Advanced Technology for Children: What Works When

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Introduction:

When counseling parents of children with hearing loss, audiologists must provide accurate information as to the benefits and limits of various products and features. Regardless of the funding source (parent, insurance, government etc) the value of the hearing aids must justify the expense with respect to auditory expectations, improvement over prior technology, and overall performance. This article will review the role of advanced signal processing across a multitude of acoustic environments and conditions, and how advanced signal processing can impact the incoming auditory signal.

Three advanced signal processing features (feedback cancellation, automatic directionality and noise reduction) will be reviewed. These features are common across many mid–and-advanced level hearing aids. Implementation of these features varies considerably from manufacturer to manufacturer. In this article, the decision making process employed in Oticon products will be discussed.

Feedback Cancellation

One of the most important digitally-based features is the advanced feedback cancellation algorithm. Previously, acoustic feedback was managed by reducing overall gain or reducing gain in the high frequencies. Rather than reducing gain, advanced feedback cancellation algorithms eliminate feedback by creating a new “cancellation signal” with the same amplitude and frequency content as the feedback, but 180 degrees out of phase. This cancellation signal is added to the original feedback signal to essentially “zero out” the feedback.

Feedback rarely has one stable and recurring feedback spike. Indeed, feedback usually has multiple and variable components that may present as different feedback spikes each time feedback is elicited. In Figures 1-4, KEMAR was fit with a Vigo Pro hearing aid with an open mold. Gain was increased until the device was just below its feedback threshold, at which point feedback was intentionally created by moving a hand near the ear multiple times.

The frequency characteristics of four of those induced feedback signals are depicted in the four panels in Figure 1. Each time the hand was held at a slightly different angle, a new feedback signal was induced and novel spectral components were introduced. Despite vast acoustic variation, each induced feedback signal was identified and cancelled within one second as the feedback cancellation algorithm created and applied unique feedback cancellation signals.

Although feedback cancellation approaches are available from several manufacturers, in certain hearing aids feedback is still controlled through filtering, not cancellation. An examination of the feedback components in Figure 1 demonstrates how difficult it would be to eliminate feedback via filtering. Effective feedback filtering would require multiple deep spectral notches to effectively eliminate all potential feedback components. As a result, large sections of the frequency response would necessarily be sacrificed to minimize the potential for feedback. Fortunately, when employing feedback cancellation, high frequencies are not sacrificed, as demonstrated in Figure 2.

In Figure 2, low level white noise was presented while, once again, a hand was used to initiate feedback. As the device started to feed back, the hand was secured in place. Note that the new feedback component was identified and cancelled within one second. The frequency component of the feedback (visible in the middle panel) was centered at 2600 Hz. In the lower panel, the spectrum of the output of the hearing is shown before (left panel) and after (right panel) cancellation is added to the response of the hearing aid. Note that the hearing aid output is identical before and after feedback cancellation. Filtering was not used, nor was it required, to effectively eliminate acoustic feedback.

For children with hearing loss, high frequency audibility is critical. In devices without true cancellation, gain (particularly in the high frequencies) must frequently be reduced even in the presence of well-sealed earmolds to prevent acoustic feedback. Feedback cancellation algorithms typically permit 10 to 15 dB more gain in the high frequencies. This additional high frequency audibility facilitates improved perception of speech and non-linguistic acoustic cues, such as those necessary for improved spatial perception and awareness.

Automatic Directionality

Automatic directionality is considered by many to be an inappropriate technology for use with an infant or younger child (American Academy of Audiology, 2003). However, at some point in time it does become a technology that should be considered for the older child. (A discussion of the age at which directionality is a good choice for a child is beyond the scope of this paper.) With an automatic algorithm, directionality should only be implemented if a specific set of conditions is met. Having a good understanding of the criteria used in deciding whether directionality should be implemented can help in determining at what age this feature should be enabled.

Directionality has been shown to improve speech understanding in noise in particular environments. Walden and colleagues (2004) identified conditions in which directionality was deemed useful and desirable by adult hearing aid users:

- Speech originates in front of the listener and is close by
- Noise is present and originates from the back or side of the listener
- Reverberation is not excessive

When these above conditions above were met, users judged directionality to be valuable. When these conditions were not met, users preferred omni-directional settings. The principle lesson learned from this research is that directionality should only be used when it improves speech understanding in noise. If it does not help, it should not be used.

Although Walden and colleagues (2004) used manually activated directionality systems, most commercially available hearing aid systems (2008) use automatic switching. Automatic systems employ a “decision making” function built into the hearing aid circuitry to decide when directionality should be active. These decision making functions vary enormously from manufacturer to manufacturer. The simplest systems evaluate the overall input level. As amplitude increases, directionality is activated. However, use of such a simple metric in isolation will not provide directionality consistent with Walden’s three conditions, nor will it assure the activated directionality provides benefit.

