Coordinated Adaptive Processing in the Neuro Cochlear Implant System

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Abstract: Oticon Medical's sound processing features a unique combination of hearing instrument and cochlear implant technologies. With Oticon's advanced Inium powerful platform inside, Neuro sound processors enable automatic features within a state-of-the-art model called "Coordinated Adaptive Processing". This novel approach revolutionizes conventional sound processing systems by providing the perfect balance and coordination of its full range of advanced hearing instrument sound processing features and post-processing operations: specifically Inium environment detection, Free Focus directionality, Wind Noise protection, Voice Track noise reduction and Voice Guard speech enhancement. Given the proven benefits of these individual components, Coordinated Adaptive Processing is expected to deliver greater performance, by ensuring that the user is always placed in the best listening situation for optimal speech understanding across all sound environments.

This document provides a comprehensive overview of the new sound processing model and the different advanced processing technologies on which it relies.

Introduction

Cochlear implant systems translate acoustic information sampled from the acoustic environment into electrical impulses delivered directly to the patient's auditory nervous system in an attempt to recreate functional hearing. After more than 30 years of intensive development, the efficacy of cochlear implants for the rehabilitation of severe to profound hearing loss has been well established, with no less than 400,000 patients worldwide. As a result of the challenges and the excitement of discovering or rediscovering the rich sensory modality of hearing, cochlear implant users have very specific needs in terms of sound quality and the audibility of acoustic information.

Cochlear implant users must be able to enjoy the natural richness of sounds available in the environment without compromising sound quality in speech or music. Therefore, dedicated cochlear implant sound processing strategies must be able to provide the richest sound experience possible, offering the full range of sound to stimulate the nervous system with complex and detailed signals. This will promote and encourage neural plasticity mechanisms and stimulate auditory development. Cochlear implant sound processing strategies should also aim to reduce listening effort and provide maximally intelligible speech sounds in complex listening situations such as in noisy backgrounds or where there are multiple talkers in the same room. In these situations, cochlear implant sound processors must offer adaptive and efficient directional microphones and noise reduction systems to alleviate the challenges of hearing.

In recent years, sound processing features inherited from hearing instrument innovation, have been added to the digital cochlear implant signal processing chain, with the ultimate goal of continuously improving hearing outcomes, especially in challenging listening environments. Oticon Medical is moving one step closer towards that goal with the introduction of Coordinated Adaptive Processing in Neuro sound processors.

Figure: Coordinated Adaptive Processing

I. Coordinated Adaptive Processing

Coordinated Adaptive Processing is a unique sound processing concept developed for Neuro sound processors that introduces a novel paradigm to the cochlear implant industry.

The signal processing chain implemented in Coordinated Adaptive Processing can be seen in (Figure 1).

Coordinated Adaptive Processing was especially developed to adapt the most recent technologies available in Oticon hearing instruments to sound delivery paradigms in cochlear implant systems. It aims to automatically deliver the perfect coordination and balance between its different sound processing tools in order to allow access to maximum speech audibility and sound clarity in every listening situation.

Two main ideas led the path to the development of Coordinated Adaptive Processing:

- Deliver the richest sound experience possible to cochlear implant users by capturing sounds from the environment over the widest possible dynamic range and apply sound processing algorithms without introducing or propagating distortion.
- Maximize speech and sound quality in every listening situation, by integrating a unique combination of hearing instrument algorithms and cochlear implant dedicated sound treatments, driven by continuous monitoring of the acoustic environment.

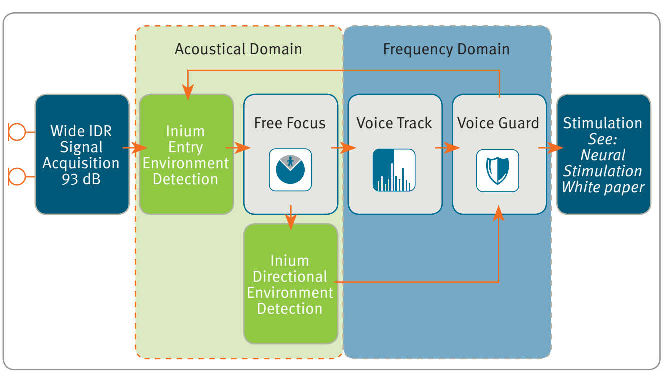

Figure 1. General processing scheme of Coordinated Adaptive Processing.
In Coordinated Adaptive Processing, sound processing starts with a wide input dynamic range (IDR) dual microphone signal acquisition, in order to benefit from the richest sound possible entering the processing chain. The Inium environment detection system then offers real-time analysis of the acoustic environment, selecting the ideal directionality mode of the Free Focus adaptive directionality technology, together with a dedicated wind noise reduction algorithm. The signal is then transformed to the frequency domain and Voice Track reduces noise in selected spectral channels. Finally, Voice Guard applies dedicated multiband instantaneous output compression, based on the analysis of the environment detection system. The entire system is constantly adapting its behaviour to the listener's actual acoustic environment. The information is then transferred by forward telemetry to the Neuro Zti Implant.

