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PRINCIPAL INVESTIGATOR: Theodore Argo, Ph.D.

CONTRACTING ORGANIZATION: Applied Research Associates, Inc., Littleton, CO

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6. AUTHOR(S) Theodore Argo, Ph.D.; Greg Rule, P.E.; Alexandria Podolski; Santino Cozza; Kiersten Reeser; Mark Espinoza; Kyndall Tatum; Nathaniel T. Greene, Ph.D.; Carol Sammeth, Ph.D. Andrew Brown, Ph.D.; David J. Audet, Au.D.; Aoi Hunsaker E-Mail: targo@ara.com				5d. PROJECT NUMBER	
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7. PERFORMING ORGANIZATION NAME(S) AND ADDRESS(ES) Applied Research Associates, Inc. 7921 Shaffer Parkway Littleton, CO 8127		Univ. of Colorado School of Medicine Department of Otolaryngology 12800 E 19th Ave. Aurora, CO 80045		8. PERFORMING ORGANIZATION REPORT NUMBER	
University of Washington Dept. of Speech and Hearing Sciences 1417 NE 42nd St. Seattle, WA 98105		University of Minnesota Duluth Department of Electrical Engineering 1023 University Dr. Duluth, MN 55812			
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14. ABSTRACT Military personnel require hearing protection for a wide variety of environments and the current method of selecting appropriate hearing protection devices (HPDs) is based largely on guesswork. Only the Noise Reduction Rating (NRR) is used as a standard HPD specification; other important characteristics of advanced HPDs are not evaluated or reported in a standardized manner. The primary objective of this effort is to verify electromechanical test methods for evaluation of advanced HPDs to reduce stakeholders' long-term dependence on time-consuming and expensive human subject testing. A second objective of this effort is to develop a software tool using these verified HPD performance metrics to enable mission planners and Warfighters to select HPDs appropriate to support specific mission profiles, thereby optimizing Warfighter performance. In Year 1 of this program the human subject evaluation protocols and associated instrumentation were developed and approved. Electromechanical test apparatus were refined in preparation for evaluation of a broad set of HPDs. Development of the software tool was initiated to support determination of a transition path. In Year 2, electromechanical and human subject measurements will be performed on an array of HPDs and compared before inclusion within the software tool database. In Year 3, comparisons of electromechanical and human subject data were reduced to metrics for each characteristic.					
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1. Introduction

Military personnel require hearing protection for a wide variety of environments, and the current method of selecting appropriate hearing protection devices (HPDs) is based largely on guesswork. The only standard HPD rating currently available is the Noise Reduction Rating [NRR; Occupational Safety and Health Administration (OSHA) 1910.95]; other important characteristics of advanced HPDs are neither evaluated nor reported in a standardized manner. HPD characteristics may be evaluated using human subjects or electromechanical test methods. While human subject testing is the gold standard for final HPD evaluation (Berger 2005), the use of human subjects is a time-consuming and expensive process due to the regulatory and scientific protocols necessary to obtain reliable results. Human subject testing is particularly impractical for the purposes of HPD research and development, compounded by now prevalent rapid prototyping capabilities (Attaran 2017) and the need for quick evaluation. Human subject testing is also impractical for operational hearing protection qualification, due to the potentially large number of device variations and performance characteristics that may be important to the Warfighter. Therefore, electromechanical test methods are necessary to guide and constrain the need for high amounts of human subject testing. To address this challenge, ARA will deliver (1) a verified suite of quantitative, sensor-based tests to quickly, inexpensively, and comprehensively evaluate candidate HPDs for military use, and (2) a software tool to be used by operational planners to identify the optimal HPD solution based on mission requirements.

In the first year of this program, the research Team completed protocol design and experiment development. A suite of electromechanical tests developed under previous funding were updated to improve the valid range. For each of these tests, we also developed a human subject protocol to compare human subject performance on auditory tasks with the calculated output of the electromechanical test. We developed and acquired the equipment necessary to support each test and began recruitment of human subjects for pilot data collection. We initiated development of the HPD optimization software tool and began discussions with various Department of Defense (DoD) organizations regarding transition planning.

In the second year of this program, the research Team completed the human subject pilot tests, updated the human test protocols and began enrolling participants in the full study. A total of 121 subjects at both study sites were enrolled with 44 completing the full protocol. We also reviewed existing acoustic testing standards to establish a baseline for writing the new electromechanical test method in accordance with ANSI requirements. Negotiations continued with the DoD, and specifically the Defense Logistics Agency (DLA) to establish a transition pathway for the standards and the optimization tool. Additional design work on the design tool was completed to include demonstrations with the Hearing Center of Excellence (HCE) and the Medical Research and Development Command (MRDC).

In the third year of this program, the research Team completed human subject and electromechanical testing. Subsequent analysis of human subject data yielded a greater understanding of the impact of hearing protection devices on perception and spatial hearing. When compared with the electromechanical data, correlations between human subject and electromechanical data were developed. The validated electromechanical measures were then incorporated into draft standards.

Details of these efforts are described in the following sections.

2. Key Words

Hearing Protection, Electromechanical Testing, Acoustics, Audiology, Hearing Conservation

3. Accomplishments

3.1. What were the major goals and objectives of the project?

The primary objective of this effort is to verify electromechanical test methods for evaluation of advanced HPDs to reduce stakeholders' long-term dependence on time-consuming and expensive human subject testing. A second objective of this effort is to develop a software tool using these verified HPD performance metrics to enable mission planners and Warfighters to select HPDs appropriate to support specific mission profiles, thereby optimizing Warfighter safety and effectiveness.

The overarching goal of the proposed effort is to verify that a battery of previously developed electromechanical test methods for evaluation of HPD performance is predictive of human auditory performance. Thus, parallel electromechanical and human subject data sets will be measured. Concurrent with these efforts, verified HPD test data will be compiled to create an advanced HPD selection software tool for mission planners and Warfighters. Collectively, these advances will increase the efficiency and effectiveness of advanced HPD development and deployment, enhancing Warfighter protection and mission effectiveness.

Specific Aim 1: Human auditory performance during HPD use will be evaluated for comparison to parallel electromechanical test methods. Live human subjects will be evaluated for performance on speech perception in noise (hSQ), sound source localization (hSL), sound level-dependent attenuation across frequency (hLD), and audibility at low sound levels (hSN). Cadaveric human subjects will be evaluated for high-level impulse noise transmission to the inner ear (hIN). Outcomes are absolute performance measures for hearing protection devices needed for verification of the analogous relative measures produced by the electromechanical test methods.

Specific Aim 2: Results of the absolute measurements of human performance will be compared to the relative metrics produced by the electromechanical methods for signal quality (eSQ), sound localization (eSL), level-dependent frequency response (eLD), self-noise (eSN), and impulse noise attenuation (eIN). These methods will be modified and refined to implement source signals (durations, levels, spacing), analysis methods, and interpretation guidelines that best reflect the range of observed human performance and enable differentiation of HPDs. The results of the human subject and electromechanical test methods will be used to support development of relevant military and civilian HPD evaluation standards.

Specific Aim 3: Both human and electromechanical tests will be applied to a range of hearing protection devices to ensure the relationships developed in Specific Aims 1 and 2 hold across device types. Electromechanical metrics (verified by human performance) will then be used to generate an expandable database of advanced HPD performance ratings. This database will be incorporated into a tool through which end-users and acquisitions personnel may select optimal hearing protection devices for specific mission profiles and military occupational specialties.

3.2. What was accomplished under these goals?

3.2.1. Specific Aim 1, Major Task 1: Submission of Human Use Protocols and Preparation of Facilities

3.2.1.1. Subtask 1.1. Develop and Submit Human Use Protocols

This subtask is complete, as summarized in the prior annual report.

3.2.1.2. Subtask 1.2: Develop and Submit Human Cadaver Use Protocols

This subtask is complete, as summarized in the prior annual report.

3.2.1.3. Subtask 1.3: Prepare Human Test Facilities

This subtask is complete, as summarized in the prior annual report.

Milestone 1: Local IRB approval

This milestone is complete, as summarized in the prior annual report.

Milestone 2: HRPO approval

This milestone is complete, as summarized in the prior annual report.

3.2.1.4. Subtask 1.4: Submit Protocols for Army HRPO Approval

This subtask is complete, as summarized in the prior annual report.

Milestone 2: HRPO approval

This milestone is complete, as summarized in the prior annual report.

3.2.2. Specific Aim 1, Major Task 2: Test Method Verification in Human Subjects

3.2.2.1. Subtask 2.1: Obtain Pilot Psychoacoustic Measures of HPD Effects

This subtask is complete, as summarized in the prior annual report.

3.2.2.2. Subtask 2.2: Analyze Pilot Psychoacoustic Measures of HPD Effects

This subtask is complete, as summarized in the prior annual report.

3.2.2.3. Subtask 2.3: Obtain Psychoacoustic Measures of HPD Effects

Full-scale human subject testing at the University of Colorado (CU) and University of Washington (UW) test sites was completed in Year 3. A description of each task setup and procedure has been summarized in previous reports.

Testing followed a pre-programmed testing matrix (Table 1) in which the ordering of testing across tasks and hearing protectors was appropriately counterbalanced to ensure even accumulation of data and account for anticipated subject attrition, with testing blocked such that subjects completed all localization trials, all QuickSIN and Modified Rhyme Test (MRT) trials, and all testing for Alternate Binaural Loudness Balance (ABL) and Real Ear Attenuation at Threshold (REAT) testing within a minimal number of sessions. This protocol likewise reduced variability of hearing protector placement across trials within each task, minimizing an anticipated source of variability.

Table 1. Final testing matrix for full-scale human subject testing. The ordering of tasks, HPD types, and test materials is counter-balanced to ensure even accumulation of data across the testing period, and to anticipate impacts of subject attrition. The matrix shown is an excerpt from the UW site; a parallel matrix was established for the CU site. Additional rows were added as-needed following the same counterbalanced ordering.

SUBJECT ID	HPD ORDER	QSLN ORDER	Enrollment	Appointment	paid	SESSION 1	X(notes)	paid	SESSION 2	X(notes)	paid	SESSION 3	X(notes)	paid	SESSION 4	X(notes)	paid	SESSION 5	X(notes)	paid	SESSION 6	X(notes)	paid	SESSION 7	X(notes)	paid	SESSION 8	paid	(SESSION 8)
W01	Order1				NSL		NSL		HSNHLQ																				
W02	Order1	Order2	Enrolled date: MMDDYY		NSL		NSL		NSQ																				
W03	Order1	Order3	Enrolled date: MMDDYY		HSNHLQ		NSL		NSL																				
W04	Order1	Order4	Enrolled date: MMDDYY		NSL		HSNHLQ			NSL		HSNHLQ																	
W05	Order1	Order5	Enrolled date: MMDDYY		NSL		NSL		HSNHLQ																				
W06	Order1	Order6	Enrolled date: MMDDYY		HSNHLQ		NSL		NSL																				
W07	Order2		Enrolled date: MMDDYY		NSL		NSL		HSNHLQ																				
W08	Order2	Order7	Enrolled date: MMDDYY		NSL		NSL		HSNHLQ																				
W09	Order2	Order8	Enrolled date: MMDDYY		HSNHLQ		NSL		NSL																				
W10	Order2	Order9	Enrolled date: MMDDYY		NSL		HSNHLQ			NSL		NSL																	
W11	Order2	Order10	Enrolled date: MMDDYY		NSL		NSL		NSL																				
W12	Order2	Order11	Enrolled date: MMDDYY		HSNHLQ		NSL		NSL																				
W13	Order2	Order12	Enrolled date: MMDDYY		NSL		NSL		NSL																				
W14	Order2	Order13	Enrolled date: MMDDYY		NSL		NSL		HSNHLQ																				
W15	Order2	Order14	Enrolled date: MMDDYY		HSNHLQ		NSL		NSL																				
W16	Order3	Order15	Enrolled date: MMDDYY		NSL		HSNHLQ			NSL		NSL																	
W17	Order3	Order16	Enrolled date: MMDDYY		NSL		NSL		HSNHLQ																				
W18	Order3	Order17	Enrolled date: MMDDYY		HSNHLQ		NSL		NSL																				
W19	Order4	Order18	Enrolled date: MMDDYY		NSL		NSL		HSNHLQ																				
W20	Order4	Order19	Enrolled date: MMDDYY		NSL		NSL		NSL																				

3.2.2.3.1. Human Subject Enrollment

Study enrollment was completed at both UW and CU study sites. Both sites recruited subjects through a combination of recruitment flyers, word-of-mouth, and email contacts. At the Washington site, a total of 96 subjects were enrolled, with 13 additional subjects tested but failing to qualify. At the Colorado site, a total of 66 subjects were enrolled, with 4 additional subjects tested but failing to qualify. The number of complete datasets per device per task varies across sites. Enrollment status is summarized in Table 2.

Table 2. Human subject enrollment status. *The number of “completed” subjects is based on the task at each site for which the fewest complete datasets exist, i.e., other tasks have accumulated a greater number of complete datasets on one or more devices.

Site	Screened	Enrolled	Completed
University of Colorado	70	66	57*
University of Washington	109	96	65*

3.2.2.3.2. Human Subject Testing Results

3.2.2.3.2.1. Sound Localization (hSL)

Human sound localization (hSL) data consist of response locations differenced from corresponding target locations leading to per-trial localization errors. Figure 1 shows an example recording (for illustrative purposes only) demonstrating the error calculation procedure. The x-axis represents the azimuth (Az), the y-axis elevation (El), and the z-axis is time within this trial. The subject's head position as a function of time is shown with a heavy black line, and is shown starting at the origin (0,0), and moving to the response location over the course of approximately 3 seconds. This response location differs from the target location in both azimuth and elevation; total angular error is calculated as the magnitude of the vector difference between response and target.

Figure 2 plots show the aggregated subject responses from the UW and CU study sites. Distributions of localization error are shown for each HPD (colors) at each azimuth (columns) and elevation (rows) location. Note, total error is bounded by 0° (no error) and 180° (the exact opposite direction). Certain errors are expected, such as a compression of responses to 0° elevation, and a front/back (hemifield) reversal. hSL data is distilled for comparison to the eSL data in Sec 2.

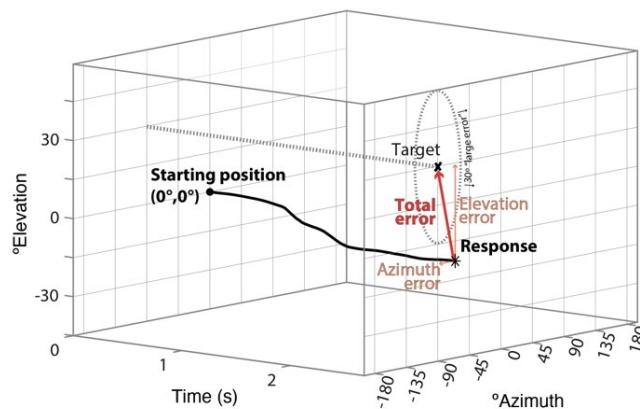


Figure 1. Head position tracking demonstrates convergence (or divergence) of responses with respect to the target across time. Errors include Azimuth and Elevation components. The trigonometric combination of these components determines the total error, an ecologically useful measure of mislocalization.

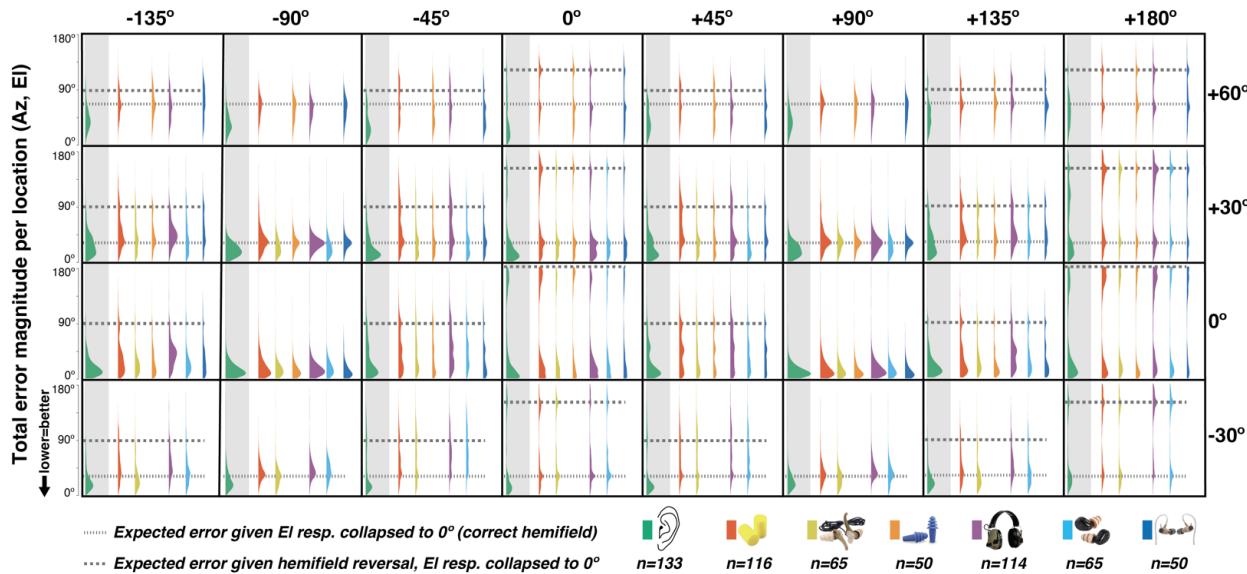


Figure 2. Sound localization (hSL) data pooled across both UW and CU study sites. Data are presented as total error histograms (combining azimuth and elevation error) for 32 different source locations (8 azimuths and 4 elevations). Within each panel, upward spread of data (nonzero bins at higher y-values) indicates larger error magnitudes (largest possible error is 180°), while greater horizontal distribution extent indicates more trials yielding that error magnitude (i.e., a prominent mode). Dashed and dotted lines reflected stereotyped varieties of error observed in the localization task – front-back reversal with elevation response collapsed to zero, and elevation response collapsed to zero (as labeled). Left to right for each location: open ear, EAR Classic, Combat Arms Open Mode, Elvex Quattro, Comtac V, TEP-200, Invisio X5.

The most complicated metric to compare between human and electromechanical results is sound localization, due to the rich data sets, numerous conditions, multiple contributing cues, and complicated analysis. The human subjects sound source localization error distributions shown in

Figure 2 of the present report reflect varied performance across devices. Compared to open ear performance, HPD-mediated performance shows increased error values, with errors at reproducible locations leading to modes (local maxima) in error distributions. To quantify the patterns of errors across devices, we have developed a Gaussian Mixture Model (GMM)

Conceptually, the GMM seeks to explain the total picture of subjects' error patterns as a mixture of a few major error types: failure to perceive source elevation, failure to discriminate forward ("front") and rearward ("back") sounds, and gross errors in the lateral dimension leading to categorical "left" and "right" responses rather than graded localization responses. Each of these errors manifests as a mode in total error magnitude, the location of which varies depending on the target location. For example, a front-back confusion for a source at $(0^\circ, 0^\circ)$ (azimuth, elevation) has an expected value of 180° , whereas a front-back confusion for a source at $(45^\circ, 0^\circ)$ has an expected value of 90° (reflecting an expected response at $(135^\circ, 0^\circ)$). Using the total distribution of response errors, the GMM fits distributions to each error term, and the area under each component Gaussian defines its interim contribution to the model. A schematic example of the layout of the GMM results for a single speaker location is illustrated in Figure 3.

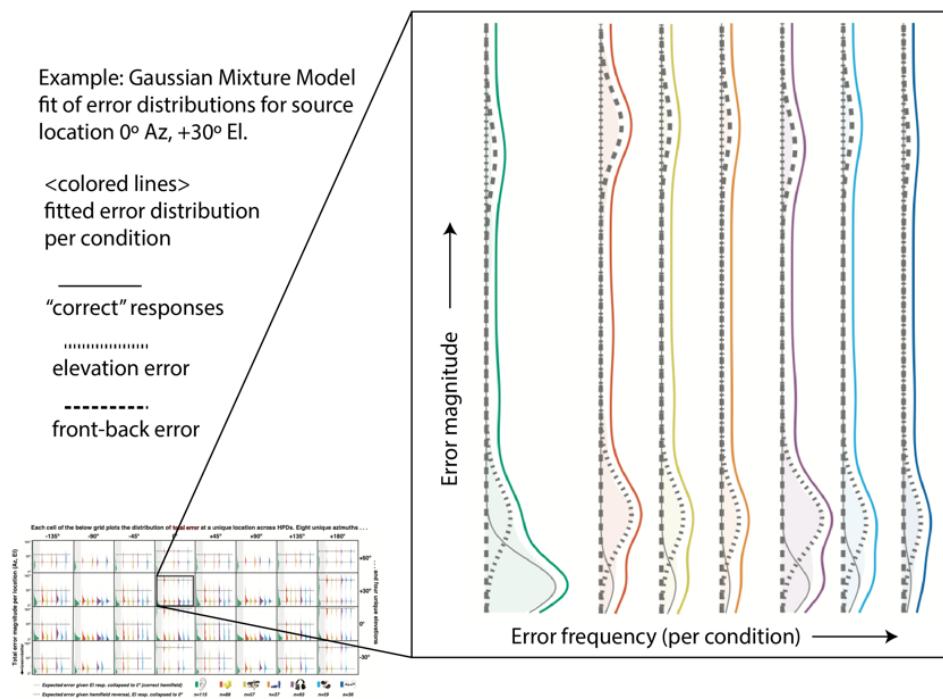


Figure 3. Layout of Gaussian Mixture Model (GMM) used to generate summary human sound localization metric hSL. A single location $(0^\circ, +30^\circ)$, cf. Figure 2, is illustrated. For all locations, the area under each of the component Gaussians is computed, this value is weighted by the magnitude of the expected error value, and the open-ear values subtracted to express a relative change for HPD condition. Values are then aggregated across locations and normalized to derive a final hSL metric.

Because the magnitude of error is ecologically important (i.e., a 180° orientation error is worse than a 90° orientation error in terms of disorientation and/or increased reaction time), the sum of the area under each curve is weighted to define its total contribution to the final model. Finally, because some target locations preclude the occurrence of some error types (e.g., a front-back

confusion is not theoretically possible for a source at (90°,0), some error terms are excluded from the fitting procedure at some locations.

In the below equation, the terms Err refer to the area under the curve for each of four possible components. The terms W refer to the weight assigned based on the expected value. Note that a “correct” term captures responses near the target. In practice, when performance is degraded by an HPD, the responses from that term are redistributed to the other terms. The summed term $hErr_x$ defines the error for condition x at each azimuth i and each elevation j .

$$hErr_x(i, j) = Err_{correct} + W_{CE}Err_{CE} + W_{CA}Err_{CA} + W_{FB}Err_{FB}$$

In order to quantify the relative effect of each HPD on performance, corresponding open-ear error terms are subtracted from HPD (closed-ear) error terms at each azimuth and elevation. This difference is then divided by the open-ear error term, such that equal error terms in closed- and open-ear conditions (i.e., no HPD-induced error increase) lead to a value of 0. Finally, to normalize the scale of the metric with the eSL metric, this value is subtracted from 1.

$$hSL = 1 - \text{mean} \left(\frac{hErr_o(\theta, \phi) - hErr_c(\theta, \phi)}{hErr_o(\theta, \phi)} \right)$$

As constructed, hSL values approaching 1 indicate performance similar to open ear and hSL values approaching 0 indicate increasingly poor localization performance.

3.2.2.3.2.2. Sound Quality (hSQ)

Human Sound Quality (hSQ) testing was completed at both UW and CU study sites. hSQ consists of two separate speech-in-noise tasks: (1) the Modified Rhyme Test (MRT), which presents target words in a background of omnidirectional pink noise and requires subjects to identify the target word using a keypad given a closed set of 6 similar (rhyming) words, which are displayed on a monitor inside the testing arena, and (2) the QuickSIN, which presents a target sentence co-located in a background of multi-talker babble and requires subjects to verbally repeat back the sentences to the experimenter. Performance is quantified, across Open Ear and HPD conditions, according to the percent of target words correctly identified as a function of signal-to-noise ratio. Lower signal-to-noise ratios make the task more difficult, resulting in lower percent-correct scores.

As described in previous reports, MRT data collection was conducted using a set of pre-recorded speech tokens of three female speakers available on the NIST website, reading the MRT word database outlined in ANSI/ASA standard S3.2-2009, and presented from the loudspeaker directly in front of the subject (and directly in front of the monitor presenting response options). Background pink-noise was presented from 7 speakers distributed equally in azimuth around the subject’s head. Subjects held a wireless number pad which they used to control stimulus presentation and record perceived responses. During each trial, subjects were presented with the six possible words spoken; subjects play the word (once) when in position and ready, and subjects may proceed to the next trial once a word is selected. Words were presented with three nominal signal to noise ratios, -3, 0, and 3 dB, where the sound pressure level (SPL) of the background noise was set to vary above and below the 70 dBA SPL of the target words (i.e. the target was fixed and the noise masker varied in level).

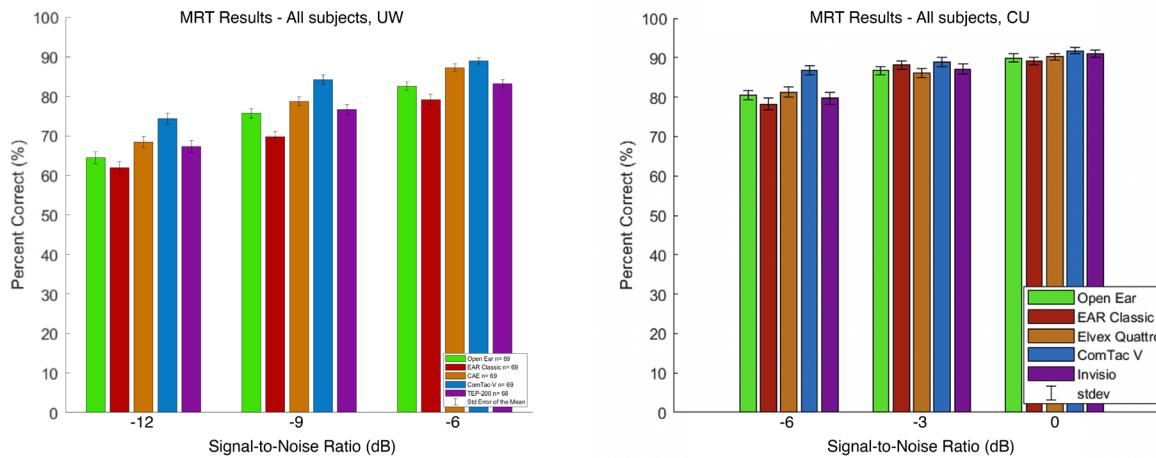


Figure 4. Summary MRT data from the UW (left) and CU (right) study sites. Major groupings are the signal-to-noise ratios defined by variation in the level of presented pink noise. Error bars give standard error of the mean. Performance improves with increasing SNR but is generally worst for the EAR Classic and best for the ComTac V. Notably, in this task, the target is presented from 0° (directly in front of the subject) while pink noise is presented from ±45°, ±90°, ±135°, and 180°. Therefore, a device with a strong forward-directional characteristic (e.g., such as the forward-directional microphones of the ComTac) should be expected to outperform comparatively omnidirectional open ears.

A disparity was noted in the percentage correct observed at the two study sites, and this effect is present in this final data set. To explore the cause of this difference, the MRT stimuli were recalibrated at both sites. During this process, the causes of some minor differences in the SNR presented to subjects at each site were discovered. First, the microphone orientation was not identical at both sites during calibration, leading the UW site to slightly overestimate SNR. Second, the computer monitor was not in position at the CU site, leading the CU site to slightly underestimate SNR. Third, the SPL of the speech stimulus was assessed by monitoring the readout of a sound level meter on fast mode, which slightly overestimated the signal level recorded. Finally, analysis of the waveforms yielded a difference in the SPL of the three talkers, leading to greater SNR variability from one subject to the next than anticipated.

To address these concerns, each individual MRT speech waveform was calibrated at each site using the same calibration protocol to determine the de facto SNR level for each speech waveform. Calibration results are shown for the CU study site in Figure 5. Results with the computer monitor rotated away from the subject seat position are shown at left, and with the monitor rotated towards the subject's seat position at right. The dBA value measured for each speech waveform (x-axis), as well as the mean dBA level of the pink noise presentation (at 0 dB SNR, horizontal grey lines) are shown in the top row, and the de facto SNR value are shown in the bottom row. Colors indicate the speaker (female 1, 3, or 4) for each word (dots) and the median across the speaker (lines), and the heavy blue line represents the median level across all words presented. The actual SNR value used for testing (at the nominal 0 dB SNR level) at CU is thus shown in the lower right panel. The median de facto SNR (heavy blue line) was -3 dB at the nominal 0 dB level, thus all CU SNR values must be adjusted downwards, and the actual (median) SNRs presented were 0, -3, and -6 dB. A similar analysis at UW yielded de facto SNR values of -6, -9, and -12 dB. Additional analysis is underway to determine subject performance on a per-trial basis, but on average, the results from the two study sites represent an overlapping but disparate range of SNR values and largely explains the study site differences.

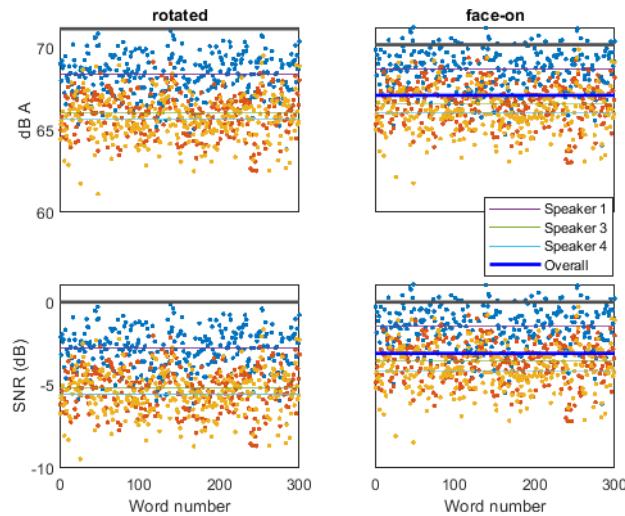


Figure 5. MRT calibration results from the CU study site. Dots represent the level (in dBA, top) and signal-to-noise ratio (SNR, bottom) measured from individual spoken word presentations for the three female talkers, their medians (colored lines), and the background pink noise level (horizontal grey line). The computer monitor used to provide the subject visual input for the MRT task was either rotated 90° away from (left), or towards (right) the subjects seating position, which increased noise, signal, and SNR levels.

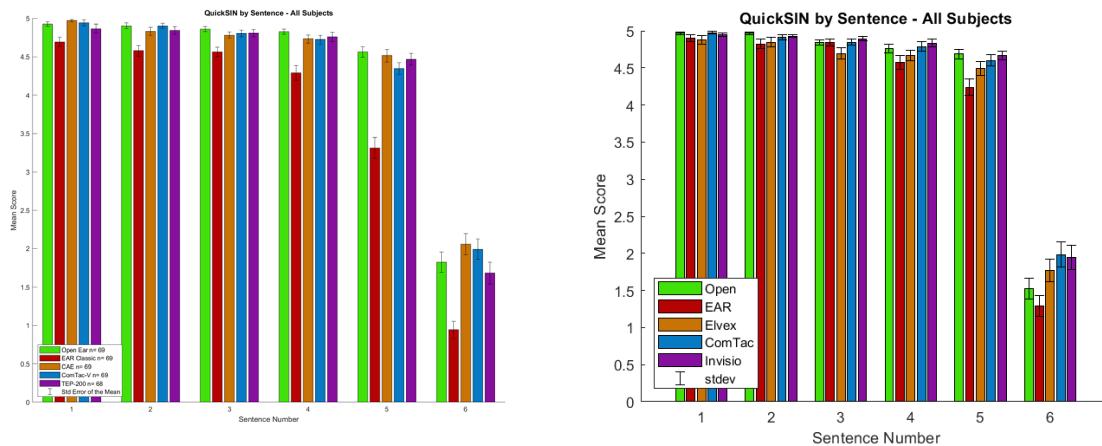


Figure 6. QuickSIN data from the UW (left) and CU (right) study sites. Major groupings are sentence numbers in the QuickSIN sequence. In this sequence, each sentence becomes progressively more difficult to hear as the level of background noise (multi-talker babble) increases (i.e., as the SNR decreases in 5-dB decrements from +25 dB in sentence 1 to 0 dB in sentence 6). Bars show mean words correct (out of 5; equivalent percent correct can be obtained by multiplying these values by 20). Error bars given the standard error of the mean (N indicated in the inset legend). The Open condition and active HPDs produce similarly good performance in many cases at SNRs of +5 dB or better. The EAR Classic (high attenuation passive device) produces the worst performance in all cases. At the lowest SNR (0 dB), performance approaches the floor, with 1-2 words (of 5) correct on average, and increased variability. In this case, the Combat Arms gave the best score.

QuickSIN data was collected using commercially available QuickSIN recordings, which include spoken sentences at a fixed level (70 dBA), combined with a multi-talker babble background (that increase in level from one sentence to the next, from 25 to 0 dB SNR in 5 dB steps), in a single loudspeaker channel. The target and noise are co-located, thus the differences in calibration noted above, which result from varying directionality of the test equipment, do not affect the SNR of the output (though the overall level may differ slightly). As a result, no corrections are required for QuickSIN output. The overall results across sites (Figure 6) are generally consistent, though performance with the EAR classic was somewhat worse in subjects at UW than CU.

3.2.2.3.2.3. Self-Noise (hSN)

Human subject self-noise (hSN) was completed following a real-ear-attenuation-at-threshold (REAT) under headphones paradigm to measure the effective change in auditory sensitivity produced by active HPDs. Specifically, while active HPDs are designed to ‘pass’ low-level sounds, hum from active electronics may serve to mask sounds near threshold. In practice, REAT in the hSN task is calculated by computing the difference between open-ear thresholds and active-HPD-affected thresholds. Positive values indicate higher thresholds for the HPD than for the open ear.



Figure 7. hSN data from the UW study site. Small upper panels show data for 65 individual subjects across 3 conditions: Open ear (green), TEP-200 (blue), and Invisio X5 (red). The TEP-200 was tested in Gain setting 2 of 4. The Invisio X5 was tested in gain setting 2 of 3. For comparison purposes, the open-ear audiogram measured by an audiologist at the intake appointment is plotted in black, generally following and often intersecting the self-determined open-ear threshold. Blue and red curves fall above green curves in most but not all cases. Subsets of data are missing for a few subjects. Lower panel: Mean computed REAT values for TEP-200 and Invisio X5 (effectively, the distance from red or blue to green curves) across subjects.

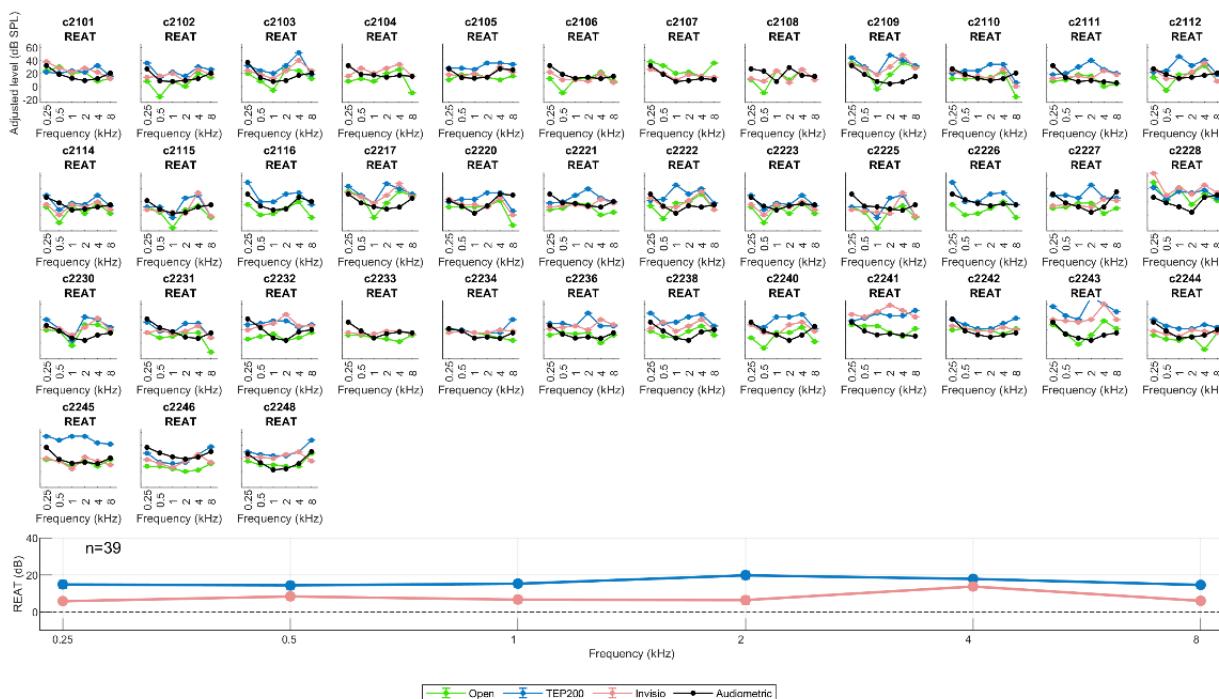


Figure 8. hSN data from the CU study site; legend as above.

3.2.2.3.2.4. Level-Dependence (hLD)

Human level-dependence (hLD) testing was transitioned from a task with live human subjects to measurements completed in Post-Mortem Human Surrogate (cadaver/PMHS) specimens due to the limited source levels achievable with human subjects. The PMHS hLD measurement generates input-output functions substantively comparable to those generated with electromechanical test fixtures but does so with human cadaveric specimens that recapitulate the material properties of live human ears, including alternate sound transmission pathways to the inner ear. These experiments were completed in tandem with the measurements made for the hIN task but were conducted first to minimize the risk of damage to anatomical structures resulting from shock wave exposure. In total, experiments were conducted in 15 whole cephalus PMHS heads, for a total of 28 ears tested; an additional several heads/ears were excluded from the study due to anatomical damage/abnormalities observed or induced during specimen preparation.

The experimental methods have been described in detail in previous reports and publications (e.g., Greene et al. 2017, 2018). Briefly, whole human cadaver heads, such as that shown in Figure 9, are prepared bilaterally with a mastoidectomy and facial recess by an experienced Otolaryngology resident or attending physician to provide access to the middle ear space. A wide incision is made in the skin behind the pinna, and retracted during middle ear preparation and measurements, so that it can be sutured back in place for the measurements. Stainless steel guide tubes are mounted to the skull with stainless steel screws and bone cement, and microscopic (300 μ m diameter) fiber optic pressure probes (FISO Inc.) are inserted into the cochlea (in the scala vestibuli near the oval window, and scala tympani near the round window) via small cochleostomies made with a sharp pick, and into the ear canal near the tympanic membrane, underwater to prevent air infiltrating the cochlea. Once inserted, pressure probes are fixed in place with dental impression material and cyanoacrylate adhesive. Velocity of the stapes

and round window are measured with a laser Doppler vibrometer (Polytec GMBH) before and after making cochleostomies and inserting the pressure probes, such as those as shown in Figure 10, and compared to normative responses available in the literature (Rosowski et al. 2007, Nakajima et al. 2009, Greene et al. 2017) to verify the condition of the specimen.