For example, consider the child working in a small group activity in the classroom. There are several such groups throughout the room and the overall noise level is high. Using overall input level alone as the criterion would likely result in the hearing instruments being in directional mode in this environment. However, this may have a negative consequence if the teacher is circulating around the room and speaking to the children from behind. Additional criteria are needed to ensure that directionality is used only when it is likely to improve speech intelligibility.

A sophisticated “decision making” approach is implemented in many Oticon products. The decision to switch among multiple directional modes is driven by the modulation characteristics (i.e., intensity change over time) of the combined speech-plus-noise signal. A more highly modulated signal is an indication of a better signal-to-noise ratio (SNR). Thus, in the Oticon directional system, the directional mode that provides the better SNR is selected, as shown in Figure 3.

In Figure 3, the upper panel shows the waveform of speech in quiet (left) and in noise (right). Speech in quiet has a modulation pattern with dramatic intensity changes occurring between 3 and 10 times per second. The modulation pattern of the noise changes over time, but in a manner different from speech. The intensity changes are not as dramatic because when the speech is absent, the background noise continues. Thus, the modulation pattern of speech-in-noise is quite different from speech-in-quiet.

In the bottom panel of Figure 3, the two signals (speech and noise) are mixed together at two different signal-to-noise ratios. The left panel has the greater SNR, and as such, the modulation pattern is more evident than in the right panel. In this example, if the left panel outcome is obtained when in a directional mode, than that mode would be selected. If the modulation pattern deteriorates when in directional mode, than the hearing instrument would remain in omni-directional mode. This would be the case even if the overall signal level was reduced when in directional mode.

Walden and colleagues (2004) further noted directionality loses effectiveness when significant reverberation is present. Figure 4 confirms directionality can be effective in non-reverberant conditions. In this example, speech was presented 2 meters in front of KEMAR's in-situ microphone, speech-shaped noise was presented 2 meters behind. The upper and lower panels show directionality active and directionality inactive, respectively. Note in the upper panel, the modulation is a bit more pronounced, consistent with a better SNR. For this reason, directionality would be chosen as the appropriate setting in this situation.

Figure 4 (bottom) demonstrates the histogram of these same two waveforms. A histogram represents quantity of occurrences per interval (i.e., loudness per frequency). When directionality was active, the distribution shifted approximately 4 dB to the left (quieter). Because directionality has no impact on sounds coming directly from the front (0 degrees azimuth) the overall loudness reduction (4 dB shift to the left) appears to be a measure of reduced speech-shaped noise originating from the rear. In this case, since the noise is coming from the rear, directionality leads to a lower overall signal level and one in which the speech modulation characteristics are more apparent.

Figure 5 demonstrates recordings of speech (from the front) and noise (from the rear) one meter from KEMAR in a highly reverberant room. Reverberation has the effect of eliminating the directional acoustic cues of sound. In highly reverberant rooms, direct sound mixes with reflections from all angles. Thus, the directional system no longer identifies a directional difference, as evidenced on histogram. Oticon's outcome-based directionality system assures activation of the directional system only when an improved SNR results.

Unfortunately, the expected benefit from directionality is often over estimated. Distance, loudness levels and reverberation impact the effectiveness of this technology. Understanding the impact of these situational factors is key to a realistic understanding of the potential benefits and limits of this technology. For example, a child should not expect to see dramatic improvements in speech understanding when using directionality in a cafeteria with very poor acoustic characteristics and high reverberation.

Noise Reduction

Another often misunderstood technology is noise reduction. Some believe noise reduction is designed to eliminate noise while allowing for amplification of speech. However, that is not the operating principle or the effect of noise reduction (Schum, 2001, 2003). Simply stated, noise reduction reduces the loudness (and often the annoyance) of some steady state competition. Gain reductions to the noise component (at a given frequency) will impact speech in the same frequency region. This potential for reduction of speech components has led to concerns about using noise reduction with younger children (American Academy of Audiology, 2003). Any noise reduction algorithm employed with children should be conservative in nature and focus first on providing the user with as much speech information as possible.

One primary challenge for noise reduction algorithms is to differentiate speech from noise. Since “noise” often includes competing speech, the task of the noise reduction algorithm is further complicated by differentiating between “good” speech and “bad” speech. Speech has characteristic modulation patterns. Amplitude modulation patterns are used to partially drive the analysis function of nearly all noise reduction systems in hearing aids. For example, when the signal changes amplitude in a manner consistent with speech, no noise reduction is applied. If the modulation pattern appears to be different than speech, gain reductions are applied.

Figure 6 shows the waveform of clean speech, speech-shaped noise and noise recorded in five different realistic situations. Notice the modulation patterns in the realistic situations demonstrate some speech-like amplitude fluctuations over time. Due to this similarity to speech, noise reduction is less likely to be engaged in acoustic environments that include competing speech.

Figure 7A shows active noise reduction in Vigo Pro for speech in a background of speech-shaped noise that contains more low frequency energy than found in a typical speech signal. Amplitude reduction of 4 to 8 dB are apparent in bands above and below the main speech frequencies (approximately 400 to 4000 Hz.) while protection of the most important speech information is apparent at remaining frequencies. Because the noise level in the lower frequencies is much more intense than the target speech signal, gain reduction can be applied without impacting speech intelligibility.

