A PARALLEL ACTIVE ORDER SOUND EQUALIZATION METHOD TO IMPROVE VEHICLES SOUND QUALITY DURING ACCELERATION

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The sound quality inside vehicles, especially different order sounds of engine, is an important factor affecting comfort sensation of passengers. Engine order sounds vary with time dramatically during vehicle acceleration so it generally couldn't be controlled by conventional active noise control system. In this paper, a parallel active order sound equalization method is proposed to compensate or eliminate the order sounds to improve sound quality inside vehicles. For tracking the change of order sound, the integral of engine speed is generated as the reference signal. And different gain parameters are used to adjust order sounds independently. Simulations are implemented in advance to estimate the performance of this system as well as determine the parameters used in practical experiments. Finally, the experiments are carried out to verify the validity of such method.
Keywords: parallel active order sound equalization; sound quality; acceleration

1. Introduction

The sound quality inside vehicles is an important factor affecting comfort sensation of passengers. Engine sound is the main noise source of vehicles, especially the order sounds influence the sound quality significantly. Sung-Hwan Shin and Takeo Hashimoto [1] have found that level control of half-order components is required to obtain the optimum order spectrum profile. Leopoldo P.R.deOliveira, Bert Stallaert, etc. [2] have proposed that pre-defined order-level vs. RPM profile should be achieved to meet a desired sound quality target. So if the order lines can be designed to meet the preference of passengers, the sound quality would be greatly improved.

The primary components of engine sound distribute in low frequency range, which is just the target frequency band of active noise control method. Active noise control (ANC), regarded as an effective supplement of passive method to control low-frequency noise, utilizes the superposition of two out-of-phase sound signals to generate a quiet zone. There are two types of ANC structure, namely feedforward ANC and feedback ANC, according to whether the reference signal is required or not. The feedforward ANC is better in terms of stability performance and wideband control effect than feedback ANC. The most widely used algorithm of feedforward ANC was proposed by Morgan [3] and Widrow [4], which is denoted as filtered-*x* least mean square (FxLMS) algorithm. This method requires reasonable reference signal because only the contained frequency components in reference signal can be controlled, but generally, many kinds of reference signals can't reflect the engine order sounds exactly. The engine order frequencies are the multiples of fundamental frequency with constant ratios, which contribute to the use of active noise equalization method on adjusting the order sounds.

Active noise equalization (ANE) method was proposed by Kuo [5] in the application of controlling harmonic sound at a given frequency by predetermined amount. This method divides the secondary control signal into two branches. The pseudo-error signal is minimized, while the actual error noise can be cancelled, amplified, attenuated or intact according to the choice of gain parameter. The essence of this method is to adjust the placements of poles and zeros of the closed-loop transfer function on the unit circle. The control system performs as a notch filter which can reduce (or amplify) the primary spectral component within a narrow band centring of the reference frequency. The broad band ANE method was also proposed by Kuo [6]. Instead of dividing the secondary control signal into two branches, a shaping filter is introduced to control the residual error noise in broad frequency. Broad band ANE can adjust the residual error noise according to the shape of shaping filter, but generally, the shaping filter is not easy to design. By contrast, the reference signal of narrow band ANE system is a pure cosine wave, which is convenient to generate from the engine speed when the order sounds need to be controlled.

In this paper, a parallel active order sound equalization (PAOSE) method is proposed to adjust the order sounds for the purpose of improving sound quality inside vehicles during vehicle acceleration. For tracking the change of order frequencies, the integral of engine speed is generated as the reference signal. Different division ratios match with different order lines. The rest of the paper is organized as follows. In section 2 PAOSE method is introduced with the computational procedure. Simulations are implemented in advance to estimate the performance of this system in section 3. And the practical implementation of PAOSE method is demonstrated in section 4. A single channel PAOSE system is established on vehicle headrest in anechoic chamber. Some key points of the practical implementation are presented in details and control results are analysed. Finally, the conclusions are given in section 5.

2. PAOSE method

The order sounds of engine need to be equalized for desired order line vs. RPM profile. Because of the order frequencies are the multiples of fundamental frequency with constant ratios, the active noise equalization method is viable for adjusting the order sounds. So the PAOSE method is proposed to synthesize several single ANE structures in parallel for multi frequency control at the same time. In order to acquire accurate reference cosine signals in real time, the reference signals are generated directly from the analog signal of engine speed.

