Active adaptive noise cancellation with a feedforward open headset

Stéphane Boucher, Bruno Paillard, André L'Espérance
Faculty of Applied Sciences, Université de Sherbrooke, Sherbrooke (Que.) Canada, J1K 2R1

Active noise-reduction headsets currently on the market use feedback-type controllers. This choice of configuration results in the active control of only low-frequency harmonics (usually under 1000 Hz). Moreover, no discrimination is made by feedback algorithms and any predictable acoustic signal, within the control band, is attenuated. Consequently, in addition to cancel the noise, the feedback-type headset will affect speech, alarms and other predictable useful signals, resulting in a discomfort sensation for the user. Also, in order to allow those headsets to attenuate higher frequencies, they are often combined with a passive attenuation shell, with the consequence that the user does not hear any useful signal and has almost the feeling of being « deaf ». A solution to this problem is to use a feedforward controller instead of a feedback controller, thus allowing the headset to :

- Actively cancel high frequency components of a harmonic disturbance,
- Cancel only the disturbance without modifying useful signals,
- Be light and comfortable (the headset can be of « open » type, since no passive attenuation is required).

Description of the controller

The figure 1 illustrates the structure of the controller. Because of the feedforward configuration, the input of the control filter « C » is taken directly at the source of noise. The control filter is adapted in order to attenuate, at the microphone, any signal that is correlated with the disturbance. This way, since useful signals are not correlated with the disturbance, they are not attenuated. The input signal (the reference signal) is filtered by « C » in order to produce an output which will, after having been modified by the transfer function between the speaker and the microphone (secondary-path transfer function « S »), reduce the contribution of the disturbance at the microphone. In order for the system to follow the changes in the transfer function between the primary source and the error microphone (the principal transfer function), « C » has to be continuously adapted. Those changes in the principal transfer function occur when the user moves. The filtered-X LMS [1] is responsible of the adaptation of « C ». This algorithm is frequently used in ANC because it compensates for the secondary-path transfer function effect in the adaptation of the control filter.

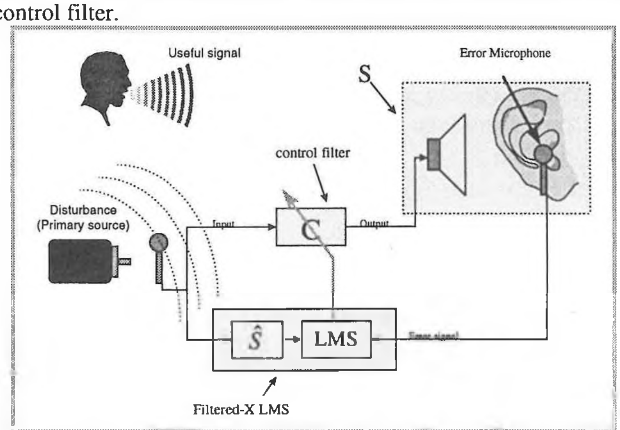

Figure 1 One channel adaptive feedforward controller

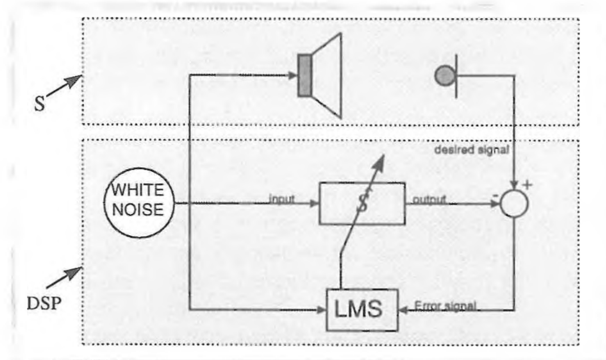

Figure 2 Off-line modeling of secondary-path

Identification of S

The FXLMS algorithm requires knowledge of the secondary-path transfer function « S ». Since this transfer function is different from one user to the other and is dependent of the position of the headset on the head, the modeling of « S » has to take place every time the headset is used and before the control starts. Since « S » is time invariant as long as the user doesn't move the headset from it's initial position on the head, off-line modeling can be used to estimate « S » prior to the control itself (no on-line modeling is necessary).

Figure 2 illustrates the off-line identification process. A finite impulse response filter (FIR) is used to model « S ». This filter is optimized by an LMS algorithm in order to make the output signal of the model as close as possible to the desired signal returned by the microphone. A white noise signal is used as input to the system because it allows an equally fast modeling for the frequency band of interest. In order to obtain a precise enough model of « S » despite the presence of an ambient noise, the duration of the identification has to be automatically adjusted, in regards to the energy of the ambient noise. The louder the ambient noise, the longer the identification phase has to be. In fact, a doubling in the intensity of the ambient noise requires a doubling in the duration of the identification. If the ambient noise is beyond a certain level, it becomes difficult to obtain a good model because of limitations in the computational precision. The range of duration to obtain a good model of « S » goes from 0,1 (low ambient noise) to 1,5 seconds (loud ambient noise).

Implementation

We chose to use two fixed-point TMS320C50 DSPs (one per channel) for the implementation in order to have enough calculation power for the algorithms to run in real time, even at relatively high sampling frequencies. This way, we could expect fast adaptation for the identification and the control process. The choice of fixed-point DSPs was encouraged by the low-cost manufacturing possibility of an eventual commercial product and the low power consumption of this kind of DSP (possibility to power the system from batteries). The down side is that fixed-point DSPs have a more limited computational dynamic and are more difficult to program (normalization) than floating-point DSPs [2].

