PHILIPS

HearLink

White Paper

Next generation sound technology

January 2021

Philips HearLink is designed to help hearing aid users connect to people. This overarching goal guides the design of the two technology areas of Philips HearLink, connectivity (SoundTie) and sound processing (SoundMap). The new HearLink 9030|7030|5030 receives significant updates in both areas.

SoundTie 2

The first generation of SoundTie offered wireless connectivity with direct audio streaming from iOS devices to Philips HearLink. While connectivity was possible with Android™ devices, an intermediate device (AudioClip) was required for streaming. Thanks to an updated wireless chipset, SoundTie 2 now offers connectivity and direct streaming from both iOS devices and Android devices to the new Philips HearLink. Media and phone calls can be streamed from either family of smart devices. See https://www.hearingsolutions.philips.com/hearing-aids/connectivity for more information on compatibility.

SoundMap 2

All areas of sound processing – the noise control, the amplification, and the feedback cancellation – have received updates in SoundMap 2.

SoundMap 2 comes with 50 % higher frequency resolution, artificial intelligence based sound processing, and the new Speech Clarifier feature.

Noise control

The most significant change to SoundMap 2 is in noise control. In the first generation of SoundMap, as in most hearing aids, speech understanding and comfort in noisy environments are improved using directionality (DIR) and noise reduction (NR). SoundMap 2 improves this architecture in three areas: a 50 % higher frequency resolution, a fundamental change in how the NR operates with the introduction of artificial intelligence (AI) based sound processing, and the new Speech Clarifier feature that complements the effect of the DIR and the AI-NR.

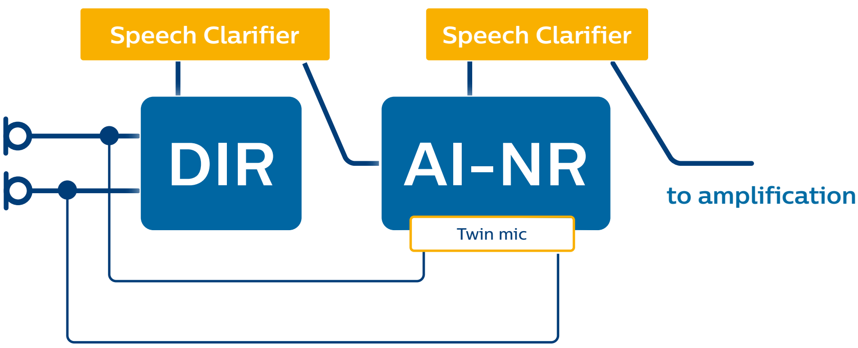

Figure 1: Noise control in SoundMap 2. The processing of the sound is now based on a 24-band architecture; the noise reduction module uses an AI sound technology; and the DIR and AI-NR are complemented by a new feature, Speech Clarifier, that makes speech stand out in noise.

In the first generation of SoundMap, the DIR and the NR operated in 16 frequency bands. Now, SoundMap 2 uses 24 frequency bands. This 50 % increase is made possible thanks to an increase in the underlying digital signal processing (DSP) capacity. In noise reduction systems, more frequency bands with smaller bandwidths are highly desirable due to the harmonic nature of speech.

As for many sounds produced by resonating devices, speech sounds, vowels in particular, consist of a low frequency narrowband sound (fundamental component) produced together with narrowband sounds at frequencies that are multiples of the fundamental frequencies (harmonics) – see Figure 2. With a greater number of smaller frequency bands, the harmonics are more likely to be isolated in individual frequency bands and separated from the noise in the frequency domain, making the DIR and the new AI-NR more efficient at making speech clear.

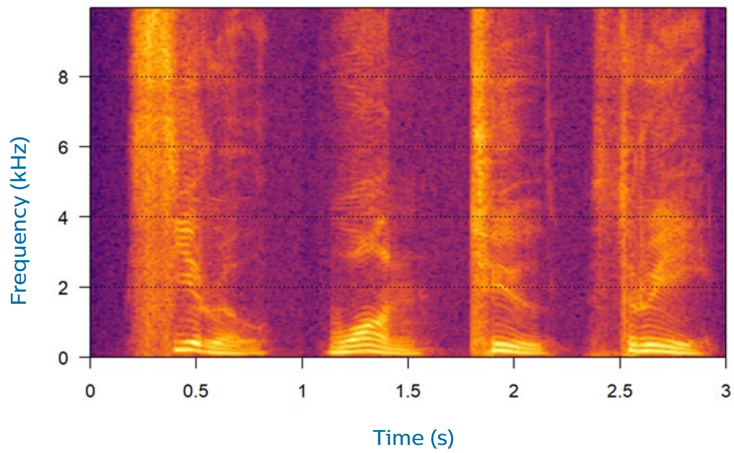

Figure 2: Spectrogram of speech showing the harmonic structure of speech, particularly visible in vowels, e.g. around 0.5 second.

