MANAGING FEEDBACK WITH DUAL STABILIZER® II DFS AND WHISTLECONTROL™

Stephen Hallenbeck, Au.D., & Jennifer Groth, M.A.

Abstract

Acoustic feedback continues to be a major hindrance in hearing instrument satisfaction. Conventional techniques for reducing feedback are effective but involve a compromise in terms of audibility, occlusion and physical comfort. Digital feedback suppression has gained acceptance in the hearing instrument dispensing community as a technique for reducing feedback while still providing gain for audibility. Dual Stabilizer® II DFS improves ReSound's feedback suppression technology while WhistleControl™ implements adaptive gain reduction in situations when the feedback path differs from the instrument model.

Acoustic feedback occurs when a portion of the hearing instrument output from the receiver returns to the input, the hearing instrument microphone. The signal is re-amplified and a portion of the re-amplified signal reaches the microphone again. This continuous feedback loop from receiver to microphone can destabilize the hearing instrument. If this feedback loop is uninterrupted, the hearing instrument will begin to oscillate (Kates, 1999) and emit loud, high-pitched squeals.

For the hearing instrument wearer, the sound of feedback can be embarrassing in quiet situations and can limit the amount of amplification available to ensure audibility. Acoustic feedback continues to be a major complaint from those wearing hearing instruments whereby only half of them report satisfaction with their hearing instruments' ability to control feedback. (Kochkin, 2005)

While many different techniques for reducing feedback exist, they can be categorized into three general methods: 1) increasing acoustic attenuation between the hearing instrument output and input; 2) reducing hearing instrument amplification; and 3) feedback suppression algorithms. Techniques related to the first two methods of feedback reduction are well known and have been routinely used for decades. Although feedback suppression is a relatively new method, it is widely available in current products. However, this technology is typically not equal across manufacturers, and may be inconsistently applied in fittings by many professionals dispensing hearing instruments. The ReSound digital feedback suppression (DFS) algorithm was the first of its kind and has continuously been enhanced to provide feedback-free gain with the best possible sound quality. The remainder of this paper will focus on the newest DFS version, Dual Stabilizer II DFS and WhistleControl, a recent update to ReSound's feedback management technology.

Managing feedback: Phase cancellation

The DFS algorithm works by introducing a phase inverted signal relative to the feedback path to the signal path of the hearing instrument. The result is reduced intensity and occurrence of feedback. Given that feedback can occur almost constantly in a poorly-fitting instrument, or sporadically as the feedback path of the instrument changes during use, a feedback suppressing algorithm should be designed to reduce feedback in both static and dynamic situations.

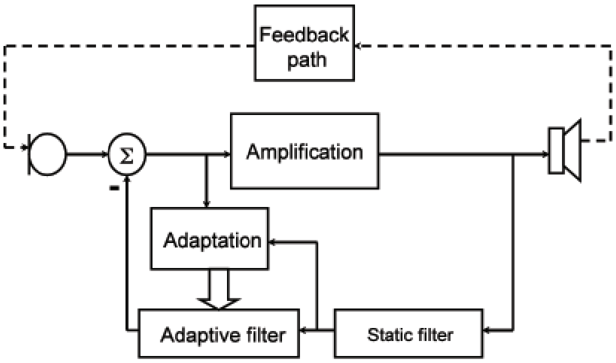

Figure 1. Schematic drawing of the hearing instrument components and the feedback path.

The DFS algorithm incorporates two filters to address these different occurrences of feedback *(Figure 1)*. The first filter focuses on the static properties of the fitting. Examples of these properties include the vent size and position, hearing instrument microphone and receiver orientation, and ear canal size and shape. The coefficients of this filter are determined via a calibration which takes place at the initial fitting. A second dynamic filter is also activated following the calibration of the static feedback suppression filter. This dynamic filter will adapt its characteristics dependent on changes in the feedback path. For example, feedback path changes occur when placing a telephone at the ear, wearing a hat or cupping the hand at the ear. In each of these situations the acoustic environment around the hearing instrument microphone is altered in such a way that the amount of signal reintroduced into the microphone is increased due to acoustic reflections from the surface of the object placed near the hearing instrument. Regardless of whether the type of feedback experienced by the patient is static or dynamic in nature, the accuracy of the feedback path model is critical for the algorithm to function as intended.

