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BENEFITS OF DYNAMIC AMPLIFICATION CONTROL IN COMPLEX LISTENING ENVIRONMENTS

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DAC™ is a key component of the new DECS™ technology developed by Bernafon. It does not belong to a classification system or a noise management algorithm. It addresses the problem of noise amplification with a sophisticated and unique solution for any listening situation. Being able to improve the output SNR while still preserving the speech signal with DAC™ has many advantages. First, DAC™ improves listening effort during word recognition tests in noise. Second, hearing aid users report more listening comfort when unnecessary amplification of noise is avoided. Finally, DAC™ estimators can track changes in any listening situation and do not rely on fixed and predefined environment categories.

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DAC™ FOR CONTROLLED AMPLIFICATION OF NOISE

The main function of any hearing aid is to provide amplification. Hearing aids use compression to ensure that the amplification provides enough audibility for the softest sounds and does not exceed the upper limit of the end user's dynamic range. The amount of amplification is traditionally defined by the fitting rationale which provides the formula that determines the amount of gain for different input levels of a speech signal. Various fitting rationales exist with formulas based on different goals and theoretical motivations, however, all of them share the following requirements: making soft speech audible while at the same time keeping loud speech comfortable.

Dynamic range compression provides more gain for the softest part of the speech signal which improves consonant discrimination in quiet (Marriage & Moore, 2003; Davies-Venn et al., 2009). This measurable benefit accomplishes the first requirement of compression, i.e., making soft speech audible. There is, however, much less consensus about the value of compressive amplification when applied to speech-in-noise situations.

When evaluating the effectiveness of compressive amplification in noise there are two main sources of variability: 1) compression parameters, and 2) test signal. Naylor & Johannesson (2009) and Alexander & Masterson (2015) suggest that there is an interaction between the design of the compression unit (level estimation speed, compression ratio, and number of channels) and the nature of the test signal (signal-to-noise ratio (SNR) and modulation characteristics of the masker) that influences the resulting output signal from the hearing aid. One common observation among all these results is that compression alone degrades the hearing aid output SNR for any speech-in-noise situation (Wu & Stangl, 2013). One reason is that the signal level is estimated without differentiating between speech and noise. Both signals receive the same amount of amplification if they are estimated at the same level. Lai et al. (2013) describe this phenomena as the "speech pause" effect that causes an SNR degradation of the hearing aid output. This occurs because noise between speech pauses is amplified as though it were soft speech. However, the result is louder noise which in turn diminishes the SNR. While dynamic compression works well in situations for which it was designed, i.e., speech in quiet, there are negative side effects in noisy situations. Being able to remove this unwanted side effect would lead to faster hearing aid acceptance and provide more listening comfort in noisy situations.

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DAC™ is a new class of signal processing algorithm that addresses the "speech pause" effect generated by dynamic compression. This is achieved by analyzing and providing information to the amplification unit about the signal. The DAC™ analysis unit not only measures the signal level but also labels the signal type (i.e., speech or noise). For a speech signal, DAC™ will not apply any changes and the resulting amplification should be as it was originally programmed and verified with real ear measurements. However, when noise is detected, DAC™ gives instructions to the level estimator and level-to-gain units to apply a specific set of rules that will reduce the amplification. The DAC™ implementation and interaction with noise management and amplification blocks are shown in Figure 1.

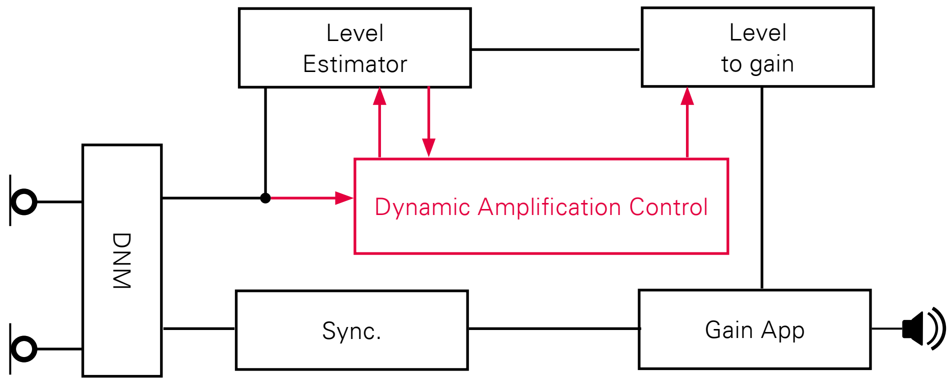

Figure 1. Block diagram of the implementation of the Dynamic Amplification Control™ algorithm between the Dynamic Noise Management™ block and the compression unit.

