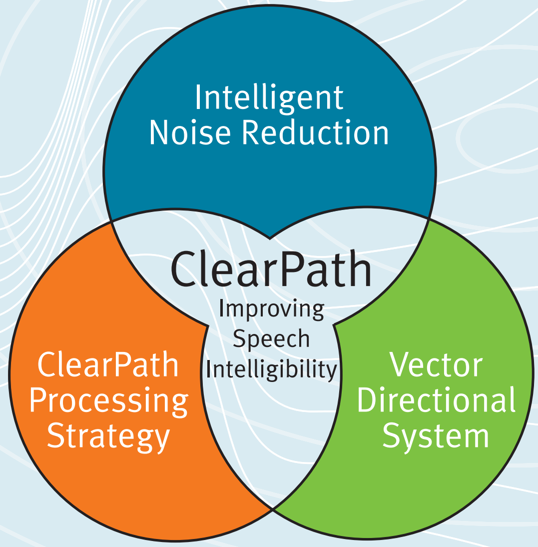
conversa™
with ClearCall

ClearPath™:

A THREE PART APPROACH TO IMPROVING SPEECH IN NOISE



Executive Summary

Hearing aid wearers need to function in environments that contain many different types of real-life audio signals often consisting of desirable sounds, such as speech or music, mixed with undesirable noise. Listening in the presence of background noise has traditionally posed the greatest challenge for hearing aid wearers and is also a primary source of dissatisfaction with hearing aids.[1,2] Recent developments in digital signal processing, including intelligent signal detection and noise reduction, as well as switchable directional microphones, work to amplify desirable audio information while suppressing undesirable noise in the signal.

Conversa's ClearPath™ technology combines 3 elements to improve speech intelligibility in noise:

1. Intelligent Noise Reduction
2. Vector Directional Microphones
3. ClearPath Processing Strategy

unitron™ hearing

Background: Early Noise Classification Systems

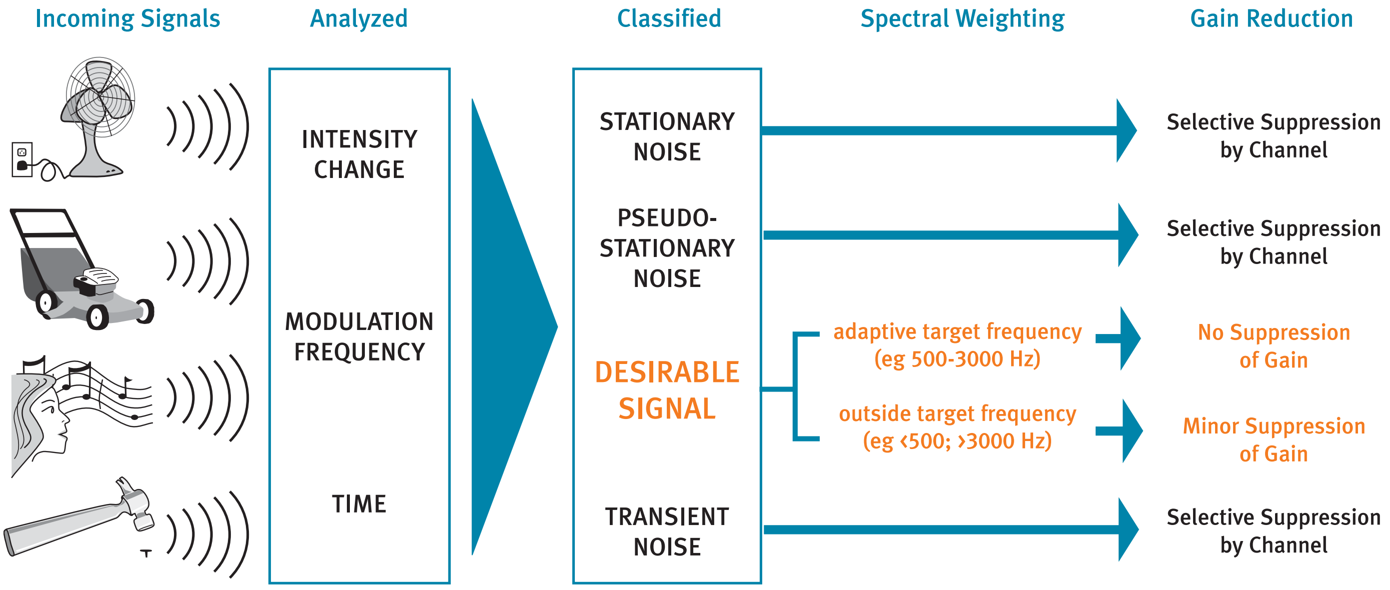
The effectiveness of a noise reduction algorithm depends primarily on the design of the signal detection system. Early noise reduction algorithms tested only one dimension of the signal to determine if a sound was a desirable signal or a noise. For example, one algorithm design categorized sound events by amplitude or intensity in each frequency region. Those sound events with the lowest amplitude were assumed to be noise. The gain for each band was calculated separately to minimize the effect of the identified noise. After applying the calculated gain in each band, the output signal was formed by recombining signals from each frequency band.