The block diagram of proposed PAOSE system is shown in Fig.1. N single structures are integrated in total, which means that N frequency components can be adjusted independently. The cosine wave reference signals $x_{11}(\mathrm{n})$, $x_{21}(\mathrm{n})$, …, and $x_{N1}(\mathrm{n})$ are generated from the fundamental frequency accompanied with their quadrature reference signals $x_{12}(\mathrm{n})$, $x_{22}(\mathrm{n})$, …, and $x_{N2}(\mathrm{n})$. The secondary control signals $y_1(\mathrm{n})$, $y_2(\mathrm{n})$, …, and $y_N(\mathrm{n})$ are calculated using the reference signals and its corresponding adaptive filter weighting coefficients $w_{i1}(\mathrm{n})$ and $w_{i2}(\mathrm{n})$, $i = 1, 2, \cdots, N$. And then every secondary control signal is divided into two branches. One branch is multiplied by $1-\beta_i$ and summed up to drive the secondary loudspeaker. Then this stimulus passes through the secondary path, the impulse response of which is denoted as $s(n)$, and is collected by the error microphone as the result of superposition of primary sound $d(n)$ and secondary sound $y'(n)$. The other branch is multiplied by β_i and filtered by the estimation of secondary path transfer function $\hat{S}(z)$, and the impulse response of $\hat{S}(z)$ is denoted as $\hat{s}(n)$. The β_i is termed gain parameter and determine whether this order sound will be compensated or eliminated. There are 2N adaptive filters in this configuration and they are updated by the reference signals, the quadrature reference signals and pseudo-error signals.

As for the *i-th* frequency component, the pseudo-error signal is expressed in Eq. (1). And the closed-loop transfer function from the desired signal to the error signal is derived in Eq. (2) [5].

$$E_i'(z) = E(z) - \beta_i \cdot Y_i(z) \cdot \hat{S}(z)\,. \tag{1}$$

$$H(z) = \frac{E(z)}{D(z)} = \left[1 - S(z)\sum_{i=1}^{N}\frac{(1-\beta_i)G_i(z)}{1-\beta_i \cdot G(z)\cdot \hat{S}(z)}\right]^{-1}. \tag{2}$$

Where $G_i(z)$ is the open-loop transfer function from the pseudo-error signal $e_i'(n)$ to the *i-th* secondary control signal $y_i(n)$. Pseudo-error signal $e_i'(n)$ in Eq. (1) is minimized by the FxLMS algorithm. The calculation procedures are listed in Table 1.

Table 1: Calculation procedures of PAOSE method

Step 1	Secondary control signal	$y_i(n) = x_{i1}(n)\cdot w_{i1}(n) + x_{i2}(n)\cdot w_{i2}(n)$
Step 2	Filtered reference signals	$x_{i1}'(n) = x_{i1}(n) * \hat{s}(n)$; $x_{i2}'(n) = x_{i2}(n) * \hat{s}(n)$
Step 3	Filtered secondary control signal	$y'(n) = \left[\sum_{i=1}^{N}(1-\beta_i)y_i(n)\right] * s(n)$
Step 4	Residual error signal	$e(n) = d(n) - y'(n)$
Step 5	Pseudo-error signal	$e_i'(n) = e(n) - \beta_i y_i(n) * \hat{s}(n)$
Step 6	Update of weighting coefficients	$w_{i1}(n+1) = w_{i1}(n) + 2\mu_{i1}x_{i1}'(n)e_i'(n)$; $w_{i2}(n+1) = w_{i2}(n) + 2\mu_{i2}x_{i2}'(n)e_i'(n)$