In each frequency band, the overall level is estimated as well as the nature of the sound, noise or speech, to derive an estimate of the signal-to-noise ratio (SNR). Based on these two estimates, the DIR and the NR attenuate the sound when it is characterized by a low SNR and a high sound level. In classic DSP audio engineering, rules (equations) define how these low-level estimates of sound properties (level, SNR) relate to an amount of noise attenuation. However, this approach is limited by:

- **Following a rule:** The attenuation applied by the DIR and the NR in each frequency band is defined by a fixed rule designed by engineers. An example could be that for an estimated level of 85 dB SPL, and an estimated SNR of 2 dB, the NR attenuates the sound in the frequency band centered at 1000 Hz by 3 dB. The process of developing these rules is oftentimes described as heuristic, meaning that the rule contributes to solving a problem, i.e., it does reduce noise effectively, but there is no guarantee that it is an optimal solution.

- **One frequency band at a time:** An important sub-optimal aspect of the architecture of current DIR and the NR is that they traditionally operate independently in each of the frequency bands. Meaning that the attenuation applied in one of the frequency bands is independent from the attenuation in another frequency band. Indeed, looking at Figure 2, one can see that speech sounds are made of coherent sounds (fundamental and harmonics) that are fairly broadband (several 1000s of Hz wide) and that they will be present in several adjacent frequency bands. The facts that (a) adjacent frequency bands are processed independently and (b) they are filled by coherent components corresponding to the same speech "element" (e.g., a vowel sound) can lead to sound artifacts – a compromise to sound quality that is becoming increasingly problematic as the number of frequency bands in DSP increases.

An AI-based system is fundamentally different from a non-AI system because it does not apply fixed rules.

More recently compared to the early days of DSP, artificial intelligence (AI) has started to show that it can be very powerful in solving technical problems. An AI-based system is fundamentally different from a non-AI system because it does not apply fixed rules: an AI system learns by sensing its environment and the information it aims to process. Applied to sound processing, this means that an AI-based sound processing system does not rely on fixed rules, but on building-up its own rules by listening to sounds.

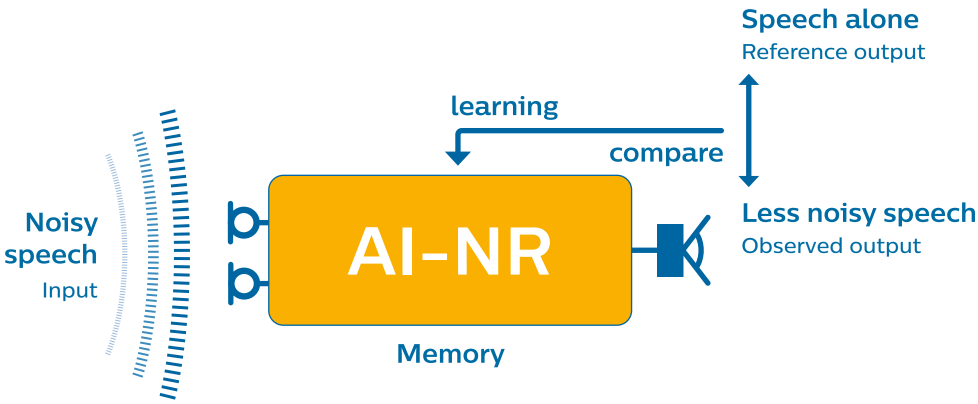

Figure 3: Representations of the training process of the AI-NR in SoundMap 2

The intelligence of the AI-NR compared to an NR system without AI is that it coordinates its effect between frequency bands.

