WHITEPAPER

Coordinated Adaptive Processing in the Neuro Cochlear Implant System

ABSTRACT

Oticon Medical's sound processing features a unique (1) 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" (1) (23). This novel approach changes conventional sound processing systems by providing the right 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 able to deliver greater performance, by ensuring that the user of Oticon Medical cochlear implant systems 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.

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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 600,000 patients worldwide (2). 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 (Figure 1) in Neuro sound processors.

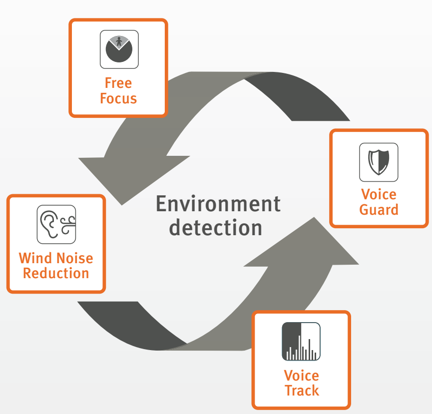

Figure 1: Coordinated Adaptive Processing

I. Coordinated Adaptive Processing

Coordinated Adaptive Processing is a unique (1) 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 2).

Coordinated Adaptive Processing is based on advanced Oticon Inium processing technology. 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 the environment by analysing the acoustic energy in different frequency bands. In Neuro system sound processing, these data are used to seamlessly adapt signal processing algorithms to maintain high audibility, maximise hearing comfort and optimise speech intelligibility levels. In addition, the environment detection system is constantly exchanging information with the directionality module, Free Focus, to ensure that the best combination of directionality parameters in any one listening environment is always selected for Oticon Medical users.

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 and create a unique (1) combination of synergies in the sound processing chain.

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.
- Maximise speech and sound quality in every listening situation, by integrating a unique (1) combination of hearing instrument algorithms and cochlear implant dedicated sound treatments, driven by a continuous monitoring of the acoustic environment.

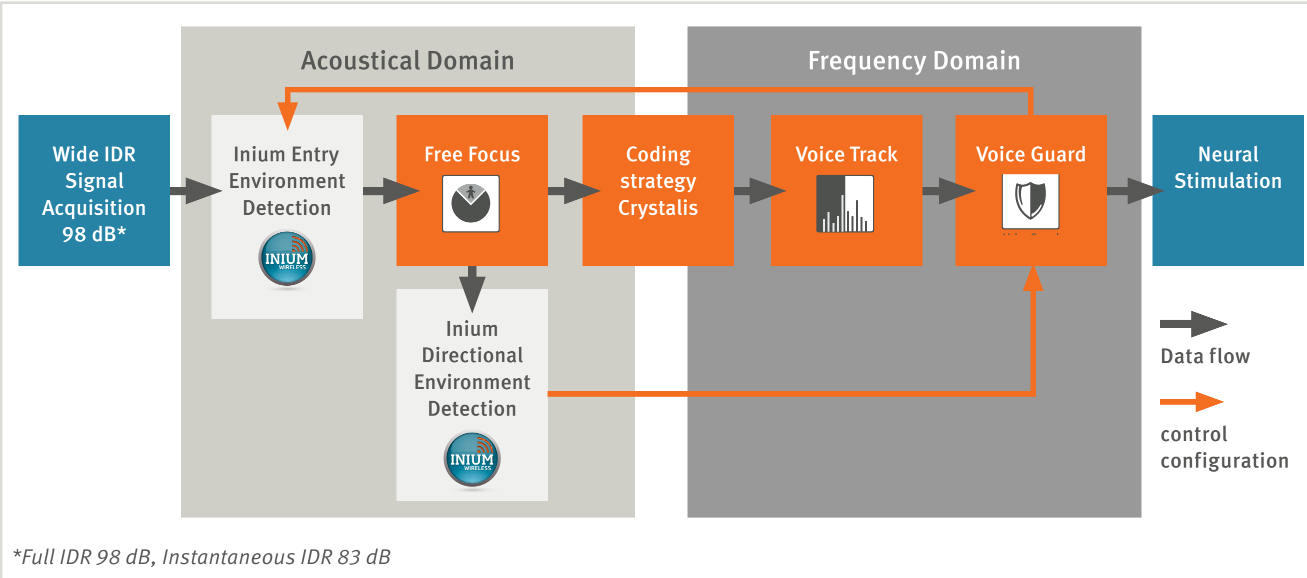

Figure 2. General processing scheme of Coordinated Adaptive Processing.
In Coordinated Adaptive Processing, sound processing starts with a wide input dynamic range (IDR) dual microphone signal acquisition, in order to benefit from the widest sound possible entering the processing chain. The Inium environment 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 using the Crystalis sound coding strategy 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 to 30 steps, where each step is 1 microsecond) (Zeng & Galvin, (3); Vargas et al., (4) and Figure 3a).

