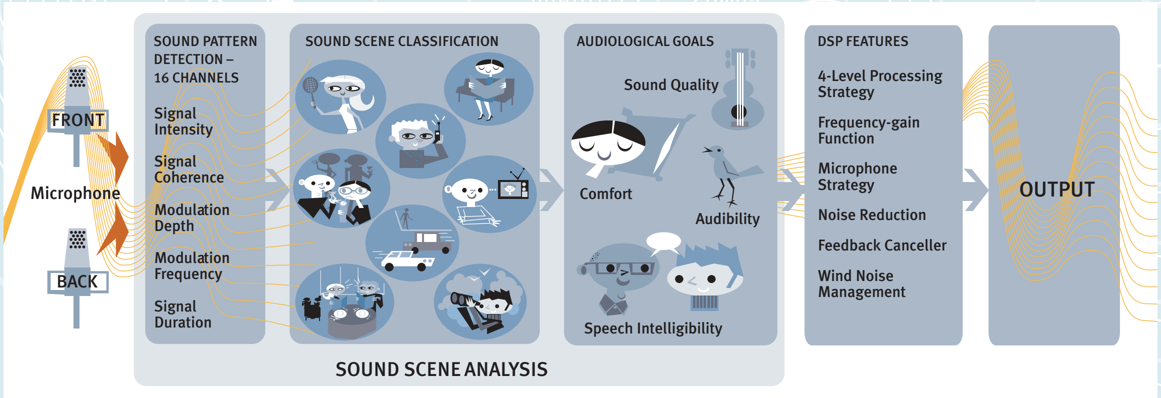
liaison™

Dynamic SoundScene™

HIGH PERFORMANCE SOLUTION FOR MULTIPLE SOUND SCENES

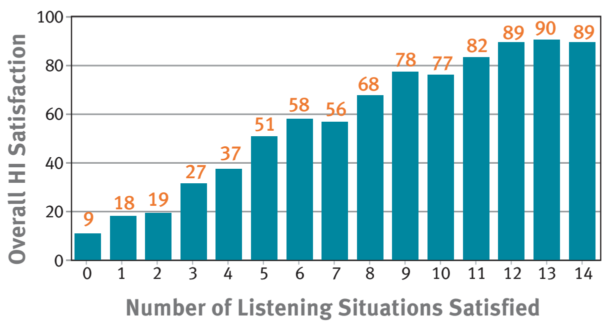


Executive Summary

Programming a new hearing instrument to provide satisfactory speech perception in quiet is a relatively straightforward process. It usually requires no more than a single mouse click to set a hearing instrument to within a few decibels of desired amplification. The real challenge is providing optimal speech perception and a more natural auditory experience across a range of listening environments. Kochkin notes that consumers expect high performance in multiple listening situations (Figure 1).[1] Only one in four hearing aid wearers are satisfied in as many as 75% of the situations they experience. An earlier survey indicates that satisfaction and performance are improved across a wider range of listening situations by using hearing aids that combine multiple features such as multiple memories, channels and microphones.[2] Combining features provides the professional with greater flexibility and the ability to tailor amplification for specific listening situations.

unitron hearing™

Figure 1
Wearer Satisfaction Increases with Multiple Environments



Source: MarkeTrak VI, Sergei Kochkin

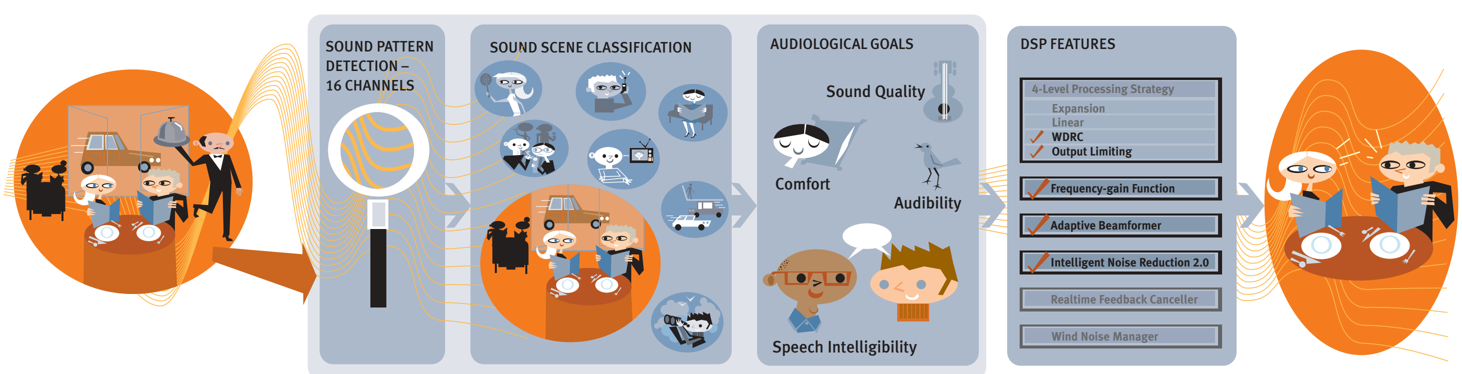
This figure shows how hearing instrument satisfaction increases as the hearing aid provides benefit in more listening situations. Wearers who are satisfied in only 1 or 2 situations report less than 20% satisfaction with their aids. Those who are satisfied in more than 10 situations report greater than 80% satisfaction.

