The Audioscan® RM500 Real-Ear Hearing Aid Analyzer: Measuring for a Successful Fit

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The goal in fitting a hearing instrument has always been to provide audibility and comfort for important environmental sounds, especially conversational speech, while ensuring that loud sounds are not uncomfortable.

With the overwhelming inundation of new technology, sometimes it is easy to lose track that this *is, always has been, and always will be*, the primary goal of hearing instrument fitting. Digital signal processing (DSP) chips provide the capability within hearing instruments for real-time manipulation of audio signals in a seemingly infinite number of ways. This technical sophistication should theoretically make our goal of providing audibility while maintaining comfort achievable for most degrees and configurations of hearing loss.

The technological advances in hearing instrument design seem so complicated that many clinicians feel they lack the knowledge to manipulate

all the variables necessary for appropriate fitting of the instrument. Confusion exists over whether the hearing instruments can be fitted using a science-based prescription fitting method or if one has to use a manufacturer's proprietary algorithm for the fitting. Clinicians are uncertain on how to verify their hearing instrument fitting. Some manufacturers have recommended that clinicians do not perform real-ear measures of hearing instrument performance when verifying their digital products. In fact, so much confusion currently exists that it appears we have entered the era David Hawkins describes as the "Something Really New, And We'll Tell You How to Fit it" method where the dispenser is often provided little in the way of explanation and is expected to follow the saying "Trust Me, It'll Work" (Hawkins, 1994). It is important that as dispensing audiologists we remain active participants in the hearing instrument selection and fitting process, otherwise we reduce one of the inherent advantages of digital instruments, that of greater flexibility in dealing with individual differences in the fitting of hearing instruments (Levitt, 1997).

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For almost 20 years, probe microphone measures of hearing instrument performance have provided audiologists and hearing instrument dispensers with an objective and versatile electroacoustic method for assessing the real-ear performance of any type of hearing instrument. An extensive body of literature has demonstrated the validity, reliability, and utility of real-ear measures (Tecca, 1994). Some important reasons to argue for the continued routine use of real-ear measurements in the fitting and verification process would include:

1. They allow individualized verification of real-ear hearing instrument performance.
2. They provide information that is essential for evaluating and/or improving a fitting based on a set of fitting criteria and the sound quality preferences of the person to be fitted.
3. They provide baseline documentation for future revisions to a fitting, if necessary.
4. They provide an excellent educational or counseling tool.
5. They are recommended by professional associations as part of their guidelines for hearing aid fittings (The Pediatric Working Group Conf on Amp for Children, 1996; ASHA, 1998).
6. They are often required for third-party reimbursement purposes.
7. They are a means to demonstrate to professionals from other disciplines that audiology is an accountable profession. They are an important means of demonstrating that audiologists are achieving clinical goals or formulating methods to improve hearing instrument fittings (Tecca, 1994).

To ensure that real-ear measurement results continue to provide meaningful information, manufacturers of real-ear measurement systems continually evaluate and improve fitting and verification procedures to keep pace with hearing instrument technology. Many real-ear measurement systems have now implemented the Desired Sensation Level (DSL) method for hearing instrument selection and fitting, and have implemented speech-like test signals for nonlinear hearing instrument verification purposes.

The Audioscan RM500 real-ear hearing aid analyzer utilizes the Speechmap®/DSL® fitting system for the fitting and verification of digital hearing instrument technology (Cornelisse et al, 1994; Sinclair, Seewald and Cole, 1996; Cole and

Sinclair, 1998). The Speechmap/DSL fitting system departs from conventional fitting and verification methods in a number of important ways. First, the concept of insertion gain is not used at all. Second, audiometric data is expressed in terms of dB SPL at the eardrum so that direct comparisons between the measured hearing instrument performance with multiple input levels and the client's residual auditory area can be made. Finally, unique test signals for soft, average and loud conversational speech, as well as loud environmental sounds are provided so that real-ear performance of digital hearing instruments can be accurately measured.

The Concept of Insertion Gain is Not Used At All

Historically, the fitting of a hearing instrument has been verified based on performance differences obtained in the unaided versus the aided condition. With the development of real-ear probe-microphone measurement systems, the behavioural measurement of functional gain was replaced with the electroacoustic measurement of insertion gain. If reliable, valid measures were obtained in each condition (i.e., behavioural and electroacoustic), the amount of real-ear gain was the same within the errors of measurement (Skinner, 1988). That is functional gain was equal to insertion gain. To date, insertion gain-based procedures are the most widely used clinical approach for linear and nonlinear hearing instrument fitting in the U.S. (Mueller, 1998).

