ReSound NoiseTracker™ II

Abstract

The main complaint of hearing instrument wearers continues to be hearing in noise. While directional microphone technology exploits spatial separation of signal from noise to improve listening in noise, a challenge for single microphone noise reduction systems is that the signal of interest is embedded in the background noise. ReSound NoiseTracker™ II uses a sophisticated algorithm to reduce amplification of noise for both single and dual microphone devices without impacting speech understanding. It accurately identifies speech and characterizes noise, operating seamlessly to reduce noise even during the pauses in running speech. In the clinic, ReSound NoiseTracker™ II can be customized for different listening preferences and/or listening situations.

Unwanted amplification of background noise is a significant factor influencing hearing instrument user satisfaction (Kochkin, 2000). Just as directional microphone technology has risen enormously in popularity as a means to combat hearing-in-noise issues, noise reduction algorithms have become an ever-present feature in digital hearing instruments.

This paper reviews the standard methods of single microphone noise reduction and describes the ReSound NoiseTracker II system, an important component of the surround sound by ReSound. NoiseTracker II is unique in its ability to reduce amplification of noise without affecting the audibility or quality of simultaneously occurring speech. This provides not only increased listening comfort but also sound quality.

Background

Even though hearing instrument users of all ages complain about background noise, it can be an especially problematic issue for older adults, who comprise the largest segment of users. The average hearing instrument user is 70 years old (Kochkin 2005). In addition to hearing loss, many older adults often experience problems with auditory processing and cognitive function, which can result in difficulty focusing, remembering or processing information they have heard (Wingfield 2005). Studies have shown that even when younger and older adults have similar hearing thresholds and speech discrimination ability in quiet, older adults will perform worse on speech discrimination tasks in noise than their younger counterparts (Kricos 2006). In addition, Tun et al (2002) suggested that higher level cognitive factors may affect the ability of older adults to process language in the presence of competing speech. In terms of hearing instrument use, understanding speech in noise is likely to be particularly challenging for the older adult, even if audibility is provided. This in turn impacts success with amplification.

Apart from issues related to processing speech in noise, other noise-related factors can influence usage and satisfaction with amplification. Nabelek and her colleagues proposed that hearing instrument usage might be determined more by wearers' willingness to listen in background noise than how well they understand speech (Nabelek et al, 1991). Using the acceptable noise level (ANL) test with elderly listeners, they found that full-time hearing instrument users accepted more background noise than did part-time users or users who had rejected their hearing instruments (Nabelek et al, 2006). These findings would suggest that eliminating or reducing background noise when listening with hearing instruments would increase the acceptance rate of hearing instruments and provide significant benefit to a hearing instrument user.

Perhaps related to the willingness to listen in noise, listening effort in background noise also has far-reaching impact on success with hearing instruments. Consider, for example, the hearing instrument wearer who is attempting to listen to speech in a noisy environment over a prolonged period of time. The concentration required to follow what is being said can be an exhausting task. Conceivably, less effort would be required by the listener if the background were less noisy, even if overall understanding of speech was not improved. Less effort required of the hearing instrument user may even allow them to perform dual attention tasks that normal listeners typically take for granted, such as listening to a talker while also paying attention to their surroundings, simultaneously noting other conversations that may be going on around them, or monitoring the activity of other people in the room. Simply eliminating a distracting or annoying sound, such as a nearby air conditioner could, in other words, provide significant benefit to a hearing instrument user because of the improved sound quality.

Noise reduction implementations

Noise reduction and dual microphone directionality in hearing instruments were both conceived as a means to improve speech understanding in noise. Directionality achieves this goal by preferentially

amplifying signals coming from a particular location, such as in the look direction of the hearing instrument user. For noise reduction to be effective in improving speech understanding in noise, it would have to separate desired speech from the competing signals after both have been picked up by the same microphone. Even then, the hearing instrument would need to be capable of reducing amplification of only the noise components of the incoming sound without disturbing the speech components. This is a daunting task, as both speech and what constitutes "noise" for the listener are similar in terms of frequency content. In fact, the undesired noise often is speech from multiple talkers. Not surprisingly, the benefit of noise reduction has been noted more in its ability to improve listening comfort and effort than for improving speech understanding (Mueller et al, 2006; Bentler et al, 2008).