The Concept of Dynamic SoundScene™

An auditory scene or sound scene is any acoustic environment in which auditory perception occurs. The success of a hearing aid fitting is directly related to the number of pertinent listening situations or sound scenes in which the device provides desired performance. More flexible hearing instruments with multiple microphones and channels or adaptive features such as noise reduction, wind noise adaptation and feedback cancelling algorithms will likely provide the best benefit. Individually, these features can improve performance and provide benefit for specific sound scenes. However, the real power of these dynamic components is best realized when they are joined together into a carefully conceived signal processing package.

For the processing package to be successful, the hearing instrument must be a feature-rich platform offering a wide range of components such as directional and omnidirectional microphones, sophisticated noise reduction, adaptive feedback cancelling, multiple processing channels and wind noise management. Advanced detection algorithms would recognize and define the listener's ever-changing auditory environment. The instrument would also contain a mechanism to obtain information from those detectors, interpret that data and adaptively adjust each component for optimal performance in the given listening situation or sound scene. In short, the hearing instrument would be able to automatically detect, classify and refine incoming signals,

Figure 2
Dynamic SoundScene Program



Dynamic SoundScene™ analyzes the sound pattern and adapts its processing strategy and advanced features to optimize auditory quality and provide the best sound experience.

thereby optimizing performance for new listening situations. The technology that incorporates these features on this advanced processing platform is known as Dynamic SoundScene™ .

The Principle Components of Dynamic SoundScene™

Five adaptive features comprise Dynamic SoundScene:

1. **Frequency-Gain Function** – adjustable gain compensation between quiet listening and background noise modes to optimize the wearer's awareness of scenes switching.
2. **Adaptive Beamformer** – adaptive directionality using paired omnidirectional microphones and digitally controlled time delays.
3. **Intelligent Noise Reduction 2.0** – 16-channel noise reduction with multiple signal and noise detectors including fitter selectable controls for degree of aggressiveness and activation threshold.
4. **Realtime Feedback Canceller** – rapid detection and suppression of acoustic feedback optimized to control oscillations caused by objects, such as a telephone, in close proximity to the hearing aid.
5. **Wind Noise Manager** – progressive multi-channel suppression of intense output transients caused by wind turbulence.

Frequency-Gain Function: Auditory Contrast Control Permits Customization

When hearing aid wearers purchase fully automatic instruments such as Liaison™, some individuals wish to hear the adaptive features at work while others prefer not to hear so much contrast as they change auditory scenes. Liaison's Dynamic SoundScene can incorporate hard or soft switching to suit individual preferences. For example, as they move from a quiet environment to a very noisy situation, they may like to hear the microphones switch from omni to directional performance or listen to the noise levels decrease as noise reduction is engaged. This type of performance is called hard switching. By using hard switching, the listener is alerted to changes occurring with the adaptive features of the hearing aid.

