Audiogram+: The ReSound Proprietary Fitting Algorithm

Abstract

Hearing instruments should provide end-users with access to undistorted acoustic information to the degree possible. The Resound compression system uses state-of-the art technology and carefully selected compression parameters to achieve this aim. Prescription of hearing instrument gain is an essential element in optimal application of Wide Dynamic Range Compression, and the Audiogram+ fitting algorithm has been developed and refined to provide the best starting point for hearing losses ranging from mild to severe. This article describes the rationale for the ReSound system and Audiogram+.

The main goal of a hearing instrument is to provide hearing impaired wearers with access to acoustic information that will enable them to listen, comprehend, and communicate with those around them. Because the human brain is by far the most superior processor of speech and speech in noise, the ReSound philosophy has continually been to deliver sound to the auditory system with minimal distortion and loss of acoustic cues. This philosophy has served as a guiding principle in all aspects of hearing instrument design, from selection of device components to sound processing to fitting methodologies. Based on the seminal work of Villchur (1973), ReSound pioneered Wide Dynamic Range Compression (WDRC), offering the first system to account for loudness recruitment by applying progressively less gain with increasing input levels. To provide a reliable starting point for use of this WDRC system, 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.

This article describes the rationale for the Resound WDRC amplification system and the accompanying Audiogram+ fitting algorithm.

Loudness compensation

Loss of sensitivity to soft sounds, also known as loudness recruitment, is an issue for individuals with sensorineural hearing losses. Figure 1 shows an example of a normal loudness growth function (solid curve) for a particular narrowband input versus an abnormal function typical of an individual with sensorineural hearing loss (dotted curve). In this example, an input of around 45dB SPL that would be perceived as “Very Soft” by the normal-hearing individual would be inaudible to the person with a hearing loss. For the hearing-impaired individual, a perception of “Very Soft” would be achieved near an input level of 70dB SPL. However, as the input level increases, the curves converge, and sounds are judged similarly by both hearing-impaired and normal-hearing individuals.

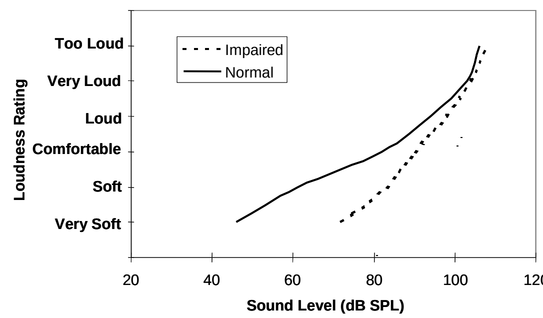

Figure 1. Normal and abnormal loudness growth functions.

Compensation for abnormal loudness growth implies that different amounts of gain will be necessary depending on the input level. Because loudness perception is most dissimilar from normal 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.

Compression parameters in the ReSound system have been carefully selected to support compensation for abnormal loudness growth specifically vis-à-vis speech. Conversational speech sounds vary up to 30dB, with the softest consonants occurring in the high frequencies at levels below 45dB SPL (McFarland, 2000). Therefore, compression thresholds are fixed at frequency dependent levels ranging from 40 to 48dB, with the highest thresholds occurring at low frequencies. With such low compression thresholds and an upper range of non-linear amplification extending above 85dB SPL, small compression ratios have immense impact on the gain provided to sounds of different intensities. For example, a compression ratio of 3:1 over a 40dB input range results in 26dB more gain for the low-level sounds than for the high-level sounds. The ReSound compression rationale allows for compression ratios of 1:1 to 3:1, which permits adequate loudness compensation for losses ranging from mild to severe. Although higher compression ratios may allow for even greater audibility of some low level speech sounds, this can obscure spectral contrast cues to a degree that degrades speech intelligibility (Moore et al, 1992).

Keeping in mind the rationale to provide audibility for speech, the dynamic compression characteristics of the ReSound system are consistent with the duration typical of speech phonemes. Also known as "syllabic compression", attack times range from 5 to 12 ms depending on frequency, and release times range from 70 to 120 ms. Release times are longer in the low frequencies, and were selected to preclude audible "pumping" artifacts which may be an unintentional side effect of too quick gain fluctuations.

