



Article Sound Quality Control Based on CEEMD Blind Source Separation and FELMS Algorithm

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Abstract: The reduction in sound pressure level is the focus of noise reduction in construction machinery, but the sound quality parameters can better describe the operator's subjective perception of noise. This paper proposes a sound quality control method for the cab, which is based on complementary ensemble empirical mode decomposition for signal decomposition and reconstruction and an adaptive control algorithm error filter. Firstly, a subjective and objective prediction model was created to identify the target parameters for the sound quality control in the cab. Secondly, the noise was reconstructed based on a complementary ensemble empirical mode decomposition method, thus evaluating the influence of each component on the sound quality and determining the frequency interval. Lastly, the active sound quality control was completed based on the variable step size filtererror least mean square algorithm. The experiments were performed in the cab of a mini-excavator to verify the method's effectiveness. It was verified that the loudness peak drops by 0.95 sones under stationary idle working conditions. The results demonstrate that the above methods play a guiding role in the actual application of sound quality control for the cab of construction machinery.

Keywords: active sound quality control; CEEMD; FELMS

1. Introduction

Recently, research into noise energy attenuation and obstruction in the cab has come into focus. Chen et al. restrained the sound pressure level (SPL) of internal cab noise by using several measures, including improved structural design, paste damping the design of the acoustic package, and the installation of vibration isolation mass [1–5]. Noise reduction for construction machinery cabs always pays more attention to the reduction of SPL, but research shows that this does not completely reflect human's subjective perception of noise. Occasionally, sounds with a higher SPL are more pleasant than ones with a lower SPL. Based on such a phenomenon, researchers have put forward the concept of sound quality and psychological parameters [6,7]. Jiang et al. summarized seven algorithms of sound quality control and application scenarios [8], and a machine learning algorithm was applied to the sound quality control [9–13]. Zhou et al. simulated the active noise reduction and conducted experiments with the neural network algorithm [14]. However, the machine learning algorithm is difficult to use due to the limitations in hardware conditions. Nowadays, the least mean square (LMS) algorithm is widely used. Kuo et al. installed a wave filter at the secondary path and error feedback path to selectively control the noise at a specific frequency. They proposed the filter-error least mean square (FELMS) algorithm for active sound quality control (ASQC) [15–18]. Som et al. verified the feasibility of loudness control using that algorithm [19]. Meanwhile, Ardekanit et al. illustrated that the convergence time of the FELMS algorithm is closely related to the frequency range [20]. Furthermore, Oliveira et al. evaluated the improvement in loudness and roughness before and after active noise control using the FELMS algorithm [21]. In addition, Lin et al.



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Copyright: © 2022 by the authors. Licensee MDPI, Basel, Switzerland. This article is an open access article distributed under the terms and conditions of the Creative Commons Attribution (CC BY) license (https:// creativecommons.org/licenses/by/ 4.0/). controlled the sound quality using the active control system and compared the speech intelligibility before and after [22]. Liu et al. improved the step size of the FELMS algorithm and increased the convergence speed and noise reduction effect [23]. The research on ASQC is mainly focused on the selection and improvement in the control algorithm, and there is little research on the target frequency band. Kuo and Bao et al. proposed the method of dividing the control band by the critical band, which achieved a good noise reduction effect [24–26]. Zhao et al. established the sound quality prediction model using LV-SVM. They determined the frequency interval by sequentially increasing the critical frequency band and validated the method's effectiveness in simulation [27]. This paper proposes a method of ASQC based on the complementary ensemble empirical mode decomposition (CEEMD) blind source separation algorithm to determine the target frequency. Firstly, a subjective and objective prediction model of cab sound quality was established based on linear regression. The mapping relationship between human subjective perception and noise parameters was formed to determine the target of sound quality control. Secondly, the noise was decomposed using the CEEMD algorithm and reconstructed through component selection, and the influence of each component on the sound quality was evaluated to determine the frequency range of noise reduction. Lastly, the variable step size FELMS algorithm was used to complete the ASQC, taking into account the convergence speed and stability, which can improve the noise reduction effect and reduce the convergence time. Thus, this paper is a significant reference for sound quality control in the cab of construction machinery.

2. Basic Theory

2.1. Variable Step Size FELMS Algorithm

The FELMS algorithm adds a sound quality filter and is based on the Filter-X Least Mean Square (FXLMS) algorithm. It controls the sound quality at a specific frequency range, and its algorithmic principle is as shown in Figure 1.