DFS calibration measurement technique

The calibration is an in situ measurement using a broadband test signal. The patient is instructed to sit in a comfortable position and remain quiet during the measurement. The test signal is then produced by the hearing instrument and received back into the instrument's microphone, thereby taking into account the whole feedback path. A successful calibration depends on a good signal-to-noise ratio (SNR) for the test signal. Thus the calibration noise must be louder at the hearing instrument microphone than the environmental noise. It is for this reason that the calibration noise (which is hearing loss dependent) is calculated to be subjectively perceived by the patient as loud, but not uncomfortable. If the SNR for the calibration signal at the hearing instrument microphone is favorable, then the software completes an analysis of the feedback path and derives a curve representing the maximum stable insertion gain before the onset of feedback. The Aventa fitting software uses this estimation of maximum stable gain (MSG) to depict a shaded area spanning approximately 10dB on the insertion gain graph *(Figure 2)*.

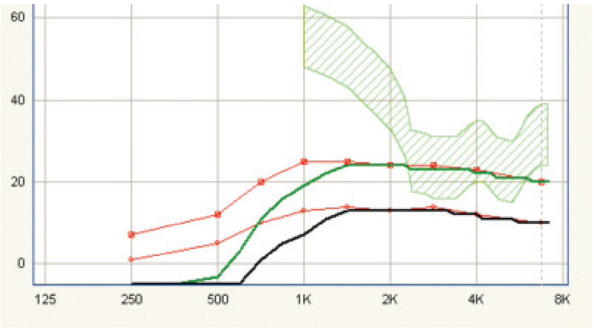

Figure 2. The green-shaded MSG area is derived from an in situ measurement of the feedback path performed during the fitting.

The MSG display reflects two things:

1. The maximum amount of gain which can be provided in the individual fitting with no DFS (bottom of display),
2. The maximum amount of gain with which the ear can be provided if the DFS system is activated (top of display).

In the original design, the calibration signal was presented at a loud level based upon the individual hearing loss and the form factor of the instrument being fit. As previously described, this loud signal was necessary to achieve an adequate SNR. In some instances this loud calibration was deemed uncomfortable by patients and as such the calibration was often not included in a standard fitting protocol, thereby eliminating any use of the feedback reduction algorithm. Recent internal research has revealed that a favorable SNR can still be attained even when the calibration signal is presented at reduced signal levels. In Aventa software version 2.9 and later, a calibration signal with a reduced initialization level has been implemented to ensure loudness comfort for patients during the calibration.

The digital feedback suppression (DFS) system implemented in ReSound hearing instruments uses the calibration data to estimate the attenuation of the feedback path as a function of frequency. A digital filter is created with the same response. This digital filter is then applied to the signal processing path in anti-phase as a mirror image in parallel to the actual feedback path. This approach eliminates the feedback signal without altering the input signal, because the signal path remains unaffected. Research has shown that approximately 8–10dB of added stable gain is achievable with this system (Groth, 1999) (Latzel, et al, 2001).

During the course of the day, changes to the feedback path occur, such as when patients bend over, move their jaws when speaking or eating, or put an object—such as a hand or phone—close to their ear. To account for the changing feedback path, an adaptive filter is also needed. The adaptive filter is designed to continually update the feedback estimation relative to changes in the acoustic environment. The most recent developments of the DFS algorithm have included enhanced modeling of the feedback path as well as tuning of the system time constants and adaptation constraints. These advances in the Dual Stabilizer II DFS algorithm allow for a more aggressive and precise approach to identifying tonal signals and suppressing without interfering with sound quality.

Assessing feedback suppression

While most hearing instrument manufacturers have feedback suppression algorithms, all are not equal. Two areas where manufacturers vary the most in their implementation of feedback control is their ability to increase stable gain and limit artifacts. Sometimes referred to as "headroom," added stable gain refers to the increased gain provided from the feedback suppression algorithm. For example, an individual wearing a hearing instrument might get 10dB of feedback-free gain without feedback suppression activated. The instrument gain could be increased to 25dB without feedback with feedback suppression activated. The additional 15dB of increased gain is the added stable gain and can provide substantial improvement in audibility.

The second area of concern is that of artifacts. Artifacts are processing errors which occur when the feedback suppression algorithm fails to accurately determine the acoustic feedback characteristics as they relate to the feedback path. These errors generally arise when patients listen to tonal inputs such as flute music, keys jingling or bells. Some systems will identify these sounds as acoustic feedback and attempt to suppress them. Because the signal occurs in the listening environment and is not due to an unstable hearing instrument, these phase-reversed signals will be heard as undesirable audible signals or artifacts.

Achieving both high added stable gain and robustness to artifacts is a difficult task. A system may have a very natural sound but provide little in the way of added stable gain. Conversely, a system could be virtually free of whistling but implement a filter strategy that delivers a highly unnatural sound riddled with high-pitched artifacts and a general warbled sound quality overall. Given that most troubling artifacts related to feedback suppression are relatively transient in nature, it seems illogical to introduce a constant degradation in sound quality for an issue that may happen infrequently.

