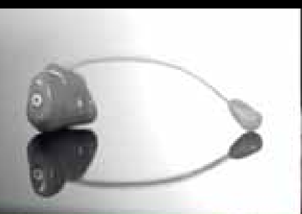
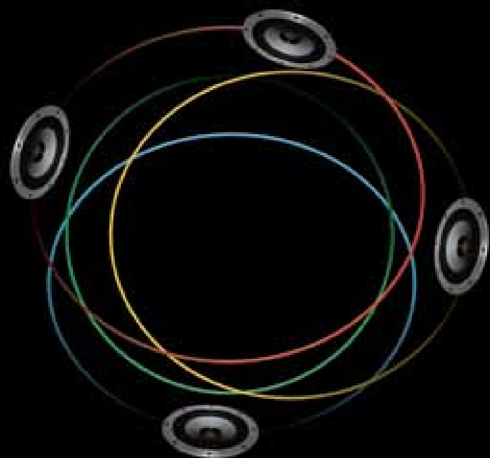
Surround Sound by ReSound

The Technical Background







SURROUND SOUND by ReSound

ReSound

rediscover hearing

THE SURROUND SOUND EXPERIENCE

“Surround sound” is a term to which many people can easily relate. Surround sound plays back recorded sound using sound enhancing techniques and encircling the listener with multiple speakers to create a virtual three dimensional sound environment. It recreates a realistic, full sound picture and gives the listener a sense of actually being in the situation. Likewise, Surround Sound by ReSound™ preserves and conveys acoustic information in such a way that hearing instrument wearers’ surroundings sound natural. However, rather than recreating sound environments, Surround Sound by ReSound helps wearers reconnect with their actual environments. This takes them to a new level of auditory awareness and natural sound perception.

Surround Sound by ReSound is a digital sound processing strategy that is inspired by surround sound at the cinema because, like that, it offers a more “true to life” listening experience. The ReSound® philosophy of using knowledge of the normally functioning ear as the basis for hearing loss compensation is exemplified in the surround sound strategy. It uses advanced technologies to emulate the function and performance of the natural ear. With Surround Sound by ReSound, hearing instrument wearers can enjoy exceptional speech understanding, enhanced awareness of their sound environments, and rich, vibrant, fully detailed sound quality.

A DIGITAL SOUND PROCESSING STRATEGY

The surround sound experience is created by the interplay of advanced technologies that model, clean, balance and stabilize the digitalized signal before it enters the hearing instrument receiver. First of all, the sound entering the hearing instrument is modeled to replicate the way the natural ear processes sound. It is then cleaned, with unwanted parts of the sound, like background noise and wind noise being removed. Another way the Surround Sound by ReSound strategy emulates the natural ear is in how the signal is adjusted to achieve the proper balance between low and high pitches, soft and loud sounds, and sounds coming from different directions. Finally, the signal is stabilized before it enters the receiver, so people don’t have to worry about annoying feedback, even when they put a phone to their ear. The Surround Sound by ReSound signal processing strategy is illustrated in Figure 1.

Figure 1:
The Surround Sound experience is created by the interplay of advanced technologies that model, clean, balance and stabilize the digitalized signal before it enters the receiver.

MODELING BASED ON THE NATURAL EAR

Sound is by nature analog, meaning that it is a continuous stream of vibrational energy. All digital processing conversely breaks the signal into discrete chunks of data in order to perform mathematical operations to treat the sound. As the foundation for any sound processing in the hearing instrument, Surround Sound by ReSound must first model the incoming sound in a digital format. To ensure the right level of detail for a natural-sounding result, the normal-functioning ear serves as the starting point for modeling.

Cochlear frequency resolution with Warp
Hearing instrument fitting procedures assume that the amplification system is capable of frequency analysis suitable for the human auditory system. ReSound solves this challenge with Warp™ processing, which emulates the way frequencies are analyzed by the natural ear. Due to logarithmic coding on the basilar membrane, the human ear’s ability to resolve sounds is best modeled by a system in which the bandwidth of the frequency analysis is nearly constant at lower frequencies and increases proportionally at higher frequencies[1]. The Warp processor uses a mathematical warping function to map frequency components logarithmically to a scale closely corresponding to the auditory Bark scale[2], which incorporates the critical auditory bandwidth as the scale unit[3]. The frequency warped ReSound compression system results in 17, smoothly overlapping frequency bands separated by approximately 1.3 Bark as illustrated in Figure 2.

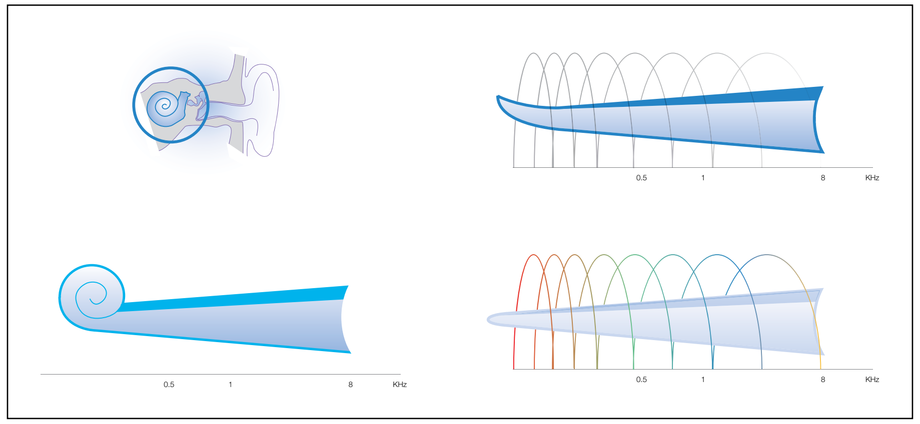


Figure 2:
The Warp™ processor models auditory frequency resolution with non-uniform band spacing approximating the auditory Bark scale. The highly overlapping bands are more closely spaced at lower frequencies and further apart at higher frequencies, which matches resolution in the cochlea.

