

Stepsize Control for Acoustic Feedback Cancellation Based on the Detection of Reverberant Signal Periods and the Estimated System Distance

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Abstract

A new approach for acoustic feedback cancellation is presented. The challenge in acoustic feedback cancellation is a strong correlation between the local speech and the loudspeaker signal. Due to this correlation, the convergence rate of adaptive algorithms is limited. Therefore, a novel stepsize control of the adaptive filter is presented. The stepsize control exploits reverberant signal periods to update the adaptive filter. As soon as local speech stops, the reverberation energy of the system decays exponentially. This means that during reverberation there is only excitation of the filter but no local speech. Thus, signals are not correlated and the filter can converge without correlation problems. Consequently, the stepsize control accelerates the adaption process during reverberation and slows it down at the beginning of speech activity. It is shown, that with a particular gain control, the reverb-based stepsize control can be interpreted as the theoretical optimum stepsize. However, for this purpose a precise estimation of the system distance is required. One estimation method is presented. The proposed estimator has a rescue mechanism to detect enclosure dislocations. Both, simulations and real world testing show that the acoustic feedback canceler is capable of improving stability and convergence rate, even at high system gains.

Index Terms: acoustic feedback cancellation, in-car communication, stepsize control

1. Introduction

Due to large background noise levels, especially at high velocities, communication between driver and rear seat passengers inside a car can be difficult. An in-car communication (ICC) system improves communication by amplifying the speech of the front seat passengers. Therefore, speech is captured by means of microphones and played back via loudspeakers close to the rear seat passengers. Since the microphones also receive the loudspeaker signal, this results in a closed electro-acoustic loop. If $H_{icc}(f)$ is the transfer function of the ICC system and $H(f)$ describes the acoustic coupling between loudspeaker and microphone, the resulting frequency response of the closed-loop system is

$$H_{res}(f) = \frac{X(f)}{S(f)} = \frac{H_{icc}(f)}{1 - H_{icc}(f) \cdot H(f)}, \quad (1)$$

where $X(f)$ is the loudspeaker signal, $S(f)$ is the local speech and f denotes the continuous frequency. To ensure that the system remains stable, the open loop gain must not exceed unity

$$|H_{icc}(f) \cdot H(f)| < 1, \quad (2)$$

as described in [1]. Thus, the system gain, included in $H_{icc}(f)$, must be adjusted in such a way that Eq. (2) is not violated, since

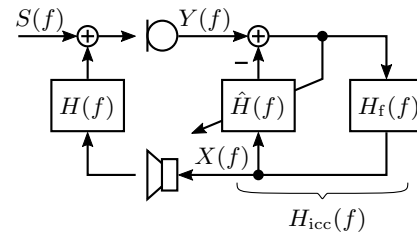


Figure 1: Block diagram of a closed electro-acoustic loop with an adaptive feedback canceler.

otherwise howling occurs. The largest possible gain that does not violate Eq. (2) is called maximum stable gain (MSG).

The aim of an adaptive feedback canceler is improving stability at high system gains. Therefore, the acoustic coupling between loudspeaker and microphone is estimated by means of an adaptive filter. The estimated feedback is then subtracted from the microphone signal, as shown in Fig. 1. The ICC system now consists of the forward path $H_f(f)$, which includes the gain, and the estimated coupling $\hat{H}(f)$. This results in

$$H_{icc}(f) = \frac{X(f)}{Y(f)} = \frac{H_f(f)}{1 + H_f(f) \cdot \hat{H}(f)}, \quad (3)$$

where $Y(f)$ denotes the microphone signal. Inserting Eq. (3) into Eq. (1) leads to the transfer function

$$H_{res}(f) = \frac{H_f(f)}{1 - H_f(f) \cdot (H(f) - \hat{H}(f))}. \quad (4)$$

If a perfect estimation can be found, i. e. $H(f) = \hat{H}(f)$, the transfer function of the system becomes $H_{res}(f) = H_f(f)$. This means that the system remains stable at arbitrary gains. However, this is not possible since the performance of adaptive algorithms, such as the normalized least mean square (NLMS), is restricted. The reason is the strong correlation between the loudspeaker signal and the local speech. Hence, advanced control mechanisms are necessary to achieve acceptable convergence of the adaptive filter.