AI systems learn how to perform a task by being trained. The training process of the AI Noise Reduction (AI-NR) is illustrated in Figure 3. It is an iterative process, whereby the AI-NR is fed with speech-in-noise sound samples. These sound samples are processed, and the result (the observed output) is compared to a "reference" result consisting of the speech of the input samples without the noise. The learning happens as the AI-NR is fed back with a measure of the difference between the speech alone (reference output), and the observed output (processed speech in noise). Large differences indicate to the AI-NR that what it is doing is "not good" and small differences indicate that whatever it is doing is "good". In other words, the AI-NR learns from trial and error. The learning happens through 100s of 1000s of training cycles taking place during development, whereby the AI-NR builds its own knowledge iteratively, and the difference between the reference output and the observed output becomes smaller.

AI technology is a bit mysterious in the sense that at the end of the training, one does not know exactly what the system has learned - it is often described as a "black box". What AI-based systems can learn, however, depends on the amount and the structure of their memory. In SoundMap 2, the memory of the AI-NR not only has access to the input sound separated in 24 frequency bands, but it also has the possibility to coordinate its effect between frequency bands.

This coordination of the processing between frequency bands reduces sound artefacts and improves the efficiency of the NR because, as mentioned above, speech sounds are broadband (i.e., they cover several of the 24 frequency bands at the same time).

AI-NR is designed to learn how to apply a coherent effect in adjacent frequency bands. The benefit is that, for instance, in the presence of a vowel sound that typically covers 1000s of Hz (see Figure 2 around 0.5 s), and thereby many frequency bands, the AI can ensure that all NR in the frequency bands filled by the vowel sound is coordinated, minimizing sound artefacts and therefore increasing the efficiency of the NR.

The new AI-NR in SoundMap 2 can deliver an increased effect of up to 10 dB of noise reduction in 24 frequency bands, which contributes to better sound quality in noise for the user and improved speech understanding in noise.

Take-away for AI sound technology:

- AI sound technology exceeds the current digital sound processing approach because it does not rely on fixed DSP rules but instead learns how to process sounds by listening to actual speech-in-noise sound samples.
- AI sound technology is introduced in the NR module of SoundMap 2.
- AI-NR has a memory structure that allows it to coordinate its effect between frequency bands and thereby more efficiently clean speech in noise.

Speech Clarifier

In addition to the AI-NR, the noise control function of SoundMap 2 receives a new feature: Speech Clarifier. As shown in Figure 1, Speech Clarifier consists of two sub-modules, each placed after the DIR and the AI-NR. Speech Clarifier enhances (and controls) how much speech stands out in noise.

