Implementation of an Online Feedback-Path-Modelling Active Noise Control System

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1.5 Implementation of an Online Feedback-Path-Modelling Active Noise Control System

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1. Introduction

Active Noise Control (ANC) systems have proved to be a very efficient way to reduce low-frequency acoustic noise. On this domain, passive techniques like enclosures, barriers and silencers tend to be relatively large, costly and ineffective.

Although many studies and articles have been published in order to improve performance and stability, the implementation of a real-time, stable and robust system still faces several theoretical and practical challenges.

2. Problem Description

A schematic diagram of a single channel feedforward ANC system is shown in Figure 1.5-1.

While the acoustic pressure produced by the noise source propagates down the duct, a small part of it is captured by the reference microphone Mic_r and so feeds the ANC system. The main function of the system is to filter the noise s(n) in order to get an anti-noise signal y(n) being similar to a replica of the original noise but with inverted phase. This causes destructive interference with the noise r(n) when injected back into the duct through the loudspeaker L_a shown in Figure 1.5-1. For the sake of simplicity the wave property of the signals has been disregarded and the noise has been depicted with a sinusoidal waveform.

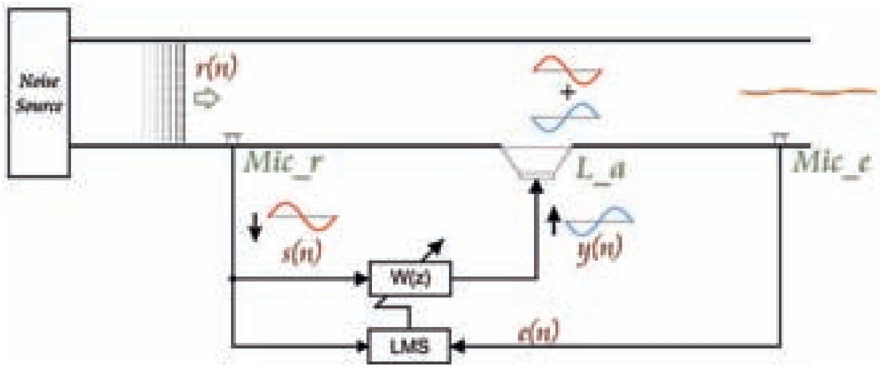

Fig. 1.5-1: *Single channel feedforward ANC*

The adaptive LMS-algorithm (Least Mean Squares) uses both the reference signal s(n) as well as the error signal e(n) resulting from the destructive interference between noise and anti-noise to adjust the coefficients of the filter W(z) in order to achieve the best system's performance, that is, the lowest noise level at the output of the duct. However, as the anti-noise signal y(n) tends to propagate not only downstream in direction of the error microphone Mic_e but also upstream towards Mic_r, it contaminates the reference signal s(n), degrading the performance of the system.

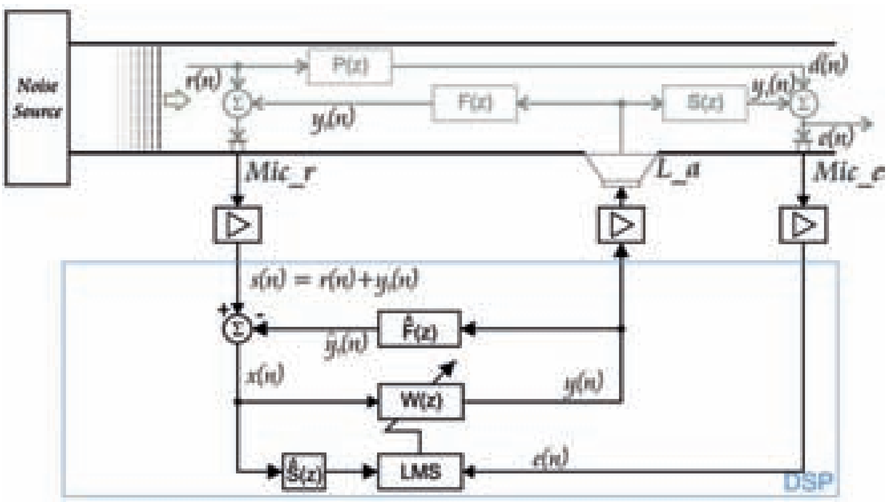

Fig. 1.5-2: *FxLMS*

Another problem is the fact that the signal y(n) is delayed and filtered by the secondary path S(z), as it travels from the anti-noise loudspeaker to the error microphone, being temporally misaligned with the reference signal s(n) in the LMS algorithm (Figure 1.5-2).

A solution for overcoming this problem was proposed by [1] and consists of the implementation of the *Filtered-X Least Mean Squares* (FxLMS) *Algorithm*.

The $\hat{S}(z)$ and $\hat{F}(z)$ filters are obtained in an offline training phase. During system's execution $\hat{S}(z)$ is used in order to align the error signal and the reference signal for the LMS algorithm, $\hat{F}(z)$ removes the acoustic feedback signal.