Insertion gain fitting and verification procedures must be reconsidered if we are to take full advantage of the versatility of modern hearing aids. First, an insertion gain fitting target implicitly assumes a soundfield audiogram was obtained during the audiometric assessment. Insertion gain fitting targets will be in error when any other transducer is used for audiometric assessment purposes because the resulting audiogram will differ from the sound field audiogram by the amount that the individual's unaided ear response deviates from the average unaided ear response.

Second, the utility of a *gain* measurement is that the *output* for any arbitrary input can be easily calculated as input plus gain, if the system is linear. For nonlinear hearing instruments, output

cannot be predicted for any arbitrary input because the relationship between gain and output is affected by the *type* of signal *and* by the input *level* of the signal. This means that different insertion gain targets are required for each different *type* (not just level) of input signal. The end result is that the use of gain can lead to fitting errors. For instance, in Figure 1, the *gain* (in dB) for a 70 dB pure tone and for a 70 dB speech signal measured with a K-AMP® hearing instrument set for a moderate hearing loss appears to be about the same across frequency, perhaps leading to the conclusion that if the aid is adjusted to make the pure tone audible, speech will also be audible. Figure 2, showing the actual dB SPL output for each signal, indicates just how inappropriate this conclusion would be. In Figure 2, the *output* (in dB SPL) for the 70 dB pure tone signal at 2000 Hz is 15 dB higher than the output in the 2000 Hz ½-octave band of the speech signal (with an overall RMS of 70 dB SPL). This is a significantly different result than what we might have predicted by looking at the curves in Figure 1. The difference relates to the fact that for a nonlinear hearing instrument the gain depends on the input signal. The pure tone input at 2000 Hz

was 70 dB and 26.5 dB of gain was measured. Input + Gain = Output. $70 + 26.5 = 96.5$ dB SPL output at 2000 Hz. However, even though we measured approximately the same amount of gain (in dB) at 2000 Hz for the 70 dB speech input, the level in the ½-octave band of speech at 2000 Hz is approximately 53 dB. $53 + 29 = 82$ dB SPL output at 2000 Hz. The gain curves of Figure 1 indicated that at most frequencies there was little difference between how the hearing instrument processed speech and a tone. The use of *gain* obscures the fact that the *output* (dB SPL) for speech is much less than the *output* (dB SPL) for the tone—and that is important information to know when fitting a hearing instrument. Therefore, in order to obtain test results that provide meaningful information and that can be used to achieve successful fittings, it is important when fitting nonlinear amplification to look at real-ear measures in terms of output, not gain.

An additional major limitation of insertion gain methods is that the insertion gain targets, and real-ear measurements that relate to these criteria, exist apart from any meaningful and/or relevant context (Seewald et al, 1993). Without being able to directly compare how the desired

Figure 1. The gain (in dB) for a 70 dB pure tone and for a 70 dB speech signal as a function of frequency (in Hz) measured with a K-AMP hearing instrument set for a moderate degree of hearing loss.

Figure 2. The output (in dB SPL) for a 70 dB pure tone and for a 70 dB speech signal as a function of frequency (in Hz) measured with a K-AMP hearing instrument set for a moderate degree of hearing loss.

performance relates to an individual's auditory area, it is virtually impossible to know, for example, where amplified speech will be placed within the listener's residual dynamic range of hearing. This issue of context has become more significant in the fitting and verification of newer technology where we have the ability to generate multiple targets for various input levels and to adjust compression thresholds and compression ratios within several independent channels. Because the real-ear aided performance of today's hearing instruments will depend on the input signal type and level used, we need to move to examining the measurement results in more meaningful terms, that of dB SPL output.

Audiometric Data are Converted from dB HL to dB SPL at the Eardrum

The Speechmap/DSL fitting system attempts to reduce the errors associated with traditional dB HL audiograms and insertion gain methods by converting all relevant variables to an eardrum SPL reference and by providing procedures to quickly and accurately measure individualized HL

to SPL transforms (Hawkins et al, 1990; Cole and Sinclair, 1998; Scollie et al, 1998). This allows for direct comparisons between the measured hearing instrument performance and the client's residual auditory area. A loudspeaker delivers test signals to the client; hearing aid output (real-ear aided response, REAR and real-ear saturation response, RESR) is measured at the eardrum using a probe microphone. This is an advantageous approach when fitting digital instruments because performance of any hearing instrument regardless of processing algorithms can be measured using multiple input signals (soft, average, loud conversational speech and loud environmental sounds). Measurement results can quickly be evaluated relative to the client's residual dynamic range of hearing.

