Hearing Aid Selection and Evaluation in the Year 2000

La sélection et l'évaluation des prothèses auditives en l'an 2000

Larry E. Humes
Department of Speech and Hearing Sciences
Indiana University

Key words: hearing aid selection, hearing aid evaluation, loudness

Abstract

This paper describes three new procedures that could prove useful for the selection and evaluation of hearing aids in the future. First, an approach to hearing aid fitting based on the restoration of normal loudness patterns is described. Next, a means of evaluating the hearing aid fit for speech sounds, referred to as the phon-gram, is illustrated. Finally, the modified Speech Transmission Index (mSTI) is offered as a potential solution to the problems of score interpretation and generalization for speech recognition scores obtained in the hearing aid evaluation. These three new procedures serve to illustrate ways in which new technology and knowledge can be applied to the solution of problems confronting the audiologist and the hearing aid wearer.

Résumé

L'auteur décrit trois nouvelles procédures qui pourraient se révéler éventuellement utiles lors de la sélection et de l'évaluation des prothèses auditives. Premièrement, il aborde une méthode d'ajustement des prothèses auditives qui est basée sur le rétablissement des niveaux normaux de sonorie. Il détaille ensuite des moyens pour évaluer l'ajustement des prothèses auditives en se basant sur les sons vocaux (graphique connu sous le nom de "phon-gram"). Enfin, il présente l'Index modifié de transmission de la parole (mSTI) comme solution possible aux problèmes de l'interprétation et de la généralisation des résultats obtenus aux examens vocaux lors de l'évaluation prothétique. Ces trois nouvelles méthodes servent à illustrer des façons d'appliquer les nouvelles technologies et connaissances à la solution des problèmes auxquels font face l'audiologiste et l'utilisateur de la prothèse auditive.

The author has been given the task of sketching out an agenda for future audiological research in the area of hearing aids. After several unsuccessful efforts to overcome the writer's block the author was experiencing as a result of this formidable task, the author decided to sketch out this agenda indirectly by describing his vision for turn-of-the-century hearing aid selection and evaluation protocols. In the process of describing various features of these ideal protocols, the problems standing between today's approaches and those of tomorrow will be identified, either directly or indirectly. The research agenda will emerge then as research designed to eliminate these barriers to progress. The author begins first with his vision or dream of the future for hearing aid selection and evaluation.

At the outset, however, several general assumptions need to be made about key players in this dream that are not directly related to the processes of hearing aid selection and evaluation. The hearing aid itself, for example, is assumed to have continued along its current path of development, which implies that the hearing aid of the year 2000 will be a 2 channel programmable ITE or canal instrument with extreme flexibility in electroacoustic performance and minimal feedback problems. It is assumed further that instrumentation permitting valid and reliable measurement of acoustic energy in the ear canal of the patient for a wide variety of simple and complex stimuli will become universally available, and that high-powered user friendly computers will be even more commonplace than today.

Given these general assumptions about the state of technology in the year 2000, it is clear that present day prescriptive approaches to hearing aid selection based on achieving target amounts of real-ear insertion gain will have vanished from all but the most archaic audiological practices. Insertion-gain measurements have evolved as an opportunistic middle man working between the theoretical objectives of the prescriptive method and the audiologist's ability to verify the accomplishment of those objectives. Few, if any, of the prescriptive methods actually describe as their theoretical objective the provision of X dB of insertion gain at a particular frequency for a given hearing impaired person. Rather, most such methods have as their objective the amplification of speech to some loudness criterion, such as most comfortable loudness (MCL), upper limit of comfortable loudness (ULCL), midway between threshold and loudness discomfort level (LDL), the 60-phon equal-loudness contour, and so on. (See Humes [1986] for a review of several of these procedures.) Unfortunately, there was no easy way to verify that these objectives were realized on a given patient and no obvious way to translate that information into the format needed to order the desired custom made, single-response analog instrument that was (and is) most commonly used. Through a series of assumptions, however, it was possible to translate the theoretical objectives of the method into recommended insertion-gain values and the emergence of relatively inexpensive computer-based probe-tube microphone systems offered a quick and reliable means of confirming the insertion-gain prescription.

In the year 2000, however, hardware and software will likely exist to verify directly the theoretical objectives of a particular prescriptive method in an efficient manner thereby obviating the need for a middle man like insertion gain. In fact, early versions of at least three such systems have already been described in the literature (Kiessling, 1987; Cox & Alexander, 1991; Humes & Houghton, 1992). Moreover, recall that the hearing aid of the year 2000 is assumed to be both programmable and extremely flexible electroacoustically so that custom circuits will not have to be selected for the patient and built at the factory. Rather, three or four instrument models covering the full range of amplification possibilities will be available from each manufacturer, and the audiologist will simply program the instrument in the office to meet the desired theoretical objectives as verified directly on that patient's ear.

What will be the theoretical objective of amplification systems in the year 2000? Although few would argue that the sole purpose of amplification is to improve the understanding of speech so as to enhance human communication, equally few would argue against this as representing the primary purpose of amplification. With this purpose in mind, an ideal objective for the hearing aid of the year 2000 would be to make all speech sounds audible, but not uncomfortably loud, by restoring normal loudness sensation for these sounds in an impaired ear. Moreover, this objective should be realized for a range of speech levels, from soft to loud.

