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An Exhaustive Study of Computer Simulation and Modeling for Active Noise Control

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ABSTRACT

Active noise control computer modeling is proposed to improve the noise reduction effect of the active noise control system. The transmission function of the adaptive control active noise reduction system is analyzed, first using the collected noise samples as the initial noise, performing a subjective assessment of the degree of fatigue, and then performing multiple linear regression experiments. Fatigue level data were used to obtain a noise-level sound quality model. Finally, the filter design is performed according to the fatigue score frequency response curve in the model. Based on this, the filter-error least mean square (FeLMS) algorithm is optimized and simulated on the computer. The simulation results show that filter-x least mean square (FxLMS) algorithm and A-FeLMS algorithm, after being controlled based on Y-FeLMS algorithm, the annoyance degree is reduced by 0.64 and 1.12 levels. Computer simulation and subjective evaluation experimental results show that the noise annoyance degree decreased significantly after the control. Compared with the traditional FxLMS algorithm, the FeLMS algorithm is more effective in improving the sound quality.

***Index Terms*—**Computer simulation, active noise control, FeLMS algorithm, acoustic quality model, filter design

I. INTRODUCTION

Active noise control (ANC) is based on the principle of destructive interference of sound waves. By canceling the sound source (secondary sound source), a radiated sound wave with the same amplitude and opposite phase as that of the canceled sound source (primary sound source) is generated, which cancels each other, thus achieving the purpose of reducing noise [1]. Compared with the traditional passive noise control, ANC has a good noise control effect in low frequency band. It has many advantages such as small size, light weight, and so on, so it has always been a research hotspot in the field of noise control (as shown in Fig. 1). Universal filter-x least mean square (fxLMS) algorithm is used to realize it. At a certain point in space, the noise control method to reduce noise by destructive interference between primary noise and secondary noise is called ANC [2-3]. Traditional ANC takes the sound power of noise after destructive interference as the objective function and achieves the noise reduction effect by minimizing the sound power of noise after cancellation. After the noise reduction, the middle and high-frequency components are dominant due to the significant reduction of the low-frequency components in the noise. Although the sound pressure level is obviously reduced, the sound quality may not be significantly improved, and it may even sound more uncomfortable [4-5]. ANC for sound quality optimization is an important development trend. The current research on sound quality control can be roughly divided into two categories. The first type is to adopt the traditional active control algorithm and then perform subjective sound quality evaluation on the controlled noise.

In the research based on sound quality control algorithm, the optimization goal in most research work only starts from the objective acoustic parameters, without considering the spectrum difference of different noises. Aiming at this defect, this paper takes cabin noise as the research object, designs the annoyance score experiment, establishes the sound quality model, and takes annoyance as the control objective. The annoyance degree is calculated in Bark domain, and a filter and a control system based on filter-error least mean square (FeLMS) algorithm are designed according to the results. The effect of improving sound quality is verified by computer simulation.

