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The Value of Speech Mapping in Hearing-Aid Fitting

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Introduction: Hearing-aid Fitting and Verification

Modern multi-channel hearing aids have many adjustable parameters. It is usually possible to adjust the compression threshold and the gain at two or more levels, separately in each frequency channel or in groups of channels. It may also be possible to adjust the crossover frequency between channels (mainly possible 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 between different forms of compression, such as dual time constant (Moore and Glasberg, 1988; Stone et al., 1999) or fast-acting (syllabic) (Villchur, 1973; Moore et al., 1992). 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 person, and to verify that the hearing aid is performing as desired.

The fitting of modern hearing aids typically involves two stages. The first stage involves setting the parameters of the hearing aid using 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) (Cornelisse et al., 1995), NAL-NL1 (Byrne et al., 2001), CAMEQ (Moore et al., 1999) and CAMREST (Moore, 2000). 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 (Swan and Gatehouse, 1995; Aarts and Caffee, 2005). 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. This is often done using "real ear" measurements based on a probe microphone (Mueller et al., 1992). Typically the measurements are made using artificial test signals, such as swept pure tones, bands of noise, or a "speech-shaped" noise (steady noise whose spectral shape 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 (Stone and Moore, 1992; Verschuure et al., 1996; Souza, 2002; Henning and Bentler, 2005; Jenstad and Souza, 2005). 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 cancelled. 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 it 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.

The Solution: Speech Mapping

A solution to all of the problems described above is provided by "speech mapping" systems such as the Aurical "Visible Speech" from GN Otometrics. The speech mapping system provides a visual display of the short-term spectrum of sounds in the ear canal of the hearing aid user, and provides a direct indication of the audibility of sounds that are important for the user, such as the speech of a partner or other relative.

This particular speech mapping system includes the following hardware (wireless SpeechLink 100 binaural neckset) and the Visible Speech software:

- Two modules each containing a reference microphone and a flexible probe microphone. A module is hung from each ear, using soft and flexible rubber. The probe microphone is initially calibrated relative to the reference microphone, and can then be inserted into the ear canal of the user, prior to fitting the hearing aid. The spectra of the signals from the microphones are calculated in real time, both for the reference and probe microphones.

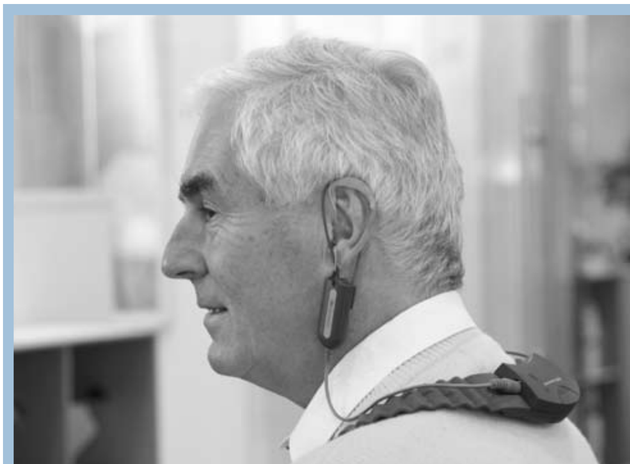

Fig. 1: Modern probes and a wireless binaural neckset.

- The spectra of the microphone signals are transmitted to a computer (PC) via a wireless link which is mounted on a flexible rubber "collar". This collar starts at the lower base of the back of the neck and runs over the shoulders. It is light and comfortable to wear.

- The PC produces a real-time running display of the spectrum in the ear canal. The peak levels can be "held" if desired. The spectrum can be displayed either in a high-resolution format or in a more familiar format based on 1/3-octave bands. In either case, the levels are adjusted so as to reflect the audibility of the ear-canal signals at each frequency. The spectral levels can be displayed either in dB Sound Pressure Level (SPL) or dB Hearing Level (HL). The display includes a plot of the hearing thresholds of the client and of the loudness discomfort levels of the client (which can be estimated from the audiogram if they have not been directly measured). The display can also include

