

Performance Improvement of Digital Hearing Aid Systems

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Abstract— Digital hearing aids addresses the issues of noise and speech intelligibility that is associated with the analogue types. One of the main functions of the digital signal processor (DSP) of digital hearing aid systems is noise reduction which can be achieved by speech enhancement algorithms which in turn improve system performance and flexibility. However, studies have shown that the quality of experience (QoE) with some of the current hearing aids is not up to expectation in a noisy environment due to interfering sound, background noise and reverberation. It is also suggested that noise reduction features of the DSP can be further improved accordingly. Recently, we proposed an adaptive spectral subtraction algorithm to enhance the performance of communication systems and address the issue of associated musical noise generated by the conventional spectral subtraction algorithm. The effectiveness of the algorithm has been confirmed by different objective and subjective evaluations. In this study, an adaptive spectral subtraction algorithm is implemented using the noise-estimation algorithm for highly non-stationary noisy environments instead of the voice activity detection (VAD) employed in our previous work due to its effectiveness. Also, signal to residual spectrum ratio (SR) is implemented in order to control the amplification distortion for speech intelligibility improvement. The results show that the proposed scheme gives comparatively better performance and can be easily employed in digital hearing aid system for improving speech quality and intelligibility.

Keywords—speech enhancement; intelligibility; musical noise; sub-band; hearing aids; noise-estimation.

I. INTRODUCTION

A hearing aid is a small electroacoustic device which can be wear in or behind the ear. It is designed for the enhancement of speech intelligibility and for correcting impaired hearing as measured by the audiometry. There are two main types of hearing aids, namely, the analog and the digital types [1], [2], [3], [4], [5]. The former operates by processing the electrical sound signal by simply amplifying both the desired speech as well as the noise signal [6]. However, noise presence normally degrades the quality of services and the information content of acoustic signal [7]. Therefore, hearing-impaired people might be experiencing some difficulties in understanding what they heard [6]. Subsequently, to address this issue, a method that can suppress the noise while maintaining the required speech quality and intelligibility is essential to enhance the performance of the hearing-aid [8]. This challenge is addressed

by digital hearing aids. With digitization, electrical sound signal can undergo different advanced signal processing such as noise reduction, filtering, acoustic feedback cancellation, and sound classification. Consequently, digitization enhances the performance and flexibility of digital hearing aids as well as improving the listening comfort of the users [7]. Hence, speech enhancement is a key part of signal processing aiming at improving the signal-to-noise ratio (SNR), intelligibility and quality of speech signal [9], [10].

There are various solutions that have been proposed in the literature to address the issues of noise estimation and speech enhancement for hearing aids, one of such is a noise estimation by minima controlled recursive averaging proposed in [11] so as to achieve higher segmental SNR and a lower level of musical residual noise. Likewise, [12] presents a noise-estimation algorithm for highly non-stationary environments to address the issue of noise estimation for the enhancement of noisy speech. In the work, the algorithm is employed to continuously update the noise estimate in every frame using time–frequency smoothing factor which is based on the speech-presence probability in each frequency bin of the noisy speech spectrum. Furthermore, [6] develops two embedded hearing aid systems with noise reduction using Kalman and Wiener filtering techniques for speech enhancement and shows that the hearing aid system based on the Kalman filter comparatively increases the rate of speech recognition and the hearing comfort in a noisy environment. Also, [9] investigates the feasibility of a generalized maximum a posteriori spectral amplitude (GMAPA) speech enhancement algorithm for hearing aids and shows experimentally that GMAPA can achieve clear improvements on long-term SNR evaluations. Also, [10] proposes a discriminative post-filter (DPF) approach that uses a non-linear function to further enhance the SNR and quality of GMAPA speech enhancement algorithm. Moreover, [13] evaluates the effects of amplification distortion on the speech enhanced by Logmmse and Spectral Subtraction algorithms and shows that in order to improve the intelligibility when speech enhancement algorithm is employed, the amplification distortion should to controlled.

