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Automatic Sound Management 3.0 with the SONNET 2 and RONDO 3 Audio Processor

Abstract

Improving listening comfort and reducing effort for cochlear implant (CI) users is a main focus of cochlear implant development. Especially background noise can be challenging for CI users, as it has been shown to impact speech understanding and increase listening effort. MED-EL's latest Audio Processors, SONNET 2 and RONDO 3, introduce two new noise reduction algorithms, as well as scene-classifier based Adaptive Intelligence (AI). The features are all part of Automatic Sound Management (ASM) 3.0, which refers to various front-end processing technologies that are designed to achieve outstanding performance in any environment. The new noise reduction algorithms Ambient Noise Reduction (ANR) and Transient Noise Reduction (TNR) attenuate distractive noise, for a more effortless listening experience. Adaptive Intelligence (AI) automatically selects the appropriate settings based on the listening environment detected by the scene classifier, providing optimal hearing experience in any situation and minimizing the need for users to change programs or settings. This paper describes MED-EL's new Automatic Sound Management (ASM) 3.0 features in terms of benefits and fitting considerations.

Background noise

Background noise and reverberation can impact speech clarity and understanding especially for hearing impaired individuals. Even though cochlear implant (CI) users are able to achieve high levels of speech perception in quiet and at high signal-to-noise ratios (SNR), listening situations with a low SNR are challenging for CI users (Fu et al., 1998; Kokkinakis et al., 2012; Holube et al., 2006). Further, noise with a similar spectral structure to the target speech signal increases listening difficulty and fatigue.

Therefore, noise reduction algorithms have been integrated in both hearing aid and cochlear implant technology to improve the signal quality and speech understanding in noise (Brons et al., 2014; Desjardins and Doherty, 2014; Bentler et al., 2008). Chung et al. (2004) showed that listening effort and speech recognition scores were significantly improved by pre-processing signals with hearing aid noise reduction algorithms in cochlear implants. In 2006, Chung et al. also demonstrated that CI users display a subjective preference and significant improvement in sound quality assessment for conditions with noise reduction. Since a variety of different noise patterns are encountered in real-life, specific algorithms are developed to target distinct noise classes. MED-EL has already successfully implemented Wind Noise Reduction (WNR) and Microphone directionality (MD) modes, which have been shown to improve speech understanding in background noise (Büchner et al., 2019; Dorman et al., 2018; Hagen et al. 2019; Honeder et al., 2018; Wimmer et al., 2016). In addition to these proven features, SONNET 2 and RONDO 3 offer two new noise reduction algorithms: Ambient Noise Reduction (ANR) and Transient Noise Reduction (TNR).

Ambient Noise Reduction (ANR) and Transient Noise Reduction (TNR)

Ambient Noise Reduction (ANR) and Transient Noise Reduction (TNR) have been integrated in the SONNET 2 and RONDO 3 Audio Processors to support CI users in everyday listening environments, particularly in more challenging situations, for example, in which speech and noise are not spatially separated. The algorithms reduce distractive noise by analyzing individual frequency bands of the audio signal, detecting if noise is present in a sub-band and attenuating the affected sub-band to a certain extent. The suppression is only applied to the sub-band that contains noise; this way, the effect on the target speech signal is minimized. Furthermore, both ANR and TNR constantly analyze the signal-to-noise ratio (SNR) and are transparent, that is, refrain from modifying the signal, in speech-only conditions. The amount of attenuation is therefore constantly adjusted based on the detected signal characteristics. The maximum amount of attenuation depends on the mode for the noise reduction chosen by the clinician through the MAESTRO fitting software (Mild or Strong).

Ambient Noise Reduction (ANR)

Ambient Noise Reduction (ANR) is tailored for situations with stationary noise, for example, fan noise, and situations with speech in noise, where the noise signal is stationary compared to the speech signal. The algorithm is able to detect and to some extent filter the steady-state noises from the noise-contaminated speech signal.

