

RESOUND ALERA™: A SURROUND SOUND CONNECTION TO THE WORLD

Tammara Stender, Au.D.

ABSTRACT

Excellent sound quality is paramount for hearing aid user satisfaction. To further enhance the hearing instrument user's listening experience, ReSound has improved on its Surround Sound by ReSound technology package. In the ReSound Alera product family, the Surround Sound package builds on its Warp™-based technology with a wide range of personalized directional options, the innovative new DFS Ultra feedback control system and Environmental Optimizer II, which automatically optimizes gain and noise reduction settings according to the listening environment. The end result of these features is a seamless listening experience that provides a strong connection between the individual hearing aid user and the auditory environment.

User satisfaction with hearing aids is directly related to the signal processing and sound quality of the devices. MarkeTrak VIII¹ reported six of the eleven parameters associated with hearing instrument satisfaction were related to the processor and sound quality of the instrument. Yet the data indicated that none of these six items received a satisfaction rating above 80%. While improvements in satisfaction ratings have been observed compared to those obtained in 2004, there is still a clear need for hearing instrument processing that provides optimal sound quality and user satisfaction.

To address this need, ReSound has developed the next generation of Surround Sound by ReSound signal processing in the new ReSound Alera product line. The aims of the advanced Surround Sound by ReSound technology package are to enhance spatial hearing and to deliver the best sound quality in the widest range of listening environments. Surround Sound by ReSound in the ReSound Alera includes Warp™ signal processing, a wide array of directional options with the Surround Sound Processor, improved feedback management through DFS Ultra and automatic gain and noise reduction adjustments through Environmental Optimizer II. The improvements in these algorithms combine to create a technology package that is designed to provide the most natural listening experience through hearing instruments to date.

WARP-BASED SIGNAL PROCESSING FROM A MORE POWERFUL CHIP

The best signal processor for sound is the human auditory system. ReSound realizes that while no machine can process sounds equally as well, it is possible to incorporate knowledge of the mechanisms of the auditory system into a signal processor that replicates it as closely as possible. Warp™-based signal processing involves a digital signal processing technique called frequency warping, which approximates the frequency resolution of the human auditory system. Frequency components are mapped logarithmically to a scale that corresponds to the auditory Bark scale² which incorporates the critical bands of the auditory system³. The end result is ReSound's compression system, which is divided into 17 smoothly overlapping frequency bands. These bands, separated by about 1.3 Bark, resolve the frequency content of sounds in a similar manner to the human ear. While Warp™ processing with Wide Dynamic Range Compression (WDRC) is not new to ReSound products, it is improved through its incorporation into a new, faster integrated circuit introduced in the Resound Alera product family. The new ReSound Range chip enables the hearing aid user to enjoy the benefits of fast signal processing and improved algorithms while experiencing the actual listening environment in a more natural, less distorted way.

ENHANCING THE SURROUND SOUND EXPERIENCE THROUGH DIRECTIONALITY

In an effort to provide the most natural listening experience possible, Surround Sound by ReSound offers a unique solution in directionality with the Surround Sound Processor. The incoming signal is differentially processed based on its frequency content to approximate the unaided human ear's natural directional characteristics. In this manner, high frequencies are processed as a directional response and low frequencies are processed as an omnidirectional response, and both are then mixed to provide the final response. By processing low frequencies as omnidirectional, the spectral characteristics of these sounds are preserved. This allows the user to take advantage of natural between-ear timing differences which aid in sound localization and promote better sound quality. In addition, other problematic side issues that traditionally have been inherent in directional hearing instrument use are alleviated through this split-band directional approach. Noise that results from the equalization, or "bass boost," of the low-frequency roll-off of directional amplification is no longer an issue, since the low-frequencies are processed as omnidirectional. This is a clear improvement, since the audible noise floor from bass boost has been suggested to detract from the overall hearing benefit^{4, 5}. A recent study also revealed that the directional characteristics in Surround Sound by ReSound doubled subjects' preference for directional processing^{6,7,8}, which may result in better hearing instrument acceptance and benefit in noise.