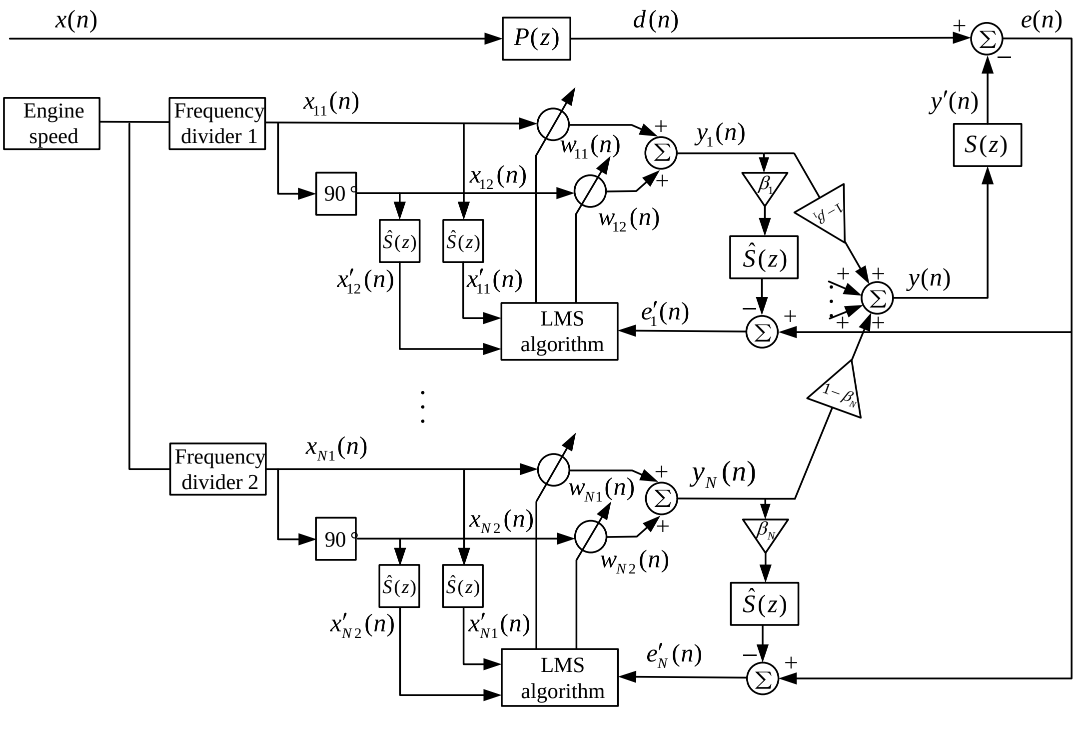

Figure 1: Block diagram of proposed PAOSE method.

3. Simulations

Simulations are conducted in advance to estimate the performance of PAOSE method. For observing the effectiveness of this method intuitively, one special control target is selected. That's elimination of all the 2nd, 4th and 6th order sounds and the control result is shown in Fig.2. The upper figure is spectrogram of primary noise and the following figure is spectrogram of the noise after control. The primary signal, speed signal and secondary path impulse response are collected from the practical setup introduced in next section. As shown in Fig.2, this method is effective for controlling the engine sounds.

