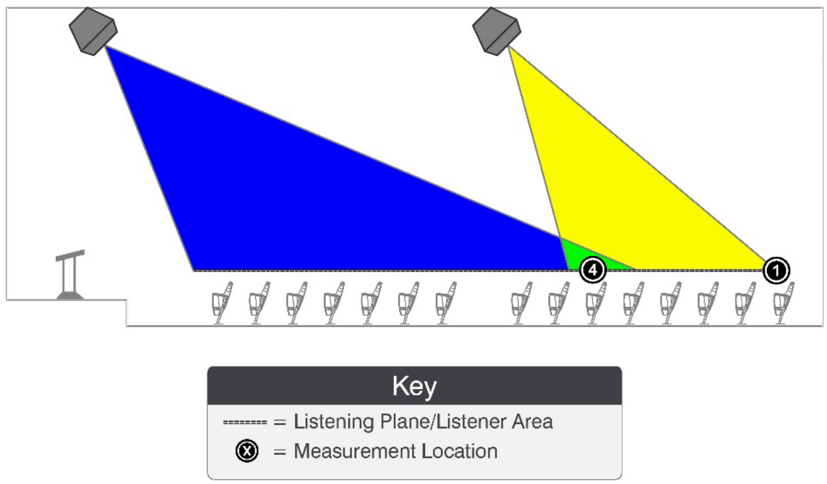


Figure 4 – Point-source/line-source loudspeaker system with fill system (over or under balcony)

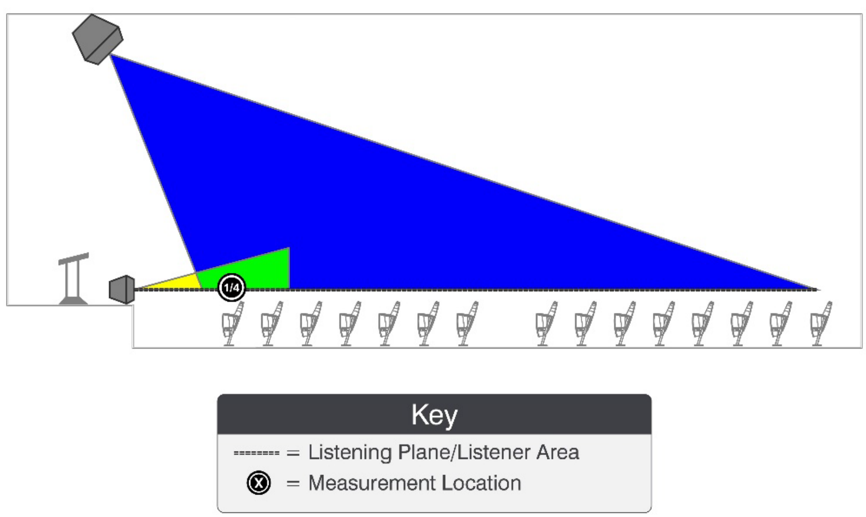


Figure 5 – Point-source/line-source loudspeaker system with stage lip loudspeakers

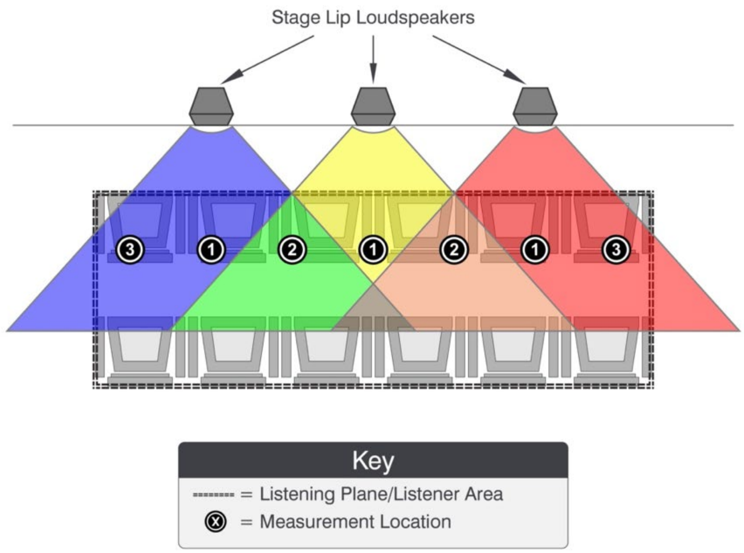


Figure 6 – Stage lip loudspeaker measurement locations

4.5.4 Sound system noise level requirements

If analog inputs are utilized, the inputs shall be passively terminated and the sound system noise level of the idle audio system at the specified listening point shall not add more than 1 dB to the lowest anticipated environmental noise level in the room at any point in the usable audio spectrum.[1]

This measurement can be performed at any time.

4.5.5 Sound system noise level contribution procedure

When determining the sound system noise level, the following procedure shall be followed:

a) Power the sound system off.

b) At the specified measurement location, measure and record the environmental noise level in octave (or narrower) bands.

c) Power the sound system on.

d) If the input is an acoustic transducer (analog microphone), temporarily terminate the input with an equivalent resistance. For other devices set them in an idle or stopped state.

e) Set the system gains such that the anticipated source produces the nominal level in the audience.

f) Measure and record as the total ambient noise level in octave (or narrower) bands.

g) Compare the total ambient noise level to the environmental noise level.

1 If the sound system noise level is 4 dB below the environmental noise floor, then the total ambient noise level will increase 1 dB or less.

Is the total ambient noise level greater than the environmental noise level by 1 dB at any point in the usable audio spectrum?

- If NO, then proceed to **Section 4.7**.
- If YES, then the sound system does not conform to this standard.

4.6 Output test and verification

4.6.1 Target SPL requirements

If the anticipated total ambient noise level is less than 51 dBA Slow, the target SPL shall be at least 66 dBA Slow (ANSI 3.5 "Raised" Speech Level).

If the anticipated total ambient noise level is between 51 and 70 dBA Slow, the target SPL shall be at least the 15dB above the ambient level.

If the anticipated total ambient noise level is greater than 70 dBA Slow, the target level shall be at least 85 dBA Slow.

Environmental noise level	Recommended SPL
<51 dBA slow	66 dBA slow
51-70 dBA slow	15 dB above environmental noise level
>70 dBA slow	At least 85 dBA slow

The sound system designer may specify a target SPL which is greater than the minimum levels given above.

