ACTIVE NOISE CONTROL USING PAIR OF SPEAKERS

Kumar Pankaj Bharti [1], Vikas Kumar[2]
M.Tech Scholar[1], Astt. Professor[2]
Electronics & Communication Engineering.Deptt.
Geeta Engineering College,Panipat-Haryana (India)
kpankajbharti@gmail.com[1], headece@geeta.edu.in[2]

***Abstract* – Low frequency noise produced in the centralized ducts is annoying causing physical and mental health issues. Passive noise control techniques have failed in low frequency noise mitigation. Active noise control (ANC), which works on the principle of destructive superposition of sound has recently evolved as an effective approach for noise control especially in the low frequency domain. LMS Adaptive Algorithm from MATLAB Simulink is utilized as the ANC techniques to model active noise control in the duct. These results show that the active noise control has considerable improvement, when duel speakers are used for noise cancellation instead of only single speaker, normally used in practical purposes. Hence, the proposed approach acts as an effective visualization tool for noise cancellation, within an arbitrary space, through minor physical changes for the implementation of ANC.**

***Keywords*- Active Noise Control (ANC), LMS Adaptive Algorithm, MATLAB, Simulink Model**

1.INTRODUCTION

Active Noise Control (ANC) is the technique in which noise is attenuated by the same sound but in the opposite phase. ANC has practical use in air conditioning, aircraft cabins, duct noise (ventilation, exhaust, air-conditioning ducts, pipe work, etc.), electronic muffler for the exhaust system, dehumidifier, interior noise, personal hearing protection, headphone, vacuum cleaner, noise attenuation inside the passenger compartment and heavy-appliances operator cabin [1, 5].

Lueg devised and patented the design of an active noise canceller in 1936[6], which used a microphone and an electrically driven loudspeaker to generate a cancelling sound. The frequency content, amplitude, phase, and velocity of the unwanted noise are nonstationary because the noise source's properties and the environment are not constant (time-varying). As a result, an adaptive active control system is required to deal with these shifting features.The error signal (the desired signal minus the actual signal; the error signal is typically defined to be zero) is minimal using adaptive digital filters. Burgess in 1981 [7] developed a duct-noise cancellation system based on adaptive filter theory. The introduction of powerful digital signal processors (DSPs) and adaptive signal processing techniques later in the 1980s had a significant impact on active noise control research[8, 9]. Active noise control's continuous progress involves developing enhanced digital signal processing hardware, transducers, and adaptive signal processing algorithms. More sophisticated adaptive filtering algorithms allow, more significant noise attenuation, more resistant to interference and faster convergence (the equalization of the phase and magnitude of the desired noise and the antinoise for the cancellation to occur) [10].

The coefficients are fixed by the stationary values in the one based on the traditional adaptive filter (i.e. filtered-U recursive LMS method).Another method involves the robust control technique where data is sampled via $H\infty$ control method [11-13]. Swainbank[14] advocated using an extra loudspeaker to cancel out the upstream sound emitted by a control source to improve system performance[15]. The method has been examined in detail under adaptive control setup [16]. Similarly, the effect of Swainbank's source has been investigated in a robust control environment[17]. However, The findings of experiments conducted on existing ventilation systems in homes have not been published.In this paper, to increase system performance, we investigate the use of active noise control systems using a pair of loudspeakers, from filtered LMS algorithm. The loudspeakers act as two independent sources, so, a single-input multiple-output (SIMO) controller was designed similar to Swinbank's source. Many worth mentioning studies, dealing with the design problem for duct active noise control systems in the adaptive control framework [18, 19] and in the robust control framework [20-23] under the simple duct setup have been reported. In contrast, the ventilation system was said to provide adaptive and strong regulation. in [24-26]. More than two speakers will definitely make the ANC more robust and faster, but the system will become bulky.

The work presented here is MATLAB Simulink simulation of ANC from the conventional adaptive filter-based method using Fx-LMS algorithm [27] adapted from the Swainbank's proposed technique [28,29]. The section I consists of an Introduction and brief literature survey. Section II has the proposed algorithm and MATLAB Simulink simulation setup.

