# Comparison of Automatic and Adaptive Directional Behavior Among Competitive Hearing Aids

**Contributors**

**Michael J. Nilsson, Ph.D. BC-HIS**
**Robert Ghent, M.S.**
**Michelle L. Hicks, Ph.D**
**Sonic Innovations, Inc.,**
**Salt Lake City, Utah**

## Introduction

Currently there exists no standard laboratory process for quantifying the behavior and measuring the effectiveness of adaptive directional microphones and algorithms in hearing aids. The present standard—ANSI S3.35-2004 (American National Standards Institute, 2004)—addresses only fixed directional systems and provides a static ratio of sensitivity between energy arriving on axis from the front and energy arriving from all other directions. The Polar Sweep Oscillograph (PSO) was developed as a means to evaluate adaptive processes that change primarily in response to dynamic spatial relationships (signals that change in their location relative to the microphone). Used in conjunction with the Noise Reduction Index (NRI), the PSO not only demonstrates the adaptive processes taking place, but also yields a numeric assessment of its effectiveness in altering the signal-to-noise ratio (SNR) in the presence of simultaneous speech and a roving noise source.

Pairs of top-of-line standard behind-the-ear (BTE) hearing aids from the seven major hearing aid manufacturers, including Sonic Innovations, were evaluated using the PSO with NRI calculations to determine the relative behaviors of their respective automatic and/or adaptive directional microphone technologies. For the purposes of this paper, automatic directionality is defined as an environmentally-triggered change from an omnidirectional polar response to a directional polar response. Adaptive directionality is defined as a directional polar response that constantly changes or is updated to maintain an optimal SNR between the signal at 0° azimuth (in this case, speech) and the strongest signal not located at 0° azimuth (in this case, Speech Shaped noise) as it moves about the azimuth. Most manufacturers utilize both automatic and adaptive processing, as defined above.

## Procedure

**The Polar Sweep Oscillograph (PSO):** The PSO fills the need to evaluate spatially-adaptive algorithms (e.g., null steering) while they are adapting, with the target and roving masker presented simultaneously. An eight-loudspeaker array was constructed within a 3.0 m x 3.0 m x 2.4 m (10' x 10' x 8') double-walled audiometric test room as shown in Figure 1. A Yamaha AW4416 digital audio workstation (DAW) was used to store and reproduce the recorded source materials as well as memorize signal routing assignments and calibrated levels. Concatenated sentences from Lists 1 and 2 of the Hearing In Noise Test (HINT; Nilsson et al., 1994) were used to create a target signal of running speech with HINT noise used as the masker. The HINT noise was made to sweep virtually around the azimuth of the hearing aid under test while the target speech was reproduced simultaneously from 0° azimuth. (Figures 1A-B.)

Uncorrelated HINT masking noise was recorded on eight tracks of the DAW with target speech on a ninth track. Each track of noise was modified with an amplitude envelope that resulted in sequential reproduction from adjacent loudspeakers around the array (Figure 2, next page). The timing and overlap of non-linear onsets and offsets of subsequent noise tracks produced

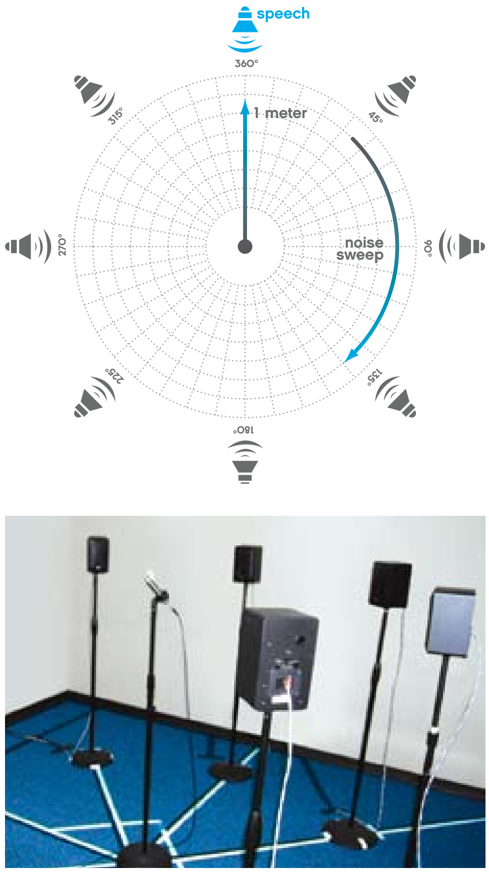

**Figure 1:** a) Schematic diagram of eight-loudspeaker array showing direction of noise sweep. b) Photo of eight-loudspeaker array with hearing aid under test located in the center.

a smooth transition of the noise from one loudspeaker to the next.

