Using digital hearing aids to visualize real-life effects of signal processing

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Digital signal processing (DSP) hearing aids have made many advances since they were first successfully introduced in 1996. Today’s premium DSP hearing aids use far more sophisticated algorithms to process sounds than those of a few years ago. Some examples are modern algorithms for compression, feedback management, noise reduction, and control of directional microphones.

The focus on finding more sophisticated ways to process sounds may be one reason that digital hearing aids accounted for 66% of all hearing aid sales in the United States in 2003. As DSP technology continues to advance, it is being extended to hearing aid applications not directly related to sound processing. Such applications are already evident in the use of DSP hearing aids for in situ threshold measurement and self-diagnostic testing. Another new application of DSP is for sound pressure level (SPL) measurement.

A DSP AID CAN MEASURE AS WELL AS AMPLIFY

There may be no obvious similarity between a sound level meter (SLM) and a hearing aid (HA). After all, an SLM measures sound while an HA amplifies it. Despite this and other differences, what both instruments have in common is using microphones to transduce the acoustic signals. What’s different is how they process the signals afterward.

Figure 1 shows the simplified functional block diagrams of an SLM and an HA. In the SLM, the level of the sound transduced by the microphone is measured within the selected filter (in dB SPL re: 20 µPa). The filter bandwidth, depending on the sophistication of the SLM, may range from the broad frequency range in filter settings like the A-scale to a narrow frequency range like 1/3-octave. The total sound pressure level within each filter channel can be displayed on an LCD panel either one channel at a time, as the specific frequency channel is selected, or, in more advanced SLMs, as a spectrum displaying the levels of all channels simultaneously.

Sounds transduced by a multi-channel hearing aid are split into channels and processed (amplified) following specific instructions or algorithms. The output of all the channels sums before the receiver transduces it back into an acoustic form. The signal processing carried out in each channel depends on many audiologic and acoustic parameters. One of these acoustic parameters is the instantaneous sound pressure level in each channel. Thus, with a suitable algorithm and proper calibration, one could display the instantaneous sound pressure level within each of the multiple channels via the programming software.

Doing this would cause a hearing aid to function, in effect, like an advanced SLM. This could offer an accurate and convenient way to quantify sounds at the wearer’s ears. As we will describe later, this capability has many potential applications in the hearing aid fitting/fine-tuning process.

HOW THE SOUNDTRACKER FUNCTIONS

Envisioning these applications, Widex included the SoundTracker in the Senso Diva hearing aid. This feature releases the previously hidden potential of digital hearing aids to measure sound pressure levels. The purpose is to measure and display the sound level of the external inputs and estimate the output of the hearing aid in the wearer’s ear canal as it is used. This feature taps into each of the Diva’s 15 channels to measure its sound level.

When the SoundTracker is activated, the hearing aid is muted in order to display the sound level information. However, all the processing functions of the hearing aid,
except the feedback path estimation, remain active. Therefore, this feature can display the effect of algorithms such as noise reduction, compression, and directional microphone. The distribution of the input sound levels is displayed in 1/3-octave bandwidth for the speech frequencies (500–4000 Hz). This is shown by the height of the light-colored bars in Figure 2. For the lowest channels and the highest channel, the bandwidth is slightly broader (approximately 2/3 octave).

The sound level is displayed in dB HL instead of the dB SPL because it reflects the sound level relative to normal-hearing individuals and thus can be directly compared to the audiogram. For example, normal-hearing individuals have a hearing threshold of 0 dB HL across all frequencies. However, if expressed in dB SPL, the sound level for normal hearing will vary depending on the test frequencies and how threshold is measured (i.e., under headphones, in the sound field). Sound levels in the dB SPL scale can be converted to the dB HL scale by applying the Minimum Audible Pressure (MAP) or Minimum Audible Field (MAF), depending on how the sound level is measured (i.e., headphones for MAP and free field for MAF).

The measurement reference is at the eardrum. To use an eardrum reference instead of the conventional microphone reference, input levels (in dB SPL) measured at the microphone are first transformed to estimated SPL at the eardrum by subtracting the average free-field-to-microphone transfer function and adding the average free-field-to-eardrum transfer function.