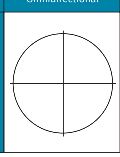
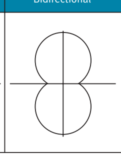
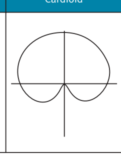
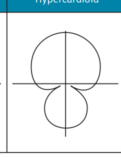
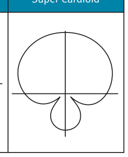
The simplest way to implement hard switching is by altering the frequency-gain function of the aid when the Dynamic SoundScene changes performance characteristics in response to new listening situations. When having a conversation with one or two people in a quiet restaurant, Dynamic SoundScene will engage omnidirectional microphones and disengage noise reduction to maintain the audibility of soft speech cues. However, upon leaving the restaurant, the wearer may step out onto a busy street where signal levels are considerably higher. After a few brief moments, Dynamic SoundScene will react to the higher signal levels by switching to directional microphones and engaging noise reduction. If wind is detected at the microphone ports, the wind noise manager will also be utilized. These adaptive changes in the hearing aid can be made very noticeable by adding a simultaneous low-frequency gain reduction. The additional frequency-gain change provides strong auditory contrast or a hard switch telling the wearer the adaptive features are engaging. The frequency-gain change has the added advantage of further reducing background noise.

On the other hand, many people prefer soft switching. For these individuals, the auditory contrast associated with the frequency-gain change can be minimized. By reducing or eliminating the low-frequency gain reduction that occurs when Dynamic SoundScene changes scenes, soft switching provides a nearly seamless flow from scene to scene as various adaptive parameters are adjusted by Dynamic SoundScene. The fitter may choose to engage very hard switching, very soft switching or something in between by adjusting the Auditory Contrast Control in Unifit™.

Adaptive Beamformer: Targets Stationary and Moving Noise Sources

A beamformer "consists of one or more microphones, which, in combination with a fixed or an adaptive filter, emphasizes desired signals while producing sharp nulls in the polar directivity pattern to cancel noise".[3] In other words, beamformer is an engineering term for a directional microphone(s) on a hearing aid. A fixed beamformer is a directional microphone with a specific polar pattern such as those shown in Figure 3.

Figure 3
Static Polar Patterns

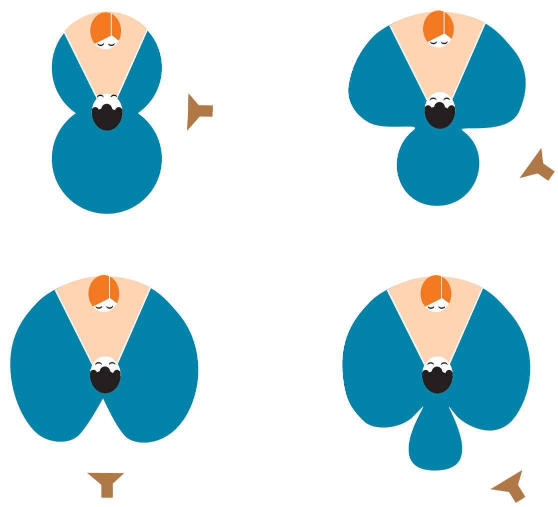
Characteristic	Omnidirectional	Bidirectional	Cardioid	Hypercardioid	Super Cardioid
Polar Response Pattern					

Paired nulls can create bidirectional, hypercardioid or super cardioid patterns depending on the position of the null in the polar plot. Fixed beamformers create a null for signals from a given direction. Adaptive beamformers are defined by the maintenance of the target area and adapt the nulls to the highest noise source.[4]

A single null (area of reduced sensitivity) at 180° is shown as a cardioid pattern. Paired nulls can create bidirectional, hypercardioid or super cardioid patterns depending on the position of the null in the polar plot. The azimuth at which the null occurs is determined by either an acoustic or digital delay applied to one microphone input also known as the fixed filter. In a fixed beamformer, the delay is applied to the rear microphone or the rear microphone port in the case of a single microphone system. The positions of the nulls never change in such a device because the time delay between the microphones never changes. Therefore, fixed beamformers can be described as having an area of reduced sensitivity for signals from a given direction.

Conversely, an adaptive beamformer does not place a null at a specific azimuth. Instead, the nulls are free to move while tracking noise sources. The nulls are limited to areas beside or behind the hearing aid and not the area in front of the aid referred to as the target area. Since the nulls can occur at many different azimuths, an adaptive beamformer is not defined by the positioning of the nulls so much as the maintenance of the target area where nulls will not occur. Liaison has a 60° target area in front of the listener as shown in Figure 4.

Figure 4
Continuous Adaptive Beamforming



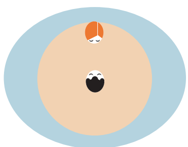
The orange cone of each polar plot is a representation of the target area for the listener, whose head is in the middle of the plot. As the sound source moves around the off-target azimuths, a null follows it, reducing sensitivity in the direction of the sound source.

