SUBBAND ADAPTIVE MODELING OF DIGITAL HEARING AIDS

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1. INTRODUCTION

The intelligibility and quality of processed speech are central concerns for designers and manufacturers of hearing aids, clinicians who prescribe and fit hearing aids, and importantly the end users. Conventional standardized and other widely-available test procedures [1] have been directed at manufacturing quality control, and provide very limited information on how hearing aid processing affects the intelligibility and quality of speech and other sounds of importance to the user. In particular, these procedures often neglect the fact that many modem hearing aids are adaptive, changing their electro acoustic characteristics in response to changes in the acoustic environment.

Researchers have recently begun to examine the potential for evaluating hearing aids using more complex stimuli that better approximate "real world" signals. For example, the ANSI standardized Speech Intelligibility Index (SII) [2] has been applied to predict the speech intelligibility improvement obtained through a hearing aid. Coherence measures, representing the linear relation between a broadband noise input and the resulting hearing aid output, have been studied as ways to predict the sound quality and speech intelligibility performance of hearing aids [3]. An alternative approach to quantifying the performance of digital hearing aids is to dynamically model their behavior using a system identification approach as this offers the flexibility of testing a hearing aid with speech and music stimuli [4, 5]. In this method, the hearing aid is modeled as a linear time-varying system and its response to speech and music stimuli is predicted using a linear adaptive filter. It is assumed that the model residual is mainly composed of distortion and noise components of the hearing aid under test [4]. The relative level of the distortion and noise can be quantified using a simple metric such as the Signal-to-Noise Ratio (SNR) or using a more sophisticated metric such as the Perceptual Evaluation of Speech Quality (PESQ) [6] which incorporates models of auditory perception.

Previous studies with analog hearing aids have shown that the speech quality metrics derived from using the system identification approach correlated well with perceptual judgments of speech quality, both by normal and hearing impaired listeners [4]. However, this method has not been tested with modern digital hearing aids, majorities of which employ multi-channel compression among other advanced signal processing features. This necessitates the need for subband models to correctly model modern digital hearing aids [5].

2. SUBBAND ADAPTIVE MODEL

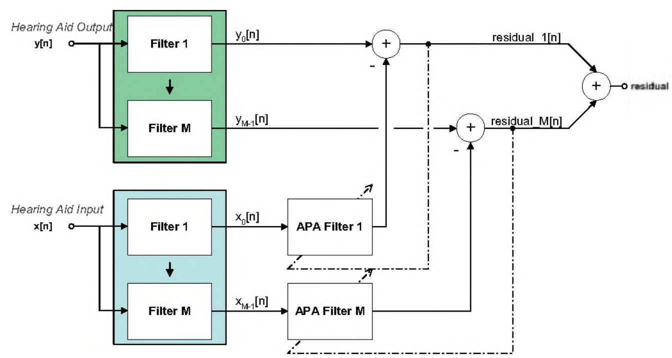

Figure 1. Block Diagram of the Subband Adaptive Model

Figure 1 illustrates the subband architecture considered in this paper. The recorded hearing aid output and input sequences, y[n] and x[n], are filtered using uniform, M-band analysis filter banks. The resulting subband output sets, $y_0[n]$, ..., $y_{M-1}[n]$ and $x_0[n]$, ...,$x_{M-1}[n]$, form the desired and reference sequences to the adaptive filter blocks (APA Filter 1, ... , APA Filter M), respectively. Each of the constituent adaptive filter blocks is implemented as a finite impulse response (FIR) filter whose coefficients are updated using a complex affine projection algorithm based on recursive matrix updating.

3. METHOD

We have previously studied the performance of the subband adaptive model in characterizing the behavior of multi-channel compression hearing aids using computer simulations [5]. In this paper, we report initial results from subband modeling of two commercial hearing aids - Oticon Syncro and the Bernafon Symbio. The Syncro is a multi-channel compression hearing aid with eight "voice aligned" compression channels, while the Symbio is a "channel-free" digital hearing aid.

Each of these aids was first programmed to fit a steeply sloping, moderate to severe hearing loss profile. In order to focus on the compression characteristics of each hearing aid, all unrelated signal processing features of the device were deactivated. Each aid was placed in a Brüel & Kjaer anechoic test box, Type 4232, with accompanying microphones and preconditioning amplifiers. Ten Hearing In Noise Test (HINT – House Ear Institute of Los Angeles, CA, USA) speech sentences were concatenated and played back at 65 dB SPL. The hearing aid output was recorded through a 2 cc coupler, while a separate reference microphone was used to record the unprocessed speech. These two signals were then applied to the subband adaptive model where the number of analysis bands was altered over a range including the number of channels in the hearing aid being tested. For each analysis band, a Signal-to-Error Ratio (SER in dB) and the PESQ – Mean Opinion Score (MOS) were calculated.

4. RESULTS & CONCLUSIONS

Figure 2 illustrates the results of subband modeling of Syncro and Symbio hearing aids for HINT 1-1 sentence. The FIR filter length in the adaptive filters was set to 256 taps and the projection order for the APA algorithm was 15. It can be seen that lower number of bands in the analysis filter bank results in lower SER values as the model cannot adequately characterize the complex, multi-channel compression of the hearing aid. An overall asymptotic trend can be observed for the SER for increasing analysis bands. It can also be noticed that an increase in the number of subbands results in much better performance for Syncro. Since Syncro has eight independent compression channels, at least 8 bands are required in adequately characterizing its dynamic behavior. On the other hand, Symbio is marketed as a channel-free compression hearing aid, and the results show that the performance improvement with an increasing number of subbands is not as significant.

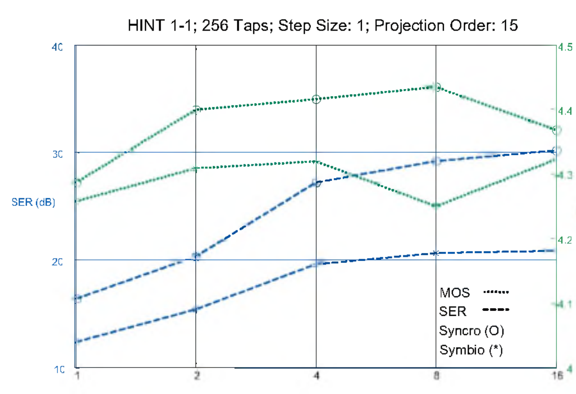

Figure 2. MOS and SER Values

Similar conclusions can be drawn from the PESQ MOS results. For the Syncro, the subband model does a poor job of characterizing the compression behavior of the hearing aid when it has either too few or too many analysis bands. The largest MOS value occurs when the model has eight analysis bands. For the Symbio, the largest MOS value occurred with sixteen analysis bands.

In conclusion, to properly characterize the complex and dynamic behavior of multi-channel amplitude compression strategies used in current generation hearing aids, a subband adaptive model is necessary. Our previous simulation results [5] and the experimental results shown in this paper suggest that as the number of bands in the subband adaptive model increases to match or exceed the number of compression channels in the hearing aid being modeled, the effectual SER value improves in a positive, asymptotic manner. In addition, the PESQ MOS scores support the generalized behavior of the subband model. It also appears that the subband adaptive model is able to characterize devices which process speech in either in the temporal domain (Bernafon Symbio), or the frequency domain (Oticon Syncro).

5. REFERENCES

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6. ACKNOWLEDGEMENTS

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