2. Previous Work

Acoustic feedback control has been researched intensively in the domain of public address systems, hearing aids as well as ICC systems. An extensive collection of control mechanisms for these applications can be found in [2]. In this paper, the focus lies on ICC systems. Their specific challenges are for example described in [3]. Besides acoustic feedback, ICC systems have to deal with high background noise levels. Therefore, in [4, 5] a time domain adaptive filter is combined with a noise reduction to reduce both acoustic feedback and background noise.

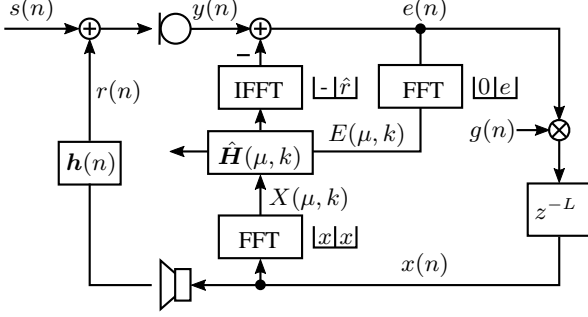


Figure 2: Block diagram of an acoustic feedback canceler implemented within an overlap-save filterbank.

However, since a standard NLMS algorithm is used for system identification, with this approach convergence and stability are limited. In contrast to that, in [6] a frequency domain adaptive filter is used for feedback cancellation. The performance of this filter can be improved by applying decorrelation methods such as frequency shift [7] or prewhitening with linear prediction [8]. Further improvement can be achieved by combining linear prediction with a variable stepsize of the adaptive filter [9, 10].

A completely different approach is presented in [11]. Here, the acoustic coupling between loudspeaker and microphone is described by an energy decay model. This model is based on the reverberation properties of the acoustic environment. The feedback is suppressed by spectral subtraction of the modeled feedback power spectral density by means of a Wiener filter.

The reverberation of the acoustic environment can also be exploited to control the stepsize of a frequency domain adaptive filter without the need of decorrelation methods [12]. There is no local speech during reverberation, and thus signals are not correlated. Hence, the stepsize can be large. It is further shown in [13] that by controlling the system gain with the inverse system distance, the reverb-based stepsize becomes an approximation of the theoretically optimum stepsize. However, the system distance is not directly accessible in a real system and must therefore be estimated. For this reason, in the present paper a system distance estimator is combined with the reverb-based stepsize control.

3. Acoustic Feedback Cancellation

The acoustic feedback canceler is realized in the discrete frequency domain, as described in [14]. The block diagram of the overall structure is shown in Fig. 2. In this work, a sampling frequency of $f_s = 32$ kHz is used for processing. An overlap-save filterbank transforms the signals to the subband domain. The FFT-length is $N = 512$ and the blocks are half overlapping, i. e. the frameshift is $L = N/2 = 256$. The discrete time index is denoted by n , the sub-sampled block index is $k = n/L$ and μ is the index for the discrete frequency bins.

The coupling between loudspeaker and microphone is described by the impulse response vector $\mathbf{h}(n)$. The impulse response may vary over time, caused for example by movements of the passengers. The microphone signal $y(n)$ is the superposition of local speech $s(n)$ and feedback $r(n)$. In the forward path, the error signal $e(n)$ is amplified by the system gain $g(n)$ and delayed by one frame (8 ms), resulting in the loudspeaker signal $x(n)$

$$x(n) = g(n) \cdot e(n - L). \quad (5)$$

The delay is caused by block processing. Vector $\mathbf{X}(k)$ contains

the discrete frequency samples at time instant k , obtained by FFT of the corresponding time domain block

$$\mathbf{X}(k) = \text{FFT} \left\{ [x(n - N + 1), \dots, x(n)]^T \right\}. \quad (6)$$

The same applies to $\mathbf{E}(k)$, with the difference that the first half of the time domain block must be set to zero to avoid errors caused by circular convolution. In the following equation, this is marked by a null-vector $\mathbf{0}_L$

$$\mathbf{E}(k) = \text{FFT} \left\{ [\mathbf{0}_L, e(n - L + 1), \dots, e(n)]^T \right\}. \quad (7)$$