DAC™ uses two specific built-in estimators to measure the local or phonemic SNR as well as the global or long-term SNR. Having one fast and responsive estimator is mandatory to follow changes within a highly modulated signal like speech. A set of rules is applied to avoid over-amplification of noise when the local SNR is below a defined threshold, indicating an absence of speech. The correction is applied to the level estimator and also to the level-to-gain function. The aim is to control the amplification of noise while still providing the programmed amplification for any incoming speech signal.

HEARING AID OUTPUT SNR INCREASED WITH DAC™

DAC™ was designed to preserve the audibility of the speech signal while at the same time, reducing the amplification of noise. The most efficient way to understand and quantify DAC™'s effect is by measuring the hearing aid output SNR. This measure is possible by separating signal and noise from recordings of hearing aid output using the inversion technique described by Hagermann & Olofsson (2004).

The idea is to generate two test signals containing one speech and one noise component with the same spectral characteristics. The first signal is a mix of the original speech and original noise (SoNo) and the second signal has also the original noise but the speech is inverted (SiNo). One signal recording of each aided SoNo and aided SiNo are needed for each test condition. Speech and noise signals are then separated with a post-processing routine, i.e., estimated noise signal is obtained by adding SoNo and SiNo while estimating the speech signal is obtained by subtracting SoNo and SiNo. The hearing aid output SNR is finally computed as the differences between the estimated signal and noise levels. This method was successfully used to evaluate the effect of wide dynamic range compression (Naylor & Johannesson, 2009; Alexander & Masterson, 2015), noise reduction (Hagerman & Olofsson, 2004), directionality (Wu & Bentler, 2007), and the combination of those features (Wu & Stangl, 2013).

To evaluate the output SNR with DAC™, a Zerena 9 miniRITE hearing aid was placed on a KEMAR and fitted according to the NAL-NL2 fitting rationale for a moderate sloping hearing loss. The prescribed acoustical coupling of an 8 mm Bass Dome, Double Vent and 85-Speaker unit was used. Three listening programs were assigned with the same gain but different combinations of adaptive features: the first program had directionality and DAC™ deactivated in order to show the effect of compression only, the second program with directionality activated, and finally a third program with directionality and DAC™ activated. The ISTS speech and noise signals (Holube et al., 2010) were combined to produce the test signal with electrical SNRs ranging from -10 to +10 dB SNR in 5 dB steps. Speech was presented from the front at 0° and four noise sources were spatially separated at 45, 135, 225, and 315°. The hearing aid's output SNR is presented in Figure 2 for all the listening conditions as a function of the estimated SNR at the ear position.

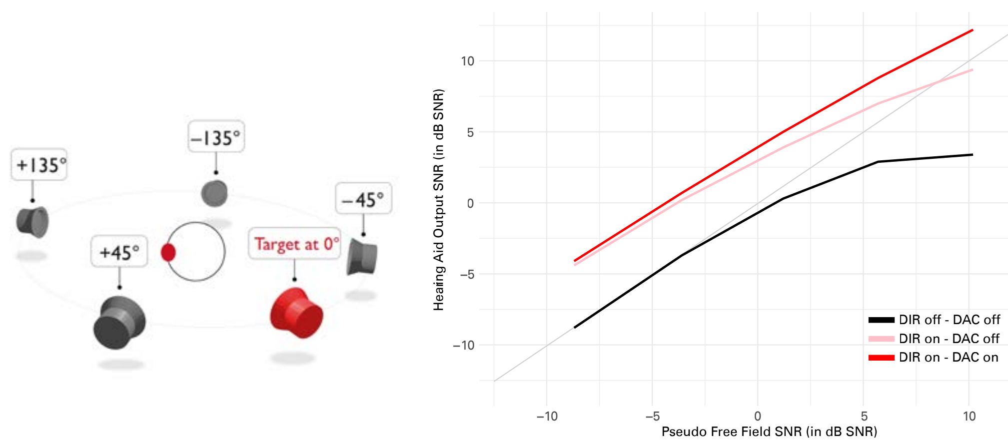

Figure 2. On the left, spatial distribution for speech, the target signal, in red and noise sources in gray for output SNR measures. On the right, a graph of the output SNR with compression plus three different feature configurations: black, compression only; pink, directionality and compression; and red, directionality, DAC™, and compression activated. The straight light gray curve shows values where the input SNR on the horizontal axis equals the output SNR on the vertical axis.