The net effect is that of a source of noise moving in a smooth arc in the azimuth around the hearing aid under test. Uncorrelated noise sources must be used to avoid possible phase interactions when the noise is coming from two adjacent loudspeakers, and the timing/shape of the envelope must be controlled to avoid a 'scalloped' level effect during the transition from one loudspeaker to the next when listening at a distance greater than the distance between adjacent loudspeakers. Executed properly, a consistent noise level can be measured at the listening position (Figure 3).

The masking noise was presented at a constant 65 dB(A) and the target speech was presented at an $L_{EQ}$ of 65 dB(A), achieving a 0 dB SNR at the center of the listening position. The hearing aid under test was connected to a 2 $cm^3$ coupler that was mounted on a stick. The coupler was mounted to a 1-inch measurement microphone that was routed to a sound level meter. The output of the sound level meter was then routed to the input of a sound card on a computer running Adobe Audition. Adobe Audition was used to digitally record and edit the output of the hearing aid under test.

The hearing aids were measured as pairs (two devices were recorded) with both hearing aids programmed to the same parameters. Results were averaged across the measurements of both hearing aids in the pair. All of the hearing aids were programmed for a flat, linear, 15 dB of gain per the respective manufacturers' fitting software. The front port of the test hearing aid's directional microphone was centered at the listening position with the axis through the microphone ports directed toward the "front" loudspeaker and parallel with the floor of the test room.

During recording, masking noise originated at 45° azimuth relative to the 'front' loudspeaker. The speech began after a four-second onset of the masking noise, and the sweep transition began one second after the speech began. Owing to a sweep rate of 11.25° per second, the locus of the masker was centered at individual loudspeakers every four seconds until arriving at the 'front' loudspeaker at approximately 33 seconds. The masking noise remained at the 'front' loudspeaker for the remainder of its presentation.

**The Noise Reduction Index (NRI):** The NRI is a method for determining the change in SNR through an audio device. Target speech (from the front) and competing noise (from the same or another location) are presented simultaneously at a known SNR (usually 0 dB) to the input of the device under test (in this case, a hearing aid). Recordings of the waveform from the output of the hearing aid under test are processed arithmetically to separate the speech and noise energy remaining in the mixed output signal. Details about the derivation, operative techniques, and validation of the NRI have been described elsewhere (Hagerman & Olofsen, 2004; Nilsson & Bray, 2004; Ghent, et al., 2007) and will not be reviewed here.

**Combining the PSO and the NRI:** Because the processed speech and noise can be separated in the recorded output of the hearing aid under test, the combined NRI and PSO recordings allows several types of analyses of temporally- and spatially-adaptive processes; among them: 1) time-waveforms can be used for a visually qualitative analysis of both the target speech and competing noise; 2) spectrographic analysis can be used to evaluate changes in the frequency domain; and 3) graphical reconstruction of the NRI yields a quantitative representation of the impact of adaptive processes on SNR. The latter is accomplished by converting the time scale on the time-waveform display to degrees

**Multi-track View**

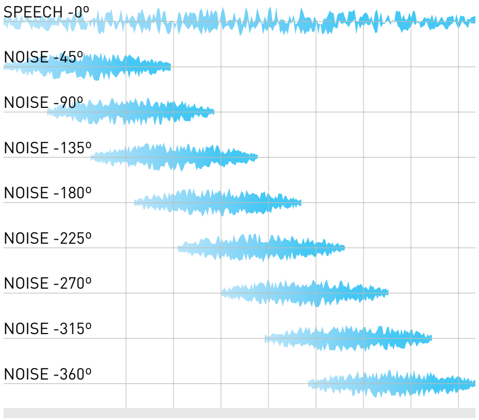

**Figure 2:** Multi-track view of PSO source materials showing timing and envelopes of maskers played back sequentially through adjacent loudspeakers in the circular array.