Section III has results and discussion. Section IV consists of conclusion and simulation inferences.

II. SIMULATION ALGORITHM AND SETUP

1) Fx-LMS Algorithm and ANC schematic description

This section describes the preferred filtered Fx-LMS algorithm, along with ANC system setup from MATLAB Simulink. The detailed analysis of the proposed ANC schematic applying LMS Algorithm is depicted in Figure 1. The least-mean-square (LMS) algorithm [30] is a simple mathematical algorithm, presented by Widrow and Hoff in 1960. It is used widely in many different engineering areas and finds special applications in areas where a fixed optimal filter is not suitable due to non-stationary signals and environments. In the LMS algorithm, the steepest descent method is modified by using the gradient of the instantaneous squared error in the weight-vector update equation to iteratively find an estimate of the solution to the Wiener-Hoff equations. The LMS algorithm aims to minimize the error signal e(n) by adjusting the FIR filter weights. After convergence, the FIR filter gives a very close estimate of the physical forward path. The feed forward ANC for the long duct shown in Figure 1, has a reference signal (i.e., sine wave) x(n) which is sensed by an input microphone close to the noise source before it passes a loudspeaker. The noise canceller modulates the reference input signal to generate a signal y(n) of the same magnitude but exactly reverse in phase. This reverse-phase signal was used to drive the loudspeaker to produce a cancelling signal, attenuating the duct's primary noise. The feed forward approach's basic principle involves the propagation time delay between the upstream noise sensor (input microphone) and the active control source (speaker), which offers an opportunity to reintroduce noise electrically at the position in the field where it causes cancellation.

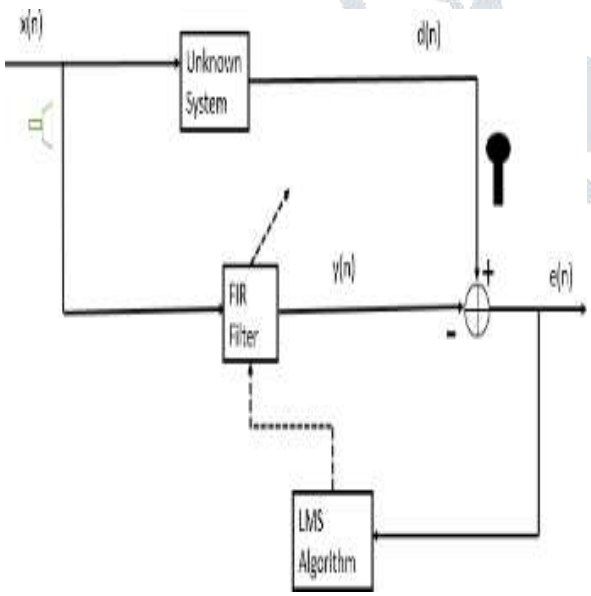

Figure 1: Illustrates a general schematic ANC block diagram for system identification using the LMS algorithm.

The distance between the microphone and the speakers must adhere to causality and high coherence rules.The speaker's noise signal must also be very similar to the input microphone's measured noise, meaning the acoustic channel cannot significantly change the noise. The noise canceller generates an equal-amplitude signal y(n) that is 180 degrees out of phase with using the input signal x(n).This noise is output to a loudspeaker and used to cancel the unwanted noise. The error microphone measures the error (or residual) signal e(n), which adapts to the filter coefficients to minimize this error.Using a downstream error signal to adjust the adaptive filter coefficients does not constitute feedback due to the reference input's error signal. Still, the actual implementations may require additional considerations to handle acoustic effects in the duct.

2) Simulink ANC Setup using Single and Dual Speaker

The MATLAB Simulink setup displaying various components of the dual speaker ANC simulation is depicted in the Figure 2. The Adaptive Fx-LMS algorithm technique used for ANC is represented as LMS block in Simulink.