There are many theoretically based prescriptive methods in use today that share this same general objective. Some methods, for example, select as target a criterion loudness value (60 phons, MCL, etc.), estimate these values for narrowband stimuli, and then verify the fit with narrowband stimuli presented at levels representing the longterm spectrum of speech. Speech itself, however, is broadband. Moreover, the hearing aid of tomorrow, like many today, is not likely to function the same way for both narrowband and broadband stimuli. The most valid information about the performance of the hearing aid will be obtained using a broadband stimulus. In the year 2000, if present trends continue, it is likely that most hearing aids will be evaluated with broadband speech signals themselves.

The use of broadband speech or speech-like stimuli in the selection and evaluation of hearing aids will present some new challenges to the audiologist. Successful realization of targets or objectives based on measures of overall loudness, such as MCL, can be deceptive, especially for broadband stimuli. Consider an extreme example for the MCL of a broadband speech signal, such as a vowel. When such a vowel is presented at MCL to a normal hearing listener, excitatory activity is distributed across a broad region of the cochlea, and the total loudness can be represented by the integral of that excitatory activity over cochlear place (or frequency). Now this same vowel is presented to a hearing impaired listener having a steeply sloping high-frequency sensorineural hearing loss. When the vowel's amplitude is adjusted to the MCL of the impaired listener, the underlying pattern of cochlear activity is likely to be quite different from that in the normal ear. The vowel will be at MCL in both the normal and the impaired ear, but the impaired ear can not be considered to have restoration of normal loudness sensation for this speech sound.

What is critical here is the notion that what should be restored to normal in the hearing impaired ear is not just the overall loudness of a sound, but the way in which that loudness is distributed across frequency (or place) within the auditory system. Let me illustrate this notion with an analogy. Consider a loudness matching task performed by a normal listener between a narrowband of noise centered at 1000 Hz and a broadband noise. Listeners can match the overall loudness of these two stimuli without much difficulty. At equal loudness, however, the pattern of activity created in the auditory system is considerably different for the narrowband noise and the broadband noise. Whereas the narrowband noise will evoke a strong cochlear response in a narrow region, the equally loud broadband noise will result in a broader spread of activity with a lower response magnitude. Overall loudness is related to the total power of the stimulus, which is affected both by its spectral density and its bandwidth. If broad bandwidth speech or speech-like stimuli are to be used to evaluate the function of hearing aids in the year 2000, electroacoustically and perceptually, then targets based on the restoration of normal loudness patterns must be established.

Loudness patterns, graphic displays of the distribution of loudness across frequency or cochlear place, have a long and significant history in psychoacoustics (Munson & Gardner, 1950; Zwicker & Feldtkeller, 1967; Zwicker, 1975, 1982; Moore & Glasberg, 1987; Glasberg & Moore, 1990). Figure 1 illustrates graphically the conversion of acoustic amplitude spectra for a 1000 Hz tone and a broadband noise to underlying patterns of specific loudness using the framework of Zwicker. The top panel in this figure illustrates the amplitude spectra for both sounds (solid lines = pure tone; dashed lines

Figure 1. Schematic illustration of amplitude spectra for a 1000 Hz pure tone and a spectrally shaped broadband noise (A). The corresponding excitation patterns (B) and specific loudness patterns (C) for each stimulus also are shown.

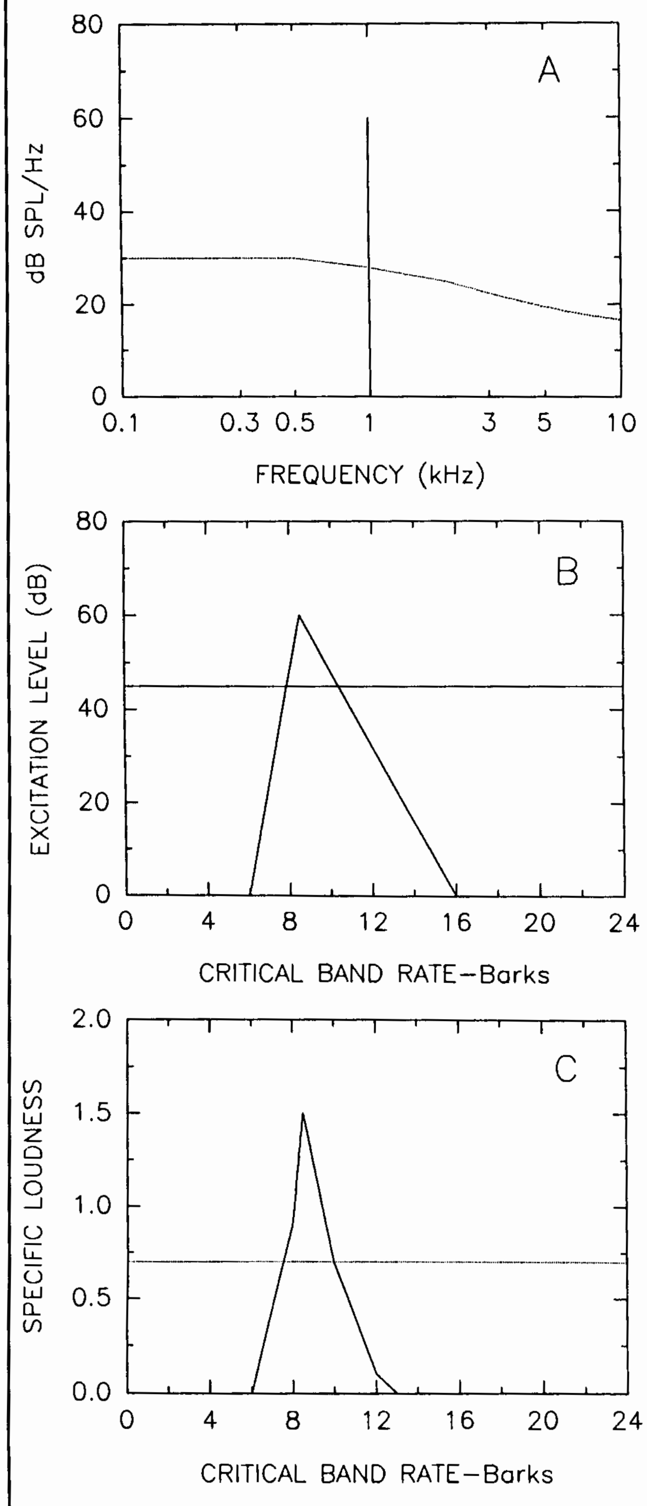
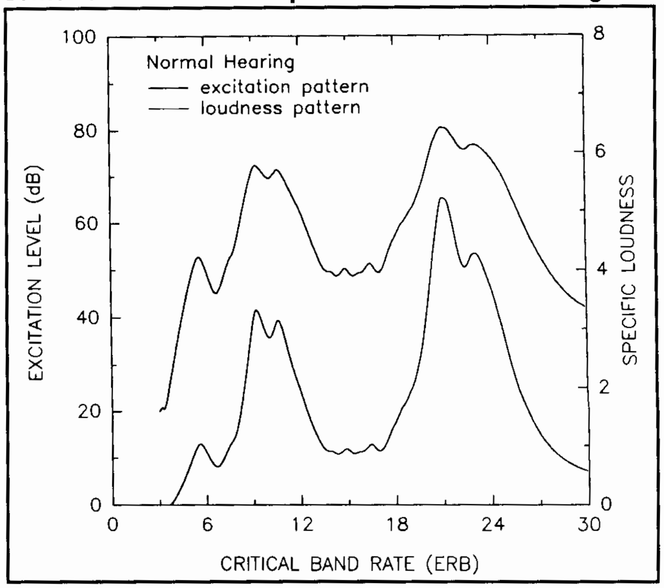