**Consistent Noise Level**

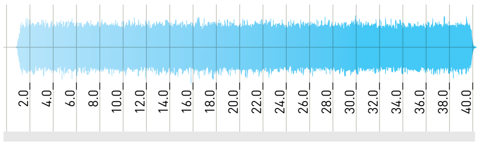

**Figure 3:** Noise sweep from the eight-loudspeaker array recorded at the listening position.

azimuth, using the 11.25°-per-second sweep rate. NRI calculations are made in four-second-wide measurement windows that slide in two-second intervals over the duration of the speech sample, resulting in a smoothed moving average. Since the noise sweep starts at 45° azimuth, this process yields 15 NRI measurements centered at 22.5° intervals from 45° to 360°, inclusive, that can be graphed for comparison.

## Results and Discussion

The competitive hearing aids compared against Velocity 24 were labeled A through F. The NRI for adaptive directionality with no other adaptive processes engaged is graphed as a weighted function of degrees azimuth in Figure 4 for hearing aids B through E, in comparison to Velocity 24. The performance of hearing aid A is not included in this graph because it relies solely on automatic switching from omnidirectional to fixed directional mode and has no adaptive directional feature; its behavior is discussed later in this paper. Hearing aid F's performance is also not included in Figure 4 since it was not possible to disengage the noise reduction (NR) feature in that device for these measurements. The performance of hearing aid F is compared to that of Velocity 24, with NR engaged, in a subsequent discussion.

In Figure 4, higher NRI values are better, indicating that the 0-dB SNR at the input of the hearing aid under test has been improved at the output of the device due to the operation of the directional microphone. The first observation to make from Figure 4 is that there are large variations in auto/adaptive directional performance across the various manufacturers. These variations are apparent in the following ways:

1. The time taken for the directional algorithm to begin automatically switching to adaptive directional mode (i.e., the point at which the NRI begins to improve).
2. The time taken for the directional algorithm to reach its optimum polar response (i.e., the steepness, or how quickly the curve rises to its highest NRI).
3. The maximum NRI attained (i.e., where the curve peaks).
4. The total mean NRI achieved while the roving noise is in the "rear hemisphere" (i.e., between 90° and 270°).

Another observation is that all of the adaptive algorithms cease adapting once the noise sweep reaches 270°. From this point on, the noise is moving in the "front hemisphere" toward 360°, the position from where the speech is being reproduced. This is expected behavior for all adaptive directional microphones in this comparison.

**Hearing Aid A.** Hearing aid A was not included in the comparison in Figure 4 because it does not incorporate an adaptive directional algorithm, having only automatic switching to a fixed (hypercardioid) directional polar response. The auto-switching feature, however, exhibits some peculiar behaviors.

In order for the algorithm to engage, the competing noise needed to be present for 40 seconds before it began to switch (Figure 5). This "pre-conditioning" noise was presented from 90° azimuth so switching to a directional response could be easily detected.

Once the response started to switch, a pause in the noise (a characteristic of the measurement process) caused the microphone to reset to an omnidirectional response (Figure 5 shows a 1-second pause). The hearing aid was omnidirectional again for another 15 seconds before switching to a directional response for a second time.

**Polar Sweep Oscillograph Results**
(HAS B - E & V24)

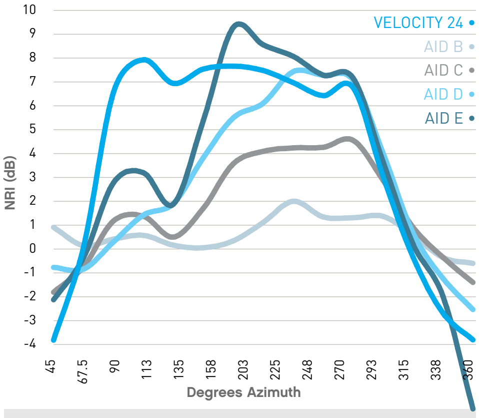

**Figure 4:** Auto/Adaptive directional NRIs versus degrees azimuth for Hearing Aids B – E and Velocity 24 with no NR. Each curve represents results averaged over measurements on two identical hearing aids. The bump in the curves between 90° and 100° azimuth is an artifact of the recording environment.

