An Audio Integrated Noise Control System using Variable Step Size MFxLMS Algorithm with Online Secondary Path Modeling

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ABSTRACT

Active noise control (ANC) system has been a popular solution in last few decades for low frequency noise issue. The FxLMS algorithm developed in 1985 is the cornerstone for most ANC system due to its secondary path estimation which solves the problem of time delay in the system. Various approaches have also been proposed to resolve and improve the performance of the system. In this paper, several techniques of active Noise control with online secondary path modeling have been reviewed and tested. A detailed study was conducted which focused on reducing residual noise in a psychoacoustic approach and finally leads to design of an audio integrated ANC system.

1. INTRODUCTION

1.1 Background

Noise has been more and more common and intolerable with the advance of our modern society, and it may cause unpleasantness and disturbance for human being both physically and mentally. Recently, an uprising trend of evidence has shown that the acoustic noise is the key factor to a series number of human disease and disorder, such as insomnia, stress, hearing loss and high blood pressure. The passive noise control is a conventional noise control technique and has been proven effective in reducing high frequency noise. This technique basically utilizes sound absorbing or blocking materials to mitigate the sound level of noise. However, due to the limitation and high expenses of passive noise control and the dramatic increase of presence of low frequency noise generating from all kinds of electronic device and machine in the working environment, passive noise control is no longer the optimal solution. The active noise control technique, which by generally generating an anti-noise signal which acquires the same magnitude and opposite phase to the original noise, is a more affordable and efficient approach to mitigate the noise in low frequency range.

1.2 Review of basic ANC system

The design of ANC system which uses a loudspeaker and a microphone to generate the canceling noise was proposed in 1936. Since the cancelling noise has to be adjusted in real time according to the variation of noise in the environment, an adaptive filter that can adjust its own coefficient to minimize the error signal is the ideal option to deal with the variation in the environment. The most frequently used combination in a ANC system is finite response filter (FIR) and Least-mean-square algorithm. The object of the adaptive filter in the ANC system is to minimize the error signal $e(n)$. Ideally, the perfect cancellation can be achieved when $e(\boldsymbol{n}) = d(n) - y(n) = 0$, where $d(n)$ represents the primary noise and $y(n)$ represents the canceling noise. The formula for the LMS algorithm is expressed as

$$w(n+1) = w(n) + ue(n)x(n) \qquad (1)$$

The expression of weight for adaptive filter can be expressed as $w(n) = [w0(n), w1(n), \ldots, wL-1(n)]$ .The weight of the adaptive filter $w(n)$ is updated during each iteration by the step size u, input signal $x(n)$ recorded by reference microphone and the instantaneous error $e(n)$ given by equation 2.

$$e(n) = d(n) - y(n) \qquad (2)$$

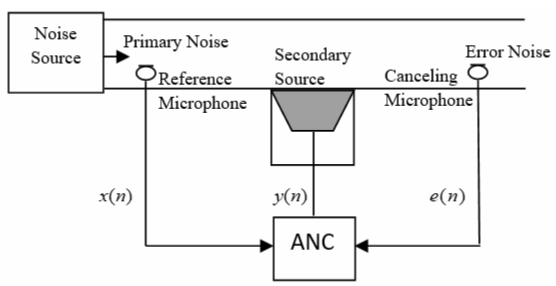

Fig. 1. General diagram of ANC system

1.3 FxLMS Algorithm

The canceling noise of the ANC system is generated and goes through secondary path $S(z)$ which consists of amplifier, DAC, reconstruction filter and the loudspeaker, etc. The limitation of LMS algorithm is the time delay between the secondary path $S(z)$ and the primary path $P(z)$, which could greatly affect the performance of the ANC system and cause instability. The solution to this issue is called Filtered-x Least Mean Square (FxLMS). A $\hat{S}(z)$ acting as an estimation of the secondary path is placed to filter the reference signal $x(n)$. Comparing to LMS algorithm, FxLMS algorithm achieves a better convergence rate and reduces the error caused by time delay between two paths. A plot of FxLMS algorithm based ANC system is shown in figure 2. The weight of adaptive filter is updated by using the filtered signal $x'(n)$ instead of $x(n)$, where the equation of $x'(n)$ is given as

$$x'(n) = x(n) * \hat{s}(n) \quad (3)$$

$$w(n+1) = w(n) + u * e(n) * x'(n) \quad (4)$$

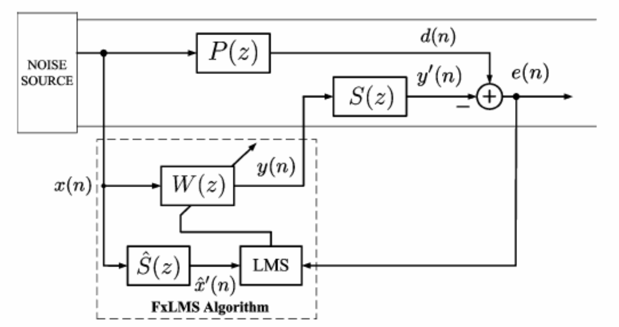

Fig. 2. Block diagram of feedforward ANC system. $P(z)$ is the primary path, $S(z)$ is the secondary path, $\hat{S}(z)$ is the estimation of secondary path and $W(z)$ is the adaptive filter..

