Topics in Amplification

DYNAMIC AMPLIFICATION CONTROL™ – INTELLIGENT AMPLIFICATION

Providing the right amount of amplification is the essential function of any hearing aid but also the main challenge for the hearing care professional (HCP). The fitting process might be prolonged and cumbersome if the amplification does not meet the hearing aid user's needs. There is no off-the-shelf solution that will guarantee a successful fitting. Instead, clinical experience combined with advanced technology, are the tools upon which the HCP should rely to optimize each individual fitting.

Traditional amplification, defined by fitting rationales, is designed for speech in quiet. It is quite straightforward to verify that amplification matches the prescribed gain with real ear measurements and to validate its benefit with speech tests. The difficulties start as soon as amplification is confronted with noisy real-life listening situations because not all the encountered signals are as significant as speech. Without an additional algorithm, the amplification will give all sounds equal importance and amplify them based only on the input signal. Noise management algorithms, like directionality and noise reduction, are traditional solutions that address these difficulties. However, each system wants to independently solve one part of the equation using amplification for audibility and noise management for comfort. Each of these systems acts in an opposite direction. It is like driving a car and trying to brake while at the same time constantly accelerating. The noise management reduces the unwanted signal that is later inappropriately amplified by the amplification unit. For this reason, traditional technology has always resulted in a compromise between audibility and comfort, as neither specification could be simultaneously achieved. Finding the best compromise acceptable for each individual takes time, and can potentially result in disappointment when the appropriate balance between audibility and comfort cannot be found. In response to the constant negotiation between audibility and comfort, Bernafon introduces Dynamic Amplification Control™ (DAC™).

DAC™ is one of the elements comprising the new Dynamic Environment Control System™ (DECS™). Combined with Continuous Environment Detection, Dynamic Noise Management™, and Dynamic Speech Processing™, the entire system works to seamlessly improve each listening situation for the end user. DAC™ is a new class of algorithm developed with an intelligent processing unit that uses information from the noise management block to control the amplification. It aims to improve comfort without making any compromise on speech amplification. Within this paper, the limitations of traditional amplification will be explained, a detailed description of DAC™ will be given, and helpful insights to the fine-tuning of DAC™ will be described.

Traditional amplification limitations

While there are many possible technical implementations of traditional amplification systems, some functions are indispensable. The following processing units are required for any amplification system: a level estimator unit which measures the input signal, a level-to-gain function (programmed with the fitting software) which determines how much gain to be added to the signal, and a gain application unit that finally combines the signal plus the defined gain. Their implementation and relationships are shown in Figure 1.

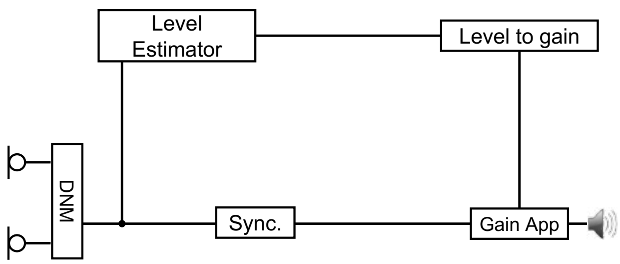

Figure 1: Block diagram of the implementation of a conventional amplification unit in a hearing aid.

One challenge is to design a level estimator unit that will give the appropriate information to later stages of the amplification block. There are a lot of articles and research which try to understand the implication of using different parameters for the level estimator. All these efforts which aim to improve the hearing aid performance, consider amplification application only as a quantitative problem, i.e., how loud is the input signal level? These systems apply the same gain based on the input signal level without analyzing the signal type. Therefore, the same gain is applied to a speech signal as that applied to a noise signal for a given estimated level. The level of amplification may be the correct amount required for speech but, it might not be optimal in some circumstances when it increases the noise level. Lai et al. (2013) describe this local over-amplification of noise as the speech pause effect. This effect is prominent in listening situations where noise is at the same level as the softest phonemes of speech. Compressive amplification will reduce the ratio between the speech and noise signals at positive input SNRs by over amplifying the noise. The consequence of this noise over-amplification is a degraded hearing aid output signal-to-noise ratio (SNR) as described by Naylor & Johannesson (2009).

The second limitation, for implementing an amplification system that is based on quantitative estimation only, is its possible interaction with noise management algorithms. This is particularly important for serial implementation of directionality and amplification (Chung, 2007). This implementation means that each block is placed and processed one after the other. While directionality can achieve effective attenuation of spatially separated noise sources, its benefit can be negatively affected by amplification. In compressive amplification, more gain is given to the softest portion of the signal in order to restore audibility. The side effect is that noise, attenuated by directionality, will be estimated at a lower input level and might receive an additional amount of amplification. A system that could reduce this noisy effect of amplification would provide many benefits for the hearing aid user: improved acceptance, more comfort, and guaranteed audibility.

Qualitative analysis with DAC™

DAC™ tackles amplification limitations by controlling the amplification. This is achieved by providing information to the amplification unit about the signal coming from the noise management block. The DAC™ analysis unit characterizes the type of the signal i.e., is the signal speech or noise. For a speech signal, no changes will be applied and the amplification should be kept as it was programmed and verified. However, when noise is detected, instructions are given 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 2.

