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USE OF ADAPTIVE FILTERS IN ACTIVE EAR DEFENDERS

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1. INTRODUCTION

One of successful applications of active noise control is active ear defenders. Recently, commercial active ear defenders became available in the market. Since those devices are equipped with non-adaptive analog control circuits, their performance has to be compromised to accommodate possible changes in the system. Such changes may arise from air leaks around the edges of an ear cover, or by relative motion between the device and the ear canal. Adaptive control algorithms implemented in a digital processor chip are expected to provide a better performance [1]. So far most efforts on digital ear defenders have been towards the adaptive feedforward control [2]. In this paper, an active ear defender based on adaptive feedback control is presented. One obvious reason of using feedback control is its less complex in hardware due to the removal of the feedforward path.

2. ADAPTIVE FEEDBACK CONTROL ALGORITHM

Fig.1(a) shows the schematic representation of a digital feedback ear defender. In feedback control, the pressure signal picked up by the error microphone is feedback to the digital controller which in turn drives the loudspeaker to minimise that same signal. The resulting electro-acoustic system is assumed to be linear and can be represented by a simplified block diagram as shown in Fig.1(b). In the figure, H_s is the transfer function of the so called secondary path comprised of the microphone, the loudspeaker and the acoustic path, and H_C the controller transfer function. The signals e, d and y are related to the sound pressures of controlled and uncontrolled, and the one generated by the loudspeaker respectively, and u the controller output. Using adaptive filtering to realise H_C [3] results in an adaptive feed-back control structure depicted in Fig.2, where $\hat{H}_s$ is the estimate of H_s, z^{-1} one sample time delay, and W a FIR (Finite Impulse Response) filter, the coefficients of which are optimally adjusted in real time according to the error signal e.

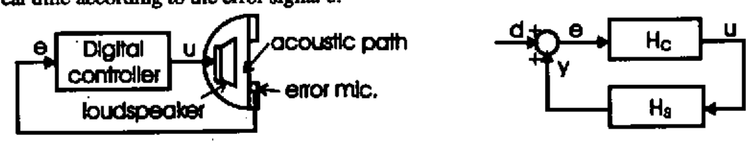

Fig.1. (a) Schematic representation of a feedback ear defender and (b) its block diagram.

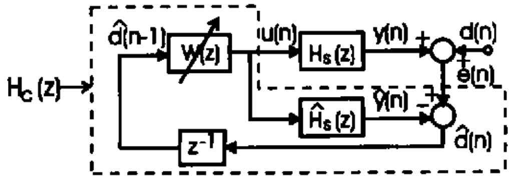

Fig.2. Block diagram of the adaptive feedback ear defender.

3. STABILITY AND ROBUSTNESS ANALYSIS

The instability of the adaptive feedback control of Fig.2 may result from three factors, ie, the divergence of W, the existence of feedback loop and the inaccurate estimate of H_s. The divergence of W depends largely on adaptive algorithms and will not be discussed here. To concentrate on the latter two factors, W is assumed to be time-invariant in the analysis. Thus, from Fig.2, the close loop transfer function between e and d will be

$$H_p(z)=\frac{E(z)}{D(z)}=\frac{1+\hat{H}_s(z)W(z)z^{-1}}{1+\left(\hat{H}_s(z)-H_s(z)\right)W(z)z^{-1}}. \tag{1}$$

In view of Eq.(1), the control objective can be stated as to adjust W in such way that the numerator of H_p is zero while the poles of H_p are kept within the unit circle.

From Eq.(1), if $\hat{H}_s=H_s$, which represents the situation of exact modelling, the close loop transfer function becomes $H_p(z)=1+H_s(z)W(z)z^{-1}$. Since H_s is a stable system in reality and W a FIR filter, H_p is guaranteed to be stable. Actually in this case, the control structure of Fig.2 can be equivalent to a pure feedforward system and hence stable.

If $\hat{H}_s \neq H_s$, the stability of the control structure of Fig.2 is no longer guaranteed. Thereafter we shall show how the inaccurate modelling of H_s affects the stability of the controller. In the analysis, H_s is assumed as a pure delay and W a two-tap FIR filter, ie:

$$H_s(z)=z^{-1},\quad W(z)=w_0+w_1z^{-1}. \tag{2}$$

It should be noted that the above assumption is purely for easy analysis and will not affect the conclusion derived. For a given frequency, the modelling error can be separated as the gain and the phase errors. Consequently, the effect of errors may also be treated separately.