NOTE: If a target SPL of greater than 85 dBA is specified, the designer should tell the venue how long the sound system may be operated at full power before action must be taken to protect hearing.

4.6.2 Max SPL procedure

When testing for max SPL, the following procedure shall be followed:

a) Prepare by documenting the following:
   1) The target SPL,
   2) The frequency range of the system that is assumed for the purposes of this test,
   3) The calibration of test equipment used to measure sound levels,
   4) The measurement microphone position, and
   5) The interruption point.

b) Recall the stored total ambient noise level at the microphone position in octave bands, as measured in 4.6.2.

c) Inject the chosen reference signal into the interruption point.

d) Document the linear response of the system at the measurement microphone position:
   1) Choose an initial level that produces a SPL that is at least 3 dB above the recalled total ambient noise level, as measured in 4.6.2.
   2) Measure the frequency response at the initial level.[2]
   3) Increase the reference signal level by at least 3 dB. This is the confirmation level.

2 John Meyer, “Precision Transfer Function Measurements Using Program Material as the Excitation Signal”, Proceedings of the 111th International AES, p. 354.

4) Measure the frequency response at the confirmation level.

5) Confirm the frequency response at initial and confirmation levels are within +/- 1 dB. This is the linear frequency response of the system. Document this.

e) Test and document whether the system can sustain one minute of reference signal at the maximum linear SPL by increasing the reference signal level until the system reaches any of the following stop criteria:

1) The target SPL

2) A change in the frequency response of greater than 2dB within the frequency range of the system.

3) A coherence lower than 91% (corresponding to 10 dBSNR)

f) Record the reference signal level at the interruption point and at the measurement mic position by measuring the slow weighted SPL using the appropriate frequency weighting.[3] If a weighting other than the default defined in Section 4.3 is used, it shall be noted. At the interruption point this is the maximum interruption point signal level; at the mic position, this is the maximum linear SPL.

4.7 Input Test and Verification

The input test is intended to verify the capability of the designated input signal chain in providing sufficient reference signal level to the output signal chain and achieve the max SPL target value.

a) Prepare by documenting the following:

1) Which input in the system will be used for the test.

2) If a simulated talker or artificial mouth is used, document its type and location.

b) Measure and document the electrical noise floor at the interruption point.

c) Insert a reference signal at the input of the system (using a simulated talker or artificial mouth if necessary).

d) Document the linear response of the system:

1) Choose an initial level that produces a signal level at the interruption point that is at least 3 dB above the electrical noise floor.

2) Measure frequency response at the initial linear frequency response level. [4]

3) Increase the reference signal level by at least 3 dB. This is the confirmation level.

4) Calculate the frequency response in a manner compatible with reference signal.

5) Confirm the frequency response at initial and confirmation levels are within +/- 1 dB. This is the linear frequency response of the system. Document this.

e) Increase the reference signal level until the system reaches any of the following:

1) The maximum interruption point signal level

2) A coherence lower than 91% (corresponding to 10 dBSNR)

f) Record the level at the input and at the interruption point.

3 ANSI/ASA S1.13:2010, Measurement of Sound Pressure Levels in Air.

4 John Meyer, “Precision Transfer Function Measurements Using Program Material as the Excitation Signal”, Proceedings of the 111th International AES, p. 354.

5 Conformance

Dynamic Range is the difference between the noise floor of the system and the loudest sound that can be linearly reproduced. A system conforms with the standard if the system's noise floor minimally increases the ambient noise floor in the room and the system's maximum linear SPL meets or exceeds the target SPL.

5.1 Reporting:

To conform to this Standard, the above procedures shall be followed, and a test report shall be generated which at a minimum contains:

a) Reference Signal Used (if other than the default)

b) Measurement Location(s)

c) Input Tested

d) Environmental Noise Floor

e) Total Ambient Noise Level

f) Frequency Range of the System

g) Target Max SPL (noting whether weighting is other than default)

h) Actual Max SPL of the System (noting whether weighting is other than default)

The Total Ambient Noise Level should be no more than 1dB greater than the Environmental Noise Floor. The Measured Maximum Linear SPL should meet or exceed the Target Max SPL.

Below is an example of a test report that may be used to demonstrate conformance.

5.1.1 Test report example

Venue	**Name:**	
	Location:	
Date:		
Evaluator:		

Measurement equipment and location	
Microphone calibrator:	Calibration date of microphone calibrator:
Computer/measurement device:	Measurement software:
Measurement tools:	Microphone model:
Stimulus signal:	Pre-amplifier:
Measurement location(s):	
Other:	