The SNR degradation caused by the compression for positive input SNRs is clearly visible from the black curve in Figure 2. At a positive input SNR, compression applies more amplification on the softest part of the signal which is mainly noise. The result is that the output level difference between speech and noise is reduced by compression. Similar results were found by Naylor & Johannesson (2009) and Alexander & Masterson (2015).

Activating the directionality from the Dynamic Noise Management™ partially solves the SNR degradation caused by the compression. When noise is spatially separated from the speech source, a high benefit from directionality is observed with the pink curve which shows an improved output SNR. However, despite this improvement, the same degradation effect due to the compression is visible at positive input SNRs. This is demonstrated by the slope of the curve which becomes flatter as the input SNR increases.

DAC™ will prevent noise, which is filling the pauses of the speech signal, from receiving the same amplification as speech at the same estimated level.

Zerena's output SNR with DAC™ and directionality activated is shown with the red curve. This condition provides the best output SNR at any input SNR compared to the other test conditions. At negative input SNRs, the benefit is similar to that with directionality alone. The benefit from DAC™ is more pronounced at higher input SNRs. DAC™ will prevent noise, which is filling the pauses of the speech signal, from receiving the same amplification as speech at the same estimated level. The red curve in Figure 2 demonstrates this effect as it is much steeper than the other curves where DAC™ was deactivated. DAC™ was designed specifically for environments with positive input SNRs, which represent most listening environments as reported by Smeds et al. (2015). They estimated the SNR in various daily life situations with an A-weighting on the best ear and determined that it is primarily above 0 dB SNR. This positive SNR corresponds to the threshold where DAC™ gives the bigger benefit. Before running any speech tests to evaluate the performance of DAC™, it is important to be aware that the optimal SNR level at which DAC™ will show an improvement is above 0 dB SNR.

IMPROVED LISTENING EFFORT IN SPEECH-IN-NOISE SITUATIONS WITH DAC™

DAC™ reduces noise amplification in speech-in-noise conditions and its effect can be measured with the output SNR. The relationship between differences in output SNR and speech perception in noise has been evaluated by Miller et al. (2017), Gustafson et al. (2014), and Wu & Stangl (2013) using different measurements tools. These studies show that a change in output SNR cannot be systematically predicted and measured with speech intelligibility tests like speech reception thresholds or phoneme recognition. They suggest that a change in output SNR might affect other aspects of speech perception like acceptable noise levels or listening effort measured with response time. With this in mind, Zerena was tested using an adapted version of the WAKO word recognition rhyme test (v. Wallenberg & Kollmeier, 1989) which simultaneously measures the listener's answer and response time.

Thirty experienced Juna 9 users with a moderate to severe hearing loss were tested to evaluate DAC™ technology in a controlled environment. Binaural amplification was applied and verified with real ear measurements to fit targets delivered by the NAL-NL2 fitting rationale. Speech and noise were presented from one single loudspeaker in the front at a fixed SNR of +5dB SNR and speech level at 65 dB SPL. The test presentation was automated so that for each word the response as well as the response time were recorded. The test results in terms of word recognition and response time are shown in Figure 3.

