Consumer-Based Electroacoustic Hearing Aid Measures

Mesures électro-acoustiques des prothèses auditives axées sur le consommateur

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Key words: hearing aid measurement, electroacoustic testing, complex signals, signal processing hearing aids

Abstract

The standardized procedures established for the measurement of linear and automatic gain control (AGC) hearing aids, available on commercially available measurement systems, make it easy to conduct electroacoustic analyses. However, these procedures fail to provide an adequate characterization of the electroacoustic performance of a hearing aid when it is worn in real life by a hearing impaired listener. This paper reviews a variety of existing and proposed procedures for the electroacoustic measurement of hearing aids with respect to their ability to provide answers to five questions: (1) To what extent will desirable, target sounds (e.g., speech, music) be audible? (2) Will the hearing aid make use of the listener's full dynamic range? (3) Will sounds from everyday life cause discomfort while wearing the hearing aid? (4) How will speech intelligibility/quality degrade in various listening environments? and (5) How will changes in hearing aid processing affect listeners' performance and their perception of sound quality over the normal operating range of the hearing aid? After reviewing existing and proposed hearing aid measurement standards and published descriptions of experimental hearing aid test systems, it is concluded that new test protocols are required to answer most of the questions. These protocols would be consumer-based in the sense that they would be oriented to the hearing aid wearer and would provide measures that are meaningful in terms of real-life experiences. An experimental hearing aid test system (HATS) developed at the Hearing Health Care Research Unit provides a platform for implementing and evaluating such consumer-based tests.

Résumé

Les procédures étalonnées pour la mesure des prothèses auditives avec système d'amplification linéaire et système de compression d'entrée (AGC) disponibles sur des systèmes de mesure commerciaux facilitent l'expertise électroacoustique. Cependant, ces procédures ne fournissent pas une synthèse des manifestations électroacoustiques d'une prothèse auditive portée par un malentendant dans diverses situations d'écoute. Le présent document se penche sur des protocoles existants et proposés par la mesure électroacoustique des aides auditives et révise leur capacité à répondre à cinq questions : (1) Jusqu'à quel point les sons-cibles désirés (parole, musique) seront-ils audibles? (2) Est-ce que la prothèse auditive utilisera tout le champ dynamique de la personne qui écoute? (3) Est-ce que les sons de la vie quotidienne causeront de l'inconfort lors du port de la prothèse auditive? (4) Comment l'intelligibilité et la qualité de la parole se dégraderont-elles dans différents milieux d'écoute? (5) Comment le changement du traitement des prothèses auditives touchera-t-il le rendement de la personne qui écoute et sa perception de la qualité du son sur les modes normaux de fonctionnement de la prothèse auditive? Après avoir examiné les normes de mesures existantes et proposées pour les prothèses auditives et publié les descriptions de systèmes d'évaluation expérimentaux des prothèses auditives, l'auteur termine en disant qu'il est nécessaire d'élaborer de nouveaux protocoles d'expertise électroacoustique afin de répondre à ces questions. Ces protocoles seraient axés sur l'utilisateur de l'aide auditive fournissant ainsi des mesures significatives sur le plan de l'expérience réelle. Un système d'évaluation expérimental des prothèses auditives (HATS) conçu par la Hearing Health Care Research Unit, fournit une plate-forme pour la mise en oeuvre et l'évaluation de tels tests axés sur les consommateurs.

Audiologists, engineers, and others who work with hearing aids are familiar with the standardized procedures established for the measurement of linear and automatic gain control (AGC) hearing aids (e.g., ANSI, 1987; IEC, 1983). Familiar too are the commercially available measurement systems which make it easy to conduct such standardized tests on hearing aids. Unfortunately, neither the publication and acceptance of these standards nor the widespread availability of such inexpensive, automated electroacoustic measurement systems permits audiologists to be confident that a given hearing aid will perform satisfactorily when it is worn in real life by a given hearing impaired listener. This situation has been recognized both by audiologists involved in hearing aid evaluation and by those who write measurement standards:

> "the widely used standard for specifying hearing aid performance ... does not relate to performance on the user" (Beck, 1991, p. 96S). "The results obtained by the methods specified in this standard express the performance under the conditions of the test and may deviate substantially from the performance of the hearing aid under practical conditions of use" (IEC, 1983a, p. 11).

This paper summarizes the need for new test protocols to characterize the electroacoustic characteristics of modern hearing assistive devices and describes some current efforts to that end. These protocols are "consumer-based" in the sense that they are oriented to the hearing aid wearer and are intended to provide measures that are meaningful in terms of real life experiences. A fundamental premise of this work is that the performance data provided by existing electroacoustic test systems differs from the information needed by audiologists. Audiologists need hearing aid characterizations that answer the following questions:

1. To what extent will desirable, target sounds (e.g., speech, music) be audible?

2. Will the hearing aid make use of the listener's full dynamic range?

3. Will sounds from everyday life cause discomfort while wearing the hearing aid?

4. How will speech intelligibility/quality degrade in various listening environments?

5. How will changes in hearing aid processing affect listeners' performance and perception of sound quality, over the normal operating range of the hearing aid?

Available electroacoustic measurement systems fail to answer these questions for much the same reason that conventional hearing assessment procedures do not adequately characterize the listener's residual auditory function for aural rehabilitation purposes: the procedures that are routinely available to audiologists were developed to meet other needs. In the present case, the electroacoustic measurement standards are in place to meet the need for a manufacturing standard. As a consequence, audiologists who seek to characterize the performance of hearing aids using available procedures typically encounter the following problems: (1) conventional, standardized testing procedures are not designed to address real-life performance; (2) published specifications are stated in terms of these conventional, standardized testing procedures so that the opportunities to assess the suitability of a given hearing aid for a given hearing impaired listener are limited; and (3) procedures that could meet the needs of audiologists are not available clinically and need to be developed and/or implemented in hearing aid test systems.

