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SIGNAL PROCESSING, HEARING AID DESIGN, AND THE PSYCHOACOUSTIC TURING TEST

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ABSTRACT

A psychoacoustic Turing Test is proposed as a means for assessing hearing aid signal processing: a hearing aid is considered successful if the aided impaired listener is indistinguishable from a normal hearing listener using psychoacoustic measures. Forward masking, loudness summation, and frequency resolution tests are examined within this construct. The performance of hearing impaired listeners using multiband compression is predicted and compared to the performance of normal hearing listeners. The results provide a measure of hearing aid performance and a method for selecting between different signal processing designs.

1. INTRODUCTION

Hearing aid design has changed dramatically since the end of the 1980s, when nonlinear hearing aids were developed to compensate for the change in the nonlinear function of the auditory system caused by damaged outer hair cells. The signal processing and fitting of these hearing aids were the first significant attempt to create something akin to a cochlear model in the hearing aid design. Today, these designs are still primarily validated using results from aided speech tests and subjective judgments of the hearing aid wearer. Because of the robustness of speech recognition to signal distortion and the general benefit of amplification to auditory perception, there is no consistency among the specific designs of the sophisticated hearing aids on the market today. They vary significantly in the signal processing structure of the hearing loss compensation, the filterbank structure, and the time constants of the signal processing components. Other aspects of design are completely ignored, such as phase response. The following proposes a theoretical structure for evaluating hearing aid design based on the psychoacoustic aspects of auditory perception.

A variation of the Turing Test [1] is proposed for assessing hearing aid signal processing. The most common interpretation of the classic Turing Test is that a computer program passes the definition of intelligence if a remote tester cannot distinguish between the algorithm and a human. This concept has previously been applied to the assessment of clinical treatment, where the treatment is successful if the post-treatment patient can not be distinguished from a healthy untreated subject through a series of relevant tests [2].

Similarly, the Turing Test can be applied to hearing aids as follows: if a tester is allowed to conduct any number of psychoacoustic tests on an undisclosed subject and the tester cannot determine if the person is a normal hearing subject or a hearing impaired subject wearing a hearing aid, then the aid has passed the Turing Test. The underlying hypothesis of the Turing Test's application is that any listener that demonstrates normal hearing acuity as measured with basic psychoacoustic tests is perceiving sound normally and will function as a normal hearing individual. The link between basic psychoacoustic ability and higher level perception exists in many areas, including speech perception [3] and auditory streaming [4]. This approach can be used to discriminate between signal processing designs that cannot be differentiated using basic speech metrics such as speech reception thresholds or using quality assessments such as user acceptance questionnaires. This theoretical construct can also be used in an *a priori* way such that the signal processing design criterion is to produce normal psychoacoustic behavior in anticipation of a Psychoacoustic Turing Test.

2. THE PSYCHOCOUSTIC TURING TEST

The following sections will examine three aspects of auditory perception that are significantly altered by sensorineural hearing loss. They are relevant to aided perception in that they represent the perception of signals more complex than the standard single tones and narrowbands of noise that are commonly used in clinical characterizations of the hearing of hearing loss patients. Alterations to these aspects of hearing described below, either due to cochlear damage or hearing aid signal processing, could have significant consequences for the way in which speech cues are coded, in which signals are grouped in to auditory objects, or in which sound is categorized at a high level. As such, they are reasonable initial candidates for components of a Psychoacoustic Turing Test. While considerable variation exists for how these three phenomenon can be measured, manageability dictated the selection of a single paradigm for each phenomenon using the single criterion that the test gives a reasonable representation of the percept being measured.

The hearing aid signal processing applied to the Turing Test in these sections is multiband compression. Many of the performance differences between normal and hearing impaired subjects in the psychoacoustic tasks discussed may be due to the lost cochlear nonlinearity caused by sensorineural hearing loss [5]. The traditional use of compression to compensate for reduced dynamic range may also compensate for differences in the normal and impaired psychoacoustic performance in the tasks described. Some form of multiband compression exists in all current high-end hearing aids, and these compressors vary considerably in their signal processing structure, differing in the numbers of subbands, filter bandwidths, time constants and compression ratios. The following will investigate these variations when examining the ability of a multiband compressor to aid a hearing impaired subject to pass the Psychoacoustic Turing Test.

