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Survey on Methods for Speech Intelligibility Enhancement in Hearing Aids

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Abstract

Communication system involves a speaker, listener and a communication device. The speech signal should be transmitted such that it reaches the listener with good quality. In real life scenario, the speech signal reaches the listener with various disturbances. Physiological and pathological disabilities of human auditory system reduce speech intelligibility under noisy environment. Hence hearing aids need to integrate noise reduction algorithm with amplification. The prime objective of noise reduction algorithm is to raise the signal-to-noise ratio of the received signal to be fed to the ear drum. The binaural setup provides significant increase in speech intelligibility than monaural setup. The binaural noise reduction algorithm should be designed such that the rhythm pattern and the Interaural Coherence are preserved. The time delay and complexity of the algorithm should be minimalistic so as to have better performance.

Keywords: Speech intelligibility, noise reduction, binaural, and hearing aids etc.

Introduction

Hearing aid is electronic equipment which amplifies sound signal and makes speech audible and understandable. It is designed to receive sound signals with a microphone, amplifies weaker sounds and transmits them to the eardrum through a tiny speaker. It helps hearing impaired persons to perceive sound and improve their intelligibility. With the advance in digital signal processors and microchips hearing aids have got compact and have significant improvement in quality.

The real life listening scenario usually includes multiple sound sources overlapped in time and frequency or with single speech source and background noise. Hearing impaired persons subjected to these noisy inputs will have reduced speech intelligibility. Hence methods to enhance speech intelligibility are inevitable. The two methods used with this objective are: directional microphone and noise reduction algorithm [1]. The first method is designed on the basis of differences in speech and noise characteristics in space and utilizing the beamforming technology to enhance the directionality of speech signal. But this method makes the hearing aid complex and bulkier. The second way is by using a noise reduction algorithm and there by separating the noise from noisy speech signal.

This paper studies the various factors influencing the speech intelligibility and methods adopted to enhance speech intelligibility. The various performance evaluation methods are also discussed here.

Speech Intelligibility Enhancement

Thomas et al. [2] introduced the method of filtering and clipping to enhance the intelligibility of speech in noise. A high pass filter is used to attenuate the low frequency components and to pass all the high frequency components in the required noisy speech and thus enhancing the second formant frequency with respect to first formant. The consonants which have weaker power than vowels convey more information and make the speech intelligible [3]. But the noise components mask the consonants more than vowels. So true speech need to be reprocessed such that the power of consonants is high and can be done by an amplitude clipping circuit. Even though clipping improves speech intelligibility, it introduces harmonic distortion in the processed speech. These distorted components usually fall within the high frequency range. This appears as a disturbance to the listeners which in turn reduces the speech intelligibility.

Russel and James [4] presented the speech enhancement method of rapid amplitude compression followed by high pass filtering to preprocess speech signal. This method replaces amplitude clipping circuit by an amplitude compressor in which the output signal amplitude is maintained at a constant level irrespective of the input for some input amplitude range. The basic idea behind this method is that the amplitude of speech spectrum lies within a given range and the spectrum outside this amplitude range corresponds to that of noise signal. The two parameters of an amplitude compressor are the attack and release time and the normalization range [4]. For improving the intelligibility of processed speech, the speech signal should have phonetic events length greater than the attack and release time and the speech signal dynamic range should be less than the normalization range. This method results in a drastic improvement in speech intelligibility.

D.Deepa et al.[5] suggested spectral subtraction method to process speech signal with background noise. The magnitude spectrum of background noise is subtracted from noisy speech signal to obtain a clear speech. The noise estimate is measured during non-speech activity. The enhancement in speech intelligibility depends on the accuracy of noise estimate. In the proposed method the quality of processed speech is measured in terms of signal to noise ratio. This technique can be implemented in hearing aids to impart a significant increase in SNR of 5 dB to 10dB.

C. H. Taal et al. [6] introduced perceptional distortion measure based pre-processing algorithm to enhance speech quality for the near-end listener. The algorithm is based on optimal speech energy redistribution over frequency and time domain. Since distortion measure is used as the parameter speech signal. Thus the transient part of speech gains more amplification than vowels which in turn increases the speech intelligibility.

