Baltimore, Maryland
NOISE-CON 2010
2010 April 19-21

Optimal microphone placement for active noise control in a forced-air cooling system

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This paper discusses the placement of a microphone for feedforward based active noise control in a forced-air cooling system. Incorrect placement of microphones may lead to poor signal correlation that limit the ability of active noise suppression. Microphone placement can be by facilitated by direct on-line optimization of the Active Noise Control system. The feedforward based noise cancellation algorithm is applied to a data projector and uses the optimization results of a linearly parametrized filter to select microphone location. Successful implementation of the feedforward based active noise controller for this application shows a 20 till 40Db reduction of external noise over a broad frequency range from 1kHz till 5kHz.

1 INTRODUCTION

In applications where external sound disturbances interfere with the environment, passive or active attenuation can be used to control sound emission of forced-air cooling systems[9]. Passive noise control is effective at reducing high frequency sound components but requires large amounts of absorption material to reduce low frequent noise signals[3,10,13]. Especially for small electronic systems where forced air-cooling is required to control the temperature of large power sensitive components in the system, this is not a viable solution.

In active noise cancellation systems with relatively small acoustic coupling, feedforward compensation is an effective methodology to create a controlled emission for sound attenuation[16,17]. The basic principle and idea behind Active noise control (ANC) is to cancel sound by a controlled emission of a secondary out-of-phase sound signal[6,18]. In feedforward based ANC systems, the out-of-phase sound signal is created by filtering and amplifying the signal from a noise source microphone[2,4,7,8]. Filtering and amplification is used to provide an opposing sound signal with the right phase shift and amplitude to cancel the undesired noise.

Algorithms based on recursive (filtered) Least Mean Squares (LMS) minimization[11] can be quite effective for the estimation and adaptation of feedforward based sound cancellation[5]. These methodologies use both a noise source microphone and an error microphone to recursively tune and adapt the filtering and amplification needed for the ANC system. Crucial for the

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performance of the ANC system is the placement of the noise source microphone as incorrect placement may lead to poor signal correlation and limitations on the noise suppression. Poor signal correlation may occur when the noise source microphone is placed in the airflow or is subjected to acoustic coupling from the ANC system.

In this paper we discuss the placement of the noise source microphone for feedforward based active noise control in a forced-air cooling system. Noise source microphone placement is directed by the ANC performance of an on-line output-error based affine optimization of a linearly parametrized generalized finite impulse response filter[19] for sound compensation. Generalized or orthogonal FIR models[12] exhibit the same linear parametrization as a standard FIR filter. The procedure is demonstrated on a small portable NEC LT170 data projector, equipped with a shielded internal directional pick-up microphone to measure the sound created by the forced-air cooling of the projector's light bulb. Non-invasive small directional speakers located at the inlet and outlet grill of the data projector are used to minimize acoustic coupling.

2 ACTIVE NOISE CONTROL

2.1 Analysis of Feedforward Compensation

In a scalar feedforward noise compensation, an (amplified) signal $u(t)$ from an internal pick-up microphone is fed into a feedforward compensator or discrete-time filter $F(q)$ that controls the signal $u_c(t)$ to a control speaker for sound compensation. In order to analyze the design of a feedforward compensator $F(q)$ for active, noise cancellation, consider the signal $e(t)$ that reflects the combined effect of sound due to external disturbance and control speaker.

The objective of the Active Noise Control (ANC) system is to minimize the signal $e(t)$ and an error microphone can be used to measure $e(t)$ and monitor the performance of the ANC system. The dynamical relationship between the discrete time sampled signals in the ANC system can be characterized by difference equations, where the operator q is used to denote a unit sample delay $qu(t) = u(t+1)$. The dynamic relationship between the control input $u_c(t)$, the sound disturbance $u(t)$ and the error signal $e(t)$ is characterized by

$$e(t) = H(q)u(t) + G(q)u_c(t) \tag{1}$$

where $H(q)$ is a stable filter in the 'primary path' and $G(q)$ is a stable filter in the 'secondary path' of the ANC system. Both $H(q)$ and $G(q)$ characterize the discrete time dynamic aspects of the sound propagation through the ANC system to the error signal $e(t)$.

