Envelope enhancement for improving hearing in reverberant spaces

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Summary and Introduction

Cochlear implants are neural prostheses which give a sense of hearing to deaf people by stimulating the auditory nerve. In many users they restore the ability to understand speech in quiet, but noise and reverberation cause severe problems. It seems that implant users have difficulties hearing out one source in a potpourri of sources. This so called "auditory scene analysis" relies, amongst other cues, on binaural information which is degraded with implants. Sound localization ability, directly dependent on access to binaural cues, is highly sensitive to the presence of noise or reverberation [1, 2]. These studies have shown that noise and reverberation affects localization ability of cochlear implant users already at a signal-to-noise ratio (or direct-to-reverberant ratio) about 6-12 dB more positive than that found with normal hearing listeners. Here we focus on the coding of binaural information to improve the perception of sound direction – an important aspect for locating and understanding speakers. Several ways exist to improve the coding of binaural cues in reverberation in cochlear implants. Most past approaches have attempted to reduce the energy of the interfering reverberation or noise. Our novel approach instead aims to increase the perceptual saliency of the cues used to locate sounds in reverberation. Studies of sound localization and the precedence effect have shown that information in the sound onset is important for correct localization in reverberant space [3, 4, 5, 6]. Onset information is transmitted in the modulated envelope of the electrode signals in cochlear implants. However, due to the bandbass filtering and compression stages in the implant, modulation depth and onset gradient inherent in envelope fluctuations are reduced – both are important factors for transmitting interaural time difference cues (ITDs) in the envelope [3].

Here we present a new method to enhance the envelope signal by shaping it. The implementation of the algorithm is discussed and results of evaluation experiments are presented. In simulations of implant use with normal-hearing listeners we demonstrate that by changing the coding of the target sound to better transmit its binaural cues, it can be localized better in reverberation. Despite the envelope alteration, speech understanding is not affected. The new approach only alters the transmitted envelope signal and can thus be implemented in commercial devices without changing the implanted part.

Methods

Onset enhancement algorithm

Monaghan et al. 2013 [3] show that the discrimination of envelope ITDs depends crucially on the onset gradient, modulation (onset) depth, repetition rate (the duration of the gap before an onset) and interaural coherence of the envelope. The present algorithm manipulates onset gradients and modulation depth with the aim to improve ITD discrimination and localization. For a first proof of principle the algorithm is implemented for and evaluated with normal-hearing listeners instead of bilateral cochlear implant users since the latter are hard to access. The impact of the cochlear implant signal processing is mimicked with a vocoder [7]. The processing schematic is depicted in Figure 1. Since the normal-hearing auditory system is particularly sensitive to envelope ITDs at high frequencies, and since ITDs carried in the signal's fine structure at low frequencies should not interfere with the evaluation [8], the vocoder is restricted to frequencies above 2 kHz. The signal is filtered into 6 frequency bands. From each band-passed signal the envelope is extracted by rectification and low-pass filtering before it is sent to the enhancement stage. After altering the envelope it is modulated on a sinusoidal carrier, the modulated signals for each channel are bandpass-filtered, summed and presented via headphones.

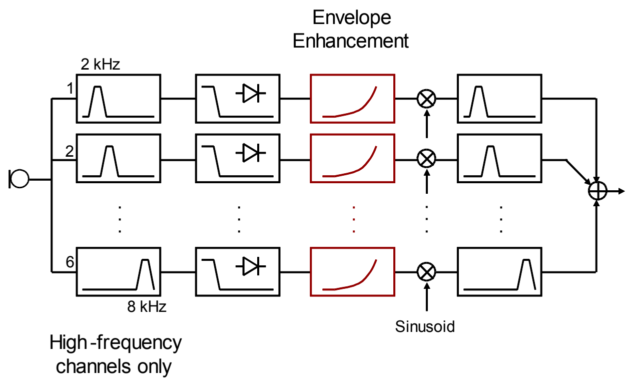

Figure 1: Vocoding approach to evaluate the onset enhancement algorithm. The signal is filtered into 6 frequency channels with high center frequency. In each channel, the envelope is extracted and passed through the enhancement stage before it is used to modulate a carrier tone (sinusoidal vocoder). The modulated tones are bandpass-filtered again before being summed for headphone presentation.