The response patterns in Figure 4 have two nulls symmetrical to the vertical plane between the two talkers with the exception of the cardioid plot (lower left). If the hearing aid is modifying the digital delay between the microphones to adjust the position of the null, this modification will affect signals from the left and right equally. In reality, the presence of the speaker's head will significantly alter the sound field around the hearing aid microphones and the null for signals from the opposite side of the head will be wider and shallower.

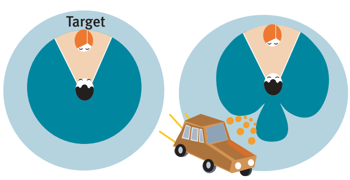
The basic premise of adaptive beamforming is to maintain the amplification within the target area and to place a null along the azimuth of the most intense off-target signal. The intensity of the off-target signal is inconsequential; an adaptive beamformer will attempt to suppress it. If the most intense off-target signal is an undesirable noise, such as a lawn mower, this is very effective. However, if the off-target signal is something the wearer wishes to hear, like someone speaking to them, this is undesirable. Therefore, adaptive directionality can be very effective in a noisy sound scene, but it can also mistakenly attenuate soft speech or environmental sounds in a quiet environment, leading to missed conversations and sounds.

Figure 5

Dynamic SoundScene's Adaptive Directionality



In quiet settings, Liaison remains in omnidirectional mode.



In the presence of an off-target, background noise source, the adaptive beamformer engages, positioning a null to reduce the amplitude of the noise. The target zone in front of the listener is maintained even as noises outside the zone are adaptively suppressed.

Signals from the listener's environment are continuously sampled in a 360° radius. When background noise is detected from any direction outside the target zone, Liaison responds with rapid adaptive suppression. Optimal suppression of background noise is maintained by refreshing directivity every 30 milliseconds to provide the lowest sensitivity for off-target, non-desirable signals. The adaptive beamformer rapidly updates off-target suppression to track moving sources, such as a car driving by, and maintain a favorable signal-to-noise ratio.

Liaison's adaptive beamformer in Dynamic SoundScene analyzes the sound pattern and automatically switches between omnidirectional and directional modes based on the input level as shown in Figure 5.

In quiet environments, Liaison provides omnidirectional performance at low input levels. In louder environments, Liaison engages the adaptive beamformer to produce maximum possible sensitivity for sounds from the target direction while suppressing off-target noise.

This maintains audibility for soft sounds in quiet (omnidirectional) and is particularly important for sounds like the telephone or a family member speaking from another room. However, at sustained high-level inputs Dynamic SoundScene provides suppression of off-target high-level noise sources (adaptive directional). In noise, Liaison uses continuously adaptive polar patterns to track moving noise sources, thus improving comfort in noisy sound scenes.

Intelligent Noise Reduction 2.0: Specify Activation Levels and Degree of Noise Reduction

The primary goal of a noise reduction system is to improve wearer comfort in the presence of background noise. Noise reduction systems achieve this by applying reduced gain to undesirable signals relative to speech or music. The challenge for an adaptive noise reduction system is that no two individuals define undesirable or desirable sounds in quite the same way. This is why Liaison's Intelligent Noise Reduction 2.0 allows customization.

Liaison's Intelligent Noise Reduction 2.0 is based on the same noise reduction strategies successfully employed in earlier products, such as Conversa™, with the added ability of improving customization during the fitting process.

Liaison's Intelligent Noise Reduction 2.0 uses multi-dimensional signal detection and classification to examine several aspects of the incoming signal simultaneously.

This system more accurately differentiates the different types of noise from desirable signals such as speech or music. The algorithm analyzes the signal for:

1. Intensity change or amount of modulation,
2. Modulation frequency, and
3. Time or duration.

Using this information, sound is categorized as:

1. Stationary noise (e.g. engine or fan),
2. Pseudo-stationary noise (e.g. traffic or crowd of people),
3. Transient noise (e.g. door slam or hammer) or
4. Desirable signals (e.g. speech or music).

Intelligent Noise Reduction 2.0 responds quickly enough for optimal reduction of unwanted noise while preserving speech integrity.

Utilizing optimized attack and release times, Intelligent Noise Reduction 2.0 ensures important speech information is not lost and noise is reduced quickly enough to improve comfort.

Intelligent Noise Reduction 2.0 incorporates a spectral weighting factor.

