Feedback Cancellation in Speech Signal using LMS Algorithm

Dr. Bassam Lala[*]

Abstract

Acoustic feedback is a well-known problem in reinforcement systems, which is caused by the undesired acoustic coupling between the loudspeaker and the microphone. Acoustic feedback limits the maximum amplification used in the reinforcement systems without making it unstable. Feed forward suppression techniques are widely used in this context to alternate the forward path in the reinforcement systems and reduce the feedback signal. However, they limit the amplification in the forward path and can lead to severe sound quality distortions in loudspeaker signal.

This paper focuses on the adoption of the Least Mean Square (LMS) Algorithm of adaptive filtering as a solution to reduce this unwanted signal in speech signal and thus increase speech quality. LMS algorithm of adaptive filtering in the cancellation system is then validated by computing the filter coefficients using discrete signal processing in MATLAB. Dr. Bassam Lala[*]

Keywords: Speech Signal, Digital Filters

[*]**Biomedical Engineering, Damascus University.**
[*]**AIU, Arab International University.**

إلغاء الصفير في إشارة الكلام باستخدام خوارزمية LMS

د. بسام لالا*

الملخص

تعد الإشارة الصوتية الراجعة مشكلة شائعة في المعينات السمعية والتي تنتج عن ازدواج صوتي غير مرغوب به بين مكبر وميكرفون المعينة السمعية. تحد الإشارة الصوتية الراجعة من التضخيم الأعظمي المستخدم في المعينات السمعية دون أن تجعلها غير مستقرة.

تستخدم تقنيات تخميد التغذية الأمامية بشكل واسع في هذا السياق لتناوب الطرق الأمامية في أنظمة المعينات السمعية وإنقاص الإشارة الراجعة. إلا أن ذلك يحد من تضخيم الطرق الأمامية ويؤدي إلى تشوه كبير في نوعية إشارة ميكرفون المعينة السمعية.

يركز هذا البحث على خوارزمية LMS المتبناة في المرشح الموائم كحل لإنقاص الإشارة الغير مرغوب فيها في إشارة الكلام مما يؤدي إلى زيادة نوعية الكلام.

تقوم خوارزمية LMS للمرشح الموائم في نظام الحذف بحساب معاملات المرشح باستخدام معالجة الإشارة المتقطعة بواسطة برنامج ماتلاب.

الكلمات المفتاحية: إشارة الكلام، المرشحات الرقمية.

* قسم الهندسة الطبية، جامعة دمشق.

* الجامعة العربية الدولية الخاصة (AIU).

1. Introduction:

A major complaint of reinforcement system users is acoustic feedback, which is perceived as whistling or howling. This feedback occurs, typically at high gains, because of leakage from the receiver to the microphone. Acoustic feedback suppression in hearing aids is important since it can increase the maximum insertion gain of the aid. The ability to achieve target insertion gain leads to better utilization of the speech bandwidth and, hence, improved speech intelligibility for the hearing-aid user [1-6].

In order to minimize the effects of acoustic feedback, a vast range of techniques for feedback control have been developed in the past [7-13].

The feed forward suppression techniques are proposed as in Fig.1.

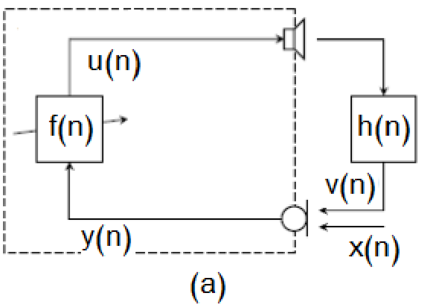

Fig. 1 the feed forward suppression technique

These techniques modify the forward path f(n) from the microphone signal y(n) to the loudspeaker signal u(n) for suppressing the feedback effect. The feed forward suppression techniques can be further divided into two categories: gain reduction and phase modification methods [2,3,14,15]. The feed forward suppression techniques have significant limitations. The gain reduction techniques limit the amplification in the forward path f (n), which is contradicting the main purpose of audio reinforcement systems including hearing aids [3,14]. Phase modification techniques can lead to severe sound quality distortions in loudspeaker signals u(n)[15]. In addition, the acoustic path transfer function can vary significantly depending on the acoustic environment. Hence, effective acoustic feedback cancellers must be adaptive.

