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A robust adaptive hybrid feedback cancellation scheme for hearing aids in the presence of outliers

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ABSTRACT

It has been demonstrated that the Filtered-x Wilcoxon LMS (FxWLMS) based adaptive filter mitigates the effect of the outliers acquired by the microphone signal of hearing aids by minimizing the Wilcoxon norm and hence shows better cancellation performance than the existing Filtered-x LMS (FxLMS) algorithm. The prediction error method based adaptive feedback canceller (PEMAFC) reduces the bias present in the estimate of the feedback path due to the continuous adaptive filtering (CAF). However, the impulse response of the measured feedback path is close to zero for the first many samples due to the delay introduced by ADC converters and then contains few significant values, which results in slow convergence rate when an adaptive filter is used to model the same. To overcome this limitation, we propose a proportionate normalized WLMS (PNWLMS) algorithm based PEMAFC (P-PNWLMS) for feedback cancellation in hearing aid in the presence of outliers. Further, with an objective to improve the convergence rate and performance accuracy simultaneously, this paper proposes a novel convex PNWLMS (CPNWLMS) algorithm which incorporates convex combination of PNWLMS and WLMS algorithms. The weight update equations are derived for PEMAFC trained by PNWLMS (P-PNWLMS) and CPNWLMS (P-CPNWLMS) algorithms respectively. The results of the simulation study show improved performance of the proposed CPNWLMS based adaptive filter over its component filters.

1. Introduction

Hearing aids are extensively used devices by hearing impaired people which assist in listening for better sound perception [1,2]. Hearing aids help to mitigate the effect of the increased hearing threshold of an impaired person by amplifying the desired incoming signal. It helps in improving speech intelligibility, listening comfort and quality of life of the user. However, acoustic feedback is one of the major drawbacks of this device. Acoustic feedback occurs when a portion of the amplified sound from the speaker is again picked up by the closely located microphone. The presence of vent, leakage from the tubes and structural transmissions from the shell are the major factors responsible for acoustic feedback in hearing aids. This acoustic feedback limits the maximum gain and hence degrades the quality of perceived speech. As a result the hearing aid user become dissatisfied with the device.

The overall impulse response of the feedback path in hearing aid includes the characteristics of the digital to analog converter (D/A), speaker, acoustic path between the speaker and the microphone and analog to digital converter(A/D). In practice, the characteristics of the feedback path is subjected to variations because of the changes in the nearby environment or jaw movement of the user [3]. The most effective method of suppressing the effects of acoustic feedback employs an adaptive filter to produce an estimate of the feedback signal which is then subtracted from the microphone signal.

Fig. 1 shows the basic block diagram of a hearing aid with feedback cancellation scheme where $x(n)$ denotes the input signal to the hearing aid and $y(n)$ represents the final processed sound delivered to the users. A portion of the output sound is again picked up by the microphone giving rise to feedback phenomenon. Therefore, the microphone signal $v(n)$ contains both the desired input $x(n)$ and the feedback signal $u(n)$. $D(z)$ represents the signal processing in the forward path of the hearing aid. The adaptive filter $\hat{W}(z)$ generates an estimate of the feedback signal which is then subtracted from the microphone signal to produce error signal.

The adaptive feedback cancellers (AFC) used to suppress the effect of acoustic feedback are broadly divided into non continuous [4,5] and continuous mode of adaptation [5]. The continuous adaptive filtering (CAF) produces a biased estimate of the feedback path because of the presence of forward path which makes the

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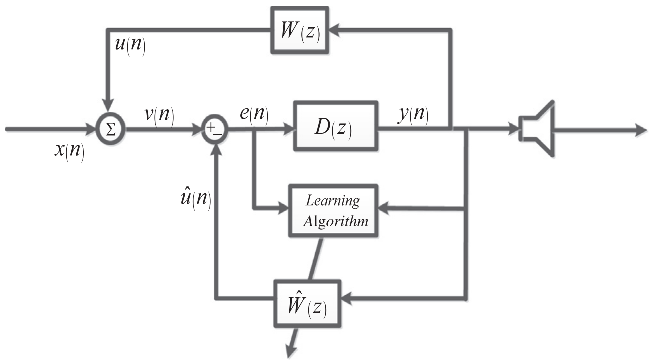

Fig. 1. Basic block diagram of hearing aid with feedback cancellation scheme.

whole system as closed loop. The forward path introduces correlation between the incoming desired signal and the output of the speaker when the desired signal is spectrally colored. This causes bias in the estimate of the feedback path which leads to cancellation of both the desired and the feedback signal. Many solutions have been proposed to tackle the problem of biased feedback path estimate such as using delays or nonlinearities in the forward path or cancellation path [6], constrained [7], band limited adaptation [8], Filtered-XLMS(FXLMS) [9], prediction error method (PEM) based approach [10,11], two stage methods for unbiased feedback cancellation [12], PEM with row operations (PEM-AFROW) [13], transient mean square analysis of PEMAFCs [14], clipping used in the feedback path by [15], physical performance based evaluation [16], sound quality measurement based on subjective measure [17], AFC based on projection sub gradient [18]. Further other methods like using all pass filters having time varying poles [19], a band limited linear predictive coding based approach [20], PEM based AFC with two signal model [21], insertion of specifically designed probe noise signals [22,23], improved prediction error filter [24], pitch and formant estimation based method [25], using two microphone based approach [26] have been developed for reducing the bias while estimating the feedback path.