Figure 1 shows how ANR optimizes a speech-in-noise signal. Figure 1(a) depicts the microphone input signal, consisting of speech and car noise. In Figure 1(b)-(d) the signal without ANR (blue) is compared to the resulting signal with ANR (red) for channels of different frequency ranges. Since car noise is a low-frequency noise, most noise is prevalent on the low frequency channel (CH1), only some noise is present on the mid frequency channel (CH6) and there is almost no noise on the high frequency channel (CH12). Accordingly, the signal after ANR (red) on the low frequency channel (CH1) is visibly attenuated. Since there is only little noise in the mid to high frequency channels (CH6 and CH12), the signal stays mainly unchanged. The transparency of the algorithm for speech is shown in Figure 1(b), where the signal with ANR (red) corresponds to the input signal (blue) for the sections with predominantly speech.

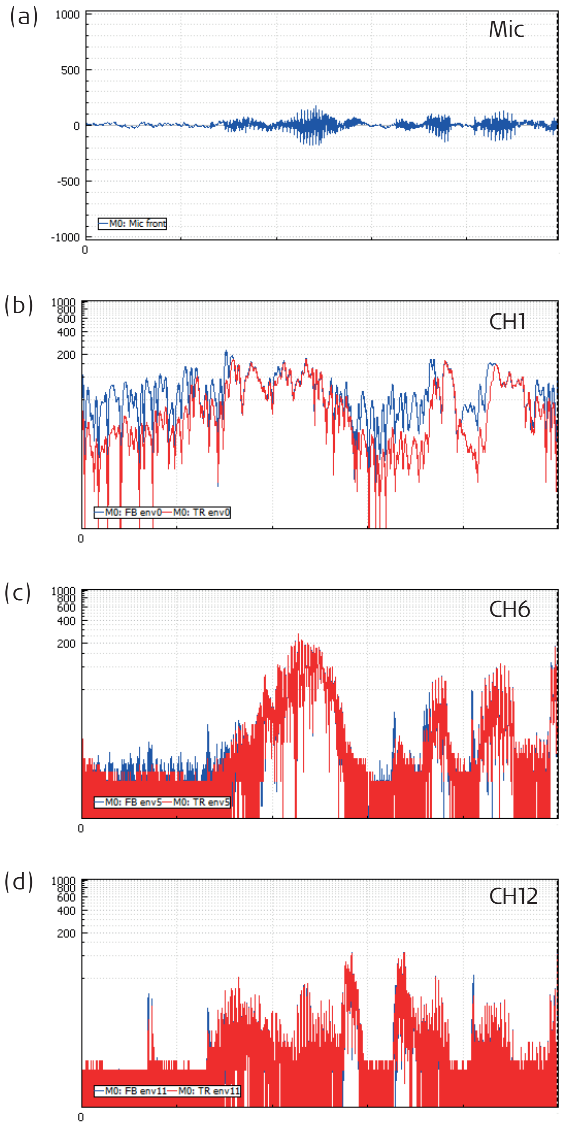

Figure 1: The effect of Ambient Noise Reduction (ANR) on a speech in noise signal. The input signal (blue) is shown in Figure 1(a), the resulting signal modifications (red) for different channels in Figure 1(b) to (d).

Transient Noise Reduction (TNR)

Transient Noise Reduction (TNR) has specifically been developed to reduce transients, which are sudden, loud noise signals, such as rattling dishes or slamming doors, in noise-contaminated speech signals. In contrast to ANR, which targets steady-state noise on all frequencies, TNR is only active on the higher frequency channels, since transient noise contains predominantly high frequency energy. This allows to better preserve the target speech signal in the lower frequencies. TNR complements the Automatic Gain Control (AGC), which also provides some reduction of transient noises, but acts slower in comparison to TNR.

