WHITEPAPER
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An introduction to **OpenSound Navigator™**

HIGHLIGHTS

OpenSound Navigator™ is a new speech-enhancement algorithm that preserves speech and reduces noise in complex environments. It replaces and exceeds the role of conventional directionality and noise reduction algorithms. Key technical advances in OpenSound Navigator:

- Holistic system that handles all acoustical environments from the quietest to the noisiest and adapts its response to the sound preference of the user – no modes or mode switch.
- Integrated directional and noise reduction action – rebalance the sound scene, preserve speech in all directions, and selectively reduce noise.
- Two-microphone noise estimate for a "spatially-informed" estimate of the environmental noise, enabling fast and accurate noise reduction.

The speed and accuracy of the algorithm enables selective noise reduction without isolating the talker of interest, opening up new possibilities for many audiological benefits.

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The daily challenge of communication in noise

Sounds occur around us virtually all the time and they start and stop unpredictably. We constantly monitor these changes and choose to interact with some of the sounds, for instance, when engaging in a conversation with a familiar voice (Gatehouse and Noble, 2004). Although a daily task, navigation and communication in complex acoustical environments is both the greatest difficulty and the largest unmet need of people with a hearing impairment (Gatehouse and Akeroyd, 2006).

The difficulty can be so great that some people with a hearing impairment may choose to avoid the most challenging environments and limit their social participation (Crews et al., 2004). Recent studies suggest that social isolation associated with hearing loss, accelerates cognitive decline if the hearing loss goes untreated (Lin et al., 2013, Kiely et al., 2013, Amieva et al., 2015). Helping people with a hearing impairment maintain good communication in noisy environments is therefore not only a matter of comfort and life-quality, but also a matter of health.

While sound amplification is efficient for improving speech understanding in quiet environments, it has limited benefits when multiple sounds are present. This is because the sound of interest, speech in a communication scenario, is acoustically mixed with all other interfering sounds. To make sense of this acoustical mixture, we use a cognitive process to focus our attention selectively on a sound of interest and put the other sounds in the background. However, sounds are of changing interest and importance. To successfully navigate complex acoustical environments, we therefore need to have access to all sounds, to be able to switch attention when the need arises, e.g., when we hear our name (Shinn-Cunningham, 2008, Shinn-Cunningham and Best 2008).

To make sense of a complex acoustical mixture, the brain organises the sound entering the ears into different auditory "objects" that can then be focused on or put in the background. The formation of these auditory objects happens by assembling sound elements that have similar features (e.g., Bregman, 1990). These features can be spectral (e.g., frequency content, pitch), temporal (e.g., amplitude modulation, onset synchronicity), or spatial (interaural differences). For instance, all sound elements starting in the same time-instant will predominantly be assigned to the same auditory object.

These sound features are unfortunately less robustly encoded in the auditory periphery (middle ear, cochlea) of people with a hearing impairment. For instance, the loss of audibility may prevent the detection of certain frequencies or the broadening of cochlear filters may impair the ability to resolve spectro-temporal information. As a result, the formation of auditory objects and the switch between them become slower for people with a hearing impairment. This slower process causes difficulty, especially in fast-changing dynamic situations, such as lively family dinners (Shinn-Cunningham and Best, 2008).

Conventional technology

Current hearing-aid technology support communication in complex acoustical environments by attenuating noise and create a focus towards the front of the user. This effect is typically realised by two independent processes: directionality and noise reduction (see Fig. 1). First, directionality, realised by a two-microphone adaptive beamformer, is applied to suppress noise sources originating from directions away from the target speaker. Subsequently, noise reduction is applied to the resulting beamformed signal, in order to further reduce the noise remaining in the signal from the beamformer.

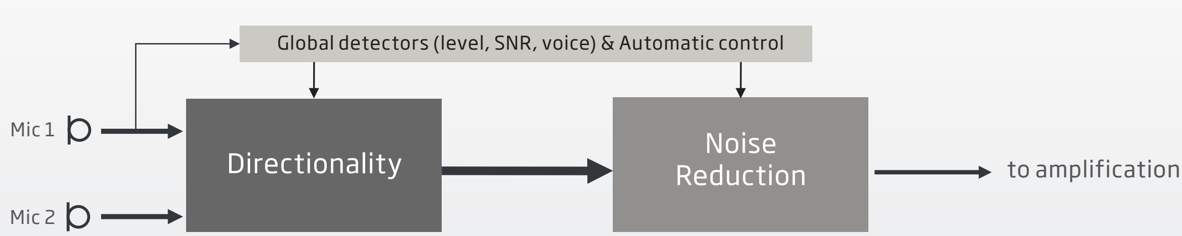

Figure 1: General structure of a speech-enhancement system with directionality, noise reduction, automatic system, and global detectors.