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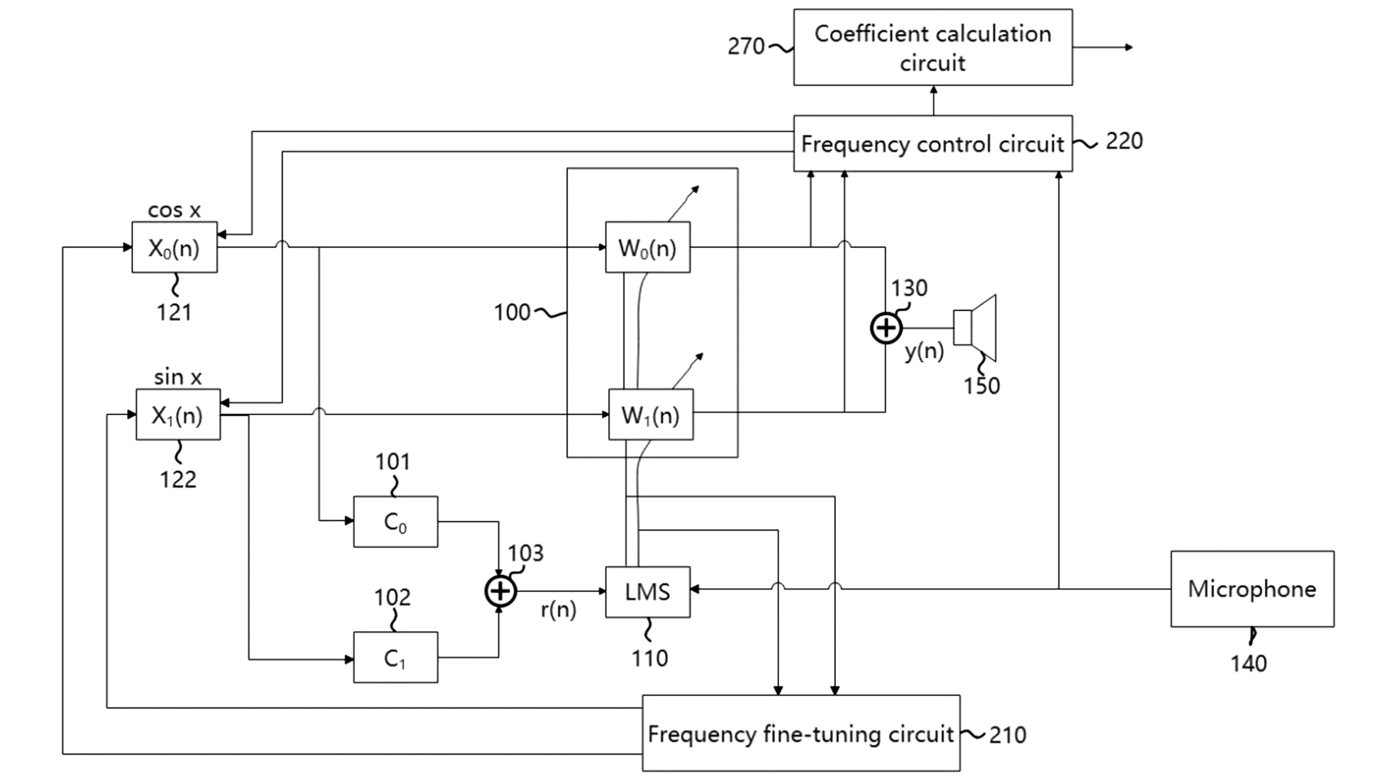

Fig. 1. Active noise suppression device.

II. LITERATURE REVIEW

ANC for sound quality optimization is an important development trend. At present, the research on sound quality control can be roughly divided into two categories [6]. The first one is to adopt the traditional active control algorithm and then evaluate the subjective sound quality of the controlled noise. Le et al. evaluated the loudness, sharpness after control, and the change of the comprehensive index generated based on these two evaluation indexes [7]. Wang et al. evaluated the change in loudness and roughness of noise after active control [8]. Zhong et al. evaluated the improvement of loudness, annoyance, and comfort after active control [9]. Xiao jun evaluated the semantic clarity after noise reduction. The other was to improve the control algorithm and then change the control system structure to improve the sound quality [10]. Guo et al. proposed FeLMS algorithm and active noise equalizer (ANE) system. ANE system could attenuate or amplify the secondary noise, which provided a method and idea for improving the sound quality. The FeLMS algorithm added a filter to the FxLMS algorithm to filter the noise after cancellation, thus improving the sound quality. Follow-up studies on sound quality control algorithms were mostly based on these two algorithms for optimization and expansion [11]. Song et al. filtered the error signal and reference signal based on the A-weighted curve [12]. Yoon studied the gain coefficient of the ANE system, divided the car interior noise into 24 frequency bands according to the critical frequency band, and set the gain coefficient separately to reduce the loudness and sharpness of the car interior [13]. Aslam et al. studied the effectiveness of FeLMS algorithm for reducing loudness [14]. Saravanan and Santhiyakumari filtered the error signal and the reference signal based on the isoacoustic curve [15]. Zhao et al. put forward some fast algorithms for FxLMS algorithm, which mainly reduce the number of iterations of weight coefficients in the sampling period or change the implementation of the algorithm to reduce the computational complexity [16].