Estimating the unaided SPL at the eardrum by measuring the input to the hearing aid via the hearing aid microphone is straightforward. However, estimating its output may be more difficult. Although one can insert an external probe microphone into the ear canal for measurement, that may be inconvenient. Also, variability from probe-tube placement, depth of insertion, compression of probe tube, etc., would affect the accuracy of the measurement.

The other approach is to simulate the output using knowledge of the input levels to the hearing aid as a reference. The output may be estimated by summing the instantaneous input level with the associated simulated in situ gain. Thus, the output reflects the effect of the signal processing performed by the hearing aid, including compression and noise reduction. However, effects such as earmold plumbing (e.g., vents) are not included in the simulated output because the actual output of the hearing aid is not measured (these effects are reflected in the magnitude of the in situ threshold or Sensogram).

The magnitude of the applied insertion gain is shown by the height of the darker-colored bars on top of the lighter-colored bars in Figure 2. The overall output is shown by the height of the bar (dark plus light color). The maximum peak level registered in the preceding second is displayed as the line connecting the different frequencies.

Unles the individual real-ear-to-coupler difference (RECD) corrections are applied, the input and output levels are reported with the age-appropriate RECD corrections and FF2Mic corrections (for the different Diva models) to simulate real-ear SPL. To enhance the usefulness of the display, in situ thresholds (or Sensogram) are included.

The advantage of determining in situ thresholds is that hearing aid factors such as ear canal volume and vent effects are reflected in the magnitude of the Sensogram to yield a more accurate specification of gain. Because both the Sensogram and the input levels in the SoundTracker have the same reference, the measured input level and the simulated output level can be compared with the Sensogram. The difference between the sound level and the Sensogram (or in situ threshold) is the sensation level, or the number of dB a particular sound is above the threshold of the individual.

Figure 2. The visual display (level versus frequency channels) of the SoundTracker. The light-colored bars represent the input level in dB HL. The dark-colored bars represent the instantaneous insertion gain at the specific channel. The simulated output is the sum of the input level and gain. Sound levels above the Sensogram (in situ threshold) are audible, and represent the dB sensation level (SL) of the particular sound. Sound levels below the Sensogram are inaudible.

The hearing aid accurately captures small and rapid input level changes. However, for the changes within the hearing aid to be displayed rapidly onto a computer, a speedier interface is necessary. HIMSA has introduced such an interface, the NOAHlink, which is used in the SoundTracker to speed up data communication. Currently, the screen displaying the frequency level information is updated 10 to 15 times a second.

VALIDATING THE MEASUREMENTS

To ensure that the SoundTracker reports the actual sound level, we compared the levels of stationary signals measured by...
the SoundTracker with those measured on a Quest SLM at the same location 2 feet in front of a Bose loudspeaker. The signals included a white noise (equal spectral level measured with a fixed 1-Hz bandwidth), a pink noise (equal level measured with a logarithmic bandwidth, e.g., 1/3-octave), the ANSI speech noise, and modulated pure tones at 500, 1000, 2000, and 4000 Hz.

We taped the hearing aids to the sides of the SLM microphone to achieve similar microphone location. We set the microphone to 1/3-octave bandwidth measurement on the fast scale.

The results showed excellent agreement for all the stimuli attempted, with deviations of about 1-2 dB, mostly in the higher frequencies. This may be due to the shorter wavelengths of the higher frequencies and the separation between the SLM and hearing aid microphone.

APPLICATIONS OF THE SOUNDTRACKER

Various applications are possible with the SoundTracker. The following section discusses some of these.

Why use a hearing aid to function as a sound level meter?

Knowing the spectral-level content of the input signals at the wearer's ears has significant implications for counseling, verification, and trouble-shooting/fine-tuning of hearing aids. This feature effectively offers this capability. It provides a more accurate measurement of the SPL at the wearer's ears because it reflects the wearer's exact acoustic condition. It is a more convenient and cost-effective way to estimate the SPL in the ear canal because it uses the wearer's own hearing aids and does not require an external probe microphone or cooperation by the patient. We believe its use is justified as long as one realizes the limitations of this method (different I/O scales from conventional, simulated output) and finds ways to ensure its validity (such as making sure that the hearing aid is functional).

