

New Technology for Effortless Hearing: A “Unique” Perspective

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The UNIQUE, a new hearing aid from Widex, is designed to provide effortless hearing by addressing the nuances hearing aids need to capture and process for patients' unique, individualized listening needs.

Numerous studies were available in the last few years to support a strong relationship between cognition and hearing.¹ What is noteworthy from these studies is that the type of amplified sounds could place different levels of cognitive demands on the impaired ears.^{2,3} Thus, a reasonable objective in hearing aid design is to deliver amplified sounds that require the least effort from the hearing aid wearers to hear satisfactorily in all listening environments, regardless of their cognitive capacity. We term this design objective Effortless Hearing. The goal is that hearing-impaired people hear more of what they want (to hear) and less of what they don't.

To achieve effortless hearing, it is logical to expect that the hearing aids are able to capture all the necessary nuances of the incoming sounds (ie, natural hearing). Because not all natural sounds are desirable (eg, refrigerator noise), the hearing aids must be able to “purify” and process the incoming sounds in such a way that they retain all the necessary nuances without the distractions that may make them more effortful to listen.

This paper provides a review on what nuances hearing aids need to capture, what sounds need to (and can) be purified, and how processing should be done to individualize the input sounds. A description about how the new Widex UNIQUE hearing aid performs these tasks to achieve the objective of effortless hearing is also provided.

Basis of Effortless Hearing: Capturing the Natural Nuances

Effortless hearing requires that all the natural nuances within the input sounds are captured accurately. This includes information in the frequency, intensity (including intensity changes and relative intensities between ears), and temporal domains. In a digital hearing aid, the sampling frequency of the analog-to-digital converter (ADC) determines the frequency range of sounds that can be captured. For example, a sampling frequency of 33 kHz (as used in all Widex digital hearing aids) captures sounds up to 16.5 kHz, whereas a sampling frequency of 16 kHz only captures sounds up to 8 kHz.

The number of bits used in the ADC also determines the range of sounds and the maximum SPL that can be captured by the hearing aid. To preserve all the natural nuances of the input sounds, the ADC must be able to capture sounds as soft as 0-10 dB SPL and as loud as 110 – 115 dB SPL.⁴ The ability to preserve the linearity of the inputs at these levels prevents the risk of distortion at the input stage which may degrade the effectiveness of subsequent signal processing algorithms.⁵

Because the auditory system uses input from both ears to assist in spatial awareness and sound localization,⁶ bilateral hearing aid fitting is a fundamental requirement for effortless hearing. In addition, preserving the interaural intensity difference (IID) and interaural timing difference (ITD) of the inputs from both ears is critical in preserving the temporal characteristics of the input for subsequent processing.

Purifying the Input Sounds—Hearing Less of What You Don't Want to Hear

Not all sounds that are captured by hearing aids are desirable for hearing aid wearers. Having only the desirable inputs could improve the ease of listening and lessen the use of cognitive resources so that the limited resources can be allocated to other relevant tasks. Two types of sounds are especially relevant to remove.

Internal noise. The internal noise from a hearing aid originates from its analog input stage and the ADC. People with a mild, sloping hearing loss are more likely to perceive the internal noise while receiving the needed amplification from the hearing aids. A mechanism to purify the internal noise while preserving the audibility of soft speech is desirable.

Wind noise. Wind turbulence at the hearing aid microphones results in wind noise. The amount of noise is dependent on the wind speed and the position of the microphone opening. MarkeTrak VIII⁷ shows that only 58% of hearing aid wearers are satisfied with the performance of their hearing aids in windy situations. Minimizing the impact of wind (subjective annoyance, speech understanding) could result in effortless hearing in those situations.

Processing the Input Sounds – Hear More of What You Want to Hear

A basic goal in hearing aid amplification is to apply appropriate gain to the frequencies where the hearing is deficient (ie, hearing loss compensation). In addition, such changes must also account for the individual differences with changes in acoustic environments. Fundamentally, it requires that the hearing aid provide adequate gain and output headroom (MPO), such that the processed sounds can be delivered without spectral and/or temporal distortion. Kuk et al⁸ reported that a reduced MPO degraded SNR by 2 dB over a hearing aid with an optimal MPO setting. Other important considerations include:

Appropriate compressor characteristics. The compressor is the core processing unit of modern hearing aids, and its parameters must be selected carefully.⁹ This includes the determination of the optimum number of compression channels,¹⁰ the appropriate compression ratio (CR) at each channel, and the time constants used to regulate gain changes.