In the ReSound Alera, the Surround Sound Processor is personalized for each individual patient. The "blending point," or the optimized frequency at which the processing changes between omnidirectional and directional, is automatically calculated for each individual's hearing thresholds by the Aventa3 fitting software. The blending point is derived by averaging the person's hearing loss thresholds for 250 and 500 Hz, and is also dependent on the hearing instrument model selected. In the fitting software, the blending

point is expressed as the "Directional Mix" setting. As suggested by the name, a high directional mix indicates that the setting provides directionality over a relatively broader frequency range than a low directional mix. Directional Mix settings are Very Low, Low, Medium, and High. The Directional Mix is automatically set to the same level for each program, but may be changed in an individual program through the Directional Mix pull-down menu in Aventa3. In Figure 1, a higher Directional Mix setting would have a greater proportion of directionality than a lower setting.

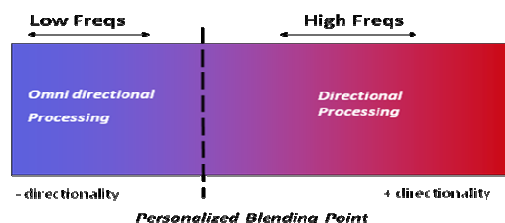


Figure 1. The Surround Sound Processor features a customized blend of directionality and omnidirectionality.

DIRECTIONAL OPTIONS FOR PERSONALIZED FITTING

There are five directional options offered in the ReSound Alera. All are based on the Surround Sound Processor and are designed to be fitted according to the hearing instrument user's needs. These options include omnidirectional, fixed directionality, MultiScope Adaptive Directionality with AutoScope, SoftSwitching Adaptive Directionality and Natural Directionality II. Omnidirectional settings are best for quiet situations, and may be useful in Music and Traffic programs. Fixed directionality may be useful for noisy situations, or for patients who prefer a more traditional directional approach. MultiScope Adaptive Directionality selects omnidirectional and directional responses based on the intensity and direction of the noise. Beamwidths for the directional response include narrow, medium and wide, and with the option of AutoScope, can be automatically selected based on the characteristics of the listening situation. For clients who may not care to manually switch

programs but who may benefit from directionality, SoftSwitching Adaptive Directionality slowly transitions between omnidirectional and directional settings based on an analysis of the acoustic environment.

Natural Directionality II maximizes environmental awareness by combining the benefits of the Surround Sound Processor with asymmetrical directional processing. An omnidirectional pattern is applied to one ear, termed the “Monitor” ear, and a fixed hypercardioid directional pattern is selected for the other ear, termed the “Focus” ear. The Focus and Monitor ears are automatically designated through the Focus Ear Calculator in Aventa3. Asymmetric directionality offers improved ease of listening and awareness of surroundings as compared to bilateral directional fittings⁹ without degrading directional benefit^{9,10} due to the availability of environmental sound inputs from the monitor ear. Further, Natural Directionality II synchronizes the processing delay on both ears in order to preserve important localization cues. This unique fitting strategy represents the best solution for clients seeking improved hearing in noisy situations without losing the ability to monitor the listening environment.

The ReSound Alera offers this wide array of directional options to allow for the most personalized fitting for the individual hearing instrument user. Whether the user’s lifestyle would be best accommodated with symmetric or asymmetric fitting, or with fixed or adaptive directionality, the optimal selection is easily accessed through the fitting software. Together with the personalized directional mix, ReSound’s directional options promote individualized, effective fittings to optimize speech understanding and environmental awareness.

