STANDARDS AND INFORMATION DOCUMENTS

Call for comment on DRAFT AES standard for acoustics – Measuring loudspeaker maximum linear sound levels using noise

This document was developed by a writing group of the Audio Engineering Society Standards Committee (AESSC) and has been prepared for comment according to AES policies and procedures. It has been brought to the attention of International Electrotechnical Commission Technical Committee 100. Existing international standards relating to the subject of this document were used and referenced throughout its development.

Address comments by E-mail to standards@aes.org, or by mail to the AESSC Secretariat, Audio Engineering Society, PO Box 731, Lake Oswego OR 97034, USA. **Only comments so addressed will be considered**. E-mail is preferred. **Comments that suggest changes must include proposed wording.** Comments shall be restricted to this document only. Send comments to other documents separately. Recipients of this document are invited to submit, with their comments, notification of any relevant patent rights of which they are aware and to provide supporting documentation.

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DRAFT

AES standard for acoustics – Measuring loudspeaker maximum linear sound levels using noise

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Abstract

This standard details a procedure for measuring maximum linear sound levels of a loudspeaker system or driver using a test signal called M-Noise. In order to measure maximum linear sound levels meaningfully and repeatably, a signal is required whose RMS and peak levels as functions of frequency have been shown to be representative of program material. Various existing standards define noise-based test signals which, like M-Noise, have incorporated the knowledge that typical program material has a diminishing RMS level with increasing frequency, but M-Noise uniquely also features a relatively constant peak level as a function of frequency, so that the crest factor (peak level – RMS level) increases with frequency, which an analysis on a large variety of music and other content has revealed is an important additional characteristic of typical program material. The specified procedure determines a loudspeaker's maximum linear sound levels by incrementally increasing the Playback Level of M-Noise until a stop condition is met: either an unacceptable change in the transfer function's magnitude or an unacceptable change in the coherence of the transfer function.

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Contents

Foreword

This foreword is not part of the AES-X250 *AES standard for acoustics – Measuring loudspeaker maximum linear sound levels using noise*.

This document was developed in project AES-X250, in the SC-04-03-A task group on measurement of maximum linear sound levels using noise, under the leadership of Merlijn van Veen and Roger Schwenke.

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2021-11-02

Note on normative language

In AES standards documents, sentences containing the word “shall” are requirements for compliance with the document. Sentences containing the verb “should” are strong suggestions (recommendations). Sentences giving permission use the verb “may”. Sentences expressing a possibility use the verb “can”.

DRAFT

AES standard for acoustics – Measuring loudspeaker maximum linear sound levels using noise

0 Introduction

0.1 General

This standard specifies a method for measuring the maximum linear sound levels of a loudspeaker system or driver. It uses a mathematically derived test signal called M-Noise that effectively emulates the dynamic characteristics of music. It measures loudspeaker maximum linear sound levels in a repeatable manner which closely represents the values determined in practice with typical program material.

In order to measure maximum linear sound levels meaningfully and repeatably, a signal is required whose RMS and peak levels as functions of frequency have been shown to represent program material. Previous standards have incorporated the idea that typical content has a diminishing RMS level with increasing frequency. In research leading to this standard, a large variety of music has been analyzed, and it has additionally been found that peak levels do not reduce, but rather are relatively constant with frequency. The M-Noise test signal features a relatively constant peak level as a function of frequency, but a diminishing RMS level with increasing frequency.

The maximum sound levels of a loudspeaker are determined by incrementally increasing the Playback Level of M-Noise until a stop condition is met: either an unacceptable change in the transfer function's magnitude, or an unacceptable change in the coherence of the transfer function.

To help clarify the relationship between the terms peak level and RMS level it is useful to consider a period of silence interrupted by a drum strike followed by more silence. The peak sound level of this signal can be measured. Now imagine the same drum being hit with exactly the same strength over and over again at an increasing rate. The peak sound level of this signal is the same as the single drum hit. However, the RMS sound level increases as the rate of the drum hits increases.

Observations like this led to the development of the M-Noise ("Music Noise") test signal used in this standard as a more appropriate signal than the commonly used pink noise signal. Even if a filter, such as the ANSI/CTA-426-B filter, is applied to a pink noise signal to shape its magnitude to more closely match the magnitude of typical music content, the resulting crest factor versus frequency will not match that of typical music like M-Noise will. The RMS magnitude spectra for M-Noise and other signals are shown in figure 0. The magnitude is shown as signal power per 1/n octave-based frequency bands. Pink noise would be a horizontal line at 0 dB.