2.1. Forward masking

Forward masking experiments measure the temporal resolving capabilities of the listener by determining the lowest level at which the listener can hear a brief probe tone that follows a masker tone or noise at the same frequency. As the temporal separation between the masker and probe increases, the listener is better able to resolve the probe from the masker, producing lower thresholds. People with sensorineural hearing loss produce higher thresholds with this task, indicating poorer resolving abilities. This deficit appears to be a result of lost compression in the damaged cochlea [6], suggesting that multiband processing in a hearing aid could restore this percept to normal if designed correctly.

The decay of forward masking thresholds can be modeled with the following set of equations:

$$\mathrm{SL}_p = k(\mathrm{SL}_m + M)e^{-t/T}, \quad (1)$$
$$T = 53.0e^{0.018(\mathrm{HL})}, \quad (2)$$

where SL_p is the probe sensation level threshold, SL_m is the masker sensation level, T is the time constant of masking recovery, HL is the hearing loss of the listener, M is a sensitivity constant, and k is a masking slope constant [7]. All levels are in dB. From Equations (1) and (2), the gain function necessary to restore aided forward masking to normal can be calculated.

For the parameter set (k, SL_m, M, HL), values of (1, 90, -5, 0) were used for normal hearing subjects and (0.7, 40, 2.21, 50) for hearing impaired subjects. These accurately represent forward masking data obtained experimentally with these two groups of subjects [7]. Figure 1 shows the forward masking functions predicted with these two configurations.

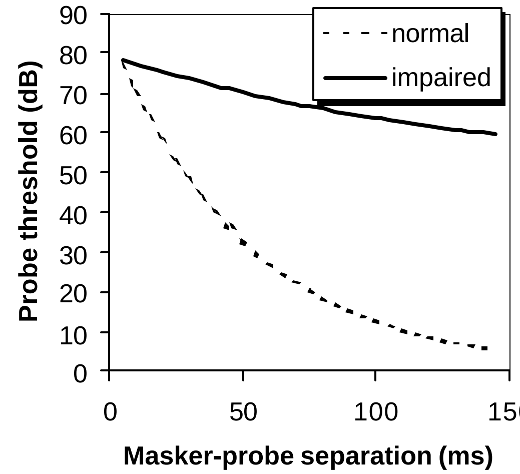

Figure 1. Forward-masked thresholds for normal hearing and hearing impaired listeners, calculated from (1) and (2) using the parameters in the text.

In order to restore forward masking to normal, the gain from a hearing aid must vary with level such that, for example, the normal threshold of 53 dB at T = 25 ms will be achieved for the hearing impaired if that level receives 21 dB of gain, while the normal threshold of 36 dB at T = 45 ms will be achieved for the hearing impaired listener if that level receives 35 dB of gain. Figure 2 shows the gain function that restores forward masking to normal given the parameter sets stated above, derived from the functions in Figure 1. Also shown is a gain function that provides 3:1 compression, set such that the gain is 0 dB for an input level of 85 dB.

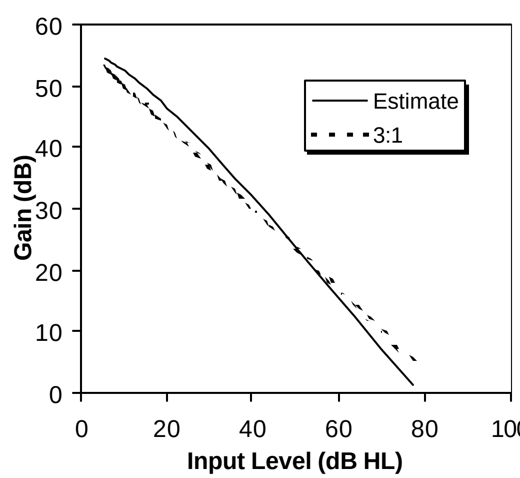

Figure 2. Estimated gain necessary for producing normal aided forward masked thresholds, and gain from a standard 3:1 compression.

In order for the hearing impaired listener to pass the Turing Test with a forward masking paradigm, the hearing aid will have to apply the gain function shown in Figure 2 to the probe signal. Additionally, the time constants, specifically the release time, will have to be fast enough to allow the gain to increase in time to reach the prescribed gain in Figure 2 for the various values of *T*. If the release time is too slow, the gain will not increase in time and the forward masked thresholds for the aided hearing impaired subject will be elevated relative to normal, identifying the subject as hearing impaired and failing the Turing Test.