Each of these technologies has been used and improved on for many years. Directionality systems are well known to improve speech understanding in noise. More recently, binaural beamforming has also been introduced. In these systems, the larger distance between the microphones placed on the left and right hearing aids enables a much narrower beam, i.e., they suppress more noise from directions others than that of the target speaker.

Noise-reduction systems, in particular those working in a single broad-frequency band, are less beneficial in this respect but the reduction of the noise level increases listening comfort (e.g., Hu and Loizou, 2007). These noise-reduction systems use long-term properties of the sound to estimate the noise, making them slow to react compared to the fast acoustical changes occuring in complex environments.

The effectiveness of these technologies in everyday environments, i.e. outside laboratory conditions, has also been criticised (e.g. Bentler 2005). Recent investigations have even shown a negative effect. Speech understanding was shown to decrease with direcitonaly systems using beams narrower than 50° (Beach et al. 2015). The speed and the accuracy of human sound localisation was also shown to be negatively impacted by directional systems (Brimijoin et al. 2014). Arguably, the limit of the current technology comes from the fact that it reduces context, i.e., it removes information that the brain naturally uses to disentangle the complex acoustical environments. Binaural beamformers are a good example of this trend. The extra benefit offered by binaural beamformers, compared to two-microphone beamformers, comes at the cost of a very narrow beam that not only suppresses all context but also requires the user to maintain their head unnaturally fixed to obtain the full benefit of the technology.

So how can a speech-enhancement algorithm better support communication in complex daily acoustical environments? Such technology would need to not only remove noise, but also preserve important speech information from all directions to facilitate the user's natural process for forming auditory objects, follow ing different talkers, and switching between them.

Oticon introduces such a technology. It reduces the noise in complex environments without isolating a single talker but, on the contrary, maintains access to all talkers. We therefore call it a Multiple Speaker Access Technology (MSAT). This has been enabled by many technological achievements and is protected by international patents (Kjems and Jensen, 2015). For the Opn family, the MSAT technology is implemented in the new feature, OpenSound Navigator™.

OpenSound Navigator

OpenSound Navigator (OSN) is a MSAT-class speech-enhancement algorithm that runs on the new platform Velox™. It replaces conventional directionality and noise reduction systems. These two technologies still exist in an advanced version in the OSN feature but they are used in a very different way. As shown in Fig. 2, the noise reduction, here called Noise Removal, is placed after the directionality, here called Balance, and importantly, both modules receive a spatially-informed noise estimation realised by a multi-microphone noise estimator in the new Analyse module. In addition, YouMatic™ LX, adapts the strength of the Balance and Noise Removal modules to the acoustical environments and the sound preferences of the individual users. The three main modules of OSN and how they interact is explained in detail in the next sections.