The sound disturbance $u(t)$ of the forced-air cooling system measured at the pick-up microphone is characterized by

$$u(t) = W(q)n(t)$$

where $n(t)$ is a zero-mean white noise signal with variance $E\{n(t)^2\} = \lambda$ and $W(q)$ is a (unknown) stable and stably invertible noise filter. The combination of a zero-mean and filtered white noise signal $u(t)$ allows for a characterization of a rich class of random sound disturbances for which the discrete time spectrum $\lambda|W(e^{j\omega})|^2$ can be modeled by a spectral decomposition[15]. In case the sound disturbance $u(t)$ itself is not influenced by the feedforward sound compensation $u_c(t) = F(q)u(t)$, the performance signal $e(t)$ can easily be described by

$$e(t) = W(q)[H(q) + G(q)F(q)]n(t) \tag{2}$$

and is a stable map, provided the feedforward compensator filter $F(q)$ is stable. Absence of a correlation between $u_c(t)$ and $u(t)$ given by $u(t) = G_c(q)u_c(t)$ requires a well-designed ANC system that minimizes the acoustic coupling $G_c(q)$. For the ANC application of the data projector discussed in this paper, acoustic coupling is minimized by small directional speakers located at the outside of the data projector. As a result, the effect of acoustic coupling is not considered in the remaining part of the paper.

In case the location of the noise source microphone and the control speaker are fixed and the transfer functions in Eqn. (2) are known, an ideal feedforward compensator $F(q) = F_i(q)$ can be obtained with

$$F_i(q) = -\frac{H(q)}{G(q)} \tag{3}$$

provided $F_i(q)$ is a stable and causal transfer function. The formulation of the ideal forward filter $F_i(q)$ in Eqn. (3) can, in general, not be used for the construction of a feedforward based ANC system for several reasons. First of all, the solution of $F_i(q)$ in Eqn. (3) assumes full knowledge of $G(q)$ and $H(q)$. Moreover, the filter $F_i(q)$ may not be a causal or stable filter due to the dynamics of $G(q)$ and $H(q)$ that dictate the solution of the feedforward compensator $F_i(q)$.

Instead, an approximation of the feedforward filter $F_i(q)$ can be made by an output-error based optimization that aims at finding the best causal and stable approximation $F(q)$ of the ideal feedforward compensator in $F_i(q)$ in Eqn.(3). The output-error based approximation can be characterized by examining the variance of the discrete time error signal $e(t)$ in Eqn. (2). Use Parseval's theorem the variance of the discrete time error signal $e(t)$ in Eqn. (2) is given by

$$\frac{\lambda}{2\pi}\int_{-\pi}^{\pi}\left|W(e^{j\omega})\right|^2\left|H(e^{j\omega})+G(e^{j\omega})F(e^{j\omega})\right|^2 d\omega$$

where λ denotes the variance of $n(t)$. In case variance minimization of the error microphone signal $e(t)$ is required for ANC, the optimal feedforward controller is found by the minimization

$$\begin{aligned} &\min_\theta \left\|L(q,\theta)\right\|_2^2 = \min_\theta \int_{-\pi}^{\pi}\left|L(e^{j\omega},\theta)\right|^2 d\omega, \\ &L(q,\theta) = W(q)[H(q)+G(q)F(q,\theta)] \end{aligned} \tag{4}$$

where the parametrized filter $F(q,\theta)$ is required to be a causal and stable filter to facilitate the real-time filtering operation in the ANC system

The minimization in Eqn. (4) is a standard 2-norm based feedback control and model matching problem[1,14] that can be solved in case $W(q)$, $G(q)$ and $H(q)$ are known. Unfortunately, for an ANC system, the discrete time dynamic relation between the signals $e(t)$, $u(t)$ and $u_c(t)$ is unknown. Moreover, even with a fixed location of the control speaker, the transfer functions $W(q)$, $G(q)$ and $H(q)$ depend on the location of the noise source and error microphones of the ANC system. Since the location of the noise source microphone is up for discussion, the optimization in Eqn. (4) will be solved as a function of the microphone location to determine the optimal location of the noise source microphone.