II. Wide-IDR Signal Acquisition

Cochlear Implant systems should account for the large difference between the acoustic dynamic range of natural sounds (i.e., the difference between the softest and loudest perceptible sounds: 90-100 dB range) and the reduced electrical dynamic range (EDR) of the electrodes (i.e., the difference between the smallest and the largest electrical stimulation applied to the auditory nerve: 10-30 dB) (Skinner et al., 1997; Zeng & Galvin, 1999; Vargas et al., 2012 and Figure 2a).

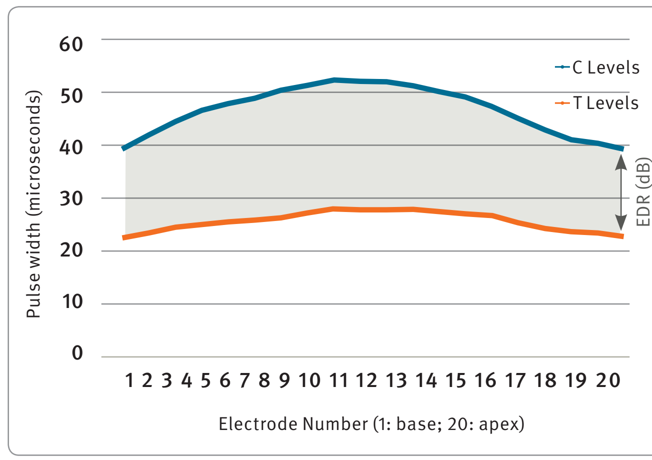

Figure 2a. Average Electrical Dynamic Ranges in Oticon Medical CI users.
An analysis performed over 523 maps obtained with the Digisonic® SP cochlear implant system revealed average EDRs between 17.2 dB and 25.9 dB, respectively on electrode 20 (most apical) and electrode 9 (1 position more basal than the middle of the array).

Input sounds must therefore be greatly compressed, potentially leading to signal distortion that can limit speech intelligibility (Hornsby & Ricketts, 2001; Neuman et al., 1994; 1998).
One way to circumvent the issue of compression in cochlear implants is to limit the range of sounds entering the system by continuously adapting the input dynamic range (IDR) of the microphones, using automatic gain control (AGC). AGCs, usually placed at the input/front-end of the processing chain, adaptively reduce the IDR, with the goal of always focusing on meaningful information in incoming sounds (Loizou, 1998).

However, reducing the IDR is a less than optimal compromise as it causes information loss and introduces signal distortion (Van Hoesel et al., 2002) propagated along the entire signal processing chain. Speech intelligibility can even be compromised if the IDR becomes too narrow, around 30 dB for example (Spahr et al., 2007), due to the large variations in the intensity of speech sounds. (Figure 2b).

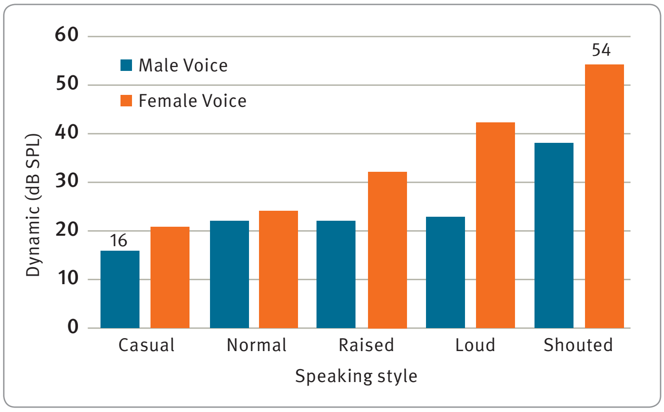

Figure 2b. Average Acoustic Dynamic Range of Speech.
The natural dynamic range of speech sounds (the difference between the softest and the loudest sounds in an utterance), can evolve from 16 up to 54 dB SPL depending on the talker and the loudness. Adapted from: Olsen, Am J Audiol. 1998, 7(2):21-25.