When noise is detected in any of the 16 channels, noise reduction is not applied equally across frequencies, but is spectrally weighted. The algorithm is most aggressive in the extreme low and extreme high frequencies, and less aggressive in the mid-frequencies where much of the important speech information is contained.

Intelligent Noise Reduction 2.0 also takes the next step to improve customization during the fitting. Most noise reduction systems are activated for all noise-like stimuli, regardless of the intensity of the signal. They attenuate any noise-like signal even if it is very soft and they do not focus solely on noisy environments. To have gain reduction for loud noise, the wearer must endure reduction of softer background sounds, sounds that some individuals may prefer to hear.

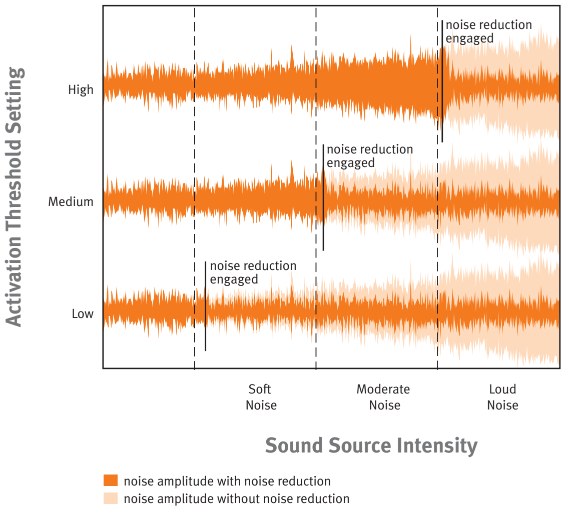
Adjustable Noise Reduction Activation Levels: Noise Reduction Only When You Need It

Dynamic SoundScene with Intelligent Noise Reduction 2.0 now offers an adjustable activation level to address this issue. Noise reduction will engage when needed most in very noisy environments. The activation level setting in Unifit™ allows the professional to adjust how the noise reduction engages for soft and moderate-level noise. If clients prefer to stay in touch with the soft sounds around them, the activation level can be increased. If they desire a more aggressive approach, the activation level can be decreased. The effect of the activation level control can be seen in Figure 6.

The bottom of the three waveforms shows the effect of selecting a low activation threshold. Noise reduction is

Figure 6

Intelligent Noise Reduction 2.0 Adjustable Activation Level



The effects of various settings of Intelligent Noise Reduction 2.0's adjustable activation level. Three waveforms are shown, each with a different activation level. The amplitude of the noise at the hearing aid microphone is increasing from soft to loud while moving from left to right in the figure.

engaged even for very soft noises. The middle waveform shows the effect of raising the activation level slightly. With Intelligent Noise Reduction 2.0, soft noises do not engage the noise reduction algorithm, but noises at a moderate level will. With the activation level set very high, as shown at the top of Figure 6, gain is reduced only when the intensity of the noise is quite high. This provides considerable flexibility when customizing the noise reduction settings. For example, you can set a:

- **Higher activation level** - for clients who prefer to hear soft noises such as air conditioning or who want noise reduction only for very loud background noise.
- **Lower activation level** - for clients who are bothered by soft environmental noise, perhaps due to normal hearing in low frequencies, or those that prefer to have noise reduction active for softer or louder background sounds.

Degree of Noise Reduction

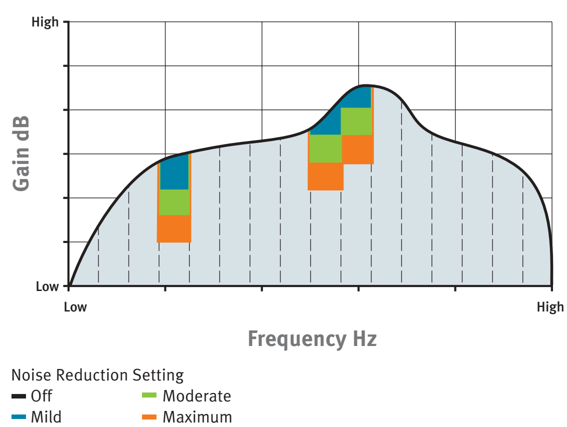
In addition to the activation level adjustment in Unifit, Intelligent Noise Reduction 2.0 also offers the ability to set the degree of noise reduction. This allows further customization to client's individual needs and preferences and is available both in the Dynamic SoundScene program and in the manual programs. Setting choices include:

- New **Mild** level - an average of 6 dB gain reduction in channels containing noise ranging up to 10 dB at some frequencies and is ideal for clients who want more subtle suppression of noise with minimal influence on their overall perception of loudness, even in loud conditions.
- **Moderate** –an average 12 dB reduction up to 16 dB in extreme low and extreme high frequencies.
- **Maximum** – an average of 18 dB reduction ranging up to 22 dB in some channels to provide maximum comfort in noisy settings.