The envelope enhancement stage functions as follows (figure 2): In each channel-wise envelope signal, minima and maxima are marked. Maxima at which the direct-to-reverberant ratio is larger than 0 dB are selected if an interaurally matching maximum can be found in temporal vicinity. On both, the left and the right ear channel, the envelope is set to zero starting at the minimum prior to the selected maximum. This leads to a highest onset gradient and modulation depth for the selected maximum. Since the selection criterion ensures that the maximum is dominated by the direct sound, it is likely that the binaural cues carried in it are congruent with the direct sound location.

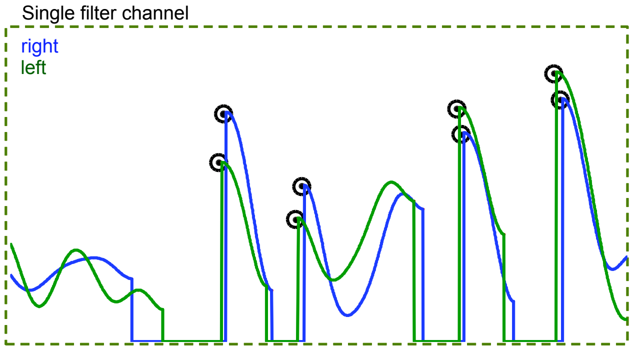

Figure 2: Envelope enhancement stage. Minima and maxima are found in the channel-wise envelope signals. The envelope is set to zero between the preceding minimum and each specifically selected maximum. This maximizes the onset gradient and modulation depth, both important for ITDs.

The effect of the algorithm was evaluated in three listening experiments. In order to evaluate localization improvements for the direct sound target, ITDs were applied in all experiments only to the direct sound component while the reverberant part was kept unchanged. Evaluation was made for different distances from the target, and hence at different direct-to-reverberant ratios. A reverberated speech signal was passed through the onset-enhancement vocoder. Seven normal hearing listeners (<20 dB HL) participated in the evaluation. Subjects were paid and the study was approved by an ethics committee.

ITD discrimination

A standard ITD discrimination test was used to assess if the algorithm improved access to ITDs in the direct sound part. ITDs were changed with a Levitt-tracker as in [3].

ITD-based lateralization

The impact of the enhancement algorithm on lateralization was studied with the stimuli of the ITD discrimination test. Lateralization was measured for fixed ITDs using the line-dissection paradigm of Seeber and Hafter (2011) [8, 9].

Speech understanding

Any improvements of localization are only worthwhile if the algorithm does not impair speech understanding. Hence, speech understanding of a single source in reverberation processed with the high frequency vocoder was measured. IEEE sentences were used in a paradigm similar to that of Wiggins and Seeber 2013 [10].

Results and Discussion

Results obtained with the onset-enhancement vocoder show improved ITD discrimination at short distances from the source. What is more, ITD-based lateralization was also improved. With the algorithm, vocoded reverberant stimuli were perceived about twice as far lateral as without vocoding, demonstrating that the aim to improve localization has been achieved. These improvements occur only at short distances from the source where the direct-to-reverberant ratio is not too negative such that onsets dominated by the direct sound can be identified. At larger distances the number of identified onsets declines, reducing the effect of the algorithm. Speech understanding is not negatively affected by the algorithm despite the severe envelope changes. However, informal listening shows that speech quality appears poorer since the speech sounds rougher.

The onset enhancement approach is a promising new way to improve sound localization ability in reverberation through changing the relative salience of the elements in the sound carrying the (binaural) information. Unlike many other approaches interfering energy is not subtracted. Instead, improved localization was demonstrated through increasing perceptual access to binaural cues by sharpened modulation.

Acknowledgements

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