NO COMPROMISES WITH DFS ULTRA

Feedback control has come a long way with digital hearing instruments, and can take part credit for the popularity of open fittings in 21st century hearing instrument fittings. However, despite the advancements, current feedback

cancellation systems impose trade-offs between gain, freedom from feedback in all situations, and sound quality. As shown in Figure 2, the system that attempts to eliminate feedback in all situations may have poor sound quality. DFS Ultra combines the most advanced feedback control systems available, allowing the optimum performance on all dimensions. ReSound’s new incarnation of its digital feedback suppression (DFS) system has been re-engineered to provide the most feedback-free sound experience to date. DFS Ultra models both the feedback path and

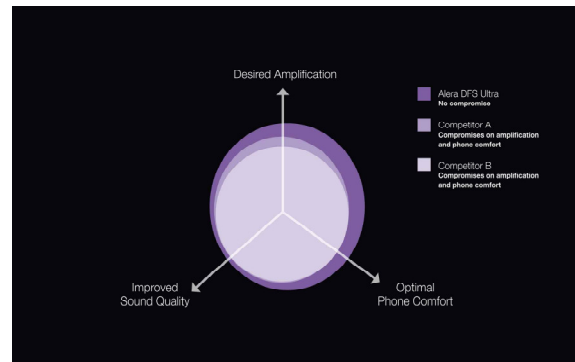


Figure 2. Feedback cancellation systems may perform well in one dimension of usage but be poor in others. DFS Ultra performance excels with no compromise.

the external input signal, to improve the accuracy in identification of feedback from other tonal input sounds. Once it correctly separates feedback from other tonal input sounds, phase cancellation is applied to eliminate the feedback alone. The result is fewer sound quality artifacts, which can occur when traditional phase cancellation feedback control systems incorrectly try to cancel out tonal input sounds instead of feedback. In addition, new improvements to the precision and scaling in the models result in a more accurate feedback cancellation signal.

Whistle Control serves as an additional layer of feedback control, by reducing the occurrence of feedback that arises from dynamic, transient situations such as hugging and using the telephone. Feedback that manages to elude the DFS system is controlled via gain reduction in the frequency range of the feedback. The signal itself is unaffected while the extra gain from the feedback spike is reduced. Thus, Whistle

Control serves as an emergency option, for the most difficult feedback-provoking events which are not completely resolved through DFS alone. These algorithms together, which comprise DFS Ultra, create a feedback control system that eliminates artifacts to provide the best sound quality without sacrificing user gain.

LISTENING COMFORT FOR EACH INDIVIDUAL WITH ENVIRONMENTAL OPTIMIZER II

Hearing aid users are thrust into a diverse set of listening environments every day. Traditionally, the only way to adjust the prescribed gain for each listening situation was for the hearing instrument user to manually change the volume control setting. This solution was tedious for some users, especially since it had to be applied with every change in listening situation. With the advent of hearing aids with WDRC, the hearing aid was able to adjust the gain based on the input level of the sound. Yet this overall change in gain dependent solely on the input level still did not resolve the main underlying issue that individuals may prefer different amounts of gain in different listening situations, and may have different tolerance levels for listening in noise. It has been proposed that success with amplification may be determined more by the wearer's willingness to listen in background noise than speech understanding¹¹. In noisy environments, automatic gain changes may be helpful in achieving listening comfort and promoting hearing instrument success. Hearing instrument usage and success may also be dependent on the effectiveness of the noise reduction strategy¹². In an effort to create the most personalized, real-world sound experience, ReSound has combined the automatic gain changes of the Environmental Optimizer with the benefits of NoiseTrackerII noise reduction to create Environmental Optimizer II.

When the Environmental Optimizer II feature is selected in the software, gain sliders are visible in 7 different environments, as listed in Table 1. The Environmental Classifier measures the signal-to-noise ratio (SNR) and

intensity level of the input sound to determine which of the 7 listening environments is most likely at that given time. The overall gain in the hearing aid can be automatically increased or decreased as the listener encounters different listening environments while using the hearing instrument. The levels to which the hearing aid adjusts to the environment can be set in two ways: they can be based on default, average data, or they can be manually set in the software based on the hearing instrument user's preferences. When a new listening environment is identified, the automatic gain change is implemented in less than 1.5 seconds.