**Auto-Switching Behavior**

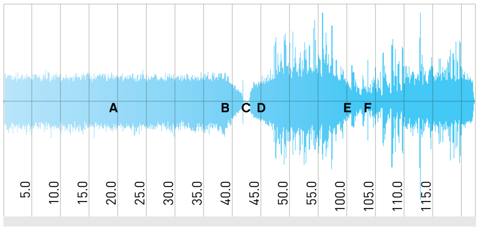

**Figure 5:** Auto-switching behavior of hearing aid A. Switching begins only after 40 seconds of noise presented at 90° azimuth. A pause in the noise causes the algorithm to 'unswitch'. It doesn't switch back to directional until another 15 seconds of noise has elapsed. **A)** Pre-conditioning noise. **B)** Microphone begins to switch from omni to directional at 40 sec. **C)** Alignment click & 1-sec. **D)** Microphone switches back to omni response. **E)** Microphone again switches to directional after additional 15 sec. **F)** Level of speech (from front) is attenuated along with the noise.

The time-waveform in Figure 5 (previous page) shows the noise sweep passing through one null of the fixed hypercardioid response (~ 225°) before sweeping toward the front where the directional microphone no longer has any effect. It is interesting to note that the level of the speech—presented from 0° azimuth—is attenuated along with the noise when the noise reaches the location of the null. The change in speech intensity is apparent in Figure 5 and is audible in the recording. This is a counterintuitive problem: The speech should remain amplified and only the noise should be attenuated by the directional microphone.

**Hearing Aid B.** Hearing aid B doesn't start to adapt until the sweep reaches 180° azimuth, after which it shows little SNR advantage in terms of the NRI, reaching a maximum of 2 dB at 225°. The reason for this may have to do with the split-band adaptive directional algorithm which maintains an omnidirectional response in the low frequencies while adapting the polar response for higher frequencies as the noise sweeps through the azimuth, as demonstrated in Figure 6.

The spectrographs in Figure 6 were created after the arithmetic processing of the NRI to separate the speech and noise energies from the mixed output signal. The level and spectrum of the speech (top tracing) remains unaffected while the noise (bottom tracing) shows that only frequencies above 1 kHz have been attenuated during the sweep. While this is an impressive technological feat, the remaining noise energy below 1 kHz continues to mask the speech energy at these frequencies, reducing the NRI. When worn by a hearing aid user, upward spread of masking from this remaining low-frequency noise energy may compromise directional benefit.

**Hearing Aid C.** Hearing aid C begins adapting earlier than hearing aid B, at about 135° azimuth. However, the NRI does not improve much more than just above 4 dB for the remainder of the adaptation period. This appears to be due to the fact that the null of the polar response tracking the noise during the sweep is not very deep. Therefore, the noise is not attenuated as much as in other hearing aids.

**Hearing Aid D.** As an alternative to a continuously variable polar response, hearing aid D appears to switch among various fixed responses as the noise sweeps around the azimuth. This is evidenced by the stepped morphology of the NRI curve in Figure 4 and can be seen in Figure 7 in the time-waveform of the noise sweep (lower tracing) after NRI processing.

The adaptive directional algorithm analyzes the input signal and changes the polar response in an attempt to reduce the output level to the statistical minimum. This is accomplished by assuming that desired portions of the total mixed signal always come from the front and undesired portions exist somewhere between 90° and 270°. The greatest disadvantage with this method is that the process of analyzing the signal and switching the polar response isn't able to keep up with the dynamic environmental changes. In this laboratory test, the most advantageous SNR isn't achieved until the noise sweep reaches 225° azimuth.

