Active Noise Control in a Duct by an Analog Neural Network Circuit

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Abstract— **Conventional active noise control (ANC) in ducts has been realized with digital signal processing. The physical size of the conventional ANC systems is usually large owing to the signal processing interval, and the cost of the system depends on the price of the digital signal processor (DSP). This paper proposes a new ANC system with an analog neural network circuit, which will process signals in short time periods without DSP. The proposed neural network circuit has a simple structure consisting of analog multipliers and an integrator, and we simulated the performance of the circuit by HSPICE. We also fabricated a circuit connected to a real duct and confirmed operation of the proposed ANC system.**

Index Terms—**active noise control, analog circuits, neural networks.**

1. INTRODUCTION

Active noise control (ANC) is one of the techniques for noise reduction in low-frequency regions [1]. A typical application of ANC is the cancelation of a noise in a duct, and the conventional ANC system with digital processing is schematically shown in Fig. 1. The duct has a noise source on the left-hand side and the ANC system operates on the right-hand side of the duct to reduce the noise. A reference microphone detects a noise signal, which is converted to a digital signal by an analog to digital (A/D) converter. A digital signal processor (DSP) forms a signal with opposite phase to the noise. A digital to analog (D/A) converter converts the digital signal formed by DSP to an analog signal, which is output by a canceling speaker as sound. The noise is superposed by the sound from the canceling speaker and its power is reduced. Usually, since noise changes its waveform between the reference microphone and the canceling speaker, perfect cancelation is not possible. A residual noise after cancelation is detected by an error microphone as an error signal, which is employed to change the processing in the DSP. Using these methods, the ANC system always operates to reduce the error signal [1,2].

Such digital ANC systems are a useful tool for noise reduction in ducts and have been employed in the industry. However, there are still several difficulties for their application in widespread fields. One of the difficulties is the cost of the system. In particular, DSPs are generally expensive compared with other electronic parts, which makes the cost of the system higher. Another difficulty is the large physical size of the system, which originates from long processing time period in A/D and D/A converters and DSP. Since, during this time period, the noise can proceed a certain distance after signal detection by the reference microphone, we need to place the canceling speaker at this distance from the reference microphone. This distance is usually 10-100 cm, and the physical size of the system is even longer than this distance. To overcome these difficulties in cost and physical size, it is effective to adapt analog circuits for signal processing. Although approaches to achieve ANC by analog circuits have been reported [3], the actual implementation for noise reduction has never been reported. Therefore, in this paper we propose an ANC system for ducts using an analog neural network circuit. This system does not require any A/D and D/A converters and DSPs, and it can process signals faster, and can be fabricated at a lower cost than digital ANC systems.

2. CIRCUITS IN THE PROPOSED SYSTEM

A. *Proposed neural network structure*

The neural network has been accepted as a useful algorithm for the ANC, and the ANC systems with this algorism for noise reduction in ducts have been reported [4-6]. The neural network is usually realized by software or digital circuit, while it can also be constructed by analog circuits [7]. The analog neural network circuit has several disadvantages compared with software or digital circuits. For example, constructing a large network structure is difficult, and there are no adequate devices to memorize information. However, analog neural network circuits can operate very fast and do not need A/D and D/A converters to process signals. Therefore, we applied the analog neural network circuit to the ANC system.

The neural network structure in the proposed system is shown in Fig. 2. The structure has input and output layers, and there is only one neuron for each layer. The input signal, I, enters the neuron in the input layer and then the signal goes to the output layer. Before the signal enters the output layer, the signal is multiplied by a weight, W. Then, the signal outputs through the

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neuron in the output layer as output signal, O. When we assume that neurons do not change the signal, i.e., a neuron has identity function, the output signal at a given time, t, is expressed as

$$O(t)=W(t)\cdot I(t). \quad (1)$$

W should be sequentially updated to minimize square of the error signal, $E\equiv I-O$. When we use the steepest descent method, the amount of the update, ΔW, is obtained by solving $-\mathrm{grad}_{w}E^2$. Therefore,

$$W(t)=W(t-1)+\Delta W, \quad (2)$$

and,

$$\Delta W=-\alpha\frac{dE^2}{dW}=2\alpha\cdot I\cdot E \quad (3)$$

where α is an arbitrary constant called learning coefficient. This neural network always operates to reduce E to a minimum value by updating W from its previous value. For the ANC system, I and O correspond to the signal obtained by the reference microphone and the signal sounded by the canceling speaker, respectively, while E is the error signal obtained by the error microphone.

