OBJECTIVE ASSESSMENT OF COMPANDING ARCHITECTURE FOR ASSISTIVE HEARING DEVICES

Farid Moshgelani [*1] and Vijay Parsa [†1]
[1]National Centre for Audiology, Western University, London, Ontario, Canada.

1 Introduction

Understanding speech in noisy environments is a significant challenge for hearing impaired listeners. The situation becomes more challenging with poorer Signal-to-Noise Ratios (SNRs). The listening difficulty arises from deficits in temporal, spectral, binaural, and/or cognitive processing. This research focuses on enhancing speech for hearing impaired listeners with poor temporal and spectral processing (e.g., individuals with auditory neuropathy spectrum disorder (ANSD) [1] . In particular, the present research evaluates the performance of a companding signal processing architecture in enhancing both temporal and spectral cues within a speech signal. Previous research has shown that this companding architecture enhances speech perception by ANSD and Cochlear Implant (CI) subjects [1,4], and the present work builds on the earlier published results.

In order to evaluate the performance of the companding architecture, subjective and/or objective quality and intelligibility measurements are required. Subjective methods require individuals to judge the quality and intelligibility of the processed speech signal. However, subjective measurements are costly and time consuming processes [2]. As a result, computer-based objective measurement techniques have been proposed to estimate speech intelligibility and quality in the presence or absence of background noise. Generally, objective measurement methods can be divided into two categories: intrusive or non-intrusive [2]. The intrusive techniques perform the measurement with respect to a reference signal (clean speech), whereas the non-intrusive methods perform the measurement independent of the reference speech signal. In the present study, the performance of the companding architecture was assessed objectively using a non-intrusive metric, the speech-to-reverberation modulation energy ratio (SRMR). The objective assessment was conducted across two experiments. In the first experiment, the noisy speech stimuli at different SNRs were processed with the companding architecture, and the processed stimuli were assessed using the objective metric. The second experiment was similar to first, except a Minimum-Mean-Square-Error (MMSE) noise reduction algorithm [3] was applied before processing through the companding architecture. Results from these experiments are expected to signify the practical application of companding architecture in assistive hearing devices.

[*] fmoshgel@uwo.ca
[†] parsa@nca.uwo.ca

The remainder of this paper is organized as follows: Section 2 presents a brief overview of the companding architecture, the noise reduction algorithm, and the SRMR metric. Section 3 reports the experimental methodology and results, and section 4 concludes the paper.

2 Companding, noise reduction, and SRMR

2.1 Companding architecture

The companding algorithm for the present study was adopted from Bhattacharya & Zeng [4] and implemented in MATLAB. Fig. 1 illustrates the block diagram for a single channel in the companding architecture. The algorithm consists of two individual blocks: compression and expansion. The input speech signal is first divided into 50 frequency channels using a bank of relatively broad band bandpass filters (BBBPFs). Next, the signal in each channel is subjected to amplitude compression. The compression index (n_1) and the output of the envelope detector (ED) determine the amount of compression. The compressed speech signal is then passed through a relatively narrow bandpass filters (NBBPFs) before being expanded in the expansion block. The amount of expansion is determined by the corresponding ED output and the ratio $(n_2-n_1)/n_1$, where n_2 is the expansion index. Subsequently, the outputs from all the channels are combined to obtain the processed signal.

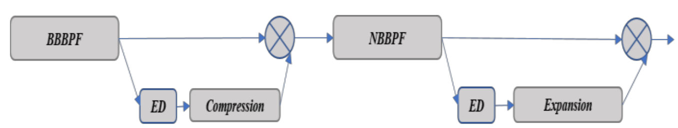

Figure 1: A single channel within the companding architecture [4].

2.2 Noise reduction algorithm

In a typical hearing aid application, the acoustic mixture of speech and background noise is received by the hearing aid microphones. However, it is known that the effectiveness of the companding algorithm reduces in the presence of background noise [1]. Hence, a noise reduction algorithm is imperative as a front-end to the companding algorithm. In the present research study, the MMSE noise reduction algorithm [3] was applied to noisy speech at different SNR conditions before applying the companding algorithm. The MMSE algorithm was chosen as it generates fewer artifacts ("musical noise") typically associated with noise reduction algorithms [3].

2.3 SRMR objective measurement

Computation of the SRMR metric involves the following steps [2]. First, cochlear processing is emulated by passing the input signal through a 23-channel gammatone filterbank. Second, temporal envelopes are extracted in each channel using the Hilbert transform and multiplied by the Hamming window. Third, the modulation spectral energy in each channel is computed as the squared magnitude of the discrete Fourier transform of the temporal envelope. Finally, the modulation energy across all acoustic frequencies is computed for lower and higher modulation frequency bands, and the ratio of modulation energy in lower/higher modulation frequency bands is the SRMR metric [2].

