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Noise injection for feedback cancellation with linear prediction

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ABSTRACT

Feedback oscillation is one of the major issues with hearing aids. An efficient way of feedback suppression is feedback cancellation, which uses an adaptive filter to estimate the feedback path. However, the feedback canceller suffers from the bias problem in the feedback path estimate. The recent progress suggests a feedback canceller with linear prediction of the desired signal in order to eliminate the bias when certain conditions are met. However, the bias still remains in many situations, for example when the input signal is voiced speech. Noise injection is investigated in this paper to help reduce the bias further and improve the system performance. Two nearly inaudible noises are proposed: a masking noise, which is tailored to the hearing-aid application, and a linear prediction based noise, which is especially efficient for feedback cancellation with linear prediction. Simulation results show that noise injection can further reduce the feedback estimation error by 1-4 dB and/or increase the stable gain by 3-4 dB, depending on the characteristics of the input signal.

INTRODUCTION

Feedback oscillation is one of the major problems with hearing aids. It limits the maximum gain that can be achieved. A widely adopted approach to feedback suppression is feedback cancellation, where an adaptive filter is used to model the feedback path. However, the closed-loop plant used in continuous-time feedback cancellation systems for hearing aids results in biased estimations of the feedback path when the input and output signals are correlated [1].

Several approaches have been proposed to reduce the bias. Classical approaches include introducing signal de-correlating operations in the forward path or the cancellation path (such as delays or nonlinearities), adding a probe signal to the receiver input, and controlling the adaptation of the feedback canceller , e.g., by means of constrained or bandlimited adaptation [2]. New solutions based on closed-loop identification theory have been investigated in the recent years [3] [4]. As a result, a feedback cancellation system with linear prediction was proposed in [5], which eliminates the bias when certain conditions are met. A combination with the classical solutions were also mentioned in [5], which showed that linear prediction combined with noise injection could improve the system performance further. However, the injected noise used in that work was an audible speech-shaped noise and therefore compromised the sound quality.

This paper proposes two kinds of nearly inaudible noises for injection in the hearing-aid output when the feedback canceller with linear prediction is used. The results show that the injected noise maintains the sound quality, reduces the bias further and increases the stable gain.

The outline of the paper is as follows: in Section , the feedback cancellation system with linear prediction is explained. In Section the adaptation structure for noise injection is discussed and the generation of inaudible noises is described. In Section , simulation results are presented. Concluding remarks are given in Section .

FEEDBACK CANCELLATION WITH LINEAR PREDICTION

Two classes of adaptive procedures for identifying the desired signal model and the feedback path were derived in [5]: a two-channel identification method and a prediction error method (PEM). The latter is found to be preferable for highly non-stationary sound signals and is therefore chosen as the method investigated in this paper.

The diagram of the prediction error model based adaptive feedback canceller (PEM-AFC) used in this work is depicted in FIG. 1. The signal processing path of the hearing aid (the so-called forward path) is denoted by $G(q)$; the acoustic feedback path is denoted by $F(q)$. The receiver and microphone signals are $u[k]$ and $y[k]$, respectively. The filter $\hat{F}_0(q)$ in the feedback cancellation path is an initial estimate of $F(q)$. It is continuously replaced during adaptation by the estimate $\hat{F}(q)$. The external input $x[k]$ is assumed to be an Autoregressive (AR) random process generated from the white noise sequence $w[k]$ and the AR model $H(q)$, which is also referred to as signal model. The FIR filter $A(q)$ is the prediction error filter of the forward-path output $g[k]$. In this diagram, the linear prediction is placed at the receiver end before noise injection which differs from the diagram in [5], where the linear prediction is located at the microphone side and operates on the error signal $e[k]$. This change of placement does not affect the steady-state performance but gives better transient convergence [6]. The prediction error (PE) of $e[k]$ and $u[k]$ are denoted by $e_p[k]$ and $u_p[k]$ respectively. The probe signal (injected noise) is $r[k]$.

