ENHANCING SOUND QUALITY AND LISTENING COMFORT WITH RESOUND NOISE REDUCTION

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Background noise is a significant factor influencing user satisfaction with hearing instruments (Kochkin, 1992, 1993, 2000). The detrimental effect of background noise on speech understanding is well-documented (Killion, 1997a, 1997b, 1997c; Moore, 1995; Peters, Moore & Baer, 1998; and Plomp, 1986), as are specific hearing instrument signal processing techniques designed to help overcome this complaint (Agnew, 2000; Kochkin & Kuk, 1997; Schum 1996; Agnew & Block, 1997; Agnew, 1999; Mueller & Ricketts, 2000; Preves, 1997). Speech understanding, however, is not the only problem experienced by hearing instrument users when listening in background noise. Comfort and listening effort along with sound quality must also be considered. Think about, for example, the hearing instrument user 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. Less effort would be required by the listener if the background were less noisy, even if overall understanding of speech were not improved. Less effort required of hearing instrument users 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.

Since hearing instrument satisfaction is not only determined by speech understanding in background noise, Nabelek and colleagues (1991) were motivated to test hearing instrument users' tolerance to background noise. In this study, the acceptable noise level (ANL) was measured for three groups of elderly hearing instrument users. The (ANL) is the difference between the most comfortable listening level for speech and the highest background noise level that is acceptable when listening to and following a story. The ANL measure assumes that speech understanding in noise may not be as important as is the willingness of a hearing instrument user to listen in the presence of background noise. Findings of this study showed that full-time hearing instrument users accepted more background noise than did part-time users or users who had rejected their hearing instruments. This study was replicated in 1994 by Lytle. In this study, successful and unsuccessful hearing instruments users were matched for lifestyle, age, hearing sensitivity, and speech understanding scores. Successful users accepted more background noise than unsuccessful users. 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 a significant benefit to a hearing instrument user.

While dual microphone directionality has been the primary technique used in digital hearing instruments to improve speech understanding, single microphone noise reduction techniques have been used to improve listening comfort and effort. The noise reduction in the Danalogic 6 represents a significant improvement over other noise reduction techniques in meeting this objective. Noise reduction is the first true spectral subtraction method used for noise reduction in a hearing instrument. The goals of the noise reduction are two fold: 1) to reduce noise during the presence of speech; and 2) to keep the overall sound quality of the signal good. This paper will describe the ReSound noise reduction algorithm and the advantages it offers in terms of increased listening comfort and effort without impacting sound quality.

NOISE REDUCTION IMPLEMENTATIONS

Separating the desired speech from competing noise after they have both been picked up by the same microphone is extremely difficult. Single microphone noise reduction systems face the seemingly unsolvable dilemma of how to separate speech from noise when both signals usually have energy at the same frequencies at the same time. A variety of approaches have been used to help with this dilemma over the years.

Early versions of noise reduction systems aimed at reducing noise in the low-frequencies either by means of high pass filters or various compression strategies. These methods, although not found to be effective at improving speech understanding (Fabry and Van Tasell, 1990) were based on the principle that noise has a greater low-frequency component than speech. The low-frequency component of noise contributes most to the loudness of the noise, and constitutes more of a problem at high levels than at lower levels. In addition, the low frequency noise may cause upward spread of masking, thereby masking the high frequency parts of speech. Digital hearing instruments have seen the introduction of more complicated strategies that are capable of reducing noise across all frequencies.

Multi-channel digital hearing instruments use noise reduction strategies that decrease noise in any frequency region where the signal-to-noise ratio is estimated to be poor. An estimate of the signal-to-noise ratio within each channel is achieved by taking advantage of the fluctuations in level that are characteristic of speech, in comparison to the more constant level that is characteristic of many background noises. A speech detection algorithm analyses the envelope in each channel. High-envelope levels are assumed to indicate that a speech signal is present. Sustained low-envelope levels are assumed to indicate absence of speech. When there is high confidence that no speech is present in the signal, the background noise levels can be reliably measured. The detector combines these estimates of signal level and noise level to estimate a modulation-index in each channel. A perceptually appropriate gain for each channel can then be calculated on the basis of these modulation-index levels. This type of approach is called modulation-based noise reduction, and works well when the wanted signal is a single talker in the presence of steady noises, like a fan or motor. However, this type of system doesn't work as well when the competing noise consists of many people talking.