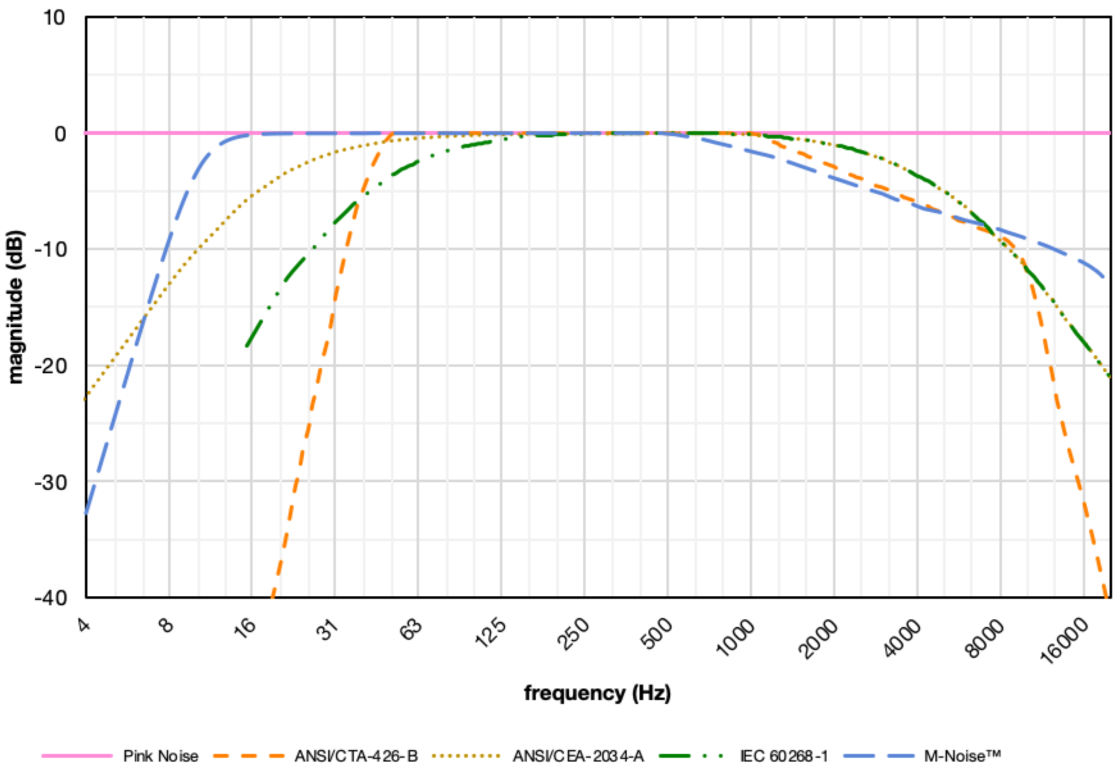

Figure 0 – RMS spectra of various Simulated Programme Content normalized at 500 Hz

0.2 Patents

The AES draws attention to the fact that compliance with this document involves the use of US patent 10,841,717 dated 2020-11-17.

The AES has no position concerning the evidence, validity and scope of this patent right.

The holder of this patent or intellectual property rights has agreed to license them under the End User License Agreement (EULA) in Annex C.

Attention is drawn to the possibility that some of the elements of this document may be the subject of patent rights other than those identified above. The AES shall not be held responsible for identifying any or all such patent rights.

0.3 Documentation conventions

Following ISO convention, decimal points are conventionally shown as commas (,).

1 Scope

This standard specifies a method for measuring the maximum linear sound levels of a loudspeaker driver or system. It uses a mathematically derived test signal that effectively emulates the dynamic characteristics of music as a function of frequency as well as its spectral content.

2 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

IEC 61672-1:2013, *Electroacoustics - Sound level meters - Part 1: Specifications,* International Electrotechnical Commission (IEC), Geneva, Switzerland, http://www.iec.ch

IEC 60942-1:2017, *Electroacoustics - Sound calibrators - Part 1: Specifications,* International Electrotechnical Commission (IEC), Geneva, Switzerland, http://www.iec.ch

3 Terms, definitions, and abbreviations

For the purposes of this document, the following terms, definitions, and abbreviations apply.

3.1 System Frequency Range

The manufacturer's stated operating frequency range of the System Under Test, whether it is a single loudspeaker, a small array, or an entire system.

3.2 Playback Level

A level adjustment to the M-Noise test signal, which is applied before the signal arrives at the reference channel of the frequency analyzer.

3.3 Provisional Linear Frequency Response Level

A relatively low Playback Level used to establish the linear frequency response of the system. It is the Playback Level of the test signal, which produces a coherence of nearly 100% when a microphone is placed close to the loudspeaker, as described by sections 3.6 and 5.3.

3.4 Confirmation Linear Frequency Response Level

A Playback Level that is at least 3 dB louder than the Provisional Linear Frequency Response Level. If the frequency responses at the Provisional and Confirmation Levels match, this is the linear frequency response of the System Under Test.

3.5 Compression Target

The stored linear frequency response, offset by -2 dB.

3.6 Minimum Linear Frequency Response Coherence Criterion

The coherence should correspond to 15 dB SNR ($\gamma^2 \geq 97\%$) or greater within the System Frequency Range when measuring the linear frequency response. See Annex A.1 for more information regarding the conversion of SNR to Magnitude Squared Coherence (γ^2).

3.7 Coherence Reduction Target

A coherence corresponding to 10 dB SNR ($\gamma^2 \leq 91\%$) or less for more than 1/3rd octave within the System Frequency Range. See Annex A.1 for more information regarding the conversion of SNR to Magnitude Squared Coherence (γ^2).

3.8 Maximum Linear Sound Levels

The maximum weighted sound levels at a documented position using the test signal when the Compression Target or Coherence Reduction Target are reached.

L_{ZSmax} is the maximum Slow time-weighted sound level measured over the duration of the test signal using Z-frequency weighting.

L_{Zpeak} is the maximum Peak sound level measured over the duration of the test signal using Z-frequency weighting.

L_{ASmax} is the maximum Slow time-weighted sound level measured over the duration of the test signal using A-frequency weighting.

3.9 "Document"

When used as a verb in this procedure, document means to record information in some way such that an audio professional who has access to the "documents" could replicate the procedure and get the same results. For example, "documenting" the microphone position could mean making a drawing, taking a picture, or writing a description in the notes field of a stored measurement.

3.10 Transfer-Function Microphone

A microphone used for the purpose of detecting changes in the frequency response, and or coherence. It must be close enough to achieve high coherence when the speaker is operating linearly, not necessarily on-axis or in the far field.

3.11 SLM microphone

A microphone placed on-axis in the far field of the SUT used to measure the maximum linear sound levels. The distance from the SUT to the SLM microphone should be sufficient so that the magnitude of the radiation from one area of the SUT is not overly emphasized compared to the other areas.

3.12 System Under Test (SUT)

The device, or collection of devices whose performance is being tested — not including the measurement equipment such as the microphone and analyzer.

3.13 True RMS Voltage Meter

Root Mean Square: The square root of the mean of the squared signal. Different waveforms with the same RMS value will deliver the same amount of power. The RMS value of a sine wave is 1/sqrt(2) times its peak value.

A rectifying averaging meter shall not be used, even if it is calibrated for a particular signal.

3.14 Sound Level Meter (SLM)

The device, or collection of devices, used to determine all published sound levels obtained from this procedure. It shall be Class 1 compliant according to IEC 61672-1 and shall also have a frequency response which varies by no more than ±2dB within the System Frequency Range. It shall be capable of measuring Slow and Peak sound levels, have a resettable maximum hold feature, and have Z-weighting capability. It should also have A-weighting capability.

3.15 M-Noise .wav File

Either of two files containing the M-Noise signal in uncompressed .wav file format. There are 96 kHz and 48 kHz .wav files. The M-Noise test signal is functionally the same in both files (other than the lower high-frequency bandwidth limit of the 48 kHz version). The needed file version should be downloaded for use with this procedure at https://www.aes.org/standards/models/ and should be verified to have the correct checksum:

48 kHz: SHA1 = 1fb296463d5442e34a8ee3f8bf3789b6e1c98e83
48 kHz: MD5 = 84d4b42975fc886ab98e3a8e87da9041
96 kHz: SHA1 = 30686cdfcb7b277e8b8953416597aa21c65cd208
96 kHz: MD5 = 6539f08317d36216c3e0c37cf68c2b38

The contents of the file shall not be altered, such as converting to MP3 or any other compressed format.