Playing it safe: Using MSG information for a feedback-free fitting

The green MSG area which appears after DFS calibration is a visual representation of the feedback path for the individual fitting. It is derived from information regarding the frequency and intensity at which the feedback is expected to occur and extrapolates upper and lower boundaries. These boundaries help account for variability in the feedback path such as subtle differences in instrument placement or the depth of insertion that will naturally occur during daily use.

Information regarding the frequency and gain level at which feedback is likely to occur is a powerful tool. It allows for an informed decision regarding feedback management. For example, an MSG curve that dips severely into the first-fit gain levels of a hearing instrument may suggest that the shell or earmold is loose-fitting or has too large of a vent for the amount of gain desired. In less severe instances, the MSG curve provides a visual cue, prompting the dispenser to proceed with caution when increasing gain into the MSG area.

Hearing instrument fittings may be fit with gain curves well within the MSG area. Such a scenario is more likely to occur in Open-fit hearing instruments where the large open vent creates a direct feedback path with little attenuation. The upper boundary of the MSG curve represents a boundary between an area where feedback will rarely occur and where it is very likely to occur. ReSound has not previously limited gain to the MSG even though the risk for feedback is very high. Aventa software versions 2.9 incorporate a "Safe Fitting" feature. When enabled, Safe Fitting prevents the dispenser from setting the gains above the MSG area, thereby reducing the likelihood of feedback. It is important to remember when using this feature that the hearing instrument may not be able to provide the desired amount of gain.

WhistleControl: A new method for feedback control

WhistleControl addresses the occasional or situational problems of feedback occurrence, even when the DFS system is activated. Although the feedback modeling performed by the Dual Stabilizer II DFS system is extremely accurate, it is not a perfect representation. This means that the suppression filter settings may over- or undercompensate for the actual feedback, which is most likely when the instantaneous feedback path differs greatly from that measured during the fitting. This could happen when the patient holds a phone up to the device or when the gain settings are well into or beyond the green MSG area. For the patient, intermittent feedback or occasional degradation in sound quality may be the result.

As a supplement to feedback suppression, WhistleControl can be selected to run in parallel with the Dual Stabilizer II DFS *(Figure 3).* WhistleControl continually compares the instantaneous estimate of the feedback path to the model obtained during DFS calibration. This allows a calculation of the feedback path model error, which is then combined with prior knowledge of the maximum stable gain limit in each band to provide gain reduction and maintain stability.

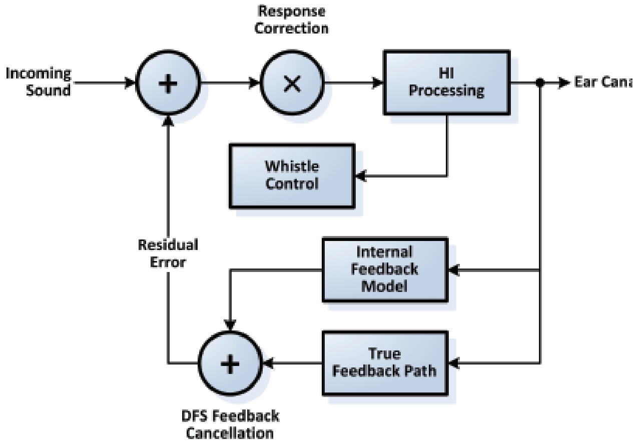

Figure 3: The WhistleControl algorithm is in parallel to the DFS and provides more options for reducing feedback.

To understand WhistleControl it is helpful to revisit some of the previously described basics of the DFS. The DFS suppresses feedback by analyzing the feedback path and configuring filters that have the same gain and frequency characteristics but are opposite in phase. There are two filters: a "static" filter which is initiated by the DFS calibration during the fitting, and an adaptive filter which changes coefficients during use of the instrument to suppress feedback resulting from changes in the feedback path. The feedback path is shown in *Figure 4* as a transfer function with frequency on the x-axis and gain on the y-axis (dark solid line). This shows that there is least attenuation of sound returning to the microphones somewhere in the high frequencies. This is the DFS static filter model.