Input sounds must therefore be greatly compressed, potentially leading to signal distortion that can limit speech intelligibility (Hornsby & Ricketts, (5); Neuman et al., (6)).

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, in order to keep it around the presentation level. (Loizou, (7)).

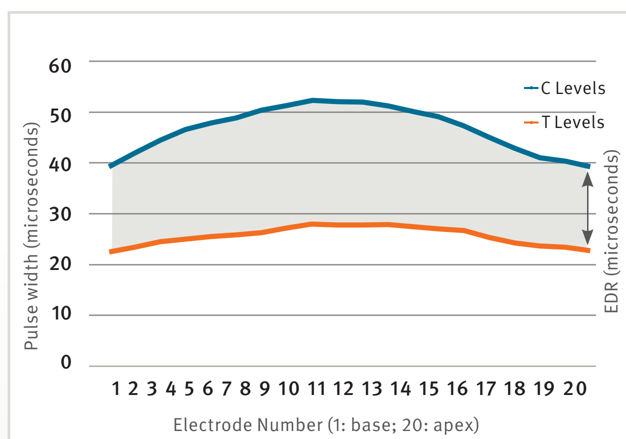

Figure 3a. Typical Electrical Dynamic Ranges in Oticon Medical CI users.

However, reducing the IDR is a less than an optimal compromise as it causes information loss and introduces signal distortion 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., (8) and Studebaker et al., (9)), due to the large variations in the intensity of speech sounds (Figure 3b).

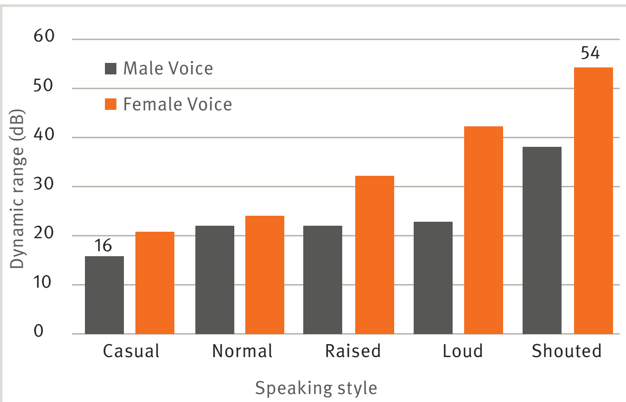

Figure 3b. Average Acoustic Dynamic Range of Speech.
The natural dynamic range of speech sounds (the difference between the softest and the loudest sounds in an utterance), can evolve from 16 up to 54 dB depending on the talker and the loudness. Adapted from: Olsen, Am J Audiol. 1998, 7(2):21-25 (10)

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., (11)). On the contrary, the use of a wider IDR can be beneficial for cochlear implant users (Zeng et al., (12)), especially if the compression system is specifically designed to account for a wide IDR (Spahr et al., (8)). Similar results were reported in hearing instrument users (Oeding & Valente, (13)), 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 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 (full IDR is 97.6 dB on average, while the Instantaneous IDR is 83 dB) combined with Voice Guard's automatic multiband output compression system. The aim is to maximise 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 and digitised at 20 kHz (18 bits) over a frequency range up to 8000 Hz.

