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PROACTIVE NOTCH FILTERING FOR ACOUSTIC FEEDBACK CANCELLATION

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ABSTRACT

We propose a method which performs acoustic feedback cancellation based on a closed loop estimate of the room impulse response and traditional notch–filtering techniques. The room impulse response estimate, provided by the PEM–AFROW algorithm, allows to pro–actively design notch filters, i.e. they can be inserted into the loop before acoustic feedback occurs. Extra feedback cancellation behaviour is provided by an adaptive controller based on the same estimate.

1. INTRODUCTION

In closed–loop electro–acoustic systems used in public address (PA) applications or hearing aids, system instability due to acoustic feedback (the so–called Larsen–effect, or *howling*) is a well known phenomenon. It occurs when the system loop gain exceeds unity on frequencies ω_i where the loop phase equals $2n\pi$ radians, with n integer. An acoustic feedback canceller is a controller for this system, which attempts to maintain stability, resulting in an 'added stable gain'. This performance measure is the difference between the maximal gain for a stable controlled system and the maximal gain for a stable uncontrolled system.
The most important disadvantage of traditional solutions [1, 2], mainly based on the insertion of notch filters centered on ω_i, is that they are *reactive* (the instability first has to occur before the notch filter can be designed and typically the detection takes 0.5 second). On the other hand, a major advantage is that the notch filtering technique can be quite robust against fast changes in the acoustic environment : typically, a small displacement of the microphone will only modestly affect the center frequencies on which instability occurs. This means that by using wider notch filters, robustness can be achieved.
In [3, 4], an alternative acoustic feedback cancelling method is derived which is based on the so–called PEM–AFROW algorithm. This method belongs to a class of acoustic feedback cancellation algorithms which are inspired on acoustic echo cancelling [5, 6, 7, 8, 9]. **Figure 1** provides a diagram, where g is the gain, $q^{-1}$is the delay operator,

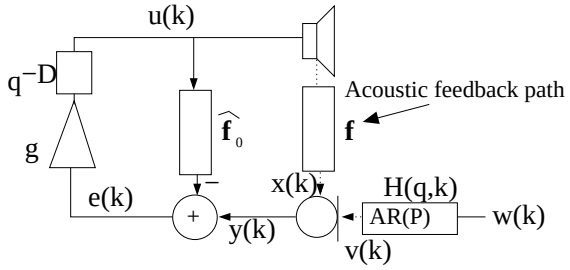

Figure 1: Acoustic feedback cancellation

hence q^{-D} is a delay of D samples. The speech source signal $v(k)$ is assumed to be a time–variant autoregressive (TVAR) process, and is written as an IIR filter

$$H(q,k) = \frac{1}{1 + a_1(k)q^{-1} + ... + a_P(k)q^{-P}} \qquad (1)$$

driven by white noise $w(k)$, so $v(k) = H(q,k)w(k)$. The PEM-AFROW algorithm allows for unbiased joint identification of a linear model $\hat{\mathbf{f}}$ of the room impulse response (RIR) $\mathbf{f}$, and of the TVAR–coefficients $\hat{\mathbf{a}}(k)$ of the denominator of the speech source signal model $H(q,k)$ in the a closed loop system of **Figure 1**. Here $\hat{\mathbf{f}}$ is a vector which contains estimates of the filter coefficients of the finite impulse response (FIR) filter which models the room impulse response, and $\hat{\mathbf{a}}(k)$ is a vector

$$\hat{\mathbf{a}}(k) = \begin{pmatrix} 1 \\ \hat{a_1}(k) \\ \vdots \\ \hat{a_P}(k) \end{pmatrix}. \qquad (2)$$

The estimated room impulse response $\hat{\mathbf{f}}$ can then be used in a so–called 'controller' $\hat{\mathbf{f}}_0$, where the loudspeaker signal (far–end signal) $u(k)$, filtered by the model of the room impulse response (the controller), is subtracted from the microphone signal $y(k)$. The resulting signal $e(k)$ has no remaining component stemming from the loudspeaker, but it only contains the so–called 'near end signal' $v(k)$.

The advantages of using such a scheme are obvious. It is *pro-active* (no feedback has to occur before identification can take place), and the room impulse response is continuously tracked using the speech signal $u(k)$ as a measurement signal. Secondly, no (notch) filtering occurs in the signal path, which means that in the ideal case, no signal distortion occurs. Third, this scheme removes the 'reverberating' sound which occurs on the edge of stability.

A disadvantage of this scheme is a robustness issue in case of changing acoustic feedback paths. The controller $\hat{\mathbf{f}}_0$ should be matched to the acoustic feedback path $\mathbf{f}$ at all times to avoid instabilities. Fast changes in the environment can often not be tracked fast enough by the adaptive filter.