Figure 1. Variable step size FELMS principle.

Here, x(n) is the reference signal estimation, W(z) is the transverse filter, C(z) is the error filter, S(z) is the acoustic path from the secondary speaker to the error sensor, y(n) indicates the output of the sound quality filter, $\hat{S}(z)$ equals S(z), e(n) is the residual noise, and e'(n) is the pseudo-error signal.

Primary noise d(n) is as follows:

$$D(n) = p(n) * x(n)$$
(1)

where p(n) is the impulse response function of the primary path P(z). The error noise is as follows:

$$e(n) = d(n) - y'(n) * s(n)$$
 (2)

The pseudo-output signal of the filter is as follows:

$$y'(n) = y(n) - y(n) * c(n)$$
 (3)

where s(n) is the impulse response function of the secondary path S(z), c(n) is the impulse response function of the sound quality filter C(z).

The pseudo-error signal is as follows:

$$e'(n) = e(n) - y(n) * c(n) * \hat{s}(n)$$
(4)

Substituting Formulae (1), (3), and (4) into Formula (2), we obtain:

$$e(n) = p(n) * x(n) - w(n) * x(n) * s(n) + w(n) * x(n) * c(n) * s(n)$$
(5)

where w(n) is the weight coefficient of the filter. Change Z in Formula (5) and calculate the weight coefficient of the filter as follows:

$$E(z) = C(z) * D(z)$$
(6)

As is indicated in (6), the design of the error filter controls the error noise.

Calculate Formula (5) with the steepest descent method, and the gradient of mean square deviation as follows:

$$\nabla \hat{\xi}(n) = 2E \begin{bmatrix} e(n) \begin{bmatrix} \frac{\partial e(n)}{\partial w_0(n)} \\ \vdots \\ \frac{\partial e(n)}{\partial w_{L-1}(n)} \end{bmatrix} \end{bmatrix}$$
(7)

Substitute Formula (2) into Formula (6):

$$\nabla \hat{\xi}(n) = -2E[e(n) \times (n)] \tag{8}$$

Therefore, the formula of the weight coefficient vector of the filter is updated as follows:

$$w(n+1) = w(n) - 2\mu e(n) \times (n) \tag{9}$$

where μ is the step size. This paper proposes a new method of updating step size based on the traditional FELMS algorithm, that is, to establish a new nonlinear function relationship among parameter variables α and β and error signal e(n), so as to not adjust the step size too violently when the algorithm enters the steady state. The function expression of the step size is as follows:

$$\mu(n) = \beta(n) \lg(2 - \exp\left(-\alpha(n)|e(n)|^2\right)) \tag{10}$$

where

$$\alpha(n) = \vartheta |e(n)e(n-1)|^m \tag{11}$$

$$\beta(n) = \rho\beta(n-1) + \gamma e^{2}(n) \tag{12}$$

where ϑ , m, ρ , and γ are constant parameters. The variable-step FELMS algorithm not only accelerates the convergence of the algorithm when the error is large but also ensures its steady-state performance when the error is small.

2.2. Signal Decomposition Theory

The CEEMD algorithm is a signal decomposition and reconstruction method that avoids the mode mixing of Empirical Mode Decomposition (EMD) in its application. The basic principle is as follows:

n groups of Gaussian white noise Ga are introduced into the original signal S, and each group has positive and negative pairs. N_1 and N_2 are the post-processed signals:

$$\begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} \begin{bmatrix} S \\ Ga \end{bmatrix} = \begin{bmatrix} N_1 \\ N_2 \end{bmatrix}$$
(13)

The original signal is decomposed into n numbers of Intrinsic Mode Function (IMF), component $c_i(t)$, and residual signal r(t):

$$x(t) = \sum_{j=1}^{k} c_j(t) + r(t)$$
(14)

where x(t) is the original sequence, and the (n + 1) IMF component is calculated as follows:

$$C_{n+1}(t) = \frac{1}{N} \sum_{i=1}^{N} E_1[r_n(t) + \sigma E_n[w_i(t)]]$$
(15)

where σ is the scale factor, and $w_i(t)$ indicates white Gaussian noise.