References
(1) Segovia Martinez, M., (2017). Cochlear implant and an operating method thereof. (Patent n°. US9844671B2, Oticon Medical A/S)
(2) European Association of Cochlear Implant Users, http://eurociu.eu/what-is-the-ci-
(3) Zeng FG & Galvin JJ. Amplitude compression and phoneme recognition in cochlear implant listeners. Ear Hear. 1999 Feb; 20(1):60-74
(4) Vargas JL et al. Long-term evolution of the electrical stimulation levels for cochlear implant patients. Clin. Exp. Otorhinolaryngol. 2012 Dec;5(4):194-200.
(5) Hornsby BW & Ricketts TA. The effects of compression ratio, signal- to-noise ratio, and level on speech recognition in normal-hearing listeners. J. Acoust. Soc. Am. 2001 Jun; 109(6):2964-73
(6) Neuman et al. The effect of compression ratio and release time on the categorical rating of sound quality. J. Acoust. Soc. Am. 1998 May; 103(5 Pt 1):2273-81
(7) Loizou PC. Mimicking the Human Ear. IEEE Signal Proc. Mag. 1998 Sep; 15(5):101-30.
(8) Spahr AJ, Dorman MF & Loiselle LH. Performance of patients using different cochlear implant systems: effects of input dynamic range. Ear Hear. 2007 Apr; 28(2):260-75
(9) Studebakes GA et al., Monosyllabic word recognition at higher-than-normal speech and noise levels. J. Acoust. Soc. Am. 1999 Apr; 105 (4), 2431-2444
(10) Olsen WO. Average speech levels and spectra in various speaking/listening conditions: a summary of the Pearson, Bennett, & Fidell (1977) Report. Am. J. Audiol. 1998 Oct; 7(2):21-25
(11) 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
(12) Zeng FG et al. Speech dynamic range and its effect on cochlear implant performance. J. Acoust. Soc. Am. 2002 Jan; 111(1 Pt 1):377-86
(13) Oeding K & Valente M. The effect of a high upper Input limiting level on word recognition in noise, sound quality preferences, and subjective ratings of real-world performance. J. Am. Acad. Audiol. 2015, 26(6):547-62

III. 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 directionality 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., (14); Weile et al., (15)).

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.

Neuro 2 sound processor offers two different omnidirectional solutions (16), amongst which audiologists and users can choose:

- Opti-omni mode: This mode is a more traditional implementation of the omnidirectional mode. All spatial directions are equally picked up, except for the front-back axis. Sounds coming from the front are slightly amplified (+3-5 dB) and sounds coming from the back are slightly attenuated (-3-5 dB) to reduce front-back confusions (Figure 4).

- Speech-omni mode: This newly introduced mode is designed to improve sound quality and provide some emphasis on speech cues above 1880 Hz. This mode exacerbates the natural sound-shaping effect of the outer ear and provides some highlight on high-frequency speech cues, while preserving a very natural and omnidirectional sensation for sounds in the low frequencies (Figure 4).

- Speech-omni and Opti-omni modes were assessed in two groups of adult Neuro CI system users. The overall aim of the study was to measure and compare speech identification scores in quiet and noise with the two modes and the results show significant benefits of beamforming in the high frequencies, as implemented in the Speech-omni mode (Bastos Cordeiro et al. (17)).

Neuro 2 also offers two different directional microphone settings optimised for listening situations with higher levels of noise.

- Split-directional mode: This mode is an extended omnidirectional configuration taking profit of partial directionality. The response is omnidirectional in the low frequencies, to effectively replicate the acoustic environment, while in the high frequencies, starting at 2 kHz, the microphones exhibit increased sensitivity to 0 degree azimuth. Besides the cut-off frequency that differs somewhat, this mode will be automatically selected for noise levels around 65 dB SPL in the automatic mode and is efficient in low to medium level noise situations. (Figure 4)

- Full-directional mode: This mode picks up sounds coming from the front and markedly reduces background noise coming from all other directions. This mode is particularly effective for speech perception in noisy environments, or where there are multiple conversations in the same room. This mode is especially useful in high-level noise situations and is automatically selected for noise levels around 75 dB SPL in the automatic mode. (Figure 4)