Therefore, this paper focuses on the use of adaptive filtering technique to reduce the feedback signal v (n), thus increasing speech quality. In Section 2, we present a description of proposed system model of the adaptive filter. Section 3 focuses on the LMS algorithm and its simulation in Matlab. Section 4 presents the simulation results obtained from the adaptive feedback cancellation algorithm and the discussion of these results is presented in the Section 5.

2. The proposed Adaptive Feedback Cancellation (AFC) system:

Figure 2 shows a block diagram of the adaptive feedback cancellation system in a simple reinforcement system,which consists of a single-microphone and single-loudspeaker.

Here the filter H (n) represents the impulse response of the feedback path between the loudspeaker and the microphone, where: u (n) is the input signal, h (n) is the feedback signal, W (n) represents the adaptive filter used to cancel the feedback signal. The forward path G (ω, n) maps the microphone signal d (n), possibly after AFC, to the loudspeaker signal s (n).

The microphone signal is given by:

$$d(n)=u(n)+h(n) = u(n)+H(\omega, n)\,.s(n) \quad (1)$$

The concept of the AFC is to place an estimated finite impulse response (FIR) adaptive filter (W) in parallel with the feedback path, having the loudspeaker signal as an input and microphone signal as the desired output. The feedback canceller (w) produces an estimation of the feedback signal h (n) which is then subtracted from the microphone signal d (n).

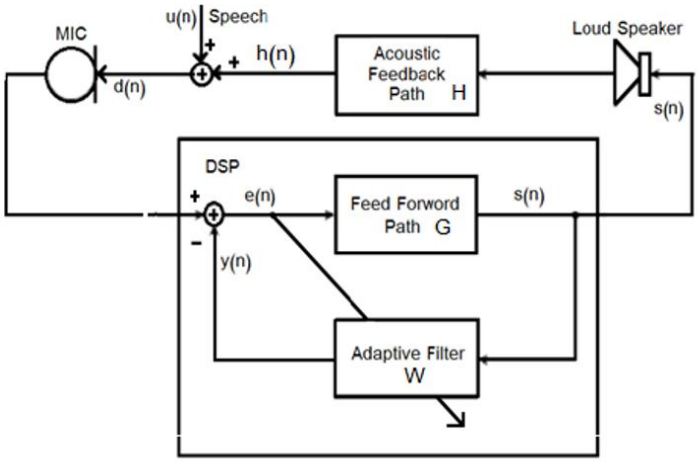

Fig (2) Adaptive feedback cancellation (AFC).

The adaptive filter aims to equate its output y (n) to the desired output d (n). At each iteration the error signal, e (n) = d (n) - y (n), is fed back into the filter, where the filter characteristics are altered accordingly.

In the case of acoustic feedback cancellation, the optimal output of the adaptive filter is equal in value to the unwanted signal. When the adaptive filter output is equal to desired signal the error signal goes to zero. In this situation the feedback signal would be completely cancelled.

3. Least Mean Square (LMS) Algorithm

The Least Mean Square Algorithm is a linear adaptive filtering algorithm that consists of two basic processes:

1. A filtering process, which involves:

a) Computation of the output of a transverse filter produced by a set of tap inputs, and

b) Generating an estimation error by comparing this output to a desired response.

2. An adaptive process, which involves the automatic adjustment of the tap weights of the filter in accordance with the estimation error.

LMS Algorithm is well known and widely used due to its computational simplicity. With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula:

$$w(n+1) = w(n) + 2\mu e(n)u(n) \qquad (2)$$

Here u (n) is the input vector of time delayed input values:

$u(n) = [u(n), u(n-1), u(n-2) \ldots u(n-N+1)]^T$.

