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# Single Microphone Noise Reduction: New Findings

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## ABSTRACT

Single microphone noise reduction as implemented in many DSP hearing aids has failed to demonstrate objective performance improvements in noise. A 9-channel DSP hearing aid implementing single microphone spectral subtraction was evaluated on 28 binaurally aided subjects. Modified HINT thresholds with a speech and noise location of 0 degrees azimuth were measured with and without noise reduction. Significant improvements in HINT thresholds were found with the noise reduction enabled

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# INTRODUCTION

Spectral subtraction, as a signal processing algorithm, has been used for many years to reduce or eliminate the level of maskers found in mixed audio signals. The level of success with various implementations of spectral subtraction has depended upon the technical application, as well as the expectations of the user.

In hearing aids, which ideally provide amplification as well as improvements in signal-to-noise ratio, spectral subtraction has not yet demonstrated measurable performance benefits in noise. Implementations have been plagued with acoustic artifacts (waterfall effect, pumping, or other distortions), and when these are controlled, the benefits are only seen with respect to self-evaluated improvements in listening comfort1.

A new implementation which combines a modified spectral subtraction technique with a unique signal processing algorithm, has been evaluated to determine if measurable performance improvements can now be identified.

# ALGORITHM

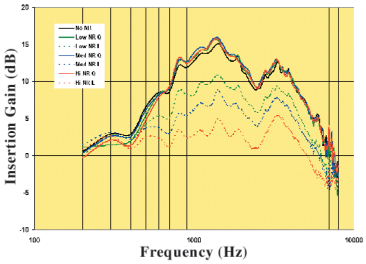

Figure 1: Real-Ear Insertion Response

All single-microphone noise reduction (NR) algorithms need a method of differentiating the signal from the noise in a single input. The current system implements NR in each channel of a unique nine-channel architecture2. Signals in each channel are separated into the amplitude envelope and vibrational component (i.e., the vocal cords create the vibration that is modulated in amplitude by the vocal tract). The noise detectors determine the portion of the entire signal that is noise based upon the modulation rate of the envelope. From this, the signal-to-noise ratio in each channel is calculated which, in turn, determines the specific amount of noise reduction applied within the channel.

The NR system (currently implemented in the NATURA™ 2 SE and CONFORMA™ 2 SE products) has dual time constants. The noise detector is characterized by an attack time of a few seconds (to accumulate sufficient information to accurately identify noise) and fast release times (to shut off quickly when the noise is no longer present). Once engaged, real-time processing with very-fast, symmetric attack and release times is implemented to respond to ongoing variations in signal-to-noise ratio.

An example of the real-world impact of this processing is shown in Figure 1. The Real-Ear Insertion Response (REIR) averaged across 14 ears shows the change in gain with different levels of NR. It also demonstrates the system's ability to maintain the prescribed gain when the noise detector is not triggered, as shown in the quick ("Q") measures which overlap the curve without noise reduction.

# METHOD

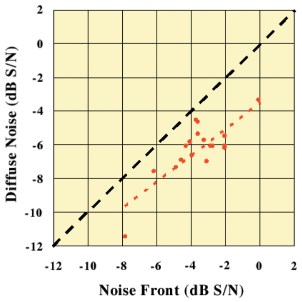

Figure 2: "Normal" HINT in noise

When evaluating a noise reduction system in the laboratory, it is important to control variables that can impact performance, such as the modulation of the masker (which increases performance variability by allowing for listening between the noise) as well as the location of the masker (which may introduce interaural cues that improve performance).

The current study used the Hearing In Noise Test (HINT)3 because of its control of materials and listening conditions. In order to evaluate the method of presentation of the materials, a small group of normal-hearing listeners were tested using two scenarios: noise presented from a single front speaker (as specified by the HINT), and the same noise presented simutaneously from four speakers positioned at 45o, 135o, 225o, and 315o azimuths (time shifted to make the noise uncorrelated). Figure 2 demonstrates differences in performance between the noise-front conditions and the diffuse noise conditions (data are for 18 normal hearing subjects). The spatial separation in the diffuse condition improved thresholds by 2.2 dB. Since the intent of the current study is to select the most difficult condition to evaluate the noise reduction (NR) system, only the noise-front condition is used.

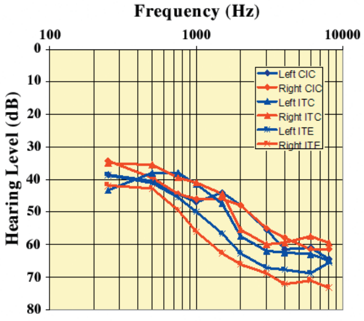

Figure 3: Mean Audiogram

Subjects: Twenty-eight subjects were fit binaurally with CIC, ITC, or ITE devices with the current noise reduction algorithm. Audiometric data are summarized in Figure 3.