Figure 3 illustrates an unaided SPLogram used in the Speechmap/DSL fitting system. Audiometric test results and target criteria for real-ear performance are displayed in terms of dB SPL at the eardrum as a function of frequency. The goal in hearing instrument fitting for this individual is to amplify speech to levels above threshold, but limit amplification so that loud sounds do not exceed uncomfortable loudness levels.

Figure 3. An unaided SPLogram: Audiometric test results and target criteria for real-ear performance are plotted in terms of dB SPL at the eardrum as a function of frequency (in kHz).

Figure 4 illustrates how measured real-ear performance is displayed on an aided SPLogram. Measured performance can quickly be evaluated relative to the target criteria developed at the selection stage for the client to be fitted. One of the main advantages to an SPLogram approach to hearing instrument selection and fitting is that when compromises or changes are made in the setting of the frequency response, gain, compression characteristics and maximum acoustic output of the hearing aid we can quickly see how the decisions affect audibility, comfort and potential benefit for the client being fitted.

Test Signals for Soft, Average and Loud Conversational Speech, as Well as Loud Environmental Sounds are Provided

Hearing aids are designed to process speech (and music and noise). While hearing instrument am-

Figure 4. An aided SPLogram: Real-ear aided performance is compared to target criteria in dB SPL near the eardrum as a function of frequency (in kHz).

plification targets are defined based on the fitting goals for speech inputs, hearing instruments have traditionally been verified using artificial test signals such as tones or noises. In a typical clinical environment, a pure tone, warble tone, or broadband noise signal is presented to the hearing instrument at a level that is similar to the overall level of soft, average or loud speech and gain or output is measured across frequencies. These measurements are then interpreted as being representative of hearing instrument performance results for speech inputs. Clinical test signals will vary in their ability to predict speech output because some signals will be more or less speech-like than others. For instance, real speech at average levels is low-frequency weighted, and varies in amplitude and frequency content with time. Presumably, the more closely test signals mimic these real speech characteristics (e.g., overall RMS level, $\frac{1}{3}$ octave band levels, modulation characteristics), the more successful they will be at predicting real speech output.

With the introduction of digital processing hearing instruments, concerns regarding the predictive validity of clinical test signals have only increased. A common misconception is that digital hearing instruments cannot be verified using probe microphone measurements because the clinical test signals (e.g., pure tones, composite noise) used to predict speech output will be interpreted as noise. Consequently, the hearing instrument gain will be reduced relative to the gain for speech inputs, thereby preventing accurate estimates of real speech output. When addressing this concern, it is important to consider a couple of important factors. First, not every digital instrument will incorporate noise reduction processing features. Thus, with these non-noise reduction digital instruments, steady-state test signals may still be used to evaluate hearing instrument performance. Second, whether digital or analog, nonlinear instruments are best evaluated using time-varying, speech-weighted signals. As Fabry (1998) stated, "While evaluation of digital hearing aids is not feasible by conventional measurement techniques (e.g., swept sinusoids or steady-state composite-noise signals), it is not theoretically impossible to do so . . . will require the use of broad-band, temporally varying stimuli to simulate 'real-world' performance" (p. 32).

In an attempt to successfully determine the response of all types of digital hearing instruments (e.g., noise reduction and no noise reduction) for real-speech signals, Audioscan has developed some unique test signals called SWEPT burst-biased tones and the DYNAMIC roving-tone. The SWEPT burst-biased test signal consists of 16 or 64 tones presented in sequence, with each tone constructed as in Figure 5 (this figure shows the construction of the SWEPT *average* test signal). The 75 ms burst preceding each tone is designed to set the gain of the compression circuit as it would be set by repeated speech peaks while the measurement of hearing aid output occurs during recovery, with the input at the $\frac{1}{2}$ octave band level for the speech being simulated (soft, average or loud). The measurement takes about 30 ms after a delay of approximately (40/frequency) seconds. The frequency weightings used for soft (55dB) average (70dB) and loud (85dB) speech are shown in Figure 6. The DYNAMIC roving-tone test signal consists of 64 tones at $\frac{1}{2}$ octave frequencies presented at 12 dB above and 18 dB below the long-term average band levels of Figure 6. The order of presentation and the duration of

Figure 5. Construction of the Audioscan burst-biased tone average level (70 dB SPL) as a function of time (msec).

each tone is based on the order and duration of significant $\frac{1}{2}$ octave band levels in the phrase "Joe took father's shoebench out. She was waiting at my lawn". Figure 7 shows how the DYNAMIC test signal simulates some of the temporal and duration characteristics of a real-speech signal. The maximum output (MPO) test signal consists of a series of 85 ms tone bursts at 85 or 90 dB and is used to estimate worst case output in response to loud environmental sounds.

