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## Methods for audio demonstration of digital hearing aids to listeners with normal hearing.

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### Abstract

A set of audio demonstrations of digital hearing aid operation were developed using recordings of hearing aid output. The demonstrations were used to compare an ADRO™ hearing aid to other top of the range digital hearing aids in common listening scenarios. Each hearing aid was fitted using the manufacturer's recommendations, for a moderate hearing loss with 50 dB HL pure tone thresholds. For the purposes of demonstrating audibility and intelligibility differences to listeners with normal hearing, it is essential to calibrate the absolute levels of input and output of the hearing aid, and to include a simulation of the sensorineural hearing loss. The hearing loss simulation was based on previous studies examining speech intelligibility, and used dynamic expansion by multichannel automatic gain control to reproduce the elevated thresholds and loudness recruitment observed in hearing impaired listeners. For the purposes of demonstrating sound quality and adaptive response of the hearing aid output it is important that the hearing loss simulation does not affect these properties of the sound, and a static attenuation model was found to be more suitable than the dynamic expansion model. The demonstrations exposed important differences in audibility, speech intelligibility, sound quality, and adaptive response for the variable conditions found in common listening scenarios. Most hearing aids performed well for speech at average levels in quiet. The differences between the aids were most marked at high or low levels, for speech in background noise, or for environmental sounds. The ADRO™ hearing aid was not always the best performer in every scenario, but was the best all-round performer of the hearing aids tested.

## Introduction

Listening comparisons are an important part of the development and evaluation of innovative hearing aid processing strategies. Although audio demonstrations cannot replace objective trials with hearing aid users, they do help people to understand the differences between hearing aids and illustrate the effects that underlie the clinical results. The goals of the present study were to demonstrate the audibility, comfort, speech intelligibility, sound quality and noise reduction differences between an ADRO™ hearing aid [1, 2] and other top of the range digital behind-the-ear (BTE) hearing aids. Many of these differences had already been established in clinical trials with hearing aids [2, 3] and cochlear implants [4].

Key issues to be resolved in the development of audio demonstrations are the choice of a suitable hearing loss, proper configuration of each hearing aid, use of an appropriate recording process, and choice of useful listening scenarios. The audience for audio demonstrations usually have normal hearing, so it is essential to include a simulation of the hearing loss in order to make audibility, comfort, and intelligibility differences evident. Attenuation is the most fundamental requirement for the hearing loss model, in order to keep the high intensity hearing aid output within the normally-hearing listeners' comfortable range. However, fixed attenuation does not accurately model the hearing-impaired listeners' reduced dynamic range and loudness growth function. Three types of nonlinear processing have been used to model these aspects of hearing loss: dynamic expansion by automatic gain control (agc), addition of masking noise, and centre clipping. Of these approaches, the expansion method has been found to be preferable for audio demonstration, since both masking noise and centre clipping introduce ever-present noise or audible distortions which sound less realistic [5, 6]. Although the expansion approach has

been investigated in comparison studies of speech intelligibility for simulated and actual hearing impaired listeners, such studies have generally not examined broader sound quality issues in detail. Therefore, the output of expansion hearing loss simulations may not be suitable for demonstrations where criteria other than speech intelligibility are used to assess the sounds. In particular, it is important that the hearing loss simulation does not introduce dynamic changes in the spectral shape or level of the signal if the purpose of the demonstration is to show differences in sound quality or the dynamic response of the hearing aid.

## Hearing Loss

The hearing aids were fitted to a moderate hearing loss with 50dB HL pure tone thresholds. Moderate hearing loss was chosen because it is the most common degree of hearing impairment worldwide. Statistics for the United States indicate 50% of people with a hearing impairment have a moderate hearing loss [7]. A flat loss was used for three reasons: It is technically easier to simulate than a sloping loss; a flat loss does not distort spectral shapes, making sound quality judgements easier for people with normal hearing; and speech intelligibility results for simulated hearing loss best match those of moderately hearing impaired listeners for flat frequency-gain characteristics [5].

## Hearing Aid Fittings

All hearing aids used in the audio demonstrations were fitted using the manufacturer's recommendations and default settings, and programmed using the manufacturer's current fitting software via Noah 3.1 and the Aurical Hi-Pro. The ADRO™ hearing aid was programmed using the ADROFit software with no changes from the predicted fitting. In all cases, the only input to the fitting was the assumed audiogram. An experienced non-linear user profile was used to ensure adequate gain was prescribed in the aid. To ensure accurate fittings, all hearing aids were tested

via a 2cc coupler (type HA2) in either a Verifit or Aurical test box. Depending on manufacturer, a swept tone function or speech modulated noise was used to verify the hearing aid response.

