

WIDEX ALLURE™

# COMPARING WIDEX ALLURE WITH FOUR DNN-BASED HEARING AIDS ON SPEECH IN NOISE PERFORMANCE

Authors: Petri Korhonen, MSc, Francis Kuk, PhD, Christopher Slugocki, PhD

A key design objective of the sound philosophy that underlies the Widex Allure platform is maintaining the naturalness of the sound to align with the brain's natural expectations. Preserving the naturalness of sounds ensures exceptional hearing aid experience including speech-in-noise comprehension.

In this study, speech-in-noise performance with Widex Allure RIC was evaluated against four competitors' premium devices that market deep neural networks (DNNs) as their noise management strategy. Twenty nine adults with symmetrical sensorineural hearing loss completed speech in noise testing in a conversation environment designed to reflect real world listening.

Widex Allure's natural processing approach matched or exceeded the performance of all four leading competitors, delivering between 10 and 14% better word recognition for mild-to-moderate hearing losses and between 18 and 26% better word recognition for moderate-to-severe hearing losses than three DNN-based hearing aids. Because all devices in the current study, except Widex Allure, use DNNs as a noise mitigation strategy, these findings support our belief that natural processing in the Widex Allure can be as, if not more, effective for speech-in-noise performance than the current generation of DNN-based devices.

## Key findings

The natural processing philosophy implemented in Widex Allure RIC delivers:

- Up to 14% and 26% better speech-in-noise performance than three leading DNN-based hearing aids, for mild-to-moderate and for moderate-to-severe hearing losses, respectively.
- Comparable speech-in-noise performance with the best performing DNN-based hearing aid.
- Speech-in-noise improvement in listening conditions that are realistic and meaningful for everyday communication.

## Introduction

A common goal in commercial hearing aids is to restore audibility and improve communication in noisy environments. Despite the same common goal, hearing aid manufacturers have different sound design philosophies that lead them to approach the problem from different angles. Some sound designs aim to preserve as much of the natural speech cues while introducing as little distortion as possible. Others may emphasize performance in one-on-one conversations in very loud and difficult noisy settings, even at the expense of degraded overall sound quality. Some approaches aim at supporting natural speech communication in group conversations and dynamic environments where speech and noise characteristics are constantly changing. Another approach may emphasize maintaining auditory awareness of listeners' surroundings regardless of the type or direction of sound.

Widex's sound design philosophy distinguishes itself from the rest of the hearing aid industry. It centers on creating the most natural listening experience, with minimal distortions to the auditory cues present in the original sounds. The aim is to allow listeners to hear more clearly and to experience sounds in their most authentic form. Such an approach to delivering life-like high-quality natural sound should not be viewed as an isolated end goal, but as a means for enhanced listening performance and greater satisfaction with the hearing aids across diverse listening situations. In fact, on a survey on the importance of naturalness and sound quality, 94% of hearing aid users (N = 3,877) reported the sound quality of their hearing aids to be very or extremely important (Ziegler et al., 2023). In the same study, 88% of the listeners were satisfied with the naturalness and 90% with the sound quality of their current Widex hearing aids. From the hearing aid manufacturer's perspective, these results emphasize the importance of considering sound quality and naturalness when designing a hearing aid that meets the consumers' needs.

## Clinical implications

Widex Allure provides wearers with better speech-in-noise performance than three DNN-based hearing aids.

Preserving natural acoustic cues in hearing aids is not merely a matter of sound quality preference but it also affects how listeners understand speech. The human auditory system relies heavily on prediction. The brain constantly anticipates the next word, where the voice is coming from, and how loud or soft a sound should be (Best et al., 2008; Sohoglu et al., 2012). When hearing aids maintain the natural acoustic cues present in the original signal, they provide stable, predictable information the brain expects, thus reducing the brain having to reconstruct missing or distorted acoustic details. In contrast, hearing aids that distort temporal, spectral, and/or spatial cues may require the brain to work harder to interpret the speech. Understanding acoustically degraded speech requires engagement of additional cognitive resources, which can increase listening effort and leave the listener feeling fatigued after social interactions, even if speech is audible (Peelle, 2018).

### **Key features to achieve natural sounds ...**

Preserving the natural details of the original sound does not imply that no hearing aid processing is applied. In fact, the opposite is true. It requires sophisticated sound processing to keep deviation from the natural sound to a minimum, while simultaneously overcoming the hearing sensitivity loss of the individual listener. Widex hearing aids achieve natural and high-quality sound through careful interplay of several sophisticated signal processing features, each designed with the goal of preserving the details of the original sound.