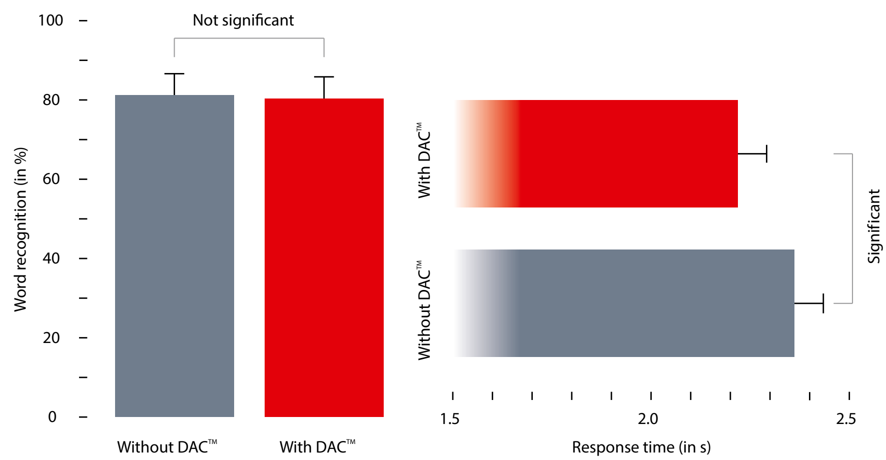

Figure 3. Results from word recognition tests with DAC™ deactivated in gray and DAC™ activated in red. Performance is measured with word recognition percentage on the left side and with response time in seconds on the right side. Average performance and one standard error are shown on the graphs.

DAC™ estimators are accurate enough so that attenuation is only applied to the noise signal, leaving the speech signal unchanged.

The first interesting and prominent result involves the word recognition performance with DAC™. Enabling DAC™ should only reduce the amplification of the noisy parts of the test signal, leaving speech unchanged. If DAC™ was not fast or accurate enough, then we would expect that some speech portions would be also attenuated. This unwanted effect would lead toward a decrease of speech audibility causing a possible negative effect on word recognition. The average difference in word recognition between both test conditions is less than 1% and not statistically significant. This result suggests that DAC™ estimators are accurate enough so that attenuation is only applied to the noise signal, leaving the speech signal unchanged.

Response times are also measured with and without DAC™ for all the tested words. The average response time is significantly faster with DAC™ activated (145 ms, $p = 0.03$). This indicates that listeners needed less time to give their answer at the same intelligibility level when DAC™ is activated.

The output SNR measures have shown that DAC™ reduces the level of noise for a speech-in-noise signal which ultimately improves the hearing aid output SNR. This attenuation is precise enough so that speech intelligibility is not affected by this processing, however it makes the response time faster during the test. The interpretation of changes in response time can be found in the ease of language understanding (ELU) model (Rönnberg et al., 2013; see Figure 4) and should reflect changes in listening effort (Houben et al., 2013 and Pals et al., 2015).

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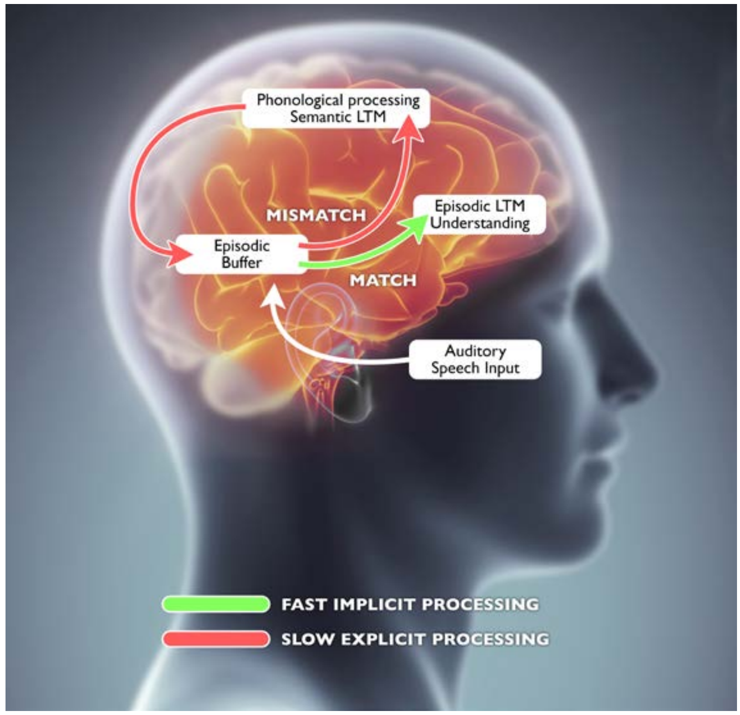

Figure 4. The ease of language understanding (ELU) model from Rönnberg et al. (2013). The match between the speech input and phonological attribute is fast and automatic in ideal listening conditions. Extra working memory, like phonological processing and semantic long-term memory (LTM), is needed when the input signal is degraded by noise or a hearing loss leading to a mismatch. This mechanism takes more time but can be reduced by improving the quality of the incoming speech signal.