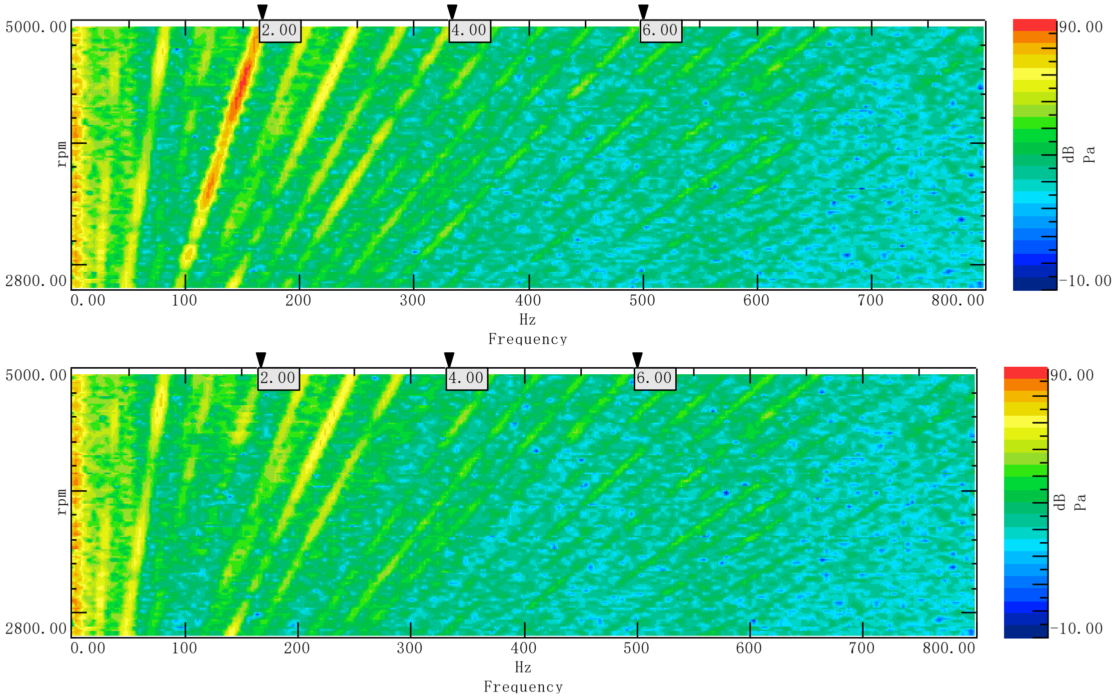

Figure 2: The spectrogram of primary noise and the noise after control in simulation.

4. Practical implementation of PAOSE method on vehicle

In order to eliminate the influence of road noise and wind noise, the vehicle is accelerated in neutral gear in anechoic chamber. The practical implementation setup is shown in Fig.3. Error microphones are positioned near the headrest of the rear chair, to represent the perception of human ears. Two loudspeakers are used as secondary control sources and located behind the headrest with independent channel and power amplifier. But in this paper, only the right microphone and right loudspeaker are used for the sake of simplicity. The MircoLabBox of Dspace Company is used for data acquisition, real-time control and generation of secondary signal. The key points of this method are construction of reference signals and secondary path modelling which are illustrated in later subsections.

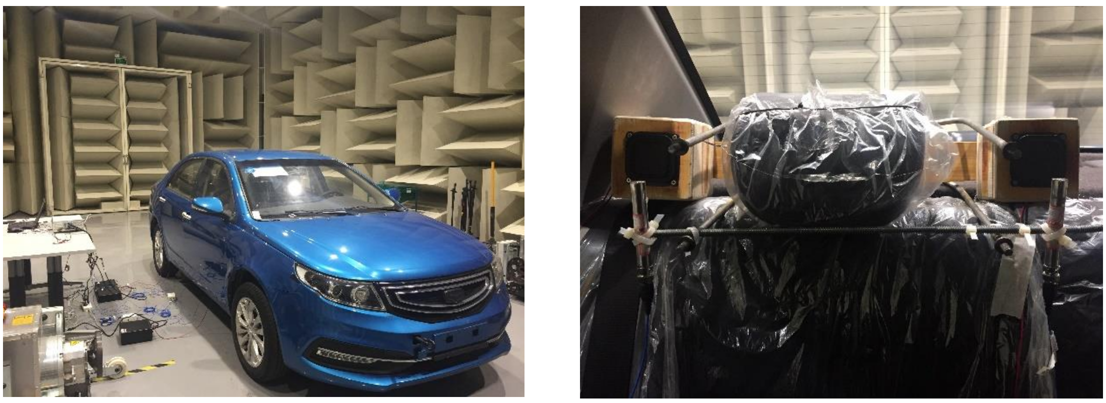
Figure 3: The experimental setup of proposed PAOSE system.

4.1 The construction of reference signal

The frequencies of order sounds vary with engine speed but have fixed ratios with the fundamental frequency which determines that order lines can be controlled using PAOSE method. An important point is to generate the reference signals. The RPM-8000-OBD2 of KMT Telemetry Company can read the engine speed and vehicle speed in cars via CAN "on board diagnostics" interface with analog and pulse outputs with low time delay and small fluctuations. The RPM analog output is 0.5V corresponding to 1000r/min of engine speed. Connecting the analog output of RPM-8000-OBD2 with MiroLabBox, the engine speed can be acquired in real time.

The reference signal of the *i-th* (i =1, 2, …, N) frequency component is constructed as Eq. (3).

$$x_{i1}(t) = A_i \cos\left[\theta_i(t)\right]. \tag{3}$$

Where A_i is the amplitude of the *i-th* reference signal. And $\theta_i(t)$ is generated using Eq. (4).

$$\theta_i(t) = \int_0^t \omega_i(t)dt = \int_0^t 2\pi f_i(t)dt = \int_0^t 2\pi \frac{\mathrm{RPM}}{(\bullet)}dt. \tag{4}$$

Where RPM is the real-time engine speed with the unit of r/min and $(\bullet)$ denotes the frequency division coefficient. Different $(\bullet)$ s are used in terms of different order sounds. Take a 4-stroke, 4-cylinder engine as an example, $(\bullet)$ is 30 for the 2nd order sound, while for the 4th order sound, $(\bullet)$ is 15 and so on.