III. RESEARCH METHODS

A. Adaptive Active Control Algorithm

Adaptive filtering is a Wiener filter that can automatically adjust its unit impulse response to achieve optimization. When designing an adaptive filter, it is unnecessary to know the autocorrelation functions of signal and noise in advance, and even if these autocorrelation functions change slowly in the filtering process, they can be adjusted adaptively and automatically to meet the requirement of minimum mean square error [17]. The method of adjusting the weight coefficient of adaptive filter is called adaptive algorithm.

1) Least Mean Square and FxLMS Algorithm

The ultimate goal of adaptive filtering is to find the best weight vector W_0. The least mean square (LMS) algorithm is a simple and effective recursive method for W_0. The recursive formula of weight coefficient is obtained by LMS steepest descent method. The weight vector $W(n+1)$ at the next moment is equal to the current weight vector $W(n)$ minus one proportional $\nabla(n)$. That is, as shown in (1):

$$W(n+1) = W(n) - \mu\nabla(n) \tag{1}$$

In this formula, μ is the coefficient of convergence.

In practical application, the square of the error signal is generally taken. $e^2(n)$ can be used as the estimation of the mean square error gradient. $\hat{\nabla}(n)$ is the true value. As shown in (2):

$$W(n+1)=W(n)+2\mu e(n)(n) \quad (2)$$

The updating method of the above weight vector is called LMS algorithm. FxLMS algorithm is an algorithm based on LMS algorithm, which is adjusted by considering the problem of secondary path delay. The algorithm adds a secondary path. $s(n)$ is the filter output. Through the response after the secondary path, the filter weights $y(n)$ are adaptively adjusted and optimized.

The iterative formula of filter weights is as shown in (3):

$$W(n+1)=W(n)-2\mu e(n)r(n) \quad (3)$$

where r is the filtered -x signal. The relationship between the filter -x signal vector and the reference signal vector is shown in (4):

$$W(n+1)=W(n)-2\mu e(n)r(n) \quad (4)$$

where $h_s(n)$ is the impulse response of the secondary path. The block diagram of the algorithm is shown in Fig. 2.

2) FeLMS Algorithm

The FeLMS algorithm adds an error filter to the FxLMS algorithm to filter the error signal E. $h_w(z)$ is the added error filter. $r'(n)$ is error filter. $e'(n)$ is an error signal. The iterative formula of filter weights is shown in (5):

$$W(n+1)=W(n)-2\mu e'(n)r'(n) \quad (5)$$

Using error filter $h_w(z)$, error signals entering the controller can be analyzed. The control system based on FeLMS algorithm can control the noise in a specific frequency band. The FeLMS algorithm is helpful to solve the problem that the low-frequency components of noise are greatly reduced after noise reduction, which leads to insignificant improvement of sound quality, and can improve sound quality on the basis of active noise reduction. On the error filter, in terms of setting, most of the research work only aims at setting the objective parameters of sound quality. In this paper, considering the characteristics of human ear and noise spectrum, the model of noise sound quality is established through the annoyance evaluation experiment to set the error filter [18, 19].