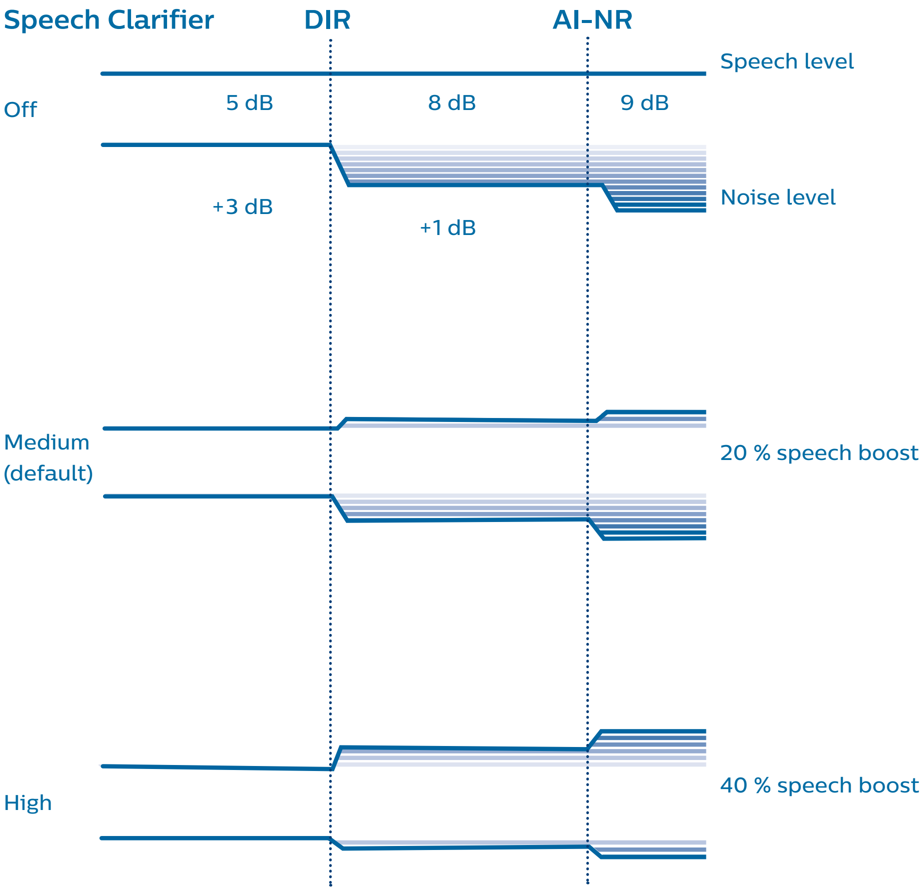

Figure 4: Illustration of the effect of Speech Clarifier (between 1 and 5 kHz – see text) in combination with directionality (DIR) and the noise reduction (AI-NR).

Speech Clarifier complements the effect of DIR and AI-NR to enhance the clarity of speech by considering individual differences to noise preference and tolerance.

Traditionally, DIR and NR aim to preserve speech and attenuate noise. This is illustrated in the top panel of Figure 4 (Speech Clarifier Off), whereby the level of the speech is maintained, and that of the noise is attenuated by 3 and 1 dB in the DIR and the AI-NR, respectively, in this illustration. While this approach is widely used in the sound processing of hearing aids, the perception and tolerance for noise varies among people with hearing loss.[1] Speech Clarifier further refines the effect of the DIR and the NR in hearing aid sound processing by acknowledging these perceptual differences to enhance the clarity of speech for individual users.

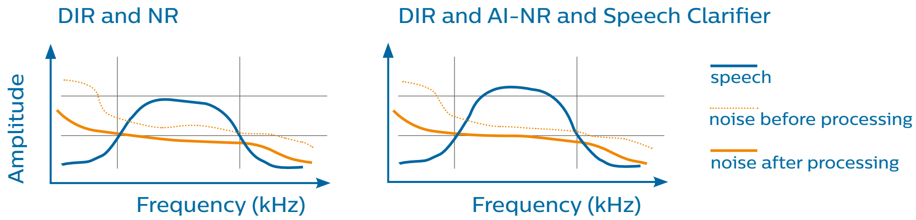

Figure 5: Illustration of level of speech and noise as a function of frequency after the processing by DIR and NR (left) and after the processing by DIR, AI-NR, and Speech Clarifier (right).

As shown in the middle panel of Figure 4, the default net effect of the DIR, the AI-NR and Speech Clarifier is that while the noise is attenuated, the level of the speech is slightly increased. The effect of Speech Clarifier across frequencies is further illustrated in Figure 5. The left panel shows the effect of conventional DIR and NR, i.e., an attenuation of the noise, which is considered constant across the bandwidth for the simplicity of the illustration. The right panel shows the net effect of the DIR, the AI-NR and Speech Clarifier, which can be compared to the left panel.

Speech Clarifier works by redistributing a fraction of the attenuation applied on the noise at the output of the DIR and the AI-NR, in the frequencies between 1 and 5 kHz within that range. This provides an increase in the contrast between speech and noise (perceived as an increase in clarity) because the noise level below 1 kHz and above 5 kHz is unaffected, and because speech is dominant over noise between 1 and 5 kHz for the environments typically encountered by people with hearing loss. The latter is due to the fact that the long-term spectrum of

[1] A.K. Nabelek, M. C. Freyaldenhoven, J.W. Tampas, S.B. Burchfield, and R.A. Muenchen, 'Acceptable Noise Level as a Predictor of Hearing Aid Use', J Am Acad Audiol, 2006, 17:626–639

speech is concentrated between 1 and 5 kHz and that people rarely venture into environments with a negative broadband SNR.[2]

Several important aspects to understand the benefit (and limits) of Speech Clarifier are shown in Figure 5:

- The noise level between 1 and 5 kHz is higher with Speech Clarifier compared to processing with DIR and AI-NR only, yet lower than the level of the unprocessed sound. In other words, the net effect of the DIR, the AI-NR and Speech Clarifier is always a reduction of noise.
- The acoustical contrast between the speech and the noise, i.e., the SNR, is not changed by Speech Clarifier, but Speech Clarifier helps speech perceptually stand out from the noise because the effect is only applied between 1 and 5 kHz where speech is acoustically dominating.
- Sounds below 1 kHz and above 5 kHz are not modified by Speech Clarifier.
- Speech Clarifier works in sync with DIR and AI-NR, which have reaction times of 8 and 2 ms, respectively. Speech Clarifier is therefore providing a dynamic level adjustment between 1 and 5 kHz, which cannot be matched by a change in static gain.