4.2 The adaptive off-line secondary path identification

The secondary path has an important effect on the system stability and control performance. Generally, there are two methods for the modelling of secondary path [7]. The off-line method is effective in terms of stable secondary path, but if the secondary path changes in the course of control process, the on-line modelling method has to be used with the sacrifice of signal to noise ratio because a white noise will be introduced throughout the control process. Researchers [7] have testified that if the modelling error between the phases of modelled path and physical path is less than 90°, the control performance is still guaranteed.

Considering that the vehicle is accelerated in neutral gear in the process of the experiment, so the adaptive off-line secondary path identification method is adopted. Band pass white noise is used as the stimuli of loudspeaker. The time domain history of error signal and the impulse response of the estimation of secondary path after convergence are shown in Fig.4. The length of impulse response is chosen as 80. It can be seen that the error signal converges to a small value within 5 seconds.

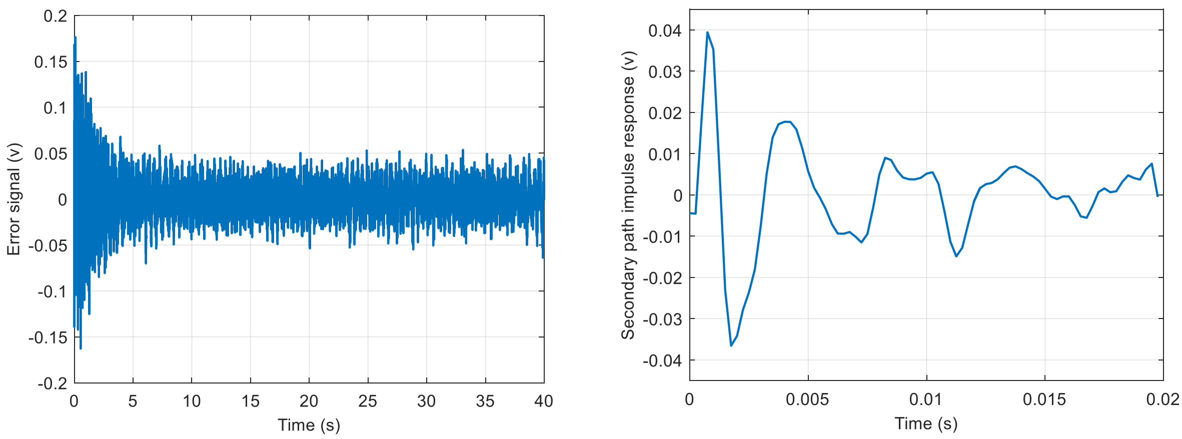

Figure 4: Time domain history of error signal and impulse response of the estimation of secondary path.

4.3 The control results of PAOSE method

Two control targets are verified using PAOSE method and aforementioned experimental setup. The first target is mitigation of the transient sound pressure level (SPL) of the 2nd and 4th order sounds and enhancement of the transient SPL of the 6th order sound, and the denominators of Eq. (4) are 30, 15 and 7.5 respectively for construction of reference signals. The second target is mitigation of the transient SPL of the 2nd order sound and enhancement of the transient SPL of the 4th and 6th order sounds. The reason why we choose these two control targets is such modification of order sounds will increase the pleasant sensation of passengers from the subjective evaluation test. The sound should be quiet and rich by mitigation of lower order lines but with the existence of higher order lines.

The gain parameters used in the first target for three order sounds are $\beta_1 = 0$, $\beta_2 = 0$, $\beta_3 = 2$ and results are shown in Fig.5 and Fig.6. In Fig.5, the left graph is the spectrogram of primary sound measured at the position of right microphone, while the right graph is the spectrogram of sound after control using the proposed PAOSE method. The sound pressure level of each order sound is extracted in Fig.6. From the spectrograms and SPL maps, it can be seen that the 2nd order line is mitigated, the 4th order line is almost eliminated, and the 6th order line is enhanced. But the mitigation of 2nd order line is not so obvious. That's because of the harmonic components of loudspeaker. The coupling of three structures imposes influence on each other. But the control effect is still considerable considering the 4th and 6th order sounds. Meanwhile, the change of average SPL levels of each order sound are listed in Table 2. As shown in case 1, about 4dB mitigation of the 4th order sound and 4dB enhancement of the 6th order sound can be achieved using such method.