This paper reviews a variety of existing and proposed procedures for the electroacoustic measurement of hearing aids with respect to their ability to provide answers to the five questions listed above. Those considered include the present ANSI standard (ANSI, 1987), a revised ANSI standard (ANSI, 1992), the current IEC standard (IEC, 1983a) and its supplement for AGC hearing aids (IEC, 1983b), a description of hearing aid testing by Bruel and Kjaer (Bareham, 1990), the National Institute of Standards and Technology's (NIST) procedures for hearing aid measurement as described by Burnett (1991a,b), a hearing aid measurement system developed by Kates (1991), and the experimental hearing aid test system (HATS) developed at the Hearing Health Care Research Unit.

These various procedures can be viewed as being located on a continuum with one extreme represented by the tightly controlled, manufacturer-oriented procedures associated with the published standards. These procedures use synthetic signals under artificial but highly reproducible conditions, and they are relatively easy to implement. Such procedures are readily available to audiologists in inexpensive, commercially available hearing aid test systems, but they may be misleading regarding how a hearing aid will respond when worn in real life. At the other extreme are the real-life environments, which hold the greatest importance for clients and clinicians (and in reality for manufacturers as well). Here, the signals occur naturally, but are highly varied. Natural acoustic environments are also varied and uncontrolled; they are less amenable to reproducible testing and are quite difficult to approximate in a test system. Until commercial hearing aid test systems provide appropriate procedures, it will not be possible for most audiologists to test hearing aids under realistic listening conditions.

Stimulus Considerations

In general, the signals of interest for listeners tend to be complex and relatively broadband with variable amplitude envelopes, such as conversational speech and music. They also tend to be highly variable in time. For example, a substantial portion of a speech message contains intervals of silence, and many of the important cues to speech sounds are brief and/or of relatively low energy (see Burnett, 1991a, for a discussion in relation to hearing aids). The signals that are potentially annoying or uncomfortable for listeners are also highly variable. They may be broadband (e.g., a running faucet), narrowband and pulsed (e.g., alarms, a telephone, a fax beeper), or high frequency and transient (e.g., the hiss of an air brake). The signals specified in hearing aid testing standards have properties that are quite different from such real-life signals.

In view of these stimulus factors, hearing aid testing systems could use a hierarchy of "special" signals: continuous speech, syllabic speech (analytic tests), other broadband signals (noise, speech-weighted noise, composite noise), tone sequences, swept tones, stepped tones, or combinations of these. Of course, each type of signal will differ from the others in certain ways, and these differences may significantly influence the way the hearing aid functions and thus, the outcome of the measurement. Neither the desirable (e.g., speech) nor the undesirable (e.g., background noise) signals are well approximated by the swept or stepped pure tones specified for use in the existing standards and used in most commercial and proposed hearing aid test systems.

Table 1. Examples of decisions regarding test stimuli, couplers, and test environment associated with alternative measurement procedures, and the consequences of such decisions.

	Stimuli	Acoustic Environment	Coupled to	Possible Effects
ANSI	Swept tones	Quiet	Coupler, closed	Controlled
IEC	Swept tones	Quiet	Coupler, closed	Controlled
ANSI 1992	Shaped noise	Quiet	Coupler, closed	Controlled
Bareham	Shaped/ pulsed noise	Quiet & noise	Coupler, closed	Controlled
Kates	Shaped noise	Quiet & noise	Coupler, closed	Controlled
Burnett	Shaped noise	Anechoic & noise	Manikin, simulator	Controlled
Real Life	Speech, music various noises	Reverberant & noisy, with a wide range of levels	Ear, sound field, open	Feedback; stimulus interactions

Towards More Realistic Conditions

Several steps have been taken towards the use of more realistic test signals in hearing aid testing. One approach is to use broad band test signals. White noise is a familiar example of such signals; it has a uniform distribution of energy across frequencies. White noise is undesirable for hearing aid testing, however, because it has more high-frequency energy than will normally be encountered in real life. This energy can affect hearing aid functioning, causing the hearing aid to saturate or go into compression prematurely (relative to signals with a more typical energy distribution). Alternatively, pink noise (where there is additional weighting of low-frequency sounds relative to highs) or speech-shaped noise (where a white noise is both high- and low-pass filtered so that the resulting random spectrum approximates the long-term average spectrum of speech [LTASS]) may be used. Even speech-shaped noise fails to convey many of the important characteristics of speech, however, as the LTASS is naturally dominated by the longer duration, higher energy, voiced vowel sounds, which typically do not pose much difficulty for hearing impaired listeners. Specifically, it is emphasized that the LTASS is an ergodic statistic, so that it is not like any individual speech sound.

In view of these concerns, several other alternative test signals have been proposed (see Table1). The most familiar of these to audiologists are the speech-weighted composite noise, used in the Fonix 6500 system (Frye, 1986), and the "Newby" noise, used in the AudioScan system (EDI, 1992). These signals are composite noises formed from a tonal sequence constrained to have the range of peaks in the signal approximate the peaks of real speech (Frye, 1987), or tonal or noise sequences having both certain temporal and long-term spectral characteristics similar to speech (e.g., Bareham, 1990c; EDI, 1992). Others have been proposed. Bareham (1990c) describes how a test signal created by pulsing the LTASS at a speech-like rate can generate informative test results in certain hearing aids having complex speech-sensitive circuits. Clearly, these signals capture certain aspects of the temporal structure of speech that are desirable. However, they each represent compromises of different sorts dictated by the limitations of the hardware devices on which the hearing aid testing system has been based. While each of these approaches may capture one or more important aspects of real speech, they fail to capture the range of features found in speech and to characterize the range of listening conditions encountered by the listener in everyday life. For example, real speech is typically broadband or multi-band at any point in time and may be voiced or unvoiced with varying spectrotemporal characteristics. In addition, signals and their backgrounds occur at various levels and signal-to-noise ratios (SNRs); the listening environment may not be anechoic or even quiet, causing reverberation and/or background noise to interfere with the intended signal. Finally, hearing aids are worn on or in the ear, not coupled in a test chamber.