By default, Speech Clarifier redistributes 20 % of the noise attenuation obtained in the DIR and the AI-NR, a value that has proven to be the preferred choice for most users during development. The effect can be increased to 40 % or turned off using the Speech Clarifier control in the HearSuite 2020.2 fitting software. In this latter case, the noise control in SoundMap 2 behaves as the traditional approach that is solely relying on DIR and AI-NR.

The clinical application of Speech Clarifier can be summarized as follows:

- For patients with a low noise tolerance who want maximum noise reduction, clinicians should increase the effect of DIR and AI-NR first, by selecting higher performance settings in the software. As a second measure, clinicians can set Speech Clarifier to a lower setting, to buffer them against the relative effects of noise that the control provides.
- For patients with a higher noise tolerance who want

[2] K. Smeds, F. Wolters, and M. Rung, 'Estimation of Signal-to-Noise Ratios in Realistic Sound Scenarios', J Am Acad Audiol 2015; 26(02): 183-196 DOI: 10.3766/jaaa.26.2.7

maximum speech clarity, clinicians can increase the effect of the DIR, or set Speech Clarifier to a higher setting. This is because most of the increase in speech understanding in noise is due to the off-axis noise attenuation obtained by the DIR. Speech Clarifier can further refine the perceptual salience of speech, and in some low-frequency noise conditions, it can reduce the upward spread of masking due to the emphasis of the level increase on frequencies above 1 kHz.

Take-away for Speech Clarifier:

- DIR and the AI-NR are the primary controls to adjust the sound in noisy environments, because only DIR and AI-NR can increase the SNR at the ears of the patient. Speech Clarifier acts as a secondary order effect, whereby it controls the perceptual clarity of speech.
- Speech Clarifier complements the effect of DIR and AI-NR to enhance the clarity of speech by considering individual differences to noise tolerance.
- Speech Clarifier increases the perceptual clarity of speech in noise by redistributing the SNR gained by DIR and the AI-NR between 1 and 5 kHz, where speech dominates.

Amplification

As for the remaining modules of SoundMap 2, the amplification module now operates in 24 frequency bands. The moment-to-moment gain in the compression function is calculated over 24 frequency bands, and it is now possible to fine tune the gain with up to 24 frequency bands in HearSuite 2020.2 for HearLink 9030.

Feedback cancellation

The feedback cancellation introduced in the previous generation of Philips HearLink was a novel solution whereby feedback detectors were directly placed in the amplification unit. This architecture allows the feedback loop to be detected as it is still building up. Upon feedback build-up detection, the system uses a breaker signal to break the feedback loop and prevent further growth of the feedback - see the Philips HearLink 20.1 White Paper.

This leading technology is ported to SoundMap 2 and the overall feedback cancellation solution is further improved by the introduction of a new acoustical earpiece. The new dome, called the OpenBass dome, replaces the current Open dome.

Figure 6 shows the legacy Open dome on the left, which is still used for previous generations of Philips HearLink hearing aids, and the newly designed OpenBass dome only available for HearLink 9030|7030|5030, on the right. The main difference between the two lies in the design of the acoustical venting.

The new OpenBass dome comes with SoundTunnels™ realized by 3D grooves.

Figure 6: Legacy Open dome with acoustical venting realized by holes (left) versus the newly designed OpenBass dome with acoustical venting realized by SoundTunnels™ (right).

In the legacy Open dome, the acoustical venting is realized by "flat" holes that are perforated on the surface of the silicon dome. In the new OpenBass dome, the acoustical venting is realized by 3D grooves through the silicon dome that are formed by a curved floor, and a ceiling towards the base of the dome. This structure is three dimensional, as opposed to flat holes in the legacy Open dome, and it is referred to as SoundTunnels™.