Measurements were made in scala vestibuli (P_{SV}) and scala tympani (P_{ST}), and the differential pressure ($P_{Diff} = P_{SV} - P_{ST}$), which provides the acoustical drive to the basilar membrane (Dancer and Franke, 1980), was calculated as shown in Figure 11. Measurements were made both during hearing protector use, and when the ear was unoccluded. Insertion loss was typically low for low frequencies, and increased for frequencies > 1 kHz, but varied across HPDs.

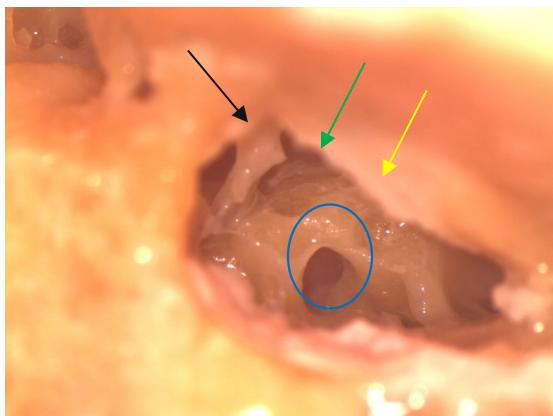


Figure 9. Photograph of the temporal bone preparation (left), and PMHS positioning (right) during hLD testing. Each ear is prepared with a canal-wall-up mastoidectomy and facial recess to expose the middle ear. Here, the posterior crus of the stapes is visible (black arrow), as well as the round window (blue circle) and small depressions drilled into the cochlear promontory bone in preparation for pressure probes to be inserted into the scala vestibuli (P_{SV} ; green), and scala tympani (P_{ST} , yellow). PMHS are suspended upside down from the c-spine to allow skull vibration and prevent loss of cranial contents during testing.

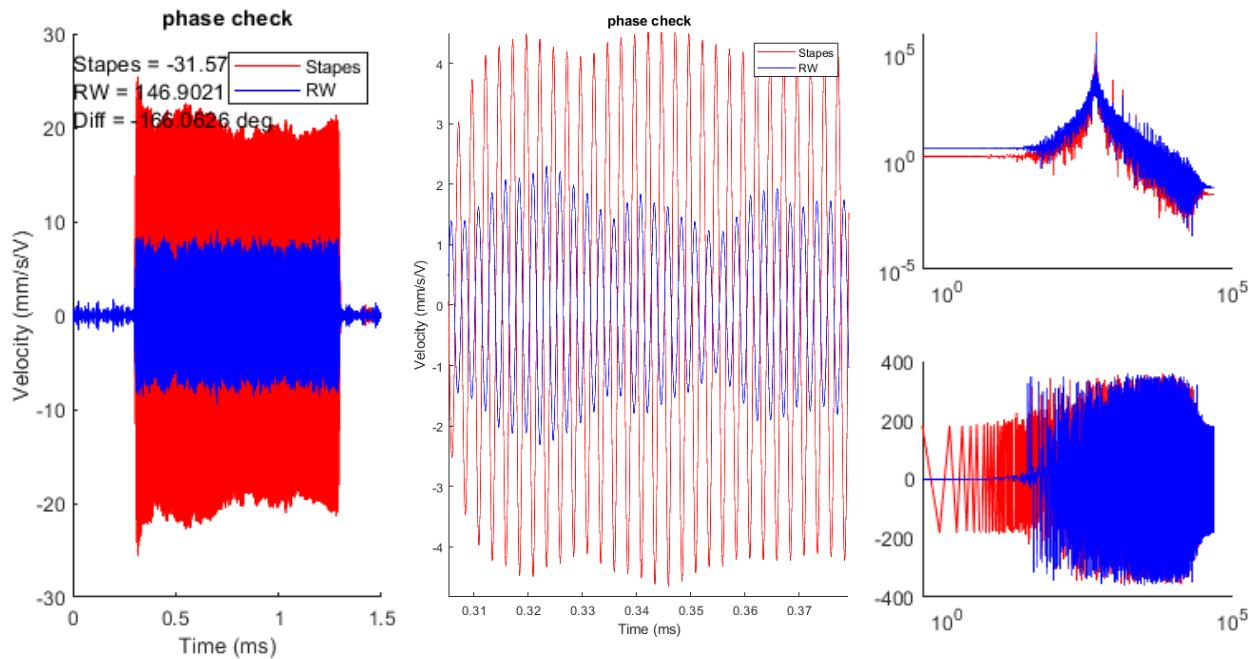


Figure 10. Stapes (red) and round window (blue) velocity measurements made for a 1 second duration, 400 Hz tone pip before making cochleostomies in the bone over the scala vestibuli and scala tympani. The left panel shows the velocity as a function of time for the full recording duration, the center panel shows a zoomed in section of the recording demonstrating the anti-phasic responses of the two measurements, and the right panel shows the the amplitude (top) and phase (bottom) spectra of each recording. The phase angle of each recording, as well as the difference between the two, derived from the phase spectra at right, is provided as text in panel 1. High level sound stimuli were generated by custom MATLAB software, amplified (Crown power amplifier), and presented from an 8" coaxial loudspeaker (B&C 8CXN51) inside a stainless-steel concentrating horn with a 2" diameter outlet. The loudspeaker outlet was placed approximately 6" away from the specimen's ear and pointed directly at the specimen's interaural axis. A probe tube microphone was placed adjacent to the specimen's ear to measure the incident sound pressure level. Pressure probes in the ear canal, scala vestibuli, and scala tympani measured sound arriving at each point in the auditory transduction pathway. Sound was presented at full scale (approximately 120 dB SPL) and attenuated in 10 dB increments to investigate the level dependence of ossicular transmission, and attenuation provided by HPDs.

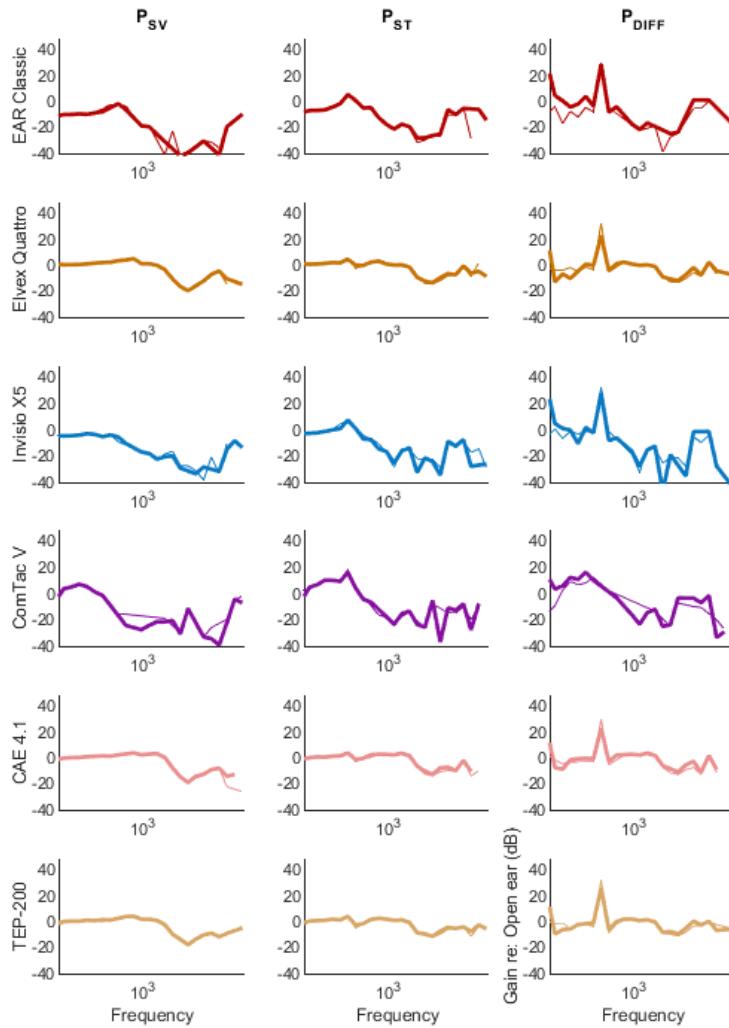


Figure 11. The attenuation provided by each hearing protector during high sound presentation levels is assessed in human cadavers. Responses were measured during presentation of approximately 110 dB SPL (thin) and 120 dB SPL (thick) tones at frequencies between 100 Hz and 10 kHz (in $\frac{1}{4}$ octave steps). Responses recorded in the scala vestibuli (P_{SV}) and scala tympani (P_{ST}), as well as the differential pressure ($P_{Diff} = P_{SV} - P_{ST}$) were recorded with each hearing protector, and the responses compared to the response when the ear was unoccluded to determine the insertion loss (in dB). Analysis continues, and these results should be considered preliminary.

3.2.2.4. Subtask 2.4: Analyze Pilot Human Cadaver Measures of HPD Attenuation to Impulse Noise Exposures

This subtask is complete, as summarized in the prior report.

3.2.2.5. Subtask 2.5: Obtain Human Cadaver Measures of HPD Attenuation for Impulse Noise Exposures

To conduct shock wave transmission measurements, the specimen is suspended up-side-down from the neck to enable free vibration of the skull since we anticipated a substantial bone-

conducted component will contribute to the overall cochlear response; see Greene et al. 2018. To do so, skin and soft tissue are removed from the lowest cervical spine, a 4" section of ABS pipe is fit over the C-spine and held in place with 4" screws, and any cavity left between the tissue and pipe is filled with polyester automotive body filler. This pipe is inserted into a custom steel/aluminum frame and held in place in a larger PVC pipe with a concentric series of set screws. This frame is placed on a rolling stainless-steel cart, allowing the specimen to be positioned directly in front of the shock tube located inside a double-walled sound attenuation booth, and the shock tube is operated manually from outside the booth. Plastic sheeting and acrylic plates are placed around the booth to limit any potential contamination from the specimen. See Figure 9 for a typical test setup.

Measurements were made with each of the hearing protectors for at least 5 repetitions, to allow averaging to improve signal-to-noise, as described in the manuscript published this year (Anderson, Argo, & Greene, 2023).¹ Foam earplugs were placed once and left in place due to the long expansion time, but all other devices were re-placed before each exposure (particularly important for the ComTac-V as it was often removed by the shock wave). Measurements were made by making 10-second-long recordings, with the shock wave captured approximately halfway through the recording, to provide ample time before and after the impulse to estimate background noise. Preliminary analysis was presented in the previous annual report, as developed in the same manuscript accepted for publication this year (Anderson, Argo, & Greene, 2023). Further development of these analyses is ongoing and are aimed at both meeting and extending standard assessments in the blast literature.

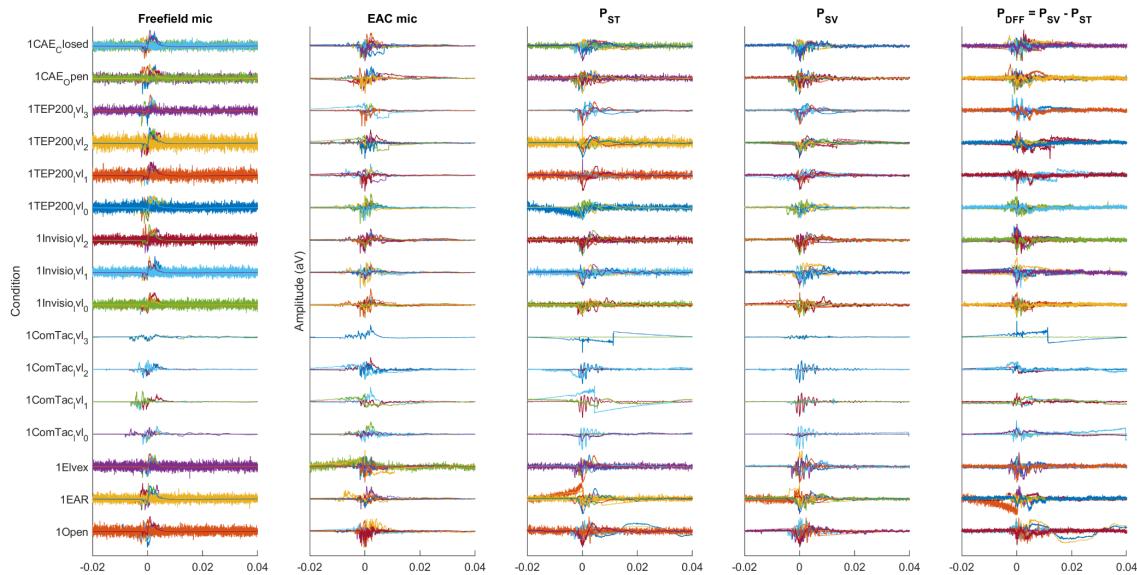


Figure 12. Exemplar data for impulse noise exposures in PMHSs. Pressure recordings from the microphone adjacent to the PMHS in the freefield, the microphone/pressure probe in the ear canal/external auditory canal (EAC), pressure probes in the scala tympani (P_{ST}) and vestibuli (P_{SV}), as well as the differential pressure (P_{Diff}) are shown as a function of time relative to the arrival of a ~1 psi shock wave for each of the hearing protector conditions. At least 5 repetitions were collected for each condition, and all mode/gain settings were assessed (e.g. ComTac V level 0-3, Combat Arms open/closed).

¹ Anderson, D. A., Argo, T. F., 4th, & Greene, N. T. (2023). Occluded insertion loss from intracochlear pressure measurements during acoustic shock wave exposure. Hearing research, 428, 108669..

Impulse noise exposures are analyzed using two metrics: Impulse Peak Insertion Loss (IPIL; ANSI S12.42-2010), and Impulse Spectral Insertion Loss (ISIL; Fackler et al. 2017²). Example IPIL and ISIL calculations are shown for a single specimen in Figure 12 and Figure 13. As with hLD, analyses are ongoing.

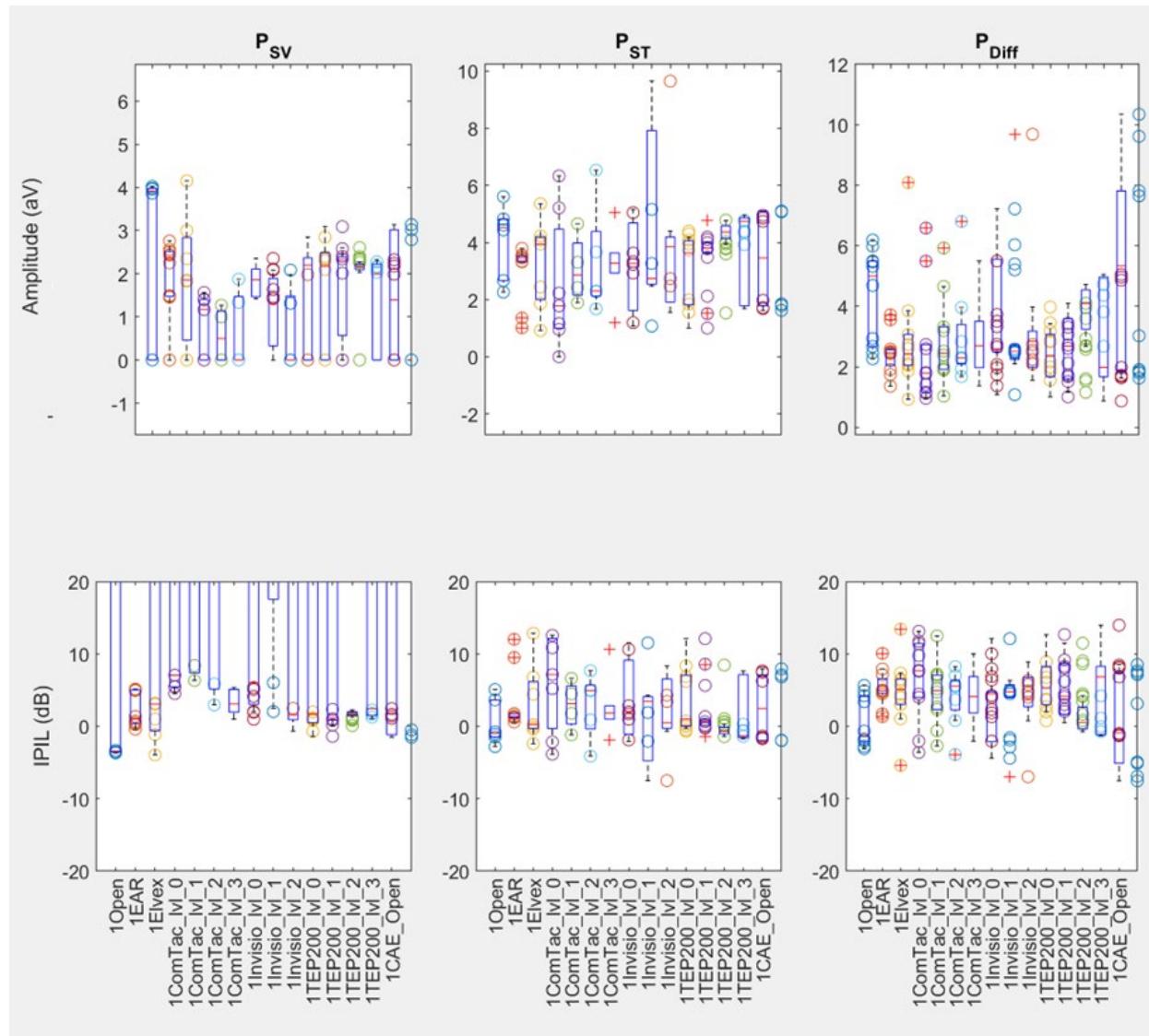


Figure 13. The peak pressure for each impulse noise measurement was found for each location (P_{sv} , P_{st} , and P_{diff} pressures), each repetition, and each hearing protector condition (top row). IPIL (bottom row) was then calculated from the ratio of the mean open ear peak pressure to the peak pressure in each HPD condition. The free-field pressure sensor shows consistent peak pressures and no IPIL gain as expected. The other sites show varying degrees of insertion losses, depending upon both the HPD type and the sensor location. Considerable variability is observed due to sensor failures, as well as insertion variability.

² Fackler, C.J., Berger, E.H., Murphy, W.J. and Stergar, M.E., 2017. Spectral analysis of hearing protector impulsive insertion loss. *International journal of audiology*, 56(sup1), pp.13-21.

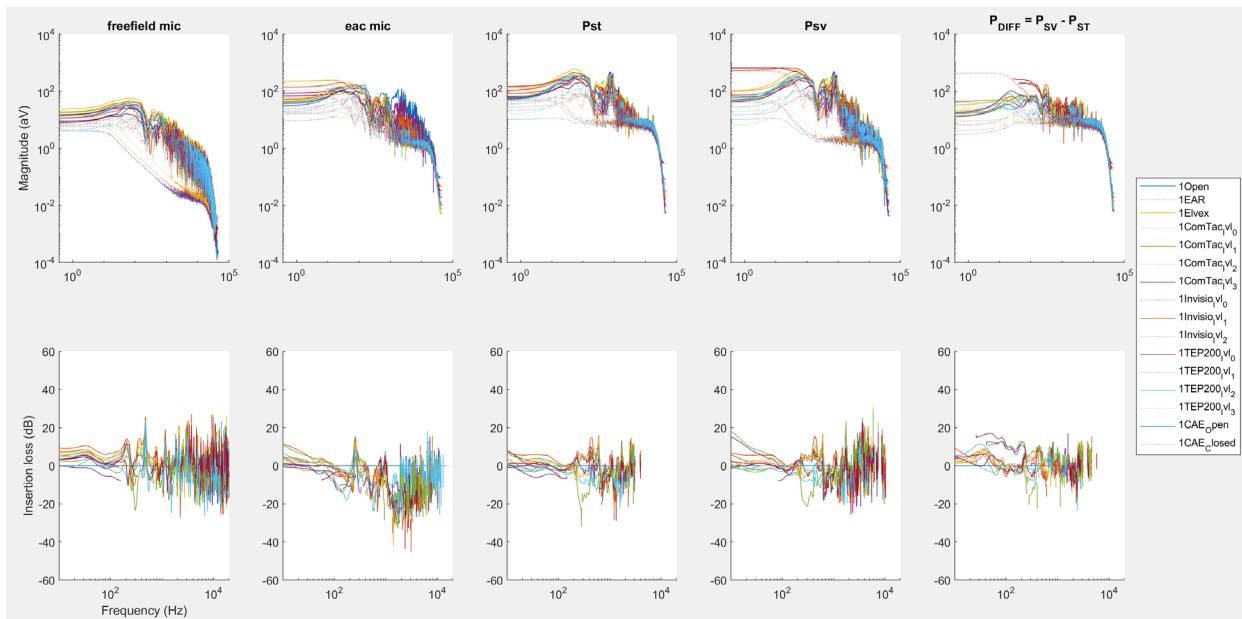


Figure 14. The impulse noise amplitude spectrum (top) and ISI (bottom) was found for each measurement ($P_{\text{freefield}}$, P_{EAC} , P_{SV} , P_{ST} , and P_{Diff} pressures), averaged across repetitions, and shown for each hearing protector (colors). ISI was calculated from the ratio of the mean open ear peak pressure to the peak pressure in each HPD exposure. Average ISI values are somewhat lower than IPIL values, though general trends across HPDs remain. In general ISI is lower in intracochlear pressure than EAC measurements, likely due to the effects of alternate sound transmission pathways to the inner ear not captured in the EAC sensor.

3.2.2.6. Subtask 2.6: Conduct Ongoing Quality Assurance Review of Psychoacoustic, Cadaver, and Associated Data

Completed with final data collection.

Milestone 3: Manuscript on impulsive noise measurements on cadaveric subjects with HPDs

This milestone is complete, as summarized in the prior annual report.

Milestone 4: Manuscript on HPD effects on perception

Multiple manuscripts, describing the effects of HPD use on each major perceptual effect investigated in this program, are in preparation and will be completed by the end of the period of performance.

3.2.3. Specific Aim 2, Major Task 1: Verification of Electromechanical Test Methods

This subtask is complete, as summarized in the prior annual report.

3.2.4. Specific Aim 2, Major Task 2: Develop Prototype Standards for Appropriate HPD Evaluation Methods

Using key ANSI standards as a guide, draft standards were created for the eSQ, eSL, eSN, and eLD. The full draft standards are included in Appendix C. A memo outlining updates to ANSI standards for eIN is also included in the appendix. As data analysis continues, the original calculations for each metric in the draft standards may be updated. A further discussion of sources of uncertainty and estimations of uncertainty may also be incorporated as calculations are revised.

Dr. Argo attended the Acoustical Society of America Standards Working Group 11 meeting where changes to ASA/ANSI S12.42 governing evaluation of HPDs under impulsive noise were debated. Changes to S12.42 discussed during that meeting and the subsequent meeting to be held in early February at the National Hearing Conservation Association convention will be used in the evaluation of final data and update the guidance on the eIN metric.

Milestone 5: Deliver standards input to appropriate standards committees

Draft standards are included as appendices to this report and may be updated as data analysis continues.

3.2.5. Specific Aim 3, Major Task 1: Hearing Protection Device Evaluations

As previously reported, 20 unique hearing protection devices have been evaluated using the electromechanical test battery, some of which were tested in multiple “modes” (e.g., Combat Arms Open vs Combat Arms Closed). After preliminary metric calculations, some adjustments have been made not only to the methods used to calculate the metrics, but also the test methodologies.

3.2.5.1. Signal Quality (eSQ)

The current eSQ metric that has been selected is based on the Speech Intelligibility Index (SII) established in ANSI/ASA S3.5-2020, *Methods for Calculation of the Speech Intelligibility Index*.

The eSQ metric is a combination of the band importance function (I), the speech level distortion factor (L), and the band audibility factor (K) over all frequency bands. Theoretically, the SII quantifies the ability to distinguish a source signal from background noises for each octave band essential for speech recognition. Scores fall in the range between 0 and 1, where 1 indicates perfect signal reproduction and 0 indicates complete dissimilarity.

$$S = \sum I * L * K$$

Modifications to the SII calculation established in the standard were primarily in the signal to noise ratio (SNR) and hearing threshold. The SNR values varied greatly from those expected for standard calculations; the electromechanical testing had an SNR value of about 50 dB while human testing SNR values were between -12 and 0 dB. The spectrum level of speech (or spectrum level of the signal for electromechanical tests) was a constant for the electromechanical tests but varied for each frequency band in the human tests. The threshold of hearing value also changed between humans and electromechanical testing; the absolute hearing threshold for humans is approximately 0 dB, whereas in the electromechanical tests the value depends on the theoretical lower limit of the dynamic range of the microphones (18 dB).

An example calculation of the modified SII is shown in Figure 15 below. Data is presented for those devices used in the hSQ testing. As expected, the open ears have the highest value since the acoustic field is unperturbed. The next highest devices were active devices, which exhibit a higher signal to noise ratio leading to a higher signal quality at the levels tested. The passive devices produced the lowest signal quality due to the lower signal to noise and greater alteration of the frequency spectrum exhibited through insertion loss measurements.

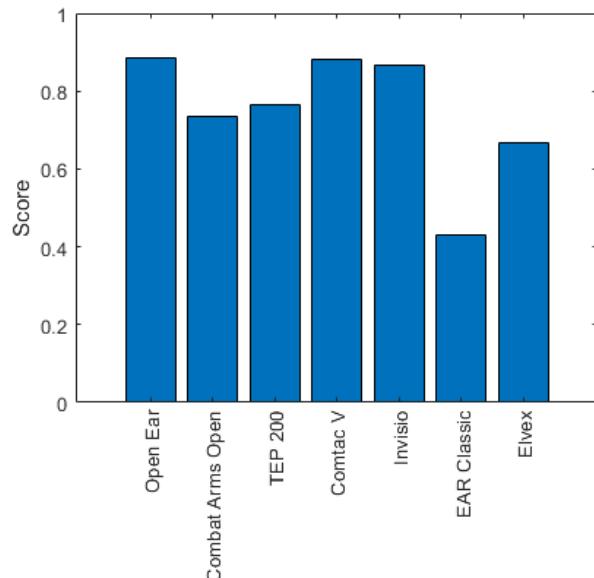


Figure 15. The eSQ represented by modified SII calculations for the HPDs evaluated with human subjects.

3.2.5.2. Localization (eSL)

The laboratory setup and procedure has been described in detail in previous reports; a summary is presented here with relevant changes.

An ANSI S12.42-compliant acoustic test fixture (ATF) was exposed to a 1-second logarithmically swept sine wave stimulus spanning 250 Hz to 16 kHz. Twenty stimuli were presented at a stimulus level of 90dB for passive HPDs and 70dB for active HPDs to avoid activating the signal-limiting electronics present in the active systems. The ATF was mounted within a virtual hemispheric speaker array to allow for stimulus presentation from a controlled grid of azimuth and elevation positions. Responses were recorded for each HPD included in the human subject testing (Combat Arms 4.1 open mode, Combat Arms 4.1 closed mode, EAR Classic, Elvex Quattro, 3M TEP-200, 3M Comtac V, and Invisio v50) as well as the open ear (unoccluded) condition.

Processing techniques were developed to reduce the influence of the measurement space on the recorded signals. Data at each source location for each device was converted to an impulse response waveform and truncated to remove any room reflections, then zero-padded back to the original signal length. The 20 impulse response iterations were averaged in the time domain to increase the measurement SNR.

The four types of electromechanical error terms described in the following sections were then calculated from the head-related impulse response (HRIR) data: directional transfer function (DTF) distortion error terms (eSL_{DTF}), interaural time delay distortion error terms (eSL_{ITD}),

interaural level difference distortion error terms (eSL_{ILD}), and insertion loss error terms (eSL_{IL}). These error terms will be discussed in the following subsections.

3.2.5.2.1. eSL_{DTF} Error Terms

The overall DTF error term is composed of three sub-error terms: collapse-in-elevation (CE) errors, collapse-in-azimuth (CA), and front-back confusion with collapse in elevation (FBCE) errors. Each type of error term is derived from an observed type of error in human subject data. At each unique elevation and azimuth stimulus point k , the head-related transfer functions will be referred to as H^O for the open condition and H^C for each occluded condition (each HPD) and have been converted to a dB scale. The term H^X will refer to an HRTF function without specifying open or occluded.

CE DTF terms for either the open or occluded condition at a given stimulus point (az, el) are given by:

$$DTF_{az,el}^{CE,X} = H_{az,el}^X - H_{az,0}^X$$

CA DTF error terms are given by:

$$DTF_{az,el}^{CA,X} = \begin{cases} H_{az,el}^X - H_{90,el}^X & \text{for } 0 < az < 180 \\ H_{az,el}^X - H_{270,el}^X & \text{for } 180 < az < 360 \end{cases}$$

FBCE DTF error terms are given by:

$$DTF_{az,el}^{FBCE,X} = \begin{cases} H_{az,el}^X - H_{180-az,0}^X & \text{for } 0 < az < 180 \\ H_{az,el}^X - H_{540-az,0}^X & \text{for } 180 < az < 360 \end{cases}$$

The DTF distortion is defined as the difference between the open DTF and the occluded DTF ($DTF_{az,el}^{\{error type\},O} - DTF_{az,el}^{\{error type\},C}$) at each stimulus point and referred to as $\Delta DTF_{az,el}^{\{error type\}}$.

The error value for an error type and stimulus point is the RMS value of the DTF distortion over the bandwidth 2kHz – 16 kHz. Location-dependent DTF error scores for all three error types are shown below in Figure 16 and Figure 17 for the EAR Classic and Comtac devices, with colors representing error normalized to 10dB. Scores are averaged over the left and right ears.

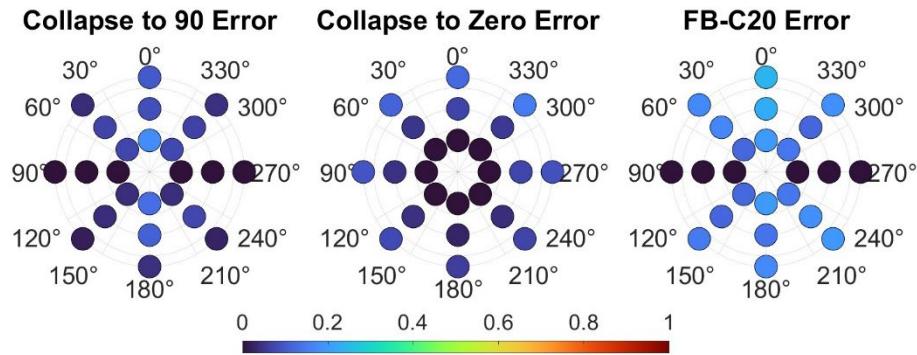


Figure 16. EAR Classic DTF error terms with normalized colors representing the error magnitude at each stimulus point. The innermost circle is 0 degree elevation and the outermost circle is 60 degree elevation.

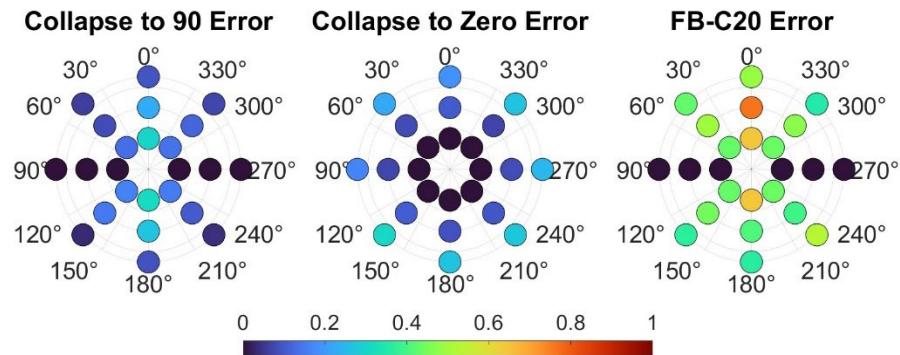


Figure 17. Comtac V DTF error terms with normalized colors representing the error magnitude at each stimulus point. The innermost circle is 0 degree elevation and the outermost circle is 60 degree elevation.

The total error term for a specific error type (CE, CA, FBCE) is the mean over all stimulus points in dB and is referred to as $\Delta DTF_{mean}^{\{error\ type\}}$. Finally, the combined DTF error term eSL_{DTF} is defined as

$$eSL_{DTF} = \frac{1}{1 + \Delta DTF_{mean}^{CE} + \Delta DTF_{mean}^{CA} + \Delta DTF_{mean}^{FBCE}}$$

The resulting values of eSL_{DTF} trend to zero for large DTF disruptions and trend to one for a DTF with no distortion consistent with the open ear.

3.2.5.2.2. eSL_{ITD} Error Terms

At each stimulus location, HRIRs are bandlimited to 500-1500 Hz, and the autocorrelation function of the left and right ear HRIRs is used to find the time delay between the left and right channels, referred to as the interaural time delay (ITD). ITD is calculated at each stimulus location for the open ear condition ($ITD_{az,el}^O$) and each occluded condition ($ITD_{az,el}^C$). A comparison of open and

occluded ITD curves as a function of stimulus azimuth and elevation are shown in Figure 18 and Figure 19.

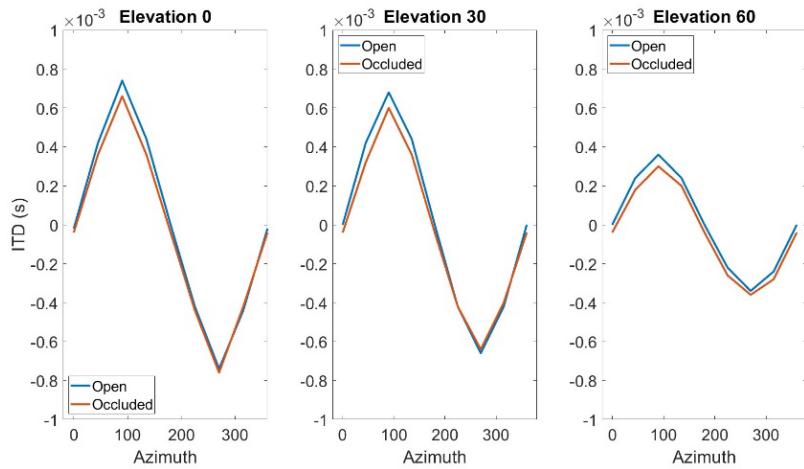


Figure 18. EAR Classic ITD curves as a function of azimuth and elevation, compared with open ear ITD curves.

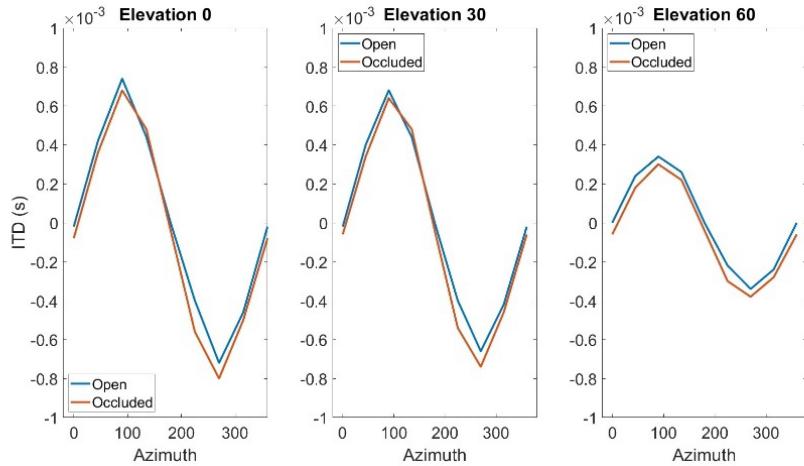


Figure 19. Comtac ITD curves as a function of azimuth and elevation, compared with open ear ITD curves.

The ITD distortion curve for a given HPD is calculated as $\Delta ITD_{az,el} = ITD_{az,el}^O - ITD_{az,el}^C$. The error term is then calculated as

$$eSL_{ITD} = 1 - \frac{\sum |\Delta ITD_{az,el}|}{\sum |ITD_{az,el}^O|}.$$

The eSL_{ITD} spans from zero for a high time difference and one for a zero time difference relative to the open ear.

3.2.5.2.3. eSL_{ILD} Error Terms

At each stimulus location, HRTFs are calculated for the open ear case $H_{az,el}^O$ and each occluded case $H_{az,el}^C$ over the bandwidth 1500 Hz – 16 kHz. In each case, there is a left ear and right ear HRTF, e.g. $H_{az,el}^{OL}$ and $H_{az,el}^{OR}$. The ILD for each case is defined as:

$$ILD_{az,el}^{\{condition\}} = H_{az,el}^{\{condition\}L} - H_{az,el}^{\{condition\}R}.$$

The single ILD value at a given source location is then calculated as the median value over all frequency points, referred to as $\overline{ILD}_{az,el}^{\{condition\}}$. Median ILD curves as a function of source azimuth and elevation are shown in Figure 20 and Figure 21.

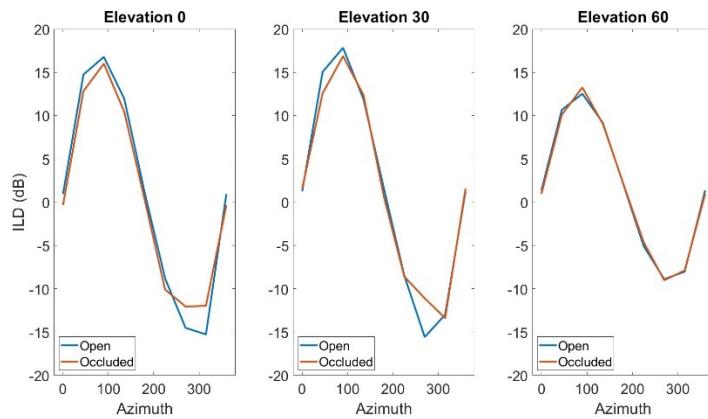


Figure 20. EAR Classic ILD curves as a function of azimuth and elevation, compared with open ear ILD curves.

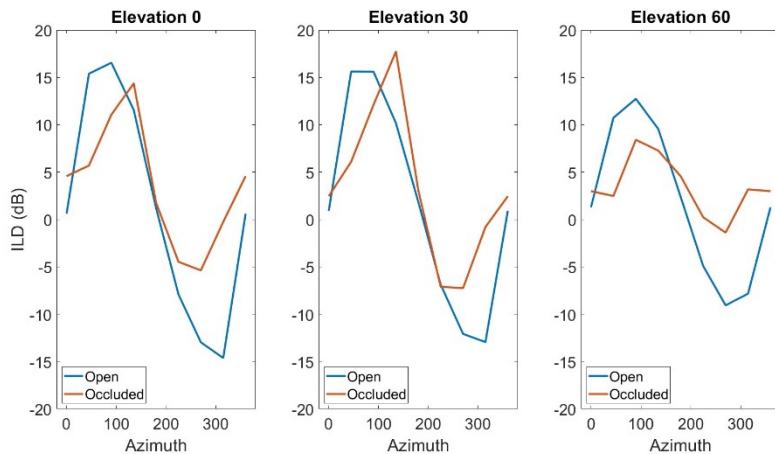


Figure 21. Comtac ILD curves as a function of azimuth and elevation, compared with open ear ILD curves.

The ILD distortion curve for a given HPD is then calculated as $\Delta ILD_{az,el} = \overline{ILD}_{az,el}^O - \overline{ILD}_{az,el}^C$. The error term is then calculated as

$$eSL_{ILD} = 1 - \frac{\sum |\Delta ILD_{az,el}|}{\sum |ILD_{az,el}^0|}.$$

The eSL_{ILD} spans from zero for a high level difference and one for a zero level difference relative to the open ear.