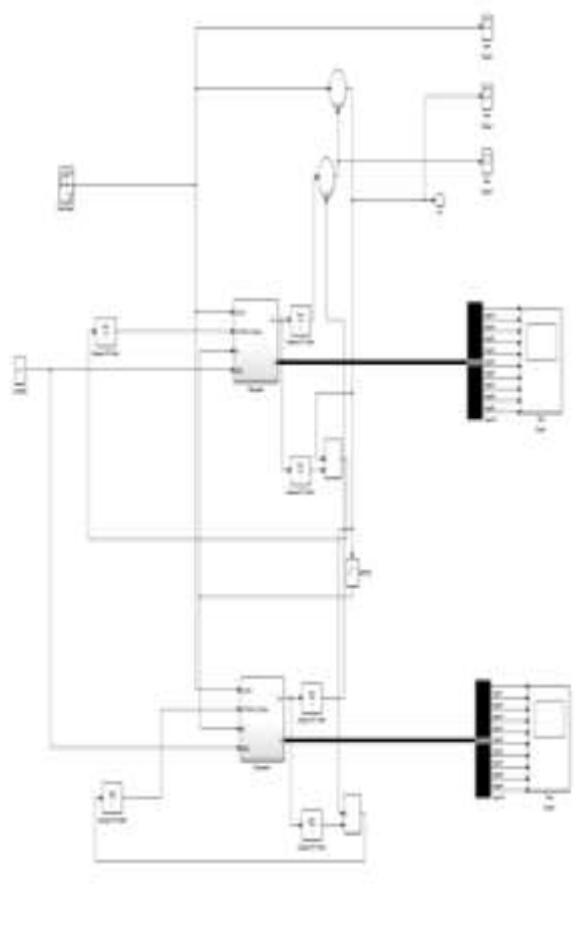

Figure 2: Depicts MATLAB Simulink Model for dual speaker ANC

The LMS Block has sine wave input (Figure 3) via its signal input. The default constant value of $\mu = 0.1$ is adopted through the Alpha port. The dual speakers were placed at diagonally opposite directions (180°) in the straight line. The output signal serves as the error signal as a part of it is used as filter signal after using LMS blocks. Adaptive filters are the filters whose coefficients/ weights change over time in order to adapt to the properties of the measured signal. The Simulink model created has produced two output signals. The signal output of the exterior mic port has white noise while at the mic port it has colored noise output. The filter weights for the coefficient A(z) and B(z) and time step (Ts) is provided via the MATLAB command window. The value is being 0, the Gaussian noise in the signal gets filtered by a lowpass filter. If the value is 1, the noise is being filtered via a bandpass filter. The initial value of the filter weights is zero. The reset port is adjusted as non-zero sample. The filter weights are displayed in the scope remain constant, if the input to the adapt port is zero and the filter weights are not updated. The corresponding output and weights signals are viewed in scope window. If the input to the adapt port is changed manually to 1, the LMS filter weights get updated Thus, the model built is capable of noise cancellation using adaptive filtering techniques.