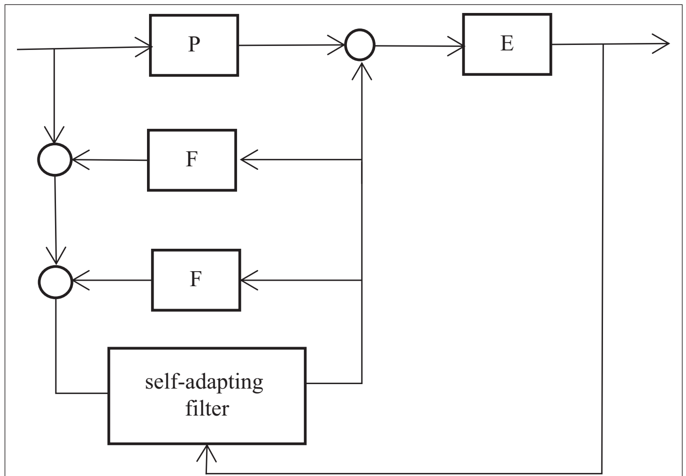

Fig. 3. Transfer function block diagram of adaptive system.

B. Computer Simulation of Adaptive System of Active Noise Control

The adaptive system shown in Fig. 3 is used for computer simulation. For this system, it is assumed that the transfer functions of the detection microphone and the error microphone have ideal flat frequency response without any phase shift, which is consistent with the actual situation [20]. The transfer function of the secondary loudspeaker adopts the amplitude and phase characteristics of airflow loudspeaker, which has the characteristics of low-pass filter. The cut-off frequency is assumed to be 500 Hz, which is replaced by finite impulse response (FIR) filter designed according to second-order dynamic system in computer simulation. The transfer function of the controller adopts LMS algorithm with FIR filter structure. In the following computer simulation, this paper takes $l_1 = 5m$. The stability l_2 of the algorithm should be taken into account, and it should be as small as possible when the actual installation allows. In the simulation, $l_2 \approx 0$.

The purpose of simulation is to explore and study:

a. stability and convergence rate of the algorithm;
b. noise reduction characteristics of active noise elimination system:

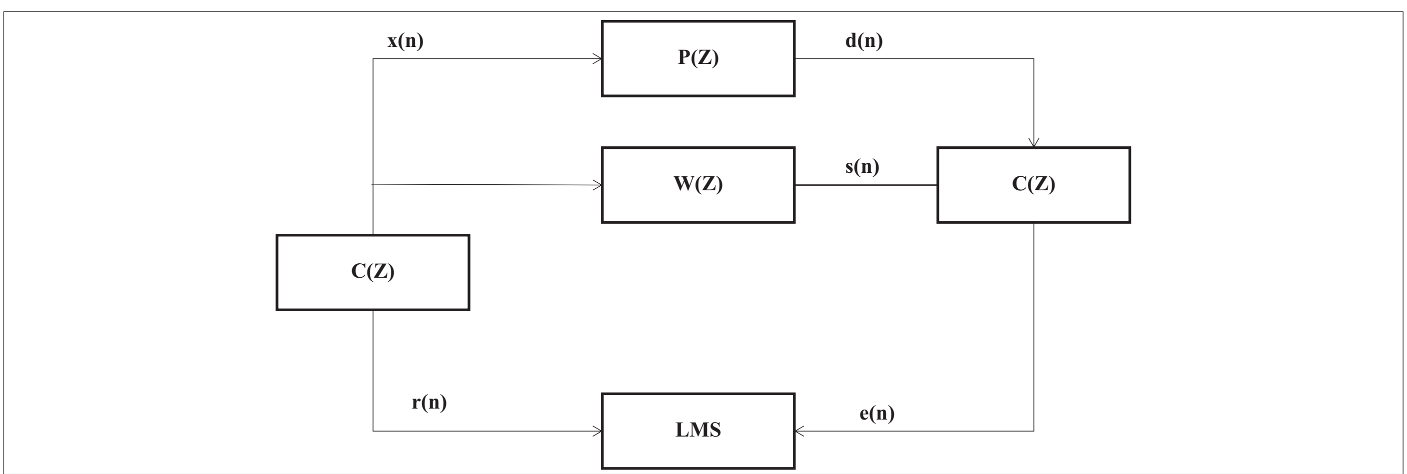

Fig. 2. Block diagram of the FxLMS algorithm.

c. compare the noise reduction characteristics of fixed-parameter controller and adaptive filter;
d. observe the influence of acoustic feedback on the noise reduction of the system.