The vector $w(n) = [w_0(n)\ w_1(n)\ w_2(n) \ldots\ldots w_{N-1}(n)]^T$ represents the coefficients of the adaptive FIR filter tap weight vector at time n. The parameter (μ) is known as the step size parameter and is a small positive constant. This step size parameter controls the influence of the updating factor. Selection of a suitable value for (μ) is imperative to the performance of the LMS algorithm, if the value of the (μ) is too small, the adaptive filter's time will be too long; if (μ) is too large the adaptive filter becomes unstable and its output diverges [5,6,16].

Each iteration of the LMS algorithm requires three distinct steps in this order:

1. The output of the FIR filter, y (n) is calculated using equation 3.

$$y(n) = w^T(n)u(n) \qquad (3)$$

2. The value of the error estimation is calculated using equation 4.

$$e(n) = d(n) - y(n) \qquad (4)$$

3. The tap weights of the FIR vector are updated in preparation for the next iteration, by equation 5.

$$w(n+1) = w(n) + 2\mu e(n)u(n) \quad (5)$$

The main reason for the LMS algorithms popularity in adaptive filtering is its computational simplicity, making it easier to implement than all other commonly used adaptive algorithms. For each iteration the LMS algorithm requires 2N additions and 2N+1 multiplications (N for calculating the output, y(n), one for 2μe(n) and an additional N for the scalar by vector multiplication)[5,6].

4. Simulation and Results of LMS Adaptive Filtering Algorithm:

The LMS algorithm was simulated using Matlab. The input signal is a speech signal (i.e., an English-speaking male voice) sampled at 16 kHz because the speech bandwidth is up to 8 KHz. The feedback signal was generated by FIR filter then convolving this with a speech input wav file in Matlab. The LMS adaptive filtering algorithm outlined in Section (3) was implemented using Matlab with the step size of 0.0005 to reduce the adaptive filter's time. We proposed that the adaptive filter and the real feedback path are both FIR and of the same order to be insured that the adaptive filter becomes stable.

The results are shown below:

Figure 3 shows the input signal (an English-speaking male voice sampled at 16 kHz) in the time domain.

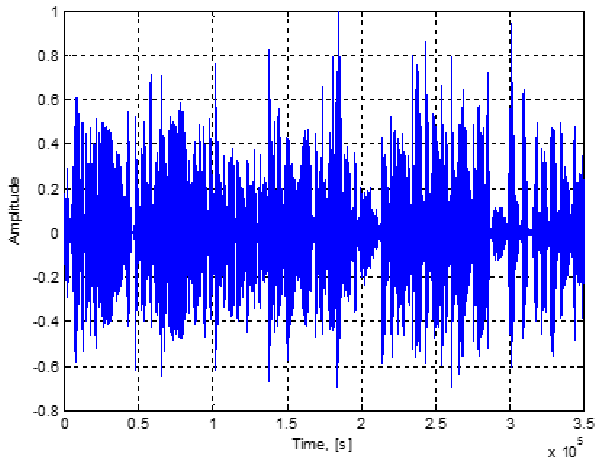

Figure 3 Input signal

Figure 4 shows the estimated error, output filter and mixed signals for using adaptive LMS filter in the time domain.

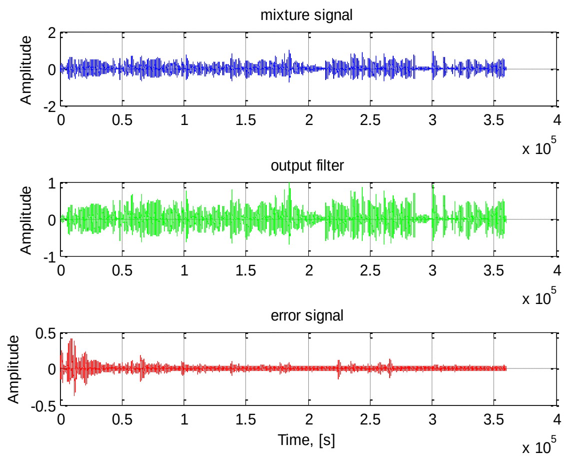

Figure 4. LMS algorithm outputs for speech signal input, μ=0.005.

