running even though the sound input levels in these situations could be similar.

In order to achieve automatic, situation-dependent gain, the first step was to construct a method to analyze the environment upon which inferences about user preferences can be based. Tools developed for data-logging became the foundation for the Environmental Classifier™, which analyzes the input signal and classifies it into one of seven different listening situations. Subsequently, the Environmental Optimizer can apply volume adjustments based on the environmental characteristics. Default settings can be applied based on hearing loss while individual adjustments can also be made based upon patient needs.

Discussion

The primary need of any individual with hearing loss is a bridge connecting that person to their friends, family and surroundings. A simple amplifier, while meeting the most basic component of rehabilitation, audibility, will do little to inspire the patient to reconnect to a world previously shrouded in a veil of silence. An integrated processing strategy is needed to deliver a hearing instrument fitting that will re-engage the listener. Surround Sound by ReSound uses the combined processing of several algorithms to ensure a seamless listening experience with the flexibility to meet the complex needs of those with hearing loss.

References

Bentler RA, Egge JLM, Tubbs JL, Dittberner AB, Flamme GA. (2004). Quantification of directional benefit across different polar response patterns. *Journal of the American Academy of Audiology, 15,* 649–659.

Cord MT, Walden BE, Surr RK, Dittberner AB. (2007) Field evaluation of an asymmetric directional microphone fitting. *Journal of the American Academy of Audiology, 18*(3), 245–56.

Thompson SC. Tutorial on microphone technologies for directional hearing aids, *The Hearing Journal, 56*(11), 14–21.

Walden BE, Surr RK, Cord MT, Dyrlund O. (2004). Predicting hearing aid microphone preference in everyday listening. *Journal of the American Academy of Audiology, 15,* 365–396.

ReSound North America • 8001 Bloomington Freeway, Bloomington, MN 55420 • 1-800-248-4327
ReSound Canada • 303 Supertest Road, Toronto, Ontario M3J 2M4 • 1-888-737-6863
gnresound.com • customerexperience@gnresound.com

MK602527 Rev.A

SURROUND SOUND BY RESOUND™

Stephen A. Hallenbeck, Au.D.

Abstract

The individual needs of those with hearing losses continue to grow. Professionals expected to treat these individuals using amplification must be provided advanced yet seamless solutions. With proven Warp™-based technology as its foundation, the Surround Sound by ReSound feature set consists of the surround sound processor in conjunction with Dual Stabilizer® II DFS with WhistleControl™, NoiseTracker™ II and the Environmental Optimizer™. A brief description of these individual components follows to provide a broad overview of this technology.

The main goal of hearing instruments is to provide audibility for sounds to which the patient otherwise would not have access. ReSound's further aim has been that this amplified sound should be delivered with minimal distortion and loss of acoustic cues. As technology in general has advanced, patients understandably have increased expectations for high-fidelity sound processing. It is paramount that new technology delivers sound quality aimed at a seamless listening experience with flexible fitting capabilities. ReSound has continually risen to the challenge, delivering several technologies that have raised the bar and redefined patient expectations regarding open fit, directionality, and cosmetic appearance. While these singular technological developments provide excellent solutions for specific problems, modern instrumentation must meet the needs of hearing-impaired people through a holistic approach. Surround Sound by ReSound, built on the Warp platform and driven by the surround sound processor, combined with Dual Stabilizer II DFS with WhistleControl, Noise Tracker II and the Environmental Optimizer integrates sound processing to deliver the real world connection sought after by hearing-impaired people. It is the effective combination and interplay of these algorithms which moves beyond independent algorithms and a solution-driven approach to an encompassing design mimicking the human auditory system.

ReSound Warp-based processing

Frequency representation similar to that of the human auditory system is a prerequisite for good sound quality. As digital hearing instruments are expected to operate for various lengths of time ranging from several days to several weeks on 1.3 volt battery cells, designers must work within strict constraints regarding the complexity of the processing. Frequency warping is an efficient technique to attain frequency resolution similar to the human auditory system. It is virtually distortion-free and associated with minimal processing delay. With 17 smoothly overlapping frequency bands, Warp processing is the foundation providing the highest sound quality in modern hearing instrument processing.

Surround sound processor

The prevailing trend when fitting amplification has been to address hearing in noise using directional microphones. When speaking to patients about hearing instrument programs, we typically describe the first program (omni-directional) as providing an ability to hear sounds from all around the listener. This is contrasted with the second program (directional) in which we describe how sounds from in front of the listener receive more amplification than sounds from behind the listener. This explanation typically provides enough information for a basic understanding of the technology. What we typically do not explain are the unexpected side effects of directional microphones. As a result, problems in the directional mode due to phenomena related to low-frequency noise and diminished localization are reported frequently.