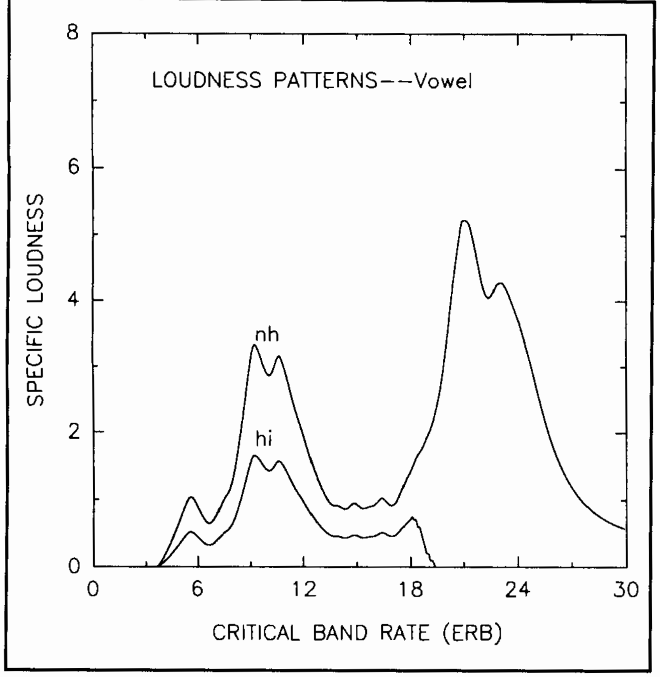
Figure 2. Illustration of the excitation pattern (top) and specific-loudness pattern (bottom) for the vowel /e/ presented at 77 dB SPL to a person with normal hearing.



= broadband noise). With this as input, the second panel illustrates the result of converting the amplitude spectra into so-called excitation patterns. Excitation patterns are based heavily on psychoacoustic masking patterns and represent these patterns in a plot of excitation level as a function of critical-band-rate in Barks or ERBs. The basic notion is that the encoding of sound in the auditory periphery can be likened to analysis by a parallel bank of critical-band filters that span the range of audible frequencies, with the excitation level at a particular critical-band value reflecting the output of that particular critical-band filter. Computer programs now exist that calculate excitation patterns for complex stimuli, such as vowels, from an FFT of the stimulus (Moore & Glasberg, 1987; Glasberg & Moore, 1990). Examples of an excitation pattern and specific-loudness pattern for the vowel /e/ presented at 77 dB SPL are shown in Figure 2.

The bottom panel of Figure 1 illustrates the specific-loudness patterns derived for the 1000 Hz tone and the broadband noise whereas the lower curve in Figure 2 illustrates such a pattern for the vowel /e/. Loudness patterns such as these are derived by using the corresponding excitation levels as input and applying a nonlinear transform, similar to Stevens' power law for loudness (Stevens, 1975), to convert excitation levels to specific loudness in sones per Bark. The result is a perceptually meaningful representation of a pattern of activity that is akin to the physically meaningful representation in terms of spectral density. To derive the overall loudness of a particular sound, one just integrates specific loudness over critical-band-rate (just as spectral density is integrated over frequency to derive total power in the physical domain).

Figure 3. Specific-loudness patterns for the vowel /e/ at 77 dB SPL in an ear with normal hearing (nh) and one with a high-frequency sensorineural hearing impairment (hi).