3.16 Media Player

A device, or collection of devices, capable of reading one of the M-Noise .wav files and playing its signal continuously in a seamless loop as an unprocessed audio waveform (without any filtering, dynamic processing or other effects). The Media Player should have an adjustable Playback Level control.

3.17 Analyzer

A signal processing device with two input channels, one for a reference signal and one for a measurement signal. The Media Player's output shall be connected to the Analyzer's reference input, and the Transfer-Function microphone shall be connected to the Analyzer's measurement input. The Analyzer shall be capable of calculating the real-time transfer function between the measurement and reference signals, including the coherence, and showing their levels versus frequency as lines on a graph. (See Annex A for more information on the coherence requirements.) The Analyzer shall have a delay compensation adjustment to compensate for acoustic propagation and other forms of delay between the measurement and reference signals. The Analyzer shall also be capable of saving the magnitude of measured transfer functions, displaying saved magnitude lines on the graph at the same time as the real-time transfer function magnitude line is shown, and allow adjustment of saved magnitude lines up or down in 1 dB steps on the graph.

A single device may be used as both Sound Level Meter and Analyzer if it meets the requirements for the SLM stated in 3.14 and also the requirements for the Analyzer stated above. Either the SLM, the Analyzer, or a combined SLM/Analyzer device may also be used as the Media Player if it meets the requirements stated in 3.16.

4 Setup

4.1

Choose either the 96 kHz or the 48 kHz version of the M-Noise .wav file based on the SUT and/or supporting playback equipment. The 96 kHz version is preferred if the Media Player and any downstream devices ahead of the first conversion to the analog domain all support that sampling rate. If any device upstream of the first conversion to the analog domain only supports a 48 kHz sampling rate, the 48 kHz M-Noise .wav file should be used.

4.2

Consult the documentation for the SLM- and Transfer-Function microphone(s) being used to verify that they are capable of handling the maximum Peak sound level expected from the SUT considering their respective measurement location(s). Put an IEC 60942-1 compliant calibrator on the SLM microphone and document that the microphone reads the correct SPL (most calibrators produce 94 dB). Please refer to section A.2 for more information.

4.3

Look up the maximum electrical input and output voltages of all the audio devices used to ensure the loudspeaker can be driven to its maximum linear output, typically at least +18 dBu, and sometimes as much as +24 dBu. If the Media Player's maximum undistorted output level is too low to drive the SUT up to at least its maximum peak linear output, insert a preamplifier which has this ability and adjust it to provide the missing gain. Do not change this gain during the procedure.

4.4

Connect the devices as shown in one of the example diagrams (figure 1). Document the audio devices used and the signal path between them.

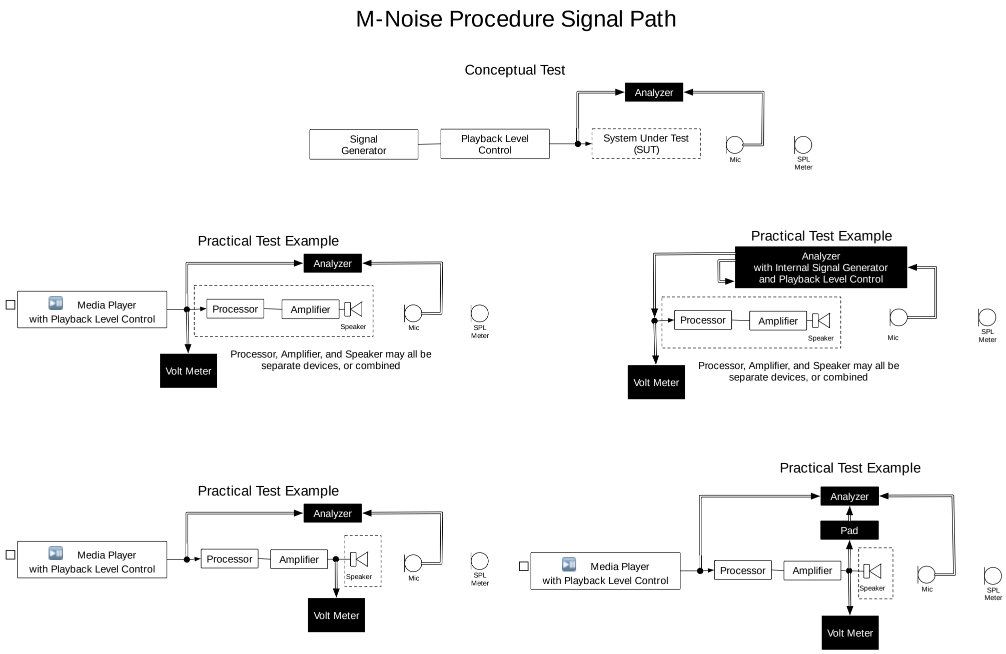

Figure 1 – Procedure Signal Path

4.5

A SUT that allows any user-adjustable signal processing shall have the setting(s) of the signal processing documented. This shall include the name and version number (if applicable) of the processing preset used for the testing.

4.6

It is possible that the M-Noise signal might contain significant energy outside of the low-frequency and/or high-frequency limits of the SUT. For such cases a high-pass filter and/or low-pass filter may be used to band limit the excitation signal presented to the SUT. This allows the excitation signal to be constrained to the operational bandwidth of the SUT. See appendix A.4 for more information on the Signal Bandwidth, RMS level, and other properties of M-Noise.

In the interest of maximizing comparability of the test results, the corner frequencies for external high-pass filters and/or low-pass filters should be set to ISO preferred one-third octave frequencies. Additionally, the slope and alignment of the high-pass and/or low-pass filters should be 24 dB/octave Butterworth.

All filtering that is not part of the actual SUT itself shall be fully documented. The filter details shall include the corner frequency, the slope (e.g., 24 dB/octave), and the alignment (e.g., Butterworth) or the quality factor (i.e., Q) of the filter. If the filtering is other than Butterworth or Linkwitz-Riley high-pass and/or low-pass filters, the manufacturer and model of the DSP used for the filtering shall also be documented.