#### *Minimize distortion at the input - True Input technology*

The performance of any sound processing system is ultimately limited by its weakest link. Because processing begins the moment sound enters the hearing aid, it is essential that the device captures the most accurate input possible before applying any subsequent digital signal processing such as compression, noise reduction, or directionality. The input stage converts the analog input signal from the hearing aid microphone into a digital signal by the analog-to-digital converter. If the incoming sound level exceeds the converter's upper limit, the signal may be peak clipped or artificially compressed. Both peak clipping and input compression can introduce audible artifacts such as crackling, popping, or muffled speech, which become more pronounced as input levels exceed the converter's upper limits. Widex's True Input technology was the first in the hearing aid industry to offer an extended input dynamic range, delivering 108 dB range with an upper limit of 113 dB SPL and a low noise floor of just 5 dB SPL. By ensuring a cleaner, more accurate input signal, features such as the adaptive directional microphones and noise reduction algorithms can operate more optimally. This has been demonstrated to improve speech understanding in loud noisy situations (Kuk et al., 2015).

#### *Minimize comb-filter effect (Low delay signal processing)*

A common reason why first-time hearing aid wearers and/or those with a milder hearing loss reject their hearing aids is poor sound quality or that "the hearing aids sound like hearing aids" (Kochkin, 2007; Picou, 2020). This is due to the audible artifact known as comb-filter effect (Lelic et al., 2022). This refers to the spectral distortion resulting from a timing mismatch between the direct unamplified sound reaching the listeners' ear canal via the open or vented ear-tip and that of amplified sound, which is delayed by the signal processing within the hearing aid. The comb-filtering results in peaks and valleys in the frequency spectrum at the eardrum. Perceptually, comb-filtering results in a bothersome "hollow" and "metallic" sound. Thus, a goal in hearing aid processing, particularly in open fittings, is to have as short of a processing delay as possible. The typical delay of most commercial hearing aids is between 5 and 8 ms (Balling et al., 2020). The recently introduced, computation-intensive machine learning algorithms in hearing aids are especially prone to further increase processing delay.

A major source for processing delay in all hearing aids is the signal processing block called the 'filter bank.' A filter bank divides the sound into several frequency bands to enable frequency-specific processing algorithms such as compression and noise reduction. Filter banks operate either in the time- or frequency-domains. These two filter bank types differ in their time and frequency resolution. A time domain filter bank processes the incoming sound continuously as soon as sound enters the hearing aid, which keeps delay to a minimum. In contrast, a frequency domain filter bank first collects a short segment of sound, then converts that entire segment into the frequency domain before any processing can occur. Because the system must wait for segments of sound to be fully captured before computation can begin, this approach inherently introduces processing delay (Kates, 2008). A characteristic of frequency-domain filter banks is that they have the same bandwidth across the entire frequency range. The filter bandwidths needed for meaningful processing are determined by the resolution of the human ear, which has fine frequency resolution at low frequencies and progressively coarser frequency resolution at high frequencies. Thus, to maintain the fine resolution at the low frequency, the frequency domain filters have to apply the same (narrow) bandwidth to the high frequencies as well.

In contrast, time-domain filter banks allow bandwidths to vary across frequencies. In Widex hearing aids the filter banks are designed with narrower frequency bandwidths (=higher frequency resolution) in the lower frequencies and broader frequency bandwidths (=lower frequency resolution) in the higher frequencies (Balling et al., 2022). Such design mimics the resolution of the human ear (Yost, 2006) and results in the lowest possible delay across all frequencies.

Time domain filter banks are typically selected in hearing aids when low latency and preservation of natural temporal and fine structure cues are the priority (Kates, 2008), whereas frequency domain filter banks are chosen when the goal is more flexible spectral processing for noise reduction or beamforming, despite the added delay and potential reduction in naturalness (Loizou, 2013). Widex hearing aids achieve shorter processing delay than the typical 5–8 ms found in premium devices, with delays of ~2–5 ms depending on frequency (Balling et al., 2020).

Widex ZeroDelay technology used in the PureSound program further reduces the processing delay to 0.5 ms. This dramatically reduces the comb-filter effect and delivers noticeably more natural sounds for users who wear open or vented earmolds. Its effect has been seen in improvement of listening preference (Balling et al., 2021) and spatial perception (Korhonen et al., 2022). It also better preserves natural voicing cues in stop consonants (Korhonen et al., 2024), promotes better fidelity of neural speech signal (Slugocki et al., 2020; Zhou et al., 2024), and reduces coloration-pitch distortions for vowel sounds (Slugocki et al., 2023). Reduced processing delay time is particularly beneficial for listeners with mild and moderate hearing losses (Balling et al., 2020; Lelic et al., 2022; Stiefenhofer G. 2023).