It is also apparent from Figure 7 that there is a secondary gain-changing process taking place that was not accounted for in the linear programming of the hearing aids. When the noise sweeps back to the front of the hearing aid, its intensity is lower than when the noise was at 45° off-axis at the beginning of the recording and the intensity of the speech is lower as well. In fact the mean level of the speech-plus-noise, averaged over recordings from the two hearing aids,

**Spectrographs**

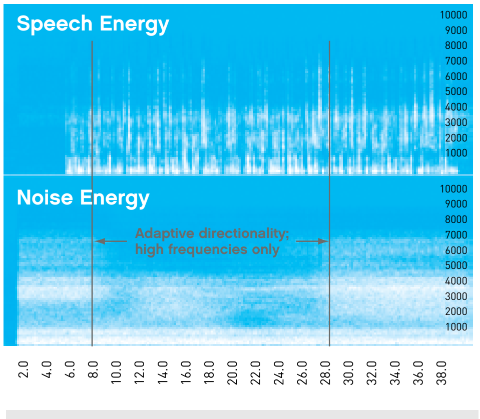

**Figure 6:** Spectrographs of speech and noise energy remaining in output of split-band adaptive directional hearing aid after NRI processing. Only the high frequencies (> 1 kHz) in the noise are attenuated by the directional response while the low frequencies remain unaltered.

**NRI Curve**

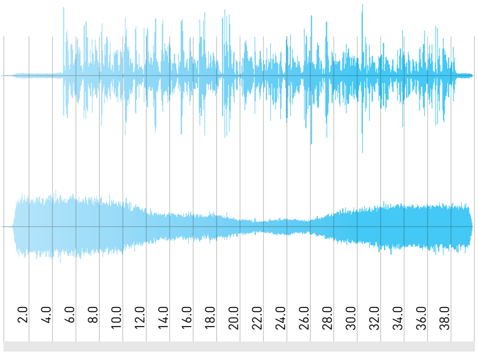

**Figure 7:** Post-NRI separation of mixed speech and noise from hearing aid D showing stepped switching to different polar responses in order to minimize output level.

is about 2.75 dB lower during the last four seconds of the recording than at the first four seconds where speech is present.

**Hearing Aid E.** The adaptive directional algorithm for hearing aid E begins adapting when the noise sweep reaches about 130° azimuth. This is earlier than the other hearing aids previously described. The adaptation rate (slope) is also steeper than for the other, previous hearing aids, and a peak NRI is attained before 180°.

However, there are some serious weaknesses in the adaptive directional processes for this hearing aid. First of all, the algorithm would not initiate directional adaptation at an input SNR of 0 dB; the SNR had to be improved to +3 dB before adaptation would take place. In order to gauge a fair comparison to the other hearing aids, 3 dB should be subtracted from the NRI curve in Figure 4.

The second problem is the observation that the maximum NRI could not be sustained for this hearing aid. Note that the NRI curve in Figure 4 also begins to slope downward after the peak NRI is attained, at about 175° azimuth. This is due to the fact that the intensity of the speech coming from 0° azimuth is attenuated even while the noise is being attenuated in the "rear hemisphere" by the directional microphone, as was the case for hearing aid A. This phenomenon could be clearly heard in the recording and is demonstrated in Figure 8. The speech remains attenuated after the noise

**Velocity 24.** The NRI curve in Figure 4 demonstrates the auto/adaptive directional behavior of Sonic Innovations' Velocity 24 hearing aid. The algorithm switches the polar response from omnidirectional to a dipole directional pattern within seconds of detecting noise in the environment—even while the noise is still in the "front hemisphere." The null at 90° attenuates the noise once the sweep reaches that position and begins tracking it consistently until the sweep reaches 270°, where the polar response remains dipole once again. Velocity 24 attains an NRI of over 7 dB and maintains it fairly consistently while the noise sweep is active in the "rear hemisphere."

**Hearing Aid F Compared to Velocity 24.** Hearing aid F could not be fairly compared to the others in this investigation because its NR feature could not be disengaged. However, it can be compared with Velocity 24 when both pairs are configured the same. Figure 9 shows the NRI calculations on the PSO recordings of hearing aid F and Velocity 24.