Another drawback to modulation-based systems is related to the dynamic characteristics. The time constants specified for the system to begin reducing gain when noise is identified and to restore gain when the estimated signal-to-noise ratio improves are fixed in existing systems. These time constants necessarily represent a compromise in providing listening comfort without degrading speech intelligibility or sound quality. Slow time constants may reduce audibility of speech as the listening environment changes from predominantly noise to speech. Fast time constants may cause disturbing perceptual effects as the gain is rapidly decreased and increased. As a general rule, slow time-constant systems make listening in noise quite comfortable, but they do not try to reduce noise during speech segments. Noise is reduced only during noise-only periods. In fast time-constant systems, noise is reduced at all times, also during speech. Unfortunately, strong levels of fast time constant systems introduce artifacts that diminish the overall sound quality of the signal.

NOISE REDUCTION

Like the modulation-based noise reduction systems just described, the basic goal of the noise reduction is to reduce gain in frequency areas where little or no speech signal is detected. However, the similarity ends there. The noise reduction differs in 3 significant ways, each of which serves to maintain listening comfort and good sound quality without removing important speech information. These distinguishing aspects include: 1) more accurate identification of speech, 2) precise characterisation of noise, and 3) adaptive time constants to maintain noise reduction without affecting sound quality. Together, these differences set noise reduction apart as a system capable of true spectral subtraction, such that the spectral content of noise can effectively be diminished and the envelope of the speech signal preserved. Spectral subtraction is a popular method of reducing the effect of additive, un-correlated noise in a signal (Boll, 1979).

The concept of spectral subtraction, illustrated in Figure 1, is simple: measure the noise, subtract it from the total signal, and you're left with the desired signal alone. Clearly, the success of this strategy hinges on being able to identify speech and to precisely characterise noise. An additional challenge is to keep up with the dynamic speech and noise make-up of real listening environments.

Figure 2 illustrates the noise reduction system, which operates in each of the 17 Warp signal processing bands (Groth and Nelson, 2005). As can be seen, the analysis of the input consists of three components: a Signal Power Tracker, a Signal Presence Indicator, and a Noise Power Tracker. The Signal Power Tracker keeps a running tab on the overall signal entering the system. This is the part from which any noise will ultimately be subtracted. The Signal Presence Indicator is the component which identifies whether speech is present in the overall signal. To achieve this, spectral and temporal characteristics are derived from the input spectrum at 1 millisecond intervals to determine the probability that speech is present. This method constitutes a more exact way of identifying speech than modulation alone. Such an accurate identification of speech allows noise estimation to be restricted to frequency regions and points in time where speech is not mixed in. For any interval where speech is not found to be present, the Noise Power Tracker calculates a noise estimate from the input spectrum.