Figure 2 shows how the AGC and TNR complement each other. In Figure 2(a) the effect of the AGC on an input signal containing transient noise (red) is depicted. The input signal after the AGC (blue) demonstrates, that the AGC attenuates loud sounds, but due to its reaction time there is a window which can allow transients to pass. TNR acts after the AGC and is able to detect if the signal still contains residual transients that have passed the AGC due to the mentioned reaction time. Figure 2(b)-(d) show how TNR optimizes the input signal from the AGC (blue) for high frequency channels (CH8, CH10 and CH12). TNR provides an attenuated signal (red) without any of the sharp peaks that are still left after the AGC. After attenuating these residual transients, TNR becomes inactive since the later parts of the initial transient have been dampened by the AGC already.

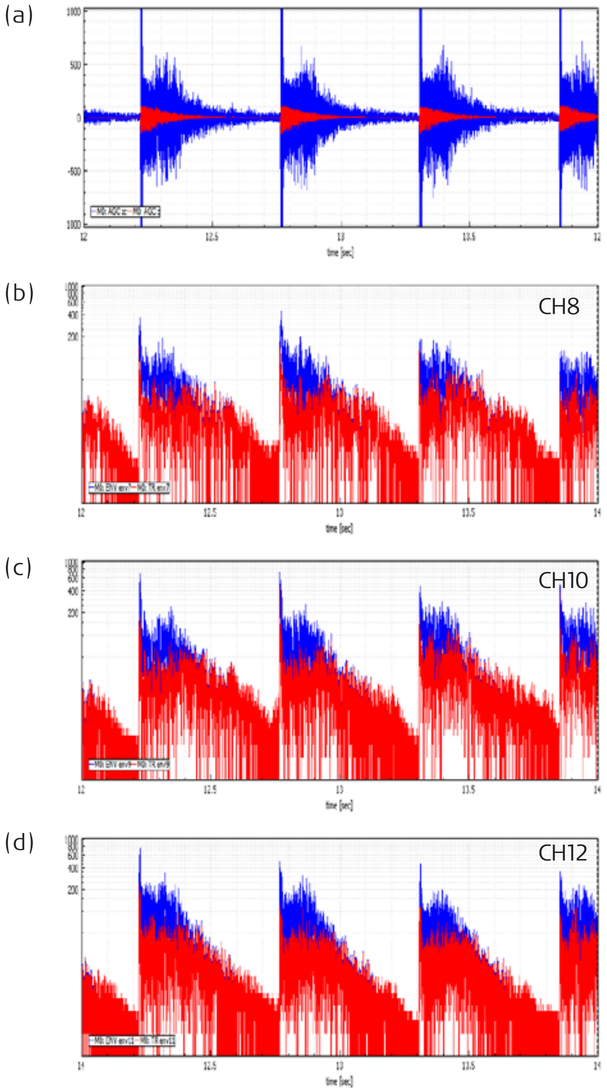

Figure 2: The effect of Transient Noise Reduction (TNR) on a signal containing transients. The input signal (red) and the resulting signal after AGC are shown in Figure 2(a). The signal after AGC (blue) is compared to the signal processed by TNR (red) in Figure 2(b) to (d).

Clinical Considerations for Ambient Noise Reduction (ANR) and Transient Noise Reduction (TNR)

Ambient Noise Reduction (ANR) and Transient Noise Reduction (TNR) can, but do not necessarily have to, be the everyday settings - they can help CI users in more challenging situations and with listening comfort, while preserving the speech signal. Both ANR and TNR can be set per fitting map to either Strong, Mild, or Off. It is recommended to turn off both ANR and TNR for dedicated music maps.

The noise reduction and Microphone directionality technologies in SONNET 2 and RONDO 3 complement each other, and their effects are additive. While Microphone directionality has been shown to improve speech understanding in background noise that originates from the back hemisphere of the user, the effect of Microphone directionality is limited for situations, where noise and the target signal come from the same direction. The direction-independent noise reduction algorithms ANR and TNR can further add to the effect of Microphone directionality, irrespective of the origins of the signals, which can help in reducing listening effort. ANR and TNR are not expected to have an equally strong effect as Microphone directionality, and it is not recommended to replace Microphone directionality with noise reduction. Still, the noise reduction can improve the comfort of the user.