The maximum linear RMS level of the excitation signal shall be measured at the input of the SUT. If any high-pass and/or low-pass filtering that is not part of the actual SUT has been applied to the M-Noise signal (e.g., Band Limiting), the measurement point for the maximum linear input level shall be after this filtering.

4.8

Document the loudspeaker position and the SLM microphone position.

The distance from the SUT to the SLM microphone should be sufficient so that the magnitude of the radiation from one area of the SUT is not overly emphasized compared to the other areas

4.9

Scale coherence so it takes up as much vertical space on the screen as possible. The number of averages should be set to 8, and overlap should be no more than 50%. Set the smoothing to 1/6 octave or narrower. See Annex B.4 for more details.

5 Procedure

5.1

Load the 96 kHz or 48 kHz M-Noise .wav file chosen in step 4.1 into the Media Player. Play back the M-Noise .wav file at a low level, so the acoustic output is about 20 dB less than the expected maximum sound level.

5.2

In the analyzer, apply delay compensation to synchronize the reference and measurement signals (channels).

5.3

Using the analyzer, confirm that the Linear Frequency Response Coherence Criteria has been met (SNR $\geq$ 15 dB, $\gamma^2 \geq 97\%$). If not, then take actions to increase the coherence, such as moving the Transfer-Function microphone closer, or eliminating sources of background noise. When moving the Transfer-Function microphone closer, find a position whose frequency response is as close as possible to the far field anechoic response. If the Minimum Linear Frequency Response Coherence Criterion (3.6) and the minimum distance from the SLM microphone definition (3.11) are met, it is acceptable to use the same microphone as the SLM microphone and the Transfer-Function microphone.

5.4

Document the Transfer-Function microphone position.

5.5

Set the Playback Level of the test signal to the Provisional Linear Frequency Response level.

5.6

Using the analyzer, store a provisional measurement of the frequency response at the Provisional Linear Frequency Response Level.

5.7

Increase the Playback Level to the Confirmation Linear Frequency Response Level (at least 3 dB louder) and store a confirmation measurement.

5.8

Both provisional and confirmation measurements — throughout the system frequency range — shall:

- have matching magnitude values within ±1 dB, and;
- have coherence (γ^2) values of at least 97% (SNR $\geq$ 15 dB).

The confirmation measurement defines the linear frequency response of the system.

5.9

Offset the confirmation measurement down 2 dB, hereafter referred to as the Compression Target.

Caution: Use hearing protection or take other hearing protective measures — the next step can produce dangerous sound levels.

5.10

Increase the Playback Level in steps slowly (to avoid overshooting the verification criteria in step 5.14) until one of the stop conditions below is reached. Once within 10 dB of the expected max input level, a rate no faster than 1 dB/minute is recommended:

- the live measurement differs from the linear frequency response by at least 2 dB over at least two octaves (matches or goes below the Compression Target, see figure 2), or;
- the live measurement differs from the linear frequency response by at least 3 dB anywhere (figure 3), or;
- the Coherence Reduction Target is met (SNR ≤ 10 dB, $\gamma^2 \leq 91\%$).

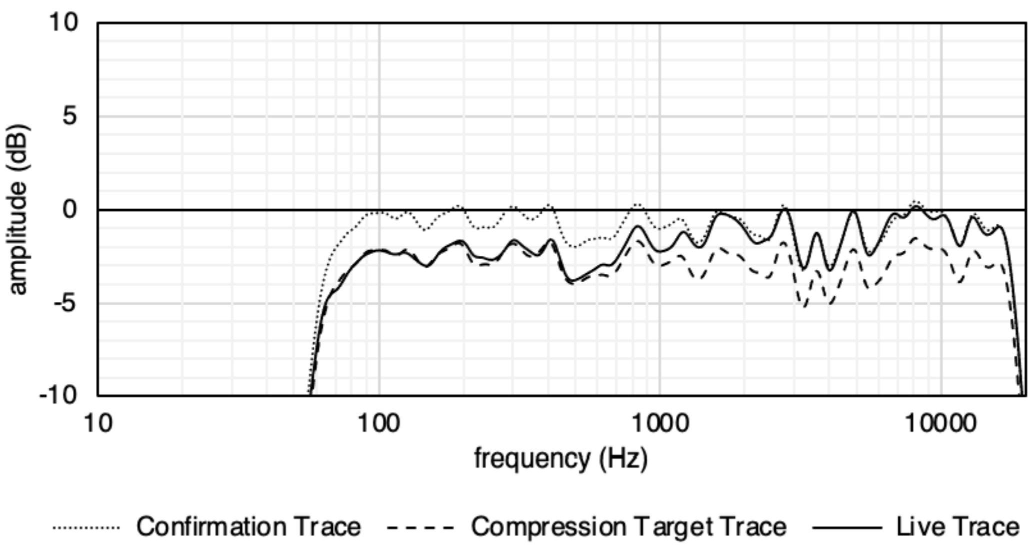

Figure 2 – Two dB compression over at least two octaves

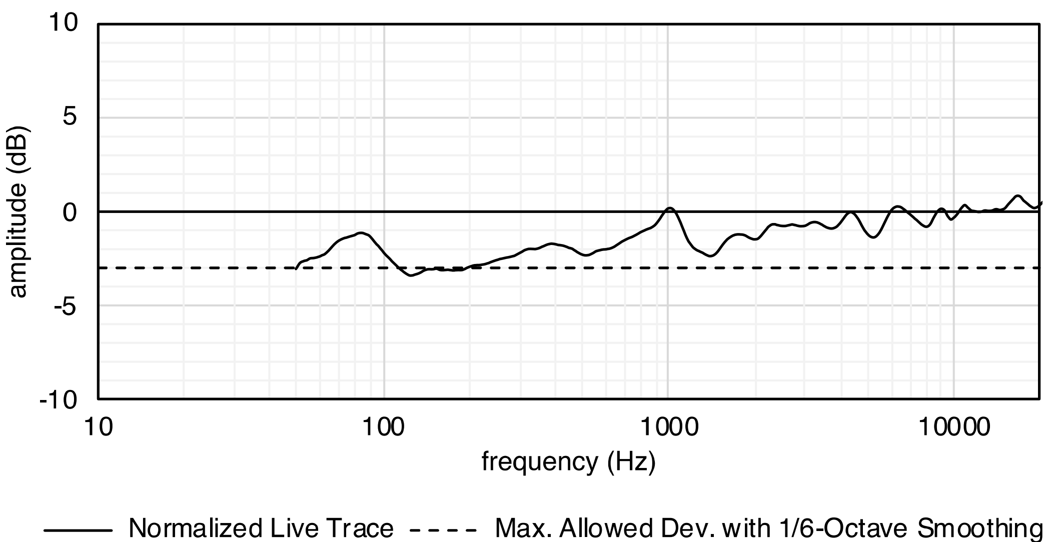

Figure 3 – Three dB compression anywhere (normalized response)

5.11

Store the live measurement for documentation.