The effects of the noise reduction degree adjustment can be seen in Figure 7.

Figure 7

The Effect of Liaison's Intelligent Noise Reduction 2.0 Degree Adjustment



The gray area of the figure shows the basic frequency response of the hearing aid without any noise reduction. The dotted lines represent the edge frequencies of the 16 noise reduction channels. The gain reduction required for noises in three different channels is shown for the three noise reduction degree settings.

If the hearing aid detects the same noise at each of three settings on the degree control, varying amounts of gain reduction are applied. Progressively more gain reduction is applied at each setting as the degree shifts from Off to Maximum. The effect also varies across different channels because the absolute gain reduction is further influenced by spectral weighting.

Realtime Feedback Canceller: Activates Without Deteriorating Speech

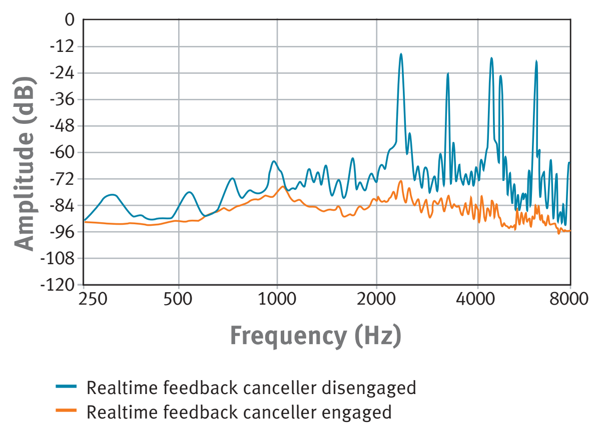
The purpose of the realtime feedback canceller is to provide rapid detection and suppression of acoustic feedback. Given enough time and resources, feedback suppression can be attained using other approaches such as phase cancelling. However, the time required is a limiting factor for success in any situation where a moving object such as a hand or telephone is placed in close proximity to the hearing aid. Sound reflecting from a nearby surface can drive the instrument to generate multiple feedback oscillations in few milliseconds. Those oscillations will build to self-sustaining feedback peaks that can put the hearing aid into saturation in as little as 200 milliseconds. For a hearing aid wearer to use a telephone naturally, as they would without the hearing aid, feedback suppression must be nearly instantaneous. The realtime feedback canceller can detect and suppress feedback in a little as 60 milliseconds.

Aside from rapid suppression, effective feedback cancellation across a wide range of listening environments including telephone use, must be able to:

1. Detect undesirable feedback,
2. Adapt to changes in the feedback pathway,
3. Suppress multiple oscillations simultaneously, and
4. Minimize disruptions of the desired signal.

Liaison's realtime feedback canceller within Dynamic SoundScene meets all four conditions using the same feedback system as Conversa. The system employs 12 independent narrowband detectors. Each detector has a bandwidth of only 500 Hz. All 12 independent detectors can suppress feedback simultaneously and each detector can suppress feedback oscillations in as little as 60 milliseconds. This elegant multi-channel approach to feedback suppression requires limited digital processing

Figure 8
Liaison's Realtime Feedback Canceller Suppresses Multiple Feedback Oscillations



This recording shows the effectiveness of the realtime feedback canceller suppressing multiple feedback oscillations caused by a telephone handset next to the hearing aid.

power and conserves battery life. An example of multipeak suppression can be seen in Figure 8.