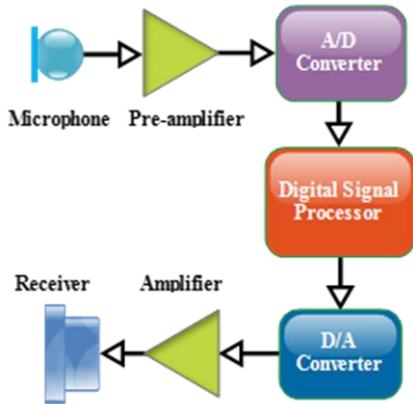
This paper focusses on noise reduction aspect of digital hearing aid systems in order to suppress noise that maintains a relative constant sound pressure level (SPL) over time. In our previous work in [7] and [8], adaptive-control factor and multi-band based spectral subtraction methods are implemented to

enhance the performance of communication systems and to address the issue of the associated musical noise generated by the conventional spectral subtraction algorithm. In this work, the signal to residual spectrum ratio (SR) presented in [13] is implemented with multi-band based spectral subtraction methods in order to control the amplification distortion for speech intelligibility improvement. Also, noise-estimation algorithm proposed in [12] for highly non-stationary noisy environments is employed. The proposed approach maintains high speech quality and mitigates anomalies which are related to the linear subtraction technique in order to enhance speech intelligibility.

The subsequent section gives an overview of a digital hearing aid system. Section III, presents the signal model and speech enhancement algorithms for performance improvements as well as the proposed algorithm. Also, in Section IV, experimental results and analysis which are based on objective and subjective tests are discussed. Conclusions are drawn in Section V.

II. OVERVIEW OF DIGITAL HEARING AIDS

The main components of a digital hearing aid are the microphone, analogue-to-digital converter (ADC), digital-to-analogue converter (DAC), digital signal processor (DSP), pre-amplifier, amplifier and receiver [2]. The incoming acoustic waves picked up the microphone are transformed into an electrical signal. The electrical signal is then converted to digital representation for further processing by the ADC [2], [3], [5]. The resulting signal is then processed by the DSP in which the signal undergoes different advanced signal processing such as noise reduction, adaptive filtering, acoustic feedback cancellation, and sound classification [5]. Moreover, the processed signal is transformed into analogue electrical signal by the DAC whose output drives the power amplifier that feed the receiver which in turn converts the signal back to the sound [3]. Fig. 1 depicts a block diagram of a generic digital hearing aid. The challenge is, designing an effective DSP while not exceeding a power budget that allows reasonable battery life [5]. This work focusses on the noise reduction aspect of the DSP in order to suppress noise that maintains a relative constant SPL over time. Therefore, noise reduction for hearing aids requires an effective signal model



for the system.

Fig. 1. Block diagram of a digital hearing aid system (adapted from [1], [5])

III. SIGNAL MODEL

Assume n point sources in a noisy environment, the received signal Y_m at microphone m in an individual frequency band is modeled as [14], [15]

$$Y_m = y_d + y_r + d_m \quad (1)$$

where y_d and y_r represent the wanted signal and the reflected signal received by the microphone respectively and d_m denotes the additive noise.

However, the received signal at the microphone basically consist of direct and the reflected sounds. The reflected sound y_r is also divided into the early reverberation y_e and the late reverberation y_l and is modeled as [15]:

$$y_r = y_e + y_l \quad (2)$$

Equation (2) can be written as

$$Y_m = y_d + y_e + y_l + d_m \quad (3)$$

Which can be expressed as

$$Y_m = H_{m,n}^d S_n + H_{m,n}^e S_n + H_{m,n}^l S_n + d_m \quad (4)$$

where S_n denotes source signal, $H_{m,n}$ represents the transfer function from the source n to the microphone m , $H_{m,n}^d$, $H_{m,n}^e$ and $H_{m,n}^l$ are the direct, early reverberation and late reverberation components of $H_{m,n}$ respectively.

According to [15], the detrimental perceptual effects of the reflected sound are mainly associated with the late reverberation. However, early reverberation aids in speech intelligibility improvement and it is known as the early speech component when combined with the direct sound. Therefore, (4) can be written as

$$r(k) = s(k) + z(k) \quad (5)$$

where $r(k) = Y_m$, $s(k) = H_{m,n}^d S_n + H_{m,n}^e S_n$ is the wanted signal received by the microphone, $z(k) = H_{m,n}^l S_n + d_m$ is the interfering signal received by the microphone and k is the discrete time index. The interfering signal is suppressed by the adopted speech enhancement technique in order to improve the signal quality for hearing aids.