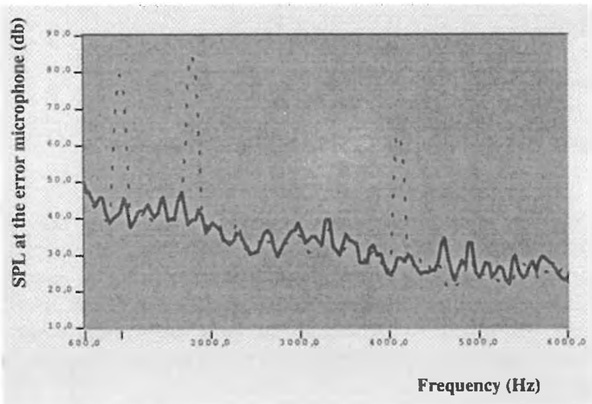

Figure 3 Spectrum of a 3 frequency components signal, before and after control

Performances
The use of two fixed-point DSPs allowed us to reach real time performances for sampling rates up to 20 kHz, thus allowing the control of harmonics as high as 8 kHz. However, even if harmonics up to 8 kHz can be attenuated at the error microphones, the attenuation is only subjectively felt for harmonics under 4 kHz. The reason is that the microphone is too far from the eardrum, compared to the wavelength of the frequency components higher then 4 kHz. To improve this point, we are developing a headset with a microphone that extends to the entrance of the ear canal.
Figure 3 shows the spectrum of a 3 frequency components signal at the error microphone without and with control. We observe an attenuation of 30 to 40 dB per component. The speed at which this attenuation is reached for the 1800Hz component is illustrated in figure 4. It shows a 100 ms segment of the signal at one error microphone when the control is started. We notice that it takes approximately 50 ms for the controller to converge. Perceptually, the transition seems to be instantaneous. This convergence speed corresponds to an adaptation step size small enough to insure almost no sensitivity to useful signals. Actually, even if the controller has a feedforward structure, useful signals can have a negative impact on the adaptation of the coefficients of « C ». A step size too high results in a faster convergence speed but allows useful signals to misadjust the control filter, thus increasing the noise. A compromise was made between the capacity of the controller to follow user's displacements and the misadjustment due to useful signals.

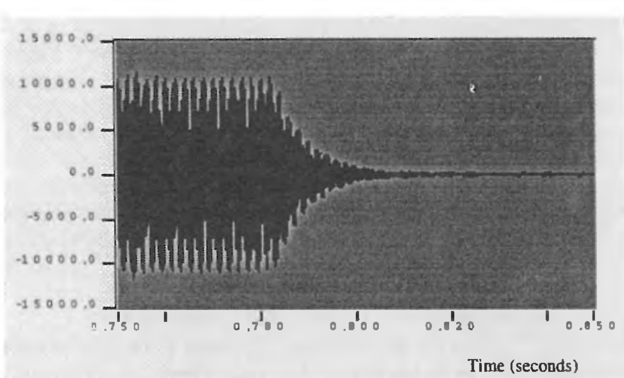

Figure 4 Convergence speed for the 1800Hz component

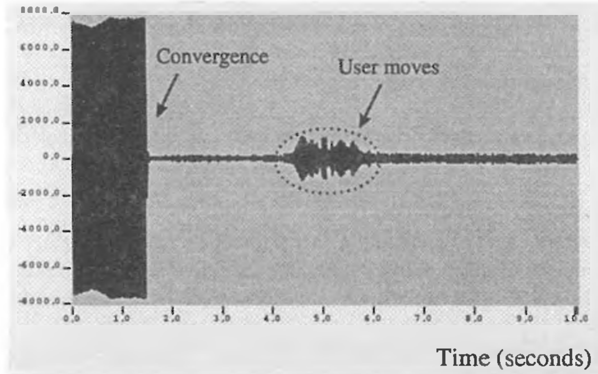

Figure 5 Misadjustment of the controller when the user moves

When the user moves, another misadjustment can sometimes be felt due to the limited tracking speed of the adaptation algorithm. The importance of this misadjustment depends on the convergence speed of the control algorithm, the rapidity of the user's movement, the highest frequency component in the disturbance and the number of frequency components in the disturbance. The highest frequency component present in the disturbance will influence the spatial complexity of the acoustic field. The more the acoustic field is complex, the more the controller has to « follow » rapid changes. Also, the number of frequency components in the disturbance will influence the adaptation speed of the controller. The control filter will adapt more slowly if there are many frequency components in the disturbance. As we can see, the capacity of the controller to follow user's displacement varies with the disturbance characteristics. Figure 5 illustrates the misadjustment of the controller when the user moves at about 2m/s. We notice that despite the misadjustment, the noise level at the microphone while the user moves stays under the initial noise level (before convergence).

Conclusion
The performances achieved in laboratory with the active adaptive feedforward open headset prototype are very encouraging for the future. This kind of headset seems to be a promising solution for many applications, where source noise reduction is not possible.

ACKNOWLEDGEMENT

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[2] DSP Processor Fundamentals, architectures and features, Phil Lapsley, Jeff Bier, Amit Shoham, Edward A. Lee, IEEE Press, 1997