ReSound Warp-based processing

Hearing instrument fitting procedures assume that the amplification system is capable of frequency analysis that is suitable for the human auditory system. As part of the development of hearing instruments, it is necessary to continually evaluate and, if advantageous, update the technology used to implement such things as compression. 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 (Moore & Glasberg, 1983). Frequency warping is en efficient technique used in ReSound hearing instruments to emulate the frequency resolution of the human auditory system with virtually no distortion and minimal processing delay. The ReSound system uses a mathematical warping function to map frequency components logarithmically to a scale closely corresponding to the auditory Bark scale (Smith & Abel, 1999). The Bark scale incorporates the critical auditory bandwidth as the scale unit (Zwicker et al, 1957). The frequency warped ReSound compression system results in 17, smoothly overlapping frequency bands separated by approximately 1.3 Bark as illustrated in Figure 2.

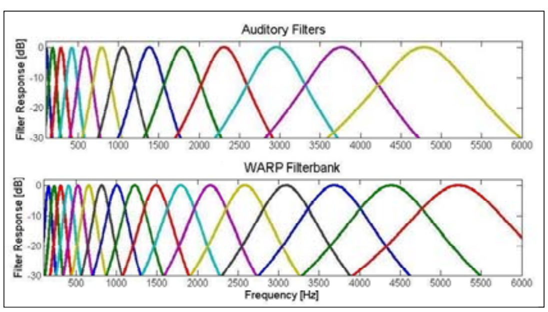

Figure 2. Comparison of a model of the auditory system filters and the ReSound Warp-based system.

Rationale

The Audiogram+ target prescription is grounded in a loudness normalization rationale; although the actual gains prescribed do not realize this objective. In order to achieve loudness normalization, frequency and input level dependent gain would be applied such that the hearing instrument wearer perceived the loudness of narrowband signals similar to the way a normal-hearing individual would. An implicit assumption of loudness normalization is that loudness summation will be similar for the hearing-impaired and normal-hearing listener, and will lead to adequate and satisfactory loudness of real-world sounds for the hearing instrument user. However, hearing instrument users tend to prefer less gain than a loudness normalization rationale would prescribe (Smeds, 2006; Keidser & Grant, 2003). The analysis of fittings based on individual loudness scaling procedures that led to the development of Audiogram+ revealed similar findings. For this reason, Audiogram+ prescribes less gain than a strict loudness normalization rationale would indicate. Compared to the generic NAL-NL1 fitting rule, which aims to maximize speech intelligibility, Audiogram+ prescribes 3 to 10dB less insertion gain depending on frequency and hearing loss configuration.

Influence of audiometric data

Audiogram+ calculates insertion gain targets for narrowband inputs of 50 and 80dB SPL at 11 audiometric octave and interoctave frequencies from 125 Hz through 8 kHz. It bases the prescription on the hearing threshold levels at these audiometric frequencies, although it requires a minimum of only one audiometric threshold as input to the formula. Any missing hearing threshold data is estimated via interpolation from the existing datapoints. A bilateral fitting is always assumed.

Apart from the hearing threshold levels at the individual frequencies, four audiometric factors are taken into consideration. These include the severity of the hearing loss, the configuration of the hearing loss, the individual Uncomfortable Loudness (UCL) levels and whether the hearing loss has a conductive component.

Hearing loss severity

For severe-to-profound hearing losses, there is less high frequency emphasis and relatively more low frequency gain in the prescribed response than for mild-to-moderately severe losses. These empirically derived accommodations for severe-to-profound hearing losses are consistent with the observations of other investigators. For example, Byrne et al (1990) estimated an optimal frequency response and measured insertion gain at the preferred volume with this response for a number of listeners with severe-to-profound hearing losses. They found that relatively more low frequency gain than prescribed by the NAL formula was optimal, and that preferred gain as a rule was 10dB higher. The effect of hearing loss severity for Audiogram+ is typically a low frequency gain increase of 4 to 5dB and a high frequency reduction of 4 to 5dB compared to what would have been prescribed without the correction for severity. Although the exact conditions imposed in the Audiogram+ formula are somewhat more complicated, these changes are generally applied when the PTA exceeds 65dB HL.

Hearing loss configuration
The ability of hearing instrument wearers to make use of high-frequency speech information has been observed to decrease as hearing threshold levels exceed about 60dB HL (Hogan & Turner, 1998). An explanation for this could be the presence of “dead regions”, which are portions of the basilar membrane having complete loss or damage of both outer and inner hair cells (Moore et al, 2000). The probability of a high frequency dead region is greatly increased when high frequency hearing threshold levels exceed 80dB HL. Audiogram+ avoids prescribing excessive high frequency gain in cases of steeply sloping high frequency hearing losses. If the slope of the hearing loss above 1 kHz exceeds 20 dB and high frequency thresholds are worse than 80 dB HL, then the formula will prescribe high frequency gain as if the thresholds above the maximum audiogram slope were at 80dB HL. For example, an individual with the hearing loss shown in Figure 3 would have identical targets prescribed for the two ears.