A. Speech Enhancement

It has been shown in the literature that, noise suppression techniques are essential for effective operation of communication systems, because, the presence of noise often result in erroneous and unreliable systems [8]. Therefore, it is important to employ a noise suppression techniques that can maintain the required sound quality [16]. Speech enhancement techniques such as Kalman filter and Wiener filter have been presented in the literature and their effectiveness have been demonstrated [6]. In [17], Boll proposed spectral subtraction (SS) technique for suppressing effect of noise that is acoustically added to the speech signal. The approach is popular mainly because of its simplicity and flexibility in concept as well as the effectiveness in enhancing speech

degraded by additive noise [18]. This technique is viable under the assumption that noise signal is uncorrelated and remains relatively constant prior to and during voice activity. Implementation of SS requires the estimation of the spectral magnitude of the received noisy signal and the noise spectrum for regions that are considered as “noise-only” using voice activity detection (VAD). Then, the magnitude spectrum of noise is subtracted from that of the noisy signal in order to achieve an enhanced signal [7]. However, implementation of SS leads to musical noise through linear subtraction of noise across the entire speech spectrum [19], [20].

In other to address the associated anomaly of linear subtraction, recent researches focus on nonlinear subtraction process which is justified by the variation of signal-to-noise ratio across the speech spectrum. Moreover, the nonlinear method shows that, noise signal does not affect the speech signal uniformly over the whole spectrum because certain frequencies are affected more adversely than others [21], [22]. To prevent the variation of signal-to-noise ratio across the enhanced speech spectrum as well as destructive subtraction of speech while removing most of the residual noise, it is necessary to develop an appropriate scheme that will subtract only the necessary amount of noise spectrum from each frequency bin [7], [8]. In [12], several noise-estimation algorithms are discussed and a noise-estimation algorithm for highly non-stationary noisy environments is proposed. In this paper, a multi-band approach which is based on SS technique is implemented using the noise-estimation algorithm proposed in [12] instead of the VAD employed in our previous work due to its effectiveness.

B. Multi-Band Spectral Subtraction (MBSS)

The multi-band spectral subtraction (MBSS) scheme divides the spectrum into frequency sub-band based on the nonlinear multiband frame. For each sub-band, the updated average noisy speech power in the past and present time frames are compared to the statistics of the noise power in order to improve the noise-estimation algorithm in the sub-band. Suppose a wanted signal $s(k)$ is corrupted by an additive noise $z(k)$, the received signal $r(k)$ is the sum of $s(k)$ and $z(k)$. The noisy speech signal model in the time domain for the discrete time index k is expressed as [8]:

$$r(k) = s(k) + z(k) \quad (6)$$

It should be noted that (6) is in line with the expression obtained in (5) for hearing aid signal model. Employing a frequency variable ω , the implementation of SS depends on the estimated noise spectrum, $Z(\omega)$ which is subtracted from the received signal spectrum, $R(\omega)$. The estimated output spectrum, $\hat{S}(\omega)$ can be transformed by the inverse fast Fourier transform (IFFT) to $\hat{s}(k)$. The Fourier transforms of (6) in frequency domain is:

$$R(\omega) = S(\omega) + Z(\omega) \quad (7)$$

where $R(\omega)$, $S(\omega)$ and $Z(\omega)$ are the Fourier transforms of the noisy signal, the wanted signal and the noise signal respectively.

The received signal $r(k)$ is buffered and divided into segments that consist of N samples length. Then, each segment is Hamming windowed and transformed through discrete Fourier transform (DFT) to N spectral samples. Windowing has the advantage of alleviating the effects of discontinuities at the endpoints of each segment and also suppresses glitches; therefore, it avoids the broadening of the frequency spectrum caused by the glitches [7], [8]. In frequency domain, the windowing operation can be represented by:

$$R_w(\omega) = S_w(\omega) + Z_w(\omega) \quad (8)$$

Therefore, expression describing SS is expressed as:

$$|\hat{S}(\omega)|^p = |R(\omega)|^p - \alpha |Z(\omega)|^p \quad (9)$$

where $|\hat{S}(\omega)|^p$ is an estimate of the original signal spectrum, $|Z(\omega)|^p$ is the time-averaged noise spectra and p is the power exponent which for the magnitude SS, $p = 1$, while for power SS, $p = 2$. The parameter α is for controlling the amount of noise subtracted from the noisy signal. For full noise subtraction, $\alpha = 1$ and for over-subtraction $\alpha > 1$. A novel noise element suppression (NES), α_k is proposed in [8] which can adaptively control the level of noise to be subtracted from each k^{th} sub-band. The magnitude SS of the estimated signal is given as:

$$|\hat{S}(\omega)| = |R(\omega)| - \alpha |Z(\omega)| \quad (10)$$

A multi-band approach to SS is proposed since noise has non-uniform spectral distribution and its equivalent band consists of K parallel subbands [8]. Moreover, the signal spectrum is divided into K non-overlapping sub-band and the segmentation makes it easy to performed SS independently across multiple frequency sub-band. Consequently, the estimate of the signal spectrum in the k^{th} channel is obtained by:

$$|\hat{S}_k(\omega)| = |R_k(\omega)| - \alpha_k |Z_k(\omega)| \quad 0 \leq k \leq K - 1 \quad (11)$$

Evaluating the expectation of (11), implies [8]:

$$E[|\hat{S}_k(\omega)|] = E[|S_k(\omega)|] + E[|Z_k(\omega)|] - \alpha_k E[|Z_k(\omega)|] \quad (12)$$

$$\approx E[|S_k(\omega)|] \quad (13)$$

For signal restoration, the magnitude estimate of the required signal, $|\hat{S}_k(\omega)|$ is combined with the phase of the noisy signal and then inverse Fourier transformed. The resulting signal is overlapped added to reconstruct the enhanced

output sequence. Also, to further improve intelligibility, signal to residual spectrum ratio scheme is employed.

C. Signal to Residual Spectrum Ratio (SR)

It has been shown in [23] that, speech enhancement algorithms implemented in hearing aids improve listening comfort but they are not optimal for intelligibility improvement. Besides, it is shown in [23] that, when the amplification distortion is controlled, the intelligibility improves significantly. This parameter is employed to estimate the amount of attenuation and amplification distortions in the enhanced speech to obtain more effective speech enhancement algorithm with better performance in terms of quality and intelligibility. The *SR* is expressed as [23]:

$$SR(\omega) = \frac{SNR(\omega)}{[\sqrt{SNR(\omega)} - \sqrt{SNR_e(\omega)}]^2} \quad (14)$$

where $SNR(\omega) \triangleq S^2(\omega)/Z^2(\omega)$ is the true instantaneous SNR at frequency bin ω , $SNR_e(\omega) \triangleq \hat{S}^2(\omega)/Z^2(\omega)$ is the enhanced SNR and $S(\omega)$, $Z(\omega)$, and $\hat{S}(\omega)$ are wanted signal magnitude spectrum, the noise signal magnitude spectrum, and the estimated magnitude spectrum of the wanted signal respectively.

IV. EXPERIMENTAL RESULTS AND ANALYSIS

In this work, the sampling frequency of speech signal obtained from the BBC global news podcast is set to 8 KHz and different samples of environmental noise signal are added to the speech signal to produce the noisy signal. The resultant noisy signal is then windowed using a 20 ms (160 samples) window and 50% overlap between frames. The magnitude spectrum of the windowed signal is then estimated using a 256-point fast Fourier transform (FFT). The obtained noisy signal spectrum is divided into K sub-bands, and updated average value of the segmental signal-to-noise ratio (SEGSNR) is calculated over each preceding and succeeding $k - th$ sub-band. Furthermore, spectral subtraction is implemented independently across multiple sub-band by subtracting the average value of the estimated magnitude of the noise spectrum in each $k - th$ sub-band from the noisy signal spectrum. Moreover, the estimate of the enhanced output signal is obtained by the inverse fast Fourier transform (IFFT) of the enhanced spectrum using the phase of the original noisy signal spectrum. The resulting signal is overlap added to reconstitute the enhanced output signal and experimental results are analyzed.

The analysis in this work is divided into two parts. In the first part, to compare performances of the conventional SS and the MBSS speech enhancement-based hearing aid systems on the real-time signal employed, objective test is performed while in the second part, subjective test is implemented. Then, the respective enhanced speech signal for each part is saved for further analysis.