A recent study (Scollie et al, 1999) reported on the accuracy of the SWEPT and DYNAMIC test signals in predicting the levels of aided speech over a wide range of hearing aid circuit types (linear, nonlinear, digital) fitted for varying degrees of hearing loss. Results indicated that, on average, the test signals (at soft and average speech-weighted input levels) predicted aided speech output levels with a high level of accuracy (within 6 dB RMS) across a large number of hearing instruments. A second finding of this study indicated that digital hearing instruments with noise reduction algorithms were also, on average, successfully tested within the 6 dB RMS criterion range with the test signals. This is likely related to the fact that the temporal modulations in the SWEPT and DYNAMIC signals prevented the test

Figure 6. Test levels (sound-field SPL) for Audioscan's soft (55dB), average (70dB) and loud (85dB) swept burst-biased tones as a function of frequency (kHz).

signal being processed as noise. Figure 8 plots the degree of agreement between the aided average speech levels and the levels of the SWEPT average speech test signal for 34 hearing instruments. The X axis indicates the measured level of aided speech. The Y axis indicates the measured level of the test signal. The diagonal line indicates the point of perfect agreement between the two. If the measured points fall on the diagonal line, the test signal has provided an accurate estimate of the hearing instrument's performance for speech. As shown in Figure 8, the clustering of data points around the diagonal line suggests that the test signal was accurate in predicting average aided speech levels.

Figures 9 and 10 demonstrate how well the Audioscan SWEPT and DYNAMIC test signals estimate aided performance as compared with real speech for two hearing instruments. The dashed lines show the hearing aid output levels measured in 1/3-octave bands for 2 minutes of running speech

Figure 7. Comparison of the Audioscan DYNAMIC test signal with a real speech test signal as a function of time (secs).

Figure 8. The $\frac{1}{3}$ octave band levels of aided speech as predicted by the Audioscan SWEPT average test signal are plotted against the measured $\frac{1}{3}$ octave band levels of the “rainbow” passage processed by 34 hearing instruments. If measured points fall on the diagonal line, the test signal provides an accurate estimate of the hearing instrument’s performance for speech.

Figure 9. Predicted real-ear hearing aid performance (dB SPL) as a function of frequency (Hz) for the Widex Senso™ hearing instrument using Audioscan Speechmap/DSL test signals (DYNAMIC and SWEPT) compared with real speech.

Figure 10. Predicted real-ear hearing aid performance (dB SPL) as a function of frequency (Hz) for the Siemens Prisma™ hearing instrument using Audioscan Speechmap/DSL test signals (DYNAMIC and SWEPT) compared with real speech.

(the rainbow passage), while the solid lines show levels measured using the SWEPT and DYNAMIC test signals. The lower dashed curve is the level exceeded by 70% of the speech samples in each band, the middle dashed curve is the average level over the entire speech passage and the upper dashed curve is the level exceeded by 1% of the samples in each band. The upper and lower solid curves (MAX and MIN) show the dynamic range as measured using the DYNAMIC test signal while the middle solid curve (AVG) is the long-term average speech spectrum as measured by the SWEPT test signal. Results suggest that the test signals accurately predict both average aided speech levels and the peaks and valleys of average aided speech.

The technical sophistication of digital hearing instruments allow input signals to be manipulated and modified in a seemingly infinite number of ways. The important thing to remember is that the goal of fitting a hearing instrument remains the same—to ensure audibility of important environmental sounds, especially speech, while limiting loud sounds so that they are not uncomfortable. The digital ‘advantage’ is the flexibility avail-

able to help achieve this goal. The real-ear system ‘advantage’ is the ability to directly measure what is being delivered at the eardrum, regardless of the hearing instrument being fitted, to ensure the goals of audibility, comfort and safety have been achieved.

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