### Recording Process

The following recording setup and procedure were designed to produce high-quality, reproducible recordings with calibrated output levels. The recordings were done in the free field of an audiometric test booth. Each input stimulus was generated from a desktop personal computer using Goldwave audio editing software through a Creative Soundblaster Extigy sound card, Sherwood AX-5010R amplifier and Interdyn pee-wee studio monitor. For omnidirectional microphone recordings, the hearing aid was placed in the centre of the booth, 1 metre from the speaker, mounted on a microphone stand with forward axis pointing directly towards the speaker face. For recordings testing a directional microphone response, the hearing aid was mounted on a dummy head, with the noise source speaker mounted at 135° from the hearing aid forward axis, also at a horizontal distance of 1 metre. Both speaker and hearing aid were approximately 1 metre from the floor surface.

The output of the hearing aid was recorded via a Brüel & Kjaer 2cc coupler and Investigator 2260 sound level meter, at a sample rate of 22 kHz through the sound card and editing software. The arrangement for the recording process is depicted in Figure 1.

The amplifier and software gains were adjusted so that the output to input loop gain at the computer was unity without the gain of the hearing aid. The gains of the amplifier, sound card and software were set well below maximum to avoid any high end distortion.

Calibration of hearing aid output sound levels was needed to determine the level-dependent attenuation to be applied in the hearing loss simulation. This level calibration was found using a 1kHz, 1/3 octave calibration noise and slow, linear weighting mode on the B&K sound level meter. The probe microphone was placed at the hearing aid location, and a steady state calibration noise level of 86 dB SPL, linearly weighted, corresponded to a long term root-mean-square (rms) average signal level of -30dB in the initial calibration recordings at the computer. This calibration was verified using a sound level range of 30dB (66 to 96dB SPL). The long term rms average sound level in dB SPL for each hearing aid recording via the 2cc coupler was then confirmed by reference to this level calibration.

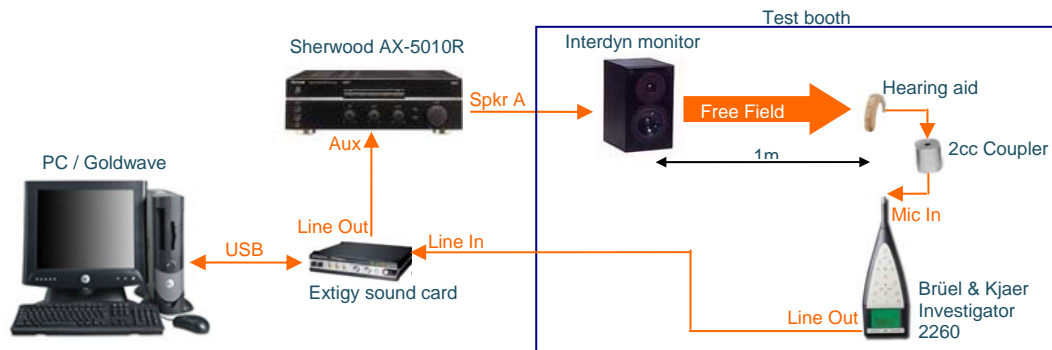


Figure 1 – Recording equipment and setup

## Hearing Loss Simulations

Two hearing loss simulations were used in the audio demonstrations: a static attenuation model, and a dynamic expansion model. Both models were based on the input-to-output function shown in Figure 2.

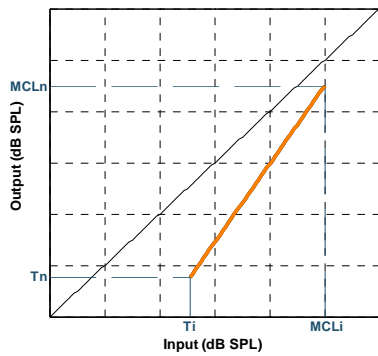


Figure 2 – Hearing loss simulation input to output expansion levels

The purpose of the static attenuation model was to maintain the overall loudness differences between the recorded hearing aid output signals without introducing any spectral shape or dynamic processing artefacts. Static simulations are suitable for demonstrations of sound quality and dynamic processing characteristics of the hearing aid. The dynamic expansion model was based on previous studies of speech intelligibility [5,6,8]. Although the dynamic model introduces both spectral and temporal distortions into the signals, it is most suitable for demonstrations where the relative intelligibility or audibility of the signals is important.