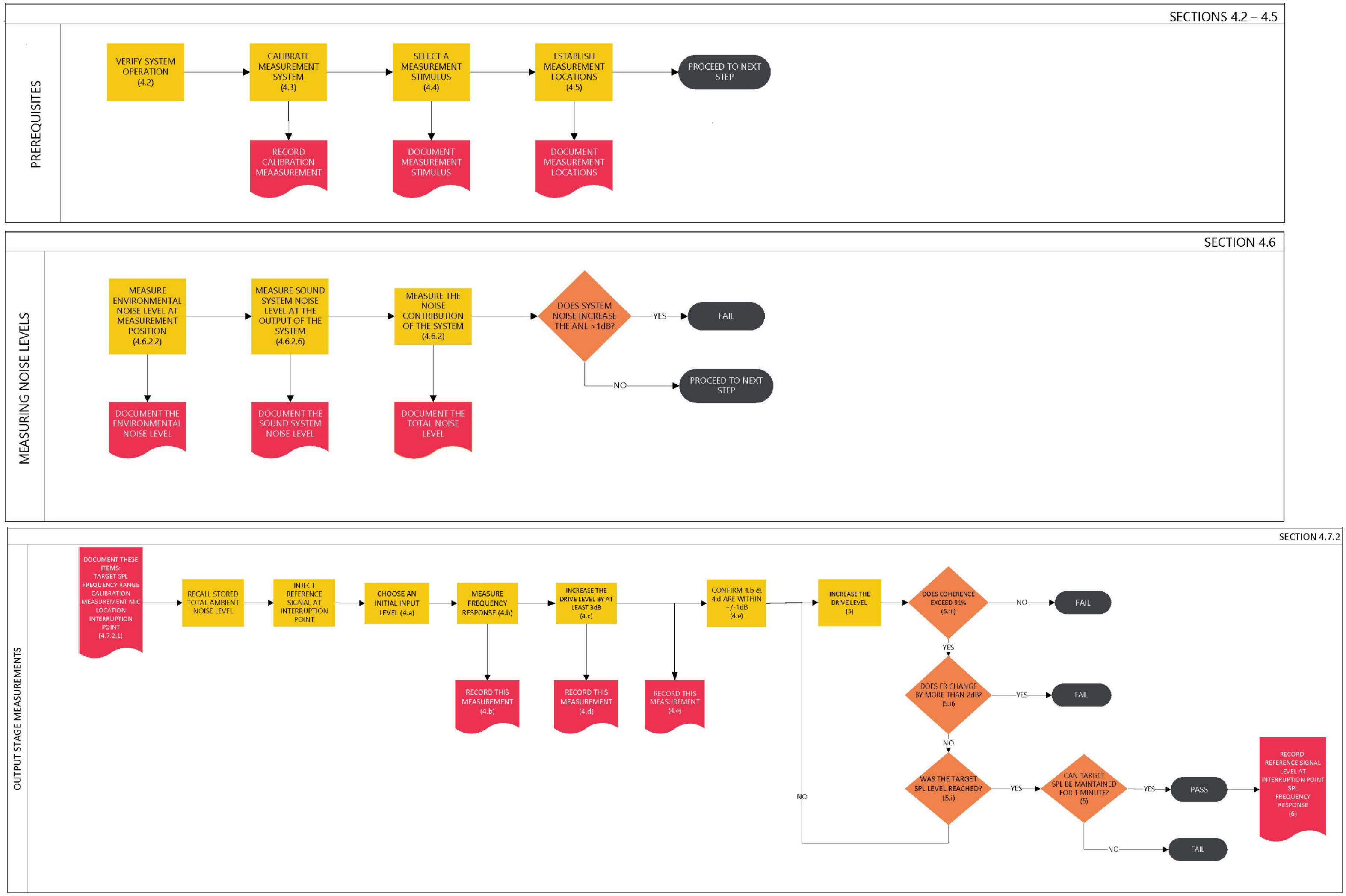
Sound system noise level measurement:	
Environmental noise level:	
Total ambient noise level:	
Is the total ambient noise level greater than the environmental noise level by 1 dB at any point in the usable audio spectrum? If Yes, then this sound system does not conform to this standard.	Yes (Fail)/ No (Pass)

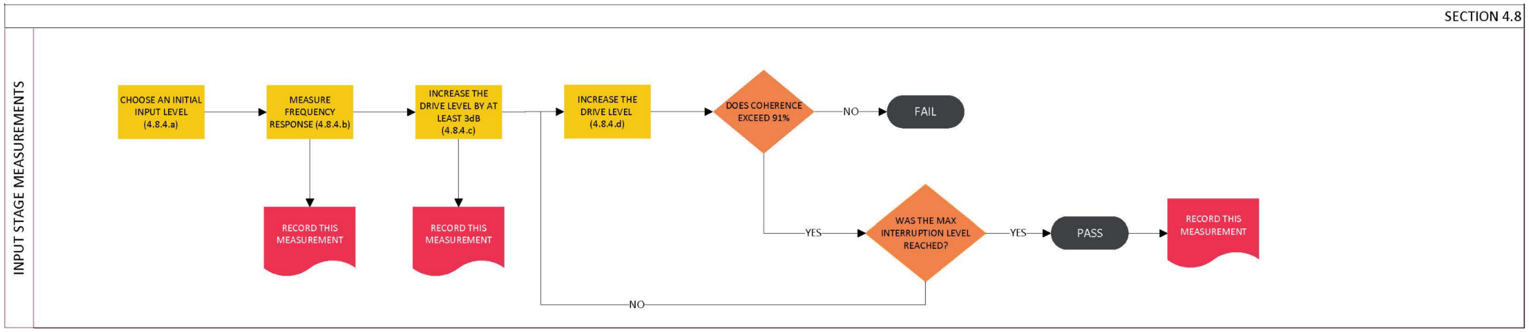
Max SPL measurement:	
Target SPL:	
Frequency range of the system:	
Interruption point:	
Initial linear frequency response level (at least 3 db above Total Ambient Noise Level):	
Initial level frequency response:	
Linear frequency response confirmation level (at least 3 dB above the initial reference signal level):	
Confirmation level frequency response:	
Are the frequency responses at initial and confirmation levels within +/- 1 dB of each other?	Yes / No
Maximum linear SPL sustained for one minute:	
Maximum interruption point signal level:	

Input test and verification:	
Input under test:	
If used, type of simulator talker or mouth:	
If used, location of simulator talker or mouth:	
Electrical noise floor:	
Initial reference signal level (at least 3 dB above the electrical noise floor):	
Initial level frequency response:	
Confirmation reference signal level (at least 3 dB above the initial reference signal level):	
Confirmation level frequency response:	
Are the frequency responses at initial and confirmation levels within +/- 1 dB of each other?	Yes / No
Max level at input:	
Level at the interruption point corresponding to max level at input:	
Is the level at the interruption point corresponding to the maximum input greater than or equal to the level at the interruption point corresponding to max SPL?	Yes / No

Annex A — **Process map**
(informative)

NOTE: The map below is a mock-up graphic for review purposes. A final version will be created for publication.




SECTION 4.8
INPUT STAGE MEASUREMENTS
CHOOSE AN INITIAL INPUT LEVEL (4.8.4.a)
MEASURE FREQUENCY RESPONSE (4.8.4.b)
INCREASE THE DRIVE LEVEL BY AT LEAST 3dB (4.8.4.c)
INCREASE THE DRIVE LEVEL (4.8.4.d)
DOES COHERENCE EXCEED 91%
NO
FAIL
RECORD THIS MEASUREMENT
RECORD THIS MEASUREMENT
YES
WAS THE MAX INTERRUPTION LEVEL REACHED?
YES
PASS
RECORD THIS MEASUREMENT
NO

Annex B — Noise criteria ratings
(informative)

Table B.1 – Noise criteria ratings and AV use cases

Types of spaces	Preferred range of Noise Criteria (NC)	Equivalent dBA
Concert halls Opera houses Broadcasting and recording studios Large auditoria Large houses of worship Recital halls	< NC-20	< 30 dBA
Small auditoria Theaters Large meeting rooms Teleconference rooms AV facilities Large conference rooms Small houses of worship Court rooms Chapels	NC-25 to NC-30	30 to 38 dBA
Small conference rooms Classrooms	NC-30 to NC-35	38 to 42 dBA
Reception areas Retail shops and stores Cafeterias Restaurants Gymnasiums	NC-35 to NC-40	42 to 47 dBA
Lobbies Malls	NC-40 to NC-45	47 to 52 dBA

