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BENEFITS OF DYNAMIC AMPLIFICATION CONTROL IN COMPLEX LISTENING ENVIRONMENTS

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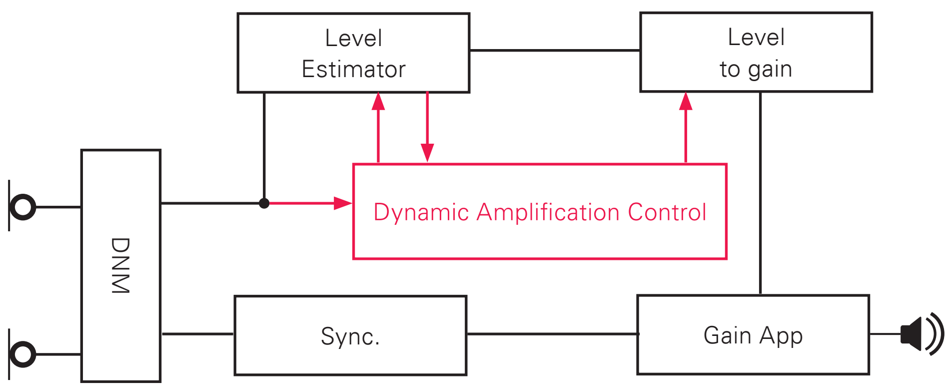
DAC™ is a key component of the new DECS™ technology developed by Bernafon. It does not belong to a classification system or a noise management algorithm. It addresses the problem of noise amplification with a sophisticated and unique solution for any listening situation. Being able to improve the output 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.

DAC™ is a new class of signal processing algorithm that addresses the "speech pause" effect generated by dynamic compression.

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 below.



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 overamplification 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.

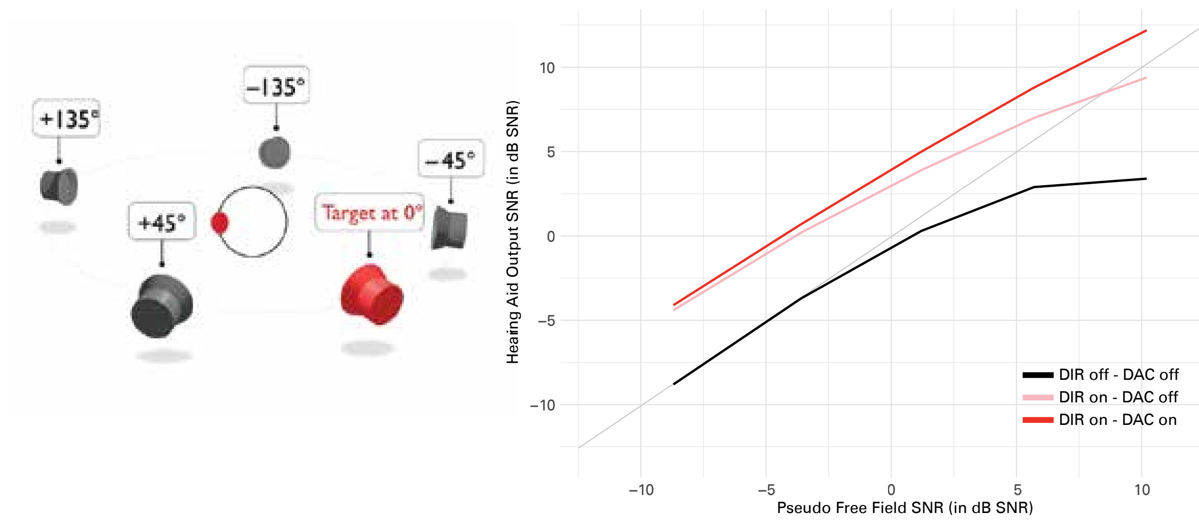
HEARING AID OUTPUT SNR INCREASED WITH DAC™

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 postprocessing 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.

the estimated SNR at the ear position.



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, DACTM, 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.