A fast time constant ensures audibility, but it can distort the spectral contrasts of the input signals. A longer time constant has the advantage of maintaining the linearity (or temporal nuances) of the input signal.¹¹ Souza¹² also reported that longer time constants have the advantages of speech intelligibility and sound quality, especially for the individuals with cognitive challenges.

Unfortunately, a fixed long time constant may not always be able to follow the fast intensity changes so that loss of audibility may occur. While solutions like adaptive release time (also called adaptive compression) are adequate for a single speed compressor in managing the changing sound pressure levels, they may not be as responsive as compressors having variable speeds.

Automatic program changes. Dedicated listening programs, such as music and party settings, should be available to enhance sound appreciation and listening comfort in special environments. While wearers can select and use these programs manually, it can be effortful for two reasons. First, wearers need to know the nature of their listening environments so they can select the right listening program. Secondly, wearers have to switch to the right program. It is desirable that hearing aids analyze the acoustic environments and automatically adjust the parameters to the most appropriate settings without the need of wearer manipulation. An automatic sound classification and processing system could ensure that the wearers hear more of what they want and less of what they don't.

Managing background noises. Background noises contaminate speech sounds and make their identification more difficult. If the desirable sounds are spatially separated from the undesirable sounds, a directional microphone can provide relative enhancement of the speech signal to improve listening comfort and speech intelligibility.¹³

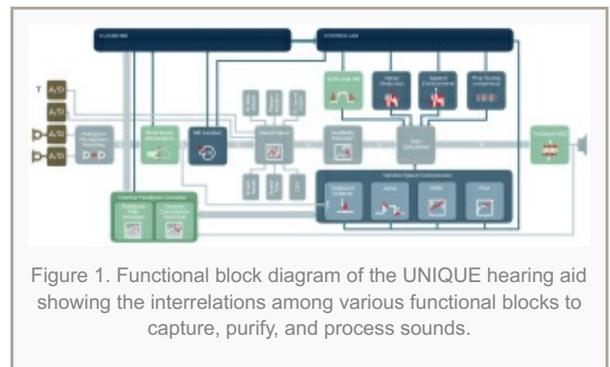
Noises that are spectrally different from the desirable speech sounds (eg, airplane and car noise) can be managed with noise reduction algorithms. On the other hand, noises that are spectrally and spatially similar to the desirable sounds are more difficult to remove. These situations require separate post-input processing based on certain assumptions of “speech” and “noise” characteristics. The patented Speech Enhancer feature in Widex hearing aids performs gain reduction while considering the configuration of the wearers’ hearing loss.¹⁴ In addition, where gain increase is desirable and permitted, it will do so to optimize the individual speech intelligibility index (SII) for that specific listening environment. Such an algorithm has also demonstrated an improvement of speech in noise ability by 2 dB in a steady-state noise.¹⁵

Preservation of inter-aural cues. The central auditory system uses input from both ears to form spatial judgment, such as spatial awareness, localization, distance estimation, etc. The use of independent fast-acting compression in each ear can disrupt the loudness and timing relationship of sounds between ears (IID and ITD cues). This may alter the spatial relationship of sounds and lead to poorer spatial performance. The use of algorithms, such as the digital pinna, are designed to preserve the front/back localization cues.¹⁶ The use of wireless connectivity between two hearing aids of a bilateral pair allows the preservation of such cues.¹⁷

Amplification for the “unaidable” ear. Amplification for hearing losses that result from a complete loss of inner hair cells or “dead” regions¹⁸ can result in a distorted perception and a decrease in speech understanding. In those cases, the use of frequency lowering in addition to the core processing may be necessary.^{19,20}

UNIQUE Capturing, Purifying, and Processing

The UNIQUE is the newest wireless hearing aids from Widex. Through hardware and software enhancements, the UNIQUE aims to provide Effortless Hearing as the ultimate wearer benefit. This paper only provides an overview of the new features in the UNIQUE that are responsible for capturing, purifying, and processing sounds so hearing aid wearers can hear more of what they want and less of what they don’t. Figure 1 shows a functional block diagram of the new hearing aid. Subsequent papers will provide more details and evidence to support the features.