Since they limit the acoustic signal at the entry-point of the system, front-end AGC systems may randomly affect all regions of the sound spectrum and tend to weaken the available signal for further analyses and processing (Khing et al., 2013). On the contrary, use of a wider IDR can be beneficial for cochlear implant users (Zeng et al., 2002), especially if the compression system is specifically designed to account for a wide IDR (Spahr et al., 2007). Similar results were reported in hearing instrument users (Oeding & Valente, 2015), where increasing the IDR of hearing instruments showed improvement in both speech and speech in noise perception scores.

The Neuro sound processing chain does not implement a traditional AGC system in order to avoid limiting the instantaneous IDR and to avoid introducing transition artifacts into the acoustic signal. Instead, Coordinated Adaptive Processing uses the full range of input sounds, acquired across a large dynamic range (from 23 to 115 dB – 93 dB IDR); combined with Voice Guard's automatic multiband output compression system. The aim is to maximize the available acoustic information in the system to perform detailed analyses and modifications, and to ensure maximal audibility without any signal loss.

In Neuro sound processors, sound is captured from the environment by two microphones across a 93 dB input dynamic range and is digitized at 16,667 Hz (18 bits) over a frequency range up to 8268 Hz.

References on Dynamic Ranges (IDR & EDR)

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Khing PP, Swanson BA & Ambikairajah E. The effect of automatic gain control structure and release time on cochlear implant speech intelligibility. PLoS One. 2013 Nov 28; 8(11):e82263
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III. Inium Environment Detection

Coordinated Adaptive Processing is based on advanced Oticon Inium processing technology. Built on a powerful quad-core processor, Inium offers high performance signal processing in a small and power efficient chip. Inium includes a dual-microphone acoustic environment detection system which continuously monitors and analyzes the environment by analyzing the acoustic energy in different frequency bands. In Neuro sound processing, these data are used to seamlessly adapt (time-constant of 2s) signal processing algorithms to maintain high audibility, maximal hearing comfort and optimal speech intelligibility levels. In addition, the environment detection system is constantly exchanging information with the directionality module, Free Focus.

The environment detector will constantly analyze the three main directionality modes in all 64 channels to ensure that the best combination of directionality parameters in any one listening environment is always selected.

The result of this combination is also sent to the output adaptive compression system, Voice Guard, to coordinate the compression system and the directionality configuration. This creates a unique combination of synergies in the sound processing chain, in which each step is modulated by the state of all other modules inside the processing stream.

IV. Free Focus: Intuitive Automatic Adaptive Directionality

Our auditory system constantly adapts its sensitivity to different locations in space to be able to focus on the most important signals in a dynamically changing acoustic environment, For example, we are able to concentrate on one person talking to us, even if there are others having a discussion behind or beside us. The Free Focus automatic adaptive multiband directionality system was designed to mimic this ability. Free Focus for Neuro sound processors is based on proven Oticon directional technology, which has been shown to improve speech intelligibility scores in noise as well as reducing listening effort in noise in groups of hearing instrument users (Weile et al., 2011; Weile et al., 2013).

Free Focus was adapted to offer the benefits of directional microphones to cochlear implant users. It includes a noise reduction component which specifically addresses wind noise by choosing a custom directionality configuration to automatically reduce wind noise without loss of important speech cues. Neuro sound processors can operate in three main directionality modes: Omnidirectional, Split-directional and Full-directional. Each mode is based on four different frequency clusters to reshape the spatial sensitivity of the system. Free Focus is designed to automatically offer the best signal to noise ratio, even in the most challenging conditions, so that the user is always placed in the best listening situation.

In Omni mode, all spatial axes are equally represented, with the exception of 0-180 degree azimuth, in which sounds coming from the back are slightly attenuated allowing a focus on sounds originating from the front (Figure 3a). In this mode, the normal forward-facing listeners have an advantage due to the position of the pinnae. This corresponds to the most natural listening configuration and is used whenever possible to enhance natural sound awareness and to reduce unwanted directional effects.

The Split-directional mode is an extended omnidirectional configuration. The response is omnidirectional in the low frequencies, to effectively replicate the acoustic environment, while in the high frequencies, starting at 2000 Hz, the microphones exhibit increased sensitivity to 0 degrees azimuth (Figure 3b). This mode highlights speech cues that are less resistant to background noise which usually originate from the front. This mode is perfectly suited to improving speech understanding or music perception in low to medium-noise situations.