The estimated subband impulse responses are given by vector

$$\hat{\mathbf{H}}(\mu, k) = [\hat{H}_0(\mu, k), \hat{H}_1(\mu, k), \dots, \hat{H}_{M-1}(\mu, k)]^T. \quad (8)$$

For the proposed ICC system $M = 8$ coefficients are necessary to cover the relevant length of a typical car's impulse response. In the same manner, the M previous samples of the loudspeaker signal are summarized in vector $\mathbf{X}(\mu, k)$

$$\mathbf{X}(\mu, k) = [X(\mu, k), \dots, X(\mu, k - M + 1)]^T. \quad (9)$$

With this notation, the update of the adaptive filter becomes

$$\hat{\mathbf{H}}(\mu, k+1) = \hat{\mathbf{H}}(\mu, k) + \alpha(\mu, k) \cdot \frac{E(\mu, k) \cdot \mathbf{X}^*(\mu, k)}{\|\mathbf{X}(\mu, k)\|^2}, \quad (10)$$

where $(\cdot)^*$ means complex conjugate and $\|\cdot\|$ is the Euclidean norm of the vector. The stepsize is denoted by $\alpha(\mu, k)$. Due to circular convolution, the right half of gradient of Eq. (10) must be constrained with zeros in the time domain [14].

One measure to evaluate the adaptive filter's convergence rate is the system distance. It can be calculated as

$$\|\mathbf{H}_\Delta(\mu, k)\|^2 = \|\mathbf{H}(\mu, k) - \hat{\mathbf{H}}(\mu, k)\|^2, \quad (11)$$

where $\mathbf{H}(\mu, k) \bullet \circ \mathbf{h}(n)$ is the vector of the real subband impulse response. The system distance is reduced, if the estimation $\hat{\mathbf{H}}(\mu, k)$ converges towards the real system $\mathbf{H}(\mu, k)$. It was shown in [15] that optimal convergence can be achieved, if Eq. (11) is minimized with respect to $\alpha(\mu, k)$. This results in the optimal stepsize

$$\alpha_{\text{opt}}(\mu, k) = \frac{E \{ |E_u(\mu, k)|^2 \}}{E \{ |E(\mu, k)|^2 \}}. \quad (12)$$

$E \{ \cdot \}$ is the expected value and $E_u(\mu, k)$ denotes the undisturbed error signal, i. e. the error in absence of local speech. It was also found in [15] that the expected value of $E_u(\mu, k)$ can be approximated as

$$E \{ |E_u(\mu, k)|^2 \} \approx E \{ |X(\mu, k)|^2 \} \cdot E \{ \|\mathbf{H}_\Delta(\mu, k)\|^2 \}, \quad (13)$$

if the assumption is made that $|X(\mu, k)|^2$ and $\|\mathbf{H}_\Delta(\mu, k)\|^2$ are not correlated. The optimal stepsize then becomes

$$\alpha_{\text{opt}}(\mu, k) \approx \frac{E \{ |X(\mu, k)|^2 \}}{E \{ |E(\mu, k)|^2 \}} \cdot E \{ \|\mathbf{H}_\Delta(\mu, k)\|^2 \}. \quad (14)$$

According to [16], the expected values can be approximated by first order IIR smoothing of the squared spectral magnitude. In the following, smoothing is denoted by an overline $\overline{(\cdot)}$

$$\alpha_{\text{opt}}(\mu, k) \approx \frac{\overline{|X(\mu, k)|^2}}{\overline{|E(\mu, k)|^2}} \cdot \overline{\|\mathbf{H}_\Delta(\mu, k)\|^2}. \quad (15)$$

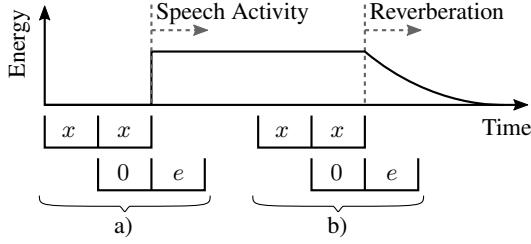


Figure 3: Detection of a) speech activity and b) reverberation by comparing the energy of the loudspeaker and error blocks.

