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Objective and subjective assessment of envelope enhancement algorithms for assistive hearing devices

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ABSTRACT

Speech perception in a noisy environment is a significant challenge for individuals with auditory processing deficits. Evidence exists that exaggerating the slow temporal modulations may enhance speech perception for these individuals in the absence or presence of background noise. Nevertheless, a comprehensive assessment of envelope enhancement algorithms is lacking. In the present research study, two different schemes of envelope enhancement (dynamic and static) were evaluated subjectively and objectively for different types and levels of background noise with and without applying a noise reduction algorithm. In the subjective assessment, the dynamic envelope enhancement algorithm was evaluated with three different subjective groups including, twelve normal adults, twelve normal children, and eleven children with suspected auditory processing disorder (APD). The subjective results revealed that the speech intelligibility scores were lower for APD subjects compared to both normal adults and normal children. The subjective results also demonstrated that enhancing the temporal envelope is much more beneficial for subjects with suspected APD when compared to normal adults and children participating in the subjective experiment. In the objective assessment, the Hearing Aid Speech Perception Index (HASPI) was employed to predict the speech intelligibility scores which correlated highly with the subjective data. Comprehensive objective experiments demonstrated that both dynamic and static envelope enhancement algorithms are only effective in improving speech perception under certain processing conditions that depended on the type, level and location of the background noise. It is also shown that the application of a noise reduction algorithm prior to the envelope enhancement algorithms will increase their range of effectiveness.

1. Introduction

Hearing assessment typically involves the measurement of hearing sensitivity in different frequency regions resulting in an audiogram [1]. While the audiogram is the front-line measurement of hearing loss, it does not adequately describe the functioning of the impaired auditory system. In particular, it does not capture the auditory processing capabilities such as auditory discrimination, auditory pattern recognition, auditory performance in noisy and reverberant environments, and performance in the presence of competing signals [2]. It is estimated that about 3–5 % of children suffer from auditory processing disorder (APD), which directly impacts their ability to learn from what is heard and to communicate with others [3]. Auditory processing deficits are also prevalent in adults, particularly in adults aged over 60 [3]. In addition, the Auditory Neuropathy Spectrum Disorder (ANSD), which affects up to 15% of children with permanent hearing loss [4], can be considered as a subset of the broader APD category wherein disruptions in auditory nerve and central auditory pathways significantly degrade auditory processing capabilities [5].

Conventional hearing aids and assistive listening devices offer little benefit to listeners with auditory processing deficits. For example, Mathai and Appu [6] investigated the effect of four different hearing aid settings on speech perception of seventeen adults with late onset ANSD. Results showed no significant differences between unaided and aided performance, indicating the lack of benefit from conventional amplification. Walker et al. [4] compared the speech perception capabilities of children with ANSD and children with sensorineural hearing loss, and found that the ANSD children demonstrated inferior speech recognition in noisy

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environment even with the provision of a hearing aid. In a similar vein, Kuk [7] reported data on hearing aid benefit collected from fourteen children diagnosed with APD, which showed that some but not all participants demonstrated improved performance with hearing aids. Researchers have also investigated the effectiveness of remote microphone (RM) assistive listening devices for APD and ANSD populations (e.g [8,9].). A recent systematic review by Reynolds et al. [10] found moderate benefit from RM systems for children with APD, although the practicality of using an RM in a number of ecologically valid situations has been questioned by Kuk [7].

Given the lack of improvement with conventional hearing aid technologies, researchers have investigated alternate speech processing methodologies. For example, it is known that listeners with ANSD exhibit abnormal modulation transfer function (MTF) thresholds indicating reduced ability to follow slow temporal speech modulations (Zeng et al. [11]). Similarly, Sharma et al. [12] reported significantly poorer thresholds for 4 Hz amplitude modulation detection by children with reported listening difficulties. Thus, envelope enhancement (EE) algorithms which attempt to mitigate temporal modulation processing deficiencies by enhancing the temporal peaks and valleys of a speech signal are an attractive option, and evidence exists that such a strategy can indeed be effective. For example, Narne and colleagues ([13–15]) have investigated EE algorithms with ANSD subjects and demonstrated improved speech perception benefits. Similarly, the impact of a EE algorithm on phrase and consonant identification was evaluated in individuals with normal hearing, sensorineural hearing loss, and ANSD under quiet and background noise conditions with promising results ([16,17]).