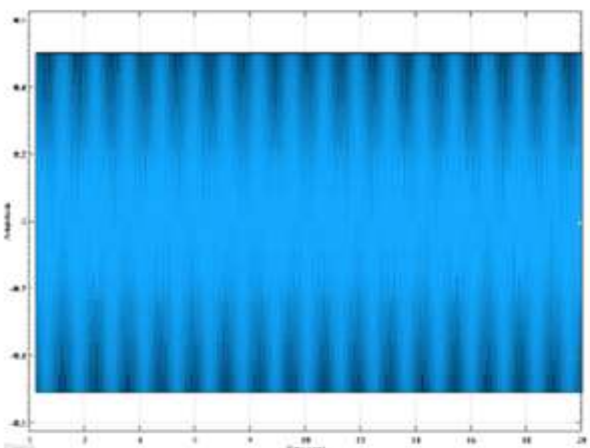

Figure 3: Sinusoidal Chirp Signal used for System Identification

III.RESULT AND DISCUSSION

1) Intercomparison of ANC systems through Single and Dual Speaker systems

MATLAB Simulink version 2015 was applied for the simulation of ANC in the ducts via the Fast Block LMS algorithm in Simulink. The simulation experiment was performed for single as well as dual speaker noise control systems. The dual speaker Simulink model is depicted in Figure 2, single speaker system is much simpler in comparison. The results presented here were performed for μ = 0.1. However, the simulations can be performed with other values of μ as well. The sampling rate of Ts is taken as 1/4000. The input for the filter weight coefficients and time step (Ts) is provided via the MATLAB command window. Both the simulations are performed for upto 20 seconds.

Figure 4 depicts the results from the active noise control simulation via single speaker. The original signal used for the experiment is a sine wave having the amplitude of 0.5 and three different sampling rates i.e., 100 Hz, 150 Hz and 200Hz. The secondary path is identified using the input signal, a sine-sweep signal, as seen in Figure 3.

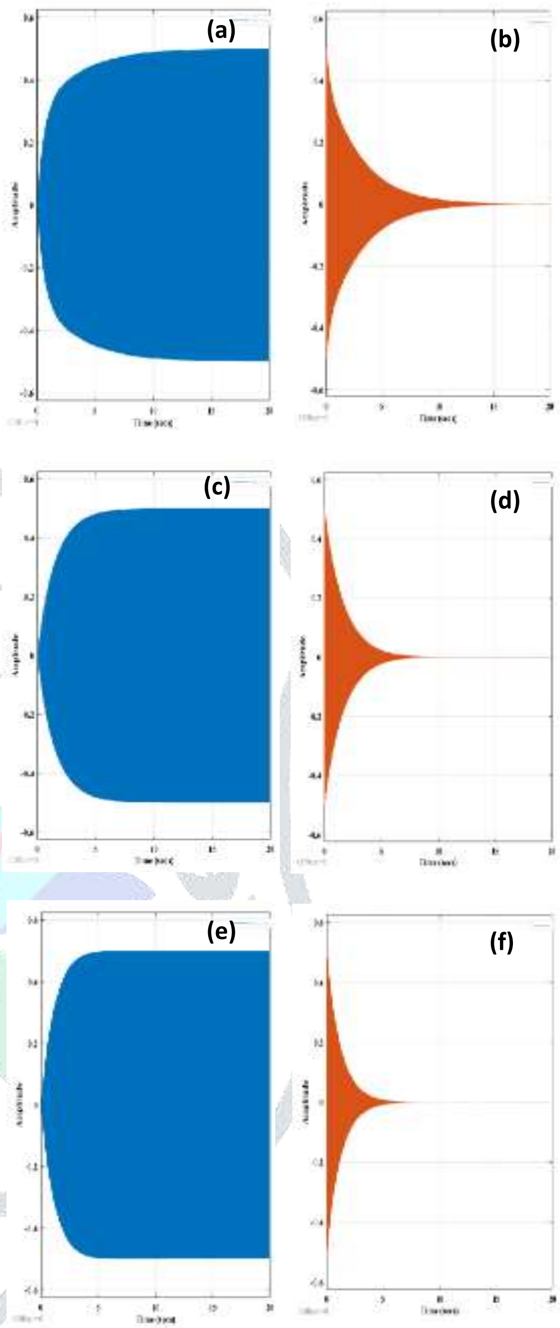

Figure 4: Depicts the Simulink model output (μ=0.1) from single speaker ANC system, (a) control speaker signal at 100 Hz; (b) corresponding error signal convergence at 100 Hz; (c) control speaker signal at 150 Hz; (d) corresponding error signal convergence at 150 Hz; (e) control speaker signal at 200 Hz; (f) corresponding error signal convergence at 200 Hz.