When the legacy Open dome is inserted in an ear canal, it will be squeezed to ensure a good fit and retention. Because of the "flat" design of the venting holes, the shape of the holes will be changed, and the size of the holes will be effectively reduced because they will likely be partly occluded. Furthermore, the magnitude of these changes is variable given the vast variety of ear canal sizes and shapes. This makes the acoustical coupling of the speaker unit on the individual ear canal hard to predict, which creates potential feedback instability.

When the OpenBass dome is inserted into an ear canal, the shape of the dome will also be changed due to the squeeze to ensure the tight fit. However, the three dimensional structure of the SoundTunnels™ will provide support and ensure that their shape and size remain as intended. This means that the acoustical venting in individual ears is markedly easier to predict, despite the individual differences in ear canal size and shape.

Compared to the legacy Open dome, the benefit of the SoundTunnels™ is that engineers were able to reduce the acoustical venting area of the OpenBass dome by 70 %, while maintaining the same “openness” perception. The combination of a reduction in venting area and a higher predictability of the effective venting once inserted in individual ear canals has several benefits, including that it creates a better acoustical environment for the feedback canceller, which also facilitates the match of the prescribed gain on expected target gain values.

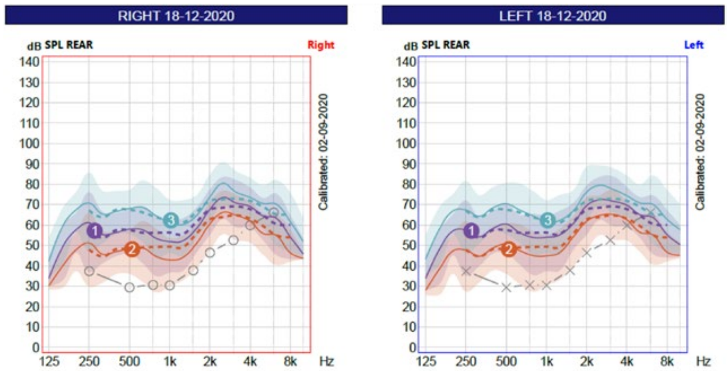

Figure 7: Illustration of the target match accuracy of the HearLink 9030 with the new OpenBass Dome. The target match is shown for 3 input levels, 55, 65 and 75 dB HL. The devices were programmed with an N2 audiogram[3], for which the OpenBass dome is the recommended dome, and the target gain was prescribed for the NAL-NL2 fitting rationale. All measurements were carried out in a Canadian Audiology Simulator for Research and Learning[4] mannequin, with two different ear canals for each side. The left ear canal was a standard ear canal while the right ear canal was a variant with a similar volume but is not as straight, thus providing a different REUR.

A qualitative evaluation of the accuracy of the target match was conducted by measuring the average RMS error (RMSE) between the expected target gain and the measured prescribed gain across frequencies and input levels. RMSE’s were measured in conditions of a first fit, i.e. with the gain as prescribed at the initial fitting of the hearing aid (FirstFit condition) and after using the auto-fit feature (AutoFit condition).

In the FirstFit condition, the averaged RMSE was 6 dB for the HearLink 9030 with the OpenBass dome, and 6.7 dB for the HearLink 9010 with the legacy Open dome, in otherwise similar

[3] Bisgaard N, Vlaming MS, Dahlquist M. Standard audiograms for the IEC 60118-15 measurement procedure. Trends Amplif. 2010 Jun;14(2):113-20. doi: 10.1177/1084713810379609. PMID: 20724358; PMCID: PMC4111352.

[4] Koch, R. W., Saleh, H., Folkeard, P., Moodie, S., Janeteas, C., Agrawal, S. K., Ladak, H. M., & Scollie, S. (2020). Skill Transference of a Probe-Tube Placement Training Simulator. Journal of the American Academy of Audiology, 31(01), 040–049. https://doi.org/10.3766/jaaa.18054

fitting conditions. The clinician has then the possibility to improve the target match either by manually adjusting the gain levels, or by using an auto-fit feature. Using the auto-fit feature, the RMSE for the HearLink 9010 decreased to 3.9 dB, and that of HearLink 9030 reached 4.1 dB.

This clinical case is rather simple and cannot represent the full diversity of fitting cases, but it does illustrate the expected benefit that the small acoustical venting area of the new OpenBass dome provides a better "first fit" of the HearLink 9030, compared to devices with legacy Open domes. The case also illustrates that the auto-fit feature of REM equipment provides a quick way of significantly improving the target match.