Annex C — Optimizing dynamic range
(informative)

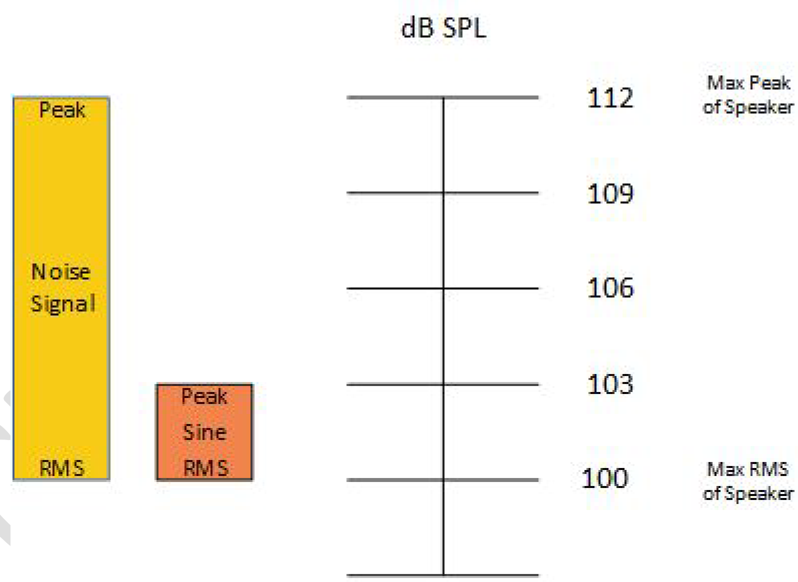
C.1 Optimizing digital dynamic range with loudspeaker dynamic range

Crest Factor is the ratio of the peak level of a signal to its RMS level. Certain signals, like a sine wave or square wave, have a crest factor of 3dB or less. Noise signals typically have a crest factor of about 12dB. Typical content, like speech and music, has recently been shown to have an even higher crest factor.

The maximum peak level that can be represented digitally is the same regardless of the signal. The maximum digital peak level of a sine wave is the same as the maximum digital peak level of speech or music. Consequently, the maximum digital RMS levels of different signals are different. The maximum digital RMS level of a sine or square wave will be significantly higher than the maximum digital RMS level of speech or music.

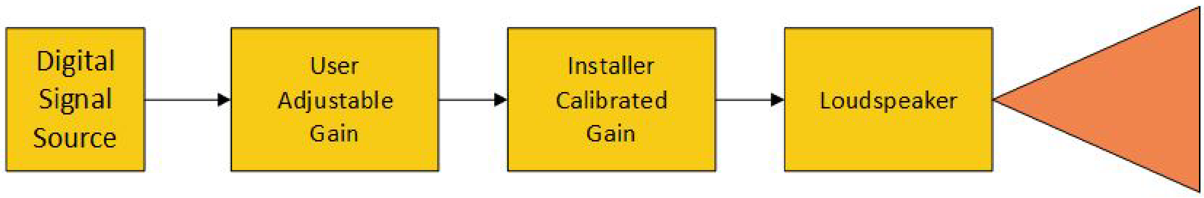
In contrast, the maximum SPL of a loudspeaker is determined in large part by the RMS level of the signal.

The maximum SPL of a loudspeaker is determined in large part by the RMS level of the signal it is asked to reproduce. The maximum RMS SPL a loudspeaker can reproduce using a sine wave test signal is often similar to the maximum RMS SPL a loudspeaker can reproduce a pink noise, speech, or music signal, although those signals will result in much higher peak SPLs due to crest factor of the signal.



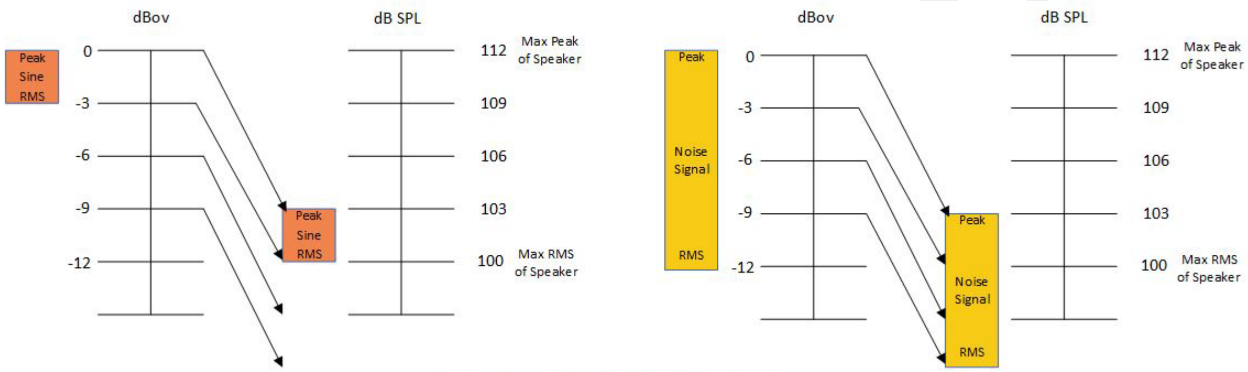
System Calibrated so largest high-crest-factor-signal corresponds to maximum SPL.

Imagine a system that is professionally installed and has several stages of gain, most of which are calibrated during commissioning. The installer leaves access to one stage of gain for the user (usually at the mixing console input faders).



Stages of Gain.

Imagine a loudspeaker with a maximum RMS SPL of 100 dB and a maximum peak SPL of 112 dB. It is possible for the mixing engineer to operate the system in one of two modes. In the first scenario, if a system is set up so that setting the user adjustable gain to maximum results in a sine wave reaching the maximum SPL of the loudspeaker, then the system will never be able to reach max SPL with a signal that has a more typical crest factor found in noise, speech, or music signals.