Figure 7B shows similar recordings obtained when speech is presented in a background of playground noise. In this example, the modulation characteristics of the multiple voices in the background noise were similar enough to speech (in quiet) that no effect from noise reduction is noted. Advanced technology can be implemented using a conservative criterion to minimize the potential for a reduction in speech intelligibility while still providing increased listening comfort in specific environment.

Another consideration to help protect the speech signal is to make the noise reduction algorithm loudness level dependent. For example, as the overall input level increases, the greater the potential amount of gain reduction which may be applied. Loudness level dependency helps ensure softer level speech is not unintentionally made inaudible.

Figure 8 demonstrates the combined effects of Oticon’s speech protection mechanisms with regard to our noise reduction systems and loudness level dependency across four conditions (note - the more the bar extends downward, the more gain reduction is applied). In the upper left panel, the input was 80 dB of speech-shaped noise without speech, resulting in approximately 10 dB (or more) gain reduction. However, when the input level was dropped to 50 dB SPL (lower left panel) much less noise reduction was applied. On the right, speech-shaped noise was mixed with speech. However, even at an overall level of 80 dB SPL (upper right panel) the noise reduction effect was modest. Further, noise reduction was non-existent at the lower input level (lower right panel).

Final Remarks:

Parents and audiologists want the very best amplification systems for children with hearing impairment. Therefore, it becomes important to understand how advanced technologies work to maximize speech information. Unfortunately, terminology such as “noise reduction” and “directionality” tend to transmit hard and fast images and incorrect assumptions about how the technology works and how these features impact speech can arise.

This article addressed three advanced signal processing features (feedback cancellation, automatic directionality and noise reduction) and how these features actually function in Oticon products. Technologies discussed in this paper can provide significant benefits for children (and adults) in many challenging acoustic environments.

References:

American Academy of Audiology (2003): Pediatric Amplification Protocol
October, 2003

Schum, D. (2001). Annoyance and hearing aids. Audiology Online, January 17, 2001. http://www.audiologyonline.com/audiology/newroot/articles/arc_disp.asp?id=247&catid=1

Schum, D. (2003). Noise-reduction circuitry in hearing aids: (2) Goals and current strategies. The Hearing Journal., *56*(6), 32-41.

Walden BE, Surr RK, Cord MT, Dyrlund O. Predicting hearing aid microphone preference in everyday listening. *J Am Acad Audiol* 2004;15(5):365-96.

Figure 1:

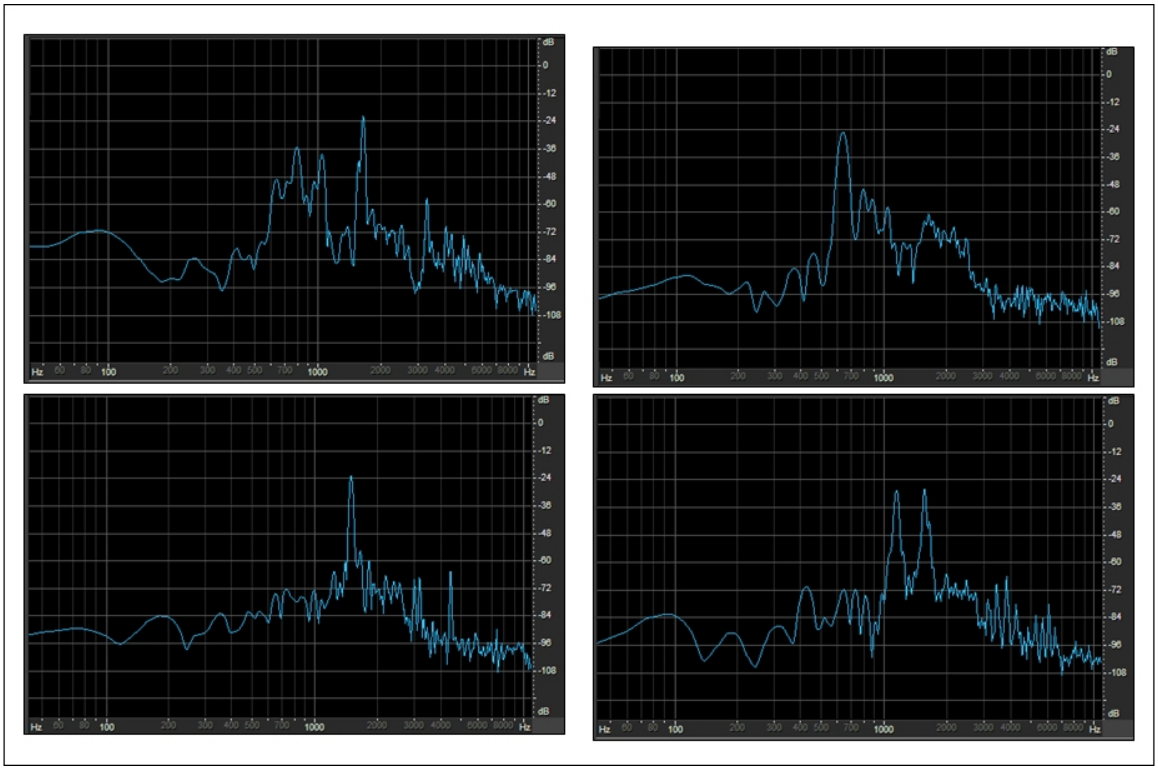

dB
0
-12
-24
-36
-48
-60
-72
-84
-96
-108
dB
Hz
60
80
100
200
300
400
500
700
1000
2000
3000
4000
6000
8000
Hz