2.2 Optimal Location of Noise Source Microphone

To minimize acoustic coupling in the ANC system of the data projector discussed in this paper, a directional control speaker is placed at the external opening of the projector and directed outwards. Due to space limitations in the small portable projector, only limited space is available for the control speaker. Furthermore, the error microphone is used only to temporarily measure the error signal $e(t)$, but will not be part of the ANC as only feedforward compensation will be used for sound compensation.

If design freedom is available for the location of the noise source or pick-up microphone, then it should be exploited to develop a well-performing ANC system. In order to study the performance of the ANC system, consider a certain location of the pick-up microphone in the ANC system. For that specific location, the transfer functions $H(q)$ and $G(q)$ in Eqn. (1) are fixed, but unknown. As a result, the performance of the ANC system solely depends on the design freedom in the feedforward compensator $F(q,\theta)$ to minimize the error signal $e(t)$.

The design freedom of the feedforward compensator $F(q,\theta)$ is restricted by the parametrization of $F(q,\theta)$ in terms of the filter coefficients θ. A direct optimization of the feedforward compensator $F(q,\theta)$ can be performed by considering the parametrized error signal $e(t,\theta)$ given by

$$e(t,\theta) = H(q)u(t) + F(q,\theta)G(q)u(t) \tag{5}$$

whereas definition of the signals

$$y(t) = H(q)u(t), \quad u_f(t) = -G(q)u(t) \tag{6}$$

reduces Eqn. (5) to an Output Error (OE) format

$$e(t,\theta) = y(t) - F(q,\theta)u(t) \tag{7}$$

With $e(t,\theta)$ in Eqn. (7) written as a function of $F(q,\theta)$, the minimization of the variance of $e(t,\theta)$ estimated on the basis of N time observations

$$\min_\theta \frac{1}{N}\sum_{t=1}^{N} e(t,\theta)^2 \tag{8}$$

can be used to compute the optimal feedforward filter $F(q,\theta)$ via a standard output-error (OE) minimization problem in a prediction error framework[15]. This makes the choice of the optimal feedforward compensator $F(q,\theta)$ for any possible location of the noise source microphone a standard OE identification problem. Moreover the minimization in Eqn. (8) matches the desired 2-norm minization in Eqn. (4).

Using the fact that the variance of the pick-up microphone signal $u(t)$ satisfies $\|u\|_2^2 = \lambda \left|W(e^{j\omega})\right|^2$, the minimization of Eqn. (8) for $\lim_{N\to\infty}$ can be rewritten into the frequency domain expression

$$\min_\theta \frac{\lambda}{2\pi}\int_{-\pi}^{\pi} \left|W(e^{j\omega})\right|^2 \left|H(e^{j\omega}) + G(e^{j\omega})F(e^{j\omega})\right|^2 d\omega \tag{9}$$

using Parseval's theorem[15]. It can be observed that Eqns.(9) and (4) are equivalent and the standard output-error (OE) minimization problem in Eqn. (8) can be used to compute the optimal feedforward filter $F(q,\theta)$ for any specific location of the noise source microphone, provided $y(t)$ and $u_f(t)$ in Eqn. (6) are available.

2.3 Data-Based Computation of Optimal Noise Source Microphone Location

For a specific location of the noise source or pick-up microphone, the signals $y(t)$ and $u_f(t)$ in Eqn. (6) are easily obtained by performing a series of two experiments. Once the signals $y(t)$ and $u_f(t)$ are available, they can be used in the optimization of Eqn. (4) or (9) to compute the optimal filter $F(q,\theta)$ to evaluate the quality of the ANC system on the basis of the chosen location of the pck-up microphone. The two experiments simply measure the pick-up and error microphone signals $u(t)$ and $e(t)$ and are used as follows.

1. The first experiment is done without feedforward compensation. Hence, $F(q,\theta) = 0$ and the error microphone signal $e_1(t)$ during this first experiment satisfies

$$e_1(t) = H(q)y(t) \tag{10}$$

   and the signal from the pick-up microphone satisfies

$$\tilde{u}(t) = u(t) + v(t) \tag{11}$$

   where $v(t)$ indicates possible measurement noise on the pick-up microphone signal. Such noise $v(t)$ might arise in case the pick-up microphone is placed in the air flow of the forced-air cooling system. This results in additional disturbances on the pick-up microphone signal that need to be considered in the optimal location of the microphone.