The Figure 4 (a), (c) and (e) show improvement in the control speaker signal as the sampling frequency progresses from 100 Hz to 200 Hz. The output from the error signal converges faster as the input signal sampling frequency increases from 100 Hz. to 200 Hz which is observed in the Figure 4 (b), (d) and (f) respectively.

Figure 5: Depicts the Simulink model output (μ=0.1) from dual speaker ANC system, (a) control speaker signal at 100 Hz; (b) corresponding error signal convergence at 100 Hz; (c) control speaker signal at 150 Hz; (d) corresponding error signal convergence at 150 Hz; (e) control speaker signal at 200 Hz; (f) corresponding error signal convergence at 200 Hz.

Figure 5 depicts the results from the active noise control simulation via dual speaker. The original signal used for the experiment is a sine wave having the amplitude of 0.5 and three different sampling rates i.e., 100 Hz, 150 Hz and 200Hz in order to maintain the homogeneity of the experiment.
Similar results are obtained when the output from dual speakers is analyzed. The trend of decreasing error convergence time and improving control signal is followed here also, with increase in sampling frequency from 100 Hz to 200 Hz. Though, in the dual speaker system the improvement in the error signal convergence with the increasing sampling frequency is not very significant and is below 3s. This was very much evident in the single speaker system. This is another advantage of applying dual speakers for noise control.

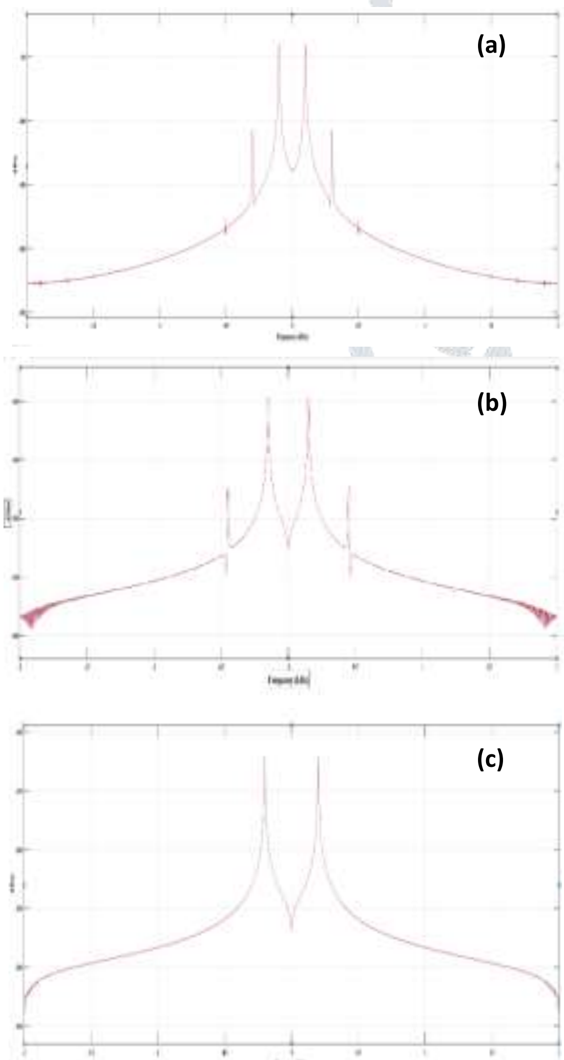

Figure 6: Depicts the power spectrum output (μ=0.1) from single speaker ANC system, (a) at 100 Hz; (b) at 150 Hz; (c) at 200 Hz.