Analyzing the listening environment

Another way that Surround Sound by ReSound models the sound is via environmental classification. Nearly all modern hearing instruments incorporate functionality that depends on environmental recognition. Therefore, it is important that the system which represents the hearing instrument user's surroundings is consistent with how the user would perceive them. The environmental classification model ReSound has developed considers acoustic factors which are of importance to all hearing instrument wearers: Is there sound above an ambient level?, Is there speech present?, Are there other background noises present?, and What are the levels of the sounds present? In order to answer these questions, the Environmental Classifier employs sophisticated speech and noise detection algorithms based on frequency content and spectral balance, as well as the temporal properties of the incoming sound to determine the nature of the acoustic surroundings. Furthermore, the classification does not occur on the basis of stringent predetermined criteria, but rather on the basis of probabilistic models, resulting in classification of listening environments which has shown a high degree of consistency with listener perception. Acoustic surroundings are categorized dependent on signal-to-noise ratio and sound level into a scheme of seven environments as indicated by the dark ovals in Figure 3. In many real-world situations, the acoustic environment often falls between categories or shifts among closely related environments. The Environmental Classifier takes these frequently occurring situations into account by assigning combinations of environments to the most probable candidates. This means that any adaptation of hearing instrument sound processing based on the environmental classification has a "behind-the-scenes" rather than pronounced or intrusive character, allowing the wearer to always experience transparent and comfortable sound transitions.

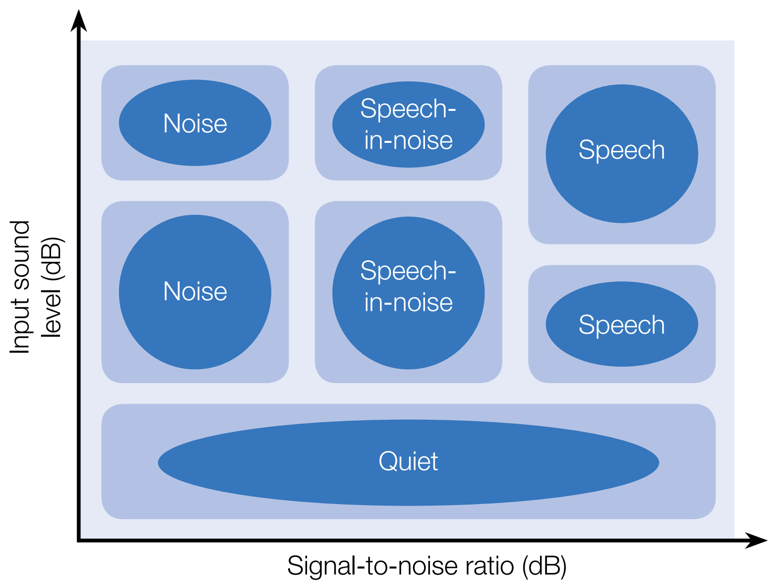


Figure 3:
Sound input to the hearing instrument is analyzed to model the acoustic environments in a digital format. Classification is based on probabilistic models, and ensures that transitions in hearing instrument settings based this information are seamless for the user.

CLEANER, CLEARER SOUND

One facet of successful hearing instrument use is the wearer's acceptance that not all amplified sounds are desirable, interesting or pleasant. Hearing instrument users have traditionally had to resign themselves to hearing sounds not of interest in order to hear those that are of interest, like speech. Examples of sounds that are bothersome for hearing instrument wearers include amplified environmental sounds such as ventilation systems, water running, the background babble of many talkers, and wind blowing across the hearing instrument. Noise generated by components in the hearing instrument itself can also be amplified to the point where it is annoying for users. Surround Sound by ReSound uses processing to diminish or remove sounds that users find distracting or bothersome, in essence to "clean" the sound.

Comfortable listening with less effort

Listening in background noise has far-reaching impact. 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 arduous and exhausting task indeed. Conceivably, 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 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 also provide significant benefit to a hearing instrument user because of the improved sound quality.

Digital hearing instruments have typically aimed to improve listening comfort and sound quality through the use of single microphone noise reduction. NoiseTracker™ II also follows this strategy in order to clean the sound for more effortless listening. This unique algorithm accomplishes a seemingly unsolvable challenge: how to separate speech from noise when both signals usually have energy at the same frequencies at the same time. Complicating matters further is that many noises have speech-like characteristics and many parts of speech have noise-like characteristics. The NoiseTracker II system uses spectral subtraction, one of the most widely used methods for enhancement of noisy speech in audio applications[4]. The concept of spectral subtraction is to subtract the short-term noise spectrum from the total signal, leaving only the speech portion. 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.