2. The second experiment is done with the forced air-cooling system turned off, eliminating the presence of the external sound disturbance. Subsequently, the measured input microphone signal $-\tilde{u}(t)$ in Eqn. (11) from the first experiment is applied to the control speaker, yielding a error microphone signal $e_2(t)$ during this second experiment that satisfies

$$e_2(t) = -G(q)\tilde{u}(t) = -G(q)u(t) - G(q)v(t) \tag{12}$$

   where it can be seen that a signal filtered by the 'primary path' $G(q)$ is obtained.

The result of the abovementioned two experiments allows us to write the error signal $e(t,\theta)$ in Eqn. (7) as a function of $e_1(t)$ and $e_2(t)$ via

$$e(t,\theta) = e_1(t) - F(q,\theta)e_2(t) - F(q,\theta)G(q)v(t) \tag{13}$$

and in the absence of the noise $v(t)$ on the input microphone during the first experiment, the minimization of $e(t,\theta)$ in Eqn. (13) is equivalent to the minimization of $e(t,\theta)$ in Eqn. (7). As a result, the obtainable performance of the ANC for a specific location of the pick-up microphone can be evaluated directly on the basis of the error microphone signals $e_1(t)$ and $e_2(t)$ from the first and second experiment respectively. With noise $v(t)$ on the input microphone during the first experiment will only lead to a larger variance of $e(t,\theta)$, also influencing the choice for the optimal location of the pick-up microphone based on two experiment.

Alternatively, both experiments can be combined by using a filtered input signal $u_f(t)$ that is based on an estimated model of $G(q)$. Because $G(q)$ is fixed once the location of the control speaker is determined, an initial off-line estimation can be used to estimate a model for $G(q)$ to construct a filtered input signal $u_f(t)$. The use of an estimated transfer function for filtering purposes in recursive and adaptive estimation is common practice in most filtered least mean squares algorithms[5]. Similar approaches are also found in identification algorithms that provide unbiased estimates of models on the basis of closed-loop experimental data.

The usage of both experiments to minimize the error signal $e(t,\theta)$ in Eqn. (7) as a function of $e_1(t)$ and $e_2(t)$ and decide on the optimal location of the noise source or pick-up microphone is summarized in the following proposition.

Proposition 1: The performance of the feedforward ANC system for a specific location of the pick-up microphone is characterized by $V_N(\hat{\theta})$. The numerical value of $V_N(\hat{\theta})$ is found by measuring $e_1(t)$ and $e_2(t)$ for t = 1, . . . ,N as described by the experiments above, and solving the Output Error (OE) model estimation problem

$$\begin{aligned}&\hat{\theta} = \arg\min_{\theta\in R^d} V_N(\theta), \quad \text{with}\\ &V_N(\theta) = \frac{1}{N}\sum_{t=1}^{N}\varepsilon(t,\theta)^2 \quad \text{and}\\ &\varepsilon(t,\theta) = e_1(t) - F(q,\theta)e_2(t)\end{aligned} \tag{14}$$

for a finite size d real-valued parameter $\theta \in R^d$ that represents the coefficients of a finite order feedforward filter $F(q,\theta)$. ◊

A finite number d of filter coefficients is chosen in Proposition 1 to provide a feasible optimization of the filter coefficients. It should be noted that the OE minimization can be further simplified to a quadratic convex optimization problem with a unique computable minimum for the finite dimensional parameter θ in case the feedforward filter $F(q,\theta)$ is parametrized using an Finite Impulse Response (FIR) filter

$$F(q,\theta) = \sum_{k=0}^{p-1}\theta_k q^{-k} \tag{15}$$

Although FIR filter representations require many filter coefficients θ_k for an accurate realization of the optimal feedforward filter, the FIR filter is used solely to evaluate the possible performance of the ANC for a specific pick-up microphone location. The FIR filter representation can also be replaced by a generalized FIR filter that uses an orthogonal expansion of basis functions other than the standard time shifts in a FIR filter to construct a linearly parametrized filter[12,19]. The proposed procedure summarized in Proposition 1 also works for generalized FIR filters, but would require additional knowledge on pole locations to generate the orthogonal expansion of basis functions.