Prior to digital hearing instruments, noise reduction systems limited low frequency amplification by means of a high pass filter or by means of different compression strategies. The rationale behind this approach was that noise – including multitalker babble - contains more energy at low frequencies than the speech of a single talker, which could be expected to contribute significantly to excessive overall loudness of the signal. Also, low frequency noise may cause upward spread of masking, thereby making the high frequency parts of speech inaudible.

The advent of digital technology enabled more advanced means of determining the composition of the input signal. A better characterization of the sound entering the hearing instrument microphone could make it possible to limit gain reduction to periods of time and spectral regions where important speech information was occurring. The first digital noise reduction algorithms to appear

were based on signal modulation and are in fact still the most common type of system today. Modulation-based noise reduction systems analyze the level fluctuations of the input signal in different frequency bands. Because single-talker speech can fluctuate in level more than 30dB, the assumption is made that large fluctuations in the input sound indicate good signal-to-noise (SNR) ratios, and that SNR progressively worsens as the level fluctuations decrease. If the SNR is estimated to be poor in a particular frequency band, gain is reduced in that band.

Although the underlying principle is the same, modulation-based systems differ in a number of important ways. One obvious difference is the amount of gain reduction. Some systems may allow more than 10dB reduction in gain while the maximum reduction for others can be 5dB or less. Another difference is that some algorithms may consider other acoustic properties of the input than just modulation in determining the amount of gain reduction. For example, they may reduce the gain differentially depending on the rate of modulation or on the overall input levels. Finally, systems differ in how long they take to decrease gain or restore it back to its original value in a given channel. As in describing the dynamics of compression systems, these are collectively called the time constants. Systems which reduce and restore the gain slowly can make listening in situations with stationary noise sources more comfortable, but may degrade the audibilility of speech. For this reason, most systems incorporate faster release times so that the gain is restored quickly when a highly modulated signal such as speech is detected. Systems which also have fast attack times may negatively affect the overall sound quality as gain is rapidly increased and decreased.
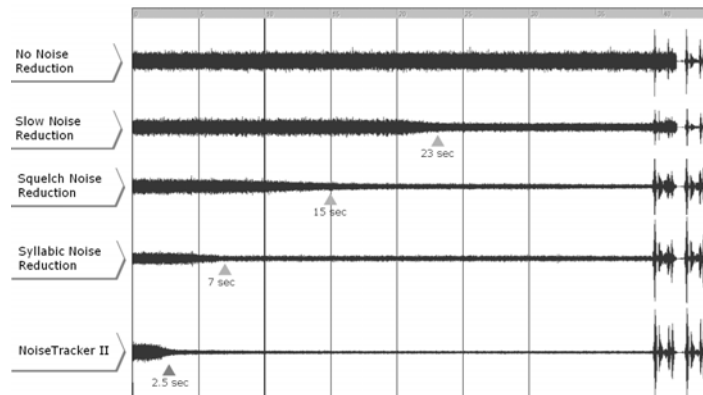