In this paper we propose to combine the pro–activity of the PEM–AFROW scheme with the robustness of the notch–filtering based approach. In section 2, the method is described, and in section 3, simulations are provided. Conclusions are geven in section 4.

2. NOTCH FILTERING BASED ON PEM–AFROW ESTIMATES

It can be observed that while PEM–AFROW is quite robust against changes in the acoustic feedback path due to moving objects (even in the vicinity of loudspeaker or microphone), it is less robust than notch filtering based techniques when the loudspeaker or microphone themselves are moved. We have simulated a room impulse response [10] for a room of 10x5x4 meter, reflection coefficients of the walls are 0.8, with a loudspeaker on position (3.21, 3.21, 2.5) and a microphone on positions (1,2,1) and (1,1.95,1), hence 5 cm apart. The room impulse response for the first microphone position and the difference between the room impulse responses for both microphones is plotted in **Figure 2**. The difference has the same order of magnitude as the impulse responses themselves. It is obvious that a PEM-AFROW based approach, which has to track the impulse response, will need a certain amount of time before it has compensated for the difference between the impulse responses. On the other hand, the position of many of the peaks in the loop gain frequency spectrum shown in **Figure 3**, does not change much, which indicates that a notch filtering based approach can be more robust against the microphone displacement.

Depending on the room acoustics, only a limited number of modes is available for instabilities to occur, and a small change in the distance between loudspeaker and microphone may lead to a small change in the center frequency of the oscillation (unless a completely different mode becomes active), but to large changes in the room impulse response. Since notch filtering is robust against small changes in the center frequency, it is desirable to

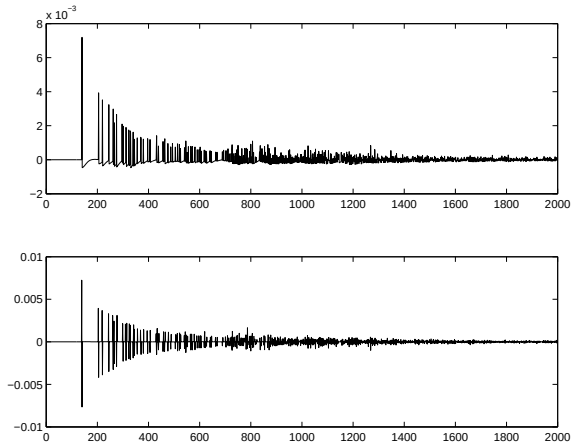

Figure 2: RIR vs. time index for first microphone position, and the (large) RIR–difference for both positions.

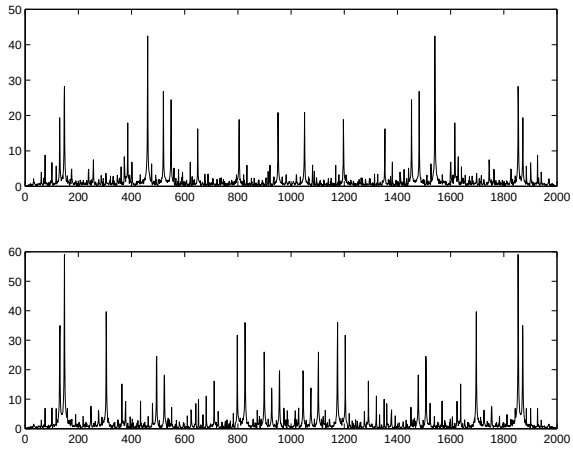

Figure 3: Loop gain vs. frequency index for two slightly different microphone positions (10 cm apart)

combine the PEM–AFROW provided estimates of the room impulse response (pro–active) and the notch–filtering technique (robust).

The condition for instability is fulfilled when the loop gain $G_l(\omega) > 1$ on frequencies where the loop phase $\varphi_l = k2\pi$, with k an integer number. The frequencies ω_i for which this condition holds, can easily be found from the estimate of the room impulse response which is provided by PEM–AFROW, and the knowledge of the forward gain G :$G_l(\omega) = G$.abs(fft($\hat{\mathbf{f}}$)).

The algorithm we have used to design the notch–filters is given in **Algorithm 1**. It places notch–filters on frequencies ω_i where $G_l(\omega_i) > T_{max}$ with threshold $T_{max} < 1$, independently of the phase characteristic (allthough this would be easy to incorporate), and it removes notch filters

Algorithm 1 Notch filter design, starting from the PEM–AFROW room impulse response estimate.

