R-HINT-E: A REALISTIC HEARING IN NOISE TEST ENVIRONMENT

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ABSTRACT

With the advent of cheap, low-power digital signal processors, it has been possible to develop hearing aids that utilize DSP technology. It is essential that the algorithms instantiated on these hearing aids be evaluated in a standard fashion with regards to both engineering and perceptual criteria. However, currently available hearing in noise tests do not allow for the range of signals or environments that are of interest to the designer of hearing aid algorithms. In addition, virtual audio environments have recently been suggested [1] as a potential tool for the auditory science community. To address the needs of both engineering and clinical evaluation, we propose a flexible, software based virtual acoustic environment capable of realistically simulating a wide variety of scenarios.

1. INTRODUCTION

In a real acoustic environment, the sound received at our ears is the sum not only of impinging sounds from multiple, independent sources, but also of received echoes of these sources. This, in a nutshell is the cocktail party problem. While a person with a healthy auditory system can somehow manage to extract a single source from this sort of background interference, a person suffering from hearing loss often cannot do so well [2]. The promise of digital hearing aids offers some hope in this regard, but our ability to effectively test such devices is falling behind our ability to conceive of them. A successful hearing aid algorithm is one that will perform well in real acoustic environments. However, there needs to be a standard testing environment that will allow ready comparisons, and easy simulation. R-HINT-E offers this, as well as flexibility and user customizability.

2. LIMITATIONS OF CURRENT METHODS

Two of the most popular hearing in noise tests currently in use are SPIN [3] (Speech in Noise) and HINT [4] (Hearing in Noise Test). Both of these tests require a subject to repeat back all or part of a speech sentence played in noise. From the point of view of algorithm testing, neither of the two systems are sufficient, since they do not permit testing in different background noises. SPIN only permits a multi-speaker babble noise, and HINT only uses LTASS noise. Additionally, it is not possible to vary the amount of room reverberation (both systems assume low reverberation), the number of interfering sources (SPIN and HINT assume a single source), or the type and direction of either the target or the interferers. This lack of range is clearly a problem for some testing some spatial processing algorithms such as beamforming and ICA, as both may perform very differently depending on the placement and number of sources. At the same time, it would be useful to test algorithms not only with speech, but also with music or traffic noise as interfering sources. It is thus necessary to develop a new system that incorporates this sort of flexibility.

In addition to the aforementioned desire for flexibility in determining experimental conditions, it is also desirable to avoid the costs associated with instantiating a design on a DSP chip before thorough testing. Instead, it would be far preferable both to simulate the acoustic environment and run the processing algorithm entirely in software. In this case, the outputs of this virtual hearing aid can be fed directly into a subject's ear, or applied to some quantitative testing metric (SNR, AI, etc.).

3. IDENTIFIED NEEDS

Following this, we identified a set of needs for a new testing environment that would form the basis of the design for R-HINT-E [5].

1. To be able to test algorithms under a variety of realistic reverberation and noise conditions.

2. To ensure experimental control over all conditions. This ensures repeatability, and the ability to make meaningful comparisons between algorithms.

3. To have available a large database of test sounds. This should include as many examples of human speech as possible. In addition, other sounds such as music, traffic noise, and so forth should also be incorporated into the range of available sources.

4. To ensure user control over the acoustic conditions. The user should be able to change the room reverberation characteristics, as well as the source positioning. This would include control over issues such as source loudness and timing.

6. To be customizable, so that the user can easily add in new acoustic environments and processing algorithms.

4. CONCEPTUAL MODEL

Given these needs, we have developed a system that incorporates the measured binaural acoustic transfer functions of real rooms with separately recorded audio signals. The conceptual model of the R-HINT-E system can ultimately be broken down into four aspects:

1. Sound stimuli recorded in quiet. This may include speech, music, or some other type of sound source.

2. Characterizing room environments by acoustic IR functions. In order to accurately judge how impinging sound waves will be perceived, it is necessary to measure the impulse responses at the entry to the ear canal. The pre-recorded sound signals may then be convolved with these impulse responses to simulate a real environment.