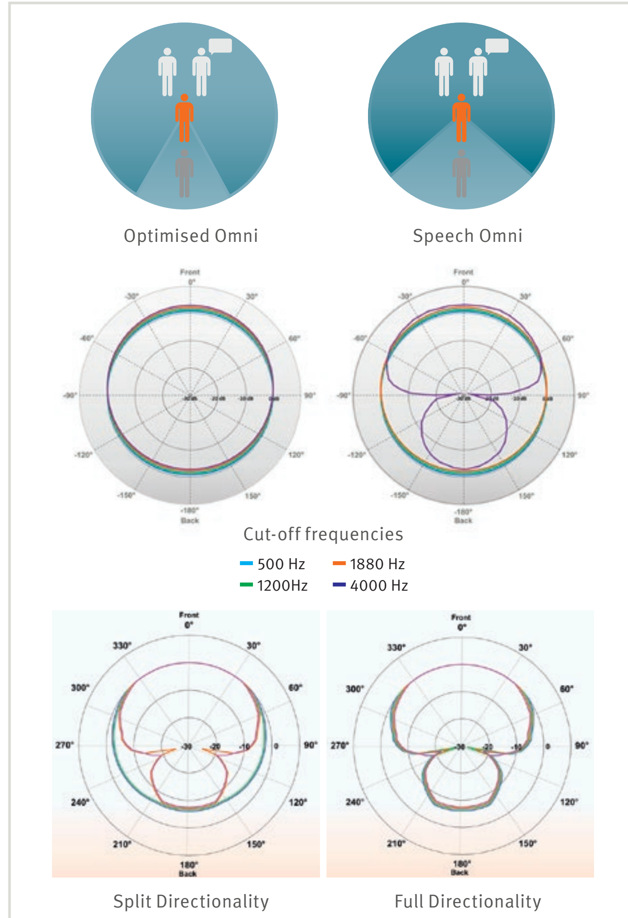

Figure 4. *The omnidirectional microphone choice in Free Focus for Neuro 2, Opti-Omni and Speech-Omni modes (16)*

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.

The modification of directional sensitivity in three dimensions: space, time and frequency, provides Oticon Medical 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.

References
(14) Weile JN, Behrens T & Wagener K. An improved option for people with severe to profound hearing loss. Hearing Review. 2011; 18(10):32-45.
(15) Weile JN et al. A broader look at performance and personalization in hearing aid fittings. Hearing Review. 2013, Aug.
(16) Neuro users say it: the everyday sounds better with speech-omni. Antonio Caruso, M.D. et al., 2020, Oticon Medical white papers series, retrieved from: https://www.oticonmedical.com/for-professionals/cochlear-implant/clinical-results.
(17) Bastos Cordeiro B. et al. Adult users of the Oticon Medical Neuro cochlear implant system benefit from beamforming in the high frequencies. Audiol.Res. 2021, 11, 179-191

IV. Voice Track

In noisy environments, important speech signals are often masked by disturbing 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 multi-band noise reduction system, based on the modified Wiener filter technology. Compared to other single-channel noise reduction algorithms, Wiener-filter have often been reported as superior in preserving speech intelligibility (Li et al., (18); Mahdu et al., (19)) 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 recognised as noise, is relatively long (2s), 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 (18ms), to minimise signal loss at speech onset. When noise is detected, it is suppressed by frequency subtraction in each band. Reduction factors can be customised to a preset marker allowing reductions in signal energy to 20% (Weak), 50% (Medium) and 70% (Strong) (Figure 5). Unwanted frequencies are therefore mostly attenuated and not suppressed altogether, leading to more natural, audible, and comfortable sound.

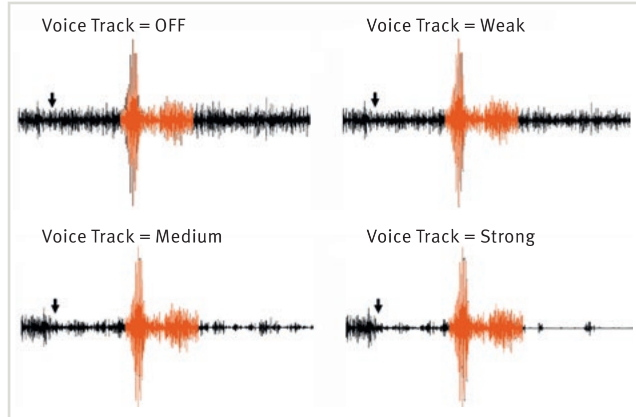

Figure 5. Effect of Voice Track on noisy speech.
Effect of Voice Track set to the Weak (0.25 attenuation), Medium (0.5 attenuation) or Strong setting (0.75 attenuation) on the single French word 'bateau' ('Boat'), embedded in static speech shaped noise at 0dB signal-to-noise ratio. The black arrows show the end of the attack time, when Voice Track starts suppressing noise.

Clinical evaluation of the Voice Track feature

Voice Track was evaluated in thirteen 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. (20)) (Figure 6). The activation of the noise reduction system allowed average improvements of +10.8 % in static noise at +5 dB 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.