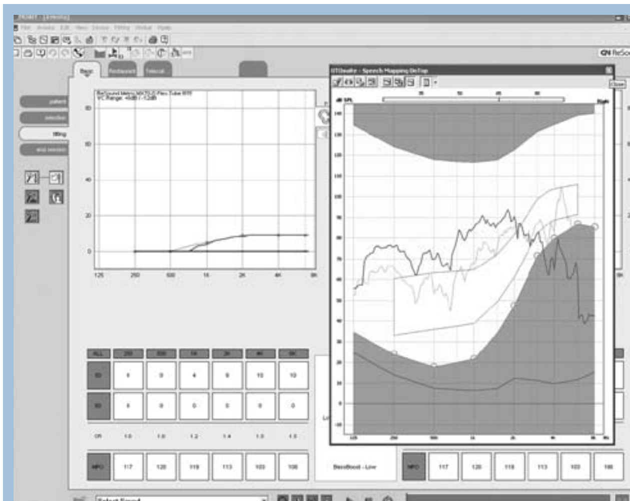

Fig. 2: Speech Mapping with "On top" mode.

the “speech banana” showing the range of frequencies and levels occurring in speech at a typical conversational level. The display can be set to “On top” mode, which means that it remains visible on the PC screen even if other software, such as hearing-aid fitting software, is running.

- If desired, the system can be used with a NOAHlink unit, so that the setting of the hearing aid can be adjusted while the Visible Speech system is in place and running and without any wires connecting the client to the PC. The NOAHlink unit attaches easily to the front part of the collar described above.

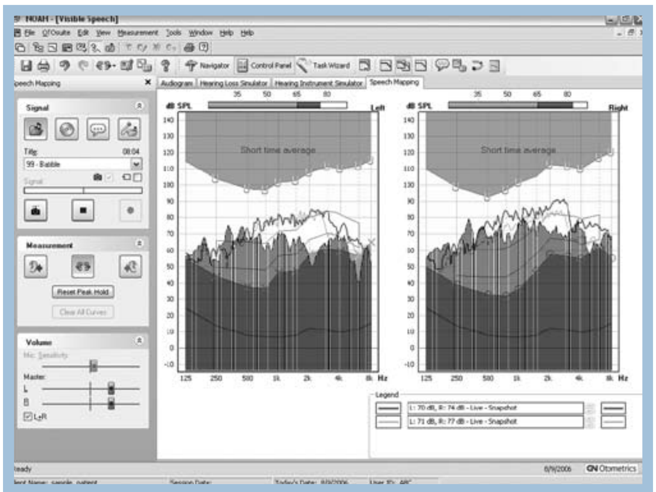

Fig. 3: Live sound and built-in sound library.

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, and so on 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 completely avoided.

The test signals used to assess the performance of a hearing aid or aids can either be generated “live” (for example a spouse or parent can talk in a normal conversational way), or sounds can be generated via the sound system of the PC. The software includes a large 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 display incorporates 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 ability of the client 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 speech banana for some frequencies, this may indicate that the aid is providing too much amplification for those frequencies, and the aid may be re-programmed to give lower gain.

Some other advantages of speech mapping and Visible Speech 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 analogue-to-digital converter. Traditional real-ear probe microphone measurements are typically made using test signals with levels ranging from about 50 dB SPL 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 realised 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 (Moore et al., 1999). If the expansion threshold is above this range, then expansion may be applied at high frequencies, leading to reduced audibility and reduced intelligibility (Moore et al., 2004). Speech mapping can immediately reveal such an effect, if it is happening. If the client has a need to understand low-level speech, then the threshold for the low-level expansion may be reduced, or the expansion may be disabled altogether.

- The client and their 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 their 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 provide an indication of whether 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 whether 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 so as to avoid loudness discomfort. In this way, return visits for re-adjustment of the aid(s) are likely to be minimised (Cunningham et al., 2002).

- The Aurical Visible Speech system is entirely wireless, so the client can move around while wearing the system, with the results displayed on-screen. This can allow the assessment of speech audibility when listening to a person from the opposite side of the room, or even in the next room.

Apart from the benefits of speech mapping and Aurical Visible Speech discribed above, this particular system includes several extra features which can be very useful. These include:

- A hearing loss simulator. This tool allows relatives of a hearing-impaired person to experience what it is like to have a hearing loss.

- A hearing instrument simulator. This tool simulates the effect of multi-channel amplification and can be used to demonstrate the benefits of amplification for a client who has never worn a hearing aid. The tool can also be used with the hearing loss simulator to demonstrate the benefits of a hearing aid to a relative of the client.

Concluding Remarks

Visible Speech 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 signal such as speech, including the speech of relatives, it makes it possible to adjust the parameters of hearing aids to optimise the audibility of speech while avoiding loudness discomfort, it involves the client and their relatives in the fitting process, leading to greater understanding and satisfaction, and it is likely to reduce the number of post-fitting visits, saving time and money.

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