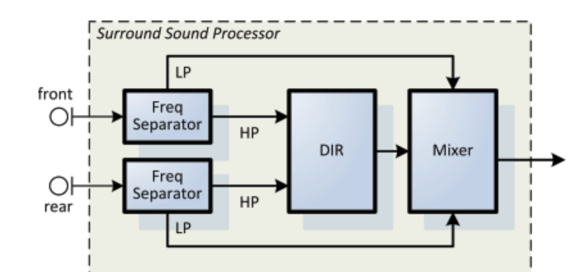

Figure 1. The ReSound surround sound processor analyzes the input from both microphones and separately routes the signal according to high- and low-frequencies, where directional processing is applied to high frequencies and omni-directional to low frequencies.

The ReSound surround sound processor specifically addresses these unwanted bi-products of directional processing technology. Separate processing applied to low and high frequencies drive the architecture of the ReSound surround sound processor *(Figure 1).* High frequencies receive directional processing, to aid in the recognition of speech in the presence of background noise, while low frequencies receive omni-directional processing to preserve between ear phase differences providing improved localization and sound quality without amplifying the noise floor.

ReSound hearing instruments featuring the surround sound processor use it in all directional options to improve sound quality while maintaining a flexible directional fitting strategy. During the initial fit of an instrument with surround sound processing, a default “directional mix” is calculated based upon characteristics of the hearing loss. The directional mix is a function of how much of the frequency bandwidth is processed using a directional strategy and how much is processed using an omni-directional setting. While it is recommended that the directional mix of the processor is not altered as a first step in the fitting process, reports of too much noise from the instrument or dull sound quality are red flags that the directional mix may require attention. Another layer of fitting flexibility is provided through the directional mix control.

Directional options

Even as technological developments improve hearing healthcare, the relationship between our increased understanding of hearing loss and our ability to address the needs of hearing-impaired people in background noise leaves professionals dispensing hearing instruments as if painting with a broad brush. The unfortunate reality is that while this might work for some, it will leave the needs of others unmet. Therefore, our ability to accurately address the needs of patients falls on our flexibility to adapt to as many needs as possible.

It has been demonstrated that patients prefer omni-directional microphone processing over directionality in a greater number of listening situations (Walden et al, 2004). These are situations in which directional microphones provide little benefit in terms of better speech-to-noise ratio. Although the reasons behind the preference for omni-directional microphones was not specifically investigated in that study, it is reasonable to assume that they are related to audibility of the signal of interest—which may not always be in front of the listener—as well as sound quality disadvantages associated with directional microphone processing (Thompson, 2003). Unfortunately, while omni-directional microphones might provide the best sound quality, the optimal performance relative to background noise may be achieved with a directional microphone. Multi-program devices would seem to be a simple solution to this dilemma. However, the confounding variables of manual dexterity and cognition have limited its effectiveness in that many patients simply do not switch programs (Cord et al, 2002). Intelligent processing in which the device analyzes the environment and automatically chooses a directional pattern has been implemented to side-step variables such as dexterity or cognition. Unfortunately, hearing instrument user intentions are not always in agreement from what can be predicted on the basis of environmental characteristics. This can result in a mismatch between the preferred and the selected microphone mode. For the patient, this mismatch can either mean difficulty perceiving environmental sounds that are outside of the directional beam if a directional response is incorrectly selected or lack of directional benefit if an omni-directional response is activated.

It is apparent that a single directional solution is inadequate to meet the needs of all patients in all situations. For this reason, the Surround Sound by ReSound technology package offers five microphone response options to provide maximum flexibility when fitting patients. Included among these options is Natural Directionality™ II, the asymmetric approach which enables hearing instrument users to enjoy the benefits of directional and omni-directional signal processing simultaneously. As patients typically desire improved speech understanding in noise and environmental awareness, independent research has demonstrated that a fitting approach combining omni-directional and directional microphones provides similar directional benefit as a binaural fitting with directional microphones (Bentler et al, 2004; Cord et al, 2007). Additional options including MultiScope™ adaptive directionality with AutoScope™, SoftSwitching™ or automatic switching between omni-directional and directional, fixed directionality, and omni-directional responses are also available to offer fitting flexibility. Keep in mind that the directional response is achieved via the surround sound processor in conjunction with all of these directional options to provide optimal sound quality regardless of directional application.