However, as $\hat{S}(z)$ and $\hat{F}(z)$ are kept fixed during the execution of the system, they cannot follow any changes in the acoustic paths S(z) and F(z) caused by e.g. variation in temperature, pressure, humidity. This might cause the system to become unstable because the digital filters do not accurately represent the acoustic paths, especially for the feedback path F(z).

In order to make the algorithm robust against these variations, the system under development in the DSP Lab of Hochschule Offenburg implements the method proposed by [2], where the digital feedback-path filter $\hat{F}(z)$ is constantly updated, following any changes in the physical path. A diagram of the whole system is shown in Figure 1.5-3.

Figure 1.5-3 includes an auxiliary noise source producing a low-level noise v(n), uncorrelated with r(n). It is fed at the same time to the adaptive filter $\hat{F}(z)$ and to the physical system through the anti-noise loudspeaker L_a, being an additional LMS-system that tries to achieve $\hat{F}(z)=F(z)$.

Now the reference signal s(n) captured by the microphone Mic_r is composed by the noise r(n) to be attenuated and both upstream signals $y_f(n)$ and $v_f(n)$. The subscript f indicates that they have been filtered by the acoustic feedback path.

If the whole system converges, the digital filters are now able to remove $y_f(n)$ and $v_f(n)$ from the reference signal and the ANC system again has only the noise r(n) on his input.

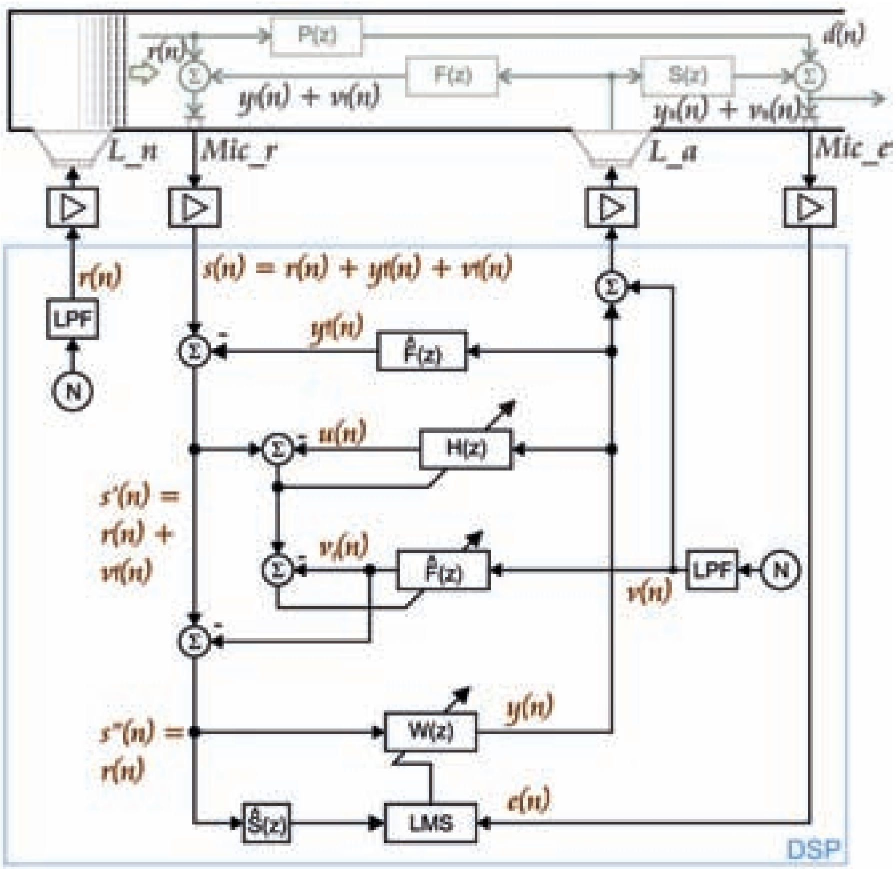

Fig. 1.5-3: *Online Feedback Modelling FxLMS*

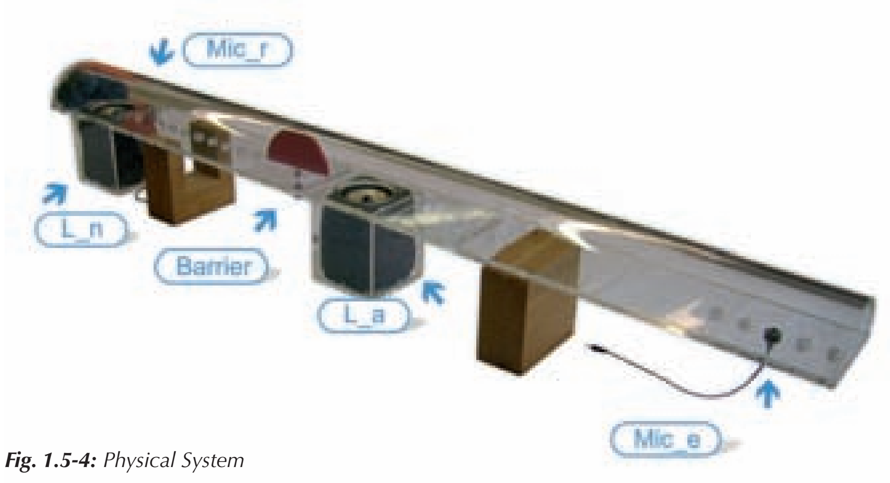

Fig. 1.5-4: *Physical System*

3. Practical System

The physical system, shown in Figure 1.5-4, consists of a duct made of Perspex (Plexiglas) with a semi-circular cross-section.