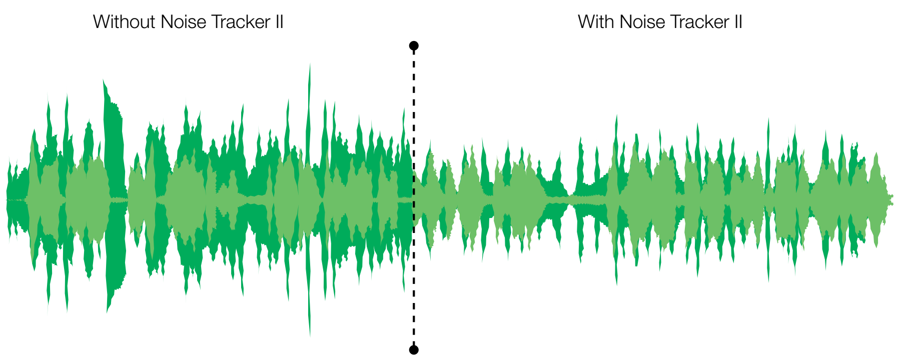


Figure 4:
Speech in speech babble is the most difficult situation for noise reduction systems to handle. The NoiseTracker II spectral subtraction approach can effectively reduce the level of the background noise (dark green) without affecting audibility of the desired speech signal (light green).

Whereas NoiseTracker II cleans sound in the hearing instrument user's surroundings, expansion takes care of sound produced by the hearing aid itself in quiet situations. Expansion reduces audibility for very soft sounds. With this type of processing, the amount of gain decreases as the input level decreases, as illustrated in Figure 5. Also commonly known as "squelch" or "microphone noise reduction", expansion is intended to keep the hearing aid from amplifying very soft sounds which are not of interest to the wearer, such as internally generated noise or very low level environmental sounds. In this way, the hearing instrument itself sounds quiet, providing the best backdrop for pleasurable listening.

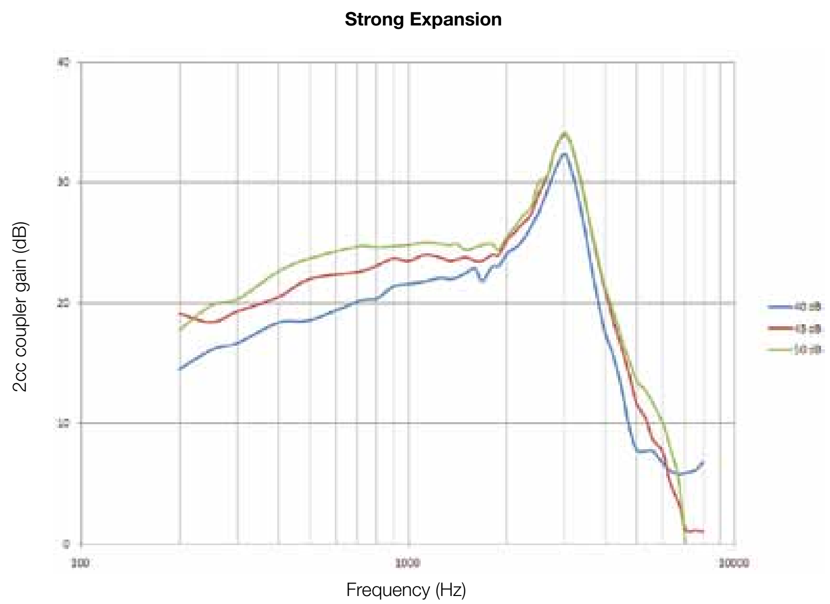


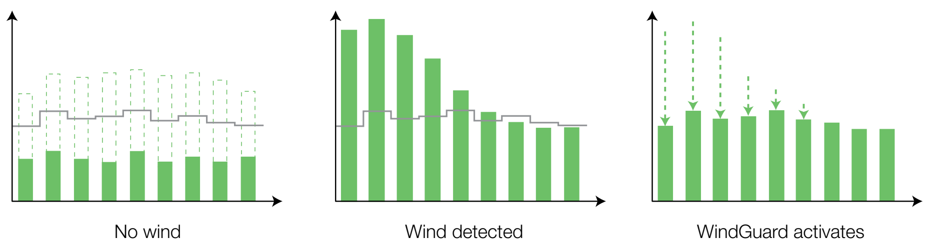
Figure 5:
Coupler gain for the Alera 61DW with expansion set to "Strong". This shows how gain decreases with decreasing input level. The effect is most apparent at lower frequencies, where the expansion kneepoints are highest.

Easier listening - even in wind

On-the-scene news reporters are often filmed speaking into microphones covered with large, fuzzy caps. Without this effective mode of protection, any breeze blowing across the microphones would be amplified and severely degrade the quality of the recording. Hearing instrument wearers who venture outside endure the same degradation in sound quality and masking of desired sounds when wind hits the instrument microphones. While a protective cap or cover such as the fuzzy ones used on handheld microphones could prevent wind from reaching the hearing instrument microphones, such a solution is obviously both impractical and cosmetically unacceptable. Another line of defense against wind noise is WindGuard™. This processing, available in dual microphone hearing instruments, effectively cleans the signal for wind noise without impacting audibility of other sounds or causing the overall sound quality to be unnatural.

The goal of WindGuard is to accurately detect the presence of wind noise, and then apply enough gain reduction in the frequency bands where wind is detected to provide listening comfort for the hearing instrument user, without disrupting the gain levels of the frequency bands that are unaffected by wind. The wind noise detector ingeniously relies on the fact that turbulence caused by wind blowing over the microphones of a dual microphone instrument will be uncorrelated, unlike desired sounds. Whenever wind noise is not found to be present, the instantaneous level of the sound input levels in each of the Warp bands is updated and stored. The amount of gain reduction applied depends on the stored level. Thus, it varies with the environment and the level of the wind noise, making the reduction as personalized as possible to the situation without sacrificing audibility for other sounds. The end result: the hearing instrument user has a very natural sounding experience, with soft wind noise in the background and preserved audibility for other sounds in the environment.

Figure 6:
WindGuard reduces gain to the continuously updated level of environment sound without wind noise. This approach helps preserve audibility for desired sounds and natural sound quality.