Scene classification

Cochlear implant (CI) users experience a variety of listening environments in their everyday life (Wu et al., 2018). Ideally, the benefit with the CI is equal for all situations, whether the user is listening to music, having a conversation in a restaurant, or while in a car. However, studies have shown that noise, and especially speech in noise, is a challenge for CI users (Kokkinakis et al., 2012). Different sound processing algorithms and settings attempt to ease the listening effort for the user and improve speech understanding in different environments.

Dedicated programs for certain environments provide optimized settings, but also increase the handling complexity for the users. Studies have shown that many hearing aid (HA) users do not change programs, but rather use the default program irrespective of the listening situation (Wu et al., 2018; Kuk, 1996; Cord et al., 2002). Furthermore, users struggle to determine the most beneficial program for a situation (Übelacker et al., 2015), and the task of switching between programs itself has been shown to be one of the most difficult operations when handling a HA, even for experienced users (Desjardins and Doherty, 2009).

Scene classification systems have been developed to evaluate listening environments and automatically switch to the optimal parameter settings for detected situations. Such systems have been implemented in the hearing-aid industry for over a decade (Allegro et al., 2001 and Büchler et al., 2005), and more recently in cochlear implant audio processors. Similar to the human auditory system, the environment is classified by analyzing the sound and identifying characteristic features and attributes (Bregman, 1990). In order to manage the variety of different hearing situations that are encountered in real life, classification systems differentiate between a certain number of classes, for example, speech, noise, quiet or music, defined by specific audio characteristics (Cui and Groth, 2017). The classification algorithm is trained to distinguish between the classes and assign any audio signal to the appropriate class. The classification can then be used for automatic adjustments of sound processing parameters according to the encountered situation – leading to reduced effort for the user and appropriate listening settings for the encountered situation (Hamacher et al., 2005).

For similar automatic classification systems in HAs, a subjective preference for the automatically selected settings over the manual setting has been shown, as well as better speech recognition performance (Rakita and Jones, 2015). Studies on the perceived value of automatic adjustments for HA users showed that the majority (range 75–89%) was satisfied with the functionality (Graaff et al., 2017; Buechler, 2001; Gabriel, 2002; Olson et al., 2004).

Adaptive Intelligence (AI) and Scene Classifier

With SONNET 2 and RONDO 3, MED-EL has implemented a scene classifier that automatically recognizes five different environment classes: Speech, Speech in noise, Noise, Music and Quiet. Based on the detected signal, the Adaptive Intelligence (AI) system then uses the appropriate sound management settings for the environment. The following front-end processing features are automatically set by Adaptive Intelligence: Microphone Directionality (MD), Wind Noise Reduction (WNR), Ambient Noise Reduction (ANR) and Transient Noise Reduction (TNR).

Once an environment change is detected, the Adaptive Intelligence system smoothly switches front-end features, to allow a fluent transition between the settings. Switching settings as a result of a scene change takes approximately 30 - 60 seconds. The frequency of switching classes is balanced between quickly providing the appropriate settings for the detected environment and avoiding changing settings too often. The decision whether the class needs to be changed includes postprocessing steps, where smoothing algorithms take past decisions into account to improve the classification and reduce potential misclassifications.

Clinical Considerations for Adaptive Intelligence (AI)
Adaptive Intelligence (AI) is intended to give recipients the peace of mind of knowing that they can effortlessly benefit from optimized settings in every situation. The feature can be set to either Mild, Strong, or Off, for each individual fitting map, which in turn influences the settings of the controlled parameters.