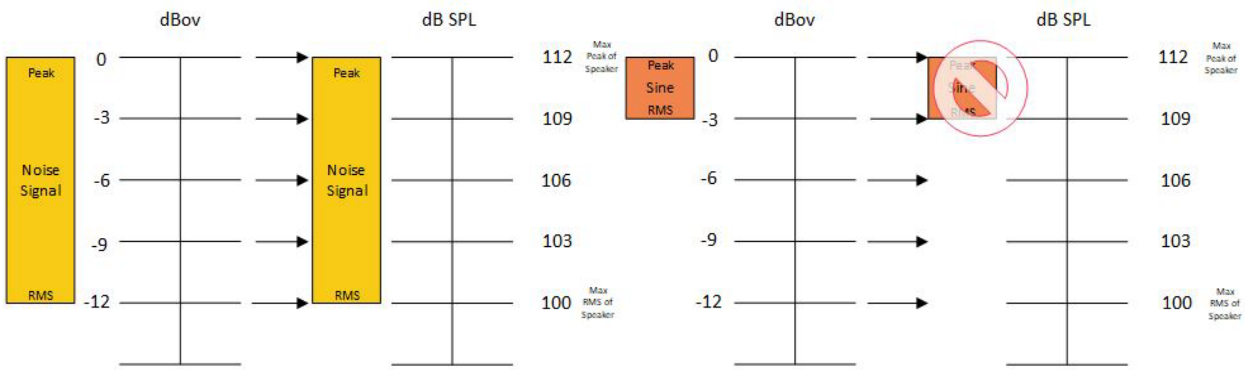


If largest noise corresponds to max SPL for noise, then largest sine wave exceeds maximum SPL.

* dBov is dB relative to overload such that a sine wave has a maximum RMS level of -3.01 dBov as defined in ITU-T G.100.1

In the second scenario, imagine that the system has been configured so that setting the user adjustable gain to maximum setting results in noise, speech, or music that exceeds the capability of the loudspeaker system(s) to reproduce the signal. However, the mixing engineer/operator can adjust the channel control to avoid this issue.

In the second scenario, imagine that the system has been configured so that setting the user adjustable gain to maximum setting results in noise, speech, or music reaching the maximum SPL. In this case, the maximum sine wave that can be digitally represented will be beyond the capability of the loudspeaker to reproduce the signal. However, the mixing engineer/operator always has the ability to turn the level down. In fact, user adjustable levels are often set less than maximum.



If largest noise corresponds to max SPL for noise,
then largest sine wave exceeds maximum SPL.

C.2 Optimizing digital dynamic range with microphone dynamic range

Strictly speaking, the terms “gain” and “level” have distinct meanings:

Level is a property of a signal at a single point in the signal path. Levels are often expressed in decibels compared to a standard reference value (such as, 0 dBV relative to 1 VRMS or 0 dBFS digital full scale or 0 dBu). A signal which is large compared to the reference value has a high level, a signal which is small compared to a reference value has a low level.

Gain is the CHANGE in level between two points in a signal path. Gains can ALSO be expressed in decibels (and this may contribute to some confusion between gain and level). Negative gain is often referred to as attenuation. However, a gain in decibels is NOT calculated in comparison to a standard reference value. A gain in decibels is calculated from the ratio of the output to the input of the gain stage.

Most gains are fixed: they do not change in time in response to the signal level or other characteristics. Gains which are not fixed are usually labeled differently in some way, such as "automatic gain" or "dynamics".

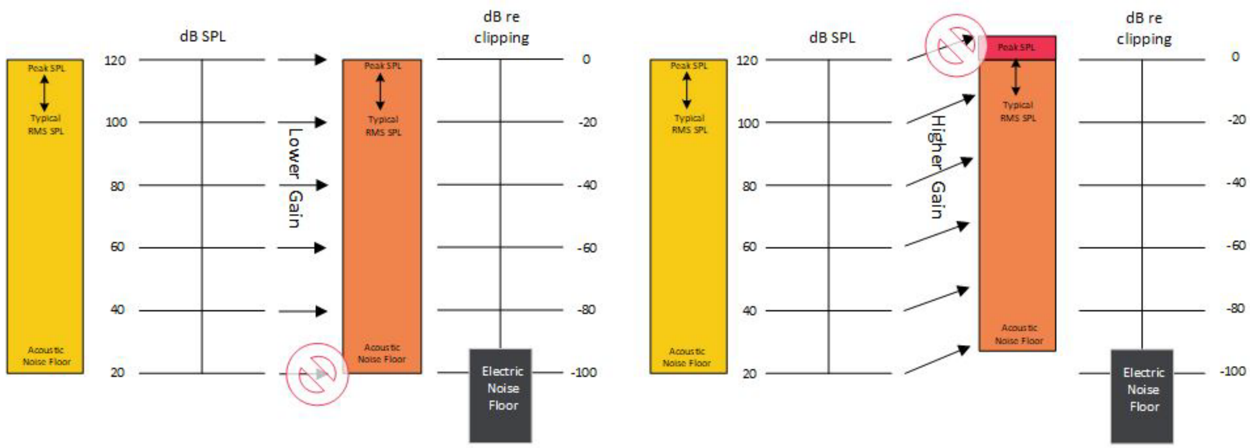
A positive fixed gain (in deciBels) will increase the level of a signal: a signal with small level is changed into a signal with a moderate level, and a signal with a moderate level into a signal with a high level. A negative fixed gain (in deciBels) decreases the level of a signal, a signal with a large level is changed to having a moderate level, a signal with a moderate level is changed to have a low level.

Gain almost always occurs when a signal is transformed from one domain (such as acoustic SPL) to another domain (like voltage or digital level).

Gain applied to a microphone signal determines which peak level in SPL corresponds to electronic clipping and determines whether the environmental noise level can be heard above the electronic noise floor. Raising the gain so that the environmental noise level can be heard above the electronic noise floor unavoidably means that a lower peak SPL corresponds to electronic clipping. Similarly, lowering the gain so that a higher SPL corresponds to electronic clipping can sometimes lower the acoustic noise below the electronic noise floor.

Audio input channels often provide a very large range of manually-selected gain, and this is useful to accommodate the large range of microphones available with different sensitivities, peak SPLs, and noise floors. However, for a specific microphone the range of gain which makes sense is likely much smaller. There is a low gain which scales the acoustic noise floor to be just above the electronic noise floor, and a higher gain which scales the mic peak SPL to electronic clipping. The optimum gain value for a given application is somewhere between these two values, and this range is likely much smaller than the range of gain available.

Gain scales the levels in one domain (like acoustic SPL) to another domain (like voltage or digital level). If the maximum expected peak SPL is scaled to clipping, then the electronic noise floor may be heard above the acoustic noise floor. If the acoustic noise floor is scaled to be greater than the electric noise floor, then a lower peak SPL will result in clipping.



Effect of Gain on noise floor and clipping.