3.2.5.2.4. eSL_{IL} Error Terms

At each stimulus location, HRTFs are calculated for the open condition $H_{az,el}^0$ and occluded conditions $H_{az,el}^C$ over the full measurement bandwidth 250 Hz – 16 kHz and is converted to a dB scale. The insertion loss at each measurement location is given by $IL_{az,el} = H_{az,el}^0 - H_{az,el}^C$. The averaged insertion loss as a function of frequency IL is calculated by averaging over all stimulus locations. Insertion loss is bounded by the bone conduction limit, at which point acoustic energy is dominated by bone conducted sound rather than airborne sound energy. The frequency-dependent bone conduction limit is referred to by BCL . Example insertion loss curves, with the bone conduction limit for reference, are shown in Figure 22.

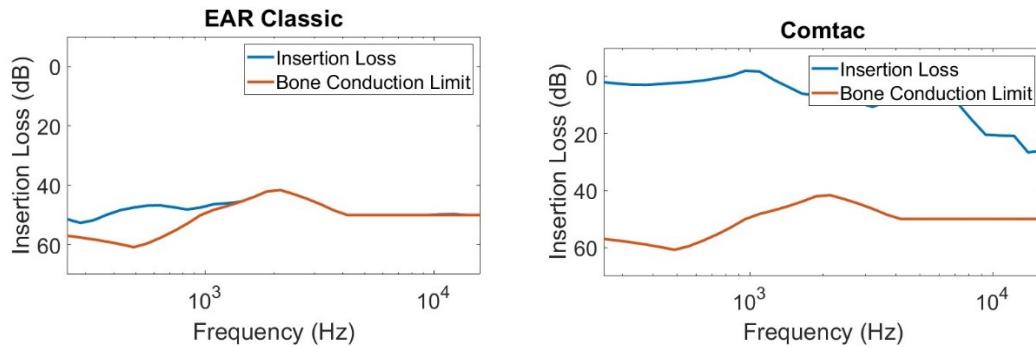


Figure 22. Insertion loss curves (with bone conduction limit for reference) of EAR Classic and Comtac V.

The eSL_{IL} error term is then calculated as

$$eSL_{IL} = \text{mean} \left(\frac{BCL - IL}{BCL} \right).$$

The eSL_{IL} term trends to zero for insertion loss approaching the bone conduction limit and trends to one for insertion loss approaching zero.

3.2.5.2.5. Calculation of eSL terms

For each device, the complete error term is then calculated as:

$$eSL = eSL_{DTF} \times eSL_{ITD} \times eSL_{ILD} \times eSL_{IL}.$$

This metric gives scores in the range of zero to one, where zero refers to a device that significantly distorts spatial cues and one refers to a device that preserves a range of spatial cues. Based on the current and preliminary dataset, eSL scores for all devices are given in Table 3.

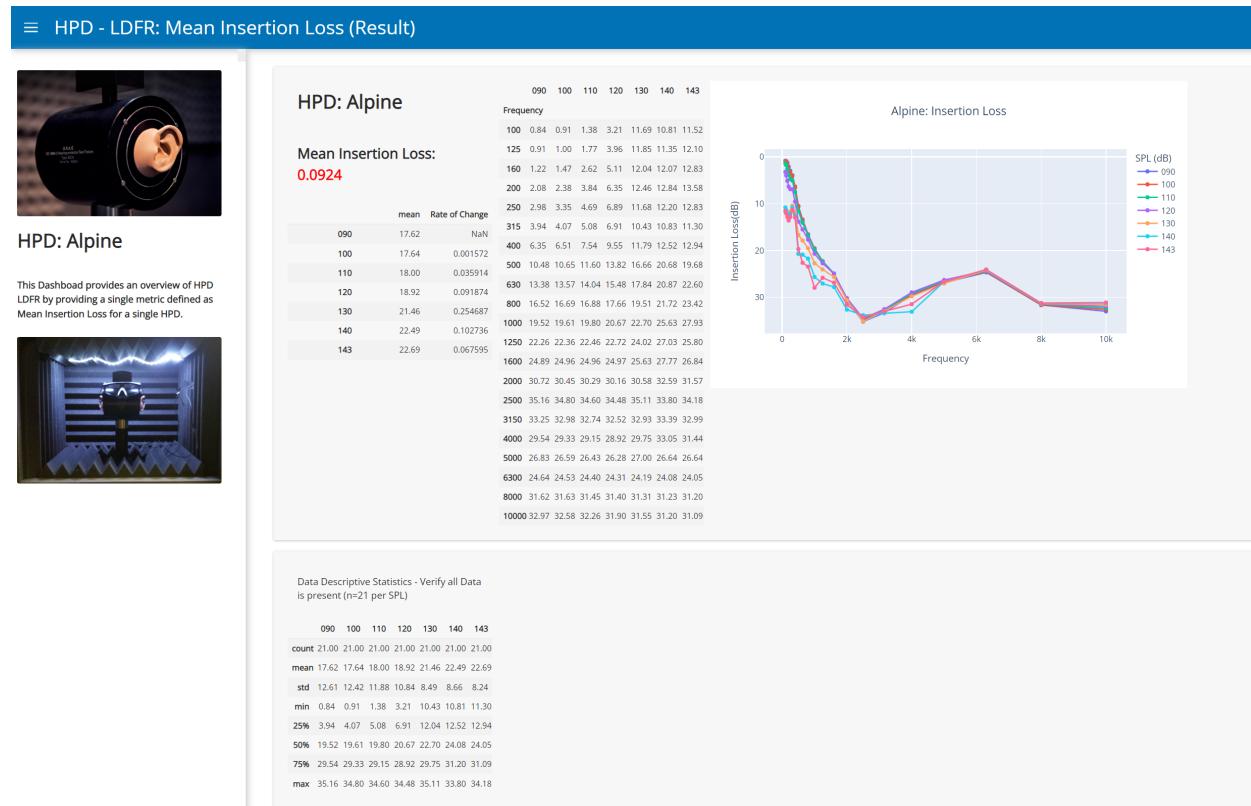
Table 3. eSL Scores for tested devices based on current calculation rubric.

Device	EAR Classic	Combat Arms 4.1 Open Mode	Elvex Quattro	TEP-200	Comtac V	Invisio V50
eSL	0.0152	0.1298	0.0302	0.0905	0.0772	0.0111

3.2.5.3. Level Dependence (eLD)

3.2.5.3.1. Data Dashboard Prototype

The results for eLD were refined to a single value metric of level-dependent insertion loss. The analyzed data is then incorporated into a detailed dashboard representing the data analysis as shown in Figure 23. This dashboard is being used for internal data triage and analysis and is not a deliverable of this program. Further refinement of this dashboard would enable data to be more broadly shared throughout the community and will be a target of follow-on funding proposals.

**Figure 23. eLD data analysis dashboard.**

3.2.5.3.2. Data Quality Analysis

A repeatability analysis was conducted on the collected data from the LDFR test results. To determine the degree of repeatability for each device, multiple iterations were run, spanning SPLs of 70dB to 143dB. Each SPL tested had a corresponding frequency range at max spanning 100 Hz to 10 kHz. The data from five iterations of occluded ear with HPD placement was analyzed, comparing the presented frequency to the recorded noise level. The results from each iteration were averaged. A root mean square dissimilarity score was then calculated, comparing

each individual iteration to the average of the five iterations. If any of the iterations had a dissimilarity score greater or equal to five, an additional five iterations were conducted.

Data from all passive and active HPDs were evaluated following these guidelines to determine individual repeatability of each device as well as trends in device type. The dissimilarity scores of each iteration for a single HPD at a set SPL level were averaged to determine the average dissimilarity score (ADS). The ADS were then compared across SPL levels both to the singular device and collection of HPDs as presented in Table 4.

Across all sound presentation levels, the average ADS were smaller for the active devices than the passive devices, and the active devices had consistently lower standard deviations for the ADS than the passive devices (Figure 24). From this, it can be concluded that the active devices have more predictable and consistent repeatability between iterations regardless of sound presentation level when compared to their passive device counterparts.

For active devices (Figure 25), trends were observed with respect to source level. The Clarus XPR, IEEP, Invisio, and Tactical Pros had a downward slope in the change in the ADS values as SPL increased. Invisio has the most notable change with the highest ADS out of active devices from 70 dB – 90 dB to one of the smallest at the higher SPLs. Tactical Pros were the most consistently low in their ADS across all SPLs, only failing to be the lowest SPL out of active devices at 130 dB – 143 dB. Comtac IV, Comtac V, TEP100s, and TEP200s had an upward slope with downturn creating a peak ADS around 130 dB – 140 dB. TEP 200s were consistently the highest ADS at all SPLs greater than or equal to 110 dB. A unique pattern specific to these devices is the similar shape between similar models. Comtac IV and Comtac V had similar results in both magnitude and direction of change. The same is true for TEP100s and TEP200s; however, TEP200s had a higher ADS than TEP100s consistently.

Table 4. Average dissimilarity scores at various sound presentation levels for active and passive hearing protection devices. *bold indicates an outlier was removed to obtain the respective ADS.

Device	ADS 70dB	ADS 80dB	ADS 90dB	ADS 100dB	ADS 110dB	ADS 120dB	ADS 130dB	ADS 140dB	ADS 143dB
Airsoft	X	X	17.7236	17.5169	17.4738	16.3158	14.2334	12.3284	6.9399
Alpine	X	X	2.2093	2.1908	2.1421	1.3126	0.97231	0.40464	0.14674
CAE Closed	11.225	12.032	12.5279	13.1088	13.7044	13.8169	13.4032	9.9267	6.2576
CAE Open	1.8594	1.7788	1.7206	1.7087	1.6563	0.92786	0.16433	0.51047	0.3092
Ear Classic	7.5164	8.3667	10.3314	11.2277	11.6127	10.9717	7.3992	5.8409	2.0608
Elvex Quattro	2.6718	2.4281	2.44	2.4261	2.4271	2.2687	1.7756	1.0611	0.42038
ER-15	2.1798	1.9344	1.8547	1.8367	1.4809	0.21616	0.058985	0.18987	0.046342
Ety Plug	X	X	18.3536	18.2433	17.958	15.218	13.2349	12.7409	6.1196
Honeywells	X	X	13.3986	14.8709	14.9881	13.139	10.3751	5.573	2.8682
LaserLite	X	X	8.2185	7.0747	7.7714	7.0841	6.8666	9.5956	3.8261
Moldex Closed	4.5843	4.72	4.7411	4.7625	4.719	2.5513	0.90336	1.6741	0.64512
Moldex Open	4.0874	4.0602	4.0635	4.0254	3.9914	2.2291	0.418	1.2642	0.49211
Shooters	X	X	11.0291	10.6957	8.7465	9.2962	16.0252	8.9645	5.5891
Surefires	X	X	8.5652	8.9271	9.0368	8.372	7.3286	6.3602	3.4243
Vibes	X	X	3.6551	3.6468	3.6532	3.4812	3.3946	2.2978	0.89795
Westone16	X	X	8.1197	9.7412	7.9862	5.1127	10.21	7.5344	2.7208
Westone20	X	X	11.3881	11.3303	9.8847	5.3145	7.7513	6.5448	3.242
Westone25	X	X	6.4455	6.4857	6.3393	5.6843	8.096	7.7742	4.1369
Clarus XPR	1.8171	1.7336	1.5339	1.4499	1.2128	0.9379	1.2494	0.26097	0.068329
Comtac IV	1.6122	0.96553	1.2056	1.6265	2.3215	2.5659	2.8552	2.4608	0.6228
Comtac V	1.1473	1.0422	1.2004	1.4396	1.3476	1.7092	2.2736	3.264	2.4281
IEEP	3.3634	3.3719	2.5929	2.4334	2.9082	2.0004	3.396	1.6897	1.3007
Invisio	7.848	8.0592	7.0622	5.5307	4.0252	1.7883	0.51273	1.5082	0.84622
Tactical Pros	0.92474	0.73345	0.55207	0.55763	0.62056	0.7493	0.66768	0.60608	0.20163
TEP100s	1.5248	1.5367	1.5539	3.5447	3.7719	4.3677	4.8344	1.0472	0.70329
TEP200s	2.5051	2.7486	3.8839	6.1177	6.2984	6.3431	9.9462	6.4371	4.8134

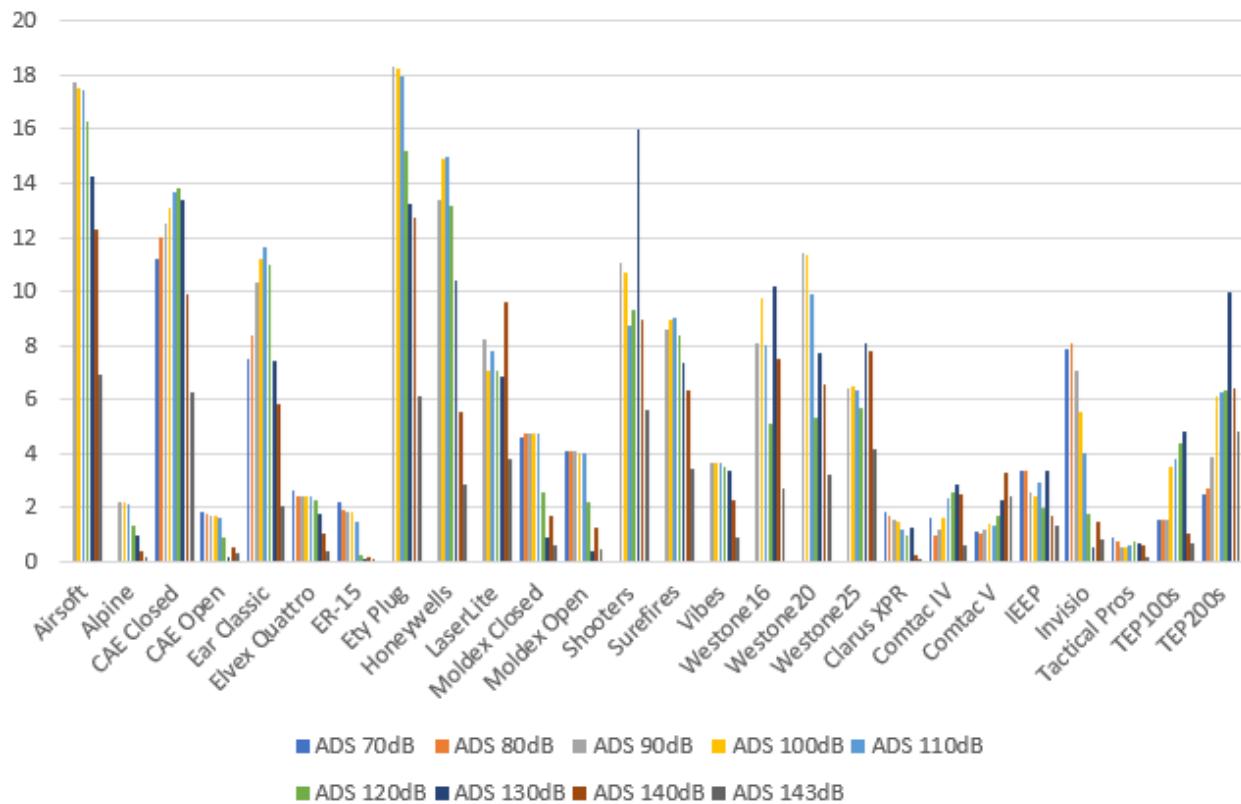


Figure 24. Average dissimilarity scores at various sound presentation levels for active and passive hearing protection devices.

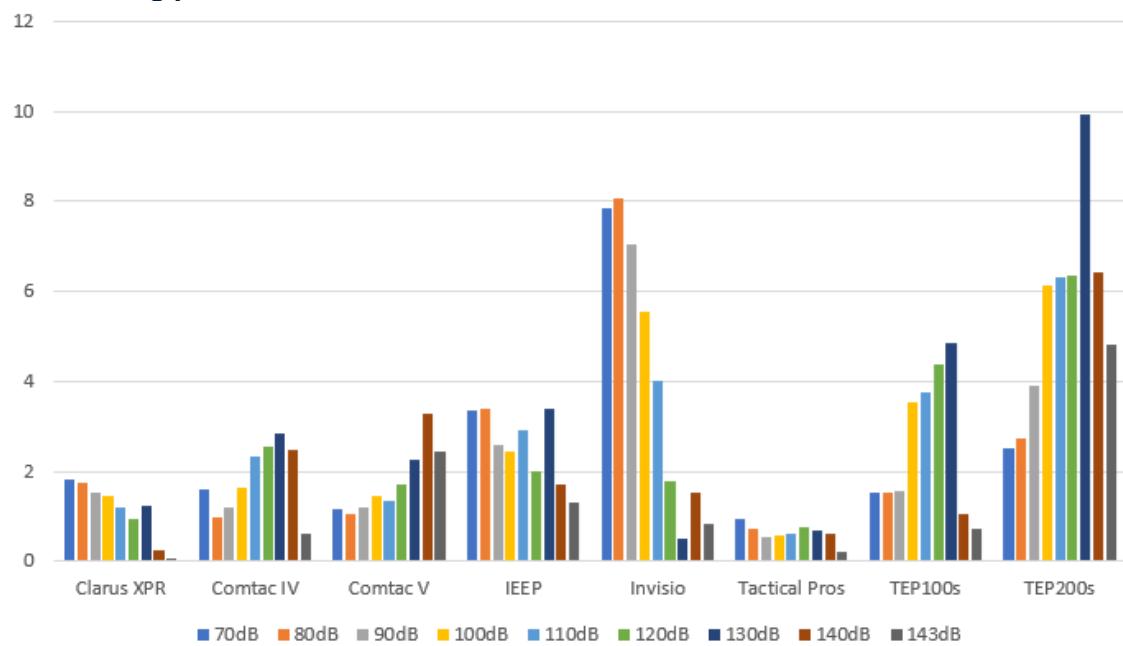


Figure 25. Average dissimilarity score across various sound presentation levels for active devices.

The passive device (Figure 26) tests are split into two categories, those tested on a full SPL range from 70 dB – 143 dB, and those tested on a limited range from 90 dB – 143 dB. The devices that were tested on the full range had similar behavior to each other CAE Open, ER-15, Elvex Quattro, Moldex Open, and Moldex Closed had similar trends with minimal change in ADS at SPL levels from 70 dB – 110 dB. At SPLs greater than or equal to 120dB, there was a sharp decreased in the determined ADS suggesting that these devices are more repeatable and consistent in the data collected at higher SPL levels. Ear Classic and Moldex Closed had the highest ADS out of the passive devices tested on the full SPL range. Both devices exhibit a local maximum at 110 dB (Ear Classic) and 120dB (CAE Closed). Similarly, to the other devices tested at these levels, 130 dB – 143 dB had a steady decline in the associated ADS. Ear Classic and CAE Closed have the similar increased repeatability at the higher SPL levels like the other devices; however, their respective repeatability is much lower than those of the other passive devices tested on the full SPL range.

A selection of passive devices was tested at SPL levels ranging from 90 dB – 143 dB (Figure 27). These devices possess majority of the high ADS and low corresponding repeatability. Alpine and Vibes present the lowest ADS out of this category of HPDs, resulting in their performance being closer to that of some of the active devices and the other passive devices tested on a full SPL range. Both have a downturn trend as when the SPL increases, the ADS decreases. Airsoft and Ety Plug have a similar shape and trend between SPL and ADS, but at a much higher magnitude for their respective ADSs. Honeywells and Surefires have more of a bell shape, both peaking at 110 dB with their end behavior closely mimicking that of the Airsoft and Ety Plug. The Westone devices all have similar behavior. Westone 16 and Westone 20 are the most similar with high ADS values for SPLs 90dB-110dB and 130dB-140dB with large decreases in SPLs at 120 dB and 143 dB. Westone 25 had similar behavior with the lowest ADS at 120 dB and 143 dB; however, its respective magnitudes were smaller and closer together in value. The Shooters and LaserLite both experienced large peaks later in the SPL levels- 140 dB for LaserLite and 130 dB for Shooters. Their behavior otherwise was like that of the Ety Plug and Airsoft.

A notable trend between all the devices, active and passive, is the decrease in ADS at higher SPL levels. Levels of 120 dB – 143 dB were tested at limited frequency ranges due to the system limitations of the speakers and noise floor levels. A source level of 120 dB was the least restricted in frequency being tested from 100 Hz to 6300 Hz. A source level of 143 dB was the most limited in frequency being tested from 630 Hz to 1000 Hz. The increased repeatability and decrease in ADS at higher SPL levels could be contributed to the device design and performance, but it is important to note the possibility that the increase in repeatability and decrease in ADS is due to the decrease in frequency levels. This limitation in frequency levels reduces the amount of recorded SPL noise levels that need to be like cause a low ADS score. This would explain the general trend behind all the devices having a decrease at most if not all the higher end SPL levels. Some devices demonstrated larger decreased in ADS at these levels which can be attributed to the device design and performance.

The average of the ADS at each SPL were found and compared to the other average ADSs of each device (Figure 28). Based on the mean ADS, Tactical Pros are the most repeatable device across all SPL levels. Comparatively, Airsoft is the least repeatable device across all SPL levels. Most of the active devices are concentrated near the small mean ADS end of the spectrum with the passive devices having larger mean ADSs. This mean ADS analysis supports the data seen at each SPL level where active devices are more consistent and repeatable than the passive devices.

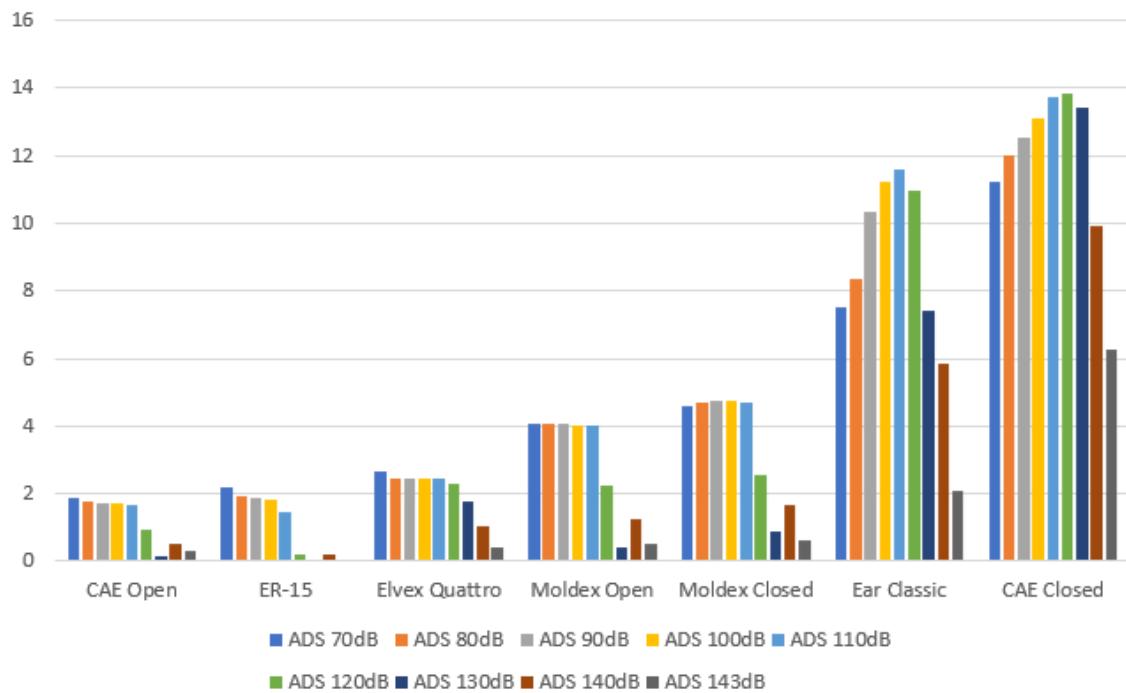


Figure 26. Average dissimilarity score across various sound presentation levels for passive devices.

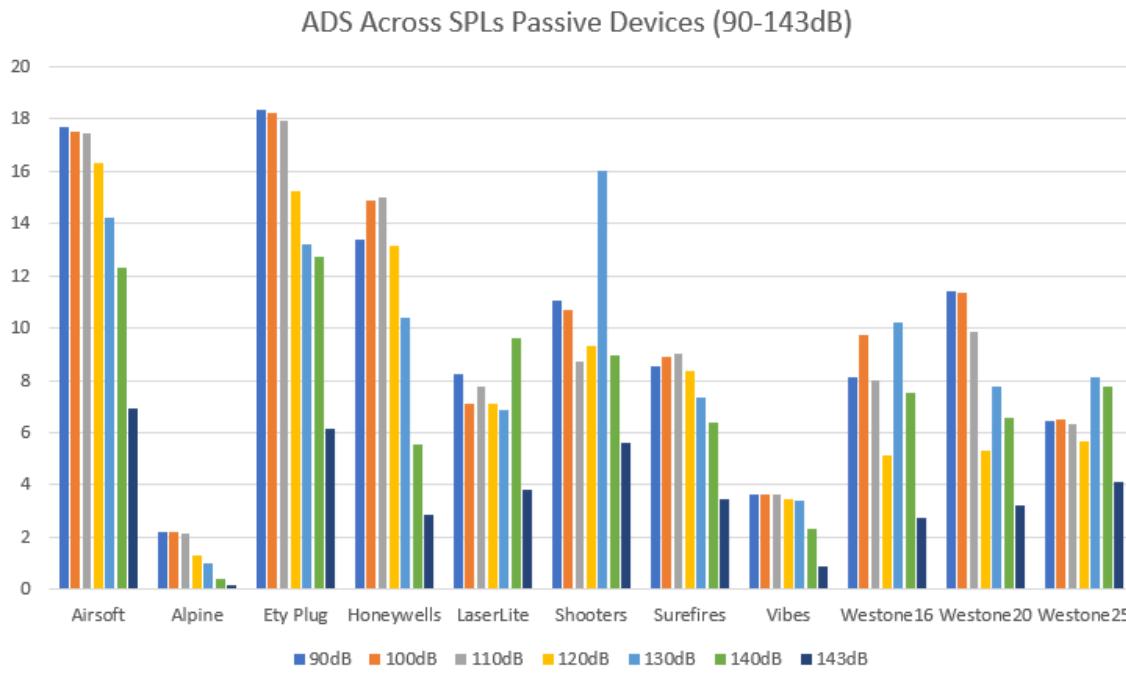


Figure 27. Average dissimilarity score across limited sound presentation levels for active devices.

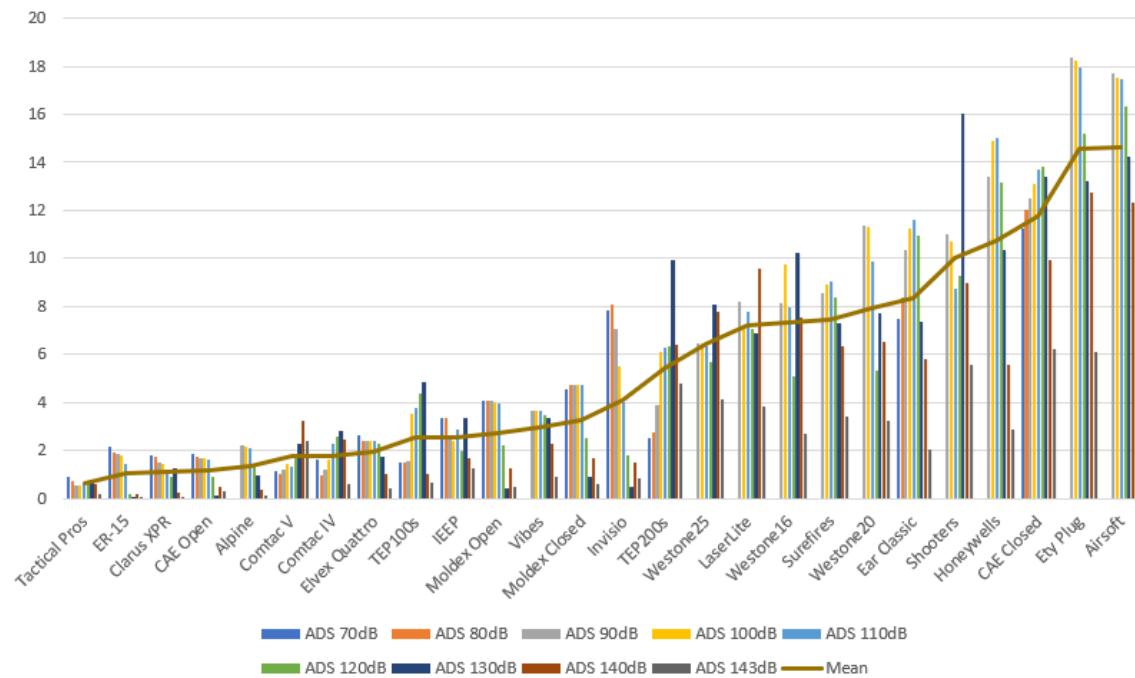


Figure 28. Average dissimilarity score and mean average dissimilarity for each hearing protection device at various sound presentation levels.

3.2.5.4. Self-Noise (eSN)

Each active HPD was tested both in the occluded-on and occluded-off status, as seen in Figure 29. When looking at the occluded-off, all of the devices except for the Clarus XPR have initial an SPL below that of the Open Ear data; the Clarus XPR is not much higher than the open ear data and quickly returns to below the Open Ear SPL values. The main trend observed between the occluded-off eSN results is that the occluded-off SPLs fall below the Open Ear SPLs at all frequencies. This is expected, as when an active device in the power-off status occluded the ATF, the noise from the testing environment should be reduced, with no additional noise added to the system from the device itself. That is, in this status the device functions like a passive device. Invisio, IEEP, and Clarus are the exceptions to this with various frequencies' SPLs above the Open Ear's SPLs, with Invisio and IEEP most notably having the largest magnitude in difference between the device and Open Ear SPLs. While this does not follow the expected trend, the standard deviation on the points above the Open Ear data is normally very small, suggesting this increase in measured SPL is due to device performance.

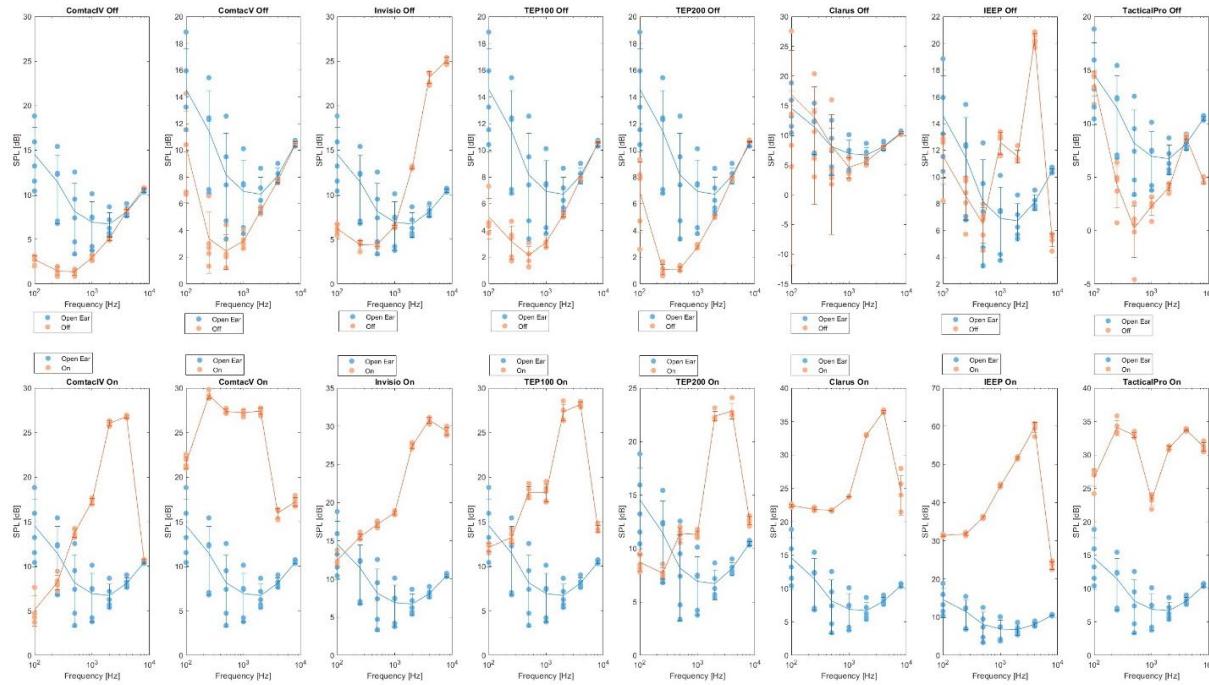


Figure 29. Self Noise for Each Active HPD for Occluded-On and Occluded-Off Status.

The SPLs for the occluded-on data are consistently higher than that of the Open Ear data since the device introduces electronically-generated noise to the system. While still blocking some of the testing environment's noise, measurable increase in noise from the device is observed. Comtac IV, Invisio, and both TEP100 and TEP200 SPL levels at low frequencies are slightly below those of the Open Ear data, suggesting the device's noise is very low at those frequencies.

The standard deviations for the occluded-on devices are much smaller when compared to the occluded-off devices. This suggests the testing noise, while minimal, does fluctuate to a degree which is more noticeable when the device's own noise is not masking some of the environmental noise. These fluctuations in noise can be due to a variety of reasons, such as the HVAC system in the laboratory turning on.

3.2.5.5. Impulse Noise (eIN)

Two shock tubes were used to collect Impulse Noise data- an ANSI shock tube and a short duration shock tube. Both the ANSI shock tube and short duration shock tube subjected the HPDs to three levels of blasts, with low-level blasts with a maximum SPL of 130 dB – 134 dB, mid-level blasts with a maximum SPL of 148 dB – 152 dB, and high-level blasts with a maximum SPL of 166 dB – 170 dB. A total of ten iterations at each blast level were combined to yield a mean peak pressure (Pa) and respective standard deviation for each blast level. These averaged pressures were then compared across all devices.

3.2.5.5.1. ANSI Shock Tube

For the ANSI shock tube there were multiple considerations when determining if data collected for each iteration was accurate. If the pressure probe recorded the peak pressure within the shot level's SPL range, the iteration was successful and counted towards the total of ten iterations

evaluated. If there was too much noise in the recorded signal such that it was not possible to see a visible peak in the recorded pressure, the iteration was removed. Once a total of ten iterations were collected that met the test requirements, they were analyzed collectively for peak pressure.

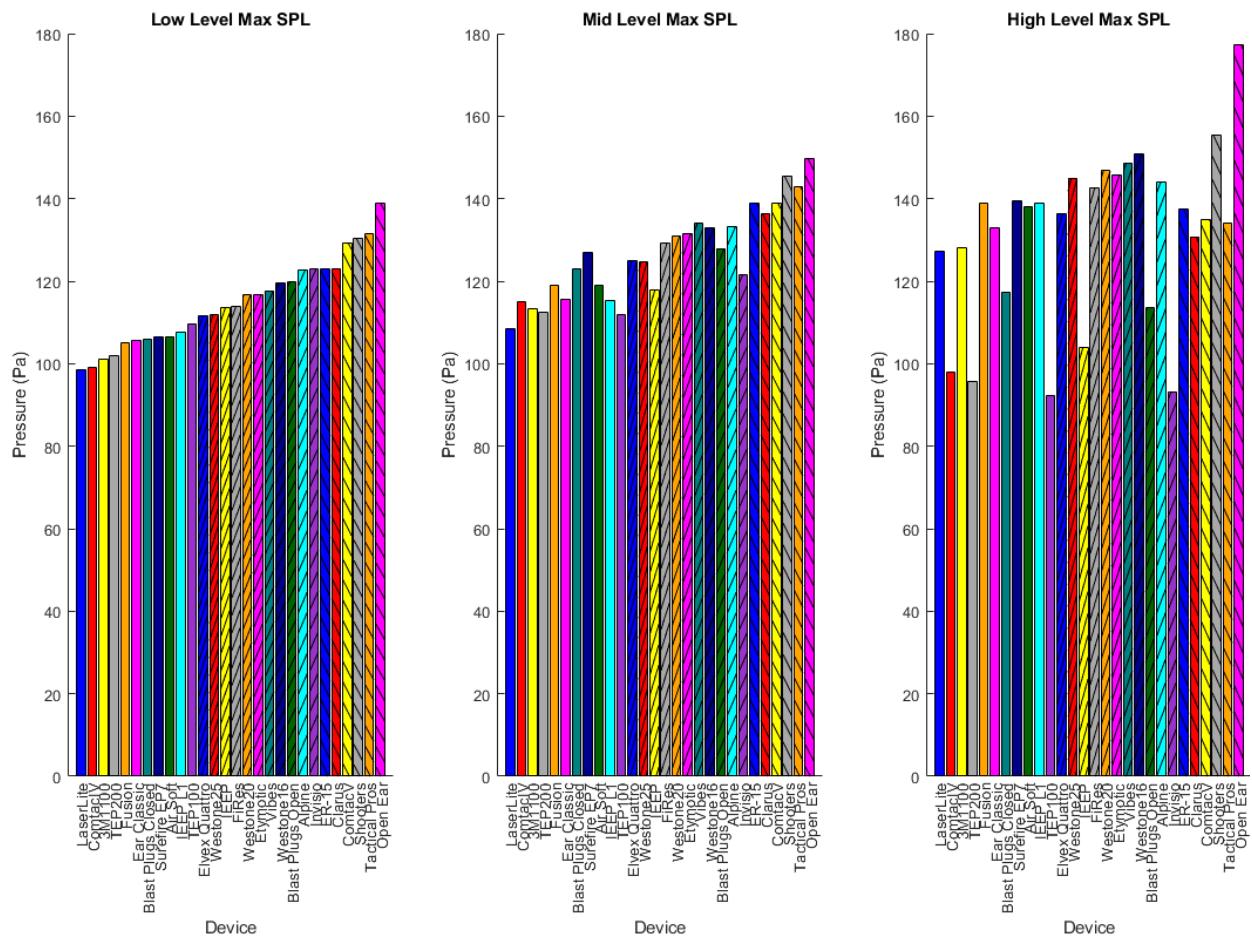


Figure 30. Average peak pressures (Pa) at low, mid, and high-level max SPL using the ANSI shock tube for all HPDs.

The devices were sorted from lowest peak pressure to highest peak pressure according to their low-level data, as seen in Figure 30. There was no distinguishable relationship between active and passive devices and their response to low level blast exposures. Multiple active devices exhibit high average peak pressures, but the active devices are generally spread randomly throughout the ordering of all of the devices.

When the devices were sorted from lowest to highest in response to their average peak pressure experienced from the mid-level blast, a more obvious trend emerged. The active devices are concentrated at low pressures. Passive devices lie predominantly in the middle of the distribution with a mixture of passive and active devices at the higher pressures.

If the devices were sorted from lowest to highest in response to their average peak pressure from the high-level blast, a similar trend occurs to that of the mid-level blast with active devices largely exhibiting low pressures, passive devices exhibiting moderate pressures, and a mixture of passive

and active devices exhibiting high pressures. The passive devices are less concentrated at moderate pressure measurements than at the moderate or low source levels, however.

In general, the active devices tend to experience lower peak pressure in response to a shock tube blast. The variation the ranking the devices from lowest to high peak pressure across the three blast levels can be seen as trends in Figure 31. Between the low level and mid-level, the general linear upward trend of the graph is the same. Some devices increase or decrease disproportionately with respect to the pattern exhibited for low-levels; these variations are mild for moderate exposures and greater for high level exposures.