In the Full-directionality mode, sounds coming from the front are captured in all frequency bands. This is the equivalent of a full dual directional microphone (Figure 3c). In this mode, sounds coming from the front are picked up and background noise coming from the side or from behind are filtered out. This

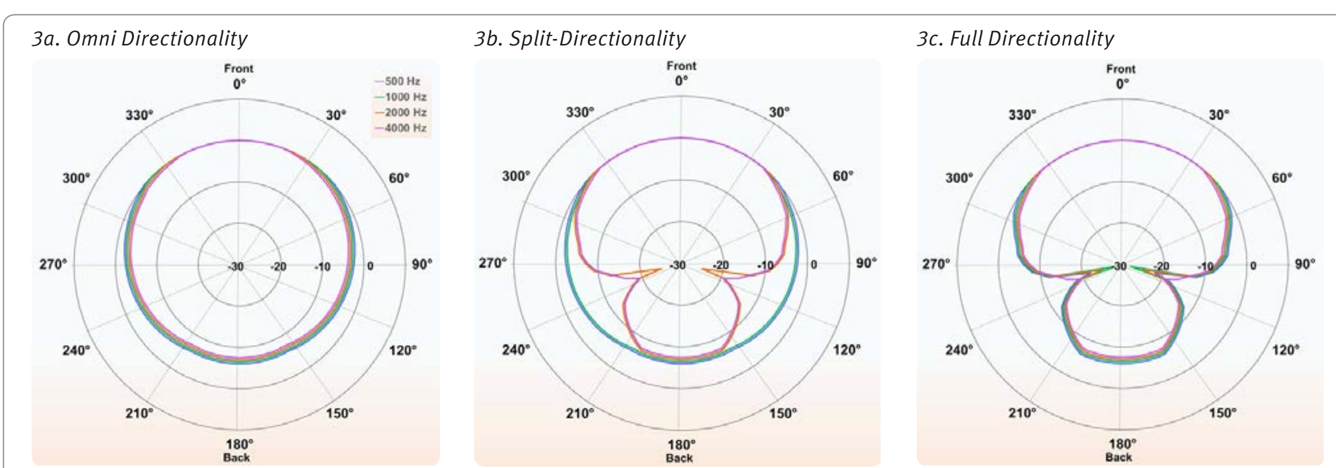

Figure 3. The different directional patterns of the three different Free Focus modes.

mode will be particularly effective for speech perception in noisy environments, or for speech discrimination where there are multiple conversations in the same room.

In Coordinated Adaptive Processing, input sound energy is constantly monitored by the environment detector in the different frequency bands and this drives the directionality mode. As a result, Free Focus is instantaneously adapting as a combination of the three main directionality modes, creating a fully adaptive directional mode.

References on Free Focus
Weile JN, et al. A Broader Look at Performance and Personalization in Hearing Aid Fittings. Hearing Review. 2013, Aug.
Weile JN, Behrens T & Wagener K. An improved option for people with severe to profound hearing loss. Hearing Review. 2011; 18(10):32-45.

The modification of directional sensitivity in the three dimensions -space, time and frequency provides cochlear implant users with the most natural hearing experience possible in every listening environment. The changes between directional modes are relatively slow (time constant = 1.5 s) to ensure stability of the acoustic environment as well as optimal hearing comfort for the user. Free Focus for cochlear implants will adapt to directional mode only in situations when it will be beneficial, so as not to hinder the effect of other synergies.

V. Voice Track

In noisy environments, important speech signals are often masked by unpleasant background noise. Voice Track aims to preserve speech signals by reducing this noise, while still allowing the listener to detect important background information. Voice Track is a multiband noise-reduction system, based on the modified Wiener-filter technology (Arslan, 2006). Compared to other single-channel noise reduction algorithms, Wiener-filter have often been reported as superior in preserving speech intelligibility (Li et al., 2011; Mahdu et al., 2013) because they preserve the temporal envelope of speech better than other noise reduction algorithms and produce less distortion.

The Voice Track strategy is based on a dynamic, time-varying signal-to-noise ratio (SNR) estimation performed in the 64 independent spectral channels. In each channel, the attack-time, defined as the time necessary for a new signal to be recognized as noise, is relatively long: 2 s, to avoid fast fluctuations in speech from becoming distorted. The release-time, on the contrary, defined as the time a signal can eventually be identified as a signal is very fast: 18 ms, to minimize signal-loss at speech onset. When noise is detected, it is suppressed by frequency subtraction in each band. Reduction factors can be customized to a preset marker allowing reductions in signal energy to 20% (Weak), 50% (Medium) and 70% (Strong) (Figure 4). Unwanted frequencies are therefore mostly attenuated and not suppressed altogether, leading to more natural, audible and comfortable sound.