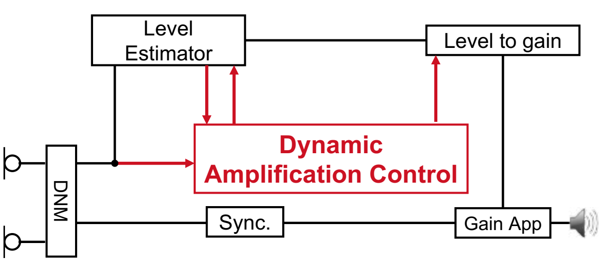

Figure 2: Block diagram of the implementation of the Dynamic Amplification Control™ algorithm within DECS™ technology.

DAC™ uses information from the Dynamic Noise Management™ block but also from the level estimators of the amplification unit. It has specific built-in estimators to monitor the SNR with a high temporal precision. Having fast and responsive estimators is mandatory in order to follow changes in everyday listening situations. When the local SNR, estimated with such precision, is below a defined threshold, then a decreased amplification is applied. The correction is applied on the level estimator and also on the level-to-gain function. The aim is to moderate the amplification applied to the noisy portion of the signal.

One advantage of the DAC™ technology is that it measures the SNR on a continuous scale instead of categorizing the listening environment into distinct static classes. DAC™ can detect subtle differences in changes between listening environments, for example, if the noise level slightly increases. Even an advanced environment classifier might not detect any differences if the change in SNR is too small. However, in certain situations a small change in SNR might have dramatic consequences on speech perception in noise.

The qualitative analysis from DAC™ provides an intelligent application of amplification. This is possible by distinguishing between the types of incoming signals. The positive effect of DAC™ is that noise attenuated by the noise management block will be detected and labelled so that its amplification will be carefully adapted. The amplification with DAC™ is still level dependent but has been improved in order to also be situation dependent without restrictive classification.

Benefit of situation dependent amplification

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 visualize DAC™'s effect is by separating the signal and the noise from recordings of hearing aid output (Hagerman & Olofsson, 2004). The extracted noise and speech signals can be aligned and presented in the same illustration. Figure 3 shows two hearing aid output recordings with and without DAC™.

The previously described speech pause effect is visible on the top waveform at the start and at 3 seconds without DAC™. In this measurement condition, the noise signal comes quite close to the speech peaks. In the short speech pause at 3 seconds, noise gets as much amplification as the softest parts of speech. This kind of amplification is perceived as very noisy (Kates, 2010).
The main effect of DAC™ emerges when reducing the noise signal. DAC™ is fast enough to reduce noise amplification within a continuous speech signal. The difference in output between speech and noise is therefore increased and can be expressed as an improved hearing aid output SNR. same illustration. Figure 3 shows two hearing aid output recordings with and without DAC™.

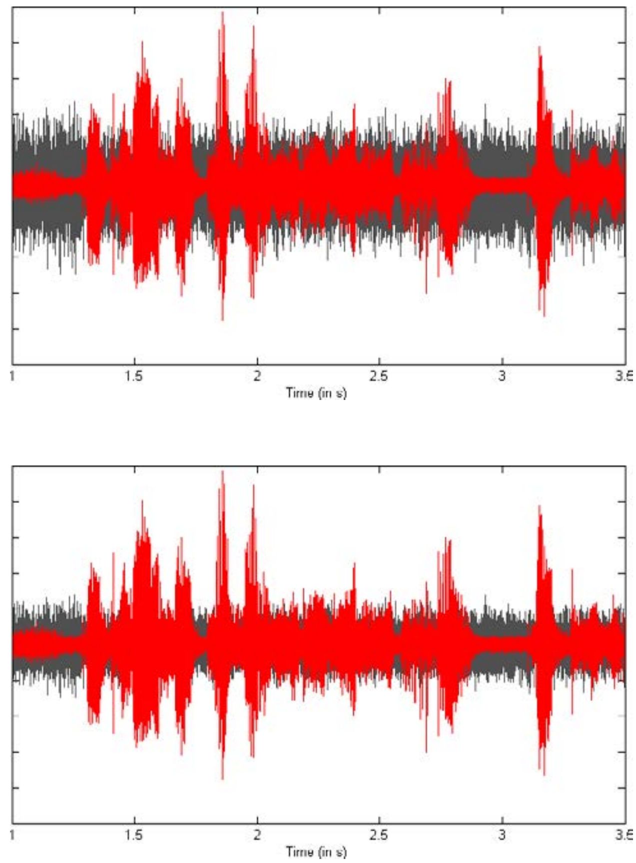

Figure 3: Hearing aid output signal without DAC™ on the top and with DAC™ on the bottom. ISTS speech (red) and ISTS noise (gray) signals were separated using a post processing inversion technique.