In (1), the operation is only multiplication, while, in (2), the operations are multiplication and addition. This addition is employed to obtain W by adding the second term to the previous W itself, and then this equation can be expressed by time integration of the second term. Therefore, we need only multiplication and integration circuits to realize these equations for constructing the neural network structure in Fig. 2.

B. *Analog operational circuits*

We adopted the wide range Gilbert multiplier (WRM) shown in Fig .3 [8], as the multiplication circuit. To suppress the common mode noise in the circuit, we employed the fully differential structure. The operation of this circuit is shown in the Appendix and its output can be expressed as

$$V_{out+}-V_{out-}=(V_{in1+}-V_{in1-})\cdot(V_{in2+}-V_{in2-}). \quad (4)$$

On the other hand, for the integration circuit (INT), we adopted an operational amplifier circuit shown in Fig. 4(a). Although this figure shows a single-end integration circuit, we placed this circuit for both the positive-phase and reverse-phase signals to realize the fully differential structure. The values for resistance and capacitance shown in this figure were employed in simulations and actual measurements. Fig. 4(b) shows a high-pass filter (HPF) employed in the circuit. In fact, this HPF is not necessary in the ideal case. However, we usually have an unwanted DC component in the signal due to offset voltage of the circuit. Thus, we employed the HPF to eliminate this DC component, and its cut-off frequency is 10.6 Hz, which is much lower than that of the sound signal to be canceled.

C. *Analog neural network circuits for ANC*

The analog neural network circuit for the proposed ANC system can be shown as a block diagram (Fig. 5(a)). The input signal, I, from the reference microphone enters WRM_1 and is multiplied by W. Then the multiplied signal, O, goes to the canceling speaker and is output to the duct as sound. This multiplication corresponds to (1). The output sound is superposed on the noise coming from the direction of the reference microphone, and the power of the noise is reduced. The residual noise is detected by the error microphone and introduced to WRM_2 as the error signal, E. WRM_2 multiplies I by E, which corresponds to (3) with $\alpha=1/2$. Then, the multiplied signal enters into INT, and INT integrates this signal to generate W. This operations correspond to (2), because (2) is equivalent to time integration of ΔW. HPFs are placed in both the input and the output sides of WRMs for elimination of DC components.

In this figure, we also show the virtual circuit blocks, “Delay” and “Add”, employed in the simulation. There is a time difference between the noise signal at the reference microphone, I, and the noise signal at the canceling speaker, I', due to the distance between them. We emulate this time difference by a delay operation and indicate it as “Delay” in the figure. At the position of the canceling speaker, I' is superposed by output sound. We emulate this superposition by addition of O to I' and indicate it as “Add” in the figure. This addition is achieved by an adder circuit using an ideal operational amplifier. In fact, the delay operation is not enough to emulate difference in signals between the reference microphone and the canceling speaker, because the duct, microphones, speaker and amplifiers have the respective delay depending on frequency. However, the emulation with frequency-dependent delay is complicated, and thus, in this paper, we adopted constant delay for simplicity.

3. VERIFICATION OF THE PROPOSED SYSTEM

A. Circuit simulation

We simulated operation of the circuit shown in Fig. 5 by HSPICE. Since general ANC systems are targeting reduction of noise power in frequency of less than 500 Hz, we employed white noise passing through a low pass filter with 500 Hz cut-off frequency as the input signal. We define the input signal, I, as the noise without control and the error signal, E, as the noise with control. Fig. 6 shows fast Fourier transformation (FFT) spectra for simulated noises with and without control at a delay time of 0.029 ms between I and I'. This delay time corresponds to a distance L = 1 cm between the reference microphone and the canceling speaker. The noise power below 500 Hz decreases by control. Although the noise power above 800 Hz increases by control, there should be negligible effects on the total noise power, since the noise level above 800 Hz is lower than that below 500 Hz. To estimate the total noise power below 500 Hz, we define the canceling effect, C_{eff}, as

$$C_{eff} = 20\log_{10}\frac{\text{total power in signal without control}}{\text{total power in signal with control}}. \qquad (5)$$

In this definition, we obtain positive (and large) C_{eff}, when noise cancelation is effective. The estimated C_{eff} from Fig. 6 is 6.49dB, and thus the circuit in Fig. 5 reduces the noise power in the simulation.