The system noise reduction amount or noise reduction amount is defined as the difference between the noise levels in the downstream and in the pipeline when the active noise reduction system is not used and when the active noise reduction system is used, expressed in decibels. Some simulation results are given below:

1) Active Noise Reduction System with Silent Feedback

Under the condition of silent feedback, the noise reduction of the two systems is calculated when the input signal is 0–500 Hz broadband noise: (1) The controller adopts the fixed parameter FIR filter and the actual transfer function of the secondary speaker, and the average noise reduction is 11.1 dB, and the maximum noise reduction is 20.6 dB. Because it is assumed that there is no acoustic feedback in the system, the system can still work stably when the loop gain of the controller is greater than L. Obviously, the amplitude and phase distortion added by the fixed parameter controller to the loudspeaker results in the reduction of noise reduction. (2) The controller adopts the adaptive filter with FIR structure and LMS algorithm and transmits the function in the secondary loudspeaker. For the ideal response, the average noise reduction of broadband noise is 0.4 dB. For the actual transfer function of the secondary speaker, the noise reduction characteristics of the system are shown in Fig. 4, and the average noise reduction is reduced to 27.5 dB. Thus, when the transfer function is non-ideal, the noise reduction of the system will decrease, and the noise reduction will still be 15 dB in the frequency band with large phase change and gain coefficient greater than 1. Compared with the system with fixed parameter controller, the adaptive system can compensate [21, 22].

2) Active Noise Reduction System with Audible Feedback

When there is audio feedback, the adaptive FIR filter of LMS algorithm is used to calculate the noise reduction of the system, and the smart signal is assumed to be broadband noise. The results are as follows.

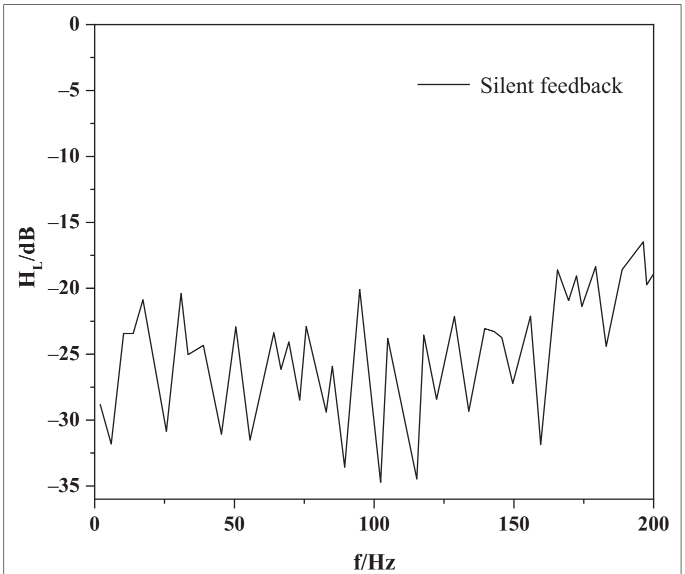

Fig. 4. Noise reduction characteristics of adaptive system (silent feedback).