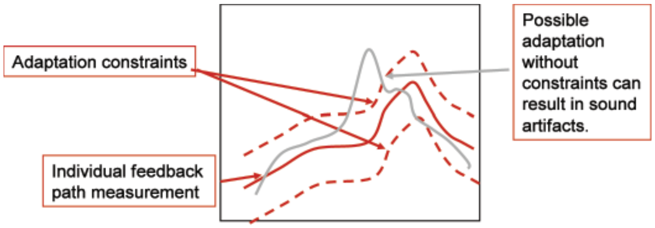

Figure 4. Model of a static feedback path (dark solid line) and the adaptive filter coefficients (dark dotted lines). The grey line represents a tonal input beyond the adaptive constraints. (unscaled)

The adaptive filter places constraints on adaptation such that the adaptive filter is not allowed to stray too far from what was modeled by the static filter. This is to prevent the system from trying to suppress sounds which probably are not feedback, but share similar tonal characteristics. These signals are everyday sounds such as music, bells and clanking silverware. The DFS strikes a balance between feedback suppression and preservation of sound quality. Recent evidence has suggested that feedback suppression algorithms do not significantly influence subjective sound quality ratings. (Johnson, 2007)

The adaptive constraints are about 10dB away from the static feedback path model. This is represented by the dotted lines in *Figure 4.* If there is an input which could be feedback and that would require the adaptive filter coefficients to be set outside of these constraints to suppress it (represented by the grey curve), it is not allowed. The adaptive filter does not adapt to signals that are very disparate from the feedback path model. While this improves the immunity of the system to tonal artifacts, it can also prevent it from always suppressing feedback in extreme situations, such as phone usage.

WhistleControl reduces gain in situations where the DFS cannot prevent feedback. It can be set to work within the DFS constraints or only to act when feedback path changes are extreme. The degree ("Mild," "Moderate," "Strong") corresponds to how big its area of operation is relative to those constraints. A "Mild" WhistleControl setting would mean that it only takes action for a feedback path change that would require adaptation outside the DFS constraints. Therefore gain reduction would occur infrequently, and only in extreme conditions when the monitored feedback path is very different from the initial model. For stronger settings, the WhistleControl will reduce gain even when adaptation is within the DFS constraints.

The amount of gain reduction applied is unknown from the setting. The amount of gain reduction will be situational and depends on how much adaptation would have been required to suppress the feedback. Consider a case where the patient places a phone up to the hearing instrument microphone. Theoretically, the introduction of the phone handset near the instrument should cause a big change in the feedback path and a large degree of adaptation would be needed to suppress it, most likely more than the DFS constraints would allow. Regardless of whether WhistleControl is set to "Mild," "Moderate," or "Strong," it would reduce the gain. Furthermore, gain will be reduced by the same amount for all settings.

Now consider a situation in which a patient receives a hug. The hugger's head comes near the hearing instrument, but it may not change the feedback path as much as holding a phone near the instrument does. In fact, the adaptation required to suppress the feedback may be within the DFS constraints. In this case, a "Mild" WhistleControl setting will probably not result in any gain reduction. However, a stronger WhistleControl setting may result in gain reduction even though the DFS alone might have been sufficient to suppress the feedback. Regardless of whether the setting is "Moderate" or "Strong," the actual gain reduction would be the same.

Given the description of the algorithm function and these two situations, two primary concepts should be apparent. First, feedback reduction through the use of DFS technology is built upon the accuracy of the modeled feedback path as well as its ability to adapt to subtle changes to the actual path relative to the modeled path. Therefore, a calibration that is representative of how the hearing instrument will be worn on the ear of the patient is necessary to the successful performance of the DFS and WhistleControl algorithms. Second, the "Mild," "Moderate," and "Strong" WhistleControl settings refer to how often the gain reduction will occur. The "how much" component of gain reduction depends on the specific situation, not the WhistleControl setting.

Summary

Enhancements to the Dual Stabilizer II DFS algorithm, as well as Safe Fitting and WhistleControl, are tools that address the vexing problem of feedback and provide greater flexibility when fitting hearing instruments. This feedback management "package" ensures that patients can enjoy feedback-free use of their hearing instruments in virtually all situations.

References

Groth J. (1999). Digital signal processing has made active feedback suppression a reality. *The Hearing Journal, 47*(10), 45–48.

Hellgren J, Lunner T, Arlinger S. (1999). System identification of feedback in hearing aids. *Journal of the Acoustical Society of America, 105,* 3481–3496.

Johnson EE, Ricketts TA, Hornsby BW. (2007). The effect of digital phase cancellation feedback reduction systems on amplified sound quality. *Journal of the American Academy of Audiology, 18*(5), 404–416.

Kates JM. (1999). Constrained adaptation for feedback cancellation in hearing aids. *Journal of the Acoustical Society of America, 106,* 1010–1019.

Kochkin S. (2005). MarkeTrak VII: Consumer satisfaction with hearing instruments in the digital age. *The Hearing Journal, 58*(9), 30–43.

Latzel M, Gebhart TM, Kiessling J. (2001). Benefit of a digital feedback suppression system for acoustical telephone communication. *Scandinavian Audiology Supplement, 52,* 69–72

MK602529 Rev.A

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

ReSound
rediscover hearing