1. for i=1:NrActive
   if $G_l(\omega_i) < T_{min}$ decrease depth on ω_i
2. While maxima can be found
   (a) find ω_{max} = argmax$G_l(\omega)$ where $\omega_{max} > T_{max}$
   (b) if ω_{max} close to existing notch ω_F: $\omega_F = \omega_{max}$; increase depth on ω_F
   elseif NrActive < Max : new notch on ω_{max}

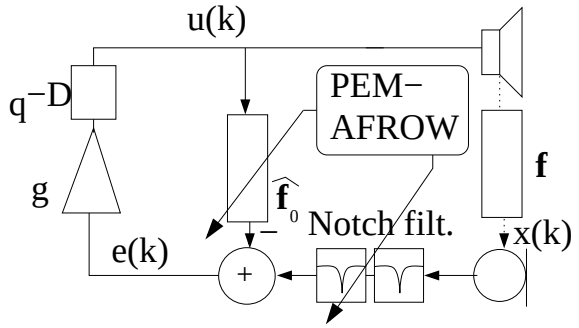

Figure 4: Combined system with proactive notch filtering and a controller $\hat{\mathbf{f}}_0$.

on frequencies where $G_l(\omega) < T_{min}$. The notch filter design procedure is repeated regularly (a few times per second), and each time only a limited number of notch filters are designed. Then the PEM–AFROW algorithm is allowed to adapt to the new acoustic impulse response *with the notch filters included*. This results in an update of $G_l(\omega)$. The notch filter depth on a certain frequency may then be increased or decreased, depending on the new value of $G_l(\omega)$.

The notch filters are IIR filters with two poles and two zeros each. From a numerical point of view, it is important not to combine all of the notch filters into one IIR filter, but to apply all of the so–called 'biquads' individually to the signal, otherwise instability may result if the poles are close together.

The notch filtering based acoustic feedback controller is now combined with a PEM-AFROW based controller, as shown in **Figure 4**. The PEM–AFROW algorithm estimates the cascade of the notch filters with the room impulse response, this estimate can be used in a controller $\hat{\mathbf{f}}$ which then cancels the remaining loudspeaker component in the microphone signal. The advantage compared to the PEM–AFROW–only algorithm [3, 4] is that the notch filters — when chosen appropriately wide — guarantee robustness for path changes (in exchange for some distortion), and the controller removes the reverb–like sounds which remain in the system of previous section.

If T_{max} is chosen lower than 1, the above technique leads to a pro–active system, because new notch filters will be designed before the system becomes unstable. The system can be made robust to path changes by choosing an appropriate width for the notch filters. The parameters for the notch filters should be chosesn to avoid audible distortion. It is important to note that due to the additional PEM–AFROW based controller $\hat{\mathbf{f}}_0$, which removes residual loudspeaker components from the microphone signal, the loop gain on the frequencies ω_i is not limited to T_{max} as in traditional, notch–filtering only based techniques.

3. SIMULATIONS

Real–time experiments at 16 kHz sampling rate in a real environment were performed with added gains of about 5 dB, using only the notch filtering section (pro–active, designed based on the PEM–AFROW estimate), and without the additional controller $\hat{\mathbf{f}}_0$. It is known that this is a typical stable added gain for notch filtering based feedback suppression. It was shown that fast movements (about 1 Hz) of the microphone over a distance of 30 cm, at 1 m distance from the loudspeaker, do not result in instability for this proactive notch filtering approach, while for the PEM-AFROW based feedback suppression algorithm in [3, 4], instability does indeed sometimes occur under these conditions. Note though that, for constant microphone-loudspeaker distances, much higher added gains can be achieved with the approach in [3, 4] (up to 15 dB), even with large changes in the room impulse response due to movements of e.g. people in the vicinity of the loudspeaker or microphone.

In order to provide reproducible results, simulations were performed. They use the room and geometry setup as detailed in section 2. Sudden changes between the two room impulse responses are applied (which are worse conditions than the 1Hz movements in the real–environment experiment described above). Then it is evaluated for which maximum gain level, no instability occurs on changes of the impulse responses. The results are provided in **Table 1**. The stepsize for the NLMS–filter in the PEM–AFROW algorithm is 0.05. For comparison : the stable added gain when only the notch–filtering section is used is 5 dB.

4. CONCLUSION

PEM-AFROW based feedback control is pro–active, removes reverberation on the edge of stability, and does not introduce distortion. On the other hand, it sometimes lacks robustness to movements of loudspeaker or micro-

Bandwidth	Depth	Added gain
3dB BW = 1/20	12 dB	10 dB
3dB BW = 1/10	12 dB	13.9 dB
1dB BW = 1/5	3 dB	12.9 dB

Table 1: Simulation results for different notch filter parameters. Bandwidths are normalized (1 = Nyquist frequency).

phone. Notch filtering exhibits a more robust behaviour in that case, but in its traditional form, it is reactive, and it introduces distortion. Since notch filtering is robust to small changes in the center frequency, the combination of an adaptive filtering approach (based on the PEM-AFROW algorithm) and notch filters, leads to a proactive and robust design, which is a combination of the advantages of both techniques. The simulations show that the best results are obtained with wide notch filters with a stop band attenuation of less than 3 dB.

5. ACKNOWLEDGEMENTS

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