#### *Minimize distortion of the signal envelope (variable speed compression)*

Another central feature in the preservation of natural sound is slow-acting compression, for which Widex is known since its introduction of the Senso, the first digital in-the-ear hearing aid (Ludvigsen & Tøpholm, 1997). Compression in hearing aids adjusts the hearing aid gain based on the input level so soft sounds are audible and loud sounds are comfortable. Slow-acting compression reacts to changes in the hearing aid input level gradually, unlike fast-acting compression, which responds rapidly to every fluctuation in sound level. The benefit of slow-acting compression is that it better retains the temporal characteristics of the sound by providing more stable gain. While fast-acting compression may ensure audibility of sudden soft sounds, it could smear the temporal envelope of the input signals (Kuk, 1996). A substantial body of evidence supports the use of slow-acting compression (Kuk 1998; Windle et al., 2023).

The Variable Speed Compression (VSC) system implemented in Widex hearing aids uses both slow and fast compressors operating at the same time with the final output depending on the contributions from both compressors. At or above a conversational speech level, the overall gain is dictated primarily by the slow compressor. The fast compressor becomes more influential for softer sounds, highly modulated sounds, or when there are large level changes in the input to ensure audibility for softer speech elements. With this dual approach, the slow-acting compressor helps

maintain the temporal envelope of speech while the fast-acting component ensures audibility in situations where speech level changes rapidly (Kuk et al., 2019).

#### ***... while achieving noise reduction and enhancing speech intelligibility***

Preservation of natural sound supports successful communication only when the speech cues are audible. Noise reduction and directional microphone features, when implemented appropriately, can improve the signal to noise ratio while preserving the acoustic cues that the auditory system uses for speech perception.

The new Speech Enhancer Pro noise reduction system in the Widex Allure is built around the legacy Widex approach to noise reduction with new enhancements. It is comprised of three primary components designed to reduce noise, enhance speech intelligibility, and increase listening comfort.

The first component is a new mechanism that analyzes the input signal across 52 frequency bands. This represents a substantial increase in resolution compared to the 15 channel system used in earlier Widex platforms, enabling more precise estimation of noise characteristics. The analysis is carried out in the time domain, preserving the advantages of low processing delay and filter characteristics that closely resemble those of the human cochlea. The system classifies the input into speech and noise by evaluating its modulation characteristics, enhances modulated speech like components and attenuates unmodulated noise. This part of the system operates relatively fast.

The second component shapes the amplified signal to optimize the Speech Intelligibility Index (SII) for the individual wearer under the specific listening environment, yielding both objective and subjective benefits (Peeters et al., 2009). The SII predicts speech intelligibility based on how much speech spectrum exceeds the listener's hearing thresholds in the presence of masking noise. The SII based adjustments are carried out more slowly, thus preserving the temporal envelope of speech and contributing to a stable and calm sound experience.

The third component of the Speech Enhancer Pro is the Hearing Threshold Level Optimizer (HTLOpt) which places the final hearing aid output optimally relative to the wearer's residual dynamic range at each frequency band. This feature is needed because the various signal processing algorithms such as compression, noise reduction, adaptive directionality operate simultaneously and independently so it can be difficult to control where the final hearing aid output is placed relative to the user's residual dynamic range. The HTLOpt considers the output from all processing algorithms within the hearing aid and ensures that the whole dynamic range of the target signal is placed above the

wearer's hearing threshold at frequencies with little noise, while placing the aided noise floor below the threshold at frequencies where noise is detected. This ensures that the overall output dynamic range is optimized with precision at each frequency band based on the individual's residual dynamic range. This feature has been demonstrated to better preserve speech loudness when listening to speech in noise (Korhonen et al., 2025). The real-life benefit of this feature is that speech loudness would be more similar across more noisy conditions, improving the perceived naturalness of the desirable target speech sounds.

In addition to the Speech Enhancer Pro, the Widex HD Locator directional microphone system reduces unwanted background noise. HD Locator uses two omnidirectional microphones arranged to form an automatic, adaptive directional system. The system continuously analyzes the acoustic environment and selects the microphone polar pattern that provides the best signal-to-noise ratio (SNR) for the listener. In quiet the system uses an omnidirectional pattern that picks up sounds from all directions, maintaining environmental awareness around the listener. On the other hand, when noise is present, the system selects a directional pattern that picks up the least amount of noise around the listener (Kuk et al., 2005; 2017).

#### ***What about machine learning in hearing aids?***

Machine learning (ML) has recently emerged as a commercially available signal processing strategy in hearing aids. ML systems use algorithms that learn how to solve problems by observing large sets of example data. In hearing aids, ML based noise mitigation approaches to the speech-in-noise (SiN) problem typically rely on deep neural networks (DNNs), which are computational networks made up of simple interconnected elements known as artificial neurons. Each neuron within the network receives input from several other neurons, makes a simple calculation to produce a single output, and passes that output forward to other neurons. The neurons are arranged in multiple layers. During the DNN algorithm's development, the network is trained to distinguish between speech and noise. The hearing aid later applies this learned knowledge to reduce noise in everyday listening. The strength of connections between any pairs

of neurons across layers are optimized by exposing the network to massive training datasets of real-world recordings of sounds. These datasets include a wide range of talkers and accents, and numerous noise types and environments. Generally, the larger, more diverse, and realistic datasets the network encounters during algorithm development, the better it learns to generalize to real-life listening situations. Conversely, networks that are trained on smaller and/or artificial datasets may perform acceptably under the trained conditions, but poorly in real-world listening. Developers can also incorporate advanced optimization techniques to improve generalization or limit unwanted signal distortion. These optimization methods may integrate auditory models into the optimization to better reflect the actual listening task.