A. Objective Test

Signal enhancement algorithms are implemented on the noisy signal using conventional SS and the proposed technique. For the conventional SS approach, the corresponding plots of

the clean signal (green), the noise signal (red), the noisy signal (blue) and the enhanced signal (magenta) are shown in Fig. (2a), (2b), (2c) and (2d) respectively while the corresponding spectrograms are shown in Fig. (3a), (3b), (3c) and (3d) to analyze frequency and level properties of the signals. With reference to Fig. (2d) and (3d), the results of signal processing based on the conventional SS show some residual noise in the enhanced signal which give rise to signal distortion and eventually listening discomfort for hearing aid users.

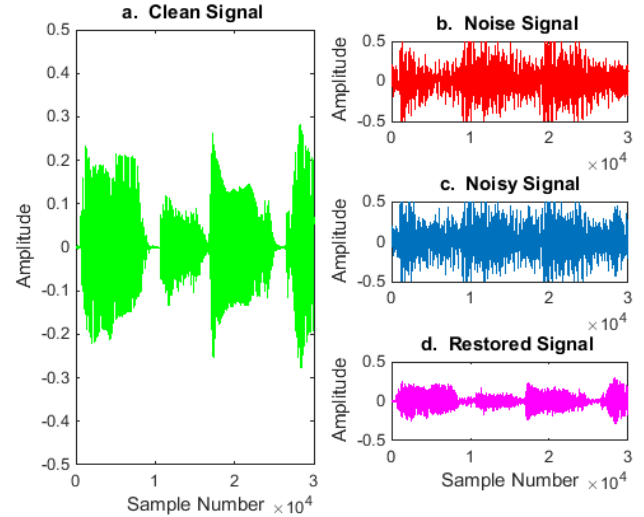


Fig. 2. (a) clean signal, (b) noise signal (c) noisy signal and (d) restored signal.

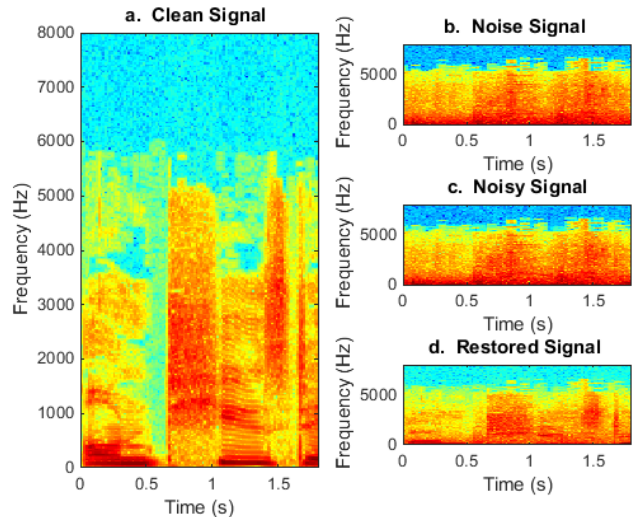


Fig. 3. Spectrogram of (a) clean signal, (b) noise signal (c) noisy signal and (d) restored signal.

Also, using the same noisy signal, the proposed multi-band SS technique which is based on amplification distortion control and new noise-estimation is implemented. With this approach,

results show that there are no notable anomalies as experienced in the conventional SS. The plot of clean signal (green), the noise signal (red), the noisy signal (blue) and the enhanced signal (magenta) using MBSS are shown in Fig. (4a), (4b), (4c) and (4d) respectively while the corresponding spectrograms are shown in Fig. (5a), (5b), (5c) and (5d). The efficiency of the proposed approach is also confirmed by visual inspection of the plot and spectrogram of the enhanced speech shown in Fig. (4d) and Fig. (5d) respectively. Furthermore, segmental SNR and the log-likelihood ratio (LLR) measurements are employed to confirm the performance of the proposed approach.

