ICIC Express Letters
Volume **16**, Number **6**, June **2022**

ICIC International ©2022 ISSN 1881-803X
pp. 629–637

CONVERGENCE PROPERTY OF ADAPTIVE ALGORITHMS FOR ENGINE SOUND CONTROL

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Received October 2021; accepted December 2021

Abstract. *Engine sound is one of the most important factors affecting the product value of motorcycles. Therefore, there is a need to develop an active noise control (ANC) system that can change the sound quality and enable the design of an ideal engine sound. The filtered-x least mean squares (FxLMS) algorithm has been widely used in ANC systems. However, it is incapable of following fast-changing engine noise. By contrast, the single-frequency adaptive notch (SAN) algorithm enables high-speed processing using scalar operations. Therefore, we conducted a comparison of control performance between LMS and SAN algorithms and a simulation of accelerated engine noise control using the SAN algorithm. The results indicated that the FxSAN algorithm has an excellent tracking performance and can be controlled for multiple frequency components.*

Keywords: Active noise control, Active sound quality control, Engine sound, Filtered-x LMS algorithm, Filtered-x SAN algorithm

1. **Introduction.** In the modern world, automobiles have become essential to our lives. However, with the widespread use of automobiles, noise pollution due to engine noise has become an issue. Although the engine is required to be quiet, the perception of engine noise is also an essential element of an automobile that gives a sense of acceleration and sportiness [1]. Therefore, focusing solely on engine quietness might lower the value of the product. Although every company is particular about the sound quality of its products, the definition of "the ideal engine sound" differs from person to person [2]. Therefore, it is necessary to realize a sound quality control system that adapts the sound to individual preferences without compromising product value while also ensuring environmental friendliness.

The design of engine sound can be achieved by increasing or decreasing an arbitrary frequency range. For this purpose, a system that can follow and control rapidly changing engine noise is required. The filtered-x least mean squares (FxLMS) algorithm has been used in conventional active noise control (ANC), but its computational complexity makes it difficult to follow fast-changing engine speed [3]. In a previous study [4], the single-frequency adaptive notch (SAN) algorithm was used to control the narrow-band noise generated during the run of a four-wheeled vehicle. The results confirmed that the algorithm was effective in reducing the targeted degree components and had a high tracking performance.

In this study, we considered the possibility of sound design for engine noise by increasing or decreasing an arbitrary frequency using the SAN algorithm. First, we compared the control performance of the FxSAN algorithm with that of the FxLMS algorithm to confirm the effective noise reduction effect of the FxSAN algorithm. Then, to control the engine

DOI: 10.24507/icicel.16.06.629

noise, which changes in frequency with a change in speed, we investigated a control method that can follow the engine noise faster.

2. **Methods.** A block diagram of the filtered-x LMS algorithm is shown in Figure 1. The control signal $y(n)$ is the output obtained by the convolution of the reference signal $x(n)$ and adaptive filter $h(n)$; the output $z(n)$ is obtained by the convolution of $y(n)$, and c is the error path characteristics up to the control point. In the LMS, $h(n)$ is updated according to the steepest descent method to minimize the error $e(n)$ between $z(n)$ and the desired signal $d(n)$. $\hat{c}$ is the modeling error path measured previously when the ANC system is configured [5].