Voltage is equal to the sensitivity (s) of a microphone times the pressure (p) at the microphone in units of pascals.

$$V = sp$$

Pressure is usually stated as a level (L) in decibels relative to the minimum audible pressure (p_0)

$$V = sp_0 10^{L/20}$$

For example, a microphone with a sensitivity of 10 mV/PA and a rated maximum SPL of 140 dB will produce a maximum voltage of:

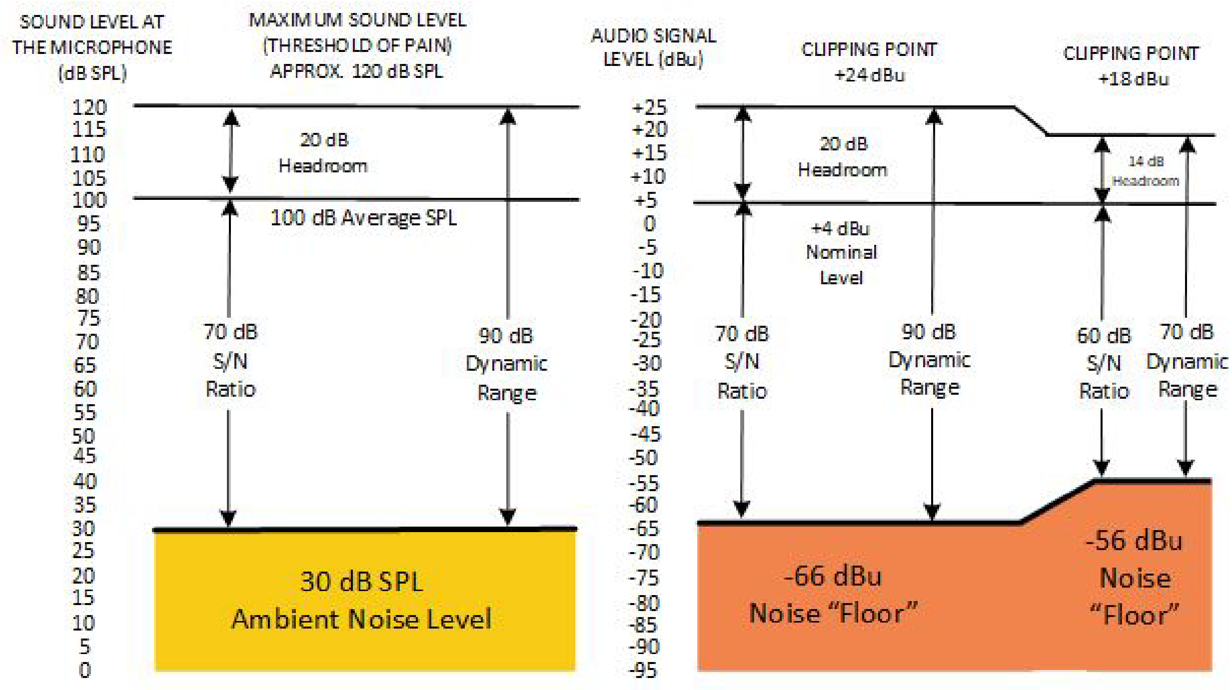
$$(0.010\, Volts/Pascal)(0.00002\, Pascals)10^{\frac{140}{20}} = 2\, Volts$$

The same equation with the same sensitivity is used to determine the electrical noise floor. Suppose the same microphone had a noise floor of 20 dB SPL.

$$(0.010\, Volts/Pascal)(0.00002\, Pascals)10^{\frac{20}{20}} = 2\, microVolts$$

Suppose there is an analog gain stage followed by an analog to digital (A/D) converter which has a max peak of 10 Volts and a noise floor of 200 microVolts. A gain of 14dB will match the max peak of the microphone to the max peak of the A/D. Whereas a gain of 34dB will match the noise floor of the microphone to the noise floor of the A/D. Applying a gain outside of this range will reduce the dynamic range of acoustic sounds which are digitally represented.

Annex D — Dynamic range infographic
(informative)



Dynamic Range and Headroom.
"Yamaha Sound Reinforcement Handbook"
Gary Davis & Ralph Jones

Yamaha Sound Reinforcement Handbook, Page 48, Yamaha Corporation of America. Reprinted with permission from Yamaha Corporation of America.

Annex E— Acoustic source to bit stream translation, end to end in conferencing systems
(informative)

There is a need in any interactive acoustic environment to allow the easy interconnection of a variety of remote sites to a host site without the need to make drastic alterations to the playback levels or transmission levels. An example of an interactive acoustic environment application is a “soft-codec” videoconference using a dedicated audio system consisting of DSP, room microphones, and loudspeakers. Disparate computer-based and dedicated hardware-based conference systems are often expected to seamlessly interoperate without requiring user adjustments.

Similarly, VoIP and PSTN may be transmission media for voice-only conferences and the same need for easy interconnection exists in that environment. The intent of this translation definition is to facilitate the notion of easy interconnection.

Human voice, in a conversational setting, is typically between 65 and 75 dB SPL at 1m distance. (ANSI S3.5)

The intention of a videoconference or teleconference is to allow people located remotely from each other to converse and interact without undue burden. So, conversational speech levels should be encouraged, with the caveat of intent. One should *intend* to be heard as a qualifying factor. In view of that intent to be heard, we shall choose the higher range of conversation speech levels as the source SPL.

The only consistent signal definition available in the signal chain of conferencing, is the bit-stream of the audio transmission path which could be encoded using a variety of algorithms (e.g., G.711a, G.728, G.722.1c, G.719, SILK, OPUS, etc.) and the only viable reference is relative to full-scale digital (i.e., all bits on, or digital saturation) often referred to as 0dBFS.

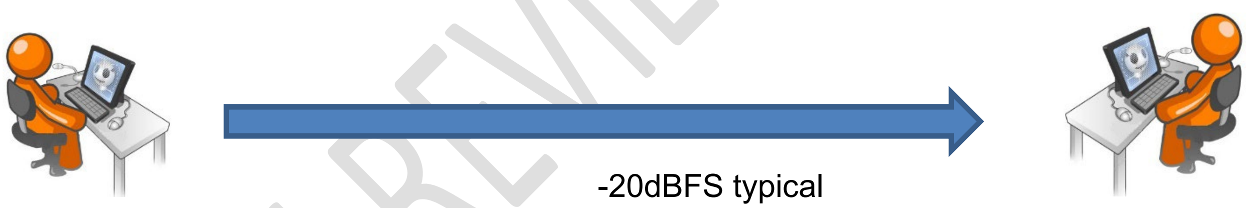


Figure E.1 – Bit stream translation

74 dB SPL long term at the microphone causes 0 dBu analog signal level which encodes at -20 dBFS which decodes to 0 dBu which plays back into the remote room at 74 dB SPL at the listener’s ear.