However, the incoming signal cannot be properly classified as speech or noise based on a single dimension. Sound events cannot be accurately categorized solely on the basis of their amplitudes. Furthermore, a single dimension does not take into account changing conditions in which noise is more or less prevalent at different times. For example, when listening to the radio while riding in a car, the amplitude of the noise from the car rises and falls proportionately as the car changes speed, yet, the speech from the car radio remains constant. The result is that the more intense noise sound events, such as engine noise when the car speeds up, will not be categorized as noise and many non-noise sound events, such as the car radio, will be categorized as noise.[3]

The modulation index is another dimension that can be used to categorize the incoming signal as either speech or noise.[4-9] The modulation index is defined as the rate of change of the signal amplitude in each band. Speech exhibits rapid and frequent amplitude changes and, therefore, has a high modulation index. Steady state noise, on the other hand, has a low modulation index. If noise is

Figure 1

Conversa's Intelligent Noise Reduction Approach



Incoming signals are analyzed along three dimensions in 16 separate channels, then classified. Spectral weighting is applied to the desired signals and gain reduction is applied.

detected in a single band, the gain of that band is reduced relative to the other non-noise bands. Thus, bands with noise are suppressed in favor of bands without noise.

Classifying noise solely on the basis of its modulation index assumes that the frequency spectrums for all noise demonstrate little or no variation over time. Such a system is accordingly limited to detecting stationary or slowly changing pseudo-stationary noise, which is not typical of most environments. Consequently, just as you cannot classify noise based only on its amplitude, the incoming signal cannot be properly classified as speech or noise based solely on its modulation index.

Conversa's Intelligent Noise Reduction Approach

A more effective method of signal characterization and noise reduction examines several dimensions of the signal simultaneously. The system is then able to adapt to signals with different noise content over time and, therefore, is capable of detecting and suppressing many different types of noise.

With ClearPath, incoming signals are analyzed in 16 separate bands along 3 different dimensions rather than using a unidimensional approach. This detailed information is used to categorize signals into 4 different noise or desirable signal categories. Amplification or suppression is applied differently and on a weighted basis to the signal based on the categorization (see figure 1).

Signal Analysis and Classification

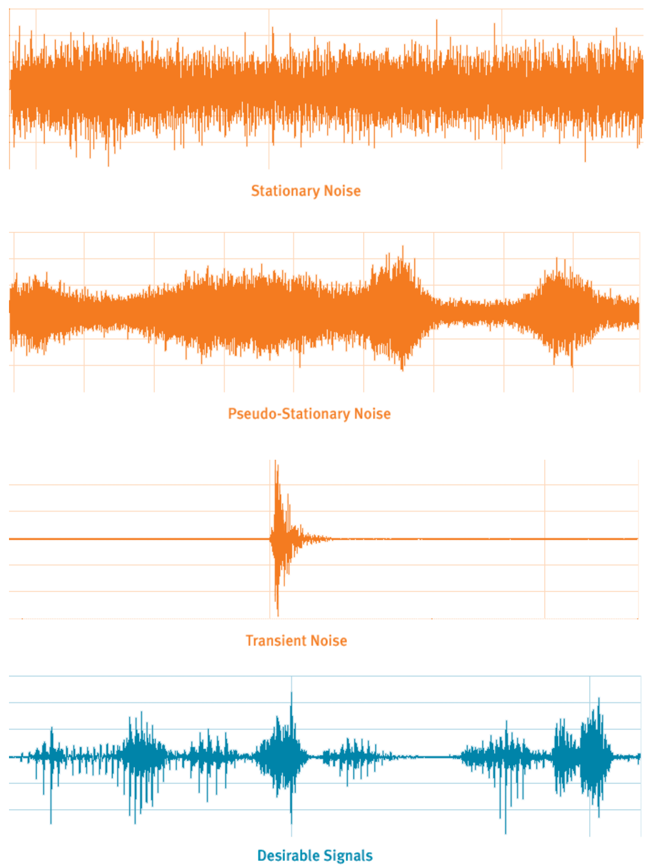
Noise can generally be classified into three major categories based on its characteristics:

1. Stationary noise (Example: air conditioner or motor/engine)
2. Pseudo-stationary noise (Example: traffic or crowd of people)
3. Transient noise (Example: hammering or door slam)

An effective signal detection and noise reduction system should be able to accurately classify signals as either one of these 3 types of noise, or as a

4. Desirable signal (Example: speech or music).

Figure 2
Waveform Envelopes of Noise and Speech



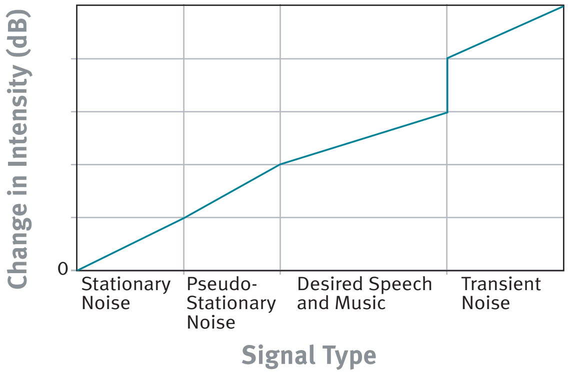
Each of the 4 signal categories shows distinct characteristics.

Conversa's ClearPath noise reduction technology analyzes and classifies sounds by simultaneously examining three specific characteristics inherent in the signals:

1. **Intensity Change** – defined as the change in the intensity of the audio signal over a selected time period. Using this dimension, the four types of audio information (3 types of noise plus desirable signals) are placed onto a continuum in which:

- stationary noises exhibit the smallest changes in intensity;
- pseudo-stationary noises exhibit larger changes in intensity than stationary noises but smaller changes than desirable sounds;
- desirable sounds exhibit a moderate amount of intensity changes; and
- transient noises exhibit the largest changes in intensity

Figure 3
Classification of Signals by Intensity Change

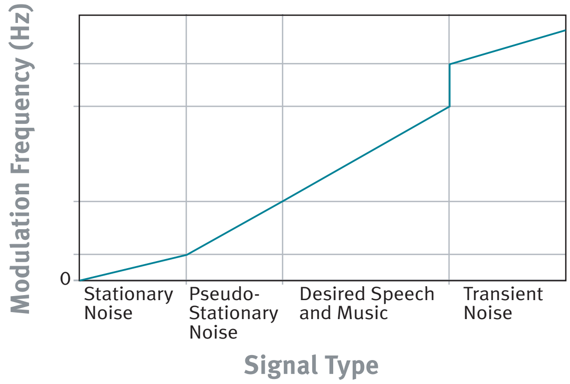


Stationary noises show smallest changes, transient noises largest

2. **Modulation Frequency** – defined as the rate at which the signal's intensity changes over a selected time period. For example, if the intensity of an audio signal reaches peak levels 30 times in one second, the modulation frequency is 30 Hz. Individual peaks need not have equal intensities, and may vary substantially over time. The four signal types are distributed in the modulation frequency dimension in which:

- stationary noises exhibit the lowest modulation frequency;
- pseudo-stationary noises exhibit higher modulation frequencies than stationary noises, but lower modulation frequencies than desirable sounds;
- desirable signals exhibit a moderate amount of modulation frequency; and
- transient noises exhibit higher modulation frequencies than desirable signals.