5.12

Reset the SLM's hold feature and measure L_{ZSmax} and L_{Zpeak} over the duration of the test signal (53 seconds).

5.13

Continue the test at the maximum linear playback level for at least five minutes. Making a screen recording during the five-minute test is a good way of documenting the transient coherence reductions.

5.14

Using the analyzer, store a measurement of the frequency response five minutes after the stop criteria were met in step 5.10. This measurement shall match the one stored in step 5.11 within ±1 dB. If not, drop the Playback Level down to the Provisional Linear Frequency Response level, wait for the SUT to cool down (e.g., 10 minutes), and return to step 5.5, increasing the Playback Level more slowly in step 5.10.

5.15

Reset the SLM's hold feature and measure L_{ZSmax} and L_{Zpeak} over the duration of the test signal (53 seconds). These levels shall match (±1 dB) those from Step 5.12. If not, drop the Playback Level down to the Provisional Linear Frequency Response level, wait for the SUT to cool down (e.g., 10 minutes), and return to step 5.5, increasing the playback level more slowly in step 5.10. If the levels do match, L_{ZSmax} and L_{Zpeak} shall be documented as the maximum linear sound levels. L_{ASmax} should also be documented as the maximum linear A-weighted sound level. Other frequency-weighted and/or time-weighted values may also be documented.

5.16

An approximate crest factor may be calculated by subtracting L_{ZSmax} from L_{Zpeak} and comparing to expectation. See section A.4 for information about the expected crest factor.

5.17

Once the SUT has reached the threshold for determining its maximum linear sound level, the RMS level at the input of the SUT shall be documented.

For an SUT having an analog input, the RMS level shall be in dBV. For an SUT having a digital input, the RMS level shall be in dBFS.

Note that for a powered loudspeaker system, whether the power amplifier and any associated signal processing (i.e., crossover, equalization, dynamics, etc.) are integral to the loudspeaker chassis or they are located remotely, the input of the SUT shall always be at the input to system and not to any individual subsystem or component of the SUT.

Refer to Figure 1 and the location of the Level Meter for the various SUT types.

6 Reporting

6.1

All publication of results conforming to this standard shall be reported as follows. These results shall be accompanied by the phrase, “AES75 max. linear sound levels”.

6.2

L_{ZSmax} and L_{Zpeak} documented in 5.15 shall be reported as the maximum linear sound levels. L_{ASmax} should also be reported as the maximum linear A-weighted sound level. Other frequency-weighted and/or time-weighted sound levels may be reported.

The RMS input level documented in 5.17 shall also be reported. For an analog input, the equivalent voltage may also be reported.

When the reporting location requires brevity, the reporting should be in the following format which uses example L_{ZSmax}, L_{Zpeak}, L_{ASmax}, and the corresponding input levels:

Example 1: Passive loudspeaker system

AES75 max linear sound levels: 126 dBZ, 144 dBZpk, 120 dBA,
at an RMS input level of 31 dBV (35,5 V)

Example 2: Powered loudspeaker system using digital input

AES75 max linear sound levels: 126 dBZ, 144 dBZpk, 120 dBA,
at an RMS input level of -22 dBFS

6.3

Sound levels shall be referred to 1-meter free field. If the actual SLM microphone position and boundary loading differ—then it shall be reported—for example, 4 meters Ground Plane referred to 1-meter Free Field.

6.4

If the system under test is a loudspeaker driver/receiver component, then that component’s front & rear acoustic loading shall be reported.

6.5

The setting of the user-adjustable processing shall be reported as part of the maximum linear sound levels and/or maximum linear input level specification using M-Noise.

It is not required that the details of any signal processing, that might be considered proprietary, be disclosed. However, the name and version number (if applicable) of the processing preset used for the testing shall be reported. This allows others to perform an identical test in order to duplicate the results.

6.6

All filtering that is not part of the actual SUT itself shall be reported as part of the maximum linear sound levels and/or maximum linear input level specification using M-Noise. The filter details shall include the corner frequency, the slope (e.g., 24 dB/octave), and the alignment (e.g., Butterworth) or the quality factor (i.e., Q) of the filter. If the filtering is other than Butterworth or Linkwitz-Riley high-pass and/or low-pass filters, the manufacturer and model of the DSP used for the filtering shall also be reported.

Annex A (informative): Notes on measurement technique

A.1 Determining Coherence Regime

"Coherence", in relation to dual channel transfer function analyzers, is a colloquialism, used indiscriminately, and typically refers to one of two common definitions: magnitude squared coherence or γ^2, and magnitude coherence or γ (footnote / bibliography).

Use table A.1 to find the correct Minimum Linear Frequency Response Coherence Criterion and Coherence Reduction Target for the analyzer used.

Table A.1 – Conversion between Coherence Definitions

	SNR	γ^2	γ
Minimum Linear Frequency Response Coherence Criterion	15 dB	97%	98%
Coherence Reduction Target	10 dB	91%	95%
	0 dB	50%	71%

Where Noise in SNR is loosely defined as the "lump" sum of all non-coherent output signal power — including distortion — that is not part of the original excitation signal.

It is useful to set coherence blanking to these values when possible.

When the analyzer's coherence regime is unknown, please use the complementary pre-mixed stereo wav files listed in table A.2 to determine the analyzer's coherence regime. These are packaged with the M-Noise™ test signals referenced in section 3.15.