Annex F — Gain structure remediation
(informative)

This standard does not alter or impact how the system gain structure is initially established. The user should perform this work in accordance with other AVIXA publications as part of the procedure for ensuring the system is in its intended operating condition, prior to performing these tests.

If a system fails Input Test step 5 and is unable to reach the maximum interruption point signal level, then consider:

- increasing the gain in the input device
- increasing the gain somewhere else before the interruption point
- changing any dynamics processing somewhere before the interruption point

If a system fails Step 5 of the Input test procedure as a result of unacceptable coherence reduction then decrease the gain in the input device or at some point in the signal chain before the interruption point.

If a system fails Step 5 of the maximum SPL Procedure because of unacceptable coherence reduction, make sure no clipping is happening between the interruption point and the loudspeaker.

If a system fails Step 5 of the Maximum SPL Procedure by reaching or exceeding the unacceptable compression criteria, then review the settings of any user-adjustable dynamics processing between the interruption point and the loudspeaker.

Annex G — Effect of distortion on coherence
(informative)

Coherence is a statistical measure which indicates contamination of the measurement. As further background on the topic of coherence, the following information is included for contextual purposes.

Distortion (acoustic): Audio lag created by loudspeaker signals.

Distortion (electronic system): Contamination of an audio signal. Some types of electronic system distortion include, but not limited to, total harmonic distortion, intermodulation distortion, unintended deviation from desired frequency response.

To learn more about the relationship between coherence and distortion, please refer to the following publications:

M. van Veen, and R. Schwenke, "Coherence as an Indicator of Distortion for Wide-Band Audio Signals such as M-Noise and Music," Engineering Brief 559, (2019 October). https://www.aes.org/e-lib/browse.cfm?elib=20582

Schwenke, Roger W., Van Veen, Merlijn, "Determining the Source of Coherence Reduction using Playback Level of M-Noise", Institute of Acoustics, Reproduced Sound, November 2020.

Annex H — Interrupted signal path measurement points
(informative)

The figures below represent conceptual and practical test examples of interrupted signal path dynamic range measurement with input and output subsystems.