Dual Stabilizer II DFS with WhistleControl

Controlling hearing instrument feedback is a major factor for a successful fitting. Techniques for reducing feedback are varied and all have inherent benefits and limitations. In recent years, digital feedback cancellation has become the gold standard for reducing feedback while still providing adequate gain. However, unwanted sound quality artifacts from digital phase cancellation can occur when tonal and transient inputs are wrongly identified as feedback by the system, and are the primary limitation of advanced feedback systems. Artifact-free feedback cancellation is partly dependent on the ability of the system to accurately model the feedback path. Dual Stabilizer II DFS utilizes a complex model of the feedback path measured during the fitting, which allows it to dynamically constrain adaptation of the system to increase its immunity to non-feedback sounds. To ensure feedback-free wear to the greatest extent possible, the WhistleControl feature also uses information regarding the feedback pathway to apply an adaptive gain control for situations when feedback cannot be cancelled within the constraints of the maximum stable gain (MSG) curve.

Dual Stabilizer II DFS builds on the ReSound digital feedback suppression (DFS) technology. The operating principle of this processing is to introduce an inverted signal, relative to the phase of the feedback pathway, into the signal pathway of the hearing instrument. This phase inverted signal mirrors the feedback pathway to reduce the intensity and occurrence of feedback. Given that feedback can occur almost constantly in a poorly fitting instrument, or sporadically as the feedback pathway of the instrument in situ changes, the feedback canceling algorithm is designed to reduce feedback in both static and dynamic situations.

Within each fitting, constraints around the modeled feedback pathway are derived to ensure that both static and dynamic feedback is reduced.

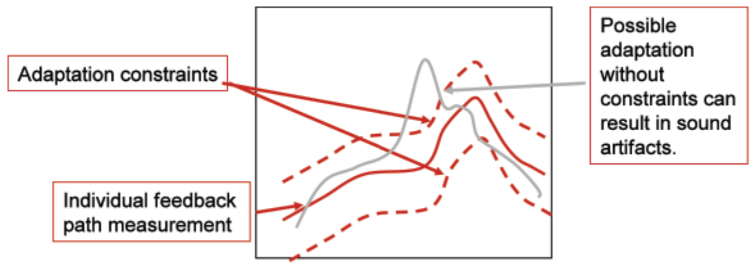

Figure 2. Visual comparison of the feedback pathway generated during DFS calibration and the actual feedback pathway as it changes during daily use. As the actual feedback pathway moves beyond the constraints of the model, artifact and feedback may occur.

Enhancements to the feedback path modeling, as well as tuning of the system time constants and adaptation constraints in the Dual Stabilizer II DFS relative to earlier implementations reduce the occurrence of sporadic feedback. In situations where the feedback cannot be controlled by DFS processing alone, the actual feedback pathway in the given situation will differ from the modeled pathway obtained during the fitting. WhistleControl relies on adaptive gain reduction to preserve patient comfort in these critical situations, and quickly restores gain as things return to normal.

NoiseTracker II

As the insidious effects of hearing loss take hold, the world of the hearing-impaired listener becomes increasingly quiet. Consequently, hearing-impaired individuals may be less tolerant of environmental sound when they are again made audible by hearing instruments. This is a major problem for both practitioner and patient as the very acoustic stimulation sought after from amplification will undoubtedly be accompanied by unwanted background noise. A patient’s own ability to tolerate such noise weighs so heavily on the successful outcome with amplification that tests to determine a personal level of acceptable noise are being included in pre-fitting test sessions.

NoiseTracker II is an innovative noise reduction system designed to specifically address this primary complaint of bothersome background noise. NoiseTracker II is differentiated from the modulation-based noise reduction systems used in other hearing instruments by its accurate method of identifying speech. This key capability enables the system to limit analysis of the background noise to rhythmic pauses of speech embedded in the input signal. In combination with the system’s fast time constants to quickly adapt to speech signals, reduction of noise with limited impact to audibility or sound quality for speech is made possible. The amount of gain reduction applied is dependent on an estimated speech-to-noise ratio, and is limited so as not to create annoying sound quality artifacts associated with large gain fluctuations.

Environmental Optimizer

When first introduced, wide dynamic range compression (WDRC) provided automatic gain adjustments based on the intensity of the input signal. This technology was a giant leap forward in that it provided more gain for soft sounds while maintaining comfortable listening levels for loud sounds. Although the benefits of WDRC were quickly realized, the gain preferences of any individual still are influenced by variables other than input level alone. The nature of the signal is one variable that affects gain preferences. For example, patients may want the sound louder when watching a movie on television than when folding laundry with the dryer