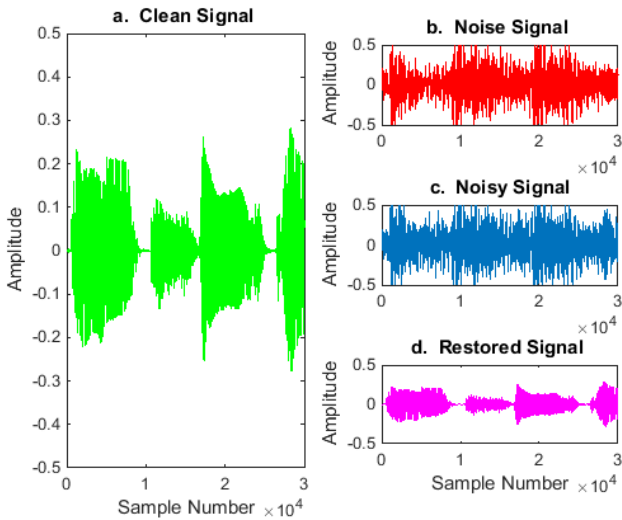


Fig. 4. (a) clean signal, (b) noise signal (c) noisy signal and (d) restored signal.

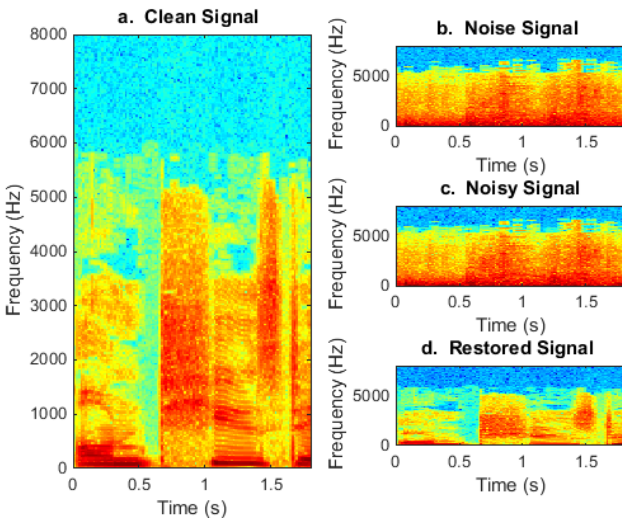


Fig. 5. Spectrogram of (a) clean signal, (b) noise signal (c) noisy signal and (d) restored signal.

B. Subjective Test

It has been observed in [7] and [8] that, digital speech processing specialists and radio broadcast professionals do not rely on simple mathematical error criterion to confirm the efficiency of enhancement algorithms. This leads to further confirmation of the effectiveness of the proposed scheme by the listening test which is one of the recommended methods by IEEE Subcommittee on Subjective Methods and International Telecommunication Union (ITU) using Mean Opinion Score (MOS) method [6]. To achieve this, the podcast and different environmental noises such as street, car, train and playground are analyzed under different SNR levels. Then, conventional SS and the MBSS speech enhancement algorithms are implemented. Then, 10 participants in the MOS rated the quality and intelligibility of the speech signals obtained by both approaches subjectively by listening to the enhanced speech obtained through a stereo headphone at appropriate volume level. The participants consists of four radio broadcast professionals who have about 10-year experience in both analogue and digital speech processing and are in their late thirties and six students working in digital speech processing area and in their twenties. The results obtained are illustrated in Table 1. The results show that the proposed method outperforms the conventional approach.

TABLE I. MEAN OPINION SCORE RESULTS

Noise Environment	SNR Level (dB)			Algorithm
	-5	0	5	
Car	1.8	2.6	3.9	Conventional Spectral Subtraction
Train	1.5	2.3	3.5	
Playground	1.3	2.1	3.2	
Street	1.1	1.6	3.0	
Car	2.3	3.8	4.7	Multi-Band Spectral Subtraction
Train	2.1	3.5	4.5	
Playground	1.7	3.2	4.2	
Street	1.5	3.1	4.0	

V. CONCLUSION

This paper presents speech enhancement algorithm with an improved intelligibility for digital hearing aid systems. The employed algorithm prevents the associated musical noise that is generated by the conventional spectral subtraction algorithm and offers comparatively higher signal-to-noise ratio. This is attributed to the multi-band scheme, the better noise-estimation algorithm and the amplification distortion control for speech intelligibility improvement being implemented. Also, the results show that the proposed scheme gives comparatively better performance for all SNRs observed with no adverse effect on the processed signal and can be easily employed in digital hearing aid system for improving speech quality and intelligibility in the noisy environment.

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