Assuming that the input signal $x[k]$ is an AR random process, it has been shown in [5] that when the delay in the forward path is longer than the order of the signal model $H(q)$ or when the probe signal $r[k]$ is introduced, the minimization of $e_p[k]$ leads

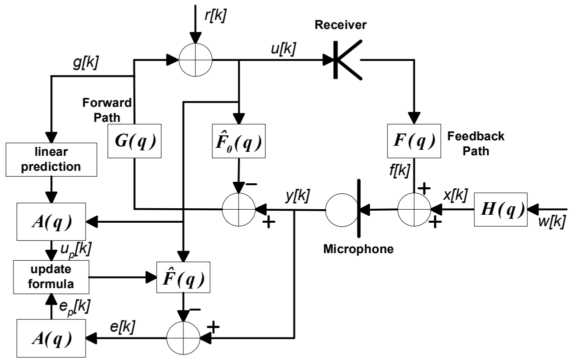

Figure 1: Diagram of the feedback cancellation system with linear prediction (PEM-AFC). Modified from [5].

to:

$$A(q) = H^{-1}(q); \hat{F}(q) = F(q) \tag{1}$$

$$H(q) = \frac{1}{1 + h_1 q^{-1} + \cdots + h_2 q^{-2} + h_{L_H-1} q^{-L_H+1}} \tag{2}$$

where q^{-1} is a discrete-time unit delay operator, $L_H - 1$ is the order of the all-pole filter $H(q)$. $A(q)$, $F(q)$ and $\hat{F}(q)$ are all FIR filters.

Although many signals can be modeled by AR random process, a large set of real-life signals, such as voiced speech and tonal music, can hardly be modeled by a low-order $H(q)$. For these signals, PEM-AFC still suffers from the bias problem due to the remaining spectrally-colored signals[1]. In addition, under-modeling which generally occurs in practice introduces the bias into the estimation. In both cases, a reduction of the remaining bias is necessary.

NOISE INJECTION FOR PEM-AFC

To reduce the bias, the probe signal $r[k]$ can be injected into the signal $g[k]$ before it is sent to the receiver. The injected noise $r[k]$ is generally uncorrelated with the input signal $x[k]$. In Fig. 1, the feedback canceller takes $u[k]$ as the input signal for adaptation. An alternative approach is to use $r[k]$ instead of $u[k]$ in the adaptation of the feedback canceller [7]. However, the low level of the probe signal $r[k]$ (to maintain the sound quality) results in a large excess error or slow adaptation of the adaptive filter $\hat{F}(q)$ [2], which will not be suitable for dynamic situations in the daily life. Therefore only the first approach as shown in Fig. 1 is considered in this paper.

Bias reduction with noise injection

The steady-state analysis of feedback cancellation has revealed that the bias problem is related to the autocorrelation of the input signal $x[k]$ [1]. In PEM-AFC, a similar steady-state analysis can be performed as follows:

$$\hat{\mathbf{f}} = \mathbf{R}^{-1}_{\mathbf{u}_p\mathbf{u}_p} \mathbf{r}_{\mathbf{u}_p y_p}, \tag{3}$$

$$= \mathbf{R}^{-1}_{\mathbf{u}_p\mathbf{u}_p} \mathbf{r}_{\mathbf{u}_p f_p} + \mathbf{R}^{-1}_{\mathbf{u}_p\mathbf{u}_p} \mathbf{r}_{\mathbf{u}_p x_p}, \tag{4}$$

$$\mathbf{R}_{\mathbf{u}_p\mathbf{u}_p} = E\left\{\mathbf{u}_p[k]\mathbf{u}_p^H[k]\right\}, \tag{5}$$

$$\mathbf{r}_{\mathbf{u}_p y_p} = E\left\{\mathbf{u}_p[k] y_p[k]\right\}, \tag{6}$$

$$\mathbf{u}_p[k] = \left[u_p[k], u_p[k-1], \cdots, u_p[k-L+1]\right]^T, \tag{7}$$

$$f_p[k] = A(q)F(q)u[k], \tag{8}$$

$$\hat{F}(q) = \hat{f}_0 + \hat{f}_1 q^{-1} + \cdots + \hat{f}_{L-1} q^{-L+1}, \tag{9}$$

[1]Spectral coloring refers to the fact that certain spectral components are stronger than other spectral components. A spectrally colored signal may be a broad-band (e.g., a speech signal) as well as a narrow-band signal (e.g., a sinusoid or alarm signal) [2].

$$\hat{\mathbf{f}} = \left[\hat{f}_0, \hat{f}_1, \cdots, \hat{f}_{L-1}\right]^T, \tag{10}$$

where $\hat{\mathbf{f}}$ is the coefficient vector of $\hat{F}(q)$, L is its length, the symbol with subscription p denotes the PE of the corresponding signal. The symbol E denotes the expectation operator, and $\mathbf{r}_{\mathbf{u}_p f_p}$ and $\mathbf{r}_{\mathbf{u}_p x_p}$ are defined similarly to equation (6). Suppose that the filter length L is sufficient. The first term in equation (4) approximates the true feedback path, whereas the second term, which represents the correlation between the PE of the desired input signal $x[k]$ and the PE of the processed hearing-aid signal $u[k]$, introduces a bias into the estimate. When the signal $u[k]$ includes the injected noise $r[k]$, the bias term can be reduced. The effectiveness of bias reduction depends on the prediction-error strength of the injected noise $r[k]$ relative to the prediction-error strength of the signal $g[k]$.