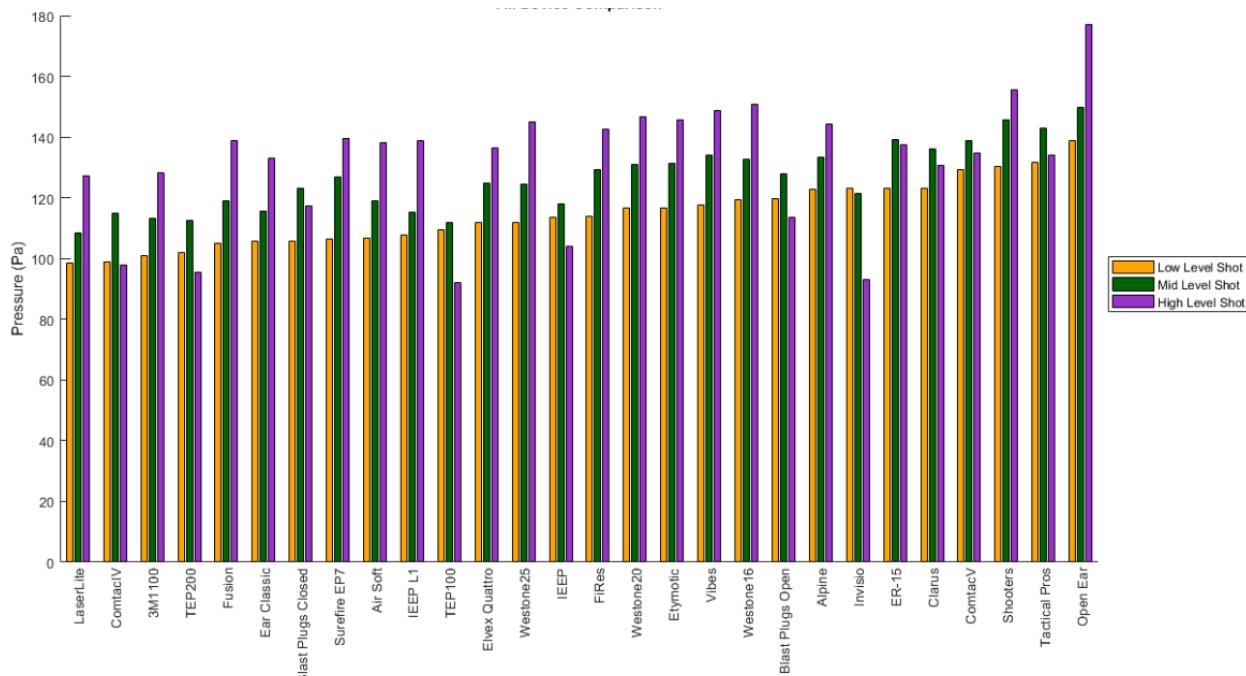


Figure 31. Average peak pressures (dB) at low, mid, and high-level max shots for each HPD using the ANSI shock tube.

Anomalous behavior was noted for multiple devices. As shown in Figure 31, most of the HPDs experience a within-device upward trend between the devices' respective low, mid, and high level shots. There are eleven devices that have experience an increase in pressure from the low to the mid-level shot, but then break the trend seen in majority of the data and have a lower pressure for the high-level shot. There is one device (Invisio) were the experienced pressure decreases from the low level to the high-level shot. It is expected that the devices will experience greater peak pressure as the shock level increases, therefore the devices that experience lower peak pressures at the high or mid-level are outliers.

A possible explanation for the outlier behavior is the equipment used when testing those devices. After conducting the trials and analyzing the data from these devices, it was determined the power modules/preamplifiers failed to operate correctly during testing. Two power modules were used during testing, a GRAS 12AA power module used for the pencil probe and a GRAS 12AQ power module used for the left and right ear simulators. The GRAS 12AA power module was determined to be functional, yielding accurate pressure measurements from the pencil probe for known exposures. The GRAS 12AQ power modules ceased yielding accurate pressure measurements

for some signal gain levels when compared with known exposures. Therefore, the equipment will be repaired and data will be recollected for the inaccurate measurements.

3.2.5.5.2. Short Duration Shock Tube

The same methodology of testing was repeated for the short duration shock tube, though not all devices were tested using the short duration shock tube due to the aforementioned equipment failure. Similarly to the ANSI shock tube data, the devices were sorted from lowest peak pressure to highest peak pressure according to their low-level data, as seen in Figure 32. As for the order of the low-level peak pressure measurements, there is no distinguishable trend for active or passive devices; active and passive devices are both exhibit a range of pressures.

When sorted from lowest to highest in response to average peak pressure experienced from the mid-level blast, there is a obvious trend again emerged. The active devices exhibit the majority of the lowest peak pressures with only three of the active devices exhibiting high peak pressures and passive devices again concentrated at moderate peak pressures.

When sorted from lowest to highest in response to their average peak pressure experiences from the high-level blast, a similar trend emerged to that of the mid-level blast. The distribution is less concentrated in comparison to that of the mid-level results, with passive devices more randomly distributed between the low-pressure active devices. In general,

The variation the ranking the devices from lowest to high peak pressure across the three blast levels can be seen as trends in Figure 33. For the low level and mid level exposures, the linear upward trend is similar. There are some devices that increase or decrease disproportionately to the expected pattern seen at low-level levels, but again these variations are mild. This same behavior of some devices breaking the expected ordering and trend is seen to a more severe degree in the high-level blast graph. There still is a general upward trend, but the higher or lower values are more extreme than those in the mid-level, disrupting the expected trend further.

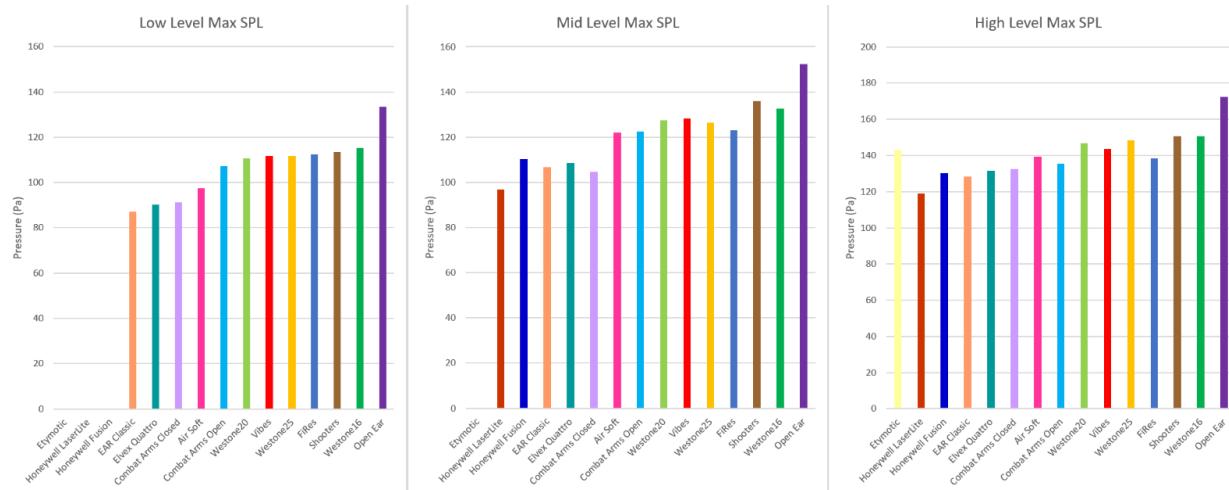


Figure 32. Average peak pressures (dB) at low, mid, and high-level max SPL using the ANSI shock tube for all HPDs

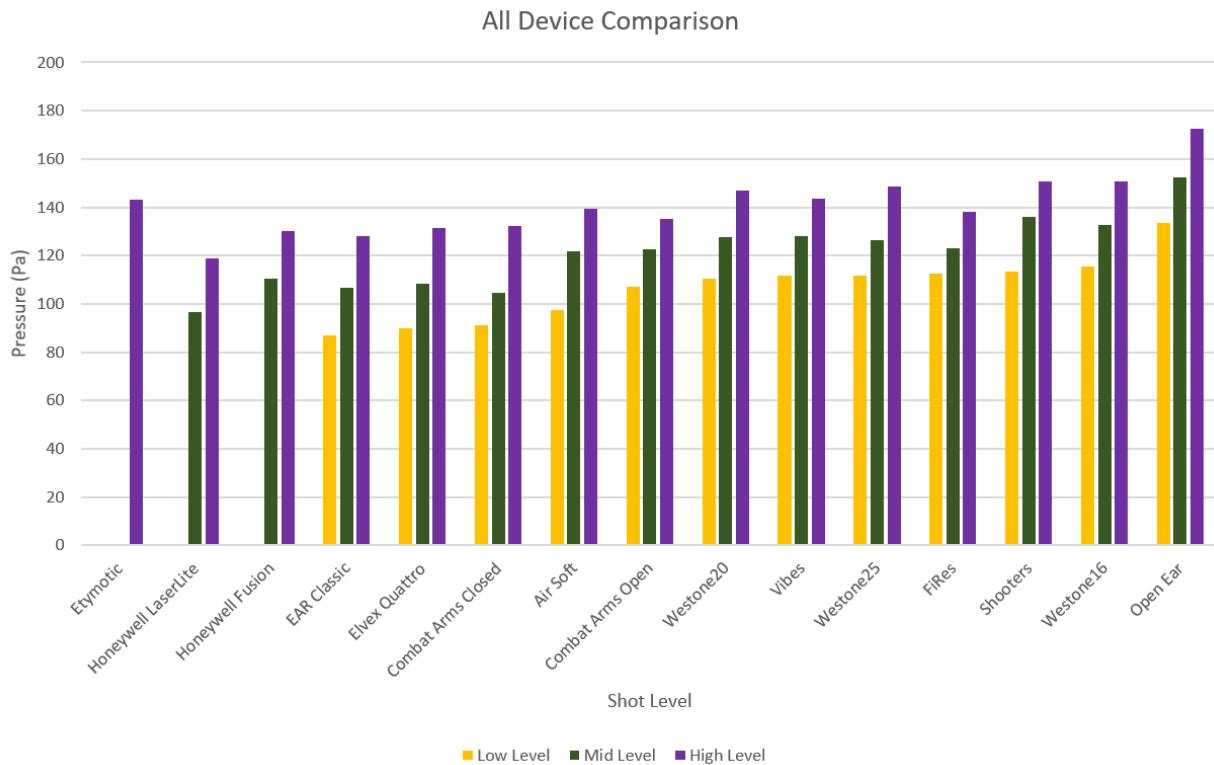


Figure 33. Average peak pressures (dB) at low, mid, and high-level max shots for each HPD using the short duration shock tube.

3.2.5.6. Comparison of Electromechanical Data with Human Performance

Comparisons between human and electromechanical tests were completed for each pair of measurements.

3.2.5.6.1. eSQ/hSQ Signal Quality

For the comparisons of the eSQ and hSQ in Figure 34 and Figure 35, the following data was considered:

- Modified Rhyme Test at -6dB SNR with 60 words/test. Word identification accuracy is normalized to 100% correct with 0 being no correct answers and 1 being 100% correct answers.
- Speech Intelligibility Index based on octave band insertion loss with modifications for source level and signal-to-noise ratio. The index ranges from 0 for unintelligible speech to 1 for perfect speech intelligibility.

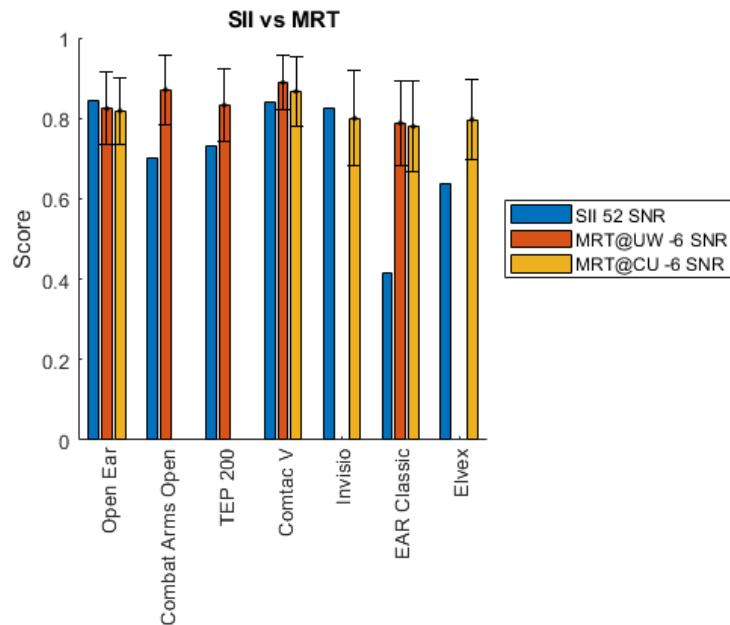


Figure 34. SII calculations for electromechanical data compared to human data for different devices summed over the octave bands.

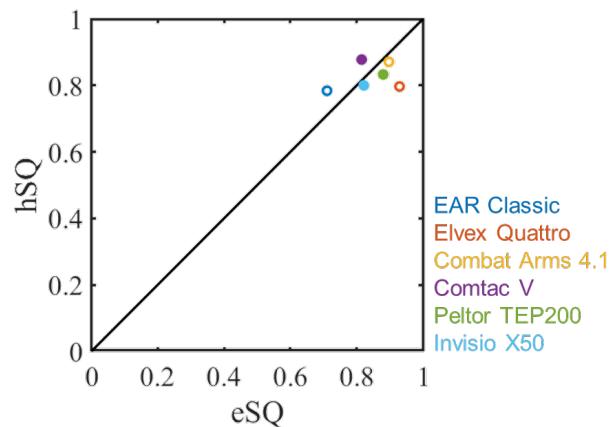


Figure 35. Comparison of eSQ and hSQ.

3.2.5.6.2. eSL/hSL Sound Localization

For the comparisons of the eSL and hSL in Figure 36, the following data was considered:

- hSL metric as derived in Section 3.2.2.
- eSL metric as derived in Section 3.2.5.

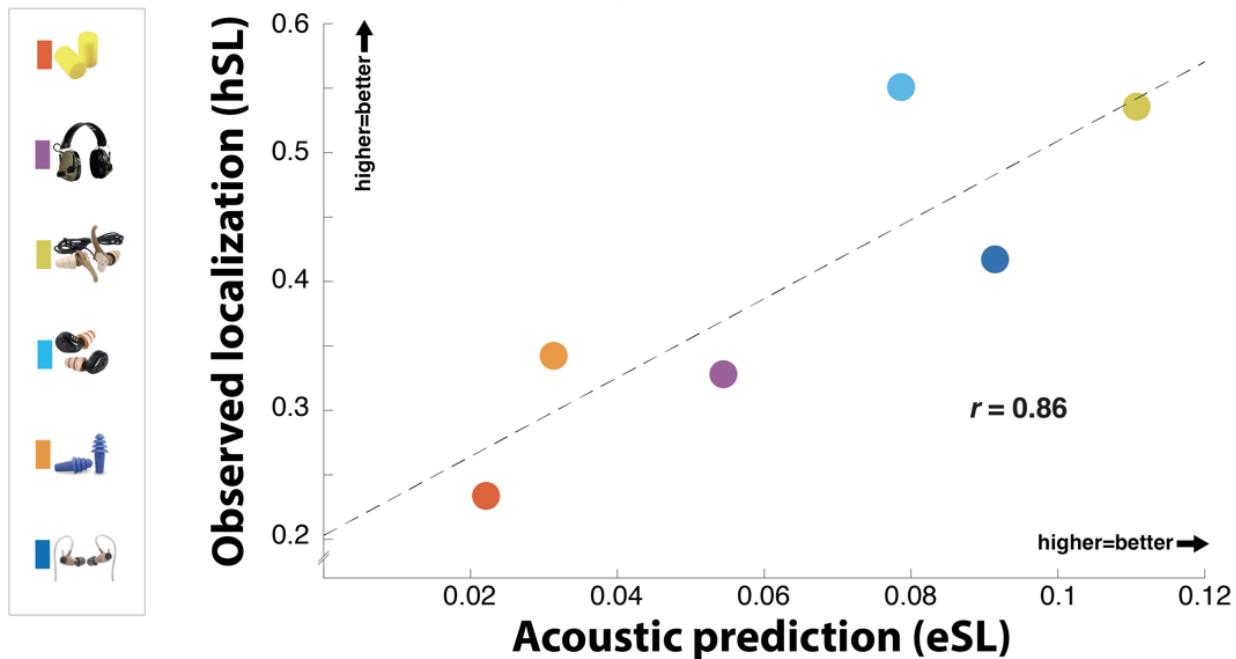


Figure 36. Comparison of eSL and hSL. Correspondence between the two measures has a correlation coefficient of 0.91. The most divergent device is the Elvex Quattro (orange); acoustic data for this device was collected prior to the improvement of the acquisition protocol.

3.2.5.6.3. eLD/hLD Level-Dependence

For the comparisons of the eLD and hLD in Figure 37, the following data was considered:

- Measurements of insertion loss in cadavers at $\frac{1}{4}$ octave band increments collapsed to a single average insertion loss value. The difference between the single values at both 110 and 130 dB exposure levels yields the eLD through:

$$hLD = \sqrt{\frac{IL_{130\text{dB}} - IL_{110\text{dB}}}{130\text{dB} - 110\text{dB}}}$$

- Measurements of insertion loss in an acoustic test fixture in $\frac{1}{3}$ octave bands collapsed to a single average insertion loss value. The rate of change across exposures from 70 dB to 130 dB yielded the eLD through:

$$eLD = \sqrt{\frac{\Delta IL}{\Delta SPL}}$$

In both calculations, a value of 1 indicates protection increases at the same rate as the source level increases and a value of 0 indicates no increase in protection with increased source level.

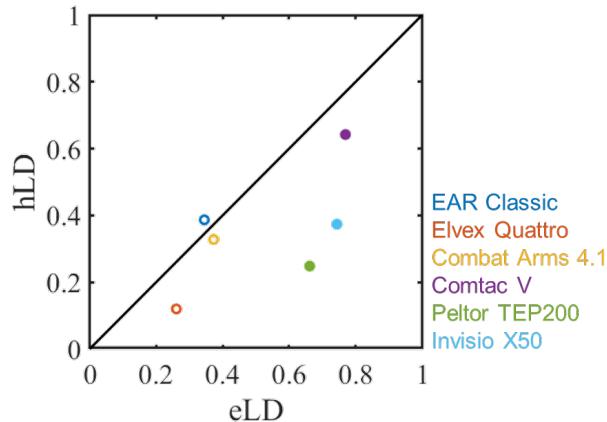


Figure 37. Comparison of eLD and hLD. Passive devices follow the equivalence trend line. Active devices exhibit saturation in eLD at approximately 0.8.

3.2.5.6.4. eSN/hSN Self Noise

For the comparisons of the eSN and hSN in Figure 38, the following data was considered:

- Mean hearing thresholds from REAT under headphones. Values are hearing threshold in dB.
- Mean of octave band noise radiated by the devices into the ears of a test fixture. Values are sound pressure level in dB.

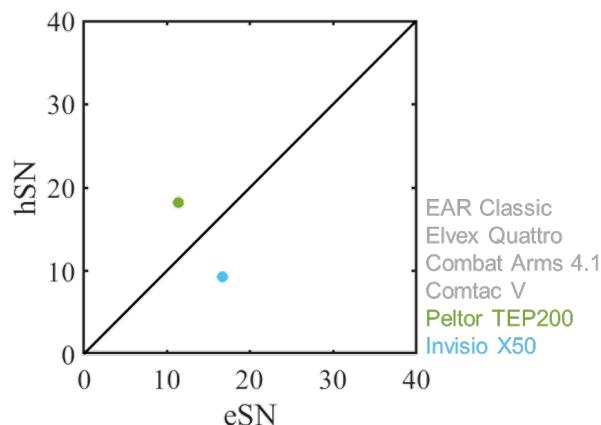


Figure 38. Comparison of eSN and hSN.

3.2.5.6.5. eIN/hIN Impulse Noise

For the comparisons of the eIN and hIN in Figure 39, the following data was considered:

- Impulse peak insertion loss for a 168 dB peak pressure measured as measured in the cochlea of PMHS subjects normalized to the bone conduction limit.

- Impulse peak insertion loss for a 168 dB peak pressure as measured in an acoustic test fixture normalized to the bone conduction limit.

In both cases, a value of 0 indicates no protection from impulsive noise and a value of 1 indicates protection from impulse noise at or above the bone conduction limit.

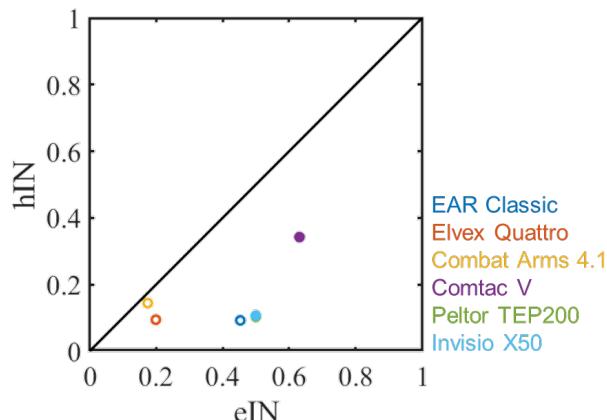


Figure 39. Comparison of eIN and hIN. All devices exhibited lower impulse noise protection in PMHS subjects compared with an acoustic test fixture.

Milestone 6: Conference presentations on comparative HPD performance

- ARO 2023: Poster presentations on eSL vs hSL, and eIN vs hIN.
- ASA Spring 2023: Poster presentation at on eSL vs hSL
- IFBIC 2023: Oral presentation on eIN vs hIN
- MHSRS 2023: Poster presentation at on eSL vs hSL, and an Oral presentation on eIN vs hIN.

3.2.6. Specific Aim 3, Major Task 2: Hearing Protection Selection Tool

We continued to coordinate with the Defense Logistics Agency (DLA) for the transition plan for the Hearing Protection Device Optimization Selection Tool. The Defense Health Agency is taking the lead in coordinating with the DLA J6 for specific requirements to receive and host the HPOT software. Specifications were provided to J6 and a path forward to complete the Risk Management Framework has been proposed to DHA and DLA for consideration.

We submitted a request for an extension of the period of performance to allow for the time necessary to complete coordination with DLA and the appropriate cybersecurity certifications of the HPOT software prior to transition.

Milestone 7: Deliver HPD optimization tool to sponsor

Anticipated at the end of the project period of performance. A one year no-cost extension was obtained to provide more time for this transition.

3.3. What opportunities for training and professional development did the project provide?

Kyndall Tatum was hired as an intern to support this program. Kyndall is pursuing a BS in Chemical and Biomedical Engineering from Carnegie Mellon University and supported the program to primarily through device testing and data analysis efforts. She has returned to school for the Fall 2023 term.

3.4. How were the results disseminated to communities of interest?

Program progress was consistently presented to communities of interest through video and audio teleconferencing, participation in scientific conferences, and peer-reviewed journal publications.

3.5. What do you plan to do during the next reporting period to accomplish the goals and objectives?

The principal period of performance for this program has elapsed, thus no new data collection will be conducted under this contract in the next reporting period. Nevertheless, analysis and dissemination of results will be conducted to support the aims of the program in the extended period of performance.

4. Impact

This research will produce the knowledge products of data and prototype standards suitable for direct incorporation into relevant military and civilian hearing protection standards. These standards will guide the testing of HPDs in a manner that augments current human subject evaluations by providing an electromechanical approximation of human subject performance. By transitioning these methods into standards, time-consuming, repetitive, and expensive human subject testing may be minimized, thereby bringing new technology to the Warfighter much faster than is presently possible. By providing simple, electromechanical testing methods to potential vendors to validate their products, the risk of future product failures and subsequent loss of hearing by warfighters will be significantly reduced.

It is noted, however, that changing existing standards or creating new standards is a time-intensive process. Therefore, the results of this research can be readily transferred directly to the Warfighter through a hearing protection device selection and optimization software program. By allowing the Warfighter to directly access the information from this program, as it relates to the HPDs they have already been issued, the time to develop standards and observe their use is lessened. By enabling mission planners to identify the optimal hearing protection equipment based on their mission requirements, they will reduce risk to Soldiers and increase their lethality on the battlefield through improved situational awareness and communications ability.

Fewer personnel experiencing hearing injury will also reduce the burden on the entire military health care system, from recruitment through Veterans Affairs medical support. The reduction in cost of billions of dollars (Saunders 2009) will ensure more funding is available for critical care situations, further research and development, and/or simple reduction in overall budget. Due to the nature of hearing loss caused by repetitive noise exposure, observing optimal protective strategies will make certain that overall healthcare costs decrease, and quality of life for veterans will increase.

5. Changes/Problems

We requested and received a modification to extend the Period of Performance for 12 months to allow additional time to complete negotiations and discussions with the DLA for final disposition of the HPOT. Additionally, further analysis and dissemination of results will continue.

6. Products

6.1. Abstracts/Presentations

Year 3:

1. Nathaniel Greene, David A. Anderson, Theodore F. Argo. Occluded insertion loss from intracochlear pressure measurements during acoustic shock wave exposure. Association for Research in Otolaryngology, Orlando, FL. February 2023.
2. Andrew D. Brown, Nathaniel Greene, David J. Audet, Aoi A. Hunsaker, Carol A. Sammethyl, Mallory A. Butler, Alexandria Podolski, Jennifer Jerding, David A. Anderson, Gregory T. Rule, Theodore F. Argo; Quantifying impacts of hearing protection devices on sound localization in azimuth and elevation: Refinement of acoustic predictors. Acoustical Society of America, Chicago, IL. May 8-12, 2023.
3. Occluded insertion loss from intracochlear pressure measurements during acoustic shock wave exposure" accepted for a podium presentation at International Forum on Blast Injury, Tokyo, Japan. May 17-19, 2023.
4. Nathaniel Greene, David A. Anderson, Theodore F. Argo. Occluded insertion loss from intracochlear pressure measurements during acoustic shock wave exposure. Military Health Science Research Symposium, Orlando, FL. August 14-17, 2023.
5. Andrew D. Brown, Nathaniel Greene, David J. Audet, Aoi A. Hunsaker, Carol A. Sammethyl, Mallory A. Butler, Alexandria Podolski, Jennifer Jerding, David A. Anderson, Gregory T. Rule, Theodore F. Argo. Quantifying Impacts of Hearing Protection Devices on Sound Localization in Azimuth and Elevation: Toward Predictors of Performance. Military Health Science Research Symposium, Orlando, FL. August 14-17, 2023

Year 2:

1. Argo, T. F., Anderson, D. A., Brown, A. D., Greene, N. T., McCallick, C., Rule, G., & Sammethyl, C. (2022) Validation of Electromechanical Hearing Protection Evaluation Methods. National Hearing Conservation Association Conference.
2. Argo, T. F., Greene, N. T., Brown, A. D., McCallick, C., Sammethyl, C. A., Anderson, D. A., Rule, G. T. Validation of Electromechanical Hearing Protection Evaluation Methods. Military Health System Research Symposium, September 12-15, 2022.
3. Andrew Brown, Nathaniel Greene, David Audet, Caylin McCallick, Carol Sammethyl, David Anderson, Gregory Rule, Theodore Argo. Quantifying impacts of hearing protection devices on sound localization in azimuth and elevation: Toward predictors of performance, Military Health System Research Symposium, September 12-15, 2022
4. Brown, A. D., Greene, N. T., Audet, D. J., McCallick, C., Sammethyl, C. A., Anderson, D. A., Rule, G. T., Argo, T. F. Quantifying impacts of hearing protection devices on sound localization towards identifying predictive patterns. Acoustical Society of America, April 2022.

5. Theodore F. Argo, David A. Anderson, Andrew D. Brown, Nathaniel Greene, Jennifer Jerding, Development of an electromechanical test system and acoustical metrics to predict impacts of hearing protection devices on sound localization. Acoustical Society of America, April 2022.
6. David A. Anderson, Andrew D. Brown, Nathaniel Greene, Theodore F. Argo, Bruno Mary. Development of an in-ear microphone for individualized measurement of hearing protection device output. Acoustical Society of America, April 2022.

Year 1:

1. Argo, T., Greene, N.T., Brown, A., McCallick, C., Sammeth, C., Anderson, D., and Rule, G. (2021) "Validation of Electromechanical Hearing Protection Evaluation Methods". Military Health Science Research Symposium, Orlando, FL. *CANCELLED, abstract accessible through MHSRS Website.
2. Anderson, D., and Argo, T. (2021) "Evaluating the Relationship Between Kurtosis Loss and Spectral Insertion Loss for Musicians' Hearing Protection Devices". 151st International Audio Engineering Society Convention, Las Vegas, NV. October 11-13, 2021.

6.2. Manuscripts/Papers

Year 3:

1. Anderson, D. A., Argo, T. F., & Greene, N. T. (2023). Occluded insertion loss from intracochlear pressure measurements during acoustic shock wave exposure. Hearing research, 428, 108669.

Year 2:

1. Anderson, D. A., & Argo, T. F. (2022). Kurtosis loss as a metric for hearing protection evaluation in impulsive noise environments. JASA express letters, 2(3), 033603.

6.3. Other Products

None to report.

7. Participants and Other Collaborating Organizations

7.1. Participants

Name: **Ted Argo, Ph.D.**

Project Role: Principal Investigator

Researcher Identifier: NA

Nearest person month worked: 10

Contribution to Project: Wrote reports, conducted planning meetings with subcontractors, development/review of test plans and apparatus. Conducted stakeholder meetings, reviewed and submitted protocols.

Name: **Gregory Rule**

Project Role: Program Manager

Researcher Identifier: NA

Nearest person month worked: 5

Contribution to Project: Contributed to and reviewed quarterly report, coordinated and supported planning meetings with subcontractors, development/review of test plans. Conducted stakeholder meetings.

Name: **Dave Anderson**

Project Role: Senior Engineer

Researcher Identifier: NA

Nearest person month worked: 12

Contribution to Project: Experimental design, data analysis, electronics design.

Name: **Nick Brunstad**

Project Role: Staff Scientist

Researcher Identifier: NA

Nearest person month worked: 2

Contribution to Project: Testing operations support for EM testing.

Name: **Santino Cozza**

Project Role: Senior Scientist

Researcher Identifier: NA

Nearest person month worked: 3

Contribution to Project: Standards development.

Name: **Mark Espinoza**

Project Role: Data Scientist

Researcher Identifier: NA

Nearest person month worked: 2

Contribution to Project: Data analysis.

Name: **Summer Graham**

Project Role: Intern

Researcher Identifier: NA

Nearest person month worked: 1

Contribution to Project: Laboratory preparation.

Name: Jennifer Jerding**Project Role:** Senior Engineer**Researcher Identifier** NA**Nearest person month worked:** 3**Contribution to Project:** Review and analysis of existing test standards.**Name: Bruno Mary****Project Role:** Staff Scientist**Researcher Identifier** NA**Nearest person month worked:** 3**Contribution to Project:** Electronics development and testing operations support for EM testing.**Name: Kaleb Morgan****Project Role:** Software Developer**Researcher Identifier** NA**Nearest person month worked:** 2**Contribution to Project:** Development of HPD optimization software tool.**Name: Alexandria Podolski****Project Role:** Laboratory Assistant**Researcher Identifier** NA**Nearest person month worked:** 13**Contribution to Project:** Testing operations support for EM testing.**Name: Kiersten Reeser****Project Role:** Junior Engineer**Researcher Identifier** NA**Nearest person month worked:** 1**Contribution to Project:** Test and evaluation, data analysis.**Name: Luke Runyon****Project Role:** Software Developer**Researcher Identifier** NA**Nearest person month worked:** 1**Contribution to Project:** Development of the HPD selection tool user interface.**Name: Kyndall Tatum****Project Role:** Intern**Researcher Identifier** NA**Nearest person month worked:** 2**Contribution to Project:** Test and evaluation, data analysis.

7.2. Collaborating Organizations

Organization: The University Colorado Anschutz Medical Campus**Principal Investigator:** Prof. Nate Greene, Ph.D.**Organization:** The University of Washington**Principal Investigator:** Prof. Andrew Brown, Ph.D.**Organization:** The University of Minnesota at Duluth**Principal Investigator:** Prof. Dave Anderson, Ph.D.

8. Special Reporting Requirements

Quad Chart. See Section Appendix B.

Appendix A. Summary of Activities Accomplished

Description	Timeline (Months)	Complete (Percent)	Start (Date)	Completed (Date)
Specific Aim 1: Human Subject Evaluations.				
Major Task 1: Submission of Human Use Protocols and Preparation of Facilities				
Subtask 1: Develop and submit human use protocols	1-3	100%	28 Jul 20	13 Jan 21
Subtask 2: Develop and submit human cadaver use protocols	1	100%	28 Jul 20	30 Sep 20
Subtask 3: Prepare human test facilities				
• Facility setup complete	2-4	100%	28 Jul 20	30 Apr 21
• Hearing protection devices for testing acquired				
• Evaluation by lab personnel underway				
<i>Milestone #1: Local IRB Approval</i>	4			COMPLETE
Subtask 4: Submit protocols for Army HRPO approval	5	100%	28 Jul 20	13 Jan 21
<i>Milestone #2: HRPO Approval</i>	6			COMPLETE
Major Task 2: Test Method Verification				
Subtask 1: Obtain pilot psychoacoustic measures of hearing protective device (HPD) effects	6-12	100%	28 Jan 21	31 Jan 22
• Pilot measurements are underway				
Subtask 2: Analyze pilot psychoacoustic measures of hearing protection device effects	10-12	100%	28 Apr 21	31 Jan 22
• Analysis of pilot measurements are underway				
Subtask 3: Obtain psychoacoustic measures of hearing protective device (HPD) effects	12-30	100%	28 Jul 21	27 July 23
Subtask 4: Analyze pilot human cadaver measures of HPD attenuation to impulse noise exposures	1-5	100%	15 Jan 21	31 Jan 22
• Re-analysis of existing data underway				
• Manuscript in preparation				
Subtask 5: Obtain human cadaver measures of HPD attenuation to impulse noise exposures	6-24	100%	01 Apr 21	27 July 23
• Three PMHS subjects obtained and prepared				
Subtask 6: Conduct ongoing quality assurance review of psychoacoustic, cadaver, and associated data	6-36	100%	28 Jul 21	27 July 23
<i>Milestone #3: Manuscript on impulsive noise measurements on cadaveric subjects with HPDs</i>	24	100%	28 Jul 21	27 July 23
<i>Milestone #4: Manuscript on HPD effects on perception</i>	30			COMPLETE
Specific Aim 2: Refinement of HPD Evaluation Methods.				
Major Task 1: Verification of Electromechanical Test Methods				
Subtask 1: Refine the electromechanical metrics				
• Equipment interfaces refined for eLN tests	1-24	100%	21 Dec 20	27 July 23
• New source for eLD developed				
• Preliminary testing of some devices completed to refine metrics				
Major Task 2: Develop prototype standards for appropriate HPD evaluation methods				
Subtask 1: Develop data and prototype standards	25-33	100%	1 Nov 21	27 July 23
<i>Milestone #5: Deliver standards input to appropriate standards committees</i>	36			COMPLETE
Specific Aim 3: Application of HPD Evaluation Methods.				
Major Task 1: Hearing Protection Device Evaluations				
Subtask 1: Evaluate a broad array of HPDs using the refined electromechanical test methods	18-30	100%	1 Nov 21	27 July 23
<i>Milestone #6: Conference presentation on comparative HPD performance</i>	30			COMPLETE
Major Task 2: Develop Hearing Protection Optimization Tool				
Subtask 1: Incorporate HPD test results	24-36			
Subtask 2: Compile HPD optimization tool on multiple platforms				
• Software tool prototype is in development to be implemented on an Amazon Web Services server for remote access	30-36	95%	1 Aug 21	
• Software tool is being developed for desktop and mobile devices				
<i>Milestone #7: Deliver HPD optimization tool to sponsor</i>	36			

Appendix B. Quad Chart

Appendix C. Draft Test Standards

The draft test standards for eSQ, eSL, eLD, and eSN, as well as the memo outlining recommendations for eIN testing, are enclosed in this appendix.

Standard eSQ

Draft

27 July 2023

Prepared for:

US Army Medical Research and Materiel Command

Prepared by:

Applied Research Associates, Inc.

Ted Argo, PhD

7921 Shaffer Parkway

Littleton, CO 80127

Phone: 303.795.8106

targo@ara.com

Performed under:

Contract W81XWH-20-C-0077, Correlation of Electromechanical Hearing Protection Test

Methods with Human Performance

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1 Scope

This standard establishes electromechanical test methods for the measurement of the signal quality of hearing protection devices (HPDs). This standard is intended to be part of a series of standards that utilizes electromechanical methods to provide uniform metrics to evaluate the performance of HPDs.

Signal quality refers to the fidelity of sound transmitted through a HPD when compared with an unprotected ear, regardless of source level. HPDs alter sound as it interacts with the device, reducing the quality of the received signal. Consequently, HPDs create confusion in auditory perception by allowing unnatural sounds to reach the ear. The distortion in signal caused by the HPD degrades situational awareness, and can significantly impact the ability to communicate, localize sound, and effectively respond to sounds in the environment.

The electromechanical test method of signal quality (eSQ) measures the effects on the perception of verbal communication of sound as it passes through a HPD by applying the Speech Intelligibility Index (SII) established in ANSI/ASA S3.5-2020, Methods for Calculation of the Speech Intelligibility Index. The results collapse to a single metric that quantifies the effects HPD reductions in sound quality has on communication.

This standard develops a hearing protection evaluation method to describe signal quality that can discriminate relative performance between devices, provide a basis to develop performance requirements, and maintain quality assurance of the devices over time. This standard establishes uniform instrumentation requirements, procedures for the measurement of signal quality, and develops the computation to generate the single value metric that is correlated with human performance.

This standard is not intended to replace current standards, or the use of human subjects to evaluate HPDs. Rather, this standard focuses on supplementing current methods by providing an evaluation tool to characterize a dimension of HPD performance not addressed by current standards.

2 Normative references

The following referenced documents are useful for the application of this standard.

ANSI S1.1, American National Standard Acoustical Terminology

ANSI/ASA S3.20, American National Standard Bioacoustical Terminology

ANSI/ASA S3.25, American National Standard for an Occluded Ear Simulator

ASTM D2240-05, Standard Test Method for Rubber Property – Durometer Hardness

ANSI/ASA S12.42, Methods for the Measurement of Insertion Loss of Hearing Protection Devices in Continuous or Impulsive Noise Using Microphone-in-Real-Ear of Acoustic Test Fixture Procedures

3 Terms and definitions

For the purposes of this standard, the terms and definitions given in ANSI S1.1, ANSI S3.20, and the following apply.

3.1 acoustic test fixture (ATF). An inanimate device that approximates certain physical characteristics and dimensions of a representative human head, pinnae, and ear canal, and is used for measuring the insertion loss of a hearing protection device.

3.2 active hearing protection device. A hearing protection device that contains electronic components including transducers (i.e., speakers and microphones) to increase or decrease the transmission of sound into the ear canal.

3.3 earmuff (over-the-ear HPD). A hearing protection device usually comprised of a headband and earcups with a soft cushion to seal against the head and intended to fit against the pinna (supraaural) or the sides of the head enclosing the pinna (circumaural). The earcups may also be held in position by attachment arms mounted on a hard hat or helmet.

3.4 earplug (in-the-ear HPD). A hearing protection device that is inserted into or that caps the ear canal.

3.5 electromechanical evaluation method. A laboratory sensor-based test to evaluate HPDs without the use of human subjects.

3.6 gain control of hearing protection device. Amount of amplification provided by an active HPD that suppresses or amplifies noise from the surrounding environment.

3.7 hearing protection device (HPD). A device, also called a hearing protector, worn to reduce the sound level in the ear canal.

3.8 insertion loss. The arithmetic difference in decibels between the sound pressure levels measured at a fixed position in the ear, measured under two different conditions (e.g., with and without the HPD in place).