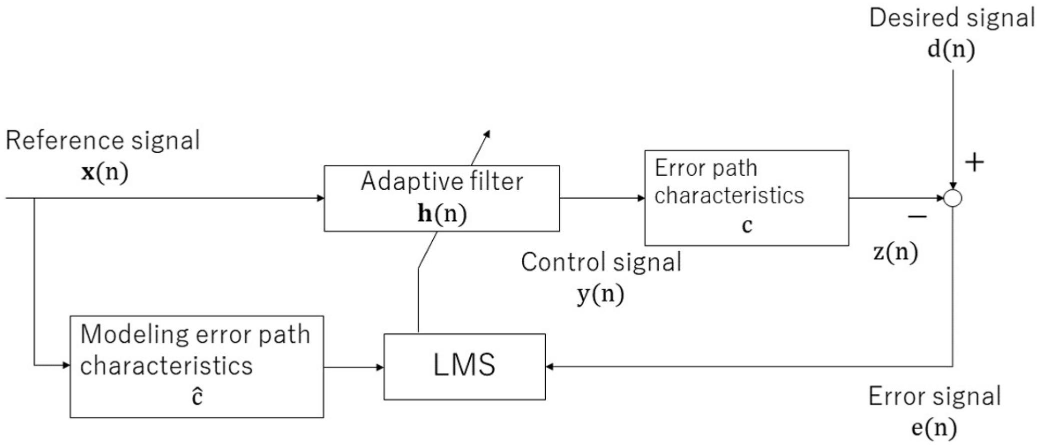

FIGURE 1. Block diagram of the filtered-x LMS algorithm [5]

SAN is an algorithm that uses an adaptive notch filter for a single frequency [6]. The reference signal in the adaptive notch filter is a sinusoidal wave, and two filters are used for each target frequency. Usually, $x_c(n)$ is generated by passing $x_s(n)$ through a 90° phase shifter. Its multiplexing capability enables the control of multiple frequencies.

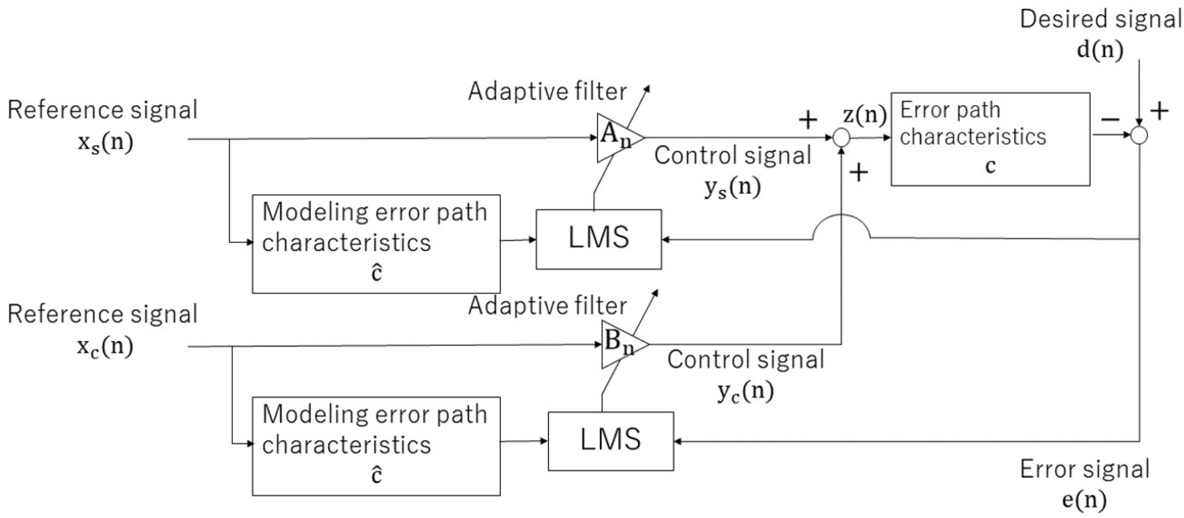

FIGURE 2. Block diagram of the filtered-x SAN algorithm [6]

3. **Comparison of Control Performance Using a Steady Running Sound.** Noise control simulations using the FxLMS algorithm and FxSAN algorithm were conducted using the sound of a motorcycle engine. The sound data were obtained at 3000 rpm during steady and acceleration runs. Figure 3 shows the frequency characteristics of the steady

running engine sound operated from 0 Hz to 300 Hz. In this experiment, three frequency components (50 Hz, 100 Hz, and 150 Hz) were simulated as control targets.

Figures 3 and 4 show the frequency characteristics after control by the FxLMS and FxSAN algorithms, respectively; Figure 5 shows the mean square error with respect to time.

