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A FAST ADAPTIVE FEEDBACK-CONTROLLER FOR ACTIVE NOISE CONTROL

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1. INTRODUCTION

In a duct (of infinite length) sound propagates in the form of plane waves if the signal does not contain frequency components above the duct's cut on frequency. In this case a single channel Active Noise Control system (ANC) is sufficient because the effective noise reduction at the sensor position is equal to the reduction over the whole cross section.

In this paper experiments on a duct are described in which the controller is build as an adaptive feedback-controller. A Fast Recursive Least Squares algorithm (FRLS) is used for the adaptation process. In numerical simulations, e.g. [4], and experiments with feedforward systems, e.g. [5], these algorithms have shown a better performance than the known LMS-algorithm in many cases. Therefore the performance of the FRLS-controller is compared with a LMS based controller.

2. DESCRIPTION OF THE EXAMINED CONTROLLER

Both control systems follow the principle of Internal Model Control (IMC). For this an estimated model of the so called secondary path is necessary. The secondary path describes the transfer function between the digital output side of the controller and its digital input side. Within the IMC the output signal of the controller is filtered with the model and subtracted from the sensor signal. By doing so the original noise signal is reconstructed if the model of the secondary path is perfect. In any other case one only gets an estimate of the noise signal whose quality depends on the quality of the model. The task of the controller can be understood as a forward predictor (for more information about IMC see e.g. [1]).

The adaptation process also requires a model of the secondary path [6]. For algorithms like the LMS the common approach is the Filtered-X method in which one assumes that the transfer function of the adaptive filter changes only slowly in time [3]. But this assumption is not fulfilled by the FRLS-algorithms. For this reason a different approach is used by the FAIFC-scheme [4] which stands for "Fast Algorithms In Feedback Control". Its main idea is the decoupling of the adaptation process and the controller. The adaptive filter is used in a forward prediction configuration and its filter coefficients are copied to the controller filter in the IMC configuration.

3. THE EXPERIMENTAL SETUP

In Fig.1 a sketch of the experimental setup is shown. The duct in our experiments had a length of 4,80m and a rectangular cross section of 30cm×13cm. Therefore the cut on frequency f_g lies under 600Hz. The loudspeaker S1 is

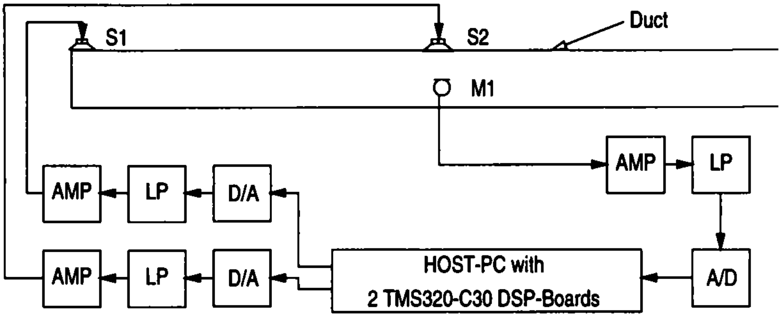


Figure 1: The experimental setup at the duct

the primary source for the noise signal and the loudspeaker S2 the secondary source of the ANC-system. The sensor microphone M1 was placed in the middle of the duct's cross section opposite to the center of S2. For the signal processing, signal generation and data collection we used a PC with two TMS320C30 DSP-boards. The AD/DA converter had a resolution of 16bit. All signals were filtered by analogous low pass filter and amplified. The cut off frequency of the anti aliasing filter was set to 400Hz and the sampling frequency f_s=1KHz. Also the low pass filtering of the test signals guaranteed that only plane waves were able to propagate in the duct.

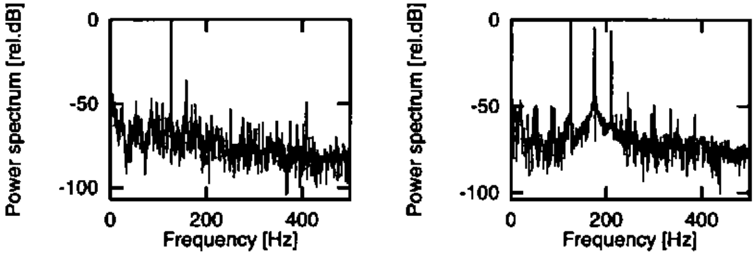
Before the start of the ANC-systems the secondary path was identified in a system identification and modeled by a FIR-filter. An FIR-filter with N_c=200 coefficients was sufficient as model for our experiments. Further the transfer function of the secondary path stayed time independent during the experiments so that an update of the model was unnecessary.

