Implementation of active feedback noise control in an acoustic cavity mimicking supra-aural headphone

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Abstract

The goal of this study is to design and implement active feedback control of noise (ANC) in a small rigid-walled acoustic cavity that mimics the size of a supra-aural headphone. A small 'noise-speaker ‘delivers noise inside the rigid-walled cavity. The sensor microphone is placed inside the cavity flush with a wall, and the control speaker is placed flush with another wall of the cavity. The frequency-domain method is used to design the feedback loop gain. The transfer function from the control speaker to the microphone sensor is measured and fitted with poles and zeros to obtain the plant transfer function. The plant transfer function without delay is inverted and multiplied with the designed loop gain to determine the controller. The controller is implemented in dSPACE as a digital filter, and the output of the controller drives the control speaker. The pressure measured by the sensor microphone for ANC ON and ANC OFF and the closed-loop sensitivity are presented and shown to agree well with predicted values. Noise reduction of about 10 dB in the frequency range of 100-300 Hz is reported. Further steps to improve the ANC are also discussed.

Keywords: Active noise control; noise reduction; control system; sound absorption;

1. Introduction

Acoustic noise is a severe problem caused due to rapid industrialization and urbanization. Noise can be defined as any undesirable disturbance from acoustic, electrical, vibration, or other media. Noise is one of the most common physical hazards present in the current situation of human settlement. This results in many medical and psychological problems for a person. Prolonged noise exposure may lead to anxiety, racing pulse, headache, high blood pressure, gastritis, depression, and hearing loss [1]. Many noise cancellation techniques are applied to avoid this issue, such as using noise-absorbing porous materials like foam are traditional passive noise reduction methods [2][3]. Passive systems are more effective in the higher and middle-frequency ranges; thus, we need a system that could also work in the lower-frequency ranges, active noise control can work well in lower-frequency ranges compared to passive noise control methods.

Active noise control (ANC) is a technique used to reduce unwanted background noise by generating a signal intended to be the opposite, or "negative," of the unwanted noise. The earliest inventor of active noise control techniques was Paul Lueg, who filed a patent in 1933[4]. Lueg outlined in his patent the principle that microphones can

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pick up unwanted sound and, after amplification, can be sent to a speaker-phone with phase-opposition to cancel the unwanted sound [5]. Other notable figures in the development of ANC include Harry Hollerman, who proposed ANC using adaptive filtering in the 1970s [6]. Hollerman presented the theory of adaptive noise canceling and discussed its potential applications. He also proposed a specific algorithm for implementing adaptive noise canceling based on the least-mean-square (LMS) method. LMS adaptive filtering is considered a standard method to develop ANC, and in 1993 a practical ANC system for use in aircraft was designed by Elliot [7] using multiple references and multiple output references in a 50-seater airplane.

Other methods of achieving ANC is through feedforward and feedback method. In a feedforward method for headphones, the microphone is placed outside the ear cup so that the microphone can pick up the signal before it reaches the ear pinna and then process the noise and create an anti-noise signal to filter it. Whereas in a feedback ANC system, the microphone is placed inside the ear cup of the headphone or inside the headphone cavity such that it gets to hear how the listener perceives the resulting sound signal and make adjustments and provide feedback to the driver speaker. Although adaptive filtering techniques are standard in developing ANC, the first practical setup of feedback ANC was proposed by Olson and May [8] in 1953, where the microphone, loudspeaker, and digital controller were used as a feedback system to cancel the noise. Much before that analog controller were used to develop feedback ANC[9], analog feedback ANCs were capable of achieving satisfactory broadband noise reduction based on speaker-to-mic distance delay as the delay reduced, so the broadband noise reduction increased [10] but with feedback analog ANC reducing narrowband noise was hard to attain such as noise from airplanes, and helicopters [11]. The analog feedback was incapable of adjusting the positioning of the ear cup of a headphone.

One of the main disadvantages of active noise control is that achieving stupendous noise reduction over a wide range of frequencies can be challenging, especially in a complex, real-world environment. Additionally, adapting the anti-noise signal to changing noise conditions can be computationally intensive and challenging when implementing ANC in a real-time system. Numerous Digital Signal Processing(DSP) real-time systems are used to develop real-time ANC. A non-linear ANC is discussed using a generalized filter bank [12] implementation using dSpace 1104 system. W. Ciesielka 2007, built ANC using dSpace equipped DSP card TMS320C31 by Texas Instrument, consisting of two subsets, the Finite Impulse Response system, and the Adaptive algorithms using LMS and NLMS algorithm [13].