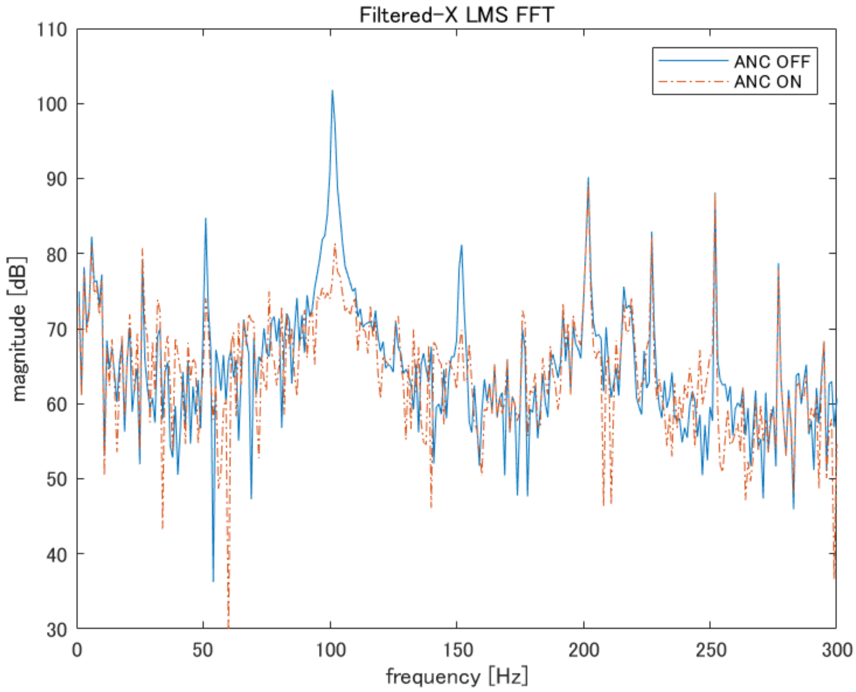

Figure 3. Frequency characteristics of a 3000-rpm steady running engine sound controlled by FxLMS algorithm

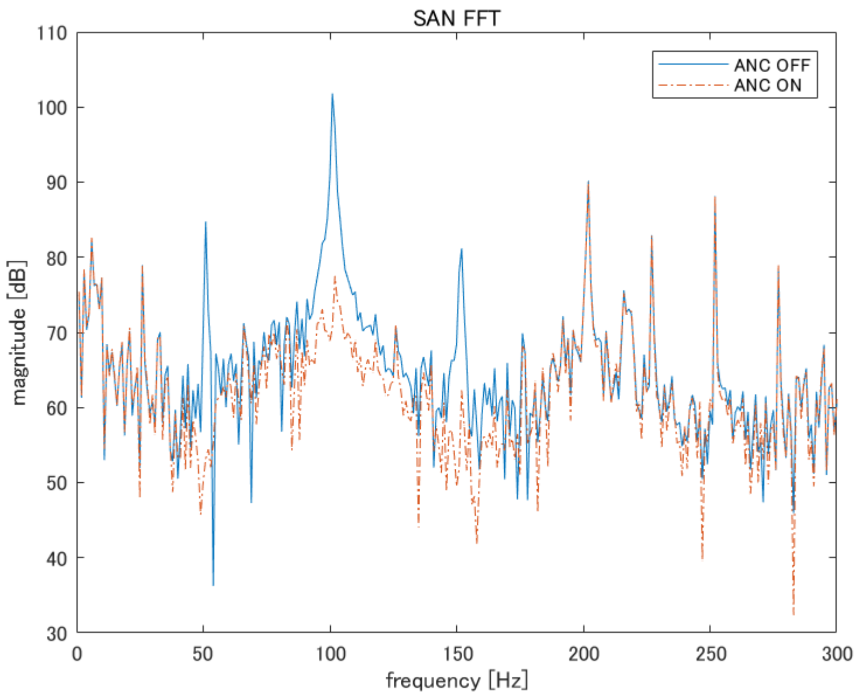

Figure 4. Frequency characteristics of a 3000-rpm steady running engine sound controlled by FxSAN algorithm