B. Actual measurements using a real duct

We fabricated the circuit simulated above and applied to ANC operation in a real duct whose photograph is shown in Fig. 7. The duct is constructed with acryl plates 6 mm thick and has a dimension of 25 cm × 25 cm × 200 cm. We placed a noise speaker of diameter 15 cm at one side of the duct, and we opened the other side as a noise exit. At the periphery of the noise exit, we attached a buffer material to prevent noise disturbance. The noise evaluation was performed with a microphone at the center of the noise exit. The canceling speaker was the same speaker as the noise speaker and placed on the lateral side of the duct, 60 cm away from the noise exit. At the lateral side opposite to the canceling speaker, many holes with diameter 4 cm (partly 6 cm) were opened up for the insertion of reference and error microphones. During measurements, we bunged up the disused holes with the buffer material. The position of the error microphone was fixed at 5 cm away from the canceling speaker toward the noise exit.

WRMs were fabricated in integrated circuits by a 0.35 μm CMOS process, while INTs were fabricated by the commercially available operational amplifier (New Japan Radio, NJM4558D). Differential to single-end signal converters (DifC) and single-end to differential signal converters (DifG) were also fabricated by the same operational amplifier. We also employed signal amplifiers (Amp) and connected them to microphones and speakers. In Fig. 5(b), we show a block diagram for actual measurements.

The noise signal was the same as that employed in simulations and was converted to sound by the noise speaker. Using the evaluation microphone, we evaluate noises with and without control (the operation of the circuit). The FFT spectra for noises with and without control for L = 1 cm are shown in Fig. 8. Although there seems no canceling effect from this figure, the estimated C_{eff} from (5) is 2.06 dB indicating a significant canceling effect.

Experimentally, the phase of sound from the canceling speaker might be accidentally inverted compared with the noise there because of the phase change during the sound flow across the distance L and the phase change in the signal in the circuit. To confirm the effect of control by the circuit, we estimated C_{eff} with various L in a range from 1 to 110 cm. We compared the experimental results with simulated results at the corresponding various delay times as shown in Fig. 9. Although the experimental C_{eff} were smaller than those obtained from the simulation, we confirmed the canceling effect at any L within the measured range.

The simulation result shows noise reduction in the entire frequency range less than 500 Hz as shown in Fig. 6, but the noise reduction in experiment is observed only for 300-500 Hz as shown in Fig. 8. This would be caused by frequency-dependent delay of the duct, microphones and speaker, which are not modeled in the simulation. However, the above frequency range (300-500 Hz) can be considered as wide enough since the total power of the noise was significantly reduced, as evidenced by the values of C_{eff}.

It has been reported that ANC systems using DSP show canceling effects of 20-30 dB [4,5], and thus the proposed system does not have a canceling effect comparable with the digital systems. The proposed system, however, can make the canceling effect even for a short L of 1cm. This is because the proposed circuit operates very fast. This fast operation is a promising characteristic for downsizing the ANC system. In addition, the number of transistors in the circuit is less than a thousand even if we take into account transistors in operational amplifiers. This small circuit size is also an advantage of the proposed system for fabrication of low cost ANC systems.

4. CONCLUSIONS

This paper proposes a new ANC system with the analog neural network circuit. We confirm the operation of the system and noise canceling effects by both simulations and actual measurements. To our knowledge, this is the first report on the ANC system based on the analog neural network circuit. The results indicate the potential of the analog neural network circuit in the field of ANC.

ACKNOWLEDGMENTS

The author thanks colleagues in his institute, Assoc. Prof. M. Yamada for discussion, Prof. M. Ichimura and Prof. emeritus E. Arai for their support, and Messrs. T. Ichikawa, T. Negami, H. Ohata, T. Kakamu, K. Onoda and T. Tanaka for their technical supports and discussion.