If only the gain error is considered, ie, $\hat{H}_s=gH_s$, where $g\geq 0$ is the gain error, substituting Eq.(2) into Eq.(1) yields

$$H_p(z)=\frac{E(z)}{D(z)}=\frac{1+gw_0z^{-2}+gw_1z^{-3}}{1+(g-1)w_0z^{-2}+(g-1)w_1z^{-3}}. \tag{3}$$

The stability issue can then be examined in the following way. First, choose the optimal W which makes the numerator zero (ie, the control objective). Then for the optimal W, examine the pole positions that in turn determines the stability. By further assuming periodic signals and the sampling rate of four times per period, the corresponding poles will be

$$p_{1,2}=\begin{cases}\pm\sqrt{1/g-1}, & g\leq 1\\ \pm j\sqrt{1-1/g}, & g>1\end{cases}. \tag{4}$$

From Eq.(4), it can be seen that (i) the system becomes unstable if $g\leq 1/2$ and (ii) the further g away from 1 (ie., the larger the gain error), the closer the poles to the unit circle hence the worse the stability and the slower the convergence. Other analyses using a different H_s and a longer W show that the system will become unstable if the gain error is beyond a certain boundary. They also show that for a given H_s, the longer the filter W, the larger the stable boundary hence the more robust the system. It is interesting to compare the above result to that of the adaptive feedforward control, in which the stability and convergence of the system are not affected by the gain error at all.

The effect of the phase error can be examined in a similar way, which shows that (i) the maximum allowed error is always somewhat below ±90° of that for the adaptive feed-forward control and (ii) the longer the filter W the larger the allowed error.

In summary, the adaptive feedback control presented here is less robust than its feedforward counterpart. The robustness can however be increased to some extent by using longer control filters.

4. SIMULATION AND EXPERIMENTAL RESULTS

The aims of simulation are two-folds: (i) to verify the analysis result and (ii) to demonstrate the convergence behaviour of the feedback control in comparison with the feedforward one. In the simulation, W is adapted using the filtered-x LMS (Least Mean Square) algorithm. The step-size of adaptation is adjusted optimally to achieve the fastest convergence for each individual simulation. The secondary path H_s is obtained experimentally from a real ear defender. In most simulation, the primary noise d is a tonal signal contaminated by additive noise (Signal to Noise Ratio is 30 dB).

Table 1 summarises the simulation results related to the effect of the gain error, where the convergence time is the time for the MSE (Mean Square Error) of the system converging below 1% of its uncontrolled value. As indicated in Section 3, contrast to its feedforward counterpart, the feedback control system is indeed affected by the gain error g and becomes unstable if g is beyond a certain value (1/2 in this set of simulation). Increasing the filter length L to 64 does improve the convergence and stabilise the system.

The simulation results with the phase error is shown in Fig.3, where the normalised convergence time of four control arrangements, ie, feedback with 2-tap and 64-tap filters and feedforward with 2-tap and 64-tap filters, are plotted against the phase error. The following observations can be obtained from the simulation.

- The convergence of the feedback control is slower than that of the feedforward control, if the phase error is large. As shown in Section 3, the maximum allowed error of the feedback control is indeed somewhat below 90° of the feedforward control; reducing to 70° in the case of 64-tap filter and 35° for the 2-tap filter.
- Increasing the filter length indeed improves the stability and convergence of the feedback control (comparing the curve of the 64-tap filter to that of the 2-tap filter).
- If the phase error is kept small (below 30° in this case), the convergence of the feedback control will be as same good as its feedforward counterpart. With a longer filter, the convergence of the feedback can still be comparable to that of the feedforward even with a relatively large phase error (eg, 55° in this case).