**Overall Comparisons.** A single value to compare devices can be calculated by averaging the NRI values while the noise is presented to the rear hemisphere (between 90 and 270 degrees). These values are shown in Figure 10 (next page), including the Velocity 24 with and without noise reduction. Traditional estimates of benefit from fixed directional microphones top-out at 6 dB, which is only slightly less than the maximum seen without noise reduction. This suggests that these results are similar to traditional laboratory measures that over-estimate real-world benefit. The change in SNR as measured by the NRI technique is not expected to transfer directly to a performance benefit for the patient, as speech recognition performance is based upon a multitude of factors that are not consistently used by all listeners. Not all listeners will be able to benefit from SNR changes at all frequencies, and therefore real-world performance will be less consistent and smaller. Instead, this methodology should be thought of as a measure showing differences between systems, which demonstrate that commercially available systems are different. In other words, null steering does not mean the same thing in all systems, and there are measureable differences between

**Mixed Speech And Noise Output**

NOTE: INPUT SNR = +3 DB

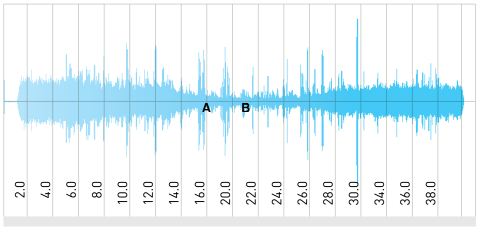

**Figure 8:** Mixed speech and noise at the output of hearing aid E showing attenuation of speech signal after the adaptive directional algorithm has engaged. **A)** Polar response begins to adapt. **B)** Speech becomes attenuated starting at this point.

**Polar Sweep Oscillograph Results**

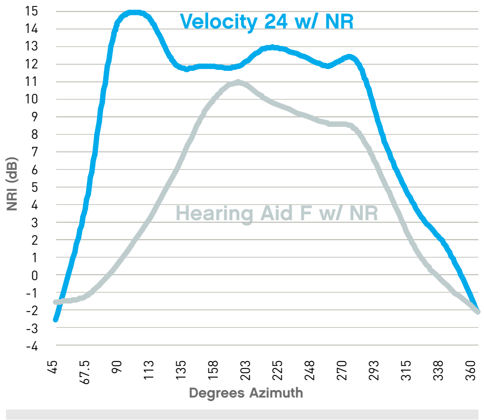

**Figure 9:** PSO results for the two devices with noise reduction enabled.

implementations. The SNR changes should be considered maximum possible benefit, and need to be correlated to measurements with actual listeners to predict real-world benefits. Other variables can be included in the analysis, including the rate at which the masker rotates around the listener, the SNR at the input, or even the bandwidth and frequency content of the stimulus. These are topics for future research.

## References

1. American National Standards Institute. (2004). Method of measurement of performance characteristics of hearing aids under simulated real-ear working conditions [ANSI S3.35-2004]. New York: Acoustical Society of America.
2. Ghent, R., Nilsson, M., and Bray, V. (2007). Validation of the Noise Reduction Index (NRI). Poster presented at 153rd Meeting of the Acoustical Society of America, June 4 – 8, Salt Lake City, UT. [Available as Document Number 5000826 from Sonic Innovations, Inc.]
3. Hagerman, B. and Olofson, A. (2004). A method to measure the effect of noise reduction algorithms using simultaneous speech and noise. Acta Acustica united with Acustica 90, 356 – 361.
4. Nilsson, M., Soli, S.D., and Sullivan, J.A. (1994). Development of the Hearing In Noise Test for the measurement of speech reception thresholds in quiet and in noise. Journal of the Acoustical Society of America 95(2), 1085 – 1099.
5. Nilsson, M. and Bray, V. (2004). The Noise Reduction Index: Bench-top estimate of SNR changes. Poster presented at International Hearing Aid Research Conference, August 25 – 29, Lake Tahoe, CA [Available as Document Number 5000634 from Sonic Innovations, Inc.]

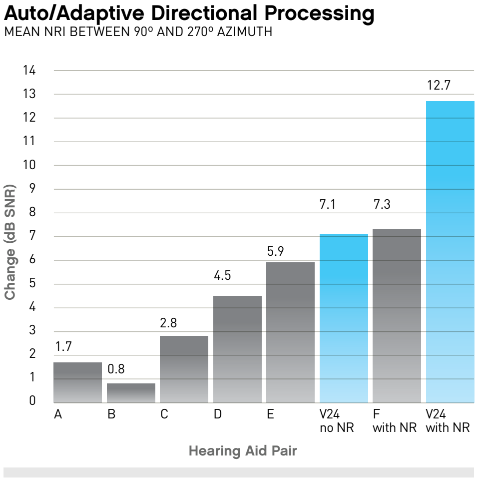

**Figure 10:** Overall comparison using averaged NRI scores.