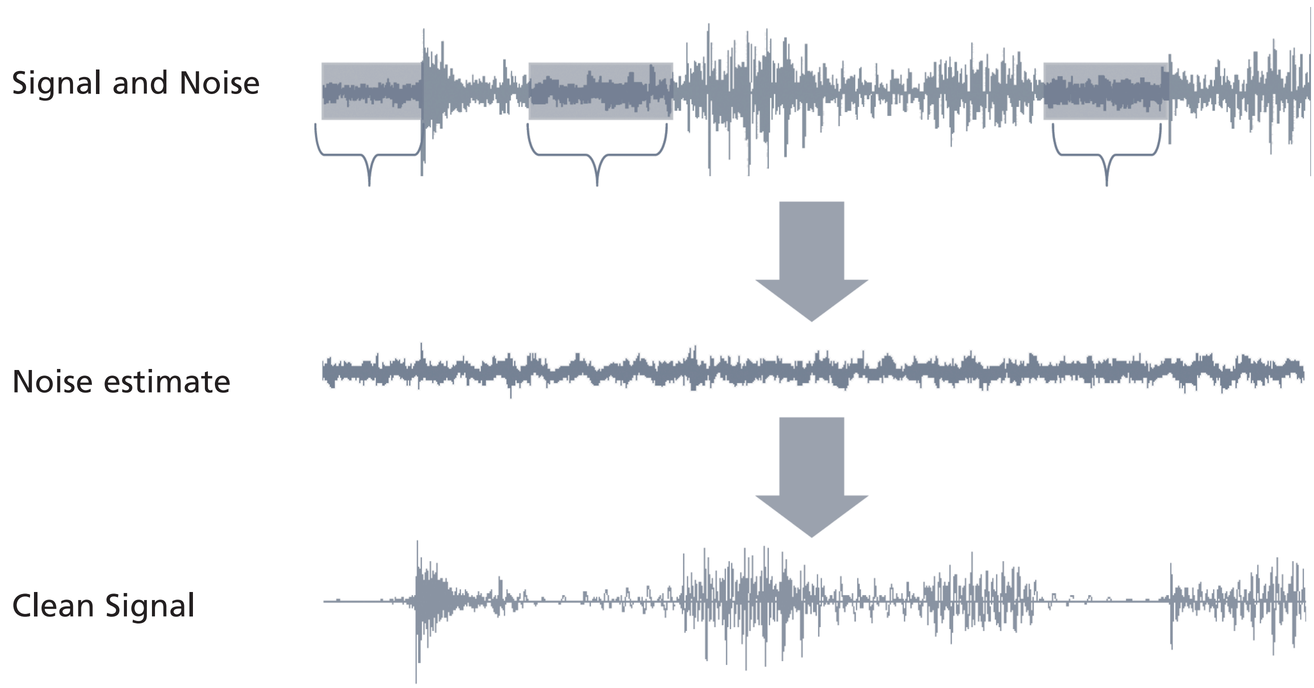

Figure 1.
Spectral subtraction removes noise from the total signal, leaving the desired signal intact.

Once the overall signal and noise are known, the signal-to-noise ratio (SNR) 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 signal-to-noise ratio estimate. Conversely, the signal-to-noise ratio estimate will be greater when the total input signal consists of both speech and noise. Depending on the signal-to-noise ratio estimate, the gain function of the noise reduction system calculates a new gain for each frequency band almost every millisecond. The gain function is mathematically derived using Wiener optimal filtering theory. This technique is preferred for preservation of the output waveform relative to the input waveform. When only noise is present, the noise spectrum estimate is subtracted from the total signal by a given amount. When noise and speech are present simultaneously, the most recent noise spectrum estimate is subtracted.

The time constants of the noise reduction system are crucial for its performance. The Signal Power Tracker employs fast time constants in order to preserve the speech envelope. However 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. That is, the system would mistake the speech for noise, resulting in speech information being subtracted from the overall input. Yet it is also desirable to have a fast update of the noise estimate in order to most efficiently react to non-stationary noise sources. To solve this dilemma, noise reduction employs a dual approach. The time constants for updating the noise remain fast when speech is not identified. When speech is detected the system is slowed down, and therefore will assume that the noise remains constant during speech. This means that during speech, the noise estimate will only be updated in pauses between words and syllables. Real world performance is not hurt by this, however, since most noise environments are stationary on the order of a few hundred milliseconds. This allows the Noise Power Tracker to catch up during pauses in the speech. Figure 3 illustrates how efficiently noise reduction is able to reduce speech babble noise without degrading speech. As is evident in this example, competitive digital noise reduction systems have little effect when the competing noise is a multi-talker background, which is so often the case in real-world listening environments.

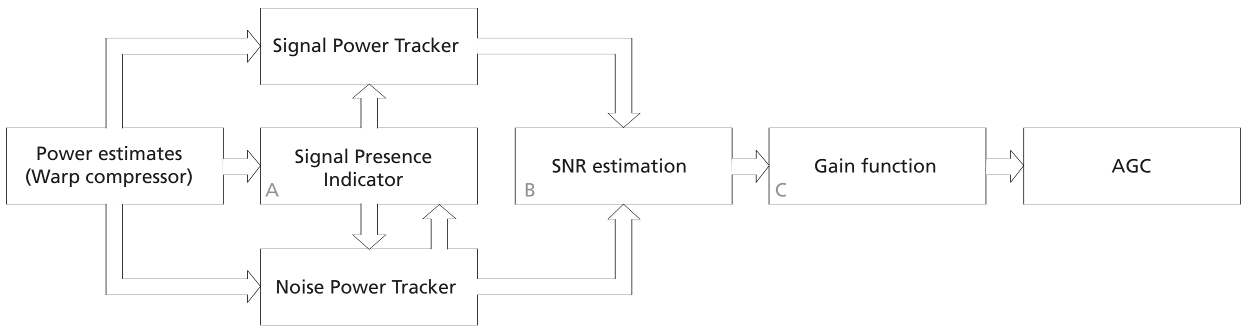

Figure 2.
Power estimates provided by the compressor structure are used by the noise reduction system to attenuate frequency bands with poor signal-to-noise ratio.