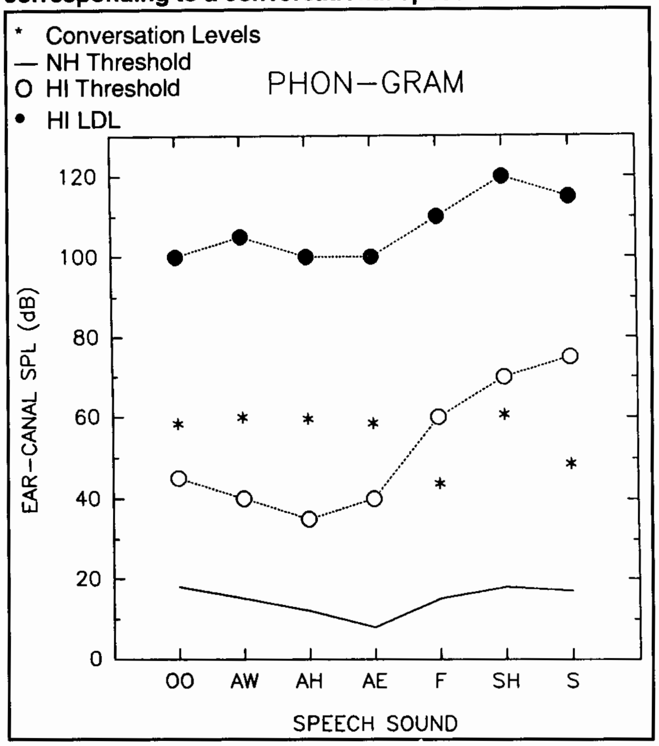
As mentioned, computer programs exist that enable rapid calculation of excitation and specific-loudness patterns from an FFT of the stimulus. Thus, the loudness pattern evoked in a normal ear by a specific speech sound can be derived quickly and easily. Over the past few years, it has been demonstrated that this same basic loudness pattern framework can be used to account for the effects of sensorineural hearing loss on auditory perception (Humes, Espinoza-Varas, & Watson, 1988; Humes & Jesteadt, 1991; Humes, Jesteadt, & Lee, 1992). Moreover, this is accomplished simply from knowledge of the individual's pure tone hearing loss. With what is currently known, we are capable of quickly calculating specific-loudness patterns for speech sounds for normal and hearing impaired listeners, with the only required input being the amplitude spectrum of the speech sound and the quiet thresholds of the impaired listeners.

How can this knowledge be used to better fit hearing aids in the year 2000? Once the normal specific-loudness pattern is calculated for a particular speech sound, it serves as the target for restoration of normal loudness perception under aided conditions in the impaired ear. This is illustrated in Figure 3 for the vowel /e/. Loudness patterns for this vowel have been calculated for a person with normal hearing (nh) and one with a sloping high-frequency sensorineural hearing impairment (hi). With knowledge of the impaired person's loudness pattern for the same stimulus, the computer can quickly determine the gain that would be required in each frequency region to restore the loudness pattern to normal. After initial adjustment for a broadband speech sound, such as the vowel /a/, additional vowels and consonants, especially high-frequency steady-state fricatives (/f, ʃ, s/), could be entered and gain values adjusted to optimize the match to targets across all speech sounds evaluated. Finally, this process could be repeated for the same set of speech sounds at speech levels that were soft, conversational, and loud. The gain values needed at each speech level in each frequency region would be useful in computer assisted selection of compression characteristics for each channel. In the end, we will have specified the gain required to restore normal loudness perception to a wide variety of speech sounds and for a wide range of speech levels. This, you'll recall, was indicated previously as our theoretical objective for hearing aid fitting in the year 2000.

Unquestionably, the most significant barrier between this dream for tomorrow and the reality of today lies in the development of computer software to assist in the optimization of hearing aid settings. This, in fact, is a serious problem today. The audiologist of today is confronted with a myriad of instrument manufacturers, models, and settings and has been provided with little assistance or guidance regarding the criteria and procedures for selecting the most appropriate settings for a given patient. Many devices today have parameters that can be adjusted, but only a very limited knowledge base exists to guide the audiologist in the selection of the most appropriate values for that parameter. Our ability to manage and use the technology has not kept pace with the ability of manufacturers to introduce new technology into the instruments. Computers will be increasingly needed to assist the audiologist in the selection of the appropriate instrument and in the initial optimization of the instrument's parameters for the patient.

If the optimization software needed to implement the loudness pattern approach were available today, the procedure could still not be put into practice without further research in additional areas. The author mentioned, for example, that loudness patterns are derived from excitation patterns through a nonlinear power law conversion and that excitation patterns are themselves closely related to masking patterns. It is well known that even normal hearing young adults exhibit considerable variability in masking patterns (Zwicker & Schorn, 1978), especially for signal frequencies higher in frequency than the masker (i.e., upward spread of masking). Normal hearing young adults also show reliable and sizable individual differences in the slopes of their loudness growth functions (Stevens, 1975). The extent to which individual differences in these functions are important contributors to similar variations in understanding speech will determine whether valid loudness pattern targets can be constructed for impaired listeners on the basis of thresholds alone. Many recent

Figure 4. Phon-gram for a hypothetical hearing impaired listener having the indicated thresholds (open circles) and loudness discomfort levels (filled circles). The asterisks indicate the ear-canal sound pressure levels of each of the speech sounds when presented at sound-field levels corresponding to a conversational speech level.



studies exploring the contributions of various psychoacoustic and cognitive factors to individual differences in speech understanding performance, however, have consistently identified the hearing loss as the primary or sole explanatory factor (van Rooij, Plomp, & Orlebeke, 1989; van Rooij & Plomp, 1990; Helfer & Wilber, 1990; Humes & Roberts, 1990; Humes & Christopherson, 1991; Jerger, Jerger, & Pirozzolo, 1991; Humes, 1992; van Rooij & Plomp, 1992; Dubno & Schaefer, 1992). Individual differences in performance on other psychoacoustic or cognitive tasks explain little of the variance in speech understanding performance. Thus, although individual differences in spread of masking and loudness growth exist, it appears that they are not critical to accomplishing the objective of improving speech understanding. Moreover, it is individual differences in the loudness growth function that may be the most critical, and these can easily be measured in a short period of time, if further research proves this to be necessary.