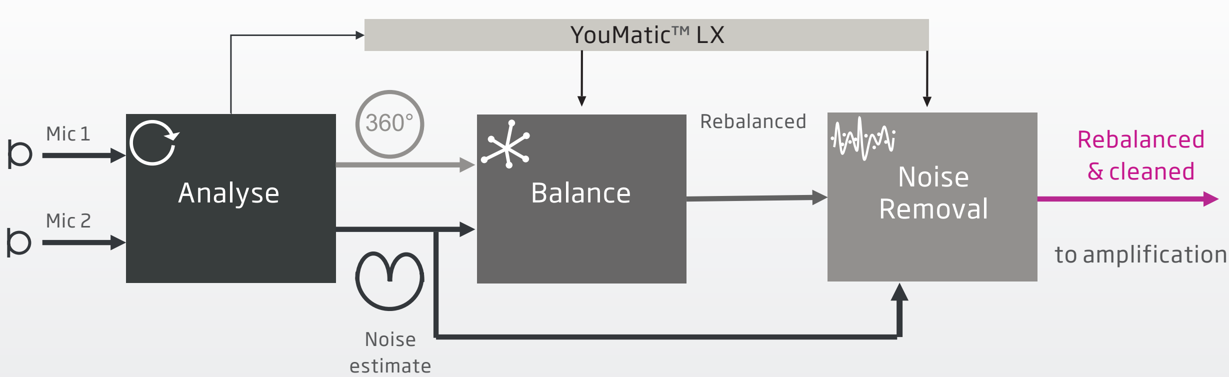

Figure 2: Functional diagram of OSN. It consists of a Balance and a Noise Removal module that replace conventional directionality and noise reduction systems. Here they are assisted by the Analyse module that ensures an accurate estimation of the acoustical conditions, e.g, noise level and location.

Analyse

The Analyse module informs the Balance and Noise Removal modules of the acoustical conditions. It uses a multi-microphone beamforming algorithm to create two fixed acoustical "views" of the environment. The first view is realised by an omnidirectional beam that captures a 360° panorama of the environment. This gives OSN a signal that captures all the sounds. The second view is realised by a back-facing cardioid beam (see Fig. 3) that captures the sounds from the side and the back of the user. The sounds captured by the back-facing cardioid form the noise estimate for the whole system. This spatial weighting of the noise makes sense audiologically, because the further to the back a sound is, the less likely it is that it forms part of the active interaction at the front. Speech from the sides and behind is an exception and it is preserved by the Voice Activity Detector - see the section *Access to Multiple Speech.*

In contrast with conventional systems that estimate noise from a single microphone channel (see Fig. 1), in OSN, the noise is estimated using the two-microphone channels (see Fig. 2). Therefore, the noise estimate in OSN reflects not only the level of the noise, but the direction-dependant sensitivity of the back-facing cardioid also captures the spatial arrangement of the noise (Kjems and Jensen 2012, Jensen and Pedersen 2015). As shown in Fig. 3, the noise estimate is sensitive to where noise sources are placed: a noise source at the back will be estimated as "noisier" than if it is on the side.

With a noise estimate updated 500 times/s, independently on each of the 16 frequency bands, this accurate two-microphone technique allows the Balance and Noise Removal modules to be more selective in their noise reduction effect.

Balance

The Balance module is essentially a directionality system that uses a minimum-variance distortionless response (MVDR) beamformer. This algorithm is widely used in different systems to improve signal-to-noise ratios (SNRs) and detectability, such as in radars. Here, it increases the SNR by constantly mixing the omnidirectional and the noise signals (see Fig. 2), and thereby creates a rebalanced soundscape where speech is made clearer by attenuating the loudest noise sources place between speech sources. The key to effecive performance of such systems is how they are informed about the acoustical conditions, here by the Analyse module and the two-microphone noise estimate.

The most important sound, the target speech in front of the user is only present in the omnidirectional signal. The disturbing sounds however, are present both in the omnidirectional and in the noise signals. The MVDR algorithm subtracts the noise signal from the omnidirectional signal to minimise the noise in the rebalanced signal. In effect, this subtraction creates strongs attenuations, called null directions, towards the dominant noise sources (see Fig. 4). One null directions is adjusted 125 times per second independently in each of the 16 frequency bands, allowing OSN to control, in principle, up to 16 sound sources in each side of the head (32 in total). The speed and the precision at which this processing is executed allows OSN to selectively attenuate noise sources between speech sources.

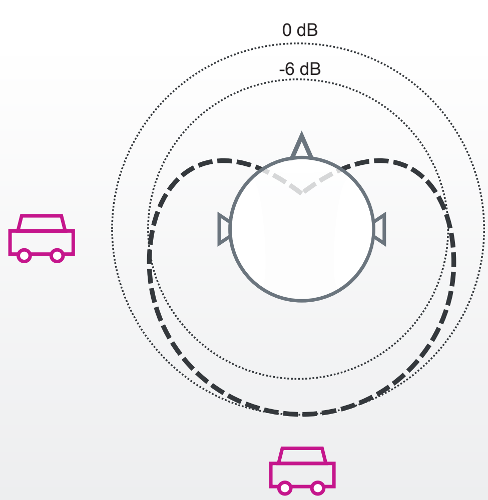

Figure 3: Top view of user and the fixed back-facing cardioid (dashed line) used to estimate the noise. The shape of the back-facing cardioid causes the car at the back to be about 6dB more "noisy" than the car on the side.