A natural extension of these approaches is to use real speech, digitized and stored on computer disk and replayed as required. One such approach used in our system employs digitized tokens of natural speech, selected to represent the spectral extremes of the range of English sounds. Following the notion of Ling's (1989) revised 5-sound approach to measuring a child's ability to hear speech sounds, we used the six sounds "sh," "oo," "m," "ah," "s," "ee". We have also used samples of digitized, continuous speech; an alternative we are investigating employs standardized spoken language databases now available on compact disk (e.g., the DARPA/NIST speech database; Pallett, 1988).

Table 2. Examples of situations where changing the test signal changes the outcome of the measurement in a significant manner.

Effect	Conditions	Description
Blooming	Pure tones with AGC	Apparent broadening of frequency response at high input levels (e.g., Klingham, 1978)
Discomfort	Broadband vs. Narrowband signals at high levels	Apparent reduction in maximum output with broad band signal (e.g., Stelmachowicz, et al. 1990)
Saturation	Broadband vs. Narrowband	Saturation earlier with broadband or peaky signals (e.g., Burnett, et al., 1987)
Frequency	Continuous vs. pulsed signals	Speech-sensitive circuits not activated by continuous, broadband signals (e.g., Bareham, 1990c)

Implications of Stimulus Choice

The selection of signals and test conditions has clear consequences for test results (Table 2). For example, when some hearing aids are tested with a swept or stepped sine wave signal, they exhibit "blooming," an apparent broadening of the frequency response as the input level is increased. When the hearing aid is tested with a broadband signal at various levels, no such broadening of the frequency response is observed. Note that blooming is a real effect, not an artifact. In fact, the response observed with the sine wave is exactly how the hearing aid responds when a sine wave is presented to it. The effect reflects a specific design characteristic of many hearing aids with automatic gain control (AGC). The AGC circuit is designed to be more sensitive at high frequencies than at low frequencies. When the test which is applied to the hearing aid uses high intensity pure tones, only one frequency is presented at a time. In such cases, the AGC circuit may be activated only in the presence of the higher frequency tones, reducing gain in that region relative to the (uncompressed) gain observed when testing at low frequencies. Moreover, one must be cautious in interpreting apparent blooming effects observed with pure tone testing of hearing aids. For example, the blooming effect observed in pure tone testing of a K-amp hearing aid can be confirmed in tests using broadband signals.

A Platform for Hearing Aid Testing and Development

As indicated above, conventional "turnkey" acoustic measurement systems are unable to characterize the response of many hearing aids when worn in real-life situations. Instead, specialized systems must be developed, employing a combination of hardware and software which has the capability of (1) stimulating the hearing aid with the desired acoustic test signal, (2) measuring the output of the hearing aid, and (3) performing the computations required for the specific test (Bareham, 1990a; Kates, 1990).

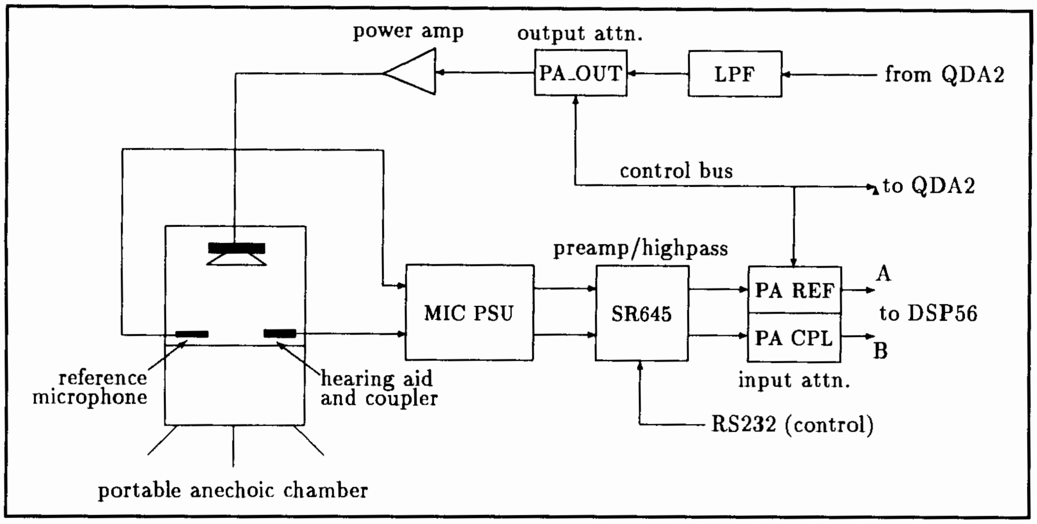
At the Hearing Health Care Research Unit, we have developed two such systems to permit us to implement and evaluate a variety of unconventional testing procedures. Both involve dual-channel testing in a small anechoic chamber, in which both the acoustic input to the hearing aid and the output are sampled digitally and subsequently compared computationally. In our first hearing aid test system (HATS I), we used a combination of special purpose hardware and a 16 bit microcomputer-based digital-to-analog board to generate the required acoustic test signals. An expensive, Bruel and Kjaer, dual-channel spectrum analyzer was used to sample the two acoustic channels (i.e., hearing aid input and output) and to perform the computations required for the tests. In our more recent system (HATS II), we have replaced the spectrum analyzer with a digital signal processing board and appropriate software to perform the computations required.

Figure 1 provides a diagram of the hardware components of the HATS II test system. The dsp-board (DSP56) and the digital-to-analog board (QDA2) both reside in a 33 MHz 80486-based EISA PC that is used for data collection and computations. The DSP56 samples both channels simultaneously and records data directly to disk. The QDA2 plays stimuli that have been uploaded from computer disk to on-board memory. A programmable output attenuator (PA OUT) is used to adjust the SPL in the portable anechoic chamber.