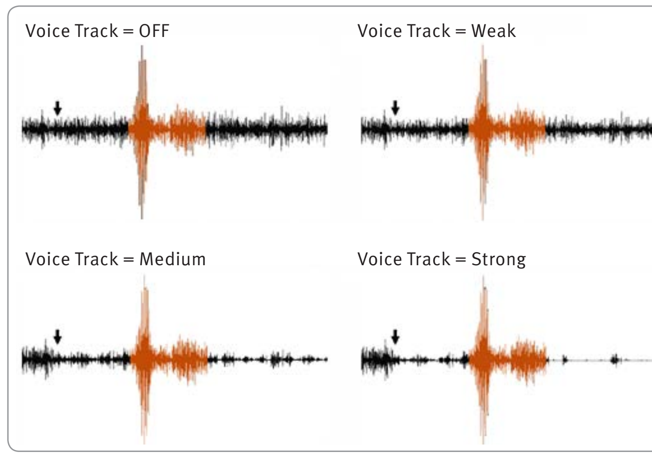

Figure 4. Effect of Voice Track on noisy speech.
Effect of Voice Track set to the Weak (0.25 attenuation), Medium (0.5 attenuation) or Strong setting (0.75 attenuation) on the single French word 'bateau' ('Boat'), embedded in static speech shaped noise at 0dB signal-to-noise ratio. The black arrows show the end of the attack-time, when Voice Track starts suppressing noise.

Clinical evaluation of the Voice Track feature

Voice Track was evaluated in 13 post-lingual deaf adult CI users, aged 50.5 +/- 17.2, with an average deafness duration of 13.2 +/- 14.7 years (Guevara et al., 2016) (Figure 5). The activation of the noise reduction system allowed average improvements of +10.8 % in static noise at +5dB signal to noise ratio (SNR) and +11.6 % in cocktail party noise at 0 dB SNR. Smaller yet positive differences were also observed in cocktail party noise at +5 dB SNR: +6.9 % and at 10 dB SNR: +5.4 %. After an adjustment period of 30 days significant benefit in speech-shaped static noise at +5 dB SNR reached statistical significance at +13.4 %; and +8.5% in cocktail party noise. Subjective evaluation of sound quality revealed an overall sound quality improvement (significant main effect of noise reduction), with pronounced effects in two conditions: the intelligibility of speech in noise and the quality of sound on the telephone.

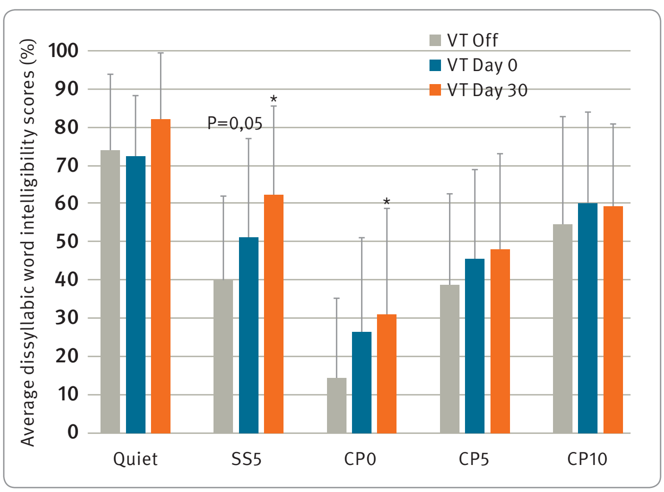

Figure 5. Clinical Evaluation of Voice Track.
Average speech identification scores measured without noise reduction (VT Off – grey), at activation day (VT Day 0 – blue) and after a 30 day habituation period (VT Day 30 – orange). Dissyllabic word recognition was measured in quiet (Quiet) and in 4 different background noise conditions and SNR: a speech-shaped broadband static noise presented at a +5dB SNR (SS5), and cocktail party noise presented at 0 (CP0), +5 (CP5) and +10 dB SNR (CP10). Error-bars represent standard deviation of mean and stars indicate a statistically significant difference (post-hoc LSD test, p ‹.05).

References on Wiener-filters and Voice Track
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Guevara N, Bozorg-Grayeli A, Bébéar JP, et al. The VoiceTrack® multiband single-channel noise reduction system for cochlear implants: individual preferences, best fitting parameters and functional outcomes. (2016). International Journal of Audiology.