A DELICATE BALANCE

It is universally acknowledged that the primary complaint of hearing instrument users is hearing in noise. Rehabilitation as well as hearing instrument design focuses on improving the hearing impaired individual's ability to understand speech in noisy conditions, assuming that resolving this primary issue will lead to success and satisfaction with hearing instruments. But the fact is that hearing instrument users may choose not to use the enhancements proven to increase speech understanding in noise because the sound picture becomes unnatural and noisy. It can be difficult to reconcile evidence that the hearing instruments are beneficial with reports that they are unappreciated or unused in many situations. However, hearing instrument wearers' overall satisfaction is closely tied to the number of their daily listening environments in which they can function satisfactorily[5] rather than benefit determined under controlled and predictable conditions. It was estimated that hearing instruments would need to be of use to wearers in at least 70% of the listening situations encountered in order for them to assign a high overall satisfaction rating.

Implicit in the Surround Sound by ReSound philosophy is to recreate natural perception of sound environments in addition to providing speech in noise benefit in the traditional sense. This involves putting the individual wearers back in touch with as many of their wide-ranging listening situations as possible. The list of situations in which an individual experiences hearing difficulties but still wants to be aware of their surroundings might include talking in small groups and at parties, communicating at meetings, hearing the instructor at a fitness class, hearing announcements over PA systems, orientation in traffic, speaking with small children, and many others. To accomplish this, the sound processing must compensate for the loss of the normal hearing experience by balancing varying aspects of the incoming sound. the soft versus loud sounds, speech versus noisy sounds, high versus low pitches and sounds coming from in front versus in back of the hearing instrument wearer must be carefully balanced. This enables and enhances speech recogning without compromising on the richness and comfort of the listening experience.

WDRC – the perfect balance

A hallmark of sensorineural hearing loss is a reduction in sensitivity for softer sounds, but preserved sensation of loudness for louder sounds. As shown in Figure 7, this can also be described in terms of reduced dynamic range. Restoring audibility for the soft sounds outside of the individual's dynamic range should be accomplished without boosting louder sounds above the individual's upper range of comfort. This implies that different amounts of gain will be necessary depending on the input level to achieve the appropriate balance between soft and loud sounds. Because loudness perception is most dissimilar from normal loudness perception at lower input levels, more gain is needed for soft sounds, and progressively less gain is required for louder sounds. In addition, loudness growth functions are frequency dependent, which also points to a need for a multichannel system replicating the natural ear's frequency analysis in order to balance loudness in a suitable way across frequencies.

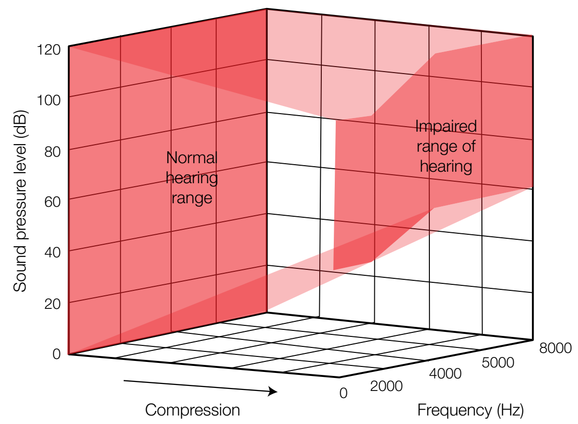


Figure 7:
WDRC balances loudness by applying frequency and input level dependent gain to compress the wide range of daily sounds into the smaller dynamic range of the hearing instrument user.

ReSound pioneered Wide Dynamic Range Compression (WDRC), offering the first system to account for the loudness growth associated with sensorineural hearing impairment by applying progressively less gain with increasing input levels. Compression parameters in the ReSound system were carefully selected to support compensation for abnormal loudness growth in a way that emulated studies of how compression is carried out in the human cochlea. These include low, frequency dependent compression kneepoints, compression ratios of up to 3:1, and syllabic time constants. An essential part of restoring loudness balance is the prescription of gains for the WDRC system. To provide a reliable starting point, a fitting algorithm based on individual psychoacoustic measures of loudness growth was initially used. Years of clinical experience encompassing thousands of fittings with this procedure provided the foundation for development and refinement of the threshold-based Audiogram+ fitting algorithm, which provides the basis for fitting ReSound hearing instruments today.

Directionality like the open ear

Another feature contributing to the surround sound experience which provides balance in the amplified sound is the Directional Mix™. This technology gives wearers the advantages of directionality without compromising on sound quality. The idea behind the Directional Mix is to promote good speech intelligibility and natural sound quality by preserving the acoustic characteristics of the open ear. This unique technology considers the individual's audiometric data and physical properties of the device in calculating a balanced, personalized mix of directional and omnidirectional processing. Incoming sound will be preprocessed, with higher frequency components delivered to the directional system while an omnidirectional response is maintained in low frequencies. This processing strategy replicates directionality patterns of the unaided ear even for BTE microphone placement, contributing to natural perception of sound for the user. Figure 8 illustrates how the Directional Mix preserves open ear directional characteristics. The left panel shows the open ear directional response for 4 frequencies measured on KEMAR. Note how the response is omnidirectional with only subtle asymmetries for the 2 lower frequencies while in the higher frequencies there is relatively more amplification for frontal incident sound than for sound coming from other directions. The right panel presents the same measurement performed with a BTE with Directional Mix. This technology ensures that the directional characteristics with the hearing instrument in situ are a good match to those of the open ear.