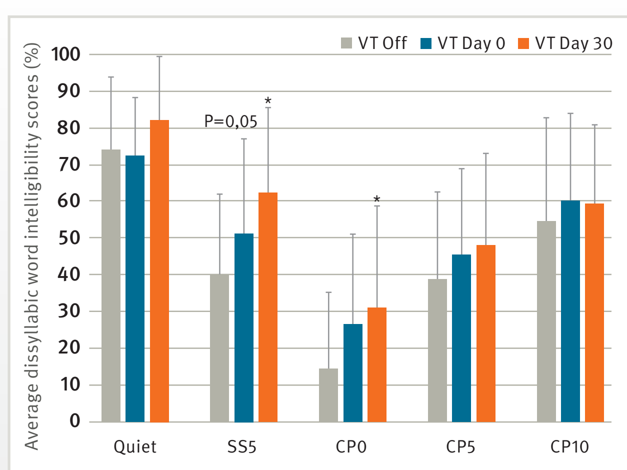

Figure 6. Clinical Evaluation of Voice Track (20).
Average speech identification scores measured without noise reduction (VT Off – grey), at activation day (VT Day 0 – blue) and after a 30-day habituation period (VT Day 30 – orange). Dissyllabic word recognition was measured in quiet (Quiet) and in 4 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
(18) Li J et al. Comparative intelligibility investigation of single-channel noise-reduction algorithms for Chinese, Japanese and English. J. Acoust. Soc. Am. 2011 May; 129(5):3291-301
(19) Madhu N et al. The potential for speech intelligibility improvement using the ideal binary mask and the ideal Wiener Filter in single channel noise reduction systems: application to auditory prostheses. audio, speech, and language processing, IEEE Transactions on. 2013; 21, 63-72
(20) Guevara N, Bozorg-Grayeli A, Bébéar JP, et al. The VoiceTrack® multiband single-channel modified Wiener-filter noise reduction system for cochlear implants: patients' outcomes and subjective appraisal. International Journal of Audiology 2016, Aug; 55(8):431-8

V. 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 they are 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 dynamics and around 40-45 dB SPL intensity – Figure 7). 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 7). Compression systems designed for cochlear implants must account for this variability, while always prioritising the optimal representation of speech information.

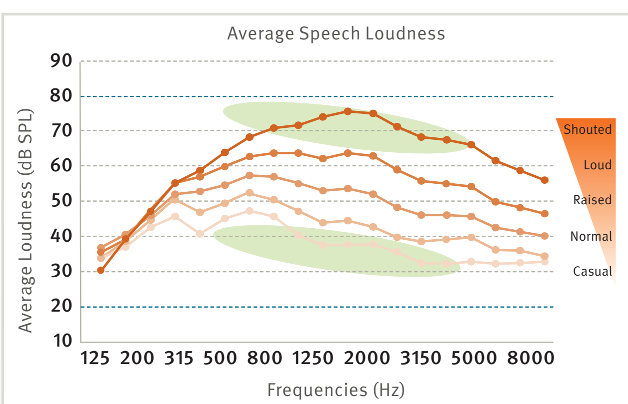

Figure 7. Average loudness – spectral content of speech in different communication modes.
Data averaged for children, women and male voices. Original data from: Pearsons, K. S., Bennett, R. L., & Fidell, S. (21). Speech levels in various noise environments (Report No. EPA-600/1-77-025). Washington DC:U.S. Environmental Protection Agency. In: Olsen WO. (10)

Voice Guard is the automatic multiband output compression system built on Crystalis signal processing strategy, associating a large IDR with the back-end multi-channel xDP compression strategy (Bozorg-Grayeli et al., (22); Langner et al., (23)) in order to maximise 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 HL space, defining two compression ratios, the first (in blue on Figure 8) is applied to softer sounds and a second (grey shading on Figure 8), where compression is usually increased, for louder sounds.