VI. Voice Guard

In everyday life, speech can exhibit an extremely variable distribution of loudness, depending on who is talking and on the environment in which he is speaking. At low-levels, for example, when speaking to someone near to us in a very quiet room, the dynamic of speech and average loudness are reduced (around 10-15 dB speech and around 40-45 dB SPL intensity - Figure 6). At the other extreme, if someone shouts in a very noisy room, the difference between the softest and the loudest sounds in speech increases with its average loudness, the dynamic increases up to 40 dB dynamic and around 65-70 dB SPL intensity and important speech information changes its position in the dynamic (see green areas on Figure 6). Compression systems designed for cochlear implants must account for this variability while always prioritizing the optimal representation of speech information.

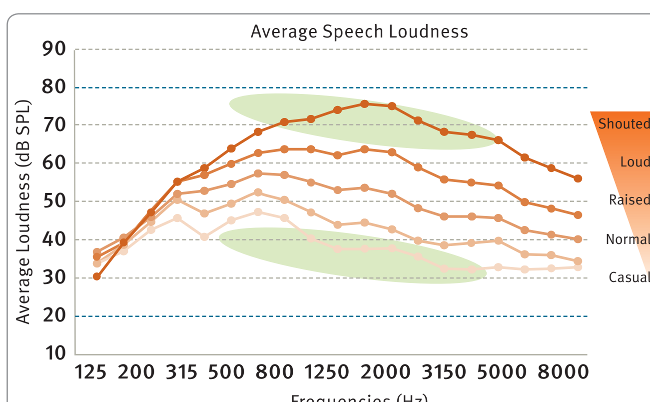

Figure 6. Average loudness / spectral content of speech in different communication modes.
Data averaged for children, women and male voices. Original data from: Pearsons, K. S., Bennett, R. L., & Fidell, S. (1977). Speech levels in various noise environments (Report No. EPA-600/1-77-025). Washington, C: U.S. Environmental Protection Agency. In: Olsen WO. Am J Audiol. 1998, 7(2):21-25.

Voice Guard is the automatic multiband output compression system building on the XDP experience (Bozorg-Grayeli et al., 2015) in order to maximize the transfer of speech information in every listening situation.

Voice Guard Transfer Function

The Voice Guard transfer function is bilinear, with a knee-point defined in the dB SPL space, defining two compression ratios, the first (in blue on Figure 7) is applied to softer sounds and a second (grey shading on Figure 7), where compression is usually increased, for louder sounds.

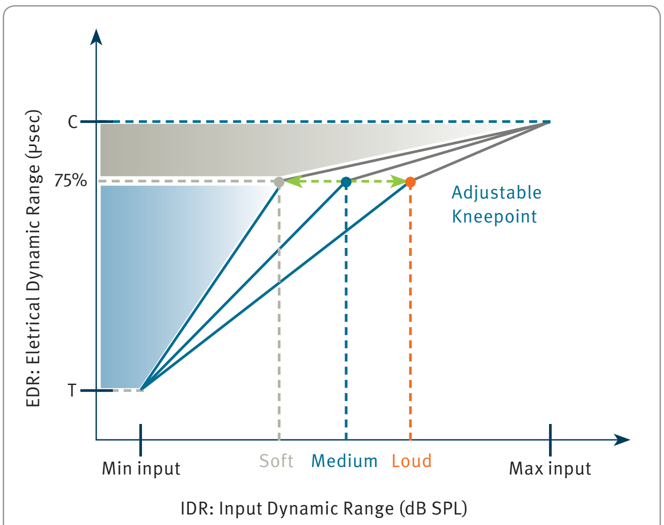

Figure 7. The Voice Guard Transfer function.
The Voice Guard output compression system builds on the XDP transfer function that can account for different sound environments thanks to the three knee-point presets for Soft, Medium and Loud sounds which must however be put in independent programs and selected manually.

Voice Guard multiband pre-sets

Preset values were defined for knee-points to maximize speech intelligibility in different sound environments. What sets Voice Guard technology apart from all other traditional AGC systems is the fact that the compression transfer function can be independently modified within four different frequency bands: 195-846 Hz; 846-1497 Hz; 1497-3451 Hz; 3451-8000 Hz (see Table 1). The boundaries of these bands were determined by statistical analysis looking for spectral energy distribution similarities across spectral bands. This analysis grouped together electrodes showing a comparable distribution of energy when exposed to a large speech database. Similarly, preset knee-point values for three different listening environments (Quiet, Medium and Loud) were statistically determined so that 95% of the speech information falls in the area under the knee-point in each setting considered.