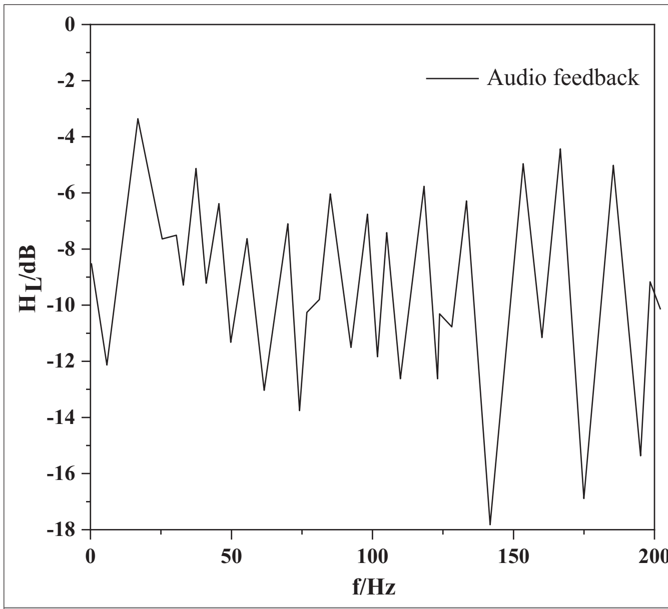

Fig. 5. Noise reduction characteristics of adaptive system (audible feedback).

For ideal function transfer, the noise reduction of the system is shown in Fig. 5, and its average noise reduction is 10.2 dB. Compared with the case of silent feedback, the noise reduction is about 30 dB lower. For the actual transmission function, the average noise reduction is only 4.8 dB. It can be seen that the acoustic feedback has a great influence on the noise reduction of the system [23]. On the other hand, in the case of audible feedback, the system will diverge without the method of adaptive control when the airflow speaker is used to transfer the function, which indicates that the adaptive control has a positive effect on the control acoustic feedback and system instability caused by non-ideal characteristics is very effective.

If the input signal is the sum of two single-frequency sinusoidal signals, its amplitude is 3:1. When the transfer function is ideal, the noise reduction of the two frequency signals is 47.8 dB and 31.2 dB. The large amplitude signal has a large amount of noise elimination, and the adaptive filter has a good cancellation effect on the large signal [24].

If the controller is implemented by infinite impulse response (IIR) filter, the average noise reduction of broadband noise is listed in Table I.

IV. RESULTS AND DISCUSSION

A. Active Noise Control Algorithm and Computer Simulation and Modeling

In this section, an acoustic sample in the subjective evaluation experiment is selected, and the filter is designed according to the frequency spectrum of the acoustic sample and the acoustic quality model. Through computer simulation, the control effect of the system before and after adding the filter is compared, that is, the change of annoyance.

TABLE I. COMPARISON OF NOISE REDUCTION OF DIFFERENT ADAPTIVE FILTERS

Explain	FIR	IIR
H_L is ideal transfer function.	10.1	15.1
H_L is the actual transfer function.	4.7	8.1

TABLE II. CALCULATION RESULTS OF ACOUSTIC SAMPLES

Objective Parameter	Loudness	Roughness	Sharpness	Fluctuation Intensity
Calculation value	40.5000	0.6946	1.5230	0.0223

1) Filter Design

In FeLMS algorithm, most of the filters are designed based on A-weighted curve or equal-sound curve. However, they only take into account the characteristics of human ears and do not take into account the spectrum differences of different noises, and the control effects of the noises with large differences are also quite different. Select an acoustic sample in the subjective evaluation experiment, and design the filter according to the frequency spectrum of the acoustic sample and the annoyance model [25].

Based on Zwicker model, the psychoacoustic objective parameters of selected acoustic samples are calculated, and the calculated results are shown in Table II:

The loudness, roughness, sharpness, and fluctuation intensity of sound samples are in each Bark domain. By substituting the calculated values of objective parameters of sound quality in each Bark domain into the sound quality model, the annoyance score of sound samples in each Bark domain can be obtained. Considering that high-frequency noise reduction accounts for a small proportion in the active control system, the control effect of the system decreases with the increase of frequency band. Therefore, select the first 16 Bark domains, that is, the annoyance score with frequency in the range of 0–3500 Hz, and fit the filter in the frequency domain according to the normalized score, and the response amplitude of the synthesized filter in each frequency band is roughly equal to the annoyance score [26, 27].