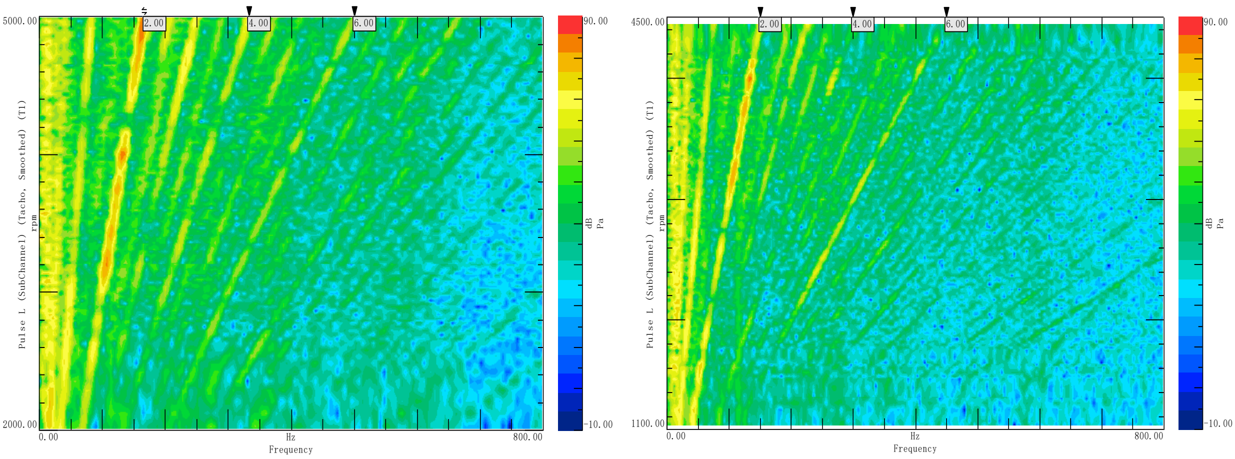

Figure 5: The spectrograms of primary sound and sound after control.

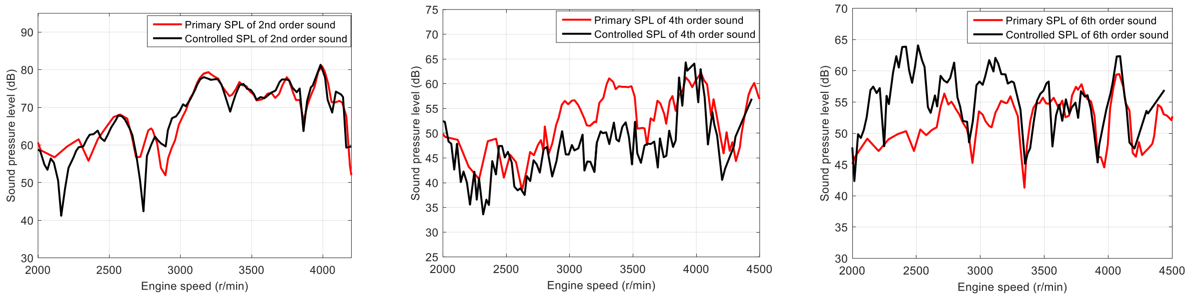

Figure 6: The SPL maps of primary sound and sound after control of each order sound.

The gain parameters used in the second target for three order sounds are $\beta_1 = 0$, $\beta_2 = 2$, $\beta_3 = 2$ and the results are shown in Fig.7 and Fig.8. In Fig.7, the left graph is the spectrogram of primary sound measured at the position of right microphone, while the right graph is the spectrogram of sound after control using the proposed PAOSE method. And the SPL maps are also extracted in Fig.8. From the spectrograms and SPL maps, it can be seen that the 2nd and 4th order lines are mitigated, and the 6th order line is enhanced. The change of average SPL levels of each order sound are listed in Table 2. As shown in case 2, about 3dB mitigation of the 2nd order sound, 7.6dB enhancement of the 4th order sound and 2dB enhancement of the 6th order sound can be achieved using such method.