3 Experimental methodology and results

3.1 Method

The companding algorithm was implemented in MATLAB, with the n_1 and n_2 parameters set to 0.3 and 1 respectively for all the experiments. In addition, the Root-Mean-Square (RMS) value of the companded signal was equated to that of the original input signal.

The clean speech sentences used in the present study were taken from the hearing in noise test (HINT) database [5]. This test contains 25 lists with each list consisting of 10 sentences which are phonetically balanced, and are equally difficult. It is pertinent to point that one list is selected randomly for each condition and/or experiments described below. Objective results are shown as the SRMR scores, averaged over the ten sentences in that randomly selected list. In conditions involving background noise, the HINT speech-shaped-noise was mixed with the clean speech at different SNRs before applying the noise reduction and/or companding algorithms.

3.2 Results

Fig. 2 displays a sample experimental result wherein the long-term averaged spectra of noisy speech (SNR = 0 dB) are compared across three processing conditions: unprocessed, companding alone, and a combination of noise reduction and companding. It can be seen that companding alone does sharpen the speech spectral peaks. However, applying both noise reduction and companding to noisy speech at the same SNR value results in a significantly better sharpening of the spectral peaks.

In addition, Fig. 3 depicts the SRMR metric computed across different SNR and processing conditions. Note that a higher SRMR value denotes a more intelligible signal. The results from Fig. 3 demonstrate a significant benefit of the companding condition over unprocessed, regardless of the SNR value. Furthermore, results from Fig 3 reveal that the application of noise reduction algorithm can improve the performance of companding architecture, once again irrespective of the SNR value. It can also be seen that the amount of improvement with the noise reduction algorithm was significantly higher for SNRs less than 15 dB.

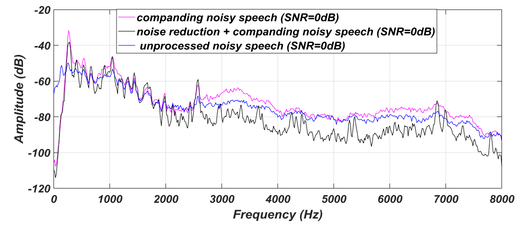

Figure 2: Comparison of long-term average power spectra

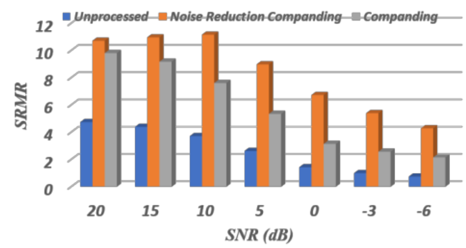

Figure 3: SRMR values for different processing conditions

4 Conclusion

Two experiments were conducted to investigate the performance of a companding architecture objectively using the SRMR metric. The first experiment was conducted to explore the performance of the companding algorithm in terms of the predicted speech intelligibility score in the presence of background noise. The second experiment was conducted to evaluate the effectiveness of the same companding architecture by incorporating an additional noise reduction algorithm. Results showed that the incorporation of the noise reduction algorithm does expand the effectiveness of the companding algorithm over a wider SNR range. These results can potentially guide the choice and activation of companding architecture in assistive hearing devices.

References

[1] V. Narne and A. Barman, "Effect of Companding on Speech Recognition in Quiet and Noise for Listeners with ANSD," *Int. J. Audiol.*, vol. 53, pp. 94–100, 2014.

[2] T. H. Falk, C. X. Zheng, and W.-Y. Chan, "A Non-Intrusive Quality and Intelligibillity Measure of Reverberant and Dereverberated Speech," *IEEE Trans. Audio,Speech, Lang. Process.*, vol. 18, no. 7, pp. 1766–1774, 2010.

[3] P. C. Loizou, *Speech Enhancement : Theory and Practice*, 2nd. 2013.

[4] F. Zeng and A. Bhattacharya, "Companding to Improve Cochlear-Implant Speech Recognition in Speech-Shaped Noise," *Acoust. Soc. Am.*, vol. 122, no. 2, pp. 1079–1089, 2007.

[5] M. Nilsson, S. D. Soli, and S. JA, "Development of the Hearing in Noise Test for the Measurement of Speech Reception Thresholds in Quiet and in Noise," *Acoust. Soc. Am.*, vol. 95, no. 2, pp. 1085–1099, 1994.