The DNN-based noise reduction systems vary across manufacturers in how they are designed, trained, and optimized. Since hearing aids have strict limits on power and processing capacity due to their small size, a DNN model that drains the battery or introduces noticeable delay is not usable. Developers need to balance between development cost, processing complexity, memory requirement, and battery consumption. These design optimizations could remove some of the natural cues in the target signals, compromise the sound quality, and introduce audible artifacts which interfere with identification of the target sounds for some situations and lead to differences in the performance across devices in varied listening situations. The specific design details are proprietary and not disclosed publicly.

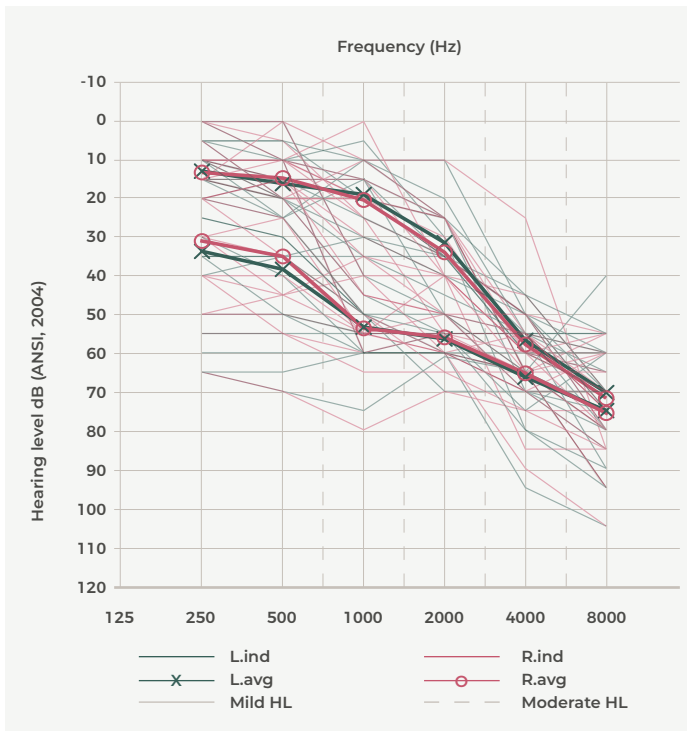
#### ***How does the Widex Allure RIC R D compare to DNN-based hearing aids?***

The Widex Allure RIC R D hearing aid does not use machine learning for noise processing; but its processing that prioritizes the naturalness of the original signals should already ensure the best hearing aid experience including speech-in-noise comprehension. In view of the various DNN approaches, which would suggest likely differences in efficacy among products, it would be meaningful to the hearing care providers to know how the Widex Allure RIC R D compares to four competitors' devices that marketed DNN as their noise management strategy.

## Methods

### Participants

Twenty-nine adults with symmetrical (within  $\pm 10$  dB at 250Hz to 4kHz for 23/29 subjects) sensorineural hearing loss participated. Participants were categorized into two groups based on air conduction four-frequency pure tone averages (4FPTAs) (Figure 1). The mild-to-moderate hearing loss (Mild HL) group (N = 14, 7 female) had average right and left ear 4FPTAs of  $\leq 40$  dB HL (range from 20 to 40 dB HL; mean = 31.3 dB HL, SD  $\pm 5.2$ ). The moderate and moderate-to-severe hearing loss (Mod HL) group (N = 15, 6 female) had average right and left ear 4FPTAs of  $> 40$  dB HL (range from 40.6 to 70.8 dB HL; mean = 53.1 dB HL, SD  $\pm 7.8$ ). Ages in the Mild HL group ranged from 67 years to 79 years (mean = 74.0 years, SD  $\pm 3.8$  years), while ages in the Mod HL group ranged from 59 years to 91 years (mean = 74.0 years, SD  $\pm 8.4$  years). Eight of 14 listeners in the Mild HL group had worn HAs of various makes and models for an average of 8.6 years, whereas 13 out of 15 listeners in the Mod HL group had worn HAs for an average of 22.2 years. MoCA (Nasreddine et al., 2005) scores in the Mild HL group ranged from 23 to 30 (mean = 26.8, SD  $\pm 2.1$ ), while scores in the Mod HL group ranged from 20 to 29 (mean = 26.6, SD  $\pm 2.2$ ).