Table A.2 – Coherence Regime Test Tracks

.wav File Name	Left Channel (pre-mix, use as measurement)	Right Channel (use as Reference)
U_15dB_SNR_97pct_MSC_98pct_MC.wav 15 dB SNR, γ^2 = 97%, γ = 98%	correlated pink noise + uncorrelated pink noise at -15 dB re. cor. pink noise	correlated pink noise
V_10dB_SNR_91pct_MSC_95pct_MC.wav 10 dB SNR, γ^2 = 91%, γ = 95%C	correlated pink noise + uncorrelated pink noise at -10 dB re. cor. pink noise	correlated pink noise
W_MSC_or_MC_0dB_SNR_1kHz.wav Variable SNR	pink noise + uncorrelated white noise 0 dB at 1 kHz re. cor. pink noise	correlated pink noise

With sufficient averaging, i.e., averaging set to accumulate or infinite, coherence is expected to converge on the indicated values.

Use the U and V test signals to capture reference coherence measurements for threshold purposes.

Test signal W is expected to produce coherence results in accordance with figure A.1.

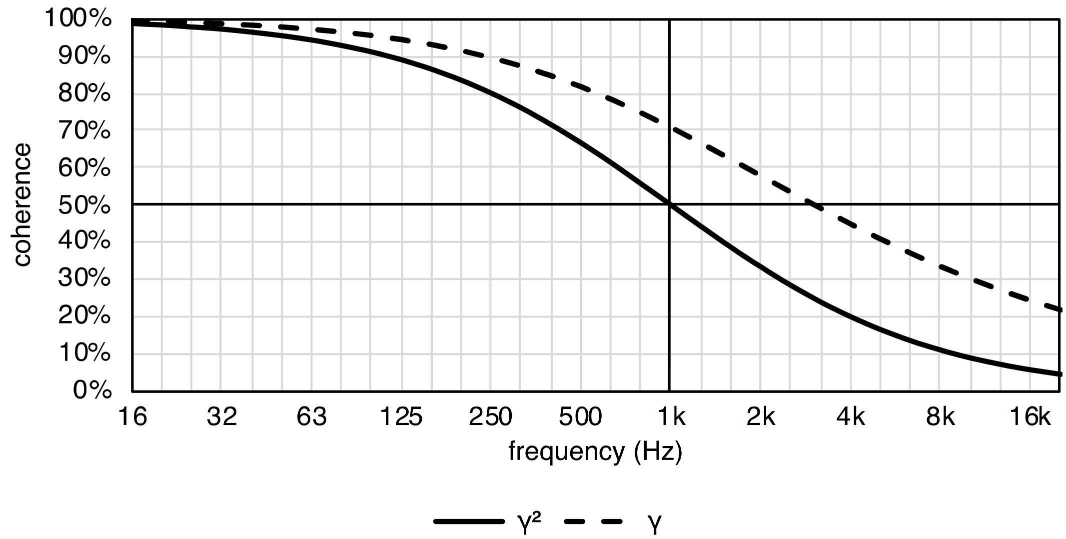

Figure A.1 – Expected Coherence for Track w

A.2 Dynamic Range for Measurement Microphones and Pre-amps

Special attention is needed in the selection process for the used measurement microphones and associated preamplifiers or conditioning amplifiers. The maximum sound levels produced by some SUTs can reach levels that can overload the input side of the measurement equipment or drive it into a region with significant distortion. Most professional microphones are specified to a sound pressure level L_p where the harmonic distortion reaches 3% for a sinusoidal input in the frequency range 160 Hz to 1 kHz (according to the IEC 61094.4 standard, 1995-11). Thus, the upper peak level corresponds to a 3 dB higher level. However, the stimulus in the test described in the present standard has a crest factor of up to 18 dB (i.e., the peak levels are up to 18 dB larger than the RMS level).

Consider the following calculation example:

Maximum input sound pressure level L_p for the microphone:	140 dB re. 20 µPa
Maximum input Peak sound level L_{Zpeak} for the microphone:	140 dB re. 20 µPa + 3 dB = 143 dBZ_{pk}
Maximum Peak sound level for the test stimulus:	143 dBZ_{pk} (like above)
Maximum Z-weighted sound level for the test stimulus:	143 dBZ_{pk} – 18 dB (Crest factor) = 125 dBZ

a) Consult the manufactures specification sheet of the microphone cartridge and verify that the upper limit of its dynamic range exceeds the expected Maximum Linear Sound Levels of the SUT and taking the above considerations into account.

b) Verify that the assembly of the microphone cartridge and preamplifier as a combination will have the needed upper limit of the dynamic range. Often it is the internal power supply voltage of the microphone preamplifier that sets the limit for the system and for battery powered portable setups this may be reduced compared to laboratory setups.

c) Make sure, that the input stage of the real time analyzer hardware can handle the voltage peak levels provided by the microphone and preamplifier.

Consider the following calculation example:

Mic. sensitivity (1/2 inch condenser mic cartridge):	30 mV/Pa	(1 Pa is L_p of 94 dB re. 20 µPa)
Preamplifier gain:	0 dB	
Maximum input sound pressure level L_p:	140 dB re. 20 µPa	

Maximum output voltage from microphone: 10^([140+0-94]/20) × 30 mV = 5,99 V_{RMS}
Maximum output peak voltage from microphone: 5,99 V_{RMS} × √(2) = 8,47 V_{peak}

A.3 Amplifier Headroom

When the SUT is a self-powered loudspeaker or a self-contained system that consists of a loudspeaker, dedicated — external — amplification and factory DSP (for driver protection, e.g., limiting), both amplifier and DSP are considered part of the SUT.

In situations where this is not the case, it is important to establish that the amplifier being used, to power the SUT, is capable of driving it to the stop conditions in step 5.10 without the amplifier itself becoming non-linear and possibly causing the procedure to reach a stop condition prematurely.

Given that the full-bandwidth crest factor of M-Noise is 18 dB, the amplifier's ability to deliver peaks well beyond a particular RMS output is of utmost concern. Like pink noise, peak levels in M-Noise — as a function of frequency — remain virtually constant throughout the entire audible band, and below 500 Hz pink noise and M-Noise are functionally the same; however, M-Noise's RMS level above 500 Hz rolls off resulting in a crest factor that rises with increasing frequency. If the M-Noise signal is high-passed, ahead of the amplifier, due to the operational bandwidth of the SUT, the crest factor can be 20 dB or even greater. In order for the amplifier to remain within its linear operating range, it would need to produce at least 20 dB (factor 10) more peak voltage relative to the SUT's expected RMS voltage handling, calculated from its power and nominal impedance ratings. For example, consider an SUT rated at 200 W and 8-ohm nominal impedance:

$$V_{RMS} = \sqrt{P \times R} \qquad [2]$$

$$V_{RMS} = \sqrt{200\,W \times 8\,\Omega} = 40\,V$$

$$V_{Peak} = CF \times V_{RMS} \qquad [3]$$

$$V_{Peak} = 10 \times 40\,V = 400\,V$$

That SUT would require about 40 V_{RMS} from the amplifier to reach its rated wattage. If the signal's crest factor at the amplifier output was 20 dB, it would require an amplifier capable of 400 V peak output. This is clearly a tall order, even when bridging two large-amplifier channels together to get twice the peak voltage. Fortunately, SUTs that can handle that much power are generally low-frequency devices that operate in the frequency range where M-Noise approximates pink noise whose crest factor — as a function of frequency as well as full range — is closer to 12 dB (factor 4). SUTs that operate at high frequencies, where M-Noise's crest factor is significantly higher, will generally have much lower wattage ratings and corresponding RMS and peak voltages.