Usually, all these methods attempt to reduce the bias in the estimate of the feedback path. However, none of these methods have considered the presence of outliers and their impact on the feedback cancellation performance. The outliers are high amplitude erroneous data which are distinct from the rest of the data. The outliers are introduced in the microphone signal due to the temporary or permanent breakdown of microphone and analog to digital converters [27]. These outliers not only cause a sudden rise in the microphone signal but also severely affect the convergence rate of the FxLMS algorithm. In addition, it may also lead to the divergence of the coefficients of the adaptive filter which significantly degrades its feedback cancellation accuracy. This results in deteriorating the sound quality which may be quite annoying for the user. Hence, the conventional learning algorithms such as LMS [28], FxLMS [29] which is based on minimization of the mean square error fail to perform satisfactorily in the presence of outliers. However, the hearing aid system should be able to work efficiently in diverse environments and its performance should not get affected by the presence of outliers. Therefore, the presence of such disturbances necessitates the development of an adaptive filter which is robust to the presence of outliers and would not let the outliers deteriorate its convergence and feedback cancellation performance.

Adaptive filter model based on minimizing the Wilcoxon norm has been successfully employed for various machine learning algorithms [30] and system identification task [28–31]. In a recent communication the Filtered-x Wilcoxon LMS (FxWLMS) has been introduced for mitigating the effects of acoustic feedback in the presence of outliers in hearing aids [32]. It has been also reported in [32] that the FxWLMS algorithm performs better than the conventional FxLMS algorithm in the presence of outliers while modelling the feedback path in hearing aids. There is an automatic gain control (AGC) mechanism used in the forward path of the hearing aid to adjust the dynamic range of the signal by controlling the gain. However, the AGC mechanism will only adjust the gain of the error signal (which contains both the desired signal and the residual feedback signal) irrespective of the presence of residual feedback component in it which is unwanted for the user. It does not take into account the negative impact of the presence of residual feedback signal due to the inefficient feedback cancellation by the adaptive filter in the presence of outliers. By adjusting the gain without properly nullifying the effect of feedback results in speaker signal containing residual feedback component in addition to desired signal. This deteriorates the speech intelligibility and sound quality perceived by the user. Ideally the speaker should provide only processed desired input signal without any residual feedback component in it. Therefore an adaptive controller based on minimization of robust Wilcoxon norm is used for effectively negating the feedback despite the presence of outliers [32].

However, the presence of few significant coefficient values and rest of the coefficient values close to zero in the impulse response of the feedback path results in slower convergence of the adaptive filter used for modelling the same. The proportionate normalized LMS (PNLMS) results in faster convergence than the normalized LMS (NLMS) in case of modelling the long length sparse impulse response as of echo cancellers [33,34]. In order to achieve fast convergence while modelling the feedback path in the presence of outliers in the microphone signal, this paper proposes a novel proportionate normalized WLMS (PNWLMS) algorithm based PEMAFC (P-PNWLMS).

The performance of any adaptive filter is measured in terms of convergence speed and steady state behavior. The algorithms which are good at yielding less steady state error, usually suffers from poor convergence whereas faster converging algorithm results in large steady state error. Therefore, appropriate combination of adaptive filters with conflicting performances (fast convergence and less steady state error) have been developed in [35,36] to enhance the overall performance of the adaptive filters. The convex combination strategy of adaptive filters has been reported to perform better than the individual component filters [37]. This method has been widely used in various applications like system identification [38,39], ANC [40–42], echo cancellation [43]. Other method using the concept of variable step size (SSV) has also been suggested in the literature to achieve faster convergence without compromising the steady state behavior [44]. However, such methods require some prior knowledge of the statistics of the incoming signal like SNR to properly adjust the filtering parameters [45]. In real life applications like hearing aid, it is difficult to estimate a priori the parameters of the incoming desired signal. In contrast to the SSV, this convex combination strategy offers the advantage of combining adaptive filters with different learning algorithms as well as of different filter lengths [40]. The hearing aid being a real time, battery operated, power and area constrained system requires algorithms which should be good at achieving conflicting objectives of fast convergence and better feedback cancellation performance simultaneously. Therefore, convex combination of adaptive filters is thought to be a potential solution to achieve two diverging goals in hearing aids. Keeping this in view, the present paper introduces a novel convex PNWLMS (CPNWLMS) algorithm based PEMAFC (P-CPNWLMS) for effective feedback cancellation in hearing aids in the presence of outliers, utilizing the concept of convex combination of WLMS and the proposed PNWLMS algorithms.