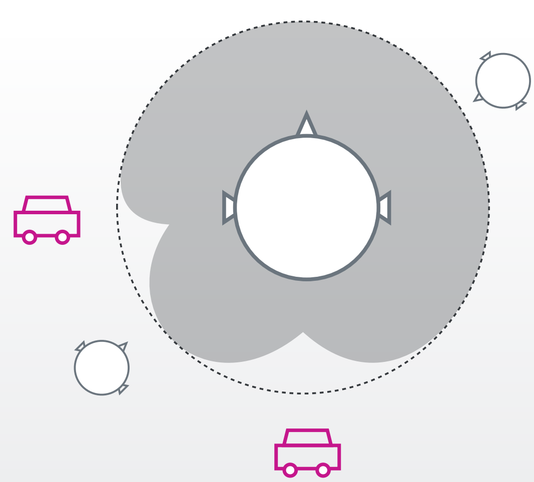

Figure 4: Illustration of how null directions attenuate dominant noise sources that are localized between speech sources. Thanks to the spatially informed noise estimate, the depth of the null direction is stronger for sounds placed further to the back.

Noise Removal
Not all noise sources have a precise location and can be attenuated by the Balance module. In many acoustical environments, the target speech in the rebalanced signal will still contain some noise. This happens for instance in environments with diffuse noise or when a noise source is directly behind the target talker. To reduce this residual noise, the Noise Removal module operates as a secondary noise cleaner. It is fundamentally a very fast noise reduction system operating independently in 16 frequency bands.

The purpose of noise reduction systems is to attenuate a sound mixture, i.e., speech+noise, at a particular moment and in a particular frequency band if the noise dominates the signal, here defined as speech, in the mixture. As for the Balance module, the major limiting factor is that the actual noise level is unknown, and must be estimated from the available microphone inputs. In OSN, the noise is estimated in the Analyse module using the novel spatially-informed two-microphone estimate. Furthermore, the level of the signal is estimated in the rebalanced signal, i.e., after the processing of the Balance module (see Fig. 2). This estimation is therefore more accurate because the signal has already been processed by the Balance module, and therefore contains less noise. As a result, the signal and the noise estimates in OSN are simply more accurate than in conventional systems, with OSN capable of estimating SNRs accurately even at low SNRs.

With an accurate estimate of the SNR, a high resolution of 16 frequency bands and time-window analysis of about 10ms (500 updates/s, with overlap), the Noise Removal module is capable of accurately removing noise between words (up to 9dB attenuation) without altering the fundamental properties of speech signals such as the temporal amplitude modulation (see Fig. 5).

Access to Multiple Speech
Multiple talkers create a common challenging situation. These talkers are placed around the user and, in particular, can be on the side and at the back. They could be interpreted as noise if present within the back-facing cardioid. To prevent the attenuation of speech information by the system, OSN is equipped with a Voice Activity Detector that operates independently in each of the 16 frequency bands. If speech is detected in one frequency band, the state of the Balance and the Noise Removal modules in the corresponding band is "frozen" to preserve the speech information regardless of the position of the talker. The detection of speech and the resulting freeze and release of Balance and Noise Removal modules is updated 500 times per second.

Perspective on Technology
OSN marks a breakthrough in the developments of speech-enhancement systems – it belongs to the MSAT class. It is not only designed to improve the acoustics at the user's ears, but also to facilitate the brain's own processing. It does not isolate the front talker but preserves access to all talkers. It is its accurate and fast spatially-informed noise estimator that allows the Balance module to selectively attenuate noise sources at given locations, between talkers, and the Noise Removal module to remove noise between words. OSN opens up many possibilities for new user benefits.