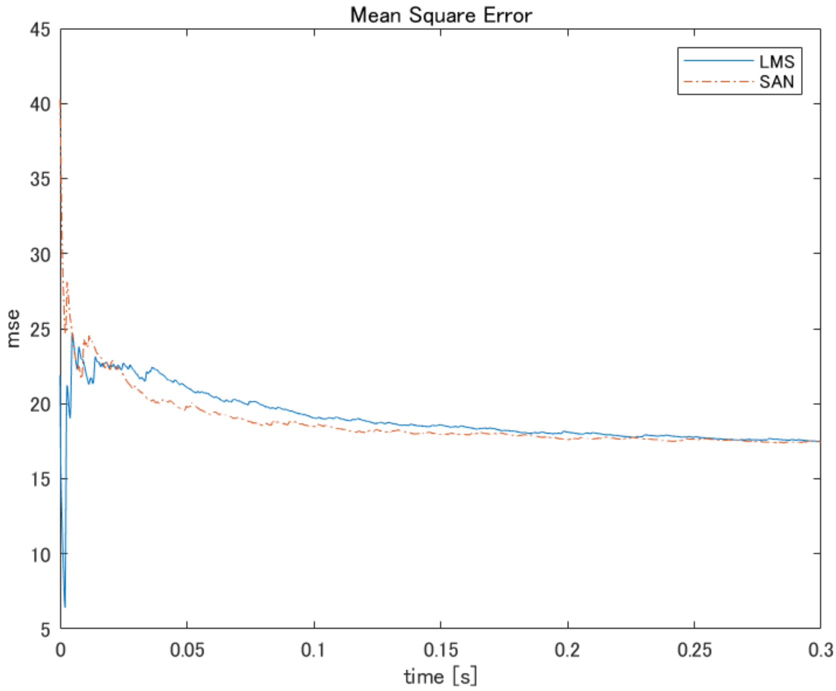

Figure 5. Mean square error of frequency characteristics of a 3000-rpm steady running engine sound controlled by FxLMS and FxSAN

Although both FxLMS and FxSAN were able to reduce the three target frequency components, reduction by FxSAN was approximately 5-10 dB greater than that by FxLMS. The convergence speed of the mean square error was also faster in FxSAN than in FxLMS by approximately 0.1 s. From these observations, we can infer that the convergence performance for a specific frequency is better with SAN as compared to LMS, and that the SAN algorithm is more suitable for noise control. If the adaptive filters used in the algorithm are vectors, all the filter elements must be accumulated and summed, resulting in a large number of operations; the SAN algorithm with a scalar filter uses a much smaller number of operations resulting in a faster convergence than that of the former.

4. **SAN Control for Accelerated Running Sound.** The actual running engine sound is constantly changing because of acceleration and deceleration. To use the changing engine noise as a reference signal for SAN control, a method to generate a sine wave from the engine pulse was investigated. In the case of a four-cylinder engine, each cylinder performs intake, compression, combustion, and exhaust during two revolutions of the crankshaft. In other words, two explosions occur for each revolution of the crankshaft. Therefore, the pulse input is half of the crankshaft revolution. Therefore, the pulse input is half that of the crankshaft revolution. Based on this relation, the following procedure was used to generate a sine wave from the engine pulse.

1) The threshold value was set, and the noise was removed from the pulse data.
2) The time instances at which the pulses rise were recorded.
3) The time for two pulses was calculated and converted to frequency.

Because this method uses the time between the pulses as the period, it cannot follow the frequency change in less than one period.

We implemented the SAN control with a sine wave generated from the pulses as the reference signal; the acceleration engine sound was used as the desired signal. Figures 6 and 7 show the spectrograms before and after the control, respectively. An overall