3.9 measurement. A single sound pressure versus time waveform interval of data collection.

3.10 model sample. A single instance of a subset of a HPD model of interest.

3.11 recording system noise. The sound produced by the electronic elements of recording systems (e.g., microphones and data acquisition systems). In the context of this standard, the recording system noise is a disturbance in the signal or quantity of interest. This is distinct from test space noise, and self-noise.

3.12 reference point. A fixed spatial location within the testing facility at which the midpoint of a line connecting the ATF's ear canal openings is located and, likewise, the point to which all objective measurements of the sound field characteristics are referenced.

3.13 self-insertion loss. The passive insertion loss of the ATF when measured with a simulator of a near-ideal HPD, normally a metal plug or cup that is machined to seal the ear canal. It represents the acoustic leakage through the flanking pathways of the ATF.

3.14 self-noise. The sound, commonly described as a “hiss,” produced by the electronic elements of active hearing protection devices. In the context of this standard, the self-noise noise is a disturbance in the signal or quantity of interest. This is distinct from recording system noise attributable to other sources and test space noise

3.15 test space noise. The sound produced by acoustic sources present in the test space (e.g., equipment fans and HVAC system). In the context of this standard, the test space noise is a disturbance in the signal or quantity of interest. This is distinct from recording system noise and self-noise.

3.16 signal quality. The fidelity of sound transmitted through an HPD when compared with an unprotected ear, regardless of source level.

3.17 speech intelligibility index. A measure of the intelligibility of speech under a variety of listening conditions

4 Applicability of test methods

The procedures outlined in this standard are applicable for measuring the signal quality of earplugs and earmuffs.

The ability to accurately reproduce the original sound is an important factor when selecting hearing protection for tasks that involve communication. HPDs can compromise perception of verbal communication due to the change of sound as it passes through the device. Workers frequently communicate to warn about different hazards, avoid unsafe practices, develop emergency response practices, and learn about concerns or other safety issues. The ability to receive natural sounds from the surrounding environment is critical to safety, communication, and efficiency.

The procedures in this standard quantify HPD reductions in sound quality by establishing a single value metric based on the SII. This standard is unique in that it consists of laboratory sensor-based tests to evaluate signal quality quickly, inexpensively, and comprehensively without the use of human subjects. Existing methods involve lengthy and qualitative assessments of speech perception in noisy or degraded signal environments [e.g., human subject tests such as the Modified Rhyme Test (MRT), or QuickSIN for use in audiological assessments].

The signal quality metric defined in this standard provides additional insight into the overall hearing protector performance. To fully characterize the HPDs, further performance specifications and testing methods must be completed in conjunction with existing standards.

5 Requirements of the test facility

5.1 Introduction

The eSQ test method requires the production and measurement of high-level sounds in a controlled laboratory setting. To confront these challenges, the methods described in this standard utilize an environment with a neutral sound field equipped with a high-powered speaker, an acoustic test fixture (ATF) with flesh-simulant ears, a microphone with a large dynamic range and frequency range, and associated data acquisition equipment.

5.2 Test site

A neutral sound field shall be maintained throughout the eSQ testing space. A neutral sound field refers to an environment in which a uniform sound signal is not affected by any reflections, standing waves, or distortions.

To achieve a neutral sound field, the following conditions shall be met:

- The sound source shall produce the source signal at a uniform amplitude over the conventional audiometric frequency range of 250 Hz to 8 kHz. Further details on this requirement can be found in Section 5.4.4.
- The first reflection from any surface arrives at least 5 ms after the original signal or is at least 60 dB down from the testing sound pressure.
- Noise in the test space, such as environmental or electrical noise, shall be sufficiently low to ensure that the test signals exceed the level of unwanted noise by more than 6 dB.

An acoustically damped laboratory may be suitable for the testing methods described in this standard.

5.3 Acoustic test fixture

Any ATF that meets the requirements specified for measurements with continuous noise in ANSI/ASA S12.42-2010 (Section 6) should be used when performing the test methods described in this standard, unless superseded by a revised standard.

The ATF shall be equipped with proper instrumentation to perform the measurements while maintaining the required signal-to-noise ratio as discussed in Section **Error! Reference source not found.** An ATF that meets the requirements described in this section is described in **Error! Reference source not found.****Error! Reference source not found.****Error! Reference source not found.****Error! Reference source not found.**

5.3.1 Microphones

Microphones shall meet the requirements of ANSI/ASA S3.25, be positioned inside of the ATF, and have a frequency range of at least 250 Hz to 8 kHz between 40 dB and 150 dB SPL to account for ear canal resonance gain, and insertion loss from the HPDs. The pressure sensitivity shall be within ± 1 dB in the frequency range 250 Hz to 8 kHz relative to the sensitivity at 1 kHz.

5.3.2 Ear simulator, coupler, and flesh simulator

Any ear simulator, coupler, and flesh simulator combination that represents the dimensions of a human ear may be used. The anthropomorphic combination must be compatible with the ATF base, meet the ATF requirements, and permit a proper placement of the HPD in accordance with the manufacturer's instructions.

5.4 Sound source

5.4.1 Output

The sound source shall be capable of producing 2.5 cycles of a sine wave spanning 250 Hz to 8 kHz up to 70 dB SPL. The onset and offset of the sound source shall be tapered with a Tukey window using a window parameter of 0.1. It is recommended to use a high-efficiency, high-power midrange speaker designed to provide high sound pressure level in a compact size.

5.4.2 Sound pressure level requirements

The sound source shall be calibrated such that the target SPL at each of the octave band center frequencies ranging from 250 Hz to 8 kHz measures ± 0.25 dB at the reference point. The calibration method is detailed in Section 7.3.3.

5.4.3 Accuracy of frequency

The accuracy of the frequency of the test signal shall be within $\pm 1\%$ of the designated value.

5.4.4 Uniform sound field requirements

With the ATF absent, the amplitude of the signal, measured using the calibration microphone at six positions relative to the reference point, ± 15 cm in the coronal, sagittal, and transverse planes, shall remain within a range of 5 dB. The difference between the measurements in the coronal plane on the axis of the ear canals shall not exceed 3 dB in each band. The orientation of the calibration microphone shall be kept the same at each position.

5.4.5 Distortion of the test signal

The total harmonic distortion of the test signal produced by the speaker shall not exceed 1% at any test frequency.

5.5 Instrumentation

5.5.1 Amplifier

The audio amplifier shall be capable of interfacing with the sound source to transmit up to 70 dB SPL in the frequency range of 250 Hz to 8 kHz without input or output clipping. A single channel amplifier is sufficient for this testing method.

5.5.2 Power module

If using an externally polarized microphone, a power module will be required.

5.5.3 Data acquisition equipment

The signal gain shall be amplified to at least 10% of the dynamic range of the data acquisition system, and less than 90% of the dynamic range to avoid clipping. The pre-amplifier may be incorporated into the power module so long as the required signal gain is achieved.

5.5.4 Pre-amplifier

The received signal shall be amplified to at least 10% of the dynamic range of the data acquisition system. The pre-amplifier may be incorporated into the power module so long as the required signal gain is achieved.

6 Test conditions

6.1 Introduction

The conditions described in this section are required to ensure that the results of the eSQ test are correlated with human performance.

6.2 Sound source and ATF orientation

The ATF shall be placed directly in front of the sound source at a distance of at least one meter, such that the sound wave is perpendicular to the frontal plane of the ATF.

6.3 Test signals

The sound source shall span frequencies from 250 Hz to 16 kHz at 70 dB. The onset and offset of the sound source shall be tapered with a Tukey window using a window parameter of 0.1. Figure 6-1 provides an illustration of the test signal. Data collected using other SPLs may be required for HPDs with high insertion loss or active electronics to ensure the minimum signal-to-noise ratio is met as discussed in Section **Error! Reference source not found.**.

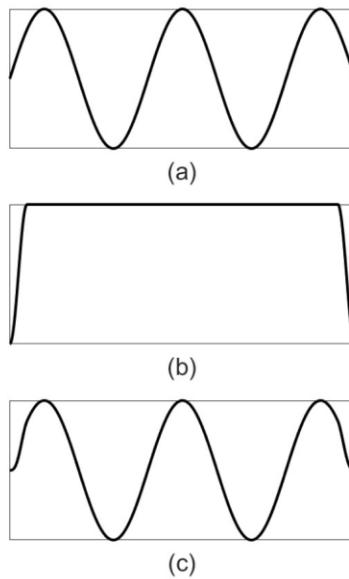


Figure 6-1. a) 2.5 cycles of a sine wave at a measurement center band frequency. (b) A Tukey window with cosine fraction 0.1. (c) The test signal formed by multiplying the signal in (a) with the Tukey window in (b).

6.4 Ambient conditions

The test procedures in this standard should be conducted with an ambient temperature between 50°F–90°F, and relative humidity between 10%–90%. Any measurements taken outside of this range shall be properly documented.

6.5 Placement of HPDs

The device under test shall be fitted on the ATF in accordance with the manufacturer's instructions and correspond to actual use. Furthermore, measurement of band force is recommended on all samples of earmuff style devices prior to testing. For further information on measuring band force, see Section 5.2 of ANSI/ASA S12.42-2010 (Section 5.2).

The eSQ test methods are only valid for earplug and earmuff hearing protection devices. These methods have not been validated with systems incorporated into helmets.

6.6 Gain control of HPD

For active HPDs that provide an ambient listening capability, the unity gain setting as described in **Error! Reference source not found.** shall be used for all measurements. Data collection at the unity gain setting is required; however, other gain settings also may be measured.

7 Measurements

7.1 Introduction

This section describes the requirements and measurement procedures for performing the eSQ test. Most sound sources do not have a flat frequency response (i.e., equal output across a range of frequencies); therefore, a fixed voltage applied at different frequencies will not produce the same SPL. Consequently, to ensure that the sound source produces a constant SPL across the frequency band, a sound source calibration must be performed. Additionally, the test method utilizes the changes in insertion loss. As a result, open ear and occluded measurements must be acquired. Finally, measurements must be made to characterize the system in terms of recording system noise of the instrumentation and sources of test space noise. To remain consistent in language this section clarifies common terms used to derive the signal quality metric.

7.2 Explanation of terms

Figure 7-1 illustrates common terms used throughout the remainder of this standard. The following sections aim to further explain each term and provide statistically relevant requirements.

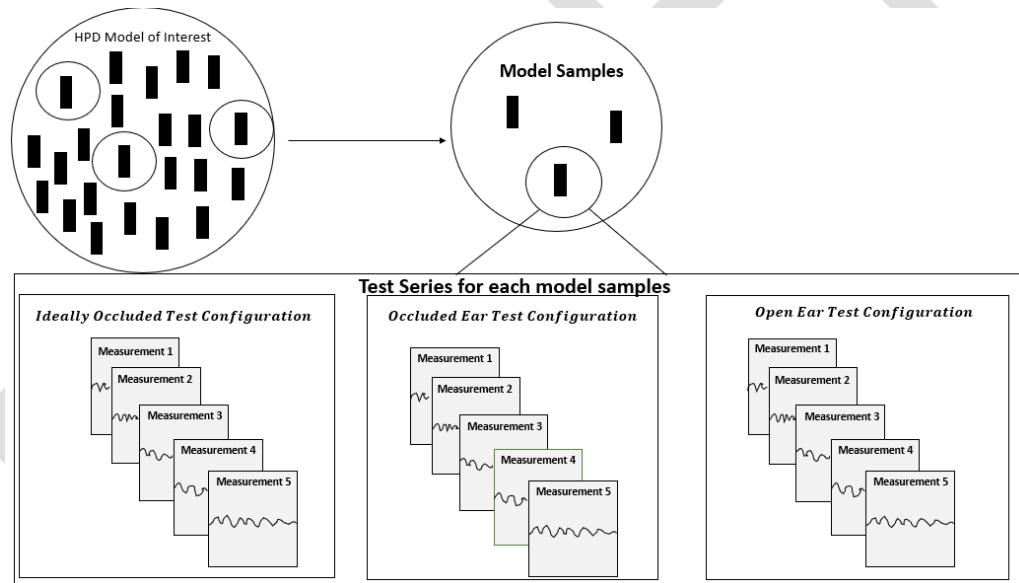


Figure 7-1. Flowchart of terms used in the described test protocol. Many individual units of an example HPD are shown in the top-left bubble; three samples are chosen from this population for characterization, as shown in the top-center bubble. The bottom graphs illustrate the test protocol for a single sample, where multiple iterations of chirp sweep responses are recorded for the sample at various spatial points.

7.2.1 Model sample

A model sample is a single instance of a subset of a HPD model of interest. At least three model samples are required to be tested to characterize the signal quality of the HPD model of interest.

7.2.2 Test series

A test series is a collection of measurements in each of the testing configurations. The different types of testing configurations are described in Section 7.4.

7.2.3 Measurement

A measurement refers to a single interval of data collection for a model sample. A measurement consists of a pressure versus time waveform, $P(t)$. Five measurements shall be recorded for each test configuration.

7.3 Requirements for measurements

7.3.1 Sampling frequency

The sampling rate is the number of samples per second required to reduce the continuous signal to a discrete signal. The signal acquired during this test shall be sampled at a minimum 44,100 samples per second (44.1 kHz).

7.3.2 Minimum signal-to-noise ratio

The signal to noise compares the spectrum level of equivalent signal to the noise floor of the electromechanical measurement. The signal-to-noise ratio between these measurements shall be at least 6 dB within octave bands spanning the range of 250 Hz to 8 kHz. Additional stimulus repetitions may be collected to achieve this requirement via signal averaging. HPDs yielding less than 6 dB SNR shall be reported as inadequate for measurement.

7.3.3 Calibration method for sound source output

To determine the SPL incident upon the ATF, a calibration microphone shall be positioned at the reference point in the plane of the test fixture ears without the ATF present. The calibration microphone shall be orientated at 45 degrees to the incident acoustic wave.

Once the calibration microphone is in place, the gain of the power amplifier should be fixed such that the target SPLs can be reached. The power amplifier output level should not be changed once the calibration has begun. To obtain a calibrated source voltage for a target SPL at the reference point, the procedures in Table 7-1 shall be performed.

Table 7-1. Order of calibration measurements

Order	Instruction
1	Position the calibration microphone at the reference point.
2	Transmit five signals at the target SPL using a single octave band center frequency.
3	Record the average SPLs of each of the five signals.
4	Adjust the voltage of the speaker (upward or downward).
5	Repeat Steps 1–4 until the average SPL of the five signals measures within 0.25 dB of the target SPL.
6	Record and use the voltage that achieved the target SPL at that frequency.

Order	Instruction
7	Repeat Steps 1–6 for each octave band center frequency from 250 Hz to 16 kHz.

The microphone used for calibration of the test signal shall be a microphone with a calibrated sensitivity and frequency response from 250 Hz to 8 kHz. The calibration of the microphone shall be verified in the 24 months prior to use for testing according to this standard.

The microphone and preamplifier shall demonstrate no more than 2% total harmonic distortion over the complete range of peak SPLs up to the maximum level to be tested. The maximum peak SPL is 12 dB over the maximum RMS SPL to be tested.

7.4 Test configurations

After the testing space is established, and all testing conditions are met, as described in Sections **Error! Reference source not found.** and **Error! Reference source not found.**, measurements shall be performed in the all test configurations.

The time-domain pressure wave shall be recorded from each ear simulator at the calibrated test signal. An appropriate gain setting should be used to ensure that the received signals are above the noise level of the measurement system.

7.4.1 Open ear (**O**)

The open-ear measurements shall be performed with the test fixture's ear canals, free of any obstructions.

7.4.2 Occluded measurements (**C**)

The occluded measurements shall be performed with the HPD under test installed into the ATF in accordance with the manufacturer's instructions.

7.4.3 Ideally occluded (**IO**)

The ideally occluded measurements shall be performed with a simulator of a near-ideal HPD, normally a metal plug or cup that is machined to seal the ear canal. The ideally occluded ear measurement quantifies the recording system noise (T') by suppressing any test space noise present in the system.

7.5 Sequence of measurements

The measurements described above shall be conducted in the sequence outlined in Table 7-2.

Table 7-2: Order of measurements

Order	Instruction
1	Set up the testing space while meeting all the testing conditions
2	Turn on the instrumentation and wait the warm-up time recommended by the manufacturer.

Order	Instruction
3	Perform five measurements in the ideally occluded test configuration.
4	Perform five measurements in the open ear test configuration.
5	Install the HPD under test into both ears of the ATF.
6	Turn on the device under test (if applicable) and confirm gain setting.
7	Perform five measurements in the occluded test configuration.
8	Repeats Steps 1–7 for each model sample.
9	Repeat Steps 1–8 for each model of interest.

8 Data analysis

8.1 Introduction

This section describes data processing and reduction techniques used to generate the electromechanical sound quality metric (eSQ). The metric is based on the Speech Intelligibility Index (SII) established in ANSI/ASA S3.5-2020, *Methods for Calculation of the Speech Intelligibility Index*. The test method outlined in this standard quantifies the alternation of the test signal's spectrum presented to the ATF and compares these measured values to standard signal spectrum that represents the signal levels prevalent in human speech levels. The result is a measure of the fidelity of the signal, representing the influence the HPD has on the intelligibility of speech under a variety of adverse listening conditions. The techniques in this section are correlated with human performance.

8.2 List of symbols

SII: Speech Intelligibility Index

eSQ: electromechanical signal quality metric

eSQ_{sample}: signal quality metric of an individual model sample

eSQ_{model}: signal quality metric of an individual model

I_i: band importance function for the *i*th octave band center frequency

L_i: spatial distortion factor for the *i*th octave band center frequency

K_i: band audibility factor for the *i*th octave band center frequency

G_i: HPD insertion gain for the *i*th octave band center frequency

E_i: spectrum level of the presented signal for the *i*th octave band center frequency

U_i: spectrum level of a standard signal for vocal effort for the *i*th octave band center frequency

X'_i: spectrum level of equivalent internal noise for the *i*th octave band center frequency

X_i: self-noise of active HPDs for the *i*th octave band center frequency

T'_i : recording system noise for the i th octave band center frequency

d : distance from the speech source to the reference point, m

d_0 : reference distance, m

E'_i : spectrum level of equivalent signal for the i th octave band center frequency

D_i : noise floor of the electromechanical measurement

N'_i : spectrum level of the test space noise for the i th octave band center frequency

SPL_i^0 : measured sound pressure level for the i th octave band center frequency band in the open ear configuration

$TFOE_i$: transfer function of the open ear for the i th octave band center frequency

8.3 eSQ metric

The eSQ metric is computed using measured electromechanical signals, and the standard signal spectrum levels for six frequency bands consistent with the octave frequency band method formulated in ANSI/ASA S3.5-2020 (Section 4). As shown in Equation (1), the metric is a combination of the band importance function (I), the speech level distortion factor (L), and the band audibility factor (K) over all frequency bands. Theoretically, the SII quantifies the ability to distinguish a source signal from background noises for each octave band essential in speech recognition.

$$SII = \sum (I_i \times L_i \times K_i) \quad (1)$$

8.3.1 Band importance function

The band importance function (I) represents the contributions of frequency bands to speech recognition. The function consists of weighted values for each frequency band that correlates to the contribution of that frequency to understanding speech. The higher the weighted value, the greater amount of significance to speech intelligibility. The individual weights are derived from ANSI/ASA S3.5-2020. For convenience, the weights for each octave band relevant to the method used in this standard are in Table 8-1.

Table 8-1. Band importance factors based on the octave band SII procedure from ANSI/ASA S3.5-2020

Nominal octave band center frequency (Hz)	Band Importance
250	0.0617
500	0.1671
1000	0.2373
2000	0.2648
4000	0.2142
8000	0.0549

8.3.2 Signal level distortion factor

The signal level distortion factor (L) represents the decrease in the signal quality. As shown in Equation (2), the factor considers the difference in the spectrum level of the equivalent signal at each octave band center frequency due to insertion loss from the HPD and spatial distortion (E'), and the standard spectrum level of the test signal comparable to a vocal effort (U). The constant values in the equation (10 and 160) are consistent with the SII computations in ANSI/ASA S3.5-2020 such that a maximum value of 1 is obtained when there is no distortion to the spectrum level of the presented signal, and its value decreases to a minimum of zero as the signal quality decreases.

$$L = 1 - \frac{E'_i - U_i - 10}{160} \quad (2)$$

8.3.2.1 Spectrum level of equivalent signal

The spectrum level of equivalent signal (E'_i) in Equation (3) incorporates the effect insertion loss and spatial distortion has on the spectrum level of the presented signal. Since the test signal is calibrated to produce a constant SPL, the values for the spectrum level of the presented signal for the i th octave band center frequency (E_i) remain constant throughout the computations. The insertion loss (G_i) is the arithmetic difference in decibels between the measured SPLs of the test signal spectrum in the open and occluded test configurations for the i th octave band center frequency. The distortion attributed to spatial propagation is calculated using the distance (d) from the sound source to the reference point, and a reference distance of one-meter (d_o) in accordance with ANSI/ASA S3.5-2020. If using the test method described in this standard, E_i will be a constant value of 70 dB SPL for each octave band center frequency, and d will be equal to the reference distance of one meter.

$$E'_i = E_i + G_i - 20 \times \log\left(\frac{d}{d_o}\right) \quad (3)$$

8.3.2.2 Spectrum level of a standard signal for vocal effort

The spectrum level of a standard signal for vocal effort (U) represents the amount of sound pressure level required in the test signal from each octave band to produce an equivalent vocal exertion. The values in Table 8-2 are based on the methods in ANSI/ASA S3.5-2020, and for the purposes of this standard, are interpolated to reflect the level of the test signal (70 dB). In the event of using another test signal level other than 70 dB for the eSQ measurements, independent interpolation of vocal efforts would be necessary.

Table 8-2. Interpolated standard signal levels to produce an equivalent effort based on values in ANSI/ASA S3.5-2020

Octave band center frequency (Hz)	Test Signal (70dB)
250	39.64
500	41.35
1000	35.98
2000	27.63
4000	18.98

8000	6.68
------	------

8.3.3 The band audibility factor

The band audibility factor (K) identifies the portion of the source signal that contributes to signal quality in noisy environments. As shown in Equation (4), the factor is the difference in the spectrum level of the equivalent signal at each octave band center frequency (E'_i) and spectrum of noise floor (D_i). The constant values in the equation (15 and 30) are consistent with the SII computations in ANSI/ASA S3.5-2020 such that a maximum value of 1 is obtained when there is a small amount of noise disturbances in the presented signal, and its value decreases to a minimum of zero as the amount of noise increases.

$$K_i = \frac{E'_i - D_i + 15}{30} \quad (4)$$

8.3.3.1 Noise floor of the electromechanical measurement

The noise floor of the electromechanical measurement (D_i) is determined by comparing the spectrum level of the test space noise (N'_i) and the spectrum level of equivalent internal noise (X'_i). Conceptually, Equation (5) identifies the source of noise in the test environment that contributes most to the degradation of signal quality. Signal quality may be impacted by internal noise from the recording system or self-noise of active HPDs (if applicable), measured by X' , and test space noise, measured by N' . The variable with the larger impact is used for final band audibility factor calculations.

$$D_i = \max (N'_i, X'_i) \quad (5)$$

X'_i in Equation (6) integrates the recording system noise (T'_i) and the self-noise of active HPDs (X_i) for each octave band center frequency. The value of T'_i is the sound produced by the electronic elements of recording systems (e.g., microphones and data acquisition systems), and is measured using an ideally occluded ear to suppress any test space noise, as described in ANSI S12.42 (Section 6.3). The value X_i is the sound, commonly described as a “hiss,” produced by the electronic elements of active hearing protection devices and is determined using the test methods described in the accompanying self-noise standard.

$$X'_i = X_i + T_i' \quad (6)$$

Equation (7) defines the test space noise (N'_i) for each octave band center frequency. The value of N'_i is the sound produced by acoustic sources present in the test space (e.g., equipment fans and HVAC system), and is measured using the results of an open ear measurement at each center octave band frequency (SPL_i^0), the transfer function of the open ear ($TFOE_i$), spectrum level of the presented signal (E_i), spatial distortion, and spectrum level of equivalent internal noise (X'_i).

$$N'_i = SPL_i^0 - TFOE_i - E_i - 20 \log \left(\frac{d}{d_0} \right) - X'_i \quad (7)$$

8.4 Methods for calculating the signal quality metric

eSQ metric is computed using the equations described in Section **Error! Reference source not found.** The eSQ metric for a given model sample is the SII that was calculated from Equation (1). The average eSQ from each model sample (Equation 8) shall be averaged across the total number of samples tested, S, to determine the signal quality for the model of interest.

$$eSQ_{model} = \frac{\sum eSQ_{sample}}{S} \quad (8)$$

8.5 Uncertainty

The measurement of signal quality described in this standard has intrinsic uncertainties of varying degrees of severity. The estimation of uncertainty is beyond the scope of this standard. In lieu of an uncertainty calculation, Table 8-1 provides various sources of uncertainty in an order of influence each source has on the overall level of uncertainty. This table can be used as an ordered guide to identify areas to improve the results of measurements.

Table 8-3. Sources of uncertainty involved in the signal quality measurements

Description of Uncertainty Source
Fitting of HPDs
Speaker positioning
Signal-to-noise ratio
Reflections in test site
Sound source calibration
Ambient conditions

8.6 Reporting instructions

The test report shall include the following information.

- a) Reference to this standard.
- b) The brands/models/specification, describing the ATF used, including a description of the microphones and ear canal couplers installed and (if applicable) pinnae variant used.
- c) The temperature and relative humidity at which the tests were conducted.
- d) The type of HPD (e.g., earplug or earmuff), its brand/model name, and the number of model samples tested.
- e) The signal-to-noise ratio of each test series. HPDs with noise levels below 6 dB SNR shall be reported as inadequate for measurement.
- f) The signal quality metric of each model sample, signal quality metric for the model of interest, comparison of the signal quality metric between models of interest (if applicable), and the uncertainty associated with the eSQ metric.

Annex A
(informative)
Acoustic Test Fixture (ATF)

A.1 G.R.A.S. 45CB acoustic test fixture

A G.R.A.S. 45CB acoustic test fixture, shown in Figure A-1, is designed and specified to comply with the ANSI/ASA S12.42 standard, and is suitable for testing according to this draft standard. It includes all of the features described in S12.42 namely pinna, circumaural and interaural flesh simulation, proper length ear canal, heated fixture, sufficient self-insertion loss, occluded ear simulator, and microphones suitable for the impulse noise testing.

This dual-walled sound isolation box is capable of:

1. Isolating high amplitude exposures from making the test room unsafe, and
2. Preventing unwanted noise from reaching the test article from the laboratory environment.



Figure A-1. G.R.A.S. 45CB acoustic test fixture.

More information on this ATF can be found on the G.R.A.S. website using the following link:
<https://www.grasacoustics.com/products/test-fixtures/for-hearing-protector-test/product/282-45cb>

Annex B
(informative)
Measurement of Unity Gain

B.1 Gain setting for active head-worn devices

The gain setting of active head-worn devices shall be measured using the same acoustic test fixture used for self-noise testing. Calibrate a speaker/amplifier combination to output a 1 kHz tone at an amplitude of 70 dB as measured in one ear of the ATF. The output level of the speaker/amplifier combination should be adjusted until 70 ± 0.5 dB is measured. This shall be designated the open-ear level.

The active HPD shall then be placed on the ATF and the gain adjusted until the frontally incident sound field level most closely matches the previously measured open-ear level. Both the open-ear level and the level under the HPD are intended to be measured at the ear simulator microphone in the ATF.

Standard eSL

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Prepared for:

US Army Medical Research and Materiel Command

Prepared by:

Applied Research Associates, Inc.

Ted Argo, PhD

7921 Shaffer Parkway

Littleton, CO 80127

Phone: 303.795.8106

targo@ara.com

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1 Scope

This standard establishes electromechanical test methods for the estimation of changes to sound localization ability while wearing hearing protection devices (HPDs). This standard is intended to be part of a series of standards that utilizes electromechanical methods to provide uniform metrics to evaluate the performance of HPDs.

Humans determine the direction of a sound source by exploiting spatial acoustic cues arising due to interactions of incident sound with the head and pinnae. Cues include binaural interaural timing differences (ITD) and interaural level differences (ILD), and monaural spectral shape (SS) cues. HPDs distort these cues, causing confusion in auditory perception and degradation of situational awareness. The extent of acoustic distortion and associated perceptual impact varies across HPDs, defining the need for a standard method to discriminate amongst the many HPDs available.

The proposed electromechanical test method of sound localization (eSL) estimates the effects HPDs on sound localization by comparing ITD and ILD, monaural SS features, and the insertion loss differences in open-ear and HPD occluded-ear conditions using an acoustic test fixture (ATF). The results, presented as a combination of the primary localization error types, collapse to a single metric that captures the effect the HPD under test has on a user's ability to localize sound.

This standard develops a hearing protection evaluation method to describe sound localization abilities that can discriminate relative performance between devices, provide a basis to develop performance requirements, and maintain quality assurance of the devices over time. This standard establishes uniform instrumentation requirements, procedures for the measurement of sound localization, and develops the computation to generate the single value metric that is correlated with human performance.

This standard is not intended to replace current standards, or the use of human subjects to evaluate HPDs. Rather, this standard focuses on supplementing current methods by providing an evaluation tool to characterize a dimension of HPD performance not addressed by current standards.

2 Normative references

The following referenced documents are useful for the application of this standard.

ANSI S1.1, American National Standard Acoustical Terminology

ANSI S1.11, Specification for Octave-Band and Fractional-Octave-Band Analog and Digital Filters

ANSI/ASA S3.71, Methods for Measuring the Effect of Head-worn Devices on Directional Sound Localization in the Horizontal Plane

ANSI/ASA S3.20, American National Standard Bioacoustical Terminology

ANSI/ASA S3.25, American National Standard for an Occluded Ear Simulator

ASTM D2240-05, Standard Test Method for Rubber Property – Durometer Hardness

3 Terms and definitions

For the purposes of this standard, the terms and definitions given in ANSI S1.1, ANSI S3.20 and the following apply.

3.1 acoustic test fixture (ATF). An inanimate device that approximates certain physical characteristics and dimensions of a representative human head, pinnae, and/or ear canals and is used for measuring the insertion loss of a hearing protection device.

3.2 sound localization (auditory localization). Ability to identify the direction of origin of a detected sound based on auditory cues.

3.3 earmuff (over-the-ear HPD). A hearing protection device usually comprised of a headband and earcups with a soft cushion to seal against the head, intended to fit against the pinna (supra-aural) or the sides of the head enclosing the pinna (circumaural). The earcups may also be held in position by attachment arms mounted on a hard hat or helmet.

3.4 earplug (in-the-ear HPD). A hearing protection device that is inserted into or that caps the ear canal.

3.5 electromechanical evaluation method. A laboratory sensor-based test to evaluate HPDs without the use of human subjects.

3.6 hearing protection device (HPD). A device, also called a hearing protector, worn to reduce the sound level in the ear canal.

3.7 insertion loss. The arithmetic difference in decibels between the sound pressure levels measured at a fixed position in the ear, measured under two different conditions (e.g., with and without the HPD in place).

3.8 interaural level difference (ILD). Difference in pressure amplitude level received in one ear relative to the other at the tympanic membrane.

3.9 interaural time difference (ITD). Travel time difference from the sound source to one ear versus the other ear at the tympanic membrane.

3.10 measurement. A single sound pressure versus time waveform interval of data collection.

3.11 model sample. A single instance of a subset of a HPD model of interest.

3.12 reference point. A fixed spatial location within the testing facility at which the midpoint of a line connecting the ATF's ear canal openings is located, and likewise the point to which all objective measurements of the sound field characteristics are referenced.

3.13 recording system noise. The sound produced by the electronic elements of recording systems (e.g., microphones and data acquisition systems). In the context of this standard, the recording system noise is a disturbance in the signal or quantity of interest. This is distinct from test space noise, and self-noise.

3.14 self-insertion loss. The passive insertion loss of the ATF when measured with a simulator of a near-ideal HPD; normally a metal plug or cup that is machined to seal the ear canal. It represents the acoustic leakage through the flanking pathways of the ATF.

3.15 self-noise. The sound, commonly described as a “hiss,” produced by the electronic elements of active hearing protection devices. In the context of this standard, the self-noise noise is a disturbance in the signal or quantity of interest. This is distinct from recording system noise attributable to other sources and test space noise

3.16 test space noise. The sound produced by acoustic sources present in the test space (e.g., equipment fans and HVAC system). In the context of this standard, the test space noise is a disturbance in the signal or quantity of interest. This is distinct from recording system noise and self-noise.

4 Applicability of test methods

The procedures outlined in this standard are applicable for measuring the distortion of acoustic cues of incident sound that enable auditory sound localization for earplugs and earmuffs. This standard quantifies the effects of these distortions by comparing the signal measured by each ear for the open-ear and HPD occluded-ear test configurations.

The ability to accurately determine the direction of arrival of a sound is an important factor when selecting operationally relevant hearing protection. Workers frequently encounter loud objects, verbal instructions, and other useful acoustic information from their entire surroundings rather than from a single direction, often requiring an orientation response. In many settings, accurate sound source localization is critical to safety, efficient communication, and the ability to perform basic functions, particularly in visually degraded environments.

The ability to locate sounds significantly changes depending on the direction of arrival of incoming sound. Generally, HPDs conserve interaural difference cues while distorting SS cues, allowing for accurate localization in the left-right dimension, and degraded localization in the front-back and up-down dimensions. The testing methods described in this standard permit for direct comparison between HPDs by establishing a single value metric based on localization accuracy across spatial location.

The sound localization metric defined in this standard provides additional insight into the overall hearing protector performance. To fully characterize the HPDs, further performance specifications and testing methods must be completed in conjunction with traditional standards.

5 Requirements of the test facility

5.1 Introduction

The eSL test method requires careful setup due to the complexity of producing a repeatable, uniform sound at a variety of incidence angles and elevations while measuring signal arrivals that correlate with human behavior. To confront these challenges, the methods described in this standard utilize an environment with a neutral sound field (equipped with a compact testing apparatus or speaker array to generate a test signal from each source location), an ATF with a representative human head, a microphone with a large dynamic and frequency range, and special data acquisition equipment.

5.2 Test site

A neutral sound field shall be maintained throughout the eSL testing space. A neutral sound field refers to an environment in which a uniform sound signal is not affected by any reflections or distortions, and the signal quality is preserved. The eSL measurements rely on high-fidelity signals across the frequency spectrum; therefore, to achieve desirable results, an emphasis must be on enhancing the quality of the test signal while avoiding any disturbances.

To achieve a neutral sound field, the following conditions shall be met:

- The sound source shall produce the test signal at a uniform amplitude over the 250-Hz to 16-kHz frequency range. Further details on this requirement can be found in Section 5.4.5.
- The first reflection from any surface arrives at least 5 ms after the original signal or is at least 60 dB down from the testing SPL.
- Noise in the test environment, such as test space or recording system noise, shall be sufficiently low to ensure that the test signals exceed the level of unwanted noise by more than 6 dB.

An acoustically damped laboratory may be suitable for the testing methods described in this standard.

5.3 Acoustic test fixture

An acoustic test fixture (ATF) that meets the requirements specified for measurements with continuous noise in ANSI/ASA S12.42-2010 (Section 6) should be used when performing the test methods described in this standard, unless superseded by a revised standard.

The ATF shall be equipped with proper instrumentation to perform the measurements while maintaining the required signal-to-noise ratio as discussed in Section 7.3.2. An ATF that meets the requirements described in this section is described in Annex A.

5.3.1 Ear coupler and microphone

Microphones shall meet the requirements of ANSI/ASA S3.25, be positioned inside of the ATF, and have a frequency range of at least 250 Hz to 16 kHz between 40 dB and 120 dB SPL. The pressure sensitivity shall be within ± 1 dB in the frequency range 250 Hz to 16 kHz relative to the sensitivity at 1 kHz.

5.4 Sound source

5.4.1 Output

The sound source shall be capable of producing frequencies spanning 250 Hz to 16 kHz of at least 90 dB SPL. A single sound source (or an array of sound sources) may be used to output the test signal as long as the required azimuth and elevation angles can be achieved.

5.4.2 Dimension

The dimensions of the sound source can potentially offset the angle of each test position and add to measurement uncertainty. To achieve the accuracy of test signal requirements in Section 6.3 and ensure all frequencies of the signal are in the acoustic far-field, a compact sound source is recommended. A description of a compact virtual hemispherical speaker array that meets the requirements of this standard is described in Annex B.

5.4.3 Sound pressure level requirements

The sound source shall be calibrated such that the target SPL measures ± 0.25 dB at the reference point for each elevation and azimuth position. The calibration method is detailed in Section 7.3.3.

5.4.4 Accuracy of frequency

The accuracy of the frequency of the test signal shall be within $\pm 1\%$ of the designated value.

5.4.5 Uniform sound field requirements

With the ATF absent, the amplitude of the signal, measured using the calibration microphone at six positions relative to the reference point (± 15 cm in the coronal, sagittal, and transverse planes) shall remain within a range of 5 dB. The difference between the measurements in the coronal plane on the axis of the ear canals shall not exceed 3 dB in each band. The orientation of the calibration microphone shall be kept the same at each position.

5.4.6 Distortion of the test signal

The total harmonic distortion of the test signal produced by the speaker shall not exceed 1% at any test frequency.

5.5 Instrumentation

5.5.1 Amplifier

The audio amplifier shall be capable of interfacing with the sound source to transmit at least 90 dB SPL or greater in the frequency range of 250 Hz to 16 kHz at least one (1) meter without input or output clipping. A single channel amplifier is sufficient for this testing method.

5.5.2 Power module

If using an externally polarized microphone, a power module will be required.

5.5.3 Data acquisition equipment

A data acquisition system capable of acquiring high-amplitude SPLs shall be used in all measurements. Channel-to-channel isolation is recommended to eliminate crosstalk between the left and right ear microphones, and to prevent one channel from interfering with the other one.

The data acquisition system shall sample at a minimum sampling rate of 44,100 samples per second (44.1 kHz) and be able to resolve voltages at a minimum of 16 bits full scale.

5.5.4 Pre-amplifier

The signal gain shall be amplified to at least 10% of the dynamic range of the data acquisition system and less than 90% of the dynamic range to avoid clipping. The pre-amplifier may be incorporated into the power module so long as the required signal gain is achieved.

6 Test conditions

6.1 Introduction

The conditions described in this standard are essential to ensure that the results of the eSL test are correlated with of human performance. If these conditions cannot be achieved, the outcome of the test may be affected.

The apparatus used to generate sound from a variety of source locations can vary widely. This can be accomplished by using multiple sound sources to form an array or by moving a single sound around a stationary ATF to each azimuth and elevation combination. Alternatively, the ATF can be moved while the sound source remains stationary.

Regardless of the configuration, it is essential that the test signal is presented to the ATF in a repeatable manner. Manual alignment of the speaker or ATF is not recommended, as it reduces the confidence in repeatable positioning.

6.2 Sound source and ATF orientation

The ATF shall be positioned at the center of a minimum one-meter hemisphere. The reference point shall be at the center point of the hemisphere and shall be coincident at the 0° elevation plane. Figure 6-1 and Figure 6-2 illustrate the azimuth and elevation angles relative to the reference point.

