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ACTIVE NOISE CONTROL IN A TWIN-ENGINE PATROL BOAT

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1. INTRODUCTION

This paper demonstrates the practical installation of a multiple reference, adaptive control system designed to reduce noise at the driver and passenger seat in a patrol boat. The dominating noise in the drivers cabin at a speed below planing was produced by the two main engines, each a 500 kW V12 engine. The spectrum was narrow band with a clearly dominating fundamental frequency. As the two engines were not synchronized, a quite annoying beating was produced in the cabin. With the two engines at idle, the fundamental frequency was approximately at 37 Hz with a linear sound pressure level of 112 dB. The background for this project is outlined in [1].

Due to the low frequency character of the noise, the A-weighted sound pressure level was fairly low, around 85 dBA. For this reason, the primary concern for reducing the noise levels was to increase comfort and safety, rather than protecting the driver from hearing loss. The high levels of low frequency noise efficiently masks higher frequencies, e.g. frequencies within the speech frequency range, which makes acoustic communication almost impossible. Thus, to minimize the risk of misunderstanding, or not hearing at all, it was necessary to attenuate the low frequency noise as much as possible.

2. THE CONTROL SYSTEM

The control system was based on a Loughborough Sound Images (LSI) PC/C31 PC-board containing a Texas Instruments TMS320C31 floating point signal processor. This board was combined with a 32 channel analog input board, a 16 channel analog output board (both from LSI) and a custom built tacho input board.

The control algorithm was implemented as a filtered-x, twin reference LMS-type algorithm with a somewhat modified nomalization procedure. Each reference signal had a separate controller whose outputs were summed together at the output terminal. Using matrix notation, the driving signal, $y_l(k)$, for loudspeaker l at time instant k can be written

$$y_l(k) = \mathbf{X}_1^T(k)\mathbf{W}_{1l} + \mathbf{X}_2^T(k)\mathbf{W}_{2l}, \tag{1}$$

where $\mathbf{X}_1$ and $\mathbf{X}_2$ are the input reference vectors for reference signal 1 and 2 respectively and $\mathbf{W}_{1l}$ and $\mathbf{W}_{2l}$ are the control filter weights for loudspeaker l and for the two references. The multiple input, multiple output LMS control algorithm has been described by other authors, e.g. [2] and [3], so we may directly write the controller update algorithm for loudspeaker l and reference r as

$$\mathbf{W}_{lr}(k+1) = \mathbf{W}_{lr}(k) - 2\mu\mathbb{E}^T\mathbf{X}^T_{F,lr}, \tag{2}$$

which is the multiple input filtered-X LMS algorithm. In equation (2), $\mathbf{X}_{F,lr}$ is a matrix, where each column contains a filtered reference vector, where the "filters" are given by the acoustic frequency response functions from loudspeaker l to each microphone, called the control paths. Thus, with M microphones in the system, $\mathbf{X}_{F,lr}$ will have M columns. It is important to notice that if a multiple output controller is desired, the update algorithm for each loudspeaker is identical to equation (2), with the noticeable exception that the "filters" are different. The only connection between the controllers occurs through the reference input, $\mathbf{X}$, and the error input, $\mathbf{E}$, signals.

With multiple input, multiple output systems of growing order it is increasingly important to use the correct normalization of the convergence factor for the controller. Widrow [4] has shown that for a single input, single output standard LMS controller the value of μ is limited by $0 \leq \mu < 1/\lambda_{max}$, where λ_{max} is the maximum eigenvalue of the auto-covariance matrix, $\mathbb{R}$ for the reference input signal. These limits are usually replaced by the requirement that $0 \leq \mu < trace(\mathbb{R})$, since $trace(\mathbb{R})$ is more easily calculated. For a filtered-X controller, $\mathbb{R}$ as used in the limit for μ is replaced with $\mathbb{R}_F$, which is calculated as

$$\mathbb{R}_F = E\left[\mathbf{X}_F\mathbf{X}_F^T\right], \tag{3}$$

which means that the convergence factor is limited by the power in the filtered reference signal.

In a multiple input, multiple output controller, the global auto-covariance matrix for the controller is calculated as the sum of the covariance matrixes for all filtered reference signals within the controller. Thus, for the control path between loudspeaker l and microphone m, the auto-covariance matrix becomes

$$\mathbb{R}_{F,lmr} = E[\mathbf{X}_{F,lmr}\mathbf{X}^T_{F,lmr}], \tag{4}$$

where r is the reference index. The auto-covariance matrix for one loudspeaker controller is calculated as

$$\mathbb{R}_{F,l} = \sum_{r=1}^{R}\sum_{m=1}^{M} E[\mathbf{X}_{F,lmr}\mathbf{X}^T_{F,lmr}]. \tag{5}$$

The global auto-covariance matrix for the complete controller is finally obtained as

$$\mathbb{R}_F = \sum_{l=1}^{L}\mathbb{R}_{F,l} = \sum_{l=1}^{L}\sum_{r=1}^{R}\sum_{m=1}^{M} E[\mathbf{X}_{F,lmr}\mathbf{X}^T_{F,lmr}]. \tag{6}$$

Thus, using global normalization, the controller for one loudspeaker in a multiple input, multiple output, multiple reference system is obtained by rewriting equation (2) as

$$\mathbf{W}_{lr}(k+1) = \mathbf{W}_{lr}(k) - \frac{2\mu\mathbb{E}^T\mathbf{X}^T_{F,lr}}{trace(\mathbb{R}_F)}. \tag{7}$$

Using this normalization, the allowed values for μ are limited to

$$0 \leq \mu < 1. \tag{8}$$

Each loudspeaker is driven by a controller which is identical to all other loudspeaker controllers and working with the same input signals. In fact, the only difference in weight updates between different controllers occurs due to differences in the control paths. Since we are using global normalization, controllers with good acoustic coupling to the microphones will have large filtered reference signal power and therefore converge more rapidly, while controllers with lower acoustic coupling will converge more slowly, or not at all.