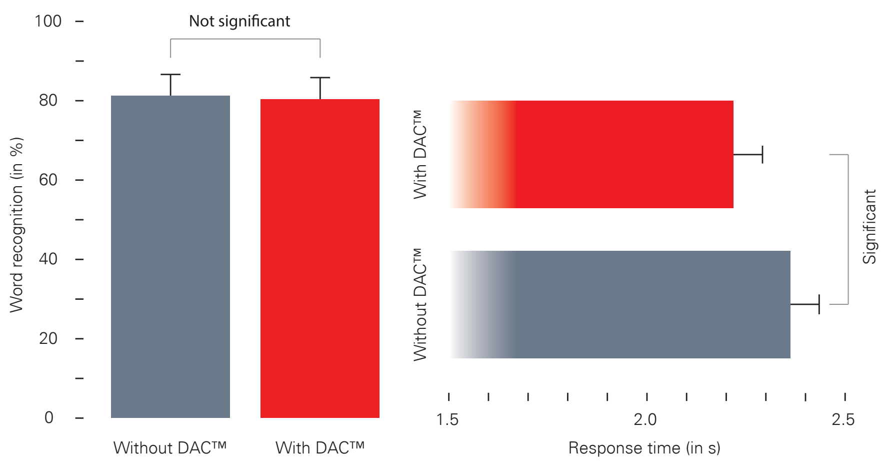
The SNR degradation caused by the compression for positive input SNRs is clearly visible from the black curve in Figure 2. At a positive input SNR, compression applies more amplification on the softest part of the signal which is mainly noise. The result is that the output level difference between speech 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.

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 below:



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.

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.

IN PRELIMINARY DATA, CELL PHONE USERS HAVE LONGER ABR LATENCIES

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The increasing use of mobile communications has raised concerns about possible interactions between electromagnetic radiation and the central auditory system, but research remains limited.

To help fill that gap, we investigated the effects of mobile phone use on the latency of peaks I, III, and V for 500 Hz, 1 kHz, 2 kHz, and 4 kHz in tone-burst auditory brainstem response (ABR).

This preliminary data suggests that prolonged use of cell phones delayed neural information transmission in the central auditory pathway. However, the functional significance of the delay is still to be determined.

SHORT-TERM VS LONG-TERM USE

Guidelines limiting exposure to electromagnetic fields from mobile phones use the specific absorption rate (SAR). The SAR corresponds to the rate at which radiofrequency energy is absorbed in the head of a wireless handset user.

The antenna is the main source of radio waves that produce SAR in the body, although there may also be leakage from the phone body shell.

Commonly, mobile phones are held near the head when in use, keeping the cochlea close to the antenna.

However, mobile phone use has not been extensively investigated in the literature as a cause of cochlear hair cell loss, with most studies examining the effects of short-term rather than chronic exposure.

In one recent study that did look at long-term use, mobile phone users of more than a year had a significantly increased risk of absent distortion product otoacoustic emissions, higher speech frequency thresholds, and lower middle latency response waves Na and Pa amplitudes compared with controls (Otolaryngol Head Neck Surg 2011;144[4]:581-585). More than three years of mobile phone usage emerged as a risk factor.

SELECTION CRITERIA

Due to the limited and controversial nature of existing evidence, combined with the widespread use of mobile phones, it is crucial to determine if the electromagnetic fields of mobile phones have adverse effects on the human auditory system.

Therefore, the current study was designed to investigate these potential effects, as determined by changes in tone-burst auditory brainstem response, and to analyze the recovery pattern in case of any aftereffects of these electromagnetic fields on click ABR.

Included in the study were 15 normal-hearing adult volunteers who did not use a cell phone—eight men and seven women between the ages of 18 and 40—comprising Group A, or the mobile nonusers.