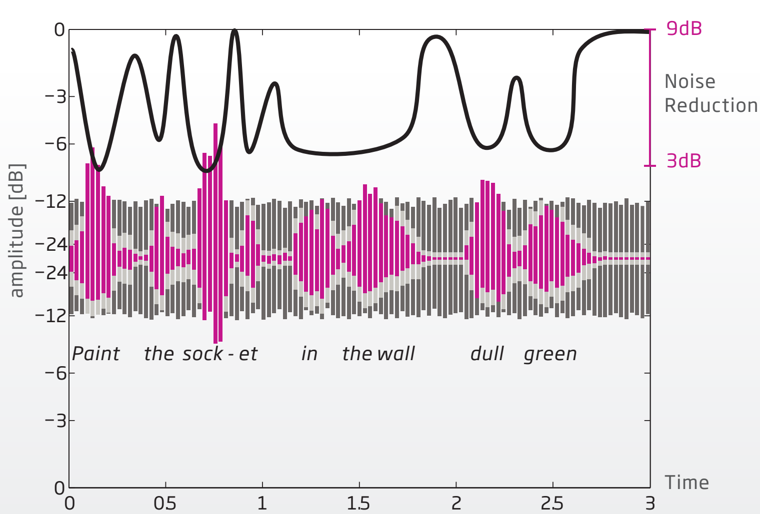

Figure 5: Illustration of the speed of the Noise Removal module. Speech sound (pink) and acoustical noise (dark grey) at an SNR of 0dB. The noise at the output of the instrument is illustrated in light grey. The black line illustrates the fast-acting noise reduction.

Fitting Opn – OpenSound Navigator

In order for clinicians and hearing aid users to take full advantage of OSN, an easy-to-use interface has been integrated in the fitting section of the new fitting software, Genie 2. The recommended way to fit and fine-tune OSN is a two-step process: (1) filling out personal information and answering five questions in Personalisation under Selection (2) fine-tuning in the new OpenSound Navigator screen, under Fitting, that includes YouMatic LX. Note that the personalisation step is optional and if not completed, the system prescribes a default setting.

Personalisation and prescription

The way OSN processes sound is prescribed and personalised based on how the five personalisation questions are answered as well as the age and the experience level of the user. This information is passed on to YouMatic LX to personalise the effect of the Balance and the Noise Removal modules of OSN.

The result of the personalisation is the assignment to one of three help profiles: Low, Medium and High, as well as the amount of noise reductions prescribed in simple and complex environments. The help profiles should be viewed as neutral, without positive or negative connotations. As an example, High help is not necessarily better than Low help. The assigned help profile is entirely dependent on the individual users' preferences and is a good starting point for the fitting.
If the personalisation step is skipped, the Medium level of help and a noise reduction level of 0 and -7dB in simple and complex environments, respectively are prescribed by default.

Fine-tuning

A new OpenSound Navigator screen has been designed in Genie 2 (see Fig. 6). Its purpose is threefold: to adjust OSN settings, to show graphically how OSN operates, and to use as a counselling tool with clients when appropriate. The OSN screen in Genie 2 has two main components: YouMatic LX, the clinician control of OSN in the bottom half of the screen and a graphical representation of OSN functionality in the upper half of the screen.

YouMatic LX ensures that the users' needs and preferences are supported in any acoustical environment. YouMatic LX replaces the Automatics Manager tab in Genie and the handles associated with this screen. Information provided in the personalisation step is used to define the default settings for OSN and is marked with default symbols as shown in Fig. 6. There are five adjustable parameters, described individually below.

Noise reduction – Simple

The word "simple" listening environment is used quite deliberately to define environments that may be quiet, but are not necessarily. Simple environments are here defined as low or medium level, with low reverberation effects, and few disturbing sound sources. Usually, there is a high SNR making it easy to hear target speech. If there are multiple sound sources, they are spatially separated making them easy to tell apart from each other. An example of a simple listening environment could be a living room where the television is on at a low level and there are two talkers, besides the hearing aid wearer, who are sitting on either side of the person.