There clearly is a burden on the operator to choose an amplifier that remains linear while driving the SUT to the stop conditions in step 5.10. Doing some calculations to estimate the peak voltage required for a given SUT and comparing that to the peak output voltage stated in the amplifier documentation is a valuable step. However, whether the amplifier actually remains within its linear operating range — until the SUT reaches a stop condition — should be verified. The illumination of "clipping" or "peak" indicators on the amplifier, usually means it has exceeded its linear range, but such indicators may not respond to short term signal peaks until the signal level is already several dB past the amplifier's linear range. Additionally, some amplifiers employ "clip protection" which can result in the front panel clip indicator remaining dim while still observing momentary reductions in coherence rather than more persistent coherence drops associated with actual clipped waveforms. To that end, it is recommended that an auxiliary transfer function — including coherence — of the loaded amplifier is monitored during the procedure to rule out the possibility of it contributing to changes in the SUT's magnitude or coherence.

PAD and Isolating Transformer

Given that the amplifier's peak voltage output could overpower the input of the analyzer, that calculates the auxiliary transfer function, an attenuator is very likely required to prevent analyzer input overload or damage.

In order to avoid accidental grounding of either the amplifier output or the analyzer input and any possible large common-mode voltage transfer from the amplifier to the analyzer input, it is recommended that a combination of a resistive pad and an isolating transformer is used between the amplifier and analyzer where the transformer is located between the resistive pad and the analyzer input. An "H" pad — using non-inductive power resistors — is recommended. Off-the-shelf solutions are often available on the market or one can be assembled out of readily available parts. Refer to the lower right Practical Test Example diagram in figure 1 for the location of the pad in the block diagram for this purpose.

There are two considerations when determining the resistive pad's required attenuation.

1. The pad should keep the signal, reaching the transformer, well below the transformer's saturation level or the transformer could contribute a non-linear response, that falsely reduces the magnitude or coherence, even though the amplifier is still within its linear operating range.
2. The combination of the resistive pad in conjunction with any gain loss from the transformer's turns-ratio should prevent the analyzer input from overloading.

A.4 M-Noise Properties

Some properties of M-Noise were specifically chosen to conform with SMPTE 2095-1.

Table A.3 – Comparison M-Noise and SMPTE 2095-1 Pink Noise

Parameter	M-Noise	SMPTE 2095-1
Signal Bandwidth	10 Hz – 22,4 kHz	
RMS Level, Full Spectrum	-18,5 dB FS	
RMS Level, 22,4 Hz – 22,4 kHz	-19,2 dB FS	-19,0 dB FS
Target RMS Level, any 1/3-octave from 20 Hz to 16 kHz	Varies with frequency	-33,74 dB FS
Crest Factor	Approx. 18 dB	Approx. 12 dB
Energy Uniformity Target	Typical Music	Pink 20 Hz – 16 kHz
Minimum Unique Signal Period	53 seconds	10 seconds

M-Noise has a crest factor of at least 17,5 dB in every 10-second interval and 18,0 dB over its entire length. M-Noise has a crest factor which increases as a function of frequency, and therefore the resulting crest factor at the SLM microphone will depend on the frequency range of the SUT. An SUT which low passes a significant range of audible frequencies will likely produce a crest factor which is less than 18 dB. In fact, the crest factor might be reduced even if the SUT is not strictly speaking "low passed" but merely has emphasized low frequencies (such as by boundary loading). M-Noise is essentially the same as pink noise below 500 Hz, so an SUT which is low passed at 500 Hz or less will have a crest factor of about 12 dB. An SUT which high passes a significant range of audible frequencies will have a crest factor which is higher than 18 dB.

A.5 Obtaining Accurate Sound Levels

To ensure the accuracy of sound level measurements described in this standard, an SLM conforming to IEC 61672-1:2013 performance class 1 is required. If a multi-purpose analyzer is used as the SLM, it should be verified to satisfy all applicable requirements of IEC 61672-1:2013 for a class 1 instrument including Peak C performance. If the SLM supports multiple sampling rates, accuracy in estimation of peak sound levels may be improved by increasing the sampling rate used to acquire audio data.

Annex B (normative): Additional measurement steps

B.1 Analyzers Without Gain Tracking

When using a software analyzer with an audio interface, the following steps may be necessary:

B.1.1

In order to ensure the microphone input of the analyzer does not clip one must estimate the maximum Peak sound level expected from the SUT. Put an IEC 60942-1 compliant calibrator on the microphone connected to the analyzer to ensure the expected maximum Peak sound level will not clip the analyzer input. For example, if the calibrator produced 94 dB and the expected maximum Peak sound level is 134 dB, the microphone input meter shall read -40 dBFS or lower.

B.1.2

Using a “Y” cable, especially between devices made by two different manufacturers, is sometimes problematic. Instead do one of the following:

a. Use an auxiliary output, set to unity gain, and use it as the reference signal for the analyzer. Add enough gain to the main output to achieve the maximum expected sound level of the SUT. Do not change these gains during the procedure.
b. Otherwise, using a processor, matrix the test signal to two outputs. Set one output to unity gain and use it as the reference signal for the analyzer. Add enough gain to the other output to achieve the maximum expected sound level of the SUT. Do not change these gains during the procedure.
c. Derive the reference signal internally. Be aware, this can add latency to the measurement path.

If the analyzer supports internal audio file playback capability, consider using the built-in generator to playback the M-Noise .wav file.

B.2 Recommendation for identifying Two Octaves of Compression

If the analyzer has 1/3rd octave tick marks, then two octaves can be identified by observing the closest third octave to the lowest frequency touching the Compression Target and counting up six tick marks.