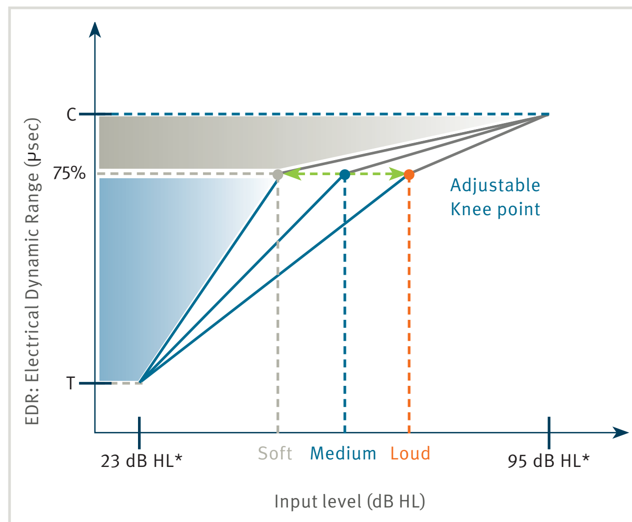

Figure 8. Principle of the back-end dynamic compression system Voice Guard.
** for very loud environments (input RMS › 93 dB SPL) the input signal is attenuated to avoid saturation . The maximum input level available is 115 dB HL without distortion. The Voice Guard output compression system builds on 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 can be selected and assigned to independent programmes.*

Voice Guard multiband presets

Preset values were defined for knee points to maximise 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: 188-812 Hz; 812-1438 Hz; 1438-3438 Hz; 3438-7938 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. The preservation of the Input DR allows us to select the transfer function knee points that follow the long-term speech average for different presentation levels. The knee points were then selected to maximise the percentage of speech information found on the pre knee point area (for different listening environments) which allows to minimise the loss of dynamic range from the acoustic input of the BTE to the electrical input of the implant.

Environment	Quiet	Medium	Loud	Frequency Range
RMS Input Level (dB SPL)	60	70	80	-
Knee point value (dB HL)	52	61	70	188-812 Hz
	52	61	70	812-1438 Hz
	47	57	66	1438-3438 Hz
	41	50	58	3438-7938 Hz

Table 1. The Voice Guard transfer function knee points presets.

Voice Guard automatic adaptive behaviour

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 9), 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 synchronised fashion, to guarantee the richest sound experience in every environment, for every Oticon Medical Neuro system user. The knee point values of Voice Guard are continuously synchronised 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 HL and the electrical stimulation level. Voice Guard offers the opportunity to directly map output levels in dB HL, providing audiologists with a much more intuitive fitting procedure.

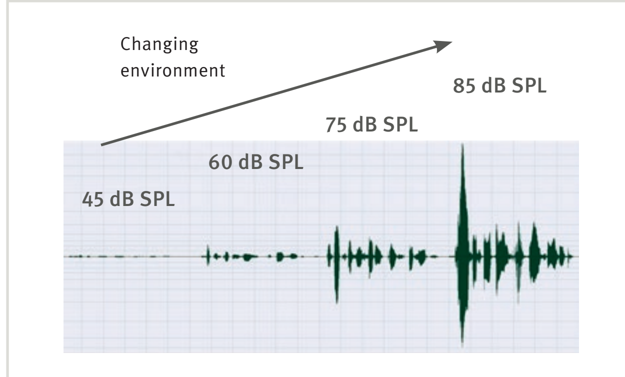

Figure 9 A.

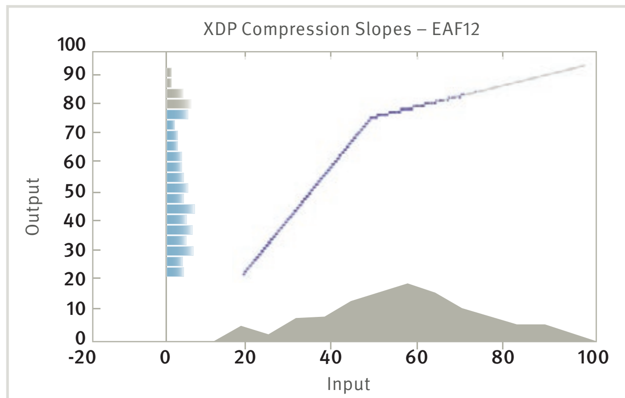

Figure 9 B.

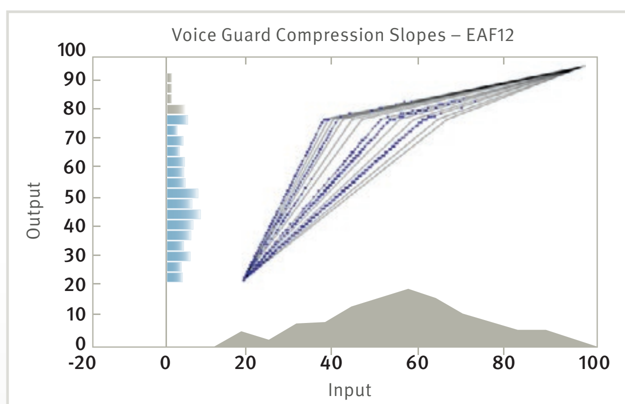

Figure 9 C.