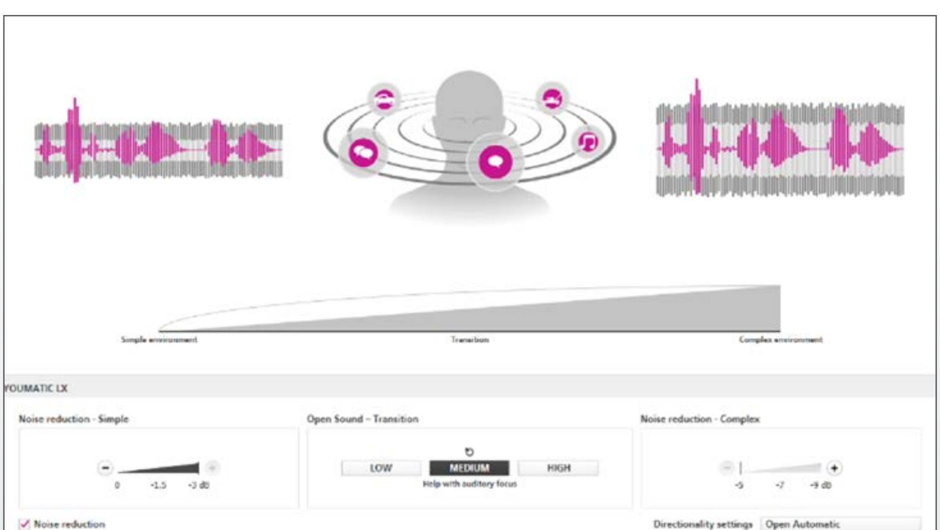

Figure 6: The OSN screen in Genie 2016: YouMatic LX at the bottom, visualization at the top.

YouMatic LX prescribes 0 or -1.5 dB of noise reduction for simple environments, depending on which help profiles the client is prescribed. The clinician can give the client up to -3 dB noise reduction for simple environments if desired. Table 1 gives an overview of prescribed noise reduction for the three help profiles. Recommendation: Fill out personalisation questions to determine your client's default setting for simple environments. Adjust setting if needed based on client comments.

Noise reduction – Complex

Complex listening environments are here defined as environments with a low SNR or a fluctuating SNR, and high sound levels. Multiple sound sources are present and are difficult to separate spatially. Reverberation and wind noise may be present and there may be noise sources that make it difficult to hear and understand target speech. An example of a complex listening environment can be a conversation between four people at an outdoor café on a busy street.

YouMatic LX prescribes -5 or -7dB of noise reduction for complex environments, again, based on the help profile of the client (See Table 1). The clinician can give the client up to -9 dB of noise reduction for complex environments if desired. Recommendation: Fill out personalisation questions to determine your client's default setting for complex environments. Adjust setting if needed based on client comments.

Noise reduction is shown graphically for both simple and complex environments above the respective handles. The dark grey colour represents noise before the effect of noise reduction and the light grey colour represents noise after the effect of noise reduction. The magenta colour represents speech. As more noise reduction is added, the band of light grey noise becomes narrower to illustrate the reduction. These representations are helpful to illustrate a point for counselling purposes, and it should not be taken literally.

Noise reduction

It is possible to deactivate the Noise Removal module of the OSN, although it is not usually recommended. This is done by unchecking the box for noise reduction in YouMatic LX. If this is done, the noise reduction handles for simple and complex environments will be greyed out. Recommendation: Leave noise reduction on, keeping in mind that noise is only removed when needed and that you are not taking useful sound away from your client.

Open Sound – Transition

This handle on YouMatic LX pertains to how the hearing aid behaves as the client moves from a simple environment to a complex environment. More specifically, is the hearing aid triggered to provide help while the client is still in a simple environment, or not until the environment is much more complex? This will depend entirely on the help profile of the client. For the Low help profile, the hearing aid will provide less help as the environment gets increasingly complex. For the High help profile, the hearing aid will provide more help. This is provided in two ways.

- The first way pertains to the Balance module; at low levels and high SNRs, the Balance module is constrained to give a response equivalent to that of the pinna of the human ear (Pinna Omni). Then, as the level increases and the SNR decreases, OSN is allowed to rebalance environmental sounds more (See Balance section). Its full effect is available for levels above 80dB SPL in the lowest frequency channels and 50dB SPL in the highest frequency channels and if the SNRs is below about 5dB in the channel.
- The second way pertains to the Noise Removal module, where more noise is removed as sound levels increase. Maximum noise reduction (-5dB, -7dB or -9dB) is applied for levels above 70 dB SPL and 40 dB SPL in the lowest and highest frequency bands, respectively. The amount of noise reduction for simple and complex is set by Noise reduction - Simple and Complex, but the way that the system transitions from one level of noise reduction to the other is shown by the Transition bar and determined by the help profile chosen.

These transitions between simple environments and complex environments are not points on a line, but rather, they are continuous and smooth. OSN cannot be described in terms of modes that the system switches between because in effect there are an infinite number of possible configurations of the system. In turn, this means that OSN does not have mode switch with the potential audible artefacts that mode switches are known to cause.

