Implementation of decentralized active control of power transformer noise

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1. Introduction

The active noise control of sinusoidal sound in free space has received considerable attention in the last few years. In a number of practical situations, the sound radiated in free space by extended sources needs to be globally reduced. This is the case of electrical power transformers, for example [1]. Nelson and Elliott have demonstrated that sound power attenuation can be achieved by active control with secondary sources, loudspeakers for example, located in the near field of an extended radiator (also called the primary source) [2]. An effective global attenuation of the sound power implies that the secondary sources must be in the vicinity of the radiator: the distance between each secondary source and the radiator must be less than $\frac{\lambda}{4}$ where $\lambda$ is the acoustic wavelength [3]. The secondary sources are driven by a controller in order to cancel the sound field measured by error microphones for example. An effective global attenuation of the sound power can be achieved by independent active controllers with collocated loudspeakers and microphones located in the near field.

The figure 1 presents experimental implementations of 24 independent control units on a real power transformers. A noise reduction of 10 dB of the first harmonics (120 Hz) have been obtained [9]. The main problem of this approach is to design a disposition of loudspeakers and microphones such that the active noise control system is efficient. A practical method is proposed in order to predict the stability and the attenuation of the system. Several experimental results with 7 control units are presented.

Figure 1 : Experimentation of decentralized active noise control on power transformer.
2. Description of the implementation

2.1 Description of the control strategy

The sound power attenuation is based on the creation of cancelling points in front of the power transformer.

In order to reduce the noise emitted at 120 Hz by a vibrating surface of 3m x 4m, 18 units are implemented as illustrated in figure 2. Each independant control unit, denoted $m$, it is composed of one loudspeaker, one microphone and one controller as illustrated in figure 2. The microphone located in front of the loudspeaker delivers the error signal, $e_m$, which results from the interference between the sinusoidal primary noise radiated by the power transformer, $d_m$, and the sinusoidal anti-noise radiated by the loudspeaker. In order to cancel the sinusoidal component of the error signal, $e_m$, the independent controller drives the loudspeaker with the command signal $u_m$.

![Figure 2](image-url) (left): typical location of 18 independent units on the power transformer, (right): one independent unit, $m$, with one loudspeaker, one microphone and one controller.

When the control is perfect, $e=0$, each cancelling point has the effect of creating a local zone of quiet; these individual zones of quiet must overlap in order to significantly reduce the noise downstream the cancellation points. By numerical simulation it is possible to evaluate different configurations in order to determine the necessary number of units. The figure 3 presents a typical result without active control and with optimal command signals for 18 units, $x_L=20$ cm, $x_m=45$ cm, at 120 Hz. In far field, 15 dB of attenuation is obtained.

![Figure 3](image-url) (left): the primary noise is presented, (right) with active noise control.
However, the resulting sound power attenuation depends on the disturbance frequency and the arrangement and density of secondary sources and the distance from cancelling points to the radiator \((x_m)\) and secondary sources \((x_L)\) [4]. The number of units and the distance \(x_m-x_L\) dramatically increase with the frequency. For a given implementation, the main practical problem is to control efficiently the system in order to obtain the perfect cancellation at the microphones.

2.2 Description of the control system

A considerable processing power is needed for multichannel systems: for \(M\) secondary sources and \(M\) error microphones the system is described by \(M^2\) loudspeaker-microphone transfer functions. A technique called decentralized control has been developed in the last years to avoid the processing power problem and facilitate the hardware and design of the control system [5]. It consists to implement an independant control system for each unit formed by a secondary source and its corresponding cancellation point. Because each independant controller does not take into account the other secondary sources only the \(M\) direct transfer functions from each secondary source to the corresponding cancelling point are needed. The main drawback of such a decentralized control approach is the risk of instability. Roughly speaking, each independant controller can work against the others.

In practice, multichannel feedforward [6] or feedback [7] adaptive controllers are typically implemented. Previous work has proved that decentralized architectures can be used to efficiently implement adaptive multichannel feedforward control [8]. However, feedforward controller needs an additional signal correlated with the primary noise. On the other hand, in the case of a periodic primary noise, a feedback controller is much attractive because it requires only sound pressure measurements at the cancelling points.

Because of the possibly changing properties of the controlled system (the loudspeakers, the microphones and the propagation path) and the primary noise, the anti-noise emitted by the secondary sources must be perfectly and continuously tuned by the controller in order to obtain the perfect destructive interference at the cancelling points. This leads to an adaptive control system. The control architecture implemented in each controller, presented in figure 4, is an adaptive Internal Model Control (IMC) [10] because it uses an internal model \(\hat{h}\) of the loudspeaker-microphone transfer function in order to estimate the primary noise, \(d\).

The filtering of the estimated primary noise, \(\hat{d}\), by the control FIR filter, \(\Gamma\), generates the command, \(u\). A normalized x-LMS algorithm [11] adapts the coefficients of the control filter.