The focus thus far has been on a means of establishing the appropriate gain and compression characteristics of the instrument. This is analogous to the "hearing aid selection" process for today's hearing aids, although we are selecting or optimizing the parameters of a very flexible programmable instrument and not selecting an instrument to be ordered.

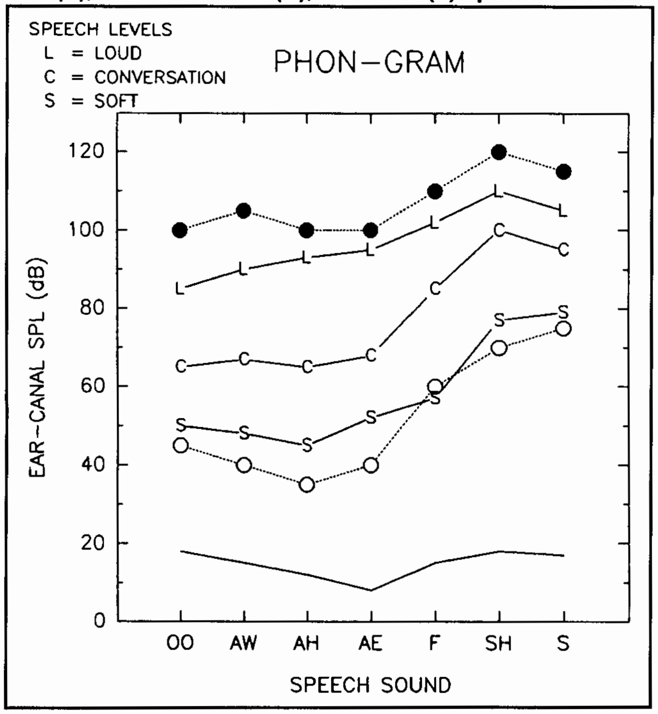
Table 1. Acoustic characteristics of speech sounds used in phon-gram. Values are adapted from Hodgson (1986).

Speech Sound	Region of Spectral Concentration	Relative Amplitude
/u/	300, 870 Hz	58.3 dB
/ɔ/	570, 840 Hz	60.0 dB
/ɑ/	730, 1090 Hz	59.6 dB
/æ/	660, 1720 Hz	58.6 dB
/f/	1000-10000 Hz	38.7 dB
/ʃ/	2000-5000 Hz	50.7 dB
/s/	4000-8000 Hz	43.7 dB

What about evaluation of the fit once it has been accomplished with the loudness pattern optimization approach? Loudness patterns, for the most part, are hypothetical constructs, not phenomena that are directly or easily measured. The theoretical objective of this hypothetical approach in the year 2000, however, was to restore normal loudness sensation for speech with the ultimate aim of making speech ranging from soft to loud audible, but not uncomfortably loud. One way in which this objective could be directly verified on the patient wearing the instrument is with the assistance of the *phon-gram* illustrated in Figure 4. The phon-gram is a plot of ear-canal sound pressure level for several speech sounds, with speech sounds arranged along the abscissa roughly from lowest to highest in frequency. All of the speech sounds shown in the phon-gram were the same ones evaluated in the initial computer optimization of the hearing aid's electroacoustic response, except that the vowels are now just two formant versions rather than the full five formant versions used in the optimization. By using just the first and second formants, these vowels can be made more frequency specific without negatively affecting perception. The frequency ranges encompassed by the various speech sounds and their relative amplitudes in conversational speech are indicated in Table 1 (Hodgson, 1986). Taken together, these speech sounds encompass a spectral range from approximately 300 to 10,000 Hz and almost the full range of amplitudes in conversational speech.

The hypothetical phon-gram method of hearing aid evaluation would proceed as follows in the year 2000. Two measures will be obtained from the impaired listener for each of these speech sounds without the hearing aid: ear-canal sound pressure levels corresponding to detection threshold and to loudness discomfort. Thresholds and LDLs from a hypothetical hearing impaired listener are shown in Figure 4. Next, the hearing aid would be inserted and the ear-canal sound pressure level measured for each of these speech sounds presented at levels representing their intensity in soft (S), conversational (C), and loud (L) speech. The measured levels for each speech sound would then be plotted with the appropriate symbol and connected.

Figure 5. Illustration of an aided phon-gram for the same hypothetical hearing impaired listener shown in the previous figure. Letters connected by solid lines represent ear-canal sound pressure levels of each speech sound when presented at sound-field levels corresponding to soft (S), conversational (C), and loud (L) speech.



A phon-gram derived in such a manner for the aided listening conditions is shown in Figure 5 for our hypothetical hearing impaired listener. We can see that, for all but the phoneme /f/ presented at a level corresponding to soft speech, we have accomplished our objective of making a wide range of speech sounds audible, yet not uncomfortably loud. We have accomplished this, moreover, using speech sounds directly rather than using artificial acoustic stimuli and making inferences about speech. Although of moderate to broad bandwidth, the speech sounds chosen here are frequency specific enough to suggest modifications in the programming of the instrument should the desired objectives not be accomplished.

