ACTIVE NOISE CONTROL

USING THE LMS AND FXLMS ALGORITHMS

MARKO STAMENOVIC APRIL 28, 2016

UNIVERSITY of ROCHESTER

HAJIM SCHOOL OF ENGINEERING & APPLIED SCIENCES UNIVERSITY of ROCHESTER

INTRODUCTION

Active Noise Cancellation (ANC) is a technique which uses destructive interference to cancel unwanted noise signals.

Machine learning algorithms are employed to learn the characteristics of the unwanted signal quickly enough for real-time attenuation.

Various applications from audio to automotive to aviation to vibration mitigation and more.

OVERVIEW

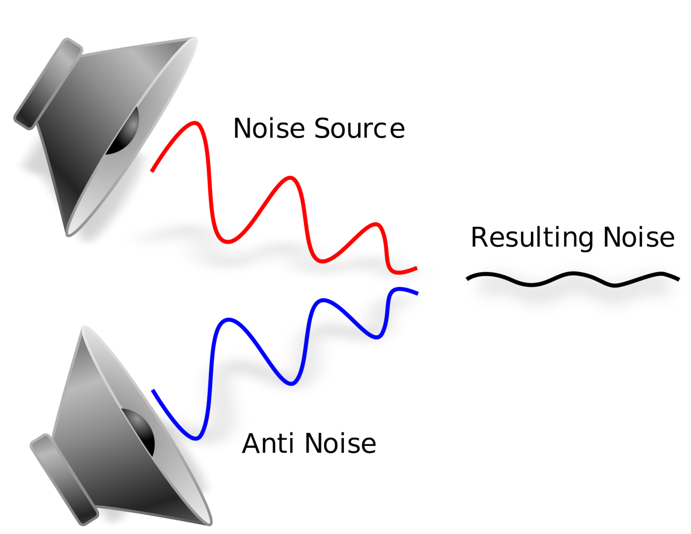


Figure 1. Conceptual diagram of Anti Noise

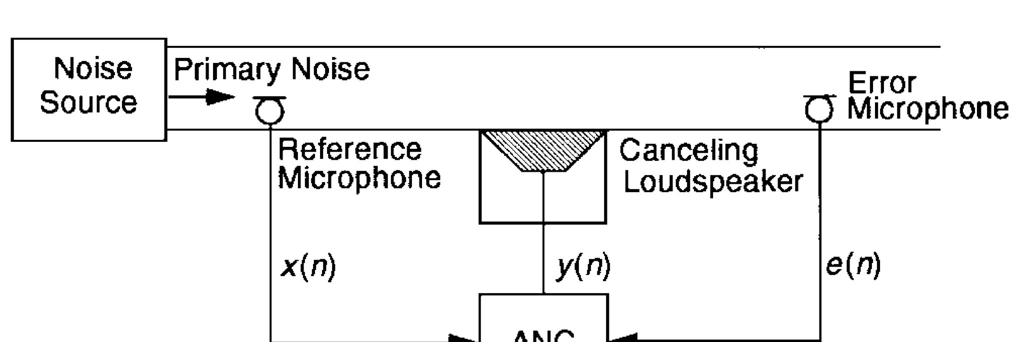


Figure 2. Conceptual diagram of ANC system setup

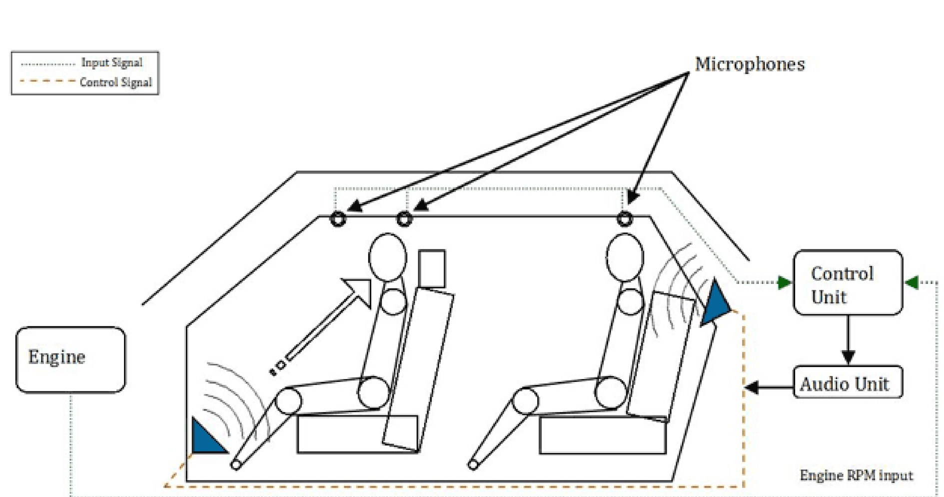
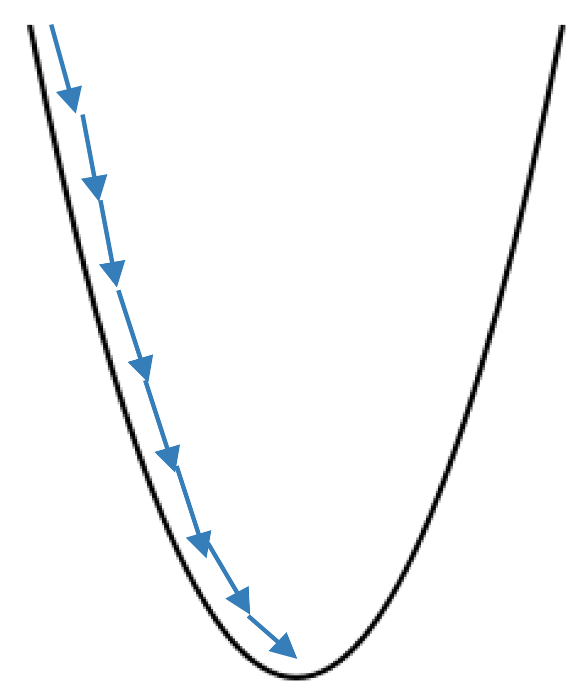
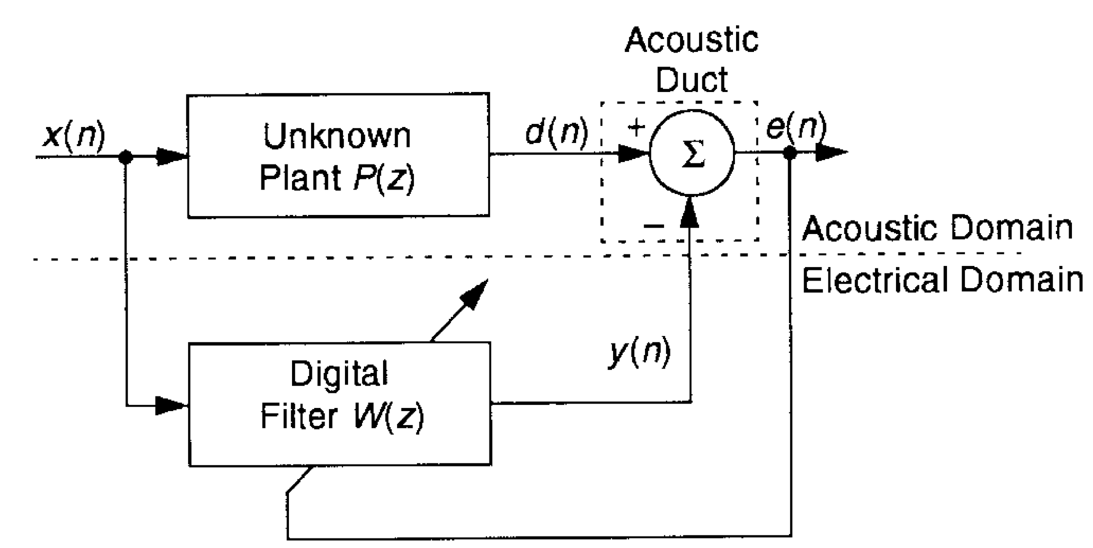


Figure 3. ANC system setup in vehicular application

ANC WITH LEAST MEAN SQUARES (LMS)





Least Mean Squares (LMS) algorithm. Uses stochastic gradient descent to minimize e(n). Simple and powerful.