Environment	Quiet	Medium	Loud	Frequency Range
RMS Input Level (dB SPL)	60	70	80	-
Knee-point value (dB SPL)	52	61	70	195-846 Hz
	52	61	70	846-1497 Hz
	47	57	66	1497-3451 Hz
	41	50	58	3451-8000 Hz

Table 1: the Voice Guard transfer function knee-point presets

Voice Guard automatic adaptive behaviour

Inherited from Oticon's highly appraised Speech Guard, Voice Guard preserves speech information for cochlear implant users while maintaining comfort at all times during Coordinated Adaptive Processing. With Voice Guard, the knee-point of the transfer function automatically adapts to the listening environment (Figure 8), with a time-constant depending on the information provided by the Inium environment detector and the directionality configuration provided by Free Focus. This is the heart of Coordinated Adaptive Processing, each sound processing module inside the pipeline senses activity in the other parts of the system in a coordinated and synchronized fashion, to guarantee the richest sound experience in every environment. The knee-point values of Voice Guard are continuously synchronized with the directionality level estimations. Previous coding strategies for cochlear implants also include output compression functions, however the mapping of the acoustic range onto the electrical range does not usually allow direct mapping between dB SPL and the electrical stimulation level. Voice Guard offers the opportunity to directly map output levels in dB SPL, providing audiologists with a much more intuitive fitting procedure.

References on Voice Guard

Bozorg-Grayeli A, Guevara N, Bébéar JP, et al. Clinical evaluation of the XDP output compression strategy for cochlear implants. Eur Arch oto-rhino-laryngol. 2015.

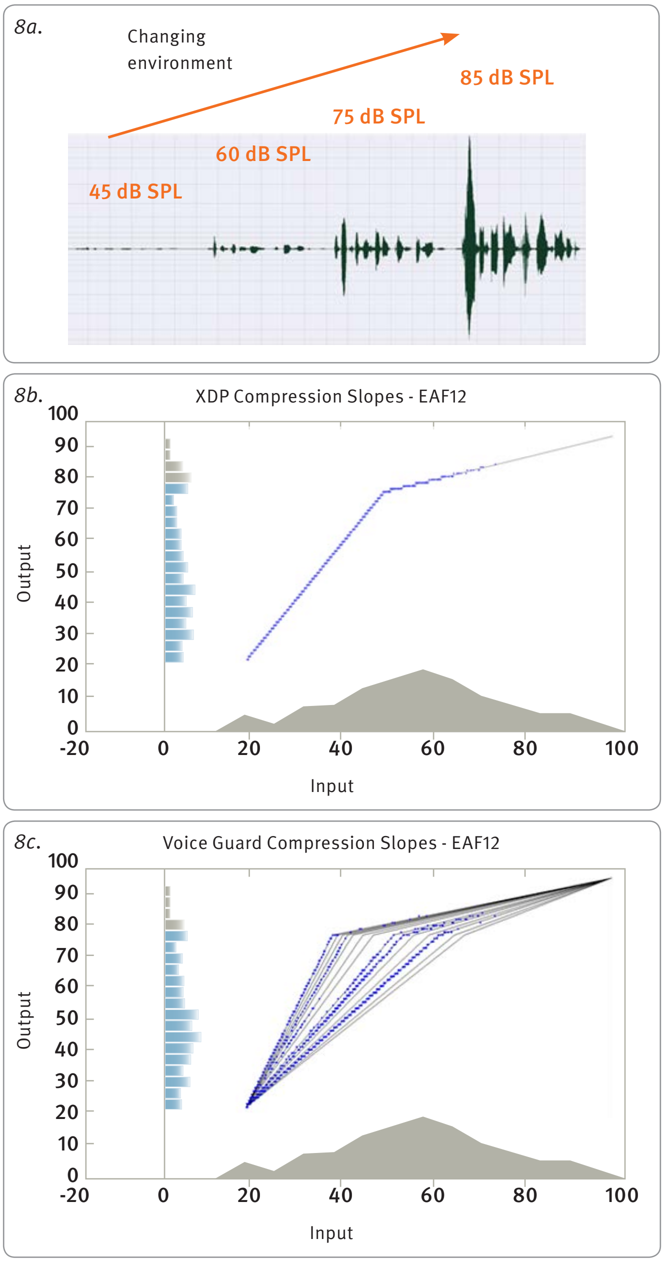

Figure 8. Comparing Voice Guard (automatic) to XDP (static) automatic multiband output compression system.
Fig 8a: An acoustic transition from a soft to a loud environment, sound is becoming louder, from 45 to 85 dB SPL.
Fig 8b: XDP is a static output compression system, placed on its default medium setting, XDP will remain in this configuration all the time, except if presets were stored in different programs, then the user can change compression by selecting another program. Medium sounds sound natural but soft and loud sounds are distorted.
Fig 8c: Voice Guard will automatically change its knee-point throughout the transition and will seamlessly accompany the environmental change by adjusting compression. Soft, medium and loud sounds are perfectly restored.