To my knowledge, there is nothing impeding the development of an evaluation tool such as the phon-gram, using today's technology. Inexpensive, standardized, real-time generation of high quality speech sounds is possible today, for example, using personal computer based versions of the Klatt speech synthesizer, and the necessary signal analysis capabilities can be found in many present day probe-microphone systems.

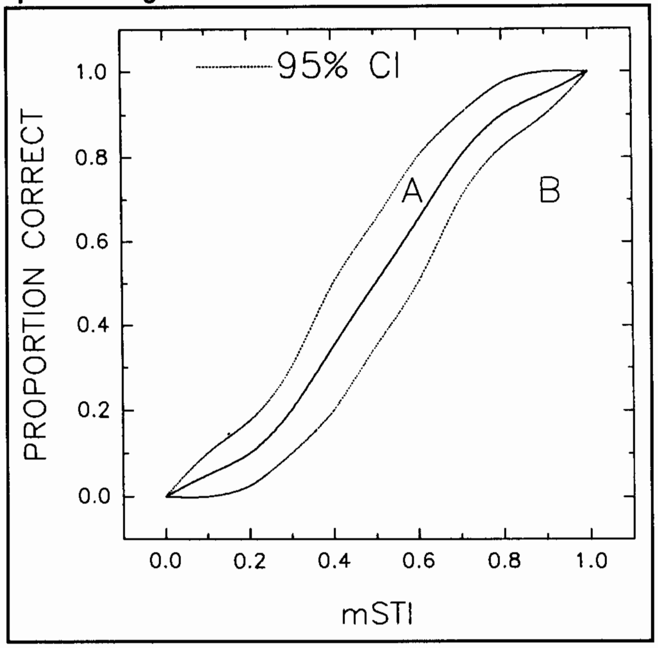
The same speech sounds used both in the optimization software to initially adjust hearing aid parameters to target and in the phon-gram could also be used to further evaluate the fitting of the instrument in a closed-set nonsense syllable identification task. Of course, the set of speech sounds could easily be expanded beyond those seven used in the phon-gram to allow a more comprehensive evaluation of performance. A closed-set nonsense syllable identification task patterned after that of the CUNY NST (Resnick et al., 1975) and administered and scored rapidly by the computer would be a nice starting point. A system having these capabilities is already in existence (Humes & Houghton, 1992). Immediate results from analyses of errors on the speech recognition task could also be used to quickly identify "problem" speech sounds for future rehabilitation or hearing aid adjustment.

With the advent of real-ear insertion-gain measures over the past decade, many audiologists have abandoned the use of speech recognition testing as part of the hearing aid evaluation and are content to confine the evaluation process to confirmation of prescribed insertion gain. Although convenience and clinical expediency are certainly two factors contributing to the disuse of speech recognition testing as part of the hearing aid evaluation, problems of score interpretation and generalization have also been major contributors to its demise. The problem of interpretation is simply one of evaluating what a speech recognition score of X% means for a particular hearing impaired listener wearing a hearing aid in a given test environment. For example, is X% better or worse than expected, given the hearing loss of the patient, the amplification characteristics of the instrument, and the test conditions and materials? Further, if it is worse than expected, what does the audiologist do with that information? The problem of generalization is also an interpretive issue, but one of broader scope. If the listener performs as expected with the instrument when listening to nonsense syllables in quiet while in a sound-treated test booth, what does that have to with listening to their grandchild in a noisy playroom?

In the year 2000, acoustic indices, such as the Articulation Index (AI; French & Steinberg, 1947) and Speech Transmission Index (STI; Steeneken & Houtgast, 1980), may offer some solutions to the problems of score interpretation and generalization. Although the AI is probably the more widely known of these two indices, the author will focus on the use of the STI because it lends itself more readily to acoustic evaluation of hearing instruments. In particular, the author will briefly describe a modification of the STI, referred to as the mSTI, which we have validated previously as an acoustic predictor of speech recognition performance in normal hearing and hearing impaired listeners (Humes et al., 1986; Humes et al., 1987; Humes, in press).

Briefly, the mSTI measures modulation transfer functions (MTFs) for modulation frequencies ranging from 0.5 to 16 Hz in octave steps in each of fifteen 1/3-octave bands ranging from 250 to 6300 Hz. The MTFs can be measured in a variety of ways (Houtgast & Steeneken, 1978; Schroeder, 1981; Polack

Figure 6. Transfer function relating proportion correct on a speech recognition task to the modified Speech Transmission Index (mSTI). Solid line represents mean function whereas the dotted lines represent 95% confidence intervals around the mean. "A" and "B" represent the aided speech recognition scores of Listener A and Listener B.



et al., 1984), but the most appropriate way for application to nonlinear devices, such as hearing aids, would appear to be through the use of a 100% intensity-modulated speech-shaped noise carrier with subsequent analysis in 1/3-octave bands. Although measurement of the mSTI in aided listening conditions with a probe-tube microphone placed in the ear canal of the hearing aid wearer is technically feasible today, the author knows of no such systems presently in existence.