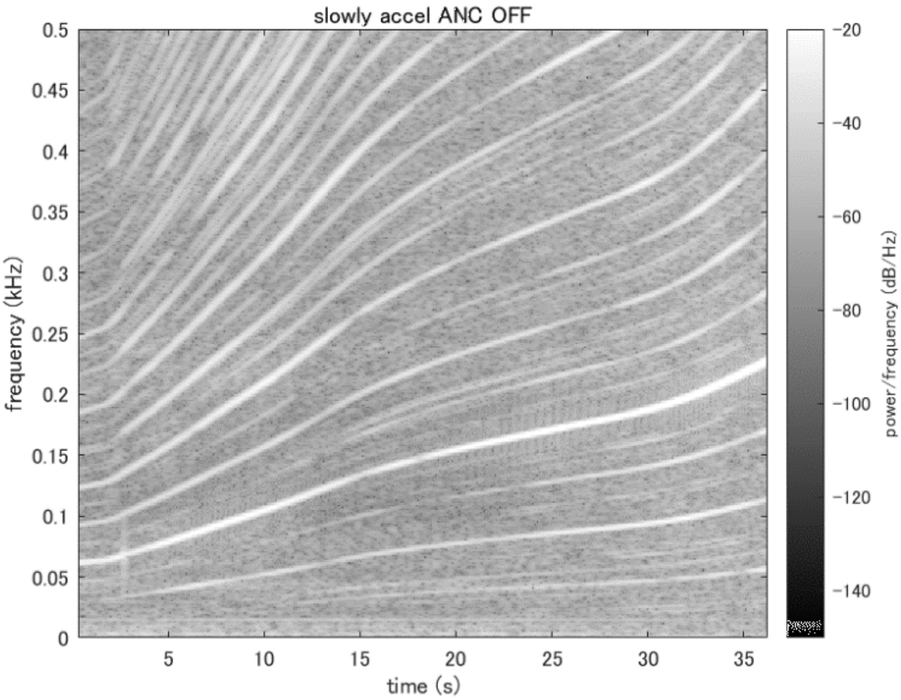

Figure 6. Spectrogram before the control

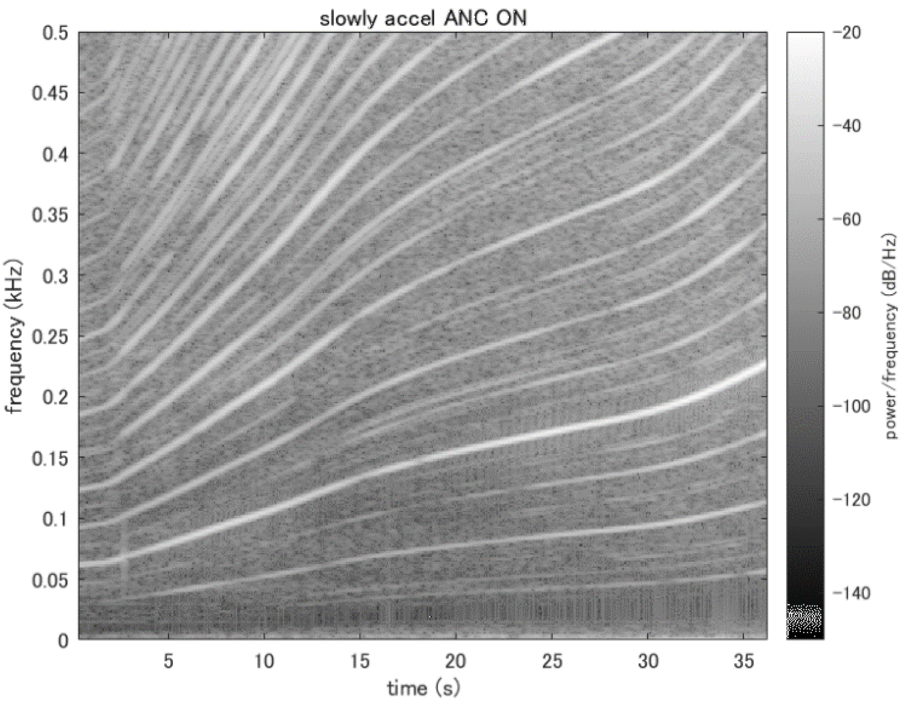

Figure 7. Spectrogram after the control

reduction of approximately 20 dB was obtained after the control, indicating that the generated reference signal was able to follow the acceleration engine sound.

However, the SAN algorithm used in this experiment was a single system, and the reference signal was a single frequency signal based on the amplitude characteristics shown in Figure 8. Therefore, it is unlikely that the generated reference signal will reduce the multiple components included in the desired signal.