![Figure 4: block diagram of an independant control unit.](image-url)
However, since each error microphone observes the anti-noise generated not only by the corresponding loudspeaker, but also by all other control loudspeakers there is an error in the estimation of the disturbance. A way to increase the stability is to improve the estimation by each unit of the disturbance noise emitted by the primary source and/or to limit the effects of the other units. Under the assumption that units locally radiate the same anti-noise, the internal model of each unit can be magnified in order to include the contributions of all nearest units: \( \hat{h} = \alpha h \), with \( \alpha \geq 1 \) a real scalar.

Experimental implementations on power transformers have demonstrated the efficiency of the decentralized adaptive feedback approach [9]. Nevertheless, stability problems due to the decentralized control strategy appeared during preliminary experiments.

### 2.3 Analysis of the stability

In adaptive feedback theory, instability of the control system may have two causes: a non-convergence of the adaptation process or an unstable feedback loop.

First, the feedback loop is assumed stable and the slow adaptation process behaviour of one controller is examined. In other words, the time-scale of the adaptation process is assumed very slow in comparison to the dynamics of the feedback loop. In this case, the behavior of the filtered-x LMS algorithm in the time domain can be evaluated in the frequency domain [12]. It can be demonstrated that it is always possible to find a value of \( \mu_0 \) which ensures the convergence of the update equation in the frequency domain to its optimum value, denoted \( \Gamma_{opt}(j\omega_0) \). Thus, because the convergence of the adaptation is always ensured, the stability of the decentralized adaptive feedback control system must be a property of the feedback loop.

To analyze the relation between the adaptation process and the feedback loop stability, a complex value called performance index is defined as follow:

\[
\beta = \frac{\Gamma(n, j\omega_0)}{\Gamma_{opt}(j\omega_0)}
\]

Assuming the stability of the feedback loop of the control system the closer the performance index is to 1, the closer the control filters are to their optimal values, the better the global attenuation at the microphones.

In order to evaluate the stability of the feedback loop, the Nyquist stability criterion is classically used [13]. It allows to define the maximal index of performance, \( |\beta_m| < \beta_{max} \) for all \( m \), such that the feedback loop is stable. If the maximal index of performance is equal to 1, a perfect cancellation can be reached at the error microphones. However, if the index of performance is inferior than 1, the adaptive decentralized feedback control will become unstable.

Thus, for a given physical implementation of the control units, it is possible to predict the maximal level of attenuation at the microphones as a function of the index of performance.
5. Simulations and experiments with 7 units

5.1 Simulations with 7 units

We now assume a control system composed of $M=7$ units as described in figure 5. The vibrating structure is a square plate (2.4m x 2.4m) of 1 mm thick. Two loudspeakers located behind the plate are used to generate the primary acoustic field.

![Configuration of the control simulation.](image)

Table I summarizes the maximum values of the performance index $\beta_{\max}$ for which this control system is stable according to the stability conditions deduced from the Nyquist Criterion condition for different distances $r$ between the control loudspeaker and the error microphone of each unit. A disturbance frequency of 240 Hz is considered in these simulations. The distance between any given unit and its neighbors was taken to be 20 cm. Several values of the gain parameter $\alpha=1,2,3,4$ and 5 were considered for the calculation of $\beta_{\max}$ according to the Nyquist Criterion condition. Without a correction gain ($\alpha=1$) the Nyquist criterion predicts the instability for $\beta_{\max}>0.33$. The introduction of a correction gain $\alpha=4$ allows to increases the stability, and thus the distance between the loudspeaker and the microphone.

<table>
<thead>
<tr>
<th>$x_m-x_L$ (cm)</th>
<th>$\alpha=1$</th>
<th>$\alpha=2$</th>
<th>$\alpha=3$</th>
<th>$\alpha=4$</th>
<th>$\alpha=5$</th>
</tr>
</thead>
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<tr>
<td>12</td>
<td>0.52</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>20</td>
<td>0.39</td>
<td>0.94</td>
<td>0.96</td>
<td>0.96</td>
<td>0.95</td>
</tr>
<tr>
<td>30</td>
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<td>0.71</td>
<td>0.92</td>
<td>0.93</td>
<td>0.93</td>
</tr>
</tbody>
</table>

Table I: maximum values of the performance index $\beta_{\max}$ for different values of the distance between each loudspeaker and microphone an different values of the gain $\alpha$ (240 Hz).
5.2 Experiments with 7 units

A set of experimental results are now presented to verify the accuracy of the theoretical results on the convergence of the adaptation, the stability of the feedback loop and the control performance.