Further details about the measurement of the mSTI are not important here. What is of importance in the present context is how an appropriately implemented mSTI can help overcome the problems of score interpretation and generalization in the hearing aid evaluation. First, regarding score interpretation, further work with the mSTI must continue to support a strong relationship between the mSTI and speech recognition score across a wide range of listening conditions, including those involving aided and unaided listening by hearing impaired listeners. If such a relationship is observed, then the mSTI can be used to predict the speech recognition score for a given hearing aid wearer obtained for a specified set of test materials and listening conditions. Confidence bounds can then be constructed around the predicted score, and the aided performance of the individual listener can be evaluated relative to those bounds. In this way, the observed score can be compared to a range of "expected" scores to determine whether the instrument wearer is performing as well as can be expected given the constraints of the test conditions. An example of this type of application of the mSTI is illustrated in Figure 6. This figure contains a hypothetical function relating the proportion correct on a word recognition task to the mSTI, with the thin dotted lines representing 95% confidence intervals constructed around the mean function (solid line). Also shown are the data points for two different listeners, A and B, who have obtained the same aided word recognition score of 72%. Based on these scores alone, one might conclude that these individuals are deriving equivalent benefit from their hearing aids. As shown in this figure, however, Listener A is performing as predicted by the mSTI, whereas Listener B is receiving poorer than expected benefit from the amplification device. Thus, the mSTI can be used to help establish guidelines for expected levels of performance for a given listener, hearing aid, and test condition. This information, together with an analysis of the types of errors or confusions made by the listener in the speech recognition testing, can be an effective guide to subsequent aural rehabilitation and counselling for those patients performing below expected levels.

For those performing at or above expected levels of performance on the speech recognition task, the mSTI offers a powerful solution to the problem of score generalization. An attractive feature of the mSTI is the multiplicative property of the MTFs underlying the mSTI (Steeneken & Houtgast, 1980; Humes, in press). A series of 1/3 octave band MTFs can be obtained for any acoustic system, including a variety of rooms and hearing aids. A catalog of such MTFs could be established, for example, for the audiometric test booth in which the testing is conducted, a typical living room, a conference room, a set of churches or synagogues, and a variety of classrooms. The aided MTFs could then be obtained in the test booth for the hearing aid wearer. This would be accomplished with a probe-tube microphone in the ear canal and appropriate adjustment of the MTFs for the hearing loss of the listener (Humes et al., 1986; Humes, in press). If the application of the mSTI is validated for a particular set of listening conditions and speech materials for the hearing aid wearer (i.e., the wearer performs as expected like Listener A in Figure 6), then the mSTI can be used to estimate how well that listener will do in any of the other rooms contained within the MTF catalog. This is accomplished quite readily at the level of the MTF by dividing the MTFs measured in the aided condition by the MTFs representing the audiometric test booth in the catalog and then multiplying the result by the MTFs representing some other room or set of listening conditions in the catalog. By establishing psychometric functions between performance and the mSTI for several different speech materials, moreover, it would be possible to generalize not only to other acoustic environments and listening conditions, but also to other types of speech material as well (e.g., nonsense syllables vs. sentences).

Much additional research is needed with the mSTI before it can become a useful solution to the problems of score interpretation and generalization. Although MTFs have been successfully measured from hearing aids using a 2 cm^3 coupler and a hearing aid test box (Ahlstrom, Boney, & Humes, 1984), they would ideally be measured in the ear canal of the listener with a probe-tube microphone. In addition, more work is needed to validate application of the mSTI to aided and unaided performance from individual listeners. It is uncertain as to whether the mSTI can be reliably used to estimate the expected performance of individual listeners in aided and unaided conditions. To the author's knowledge, moreover, the multiplicative property of MTFs that is essential to the "catalog solution" to the score generalization problem has never been rigorously evaluated. Nonetheless, the foregoing discussion of the mSTI and its application to the problems of score interpretation and generalization in the hearing aid evaluation illustrate a way in which acoustic indices such as the mSTI may be used in the year 2000 as a solution to these problems.

As with all dreams or visions, the details that seemed so clear and sensible at the time fade rapidly under closer inspection and with the passage of time. The same may certainly prove true for the loudness pattern approach to hearing aid fitting, the phon-gram method of hearing aid evaluation, and the use of the mSTI as a tool to facilitate interpretation of speech recognition scores in the hearing aid evaluation. In the present context, however, the prognostic accuracy of the details are not as important as the general features of the protocols, the problems they illustrate, and the solutions they offer. It is hoped that through discussion and careful examination of new protocols, such as those described here, we will move closer to the ultimate objective of restoring normal auditory function to hearing impaired listeners through use of amplification.

Acknowledgements
This work was supported, in part, by grants from the National Institute on Aging and the National Institute on Deafness and other Communication Disorders.

Address all correspondence to: Larry E. Humes, Ph.D., Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405

References

Ahlstrom, C., Boney, S., & Humes, L.E. (1984). *Modulation transfer functions of hearing aids*. Paper presented at meeting of the American Speech-Language-Hearing Association, San Francisco.

Cox, R.M., & Alexander, G.C. (1991). Evaluation of an in situ output probe-microphone method of hearing aid fitting verification. *Ear and Hearing, 11*, 31-39.

Dubno, J.R., & Schaefer, A.B. (1992). Comparison of frequency selectivity and consonant recognition among hearing impaired and masked normal-hearing listeners. *Journal of the Acoustical Society of America, 91*, 2110-2121.

French, N., & Steinberg, J. (1947). Factors governing the intelligibility of speech sounds. *Journal of the Acoustical Society of America, 19*, 90-119.

Glasberg, B.R., & Moore, B.C.J. (1990). Derivation of auditory filter shapes from notched-noise data. *Hearing Research, 47*, 103-138.