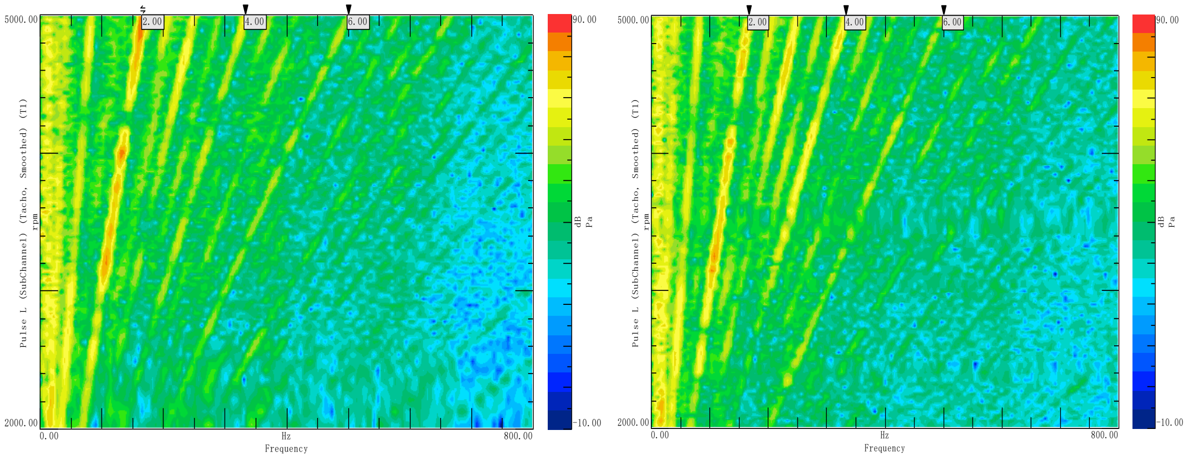

Figure 7: The spectrograms of primary sound and sound after control.

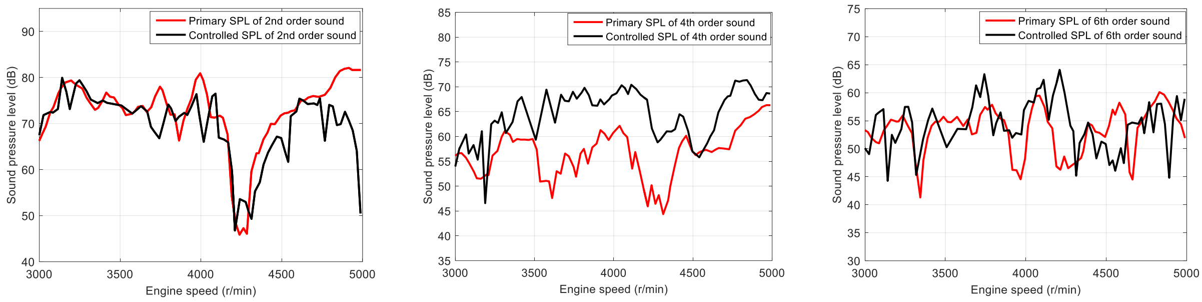

Figure 8: The SPL maps of primary sound and sound after control of each order sound.

Table 2: Average SPL of each order sound

	Order	2^{nd} order	4^{th} order	6^{th} order
Case 1	Engine speed range (r/min)	2000～4200	2000～4500	1800～4500
	SPL of primary sound (dB)	70.52	53.78	51.78
	SPL of sound after control (dB)	70.32	49.59	56.02
Case 2	Engine speed range (r/min)	3000～5000	3000～5000	1500～5000
	SPL of primary sound (dB)	74.67	57.78	52.11
	SPL of sound after control (dB)	71.98	65.36	54.12

5. Conclusions

A parallel active order sound equalization method (PAOSE) is proposed in this paper. This method utilizes the engine speed to generate the reference signal so that the order frequencies can be adjusted intentionally to meet the preference of different people. The simulation result supports this method as a useful tool for designing the order sounds. The procedures of this method are listed in details. And the practical implementations are carried out in anechoic chamber with the predefined control targets. The effectiveness of such method is verified with the experimental results.

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