Figure 1. Steady-state noise is treated differently by different noise reduction systems.
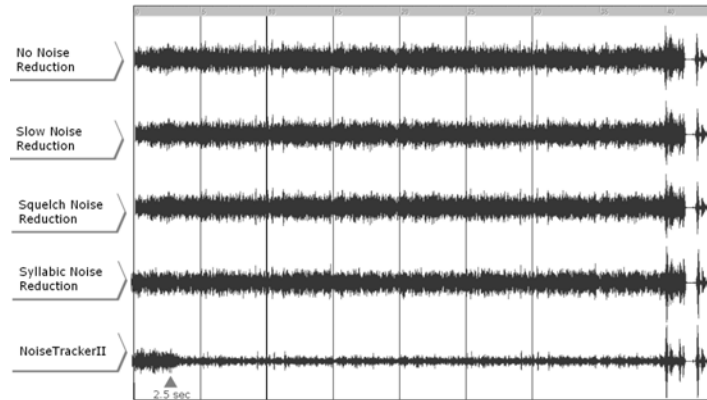
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Figure 2. For babble noise, modulation-based noise reduction systems do not affect the background noise. NoiseTracker™ II is different than traditional modulation-based systems as it is able to reduce the level of the babble while preserving the level of speech.

As a result of such differences, modulation-based noise reduction systems also differ greatly in how various types of input sounds are treated (Bentler & Chiou, 2006). An example of this is illustrated in Figure 1 which shows how different types of system react to steady-state noise with a short speech passage at the end of the timeframe. Figure 2 shows the same systems with a babble background noise and the same short speech passage at the end of the timeframe. There are clear differences in both reaction time as well as amount of gain reduction for the same input. While no evidence exists supporting the superiority of any particular system, there is one apparent shortcoming of the modulation-based approach. This is the inability of such a system to accurately identify when speech is present. As a result, modulation-based systems are more likely to make "mistakes" in terms of when and in what frequencies gain reduction would be beneficial.

NoiseTracker™ II noise reduction

Like other noise reduction schemes, NoiseTracker II has the goal of suppressing noise in frequency regions where the speech-to-noise ratio is low. However, NoiseTracker II is distinguished from the modulation-based method by its ability to reduce unwanted noise from the incoming signal without appreciably affecting the speech portion of the signal. The NoiseTracker II system is able to accomplish this because of 1) the higher degree of accuracy with which it identifies speech and noise when compared to other systems, 2) an adaptive noise estimate, 3) fast time constants, and 4) a mathematically optimal gain reduction function based on SNR rather than signal modulation.

Built on the ReSound Warp-based platform, the NoiseTracker II system uses spectral subtraction (Boll, 1979), one of the most widely used methods for enhancement of noisy speech in audio applications. The concept of spectral subtraction, illustrated in Figure 3, is to subtract the short-term noise spectrum from the total signal, leaving only the speech portion. Although the concept is easy, the implementation is not. The success of this strategy hinges on being able to identify speech and to precisely characterize noise. An additional challenge is to keep up with the dynamic speech and noise make-up of real listening environments. Finally, it is important for hearing instrument users that not all noise be removed from the signal, and that the noise characteristics be preserved. If all ambient noise were removed or if the spectrum of the noise background was altered, this would create an unnatural-sounding experience. Background sounds do need to be audible to the degree that users can recognize and orient themselves in their listening environments. Ultimately, the goal is undistorted speech at the prescribed gain, and undistorted noise at lower gain.
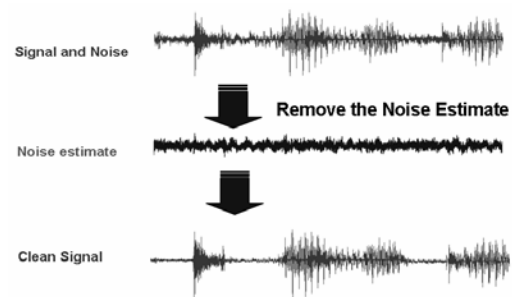

Figure 3. Spectral subtraction removes noise from the total signal, leaving the desired signal intact.

The NoiseTracker II signal analyzer itself is comprised of three main components: a signal power tracker, a speech presence indicator, and a noise power tracker. These three components provide information that is used to estimate the signal-to-noise ratio (SNR), which in turn

determines the amount of gain reduction to be applied.