Figure 5 shows the measured impulse response of feedback signal.

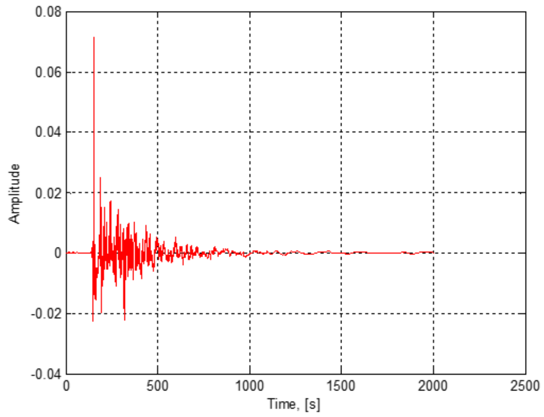

Figure 5 impulse response of feedback signal

Figure 6 shows Comparison between real feedback path and estimated feedback path using simple LMS algorithm.

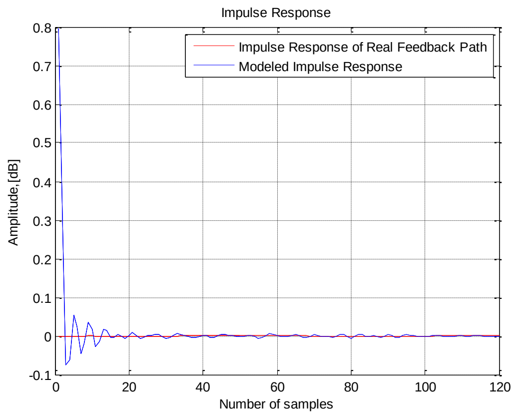

Figure 6. Comparision between real feedback path and modeled feedback path using simple LMS algorithm

5. Discussion:

The LMS algorithm proved to be effective for feedback cancellation because the error signal is very small between output filter and desired signal as it appears in Figure 4. However, a biased estimation of the feedback path was noticed as in Figure 5. It seems that there is a mismatch between the real and estimated feedback path. The mismatch can be justified if we follow the signal flow in the simulation carefully. When a speech signal u (n) is fed into the reinforcement system, it appears in the error signal e (n).The speech signal is then amplified in the reinforcement system. As a result, an amplified and delayed version of the speech signal appears in the reinforcement system output and the input to the adaptive filters s (n). Because e (n) and s (n) are both speech signal at the same frequency, they will be highly correlated, and the correlation between the two signals will be much greater than any signal correlations related to the feedback path transfer function. Because the adaptive filter coefficient update is driven to minimize the error signal, the filter will adapt by shifting the amplitude and phase of s (n). The filtered signal y (n) cancels the speech signal at the microphone output d (n).

The result is that for a speech input signal, adaptive feedback cancellation stops modeling the feedback path and instead adapts to cancel the input signal. When the microphone signal is heavily distorted, the cancelling effect will be smaller since the cross-correlation is diminished and as a result, the microphone signal increases again.

The challenge is therefore to reduce the correlation between the input signal and the loudspeaker signal.

6. Conclusion:

In this paper, the performance of the LMS algorithm was studied for feedback cancellation of speech signals.

The analysis leads us to conclude that LMS can perform well for feedback cancellation of speech signals if a small step size is chosen. Otherwise, LMS adaptive filtering algorithm will show a biased solution because of the correlation between the input signal and the loudspeaker signal.

This bias severely limits the performance of the feedback cancellation system especially when tonal signal or music signals are fed into the reinforcement systems.

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