Figure 4
Classification of Signals by Modulation Frequency

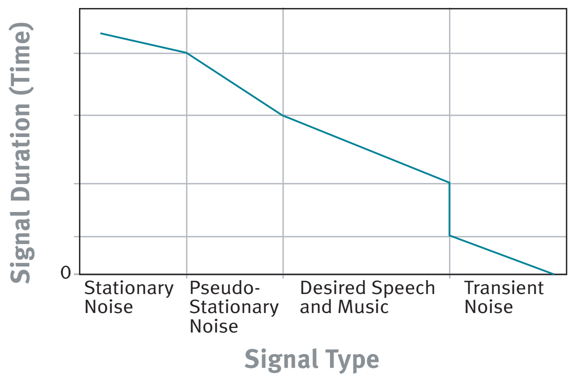


Stationary noises show lowest modulation, transient noises highest

3. **Time** - defined as the duration of the signal. For time, the four signal types are placed generally onto a continuum in which:

- stationary noises generally exhibit the longest time duration;
- pseudo-stationary noises often have shorter time duration than stationary noises, but longer time duration than desirable sounds;

Figure 5
Classification of Signals by Duration



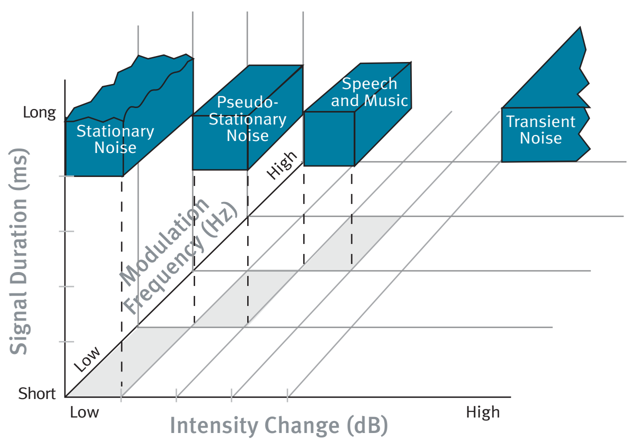
Stationary noises show longest duration, transient noises shortest

• desirable signals exhibit a duration with varying but generally moderate length; and

• transient noises generally exhibit much shorter time duration than desired signals (see figure 5).

An analysis of the intensity change, modulation frequency and time is done simultaneously in each narrow frequency band to determine subindices for each of these three dimensions. These are combined to produce a signal index on a three dimensional continuum within each frequency band. The signal in each band is classified as one of the four signal types by its three dimensional signal index (see figure 6). This determines how much the output of that band is amplified or suppressed. Consequently, the most gain is applied to bands containing desirable signals, and less or no gain is applied to bands containing noise. This results in a reliable, accurate, adaptive signal detection system which drives the ClearPath technology.[10]

Figure 6
Three Dimensional Signal Classification



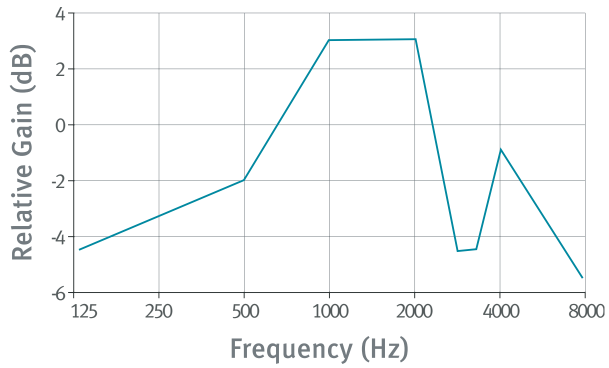
The four signal categories fall in distinct regions when plotted based on all three subindices.

Optimization of Noise Reduction

Spectral Weighting

After reliable signal detection, the algorithm reduces unwanted background noise with minimal impact on important speech cues. ClearPath technology incorporates a spectral weighting bias which tempers the degree of noise suppression across different frequencies by controlling the amount of gain reduction in each channel. For example, gain reduction may be applied vigorously in the low-frequency channels, moderately in the high-frequency channels and least aggressively in the mid-frequency channels where the most important speech cues occur.