The dynamic simulation was implemented in Matlab 6.5. The expansion level dependent attenuation was calculated and applied using an FFT/Hanning window overlap-add filterbank. Twenty-one third-octave frequency sub-bands were constructed by combination of frequency bands from a length 256 FFT. Each sub-band of the input signal was processed independently as shown in Figure 3. Short time (100ms) average sound level estimates were calculated for each frequency band, and a dynamic level-dependent attenuation was applied in each band using the expansion line of Figure 2.

Consistent with observations in the previous studies [5,6], shorter window times caused more noticeable signal distortion, similar to that audible after processing with fast-acting compression schemes.

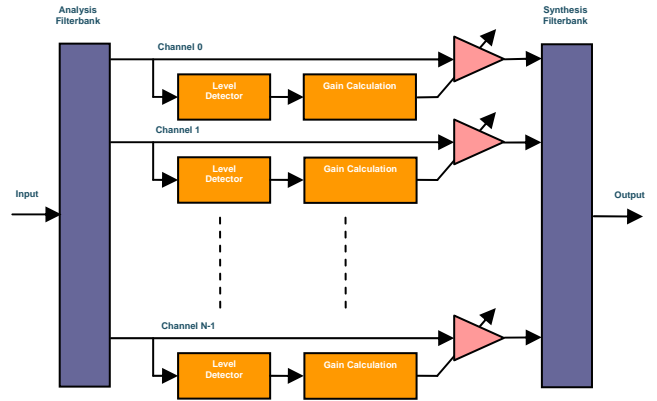


Figure 3 – Hearing loss simulation frequency sub-band processing

The static simulation was then implemented by comparing the overall levels of the dynamic simulation input and output signals, calculating the total attenuation, and applying this attenuation directly as a static loss to the input signal.

A scatter plot of average levels (low frequency, blue, to high frequency, red) calculated from a comparison of inputs and outputs for an example signal processed by the dynamic hearing loss simulation is shown in Figure 4. The figure which shows the relationship between input and output levels, has the shape expected from Figure 2.

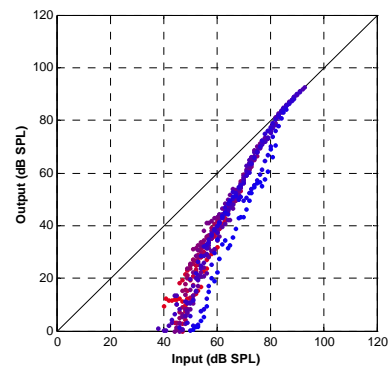


Figure 4 – Example levels from post-comparison of input and output signals

## Results

A range of audio recordings were processed through a selection of high-quality hearing aids from leading manufacturers, including an ADRO™ hearing aid. All the recordings for speech at normal input levels in quiet were similar in their intelligibility and loudness, however, the hearing aids behaved quite differently from one another for more challenging input signals. Examples of challenging scenarios included high intensity sounds, low intensity speech, music and environmental sounds with spectral shapes different from speech, and sound mixtures with low signal to noise ratio. The ADRO™ hearing aid was a consistently good performer across all scenarios, and the differences between the ADRO™ hearing aid processing and the compression processing used in the other hearing aids were consistent with the clinical trial evaluations.

All of the recordings were processed with both the static attenuation hearing loss model and the dynamic expansion model. As expected, the dynamic model produced the most obvious intelligibility differences between the hearing aids, but also produced audible distortions that affected the quality of the sounds. Thus the static model was more useful in making comparisons of the sound quality of the different aids.

The effects of the dynamic hearing loss model are illustrated in Figure 5 which compares the outputs of the ADRO™ hearing aid with a compression hearing aid for an input speech signal at 68 dB SPL in the presence of a warning siren at 94 dB SPL. The siren has a fairly narrow frequency distribution around 2 kHz, and turns on about 7 seconds into the recording, and off again 22 seconds into the recording. The rms sound level for the ADRO™ hearing aid output before the hearing loss simulation was approximately 90 dB SPL, in a 2cc coupler, compared to 97 dB SPL for the compression aid. The alarm sound was relatively loud, and was attenuated relatively little by the

simulated hearing loss in comparison to the speech. The adaptive processing in the ADRO™ hearing aid reduced the alarm level to bring it within the listener's comfortable dynamic range, while preserving speech information in frequency regions not affected by the siren. The compression aid acted more quickly when the alarm was turned on or off, but reduced the sound level across the whole frequency range, making the speech unintelligible. These differences are clearly audible in the recordings.