To reconstruct the output signal after the digital-to-analog converter, an 8th order elliptic low-pass filter (LPF) with a cutoff frequency of 12 kHz is used. The output from this filter feeds the power amplifier that drives the speaker in the anechoic chamber. Two responses to this input signal are recorded: (1) a reference channel, recorded using a measurement microphone (Bruel & Kjaer type 4134); and (2) the hearing aid processed channel, recorded using another B&K type 4134 microphone to capture the output of the hearing aid in an IEC standard occluded ear simulator (B&K Type 4157).

Two programmable input attenuators (PA REF and PA CPL) and a dual-channel programmable pre-amp/high-pass filter (SR645), all under computer control, are used to scale the input from the reference and hearing aid channels so that the signal levels are appropriate for the DSP56 analog-

Figure 1. Diagram of the Hearing Aid Test System (HATS II) described in text. The system uses an 80486-based PC (not shown) for data collection and computations. A fast DSP-board (Ariel DSP56) allows reference signals and signals from the hearing aid to be recorded directly to disk and makes it possible to complete computations quickly. A 16 bit digital-to-analog board (QDA2) plays sampled or synthetic sounds stored on the computer disk to stimulate the hearing aid as required for various tests.



to-digital converters. The high-pass filters (f=80 Hz) remove low-frequency room (e.g., ventilation) noise.

The system's software comprises three major components: level setting, data collection, and post-processing. The level setting portion of the program calibrates the output SPL in the anechoic chamber and adjusts the input attenuators to use the full dynamic range of the analog-to-digital converters, which sample the signals from the hearing aid and the reference channel. Samples from both channels are recorded directly to disk during this data collection phase. Following data collection, the data from each channel are post-processed, and the frequency response and other measures are computed.

The HATS II system provides a platform to develop and evaluate new approaches to the electroacoustic measurement of hearing aids. The remainder of this paper summarizes our present thoughts regarding such developments based on an evaluation of existing, proposed, and other possible approaches to hearing aid testing.

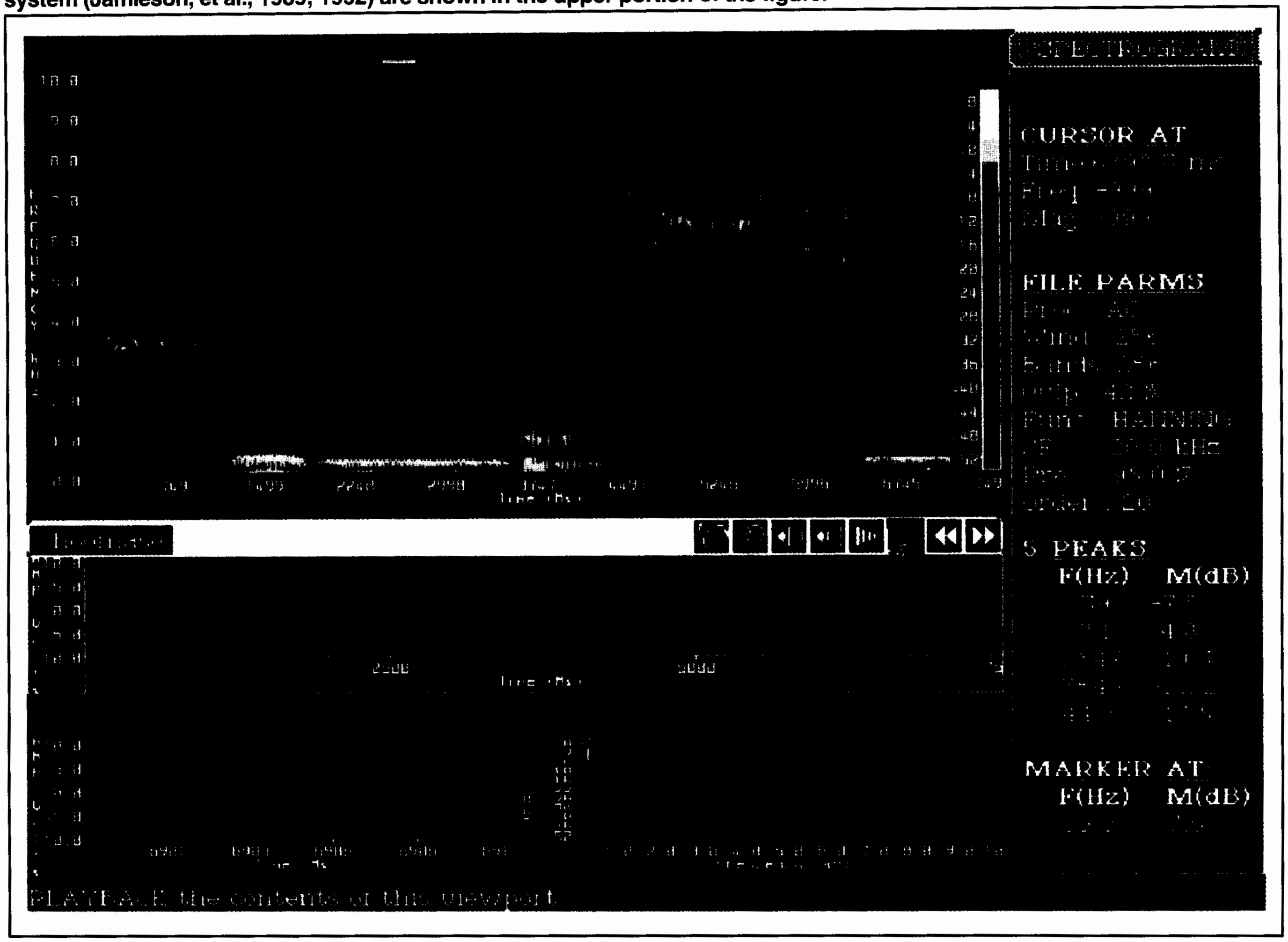
Towards Consumer-based Hearing Aid Testing