However, a loud noise will degrade the sound quality. The injection of noise therefore involves a trade-off between bias reduction and sound quality. To maintain the sound quality, two types of nearly inaudible noises are investigated in the following sections, namely the masking noise and the linear prediction based noise (LP noise).

Masking noise

The Moving-Pictures-Expert-Group (MPEG) standard for audio compression [8] utilizes frequency masking to reduce the number of bits in the intervening transmission. It assumes an audio signal with a sampling frequency of 44.1 kHz and calculates the masking threshold in each sub-band based on the instantaneous spectrum of the signal by using a 512-point FFT.

To make it possible and affordable for hearing-aid application, the calculation is modified to operate with the sampling frequency of 16 kHz and 64-point FFT. The loss in sound quality due to such a modification is found to be insignificant. In addition, since feedback whistling usually occurs below 6 kHz, the masking threshold is not calculated for those sub-bands above 6 kHz.

The masking noise is generated by shaping the white noise sequence with the calculated masking threshold so that it has the same average spectrum as the masking threshold. It should be stressed that the masking noise formed in this way is still audible because the psychoacoustic model used to calculate the masking thresholds is established for sinusoids instead of noise. Therefore the masking noise generated from the masking thresholds is attenuated by 14 dB in this work to make it nearly inaudible.

LP noise

In order to avoid the degradation of sound quality while providing as strong a force as possible to reduce the bias, the injected noise $r[k]$ should be strong in those frequency bands where the desired signal is strong and weak where it is weak. LP noise is proposed to achieve this idea efficiently.

Instead of calculating the spectrum of the outgoing signal $g[k]$, LP noise is generated by filtering a white noise sequence with the inverse of the prediction-error filter $A(q)$ as shown in the equations below:

$$g_p[k] = A(q)g[k] \tag{11}$$

$$p[k] = \frac{1-\beta}{1-\beta q^{-1}} g_p[k]^2 \tag{12}$$

$$r[k] = \alpha\sqrt{p[k]}A^{-1}(q)n[k] \tag{13}$$

where $p[k]$ estimates the smoothed prediction-error power of $g[k]$ by passing the instant power through a low-pass IIR filter configured by a smoothing factor β, which is set to 0.996 in this paper. The white noise sequence $n[k]$ has unit variance. The parameter α controls the strength of the LP noise. $\alpha = 0.3$ is found to be low enough to produce nearly inaudible noises for most signals.

Noise generated in this way has two advantages: first, the computation load is very low as $A(q)$ is ready to use after the linear prediction stage; secondly, when $r[k]$, embedded in $u[k]$, goes through $A(q)$ to form the prediction error signal $u_p[k]$, it becomes the white noise sequence $n[k]$ again, which has ideal autocorrelation properties. To ensure the stability of the IIR filter $A^{-1}(q)$, the linear prediction has to yield a stable model.

SIMULATION RESULTS

To evaluate the performance of PEM-AFC with noise injection, the system in the Fig. 1 is simulated with a sampling frequency of 16 kHz. The feedback path is 50-order, measured from a commercial behind-the-ear (BTE) device and normalized so that the maximum stable gain is 0 dB without feedback cancellation, as illustrated in Fig. 2. The feedback model $\hat{F}(q)$ is also 50-order and the linear prediction-error filter $A(q)$ is 21-order. Levinson-Durbin algorithm, which yields stable models, is used to linear predict $g[k]$ with a frame length of 168 samples. The forward path $G(q)$ consists of a delay of 24 samples and a linear gain. The adaptation algorithm for feedback canceller is the recursive-least-square (RLS) algorithm with a forgetting factor $\lambda = 1$. Three types of input signals are investigated: a 3 kHz pure tone in white noise (to simulate the background noise and the microphone noise) whose power is -40 dB below the tone, male speech signal and guitar music signal. All the signals are 12-second long.