3. Combining multiple sounds and locations. Using the pre-recorded sounds, and the measured IR's for different locations, it is possible to create virtual "soundscapes" that incorporate multiple sources in a variety of locations.

4. Signal enhancing algorithm incorporation. Using the signals created in step 3 as the inputs to an algorithm under test, it is possible to gauge its performance. Hearing aid specific amplification and compression stages can also be incorporated here as well.

5. METHODS

Impulse functions were measured for three different rooms. Room 1 (low reverberation condition) was 11'x11'x8'6" with a double row of heavy velour drapes around its periphery. Room 2 was the same as Room 1, but with the drapes open, which increased the reverberation and changed the dimensions to 12'x12'x8'6. Room 3 was a hard-walled, reverberant lecture room with dimensions 17'10"x32'9"x8'8". Numerous tables and chairs were also present in this room, contributing to a general acoustic clutter.

The recordings were made using six Knowles FG microphones, three of which were inserted into each ear of a KEMAR mannequin. In the ear, the microphones were placed 5mm apart from each other in a horizontal fashion. The microphones were hooked up to a custom-built pre-amplifier board which was provided to us by Gennum corporation., and from there to an Echo Layla Sound card onboard a laptop running Cool Edit Pro.

The recordings were made with KEMAR standing in the middle of each room, while a flat-response loudspeaker was moved to various positions throughout the room. These locations included 12 different angles (0°,22.5°,45°,67.5°,90°,135°,180°,225°,270°,292.5°,315°, 337.5°), 3 heights (1'6.5", 4'6", 5'5"), and 2 distances (3' and 6'). A swept sine stimulus[6] was presented to KEMAR at each position for the purpose of measuring the combined room impulse response and Head-related transfer function (HRTF). In addition, 6 speech signals (4 male and 2 female) were also presented to KEMAR for the purposes of validation. These recorded sentences would later be compared with the convolution of the room IR's with the pre-recorded speech sentences, which ensured that we were accurately simulating the acoustic environment. Perceptual tests were also carried out to confirm that the two signals were not audibly distinguishable.

In addition, it should be mentioned that Room 3 (the reverberant lecture room) was subject to moderate levels of HVAC noise. Since the system could not be turned off, Cool Edit Pro's "Noise Reduction" feature was used to eliminate the noise. The impulse responses were calculated, and validated using the de-noised signals.

6. THE R-HINT-E SOFTWARE

The R-HINT-E software is a virtual acoustic environment geared to the needs of the hearing aid research community. The program is written in MATLAB as a GUI application to support the ease of use demands that required the easy addition of user-developed algorithms.

Moreover, it is meant to be extremely flexible. In addition to adding in custom processing algorithms, users can readily add in their own acoustic environments, develop ‘plug-in” applications, or add in new sounds. In addition, users are not restricted to the binaural recording arrangement that has been described here. It is entirely possible to use different sensor geometries, such as linear or planar microphone arrays.

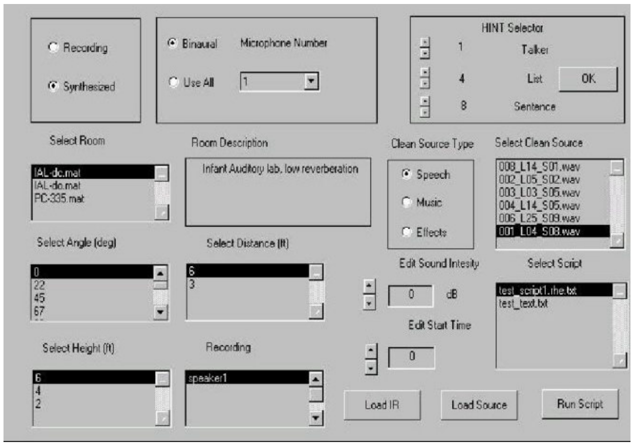

Figure 1. The loading screen.

