SCENIC™

Environmentally Aware
Feature-Packed DSP Hearing Aid Amplifier

WHITE PAPER

ACOUSTIC SCENE ANALYSIS

AUDIOLOGY BACKGROUND

IntriCon
The Body-Worn Device Company

1260 Red Fox Road · Arden Hills, MN 55112 · TEL: (651) 636-9770 · FAX: (651) 636-8944
Kesselschmiedstrasse 10 · D-85354 Freising, Germany · TEL: 049-8161-4804-0 · FAX: 049-8161-4804-18
www.intricon.com

The aim of every hearing aid is to provide the wearer with appropriate amplification so as to make speech audible while keeping loud sounds comfortable. As aids have become more sophisticated, additional technologies have been developed to complement the underlying amplification scheme and to help improve user satisfaction in diverse environments.

The difficulty with many early additions was that manual program changes were required, something that relied on users having adequate physical and cognitive abilities to correctly adjust the device. The development and inclusion of adaptive features now means that many products are capable of automatically adapting their features to suit the environment.

Scenic is a fully adaptive product whose core feature is Acoustic Scene Analysis. Acoustic Scene Analysis promotes the synergistic operation of *Scenic's* adaptive algorithms, so that every acoustic event in a given environment is defined and appropriately processed. A Live Display capability captures and presents information about the environment based on the activity of the adaptive hearing aid features.

ACOUSTIC SCENE ANALYSIS

When you take a photo, you are taking a snapshot of a moment in time. Every time the shutter is pressed, information about lighting conditions, shadows, and movement is captured and digital processing is used to ultimately generate the resulting the picture.

The same thing happens when you take a snapshot of the acoustic scene. Temporal, spectral and spatial information is captured, as is information about tonal qualities and the signal-to-noise ratio.

In *Scenic*, Acoustic Scene Analysis uses snapshots that occur every two milliseconds to activate and adapt features to optimize speech intelligibility and sound comfort.

The operation of these features is dependent not only on various acoustical indicators including input level; spectral; temporal and spatial information; and signal-to-noise ratio, but also on complex interactions with other adaptive *Scenic* features.

In other words, Acoustic Scene Analysis is the activity of the adaptive features of *Scenic* that occur through a series of complex interactions influenced not only by the original input characteristics, but also by the other adaptive features. The transitions of the adaptive features are seamless, and they provide the best conditions for high speech intelligibility and comfort.

With a system-wide architecture, Acoustic Scene Analysis controls modules based on the acoustic surrounds. Its power lies in its robust spread of information. Analogous to distributed processing, information used in Acoustic Scene Analysis is spread across multiple processing modules. This results in sound processing that has greatly improved efficiency.