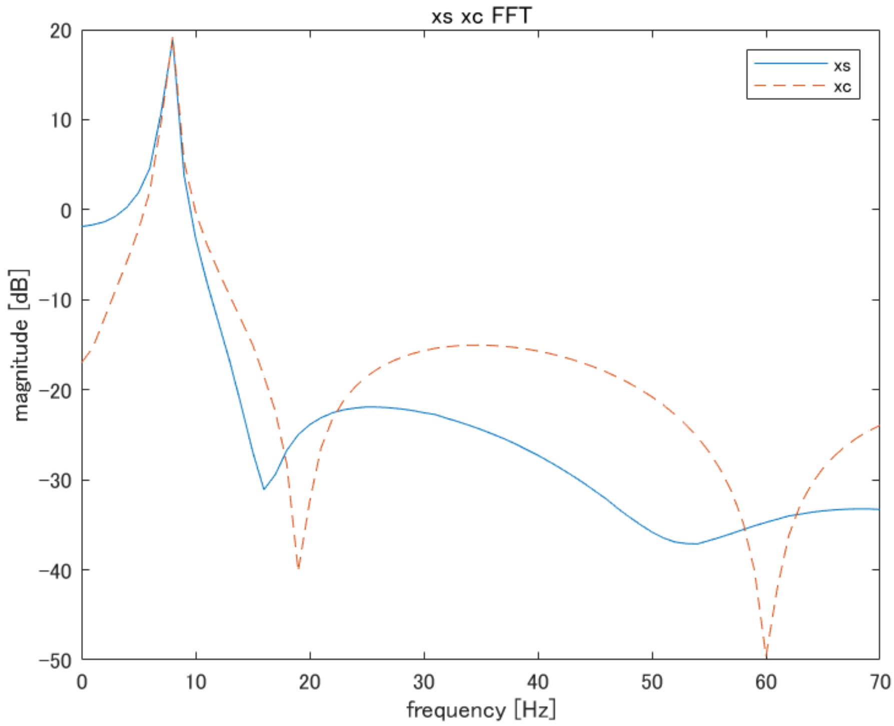

FIGURE 8. Amplitude characteristics of the reference signal

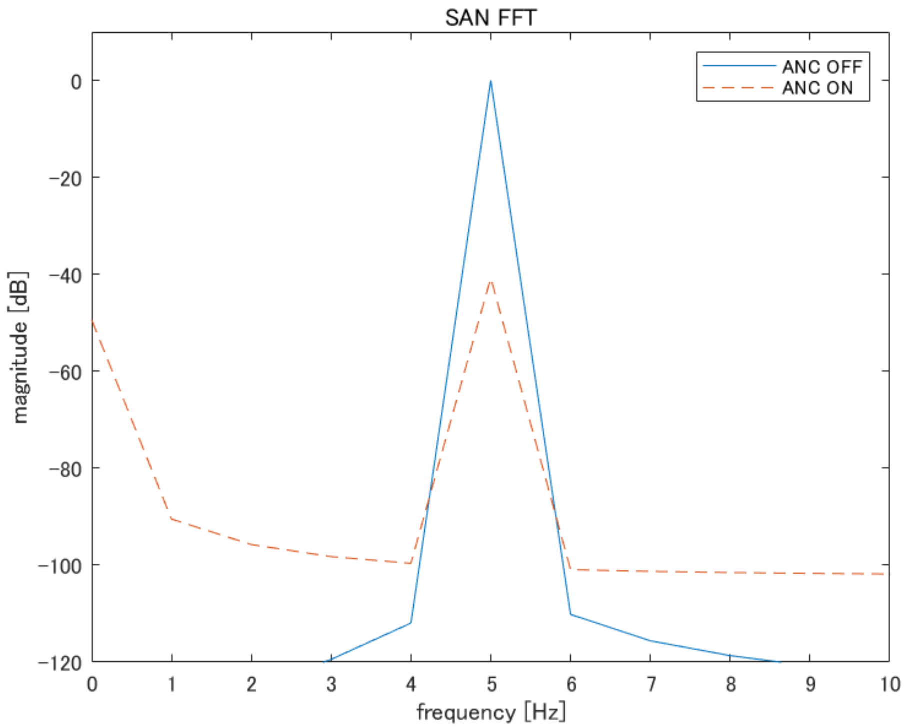

FIGURE 9. Amplitude characteristics of SAN control with uncorrelated frequency combinations

To investigate this phenomenon further, we conducted an SAN control simulation using a combination of simple and uncorrelated frequency signals. The desired signal was a 5 Hz sine wave, and the reference signal was a 2 Hz sine wave; the magnitude of the 5 Hz component was reduced, as shown in Figure 9.