To review, consumer-oriented hearing aid measures would establish: (1) the extent to which desirable, target sounds (typically speech and music) will be audible to the person being fitted; (2) that the sounds encountered in normal life will not cause discomfort for the hearing aid wearer; (3) that the listener's full dynamic range will be utilized; (4) to what extent sounds encountered in normal life will reduce intelligibility, degrade sound quality, and/or annoy the hearing aid wearer as a consequence of distortion or saturation; and (5) the likely effects of changes in hearing aid processing on the listener's performance over the normal operating range of the hearing aid. The following sections review alternative approaches to meeting each of these needs.

Assuring Audibility

Hearing aid selection strategies, such as the Desired Sensation Level Method (Seewald, Zelisko, Ramji, & Jamieson, 1991), attempt to maximize the audibility of speech as a primary objective. Typically, speech audibility is defined in terms of a measure of the LTASS. As noted previously, the LTASS does not represent the spectrum of any speech sound. Rather, the LTASS represents a mean spectral value, which is dominated by those sounds that are loud and/or long (i.e., vowels). In fact, none of the available approaches to measuring hearing aid performance with low level inputs (Table 3) provides an answer to the fundamental question of which speech sounds will be audible for a given listener wearing a given hearing aid. One way to obtain this answer is to determine, for a set of representative speech sounds, which sounds will exceed the listener's thresholds in the aided condition when they are amplified by the hearing aid. Speech sounds can be selected, as desired, to establish which target sounds will be audible when the hearing aid is worn and identify any that would be inaudible. For example, by stimulating the hearing aid with the sextet of Ling sounds and examining the output, we could determine which sounds would be audible for a given listener and hearing aid (Figure 2). The spectra of these sounds are clearly different from the LTASS used in most sensible hearing aid prescription procedures. As a consequence, even when a substantial portion of the LTASS is audible, one cannot be assured that these parts of speech will be audible. There is additional concern regarding the audibility of individual sounds in continuous speech in which the sound pattern changes rapidly and many speech cues are both brief and of low energy. All the sounds in this set are intermediate to long in duration, and all are spectrally static. Additional speech cues could be studied as desired by adding additional sounds that reflect the relatively brief and/or dynamic cues that may be

Figure 2. Acoustic analyses of the six sounds used in the Revised Ling Five-Sound Test. From left to right, the sounds are "sh," "oo," "m," "ah," "s," and "ee". Spectrograms of these sounds generated using the AutoCorrelation method in the CSRE system (Jamieson, et al., 1989; 1992) are shown in the upper portion of the figure.



misperceived by a given hearing aid wearer. An example of this approach is provided in Figure 3, where the voiced "th" fricative is related to the listener's threshold of audibility and LDL. It can be seen that no part of the consonant's spectrum would exceed the listener's threshold of hearing, so that cues from this part of the speech signal would be inaudible. The analysis approach can be generalized readily to other speech cues.

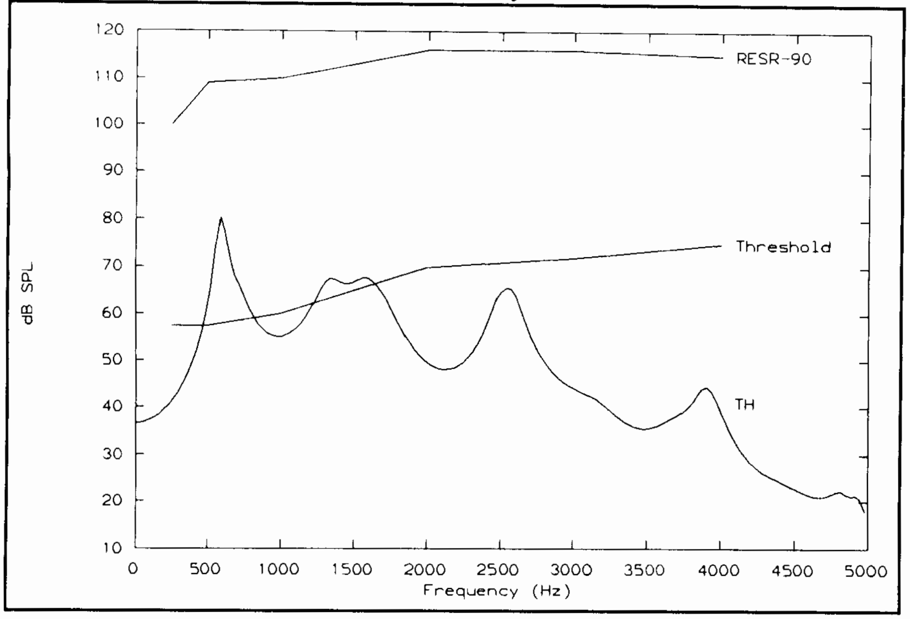
Avoiding Discomfort

An additional consumer need is to establish that the listener's full dynamic range will be utilized without causing discomfort when the hearing aid is worn in everyday life. Table 4 provides an overview of the various approaches to testing with high level inputs. Following Stelmachowicz, Lewis, Seewald, and Hawkins (1990), the preferred approach is to measure hearing aid output in the region of Loudness Discomfort Levels (LDLs), using pure tone signals and to relate these to the listener's LDLs which have been measured with comparable, pure tone signals. This approach is conservative, as it is likely to overestimate the output of the hearing aid in response to complex signals, thus reducing the possibilities for discomfort.