**Figure 1:** Individual (thin lines) and average (thick lines) air conduction thresholds of listeners in the mild (solid) and moderate (dashed) hearing loss groups.

### Study Hearing Aids

Hearing aids from five manufacturers were included in the study (see Table 1). All study aids were fit using default settings in their respective fitting software according to each manufacturer's proprietary gain targets. Ear-tips were chosen individually based on the Widex fitting software recommendation and the same eartips were used for all study devices (see Table 2 for distribution). Study aids were set to a fixed speech-in-noise program. Widex Allure RIC R D does not have a specific speech-in-noise program and was set to either the Universal (n = 18) or the PureSound (n = 11) program, depending on the Compass Cloud fitting software's recommendation (i.e., PureSound only for milder hearing losses). Feedback tests were conducted for all fittings prior to testing.

Manufacturer	Model	Fitting rationale	Noise/test program
Oticon	Intent 1 miniRITE 13	VAC+	Speech in noise
Phonak	Audéo I90-Sphere	Adaptive Phonak Digital 3.0	Spheric speech in loud noise
ReSound	Vivia 9 VI960S0DRWC (microRIE)	Audiogram+	Hear in noise
Starkey	Edge AI 24 RIC RT	e-STAT 2.0	Crowd
Widex	Allure RIC R D 440	Widex rationale	Universal or PureSound

**Table 1:** Hearing aid makes and models included in the study, listed alphabetically by manufacturer.

Ear-tip type	Hearing loss group	
	Mild (#)	Moderate (#)
Open	5	0
Sleeve, vented	4	2
Tulip	5	3
Double dome	0	10

**Table 2:** Number of participants fitted with each ear tip option in the mild and moderate hearing loss groups.

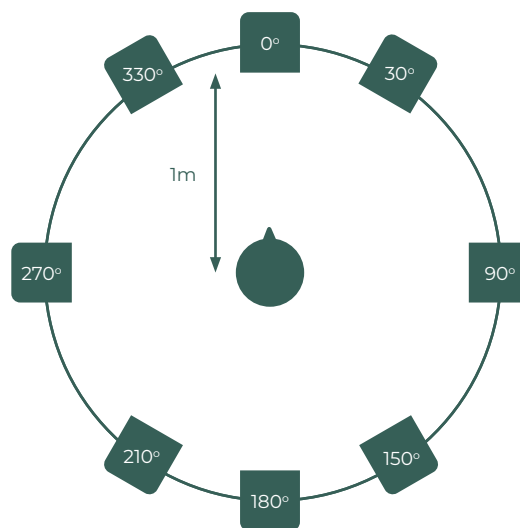
## Outcome measures

Data collection consisted of two phases. In the first phase, listeners' speech-in-noise abilities were assessed using the ezSRT test (Slugocki et al., 2026). The ezSRT test uses a Bayesian adaptive psychometric testing method known as QUEST+ (Watson, 2017) to estimate listeners' performance intensity (P-I) function using sentences from the Repeat-Recall Test (RRT, Slugocki et al., 2018). The test presents listeners with 24 sentences against the continuous background noise. Listeners repeat each sentence after it is presented. The experimenter scores the accuracy of repetition based on three or four designated target words within the sentence. The test chooses the SNR for each trial based on performance across all previous trials according to the QUEST+ algorithm, which uses entropy minimization principles to reduce uncertainty about the threshold and slope parameters that define the listener's psychometric P-I function. Estimates of the SNRs required for different levels of speech understanding can be derived from the P-I function. In this study, performance was quantified as the SNR required to achieve 90% understanding (i.e., speech reception threshold, SRT90). Because most daily successful communications demand near complete comprehension, SRT90 would likely represent the poorest/lowest SNRs needed for effective real world communication. Thus, evaluating devices at this SNR level offers clinicians clearer insight into which hearing aids are most likely to deliver realistic benefit for communication in everyday situations.

Given that the study phase #1 was carried out using an adaptive test, there is a possibility that trial-to-trial changes in SNRs could have affected how the noise mitigation algorithms in the different study HAs engaged at any given moment. To control for this possibility, the study phase #2 measured listener word recognition performance for 32 sentences from the Repeat-Recall Test (RRT) with each study hearing aid at an individually determined fixed SNR corresponding to 90% performance (SRT90) of the best-performing device in study phase #1.

