A new concept of active control of transformer noise, Part 2 : Control Algorithm

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The new concept of active envelope for the control of transformer noise[1] needed the implementation of identical independent control units. The controller used for those units is briefly described here along with the problems and solutions associated with the placement of independent units side by side.

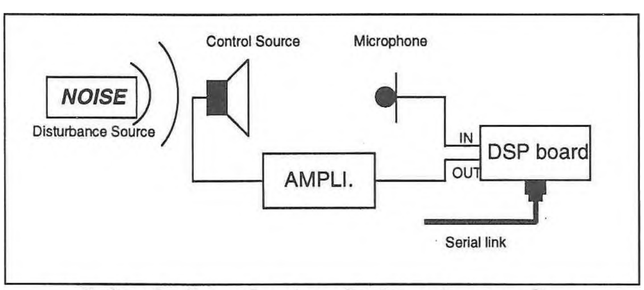
The controller

Figure 1 illustrates an independent control unit. It consists of a mono-channel feedback active adaptive noise control system and it is physically similar to the one presented by Olson and May in 1953 [2]. The choice of a feedback configuration leads to a very simple system (Fig. 1) composed of a few low cost components. The problem to be solved is the generation of the acoustic interference to obtain a reduction of the noise at the microphone and thus, to create a « zone of quiet » around the microphone. In the feedback ANC system shown in Fig. 1, the disturbance is not available because it is intended to be canceled by the microphone signal.

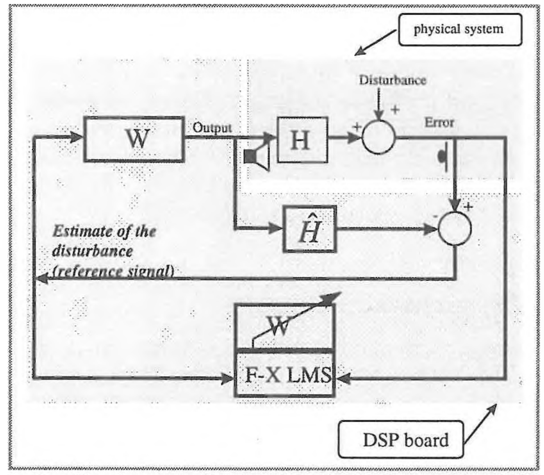
The feedback algorithm that we use (Fig. 2) is said to « use a feedforward approach » [3] because it estimates the disturbance and use it as a reference signal for the control filter. On this figure, « H » represents the transfer function of the plant, which is the relation between the signal sent to the loudspeaker (the output signal y(n)), and the signal measured at the microphone (the error signal e(n)). A model of this transfer function (denoted $\hat{H}$ in figure 2) allow to approximate the contribution of the control signal at the microphone. This approximation is digitally subtracted from the microphone signal, yielding the signal $\hat{d}$ (n), which represents an estimate of the unwanted signal alone (the contribution from the control speaker has been subtracted). This estimate can then be used as the input of the control filter « W ». The output of this filter, the control signal, is sent to the loudspeaker in order to produce the destructive interference.

The adaptive filter « W » is adapted with the filtered-X LMS algorithm [4]. This filter has two purposes :

- It predicts the value of the perturbation signal, some time in the future, from the present and past sample values. The prediction time corresponds roughly to the propagation delay between the loudspeaker and the microphone.



1. ***Single channel active adaptive noise canceller***



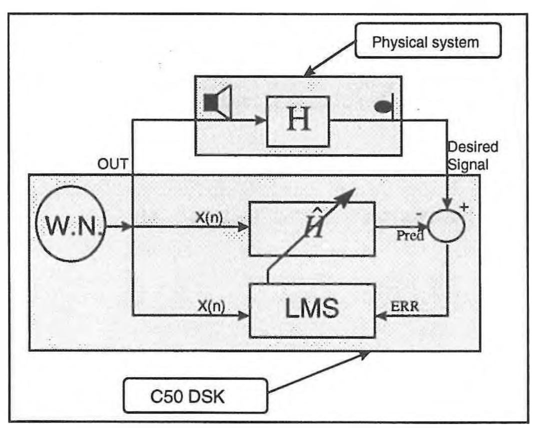
2. ***Feedback controller using a feedforward approach***

- From this predicted value, it generates the control sample to be sent to the control speaker. This generation takes into account the transfer function between the speaker and the microphone.

Actually, both filters are combined into a single control filter W.