Capturing Starts with a Powerful Building Block

The UNIQUE is based on a new chip that is almost 60% more complex than the previous chip. The hardware enhancements allow the UNIQUE increased processing complexity to capture more sounds and to tailor the environmental input sounds to the wearer’s needs in more sophisticated manners. Notwithstanding, the current power consumption remains at 1 mA.

Four analog-to-digital converters (ADC). There are 4 independent ADCs in the UNIQUE: two ADCs for the front and back microphones, one for the T-coil input, and one for the direct audio input (DAI). This means that inputs from each of the four input sources can be independently sampled and processed without any compromises.

For example, the previous MT option will leave the microphone in an omnidirectional mode while the user is using the T-coil. In the UNIQUE, the MT option would allow a directional microphone (which requires two microphone inputs) and the T-coil use at the same time. This allows the device to capture a cleaner acoustic input in noise while the user is using an MT option.

Increased input dynamic range. The new hearing aid uses a new 18-bit ADC. This higher bit depth means that the linear input dynamic range is extended to 108 dB (DR in dB = # bits x 6). The UNIQUE keeps the upper linear input

limit at 113 dB SPL, but extends its lower limit to 5 dB SPL in order to fully utilize the new linear 108 dB input dynamic range (Figure 2). This suggests that it would capture linearly the full range of input sounds permissible by current microphone technology. This minimizes input distortion and allows for more accurate processing.

Purifying: Hearing Less Unwanted Sounds

The new hearing aid uses True Input Technology to minimize distortions at the input stage. Two new algorithms are in place to purify the input further for optimal subsequent processing:

Soft level noise reduction. One of the distinguishing features of Widex hearing aids is their low compression thresholds. This has the advantage of providing more gain for soft sounds than hearing aids with a higher compression threshold. While greater gain for soft sounds is desirable for consistent speech intelligibility²¹ and for patients with tinnitus,²² it may not be appreciated by people who have a milder hearing loss or new hearing aid wearers. The soft level noise reduction (SLNR) algorithm is designed so UNIQUE wearers may hear less soft noises (ie, increased comfort) without affecting soft speech.

The design rationale behind the SLNR is that speech and non-speech (noise) signals have different temporal and modulation characteristics. By utilizing a new speech detector in the new hearing aid, sounds below 62 dB SPL are classified into speech sounds and non-speech sounds. Gain for speech sounds is maintained, whereas gain for non-speech sounds is reduced. This has the effect of minimizing soft noises in quiet environments (such as refrigerators, fans etc) and also circuit noise to enhance listening comfort without affecting the audibility of soft speech sounds. A study conducted at ORCA-USA confirmed that the feature maintained intelligibility for soft speech presented in quiet (50 dB SPL) and in the presence of fan noise while minimizing the perception of the fan noise.

Wind noise attenuation algorithm. The UNIQUE implements all the mechanical designs in Widex hearing aids to minimize wind noise. In addition, it utilizes a patented, software-based wind noise attenuation feature consisting of two major functional stages: the detection stage (involving wind noise detection and input selection), and a reduction stage (involving an adaptive filter). The detection stage exploits the fact that wind noise is largely uncorrelated at the two microphone openings of a directional microphone while speech sounds are correlated. The presence of wind noise is determined by examining a combination of correlation, frequency spectrum, and energy level from the two microphone inputs.

During the input selection part of the detection stage, the microphone signal with the better SNR is selected as the input signal to the hearing aid amplifier. The other microphone signal is used in the adaptive filter to estimate what is the wanted signal and what is wind noise. Once wind noise is identified, it is attenuated from the microphone signal with the better SNR.

The effectiveness of this algorithm was evaluated at ORCA-USA with wind blowing directly in front at a speed of 5 m/s (as in leisure biking) and speech (nonsense syllables) presented to the side at levels from 60 to 75 dB SPL in 5 dB increments. Recordings were made from KEMAR in a wind tunnel and a total of 15 subjects were tested under earphones. Figure 3 shows almost 50% improvement in phoneme scores or an equivalent improvement in SNR of 8.4 dB using a 50% criterion.