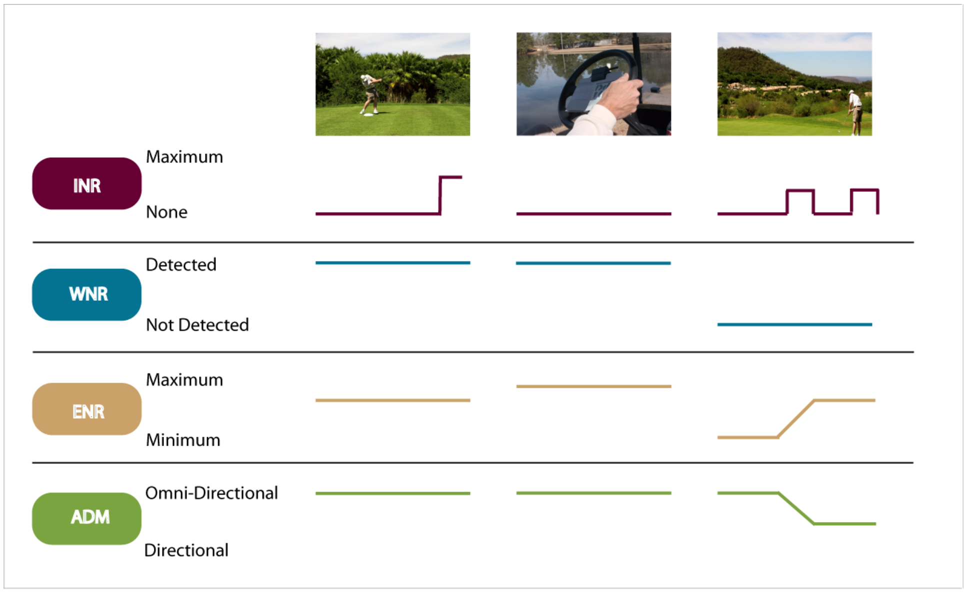

Figure 1. *The activity of the Acoustic Scene Analysis modules are influenced by the temporal, spatial, input level, signal-to-noise ratio and tonal qualities of the input signals, as well as the other adaptive modules.*

The adaptive features of *Scenic* are advanced in their own right. *Scenic* automatically suppresses loud impulsive sounds, wind and surrounding environmental noise. At the same time, *Scenic* improves the signal-to-noise ratio and provides high quality sound. *Scenic* is able to adapt to any listening environment with no user intervention.

IMPULSE NOISE REDUCTION

Everyone is exposed to loud noises numerous times each day. Hearing impairment and its associated physiological recruitment effect means that those with hearing loss are especially susceptible to greater levels of discomfort when loud sounds occur. This discomfort is compounded when hearing aids are worn. In other words, loud sounds can be “outright annoying to listen to” (Keidser, O’Brien, Latzel & Convery 2007). In particular, impulsive noises - broadband, high intensity, short duration (Henderson & Hannernick, 1986) - remain problematic because traditional algorithms, such as WDRC can’t respond quickly enough to control their loudness.

Hearing is vital as a warning for imminent danger (Dillon, 2001), and impulsive sounds are a strong indication of danger. Wearing a hearing aid can help those with hearing impairment locate the sources of danger (Moll van Charante & Mulder, 1990; Dillon, 2001; & Arlinger, 2003). The requirement therefore for an effective impulsive noise reduction algorithm is that it not only reduces the impulsive noise to a comfortable listening level, but that it also preserves its distinctive characteristics for safety.

Instant Detection and Suppression of Impulsive Sounds

Scenic's impulse noise reduction system detects and reduces the loudness of impulsive sounds almost instantly. It is able to do this because it monitors the signal levels at the front-end of the signal processing chain - before any other analysis is performed.

A large amount of the power in an impulse is at its beginning, so when the signal level quickly rises - above that of the long term peak level - an impulsive sound has been detected.

To protect the hearing aid user there is a dedicated gain function, separate to WDRC. When an impulsive sound has been detected, the gain is rapidly reduced to a safe level where it stays for the duration of the impulse. The gain returns back to its original level more slowly. This preserves the naturalness of the impulsive sound, and also ensures that speech quality is not adversely affected.

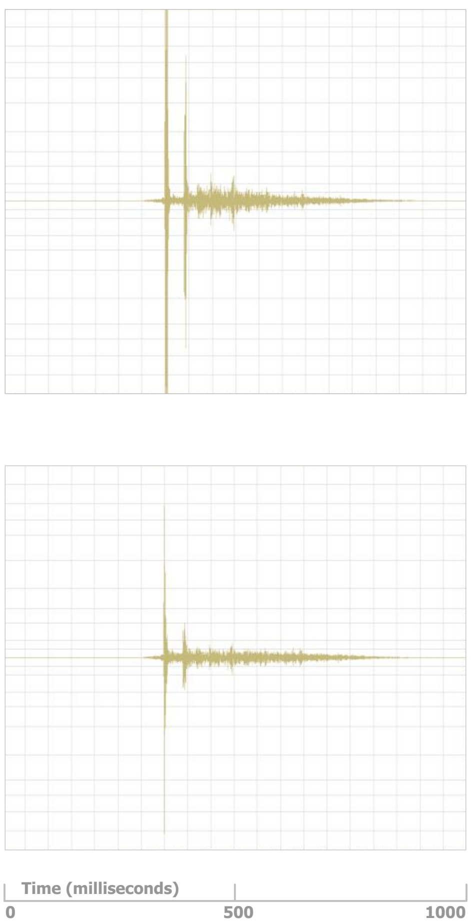

Figure 2. *Time domain responses of a door slamming. In the top image, INR is turned off, in the bottom image, INR is turned on. The integrity of the original response is maintained, while at the same time its intensity has reduced.*