Identification of H

The filtered-X LMS and the calculation of the estimate of the disturbance requires both a model of the secondary transfer function « H ». This model can be obtained prior to the control itself with an identification process. This identification process is illustrated in Fig. 3. A finite impulse response filter (FIR) is used to model « H ». This filter is optimized by an LMS algorithm in order to make the signal predicted by the model as equal 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. This white noise is generated by the DSP, and driven to the speaker for a few seconds. After this phase, the control algorithm of figure 2 is engaged, and the filtered-X LMS algorithm adapts W so as to minimize the residual energy at the error microphone.



3. ***Identification process of the plant***

Problems

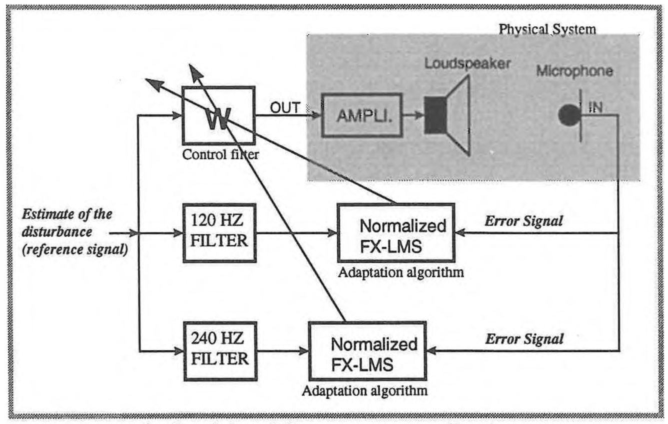
A 40 dB reduction at 120 and 240 Hz can be observed at the microphone when a unit is working on its own (no other units around trying to control). However, the concept of active envelope involves the placement of independent units side by side. The consequence of this tight network of control units is that the control signal of one unit is picked up by the microphone of neighbor units. The error signal of each unit is then influenced by the control signal of neighbors units. Therefore, the estimate of the disturbance is not only function of the disturbance but is also function of the control signal of neighbor units, resulting in a potential instability of the system and in difficulties to control more than one frequency component at a time. Changes have been made to the classical active adaptive control algorithm we used, in order to solve those problems. Spatial normalization as well as frequency normalization of the control algorithm have resulted in a drastic improvement of the behavior of the system.

Spatial normalization

For a non-normalized version of the filtered-X LMS algorithm, the convergence speed of a control unit depends on the energy of the reference signal. A unit converges very fast if the energy of the estimate of the disturbance is high and vice-versa. In our case, many units are placed side by side. Each unit has it's own estimate of the disturbance which is dependent of the noise at it's microphone and will thus converge at it's own speed. The consequence of this lack of uniformity in the convergence speed is a system that can be unstable. We solved this problem with a normalization of the adaptation algorithm (normalized FXLMS)[4]. This « spatial » normalization force the units to converge at the same speed, no matter the level of noise present at each unit.

Frequency normalization

While the spatial normalization allowed to standardize the convergence speed between the control units, it didn't allow the convergence speed between the frequency components to be uniform. Actually, the convergence speed for one frequency component depends on the energy of the reference at this frequency. The control filter is adapted faster for components with more energy in the reference and vice-versa. For example, if the energy of the frequency component at 120 Hz is 10 dB (10 times) superior to the energy of the frequency component at 240 Hz, the control filter will be adapted 10 times more rapidly for the 120 Hz component than for the 240 Hz component. This would not be a big problem if the ratio between the components could stay between 0 and 20 dB. It is usually the case when a unit works on its own since the reference signal represents well the disturbance. However, the placement of many units side by side distorts the estimate of the disturbance in such a way that the ratio between frequency components is not anymore the same as the disturbance itself. Moreover, this distorted ratio can be several dB over/under the real ratio, resulting in a convergence speed that can be very different from one frequency component to the other. Actually, we could observe for some control units a distort ratio of approximately 40 dB which resulted in a convergence speed 100 times faster for the most energetic component. In that case, the less energetic component could take several hours to converge. Sometimes, it will not converge at all, du to the limited calculation precision of the DSP.



4. Spatial and frequency normalization

The solution to this problem is to « frequency normalize » the control algorithm. This strategy, illustrated in Fig. 4, consist in making the convergence speed independent of the frequency. In our case, since we want to control two different frequencies (120 and 240 Hz), we use two filters. Each one filters the reference signal in order to separate the two frequency components. The two output signals then serve as inputs for two normalized FXLMS, running in parallel. Since the FXLMS algorithm is normalized, the control filter will be adapted in order to make the two frequency components converge at the same speed.

Conclusion

Spatial and frequency normalization has been applied to the independent control units in order to improve the stability of the overall system and to facilitate the control of more then one frequency component at a time. As anticipated, a drastic improvement of the behavior of the system resulted.

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