Unknown plant P(z) is the transfer function b/w engine and passenger in car our outside world and ears in headphones.

mu = Learning rate affects convergence rate

As e(n) approaches 0, W(z) approaches P(z)

$$e(n) = d(n) - \boldsymbol{w}^T(n)\boldsymbol{x}(n)$$
$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) - \mu\boldsymbol{x}(n)e(n)$$

ANC WITH FILTERED LEAST MEAN SQUARES (FxLMS)

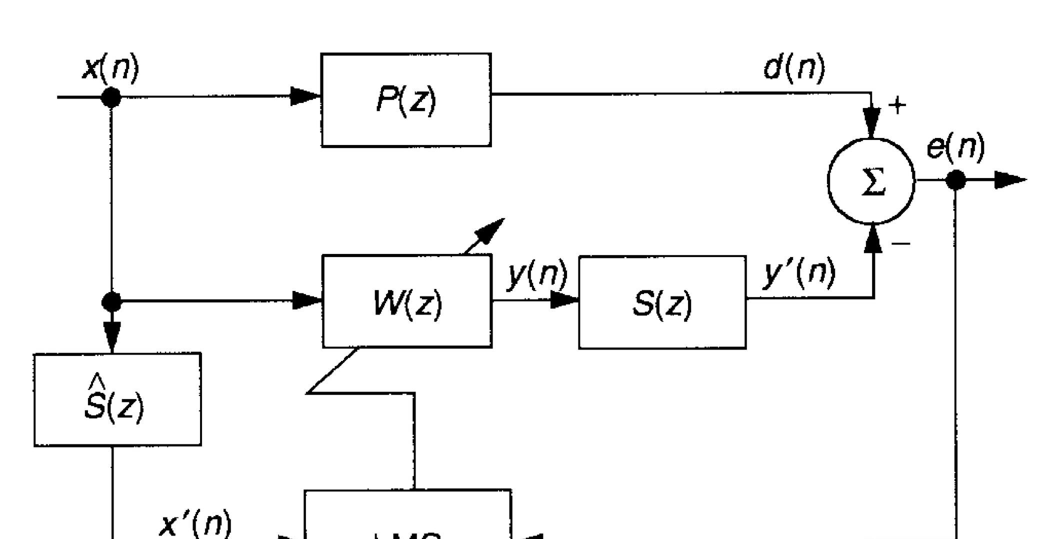
One thing missing from the LMS model in practical ANC applications is the path from the cancellation speakers to the ear for the correction signal.

This is called the Secondary Path or S(z).

Not including the secondary path in the model can cause nonconvergence and severe phase problems.

The FxLMS or Filtered LMS algorithm takes care of this issue by learning the transfer function of the secondary path and filtering the reference signal through an inverse filter $\hat{S}(z)$.

$$e(n) = d(n) - \boldsymbol{w}^T(n)\boldsymbol{x}(n)$$
$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) - \mu\boldsymbol{x}'(n)e(n)$$



EVALUATION

The system was evaluated in the digital domain using a simulated filter for P(z) and S(z). The system was implemented in Matlab as well as in embedded C on the Texas Instruments OMAP L-138 board.

The system was tested for various source signals including pink noise, idling engines and aviation sounds. The system converged for all tested steady-state sources.

The biggest factor on convergence was, unsurprisingly, the learning rate.