The second important information from this measure is about the processed speech signal. DAC™ was designed to preserve the amplification of speech. Because DAC™ estimators are fast enough to accurately differentiate the signal type, amplification corrections are applied so that the speech signal is not affected when DAC™ is enabled. This aspect is important because intelligibility relies initially on the audibility of the speech signal (Amlani et al., 2002). It is, therefore, very important to ensure that DAC™ does not reduce any portion of the speech signal.

Fitting and fine tuning DAC™

A new innovative tool allows you to specifically refine DAC™ to meet your clients' listening preferences. DAC™ adaptation and fine tuning options are implemented in Oasis^{nxt} fitting software in the feature section under the general tab as shown in Figure 4. Specific settings can be defined for each listening program with two handles: preference for speech in noise environment and comfort in noise only. Each handle addresses the hearing aid performance in the two most significant listening categories defined by the presence or absence of speech.

listening profile, distortion hater, needs more audibility and might be distracted by noticeable signal processing applications like directivity, noise reduction, and also DAC™. For this user profile, DAC™ should only work for higher noise levels, and this can be achieved by selecting the "Audibility" or "Max Audibility" setting.

DAC™ corrections for listening environments without speech cannot be based on the SNR. In this case, DAC™ uses other parameters to estimate the absence of speech and will slowly

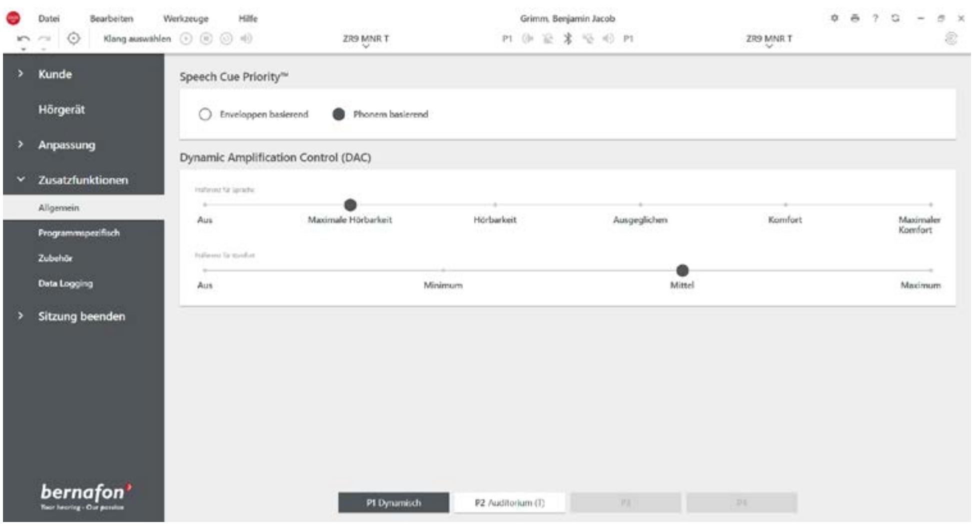

Figure 4: DAC™ feature HCP controls in the Oasis^{nxt} software

The first handle, "Preference for speech in noise environment," will influence DAC™'s behaviour if speech is detected. It covers any listening situations with speech even with soft background noise. The default setting is the optimal position for the best mix between audibility and comfort. However, Völker et al. (2016) showed that listeners might have some individual preferences regarding optimal hearing aid signal processing settings. They describe two major trends among many listening preference profiles, the "noise haters" and the "distortion haters". A noise hater will tolerate more correction from a signal processing algorithm as long as more noise is attenuated. For users with this listening profile, the setting preference for speech in noise environment can even be set to "Comfort" or "Max Comfort" if required. This will shift the threshold triggering the reduction of amplification to lower noise levels. The second

reduce the amplification in these situations. The amount of amplification reduction is controlled by the "Comfort in noise only environment" handle. It covers a wide range of noise only environments including soft fan noise, computer noise, or a loud vacuum cleaner engine. The default setting is set to "Medium" and was optimized to cover the noise sources most often encountered. However, individualization based on specific needs is also possible with this handle. Some noise sensitive users might require more amplification reduction. DAC™ offers a "Maximum" option that provides more comfort for noise only listening situations. On the other hand, some users might prefer to be more aware of changes in their environment and require less comfort in noisy situations. The "minimum" setting should be chosen to optimize audibility for these users.

Another important point for the fitting process is the verification of gain with real ear measures. There are many parameters that can be adapted for this protocol. One crucial aspect is the choice of the test signal. While it is recommended to use speech as a test signal, there is also the possibility to perform real ear measurements with a noise signal. If using a noise signal, it is recommended to disable both handles from DAC™. For measures with any speech signal, no special attention should be given to the state of DAC™ settings.

Unique seamless experience

Being able to improve the output SNR while still preserving the speech signal with DAC™ has many advantages. First, hearing aid users will have better acceptance of amplification if unnecessary amplification of noise is avoided. Second, DAC™ estimators can track changes in any listening situation and do not rely on fixed and predefined environment categories. Finally, DAC™ offers high end personalization options to adjust its performances based on your clients' listening preferences.

DAC™ is a key component of the new DECS™ feature 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.

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