WIND NOISE REDUCTION

Wind noise is created by turbulence that occurs as the air passes the aid user's head, hearing device, or other obstacles. Concentrated in the low frequencies below 500Hz, it results in unpleasantly loud sound which swamps everything else or drives the hearing device into saturation. Wind mostly affects BTE devices and some devices with directional microphones (Chung, Mongeau & McKibben, 2009). Considering that the intensity of wind at the eardrum can reach 100 dBSPL (Grenner, Abrahamsson, Jernberg & Lindblad, 2000; Roe, Dillon & Katsch, 2001), it is not surprising that users continue to be dissatisfied with the performance of hearing aids in windy environments (Kochkin, 2010).

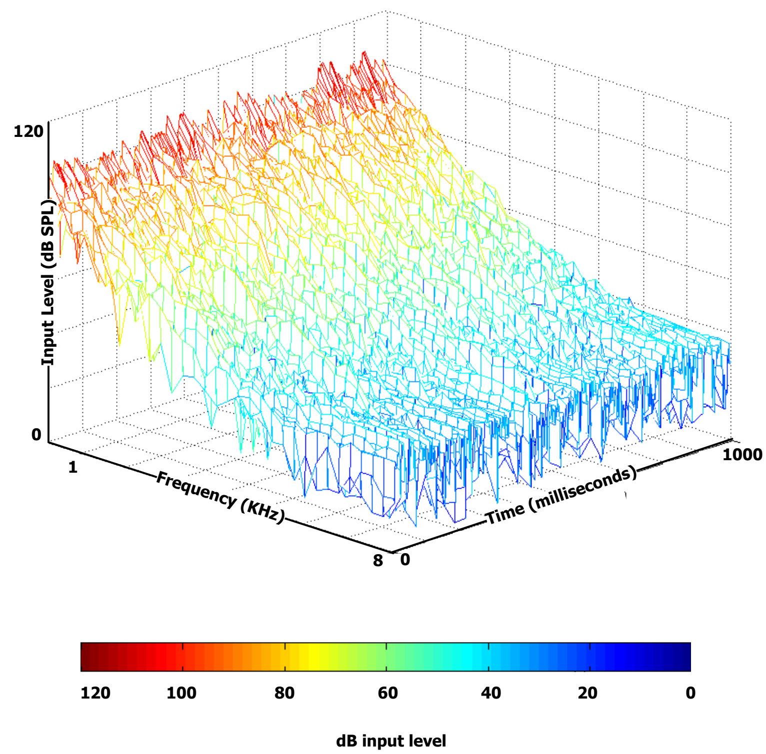

Figure 3. *3D image of wind input to a BTE hearing aid for 1 second. Wind was located at 0 degree azimuth and was at a speed of 13.4 mph (6m/s). The intensity level exceeded 100 dB SPL in the low frequencies.*

Reducing Wind Noise

Scenic's wind noise reduction algorithm uses its stored knowledge about the spectro-temporal characteristics of wind – low correlation and low frequency emphasis – to identify wind and to reduce its intensity. With variable settings that suppress different wind speeds the algorithm is able to effectively cope in a variety of outdoor environments, regardless of the microphone combination and style of hearing aid.

ENVIRONMENTAL NOISE REDUCTION

We are constantly surrounded by ambient noise of varying levels. Walking along a busy street, a meal in a restaurant, and travelling on the subway are all examples of noise sources that many people encounter daily. Despite this common occurrence, or perhaps because of it, the presence of noise continues to be the cause of the highest level of dissatisfaction for hearing aid users (Kochkin, 2010).

It is therefore essential to add a specialized noise reduction system to WDRC to compensate for the discomfort and unpleasantness that noise brings.