Noise reduction	Help		
	Low	Medium	High
Simple	0	0	-1.5
Complex	-5	-7	-7

Table 1: Overview of prescribed noise reduction for three help profiles.

A graphical representation of transition is shown by the bar in the middle of the screen, marked Transition. The grey shaded area represents help. If High help is chosen, the grey area widens across environments and if Low help is chosen, help escalates only when the client is in complex environments. Above the Transition bar, the head and sound circles represent the Balance module. In High help, the lawnmower and car icons will become quite small and in Low help, they remain somewhat larger and the speech circles are enhanced.

Recommendation: Fill out personalisation questions to determine your client's default help setting. As a follow up, you can ask your client this question, "Do you find it difficult to focus on important sounds when there is noise?" to start a conversation with your client about what their needs are as the acoustical environment gets more complex.

Directionality settings
These choices relate to the Balance module of the OSN. The Balance module can be set to Open Automatic, to provide the full benefit of the technology or it can be manually set into a Pinna Omni, or a Full Directional if desired. In Open Automatic, the Open Sound – Transition handles are active.

When the clinician selects Pinna Omni, the Balance module is constrained to give a response equivalent to that of the pinna of the human ear, regardless of the environment the listener is in. This means that signal processing does not transition in any specific way between simple and complex environments in terms of how directional the system is allowed to be – the system is fixed in Pinna Omni and Open Sound – Transition is greyed out. However, noise is still removed as sound levels increase and noise reduction can therefore still be set for simple and complex environments.

When the clinician selects Full Directional, the Balance module is constrained to give a front focus, regardless of the environment the listener is in. Again, there is no transition between environments and the Open Sound – Transition handle is greyed out. Noise reduction for simple and complex environments is active as with other directionality settings.
Figure 7 is a n overview of directionality settings availableto the clinician, in particular if the prescribed setting is not appropriate. There are five ways to set directionality in the Balance module of the OSN. In the OSN, help is defined in terms of Noise Removal and directionality of the system in the transition between simple and complex acoustical environments.

Recommendation: Choose Open Automatic to take full advantage of the Balance module of the OSN.

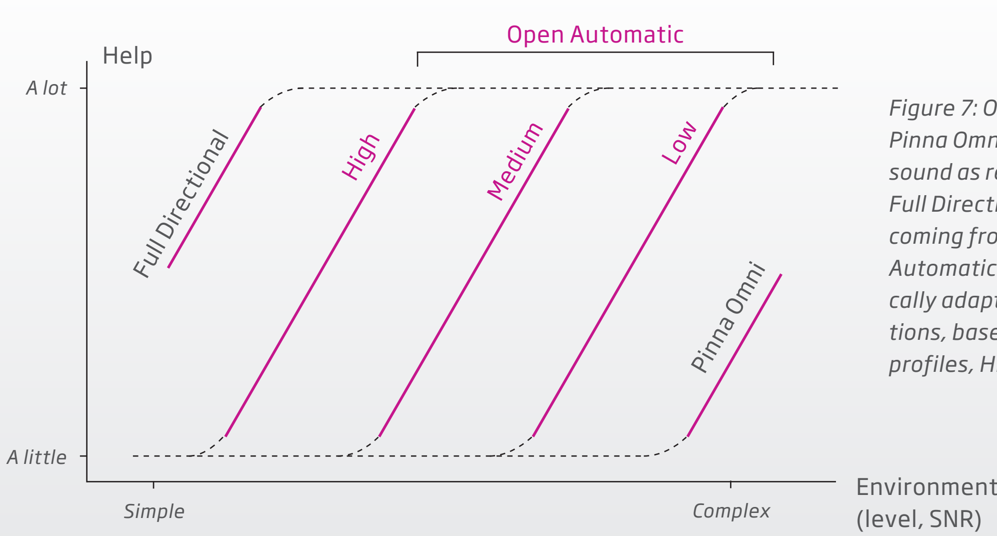

Figure 7: OSN directionality settings. In Pinna Omni, the hearing aid mimics sound as received by the human ear. In Full Directional, the focus is on sounds coming from the front. In Open Automatics, the hearing aid automatically adapts to the acoustical conditions, based on one of the three help profiles, High, Medium, or Low.

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