3. Sound Quality Control Based on CEEMD and Variable Step Size FELMS

On the basis of active control and mode decomposition theory, this paper proposes a method to improve the sound quality in the cab of construction machinery based on the CEEMD blind source separation and the variable step size active control algorithm. The process is shown in Figure 2.



Figure 2. Sound quality control based on CEEMD and variable step size FELMS.

- Step 1: Build a linear-regression-based subjective and objective prediction model of cab sound quality to form the mapping relationship between human subjective perception and objective noise parameters, thereby determining the objective parameters that need to be improved.
- Step 2: Decompose the noise using the CEEMD algorithm and then recombine the noise through the IMF to evaluate the influence of each component on the sound quality, thus determining the frequency range of ASQC.
- Step 3: Use the variable step size FELMS algorithm with balanced convergence speed and stability to complete ASQC within the determined objective parameter target and frequency range.

4. Application and Verification

The active sound quality control method proposed in this paper was applied to a mini-excavator to verify its effectiveness in the cab.

4.1. Prediction Model

4.1.1. Objective Parameters

The method of collecting the objective parameters is shown in Figure 3. A SCADAS data collector and GARS microphone were adopted to collect parameters including: SPL of cab noise, AI speech articulation, fluctuation, loudness, roughness, and sharpness. The mini-excavator was in a stationary, idle working state with its air conditioner set to level 3. The range of rotation speed was 1000–2000 r/min, the sampling frequency was 44,100 Hz, and the sampling time was 30 s. The collected data were imported into the Sound Diagnosis module to calculate each parameter value of sound quality. Figure 4 shows the SPL curves of the right ear in the cab at 1200 r/min, 1500 r/min, and 1900 r/min.



Figure 3. The signal acquisition experiment. (a) The mini-excavator cab. (b) The layout of microphone.



Figure 4. The right ear noise at different speeds.

4.1.2. Jury Testing

The jury testing of noise assesses how pleasant the sound is to a person. The pairwise comparison method requires a smaller sample size, and the evaluator must have no experience in evaluation. Combined with experimental conditions, the pairwise comparison method was selected for jury testing. There were 20 evaluators in total, including graduate students and engineers, 15 males and 5 females aged 24 to 40. First, the evaluators intercepted a 5 s stable signal for equivalent response processing as an audition sample. Where noise samples were considered to be better quality than the audition sample, the evaluator awarded 2 points, the same audition quality was given 1 point, and worse quality samples were given 0 points—the larger the numerical value, the better the subjective evaluation. To ensure that the subjective results were statistically significant, we performed consistency and reliability tests on all subjective evaluation coefficients below 0.7 were excluded. The weighted consistency coefficients and correlation coefficients are shown in Tables 1 and 2. The subjective evaluation results of 11 groups of samples are shown in Table 3. The subjective evaluation value gradually decreases with the increase in rotation speed.

Consistency coefficient	Evaluator Value	1 0.77	2 0.83	3 0.73	4 0.90	5 0.77	6 0.83	7 0.75	8 0.71	9 0.71	10 0.73
	Evaluator Value	11 0.79	12 0.94	13 0.71	14 0.77	15 0.73	16 0.77	17 0.69	18 0.79	19 0.83	20 0.81

Table 1. Consistency coefficient of evaluation data.

Table 2. Correlation coefficient between evaluations.

Correlation coefficient	Evaluator Value	1 0.91	2 0.85	3 0.89	4 0.92	5 0.84	6 0.91	7 0.92	8 0.91	9 0.88	10 0.92
	Evaluator Value	11 0.92	12 0.92	13 0.91	14 0.92	15 0.38	16 0.90	17 0.56	18 0.91	19 0.92	20 0.89

Table 3. Results of subjective evaluation.

Sample	SPL/dB	Articulation Index/%	Fluctuation Strength/Vacil	Loudness/Sone	Roughness/Asper	Sharpness/Acum	Pleasant
1	70.61	46.09	0.82	30.67	0.21	0.98	418
2	70.16	43.74	0.90	30.32	0.20	1.00	415
3	71.07	42.19	0.73	30.42	0.25	1.02	390
4	73.27	39.85	0.85	35.35	0.23	0.93	231
9	75.86	30.28	0.92	38.94	0.28	1.01	130
10	74.73	29.81	0.95	38.30	0.27	1.03	113
11	76.14	28.24	1.00	39.68	0.28	1.03	145

4.1.3. Subjective and Objective Prediction Model

It is a complicated process to evaluate sound quality using a subjective evaluation test. Related research has optimized the process by establishing a mapping relationship between psychological parameters and subjective evaluations and using objective parameters to judge subjective feelings. As there are few test samples in this study, this paper builds the prediction model using linear regression. The linear analysis was performed using SPSS, and the results are shown in Table 4.