Sound Input	Loudness	Environment
Quiet	Very Soft	1. Quiet
Speech	Soft	2. Soft Speech
	Moderate	3. Loud Speech
Speech in Noise	Moderate	4. Moderate Speech in Noise
	Loud	5. Loud Speech in Noise
Noise	Moderate	6. Moderate Noise
	Loud	7. Loud Noise

Table 1. Environmental Optimizer II environments

NoiseTrackerII uses spectral subtraction to reduce amplification in frequency regions with low signal-to-noise ratios and maintain the best listening comfort. To accurately identify speech and noise components, NoiseTrackerII looks at the "energy behavior" of the input sound. Speech has the special characteristic that it consists of periods in which it has most of its energy in the low frequencies and little in the high, alternated by periods in which it has most of its energy in the high frequencies and little in the low. The speech detector uses this temporal pattern in addition to other acoustic properties in deciding the probability that the signal is speech. Due to the inclusion of temporal characteristics in the analysis, the speech detector in NoiseTrackerII is better in distinguishing highly modulated noise (e.g., keyboard typing, paper crumpling) from speech.

Because NoiseTrackerII can accurately identify speech, it can accurately estimate the signal-to-noise ratio (SNR) at any given point in time. Based on this estimate, a speech-weighted

gain reduction function continually adjusts itself to reduce unwanted noise. This is done such that the spectral content of noise can effectively be removed and the envelope of the speech signal left virtually intact. If the probability that speech is present is 0%, a flat amount of noise reduction is applied across all 17 bands. As the probability of speech being present increases, noise reduction is primarily restricted to low and very high frequencies, not the speech frequencies, allowing for preservation of the speech signal (Figure 3). Based on the ANSI Speech Intelligibility Index Standard (ANSI S3.5-1997), the most important frequencies for speech are 1500-3500 Hz. This frequency region corresponds to the region of the least amount of noise reduction in NoiseTrackerII.

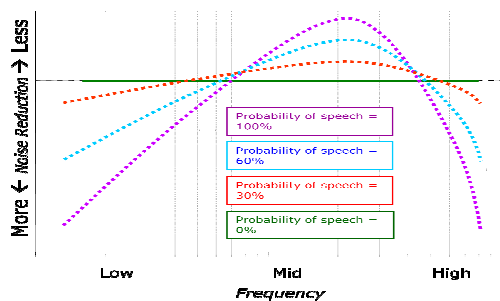


Figure 3. NoiseTrackerII noise reduction is weighted based on speech probability.

ReSound's Alera line offers a wide range of noise reduction values. Levels of noise reduction can be set per the individual environment, or can be set at up to 4 default levels: mild (-3 dB), moderate (-6 dB), considerable (-8 dB) or strong (-10 dB) (Figure 4). The level chosen within an environment indicates the maximum amount of gain reduction that could be applied. For positive SNRs, little gain reduction is applied to afford maximal audibility.

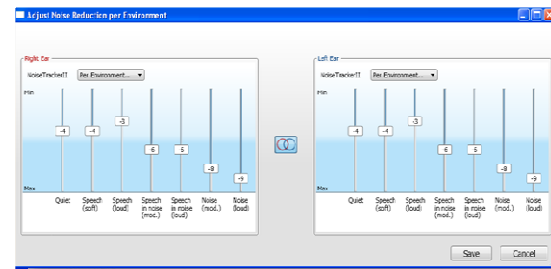


Figure 4. NoiseTrackerII levels can be set per environment in Environmental Optimizer II.

Another improvement that Environmental Optimizer II provides is seamless transitions in gain changes and noise reduction levels from environment to environment. As the hearing aid user moves throughout different listening situations, the hearing aid makes subtle changes which enhance the listening comfort while maintaining the sound quality. Further, if the classifier detects ambiguity in the listening environment's acoustic characteristics, a combination of the appropriate categories is selected, offering a higher degree of accuracy. These features help to ensure a comfortable listening experience that is personalized to the wearer's daily life.