## Stimuli

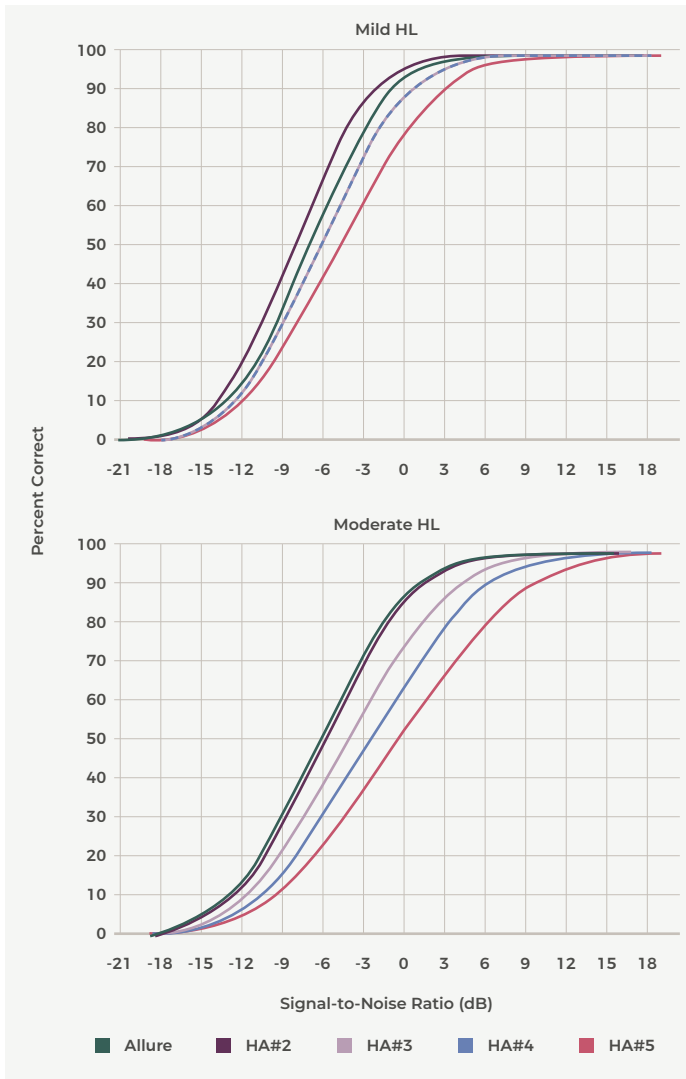
The test configuration (Figure 2) involved presenting sentences across three loudspeakers located in front of the listener simulating conversation partners in a group. Sentences were presented in a complex background noise comprising of a mixture of speech-like noise and cafeteria noise. Target speech stimuli consisted of "low context" sentences taken from the Repeat-Recall Test (Slugocki et al., 2018). The target speech was presented from  $-30^\circ$ ,  $0^\circ$ , or  $+30^\circ$  azimuth in a pseudo-random order with the target position changing from one sentence to the next. Every second sentence came from  $0^\circ$ , while the side speaker location ( $\pm 30^\circ$ ) was randomized with each side using an equal number of presentations. The target speech level on the first test trial was 83 dB SPL and thereafter varied adaptively according to the QUEST+ algorithm. Three speech-like temporally decorrelated distractor signals (ISTS, Holube et al., 2010) were presented from  $150^\circ$ ,  $180^\circ$ , and  $210^\circ$ . Additionally, an ongoing cafeteria noise was presented from  $90^\circ$ ,  $150^\circ$ ,  $180^\circ$ ,  $210^\circ$  and  $270^\circ$  at -10 dB relative to the ISTS signal. The total combined background noise level was fixed at 68 dB SPL. The experiment followed a single-blind design. The order of HA conditions was counterbalanced across participants.



**Figure 2:** The loudspeaker configuration used during the study. Participants were seated in the center of the loudspeaker array, at a distance of 1 m, with loudspeakers positioned at ear level.

## Results

The Widex Allure RIC R D is labeled as 'Allure.' The other study hearing aids were labeled HA#2 through HA#5 to ensure anonymity. Please note that the order of the HAs reported in Table 1 is not the same as the order reported in the subsequent discussion.



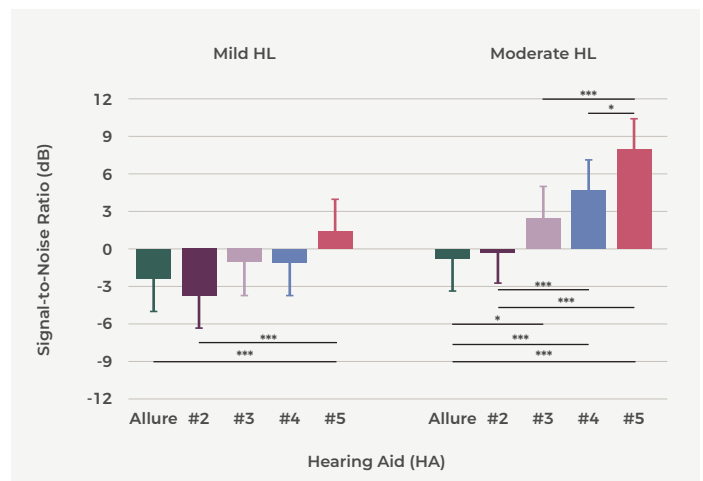
**Figure 3:** Performance-intensity (P-I) functions for speech-in-noise performance for Mild HL (top) and Mod HL (bottom) groups. Each P-I function was derived from the average participant's threshold and slope parameters as measured by the ezSRT test. Horizontal dashed line represents performance corresponding to 90% SiN understanding (i.e., SRT90). Note: the order of the hearing aids is not the same as in Table 1.