Figure 7
Spectral Weighting

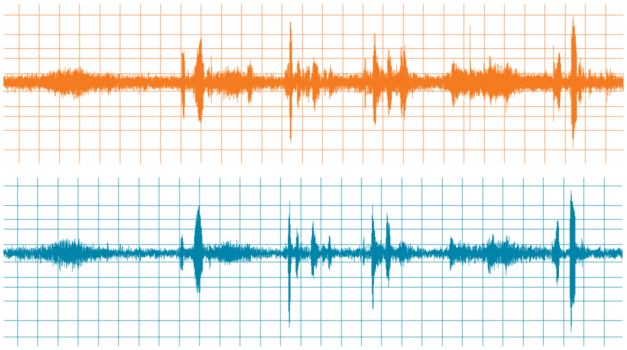


Gain reduction is increased or decreased by several dB across frequencies

Optimized Attack and Release Times

ClearPath's optimized attack and release times minimize impact on speech sounds when the noise reduction is activated. Time constants are a crucial factor in a noise reduction algorithm. If the algorithm reacts too quickly to changes in the signal index, a sudden transient burst from a brief phoneme (such as a /d/ or a /t/) could be suppressed as if it were a noise and become inaudible. When the time constants are too long, pseudo-stationary noises and transient noises would not be detected quickly enough. The gain in the frequency channels containing noise would not be reduced in time and there would be no benefit from the noise reduction algorithm. Furthermore, as the signal changes from noise to speech, the time required for the algorithm to increase the gain in each band would be too long. Therefore, important speech information may be lost (see figure 8).

Figure 8
Speech with Oscillating Fan Noise: Slow vs. Fast Time Constants

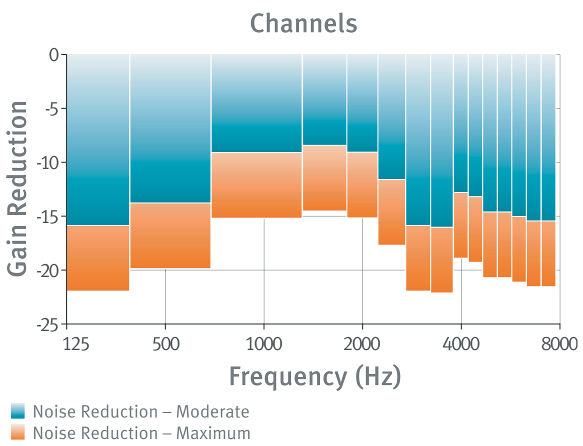


With faster time constants (bottom diagram), noise reduction is more effective in removing noise from speech than with slower time constants (top diagram).

Adjustable Noise Reduction

The amount of noise reduction applied is fitter adjustable through the Unifit™ software. Choices of moderate or maximum are available for each program allowing different settings according to the wearer's needs or preferences in different listening environments. Switching from moderate to maximum noise reduction increases the amount of available gain reduction equally in each frequency band. This means that the amount of absolute gain reduction can be changed, but the spectral weighting bias will be unaffected.

Figure 9
Moderate vs. Maximum Noise Reduction



The end result of ClearPath's multidimensional signal detection and optimized noise reduction is that the actual amount of gain reduction at any frequency over a given time frame is dependent upon five factors:

1. The mixture of signals and noise present in each band
2. The signal index as defined by the multidimensional detection parameters
3. The spectral weighting bias
4. Time constants of the noise reduction algorithm
5. Degree of noise reduction applied (moderate or maximum)

Vector Directional System

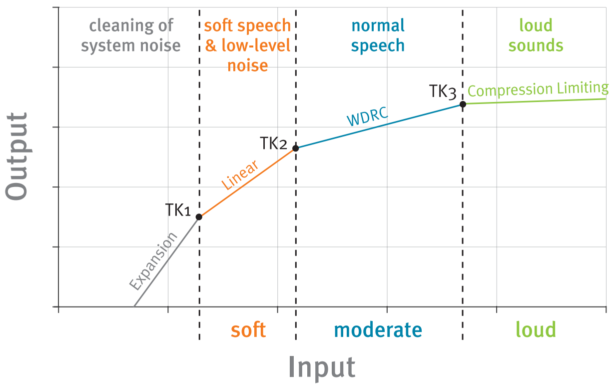
The vector directional microphone system complements intelligent noise reduction providing an effective solution for listening in background noise. Research has shown that directional microphones increase Signal-to-Noise Ratio (SNR) improving speech recognition in noise.[11, 12] The vector system provides free field AI-DI's of up to 5.3, which can translate into more than a 50% improvement in speech recognition in noise.[13] Directional microphones also increase users' overall satisfaction with their hearing aids.[14]

Vector directional systems are programmable into any of Conversa's three listening programs for customization to the wearer's needs. The vector BTE system utilizes two omnidirectional microphones. Vector custom combines one omnidirectional and one directional microphone which share 2 port openings. Due to the system's exceptionally small size, vector is available in shell styles down to the canal model allowing even more individuals to experience the benefits of directionality.