Figure 2:

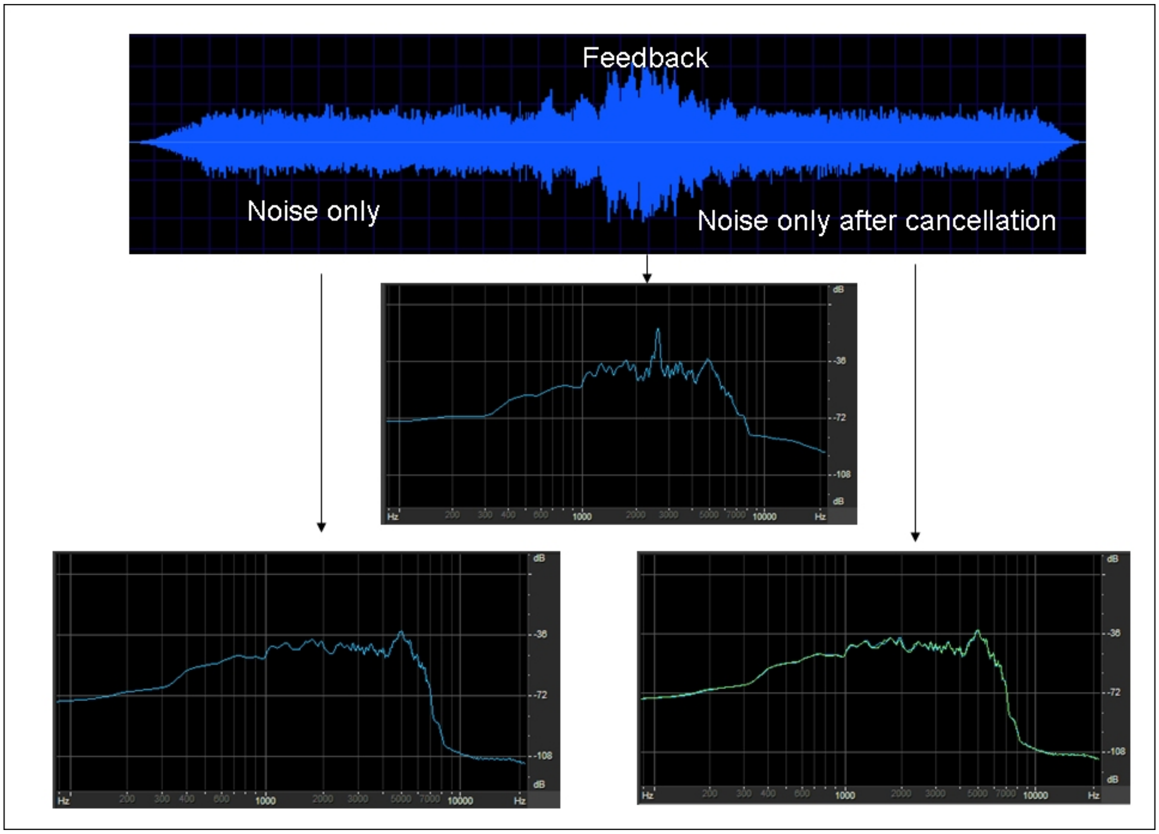

Feedback
Noise only
Noise only after cancellation
dB
-36
-72
-108
dB
Hz
200
300
400
600
1000
2000
3000
5000
7000
10000
Hz

Figure 3:

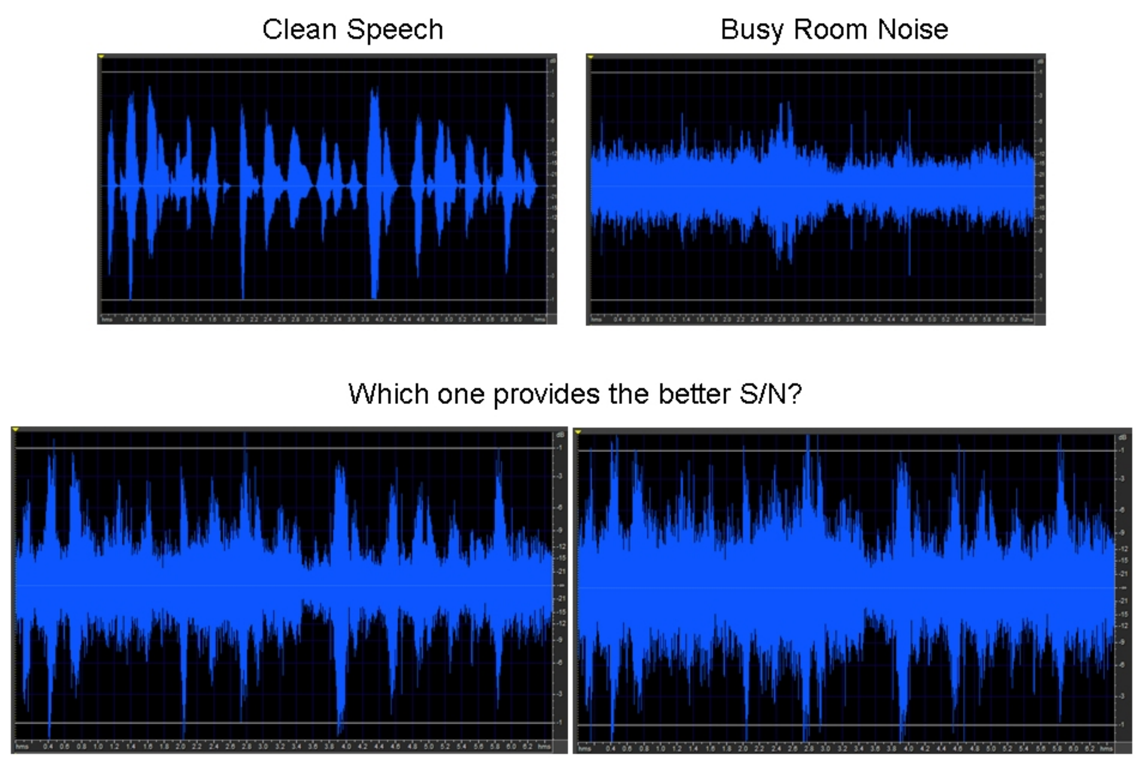

Clean Speech
Busy Room Noise
Which one provides the better S/N?

Figure 4:

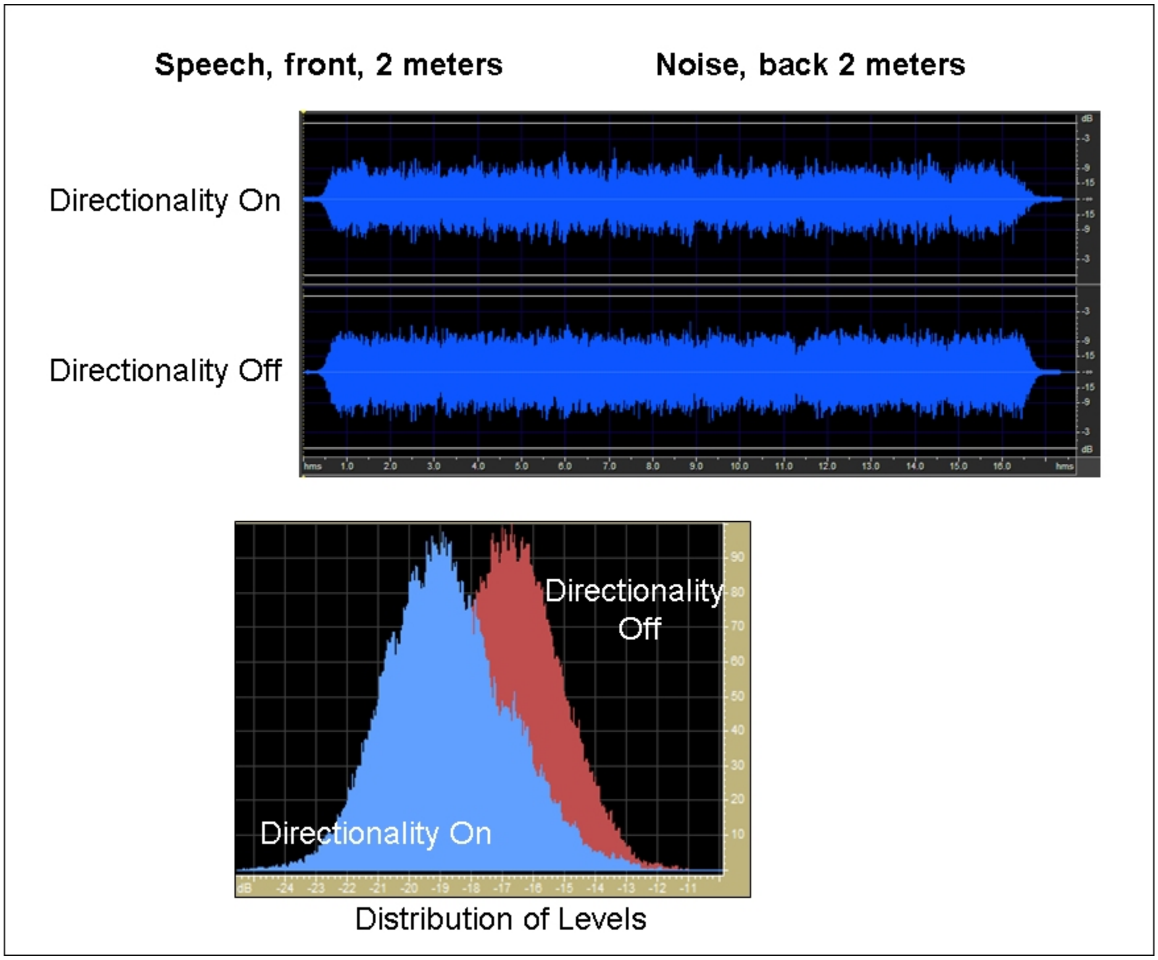

Speech, front, 2 meters
Noise, back 2 meters
Directionality On
Directionality Off
dB
-3
-9
-15
-15
-9
-3
hms
1.0
2.0
3.0
4.0
5.0
6.0
7.0
8.0
9.0
10.0
11.0
12.0
13.0
14.0
15.0
16.0
hms
Directionality Off
Directionality On
90
80
70
60
50
40
30
20
10
dB
-24
-23
-22
-21
-20
-19
-18
-17
-16
-15
-14
-13
-12
-11
Distribution of Levels

Figure 5:

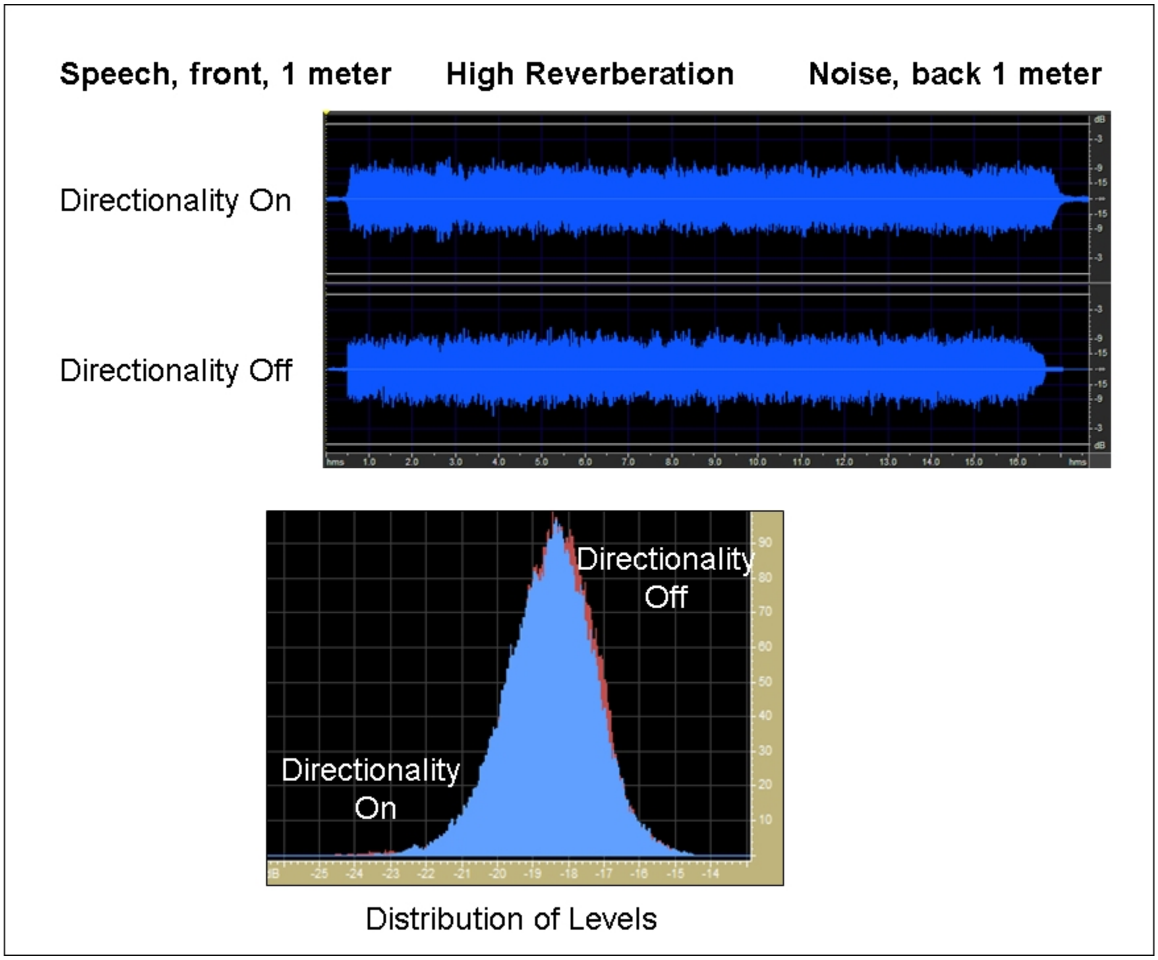
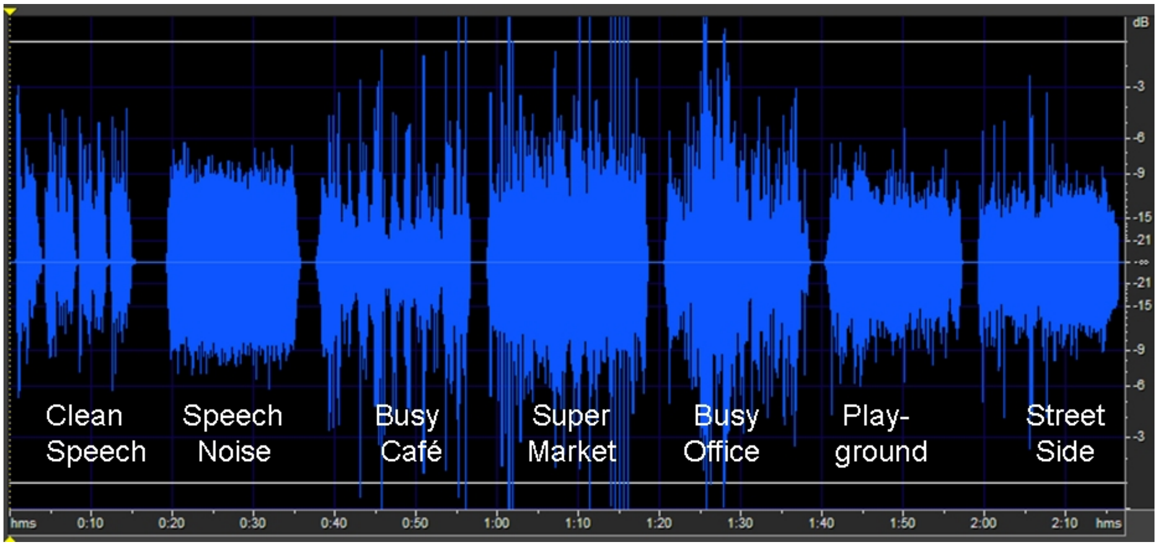