## Signal-to-noise ratios (SNR) for 90% correct identification (SRT90)

The mean performance-intensity (P-I) functions determined during phase #1 of the study for the Mild HL and Mod HL groups are shown in Figure 3. They show that Allure and HA#2 provided the best performance with their P-I curves positioned furthest to the left. HA#5 showed the weakest performance, whereas HA#3 and HA#4 delivered intermediary results between Allure/HA#2 and HA#5. There was greater separation among the P-I curves for the Mod HL group than for the Mild HL group, suggesting greater performance differences among hearing aids in listeners with greater hearing losses. The performance gap at SRT90 between the best and the poorest devices was 4.9 dB and 8.1 dB for the Mild HL and Mod HL groups respectively.

A linear mixed effects (LME) model was used to assess the fixed effects of HA Condition (5 levels: Allure, HA#2 ... HA#5) and HL Group (2 levels: Mild and Mod) on listeners' SiN performance at SRT90. The LME model analysis found that SiN performance was significantly affected by the fixed effects of HA Condition and HL Group, where the fixed effect of HA Condition was qualified by significant two-way interaction with HL Group.

Post-hoc analysis of the significant two-way interaction between HA Condition and HL Group revealed that differences across hearing aids were more evident in the Mod HL group compared to the Mild HL group (Figure 4). Allure and HA#2 did not differ significantly from one another in either HL group. In the Mod HL group, Allure outperformed all other hearing aids except HA#2 by an average of 5.2 to 8.5 dB. HA#2 outperformed HA#4 and HA#5. HA#3 and HA#4 significantly outperformed HA#5. In the Mild HL group, Allure significantly outperformed HA#5 by 3.6 dB.



**Figure 4:** Plots exploring the significant two-way interaction of hearing aid condition (x-axis; colors) and hearing loss group (panels) on SiN performance measured with the adaptive ezSRT test. Lower values indicate better performance. Bar heights represent marginal means and error bars represent the 95% confidence intervals of those means. \*p < 0.05, \*\*\*p < 0.001; Tukey adjusted for multiple comparisons.

### Individual word scores (in noise) at fixed SNRs corresponding to SRT90

The individual listener's speech in noise performance (quantified as percentage of correctly identified words) was measured at the SNRs corresponding to the SRT90 estimated from the PI curve of the best performing hearing aid. An LME model was used to assess the fixed effects of HA Condition (5 levels: Allure, HA#2 ... HA#5) and HL Group (2 levels: Mild and Mod) on word recognition performance. The LME model analysis found that HA Condition had significant effect on word scores. The effect of HA Condition on participants' word scores was further qualified by significant two-way interaction with HL Group.

Post-hoc analysis of the significant two-way interaction between HA Condition and HL Group on participants' word scores revealed that differences between the study HAs were more evident in the Mod HL group compared to the Mild HL group. Allure and HA#2 did not differ significantly from one another in either HL group (Figure 5). Allure outperformed hearing aids HA#3, HA#4, and HA#5 by an average of  $\approx 10\text{--}14\%$  for the Mild group and  $\approx 18\text{--}26\%$  in the Mod HL group. HA#2 outperformed HA#5 in the Mild HL group, and all the others (except Allure) in the Mod HL group. HA#3, HA#4, and HA#5 did not differ significantly in the Mild HL group, but HA#3 outperformed HA#5 in the Mod HL group.

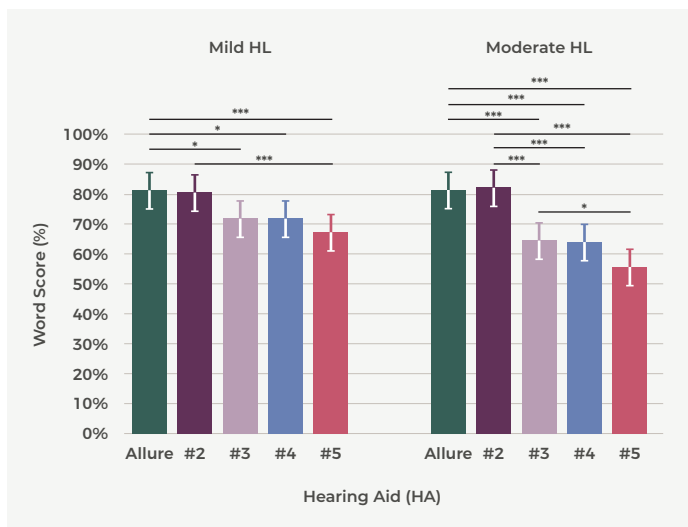


Figure 5: Plots exploring the significant two-way interaction of hearing aid condition (x-axis) and hearing loss group (panels) on word scores. Bar heights represent marginal means and error bars represent the 95% confidence intervals of those means. \* $p < 0.05$ , \*\*\* $p < 0.001$ .