The digital control system is implemented on a floating-point, 225 MHz Digital Signal Processor (DSP) TMS320C6713 from Texas Instruments.

Through the loudspeaker L_n a noise with a bandwidth of 1.5 kHz excites the system. While this noise propagates down inside the duct, the reference signal is picked up by the reference microphone Mic_r and processed on the DSP. For causality reasons the system's delay must not be higher than the time needed by the sound to travel from the reference microphone to the anti-noise source. For a distance of 0.8 m this time is slightly above 2 ms.

Through the rotation of the barrier shown in Figure 1.5-5, changes on the physical path can be simulated in order to verify the behavior of the system. As shown in Figure 1.5-6, the system is able to adapt itself to these changes. It not only stays stable but is also capable of returning to the state of the lower noise level at the output.

Fig. 1.5-5: Rotational Barrier

References

[1] S. M. Kuo; D.R. Morgan: Active Noise Control Systems – Algorithms and DSP Implementation. New York: Wiley, 1996.

[2] M. T. Akthar, M. Kawamata: On Active Noise Control Systems with Online Acoustic Feedback Path Modeling. Proc. IEEE, vol.15, no. 2, pp. 593–600, February 2007.

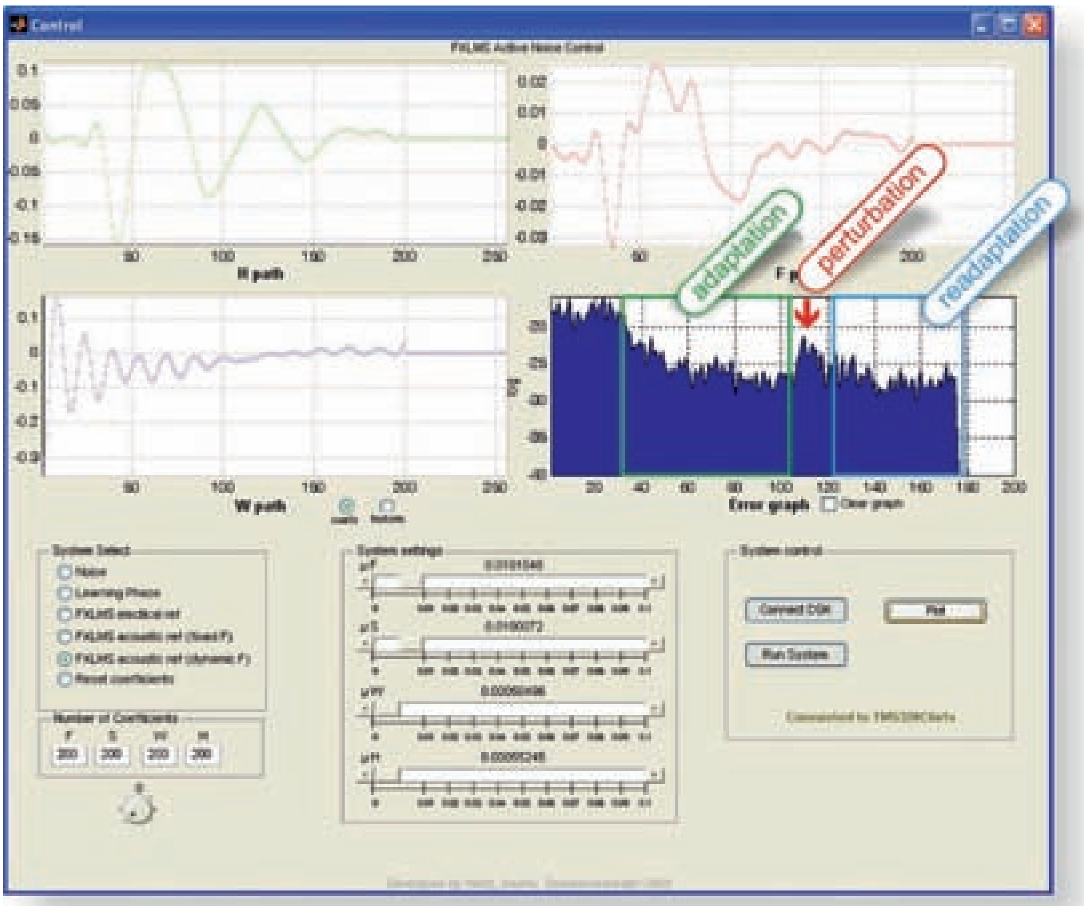

Fig. 1.5-6: Interface showing the resulting output signal