Fig. 2 shows that in the closed-loop system the only decorrelation is the delay z^{-L} . Since this delay is in an order where speech is stationary, the assumption (13) is violated. Nevertheless, it will be shown in the next sections, how Eq. (15) can be used as stepsize control also in a closed-loop system.

3.1. Reverb-based Stepsize Control

It is known from acoustic *echo* cancellation that to achieve optimal convergence the stepsize must be increased during remote single talk and decreased during double talk [17]. In contrast to that, there are no periods of pure single talk in acoustic *feedback* cancellation. However, immediately after local speech stops, there is still energy in the system, caused by reverberation of the acoustic environment. This reverberation energy decays exponentially. Since during reverberation there is no local speech but only excitation of the loudspeaker, this is similar to remote single talk. The filter can adapt undisturbed, because signals are not correlated. Thus, the reverb-based stepsize control described in [12, 13] aims at increasing the stepsize during reverberation and decrease it at the beginning of speech activity. Therefore, two properties of the overlap-save filterbank are exploited. The first is that $x(n)$ is delayed by L samples compared to $e(n)$. Second, the first half of the error block is set to zero before it is transformed to the frequency domain. If the powers of the corresponding blocks are regarded, the power of error signal follows signal changes earlier and faster than the power of the loudspeaker signal. Based on these considerations, the reverb-detection can be derived from the following ratio [12]

$$\alpha_{\text{rev}}(\mu, k) = \frac{|X(\mu, k)|^2}{|E(\mu, k)|^2 \cdot g(k)^2}. \quad (16)$$

From Eq. (5) follows that multiplication of the error signal with $g(k)^2$ is necessary to ensure comparability with the loudspeaker signal. The principle is shown schematically in Fig. 3. In the figure two time instants are marked:

- Beginning of speech activity: The error block already contains speech, but the loudspeaker does not, i.e. $\alpha_{\text{rev}}(\mu, k) < 1$.
- Reverberation: The error signal begins to decay, while the loudspeaker still contains speech, i.e. $\alpha_{\text{rev}}(\mu, k) > 1$.

Comparing Eq. (15) and Eq. (16) shows that the reverb-based stepsize equals the optimal stepsize, if the system gain is controlled with the inverse system distance

$$g(k)^2 = \frac{1}{\|\mathbf{H}_{\Delta}(\mu, k)\|^2}. \quad (17)$$

Since the reverb-based stepsize increases the stepsize only during reverberation, meaning when signals are not correlated, the

assumption made in Eq. (14) is valid. This results in an optimal convergence during reverberation. On the other hand, during speech activity, the adaption process is slowed down, which avoids misadjustment of the filter coefficients caused by correlated signals.

3.2. Estimation of the System Distance

In a real system, the system distance is unknown and must thus be estimated. From Eq. (13) follows

$$\|\mathbf{H}_{\Delta}(\mu, k)\|^2 \approx \frac{|E_u(\mu, k)|^2}{|X(\mu, k)|^2}. \quad (18)$$

While $X(\mu, k)$ is directly accessible, $E_u(\mu, k)$ is unknown. The error signal is comprised of local speech $s(n) \circ \bullet S(\mu, k)$ and undisturbed error

$$E(\mu, k) = S(\mu, k) + E_u(\mu, k). \quad (19)$$

Therefore, during speech activity, the power of the undisturbed error is smaller than the error power, while it equals the error power during speech pauses. This can be expressed as

$$|E_u(\mu, k)|^2 \leq |E(\mu, k)|^2. \quad (20)$$

As a consequence, the expected system distance can be approximated by following the minimum of the relation

$$EXR(\mu, k) = \frac{|E(\mu, k)|^2}{|X(\mu, k)|^2}. \quad (21)$$

If minimum tracking is denoted by $f_{\min}(\cdot)$, the estimation of the system distance $\gamma(\mu, k)$ can be expressed as

$$\gamma(\mu, k) = f_{\min}(EXR(\mu, k)) \approx \|\mathbf{H}_{\Delta}(\mu, k)\|^2. \quad (22)$$