Using the Full Dynamic Range

The third need — ensuring that the full residual dynamic range of the listener is utilized — is inextricably linked with the first two (assuring audibility and avoiding discomfort). Considerations regarding the dynamic range must be subsidiary to audibility and discomfort considerations. Improvements in fully utilizing the dynamic range can be expected to require the adoption of advanced audiometric measurement procedures along the lines of those outlined in this volume by Kiessling.

Figure 3. Comparison of the acoustic spectrum of a brief, relatively quiet speech sound in a format similar to that used in the DSL (Seewald, et al., 1991) hearing aid prescription system to determine the relation of the LTASS with respect to threshold and LDL. In this example, the voiced "th" fricative is related to the listener's threshold of audibility and LDL.



ning at a high level (90 dB). By reducing this signal in ~2 dB steps, while monitoring the output from the hearing aid, one can determine the point at which the hearing aid reaches saturation (Burnett, 1991B).

Another interest is to measure how much distortion a hearing aid introduces in the target signal. While the alternative procedures for hearing aid distortion measurement may be sensible from an engineering perspective (Table 6), none has been validated in terms of the degree of distortion perceived by listeners. In fact, some are known to correlate poorly with perception. As a consequence, significant future efforts need to be directed toward identifying distortion measures that are relevant to the listener's experiences. Some progress has been made in this regard in the field of telephony (e.g., Coverdale, 1983; 1989), but this has yet to be transferred to hearing aid measurement.

Table 3. Implications of alternative procedures for establishing the degree of audibility of amplified sounds in real life.

System	Signal & Conditions	Effects/Questions
ANSI 1987	Swept tones @ 60 dB	What speech sounds are audible?
IEC 1983	Swept tones @ 60 dB	What speech sounds are audible?
ANSI 1992	Speech shaped noise	What speech sounds are audible?
Kates (1991)	Speech shaped noise	What speech sounds are audible?
Bareham (1990a)	Speech shaped noise	What speech sounds are audible?
Burnett (1987; 1991)	Stepped tones @ 60 dB2	What speech sounds are audible?
HATS Proposal	Ling sounds	Relate to thresholds and LDL

Assessing Hearing Aid Performance Over the Normal Operating Range

Predicting the effects of hearing aid signal processing on listeners' performance over the normal operating range of the hearing aid requires assumptions about (1) which signals the listener is interested to hear, (2) how hearing aid performance changes as a function of the level of such an input signal, and (3) the acoustic environment in which this listening occurs. The common assumption is that the most important listening will be for speech. For the reasons discussed above, only some characteristics of this signal are approximated by the LTASS. Unfortunately, real speech is not well suited to the analytic procedures common to modern engineering. Basic analytic methods assume that the system is linear and time invariant (LTI). If we analyze a time varying system using a LTI method, we get an average response. However, using spectrograms, it is possible to analyze time-varying signals (or systems). Because speech signals are explicitly dynamic, they induce changes in the hearing aid over time that can be tracked through comparative spectrographic analyses. One alternative is to test hearing aids with signals that are consistent with available analysis methods, while also having some of the important characteristics of speech. This approach may be supplemented by testing hearing aids with real speech

Limiting Annoyance and Assuring Fidelity

A fourth consumer-oriented need is to establish the extent to which sounds encountered in real life will reduce intelligibility, degrade sound quality, and/or annoy the hearing aid wearer. Meeting this objective requires gaining an understanding of how the hearing aid responds for different types and levels of input signals in the different operating modes of the hearing aid. One interest is to determine the point at which hearing aid amplification saturates. As suggested in Table 5, a reasonable approach to answering this question for some hearing aids is to use a broadband (speech-shaped noise) signal begin-

Table 4. Implications of alternative procedures for determining whether the hearing aid will amplify real life sounds to levels that are uncomfortably loud.

System	Signal & Conditions	Effects/Questions
ANSI 1987	Swept tones @ 90 dB	May overestimate loudness of natural signals
IEC	Swept tones to 90 dB	May overestimate loudness of natural signals
ANSI 1992	Speech shaped noise	May underestimate loudness of natural signals
Kates	Speech shaped noise	May underestimate loudness of natural signals
Bareham	Speech shaped noise	May underestimate loudness of natural signals
Burnett (1987; 1991)	Speech shaped noise to 90 dB	May underestimate loudness of natural signals
HATS Proposal	Stepped tones to 90 dB	Better to overestimate loudness of natural signals

Table 5. Implications of alternative procedures for establishing when the output of a hearing aid will saturate in normal use.

System	Signal & Conditions	Effects/Questions
ANSI 1987	Swept tones @ 90 dB	Spectrum, peaks re: real-life signals?
IEC	Swept tones to 90 dB	Spectrum, peaks re: real-life signals?
ANSI 1992	Speech shaped noise @ 90 dB	Peaks re: real-life signals?
Kates	Speech shaped noise @ 90 dB	Peaks re: real-life signals?
Bareham	Speech shaped noise @ 90 dB	Peaks re: real-life signals?
Burnett (1987; 1991)	Speech shaped noise from 90 dB	Peaks re: real-life signals?
HATS Proposal	Speech shaped noise from 90 dB	Peaks re: real-life signals?

signals that are submitted to more appropriate analyses. Table 7 summarizes one such approach.

Finally, the effects of alternative operating modes of the hearing aid and of alternative listening environments need to be characterized. Here, a primary consideration is that different listening environments may change the way in which the hearing aid processes a given target input signal (e.g., speech). Combining a test signal with a bias signal offers the possibility to study the effects of such environmental factors (Table 8). Different bias signals are selected to set the hearing aid into different modes of operation, which would be experienced within different listening environments, while the test signal characterizes how the hearing aid would process the target signal within each mode. With respect to this type of testing, we are particularly interested in the possibilities of testing hearing aids using MLS analysis (see below).