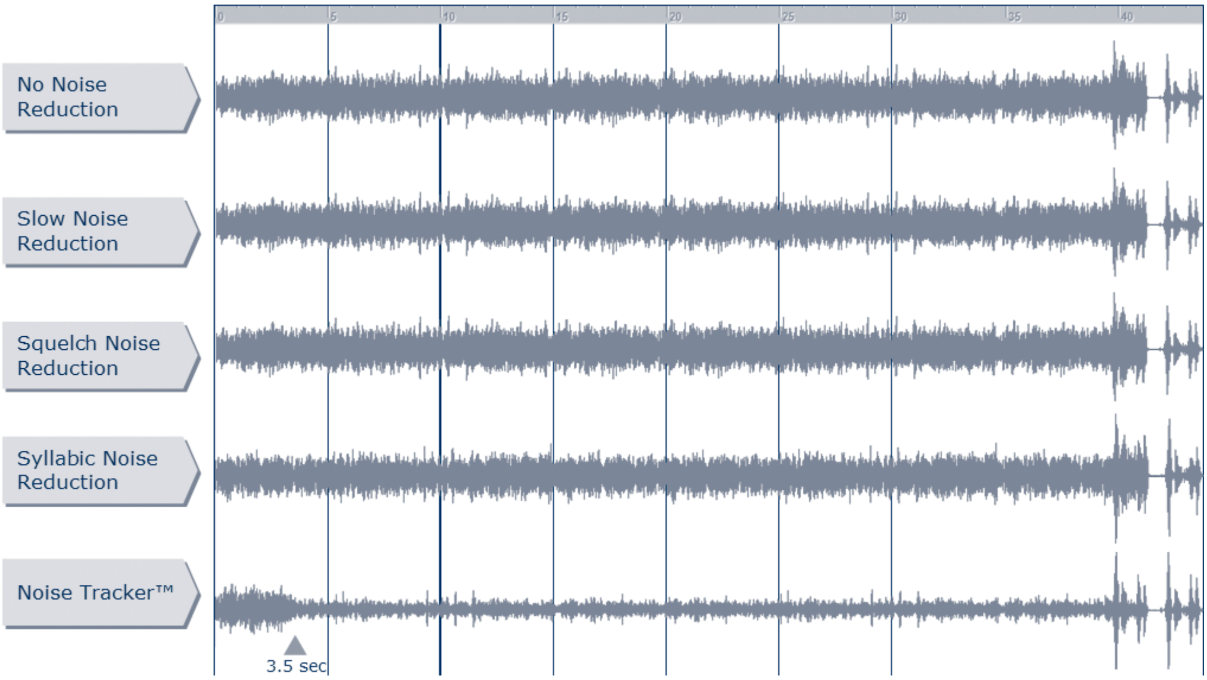

Figure 3.
Comparison of the effects of varying digital hearing instrument noise reduction schemes on speech babble waveforms. noise reduction not only reduces the amplitude of the speech babble quickly and effectively, it also preserves the single talker speech which occurs at the end of the sample.

The noise reduction system offers varying degrees of noise reduction. It is possible to fit the Danalogic 6 instruments with mild (-3 dB), moderate (-6 dB) or strong (-9 dB) noise reduction. The noise reduction value for each degree of noise reduction is the amount which would be applied if no speech signal were detected. For higher signal-to-noise ratios, the Wiener filter determines the actual gain reduction value at any given moment in each channel. All degrees of noise reduction can be applied without affecting speech intelligibility negatively. The different degrees of noise reduction are offered to increase fitting flexibility and to address individual preferences. For example, some users prefer the strong noise reduction 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 noise reduction settings in the environmental programmes. The mild degree of noise reduction is default when fitting Danalogic 6 in the Basic programme while the moderate degree of noise reduction is recommended as default when fitting the Restaurant programme.

SUMMARY

Listening comfort and effort along with sound quality are important contributors to satisfaction with hearing instruments. Both imply being able to hear desired sounds such as speech at comfortable levels without the distraction and annoyance of unwanted noise being over amplified. In addition, good sound quality requires that there are no disturbing perceptual effects of the hearing instrument processing such as audible gain fluctuations, or distorted or muffled speech. When comparing noise reduction to all other noise reduction systems on the market, slow time constant modulation systems do not reduce noise during speech. Noise reduction has the ability to do this. Fast time constant modulation systems introduce distortion while reducing noise during speech. noise reduction provides good sound quality by using adaptive time constants. The innovative techniques which accurately identify speech and noise combined with the noise reduction's adaptive time constants contribute to the excellent listening comfort and superior sound quality enjoyed by Danalogic 6 users.

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