Identifying Background Noise

To identify noise in *Scenic*, a modulation detection technique is used. In modulation detection systems, noise is identified by its shallow modulation depth (compared to speech which has a much deeper modulation depth). *Scenic's* environmental noise reduction system has multiple channels, each with a fast and a slow detection system, which are used to identify and suppress noise. To ensure that speech quality is maintained, noise reduction only occurs in the channels where noise is identified.

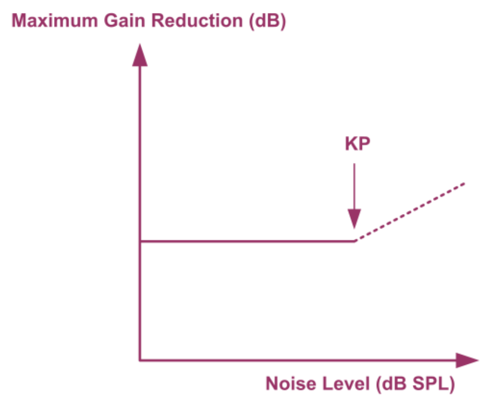

Figure 4. *Shallow modulation depths are associated with noise, and deeper ones with speech. The setting of a kneepoint, allows greater levels of gain reduction as the noise level continues to increase (right).*

User Feedback

In real-world noisy situations users prefer *Scenic's* environmental noise reduction algorithm to be on rather than off, both in situations where there is speech and noise, as well as situations where there is only noise (Zakis, Hau & Blamey, 2009). These results are consistent with other studies which have found that environmental noise reduction algorithms improve listener comfort (Boymans & Dreschler, 2000; Ricketts & Hornsby, 2005); ease of listening (Bentler, Wu, Kettel, & Hurtig, 2008); and acceptable noise levels (Mueller, Weber & Hornsby, 2006).

ADAPTIVE DIRECTIONAL MICROPHONE

The loss of frequency resolution, in addition to the elevation of hearing thresholds means that the hearing impaired require a higher signal-to-noise ratio to understand speech than listeners with normal hearing (Zwicker and Schorn, 1978, & Plomp, 1986).

Directional microphones provide the most effective method by which the signal-to-noise is improved for a hearing impaired listener. With newer generations of hearing aids, directional microphones have become more adaptive and automatic.

Automatic Adaptation to Optimal Directional Pattern

Scenic's adaptive directional microphone automatically adapts to the most effective directional pattern when in noisy environments, and automatically adapts back to an omni-directional pattern in quiet or in windy environments.

The directional microphone has a flat frequency response, meaning that low frequency compensation is not necessary. Because the directional microphone acts before any amplification is applied, it operates across a range of frequencies. Additionally, its design allows for use in open ear devices.

User Feedback

Wearer's of *Scenic's* adaptive directional microphone have found it beneficial in a variety of environments that are encountered every day (Blamey, Fiket & Steele, 2006). In the lab, the adaptive directional microphone has been shown to provide a 4.5 dB signal-to-noise ratio improvement over the omni-directional condition. This is equivalent to the level achievable with normal hearing (Blamey et al., 2006).

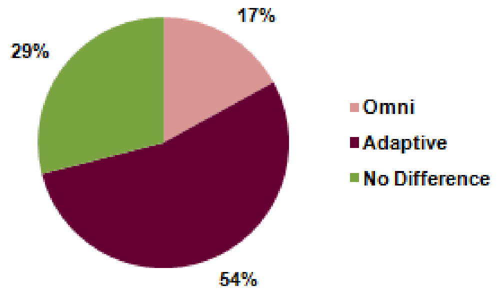

Figure 5. *When asked which microphone configuration was preferred for a collection of listening situations, the adaptive mode was the favored option (Blamey et al., 2006).*

ADAPTIVE FEEDBACK CANCELLATION

In the past, hearing aid users have had to make do with inferior performance when listening to music, often because of entrainment caused by the feedback canceller. Entrainment is the distortion or modulation of the original signal and is an audible side-effect.