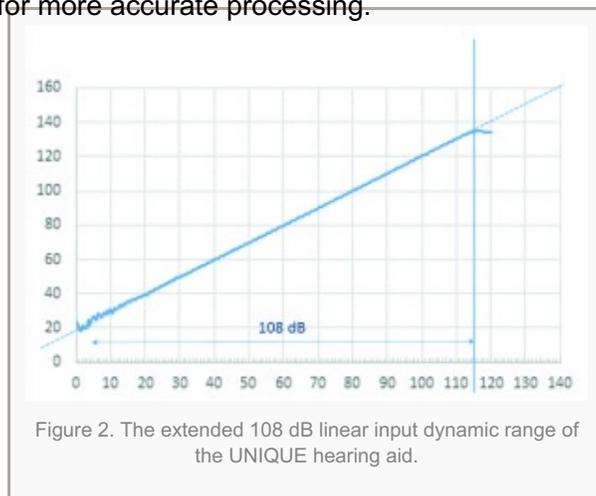


Figure 2. The extended 108 dB linear input dynamic range of the UNIQUE hearing aid.

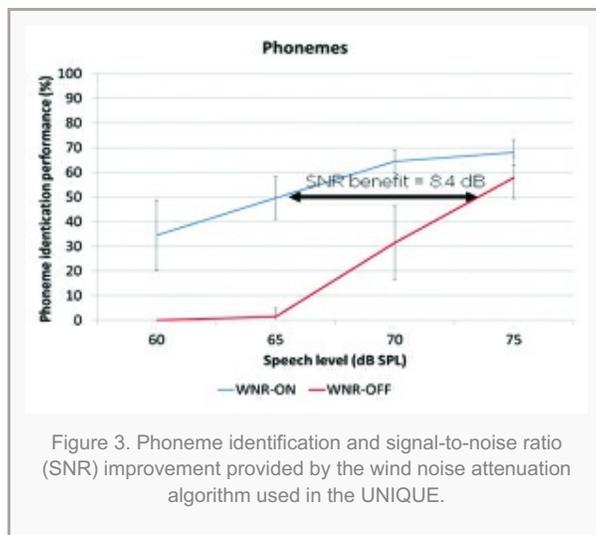


Figure 3. Phoneme identification and signal-to-noise ratio (SNR) improvement provided by the wind noise attenuation algorithm used in the UNIQUE.

Processing: Hearing More Wanted Sounds

The increased hardware complexity available in the UNIQUE allows the implementation of more sophisticated processing algorithms so that desirable sounds are heard more consistently in a wider variety of listening environments. The newest processing features that could result in the wearers hearing more wanted sounds include:

Variable speed compressor. The new hearing aid uses the sum of two compressors working in parallel to determine the final gain. The primary compressor uses longer time constants. It forms the main compressor of the hearing aid, as it adjusts the overall loudness of the input while preserving its temporal waveforms. The secondary compressor uses a shorter time constant, as it follows the rapid short-term dynamics of the inputs in situations where this can be beneficial. A Jump system is also used to facilitate smooth gain change. The benefit of the variable-speed compressor system (over the single-speed compressor) is the responsiveness of the system for all input levels in highly dynamic (ie, changing intensity) environments. This has the potential for more consistent speech understanding and listening comfort at all input levels.

Figure 4 compares the responsiveness of the single compressor system with the UNIQUE's variable-speed compressor system. The relative gain levels of the two compressors to a 50 dB SPL syllable following a carrier sentence presented at 80 dB SPL are shown. One can see that the gain level from the new variable-speed compressor system recovers faster than the single-speed compressor system. This ensures that sufficient gain is available for the softer sounds quickly. This allows more consistent audibility and requires less listener effort to identify the sound.

The opposite is true when the compressors are presented with a soft sound (50 dB SPL) followed by a loud sound (80 dB SPL). In this case, the variable-speed compressor (red) also reaches the steady state gain faster than the single speed compressor (blue) (Figure 5). This provides more comfort to the listener and promotes greater acceptance as well.

A major limitation with fast-acting compression is that it minimizes the intensity fluctuations of the input. This removes the available temporal cues and may potentially degrade speech understanding.¹² This potential limitation is minimized in the new device using two controls.

First, it is assumed that input signals that show a high degree of modulation (eg, speech in quiet) may withstand some degree of temporal smearing before a consequential degradation of its identity occurs. Input signals that show little modulation (eg, speech in noise) may not be able to withstand as much smearing. Thus, a longer time constant is used in most situations, including the more difficult ones (such as noise), to preserve as much of the temporal waveform as possible. If fast-acting compression is applied, it is only applied in the "quiet" situations where high modulation is expected.