The minimum tracker is updated recursively by weighting the previous estimation with an update parameter $\varepsilon(\mu, k)$

$$\gamma(\mu, k) = \varepsilon(\mu, k) \cdot \gamma(\mu, k - 1). \quad (23)$$

$\varepsilon(\mu, k)$ determines how fast the minimum tracker is able to rise or fall. To do so, it is controlled by the relation

$$\beta_{\min}(\mu, k) = \frac{\gamma(\mu, k - 1)}{EXR(\mu, k)}. \quad (24)$$

If $\beta_{\min}(\mu, k) > 1$, the new input is smaller than the old estimation and thus the minimum tracker must fall, resulting in $\varepsilon(\mu, k) = \varepsilon_f < 1$. Otherwise, if $\beta_{\min}(\mu, k) \leq 1$, the new input exceeds or equals the old estimation. In this case, the minimum tracker must stay at its current value. Therefore it rises only slowly, i.e. $\varepsilon(\mu, k) = \varepsilon_r \approx 1$. The problem is, that if enclosure dislocations occur, the real system distance rises fast. Due to the small rising constant, the estimator can not follow this ascent and the filter is not able to adapt to the new environment as fast as necessary. For this reason, a rescue mechanism is required, that sets the rising constant to a large value $\varepsilon(\mu, k) = \varepsilon_{rf} > 1$ if an enclosure dislocation is detected. A straightforward detection mechanism can be realized by comparison of error signal and microphone signal $y(n) \circ \bullet Y(\mu, k)$. To obtain the error signal, the estimated feedback is subtracted from the microphone. As long as the estimation is close to the real system,

the power of the error signal is smaller than the power of the microphone. A wrong estimation leads to an increase of the error power, and so the rescue mechanism is activated if the relation

$$EYR(\mu, k) = \frac{|E(\mu, k)|^2}{|Y(\mu, k)|^2} \quad (25)$$

exceeds a certain threshold β_{resc} . Taken all together, the update parameter can enter one of the following states

$$\varepsilon(\mu, k) = \begin{cases} \varepsilon_f & \text{if } \beta_{\text{min}}(\mu, k) > 1 \text{ and } EYR(\mu, k) \leq \beta_{\text{resc}}, \\ \varepsilon_r & \text{if } \beta_{\text{min}}(\mu, k) \leq 1 \text{ and } EYR(\mu, k) \leq \beta_{\text{resc}}, \\ \varepsilon_{rf} & \text{if } EYR(\mu, k) > \beta_{\text{resc}}. \end{cases} \quad (26)$$

It was found that $\varepsilon_f = 0.99$, $\varepsilon_r = 1.001$, $\varepsilon_{rf} = 1.02$ and $\beta_{\text{resc}} = 5$ dB are suitable values for the evaluated scenarios.

Finally, the optimal stepsize is approximated by the variable stepsize $\alpha_{\text{var}}(\mu, k)$

$$\alpha_{\text{opt}}(\mu, k) \approx \alpha_{\text{var}}(\mu, k) = \frac{|X(\mu, k)|^2}{|E(\mu, k)|^2} \cdot \gamma(\mu, k). \quad (27)$$

To obtain the overall stepsize for the filter update (10), $\alpha_{\text{var}}(\mu, k)$ is limited to minimum and maximum values $\alpha_{\text{min}} = 0$ and $\alpha_{\text{max}} = 1.2$ and multiplied by a fixed value $\alpha_{\text{fix}} = 0.3$

$$\alpha(\mu, k) = \alpha_{\text{fix}} \cdot \begin{cases} \alpha_{\text{min}} & \text{if } \alpha_{\text{var}}(\mu, k) < \alpha_{\text{min}}, \\ \alpha_{\text{max}} & \text{if } \alpha_{\text{var}}(\mu, k) > \alpha_{\text{max}}, \\ \alpha_{\text{var}}(\mu, k) & \text{else.} \end{cases} \quad (28)$$

While the system distance is estimated frequency selective, the gain is applied to the broadband signal. To control the gain with the estimated system distance, it is averaged over the frequency bins. The gain would increase steadily with decreasing system distance. Since this is not useful in a real application, it is limited to a maximum value g_{max} , resulting in

$$g(n) = \text{MIN} \left\{ g_{\text{max}}, \sqrt{\left(\frac{1}{N} \sum_{\mu=0}^{N-1} \gamma(\mu, k) \right)^{-1}} \right\}. \quad (29)$$