B.3 Real-Time Transfer-Function Analyzer

Some analyzers do not analyze every time sample at every frequency. These analyzers are referred to as non-real-time transfer function analyzers. Such analyzers will miss some transient distortion events, so it is important to run the test for a full five minutes and even a single transient reduction in coherence should be taken very seriously. This is why the procedure recommends making a screen recording.

B.4 Considerations Regarding Analyzer Settings

The uncertainty in a transfer function is related to the signal-to-noise ratio, coherence, the integration time, and the bandwidth of the analyzer. Analyzer settings should be chosen such that the signal-to-noise ratio is 10 dB or greater ($\gamma^2 \geq 91\%$) and such that the uncertainty is much less than the 2 dB change needed to reach the compression target. Relevant analyzer settings include the analysis time interval (or analysis window length), the analysis time interval overlap, the number of analysis time interval windows included in the averaging, and the overall integration time (sometimes called the total averaging time).

The analysis time interval — often referred to as window length or FFT length — should be long enough to achieve at least 1/6-octave resolution at the lowest frequency of interest for the SUT. For example, to get 1/6-octave resolution at 20 Hz, one should have an analysis time interval of at least 400 ms.

Some analyzers allow the setting of the amount of overlap between consecutive analysis time intervals. Different amounts of overlap are appropriate for different analysis window shapes (typically around 50%).

Setting smaller values can increase the uncertainty and setting larger values might not allow the coherence to sufficiently detect non-linearities in the SUT.

With a typical overlap of approximately 50%, one can achieve a sufficiently small uncertainty by setting the number of analysis time intervals used for averaging to 8. Setting this number much higher risks the SUT changing its behavior more quickly than the analyzer can respond, and setting the number lower might not allow the coherence to sufficiently detect non-linearities in the SUT. This can result in overshooting the stop criteria and might cause the transfer function to continually change between steps 5.9 and 5.13. (This might even cause damage to the SUT, or, at a minimum, require repeating many of the procedure steps. An overlap greater than 50% could be used to improve the analyzer display's update speed, but the number of averages must then also be increased to keep the ratio of integration time to analysis interval time greater than 4 for sufficiently accurate coherence.

The overall integration time is a combination of the analysis time interval length (T), the percentage overlap (OL) of consecutive analysis time intervals, and the number of analysis time intervals used for averaging (N). The total integration time can be calculated as $T \times N - T \times OL \times (N-1)$. With the recommended overlap of 50% and 8 analysis time intervals used for averaging, the total integration time is primarily dependent on the length of the analysis time interval that is appropriate to the lowest frequency of interest for the SUT.

Annex C (Informative)
M-Noise™ Test Signal End User License Agreement (EULA)

The Meyer Sound M-Noise test signal is being provided to you subject to the following license agreement. Please read. IF YOU DO NOT AGREE TO THE TERMS OF THIS LICENSE, DO NOT DOWNLOAD OR USE THE M-NOISE TEST SIGNAL.

BACKGROUND

The M-Noise test signal was created by Meyer Sound Laboratories, Incorporated ("Meyer Sound") for the use and benefit of the professional audio community. Subject to the provisions of the license, Meyer Sound is making this test signal freely available to all audio professionals who wish to use this signal. TO WORK PROPERLY THE M-NOISE TEST SIGNAL MUST BE USED AS INSTRUCTED. DETAILED INSTRUCTIONS ON THE PROPER USE OF THE M-NOISE TEST SIGNAL ARE PROVIDED IN THIS STANDARD.

FILE DELIVERY FORMAT AND LICENSE

The M-Noise test signal is provided to you in Waveform Audio File Format (.wav), at multiple sample rates, for use in testing loudspeaker systems ("the Purpose"). You are granted free of charge the non-exclusive, transferable, worldwide and perpetual license to copy, use and analyze the M-Noise .wav files and the test signals provided by these files for the Purpose.

USE OF M-NOISE TRADEMARK

M-Noise is a trademark of Meyer Sound Laboratories, Incorporated and signifies that the audio file used in the tests is based on an original unaltered .wav audio file supplied by Meyer Sound ("genuine M-Noise test signal"). You may use the M-NOISE trademark to identify a test signal as a genuine M-Noise test signal and in connection with test results produced by a genuine M-Noise test signal used in accordance with the AES standards instructions. Your use of the M-Noise trademark shall inure to the benefit of Meyer Sound.

You may not use the M-Noise trademark:

i. To identify a loudspeaker system test signal that is not based on a genuine M-Noise test signal; or

ii. To identify a loudspeaker system test signal based on a genuine M-Noise test signal that has been altered in any way; or

iii. To identity or describe test results that were not produced by a genuine M-Noise test signal; or

iv. To identity or describe test results that were produced by a genuine M-Noise test signal without following the standards instructions on the proper use of the M-Noise test signal.

Use of the M-Noise trademark is not mandatory:

WARRANTY DISCLAIMER

The M-Noise test signal is provided to you "as is." Meyer Sound disclaims all warranties with respect to M-Noise, including, without limitation, any implied warranties of merchantability or fitness for a particular purpose. The entire risk as to the performance of M-Noise resides with you, the user. Meyer Sound does not warrant that the functionality of M-Noise will meet your requirements or that your use of M-Noise will be error free.

LIMITATION OF LIABILITY

In no event shall Meyer Sound be liable to you or to any third party for any consequential, incidental or special damages (including lost profits) arising out of or in any way connected with this license or your use of the M-Noise test signal, regardless of legal theory, even if Meyer Sound has been advised of the possibility of such damages.

GENERAL

If a court of competent jurisdiction holds any provision of this agreement invalid, such provision shall to that extent be deemed omitted and shall not affect the remaining provisions of this agreement, which shall remain in full force and effect. This agreement constitutes the entire agreement between you and Meyer Sound with respect to the M-Noise test signal and your use thereof. Any changes to this agreement must be in writing and signed by an authorized representative of Meyer Sound. A waiver of a term of this agreement shall not be deemed a continuing waiver of such term or a waiver of any other term of this agreement.

M-Noise is a trademark of Meyer Sound Laboratories, Incorporated.

The M-Noise Test Signal Generator and Method are covered U.S. Pat. No. 10,841,717.

AES STANDARDS: DRAFT FOR COMMENT ONLY

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