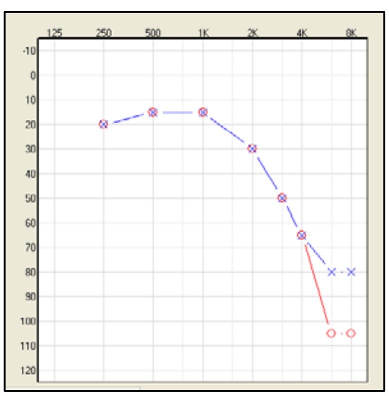

Figure 3. *Audiogram+ limits the high frequency gain prescription for steeply sloping losses. Targets would be identical for this right and left ear.*

Uncomfortable loudness levels (UCL)
Some hearing instrument wearers may have a lower or higher tolerance for loud sounds than what would be predicted by their hearing threshold levels. If UCL data is provided to Aventa, gains will be adjusted to accommodate the individual’s dynamic range into the prescription. The effect on the 50dB targets is an adjustment corresponding to approximately 1/3 of the difference between the predicted and actual UCLs. For the 80dB targets, an adjustment of approximately 2/3 of this difference is applied. If the measured UCL is lower than predicted, targets are reduced, and if higher, targets are increased.

Although the ReSound system includes compression limiting at the output stage, there is no prescription for MPO in Audiogram+ except for super power hearing instruments intended to fit individuals with severe-to-profound hearing losses. For these products, the method for prescribing OSPL90 as described by Dillon (2001) is employed. For other ReSound devices, the fast-acting WDRC effectively limits the output for high input sounds, and accommodations for UCL are made via gain adjustments. For example, if the threshold at 1 kHz is 50dB HL and the measured UCL is 95dB HL, then the target for an 80dB SPL input will be reduced from 10dB to 6dB and the compression ratio will be increased from 1.6 to 1.8. For a 90dB SPL narrowband input, this means that the output would be reduced from 96dB SPL, which is just over the measured UCL, to 92dB SPL.

Air-bone gap
A premise of Audiogram+ is that the hearing loss is sensorineural with accompanying recruitment. In pure conductive losses, loudness growth functions are normal, but shifted up by the same amount as the sensitivity loss. This means that linear amplification with an equal amount of gain for varying input levels would be appropriate. In mixed losses, it is assumed that the conductive component has a similar effect and that individuals with this type of hearing loss are most appropriately fit with more gain at higher input levels than would be prescribed for those with pure sensorineural losses. Audiogram+ increases targets if air-bone gaps are present in the audiogram. For the 50dB SPL input 1/5 of the air-bone gap is added to the target and for the 80dB input, 1/3 of the air-bone gap is added to the target, resulting in both more gain and a more linear response.

Note that corrections for measured UCL and air-bone gaps cannot be made at the same time. If the fitter chooses to account for measured UCL data, any air-bone gaps in the audiogram will be ignored by the formula. The rationale for this is that the measured UCL is expected to give a more accurate picture of the end-user’s dynamic range than what is estimated by the air-bone gap corrections. Thus a starting point based on this data is expected to be closer to the user’s preferred gain.

Influence of Non-audiometric Data

It is well-established that audiometric data is not the only determinant of hearing instrument users' gain preferences. Audiogram+ can account for experience with amplification both in terms of hearing aid use and type of amplification.

Hearing aid use

It has been reported that experienced hearing instrument wearers have altered loudness perception and different gain preferences than inexperienced users, at least for individuals with moderate-to-severe hearing losses (Olsen et al, 1999; Keidser & Grant, 2003). Audiogram+ can account for this preference if "First-time user" if selected in the Aventa fitting software "Patient" screen. This correction will decrease high frequency gain by approximately 6dB relative to the targets for experienced users, and will increase the high frequency compression ratio slightly. For end-users who are over sensitive to high frequency gain, the "Comfort" user profile results in a gain reduction ranging from approximately 10% of the hearing threshold level at 2 kHz to approximately 25% of the hearing threshold level at higher frequencies.

Type of previous amplification

By selecting "Experience - linear" as the patient profile, the targets will be adjusted to provide a slightly more linear response with more gain for both input levels. These corrections were derived from an internal study comparing fine-tuned gain preferences of 40 former users of linear hearing instruments and 67 former users of WDRC devices. The "Experience – linear" profile may be helpful in "weaning" some users from older linear technology to WDRC amplification.