4. Simulations and Implementation

The impulse response from rear seat loudspeaker to driver microphone, measured in a real sedan car, is used for simulations. The recording of a male speaker is used as local speech, the waveform is normalized to unity. Without feedback cancellation, the MSG is reached at 0 dB. Using the proposed algorithm for adaptive feedback cancellation, it is possible to increase the MSG up to +30 dB. The possible gains of the different approaches are compared in Tab. 1.

Table 1: Achievable MSGs with different approaches.

Without feedback cancellation	0 dB
Fixed stepsize $\alpha(\mu, k) = 0.02$	≈ 7 dB
Reverb-based stepsize w/o gain control [12]	≈ 20 dB
Reverb-based stepsize with gain control	≈ 30 dB

Fig. 4 shows the time curve of gain and system distance. The gain is plotted broadband in the upper figure. After approx. 10 s the desired maximum gain of 30 dB is reached. In the lower

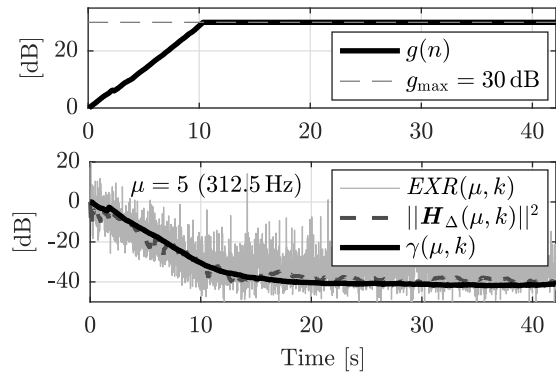


Figure 4: Time curve of gain and system distance estimation. The lower plot shows the curves at one frequency bin $\mu = 5$.

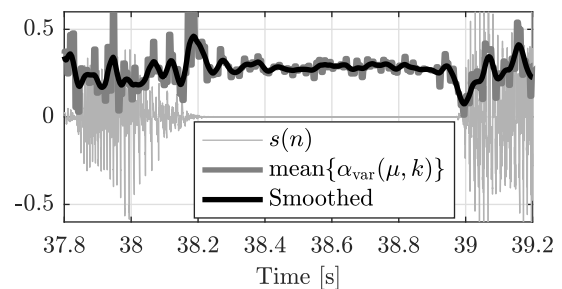


Figure 5: Mean stepsize $\alpha_{\text{var}}(\mu, k)$ during a speech pause.

figure $EXR(\mu, k)$ as well as the real- and the estimated system distance are shown. Since these quantities are frequency selective, only one frequency bin lying within the speech spectrum (312.5 Hz) is plotted. In Fig. 5 $\alpha_{\text{var}}(\mu, k)$ is plotted during a speech pause. For better presentation, it is averaged over the speech spectrum (125 Hz to 8000 Hz) and smoothed. One can observe a peak at the end of speech activity (≈ 38.2 s) and a drop at the beginning of speech activity (≈ 39 s), showing that reverb detection works as expected. Since a fixed impulse response was used for simulations, the functionality of the rescue mechanism is not shown here. However, real-time tests of the algorithm in a demonstration car showed that movements of the passengers are detected and the estimated system distance is increased accordingly. Driving tests further showed that the algorithm also performs reliable in the presence of background noise.

5. Conclusions

An adaptive feedback canceler with a reverb-based stepsize control was presented. It was shown that with the proposed approach, the MSG can be increased by 30 dB, which is approx. 10 dB above previous approaches. With the stepsize control both convergence and stability of closed-loop electro acoustic systems can be improved without the need of further decorrelation methods. Furthermore, the proposed approach is also efficient in terms of computation, since it is implemented in the frequency domain and it exploits given properties of the filterbank. Future work involves combination of the adaptive feedback canceler with a residual feedback suppression. This is necessary to suppress howling artifacts, occurring during time instants where the adaptive filter is mismatched.

6. References

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