Future Directions

Our experimental hearing aid test systems, HATS I and HATS II, permit new procedures to measure the electroacoustic performance of hearing aids to be developed and evaluated. One approach we have implemented uses Maximum Length Sequence analysis (MLS), a recently developed method for computing the impulse and frequency response of electroacoustic systems. This method has been used widely in loudspeaker testing and room acoustics (Ando, 1985; Rife & Vanderkooy, 1989; Schroeder, 1979) and holds considerable promise for application with hearing aids. The method uses a special binary sequence (maximal length sequence or MLS) whose auto-correlation is approximately a unit sample. A dual-channel approach is used in our implementation to compute the frequency response of a hearing aid. Briefly, an acoustic MLS is applied to the hearing aid, while the reference and coupler microphone signals are sampled. These two signals are each cross-correlated with the input signal to generate the periodic impulse response for each channel. The frequency response is then computed for each channel, and the result in the reference channel is compared to the result in the coupler channel to determine the frequency response of the hearing aid under test.

Our tests have shown that MLS analysis provides a reliable indication of the response of a hearing aid to a broadband input signal (Figure 4) and that it does so at least twice as quickly as comparable broadband noise-based measurements. Using MLS analysis, a complete set of frequency response curves (with input signals from 50 dB SPL to 59 dB SPL in 10 dB steps) for a hearing aid can be computed in approximately 2 minutes. Moreover, the MLS approach is relatively insensitive to low level background noise in the acoustic test environment, offering significant advantages for clinical applications. In addition, MLS analysis

Table 6. Implications of alternative procedures to predicting the perceived quality/annoyance associated with hearing aid use in real life.

System	Signal & Conditions	Effects/Questions
ANSI 1987	thd with 65 dB tones @ 800, 1600, 2500 Hz	Perceptual relevance?
IEC	thd with 70 dB tones @ 400, 500, 630, 800, 1000, 1250, 1600 Hz; ihd with 200Hz-5 kHz tones @ df = 125 Hz	Perceptual relevance?
ANSI 1992	Coherence	Interpretation? Perceptual relevance?
Kates	Comb-filtered noise @ I * 625 Hz, I= 1, 15; Coherence	Perceptual relevance? Perceptual relevance? Interpretation?
Bareham	Coherence	Interpretation? Perceptual relevance?
Burnett (1987; 1991)	thd with 70 dB tones @ 500, 800, 1600 HzHz	Perceptual relevance?
HATS Proposal	speech-like test signals (e.g., Coverdale, 1983; 1989)	Seek perceptually relevant measures

Table 7. Alternative approaches to characterizing real-life hearing aid performance over the normal operating range of the instrument.

System	Signal & Conditions
ANSI 1987	Swept tones @ 50-90 dB
IEC	Swept tones 50-90 dB
ANSI 1992	Speech shaped noise @ 60 dB
Kates	Speech shaped noise @ 50-90 dB
Bareham	Speech shaped noise @ 50-90 dB
Burnett (1987; 1991)	stepped tones 60 dB (20 bands)
HATS Proposal	MLS and real speech from standardized database

Table 8. Alternative approaches to characterizing the real-life performance of a hearing aid in its various operating modes.

System	Signal & Conditions
ANSI 1987	No special provision
IEC	No special provision
ANSI 1992	No special provision
Kates	Speech shaped noise @ 50-90 dB with bias tones
Bareham	Speech shaped noise @ 50-90 dB; may pulse, and/or use bandbass noise background
Burnett (1987; 1991)	Stepped tones 60 dB (20 bands)
HATS Proposal	MLS, with various types of bias signals

permits the response of a hearing aid to be measured in the presence of certain bias signals so that a hearing aid can be measured in the various states in which it may function when worn in real life.

Another approach we have been exploring involves the direct study of the time domain of acoustic signals at hearing aid input and output using the spectrographic analysis facilities available in the CSRE package (Jamieson, Ramji, & Nearey, 1989; Jamieson, Ramji, Kheirallah, & Nearey, 1992). Figure 5, which provides an example of a signal input to and output from a digital signal processing hearing aid with Automatic Signal processing (ASP) and compression, demonstrates the power of this technique for exploring the effects of hearing aid signal processing on various speech cues. In future work, we expect to extend these acoustic analyses by simulating the effects of neural transduction and processing in the peripheral auditory system using a model being developed at the Hearing Health Care Research Unit (Cheesman, Jamieson, Krol, & Kheirallah, 1992).