Group B, or the mobile users, consisted of 15 normalhearing adults in the same age range as the Group A participants who had used mobile phones for two hours a day over at least the past three years. Participants were right-handed postgraduate students and staff members at the Dr. M.V. Shetty College of Speech and Hearing in Mangalore, India, who met the following participation criteria:

- Thresholds of 15 dB or better for the octave frequencies 250-8,000 Hz for air conduction and 250-4,000 Hz for bone conduction, and a normal tympanogram (Type A).
- No history of audiologic or neurotologic problems, such as vertigo, tinnitus, noise exposure, etc.
- No medical problems or use of medications for any reason.
- No history of chronic narcotic or alcohol use.

Table 1. *Mean, Standard Deviation (SD), and F Values for Cell Phone Users and Nonusers*

Non Cell Phone Users, Right Ear						Frequency	Cell Phone Users, Right Ear						F values		
Peak I		Peak III		Peak V			Peak I		Peak III		Peak V				
Mean	SD	Mean	SD	Mean	SD		Mean	SD	Mean	SD	Mean	SD	Peak I	Peak III	Peak V
1.34	.24	3.45	.32	5.36	.17	500 Hz	1.49	.43	3.62	.33	6.05	.21	27.62 $P < 0.01$	9.19 $P < 0.01$	125.63 $P < 0.01$
1.44	.25	3.37	.16	5.38	.21	1 kHz	1.59	.26	3.60	.34	6.11	.55			
1.24	.20	3.43	.20	5.38	.19	2 kHz	1.78	.29	3.55	.36	6.31	.43			
1.28	.18	3.55	.26	5.37	.21	4 kHz	1.76	.29	3.84	.33	6.04	.16			

All study participants in Group B used mobile phones produced by a single manufacturer that complied with International Commission on Non-Ionizing Radiation Protection (ICNIRP) Guidelines for Limiting Exposure to Time-Varying Electric, Magnetic, and Electromagnetic Fields (up to 300 GHz; Health Phys 1998;74[4]:494-522). They also generally used cell phones in their right ears.

THE FUNCTIONAL SIGNIFICANCE OF THIS DELAY IS STILL TO BE DETERMINED.

The mobile phones all were recent models from the same year, employed Global System for Mobile communications (GSM) for networking, and had a conformed antenna housed inside the plastic case.

ALTERED ABR LATENCIES

In order to check for any potential deleterious effects on the auditory system of electromagnetic fields emitted by mobile phones, tone-burst ABR was used with two surface cup electrodes that had impedance set lower than 5 k ohms.

The inverting electrode was placed on the right mastoid, the non-inverting electrode over the vertex (Cz of international 10-20 system), and the ground electrode over the left ear mastoid.

Recorded potentials were amplified with high- and low-pass filters set at 100 and 300 Hz, respectively. At least 1,000 responses were averaged, with a maximum number of 2,000 and a repetition rate of 13.1/sec.

A 60-dB sPL, 100-μs (alternating) tone-burst (500-, 1,000-, 2,000-, 3,000-, and 4,000-Hz) sound was presented to the right ear with headphones.

We recorded ABRs twice to ascertain their reproducibility under each condition, measuring latencies and amplitudes of waves I, III, and V.

Descriptive statistics were calculated to find the mean and standard deviation of the groups, and the two-way ANOVA test was done to detect any significant differences between the groups. We confirmed normal hearing by pure-tone audiometry in both groups of participants.

The table shows the ABR results from the right ear for cell phone users versus nonusers.

For all frequencies tested, peak latencies of waves I, III,
and V in the tone-burst ABR were significantly delayed in the right ear of participants who used cell phones for more than three years.

The differences in latencies between cell phone users and nonusers were 27.62 for peak I ($P < 0.011$), 9.19 for peak III ($P < 0.01$), and 125.63 for peak V ($P < 0.01$).

No hearing loss or other symptoms were noted. Pure-tone audiometry and otoacoustic emissions examinations confirmed the absence of these conditions.

Further studies testing for longer-term effects and in larger groups of cell phone users are recommended.

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Interface SoundClip-A

*Devices streaming with Bluetooth® 2.1 or later; **BTD800 USB dongle

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