Table 4. The relationship between subjective evaluation values and objective parameters.

Objective Parameters	SPL/dB	Articulation Index	Fluctuation Strength	Loudness	Roughness	Sharpness
Related Coefficient R ²	0.903	0.883	0.531	0.931	0.394	0.023

As shown in Table 4, the related coefficient of SPL, loudness, and AI speech articulation is above 0.8. These are the main parameters in psychology and greatly influence the cab sound quality. Stepwise linear regression was used to build the prediction model. After using the F inspection rule, the Sig values of the significance index, including SPL and AI speech articulation, were 0.620 and 0.436, respectively, which are larger than 0.05, so these parameters should be removed as inapparent variables. However, the Sig value of loudness is 0, which has notable influence and should be kept. Thus, the prediction model is calculated as below:

$$Y = -31.27X + 1344.61 \tag{16}$$

where Y is the subjective evaluation of sound quality in the excavator cab; X is the loudness, involving auditory perception features that are more indicative of subjective preferences.

In terms of the goodness-of-fit, the regression model determination coefficient R^2 of the model is 0.931, close to 1, which has a high degree of fitting and good generalization ability. At the same time, the model Sig is 0, so this model is full of statistical significance.

Three groups of samples randomly reserved in the sample database were used to verify the accuracy of the prediction model. The analysis of the prediction results shows that the maximum deviation is no more than 10%, and the average deviation is 6.12% (as shown in Table 5), which has practical engineering significance. In application, loudness is directly used to assess the subjective evaluation value of sound quality in excavator cabs, saving the testing time in subjective evaluation.

Table 5. Prediction accuracy analysis.

Samples	Subjective Scores	Prediction Results	Deviation Value	Deviation Ratio	Average Deviation Ratio
2	415	411.52	3.48	0.84%	
5	185	203.48	18.48	9.99%	6.12%
9	130	139.79	9.79	7.53%	

4.2. Signal Decomposition and Reconstruction Based on CEEMD

The influence of each frequency range on loudness should be judged to determine the frequency range of active noise control. In this paper, the CEEMD algorithm was adopted to decompose and reconstruct the signal. The noise signal was decomposed into 12 groups of sub-signals, which were removed group by group successively. Then, the residual signal was recombined to calculate the loudness of the recombined signal, and the result was compared to the original signals. Finally the loudness contribution ratio was determined.

Taking the right ear noise in the cab of the excavator at 1900 r/min as an example, we decomposed it into an IMF component of 16 groups and a residual component (Res) of one group. As shown in Figure 5, the noise amplitude of stages 4 to 10 is large, while that of the other stages is small. Each component has no obvious overlap. Thus, the accuracy of the CEEMD algorithm for signal decomposition is verified.



Figure 5. Signal decomposition results with CEEMD algorithm.

Next, the above components were summed and reconstructed to obtain the reconstructed signal: $S - IMF_{\tau=1\sim 16}$. The objective parameters of the reconstructed signal were then calculated. The degree of influence of each component on the sound quality is shown in Figure 6.



Figure 6. Loudness contribution of components.

Since the IMF 4/5/6 components have a significant influence on the overall loudness, the frequency range of the sound quality filter was 81–404 Hz, which corresponds to the IMF 4/5/6 components.

4.3. Active Sound Quality Control

4.3.1. ASQC Simulation

To verify the feasibility of the CEEMD algorithm and variable step size FELMS algorithm, the active sound quality control effect was simulated in Matlab. The identification of the transfer function of the secondary path is a prerequisite for realizing the simulation and experiment of active control. As the external environment was relatively stable, this paper calculates the transfer function of the secondary path by inputting white noise for offline identification. The identification results are shown in Figure 7, and the ordinate represents the weight coefficient value of the filter.



Figure 7. Transfer function of secondary path.