S

Speaker Azimuth Angles

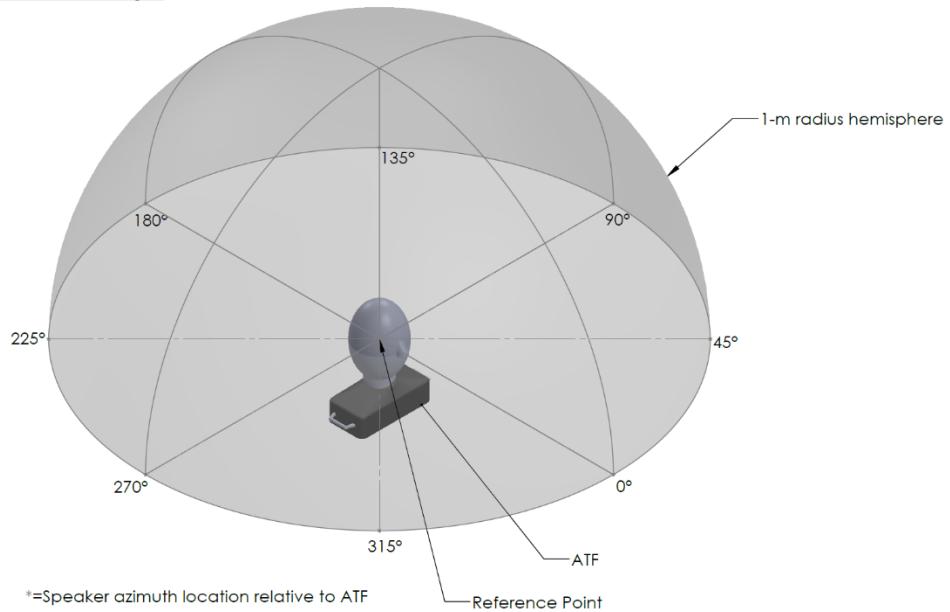


Figure 6-1. Full series of azimuth angles for the sound source relative to the reference point.

S

Speaker Elevation Angles

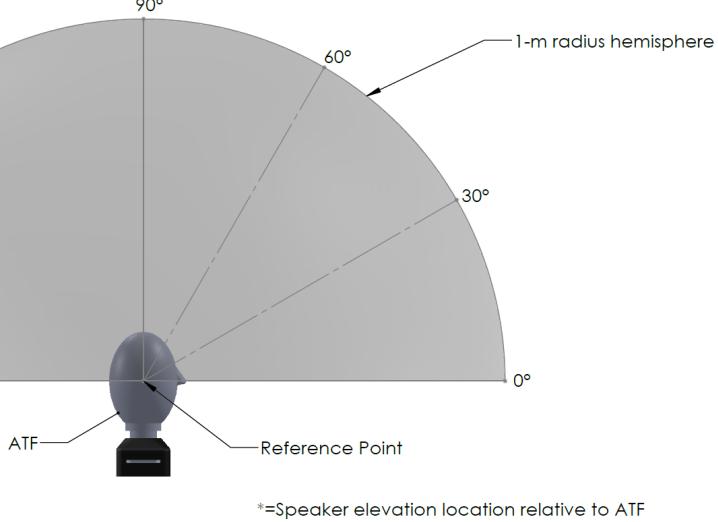


Figure 6-2. Full series of elevation angles for the sound source relative to the reference point.

6.3 Test signals

The test signal shall span frequencies from 250 Hz to 16 kHz at 70 dB SPL for active devices and at least 90 dB SPL for passive devices. Data collected using other SPLs may be required for HPDs with high insertion loss or active electronics to ensure the minimum signal-to-noise ratio is met as discussed in Section 7.3.2.

The test signal shall be presented to the ATF from a sound source located at a series of azimuth and elevation angles relative to the reference point. Azimuth angles shall range from 0° to 315° in 45° increments, and elevation angles shall range from 0° to 60° in 30° increments. Accuracy of positioning shall be $\pm 1^\circ$ in azimuth and elevation, relative to the center of the sound source.

Data collection using every combination of azimuth and elevation shown in Table 6-1 is required; however, other combinations of azimuth and elevation may also be measured. A smaller angle between measurements will increase the resolution, and potentially accuracy, of the measurement but will also increase the time necessary to perform the measurement.

Table 6-1. Location coordinates for required test position (azimuth, elevation)

Azimuth	Elevation		
	0°	30°	60°
0°	(0,0)	(0,30)	(0,60)
45°	(45,0)	(45,30)	(45,60)
90°	(90,0)	(90,30)	(90,60)
135°	(135,0)	(135,30)	(135,60)
180°	(180,0)	(180,30)	(180,60)
225°	(225,0)	(225,30)	(225,60)
270°	(270,0)	(270,30)	(270,60)
315°	(315,0)	(315,30)	(315,60)

6.4 Ambient conditions

The test procedures in this standard should be conducted with an ambient temperature between 50°F – 90°F and relative humidity between 10%–90%. Any measurements taken outside of this range shall be properly documented.

6.5 Placement of HPDs

The device under test shall be fitted on the ATF in accordance with the manufacturer's instructions and correspond to actual use. Furthermore, measurement of band force is recommended on all samples of earmuff style devices prior to testing. For further information on measuring band force, see Section 5.2 of ANSI/ASA S12.42-2010.

The eSL test methods are only valid for earplug and earmuff hearing protection devices. These methods have not been validated for other form factors (e.g., systems incorporated into helmets).

6.6 Gain control of HPD

For active HPDs that provide an ambient listening capability, the unity gain setting as described in Annex C shall be used for all measurements. Data collection at the unity gain setting is required, however, other gain settings also may be measured.

7 Measurements

7.1 Introduction

This section describes the requirements and measurement procedures for performing the eSL test. Most sound sources do not have a flat frequency response (i.e., equal output across a range of frequencies); therefore, a fixed voltage applied at different frequencies will not produce the same SPL. Consequently, to ensure that the sound source produces a constant SPL across the frequency band, a sound source calibration must be performed. Additionally, the test method performs a relative comparison at each location between the ears with and without an HPD. As a result, open ear and occluded measurements must be acquired. Finally, to remain consistent in language this section clarifies common terms used to derive the sound localization metric.

7.2 Explanation of terms

Figure 7-1 illustrates common terms used throughout the remainder of this standard. The following sections aim to further explain each term and provide statistically relevant requirements.

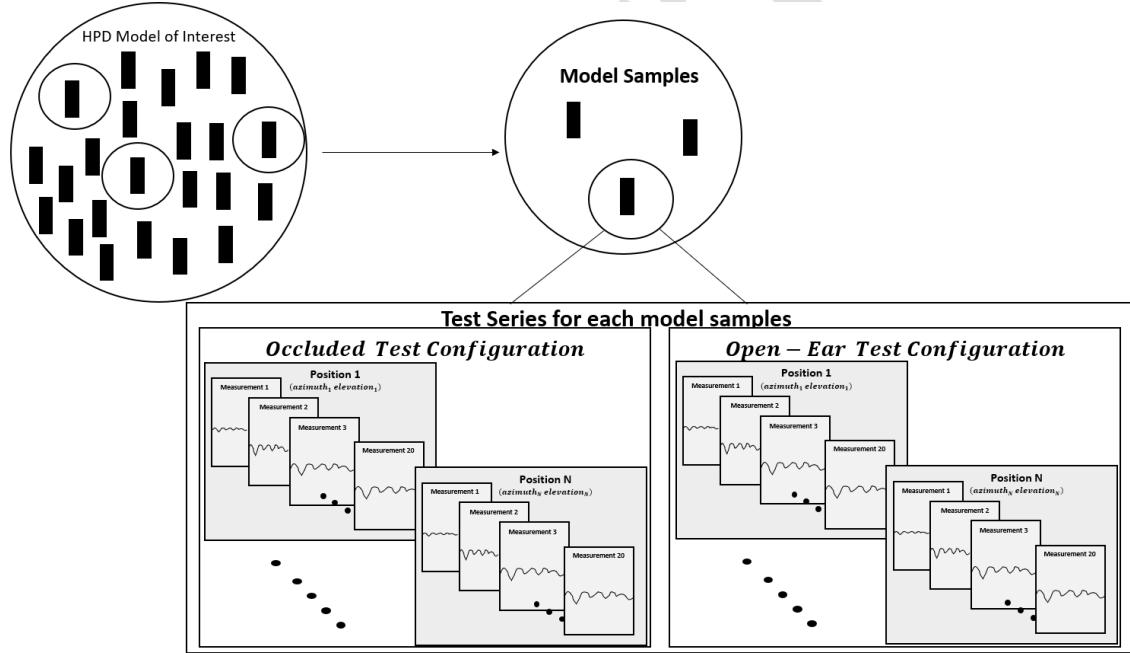


Figure 7-1. Flowchart of terms used in the described test protocol. Many individual units of an example HPD are shown in the top-left bubble; three samples are chosen from this population for characterization, as shown in the top-center bubble. The bottom graphs illustrate the test protocol for a single sample, where multiple iterations of chirp sweep responses are recorded for the sample at various spatial points.

7.2.1 Model sample

A model sample is a single instance of a HPD model of interest. At least three model samples are required to be tested to characterize the ability of the HPD model of interest to localize sound.

7.2.2 Test series

A test series is a collection of measurements in each of the testing configurations. The different types of testing configurations are described in Section 7.4.

7.2.3 Measurement

A measurement refers to a single interval of data collection. A measurement consists of a pressure versus time waveform, $P(t)$. Twenty measurements shall be recorded at each test position; therefore, 960 measurements should be collected for each model sample:

$$20 \frac{\text{measurements}}{\text{test position}} \times 24 \frac{\text{test positions}}{\text{test configurations}} \times 2 \frac{\text{test configurations}}{\text{model sample}} = 960 \frac{\text{measurements}}{\text{model sample}}$$

7.3 Requirements for measurements

7.3.1 Sampling frequency

The sampling rate is the number of samples per second required to reduce the continuous signal to a discrete signal. The signal acquired during this test shall be sampled at a minimum 44,100 samples per second (44.1 kHz).

7.3.2 Minimum signal-to-noise ratio

The signal to noise compares the output level of the occluded measurement (signal) to the output level of the open-ear (noise) measurement. The signal-to-noise ratio between these measurements shall be at least 6 dB within octave bands spanning the range of 250 Hz to 16 kHz. Additional measurements may be collected to achieve this requirement. HPDs with noise levels below 6 dB SNR shall be reported as inadequate for measurement.

7.3.3 Calibration method for sound source output

To signal to noise compares the output level of the occluded measurement (signal) to the output level of the open-ear (noise) measurement. The signal-to-noise ratio between these measurements shall be at least 6 dB within octave bands spanning the range of 250 Hz to 16 kHz. Additional measurements may be collected to achieve this requirement. HPDs with noise levels below 6 dB SNR shall be reported as inadequate for measurement.

Table 7-1. Order of calibration measurements

Order	Instruction
1	Position the calibration microphone at the reference point.
2	Transmit five signals at the target SPL using a single octave band center frequency.
3	Record the average SPLs of each of the five signals.
4	Adjust the voltage of the speaker (upward or downward).
5	Repeat Steps 1–4 until the average SPL of the five signals measures within 0.25 dB of the target SPL.
6	Record and use the voltage that achieved the target SPL at that frequency.
7	Repeat Steps 1–6 for each octave band center frequency from 250 Hz to 16 kHz.

The microphone used for calibration of the test signal shall be a microphone with a calibrated sensitivity and frequency response from 250 Hz to 16 kHz. The calibration of the microphone shall be verified in the 24 months prior to use for testing according to this standard.

The microphone and preamplifier shall demonstrate no more than 2% total harmonic distortion over the complete range of peak SPLs up to the maximum level to be tested. The maximum peak SPL is 12 dB over the maximum RMS SPL to be tested.

7.4 Test configurations

After the testing space is established, and all testing conditions are met, as described in Sections 5 and 6, measurements shall be performed in the open-ear and occluded test configurations.

The time-domain pressure wave shall be recorded from each ear simulator at the calibrated test signal. An appropriate gain setting should be used to ensure that the received signals are above the noise level of the measurement system.

7.4.1 Open ear

The open-ear (*O*) measurements shall be performed with the test fixture's ear canals free of any obstructions.

7.4.2 Occluded

The occluded (*C*) measurements shall be performed with the HPD under test installed into the ATF in accordance with the manufacturer's instructions.

7.5 Sequence of measurements

The measurements described above shall be conducted in the sequence outlined in Table 7-2.

Table 7-2. Order of measurements

Order	Instruction
1	Position the ATF/sound source at the (0°, 0°) position.
2	Turn on the instrumentation and wait the warm-up time recommended by the manufacturer.
3	Perform twenty measurements in the open-ear test configuration at the (0°, 0°) position.
4	Repeat the measurements in the open-ear test configuration for every (azimuth, elevation) position.
5	Return the ATF/speaker to the (0°, 0°) position.
6	Install the HPD under test into both ears of the ATF.
7	Turn on the device under test (if applicable) and confirm gain setting.
8	Perform twenty measurements in the occluded test configuration at the (0°, 0°) position.
9	Repeat the measurements in the HPD test configuration for every (azimuth, elevation) position.
10	Repeat Steps 1–9 for each model sample.
11	Repeat Steps 1–10 for each model of interest.

8 Data analysis

8.1 Introduction

This section describes the data processing and reduction techniques to generate the sound localization metric. The metric is constructed of four terms that quantify the relative magnitude of vertical and horizontal localization errors from aural distortions created by the HPDs. This section defines the metric, explores the impact each term has on sound localization abilities, and provides further details on the method to compute the metric. The techniques described in this section have been verified to provide a sound localization metric that is correlated with human performance.

8.2 List of symbols

f_i : individual octave band number

ϕ_j : individual azimuth test position angle

θ_k : individual elevation test position angle

O: open-ear configuration

C: occluded-ear configuration

L: left ear signal

R: right ear signal

N: total number of testing positions

CE: collapse in elevation

CA: collapse in azimuth

FB: front-back

$P(t, \phi_j, \theta_k)$: average time-dependent response for all measurements at each test position (θ_j, ϕ_k)
Subscripts can include: O, C, L, R

$P(f_i, \phi_j, \theta_k)$: average frequency-dependent response for all measurements in each octave band center frequency (f_i) at each test position (θ_j, ϕ_k)

$|P(f_i, \phi_j, \theta_k)|$: average magnitude of the frequency-dependent response for all measurements in each octave band center frequency (f_i) at each test position (θ_j, ϕ_k)

$\angle P(f_i, \phi_j, \theta_k)$: average phase angle of the frequency-dependent response for all measurements in each octave band center frequency (f_i) at each test position (θ_j, ϕ_k)

Subscripts can include: O, C, L, R

eSL: sound localization metric

ILD: interaural level difference

ILD_{tot}: total ILD between the left and right ear in each test configuration

ILD_O(f_i, φ_j, θ_k): ILD for each octave band center frequency (f_i) at each test position (θ_j, φ_k) for the open configuration (O)

ILD_C(f_i, φ_j, θ_k): ILD for each octave band center frequency (f_i) at each test position (θ_j, φ_k) for the occluded configuration (C)

ITD: interaural time delay

ITD_{tot}: total ITD between the left and right ear in each test configuration

ITD_O(f_i, φ_j, θ_k): ITD for each octave band center frequency (f_i) at each test position (θ_j, φ_k) for the open configuration (O)

ITD_C(f_i, φ_j, θ_k): ITD for each octave band center frequency (f_i) at each test position (θ_j, φ_k) for the occluded configuration (C)

eIL: insertion loss

eIL_{tot}: total insertion loss

eIL(f_i, φ_j, θ_k): IL for each octave band center frequency (f_i) at each test position

eErr: spectral error

eErr_{CE}: collapse in elevation directional error

eErr_{CA}: collapse in azimuth directional error

eErr_{FB}: front-back directional error

W: weighting term

Subscripts can include: *CE, CA, FB*

DTF: direct transfer function

Subscripts can include: *CE, CA, FB*

Superscripts can include: *O, C*

H: head related transfer function

Subscripts can include: *O, C*

8.3 eSL metric

Equation (1) is the mathematical representation of the sound localization metric. Conceptually, the equation combines relative magnitude of the primary localization error types: failure to perceive

source elevation; failure to discriminate in the lateral dimension (left, right, forward, and rearward); and gross errors due to large signal attenuation. Each term in the equation is acquired by comparing the signal arrival time and intensity between the left and right ears, and the insertion loss differences in the open-ear and HPD occluded-ear conditions. eSL values approaching one indicate the HPD performance is similar to open-ear performance, and the HPD has the potential to maintain the user's ability to locate sound. eSL values approaching zero indicate increasingly poor localization performance. By inspecting each term individually, the methods in this standard can determine the influence each localization error has on the overall metric.

$$eSL = \sqrt{ITD_{Tot}} \times \sqrt{ILD_{Tot}} \times eIL_{Tot} \times eErr_{Tot} \quad (1)$$

8.3.1 Interaural time delay (ITD)

The ITD is a binaural cue concerning the difference in arrival time of a sound between ears. ITD is critical to horizontal localization (left-right), as it provides a cue to the angle of the sound source from the head.

The average arrival time of the signal in the open-ear configuration (ITD_O) and the average arrival time of the signal in the occluded-ear configuration (ITD_C) is calculated from the difference in the average frequency-dependent phase angles for each octave band center frequency (f_i) at each test position (θ_j, ϕ_k) for the left and right ear signals (L, R) in the open and occluded test configurations (O, C), as shown in Equations (2) and (3).

$$ITD_O(t, \phi_j, \theta_k) = t_{peak}[P_{O,R}(t, \phi_j, \theta_k)] - t_{peak}[P_{O,L}(t, \phi_j, \theta_k)] \quad (2)$$

$$ITD_C(t, \phi_j, \theta_k) = t_{peak}[P_{C,R}(t, \phi_j, \theta_k)] - t_{peak}[P_{C,L}(t, \phi_j, \theta_k)] \quad (3)$$

Where the function t_{peak} returns the temporal location of the peak value of the argument. The total ITD (ITD_{Tot}), Equation (4), is the sum of magnitude of the difference between the ITD_O and the ITD_C for each octave band center frequency (f_i) and each test position (θ_j, ϕ_k). The sum of the difference in the arrival times is divided by the open-ear term, such that equal terms in occluded and open-ear configurations lead to a value of zero, resulting in an ITD_{Tot} of one (HPD performance is similar to open ear.) Conversely, an ITD_{Tot} of zero indicates the HPD significantly distorts the signal arrival between ears and, ultimately, diminishes the ability to localize sound.

$$ITD_{Tot} = 1 - \frac{\sum_{j,k} |ITD_O(t, \phi_j, \theta_k) - ITD_C(t, \phi_j, \theta_k)|}{\sum_{j,k} ITD_O(t, \phi_j, \theta_k)} \quad (4)$$

Additionally, the ITD may be calculated continuously using the phase angle difference as shown in Equations (5), (6), and (7), rather than the time difference method.

$$ITD_O(f_i, \phi_j, \theta_k) = \frac{1}{2\pi f_i} \angle [P_{O,R}(f_i, \phi_j, \theta_k) - P_{O,L}(f_i, \phi_j, \theta_k)] \quad (5)$$

$$ITD_C(f_i, \phi_j, \theta_k) = \frac{1}{2\pi f_i} \angle [P_{C,R}(f_i, \phi_j, \theta_k) - P_{C,L}(f_i, \phi_j, \theta_k)] \quad (6)$$

$$ITD_{Tot} = 1 - \frac{1}{I} \sum_i \frac{\sum_{j,k} |ITD_O(f_i, \phi_j, \theta_k) - ITD_C(f_i, \phi_j, \theta_k)|}{\sum_{j,k} ITD_O(f_i, \phi_j, \theta_k)} \quad (7)$$

8.3.2 Interaural level difference (ILD)

The ILD is a binaural cue for horizontal sound localization considering the differences in sound intensities or levels between the ears. The average intensity of the signal in the open-ear configuration (ILD_O) and the average intensity of the signal in the occluded-ear configuration (ILD_C) is calculated from the difference in the average frequency-dependent magnitudes for each octave band center frequency (f_i) at each test position (θ_j, ϕ_k) for the left and right ear signals (L, R) in the open and occluded test configurations (O, C), as shown in Equations (8) and (9).

$$ILD_O(f_i, \phi_j, \theta_k) = |P_{O,R}(f_i, \phi_j, \theta_k) - P_{O,L}(f_i, \phi_j, \theta_k)| \quad (8)$$

$$ILD_C(f_i, \phi_j, \theta_k) = |P_{C,R}(f_i, \phi_j, \theta_k) - P_{C,L}(f_i, \phi_j, \theta_k)| \quad (9)$$

The total ILD (ILD_{Tot}), Equation (10), is the sum of magnitude of the difference between the ILD_O and the ILD_C for each octave band center frequency (f_i) and each test position (θ_j, ϕ_k). The sum of the difference in the levels is divided by the open-ear term, such that equal terms in occluded and open-ear configurations lead to a value of zero, resulting in an ILD_{Tot} of one (HPD performance is similar to open ear.) Conversely, an ILD_{Tot} of zero indicates the HPD significantly distorts the signal level between ears, and ultimately diminishes the ability to localize sound.

$$ILD_{Tot} = 1 - \frac{1}{I} \sum_i \frac{\sum_{j,k} |ILD_O(f_i, \phi_j, \theta_k) - ILD_C(f_i, \phi_j, \theta_k)|}{\sum_{j,k} ILD_O(f_i, \phi_j, \theta_k)} \quad (10)$$

8.3.3 Insertion loss (eIL)

A high IL can reduce localization abilities by attenuating the perceived signal below perceptual thresholds. The total amount of IL is limited to bone conduction. At the insertion loss bone conduction limit (BCL), all the incident sound in the ear canal is attenuated, and the sound can only be transmitted to the inner ear via pathways other than the ear canal. As a result, in the reduction of signal through the ear canal, the ability to localize sound is significantly reduced. For the purpose of this standard, the IL has a lower bound of zero (i.e., no IL), and an upper bound at the BCL (i.e., max IL possible.)

The IL for each octave band center frequency (f_i) at each test position (θ_j, ϕ_k), $eIL(f_i, \phi_j, \theta_k)$, is determined from the difference in the average frequency-dependent magnitudes in the open and occluded test configurations (O, C), as shown in Equation (11).

$$eIL(f_i, \phi_j, \theta_k) = |P_O(f_i, \phi_j, \theta_k) - P_C(f_i, \phi_j, \theta_k)| \quad (11)$$

Equation (12) quantifies the effects of total insertion loss (eIL_{Tot}) with respect to sound localization by comparing the averaged insertion loss over all test positions (θ_j, ϕ_k) at each octave band center frequency (f_i) to the BCL at that frequency, with the BCL as the limit of potential insertion loss from an HPD. The BCL values have been established in ASA/ANSI S12.42.

$$eIL_{Tot} = rms \left[\frac{BCL(f_i) - \min(\frac{1}{N} \sum_{j,k} eIL(f_i, \phi_j, \theta_k), BCL(f_i))}{BCL(f_i)} \right] \quad (12)$$

As apparent in the equation, the eIL_{Tot} term is normalized by the BCL to create an eIL_{Tot} value from zero to one. An eIL_{Tot} of zero indicates the IL is equal to the BCL, and the ability to localize sound is negatively impacted. Alternatively, an eIL_{Tot} of one indicates a minimal amount of IL exists, and the ability to localize sound is not affected by the HPD (i.e., HPD performance is similar to open ear.)

8.3.4 Spectral error (eErr)

The spectral error factor is comprised as a combination of three types of directional errors: the collapse-in-elevation error $eErr_{CE}$, the collapse-in-azimuth error $eErr_{CA}$, and the front-back error $eErr_{FB}$. The equation for combining all three terms is shown in Equation (13).

$$eErr = \frac{1}{eErr_{CE} + eErr_{CA} + eErr_{FB} + 1} \quad (13)$$

The equation shown produces a value of one if there is no error magnitude from any of the three terms and approaches a value of zero as the magnitudes of the three error types increase.

Each error term is the average loss in directional transfer function (DTF) information acquired in the occluded and open test configurations over all spatial locations.

The equation for the total collapse-in-error term from Equation (13) is displayed in Equation (14).

$$eErr_{CE} = \frac{1}{jk} \sum_{j,k} W_{CE}(\phi_j, \theta_k) \Delta DTF_{CE}(\phi_j, \theta_k) \quad (14)$$

Where W_{CE} is a weighting term equal to the great circle distance of the spatial error for a given set of indices, and ΔDTF_{CE} is the difference between the open and occluded DTFs, given by Equation (15).

$$\Delta DTF_{CE}(\phi_j, \theta_k) = DTF_{CE}^O(\phi_j, \theta_k) - DTF_{CE}^C(\phi_j, \theta_k) \quad (15)$$

Where DTF^O is the open ear DTF and DTF^C is the occluded DTF. Each DTF (using the variable X to represent either O or C) is defined in Equation (16).

$$DTF_{CE}^X(\phi_j, \theta_k) = H^X(\phi_j, \theta_k) - H^X(\phi_j, 0) \quad (16)$$

Where H^X is the HRTF for either the open ear or occluded case for a given spatial source location. The weighting terms for the CE error term are calculated using Equation (17).

$$W_{CE}(\phi_j, \theta_k) = \frac{\theta_k}{180} \quad (17)$$

The equation for the total collapse-in-azimuth term from Equation (13) is shown in Equation (18).

$$eErr_{CA} = \frac{1}{jk} \sum_{j,k} W_{CA}(\phi_j, \theta_k) \Delta DTF_{CA}(\phi_j, \theta_k) \quad (18)$$

Where W_{CA} is a weighting term equal to the great circle distance of the spatial error for a given set of indices, and ΔDTF_{CA} is the difference between the open and occluded DTFs, given by Equation (19).

$$\Delta DTF_{CA}(\phi_j, \theta_k) = DTF_{CA}^O(\phi_j, \theta_k) - DTF_{CA}^C(\phi_j, \theta_k) \quad (19)$$

Each DTF (using the variable X to represent either O or C) is defined Equation (20).

$$DTF_{CA}^X(\phi_j, \theta_k) = \begin{cases} H^X(\phi_j, \theta_k) - H^X(-90, 0) & \text{for } -180 < \phi < 0 \\ H^X(\phi_j, \theta_k) - H^X(90, 0) & \text{for } 0 < \phi < 180 \end{cases} \quad (20)$$

The weighting terms for the CA error term are calculated by using Equation (21).

$$W_{CA}(\phi_j, \theta_k) = \begin{cases} \arccos(\cos(\theta_k) \cos(\phi_j + 90)) & \text{for } -180 < \phi < 0 \\ \arccos(\cos(\theta_k) \cos(\phi_j - 90)) & \text{for } 0 < \phi < 180 \end{cases} \quad (21)$$

The equation for the total front-back error term from Equation (13) is displayed in Equation (22).

$$eErr_{FB} = \frac{1}{jk} \sum_{j,k} W_{FB}(\phi_j, \theta_k) \Delta DTF_{FB}(\phi_j, \theta_k) \quad (22)$$

Where W_{FB} is a weighting term equal to the great circle distance of the spatial error for a given set of indices, and ΔDTF_{FB} is the difference between the open and occluded DTFs, given by Equation (23).

$$\Delta DTF_{FB}(\phi_j, \theta_k) = DTF_{FB}^O(\phi_j, \theta_k) - DTF_{FB}^C(\phi_j, \theta_k) \quad (23)$$

Each DTF (using the variable X to represent either O or C) is defined in Equation (24).

$$DTF_{FB}^X(\phi_j, \theta_k) = \begin{cases} H^X(\phi_j, \theta_k) - H^X(-180 - \phi_j, 0) & \text{for } -180 < \phi < 0 \\ H^X(\phi_j, \theta_k) - H^X(180 - \phi_j, 0) & \text{for } 0 < \phi < 180 \end{cases} \quad (24)$$

The weighting terms for the FB error term are calculated using Equation (25).

$$W_{FB}(\phi_j, \theta_k) = \begin{cases} \arccos(\cos(\theta_k) \cos(2\phi_j - 180)) & \text{for } -180 < \phi < 0 \\ \arccos(\cos(\theta_k) \cos(2\phi_j + 180)) & \text{for } 0 < \phi < 180 \end{cases} \quad (25)$$

8.4 Method for calculating the sound localization metric

eSL metric is computed using the equations described in Section 8.3. This section provides the basic computation steps to obtain the final eSL metric from the pressure versus time waveform measurements collected for the left and right ear at every azimuth and elevation test position in the open and occluded configurations.

8.4.1 Step 1: Average measurements

Compute the average of all measurements at each test position (θ_j, ϕ_k) for the left and right ear signals (L, R) in the open and occluded test configurations (O, C).

8.4.1.1 Step 1a: Average measurements for the left ear in the open configuration

Result: $P_{O,L}(t, \phi_j, \theta_k)$

8.4.1.2 Step 1b: Average measurements for the left ear in the occluded configuration

Result: $P_{C,L}(t, \phi_j, \theta_k)$

8.4.1.3 Step 1c: Average measurements for the right ear in the open configuration

Result: $P_{O,R}(t, \phi_j, \theta_k)$

8.4.1.4 Step 1d: Average measurements for the right ear in the occluded configuration

Result: $P_{C,R}(t, \phi_j, \theta_k)$

8.4.1.5 Step 1e: Average measurements for the open configuration

Compute the average measurements at each position for the left and right ear signals in the open configuration.

$$\text{Result: } P_O(t, \phi_j, \theta_k) = \frac{\sum_{j,k} P_{O,R}(t, \phi_j, \theta_k) + \sum_{j,k} P_{O,L}(t, \phi_j, \theta_k)}{N} \quad (26)$$

8.4.1.6 Step 1f: Average measurements for occluded configuration

Compute the average measurements at each position for the left and right ear signals in the occluded configuration.

$$\text{Result: } P_C(t, \phi_j, \theta_k) = \frac{\sum_{j,k} P_{C,R}(t, \phi_j, \theta_k) + \sum_{j,k} P_{C,L}(t, \phi_j, \theta_k)}{N} \quad (27)$$

8.4.2 Step 2: Magnitude and phase spectrum

Compute the Fourier Transform and apply a 1/1 octave band filter, in accordance with ASA/ANSI S1.11, of each average measurement found in Step 1 to obtain the magnitude and phase spectrum for each octave band center frequency (f_i) and each test position (θ_j, ϕ_k) in the open and occluded test configurations (O, C).

8.4.2.1 Step 2a: Magnitude and spectrum for the left ear in the open configuration

Result: $\angle P_{O,L}(f_i, \phi_j, \theta_k), |P_{O,L}(f_i, \phi_j, \theta_k)|$

8.4.2.2 Step 2b: Magnitude and spectrum for the left ear in the occluded configuration

Result: $\angle P_{C,L}(f_i, \phi_j, \theta_k), |P_{C,L}(f_i, \phi_j, \theta_k)|$

8.4.2.3 Step 2c: Magnitude and spectrum for the right ear in the open configuration

Result: $\angle P_{O,R}(f_i, \phi_j, \theta_k), |P_{O,R}(f_i, \phi_j, \theta_k)|$

8.4.2.4 Step 2d: Magnitude and spectrum for the right ear in the occluded configuration

Result: $\angle P_{C,R}(f_i, \phi_j, \theta_k), |P_{C,R}(f_i, \phi_j, \theta_k)|$

8.4.2.5 Step 2e: Magnitude and spectrum for the open configuration

Result: $\angle P_O(f_i, \phi_j, \theta_k), |P_O(f_i, \phi_j, \theta_k)|$

8.4.2.6 Step 2f: Magnitude and spectrum for the left ear in the occluded configuration

Result: $\angle P_C(f_i, \phi_j, \theta_k), |P_C(f_i, \phi_j, \theta_k)|$

8.4.3 Step 3: Compute ITD

8.4.3.1 Step 3a: Compute ITD_O

Input the phase spectrum results calculated in Step 2a and Step 2c into Equation (2) or (5) to calculate the ITD_O .

8.4.3.2 Step 3b: Compute ITD_C

Input the phase spectrum results calculated in Step 2b and Step 2d into Equation (3) or (6) to calculate the ITD_C .

8.4.3.3 Step 3c: Compute ITD_{Tot}

Input the results from Step 3a and Step 3b into Equation (4) or (7) to calculate the total ITD.

8.4.4 Step 4: Compute ILD

8.4.4.1 Step 4a: Compute ILD_O

Input the magnitude spectrum results calculated in Step 2a and Step 2c into Equation (8) to calculate the ILD_O .

8.4.4.2 Step 4b: Compute ILD_C

Input the magnitude spectrum results calculated in Step 2b and Step 2c into Equation (9) to calculate the ILD_C .

8.4.4.3 Step 4c: Compute ILD_{Tot}

Input the results from Step 3a and Step 3b into Equation (10) to calculate the total ITD.

8.4.5 Step 5: Compute eIL

8.4.5.1 Step 5a: Compute eIL

Input the magnitude spectrum results calculated in Step 2e and Step 2c into Equation (8) to calculate the insertion loss.

8.4.5.2 Step 5b: Compute eIL_{tot}

Input the results from Step 5a and the BCL for each octave band center frequency f_i from ANSI/ASA S12.42 into Equation (12) to calculate the eIL_{tot} .

8.4.6 Step 6: Compute eErr

8.4.6.1 Step 6a: Compute weights

Calculate weights W_{CE} , W_{CA} , W_{FB} for all azimuth and elevation measurement points (ϕ_j and θ_k) using Equations (17), (21), and (25).

8.4.6.2 Step 6b: Measure HRTFs

Measure open ear HRTFs (H^O) and occluded HRTFs (H^C) for all azimuth and elevation points (ϕ_j and θ_k).

8.4.6.3 Step 6c: Calculate DTFs

Calculate DTFs for all three error terms at all azimuth and elevation points (ϕ_j and θ_k) using the measured HRTFs from Step 6b inserted into Equations (16), (20), and (24).

8.4.6.4 Step 6d: Calculate ΔDTF error terms

Calculate DTF Errors for all three error terms all azimuth and elevation points (ϕ_j and θ_k) using the calculated DTFs from Step 6c inserted into Equations (15), (19), and (23).

8.4.6.5 Step 6e: Calculate each averaged spatial error term

Use the calculated DTF error terms and weights at all azimuth and elevation points (ϕ_j and θ_k), inserted into Equations (14), (18), and (22), to calculate $eErr_{CE}$, $eErr_{CA}$, and $eErr_{FB}$.

8.4.6.6 Step 6f: Calculate eErr

Calculate $eErr$ using Equation (13) and the error terms from Step 6e.

8.4.7 Step 7: Compute eSL

Input the results from Step 3c, Step 4c, Step 5b, and Step 6f into Equation (1) to calculate the eSL.

8.4.8 Step 8: Compare eSL

The average eSL from each model sample shall be averaged across the total number of samples tested to determine the sound localization metric for the model of interest.

8.5 Uncertainty

The measurement of sound localization described in this standard has intrinsic uncertainties of varying degrees of severity. The estimation of uncertainty is beyond the scope of this standard. In lieu of an uncertainty calculation, Table 8-1 provides various sources of uncertainty in an order of influence each source has on the overall level of uncertainty. This table can be used as an ordered guide to identify areas of measurement improvement.

Table 8-1. Sources of uncertainty involved in the sound localization measurements

Description of Uncertainty Source
Fitting of HPDs
Speaker positioning
Signal-to-noise ratio
Reflections in test site
Sound source calibration
Ambient conditions

8.6 Information to be included in the test report

The test report shall include the following information.

- a) Reference to this standard.
- b) The neutral sound field measurements of the test space of the recording system noise.
- c) The dynamic range, frequency response, and sensitivity of the microphone.
- d) The brands/models/specification, describing the ATF used, including a description of the microphones and earcanal couplers installed, and (if applicable) pinnae variant used.
- e) The temperature and relative humidity at which the tests were conducted.
- f) The type of HPD (e.g., earplug or earmuff), its brand/model name, and the number of model samples tested.
- g) The signal-to-noise ratio of each test series. HPDs with noise levels below 6 dB SNR shall be reported as inadequate for measurement.
- h) The sound localization metric of each model sample, the sound localization metric for the model of interest, and comparison of sound localization metric between models of interest (if applicable).

Annex A

(informative)

Acoustic Test Fixture (ATF)

A.1 G.R.A.S. 45CB acoustic test fixture

The G.R.A.S. 45CB acoustic test fixture, shown in A-1, is designed and specified to comply with the ANSI/ASA S12.42 standard, and is suitable for testing according to this draft standard. It includes all features described in S12.42, namely pinna, circumaural and interaural flesh simulation, proper length ear canal, heated fixture, sufficient self-insertion loss, occluded ear simulator, and microphones suitable for the impulse noise testing.



Figure A-1. G.R.A.S. 45CB acoustic test fixture

More information on this ATF can be found on the G.R.A.S. website using the following link:
<https://www.grasacoustics.com/products/test-fixtures/for-hearing-protector-test/product/282-45cb>

Annex B

(informative)

Compact Virtual Hemispherical Speaker Array

B.1 Virtual hemispherical speaker array apparatus

An apparatus that can be used to measure sound localization at all specified azimuth and elevation angles is shown in Figure B-1. This test fixture is able to simulate a virtual acoustic half-space using a rotation/translation system to move the sound source relative to a G.R.A.S. 45CB ATF to evaluate the effect of incidence direction on the perceived sound.

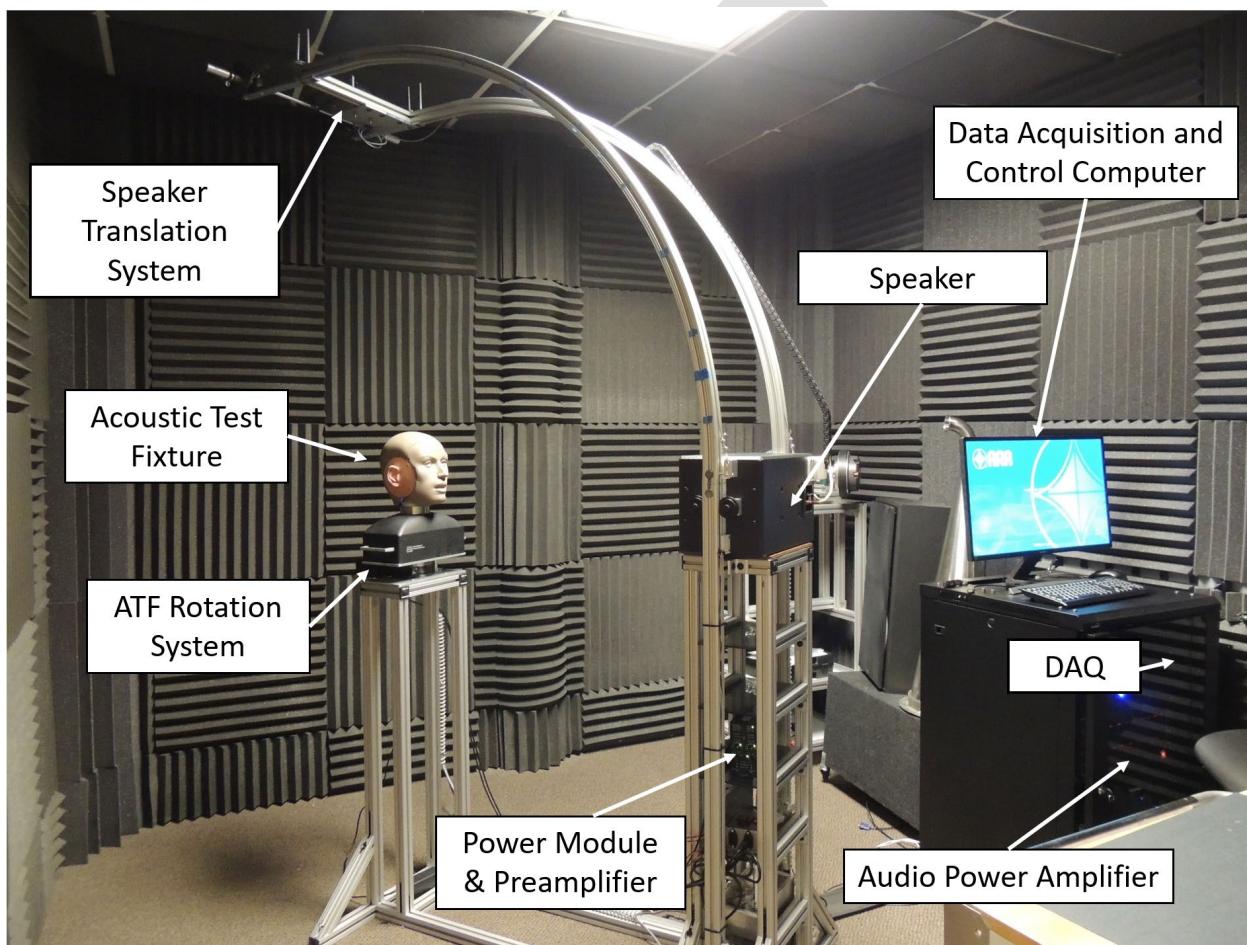


Figure B-1. The virtual hemispherical speaker array and localization measurement system

Figure B-1 shows the virtual hemispherical speaker array apparatus in a sound-isolating room. The ATF sits on a rotational stage controlled by a stepper motor. The speaker can be translated along a curved rail at a constant 100 cm offset from the reference point to achieve different elevations. This compact system can simulate sound incident upon the ATF from any ($0^\circ < \text{elevation} < 90^\circ$, $0^\circ < \text{azimuth} < 360^\circ$) position. The azimuth angle of the ATF and elevation angle of the speaker in the array is shown in Figure B-2.