As a final note, the influence of the noise $v(t)$ from the first experiment can be quantified from an identification point of view. It can be seen that the minimization of $\varepsilon(t,\theta)$ in Eqn. (14) is equivalent to Eqn. (8) with a noise $G(q)v(t)$ disturbing the output $e_2(t)$. However, the noise $v(t)$ is correlated with the input $e_1(t)$ and correlated noise on the input leads to bias estimation results in the OE minimization[15]. This directly gives rise to a performance deterioration of the feedforward ANC system in case the pick-up microphone is placed unshielded inside the air flow of the forced-air cooling system, influencing the choice of the optimal location of the microphone.

3 APPLICATION TO DATA PROJECTOR

3.1 Description of Projector

To illustrate the effectiveness of the proposed methodology for selecting the optimal noise source microphone location for feedforward based active noise control using a linearly parametrized feedforward filter, the NEC LT170 data projector depicted in Fig. 1 was used. The small and portable projector is equipped with a shielded internal directional pick-up microphone to measure the sound created by the forced-air cooling of the projector's light bulb. Non-invasive small directional speakers located at the inlet grill minimize acoustic coupling in the data projector and will be used to reduce ambient noise from the projector due to the forced air-cooling system.

The location of the inlet grill of the data projector determines the location of the control speakers. However, inside the projector there are several small areas close to the air cooling fan where a directional pick-up micro phone could be placed. An overview of the design space to place a microphone inside the projector has been indicated in Fig. 2, where the top of the projector has been removed for visualization purposes.

The error microphone is placed 25cm away from the inlet grill of the projector and experimental data of the NEC data projector is gathered in an anechoic room located at the System Identification and Control Laboratory at UCSD at a sampling frequency of 20kHz. As indicated in Fig. 2, four possible locations are considered for the location of the pick-up microphone.

3.2 Experimental Results

For each location experiments were done according to Proposition 1 to observe the error $e(t)$ and pick-up microphone $u(t)$ signals. In order to predict the performance of the feedforward based ANC system for each pick-up microphone location, 41 parameters θ of a 40th order FIR model $F(q,\theta)$ were estimated by the affine optimization Eqn.(14). A high order FIR filter was chosen to find good approximation results of the achievable optimal filter via the affine optimization. The results are summarized in Table 1.

It should be noted that the 40th order FIR model does not have to be the actual feedforward filter that should be implemented as the ANC feedforward compensator. Only the value of the criterion function $V_N(\hat{\theta})$ was used to predict the performance of the ANC system for the different pick-up microphone locations. For normalization purposes, the value of the criterion function

$$V_N(\hat{\theta}) = \frac{1}{N}\sum_{t=1}^{N}\varepsilon(t,\hat{\theta})^2, \quad \varepsilon(t,\hat{\theta}) = e_1(t) - F(q,\hat{\theta})e_2(t)$$

is compared with the variance var{$e_1(t)$} in Table 1 to indicate the improvement the simulated noise $\varepsilon(t,\hat{\theta})$ of the error microphone signal due to the feedforward compenator $F(q,\hat{\theta})$. It can be observed from Table 1 that positioning of the pick-up microphone in the center location of the projector gives far superior feedforward ANC performance than the other locations. As expected, the bad performance at location (2) is due to a large noise contribution $v(t)$ on the pick-up microphone signal in Eqn. (11) by placing the microphone in the air flow of the air-cooling system. Similarly, the close proximity of the microphone to the fan in location (4) causes the microphone to pick up residual vibrations disturbances that deteriorate feedforward performance.

3.3 Implementation of ANC

With the input microphone at the center location (3) in Fig. 2 of the data projector, experimental data of the error microphone signal $e_1(t)$ and $e2(t)$ and input microphone $u(t)$ were gathered according to the two experiments outlined in Eqns. (10) and (12). The microphone signals sampled at 20kHz is used to estimate a (generalized) FIR model $F(q,\theta)$ of order 24 vi an Output Error optimization. The estimate feedforward filter is then implemented in real-time to validate the quality of the resulting ANC system for the NEC LT170 data projector.