Joao.B.Crespo et al. [7] describes speech reinforcement technique to enhance the speech intelligibility. The author proposes the perceptual distortion measure [7] method in which the speech signal is segmented into sequences of frames and an auditory model is applied to each short time frame. The model consists of auditory filter bank followed by absolute squared and low pass filter. The processed speech signal passing through a reverberant environment reaches the listener with a good quality. To achieve this past speech frames and noise are suppressed. The drawback of the algorithm is that it acts as a multiband dynamic range compressor [6] till overlap-masking occurs during pre-processing.

V.Balakrishnan et al.[8] proposed wavelet denoising as a method for speech intelligibility enhancement. The method uses wavelet packet transforms [8] to divide the entire frequency range into numerous frequency bins. Thus the speech signal can be transformed to the wavelet coefficients by various levels of iterations to approximations and dilations. The coefficients can be threshold either by hard thresholding or soft thresholding. The threshold is adjusted such that its full magnitude is desired in silence period and is zero during periods of speech. The transformed coefficients can be reconstructed to yield speech signals with high SNR. Thus the speech signal is extracted from noise background.

Plapous et al. [9] proposed Wiener filter for noise reduction in hearing aids. Wiener filter is an optimal filter which generates an estimate of speech signal by minimizing the mean square error criteria. It is adaptive in nature in the sense that the filter coefficients are made adaptive according to the error signal. If we have a reference signal available, the various characteristics of the reference signal can be exploited to achieve noise reduction with low level of speech distortion. When no references are available, we can still achieve a better noise reduction by properly manipulating the Wiener filter, resulting in a suboptimal Wiener filter. The estimated signal will have high SNR and quality. But the presence of apriori SNR increases the correlation between the present frame gain function and the past frame. This introduces reverberations n the speech signal. To overcome this Two-Step Noise Reduction (TSNR) algorithm is used. An extension of TSNR, Harmonic Regeneration Noise Reduction (HRNR) is used to overcome the harmonic distortion created by it.

Zheng Hong et al.[10] did a comparative study on the two NR techniques based on Wiener filter and Kalman filter. Conventional hearing aid systems use an amplifier circuit to provide a gain to the input noise speech signal. Even though the amplifier circuit amplifies the input speech, it amplifies the noise along with it. The person with hearing impairment will hear but not understand the speech in a noisy environment. Hence speech enhancement based hearing aids are developed. The wiener filter and kalman filter based hearing aids were designed. The two hearing aids were subjected to objective and subjective test. The kalman filter based hearing aid outperformed wiener filter under different noise conditions.

Romain Serizel et al.[11] suggested a two stage method to improve the speech intelligibility in hearing aids. It includes a noise reduction and active noise control [11] stage. The noise reduction is implemented using a multichannel wiener filter [12]. This stage is used to preprocess the noisy speech signal within the hearing aid and thereby increasing the SNR. However the leakage signal which is prominent in open fitting hearing aid reduces the SNR gain of the pre-processed signal. Active noise control [13] is the technique used to reduce the effect of leakage signal and hence improving the final signal-to-noise ratio. The NR and ANC stages are connected in cascade and show a significant increase in SNR until the causality constraint [14] is satisfied. Parallel implementation of the two functional blocks yield a robust algorithm. However the integrated version of the two functional blocks outperforms the cascaded and parallel versions in terms of improvement in signal-to-noise ratio and improvement in intelligibility.

Romain Serizel et al.[15] proposed an integrated approach to active noise control and noise reduction in open fitted hearing aids. The algorithm is based on generating a zone of quite [15] in the ear canal. The zone of quite uses destructive interference by generating an error control signal which interferes with the leakage signal[16] and reflections and reduces the noise content in the signal reaching the eardrum. An effective feedback control scheme [17] is implemented based on an average mean squad error criterion and the system shows a significant increase in SNR on the desired zone of quite. The major drawback of this system is that the performance depends on causality and number of noise sources.

Romain Serizel et al.[18] introduced a weighting factor between the NR and ANC scheme. Based on the weighed mean square error criterion, either the ANC or the NR scheme is given the weightage. Based on this algorithm it is possible to alternate the residue noise reaching the eardrum without any feedback control system. The method succeeds in delivering a high SNR for a given weighting factor.