SUMMARY

Along with benefit and value, sound quality is one of the highest correlated parameters when measuring overall hearing aid user satisfaction¹. ReSound recognizes the need for an outstanding signal processing package to provide the best sound quality possible. The new Surround Sound by ReSound technology package offered in the Alera product family builds on the benefits of its Warp™-based technology with innovative features that better connect the hearing aid user to the world. These algorithms offer personalization of directional options, effective feedback control with DFS Ultra and optimization of gain and noise reduction settings system through Environmental Optimizer II. The combined result of these features: extraordinary sound quality that robustly connects individual hearing aid users to the world around them.

REFERENCES

1. Kochkin S. MarkeTrak VIII: Consumer satisfaction with hearing aids is slowly increasing. *The Hearing Journal*. 2010; 63(1):19-32.
2. Smith JO, Abel JS.. Bark and ERB bilinear transforms. *IEEE Transactions on Speech and Audio Processing*, 1999; 7:697-708.
3. Zwicker E, Flottorp G, Stevens SS. Critical bandwidth in loudness summation. *Journal of the Acoustical Society of America*.1957; 29:548-557.
4. Ricketts T, Henry P. Low-frequency gain compensation in directional hearing aids. *American Journal of Audiology*.2002; 11(1):29-41.
5. Wu Y, Bentler RA: Impact of Visual Cues on Directional Benefit and Preference: Part I – Laboratory Tests. *Ear and Hearing*. 2010; 31:22-34.
6. Heymans L: Dubbelblind onderzoek naar de geluidskwaliteit van verschillende directionele modi bij slechthorenden en normaalhorenden, Deel 1: Vergelijking tussen omnidirectionaliteit en vaste directionaliteit zonder bass boost (*Double blind study on the sound quality of directional processing schemes. Part 1: Omni-directional compared to Fixed directionality without bass-boost*). Unpublished thesis. Lessius University College, Antwerp, Belgium, 2009a.
7. Van De Winkel N: Dubbelblind onderzoek naar de geluidskwaliteit van verschillende directionele modi bij slechthorenden en normaalhorenden, Deel 2: Vergelijking tussen omnidirectionaliteit en vaste directionaliteit met bass boost. (*Double blind study on the sound quality of directional processing schemes. Part 2: Omni-directional compared to Fixed directionality with bass-boost*). Unpublished thesis, 2009, Lessius University College, Antwerp, Belgium, 2009b.
8. Van Der Vliet L: Dubbelblind onderzoek naar de geluidskwaliteit van verschillende directionele modi bij slechthorenden en normaalhorenden, Deel 3: Vergelijking tussen omnidirectionaliteit en vaste directionaliteit met “Bach” en tussen vaste directionaliteit met bass boost en vaste directionaliteit met “Bach”. (*Double blind study on the sound quality of directional processing schemes. Part 3: Omni-directional compared to Fixed directionality with surround sound processing and Fixed directionality with bass boost compared to Fixed directionality with surround sound processing*). Unpublished thesis. Lessius University College, Antwerp, Belgium, 2009.
9. Cord MT, Walden BE, Surr RK, Dittberner AB. Field evaluation of an asymmetric directional microphone fitting. *Journal of the American Academy of Audiology*. 2007; 18(3):245-256
10. Bentler RA, Egge JLM, Tubbs JL, Dittberner AB, Flamme GA. Quantification of directional benefit across different polar response patterns. *J Am Acad Audiol*. 2004; 15:649-659.
11. Nabelek AK, Tucker FM, Letowski TR. Toleration of background noises: Relationship with patterns of hearing aid use by elderly persons. *Journal of Speech and Hearing Research*. 1991; 34:679-685.
12. Nabelek AK, Freyaldenhoven MC, Tampas JW, Burchfield SB, Muenchen RA . Acceptable Noise Level as a predictor of hearing aid use. *Journal of the American Academy of Audiology*.2006; 17:626-639.
13. ANSI (1997). ANSI S3.5-1997, American national standard methods for calculation of the speech intelligibility index (American National Standards Institute, New York.