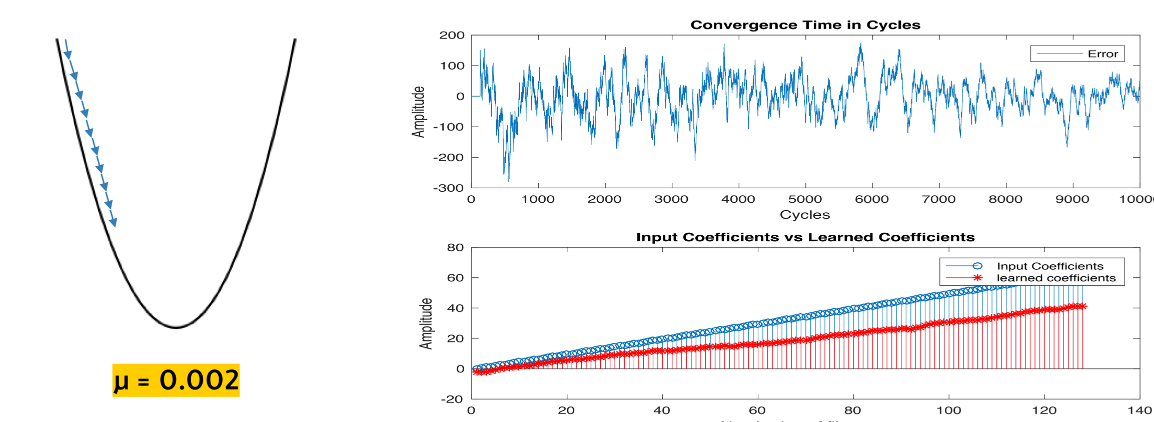


Figure 6. Example of low learning rate causing nonconvergence

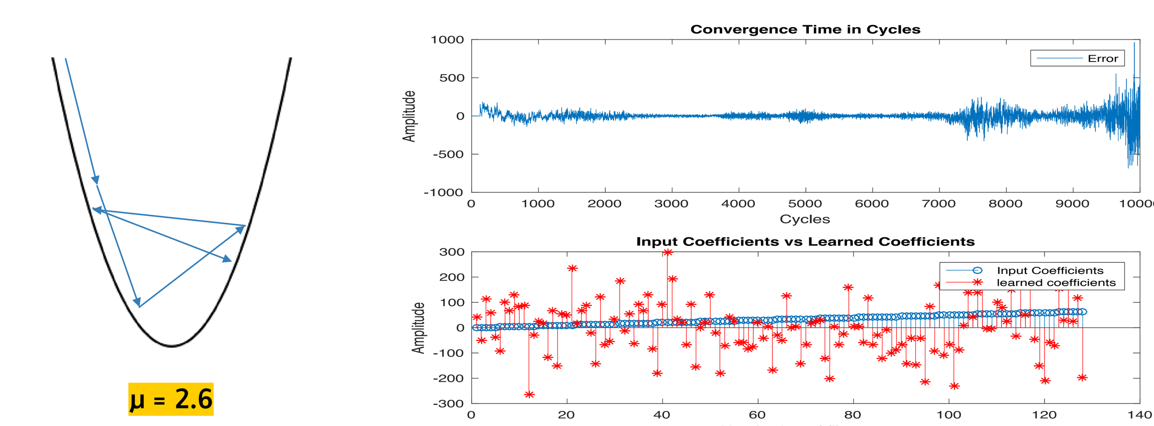


Figure 7. Example of high learning rate causing nonconvergence

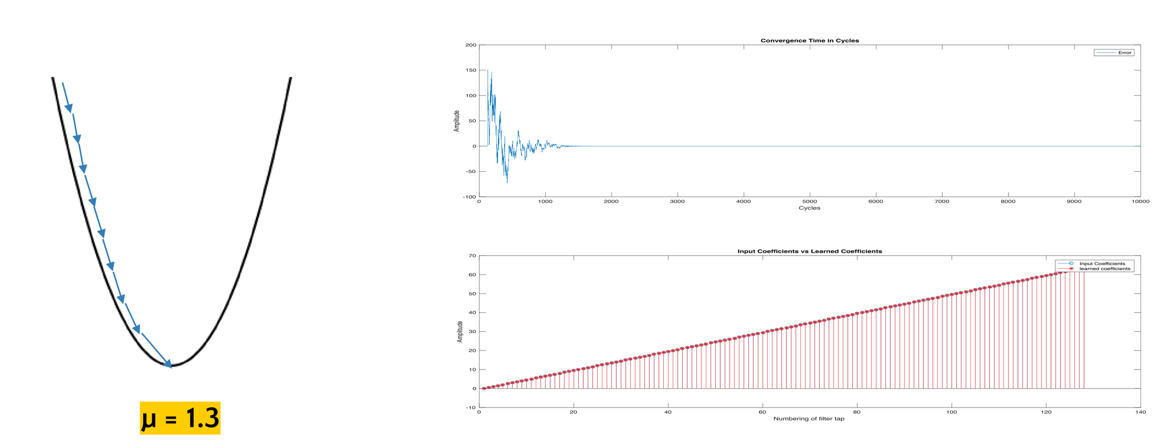


Figure 8. Example of well chosen learning. Convergence after ~1500 cycles.

Figure 9. Experimental results. Recorded error signal showing attenuation in the time domain

FUTURE WORK

Complete experiments in acoustic domain. Implement in open-source env.