Within the FAIFC-controller the SFAEST-algorithm was used for the update of the filter coefficients [2]. The numerical effort for the FAIFC/SFAEST systems is $4N_w$ higher than for the LMS based system where N_w is the number of coefficients of the adaptive filter. The LMS-based system used a normalized version of the LMS. The algorithms were always initialized in a way that an "optimal" adaptation/control performance could be observed.

4. RESULTS OF THE EXPERIMENTS

The following Figures show the results of experiments with the two adaptive systems for two different test signals. Test signal 1 is a stationary sine tone with a frequency of 125Hz whose power spectrum is shown in Fig.2(a). Test signal 2 is a mix of three stationary sine tones with frequencies at 125Hz, 175Hz and 210Hz (Fig. 2(b)). The filter length for the adaptive filter was set to $N_w = 5$ for the single tone and $N_w = 20$ for the tone mix.

Fig.3(a) and Fig.3(b) show the power spectrum of the residual error for the

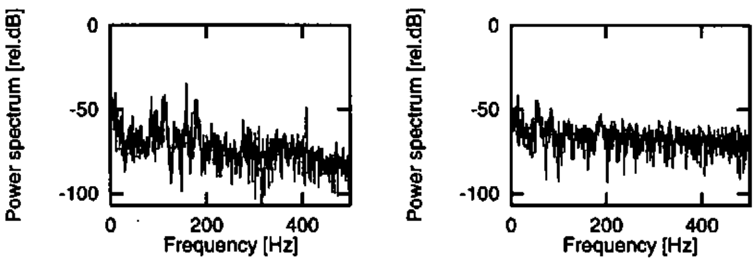


(a) Test signal 1: Pure Tone (b) Test signal 2: Tone Mix

Figure 2: Power spectra of the two test signals

two controller with test signal 1. The spectra were calculated after 10000 samples and both controller have compensated the test signal 1 at this time.

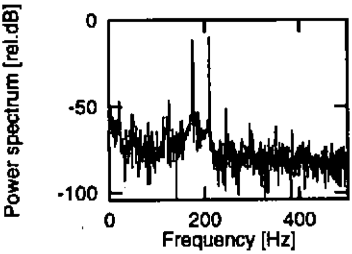
For the test signal 2 the figures of the residual error spectra show a differ-



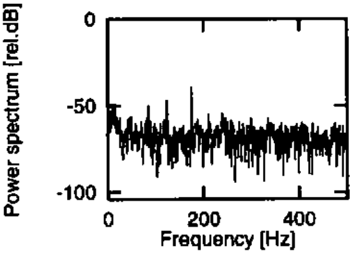
(a) Test signal 1: LMS (b) Test signal 1: FRLS

Figure 3: Power spectra of the residual error signal for both controller after 10000 Iterations (for test signal 1)

ent picture. The FRLS-controller has compensated all spectral components of test signal 2 (Fig.4(b)). The LMS-controller, however, has only reduced the first component after 10000 samples (Fig.4(a)) and needed more than 300000 samples to control all spectral components. But it must also be



(a) Test signal 1: LMS



(b) Test signal 2: FRLS

Figure 4: Power spectra of the residual error signal for both controller after 10000 samples (for test signal 2)

mentioned that the FRLS controller used a shorter model for the secondary path with $N_c = 180$ because with $N_w = 20$ the controller was not capable of real time operation.

5. SUMMARY AND CONCLUSION

The experiments on the duct lead to the following conclusions. It was shown that adaptive feedback-controller can be used for active noise control with a variety of adaptive algorithms.

In line with numerical simulations and experiences with feedforward controller, the convergence of the FRLS based FAIFC-controller was found to be widely independent of the statistics of the test signal. The LMS algorithm is relatively simple to implement but takes a long time to converge if the autocorrelation matrix of the filtered reference signal has a large eigenvalue spread, as it is the case of two sinusoids with dissimiliar magnitudes.

ACKNOWLEDGEMENT

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