Take-away for the OpenBass dome:

- Newly designed OpenBass dome replaces the legacy Open dome for HearLink 9030|7030|5030.
- Acoustical venting in the OpenBass dome uses SoundTunnels™ which retain their shape when inserted into individual ear canals.
- The OpenBass dome provides the same perception of "sound openness" to the user while ensuring better feedback stability and better sound quality, including when streaming sounds from mobile devices.

Validation test

In order to validate the performance of the new technology of HearLink 9030, and compare it to the previous generation HearLink 9010, an internal product validation trial was conducted. The goal of the trial was to measure the benefit of a combination of all the algorithms embedded within the hearing aid by assessing the speech understanding performance.

Nineteen participants participated in the trial. The average age was 68, and the hearing loss of the participants was an average pure-tone audiometry corresponding to a sloping mild to severe sensorineural hearing loss (right PTA 44.6 dB HL, right HFA 64 dB HL, left PTA 43 dB HL and left HFA 62.9 dB HL). All participants were fitted with either HearLink 9010 or HearLink 9030 miniRITE T R hearing aids that were fitted according to the default prescription and following clinical best practice. Target gain was set according to the NAL-NL2 fitting rationale and verified with real-ear measurements (ISTS signal and NAL-NL2 targets).

The speech performance was assessed in the clinic by measuring the speech reception threshold (SRT) with the Göttinger Satztest[5]. The testing took place in a sound-isolated doublewalled room with an acoustical treatment to ensure a low reverberation time. The participants were seated in the middle of the room and surrounded by multiple loudspeakers. The target speech signal was presented from the front starting at +10 dB SNR, and was adapted toward 50 % performance while the sum of all the noise sources was fixed at 67 dB(A). The noise was presented from 5 loudspeakers from the back (from 120, 180, and 240 °) and the front (from 30 and 330 °). The noise consisted of 3 steady speech shaped noise sources from the back and two unsynchronized ISTS maskers from the front.

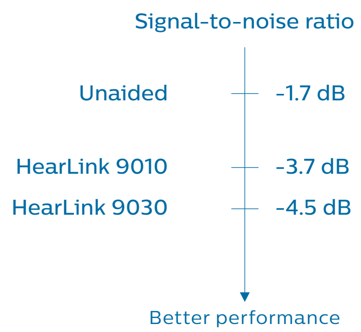

Figure 8: Average speech reception thresholds in 3 conditions.

As shown in Fig. 8, the average performance was -1.7 dB SRT when participants were unaided, -3.7 dB SRT with HearLink 9010, and -4.5 dB SRT with HearLink 9030. The mean standard error was 0.3 dB, and all differences are statistically significant ($p<0.001$). Giving a real-life interpretation to absolute clinical data, here SRTs, is a delicate matter, but a relative comparison of these numbers gives some appreciation of the performance. The 2 dB improvement from the unaided condition to using HearLink 9010 represents the overall benefit of HearLink 9010, both in terms of amplification and noise processing. The additional 0.8 dB provided by HearLink 9030 represents the added benefit of the new technology (AI-NR, Speech Clarifier, OpenBass dome, 24 frequency-band architecture). The 0.8 dB in SRT between the HearLink 9010 and HearLink 9030 conditions represents 40 % of the difference in SRT between the unaided and HearLink 9010 conditions. This illustrates the significance of the technical improvement of the new technology and how it will better help the users to understand speech and help them connect to people in noisy environments.

[5] Kollmeier B, Wesselkamp M (1997) Development and evaluation of a German sentence test for objective and subjective speech intelligibility assessment. J Acoust Soc Am 102(4):2412–2421

Conclusion

The new Philips HearLink receives major technology updates that further strengthen the overarching benefit of Philips hearing aids: helping users to create connections with people.

The new connectivity solution SoundTie 2 brings connectivity and direct audio streaming to iOS as well as compatible Android devices, so users can enjoy direct streaming of phone calls and entertainment to their ears.

The core sound technology of the new Philips HearLink also receives major updates. From the very advanced AI sound technology to the most tangible OpenBass dome, all updates contribute to making speech clearer to the user. These updates are also designed with the hearing care professional in mind with new controls (Speech Clarifier, 24 fitting bands), bringing new possibilities to address the needs of users of the new Philips HearLink.

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