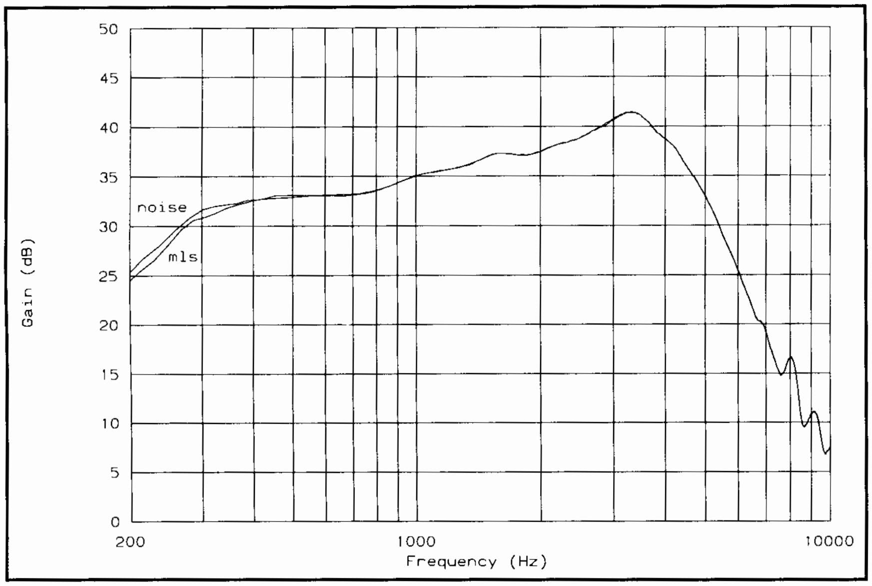
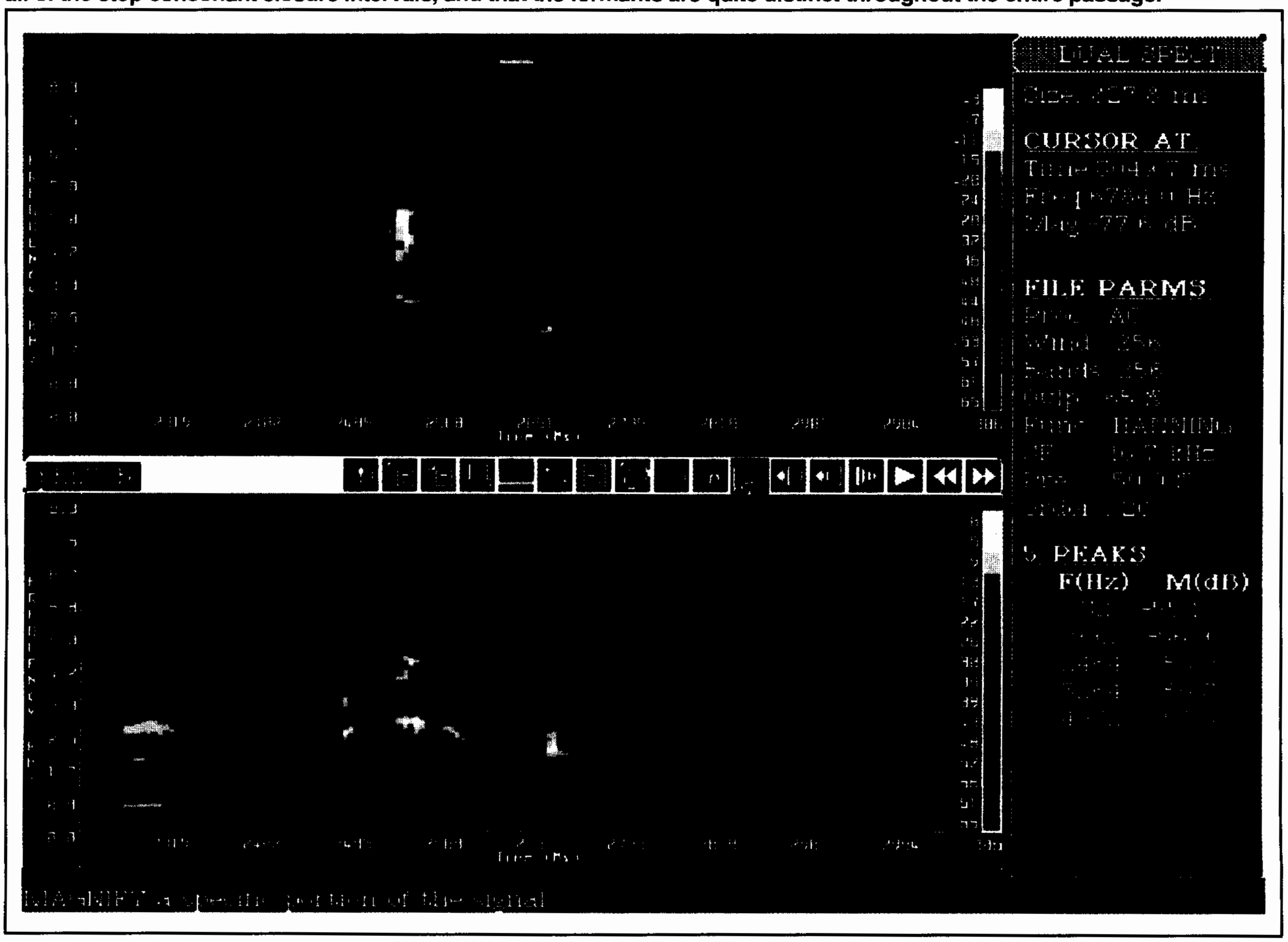


Figure 4. The response of a linear hearing aid to a broad band input signal, as measured using two approaches. In one method, the hearing aid is stimulated with speech-shaped noise. In the second, the hearing aid is stimulated using the Maximum Length Sequence (MLS) analysis procedure (with speech shaping). Both signals were presented at 60 dB SPL. The two methods show nearly identical frequency responses: The MLS response shows slightly less gain at low frequencies because it was computed from a truncated impulse response.

Figure 5. Comparative spectrographic analyses of continuous speech at the microphone input and the output from the Nicolet Phoenix digital hearing aid using the CSRE software system (Jamieson, et al., 1989; 1992). The segment displayed is the utterance "weather to come" from the the end of the passage, "The warm south wind is a reliable warning of wet weather to come." The upper panel shows that, in the original passage, energy extends to at least 8kHz, that energy is present through all of the stop consonant closure intervals, and that the formants are quite distinct throughout the entire passage.



These and other approaches offer the possibility that clinicians will soon have the tools they need to determine how modern, signal processing hearing aids will perform when their clients wear them in everyday life. Such knowledge is essential to improving the routine practice of selecting and fitting hearing aids and to increasing the level of users' satisfaction with their aids.

Acknowledgements
The contributions of Meg Cheesman, Bill Cole, Leonard Cornelisse, Jim Jonkman, Issam Kheirallah, Lucy Kieffer, Emmet Raftery, Ketan Ramji, Todd Schneider, and Richard Seewald, at various stages of this project, are gratefully acknowledged. The work was supported by the Ontario Ministry of Health, the Rehabilitation Technology Research and Development Consortium, Unitron Industries Limited, and Etymonic Design Incorporated. Correspondence should be addressed to Dr. D.G. Jamieson at the indicated address, or by e-mail (JAMIESON@UWO.CA).

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