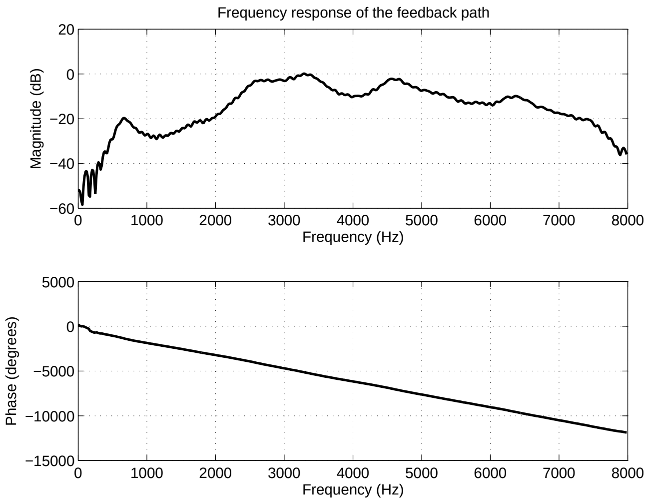

Figure 2: The frequency response of the feedback path of 50 orders based on the measurement of a commercial BTE hearing aid.

As mentioned in Section , in PEM-AFC, the amount of bias reduction actually depends on the relative strength of the prediction error of the injected noise. To illustrate the effectiveness of the two inject noises, it is first assumed that the feedback cancellation is perfect, i.e., $\hat{F}(q) = F(q)$. When the 3 kHz sinusoid in white noise is input into the system, the power spectral density (PSD) of the signal $g[k]$, the injected LP noise, the injected masking noise and PSD of their corresponding prediction errors are plotted in Fig. 3.

In the figure, it can be seen that the PSD of the prediction error of the signal $g[k]$, i.e. $g_p[k]$, exhibits a peak at 3 kHz, which indicates that $g_p[k]$ is still a highly autocorrelated signal and therefore the bias remains in the system after linear prediction. The PSD of the injected LP noise resembles the PSD of the signal $g[k]$, and the prediction error of the LP noise is white as expected. In the 3-kHz frequency band, the prediction error of the LP noise is as strong as the peak in the PSD of $g_p[k]$. Hence, the LP noise can provide a significant force to reduce the bias. In those frequency bands around 3 kHz (from 2.72 kHz to 2.97 kHz and from 3.03 kHz to 3.28 kHz), the prediction error of the masking noise has almost the same power as that of $g_p[k]$. Therefore, the masking noise should help more in bias reduction at those frequencies than the LP noise.

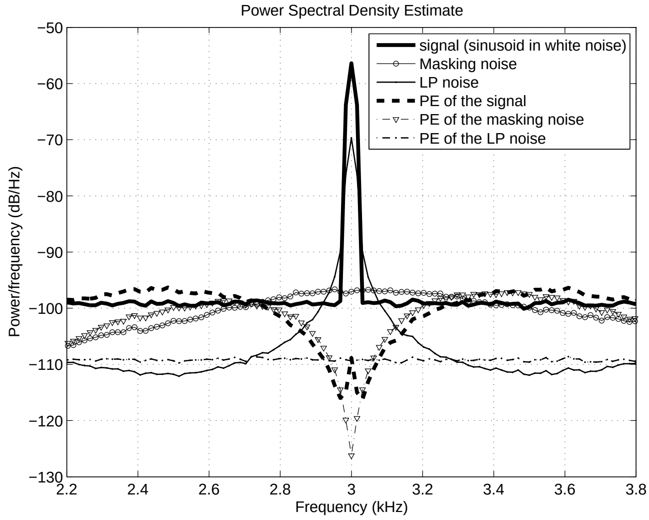

Figure 3: Power spectral density of the signal (a 3-kHz sinusoid in white noise), the injected masking noise, the injected LP noise and their corresponding prediction errors.

To assess the performance quantitatively, we used signal to feedback residual error (SFRR) as the measure:

$$SFRR = 10\log_{10}\frac{\sum_{k=1}^{N}(F(q)u[k] - \hat{F}_0(q)u[k])^2}{\sum_{k=1}^{N}(x[k])^2} \qquad (14)$$

where N is the total number of samples used in the simulation. $x[k]$ and $u[k]$ are both zero-mean. SFRR represents the ratio of the input signal to the distortion resulted from the feedback estimation error. This measure is more proper for narrow-band signals than the traditional broad-band measure maximum stable gain (MSG) because stable gains outside the signal band are not very meaningful.