This study is based on right ear noise in the cab of the excavator at 1500 r/min and 1900 r/min. The active control process of sound quality was simulated, and the performance of the variable step size of the FELMS algorithm and the FELMS algorithm were compared. The CEEMD algorithm was used to separate and recombine the signal, and the loudness contribution interval was calculated. The FIR filter was used to simulate the unknown system. The order was L = 16, and the step size was μ = 0.01. The simulation results indicate that the iteration speed of the variable step size FELMS algorithm is

2.4 times that of the conventional FELMS algorithm in the stable case. The variable step size FELMS algorithm has a more minor mean square error.

The loudness was calculated by Zwicker's method, and the results are shown in Figure 8. Better control performance can be obtained on the low-frequency loudness under two working conditions with the variable step size FELMS algorithm. In the 2–4 Bark range, the loudness peak reduced on average by 1.75 sones. Meanwhile, the sound pressure decreased by 4.4 dB, which shows the feasibility of sound quality control based on CEEMD and the variable step size FELMS algorithm.





Figure 8. Cont.



(**d**)

Figure 8. The results of ASQC simulation. (**a**) The curve of SPL at 1500 r/min. (**b**) The curve of SPL at 1900 r/min. (**c**) The curve of loudness at 1500 r/min. (**d**) The curve of loudness at 1900 r/min.

4.3.2. ASQC Experiment

The experiment on sound quality control was performed in the excavator cab to verify the effectiveness of sound quality control using the CEEMD and FELMS algorithms. After determining the parameters, such as frequency range and the variable step size algorithm, the hardware setting for ASQC is shown in Figure 9. The collected noise was stored to the TMS320VC5509A core via the MIC port on the development board, AIC23B core, and MCBSP multichannel buffered data port.



Figure 9. ASQC experiment.

When at 1900 r/min, the variable step size FELMS algorithm was used for the active control experiment. The loudness curves under different frequency ranges are shown in Figure 10, and the total loudness and loudness peaks after active control under stationary idle working conditions are shown in Table 6. According to the experiment, in the range of 2–4 Bark, although the loudness increased at high bands, the total loudness decreased by

1.12 sones. Particularly at the noise reduction range determined by CEEMD, the maximum peak decreased by 0.95 sones, and the overall loudness curve was smoother than before. Furthermore, the literature [27] calculated the influence trend of superimposed bands on loudness by sequentially increasing the critical bands. It also determined the number of superimposed bands when the loudness was optimal and then determined the frequency interval. However, this paper uses the CEEMD algorithm to decompose and reconstruct the signal to quickly determine the frequency interval, which realizes online and real-time frequency interval identification.



Figure 10. The results of ASQC experiment.

Table 6. Residual noise loudness after active control at 1900 r/min.

State	Total Loudness/Sone	Max. Loudness/Sone		
Original	38.94	5.18		
ASQC	37.82	4.23		

Errors between the experimental data and the simulation analysis results may be caused by the sound burr and howling noise of electronic hardware during the experiment, such as a loudspeaker, resulting in an amplification effect in some frequency bands during filtering. There was also some interference between the reference microphone and the error microphone. In addition, the time-varying of the reference signal is greater, and the correlation among the error signals will become weaker with the influence of the environment. Although some certain deviation exists between the control result of the experiment and the simulation result, certain control effects have been obtained in the low-frequency range, which indicates that the proposed variable step size algorithm is suitable for the active control of sound quality in the excavator cab.

5. Conclusions

This paper proposes a method for sound quality control in cabs based on the CEEMD algorithm and the variable step size FELMS algorithm, which establishes the mapping relationship between sound quality parameters and subjective pleasantness to determine which objective parameters should be optimized. The signal was decomposed and reconstructed to identify the frequency range which greatly contributes to the objective parameters. The step size of the active control algorithm was optimized to improve convergence and stability. Thus, the sound quality is improved using the above three aspects.

The method was applied to an excavator. The experiment verified that within the 2–4 Bark range determined by CEEMD, the maximum loudness peak decreased by 0.95 sones, and the convergence rate improved 2.4 times. This illustrates that the method not only improves sound quality but also increases the convergence rate of the algorithm, providing a novel method for ASQC in the cab of construction machinery.

The experimental results demonstrate that the proposed method is suitable for multiband noise sources whose spectral characteristics remain basically stable. Where the spectral characteristics of the noise change slowly, regular calculations will be performed using CEEMD to update the error filter weight coefficients at regular intervals. However, this method still has the problem of insufficient real-time performance for time-varying and non-stationary systems. In future research, the online identification of secondary channels and reduction in the time delay in hardware circuits will be considered.

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