As an initial step, the physical matrix plant $\mathbf{h}$ was experimentally identified using 128-order FIR filters to model the transfer function between each control loudspeaker and each error microphone. A discrete time-domain computer simulation of the IMC feedback structure shown in figure 2 was implemented under Simulink for each control unit. The time-domain simulation used the experimental matrix plant $\mathbf{h}$, as well as sampled values of the primary disturbance measured at the 7 error microphone locations to represent the disturbance $d$ at each error sensor. The time-domain simulation was then executed and stopped at various increasing times during the adaptation in order to assess the convergence and stability of the control system. The sum of the squared error signals obtained at the microphones is plotted as a function of time in figure 6 for $\alpha=1$ and $\alpha=4$; the oscillations of the error signal around a constant value indicate that the adaptation process has been stopped. The results of figure 6 show that the control system rapidly reaches the instability for $\alpha=1$, whereas it remains stable for $\alpha=4$. Based on these time-domain simulations, the values of the calculated performance index for a typical unit (the unit 1) and the averaged attenuation as a function of time during the adaptation process indicate that the independent controllers are able to reach their optimum value and almost perfectly reject the disturbance at each error sensor. Therefore, time-domain simulations based on the experimental matrix plant predict stability of the 7-units system. The averaged sound pressure attenuation obtained at the error microphones from the time-domain simulation is presented as a function of the performance index on table II.

| Performance index $|\beta_1|$ | 0.14 | 0.23 | 0.33 | 0.40 | 0.52 | 0.63 | 0.72 | 0.82 | 0.91 | 1.00 |
|---------------------|------|------|------|------|------|------|------|------|------|------|
| Attenuation (dB)   | 2    | 3    | 5    | 7    | 9    | 11   | 14   | 18   | 24   | 53   |

Table II: values of the attenuation versus the performance index.

Figure 6: attenuation at the microphones obtained from the time-domain simulation for $\alpha=1$ (dashed line) and $\alpha=4$ (solid line) with different frozen times of the adaptation process.
Finally, the control was experimentally tested with 7 units and appeared to be stable. The steady-state values of the local performance index for each unit and the steady-state attenuation measured at each error microphone are summarized in Table III. The experimental steady-state values of $|\beta_m|$ show that one unit actually reached its optimum ($|\beta_6|=1$), whereas the others reached slightly sub-optimal values ($|\beta_m|<1$ for $m \neq 6$). The experimental values remain however close to the maximum values deduced from the simulations with simulated transfer functions ($|\beta_m|\leq 0.94$) or identified transfer functions ($|\beta_m|\leq 1$). The reason why some units reach only sub-optimal values of $|\beta_m|$ in the experiments is not resolved yet.

<table>
<thead>
<tr>
<th>Unit $m$</th>
<th>Sound pressure</th>
<th>Attenuation with control</th>
<th>Modulus of $\beta_m$</th>
<th>Phase of $\beta_m$ (deg)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>79</td>
<td>15</td>
<td>0.89</td>
<td>-5.0</td>
</tr>
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<td>2</td>
<td>79</td>
<td>20</td>
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<tr>
<td>3</td>
<td>79</td>
<td>16</td>
<td>0.85</td>
<td>-1.7</td>
</tr>
<tr>
<td>4</td>
<td>79</td>
<td>12</td>
<td>0.83</td>
<td>-5.6</td>
</tr>
<tr>
<td>5</td>
<td>79</td>
<td>29</td>
<td>0.81</td>
<td>-6.0</td>
</tr>
<tr>
<td>6</td>
<td>79</td>
<td>25</td>
<td>1.00</td>
<td>-3.5</td>
</tr>
<tr>
<td>7</td>
<td>78</td>
<td>28</td>
<td>0.95</td>
<td>-12.6</td>
</tr>
</tbody>
</table>

Table III: steady state values of $\beta_m$ and the corresponding attenuation at the microphones for a system of 7 control units at 240 Hz and $\alpha = 4$.

**Conclusions**

This paper has presented the analysis and implementation of a decentralized adaptive feedback active noise control system of sinusoidal sound in free space. The active control system consists of an arrangement of multiple independent control units (a control loudspeaker, an error microphone and a controller) which act each to create a point of zero sound pressure at the error microphone location. The main advantage of such a decentralized strategy is an important economy in terms of processing power for large systems, as compared to a centralized control strategy. It was shown that, while it is possible to guarantee the convergence of the decentralized feedback to the optimum solution, global stability is not always satisfied. In order to quantify both the convergence and global stability of the feedback loop, a "performance index" $\beta$ was introduced. It was shown that a gain parameter $\alpha > 1$ improves the global stability of the control system.

The analytical tools developed proved to be useful in assessing the stability of the control system before it is experimentally implemented. The experiments also clearly demonstrate the effectiveness of decentralized control in situations were global stability is satisfied. Ongoing work involves the monitoring and control of the performance index of individual units in order to stop the convergence before instability is reached; also, the extension of decentralized control to structural systems is presently investigated.
References