Using a Pentium based personal computer with equipped with a MultiQ3 12 bit signed AD/DA card, the 24th order feedforward filter is implemented at a sampling frequency of 20kHz sampling Application of the feedforward filter for the active noise cancellation of sound in the data projector leads to the snapshot of the experimental data $e(t)$ of the error microphone depicted in Fig. 3. The performance of the ANC system is confirmed by the estimate of the spectral contents of the microphone error signal $e(t)$ plotted in Fig. 4. The spectral content of the error microphone signal $e(t)$ has been reduced significantly by the feedforward filter in the frequency range from 1kHz till 5kHz.

4 CONCLUSIONS

Forced-air cooling systems move air through a fan to provide an effective resource for cooling of sensitive electronic components. The drawbacks of noise due to turbulence and vibrations in the air cooling system can be reduced by deploying an active noise control algorithm to reduce the external sound. This paper presents a methodology for choosing an optimal location of the sound source or pick-up microphone. The methodology is based on an variance minimization of an error signal that is obtained by performing two consecutive experiments. The variance minimization optimization can be reduced to an convex quadratic optimization in case a linearly parametrized feedforward filter is used. The numerical and experimental results of the application of the design methodology to a NEC LT170 data projector shows the success of feedforward based noise cancellation in a small electronic system with high demands on the forced-air cooling system.

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Table 1 - Results of affine optimization (14) using a 40th order FIR model $F(q,\theta)$ for the pick-up microphone locations in Fig. 2.

location	var$\{e_1(t)\}$	$V_N(\hat{\theta})$	reduction
(1)	$1.8612\cdot10^{-4}$	$1.4171\cdot10^{-4}$	76%
(2)	$1.8535\cdot10^{-4}$	$1.7725\cdot10^{-4}$	96%
(3)	$1.8599\cdot10^{-4}$	$1.0421\cdot10^{-4}$	56%
(4)	$1.8609\cdot10^{-4}$	$1.7282\cdot10^{-4}$	93%

Fig. 1 - Dimensions of NEC LT170 data projector with inlet side grill for forced-air cooling system.

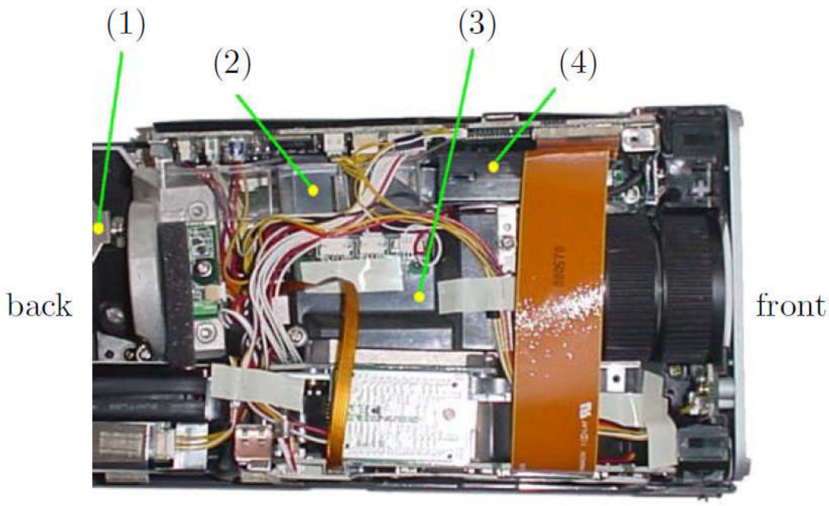

Fig. 2 - Top view of NEC LT170 data projector for ANC design with possible locations for pick-up microphone : (1) in proximity of light bulb, (2) in air flow channel, (3) in central part of projector, (4) in close proximity of cooling fan.