While these results are promising, a few research questions still remain. For example, previous studies only explored the assessment of EE algorithms after application to short segments of speech (consonants, vowels, and words). Furthermore, a comprehensive assessment of the impact of background noise on the performance of EE algorithms is lacking. To elaborate on this further, consider Fig. 1 which represents a typical RM setup where the RM is placed close to the source and the listener is wearing the hearing aid, which is connected wirelessly to the RM. Depending on where the EE algorithm is implemented (either in the RM or the hearing aid), background noise can add to the desired signal before and after the EE algorithm, and can potentially create a differential impact. In previous studies, the background noise was only added after the speech signal is enhanced and ready to be transferred to a listener. In addition to the noise location, the impact of different noise types (stationary vs. non-stationary) has not been previously investigated.

The performance of EE algorithms is typically evaluated using subjective measurements. Subjective methods require individuals to judge the quality and intelligibility of the processed speech signal. However, subjective measurements are costly and time consuming processes as they require individuals to participate in an experiment [18]. Since one of the goals of the present study is to benchmark the EE algorithms across a number of noisy conditions, subjective testing can become an onerous task. As such, objective, instrumental measures that employ computational models to predict the speech quality and intelligibility are attractive ([18–21]). A good objective metric that correlates well with subjective data can be used as a surrogate for benchmarking EE algorithm performance and allows for a more comprehensive assessment of EE algorithm across a number of noise types and SNRs.

In summary, EE algorithms have the potential to enhance speech perception capabilities for listeners with temporal auditory processing deficits. This paper contributes new results on the performance of EE algorithms by answering the following research questions: (a) does an EE algorithm enhance speech intelligibility for children with suspected APD? (b) can an objective speech intelligibility metric be derived to predict the perceptual impact of this EE algorithm? and (c) how do EE algorithms perform in a variety of noisy conditions, as evaluated using the validated objective metric?

2. Materials and methods

2.1. EE algorithms

The present study benchmarked the performance of two EE algorithms – a dynamic EE algorithm explored in Narne et al. [14], and the static EE algorithm investigated by Shetty and Kooknoor [16]. Fig. 2 illustrates the block diagram of the dynamic EE algorithm, wherein the input speech signal is split into four channels by 6th order Butterworth bandpass filters whose bandwidths range between 150–550, 550–1550, 1550–3550, and 3550–8000 Hz respectively. The temporal envelope in each frequency channel is extracted through a combination of full-wave rectification and a first order low pass Butterworth filter with a cutoff frequency of 32 Hz [14]. The extracted envelope in the ith band is exaggerated by raising it to the power k_{bi}, which is calculated in each band separately through an exponential functionas shown in Eq. (1):

$$k_{bi} = e^{\frac{(E_{bmin} - E_{bi})}{\tau}} (k_{max} - k_{min}) + k_{min} \quad (1)$$

where k_{min} = 0.3, k_{max} = 4, E_{bmin} is the minimum amplitude of the envelope in the i^{th} band, E_{bi} is the instantaneous amplitude value of the envelope in the i^{th} band, and τ is the time constant for the exponential. It is pertinent to point out that both k_{bi} and E_{bmin} are calculated for each band independently. An instantaneous correction factor is obtained next for each sample by computing the ratio of the expanded envelope to the original envelope. The correction factor is then multiplied with the original bandpass signal

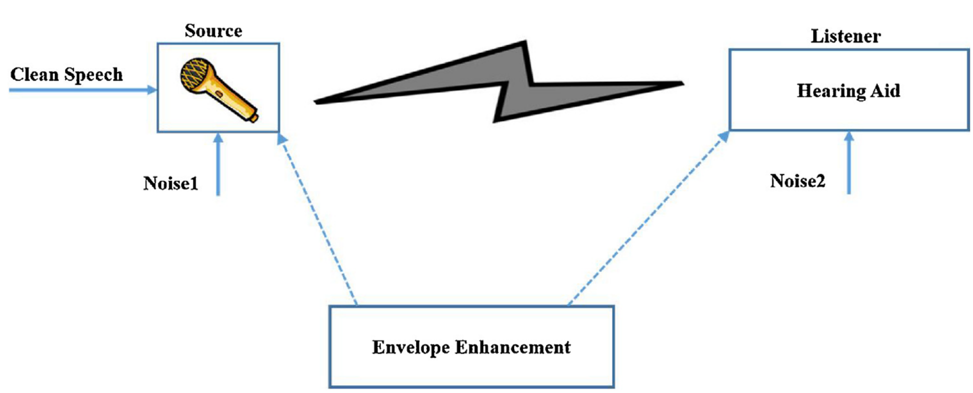

Fig. 1. Block diagram of a typical assistive listening device setup incorporating the envelope enhancement strategy.