These results show that the 5 Hz component is included in the control signal at the step of calculating the error signal. Therefore, we analyzed the signals generated at each step of the control. Figure 10 shows the time variation of the outputs from two adaptive filters.

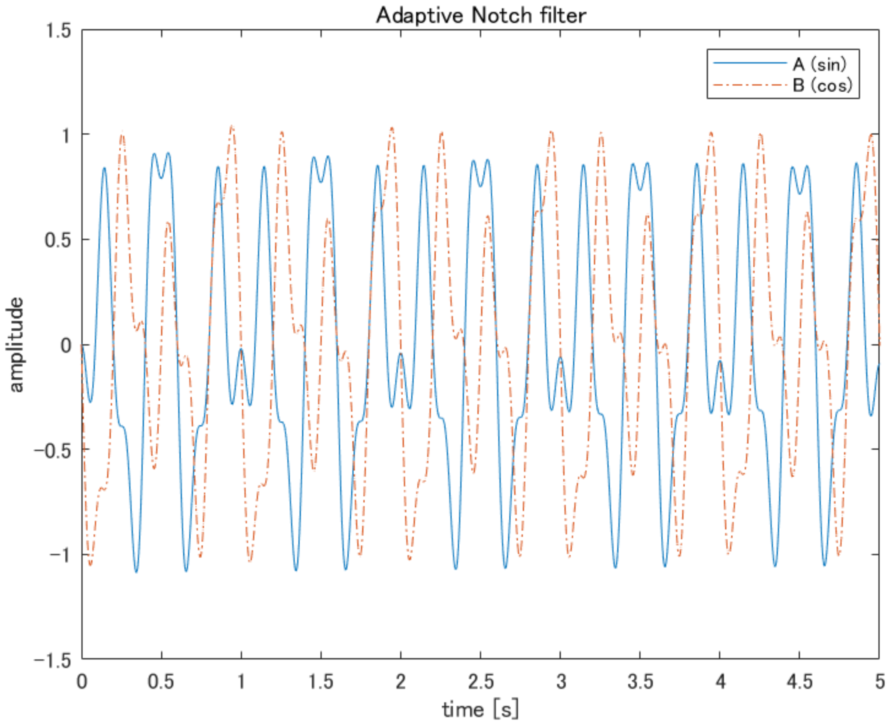

FIGURE 10. Time variation of outputs from adaptive filters

The adaptive filter output is a time-varying system that oscillates with time as it is updated. As the control signal is calculated by the product of the reference signal and the filter coefficients, the desired signal also contains multiple control target components. Subsequently, the control signal has a reduction effect on the peak components of the desired signal. Figures 11 and 12 show the control signals before and after adding sine and cosine waves, respectively.

Each signal contains multiple frequencies before addition; however, they cancel each other out after the addition, and a signal only containing 5 Hz, a component of the desired signal, is generated. We infer from the results as mentioned earlier that in the control of signals with periodic components by the SAN filter, the filter coefficients are affected by the error signal, and that the control signal contains components of the desired signal, thus realizing the control of multiple peak components.

5. **Conclusions.** In this study, we developed an algorithm for optimal control sound of a motorcycle engine by conducting noise control simulations of the FxLMS algorithm, which has been used in the past, and the FxSAN algorithm, which is effective for signals with periodic components and requires less computations compared to the former. In order to compare the control performance, control simulations were conducted for specific frequency components in the steady running engine noise. In addition, we investigated a control method that follows the noise of an accelerated driving engine and attempted to speed up the adaptation algorithm.

The comparison of control steady running engine sound led to the following observations. Although both algorithms showed a reduction effect, the control efficiency of the