Speech, front, 1 meter
High Reverberation
Noise, back 1 meter
Directionality On
Directionality Off
Directionality Off
Directionality On
Distribution of Levels

Figure 6:



Clean Speech
Speech Noise
Busy Café
Super Market
Busy Office
Play-ground
Street Side
dB
-3
-6
-9
-15
-21
-∞
-21
-15
-9
-6
-3
hms
0:10
0:20
0:30
0:40
0:50
1:00
1:10
1:20
1:30
1:40
1:50
2:00
2:10
hms

Figure 7A:

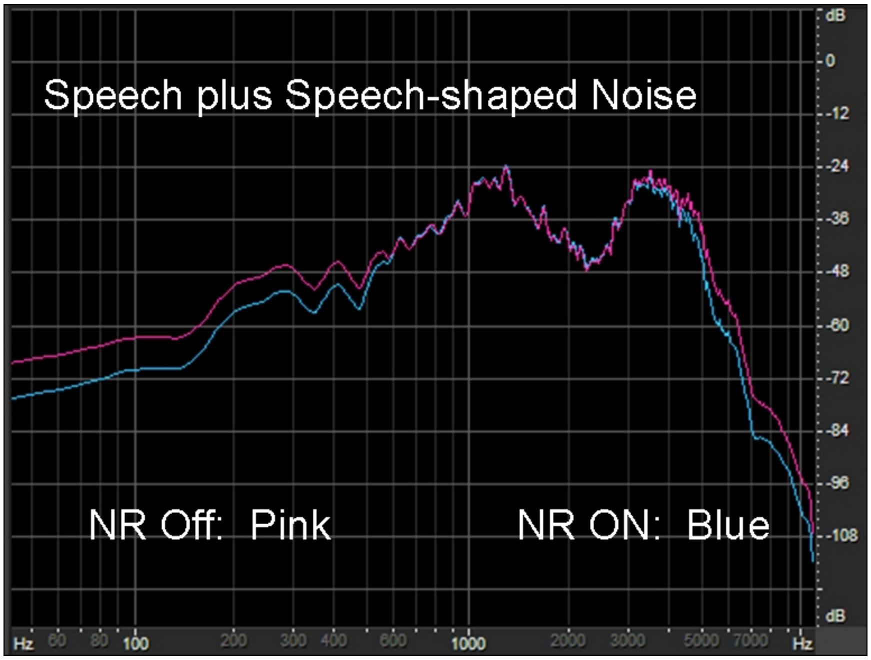


Figure 7B:

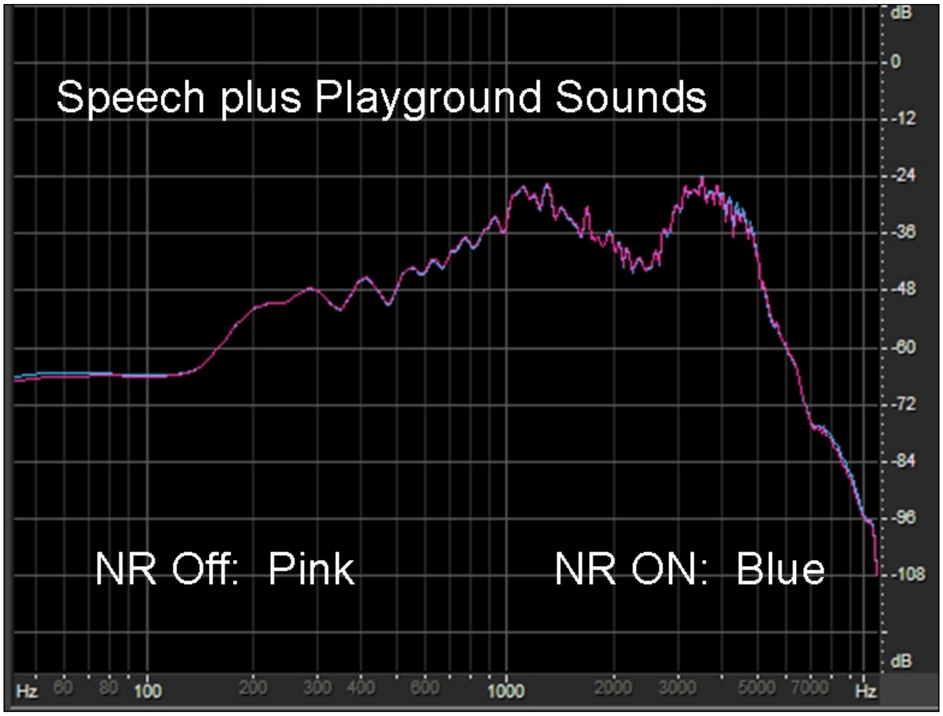
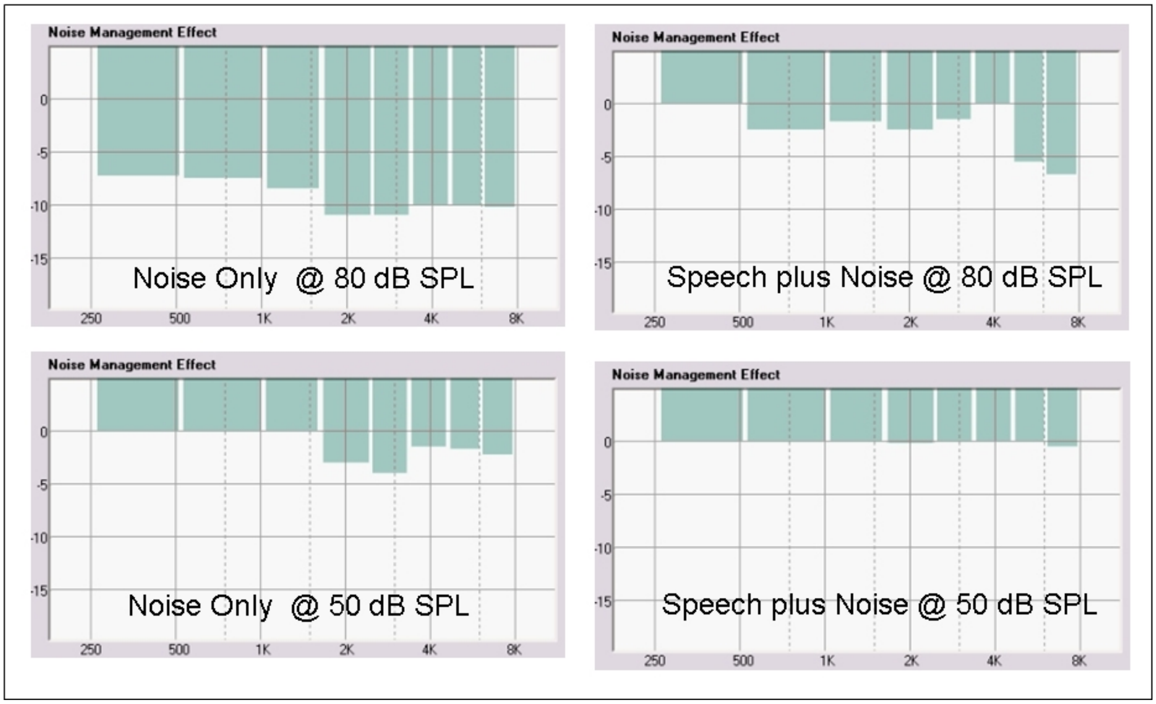


Figure 8:


Noise Management Effect
Noise Only @ 80 dB SPL
Noise Management Effect
Speech plus Noise @ 80 dB SPL
Noise Management Effect
Noise Only @ 50 dB SPL
Noise Management Effect
Speech plus Noise @ 50 dB SPL
0
-5
-10
-15
250
500
1K
2K
4K
8K