## Conclusions

Audio demonstrations of digital hearing aids were developed to compare hearing aid performance and signal processing techniques before and after a hearing loss simulation, for a moderate flat hearing loss. The static and dynamic hearing loss models were both effective in producing appropriate listening levels for people with normal hearing. The static simulation produced less spectral and temporal distortion than the dynamic model, and was therefore more suitable for comparing the sound quality of the hearing aids. The dynamic expansion model was more appropriate for demonstrating intelligibility differences between the aids. The subjective impressions gained from listening to the audio demonstrations and from spectral analyses such as those in Figure 5 supported the results of the clinical trials of ADRO™. The clinical trials indicated that ADRO™ improved the comfort of loud sounds (consistent with the lower level of the siren in Figure 5), increased the audibility of soft and medium level sounds (consistent with the greater amplitude of the speech before the siren turned on), and maintained greater intelligibility of speech in background noise (consistent with the greater spectral detail shown for frequencies above and below the siren). The siren recordings and other audio demonstrations are available on the Dynamic Hearing website, [www.dynamichearing.com.au](http://www.dynamichearing.com.au)

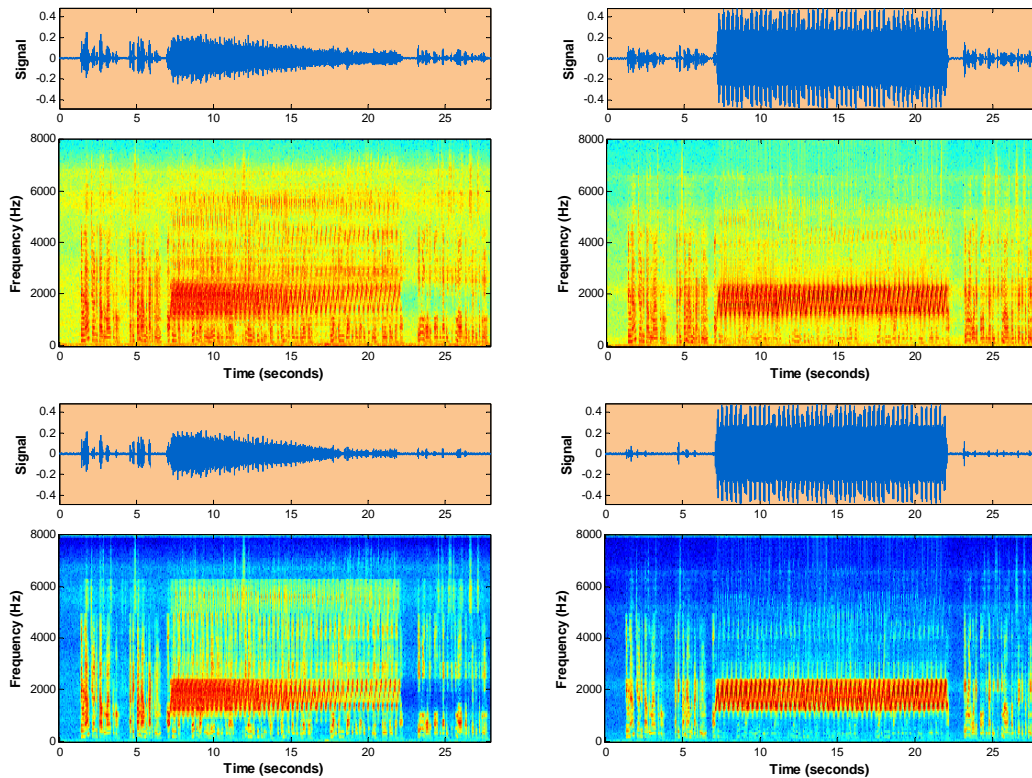


Figure 5. Output of an ADRO™ hearing aid (left panels) compared with a compression hearing aid (right), before (top panels) and after (bottom) application of the dynamic hearing loss simulation.

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