2.2. Loudness summation

Loudness summation is the phenomenon where the perceived loudness of a signal increases as the bandwidth of the signal increases while the total power of the signal is held constant. The phenomenon results from the summation of specific loudness within critical bands, where the specific loudness has been affected by cochlear nonlinearities within each band. Listeners with significant sensorineural hearing loss do not exhibit loudness summation in regions of loss because of the absence of the cochlear nonlinearity in these regions [8]. With several assumptions, the perceived loudness of a signal by these listeners is dependent only on the total power and not on how that power is distributed across the spectrum.

Multiband compression hearing aids and their fitting algorithms are typically designed to restore loudness of tones or narrowband noise to normal—the implication of the compression to the loudness of broadband signals, and to loudness summation, is usually ignored. Compression should exhibit some ability to restore loudness summation if the signal processing structure of the compressor is similar to that in a healthy cochlea.

One paradigm of a loudness summation task requires the listener to adjust the level of a single tone so that the loudness of the single tone is the same as the loudness of a two-tone complex. The level of the matching tone is plotted as a function of frequency separation of the two-tone comparison. With normal hearing listeners, the single tone must be set to a higher level as the frequency separation between the two tones in the comparison increases. Typical performance with normal listeners is shown with the dashed line in Figure 3 [9].

The aided performance of a listener with no natural loudness summation in their auditory system has been predicted with two different compressor designs and are shown in Figure 3. The two multiband compressors differ primarily in the bandwidth of the filters. The compressors have identical input-output functions for pure-tone inputs. The results show that predictions from the compressor with the narrower filters produce loudness matches that are near normal. The hearing impaired listener using this aid would most likely pass the Turing Test using this paradigm. This result indicates that, based on this test alone, the signal processing design with the narrower filters is superior better design.

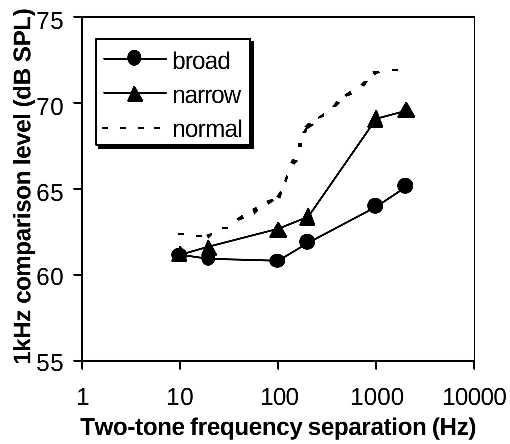

Figure 3. The loudness of a single tone necessary to equal the loudness of a two-tone complex. Predictions are given for hearing impaired listeners using multiband compression with narrow filters and broad filters. Also shown is data for normal hearing listeners [9].

2.3. Narrowband masking

One of the consequences of sensorineural hearing loss is an increase in the bandwidth of auditory filters by up to a factor of three. A wider auditory filter bandwidth results in greater off-frequency masking, where a signal at one frequency is more likely to prevent the listener from hearing a signal at another frequency. Hearing impaired listeners could be identified by their poorer frequency resolving abilities.

One technique for measuring frequency resolution is to measure the detection threshold for a tonal probe at different frequencies in the presence of a masker at a fixed frequency. As the frequency separation between the masker and probe increases, the probe threshold decreases. The amount that the threshold decreases depends on the auditory filter bandwidth. In one study, a narrowband masker was fixed at 4 kHz and the threshold of a probe was measured at multiple frequencies for both normal hearing listeners and the hearing impaired [10]. Figure 4 shows this data, demonstrating that thresholds are significantly elevated for the subjects with sensorineural hearing loss because of their increased auditory filter bandwidths.

Because multiband compression is capable of applying different gain in different frequency regions depending on the measured sound level in those regions, multiband compression can reduce the probe threshold if the compressor resolves the probe separately from the masker. More gain will be applied to the probe than to the masker if the probe and masker fall within separate filters in the compressor filterbank. Thus, the ability of mutliband compression to reduce off-frequency masking thresholds is dependent on the bandwidth and frequency resolving ability of the compressor. This is in contrast to the increase in thresholds caused by poorer frequency resolving ability in the impaired auditory system.