The ELU model assumes that the cognitive processes making auditory speech input understandable are fast and automatic within an episodic buffer. The episodic buffer matches the incoming phonological information with existing language representations stored in the long-term memory (LTM) to quickly understand speech. This process is implicit and fast only with an incoming signal that is not degraded by noise (Houben et al., 2013) or distorted by a hearing loss (Carroll et al., 2016). A degraded auditory input signal makes this association more complex and requires additional explicit cognitive processes. This effect can be measured with an increased response time for hearing impaired listeners compared to normal hearing listeners despite compensation for audibility (Carroll et al., 2016). Working memory capacity with semantic LTM and phonological processing is required to understand the degraded input signal. As seen in Figure 4, the explicit processes (red) run on a slower time scale measured in seconds while the implicit processes (in green) are faster.

A hearing aid signal processing algorithm, that improves the SNR at its output, will provide a cleaner signal that should reduce the cognitive processing time during speech recognition tasks. Gustafson et al. (2014) evaluated the effect of noise reduction with output SNR, phoneme recognition, and response time. They found a clear and systematic improvement with noise reduction in terms of output SNR and response times while the benefit in terms of phoneme recognition was less consistent. Their interpretation is that the reduced response times reflect a benefit in terms of listening effort. Our results show a similar pattern, i.e., DAC™ improves the output SNR and also response times while it preserves consonant recognition. These results suggest that DAC™ also has a positive impact in listening effort.

MORE COMFORT IN DAILY LIFE SITUATIONS

This feature evaluation would not be complete without testing DAC™ in daily life situations because controlled lab tests cannot cover the complexity of all possible listening environments. Therefore, all the participants had to gather experience with and without DAC™ before the word recognition test. They received a blank diary to report and describe their most memorable listening experiences during two weeks with two sets of hearing aids. The test order was randomized and counterbalanced so that one half of the participants wore DAC™ first and then a pair of hearing aids with conventional amplification, without DAC™, while the other half had its test order inverted. The participants were blinded as to the differences between the two sets of hearing aids and to the test purpose. They were asked to focus only on their listening experiences so that they could accurately describe what they heard. Before the test started, they were also instructed on how to express different aspects of sound quality like intelligibility in quiet and noise, spectral richness, localization of sounds, naturalness, listening effort, comfort, and audibility of soft sounds. The reported verbatim was used for the analysis that followed the qualitative research method as described by Knudsen et al. (2012). The main advantage of this method, using an open-ended approach, is to generate new and spontaneous information coming from the experience and not create data that could be partially induced by a questionnaire.

The analysis based on the verbatim conceptualizes and structures the recurring reported listener experiences with keywords. As a certain degree of interpretation is needed from the analyst, it is important to ensure that the code generated from the blinded responses is consistent and transparent. This is called the dependability of the analysis in qualitative research. The first coding needs to be reviewed and challenged by an external researcher. If some discrepancies with the initial coding are found due to different interpretations, then both analysts have to discuss each case and find an agreement on the less ambiguous response interpretation. The final representation of the results is based on the frequency of all the recurring keywords as shown in Figure 5 with one word cloud for each tested system.

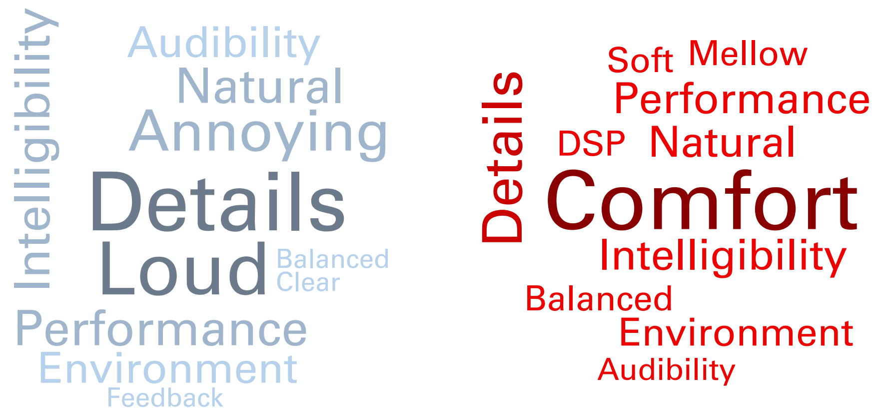

Figure 5. Reported experiences with conventional amplification in gray on the left and with DAC™ activated in red on the right. Font size is proportional to the keyword frequency.