ClearPath Processing Strategy

Through the ClearPath processing strategy, hearing aid performance is optimized for all signal input levels from very soft to very loud. In all three programs, the hearing aid applies the most appropriate strategy at any given moment: quiet mode expansion, linear function, wide dynamic range compression (WDRC) and compression limiting. Three separate, optimized kneepoint settings provide wearers with comfortable, natural sound in a wide variety of environments. As shown in figure 10, TK1 sets the upper limit of the quiet mode expansion which cleans low intensity system noise. TK1 settings in each channel are fixed at appropriate levels and are product specific to provide the quietest circuit possible. Linear processing occurs between TK1 and TK2 for soft speech and low-level environmental sounds. This linear segment provides a smoother transition from expansion to WDRC, and reduces feedback if too much gain is applied unnecessarily to soft sounds. TK2 is the kneepoint for WDRC, a soft compression applied to moderate input levels including conversational speech. At and above TK3, compression limiting occurs at a ratio of 20:1 reducing gain for very loud sounds to avoid discomfort. TK2 and TK3 are hearing loss dependent and automatically set to optimal values by Unifit fitting software.

Figure 10
ClearPath processing strategy

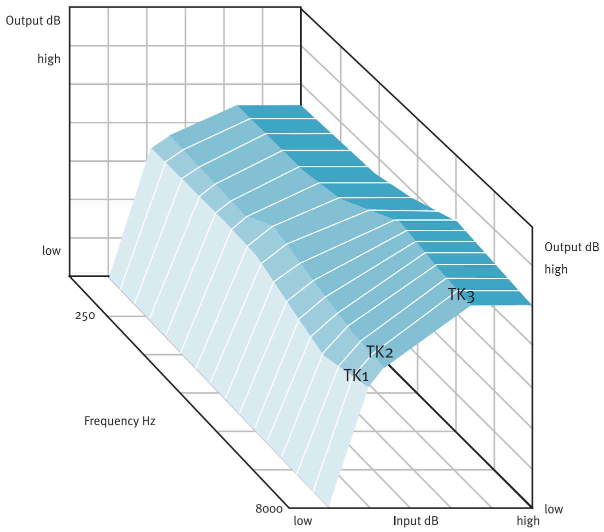


Four processing segments are separated by three kneepoints.

The ClearPath processing strategy provides a clear separation between the technical optimization of the sophisticated Conversa digital circuit, and the fitting flexibility required by hearing healthcare professionals to meet their client's audiological needs. For example, it is not necessary to sacrifice system performance by adjusting expansion kneepoint settings to provide appropriate gain for soft speech. These separate issues are dealt with by different segments of the strategy.

In addition, the ClearPath processing strategy is applied independently across all narrow frequency channels. Hearing healthcare professionals will find it provides maximum flexibility for fitting a wide range of hearing losses with appropriate gain for soft, moderate and loud inputs. For hearing aid wearers, it maintains clean, clear sound across a full range of input levels.

Figure 11
ClearPath Processing Strategy: A Three Dimensional View Across Frequencies



The strategy is working independently across all narrow frequency channels.

Summary

Individuals fit with ClearPath consistently comment on the comfort and effectiveness of the technology in noisy settings such as when driving on the highway, walking through a noisy plant, attending a large group meeting or having dinner in a restaurant. ClearPath's multidimensional signal detection, adjustable noise reduction and optimized attack and release times reduce unwanted background noise with minimal impact on important speech cues. Furthermore, ClearPath technology combines advanced noise reduction with a vector directional system to improve speech recognition in noise. The added benefit of ClearPath processing strategy optimizes hearing aid performance for all signal input levels offering hearing aid wearers a comfortable, natural sound in a wide variety of settings.

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