B. Simulation and Comparative Analysis of Results

The selected noise samples are simulated by active control in MATLAB, using FxLMS and FeLMS algorithms, respectively. Among them, in the FeLMS algorithm, the filter designed by A-weighted curve fitting and the filter designed by sound quality model fitting are set. For the convenience of description, the FeLMS algorithm based on the A-weighted curve is called A-FeLMS and the FeLMS algorithm based on the sound quality model is called Y-FeLMS. The control systems based on the three algorithms are simulated, analyzed, and compared. The noise after control is calculated, and the calculation results of psychoacoustic objective parameters of noise are shown in Table III.

It can be seen from Table III that the noise reduction after being controlled by the control system based on FeLMS algorithm is 23.1 dB and 23.4 dB, respectively, which is not superior to the noise reduction of 26.1 dB of the traditional FLMS algorithm. However, in terms of annoyance, the annoyance of FxLMS algorithm dropped to 0.3606, that of A-FELMS algorithm dropped to 0.2801, and that of Y-FELMS algorithm dropped to 0.1346. It can be seen that Y-FeLMS achieves better sound quality control than the first two algorithms [28, 29].

TABLE III. CALCULATION RESULTS OF NOISE PARAMETERS AFTER CONTROL

Noise	Noise Sample	FxLMS	A-FeLMS	Y-FeLMS
Sound pressure level (dB)	82.1	55.8	28.1	27.6
Noise reduction amount (dB)	/	26.1	23.1	23.4
Distress degree	6.1127	0.3606	0.2801	0.1346

C. Subjective Evaluation and Verification

In order to verify the accuracy of control effect calculation, the sound quality evaluation experiment was organized again according to the objective conditions of the evaluation experiment [30]. Subjects are required to score the noise samples controlled by three algorithms. The subjective evaluation results are shown in Fig. 6.

According to the experimental values in Fig. 6, compared with FxLMS algorithm and A-FeLMS algorithm, the degree of annoyance after control based on Y-FeLMS algorithm is reduced by 0.64 and 1.12 grades. The results show that the optimized FeLMS algorithm is helpful to improve the sound quality.

V. CONCLUSION

Through the above analysis and computer simulation, it is shown that the computer simulation and modeling for ANC introduced in this paper can effectively reduce the computational complexity of the original algorithm and improve the performance of the modeling part of ANC. According to the established sound quality model, the filter is designed and the FeLMS algorithm is optimized. The

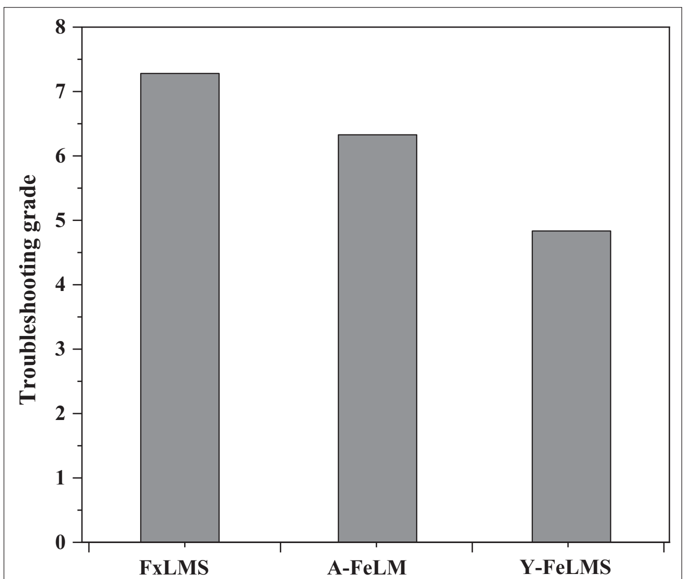

Fig. 6. Noise annoyance score after control.

active control system of Y-FeLMS algorithm based on annoyance degree is established. The simulation results show that, after control, compared with FxLMS algorithm and A-FeLMS algorithm based on A-weighting curve, the annoyance degree drops by 0.64 and 1.12 grades, the control effect is obviously improved, and better noise reduction effect is obtained at the same time.

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