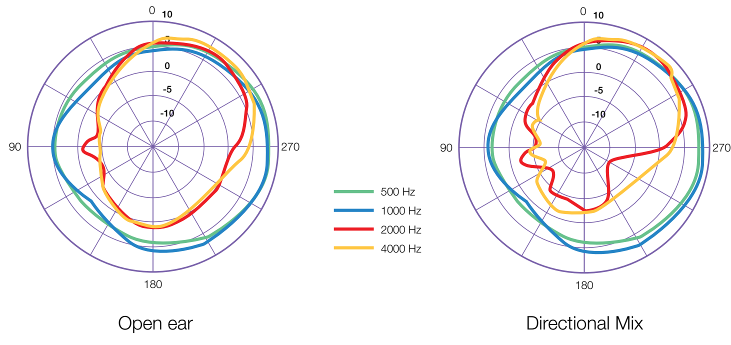


Figure 8:

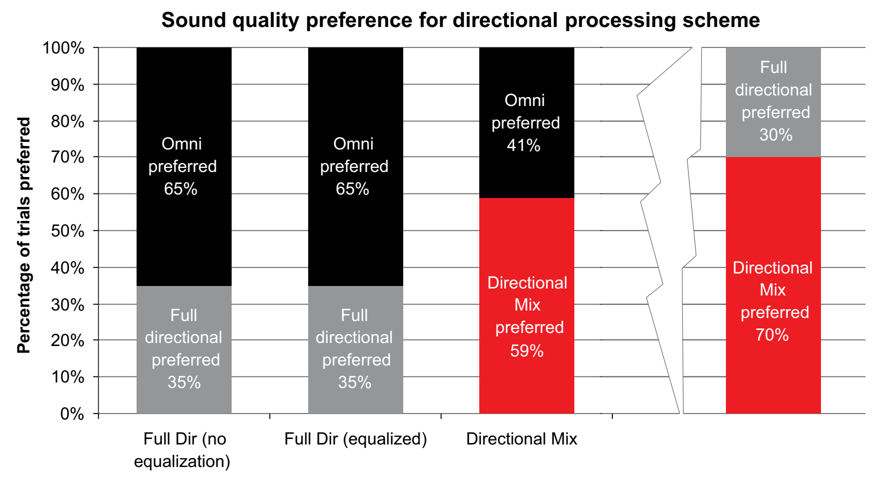
The Directional Mix provides directivity patterns that are similar to the open ear. The response is omnidirectional at lower frequencies and directional at higher frequencies.

The Directional Mix also solves byproducts of directional processing that have limited even the most advanced adaptive systems, including added noise from low frequency equalization, distortion of near-field sounds such as own voice and wind, as well as disruption of the low frequency interaural time differences which are crucial for localization. These shortcomings of directionality can be perceptually significant enough to cause wearers not to use an otherwise beneficial feature of their hearing instruments. In their laboratory study investigating the impact of visual cues on directional benefit, Wu and Bentler[6] reported that many individuals fit with an equalized directional response experienced a “hissing sound”. In a subsequent field trial[7] with the same participants and hearing instruments, these investigators determined that loudness and internal noise were the most important predictors for preference of omnidirectional microphone mode over directional. Other studies have also demonstrated strong preferences for omnidirectional microphone mode even in situations where directional processing should be of benefit[8,9]. The Directional Mix provides a directional pattern closer to a person’s own, thereby striking a natural balance between environmental awareness and directional advantage. The result is sound closer to the way hearing instrument wearers remember their acoustic world – what they describe as “natural” sounding.

Evidence has been mounting in support of the benefits afforded by the Directional Mix. Groth and colleagues[10] reported on the effects of directional processing schemes on sound quality. These investigations used a double-blind design in which hearing-impaired listeners expressed a preference for the Directional Mix, omnidirectional processing or a conventional full directional response. As illustrated in Figure 9, listeners indicated an overwhelming preference for the sound quality of omnidirectional processing over full directional with or without low frequency equalization. When comparing to the Directional Mix, listeners preferred this processing over full directionality more than twice as often.

Figure 9:

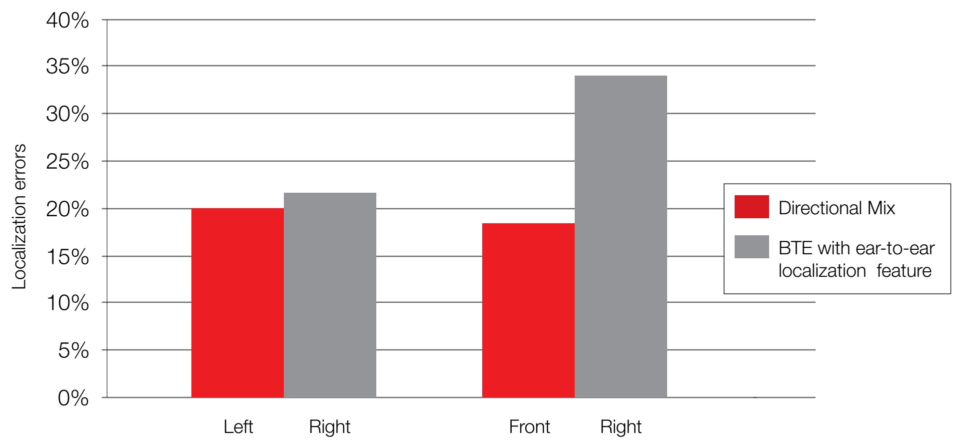
Hearing instrument wearers expressed a sound quality preference for a directional response when achieved with the Directional Mix twice as often as traditional directionality.