Figure 9 A, 9 B, 9 C. Comparing Voice Guard (automatic) to XDP (static) multiband output compression system.
Fig 9 A: An acoustic transition from a soft to a loud environment, sound is becoming louder, from 45 to 85 dB SPL. Fig 9 B: 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 programme. Medium sounds sound natural, but soft and loud sounds are distorted. Fig 9 C: 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.

Clinical Evaluation of the Voice Guard feature

Voice Guard was evaluated in ten post-lingually deafened adult CI users of the Neuro Zti / Neuro One CI system (Langner et al. (23)). All participants had a minimum of 4 months of experience with their CI and at least 60% of speech intelligibility in quiet, as well as 20% in noise at +10 dB speech-to-noise ratio (SNR) using the Hochmair-Schulz-Moser (HSM) sentence test. The study proposes to compare performance with a single-channel, dual-loop, front-end AGC and with the adaptive back-end multiband dynamic compensation system, Voice Guard (VG). Speech intelligibility in quiet and at +10 dB speech-to-noise ratio (SNR) was assessed with the HSM sentence test and a logatome discrimination task with different consonants was performed in quiet. For speech intelligibility in quiet, a significant benefit with VG over AGC of 10 percent points was observed (VG (mean [M]=69.6%, standard deviation [SD]=23.7%) and AGC (mean [M]=59.9%, standard deviation [SD]=24.7%), see Figure 10). For the noise condition, no significant difference was found. Logatome perception, although not statistically significant due to the small number of participants, increased by 9 percentage points when using VG compared with AGC.

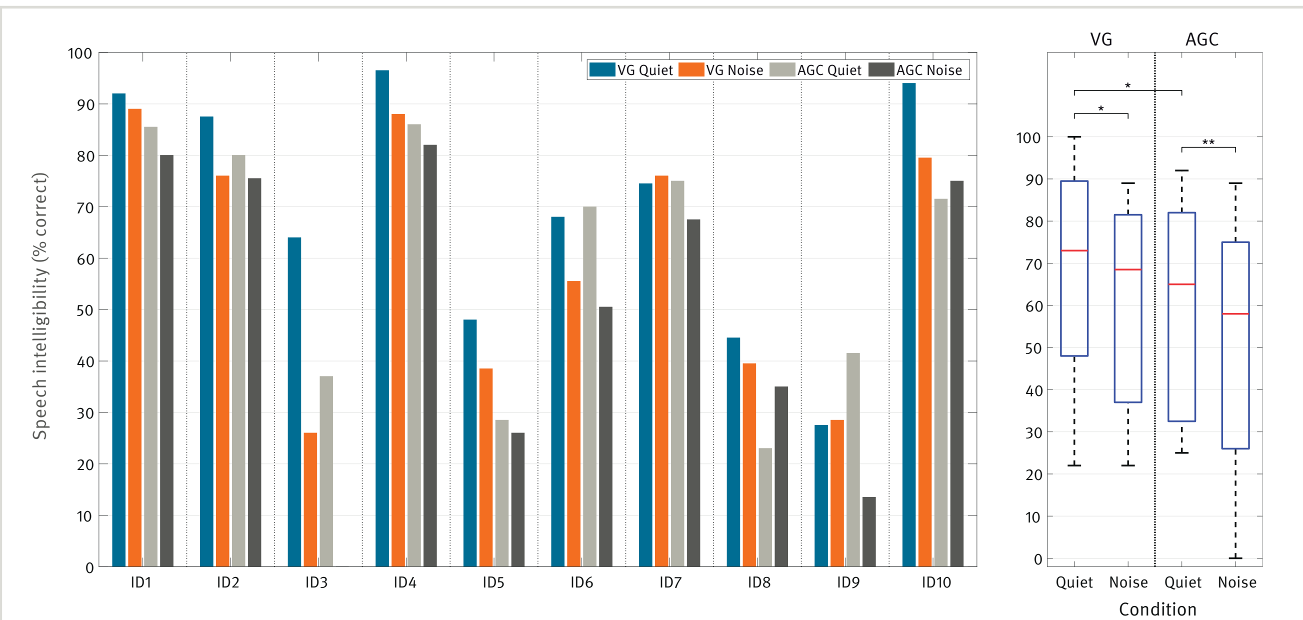

Figure 10: Speech Intelligibility results for all participants, conditions and settings, AGC vs VG. Individual results (left), Summarising box plot representation (right), median in red (Langner et al. (23))

In summary, this study revealed some benefits of an adaptive back-end dynamic compression system compared with a standard front-end AGC system with linear ATE (acoustic-to-electric) mapping. VG performed better for speech-in-quiet, at logatome recognition. The objective measures show that the adaptive knee points of the VG algorithm better preserve the dynamics of speech signals.