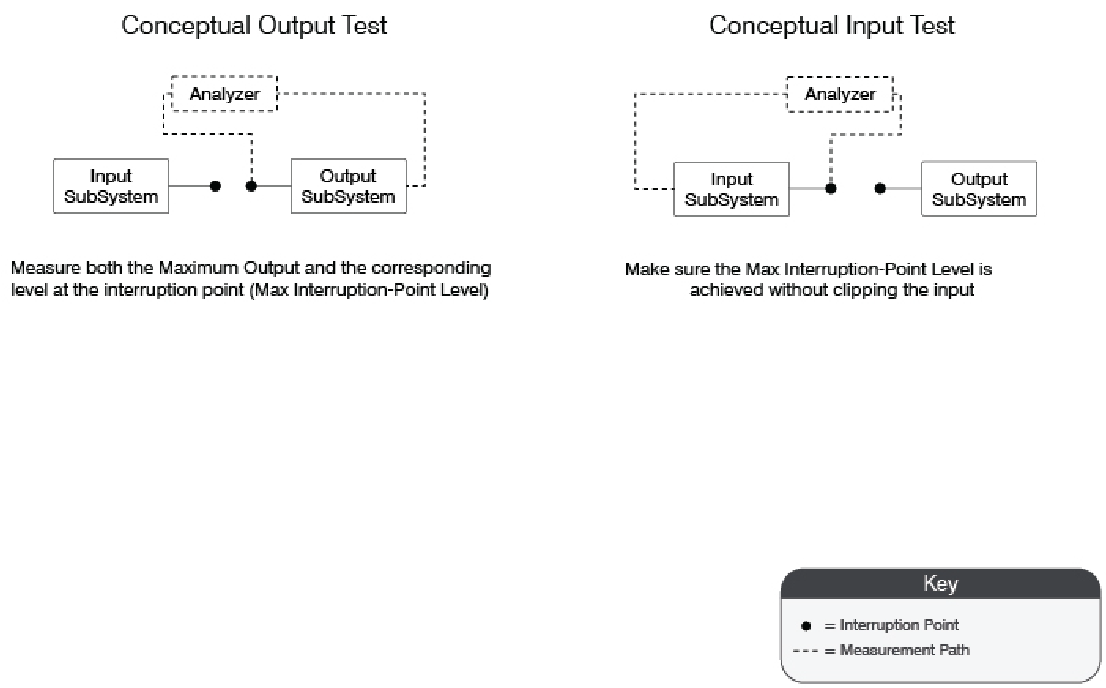


Figure H.1 – Conceptual Test Example

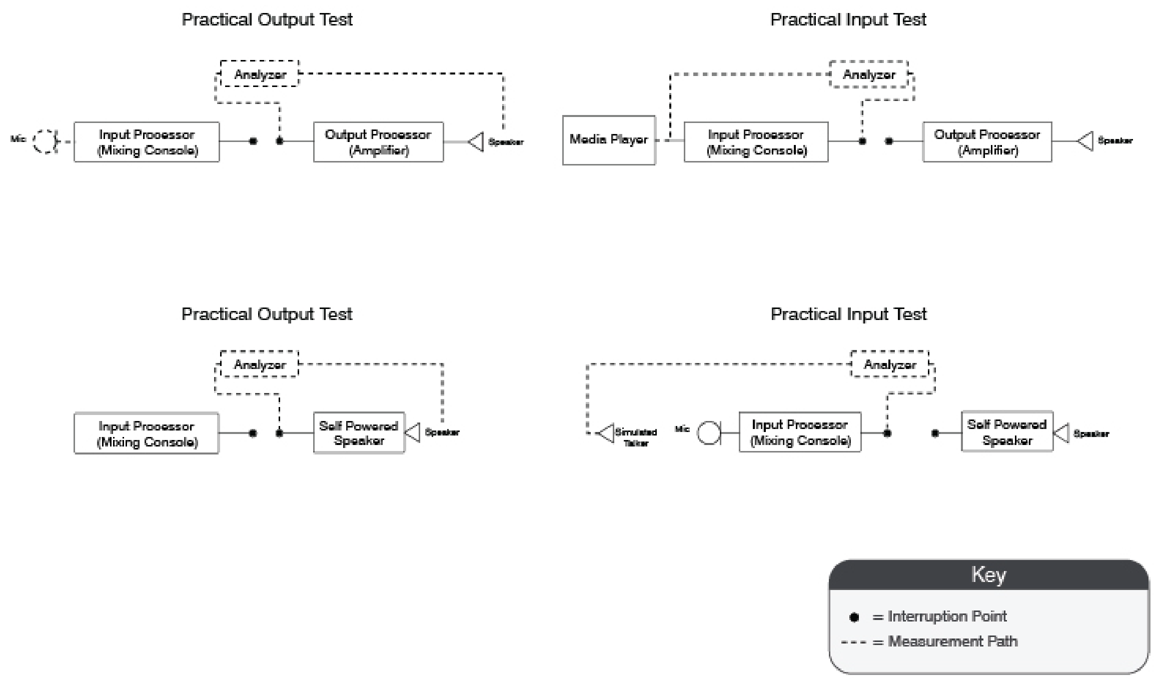


Figure H.2 – Practical test example 1

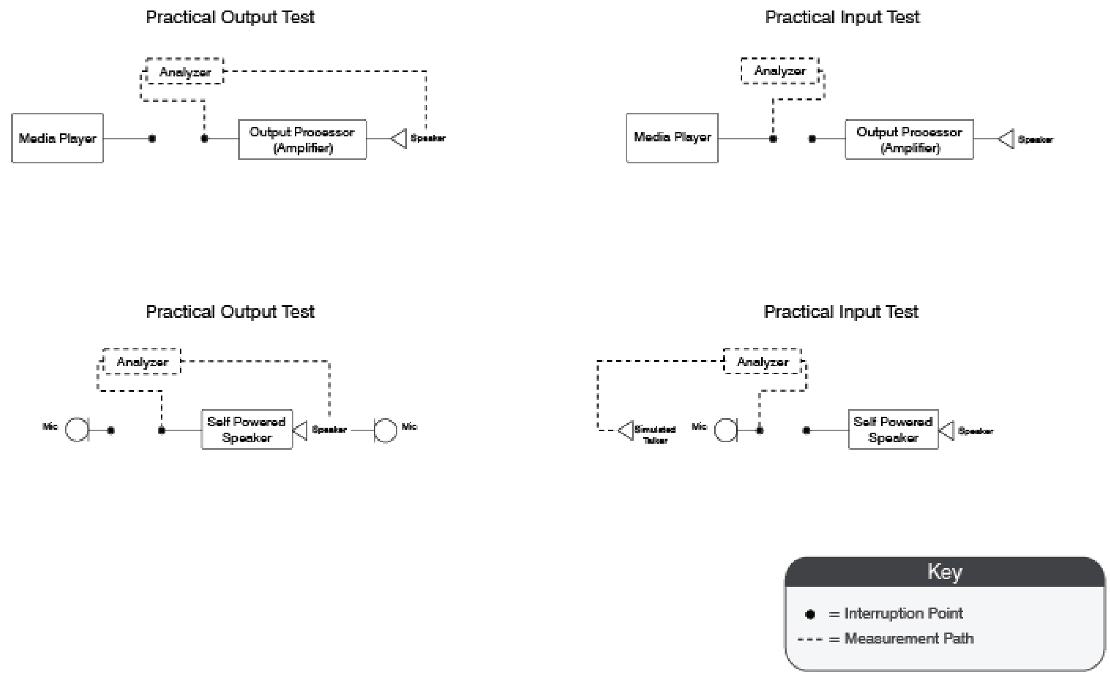


Figure H.3 – Practical test example 2

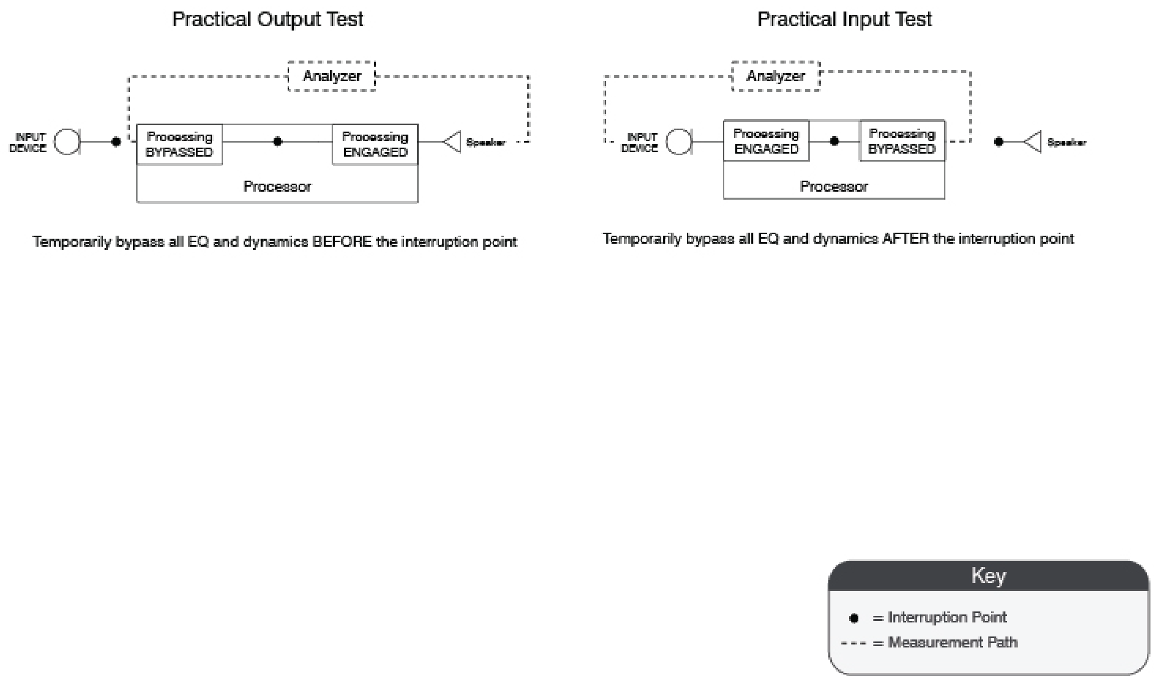


Figure H.4 – Practical test example 3

Annex I — Bibliography (informative)

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