The signal power tracker represents the overall signal including speech and noise, and is the part from which any noise will ultimately be subtracted. The speech presence indicator analyzes the acoustic characteristics of the signal at 1-millisecond intervals to determine the probability that speech is present in the overall signal. More specifically, the speech presence indicator looks for a temporal-spectral pattern of alternating high and low frequency sound which is typical of speech. This method constitutes a more precise way of identifying speech than relying on modulation alone. With such an accurate identification of speech, the noise power tracker is able to restrict analysis of the noise background to frequency regions and points in time where speech is not mixed with noise. It is critical for the system that the noise estimate not be contaminated by speech, since this would lead to distortion of the speech envelope.

Once the overall signal and noise information have been provided by the speech presence indicator and the noise power tracker, the SNR ratio is estimated by comparing the level of the noise with the level of the overall signal. When only noise is present in the total input signal the difference between the total input signal and the estimated noise will be small, as will the SNR estimate. Conversely, the SNR estimate will be greater when the total input signal consists of both speech and noise. Depending on the estimated SNR and the user-configurable NoiseTracker II level setting in the Aventa software, the gain may be reduced as illustrated in Figure 4. The gain reduction functions are mathematically derived using Wiener optimal filter theory. When noise and speech are present simultaneously, the most recent noise estimate is subtracted from the signal.
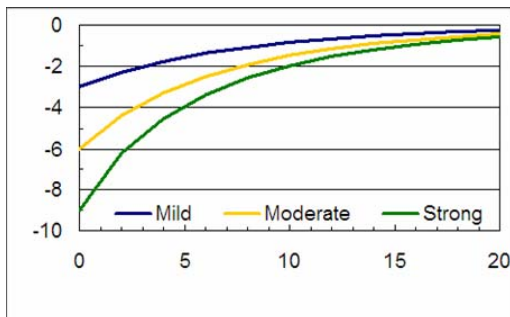

Figure 4. The NoiseTracker gain reduction function is based on the signal-to-noise ratio (SNR).

The time constants of the NoiseTracker II system are crucial for its performance. This system employs time constants for each of its tracking components as well as for the actual gain reduction and restoration. While the signal power tracker always works quickly in order to preserve the speech envelope, the noise power tracker adaptively adjusts its time constants depending on whether speech is detected. Estimation of the noise spectrum is thus limited to pauses between words and syllables. In this way, the system avoids mistaking speech for noise, and prevents speech information being subtracted from the overall input. Once noise is detected, the time it actually takes for the decrease in gain to begin is within 2 seconds. As the SNR improves or decreases further, new gain calculations are effected almost instantaneously.

Depending on the product family, the NoiseTracker II system offers flexibility in the degree of noise reduction it offers to address individual user preferences. Up to four options are available: mild (-3dB), moderate (-6dB), considerable (-8dB) or strong (-10dB) noise reduction. The noise reduction value for each degree of NoiseTracker II is the amount which would be applied when the estimated SNR is 0dB or worse. A lesser amount of gain reduction would be applied for better SNRs as illustrated in Figure 4. Some users prefer the strong NoiseTracker II setting while others prefer being able to hear more of the environmental noise. Preferences regarding degree of noise reduction may also differ from situation to situation, which is the rationale for the different NoiseTracker II settings in many of the environmental programs. For example, the "Traffic" program is set to apply a strong level of noise reduction, as it is assumed that maximum listening comfort would be desired in this situation. All degrees of NoiseTracker II can be applied without affecting speech intelligibility negatively.

Summary

ReSound NoiseTracker II overcomes the limitations of modulation-based noise reduction systems. Because of the accuracy with which is identifies speech and noise, it excels in its ability to reduce unwanted noise from the incoming signal without affecting audibility of speech or sound quality. With NoiseTracker II, the ReSound hearing instrument wearer can hear desired sounds while noise is kept at comfortable levels to allow a natural listening experience.

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