More broadly, speech intelligibility of Neuro CI users was assessed in a study describing the clinical efficiency and surgical safety of cochlear implantation with the Neuro CI system, presenting advanced sound processing features such as Voice Track, Voice Guard and Speech Omni (Schramm et al. (24)). Fifty two adult candidates with bilateral sensorineural hearing loss were involved in the clinical study. The ability to repeat HINT sentences was measured on 31 recipients before cochlear implantation and after cochlear implantation at 3, 6, and 12 months (the mean HINT scores were respectively 13%, 58%, 67%, and 72% in quiet and 13%, 46%, 53%, and 59% in noise (+10 dB SNR)). The mean improvement from baseline to 6 months post-activation was 54% in quiet and 40% in noise, see Figure 11.

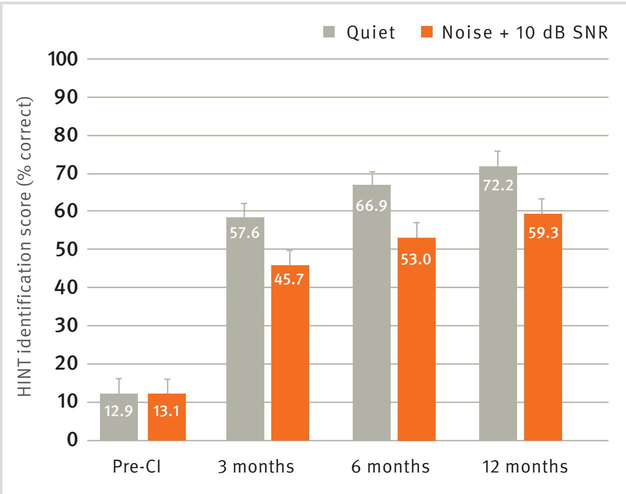

Figure 11: Speech intelligibility results with Neuro CI system (Schramm et al. (24).

As expected, the scores increased dramatically after cochlear implantation, and continued to improve over the first year following implantation. Again, scores are better in quiet than in noise, as expected. Interestingly the difference is not so large, minimising the noise disadvantage.

References

(21) Pearsons KS, Bennett R L, & Fidell S, Speech levels in various noise environments Washington, DC: U.S. Environmental Protection Agency, Report No. EPA-600/1-77-025, 1977

(22) 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. 2016 Sep;273(9):2363-71

(23) Langner F, Büchner A, Nogueira W, Evaluation of an adaptive dynamic compensation system in cochlear implant listeners. Trends in Hearing, 2020 Dec;24:1-13

(24) Schramm D, Chen J, Morris DP et al., Clinical efficiency and safety of the Oticon Medical Neuro cochlear implant system: a multicenter prospective longitudinal study. Expert Review of Medical Devices, 2020 Sep;17(9),959-967

Conclusion

Coordinated Adaptive Processing in Neuro sound processors gives cochlear implant users access to exclusive state-of-the-art sound processing (1) (23). It is defined by the fusion and coordination of advanced sound processing algorithms from Oticon hearing instruments (environment detection, automatic directionality, wind noise reduction), with unique (1) 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. Optimised sound will then be coded into electrical stimulation, delivered to the auditory nerve via the electrode array.

Notes

Because sound matters

Oticon Medical is a global company in implantable hearing solutions, dedicated to bringing the power of sound to people at every stage of life. As part of the Demant group, a global leader in hearing healthcare with more than 16,500 people in over 30 countries and users benefitting from our products and solutions in more than 130 countries, we have access to one of the world's strongest research and development teams, the latest technological advances and insights into hearing care.

Our competencies span more than a century of innovations in sound processing and decades of pioneering experience in hearing implant technology. We work collaboratively with patients, physicians and hearing care professionals to ensure that every solution we create is designed with users' needs in mind. We have a strong passion to provide innovative solutions and support that enhance quality of life and help people live full lives – now and in the future.
Because we know how much sound matters.

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