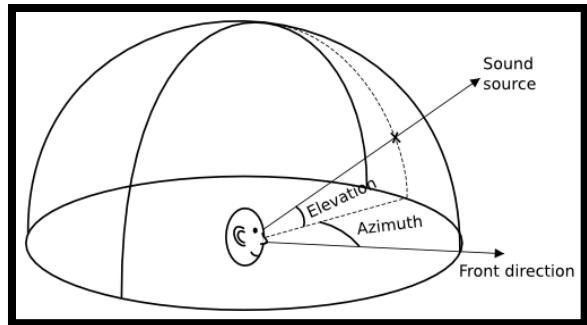


Figure B-2. Elevation and azimuthal positions defined for test apparatus

Annex C

(informative)

Measurement of Unity Gain

C.1 Gain setting for active head-worn devices

The gain setting of active head-worn devices shall be measured using the same acoustic test fixture used for sound localization testing. Calibrate a speaker/amplifier combination to output a 1 kHz tone at an amplitude of 70 dB as measured in one ear of the ATF. The output level of the speaker/amplifier combination should be adjusted until 70 ± 0.5 dB is measured. This shall be designated the open-ear level.

The active HPD shall then be placed on the ATF and the gain adjusted until the frontally incident sound field level most closely matches the previously measured open-ear level. Both the open-ear level and the level under the HPD are intended to be measured at the ear simulator microphone in the ATF.

Standard eLD

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Prepared for:
US Army Medical Research and Materiel Command

Prepared by:
Applied Research Associates, Inc.
Ted Argo, PhD
7921 Shaffer Parkway
Littleton, CO 80127
Phone: 303.795.8106
targo@ara.com

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1 Scope

This standard establishes electromechanical test methods for the measurement of level-dependent frequency response (eLD) of hearing protection devices (HPDs). This standard is intended to be one in a series of standards that utilizes electromechanical methods to provide uniform metrics to evaluate the performance of HPDs.

Level dependency refers to a characteristic of HPDs that describes the ability to provide increasing protection when subjected to increasing sound pressure levels (SPLs). This feature, also called non-linear protection, is demonstrated by attenuating higher pressures more than lower pressures. Level dependency permits the user to wear a HPD during routine tasks and retain audibility of low-to-moderate intensity sounds (e.g., speech), yet gain additional protection against high amplitude exposures without use of an additional HPD (e.g., double hearing protection) or additional engineering controls.

The eLD test method measures the insertion loss of HPDs as a function of frequency across a series of presentation levels and is used to quantify the level of protection provided by the HPD across typical sound exposures. These measurements are performed using an acoustic test fixture (ATF) inside of a facility with a speaker capable of producing a high amplitude pressure wave typical of high-level industrial exposures. The measurements are collected as frequency-resolved, level-dependent insertion losses over a range of SPLs, which are subsequently collapsed to a single metric to describe the level-dependent characteristic of the HPD.

This standard develops a hearing protection evaluation method to describe level dependent characteristics that can discriminate relative performance between devices, provide a basis to develop performance requirements, and maintain quality assurance of the devices over time. This standard establishes uniform instrumentation requirements, procedures for the measurement of sound localization, and develops the computation to generate the single value metric that is correlated with human performance. This standard is not intended to replace current standards, or the use of human subjects to evaluate HPDs. Rather, this standard focuses on supplementing current methods by providing an evaluation tool to characterize a dimension of HPD performance not addressed by current standards.

2 Normative references

The following referenced documents are useful for the application of this standard.

ANSI S1.1, American National Standard Acoustical Terminology

ANSI/ASA S3.20, American National Standard Bioacoustical Terminology

ANSI/ASA S3.25, American National Standard for an Occluded Ear Simulator

ASTM D2240-05, Standard Test Method for Rubber Property – Durometer Hardness

ANSI/ASA S12.42, Methods for the Measurement of Insertion Loss of Hearing Protection Devices in Continuous or Impulsive Noise Using Microphone-in-Real-Ear of Acoustic Test Fixture Procedures

3 Terms and definitions

For the purposes of this standard, the terms and definitions given in ANSI S1.1, ANSI S3.20, and the following apply.

3.1 acoustic test fixture (ATF). An inanimate device that approximates certain physical characteristics and dimensions of a representative human head, pinnae, and ear canal, and is used for measuring the insertion loss of a hearing protection device.

3.2 active hearing protection device. A hearing protection device that contains electronic components including transducers (i.e., speakers and microphones) to increase or decrease the transmission of sound into the ear canal.

3.3 active insertion loss. The insertion loss determined during a single fitting of an active noise reduction hearing protection device, comparing the sound pressure levels with and without the device's electronics operating. This is equivalent to the arithmetic difference in decibels between the total insertion loss and the passive insertion loss.

3.4 earmuff (over-the-ear HPD). A hearing protection device usually comprised of a headband and earcups with a soft cushion to seal against the head and intended to fit against the pinna (supraaural) or the sides of the head enclosing the pinna (circumaural). The earcups may also be held in position by attachment arms mounted on a hard hat or helmet.

3.5 earplug (in-the-ear HPD). A hearing protection device that is inserted into or that caps the ear canal.

3.6 electromechanical evaluation method. A laboratory sensor-based test to evaluate HPDs without the use of human subjects.

3.7 gain control of hearing protection device. Amount of amplification provided by an active HPD that suppresses or amplifies noise from the surrounding environment.

3.8 hearing protection device (HPD). A device, also called a hearing protector, worn to reduce the sound level in the ear canal.

3.9 insertion loss. The arithmetic difference in decibels between the sound pressure levels measured at a fixed position in the ear, measured under two different conditions (e.g., with and without the HPD in place).

3.10 level-dependent hearing protection device. A device designed to produce a change in attenuation as a function of the external sound level.

3.11 measurement. A single sound pressure versus time waveform interval of data collection.

3.12 model sample. A single instance of a subset of a HPD model of interest.

3.13 passive hearing protection device. A device that relies solely on mechanical elements to block or otherwise control the transmission of sound to the auditory system.

3.14 passive insertion loss. The insertion loss determined from the difference between the sound pressure levels with and without the HPD in place on the head or fixture, measured for a passive HPD or an active HPD with the electronics turned off.

3.15 reference point. A fixed spatial location within the testing facility at which the midpoint of a line connecting the ATF's ear canal openings is located and, likewise, the point to which all objective measurements of the sound field characteristics are referenced.

3.16 recording system noise. The sound produced by the electronic elements of recording systems (e.g., microphones and data acquisition systems). In the context of this standard, the recording system noise is a disturbance in the signal or quantity of interest. This is distinct from test space noise, and self-noise.

3.17 total insertion loss. The insertion loss determined from the difference between the sound pressure levels with and the HPD in place on the head or fixture, measured for an active HPD with its electronics turned on.

3.18 self-insertion loss. The passive insertion loss of the ATF when measured with a simulator of a near-ideal HPD, normally a metal plug or cup that is machined to seal the ear canal. It represents the acoustic leakage through the flanking pathways of the ATF.

3.19 self-noise. The sound, commonly described as a “hiss,” produced by the electronic elements of active hearing protection devices. In the context of this standard, the self-noise noise is a disturbance in the signal or quantity of interest. This is distinct from recording system noise attributable to other sources and test space noise.

3.20 sound isolation box. Enclosure engineered to isolate high-amplitude exposures from making the test room unsafe and prevent unwanted noise from reaching the test article from the laboratory environment.

3.21 test space noise. The sound produced by acoustic sources present in the test space (e.g., equipment fans and HVAC system). In the context of this standard, the test space noise is a disturbance in the signal or quantity of interest. This is distinct from recording system noise and self-noise

4 Applicability of test methods

The procedures outlined in this standard are applicable for measuring the level dependency of earplugs and earmuffs.

Level dependency is an important determining factor when selecting operationally relevant hearing protection. HPDs are necessary to protect against high exposure levels, and it is generally desirable to have high insertion loss (good protection) at these levels. For low exposure levels, high insertion loss is generally undesirable as it limits the audibility of important signals (e.g., speech). Level-dependent HPDs are designed to provide both audibility at low levels and protection at high levels. To capture the level-dependent characteristic and permit practical comparisons between various types of HPDs, the testing methods described in this standard integrate a wide range of SPLs and

frequencies. The level-dependent behavior significantly changes depending on the SPL, frequency, and the type of HPD. As a result, the expected range of exposure levels should be considered when selecting HPDs.

The level-dependent metric defined in this standard provides additional insight into the overall hearing protector performance. To fully characterize the HPDs, further performance specifications and testing methods must be completed in conjunction with existing standards.

5 Requirements of the test facility

5.1 Introduction

The eLD test method requires the production and measurement of high-level sounds in a controlled laboratory setting. To confront these challenges, the methods described in this standard utilize an environment with a neutral sound field equipped with a high-powered speaker, an acoustic test fixture (ATF) with flesh-simulant ears, a microphone with a large dynamic range and frequency range, and associated data acquisition equipment.

5.2 Test site

A neutral sound field shall be maintained throughout the eLD testing space. A neutral sound field refers to an environment in which a uniform sound signal is not affected by any reflections, standing waves, or distortions.

To achieve a neutral sound field, the following conditions shall be met:

- The sound source shall produce the source signal at a uniform amplitude over the conventional audiometric frequency range of 250 Hz to 8 kHz. Further details on this requirement can be found in Section 5.4.4.
- The first reflection from any surface arrives at least 5 ms after the original signal or is at least 60 dB down from the testing sound pressure.
- Noise in the test environment, such as test space or recording system noise, shall be sufficiently low to ensure that the test signals exceed the level of unwanted noise by more than 6 dB.

A suitable measurement system could utilize a “box-in-a-box” sound isolation chamber, as detailed in Annex A, or an anechoic chamber

5.3 Acoustic test fixture

Any ATF that has a self-insertion loss of at least 60 dB from 80 Hz to 12.5 kHz and is able to accommodate all hearing protection devices shall be used during the test procedures described in this standard. The ATF shall have circumaural bases of sufficient diameter that can fully support earmuff cushions.

The ATF shall be representative of human ear and ear canal with dimensions such that a variety of earplugs and earmuffs can be accurately tested.

The ATF shall be equipped with proper instrumentation to perform the measurements while maintaining the required signal to noise ratio. An ATF that meets the requirements described in this section is described in Annex B.

5.3.1 Microphones

Microphones shall meet the requirements of ANSI/ASA S3.25, be positioned inside of the ATF, and have a frequency range of at least 250 Hz to 8 kHz between 40 dB and 150 dB SPL to account for ear canal resonance gain, and insertion loss from the HPDs. The pressure sensitivity shall be within ± 1 dB in the frequency range 250 Hz to 8 kHz relative to the sensitivity at 1 kHz.

5.3.2 Ear simulator, coupler, and flesh simulator

Any ear simulator, coupler, and flesh simulator combination that represents the dimensions of a human ear may be used. The anthropomorphic combination must be compatible with the ATF base, meet the ATF requirements, and permit a proper placement of the HPD in accordance with the manufacturer's instructions.

5.4 Sound source

5.4.1 Output

The sound source shall be capable of producing 2.5 cycles of a sine wave spanning 250 Hz to 8 kHz up to 130 dB SPL. The onset and offset of the sound source shall be tapered with a Tukey window using a window parameter of 0.1. It is recommended to use a high-efficiency, high-power midrange speaker designed to provide high sound pressure level in a compact size.

5.4.2 Sound pressure level requirements

The sound source shall be calibrated such that the target SPL at each of the octave band center frequencies ranging from 250 Hz to 8 kHz measures ± 0.25 dB at the reference point. The calibration method is detailed in Section 7.3.3.

5.4.3 Accuracy of frequency

The accuracy of the frequency of the test signal shall be within $\pm 1\%$ of the designated value.

5.4.4 Uniform sound field requirements

With the ATF absent, the amplitude of the signal, measured using the calibration microphone at six positions relative to the reference point, ± 15 cm in the coronal, sagittal, and transverse planes, shall remain within a range of 5 dB. The difference between the measurements in the coronal plane on the axis of the ear canals shall not exceed 3 dB in each band. The orientation of the calibration microphone shall be kept the same at each position.

5.4.5 Distortion of the test signal

The total harmonic distortion of the test signal produced by the speaker shall not exceed 1% at any test frequency

5.5 Instrumentation

5.5.1 Amplifier

The audio amplifier shall be capable of interfacing with the sound source to transmit up to 130 dB SPL in the frequency range of 250 Hz to 8 kHz at least one (1) meter without input or output clipping. A single channel amplifier is sufficient for this testing method.

5.5.2 Power module

If using an externally polarized microphone, a power module will be required.

5.5.3 Data acquisition equipment

The signal gain shall be amplified to at least 10% of the dynamic range of the data acquisition system, and less than 90% of the dynamic range to avoid clipping. The pre-amplifier may be incorporated into the power module so long as the required signal gain is achieved.

5.5.4 Pre-amplifier

The received signal shall be amplified to at least 10% of the dynamic range of the data acquisition system. The pre-amplifier may be incorporated into the power module so long as the required signal gain is achieved.

6 Test conditions

6.1 Introduction

The conditions described in this section are required to ensure that the results of the eLD test are correlated with human performance.

6.2 Sound source and ATF orientation

The ATF shall be placed directly in front of the sound source such that the sound wave is perpendicular to the frontal plane of the ATF. The ATF shall be positioned on the traverse plane in a way such that the desired signal amplitude can be achieved at the reference point.

6.3 Test signals

The sound source shall be 2.5 cycles of a sine wave spanning 250 Hz to 8 kHz. The onset and offset of the sound source shall be tapered with a Tukey window using a window parameter of 0.1. Figure 6-1 provides an illustration of the test signal.

The test signal shall be presented to the ATF as at SPLs of 70 dB, 100 dB, and 130 dB in each octave band center frequency ranging from 250 Hz to 8 kHz. Data collection using this arrangement of SPLs and frequencies are required; however, other SPLs and frequencies may also

be measured, provided that they are uniformly spaced in Hz between 250 Hz and 8 kHz, and in dB between 70 dB and 130 dB, respectively.

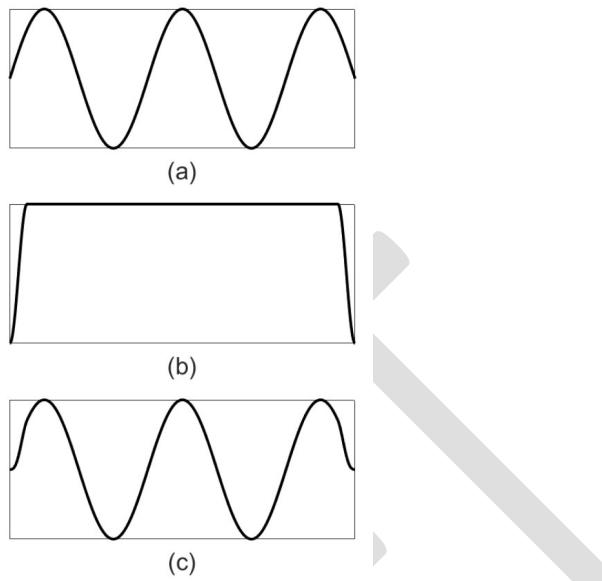


Figure 6-1. a) 2.5 cycles of a sine wave at a measurement center band frequency. (b) A Tukey window with cosine fraction 0.1. (c) The test signal formed by multiplying the signal in (a) with the Tukey window in (b).

6.4 Ambient conditions

The test procedures in this standard should be conducted with an ambient temperature between 50°F–90°F, and relative humidity between 10%–90%. Any measurements taken outside of this range shall be properly documented.

6.5 Placement of HPDs

The device under test shall be fitted on the ATF in accordance with the manufacturer's instructions and correspond to actual use. Furthermore, measurement of band force is recommended on all samples of earmuff style devices prior to testing. For further information on measuring band force, see Section 5.2 of ANSI/ASA S12.42-2010.

The eLD test methods are only valid for earplug and earmuff hearing protection devices. These methods have not been validated with systems incorporated into helmets.

6.6 Gain control of HPD

For active HPDs that provide an ambient listening capability, the unity gain setting as described in Annex C shall be used for all measurements. Data collection at the unity gain setting is required, however, other gain settings also may be measured.

7 Measurements

7.1 Introduction

This section describes the requirements and measurement procedures for performing the eLD test. Most sound sources do not have a flat frequency response (i.e. equal output across a range of frequencies). Therefore, a fixed voltage applied at different frequencies will not produce the same SPL. Consequently, to ensure that the sound source produces a constant SPL across the frequency band, a sound source calibration must be performed. Additionally, the test method utilities the changes in insertion loss as a function of frequency at a given SPL to determine the level dependent effects. As a result, open ear and occluded measurements must be acquired. Finally, to remain consistent in language this section clarifies common terms used to derive the level dependency metric.

7.2 Explanation of terms

Figure 7-1 illustrates common terms used throughout the remainder of this standard. The following sections aim to further explain each term and provide statistically relevant requirements.

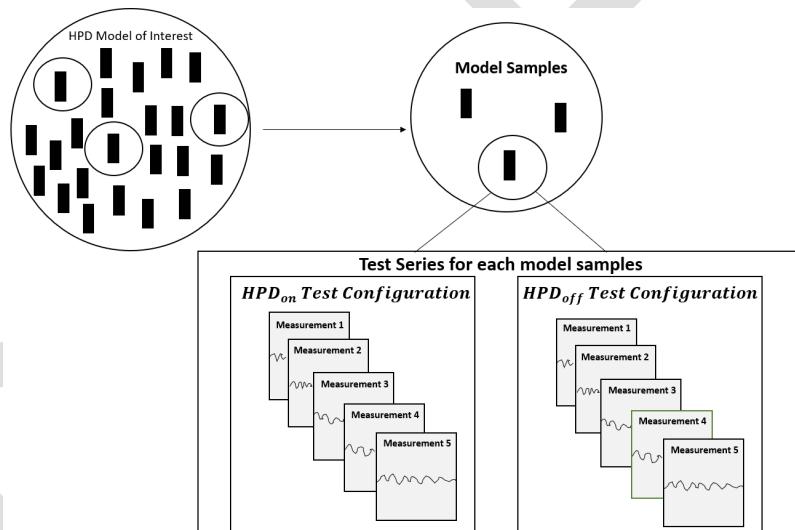


Figure 7-1. Flowchart of terms used in the described test protocol. Many individual units of an example HPD are shown in the top-left bubble; Three samples are chosen from this population for characterization, as shown in the top center bubble. The bottom graphs illustrate the test protocol for a single sample, where multiple iterations of chirp sweep responses are recorded for the sample at various spatial points.

7.2.1 Model sample

A model sample is a single instance of a subset of a HPD model of interest. At least three model samples are required to be tested to characterize the level dependent characteristics of the HPD model of interest.

7.2.2 Test series

A test series is a collection of five or ten measurements in each testing configurations, for a total of ten or twenty measurements per model sample. The different types of testing configurations are described in Section 7.4.

7.2.3 Measurement

A measurement refers to a single interval of data collection for a model sample. A measurement consists of a pressure vs time waveform, $P(t)$. The number of measurements in a given test configuration depends on the type of HPD. Five measurements shall be recorded for active devices and ten measurements shall be recorded for passive devices.

7.3 Requirements for measurements

7.3.1 Sampling frequency

The sampling rate is the number of samples of the signal per second required to reduce the continuous signal to a discrete signal. The signal acquired during this test shall be sampled at a minimum 44,100 samples per second (44.1 kHz).

7.3.2 Minimum signal-to-noise ratio

The signal to noise compares the output level of the occluded measurement (signal) to the output level of the open-ear (noise) measurement. The signal-to-noise ratio between these measurements shall be at least 6 dB within octave bands spanning the range of 250 Hz to 8 kHz. Additional stimulus repetitions may be collected to achieve this requirement via signal averaging. HPDs yielding less than 6 dB SNR shall be reported as inadequate for measurement.

7.3.3 Calibration method for sound source output

To determine the SPL incident upon the ATF, a calibration microphone shall be positioned at the reference point in the plane of the test fixture ears without the ATF present. The calibration microphone shall be orientated at 45 degrees to the incident acoustic wave.

Once the calibration microphone is in place, the gain of the power amplifier should be fixed such that the target SPLs can be reached. The power amplifier output level should not be changed once the calibration has begun. To obtain a calibrated source voltage for a target SPL at the reference point, the procedures in Table 7-1 shall be performed.

Table 7-1. Order of calibration measurements

Order	Instruction
1	Position the calibration microphone at the reference point.
2	Transmit five signals at the target SPL using a single octave band center frequency.
3	Record the average SPLs of each of the five (ten) signals.
4	Adjust the voltage of the speaker (upward or downward).
5	Repeat Steps 1–4 until the average SPL of the five (ten) signals measures within 0.25 dB of the target SPL.
6	Record and use the voltage that achieved the target SPL at that frequency.

Order	Instruction
7	Repeat Steps 1–6 for each octave band center frequency from 250 Hz to 8 kHz.
8	Repeat Steps 1–7 for each target SPL.

The microphone used for calibration of the test signal shall be a microphone with a calibrated sensitivity and frequency response from 250 Hz to 8 kHz. The calibration of the microphone shall be verified in the 24 months prior to use for testing according to this standard.

The microphone and preamplifier shall demonstrate no more than 2% total harmonic distortion over the complete range of peak SPLs up to the maximum level to be tested. The maximum peak SPL is 12 dB over the maximum root mean square SPL to be tested.

7.4 Test configurations

After the sound source has been calibrated, the testing space has been established, and all testing conditions are met, insertion loss measurements shall be performed. The insertion loss is a comparison of the signal arriving at the microphone within the ATF in the open-ear and occluded test configurations.

The time-domain pressure wave shall be recorded from each ear simulator at each calibrated SPL and octave band, as detailed in Section 6.3. A gain setting should be used which ensures that the received signals are above the noise level of the measurement system.

7.4.1 Open ear (O)

The open-ear measurements shall be performed with the test fixture's ear canals, free of any obstructions.

7.4.2 Occluded measurements (C)

The occluded measurements shall be performed with the HPD under test installed into the ATF in accordance with the manufacturer's instructions.

7.5 Sequence of measurements

The measurements described above shall be conducted in the sequence outlined in Table 7-2.

Table 7-2: Order of measurements

Order	Instruction
1	Set up the testing space while meeting all the testing conditions.
2	Turn on the instrumentation and wait the warm-up time recommended by the manufacturer.
3	Perform five for active (ten for passive) measurements in the open ear test configuration.
4	Repeat the measurements in the open ear test configuration for every SPL and frequency combination.

Order	Instruction
5	Install the HPD under test into both ears of the ATF.
6	Turn on the device under test (if applicable) and confirm gain setting.
7	Perform five (ten) measurements in the occluded test configuration.
8	Repeat the measurements in the occluded test configuration for every SPL and frequency combination.
9	Repeats Steps 1–8 for each model sample.
10	Repeat Steps 1–9 for each model of interest.

8 Data analysis

8.1 Introduction

This section describes the data processing and reduction techniques to generate the level dependency metric. The metric is based on SPL dependent changes of insertion loss. This section defines the metric, and provides instructions to compute the metric. The techniques described in this section have been verified to provide a metric that is correlated with human performance.

8.2 List of symbols

i: individual center frequency in an octave band

I: total number of bands in an octave band

j: individual sound pressure level

J: total number of presented sound pressure levels

O: open-ear configuration

C: occluded -ear configuration

L: left ear signal

R: right ear signal

N: total number of measurements

eLD: level dependent frequency response metric

eLD_{sample}: level dependent frequency response of an individual model sample

eLD_{model}: level dependent frequency response of an individual model

IL: insertion loss

Subscripts can include: i, j

SPL: measured or presented sound pressure level

Superscripts can include: p, O, C

Subscripts can include: j, j-1

RoC: rate of change in insertion loss

Subscripts can include: j

8.3 Methods for computation of level-dependent frequency response

8.3.1 Average measurements between ears in each configuration

As shown in the steps of Figure 8-1, compute the average of all N measured SPLs ($SPL_{i,j}^O$ and $SPL_{i,j}^C$) of the test signal presented at the j th SPL and i th octave band center frequency for the left and right ears in the open and occluded test configurations to obtain the average responses ($\overline{SPL}_{i,j}^O$ and $\overline{SPL}_{i,j}^C$).

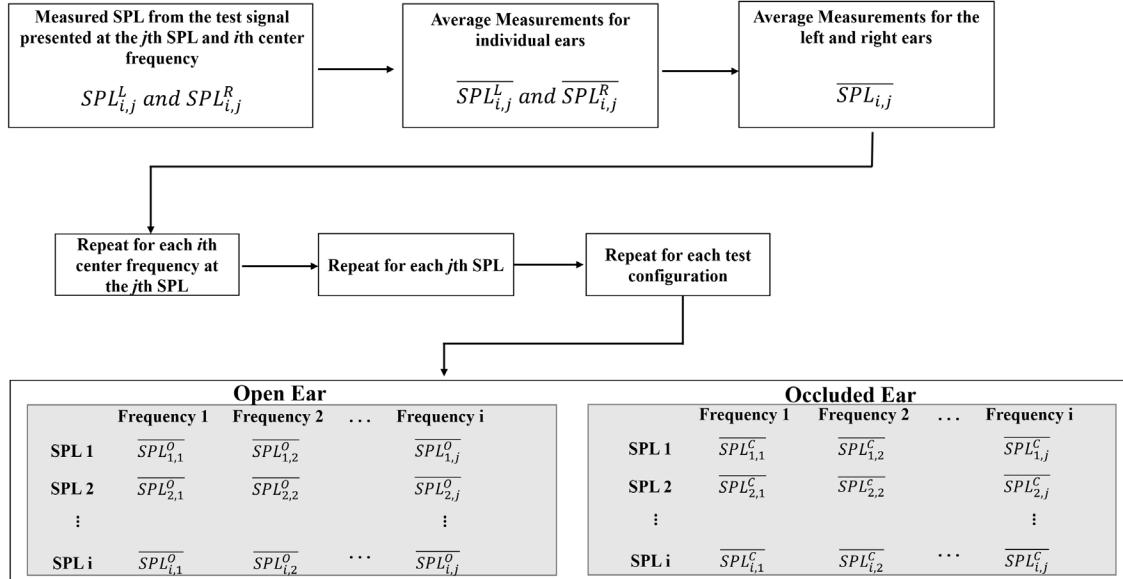


Figure 8-1. Flow diagram of the computations to average measurements for each source level and octave band in the eLD testing method.

8.3.2 Mean insertion loss

Figure 8-2 demonstrates the computations that shall be completed to calculate the mean insertion loss across the frequency spectrum at a given presented SPL.

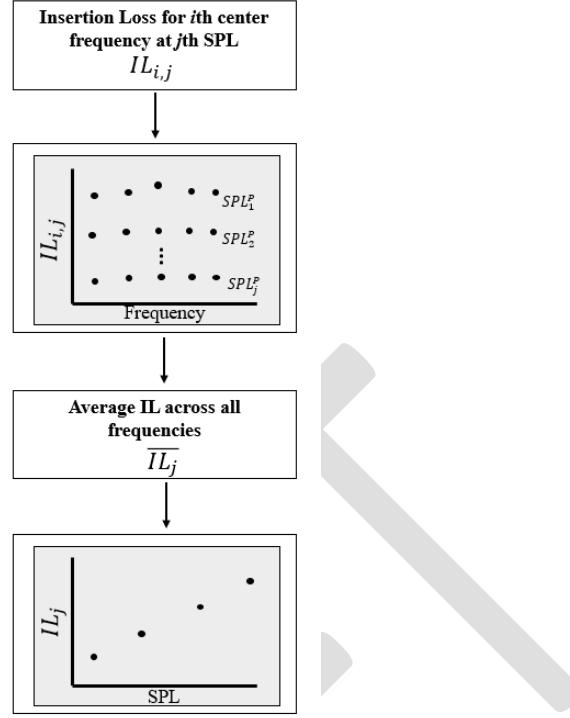


Figure 8-2. Flow diagram of the computations to calculate the mean IL with respect to source level in the eLD testing method.

Use Equation (1), and the average measurements from Section 8.3.1, to calculate the insertion loss. $IL_{i,j}$ is the arithmetic difference in decibels between the measured SPLs in the open and occluded test configurations for the i th octave band center frequency at the j th presented SPL. A positive $IL_{i,j}$ value indicates that the HPD is attenuating sound, while a negative $IL_{i,j}$ indicates that the HPD is amplifying the sound.

$$IL_{i,j} = \overline{SPL_{i,j}^O} - \overline{SPL_{i,j}^C} \quad (1)$$

The mean insertion loss, Equation (2), is the sum of the octave band insertion losses for all i , ($IL_{i,j}$) divided by the total number of octave bands, I .

$$\overline{IL}_j = \frac{\sum_i IL_{i,j}}{I} \quad (2)$$

\overline{IL}_j represents the total performance of a HPD across all frequencies at a given j th presented SPL.

8.3.3 Rate of change in insertion loss

As shown in Figure 8-3, the rate of change (RoC) describes the relationship between insertion loss and SPL across the frequency spectrum. The RoC term quantifies the average effect a change in SPL has on the amount of insertion loss provided by the HPD.

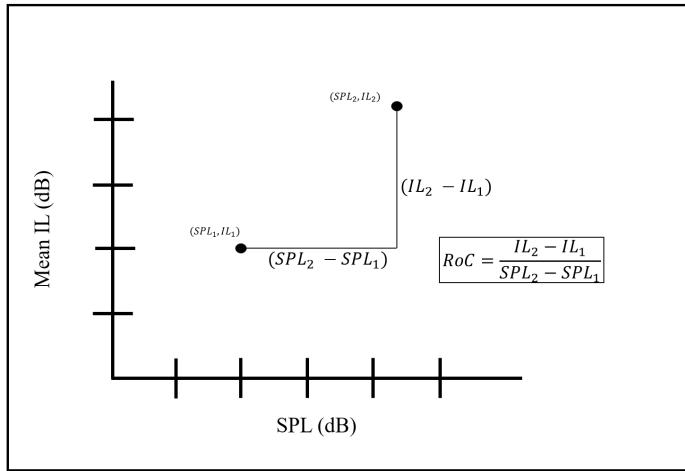


Figure 8-3. Illustration of the rate of change in IL and SPL.

Use Equation (3), and the mean insertion loss from Section 8.3.2 to calculate the square root of the rate of change. RoC is calculated by dividing the difference in mean insertion loss at a given sequential ascending order of presented SPLs (j and $j-1$) by the corresponding amount of change in presented SPL.

$$RoC_j = \sqrt{\frac{IL_j - IL_{j-1}}{SPL_j^P - SPL_{j-1}^P}} \quad (3)$$

8.3.4 Computation of eLD

As shown in Equation (4), the average RoC shall be calculated across each j th presented SPL using the results from Equation (3), and the total number of SPLs presented to ATF, J . The result shall be a single value for eLD that represents the level dependency characteristic of the model sample.

$$eLD_{sample} = \frac{\sum_j RoC_j}{J} \quad (4)$$

The average eLD from each model sample shall be averaged across the total number of samples tested, S , to determine the level dependency for the model of interest.

$$eLD_{model} = \frac{\sum_S eLD_{sample}}{S} \quad (5)$$

This single value that represents the level dependency of the HPD model of interest can be used to compare across different models of devices. An eLD value of zero indicates the HPD provides an equal amount of insertion loss regardless of the presented SPL. A positive eLD value demonstrates the HPD provides additional protection for higher level exposures relative to those at 70 dB. The higher the metric, the greater amount of protection is provided.

8.4 Uncertainty

The measurement of level dependency described in this standard has intrinsic uncertainties of varying degrees of severity. The estimation of uncertainty is beyond the scope of this standard. In lieu of an uncertainty calculation, Table 8-1 provides various sources of uncertainty in an order of influence each source has on the overall level of uncertainty. This table can be used as an ordered guide to identify areas to improve the results of measurements.

Table 8-1. Sources of uncertainty involved in the level dependent frequency response measurements

Description of Uncertainty Source
Fitting of HPDs
Speaker positioning
Signal-to-noise ratio
Reflections in test site
Sound source calibration
Ambient conditions

8.5 Reporting instructions

The test report shall include the following information.

- a) Reference to this standard.
- b) The brands/models/specification, describing the ATF used, including a description of the microphones and ear canal couplers installed and (if applicable) pinnae variant used.
- c) The temperature and relative humidity at which the tests were conducted.
- d) The type of HPD (e.g., earplug or earmuff), its brand/model name, and the number of model samples tested.
- e) The signal to noise ratio of each test series. HPDs with noise levels below 6 dB SNR shall be reported as inadequate for measurement.
- f) The level dependent metric of each model sample, the level dependent metric for the model of interest, comparison of level dependent metric between models of interest (if applicable).

Annex A

(informative)

Sound Isolation Box – General Description

A.1 Dual walled sound isolation box

A sound isolation box that can be used for level-dependency frequency response measurements is shown in Figure A-1.

This dual-walled sound isolation box is capable of:

1. Isolating high amplitude exposures from making the test room unsafe, and
2. Preventing unwanted noise from reaching the test article from the laboratory environment.

The sound isolation box shown in Figure A-1 utilizes a combination of design parameters in order to achieve high levels of acoustic and vibration isolation. The structure of the box should be such that transmission of sound in the frequency range 100 Hz–10,000 Hz is minimized. A double-walled “box-in-a-box” chamber with different inner and outer wall thicknesses provides sufficient isolation when coupled with acoustic damping materials.

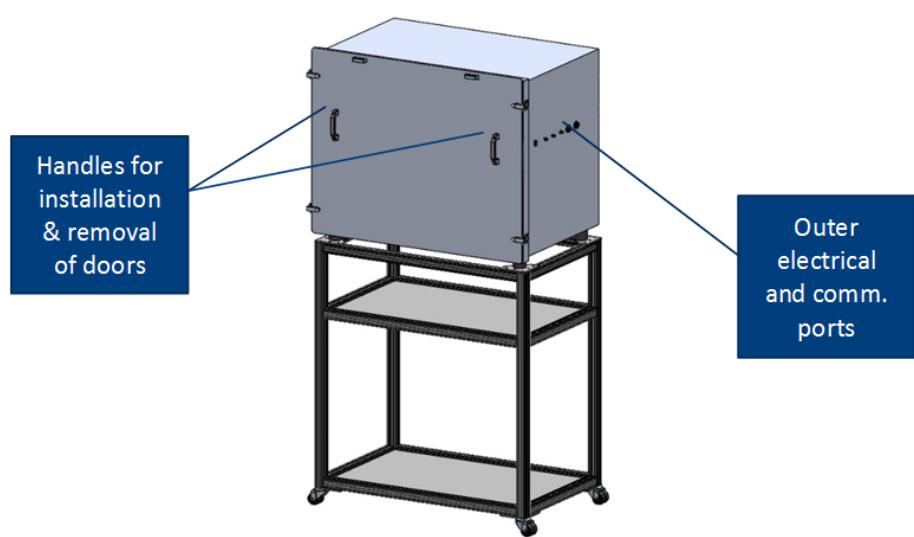


Figure A-1. A sound isolation box with solid doors can be used for level dependent frequency response testing of electronics hearing protection devices.

To prevent coincidence frequencies from causing a transmission leak through the box, two different wall thicknesses were selected for the inner box (1/4") and outer box (3/8"). The inside of the outer box is lined with 1/8"-thick mass loaded vinyl. The inner box is also lined with 1/8" mass-loaded vinyl on the outside and a combination of dual-density, 1.25"-thick Sonic Barrier foam and 2"-tall acoustic wedge foam on the inside to attenuate reflected sound within the box (Figure A-2).

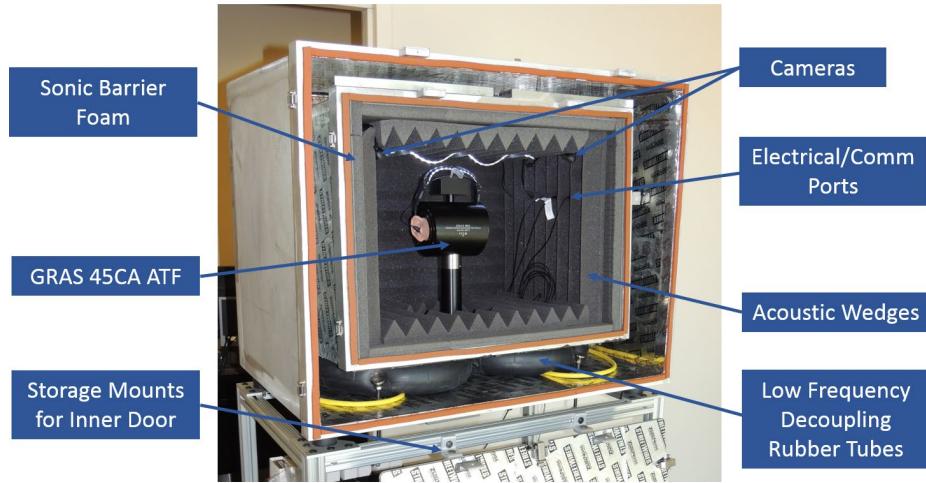


Figure A-2. Inner view of the dual-walled sound isolation box with the doors removed for full interior view. Isolation box Includes low frequency decoupling rubber tube, G.R.A.S. 45CA ATF, electrical/communication ports, camera, and all noise-reduction material.

For vibration isolation, the inner box is floated on at least one rubber inner tube, or vibration-isolating legs should be used. Electrical bulkhead connectors should be used to ensure proper acoustic and electrical isolation from the environment (Figure A-3).