In addition to sound quality, the Directional Mix has also been shown to contribute to localization ability[11]. Van den Bogaert and colleagues asked hearing impaired listeners to perform localization tasks wearing BTE hearing instruments with the Directional Mix and BTE hearing instruments featuring an ear-to-ear algorithm designed to enhance localization via wireless coordination of settings bilaterally. While performance for right-left localization was similar in both conditions, participants made more than twice as many front-back confusions when wearing the devices with the ear-to-ear feature than when wearing the instruments with the Directional Mix (Figure 10).

Figure 10:

Hearing impaired listeners fit with BTE devices with the Directional Mix performed better on a localization task than when fit with BTEs featuring ear-to-ear processing intended to improve localization.



Better speech recognition in surround sound

Given that directionality is the only proven technology to improve speech understanding in noise[12] the “more-is-better” approach of maximizing directionality across frequencies might lead one to expect better speech recognition in noise performance with full directionality than with the Directional Mix. On the other hand, articulation index theory would predict a negligible difference between the two types of processing, as added audibility in the lower frequencies should represent only a modest contribution to intelligibility[13]. Figure 11 shows results from a clinical investigation in which 18 participants were fit with devices in which they could switch between an omnidirectional, a full directional, and mixed directional response. Both full directionality and the mixed directional response provided equivalent directional benefit as predicted by the directional enhancement of speech important frequencies for both types of processing. These results support the notion that the lack of directionality in the low frequencies has virtually no impact on directional benefit. Due to the balance provided by the Directional Mix, all wearers can be provided directional benefit with no trade-off between audibility of low frequencies and noisiness.

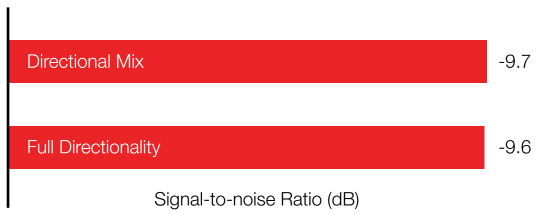


Figure 11:
Results of the Hagerman speech in noise test with 18 test subjects. Participants wore either BTE or RIE instruments fit with custom earmolds or occluding eartips. Directional benefit with the Directional Mix was equivalent to that obtained with full directionality.

Speech recognition and sound quality

It is typical to see speech intelligibility with hearing instruments discussed separately from sound quality. And yet it is unlikely that hearing instrument wearers perceive varying aspects of listening with their devices in such a compartmentalized fashion. What does seem likely is that “natural” sound quality relates to the holistic experience of hearing and perceiving sound, encompassing fidelity and perceptual dimensions such as “fullness”, “brightness” and “loudness”[14] as well as hearing and processing speech. When sound is experienced as natural, the listener can effortlessly segregate and group the continuous stream of sound in the environment. A hearing instrument achieves good sound quality by preserving spectral, dynamic, and temporal aspects of the input sound to an extent that higher level auditory and cognitive processing can interpret these physical attributes and construct auditory environments. For example, in a busy office environment, hearing instruments with good sound quality would allow the wearer not only to hear that people were talking and that others were typing, but also to perceive the spatial relationships among these sound sources and shift attention from one to another. Not only would this give the user a natural perception of the environment, it would also aid in speech intelligibility. Freyman and colleagues[15] found that the improved intelligibility observed when the speech and interfering sound are spatially separated was attributable to the perceived locations of the signal of interest and noise. In other words, knowing where to direct one’s attention makes it easier to listen to the desired signal. It is not just that the sound sources are spatially separated, but that the listener perceives them as such.

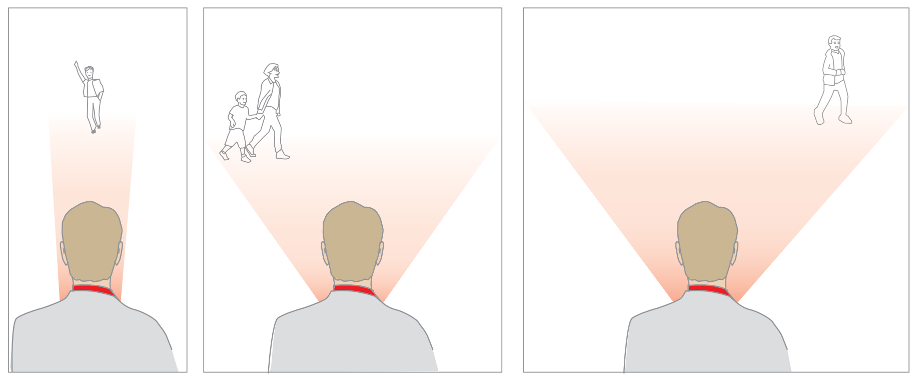
Hearing in noise: a wide array of options

ReSound hearing instruments offer a multitude of advanced directionality modes that work with the Directional Mix to improve speech understanding in noisy situations while maintaining natural sound quality for enhanced spatial hearing in all environments.

Adaptive Directionality with AutoScope

AutoScope™ is the most advanced adaptive directional algorithm available. This system analyzes the inputs to the two microphones to determine the direction of arrival of sounds and to estimate whether more focused directionality is likely to be of benefit. Like its predecessor, MultiScope™ directionality, AutoScope adaptively cancels the strongest noise source behind the user, and can simultaneously act on noise occurring at different locations when the frequency content of the noise sources differs. In addition, AutoScope monitors the relative levels of the sounds entering the front and rear microphones to automatically adjust the width of the directional beam. The stronger the signal coming from the front, the more narrow the beamwidth will be adjusted. Conversely, as the level at the front microphone decreases, the beamwidth is widened to catch more of the surrounding sounds. This effect is like zooming in on a single talker located in front of the hearing instrument wearer and zooming out when others off to the sides are speaking, as illustrated in Figure 12.