It is not possible to reasonably test Adaptive Intelligence during the fitting process. It is difficult to simulate different environments and give the user time to get used to the setting during a fitting session. In order to evaluate if Adaptive Intelligence is to the users' taste, it is therefore recommended to create a map with Adaptive Intelligence turned on and let the user experience the map at home. In order to produce stable conditions during fitting, Adaptive Intelligence is inactive during fitting (that is, if the processor is connected to the programming cable). If a map that uses Adaptive Intelligence is activated during fitting, then the front-end features are automatically set according to the “Speech” class of the respective Adaptive Intelligence setting (Mild or Strong), see Table 1 and Table 2.

Table 1: Adaptive Intelligence (AI) settings for the Mild mode.

	MD	WNR	ANR	TNR
Speech	Natural	Mild	Mild	Off
Speech in noise	Natural	Mild	Mild	Mild
Noise	Natural	Mild	Mild	Mild
Music	Natural	Mild	Off	Off
Quiet	Settings not changed			

Table 2: Adaptive Intelligence (AI) settings for the Strong mode.

	MD	WNR	ANR	TNR
Speech	Adaptive	Strong	Strong	Off
Speech in noise	Adaptive	Strong	Strong	Strong
Noise	Adaptive	Strong	Strong	Strong
Music	Omni	Strong	Off	Off
Quiet	Settings not changed			

Recommendations for fitting Adaptive Intelligence (AI) are given in Table 3. It is always advised to take the age, indication and individual circumstances of the user into account when fitting Adaptive Intelligence. Due to the fact that there are minimal changes in the signal processing and no changes to microphone directionality in the Mild setting, it is recommended as a starting point for all ages. MED-EL generally does not recommend using the Strong setting for the initial activation of Adaptive Intelligence (AI), young children or users with cognitive decline. For EAS patients, it is recommended to have an initial program with Adaptive Intelligence switched off. For SSD and bimodal patients, it is recommended to try Adaptive Intelligence. It is advised to individualize the fitting including the Adaptive Intelligence (AI) setting based on the user's feedback in the follow-up sessions.

Table 3: Fitting recommendations for Adaptive Intelligence (AI) at first fittings. P1 refers to Program position 1, P2 to Program position 2. “Use” map refers to the previously most used map without Adaptive Intelligence.

	New activation	Existing Users	
Young children (0-5 yrs)	AI Mild in P1	AI Mild in P1	“Use” map on P2
Older children (5-18 yrs)	AI Mild in P1	AI Mild in P1	“Use” map on P2
Adults (18-65 yrs)	AI Mild in P1	AI Mild in P1	“Use” map on P2
Older adults (65+ yrs)	AI Mild in P1	AI Mild in P1	“Use” map on P2
Unilateral	AI Mild in P1	AI Mild in P1	“Use” map on P2
Bilateral	AI Mild in P1	AI Mild in P1	“Use” map on P2
EAS	AI off in P1	AI off in P1	“Use” map on P2
SSD	AI Mild in P1	AI Mild in P1	“Use” map on P2
Bimodal	AI Mild in P1	AI Mild in P1	“Use” map on P2
Cognitive decline	AI Mild in P1	AI Mild in P1	“Use” map on P2

Maximum Comfortable Levels (MCLs), thresholds (THRs) or other map settings are not modified by Adaptive Intelligence (AI). The information on the encountered sound environments and loudness levels is stored in the SONNET 2 and RONDO 3 Audio Processors, and this information can be accessed by health care professionals in the form of datalogging to gain insight into the users' needs, support counselling and program adjustments.

Conclusion

MED-EL is committed to develop the most innovative and advanced hearing loss solutions. Ambient and Transient Noise Reduction further extend MED-ELs existing solutions for the most natural hearing: Microphone directionality, fine structure coding strategies, and long atraumatic electrodes. With Adaptive Intelligence, users can have the peace of mind of benefiting from optimized settings in every environment.

For more information, please contact your MED-EL representative.

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