Because people with a more severe hearing loss are more dependent on the temporal cues than people with a milder loss, fast-acting compression is not utilized for those with hearing losses above 75 dB HL and at high CRs. Second, the new compressor integrates the action of the adjacent bands (ie, one octave instead of 1/3 octave) in its fast-acting

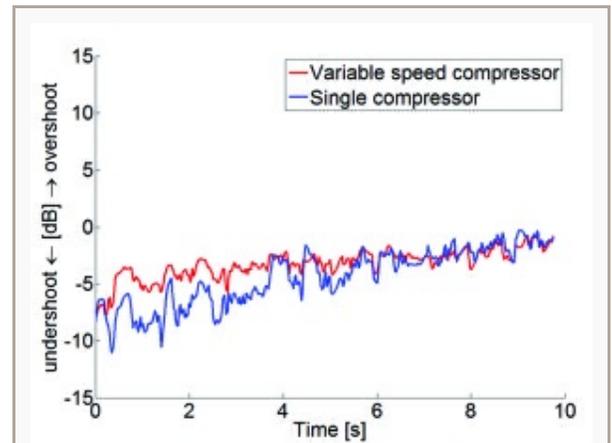


Figure 4. Responsiveness of the variable speed compressor (red) to a sudden drop in sound intensity (from 80 dB SPL to 50 dB SPL) over the single compressor (blue).

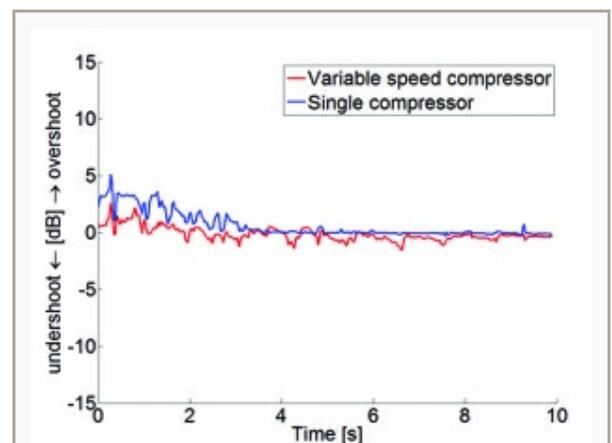


Figure 5. Responsiveness of the variable speed compressor (red) to a sudden increase in sound intensity (from 50 dB SPL to 80 dB SPL) over the single compressor (blue).

compression action. This also minimizes any potential smearing, and ensures good sound quality and consistent speech audibility.

New Sound Class Technology. The new Sound Class Technology (SCT) used in the UNIQUE is intended to provide automatically more optimal amplification settings based on the acoustic characteristics of the listening environments. The SCT has two components: a classifier and a controller. In the classifier, the input signal is analyzed on 12 attributes (eg, envelope modulation, amplitude modulation, etc). These attributes are compared to a class library (or database) consisting of a discrete number of sound-base classes. These classes are derived based on machine training (and subsequent real-life validation) of real-life bilateral recordings of over a hundred different situations that hearing aid users typically encounter. A running estimate of the correlation between each of the pre-trained base-class and the current set of attributes is performed. A separate speech detector is also available to estimate the presence of speech. The sound class that shows the highest probability of explaining the sound attributes is chosen as the current base-class.

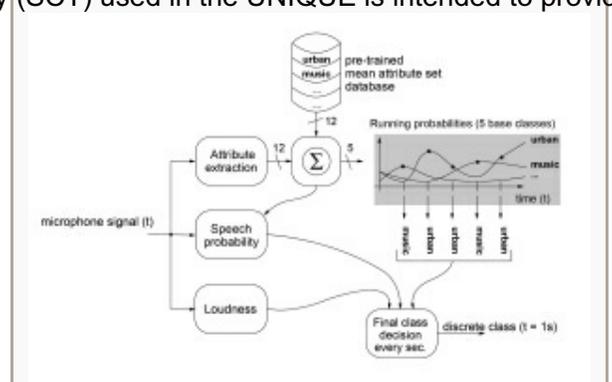


Figure 6. The classification process. The input signal is analyzed on a number of attributes and compared to a database built on a large number of real-life recordings. A separate analysis is also conducted to determine if speech is present in the signal. A probability estimate is used to determine the current class.