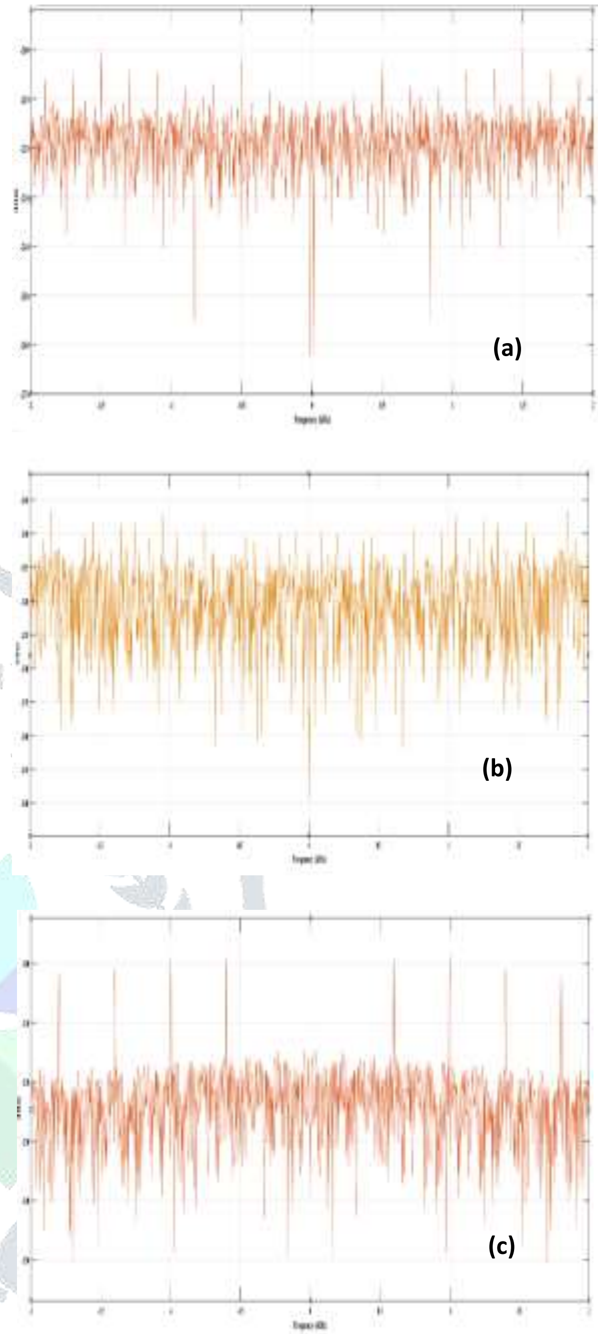

Figure 7: Depicts the power spectrum output (μ=0.1) from dual speaker ANC system, (a) at 100 Hz; (b) at 150 Hz; (c) at 200 Hz.

Figure 6 and Figure 7 demonstrates the power spectrum from the single and dual speaker systems. The above figures show how the control signal slowly grows and then reaches stability with time. The error probe placed at the geometric center of the model shows a gradual fall in amplitude. From all the different frequencies, the control signal and the error signal decreases from 100 Hz to 200 Hz at constant $\boldsymbol{\mu} = 0.1$. This tool is also very useful in identifying areas where there is destructive sound interference happening and where they add constructively.

IV. CONCLUSION

From the study conducted using either single or dual speakers to recreate an ANC problem, it can be noticed that a proper implementation of ANC depends upon various factors such as proper identification of electro-acoustic paths, targeted bandwidths, acoustic delay, electronic delay etc. So, the proper optimization and tuning of the setup is a prerequisite for this technique to actually work. The integration of simulation into the workflow of ANC can help the user to make a preliminary understanding of the problem in hand. This will act as a good visualization tool for understanding the distribution of sound in the domain of interest. The initial results obtained from this numerical analysis of ANC shows close matching trends for single and duel tone noise signals. The values obtained do not shoot up or diverge to give us physically inaccurate results. But, at the same time gives us a reasonable lower limit for the actual response of the system to be implemented. This can be helpful in making an initial feasibility study for ANC in each space. These results help in understanding the role of sampling frequency and the corresponding error convergence time.

REFERENCES

[1] Nelson, P. A. and S. J. Elliott, Active Control of Sound Academic Press San Diego, CA, 1992

[2] S. J. Elliott, Signal processing for Active Control, Volume in the signal processing and Its Applications Series,Academic Press, New York, 2000

[3] Y. Kobayashi and H. Fujioka, "Inexpensive implementation of active noise control system for one dimensional duct with application to a ventilating system," in Proceedings of the 13th International Congress on Sound and Vibration (ICSV '06), Vienna, Austria, July 2006.