Helfer, K.S., & Wilber, L.A. (1990). Hearing loss, aging, and speech perception in reverberation and noise. *Journal of Speech and Hearing Research, 33*, 149-155.

Hodgson, W.R. (1986). Speech acoustics and intelligibility. In W.R. Hodgson (Ed.), *Hearing aid assessment and use in audiologic habilitation* (109-127). Baltimore: Williams and Wilkins.

Houtgast, T., & Steeneken, H.J.M. (1978). *Application of the modulation transfer function in room acoustics* (Report No. 39). Soesterberg, The Netherlands: Institute for Perception, TNO.

Humes, L.E. (1986). An evaluation of several rationales for selecting hearing aid gain. *Journal of Speech and Hearing Disorders, 51*, 272-281.

Humes, L.E. (1991). Understanding the speech-understanding problems of the hearing impaired. *Journal of the American Academy of Audiology, 2*, 59-69.

Humes, L.E. (in press). Some temporal factors affecting speech recognition. In G.A. Studebaker & I. Hochberg (Eds.), *Acoustical factors affecting hearing aid performance*. Boston: Allyn & Bacon.

Humes, L.E., & Roberts, L. (1990). Speech recognition difficulties of the hearing impaired elderly: the contributions of audibility. *Journal of Speech and Hearing Research, 33*, 726-735.

Humes, L.E., & Christopherson, L. (1991). Speech identification difficulties of hearing impaired elderly persons: the contributions of auditory-processing deficits. *Journal of Speech and Hearing Research, 34*, 686-693.

Humes, L.E., & Jesteadt, W. (1991). Models of the effects of threshold on loudness growth and summation. *Journal of the Acoustical Society of America, 90*, 1933-1943.

Humes, L.E., & Houghton, R. (1992). Beyond insertion gain. *Hearing Instruments, 43*(3), 32-35.

Humes, L.E., Boney, S., & Loven, F. (1987). Further validation of the Speech Transmission Index (STI). *Journal of Speech and Hearing Research, 30*, 404-410.

Humes, L.E., Espinoza-Varas, B., & Watson, C.S. (1988). Modeling sensorineural hearing loss. I. Model and retrospective evaluation. *Journal of the Acoustical Society of America, 83*, 188-202.

Humes, L.E., Jesteadt, W., & Lee, L.W. (1992). Modeling the effects of sensorineural hearing loss on auditory perception. In Y. Cazals, L. Demany, & K. Horner (Eds.), *Auditory physiology and perception* (617-624). Oxford: Pergamon Press.

Humes, L.E., Dirks, D.D., Bell, T.S., Ahlstrom, C., & Kincaid, G.E. (1986). Application of the Articulation Index and the Speech Transmission Index to the recognition of speech by normal-hearing and hearing impaired listeners. *Journal of Speech and Hearing Research, 29*, 447-462.

Kiessling, J. (1987). In situ audiometry (ISA): A new frontier in hearing aid selection. *Hearing Instruments, 38*(1), 28-29.

Jerger, J., Jerger, S., & Pirozzolo, F. (1991). Correlational analysis of speech audiometric scores, hearing loss, age and cognitive abilities in the elderly. *Ear and Hearing, 12*, 103-109.

Moore, B.C.J., & Glasberg, B.R. (1987). Formulae describing frequency selectivity as a function of frequency and level, and their use in calculating excitation patterns. *Hearing Research, 28*, 209-225.

Munson, W.A., & Gardner, M.B. (1950). Loudness patterns—a new approach. *Journal of the Acoustical Society of America, 22*, 177-190.

Polack, J.-D., Alrutz, H., & Schroeder, M.R. (1984). The modulation transfer function of music signals and its applications to reverberation measurement. *Acustica, 54*, 257-265.

Resnick, J.R., Dubno, J.R., Hoffnung, S., & Levitt, H. (1975). Phoneme errors on a nonsense syllable test. *Journal of the Acoustical Society of America, 58*, S114.

Schroeder, M.R. (1981). The modulation transfer function: definitions and measurements. *Acustica, 49*, 179-182.

Steeneken, H.J.M., & Houtgast, T. (1980). A physical method for measuring speech-transmission quality. *Journal of the Acoustical Society of America, 67*, 318-326.

Stevens, S.S. (1975). *Psychophysics*. New York: John Wiley & Sons.

van Rooij, J.C.G.M., Plomp, R., & Orlebeke, J.F. (1989). Auditive and cognitive factors in speech perception by elderly listeners. I.: Development of test battery. *Journal of the Acoustical Society of America, 86*, 1294-1309.

van Rooij, J.C.G.M., & Plomp, R. (1990). Auditive and cognitive factors in speech perception by elderly listeners. II. Multivariate analyses. *Journal of the Acoustical Society of America, 88*, 2611-2624.

van Rooij, J.C.G.M., & Plomp, R. (1992). Auditive and cognitive factors in speech perception by elderly listeners. III. Additional data and final discussion. *Journal of the Acoustical Society of America, 91*, 1028-1033.

Zwicker, E. (1975). Scaling. In W.D. Keidel & W.D. Neff (Eds.), *Handbook of sensory physiology, 5*(3), (401-448). Berlin: Springer-Verlag.

Zwicker, E. (1982). *Psychoakustik*. Berlin: Springer-Verlag.

Zwicker, E., & Feldtkeller, R. (1967). *Das Ohr als Nachrichtenempfanger*. Stuttgart: S. Hirzel Verlag.

Zwicker, E., & Schorn, K. (1978). Psychoacoustical tuning curves in audiology. *Audiology, 17*, 120-140.