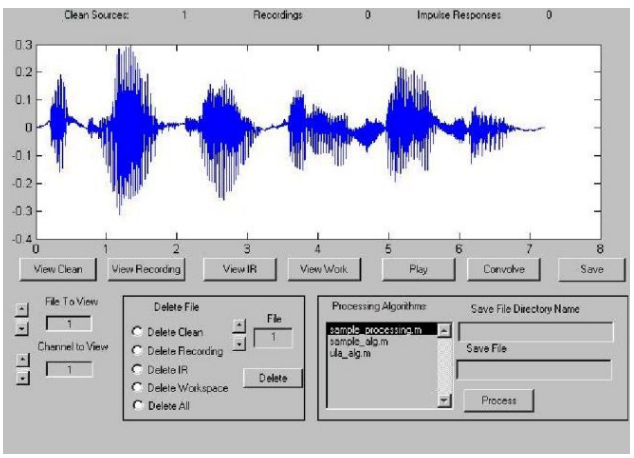

Figure 2. The processing screen.

There are two main windows in the program that allow access to R-HINT-E’s main functions. These are the loading and processing screens, which encapsulate the functions that their names imply. In the loading screen (Figure 1), users can choose the room type, the sensor arrangement (binaural or otherwise), as well as specifying the source type and location.

In addition, since this program was originally designed to work with the HINT sentences, there is also an option that allows the user to choose a specific HINT recording. R-HINT-E also supports a scripting feature that has been designed to make large trials easier to run through. A simple scripting language that encompasses R-HINT-E’s major functionality has been incorporated into the software, along with a custom interpreter. User written scripts may be accessed and run from this window.

R-HINT-E’s processing options are available through a second screen (Figure 2). These include signal viewing and playback, convolution, and the saving of processed files. In addition, the user-defined algorithms are accessed though this screen.

A final feature of R-HINT-E is the fact that it allows for the development of ‘plug-in” applications. These are applications that make use of the basic functionality of R-HINT-E, but perform other tasks that have not been included. These will likely be along the lines of clinical testing procedures, or other non-engineering tasks. To facilitate the development of these applications we have included a set of MATLAB routines that interface with R-HINT-E, and may be easily incorporated into other programs.

A sample plug-in application is shown below (Figure 3). This program simply plays speech sentences from the HINT database for a given SNR, although it illustrates some of what can be done using this feature. The sample program and its code will be included in the full release of R-HINT-E, along with a full description in the user’s manual.

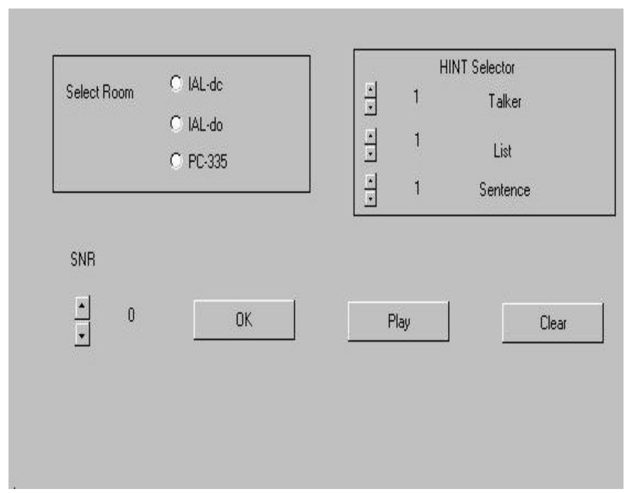

Figure 3. A sample plug-in application.

7. CONCLUSIONS

Having outlined some of the deficiencies of current testing methods with regards to hearing aid design, we have posed several new requirements that would enhance our ability to adequately assess new algorithms. Following these requirements, a new virtual acoustics program has been developed. This program, R-HINT-E (Realistic Hearing in Noise Test Environment), not only allows for a more flexible choice of testing scenarios, it is also highly user customizable, and may be used to assess other, non-hearing related acoustic signal processing algorithms.

This program will eventually be made available to the hearing aid community, in a manner that will be announced soon. In the meantime, a web-based version with reduced functionality is available at http://hearing.mcmaster.ca/rhinte. We feel that R-HINT-E, and virtual acoustics has the potential to become a vital tool both in hearing aid design, as well in signal processing as a whole. For this reason, we are continuing our investigation into applications of this nature, and will continue with further development of this software.

8. REFERENCES

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