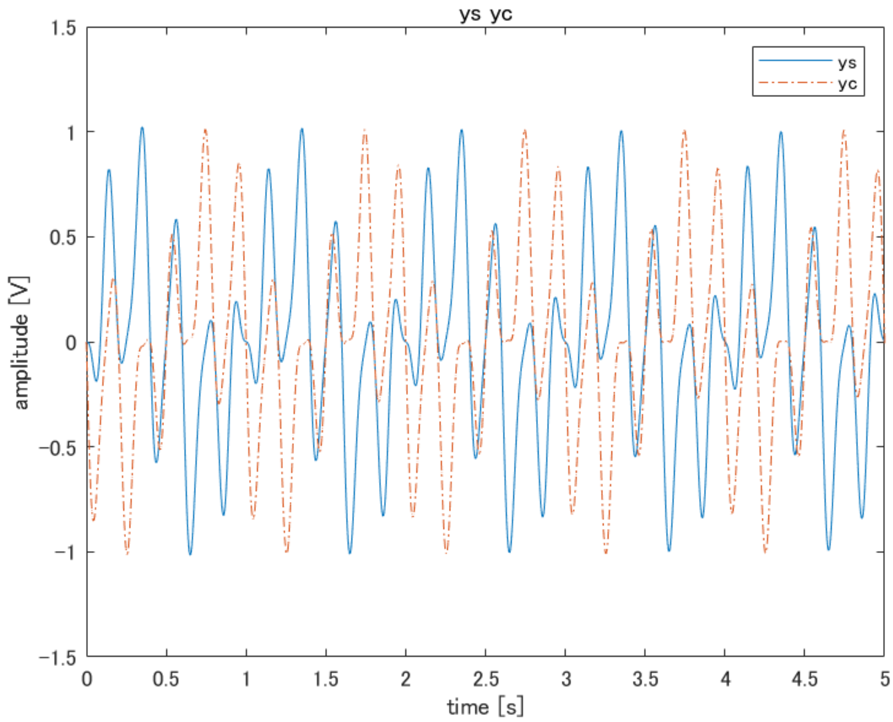

FIGURE 11. Control signals before addition

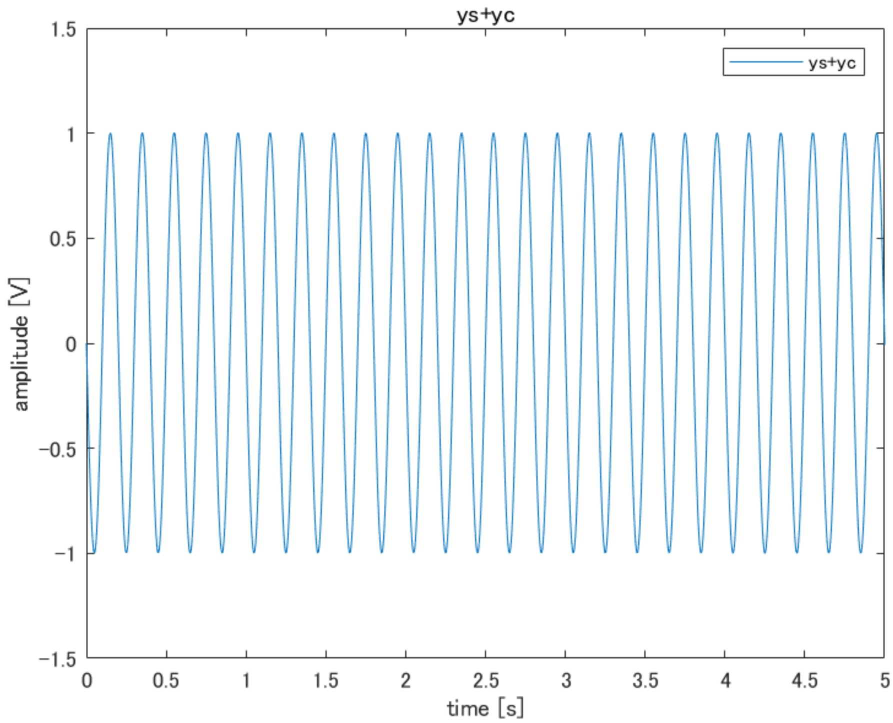

FIGURE 12. Control signals after addition

SAN filter was superior to that of the LMS filter in terms of the amount of reduction and convergence speed.

For the SAN control of accelerated engine noise, a method of generating a sine wave from engine pulses was devised, and control simulations were conducted using the generated sine wave. It was confirmed of the excellent tracking performance of the FxSAN algorithm and its control for multiple frequency components through the time-varying oscillation of the adaptive filters' outputs.

In the future, we plan to study the tracking method for engine noise that changes in frequency within one cycle and to conduct simulations using actual data to clarify the acceleration tracking performance.

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