The simulation results are shown in Fig. 4. For a sinusoid in white noise, the injected LP noise yields a 1-4 dB improvement of SFRR, and the masking noise provides 4 dB extra stable gain (the stable gain is defined as the point where SFRR begins to drop steeply). For music, the LP noise improves SFRR by 1 dB whereas the masking noise improves the SFRR slightly. For speech, the LP noise and the masking noise improve SFRR slightly and provide 3-4 dB extra stable gain.

In general, the injected LP noise performs the best (1-4 dB SFRR increase) before the system becomes instable, whereas the masking noise is better at providing additional stable gain. The reason is that when the system is stable, the LP noise has a stronger force to reduce the bias (c.f. Fig. 3). However, when the whistle is about to occur, the linear prediction in PEM-AFC tends to model the strong feedback residual signal instead of the desired signal. The LP noise generated with these wrong linear prediction coefficients does not help very much in the bias reduction. The masking noise, which on the other hand, takes into account the spectrum of the whole signal including both the desired signal and the feedback residual signal, may still provide forces in keeping the system from instability.

In the simulation, it was also found that the LP noise and masking noise injected in the traditional feedback cancellation system without linear prediction provided very little improvement. This is in agreement with the findings in [5]. Therefore the results are omitted in Fig. 4. Noise injection is more effective in feedback cancellation with linear prediction probably because of two reasons: Firstly, linear prediction reduces the bias and therefore slows down the speed that the system can go wrong at. As input signals usually have weak periods with low amplitude from time to time (due to the amplitude fluctuation and pauses), the noise injection may help the system to go back quickly onto the right track in those periods. Even when the adaptation will still be biased afterwards, the system may keep stable before the next weak period is hit. However, if there is no linear prediction, the efforts noise injection has made during the weak periods will be overruled quickly by the large bias and the whistle may occur immediately. Secondly, linear prediction is performed on noisy data, so over- and under-modeling can both occur, which averages out part of the remaining bias. In other words, the injected noise actually plays a relatively larger role than expected.

CONCLUSIONS

This paper investigates the inaudible noise injection in the feedback cancellation with linear prediction. Two inaudible noises are proposed: the masking noise, which is tailored to the hearing-aid application based on the MPEG standards, and the LP noise, which is proposed specially for the feedback canceller with linear prediction to achieve an efficient implementation. The effect of noise injection is evaluated for tonal signal, speech and music. It is shown that noise injection can reduce the feedback estimation error by 1-4 dB and/or increase the stable gain by 3-4 dB, depending on the characteristics of the input signal.

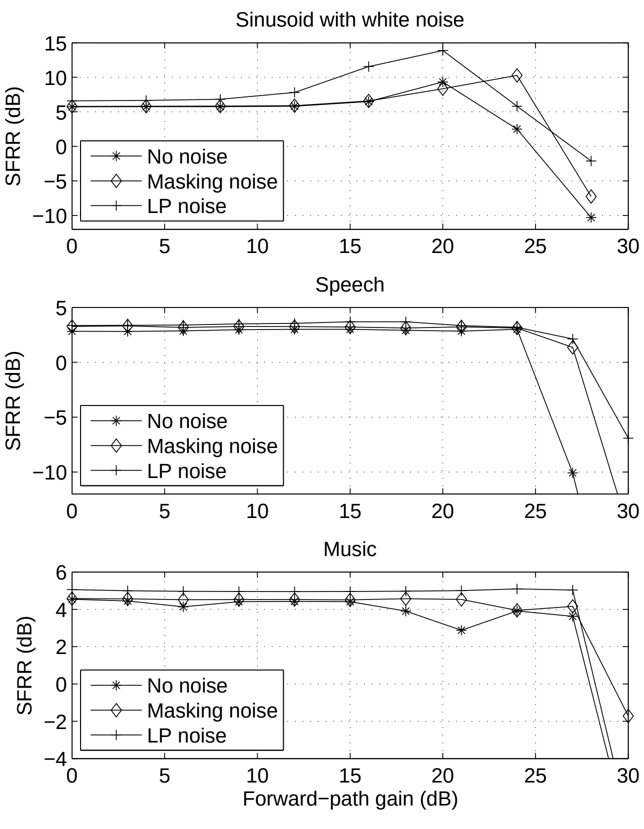

Figure 4: SFRR of PEM-AFC with no noise, LP noise injected and masking noise injected when the input is a sinusoid with noise, male speech and guitar music respectively.

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