The algorithm that was implemented for the present project uses individual normalization, which has proven to increase the convergence characteristics for the global controller. This implies a slight modification to equation (7) according to

$$\mathbf{W}_{lr}(k+1) = \mathbf{W}_{lr}(k) - \frac{2\mu\mathbb{E}^T\mathbf{X}^T_{F,lr}}{trace(\mathbf{R}_{F,l})}, \tag{9}$$

where $\mathbf{R}_F$ in equation (7) is replaced by $\mathbb{R}_{F,l}$ in equation (9). This means that the convergence factor for each loudspeaker is normalized with the filtered reference signal power within that particular controller. The convergence for each controller will then be optimized in terms of convergence speed.

3. ACTUATORS AND SENSORS

The setup in the cabin used four loudspeakers to create a secondary sound field. The loudspeakers were designed to be able to produce high levels of sound with frequencies down to 35 Hz. The final design was a band pass enclosure with a volume of 16 liters, equipped with two 12cm woofers mounted in an iso-baric fashion. At very high sound pressure levels, the high air velocity in the ports generated some turbulence which could have been avoided by using a flange at the port ends.

The microphones used were standard cheap (1$) electret microphones that were connected to a custom designed and built microphone amplifier containing all the necessary signal conditioning. The microphone amplifiers were also designed to feed the microphones with power for proper operation.

Each engine was fitted with a tacho unit for reference signal generation within the controller. Several different tacho sensors were used during the test period and the final choice fell on optical sensors. The tacho signal were fed to a custom designed PC-board that was interfaced to the processor board through an LSI specific bus, called DSPLINK.

The reference signals to the controllers were generated by reading samples from a table. By letting the tacho signal drive a counter with the same period as the reference table, the proper table index was obtained by reading the counter on the tacho-board. Since a multiple reference controller cannot be synchronized to all reference sources, this controller used a fixed sampling rate for the analog interfaces. The fixed sampling rate was 1 kHz, while the tacho frequency could vary between 400 and 1600 Hz, implying that the reference signal had to be resampled. To avoid aliasing due to periodicity in the sampling process, the reference table was oversampled with approximately 6 times the fixed sampling rate, followed by a digital low pass filter and decimation to the proper rate.

4. RESULTS

As explained above, the control system was designed for four loudspeakers, eight microphones and two references. The loudspeakers were placed at the floor, in the four corners of the cabin. The microphones were placed in groups of four in the vicinity of the drivers seat and the navigators seat.

The attenuation was evaluated at six microphone positions, separated from the control microphones. The curves presented here were measured approximately in the middle of the cabin. Figure 1 shows the sound pressure level at the evaluation position with the two engines running at idle speed. The controlled order is the 3'rd, with a frequency of approximately 37 Hz. The beating between the engines is about 1 Hz. With the controller on, the controlled frequency is attenuated more than 20 dB and the beating is completely gone.

Figure 2 shows the attenuation with one engine running at idle and the other running at approximately 1100 rpm, producing two harmonics at 37 Hz and 55 Hz respectively. The 37 Hz tone is efficiently reduced, while the 55 Hz tone is just slightly reduced with about 7 dB. There are several reasons for this, the two most important being controllability/observability and the fact that the amplitude of this harmonics is close to the overall noise floor.

5. CONCLUSIONS

A multiple reference adaptive control system has been tested in a twin engine patrol boat. Measurements have shown that the control system was able to efficiently reduce the low frequency tonal noise from the engines. The beating between the engine tones was almost entirely eliminated, since the controller could handle the two engines separately.

6. REFERENCES

[1] P.G. Sjösten, "Volumetric noise control in a twin engine patrol boat", Report, HK/R 1996
[2] P.A. Nelson & S.J. Elliott, "Active Control of Sound", (Academic Press, 1992)
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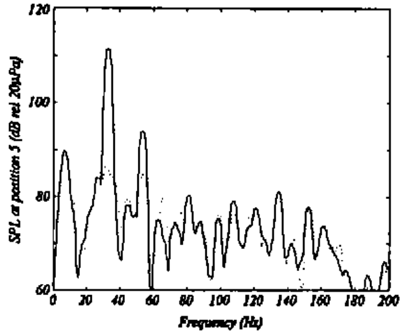

Figure 1: The two engines running at idle, with a beating frequency of about 1 Hz. Solid line: Control off. Dotted line: Control on.

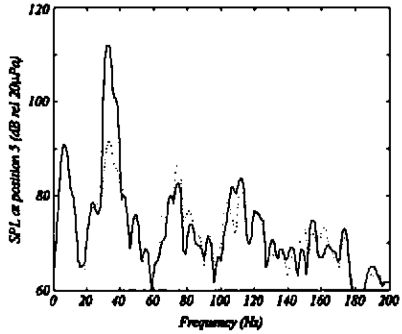

Figure 2: One engine running at idle and one engine running at 1100 rpm. Solid line: Control off. Dotted line: Control on.