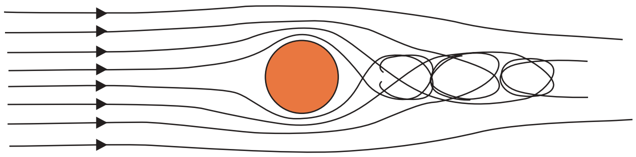
Figure 8 shows how effective the realtime feedback canceller is at suppressing multiple oscillations. The hearing aid was placed on a mannequin and a telephone handset was moved over the microphone. Although no signal was presented, enough sound was leaking from the vented earmold that the proximity of the reflective surface created a feedback loop through the hearing aid. The hearing aid was allowed to oscillate freely for several seconds resulting in the five primary oscillations visible on the blue line in this figure. The feedback canceller was engaged and after 60 milliseconds, the feedback oscillations were cancelled resulting in the orange line shown in the figure. If this had been a real hearing aid wearer, the feedback canceller would have been on from the beginning and the feedback oscillations would have been detected and suppressed before sustained oscillations could occur. In other words, the large spikes in the figure achieved those amplitudes only after at least 200 milliseconds of feedback. If the feedback canceller had been engaged when the telephone was placed next to the hearing aid, the algorithm would have reacted in 60 milliseconds and the spikes would not have had a chance to grow so large. This is a distinct advantage of rapid suppression. The realtime feedback canceller is yet another adaptive feature of Dynamic SoundScene, optimizing the performance of the Liaison hearing aid for maximum comfort in a variety of difficult listening situations.

Wind Noise Manager: Improves Comfort Without Manual Adjustments

Wind noise has traditionally caused problems for hearing aid wearers during outdoor pursuits. Today's active lifestyles involve many potentially windy conditions: golfing, cycling, walking or running outdoors, sailing, and tennis. Uncomfortable sound pressure levels in excess of 100 dB SPL are often created when outdoors. Kochkin in MarkeTrak VI states that only 59% of hearing aid wearers are satisfied with their aids' performance outdoors.[1]

Wind noise is actually turbulence caused by the normal laminar flow of air being disrupted when the wind hits a surface or sharp edge. An example of this effect is shown in Figure 9.

Figure 9
Turbulence Created by Wind

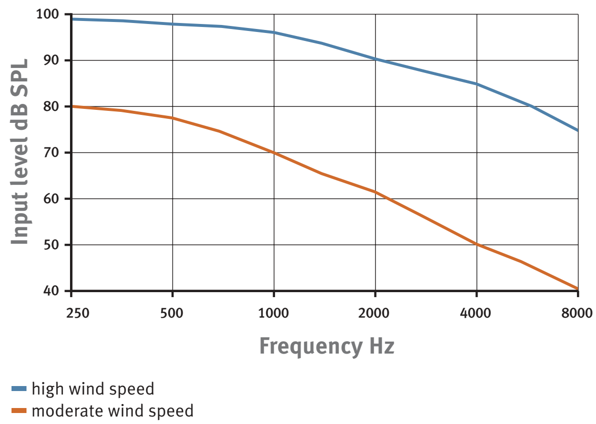


(Courtesy H.Dillon, I.Roe & R.Katsch, NAL)

The object in this figure could have a smooth surface as shown or a sharp edge. The air is traveling in a series of straight parallel lines before it encounters the object (laminar flow). The object then forces the air flow around it creating swirling eddies of high and low pressure on the back side (turbulent flow).[5]

Figure 9 illustrates how wind travelling around an obstacle causes turbulence, which translates into noise at the microphone of a hearing aid. For someone wearing a hearing aid, larger objects such as the head, medium objects such as the pinna, and smaller obstacles such as the tragus or ridges around the microphone inlets, all contribute to the noise produced by wind. Larger obstacles create low-frequency turbulence, medium

Figure 10
Effects of Wind Speed



As wind velocity increases, noise levels increase and the frequency spectrum extends upward. The orange line shows the spectrum of a moderate wind over a hearing aid microphone. The highest energy levels are found below 500 Hz. The teal line shows how the spectrum rises and spreads into the higher frequencies, flattening out all the way to 1000 Hz when wind speed increases.[5]

obstacles create mid-frequency turbulence and small obstacles create high-frequency turbulence. The larger the object, the greater the low-frequency content of the turbulence and the greater the noise that is created. The relative importance of these various obstacles that create turbulence varies depending on the shell style of the hearing aid and the orientation of the hearing aid to the wind. The sum total of noise produced tends to be weighted towards the lower frequencies, particularly in moderate wind conditions, and then spreading upward in frequency as the wind speed increases as shown in Figure 10.

The rapid and intense changes in air pressure due to turbulence surrounding the hearing aid microphone force the microphone diaphragm into extreme fluctuations. These fluctuations generate high sound pressure levels within the device.