A total of 9 different sound classes—5 where speech is absent and 4 where speech is present—are defined through this training process: 1) quiet; 2) urban noise; 3) transportation noise; 4) party noise, 5) music; and where speech is present: 6) quiet speech; 7) urban with speech; 8) party with speech, and 9) transportation with speech.

The classifier provides output in the form of a discrete class every second. When the hearing aids are turned on, they start in the “quiet” class. The classifier requires the identification of a new class 3 seconds in a row for a shift to be implemented.

Once a decision is made on a sound class, the range of parameter settings from the previous sound class will be updated automatically to optimize for the new environment. This is achieved via the controller of the SCT. This could include the various parameters (eg, ratio, time constants, gain etc) on the compression system, the amount of impulse sound reduction (sound softener), the amount of soft-level noise reduction, the speed of the Speech Enhancer, etc. Such changes can be programmed for quick or gradual implementation. The types of parametric changes are illustrated in Figure 7.

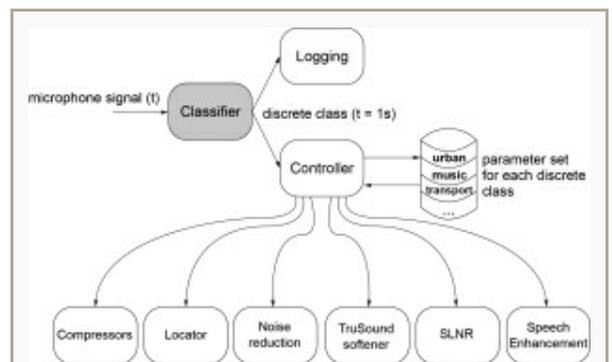


Figure 7. The controller adjusts parameters of the various features on the UNIQUE hearing aid to optimize the settings for the specific environment (or sound class).

While the SCT allows automatic treatment of sounds in various acoustic environments, some may desire more audibility in a party noise situation, while another may prefer more listening comfort in the same environment. To allow for individual preferences, the fitting software (Compass GPS) allows individual tailoring of audibility vs comfort for the overall program and each sound class. Additionally, a Preference Control (PC)—accessible via a toggle switch on the hearing aid, the remote control (RC-DEX), or using a smartphone App with the COM-DEX—is made available for the wearers to adjust for more audibility or more comfort. The PC adjusts each individual feature settings (eg, directional microphone, noise reduction settings) to optimize satisfaction. A pilot study at ORCA-USA shows improved wearer performance on speech intelligibility and listening comfort with the use of the PC. This highlights the importance of allowance for personal preferences in the design of effortless hearing.

Real-Time Speech Enhancer (RTSE). Widex introduced the patented Speech Enhancer (SE) in 2006 as a novel

way to ensure speech audibility in noise.¹⁴ The RTSE system in the UNIQUE allows much quicker speech detection and gain implementation without artifacts (<5 s). This improved responsiveness is especially advantageous over other noise reduction systems in environments where the noise characteristics change rapidly and unpredictably. When the SE is used as part of the UNIQUE SCT system, it also changes its objective from maximizing speech intelligibility in noise when speech is dominant to maximizing comfort as the noise intensity increases above the speech signal.

High frequency boost (HFB). The input-output (I-O) characteristics for frequencies above 4000 Hz in the new hearing aid are designed to provide extra audibility while ensuring subjective comfort. Figure 8 shows the in situ I-O curves at 6000 Hz between the HFB On (blue) and Off (red) for a 60 dB hearing loss. Output levels above the hearing loss (60 dB HL) is audible, and that below is inaudible. The input level where the I-O intercepts the hearing loss is the predicted aided threshold of the individual.