### Discussion

The present study demonstrated that the speech in noise performance was comparable with Widex Allure RIC R D and HA#2. Widex Allure RIC R D delivered better SiN performance than the three other market-leading devices with DNN-based noise reduction. This is reflected in lower SNRs required for 90% speech understanding (i.e., SRT90) (re: HA#3, HA#4, HA#5 in the moderate HL group, HA#5 in the mild HL group). Also, Allure had higher word scores at fixed SNRs corresponding SRT90 relative to HA#3, HA#4, and HA#5 in both mild and moderate HL groups.

Because all devices except Widex Allure RIC R D use DNN as a noise mitigation strategy, these findings challenge the notion that the current generation DNN based devices are inherently superior to non DNN options tested under the current test setup. Instead, these results demonstrated that the natural processing implemented in the Widex Allure outperforms most DNN hearing aids and is on par with the best performing DNN based hearing aid when it comes to speech-in-noise performance. Thus, dispensing clinicians evaluating hearing aid efficacy should look beyond the inclusion of DNN technology. Instead, they should understand the pros and cons of the specific technology on user benefits. Most importantly, they should examine the efficacy data that support the feature, including the design of the study and how the comparisons are made.

Earlier we noted that the test setup in the current study was designed to simulate real world conversations, and that the combination of speech like and cafeteria noise created a realistic and challenging noise background representing everyday listening demands. We selected SRT90 as the lowest SNR at which realistic and meaningful communication occurs. In this context, the study showed that Allure and HA#2 had average SRT90 values of  $-1.5$  dB and  $-1.9$  dB, respectively, across all listeners. Because meaningful real world communication situations rarely involve SNRs that are negative or near 0 dB (Smeds et al., 2015), these SNRs reflect some of the most demanding noise conditions a listener is likely to face frequently (in contrast to extremely poor SNR such as below  $-5$  dB which occurs, but infrequently). Under such challenging conditions, there is no noticeable difference between Widex Allure and HA#2 suggesting that both would be equally and highly effective in everyday communication. In contrast, HA#3, HA#4, and HA#5 showed average SRT90 values ranging from 0.8 to 4.6 dB. This significant reduction in speech-in-noise performance would be easily perceptible when compared with the two best performing devices.

It is reasonable to assume that listeners with a milder hearing loss have less auditory resolution difficulty than those with a more moderate and/or severe hearing loss. Thus, we would expect to see poorer aided performance in the moderate loss group than the milder loss group with the same hearing aids. However, when we compared the word recognition performance across mild and moderate hearing loss groups, we found that the word scores were similar for both hearing loss groups for Allure and HA#2 (around 80%) whereas they were poorer for the moderate loss than for the mild group for HA#3, HA#4, and HA#5 (70% vs 60%). This suggests that Allure and HA#2 likely offer even more benefit to those with a moderate loss than HA#3, HA#4, and HA#5.

The vast differences in measured SiN performance among the five study hearing aids (as much as 8.5 dB at SRT90 for the Mod HL group), and also between mild loss and moderate loss listeners suggest that there is a real, meaningful difference in speech-in-noise ability amongst today's premium hearing aids. This highlights the importance of conducting speech-in-noise testing during clinical hearing aid fittings in order to select the device that delivers the best speech-in-noise performance. While any speech-in-noise test may be configured for such evaluation, the ezSRT test used in the current study may offer a fast and reliable way to assess speech-in-noise performance.

DNN-based speech-in-noise solutions include a broad range of algorithms with substantial variability in their implementations. The design choices influence how well the algorithms generalize across different listening situations and what kinds of artifacts they may introduce. DNN systems trained on limited datasets may struggle in environments with unfamiliar noises, competing talkers, reverberation, or dynamic conversational scenes. As a result, improvements (or degradation) in performance demonstrated by one DNN system in a specific listening situation may not be assumed to generalize to other DNN systems or listening environments. Notwithstanding, it is evident that a non DNN-based signal processing algorithm that preserves the natural speech cues compares very favorably with DNN-based noise management systems in this study.

### **Conclusion**

A key design objective of the sound philosophy that underlies the Widex Allure RIC R D platform is maintaining the naturalness of the sound to align with the brain's natural expectations. It outperforms most other flagship hearing aids in speech-in-noise performance as measured in the current study.

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