Voice Guard expected Benefits

Voice Guard will automatically adapt the output compression ratios depending on the input intensity to maximize audibility and sound clarity in every acoustic environment. The expected performance of Voice Guard, compared to the data obtained from the clinical evaluation of XDP (Bozorg-Grayeli et al., 2015) are shown in (Figure 9). Contrary to the static version of the compression system XDP, which required the user to change settings manually according to the environment (e.g. by using a dedicated program), Voice Guard will automatically adapt compression settings to the intensity of the sound present in the environment. The compression system will therefore always be in an optimal setting to guarantee maximal audibility in every listening situation, thereby avoiding reduced intelligibility in difficult situations (soft or loud speech).

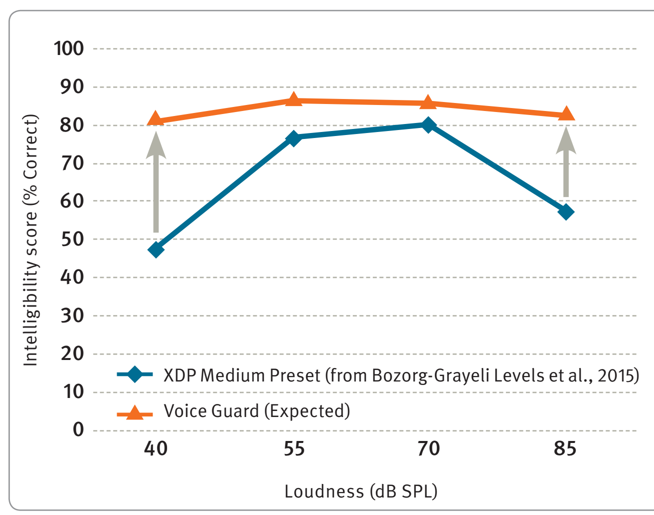

Figure 9. Voice Guard expected performance.
Blue: Speech intelligibility in Quiet measured at 40, 55, 70 and 85 dB SPL intensity, with XDP in its Medium Preset. Scores are improved for intensities corresponding to the chosen preset. In its static version, XDP processes soft (40 dB SPL) and loud (85 dB SPL) sounds with the same, non-optimal compression preset and intelligibility decreases. Original data adapted from Bozorg-Grayeli et al., 2015.
Orange: Expected performance (extrapolation) with Voice Guard, the automatic adaptive output compression system. Voice Guard will automatically adapt compression to the input level and compensate for intelligibility loss at soft (40 dB SPL) and loud (85 dB SPL) intensities. Audibility is maximized in every sound environment, without intervention by the user.

Conclusion

Coordinated Adaptive Processing in Neuro sound processors gives cochlear implant users access to exclusive state-of-the-art sound processing. It is defined by the fusion and coordination of advanced sound processing algorithms Oticon hearing instrument technology instruments (environment detection, automatic directionality, wind noise reduction), with unique dedicated cochlear implant sound processing strategies (wide input dynamic range, automatic multiband output compression system and noise reduction system). This will ensure that Neuro users will benefit from optimal sound quality and maximal speech intelligibility in every listening situation. Optimized sound will then be coded into electrical stimulation, delivered to the auditory nerve via the electrode array.

Notes

Notes

Because sound matters

Oticon Medical is a global company in implantable hearing solutions, dedicated to bringing the magical world of sound to people at every stage of life. As a member of one of the world's largest groups of hearing health care companies, we share a close link with Oticon and direct access to the latest advances in hearing research and technologies. Our competencies span more than a century of innovations in sound processing and decades of pioneering experience in hearing implant technology.

By working collaboratively with patients, physicians and hearing care professionals, we ensure that every solution we create is designed with users' needs in mind. We share an unwavering commitment to provide innovative solutions and support that enhance quality of life for people wherever life may take them. Because we know how much sound matters.

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