Verification

In verifying fittings against targets, it is important to know what assumptions are made by the prescription. For Audiogram+, the following is assumed: 1) the targets are specified in real ear insertion gain, 2) the input signal is narrowband, 3) input levels are 50 and 80dB SPL, and 4) azimuth is 0 degrees. Because of the nature of the assumed input signal, Audiogram+ targets are not appropriate for fitting devices other than Resound. While the default view in the Aventa fitting software is simulated insertion gain, the fitter can also select alternative views (e.g., 2cc coupler) to facilitate verification. Targets can also be shown assuming a speech-weighted noise as the input signal. Special signal processing such as NoiseTracker™ II and Environmental Optimizer™ may give an inaccurate impression of the device amplification in relation to the targets and should be disabled for insertion gain or coupler gain measurements. For verification, it is recommended to select the "REM" program in the Aventa fitting software.

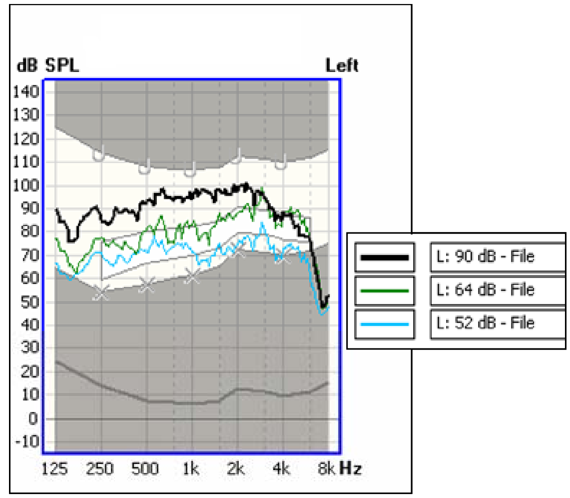

Figure 4. Speech mapping can be used to verify that speech at different levels is amplified to fit within the user's dynamic range.

Fittings with ReSound hearing instruments can also be verified by use of speech mapping with live or recorded speech as the input signal if desired. In this case, it is not necessary to disable any special signal processing. Speech mapping measures are an appealing way both to ensure that amplified speech at different levels fits into the individual's dynamic range of hearing, as well as to counsel the hearing instrument user. An example of speech mapping done with recorded speech at different levels and a ReSound hearing instrument is shown in Figure 4.

References

Byrne D, Parkinson A & Newall P (1990). Hearing aid gain and frequency response requirements for the severely/profoundly hearing impaired. Ear & Hearing, 11(1), 40-49.

Dillon H (2001). Prescribing Hearing Aid Performance. In Hearing Aids. Sydney: Boomerang Press.

Hogan CA & Turner CW (1998). High-frequency audibility: benefits for hearing impaired listeners. Journal of the Acoustical Society of America, 104, 432-441.

Keidser G & Grant F (2003). Loudness Normalization or Speech Intelligibility Maximization? Differences in clinical goals, issues and preferences. Hearing Review, 10(1), 14-22.

McFarland W (2000). Speech perception and hearing aids. In Sandlin (ed) Textbook of

Hearing Aid Amplification, 2nd edition. San Diego, CA: Singular Publishing Group, Inc.

Moore BCJ & Glasberg BR. (1983). DR & Alcantara JI (2000). A test for the diagnosis of dead regions in the cochlea. British Journal of Audiology, 34, 205-234.

Moore BCJ, Lynch C & Stone M (1992). Effects of the fitting parameters of a two-channel compression system on the intelligibility of speech in quiet and in noise. British Journal of Audiology, 26(6), 369-379.

Olsen S, Rasmussen A, Nielsen L & Borgkvist B (1999). Loudness perception is influenced by long-term hearing aid use. Audiology, 38, 202-205.

Smeds K, Keidser G, Zakis J, Dillon H, Leijon A, Grant F, Convery E, Brew C (2006). Preferred overall loudess. II: Listening through hearing aids in field and laboratory tests. International Journal of Audiology, 1(1), 12-25.

Smith JO & Abel JS. (1999). Bark and ERB bilinear transforms. IEEE Transactions on Speech and Audio Processing, 7, 697-708.

Villchur E (1973). Signal processing to improve speech intelligbility in perceptive deafness. Journal of the Acoustical Society of America, 53, 1646-1657.

Zwicker E, Flottorp G, & Stevens SS. (1957). Critical bandwidth in loudness summation. Journal of the Acoustical Society of America, 29,548-557.