For demonstration and counseling

During a patient's initial visit, a key objective is to demonstrate the potential benefits of amplification. Providing a comparison between aided and unaided hearing in reference to the patient's hearing thresholds may prove very convincing. You can use foam plugs as temporary earmolds and program BTE hearing aids with the patient's audiogram. Speak to the patient at a typical conversational distance. With the SoundTracker in the unaided mode, indicate that bars (for a frequency region) below the thresholds (or Sensogram) line are inaudible, while bars above the line are audible. The effect of amplification is to bring the bars above the patient's Sensogram in the aided condition. This is illustrated in Figure 2 also.

Similarly, the practitioner may use this tool to demonstrate why a patient has more problems understanding the voice of a specific individual (e.g., spouse, friend) or why specific phonemes are inaudible.

For linear hearing aid wearers hesitant to try compression instruments, the SoundTracker can show the advantages of enhanced dynamic range compression (EDRC) processing. You can demonstrate that EDRC processing allows more gain for soft sounds and less for loud sounds than does linear processing. To do this, program the hearing aid into linear processing by reducing gain for soft sounds and increasing gain for loud sounds while keeping gain for conversational sounds constant. For demonstration purposes, you can leave the aid in the right ear in its default processing (EDRC) and program the other aid to linear processing. When you speak to the patient softly or move farther away, the patient should see that the gain bars (darker color) for the EDRC processing grow higher than those for the linear processing. As you speak more loudly or move closer to the patient, the gain bars for the EDRC processing become shorter than those for the linear processing (see Figure 3).

One may also use this feature to examine the effects of the noise-reduction algorithm and directional microphones on the output of the hearing aids.

For verification of gain/output

One objective in a patient's second visit is typically to ensure that the chosen hearing aids provide optimal audibility and comfort. To do this, make sure the peaks of the intended amplified sounds are above the Sensogram of the wearer but do not exceed the estimated loudness discomfort levels (LDL) indicated on the SoundTracker screen. In addition, you can use...
it to verify the audibility (and comfort) of average speech spectra produced at various vocal efforts or speech produced by different individuals. Another application, especially with the pediatric population, is the use of the Ling-6 sounds (/i/, /a/, /u/, /l/, /s/, /m/) to verify that the child has audibility for the speech frequencies. In all these cases, the speech peaks should be above the Sensogram.

On the other hand, the precise sensation level (or how far the peaks should be above the Sensogram) for optimal intelligibility is less well defined. That’s because for the same degree of hearing loss, different prescriptive formulas prescribe different gain/output. In addition, several studies have shown that the frequency responses on a hearing aid may vary substantially without causing significant differences in speech intelligibility scores. This is in line with Byrne’s observations that even though real-life speech occurs at different sensation levels for different speakers, speech understanding remains unaffected as long as audibility is ensured.

These findings suggest that precise matching of a hearing aid’s output to a specific sensation level target may be unnecessary. From our early experience with the SoundTracker, it appears that the gain settings on the hearing aids should result in an output for a soft input that is above the Sensogram (i.e., >0 dB SL), but probably not greater than 20-30 dB SL. The output to conversational sounds would not need to exceed 40 dB SL for maximum intelligibility. The optimal SL will likely decrease as the hearing loss worsens. In addition, frequencies below 500 Hz and above 4000 Hz may not require as high an SL as the mid-frequencies. For loud sounds, the peaks should not exceed the wearer’s LDL. Obviously, these are general guidelines and individual preferences must be considered for a satisfactory fitting.

Some may be tempted to use the gain/output recommendations from generic formulae (e.g., NAL-NL1, DSL[i/o]) as targets in assessing if output (as seen on the SoundTracker) is optimal (i.e., if settings are appropriate). While you may use the recommendations as a general guideline, you must refrain from matching these targets to an “average” speech signal or to an individual’s voice.

There are at least two considerations in using the average speech spectrum to determine if the hearing aid output matches the target output. First is the choice of the speech signal with the “average speech spectrum.” Kuk and Ludvigsen pointed out that the term “speech-shaped” signal is imprecise. For example, both the ANSI13 and the ICRA signals14 are speech-shaped composite signals, yet their spectra are different. This suggests that output from the hearing aid would vary depending on the stimulus.