Apparatus: Previous experience with this system led to the creation of a modified set of HINT materials. To ensure the NR process-ing was engaged during the playback of the speech materials, the noise onset was increased to 5 seconds prior to speech onset4,5.

All subjects were fit using a proprietary fitting algorithm (EXPRESS*fit*™). Fittings were not altered between noise-reduction and non-noise-reduction settings.

Protocol: Subjects were placed in a double-walled sound room facing a single loudspeaker. HINT Reception Thresholds for Sentences (RTS) were measured. The RTS measures the level at which the sentences can be correctly repeated half the time and is expressed in A-weighted dB for quiet conditions or dB signal-to-noise ratio (S/N) relative to the 65 dB A noise used. RTS was measured in five conditions: unaided, aided with no NR, low NR, medium NR, and high NR. Lower RTS indicates that the same level of performance can be achieved when the sentences are presented at a lower intensity.

# RESULTS

Unaided thresholds in quiet and noise are compared in Figure 4. The data were collected on 28 hearing-impaired subjects as well as 17 normal-hearing subjects. The black line is a best fit linear regression line with a slope of 0.14 and a y-intercept of -5.93 (r2=0.71). As the level of hearing loss increases (as shown by the higher scores in quiet), the thresholds in noise also increase, and become more scattered. This demonstrates the impact of the hearing loss on the ability to extract speech from background noise. Ideally, amplification should not only improve performance in quiet (threshold loss), but also in noise (S/N loss), similar to Killion data with the SIN6.

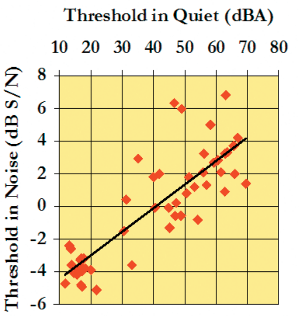

Figure 4: Unaided Quiet vs Noise

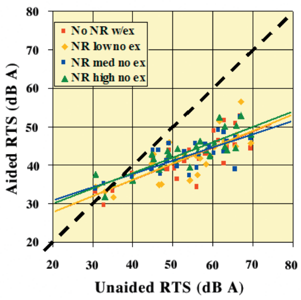

Figure 5: Unaided vs Aided in Quiet

Benefit from amplification in quiet is plotted in Figure 5. As unaided thresholds increase, aided thresholds increase at a slower rate, demonstrating improved performance with amplification. The level at which unaided and aided performance match (the points are on the dashed line) intersects with the regression lines at 30 dB A, which when expressed as HL would approximate 20 dB HL. This coincides with the definition of "normal-hearing" used in the fitting algorithm. The colored lines represent linear regression lines for each of the aided conditions. The lines are overlapping, with a predicted 4-dB increase in aided threshold for every 10-dB increase in unaided threshold (r2=0.63). The presence of NR does not significantly change performance in quiet.

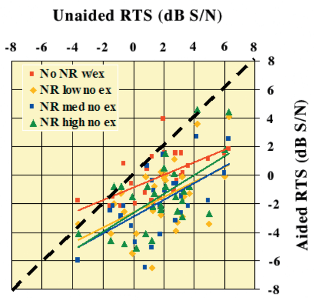

Figure 6: Unaided vs Aided in Noise

Benefit from amplification in noise is displayed in Figure 6. The points are more scattered (r2=.13 without NR, r2=.31 with NR), with a clear separation of aided scores without NR (red line) and those with NR. The mean difference between conditions is significant [F(3,75)=21.99, p<.001], with the regression equations predicting 2.7 dB improvement in threshold in noise for subjects with unaided thresholds of 0 dB S/N.

# DISCUSSION

Twenty-eight subjects have been evaluated using a new single-microphone NR algorithm. Data demonstrate the various degrees of NR available (Figure 1), the benefit from amplification in quiet (Figure 5), as well as significant improvements in speech performance in noise (Figure 6) relative to unaided and aided conditions without NR. The benefit may appear numerically small (2.7 dB between aided with and without NR), but the performance/intensity function of the HINT would predict an intelligibility improvement of between 25 and 30% in noise.

The variability in scores is greater in noise, suggesting that additional factors are influencing performance. It has been long argued that supra-threshold performance is impacted not only by a loss of sensitivity, but by an abstract quantity labeled 'distortion'. Data collection is underway in an attempt to understand what factors contribute to 'distortion'. The present data demonstrate that the current NR algorithm is capable of compensating for some portion of this disability.

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