INTRODUCTION: HEARING AID FITTING AND VERIFICATION

Modern multi-channel hearing aids have many adjustable parameters. It is usually possible to adjust their compression threshold and the gain at two or more levels, either separately in each frequency channel or in groups of channels. It may also be possible to adjust the crossover frequency between channels (especially in aids with a small number of channels).

Sometimes it is also possible to adjust the attack and release times of the compression or to select among different forms of compression, such as dual-time constant or fast-acting (syllabic).

Finally, it may be possible to adjust extra features, such as low-level expansion (described below), feedback cancellation, or noise reduction.

The large number of parameters means that it is essential to have some well-defined way of setting the hearing aid to suit the individual patient and to verify that the hearing aid is performing as desired.

Fitting modern hearing aids is typically a two-stage process. The first stage involves setting the parameters of the hearing aid according to a formula or computer-based procedure, usually using information from the audiogram, which may be combined with information on loudness discomfort levels. Examples of procedures of this type are DSL(i/o), NAL-NL1, CAMEQ, and CAM-REST. These procedures give only an initial fit; they are intended to give reasonable gains “on average,” but further adjustment is nearly always needed.

The second stage involves fine-tuning to suit the acoustic and auditory characteristics of individual ears and to suit individual needs and preferences.

It is well established that the gains programmed into a hearing aid do not correspond accurately to the gains achieved for an individual real ear. Therefore, it is essential to assess the gains actually achieved and to make adjustments when necessary to ensure that the goals of the fitting procedure are met and that signals such as speech are audible over a wide frequency range. This is often done using real-ear measurements based on a probe microphone. Typically, the measurements are made using artificial test signals, such as swept pure tones, bands of noise, or a speech-shaped noise (steady noise with a spectral shape that matches the long-term average spectrum of speech).

PROBLEMS WITH THE TRADITIONAL APPROACH

There are several limitations to the traditional way of making real-ear measurements:

❖ The gains actually achieved for real-life signals such as speech and music may differ considerably from the gains measured with steady signals, such as tones and noise. The difference depends on the number of channels in the hearing aid, the speed of the compressors, and the compression thresholds. This is the case even when features such as noise reduction or feedback cancellation are not present or are not activated.

❖ When a hearing aid incorporates feedback cancellation, pure-tone test signals may be interpreted by the aid as feedback, and a pure-tone test signal is then partially or completely canceled. The gains measured when this happens may be totally unrepresentative of the gains achieved in everyday life. In some hearing aids, it is possible to disable the feedback-cancellation system, but this may change the effective frequency response of the aid and may also limit the gain that can be achieved.

❖ Many hearing aids incorporate some form of noise reduction. If any particular spectral region appears to be dominated by noise (or by any steady sound), then the gain of the hearing aid in that frequency region is reduced. If the test signal used to assess the gain of the hearing aid is a steady noise or a tone, the gain that is
measured may be much less than actually achieved for everyday sounds such as speech and music. In some hearing aids, it is possible to disable the noise-reduction system, but this may change the effective frequency response of the aid, and the gain applied by the aid may differ from that obtained when the noise reduction is active.

The sounds that are used to make the measurements have no relevance to the sounds that the hearing aid user experiences in everyday life, for example the voice of a spouse or parent.

ADVANTAGES OF SPEECH MAPPING

A major advantage of the speech mapping approach is that the effective amplification provided by the hearing aid can be assessed using realistic signals such as speech or music and with the aid in its normal mode of operation (with features such as feedback cancellation and noise reduction enabled, if appropriate). Thus, the influence of factors such as number and bandwidth of channels, compression speed, etc., is automatically taken into account.

Furthermore, any effects of feedback cancellation or noise reduction on the performance of the aid are automatically included in the display. Effectively, what you see is what you get. The uncertainties and errors produced by the use of artificial test signals are avoided.