It is known that the noise reduction of adaptive filtering depends on two factors, ie, the coherence between the unwanted noise and the input of the adaptive filter (known as the coherence constraint) and the time lead in the system (known as the time constraint). For the feedback control presented here, the coherence constraint is always met as long as the secondary path is reasonably modelled, but the time constraint cannot be met due to the forever time lag which is the nature of feedback. Therefore, the noise reduction of the feedback control with adaptive filtering depends solely on the predicability or the autocorrelation of the noise to be cancelled. To demonstrate the effectiveness of the adaptive feedback ear defender in different noise environments, experiments are conducted. The type of noise used are tonal noise of various frequencies, ramp noise, octave band noise and pink noise. In the experiments, the ear defender is put on a dummy head, in the ear cannel of which a measuring microphone is mounted. Both feedback and feedforward control are used. Control parameters for both control are set to be same. The length of the adaptive filters is 64. The sampling frequency of the controller is 2000 Hz. The time lag of the system is measured as 3.5 ms.

Fig.4 shows the experimental results in tonal noise environment. It can be seen that for tonal noise the feedback control yields the same good noise reductions as its feedforward counterpart. This is expected, as total predicability of periodic noise removes the requirement of time lead. The results with ramp noise further confirm the strength of the feedback control in periodic noise environment. The experimental results summarised in table 2 shows the noise reduction of the feedback control in relation to the autocorrelation of the noise to be cancelled. As expected, the higher the autocorrelation (after the time lag of the system, 3.5 ms), the larger the reduction.

An interesting phenomenon in the experiment is that the feedback control copes better with the head movement of wearers. This can be explained as follows. For the feedback control, the head movement gives same impact on the input of *W* and the error, as both are picked up by the same microphone. Thus no adjustment of *W* is needed. While for the feedforward control, not only the impact is different but the transfer function between them (know as the primary path) may also changes, as they are sensed by different microphones. Thus the adjustment of *W* is required, which slows down the respond.

5. CONCLUSIONS

The main advantage of the feedback control is less complex in hardware due to removal of the feedforward path. Furthermore, the feedback control usually copes better with the head movement of wearers. As expected, the feedback control is limited by the type of noise to be cancelled. It is also less robust. The robustness can be improved however by using longer control filters. In general, if used appropriately, the adaptive feedback control can be equally effective as its feedforward counterpart.

References

[1] P.A.Nelson, S.J.Elliott, Active Control of Sound (Academic Press, London, 1992).

[2] G.J.Pan, et al, 'Application of adaptive feedforward active noise control to hearing protectors', Proc. ACTIVE 95, USA, p1319 (1995).

[3] S.J.Elliott, P.A.Nelson, 'Active noise control', IEEE Signal Processing, p12 (Oct.1993).

Table 1. Effect of the gain error

	Convergence time (iteration)					
	g=2		g=1		g=0.5	
	L2	L64	L2	L64	L2	L64
FB	83	55	21	35	∞	41
FF	21	35	21	35	21	35

*FB: feedback. FF: feedforward.

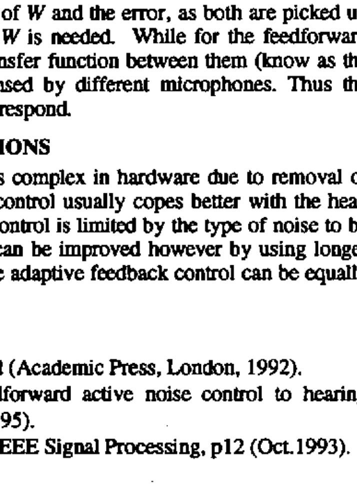

Fig.4. Experimental results with tonal noise.

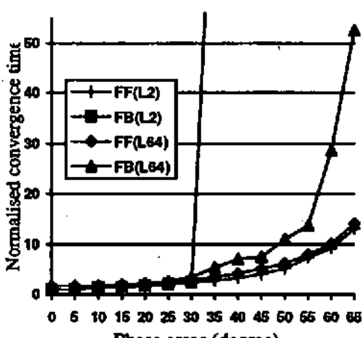

Fig.3. Effect of the phase error.

Table 2. Noise reduction vs autocorrelation.

Noise type	200Hz tone	125Hz octave	250Hz octave	Pink noise
FB(rdB)	30.0	5.0	1.1	0.9
FF(rdB)	30.0	16.0	9.8	9.0
Max autocor. (τ>3.5)	1.00	0.78	0.19	0.13

*rdB: reduction in dB.