Finally, ANC systems can be relatively expensive, requiring specialized hardware and software to generate and playback the anti-noise signal.

This work demonstrates the development of ANC in a cavity system approximately the size of a supra-aural headphones cavity. A 3d printed rigid-walled cavity of 24 x 18 x 18 mm is designed with two speakers (one to generate noise and the other is the actuator) and a sensor microphone to test the developed ANC. The foam material is also inserted into the cavity to reduce sharp resonances and nulls.

2. Methodology

The frequency-domain loop shaping method is used in this study to design the feedback loop gain.

The acoustic system is a small 3D printed cavity (Figure 1) of dimension 24x18x18 mm with two speakers and one microphone. Speaker-1 is used as a noise generator, and speaker-2 is used as the actuator (figure 2).

The sensor microphone is a GRAS 40 PP microphone with a constant current power module GRAS 12AX. The controller is implemented on a real-time RTI 1104 dSPACE system, and its output drives the actuator speaker (speaker-2).

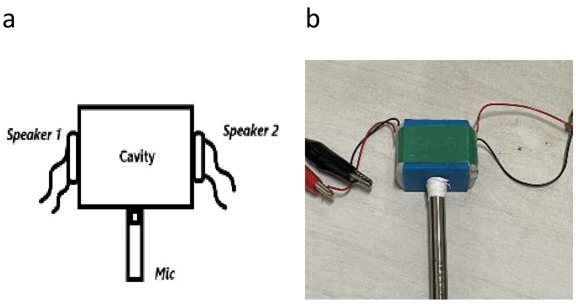

Figure 1: (a) Active noise cancellation cavity system(b) 3D printed cavity setup used for the experiment(b).

The speaker-to-microphone distance is calculated to estimate the system's delay, and the plant transfer function is acquired for further processing

3. Data Acquisition

The designed 3D-printed cavity is assembled with one microphone and two speakers, sealed using tape to avoid any leakage in the system. Further, it is filled with melamine foam of 99% porosity to reduce reflections inside the cavity. The speaker and microphone system is connected to a National Instrument NI PXIe1071 with NI PXIe4463 analog output connected to the speakerphone, and the analog input from the mic is connected to PXIe4481 for data acquisition. The microphone used in the system is a GRAS 40P CPP free field QC microphone with a dynamic range of 30dB to 135dB and a sensitivity of 47.17 mV/Pa. The plant transfer function is taken by giving an input voltage of 50mV, and a varying frequency (frequency sweep) of 25-3500 Hz is given to the actuator speaker. The output frequency response of the plant transfer function is shown in figure 2. The next step is to model a plant transfer function that mimics the same frequency response.