The test signals used to assess hearing aid performance can be generated “live” (for example, by having a spouse or parent talk in a normal conversational way), or sounds can be generated via the sound system of the PC. The software of most speech mapping systems includes a variety of calibrated test signals, including speech in quiet, speech in various types of background sounds, traffic noise, and various types of music. Thus it is possible to simulate many of the everyday situations experienced by the individual client.

Because the displays in speech mapping systems incorporate the audiogram of the client, it is easy to see which parts of the spectrum of the signal, for example speech, are audible to the client. If the peaks of normal conversational speech reach the top of the speech banana over the whole frequency range of the banana, then that speech should be fully audible. The dispenser can thus be confident that the client’s ability to understand speech will not be compromised by limited audibility.

If the peaks of normal conversational speech do not reach the top of the speech banana for some frequencies, this indicates that speech audibility is not optimal for those frequencies, and the gain programmed into the hearing aid should be increased appropriately.

If the peaks of normal conversational speech fall well above the top of the

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Speech banana for some frequencies, this may indicate that the hearing aid is providing too much amplification for those frequencies, and it may need to be reprogrammed to reduce the gain.

Some other advantages of speech mapping are listed below:

- The display may make apparent effects that would not be revealed by conventional measurements, such as effects of low-level expansion (a reduction of gain for low-level inputs, sometimes called “squelch”). Such low-level expansion is often used in hearing aids to stop the user from hearing low-level noise generated in the aid, typically by the microphone and analog-to-digital converter. Traditional probe-microphone measurements are typically made using test signals with levels ranging from about 50 to 85 dB SPL. Such measurements do not reveal the action of low-level expansion. In many hearing aids the “threshold” below which expansion operates can be programmed, and it is often set to a value between 35 and 45 dB SPL. It is not generally realized that, for weak speech with an overall level of, say, 55 dB SPL, the effective level of the speech in the higher-frequency channels of a hearing aid may be only 30-40 dB SPL. If the expansion threshold is above this range, then expansion may be applied at high frequencies, leading to reduced audibility and reduced intelligibility. Speech mapping can immediately reveal such an effect, if it is happening. If the client needs to understand low-level speech, then the threshold for the low-level expansion may be reduced or disabled altogether.

- The client and his relatives become more involved in the fitting procedure. If a hearing aid is providing only limited audibility of speech, this limitation can immediately be demonstrated and intuitively understood. This is especially important when fitting hearing aids to infants and children. If it is impossible to provide full audibility of speech, perhaps because of the severity of the hearing loss at high frequencies, the client and relatives can understand why this is so and can appreciate the consequences. If a more satisfactory fit can be achieved, the client and relatives can immediately see the difference on screen, and hopefully the client can also hear the difference.

A speech mapping system can indicate whether or not problems with loudness discomfort will occur in everyday life. The pre-recorded test signals can be used to simulate situations where discomfort might be encountered, and the display will indicate if any of the signals approach the loudness discomfort level and, if so, at what frequencies. If necessary, the parameters of the compression or output limiting in the aid can be adjusted to avoid loudness discomfort. In this way, return visits for re-adjustment of the aid(s) are likely to be minimized. In fully binaural systems, such as the Aurical Visible Speech system described earlier, it is possible to display the live spectrum of the sound at both eardrums simultaneously. This can be useful for illustrating to the client that sometimes one ear receives a better signal than the other (for example, when a talker is not in front, but to the right or left), and that the use of two hearing aids will prevent signals from the side from being missed. An effective way of illustrating this to the client is to leave the binaural display running throughout the counseling session, and to move around while talking to the client.

CONCLUDING REMARKS

Speech mapping provides a valuable tool for the hearing aid dispenser. It allows markedly improved accuracy in the verification and fitting of hearing aids. It provides an immediate indication of the audibility of important everyday signals such as speech, including the speech of relatives. Speech mapping permits the dispenser to adjust the parameters of a hearing aid to optimize speech audibility while avoiding loudness discomfort. It also involves clients and their relatives in the fitting process, leading to greater understanding and satisfaction. Finally, its use is likely to reduce the number of post-fitting visits, thereby saving time and money.

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