Figure 12:
AutoScope directionality automatically narrows the width of the directional beam to focus on a speaker in front, and expands the beam when speakers are not directly in front of the user.



Asymmetrical Directional fittings

The Directional Mix also enhances the unique asymmetrical fitting strategy introduced by ReSound. An asymmetric fitting is one in which a hearing instrument with directionality is fit on one ear, the “focus” ear, and a hearing instrument with an omnidirectional response is fit on the other, the “monitor” ear. This fitting approach is based on the idea that information presented to the two ears that differs in signal-to-noise ratio and audibility will be perceived as a unified binaural image based on which ear has the better representation of the signal of interest. This allows the wearer to take advantage of the improved signal-to-noise ratio benefit of directionality without significantly decreasing audibility for sounds arriving from behind. No manual switching of directional mode is required, which eases demands on the wearer to be able to analyze the environment, understand when directionality might be advantageous, and physically select the appropriate response. Also, no automatic switching of directional mode takes place, which ensures that one of the devices will always be in the preferred microphone mode regardless of the listening situation. Asymmetric fittings have been shown to provide directional benefit equivalent to bilateral directional fittings under laboratory conditions[16,17,18], better acceptance of background noise compared to bilateral omnidirectional fittings[18] and increased ease of listening compared to bilateral directional fittings[17]. This supports that an asymmetric fitting is an excellent way for users to effortlessly derive benefit from directional technology while maintaining full awareness of their surroundings.

Although an asymmetric fitting can be done with other hearing instruments which offer both omnidirectional and directional microphone modes, they will be prone to the same disadvantages listed earlier, including noisiness and overamplification of nearby sounds. In addition, adaptive directional processing can negate the ease of listening advantage of the asymmetric fitting, as the constantly shifting directional patterns preclude adaptation to the acoustic cues. An additional issue specific to advanced directional systems currently available is that the processing delay between the instrument in directional mode will be longer than that of the instrument in omnidirectional mode. The resulting asynchrony of the cues provided to the two ears disrupts spatial hearing and awareness. As discussed, this negatively impacts the holistic impression of the auditory environment and perceived sound quality.

Balancing volume and noise reduction

Most people encounter many dynamic listening environments throughout the day, with each offering its own unique challenges in terms of listening and communicating. The demands of hearing in the car versus the grocery store or coffee shop are different. Some environments are favorable for a hearing impaired listener as the signal-to-noise ratio is positive and sound levels comfortable, others are much more difficult. As mentioned previously, overall hearing aid satisfaction has been correlated to a hearing aid’s ability to provide improved hearing in multiple listening environments[5], which is to be expected considering that they are intended for continual wear. All hearing instrument fittings begin with some sort of prescription of settings based on client data, usually an audiogram. Yet the presumption that one set of hearing aid parameters will meet the listening needs of an individual in all conditions is clearly not met. It is reasonable to say that the goal of fitting prescriptions is to provide amplification for optimum speech understanding while ensuring comfort for loud sounds. Even if achieved for a particular individual, this would not take into account that the wearer might want to enhance or diminish different aspects of the amplified sound in different situations. For example, a hearing instrument wearer might desire more volume than prescribed in an important meeting at work, but wish for less volume when relaxing with the newspaper on the train ride home several hours later.

Individually tailored adaptive noise reduction

An automatic, personalized volume control such as the Environmental Optimizer™ solves some of the negative and impractical issues related to frequent or necessary manipulation to a manual volume control or program switch. However, this functionality combined with individually tailored adaptive noise reduction is an extraordinary solution to the complaints of hearing aid users—both the need to adapt to multiple listening environments and comfort in noise are balanced. Environmental Optimizer™ II automatically adjusts both the gain and NoiseTracker™ II settings dependent on the listening environment identified by the hearing instrument classification system. Based on the situation dependent volume preferences of hearing instrument wearers in a study at Oldenburg University[19] as well as internal research studies, the Environmental Optimizer™ prescribes volume offsets to the prescribed frequency response and optimized NoiseTracker™ II settings for 7 different acoustic environments. Each of these settings can be personalized for the individual through the ReSound Aventa® 3 software.

STABILIZE

To complete the surround sound experience, it is necessary to make sure the output from the hearing instrument is stabilized. This process corrects for any unnatural or undesirable consequences that can occur with amplification or digital sound processing. For example, stabilizing keeps users from being subjected to unpleasant and embarrassing feedback. The tiny hearing instruments of today necessarily place the sound inlet in close proximity to the sound outlet. If the sound returning to the microphone ports from the ear canal where the sound is delivered is not sufficiently damped relative to the gain applied by the amplifier, oscillatory feedback will occur. This is a dynamic issue, as the damping in the pathways from the ear canal to the hearing instrument microphone change continually during daily use situations. DFS Ultra™ with built-in WhistleControl™ provides the necessary stability to eliminate such issues.