This work is supported by VLSI Design and Education Center (VDEC), The University of Tokyo in collaboration with Cadence Corporation, Synopsys Corporation. The VLSI chip in this study has been fabricated in the chip fabrication program of VLSI Design and Education Center (VDEC), the University of Tokyo in collaboration with Rohm Corporation and Toppan Printing Corporation

APPENDIX: OPERATION OF WIDE RANGE GILBERT MULTIPLIER

Figure 3 shows a circuit for WRM consisting of MOS transistors. In a subthreshold region, the drain current conducting through a MOS transistor, I_{sat}, can be approximated by

$$I_{sat} = I_0 e^{-aV_{gs}}, \tag{A1}$$

where I_0 and a are constants, whose values depend on transistor processes [8]. Applying this equation to *Q1* and *Q2*, we obtain

$$I_1 = I_0 e^{-a(V_{in1+} - V_b)} \quad \text{and} \qquad I_2 = I_0 e^{-a(V_{in1-} - V_b)}. \tag{A2}$$

The sum of these current equals to the drain current of *Qb*:

$$I_b = I_1 + I_2 = I_0 e^{-V_b}\left(e^{aV_{in1+}} + e^{aV_{in1-}}\right). \tag{A3}$$

By solving this equation for the voltage V_b:

$$e^{-V_b} = \frac{I_b}{I_0} \frac{1}{\left(e^{aV_{in1+}} + e^{aV_{in1-}}\right)}. \tag{A4}$$

Substituting (A4) into (A2), we obtain

$$I_1 = I_b \frac{e^{aV_{in1+}}}{e^{aV_{in1+}} + e^{aV_{in1-}}} = \frac{I_b}{2}\left(1 + \tanh\frac{a(V_{in1+} - V_{in1-})}{2}\right), \tag{A5}$$

and

$$I_2 = I_b \frac{e^{aV_{in1-}}}{e^{aV_{in1+}} + e^{aV_{in1-}}} = \frac{I_b}{2}\left(1 - \tanh\frac{a(V_{in1+} - V_{in1-})}{2}\right). \tag{A6}$$

The drain current of *Q7* must be the same as I_1, and it is copied to the drain current of *Q9*. Therefore, sum of the drain currents of *Q3* and *Q4* is the same as I_1 and the drain currents of *Q3* and *Q4* are obtained by procedures similar to those for I_1:

$$I_3 = \frac{I_1}{2}\left(1 + \tanh\frac{a(V_{in2-} - V_{in2+})}{2}\right) \tag{A7}$$

$$I_4 = \frac{I_1}{2}\left(1 - \tanh\frac{a(V_{in2-} - V_{in2+})}{2}\right). \tag{A8}$$

Similarly, the drain currents of *Q5* and *Q6* are obtained as

$$I_5 = \frac{I_2}{2}\left(1 - \tanh\frac{a(V_{in2-} - V_{in2+})}{2}\right) \tag{A9}$$

$$I_6 = \frac{I_2}{2}\left(1 + \tanh\frac{a(V_{in2-} - V_{in2+})}{2}\right). \tag{A10}$$

V_{out+} is sum of I_3 and I_5 multiplied by an equivalent resistance of diode connected transistor *Q11*, R_{Q11},

$$V_{out+} = R_{Q11}\left[\frac{I_1}{2}\left(1 + \tanh\frac{a(V_{in2-} - V_{in2+})}{2}\right) + \frac{I_2}{2}\left(1 - \tanh\frac{a(V_{in2-} - V_{in2+})}{2}\right)\right]. \tag{A11}$$

When *Q12* is the same size transistor as *Q11*, V_{out-} is expressed as

$$V_{out-} = R_{Q12}\left[\frac{I_1}{2}\left(1 - \tanh\frac{a(V_{in2-} - V_{in2+})}{2}\right) + \frac{I_2}{2}\left(1 + \tanh\frac{a(V_{in2-} - V_{in2+})}{2}\right)\right] \tag{A12}$$

Subtracting (A12) from (A11), we obtain

$$V_{out+} - V_{out-} = R\left[I_1\left(\tanh\frac{a(V_{in2-}-V_{in2+})}{2}\right) - I_2\left(\tanh\frac{a(V_{in2-}-V_{in2+})}{2}\right)\right]. \qquad \text{(A13)}$$

Substituting (A5) and (A6), we obtain

$$V_{out+} - V_{out-} = R\left[I_b\left(\tanh\frac{a(V_{in1+}-V_{in1-})}{2}\right)\left(\tanh\frac{a(V_{in2-}-V_{in2+})}{2}\right)\right] \qquad \text{(A14)}$$

When $V_{in1+} - V_{in1-}$ and $V_{in2-} - V_{in2+}$ are both less than 1/α, tanh x can be approximated by x. Then, (A14) can also be approximated by (4).

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