Further complicating the comparison is the assumption of the speech spectrum that was adopted by the specific generic formula. If a target uses the international high frequencies. Consequently, if a generic formula specifies the target output using a fixed bandwidth (say 1 Hz), then using the SoundTracker (which is based on 1/3 octave) to match that target is not meaningful. The same concern arises in comparing the output measurements on the SoundTracker with the output measurements made on another device, such as a sound level meter or a real-ear test system. These are issues not only with the SoundTracker, but also with verification of non-linear hearing aids in general.

Another issue with using individual speech signals as stimuli for target gain adjustment is the large variability among voices. If you adjust the settings to optimize for a particular person (or sound), that may compromise the satisfaction for other voices (or sounds) because of the differing spectral characteristics of their voices. Therefore, one must proceed judiciously when adjusting the settings on any hearing aid to “optimize” for a specific stimulus.

**For fine-tuning**

By measuring input at the position of the hearing aid microphone, the SoundTracker accurately describes the spectral content of the external “troubling” signal. The 15 channels used in the hearing aid would identify the frequency spectrum of the troubling sound (and thus the frequency channel for adjustment) to within 1/3 octave. The input display on the SoundTracker provides information on the gain parameter to adjust.

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**Figure 4. Effect of measurement bandwidths on output.** This figure shows a higher output as the measurement bandwidth is broadened (from 1 Hz to 25 Hz) and a change of spectral tilt as a logarithmic bandwidth is used (1/3 octaves).
For example, on the Diva hearing aid, the Gain\textsubscript{soft} parameter controls sounds below 50 dB HL, increasing its effect as input level decreases from 50 dB HL. The G\textsubscript{loud} parameter controls sounds above 50 dB HL, increasing its influence as input level increases beyond 50 dB HL. The G\textsubscript{normal} parameter controls sounds between 35 and 75 dB HL, decreasing its effect as the input levels depart from 50 dB HL. This is represented by the geometric shapes on the left-hand side of Figure 5.

Consequently, you need to examine the sound level of the input signal in the unaided condition to determine which gain parameter to adjust. On the other hand, you examine the output level in the aided condition to see if the gain parameter for a specific channel warrants adjustment.

An example will illustrate this application. Figure 6 shows the output of the hearing aid to a “paper crumbling” noise. The wearer’s complaint is that the noise was “too irritating.” If you examine the input (the height of the light-colored bars), it’s obvious that the spectrum is relatively broad and includes energy from low to high frequencies. Rather than adjusting the gain parameter at all frequencies, you need first to determine which frequency region should be (and can be) adjusted to minimize the complaint without compromising the fit.

If you look at the low frequency, it’s clear that its output is significantly above the patient’s Sensogram. However, the output originates mainly from the input because only a small amount of gain is provided (see the thin dark red line). Thus, even if you reduce the low-frequency gain, that may not resolve the complaint.

If you examine the high frequency, you’ll see that even with amplification, the high-frequency output has not reached audibility, i.e., is not above the Sensogram. This suggests that the patient’s complaint does not originate from the high frequency. Reducing gain in the high frequencies will not improve satisfaction, but may compromise speech intelligibility.

On the other hand, when you examine the mid-frequency region, it’s clear that output there is sufficiently above the Sensogram and that the input level of the crumbling noise occurs mostly around 40-50 dB HL. Consequently, reducing the Gain\textsubscript{normal} parameter in the 1000-2000 Hz region may be the most effective adjustment.

**CONCLUSION**

Although the SoundTracker performs some of the same functions as an advanced sound level meter and a real-ear measurement system, it should not be viewed as a substitute for them. Many SLM applications, such as determining the overall level, or those of a real-ear system, such as determining the absolute dB SPL, are
beyond the scope of the SoundTracker. However, if you recognize its limitations and interpret its results carefully, this tool offers an accurate, convenient way of estimating the sensation level of sounds at the wearer’s ears. Such information is helpful in all stages of the hearing aid process from initial counseling to fitting/verification to fine-tuning.

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