[4] I. R. Petersen, H. R. Pota, and M. R. S. Jahromi, "System identification, uncertainty modelling and actuator placement in the robust control of an acoustic duct," in Proceedings of International Conference on Information, Decision and Control, pp. 41–46, Adelaide, Australia, February 2002.

[5] D. S. Bernstein, "What makes some control problems hard?" IEEE Control Systems Magazine, vol. 22, no. 4, pp. 8–19, 2002.

[6] Lueg, P., "Process of Silencing Sound Oscillations," U.S. Patent No. 2,043,416, June 1936.

[7] Burgess, J. C., "Active Adaptive Sound Control in a Duct: A Computer Simulation," J. Acoust. Soc. Am., Vol. 70, No. 3, Sept. 1981, pp. 715– 726.

[8] Widrow. B., and S.D. Stearns Adaptive Signal Processing Prentice Hall, Englewood Cliffs, NJ, 1985.

[9] S. M. Kuo and D. K. Morgan, Active Noise Control Systems: Algorithms and DSP Implimentations, John Wiley & Sons, New York, 1996.

[10] Kuo,S.M., and C. Chen, "Implementation of Adaptive Filters with the TMS320C25 or the TMS320C30," Digital Signal Processing Applications with the TMS320 Family, Volume 3, edited by P. Papamichalis, Prentice-Hall, Englewood Cliffs, NJ, 1990, PP. 191-271.

[11] R.S.Erwin and D.S.Bernstein, "Dicreate-time H2/H∞ control of an acoustic duct: δ-domain design & experimental results," in Proceedings of the 36th IEEE Conference on Decision and Control (CDC '97). Vol.1. PP. 281-282, San Diego, Calif, USA, Dec 1997.

[12] Q.F. Fan, K. Nonami, and M. Nakano, "Active Noise control of one-dimensional exhaust duct using H∞ control theory," Transactions of the Japan Society of Mechanical Engineers, Series C, vol.. 62, no. 597, pp. 1805-1810,1996.

[13] Y. Kobayashi and H. Fujioka, "Active noise control of one-dimensional Duct via samled-data H∞ control," in Proceedings of the 42nd Conference on Decision and Control (CDC'03), vol. 4, pp. 3900-3904, Maui, Hawaii, USA, Dec 2003.

[14] M.A. Swinbanks (1973). "The Active Control of Sound propagation in long ducts", Journal of Sound and Vibrations, vol. 27, pp. 411–436, 1973.

[15] S. Kijimoto, H. Tanaka, Y. Kanemitsu, and K. Matsuda, "Howling cancellation for active noise control with two sound sources," Transactions of the Japan Society of Mechanical Engineers, Series C, vol. 67, no. 656, pp. 52–57, 2001, (Japanese).

[16] J. Winkler and S. J. Elliott, "Adaptive control of broadband sound in ducts using a pair of loudspeakers," Acustica, vol. 81, no. 5, pp. 475–488, 1995.

[17] H. F. Olson, "Electronic Control of Noise, Vibration, and Reverberation," J. Acoust. Soc. Am., Vol. 28, No. 5, 1956. pp. 966–972.

18] Widrow, B., et al, "Adaptive Noise Cancelling: Principles and Applications", Proc. of IEEE, Vol. 63, No. 12, Dec. 1975. pp. 1692–1716.

[19] Elliott, S. J., and P. Darlington, "Adaptive Cancellation of Periodic, Synchronously Sampled Interference," IEEE Trans. on ASSP, Vol. ASSP-33, No. 3, June 1985. pp. 715– 717.

[20] J. Tichy, and G. E. Warnaka, "Effect of Evanescent Waves on the Active Attenuation of Sound in Ducts", Proc. of Inter-Noise, 1983. pp. 435– 438.