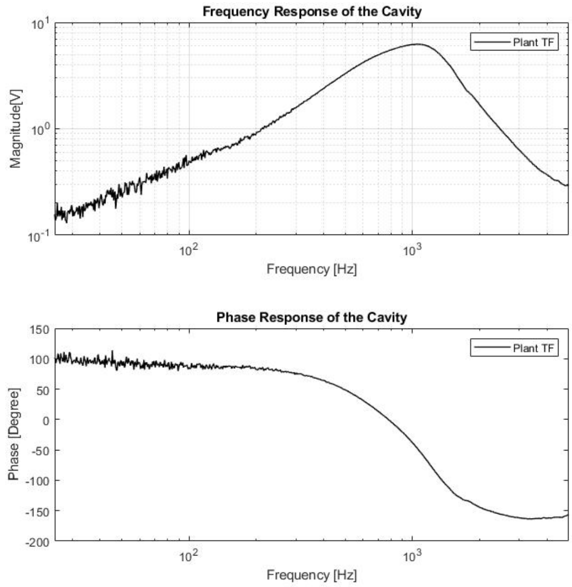

***Figure 2:** Cavity system frequency response.*

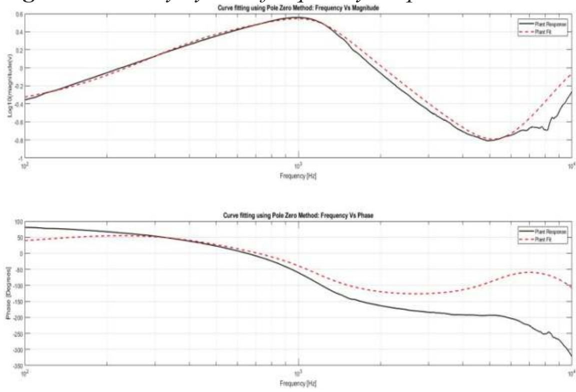

***Figure 3:** Curve fit for the plant transfer function (red) compared to the measured plant(black).*

4. Curve fitting using pole and zero transfer function

The plant transfer function magnitude and phase are used to develop a fit to the measured plant transfer function with a similar frequency response to the measured plant. To this plant transfer function, a curve fit is obtained by placing a complex conjugate of poles and zeros at frequency locations of its peaks and valleys, evaluating the position of the poles and zeros based on how sharp or smooth the peak or the valley is for the response. In this system, using (Equation 1), a set of three poles and three zeros are used to obtain the curve fitting (Figure 3). The gain represents the overall gain applied to the system and controls the magnitude of the transfer function, and **s** is the Laplace operator denoting. Delay is calculated using the speaker-to-mic distance in the cavity, assuming the speed of sound in the air$c = 340\ m/s$ equation 2.

$$Plant\ = gain * \frac{((\ s - z_1)(s - z_2)(s - z_3))}{((s - p_1)(s - P_2)(s - p_3))} e^{-s*delay} \quad (1)$$

$$delay = \frac{speaker\ to\ mic\ distance}{speed\ of\ sound\ in\ air} \quad (2)$$

4. Loop shaping

The loop transfer function is developed for a closed-loop system that contains two major components the Process and the Controller (Figure 4). The controller is a combination of feed-forward **F** and feedback block **C**. The feedback loop shown in Figure 5 consists of the **F** and **C** and the plant **P**. The reference **r**, disturbance **d**, and measured noise **n** highly influence the feedback loop. The sensitivity transfer function **S** denotes the change in the closed loop transfer function to the change in the plant transfer function, and it is obtained in this system using equation 4.

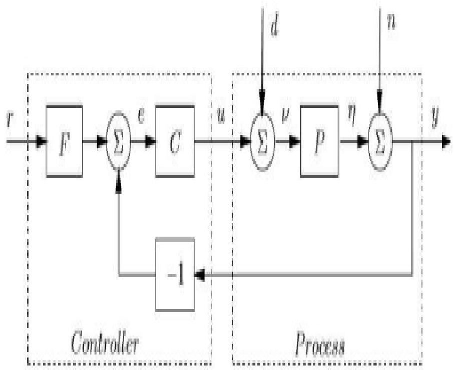

Figure 4: A simple feedback controller model.

$$G = \frac{PC}{1 + PC} \quad (3)$$

$$S = \frac{1}{1 + PC} \quad (4)$$

$Senstivity\ transfer\ function$

The output of the feedback model G is given by equation (3). Assuming $L = PC$ defines the relationship between plant and controller. The controller response can be determined as $C = P^{-1}L$for the given feedback loop.

The frequency-domain loop-shaping method is adopted in this study to design the loop transfer function. The loop is designed to target the lower frequencies because of the physical plant delay and digital delay with a maximum loop gain of 10 dB at 250 Hz and the loop characteristics mentioned in table 1.

Table 1: Characteristic features of loop transfer function.

Phase margin	ϕ_m	51.43°
Gain crossover	ω_{gc}	446 Hz
Gain margin	g_m	19.95 dB
Phase crossover	ω_{gc}	6706 Hz

The loop was developedin figure 5 using a set of 4 poles and 4 zeros in the transfer function equation 5, similar to equation 1

$$\mathbf{Loop}\ = \mathbf{gain} * \frac{((\ \mathbf{s} - \mathbf{z}_{L1})(\mathbf{s} - \mathbf{z}_{L2})(\mathbf{s} - \mathbf{z}_{L3}))}{((\mathbf{s} - \mathbf{p}_{L1})(\mathbf{s} - \mathbf{P}_{L2})(\mathbf{s} - \mathbf{p}_{L3}))} e^{-s*delay} \quad (5)$$

where

$$s - z_{l1} = (s - (r + j\omega_{z1}))(s - (r - j\omega_{z1})$$

.