Romain Serizel et.al [19] designed a binaural integrated active noise control and noise reduction system to enhance the speech intelligibility in hearing aids. A binaural hearing aid generates output signal for each ear from bottom left and right hearing aid. The speech signal received by each ear is preprocessed by a multichannel wiener filter. When the integrated scheme is implemented in a monaural or bilateral setup, eventhough there is increase in the signal-to-noise ratio, the enhancement of speech quality is less due to the degradation of binaural localization cues. In the binaural setup the microphone in both ears are synchronized and hence the system allows greater causality margin. Hence noise from large number of sources and from distinct directions can be attenuated by this method.

Daniel Marquardt et al.[20] established a theory based on preserving the Interaural Coherence(IC) in binaural hearing aid to improve the speech intelligibility. Most of the binaural hearing aid, uses a multi-channel wiener filter to reduce noise and limit the distortion in the filtered signal. But however besides noise reduction, preserving binaural cues s inevitable to make the audio signal intelligible. Hence extension of multi channel wiener filter (MWF), MWF-ITF [20] was developed. MWF-ITF preserves the interaural transfer function of the noise source. Eventhough it outperforms MWF in case of directional noise source, its performance degrades in case of spatially isotropic noise. Hence MWF-ITF was modified to MWF-IC which preserves the interaural coherence of the noise source. But there is a tradeoff between the preservation of IC and the level of noise reduction. Hence the IC preservation is limited to the coherence discrimination ability of human ear. The IC is important to localize noise sources n a reverberant and multisource environment. It has a wide range application in a hearing aid which is prone to a noisy input under reverberant condition. The MWF-IC preserves the IC of the output noise component without degrading the performance of noise reduction algorithm.

Alexander Schasse et al.[21] designed a two stage filter bank system to improving speech intelligibility in hearing aid. In most of the noise reduction algorithms, the performance highly depends on the frequency resolution. Poor frequency resolution leads to distortion in the processed signal due to residual noise artifacts. This is because the noise between the harmonics cannot be removed effectively. Here a cascaded version of filter bank system is designed. The first stage FBS does the amplification and compression task and the second stage FBS improves the frequency resolution. The second FBS is designed with small algorithmic delay and less computational complexity.

Jessica Slater et al.[22] evaluated the role of rhythms of speech intelligibility of a listener in a noisy environment. This paper is based on the concept that the speech intelligibility not only depends on the understandability of the words or sentences in noise but also depends on the sound patterns they create especially for hearing impaired persons. The timing patterns of speech provide perceptual cues. For example, under noisy environment the boundaries between words can be distinguished by analyzing durational patterns. Hence we can infer that a system sensitive to temporal patterns is important in analyzing the elements from same source sharing same temporal characteristics. There are two tests conducted to understand the influence of rhythmic pattern in speech-in-noise perception [22]. They are quick Speech-in-Noise test (QuickSIN)[23] and Word-in-Noise(WIN) [24]. Both tests calculated SNR loss such that more poor the SNR loss, better the speech intelligibility. The test was performed on three groups of people mainly non-musicians, vocalists and drummers. The drummers who have better rhythm analysis capability have more speech intelligibility in noise.

Li Zhang et al.[25] developed the objective evaluation system for analyzing the performance of noise reduction stage in hearing aid. Even though there are different subjective methods for the performance evaluation of NR, it is restricted by test conditions and are time consuming. Hence he suggested two objective methods namely [25] SNR and SegSNR. SNR indicates the noise influences on the speech signal. SNR refers to the logarithm of ratio of signal power to noise power in decibels [25]. SegSNR indicates the time average of SNR computed in each time frame. The simulations proved that the objective methods described in the paper are good estimates of performance of a NR stage.

Conclusion

In this paper a survey of various methods that can be implemented to enhance the speech intelligibility is being discussed. The noise reduction algorithm needs to discriminate between the speech and noise occupying the same spectrum. Most of the conventional hearing aids use a Wiener filter to perform noise reduction. Apparently recent studies prove that the Kalman filter provides a significant advantage compared to Wiener filter in noise reduction capability. The performance of the hearing aid is highly dependent on causality constrain, number of noise sources, rhythm pattern and interaural coherence. Advanced hearing aids uses active noise control algorithm to prevent the degradation of speech quality even in the presence of leakage signals reaching the ear drum. Various objective and subjective tests used to evaluate the performance of hearing aids are also discussed in this paper.

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