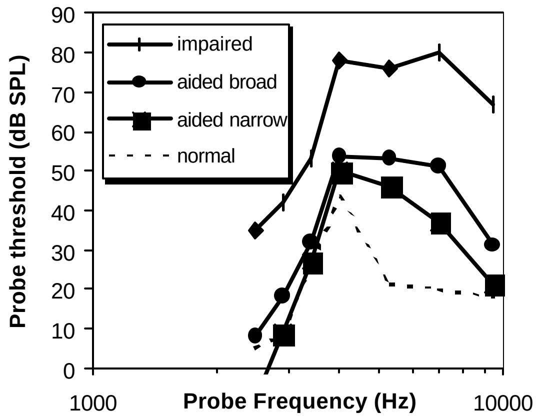

Figure 4. Detection thresholds for a pure tone probe in the presence of a masker at 4 kHz. Shown are data for hearing impaired and normal hearing listeners [10]. Also shown are predictions for hearing impaired listeners aided by multiband compression with broad filters and narrow filters.

Figure 4 shows the predicted masked thresholds of hearing-impaired listeners using two different compressors. Each compressor varies only in the bandwidth of the filters in the filterbank—one has filters that are significantly narrower than the other, resulting in better isolation between the probe from the masker and allowing the gain applied to the probe to be independent of the level of the masker. While the predicted thresholds for the impaired listener with the narrower compressor filters are still significantly higher than those for normal hearing listeners, they indicate better frequency resolving ability than with the use of the compressor with the broader filters. Theoretically, a filterbank could be designed which would restore the aided thresholds to normal, resulting in normal-appearing performance with frequency resolution tasks and allowing the impaired listener to pass the Psychoacoustic Turing Test that uses this frequency resolution paradigm.

3. CONCLUSIONS

This has been an initial exploration in the concept of a Psychoacoustic Turing Test. There are many unproven assumptions in the validity of such a test, but they are ones that seem reasonable to the author in the absence of conflicting data. The use of this test cannot strictly be applied to impairments that include inner hair cell loss where restoration of normal perception is most likely impossible. The ability of any single signal processing structure to restore all percepts to normal for solely outer hair cell loss is also questionable. But if the psychoacoustic metrics that are most important for such desired perceptual tasks as consonant identification, localization, and auditory object formation can be identified, then the concept of a Psychoacoustic Turing Test can be used in the design and verification of hearing aid signal processing.

4. REFERENCES

[1] A.M. Turing, "Computing Machinery and Intelligence," *Mind*, vol. 59, pp. 433-460, 1950.

[2] R. Sandell, E. Alfredsson, M. Berg, K. Crafoord, A. Lagerlof, T Cohn and A. Rugolska "Turing's Game and the Clinical Significance of Outcome With Borderline Patients at a Day Hospital," *J. Clin. Psychol.*, vol. 50, pp. 406-414, 1994.

[3] W. A. Dreschler, "Phoneme Perception Via Hearing Aids With and Without Compression and the Role of Temporal Resolution," *Audiol.*, vol. 28, pp. 49-60, 1989.

[4] N. Grimault, C. Micheyl, R.P. Carlyon, P. Arthaud and L. Collet, "Perceptual Auditory Stream Segregation of Sequences of Complex Sounds in Subjects with Normal and Impaired Hearing," *Br. J. Audiol.*, vol. 35, pp. 173-182, 2001.

[5] B.C.J. Moore, D.A. Vickers, C.J. Plack and A.J. Oxenham, "Inter-Relationship Between Different Psychoacoustic Measures Assumed to Be Related to the Cochlear Active Mechanism," *J. Acoust. Soc. Am.*, vol. 106, pp. 2761-2778, 1999.

[6] C.J. Plack and A.J. Oxenham, "Basilar-Membrane Nonlinearity and the Growth of Forward Masking," *J. Acoust. Soc. Am.*, vol. 103, pp. 1598-1608, 1998.

[7] D.A. Nelson and R.L. Freyman, "Temporal Resolution in Sensorineural Hearing-Impaired Listeners," *J. Acoust. Soc. Am.*, vol. 81, pp. 709-720, 1987.

[8] B. Scharf and R.P. Hellman, "Model of Loudness Summation Applied to Impaired Ears," *J. Acoust. Soc. Am.*, vol. 40, pp. 71-78, 1966.

[9] C. Zhang and F-G. Zeng, "Loudness of Dynamic Stimuli in Acoustic and Electrical Hearing," *J. Acoust. Soc. Am.*, vol. 102, pp. 2925-2934, 1997.

[10] M. Florentine, S. Buus, B. Scharf and E. Zwicker, "Frequency Selectivity in Normally-Hearing and Hearing-Impaired Observers," *J. Speech Hear. Res.*, vol. 23, pp. 646-669, 1980.