.

$$s - z_{ln} = (s - (r + j\omega_{zn}))(s - (r - j\omega_{zn})$$

$for\ all\ angular\ frequency\ for\ zero.$

$$s - p_{l1} = (s - (r + j\omega_{p1}))(s - (r - j\omega_{p1})$$

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$$s - p_{ln} = (s - (r + j\omega_{pn}))(s - (r - j\omega_{pn})$$

$for\ all\ pole\ angular\ frequences$

where ωzn is the nth frequency, and ωpn is the nth pole frequency to design the loop, and r is the magnitude for all poles and zero individually. Once the curve fit for the plant transfer function is obtained it is required to create a Simulink model and add 'transfer fcn' blocks to model the transfer function that is compiled in Matlab to get the nearest curve fit for plant transfer function invert, and loop transfer function. When the transfer function is ready it can be now run on dSpace real-time system to test for pressure with ANC on and pressure with ANC off.

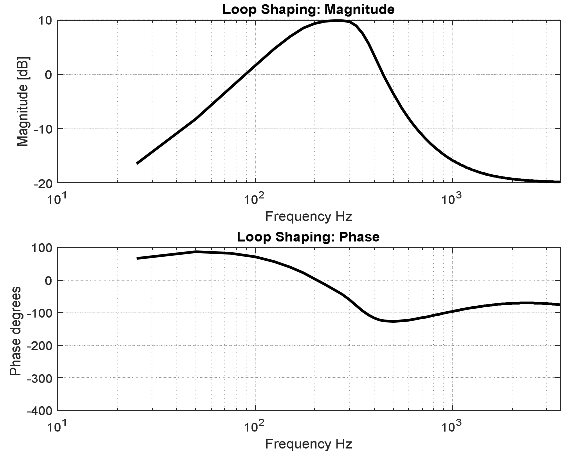

Figure 5: Loop transfer function, magnitude, and phase.

5. Results and Discussion

The ANC developed using this method is tested using a frequency generator where different sets of frequency signals are given to speaker 1 of the system, and ANC is turned ON from the control desk platform. Pressure reduction in the time domain signal is observed on the time plot graph connected to the microphone input of this signal. This developed system was tested for frequencies 100 Hz, 250 Hz, 1000 Hz, and 1250 Hz, and at a different time when ANC is turned on, the pressure reduction was observed.

Another test was done using NI DAQ to get the system's frequency response and obtain the system's sensitivity, which was done by sharing the Microphone output from the cavity with the dSPACE and NI DAQ using a T connector when connecting the BNCs. A frequency sweep signal from NI DAQ was given to speaker 1, and the pressures on the microphone for passive pressure (ANC OFF) and active pressure (ANC ON) were recorded for 100 Hz to 10 kHz Figure **6**.

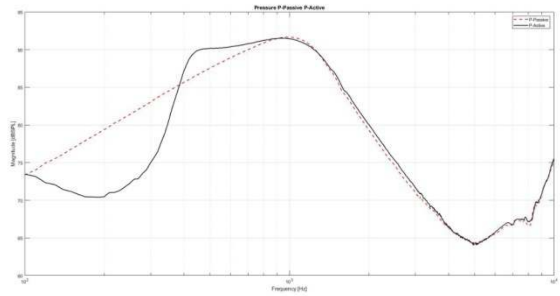

Figure 6: Frequency response of the cavity system for ANC on and ANC off

The sensitivity transfer function Figure 7from equation (5) is compared to the experimental sensitivity obtained using the equation (6).

$$S_{Exp} = 20 * \log_{10} \frac{P_{Active}}{P_{Passive}} \; [dB] \qquad (6)$$

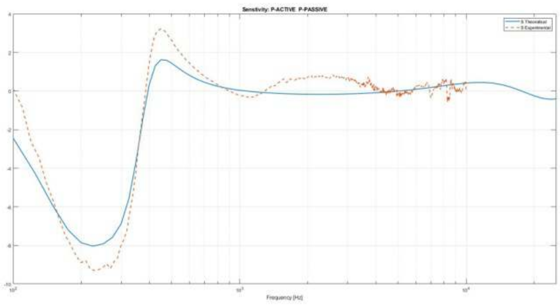

***Figure 7:** Sensitivity transfer function for theoretical sensitivity and experimental sensitivity*

6. Conclusions

Both experiments found that the ANC is working within our desired range of frequencies, and pressure reduction is observed. Up to 10 dB SPL pressure reduction is recorded. This technique can be embedded on a small-scale embedded system and can be used in the cavity of supra-aural headphones to generate low noise pressure inside the headphone cavity and isolate ambient noise from the system.

Acknowledgement

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