With *Scenic*, this is no longer the case. *Scenic's* 4th generation adaptive feedback canceller (AFC) is able to accurately detect and compensate for the tonal qualities that distinguish music from feedback. It is able to achieve this with no compromise in cancellation time and added stable gain figures. Other feedback cancellers either slow or halt the feedback canceller whenever music or tones are detected, or introduce a phase or frequency shift into the forward path signal. These methods can be effective, however they also introduce side effects. For instance, slowing a feedback canceller means that the system is unable to act effectively in situations of transient feedback, like when a phone is brought to the ear, or a hat is placed on the head. Likewise, the introduction of a phase or frequency shift to the forward signal path is audible to aid wearers, and is often just as distracting as the entrainment it is designed to avoid.

Resisting Feedback Entrainment

To overcome the limitations of these other systems, *Scenic's* feedback canceller algorithm runs continuously at full speed, but with special adaption that is tone resistant. The user is oblivious to its operation, even in the presence of fast changes to the feedback path, as the system is configured to be fast acting regardless of the input signal type. The use of this algorithm means that no entrainment artifacts are introduced and that the FBC can continue to run at its full rate even in the presence of music or tones.

User Feedback

The best people to evaluate such a feedback canceller and its performance in music are classically trained normal hearing musicians. With ears sensitive to detecting even the most subtle of changes, musicians are the most critical of listeners. Five classically trained normal hearing musicians judged the quality of the feedback canceller, by listening to the same piece of music without the feedback canceller, then with the feedback canceller. The chosen music was especially difficult for feedback cancellers as it features sustained notes likely to trigger most feedback systems. There was no significant preference for either condition.

LINKING IT ALL TOGETHER

Live Display

Live Display shows the real-time activity of *Scenic's* adaptive features. The capability can be included in fitting software and used by the fitter to observe the operation of the hearing aid in response to different inputs. Live Display can be used as an effective counselling tool as it can demonstrate the performance of *Scenic's* features under various conditions. Explaining to clients about the importance of ideal seating arrangements in busy restaurants, or positioning at a gathering, is made easier by being able to watch the effect of moving the hearing aid so that the directional microphone can work, and observing the activity of environmental noise reduction. In addition to counselling and demonstration, Live Display assists the fitter in diagnosing issues with the hearing aid fitting; evaluating the effects of fitting changes; and observing the distinctive output patterns in response to different listening environments.

Distinctive Output Patterns

A hearing aid with Acoustic Scene Analysis will react appropriately to any listening situation because the features react in distinctive ways to response to various inputs. A conversation at home in quiet has the adaptive directional microphone in an omni-directional polar pattern, while the environmental, wind and impulsive noise reduction algorithms continue running in the background, but are not actively suppressing anything. In a busy outdoor car park, the slamming of the car door, results in active suppression by the impulse noise reduction algorithm. The adaptive directional microphone will be changing its polar pattern as the cars drive past. Idling cars and the hum of nearby traffic will be stimulating the environmental noise reduction algorithm. If it is a windy day, then wind noise reduction will be active, and the adaptive directional microphone will move to an omni-directional pattern. Listening to music remains an enjoyable activity, because the adaptive feedback canceller remains free of distortions, and the other adaptive features are active.

No Changing of Programs Necessary

When Acoustic Scene Analysis is in operation, only one base listening program is required. This means that while there is a lot going on in the hearing aid, the user remains completely unaware of the activity of the various features as they go about their daily life – apart from the ongoing high quality sound. The level of amplification is fitted to the user's hearing loss, and the adaptive features need only be set once. This is because the adaptive features are designed to react to the different input conditions as appropriate. By operating in this way, the user avoids having to make changes to the volume settings or to the listening program.

In Summary

Scenic is a fully featured advanced hearing aid amplifier, setting new boundaries for automatic adaptive performance. With algorithms suitable for a wide range of hearing losses, *Scenic* can be used in any style of hearing aid for any hearing loss. With fully automatic operation, *Scenic* removes all complexity associated with changing program settings, ensuring optimal performance in any environment.

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