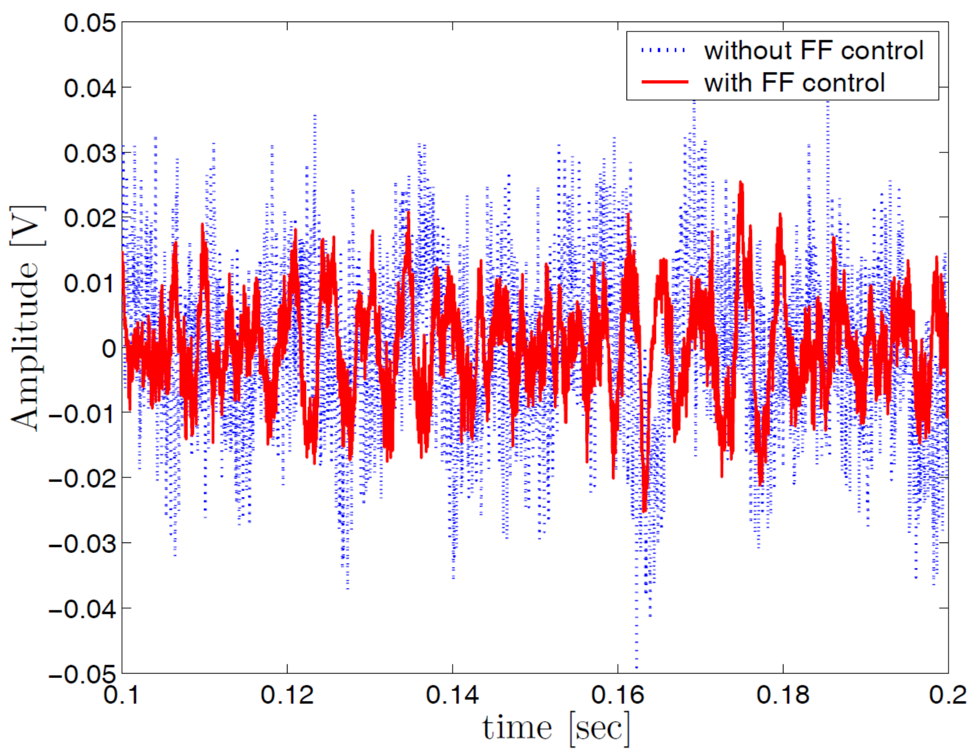

Fig. 3 - Time trace of error microphone signal without (dotted) and with (solid) feedforward based active noise control.

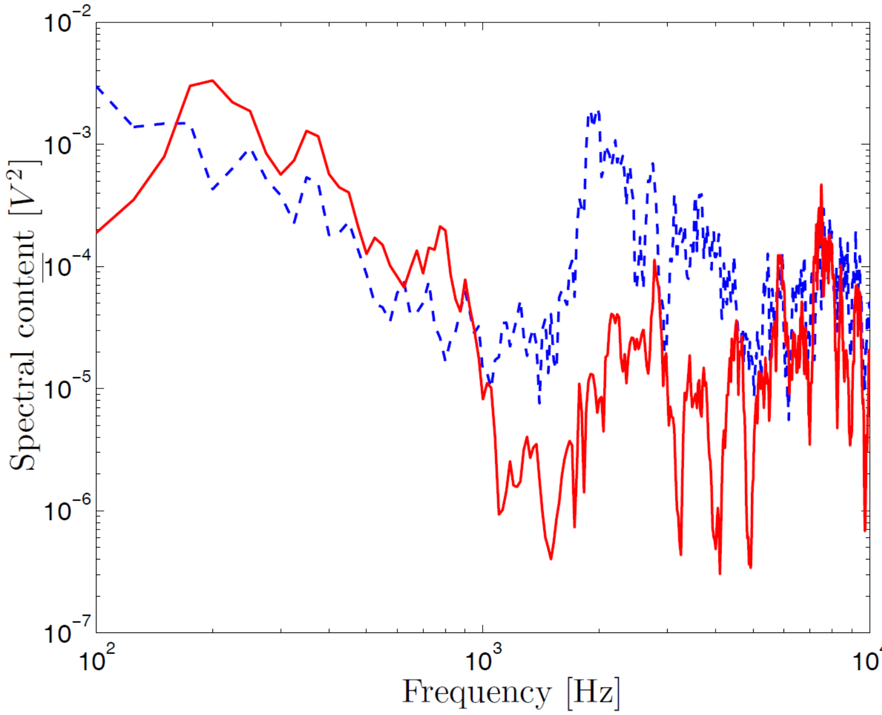

Fig. 4 - Estimate of spectral contents of error microphone signal e(t) without ANC (dashed) and with ANC using 24th order (generalized) feedforward FIR filter.