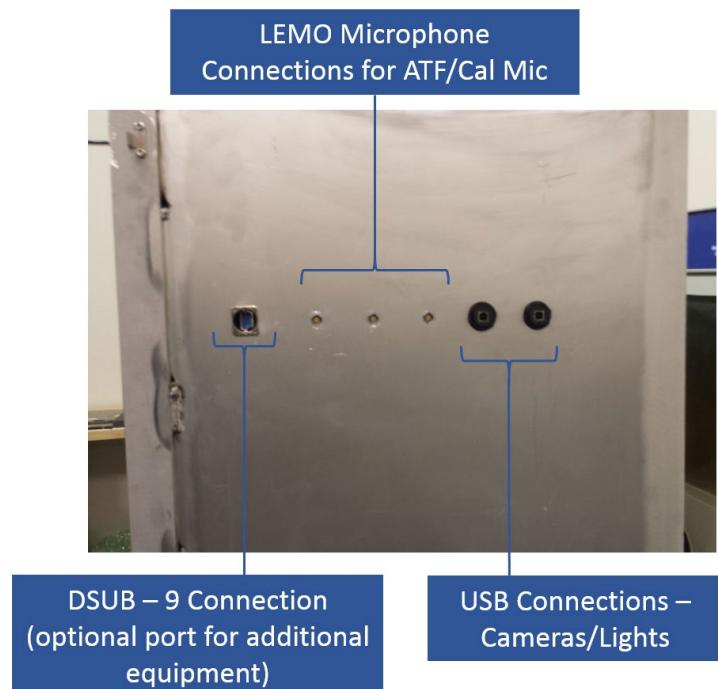


Figure A-3. The exterior of the outer box showing the electrical/communications connections for instrumentation.

Annex B
(informative)
Acoustic Test Fixture (ATF)

B.1 G.R.A.S. 45CA acoustic test fixture

An ATF that meets the requirements for this test method is the G.R.A.S. 45CA. The G.R.A.S. 45CA test fixture shown in Figure B-1 can accommodate passive and electronic hearing protection devices of over-the-ear, behind-the-ear, and in-the-ear types. When coupled with ear simulators, pinnae, and G.R.A.S. 40 BP microphones, the combined 45CA measurement system can exhibit noise floors of approximately 16–17 dBA after 100 Hz to 10 kHz pass-band filtering. The 40 BP microphones require external polarization and pre-amplification to achieve the specified dynamic ranges. For this system, a G.R.A.S. 12AQ two-channel power module with signal conditioning can provide external polarization and has a variable gain setting ranging from -10 dB to +70 dB.

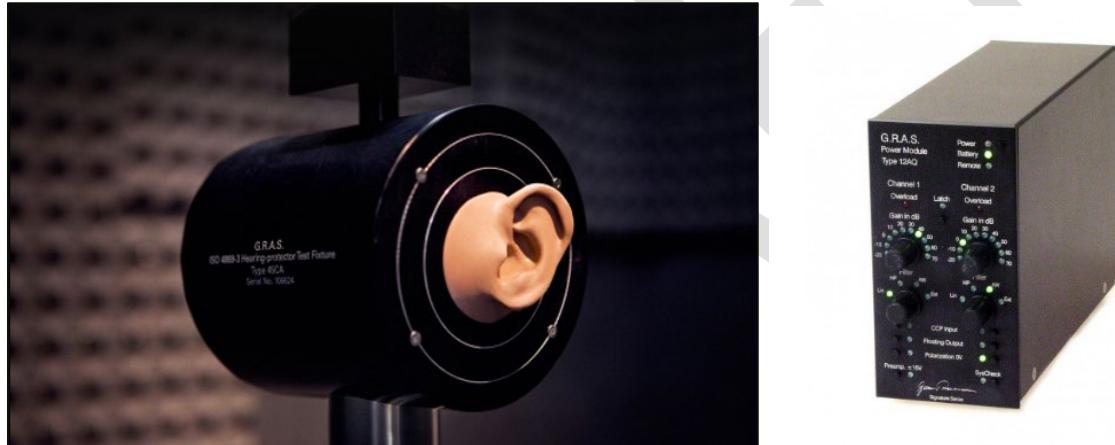


Figure B-1. Left: The G.R.A.S. 45CA ATF with pinna suitable for level dependent testing of electronic hearing protection devices. Right: A G.R.A.S. 12AQ preamplifier with variable gain.

Annex C
(informative)
Measurement of unity gain

C.1 Gain setting for active head-worn devices

The gain setting of active head-worn devices shall be measured using an acoustic test fixture that meets the requirements of this standard. Calibrate a speaker/amplifier combination to output a 1-kHz tone at an amplitude of 70 dB as measured in one ear of the ATF. The output level of the speaker/amplifier combination should be adjusted until 70 ± 0.5 dB is measured. This shall be designated the open-ear level.

The active HPD shall then be placed on the ATF and the gain adjusted until the frontally incident sound field level most closely matches the previously measured open-ear level. Both the open-ear level and the level under the HPD are intended to be measured at the ear simulator microphone in the ATF.

Standard eSN

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Prepared for:
US Army Medical Research and Materiel Command

Prepared by:
Applied Research Associates, Inc.
Ted Argo, PhD
7921 Shaffer Parkway
Littleton, CO 80127
Phone: 303.795.8106
targo@ara.com

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DRAFT STANDARD – eSN

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1. Scope

This standard defines an electromechanical method for the measurement of the self-noise of active (electronic) hearing protection devices (HPDs). This standard is intended to be part of a series of standards that utilizes electromechanical methods to provide uniform metrics to evaluate the performance of HPDs.

HPDs that incorporate active electronics generate noise due to gain, digital clocks, or other components in the electronic circuits. The electromechanical test method of the self-noise (eSN) measures the sound pressure level (SPL) of the noise, commonly described as a “hiss,” and is used to quantify potential impact on the user’s ability to hear relatively low-level sounds. This measurement is performed using an acoustic test fixture in an environment with a controlled noise floor. The waveform of the self-noise emanating from the HPD electronics is recorded and the signal is analyzed in the frequency domain to provide information on the spectral content of the self-noise. High (loud) self-noise levels can be irritating, distracting, and reduce situational awareness by decreasing the user’s ability to hear low-level sounds (low-level speech, footsteps, etc.). Passive devices do not exhibit electronic self-noise and, thus, are excluded.

This standard develops a hearing protection evaluation method resulting in a single number metric to describe self-noise that can discriminate relative performance between devices, provide a basis to develop performance requirements, and maintain quality assurance of the devices over time. This standard establishes uniform instrumentation requirements, procedures for the measurement of self-noise, and develops the computation to generate the single value metric that is correlated with human performance.

This standard is not intended to replace current standards or the use of human subjects to evaluate HPDs; rather, this standard focuses on supplementing current methods by providing an evaluation tool to characterize a dimension of HPD performance not addressed by current standards.

2. Normative references

The following referenced documents are useful for the application of this standard.

ANSI S1.1, *American National Standard Acoustical Terminology*.

ANSI S1.11, *Specification for Octave-Band and Fractional-Octave-Band Analog and Digital Filters*.

ANSI/ASA S3.20, *American National Standard Bioacoustical Terminology*.

ANSI/ASA S3.25, *American National Standard for an Occluded Ear Simulator*.

ASTM D2240-05, *Standard Test Method for Rubber Property – Durometer Hardness*.

ANSI/ASA S12.42, *Methods for the Measurement of Insertion Loss of Hearing Protection Devices in Continuous or Impulsive Noise Using Microphone-in-Real-Ear of Acoustic Test Fixture Procedures*.

3. Terms and definitions

For the purposes of this standard, the terms and definitions given in ANSI S1.1, ANSI S3.20, and the following apply.

3.1 acoustic test fixture (ATF). An inanimate device that approximates certain physical characteristics and dimensions of a representative human head, pinnae, and/or ear canals and is used for measuring the insertion loss of a hearing protection device.

3.2 active hearing protection device. A hearing protection device that contains electronic components including transducers (i.e., speakers and microphones) to control the transmission of sound into the ear canal.

3.3 earmuff (over-the-ear HPD). A hearing protection device usually comprised of a headband and earcups with a soft cushion to seal against the head, intended to fit against the pinna (supra-aural) or the sides of the head enclosing the pinna (circumaural). The earcups may also be held in position by attachment arms mounted on a hard hat or helmet.

3.4 earplug (in-the-ear HPD). A hearing protection device that is inserted into or that caps the ear canal.

3.5 electromechanical evaluation method. A laboratory sensor-based test to evaluate HPDs without the use of human subjects.

3.6 recording system noise. The sound produced by the electronic elements of recording systems (e.g., microphones and data acquisition systems). In the context of this standard, the recording system noise is a disturbance in the signal or quantity of interest. This is distinct from test space noise, and self-noise.

3.7 test space noise. The sound produced by acoustic sources present in the test space (e.g., equipment fans and HVAC system). In the context of this standard, the test space noise is a disturbance in the signal or quantity of interest. This is distinct from recording system noise and self-noise.

3.8 gain control of hearing protection device. Amount of amplification provided by an active HPD that suppresses or amplifies noise from the surrounding environment.

3.9 hearing protection device (HPD). A device, also called a hearing protector, worn to reduce the sound level in the ear canal.

3.10 measurement. A single sound pressure versus time waveform interval of data collection.

3.11 model sample. A single instance of a subset of a HPD model of interest.

3.12 noise floor. The lowest level of noise that can be measured by a device; describes the level of background noise that is present in the system without the signal of interest present.

3.13 self-insertion loss. The passive insertion loss of the ATF when measured with a simulator of a near-ideal HPD, normally a metal plug or cup that is machined to seal the ear canal. It represents the acoustic leakage through the flanking pathways of the ATF.

3.14 self-noise. The sound, commonly described as a “hiss,” produced by the electronic elements of active hearing protection devices. In the context of this standard, the self-noise is the signal or quantity of interest. This is distinct from recording system noise attributable to other sources and test space noise.

3.15 sound isolation box. Enclosure engineered to isolate the high-amplitude acoustic exposure from reaching the laboratory and prevent unwanted noise in the laboratory from reaching the test article.

3.16 test series. A collection of five measurements in each of the three testing configurations.

4. Applicability of test methods

The procedures outlined in this standard are suitable for assessing the electronic self-noise emitted from active HPDs. The procedures are appropriate for testing active earplugs and earmuffs. Passive devices do not exhibit electronic self-noise and, therefore, are not applicable to this standard.

The standard derives a single value metric, correlated with human performance, to determine the effects of self-noise on the user’s ability to maintain situational awareness. Self-noise is an important determining factor when selecting operationally appropriate HPDs so the user can simultaneously protect their hearing and identify potential low-level signals of interest. Perception of such signals may be hampered by self-noise, particularly if present in a frequency band(s) with a higher level of self-noise. As a result, the frequency-dependent characteristics of the noise must be analyzed.

The self-noise metric defined in this standard provides additional insight into overall HPD performance to include aspects related to situational awareness. To fully characterize the operational effectiveness of HPDs, further performance specifications and testing methods must be completed in conjunction with existing standards.

5. Requirements of the test facility

5.1 Introduction

The eSN test method requires low-level SPL measurements; therefore, the noise floor of the testing environment is an important requirement. As described in this standard, the noise floor of the testing environment must be lower than the device being tested. To quantify the effects of self-noise, there are specific conditions for the testing space that must be considered.

5.2 Noise floor

The maximum allowable noise floor of the testing space, including ambient noise, shall be 20 dB SPL from 80 Hz to 12.5 kHz. Any deviation from this requirement shall be documented in the test

report. A noise floor below this level is desirable to allow for accurate measurements of devices with a lower self-noise.

Typical laboratory space has an array of different noise sources that could interfere with the self-noise measurements including inherent recording system noise (e.g., microphones and data acquisition systems) and test space noise (e.g., equipment fans and HVAC systems). These sources may obscure the self-noise measurements for low-noise devices and add to the self-noise measured for higher-noise devices.

Commercially available electronic hearing protection devices are known to have a self-noise in the 20–34 dB range. Consequently, it is required to attain a system noise below 20 dB SPL. A suitable low-noise floor measurement system could utilize a “box-in-a-box” sound isolation chamber (as detailed in Annex A), an anechoic chamber, or any other space that meets this requirement.

5.3 Acoustic test fixture

Any ATF that has a self-insertion loss of at least 60 dB from 80 Hz to 12.5 kHz and is able to accommodate all electronic hearing protection devices shall be used during the test procedures described in this standard. The ATF shall have circumaural bases of sufficient diameter that can fully support earmuff cushions.

The measurements involved in this standard record the response inside of the ATF from the properly inserted / sealed active HPD. As a result, there are no incident waves, and a realistic head surrogate is not required. A representative human ear and ear canal with dimensions such that a variety of earplugs and earmuffs can be accurately tested is required.

The ATF shall be equipped with proper instrumentation to perform the measurements while maintaining the required noise floor. An ATF that meets the requirements described in this section is described in Annex B.

5.3.1 Microphones

Microphones shall be positioned inside of the ATF ear canals and have a pressure sensitivity within ± 1 dB in the frequency range 100 Hz to 10 kHz relative to the sensitivity at 1 kHz.

The microphone shall be sufficient to meet the noise floor requirements in Section 5.2 and the signal-to-noise ratio (SNR) requirements in Section 7.3.2. Microphone selection is an important factor that contributes to the ability to capture reliable self-noise measurements. If the sensitivity and dynamic range of the microphone cannot meet the requirements, the measured value of self-noise can be mistaken for the recording system noise of the microphone itself.

5.3.2 Ear simulator, coupler, and flesh simulator

Any ear simulator, coupler, and flesh simulator combination that represents the dimensions of a human ear may be used. The anthropomorphic combination must be compatible with the ATF base, meet the ATF requirements, and permit a proper placement of the HPD in accordance with the manufacturer’s instructions.

5.4 Instrumentation

5.4.1 Power module

If using an externally polarized microphone, a power module will be required.

5.4.2 Data acquisition equipment

An instrument capable of low-noise data acquisition with at least two channels (left ear and right ear) shall be used in all measurements. Channel-to-channel isolation is recommended to eliminate crosstalk between the left and right ear microphones, and to achieve low noise levels.

The data acquisition system shall sample at a minimum sampling rate of 44,100 samples per second (44.1 kHz) and be able to resolve voltages at a minimum of 16 bits full scale.

5.4.3 Pre-amplifier

The signal gain shall be amplified to at least 10% of the dynamic range of the data acquisition system. The pre-amplifier may be incorporated into the power module so long as the required signal gain is achieved.

6. Test conditions

6.1 Introduction

The conditions described in this standard are essential to ensure that the results of the eSN test are correlated with human performance. If these conditions cannot be achieved, the outcome of the test may be affected.

6.2 Ambient conditions

The test procedures in this standard should be conducted with an ambient temperature between 50°F–90°F, and relative humidity between 10%–90%. Any measurements taken outside of this range shall be properly documented.

6.3 Placement of HPDs

The device under test shall be fitted on the ATF in accordance with the manufacturer's instructions and correspond to actual use. Furthermore, measurement of band force is recommended on all samples of earmuff style devices prior to testing. For further information on measuring band force, see Section 5.2 of ANSI/ASA S12.42-2010.

The eSN test methods are only valid for earplug and earmuff hearing protection devices. These methods have not been validated with systems incorporated into helmets.

6.4 Gain control of HPD

For active HPDs that provide an ambient listening capability, the unity gain setting as described in Annex C shall be used for all measurements. Data collection at the unity gain setting is required; however, other gain settings may also be measured.

7. Measurements

7.1 Introduction

This section describes the requirements and measurement procedures for performing the electromechanical self-noise test. To directly measure self-noise, measurements must be made to characterize the system in terms of recording system noise of the instrumentation and sources of test space noise. To remain consistent in language, this section clarifies common terms used to derive the self-noise metric.

7.2 Explanation of terms

Figure 7-1 illustrates common terms used throughout the remainder of this standard. The following sections aim to further explain each term and provide statistically relevant requirements.

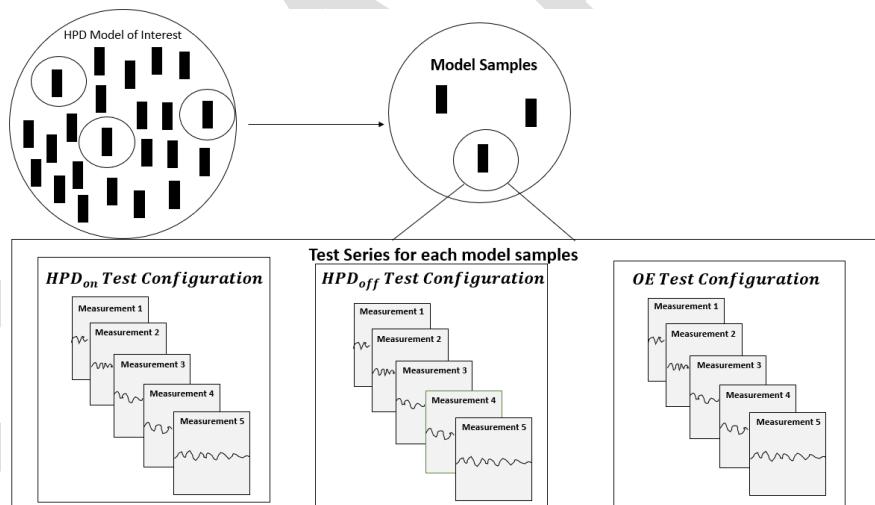


Figure 7-1. Flowchart of terms used in the described test protocol. Many individual units of an example HPD are shown in the top-left bubble; three samples are chosen from this population for characterization, as shown in the top-center bubble. The bottom graphs illustrate the test protocol for a single sample, where multiple iterations of chirp sweep responses are recorded for the sample at various spatial points.

7.2.1 Model sample

A model sample is a single instance of an HPD model of interest. At least three model samples are required to be tested to characterize the self-noise of the HPD model of interest.

7.2.2 Test series

A test series is a collection of five measurements in each of the three testing configurations, for a total of 15 measurements per model sample. The different types of testing configurations are described in Section 7.5.

7.2.3 Measurement

A measurement refers to a single interval of data collection for a model sample. A measurement consists of a pressure versus time waveform, $P(t)$. Five measurements shall be recorded for each test configuration.

7.3 Measurement requirements

7.3.1 Sampling frequency

The sampling rate is the number of samples of the signal per second required to reduce the continuous signal to a discrete signal. The self-noise signal acquired during this test shall be sampled at a minimum 44,100 samples per second (44.1 kHz).

7.3.2 Minimum signal-to-noise ratio

The signal to noise compares the output level of the HPD_{On} (signal) to the output level of the HPD_{Off} (noise) test configurations. The signal-to-noise ratio between these configurations shall be at least 6 dB within each octave band center frequency. HPDs with noise levels below 6 dB SNR shall be reported as below the noise floor of the recording system. Further details of the measurements that comprise the signal-to-noise calculation are defined in Sections 7.5.2 and 7.5.3.

7.4 Noise floor

The noise floor shall be measured with an external microphone in an empty testing space without the ATF or HPD present. The purpose of this measurement is to verify all measurements are being conducted with the appropriate noise floor, as described in Section 5.2. This is a baseline measurement and is only necessary to establish initial conditions or verify the standard is met after any changes which may affect the noise floor of the testing space.

7.5 Test configurations

After the testing space is established, and all testing conditions are met, as described in Sections 5 and 6, measurements shall be performed in the open-ear and occluded test configurations.

7.5.1 Open ear (OE)

The open-ear measurements shall be performed with the test fixture's ear canals free of any obstructions. The open-ear measurement is necessary to quantify recording system noise (N_{elc}) and other test space noise (N_{env}). This measurement defines changes to the noise floor of the testing space introduced by the ATF and other instrumentation.

$$OE = N_{elc} + N_{env} \quad (1)$$

7.5.2 Occluded off (HPD_{off})

The occluded-off measurement shall be conducted with the HPD installed into the ATF with the electronics powered off. It is assumed any test space noise will be blocked by the HPD and, as a result, the only source of background noise present in the system originates from the electronics of the recording systems.

$$HPD_{off} = N_{elc} \quad (2)$$

7.5.3 Occluded on (HPD_{on})

The occluded-on measurement shall be performed with the HPD installed into the ATF with the electronics powered on. This measurement captures the recording system noise (N_{elc}), the test space noise (N_{env}), and the self-noise of the HPD (N_{HPD}).

$$HPD_{on} = N_{elc} + N_{env} + N_{HPD} \quad (3)$$

In subsequent calculations, the self-noise is calculated by comparing the OE and HPD_{on} measurements. The signal-to-noise ratio is calculated by comparing HPD_{on} to the HPD_{off} measurement, as described in Section 7.3.2.

7.6 Sequence of measurements

The measurements described above shall be conducted in the following sequence.

Table 7-1. Order of measurements

Order	Instruction
1	Set up the testing space while meeting all of the testing conditions
2	Perform the noise floor measurement
3	Turn on the instrumentation and wait the warm-up time recommended by the manufacturer
4	Perform the open-ear measurement (OE)
5	Install the device into the ATF with the electronics turned OFF
6	Perform the occluded-off measurement (HPD_{off})
7	Install the device into the ATF with the electronics turned ON
8	Perform the occluded-on measurement (HPD_{on})
9	Repeats Steps 4–8 for each model sample
10	Repeat Steps 4–9 for each model of interest

8. Data analysis

8.1 Introduction

This section describes the data processing and reduction techniques to provide the self-noise of active HPDs. The self-noise is deduced from the time waveform recordings of each measurement. These recordings are analyzed in octave bands. The techniques described in this section have been verified to provide a self-noise metric that is indicative of human performance.

8.2 List of symbols

i: individual octave band number

I: the total number of octave bands

s: individual model sample

S: the total number of model samples

f: frequency

subscripts can include: *i*

RMS: root-mean-square

P(t): a single measurement pressure vs time waveform result

subscripts can include: *i*

P_{RMS}(f_i): RMS value for each octave band

x-bar above symbol indicates the arithmetic mean or average

superscripts can include: *HPD_{ON}*, *HPD_{OFF}*, *OE*

σ(f_i): standard deviation of the RMS value for each octave band

x-bar above symbol indicates the arithmetic mean or average

eSN(f_i): self-noise for each octave band

subscripts can include: RMS, dB

SNR(f_i): signal to noise ratio for each octave band

subscripts can include: RMS, dB

eSN_{sample}: self-noise of an individual model sample

SNR_{sample}: signal-to-noise ratio for an individual model sample

eSN_{model}: self-noise of an individual model

8.3 Computation self-noise

The following steps are used to determine the self-noise of active HPDs. Conceptually, as shown in Equation (4), the computation of self-noise involves simple subtraction between the open-ear measurement (*OE*) and the occluded-on measurement (*HPD_{on}*); however, to provide a more

comprehensive metric to compare devices, computations must be completed with the data gathered from each measurement, configuration, series, and sample.

$$eSN = HPD_{on} - OE = (N_{elc} + N_{env} + N_{HPD}) - (N_{elc} + N_{env}) = N_{HPD} \quad (4)$$

8.3.1 Computations for measurement

A measurement consists of a pressure versus time waveform, $P(t)$. The following computations shall be performed to each of the measurements for the OE , HPD_{on} , and HPD_{off} testing configurations. The results of these computations shall be the root-mean-square (RMS) value at each center frequency of the octave bands. An overview of the computations required to process each measurement is described in Figure 8-1.

- Apply an octave band filter to the $P(t)$, in accordance with ASA/ANSI S1.11, to obtain $P_i(t)$ the filtered signal for the i th octave band center frequency.
- Compute the RMS of each $P_i(t)$ to obtain an RMS value for each octave band, $P_{RMS}(f_i)$.

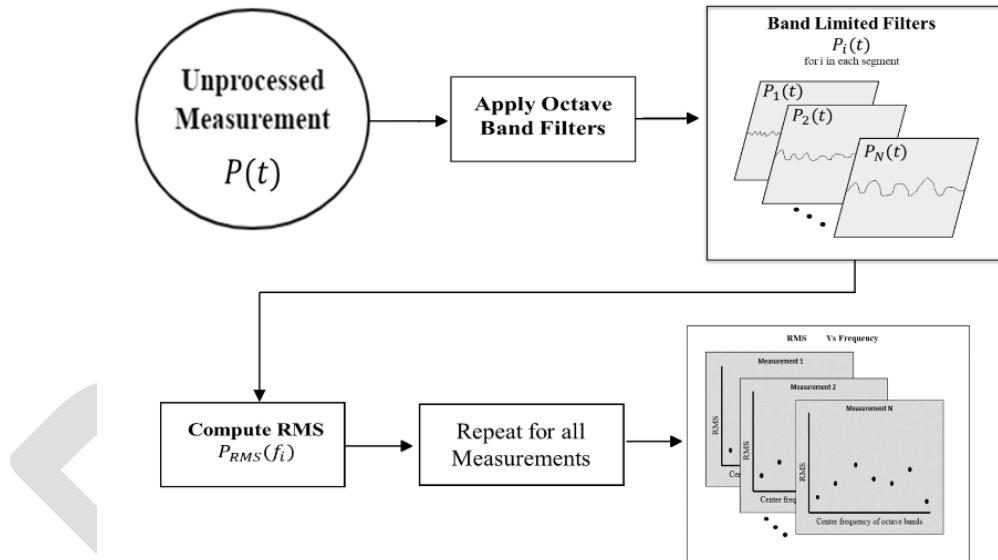


Figure 8-1. Flow diagram of the computations for measurements in the eSN testing method

8.3.2 Computations for test configuration

The average and standard deviation P_{RMS} shall be calculated at each octave band frequency for all measurements of the test configurations: OE , HPD_{on} , and HPD_{off} , as shown in Figure 8-2.

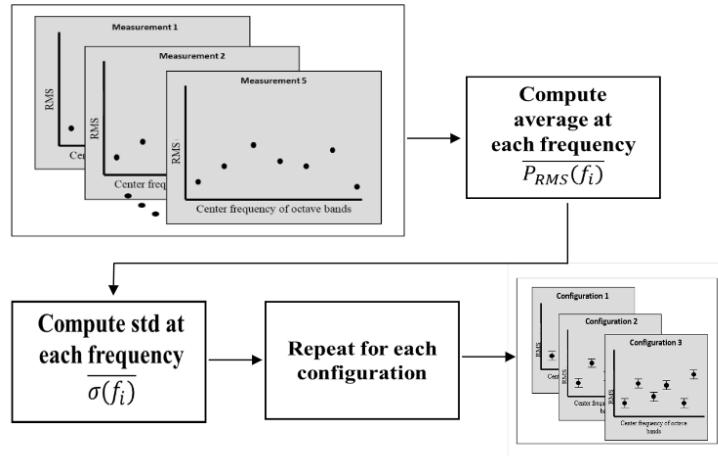


Figure 8-2. Flow diagram of the computations of test configurations in the eSN testing method

8.3.3 Computations for test series

Use Equations (5) and (6), and the results from Section 8.3.2, to calculate eSN and SNR at each center frequency of the octave band for the test series.

$$eSN_{RMS}(f_i) = \overline{P_{RMS}^{HPPD_{ON}}(f_i)} - \overline{P_{RMS}^{OE}(f_i)} \quad (5)$$

$$SNR_{RMS}(f_i) = \overline{P_{RMS}^{HPPD_{ON}}(f_i)} - \overline{P_{RMS}^{HPPD_{OFF}}(f_i)} \quad (6)$$

At this point, the eSN RMS values shall be converted to sound pressure levels in dB.

As shown in Equations (7) and (8), the average eSN and SNR shall be calculated across each octave band frequency using the eSN in dB results from equations (5) and (6), and the total number of octave bands, I . The result shall be a single value for eSN and SNR that represents the self-noise of the model sample.

$$eSN_{sample} = \frac{\sum_i eSN_{dB}(f_i)}{I} \quad (7)$$

$$SNR_{sample} = \frac{\sum_i SNR_{dB}(f_i)}{I} \quad (8)$$

8.3.4 Computation for model samples

The average eSN from each model sample shall be averaged across the total number of samples tested, S , to determine the self-noise for the model of interest. This single value that represents the self-noise of the HPD model of interest can be used to compare levels of self-noise across different models of devices. A device with a larger eSN metric will have more self-noise than a device with a smaller metric.

$$eSN_{model} = \frac{\sum_S eSN_{sample}}{S} \quad (9)$$

8.4 Uncertainty

The measurement of self-noise described in this standard has intrinsic uncertainties of varying degrees of severity. The estimation of uncertainty is beyond the scope of this standard. In lieu of an uncertainty calculation, Table 8-1 provides various sources of uncertainty in an order of influence each source has on the overall level of uncertainty. This table can be used as an ordered guide to identify areas to improve the results of measurements.

Table 8-1. Sources of uncertainty involved in the self-noise measurements

Description of Uncertainty Source
Fitting of HPDs
Signal-to-Noise Ratio
Reflections in Test Site
Sound Source Calibration
Ambient Conditions

8.5 Information to be included in the test report

The test report shall include the following information.

- a) Reference to this standard.
- b) The noise floor of the test space noise and the recording system noise.
- c) The dynamic range, frequency response, and sensitivity of the microphone.
- d) The brands/models/specification describing the ATF used, including a description of the microphones and ear canal couplers installed, and (if applicable) pinnae variant used.
- e) The temperature and relative humidity at which the tests were conducted.
- f) The type of HPD (e.g., earplug or earmuff), its brand/model name, and the number of model samples tested.
- g) The signal-to-noise ratio of each test series. HPDs with noise levels below 6 dB SNR shall be reported as inadequate for measurement.
- h) The self-noise of each model sample, the self-noise for the model of interest, and comparison of self-noise between models of interest (if applicable).

Annex A

(informative)

Sound Isolation Box – General Description

A.1 Dual walled sound isolation box

A sound isolation box that can be used for self-noise measurements is shown in Figure A-1.

This dual-walled sound isolation box is capable of:

1. Isolating high amplitude exposures from making the test room unsafe, and
2. Preventing unwanted noise from reaching the test article from the laboratory environment.

The sound isolation box shown Figure A-1 utilizes a combination of design parameters in order to achieve high levels of acoustic and vibration isolation. The structure of the box should be such that transmission of sound in the frequency range 100 Hz–10,000 Hz is minimized. A double-walled “box-in-a-box” chamber with different inner and outer wall thicknesses provides sufficient isolation when coupled with acoustic damping materials.

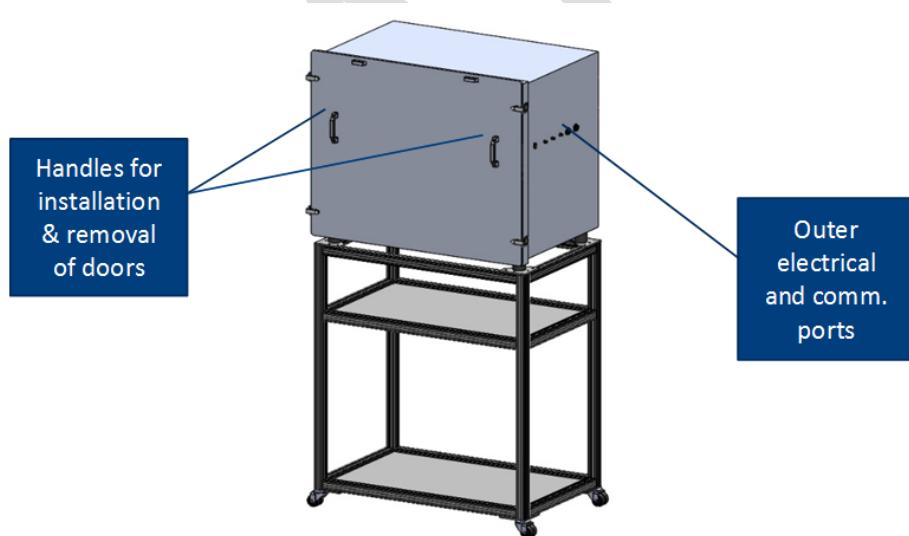


Figure A-1. A sound isolation box with solid doors can be used for self-noise testing of electronics hearing protection devices

To prevent coincidence frequencies from causing a transmission leak through the box, two different wall thicknesses were selected for the inner box (1/4") and outer box (3/8"). The inside of the outer box is lined with 1/8"-thick mass loaded vinyl. The inner box is also lined with 1/8" mass-loaded vinyl on the outside and a combination of dual-density, 1.25"-thick Sonic Barrier foam and 2"-tall acoustic wedge foam on the inside to attenuate reflected sound within the box (Figure A-2).

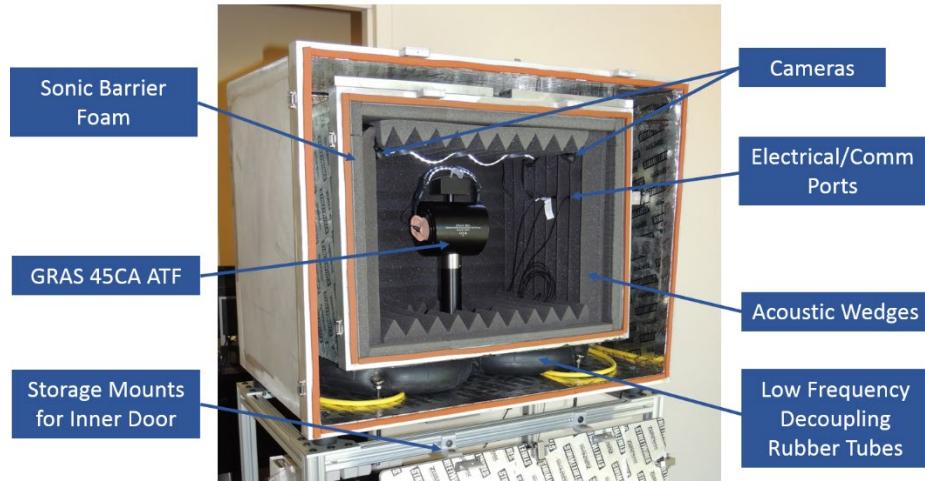


Figure A-2. Inner view of the dual-walled sound isolation box with the doors removed for full interior view. Isolation box includes low-frequency decoupling rubber tube, G.R.A.S. 45CA ATF, electrical/communication ports, camera, and all noise-reduction material.

For vibration isolation, the inner box is floated on at least one rubber inner tube, or vibration-isolating legs should be used. Electrical bulkhead connectors should be used to ensure proper acoustic and electrical isolation from the environment (Figure A-3).

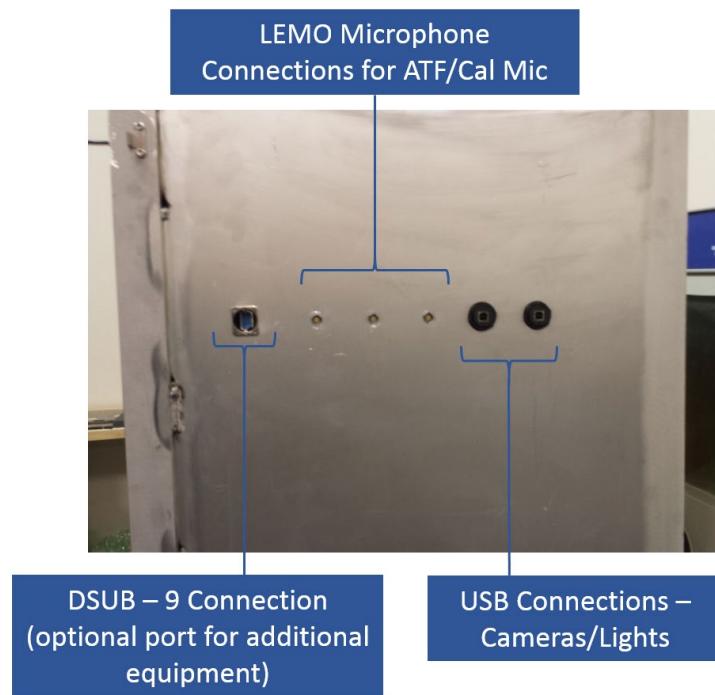


Figure A-3. The exterior of the outer box showing the electrical/communications connections for instrumentation

Annex B

(informative)

Acoustic Test Fixture (ATF)

B.1 G.R.A.S. 45CA Acoustic Test Fixture

An ATF that meets the requirements for this test method is the G.R.A.S. 45CA. The G.R.A.S. 45CA test fixture shown in Figure B-1 can accommodate passive and electronic hearing protection devices. When coupled with ear simulators, pinnae, and G.R.A.S. 40AP microphones, the combined 45CA measurement system can exhibit noise floors of approximately 16–17 dBA after 100 Hz to 10 kHz pass-band filtering, which is suitable for self-noise testing. The 40AP microphones require external polarization and pre-amplification to achieve the specified dynamic ranges. For this system, a G.R.A.S. 12AQ two-channel power module with signal conditioning can provide external polarization and has a variable gain setting ranging from -10 dB to +70 dB.



Figure B-1. Left: The G.R.A.S. 45CA ATF with pinna suitable for self-noise testing of electronic hearing protection devices; Right: A G.R.A.S. 12AQ preamplifier with variable gain

Annex C
(normative)

Measurement of Unity Gain

C.1 Gain setting for active head-worn devices

The gain setting of active head-worn devices shall be measured using the same acoustic test fixture used for self-noise testing. Calibrate a speaker/amplifier combination to output a 1 kHz tone at an amplitude of 70 dB as measured in one ear of the ATF. The output level of the speaker/amplifier combination should be adjusted until 70 ± 0.5 dB is measured. This shall be designated the open-ear level.

The active HPD shall then be placed on the ATF and the gain adjusted until the frontally incident sound field level most closely matches the previously measured open-ear level. The open-ear level and the level under the HPD are intended to be measured at the ear simulator microphone in the ATF.

RECOMMENDATIONS FOR IMPULSE NOISE TESTING REQUIREMENTS

In support of Contract No. W81XWH-20-C0077, Applied Research Associates, Inc. (ARA) developed prototype standards to establish electromechanical hearing protection device (HPD) evaluation methods that are correlated with human performance. These standards provide a means to describe operationally relevant performance characteristics of HPDs that are not fully incorporated into current standards. In lieu of developing a novel method to evaluate HPD performance in impulse noise environments, ARA has developed recommendations within the scope of the methods established in ASA/ANSI S12.42. This memorandum provides justification for these recommendations to improve the applicability of the standardized measurements to scenarios encountered by Warfighters.

ASA/ANSI S12.42 establishes a test method and analysis procedures for measuring the impulse peak insertion loss (IPIL) of HPDs in response to specific impulse noise levels and durations (0.5–2.0 ms A-duration at 132dB, 150dB, and 168 dB peak pressure). Warfighters operate in a wide range of acoustic conditions not commonly encountered by civilians including impulsive noise outside the required measurement ranges for adherence to ANSI/ASA S12.42.

Small arms fire is a common impulse noise exposure across the DOD that is not covered by the current ASA/ANSI S12.42 standard; therefore, testing at an A-Duration of 0.05–0.2 ms in addition to the 0.5–2.0 ms duration is recommended. Additionally, Warfighters are exposed to impulsive noise generated from weapons systems capable of producing pressures well in excess of 168 dB. ARA, therefore, recommends testing at 183 dB in addition to the standard 132dB, 150dB, and 168 dB. This level lies just below the tympanic membrane rupture threshold of approximately 185 dB (5 PSI), and where bone conduction of sound may become more important to hearing risk assessments.

In summary, to accurately evaluate HPD performance in environments that are representative of different types of noise exposures experienced by Warfighters (e.g., gunshots and explosions), ARA recommends measuring IPIL of each HPD at an expanded pressure range and impulse duration:

	Peak Sound Pressure Levels (dB)	A-Durations (ms)
ASA/ANSI S12.42	132, 150, 168	0.5–2.0
Recommendation	132, 150, 168, 183	0.05–0.20 and 0.5–2.0

These measurements may still be conducted using compressed gas shock tubes, explosive charges, or other means. No changes to data collection or analysis procedures are recommended.