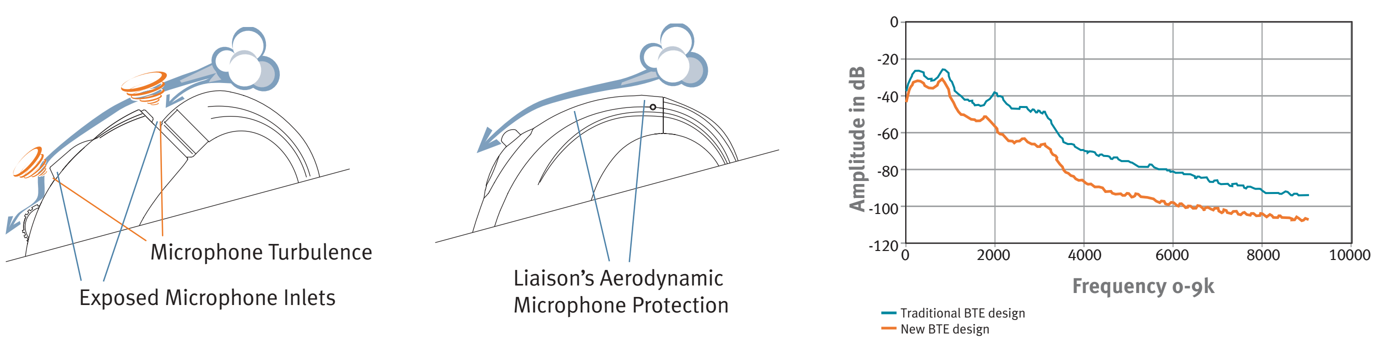
The wind noise manager in Dynamic SoundScene provides an adaptive and progressive response to changing wind conditions across 16 independent channels. Separate wind detectors in each channel engage output reduction as needed to suppress these high sound pressure levels and maintain comfort. In moderate wind, Liaison's wind noise manager responds by reducing gain in the low-frequency channels containing wind energy while preserving the quality and clarity of speech. In high winds, wind noise manager automatically and momentarily reduces output in all affected channels to prevent discomfort. As soon as the wind subsides or the wearer moves out of the windy conditions, output quickly returns to previous levels. As a result, comfort is improved in outdoor environments without the wearer having to make any manual adjustments, such as a volume or program change or turning off the hearing aid.

Liaison's New Aerodynamic BTE Design Reduces Wind Turbulence

Liaison's new BTE shell design also minimizes wind noise due to turbulence. In BTE hearing aids, any protruding edges in the shell design are prone to wind noise. BTE's with exposed microphone ports (Figure 11 – left side) create turbulence as the wind passes over the bumps and

Figure 11

Liaison BTE Designed to Reduce Wind Turbulence



The benefits of Liaison's recessed microphone ports and sleek aerodynamic design when compared to traditional BTE shell styling. Note the turbulence that builds up around the microphone ports of the traditional hearing aid on the left versus the laminar air flow over the top of the new BTE design on the right.

ridges around the microphone inlets. The sleek, aerodynamic Liaison BTE design features recessed microphone inlets (Figure 11 – middle) with a patented acoustically-transparent filter to protect the microphones from moisture and debris as well as wind. The result is an aerodynamic style that significantly reduces wind turbulence.

Figure 11 (right side) shows how Liaison's new shell design changes the wind noise spectrum. When a Liaison hearing aid was placed in front of a fan, the orange output spectrum was generated. A second hearing aid in a traditional shell, running the same Liaison algorithm, was then fixed in the same position and the recorded output is shown as the blue line in Figure 11. The difference between the two curves is due to the change in turbulence surrounding the microphones.

Summary

The success of a hearing aid fitting is related to the number of pertinent listening situations where the device can provide desired performance. More flexible and adaptive hearing instruments are most likely to provide the greatest benefit and improve overall patient satisfaction. Liaison with Dynamic SoundScene directly confronts the important unmet needs of hearing aid wearers as reported in MarkeTrak VI[1]. Dynamic SoundScene integrates and automatically optimizes performance for new listening situations through five key adaptive features:

1. Frequency-gain function
2. Adaptive beamformer
3. Intelligent Noise Reduction 2.0
4. Realtime feedback canceller
5. Wind noise manager

Individually, these features improve performance and provide benefit for specific listening situations. However, the real power of these dynamic components is best realized when they are joined together into the carefully conceived signal processing package known as Dynamic SoundScene.

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