Since last reported in 2004, user satisfaction with hearing instruments in terms of "whistling and feedback" has shown a significant 12% improvement[20]. This period of time coincides with the proliferation of feedback cancellation algorithms in digital hearing aids, which suggests that manufacturers are on the right track to improve users' experience with hearing aids with this type of processing. Feedback cancellation processing can increase the amount of gain available for a particular fitting by 15 dB or more in a static situation where there are no dynamic variations in the feedback path due to client movement or changes in the acoustics of the environment. However, feedback cancellation systems are limited in their ability to operate effectively in dynamic real-world situations. Factors such as room reverberation and nonlinearities of the hearing instrument can cause the very whistling and chirping that the processing is intended to eliminate. To complicate matters further, most systems cannot distinguish between actual feedback and external sounds that may be correlated with feedback. Such sounds include tones, beeps and musical notes, and the failure of the feedback cancellation system to recognize these sounds often results in disturbing echoing or ringing artifacts. As a result, they must compromise on balancing performance in terms of gain, sound quality and critical situations such as phone usage. For example, providing the desired amplification means the wearer may have to tolerate poor sound quality as the system erroneously attacks non-feedback sound inputs.

ReSound technology has long led the field in immunity to such processing artifacts with its use of two cancellation filters, adaptive filter constraints and separate cancellation filters for dual microphone instruments. A new generation of feedback cancellation, DFS Ultra, represents a major breakthrough in feedback cancellation processing. In addition to technical enhancements that enable even more accurate feedback path modeling, a new component of the system is input signal modeling. Unlike other systems which only attempt to model the feedback path, this unique functionality also maintains a representation of the sound entering the hearing instrument (Figure 13). The advantage of this component is that the system can more easily distinguish between feedback and non-feedback sounds, vastly improving the dynamic behavior of the system. Important everyday sounds like phone rings, alarm beeps and music can be amplified to desired levels without being mistaken for feedback.

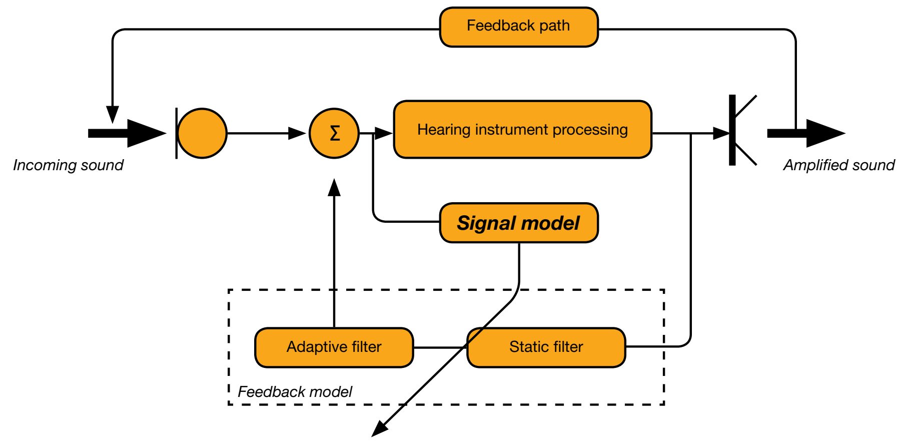


Figure 13:
DFS Ultra analyzes and models the incoming sound and uses this information to prevent the feedback cancellation filters from attacking desired sounds. This results in excellent feedback cancellation performance without creating disturbing sound quality artifacts.

DFS Ultra also incorporates WhistleControl to ensure feedback-free performance in all daily life situations, including when a phone is held up to the hearing instrument. Because the feedback cancellation is constrained to not adapt to signals that are very disparate from the feedback pathway model, it is possible for intermittent feedback to occur in situations like this. WhistleControl restores the desired response when feedback is imminent (Figure 14). The combination of DFS Ultra in effect solves the most challenging dilemma of the last decades' growth in popularity of open fittings – how to provide sufficient, undistorted high frequency gain without feedback.

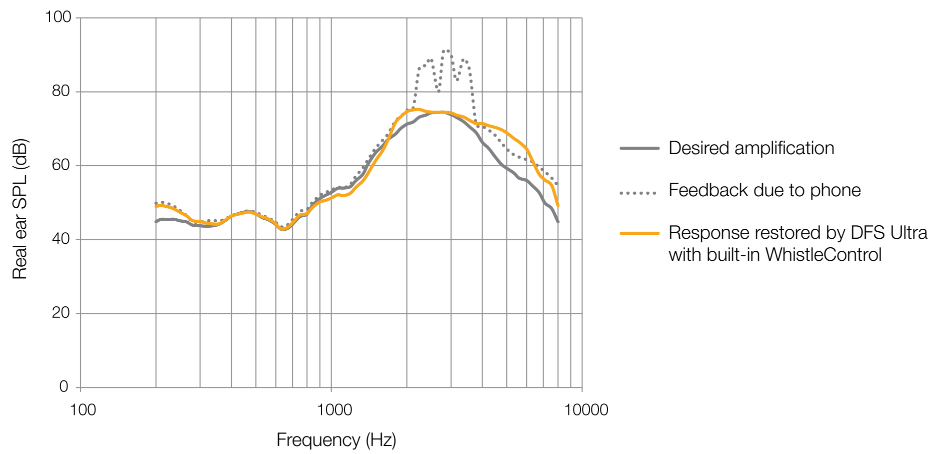


Figure 14:
DFS Ultra with built-in Whistle Control eliminates feedback in critical situations like phone use. The feedback which occurred without DFS Ultra and Whistle Control (dotted line) was eliminated and the response restored to the desired amplification when this feature was activated.

SUMMARY

Surround Sound by ReSound™ provides a natural-sounding listening experience to wearers. It can be likened to going from listening to stereo speakers to a sophisticated surround sound audio system, and translates to increased and effortless awareness and orientation in listening environments. The interplay of advanced technologies model, clean, balance and stabilize the sound in ways designed to emulate the natural ear. As a result, Surround Sound by ReSound provides full and detailed sound quality and delivers important acoustic cues for localizing sounds, which facilitates speech understanding even in challenging situations.

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