It is important to keep in mind that the size of the word mainly represents the consensus between the listeners. If a majority of them report that the hearing aids with the conventional amplification were loud and provided details of the surroundings, then the font size for these keywords will be increased. It is also possible to group some concepts like 'audibility', 'natural', 'details', or 'environment' that belong to a similar category with some subtle nuances. They evoke a better audibility and awareness of the surrounding sounds than we could expect from compressive amplification. The price to pay for this improved audibility is that amplification is perceived as being too loud or even annoying in some situations. Activating DAC™ was experienced as providing more comfort especially in noisy situations. Some participants could hear that signal processing was adapted to changes in the environment. This reporting was labelled with DSP during the coding. There are also trends in describing the amplified sound as being mellower or softer which is in a way connected to the reported comfort. While each word cloud gives a fairly good description of all the listening experiences, it might not be easy to precisely see the differences found between both tested systems. There are some keywords that are shared between systems, e.g. 'details', 'performance', or 'intelligibility'. It is quite difficult to conclude from Figure 5 that one system provides more 'details' just by comparing the font sizes. Visualizing what is different and what is common between the systems demonstrates what to expect, in general, from the tested hearing aids, but also the specific benefit to expect from DAC™. Clouds from Figure 6 plot successively the differences and the similarities between both systems.

DAC™ improves the output SNR which also gives more comfort to the hearing aid user.

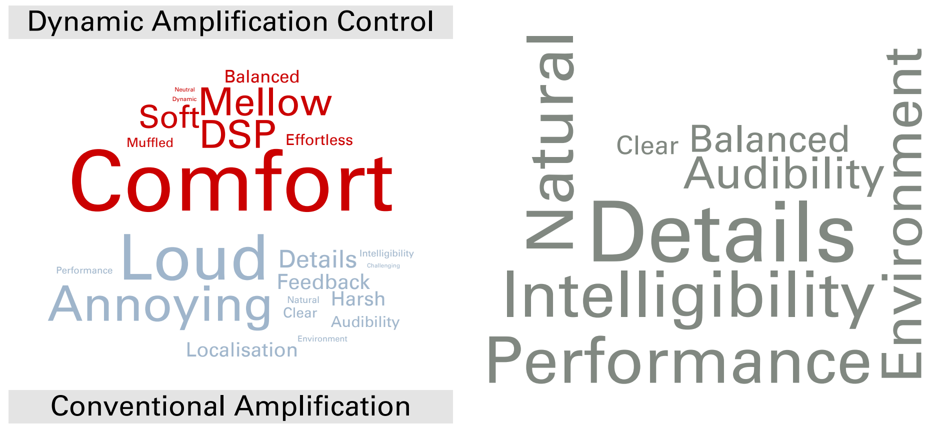

Figure 6. Comparison cloud between both tested hearing aids on the left. Keywords specific for conventional amplification are on the bottom and on the top for DAC™. On the right, the keywords that are shared with both systems represent what is the overall benefit with the tested hearing aids.

The overall impression from the field test is a general improvement in sound quality with Zerena. Independent of the DAC™ settings, Zerena's signal processing delivers a natural and detailed sound. By activating DAC™, listeners report an improved comfort in challenging situations. These reported experiences are consistent with previous lab findings, i.e., DAC™ improves the output SNR and also reduces listening effort for speech-in-noise situations.

DAC™ QUALITATIVE CONTROL OF AMPLIFICATION

Dynamic Amplification Control™ is a unique signal processing algorithm that avoids over-amplification of noise. DAC™ aims to improve hearing aid acceptance in any listening environment. It is designed as an intelligent unit that makes the link between noise management algorithms, like directionality and noise reduction, and the compression unit. The benefit of DAC™ can be measured technically with improved hearing aid output SNR but also clinically with improved response times during word recognition tests. Experience from field tests also indicates more comfort in noisy listening situations when DAC™ is activated. DAC™ should facilitate the fitting process and improve the first listening experience for any hearing aid user.

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