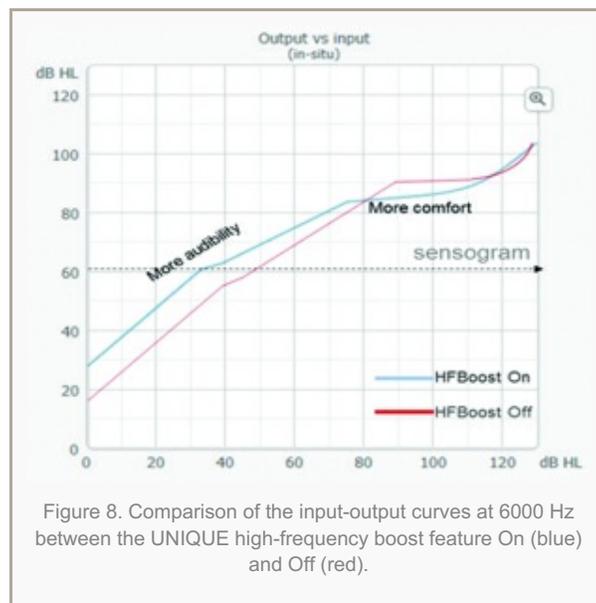


Figure 8 also shows that the I-O for the HFB-On provides more output than the HFB-Off below an input level of 80 dB HL. This means sounds below this level may be audible with the HFB-On but not with the HFB-Off. Indeed, the predicted aided threshold is 25 dB HL for the HFB-On and 45 dB HL for the HFB Off. The lower and relatively constant output from the HFB-On compared to the HFB-Off above 80 dB HL input means that the louder high frequency sounds remain comfortably loud even when the input increases. This ensures consistent audibility for speech understanding and music appreciation at a comfortable level for its intended candidates.

Enhanced Audibility Extender (AE). The AE was introduced in 2006 as a means of providing unaidable information in the high frequencies.²³ The new AE builds on the success of the current AE. It allows further precision and individualization so the wearers are more likely to accept the new sound and improve their hearing aid satisfaction.

The new speech detector is utilized in the AE to distinguish between voiced and unvoiced sounds in the source region. Voiced speech sounds are attenuated before transposition so it is audible but not distracting. Voiceless speech sounds, such as voiceless fricatives (like /s/), are transposed without attenuation to add saliency. In addition, a new and improved harmonic tracking system is used in the enhanced AE to ensure proper alignment of the harmonics between the source and target signals.

The enhanced AE also allows a variable bandwidth of amplification. This tailors the amplification/transposition characteristics more closely to the etiology of the person's hearing loss. Along with the AE acclimatization program, these new features ensure more immediate acceptance and satisfactory use of the AE.

Who Benefits from the UNIQUE?

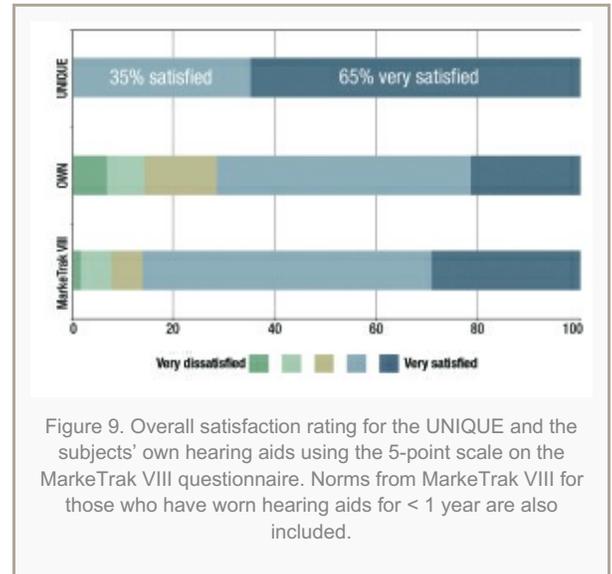
By capturing as much of the natural nuances as possible, purifying them from wind noise and soft level noises, and processing them to further enhance its consistent audibility, comfort, and SNR, the UNIQUE aims to provide as much needed information to its wearers as possible to achieve effortless hearing. This could allow successful communication in many more daily listening situations—including those where there are rapid changes in the speech and/or noise characteristics—with greater ease and confidence.

In two pilot studies at ORCA-USA totaling 25 experienced wearers of hearing aids, all subjects rated the UNIQUE to be satisfactory (35%) or very satisfactory (65%) (Figure 9). This compared favorably to the MarkeTrak VIII norms⁷ where 86% of wearers were satisfied or very satisfied with their own aids that were purchased within one year. This

reaffirms that the UNIQUE, when properly fitted, is appropriate for the majority of hearing aid wearers to provide satisfactory hearing.

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