The estimation of the secondary path $\hat{S}(z)$ can be completed offline by using white noise as an excitation signal. However, for real-time applications, the secondary path will not always be stationary, some of the applications even have significant time-varying secondary paths. Therefore, updating secondary path is required and critical for some ANC systems. Online secondary path modeling method is the desired approach to address the issue.

2.ONLINE SECONDARY PATH MODELING

During the past 20 years, various genius approaches to the online secondary path modeling issue and improvement have been proposed. In general, there are two distinct approaches to the online secondary path modeling. The first approach models the secondary part by injecting an additional random noise into the ANC system. The second method estimates the path by modeling it from the output of ANC controller, thus avoiding using the additional noise. Although the second method do not have to inject the noise into the ANC system, comparison in [14] has proven that the first method is superior to the second method in regard to computational complexity, convergence rate and response to the changes of noise in primary path. Therefore, only the system of the first approach will be introduced and tested in this paper.

2.1 Eriksson's method

Eriksson's method [4] is the first method that introduces online secondary path modeling to the ANC system and has been the cornerstone for this area. Two adaptive filters are used in the system. The first is the same control filter $W(z)$ used in FxLMS algorithm. The second is the modeling filter $\hat{S}(z)$ for the estimation of secondary path. Block diagram of ANC system based on Eriksson's method is shown in figure 3.

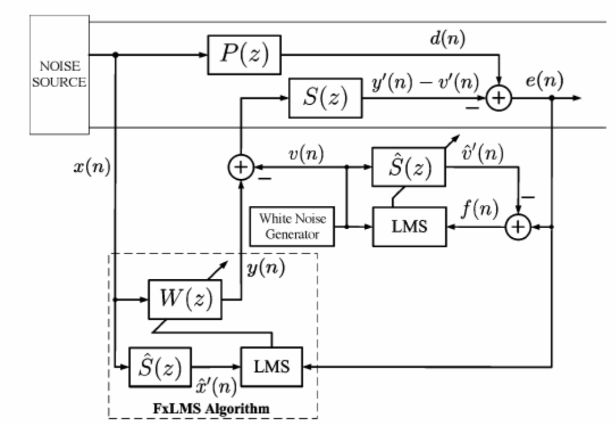

Fig. 3. Block diagram of Eriksson's method for feedforward ANC system.

The over structure resembles the basic FxLMS algorithm except the injection of white noise v(n) and the additional adaptive filter. The internally generated zero-mean white noise v(n) is first injected into the secondary path along with the secondary signal y(n) which can be expressed as

$$\mathrm{y(n)} = w^T(n) * x_L(n) \quad (5)$$

Where $w^T(n) = [w_0(n)w_1(n) \dots \dots w_{L-1}(n)]^T$ and $x_L(n) = [x(n)x(n-1) \dots \dots x(n-L+1)]^T$. L is the weight length of the FIR filter. The error signal for this structure is given as

$$e(n) = d(n) - y'(n) + v'(n) \quad (6)$$

where $d(n) = p(n) * x(n)$ represents the signal going through the acoustic path $P(z)$ and getting picked up at the reference microphone. $y'(n) = y(n) * s(n)$ and $v'(n) = v(n) * s(n)$ represent cancelling signal and modeling signal both going through the secondary path. $*$ denotes the operation of convolution and $s(n), v(n)$ represent the impulse response of each path. The obtained error signal $e(n)$ is used to update the weight of the $W(z)$:

$$\begin{aligned} w(n+1) &= w(n) + u_c e(n) x'(n) \\ &= w(n) + u_c[d(n) - y'(n) + v'(n)]x'(n) \quad (7) \end{aligned}$$

The signal $\hat{v}'(n)$, filtered by the estimated secondary path $\hat{S}(z)$, is subtracted from the $e(n)$ and the result of this operation $f(n)$ is expressed as

$$f(n) = d(n) - y'(n) + v'(n) - \hat{v}'(n) \quad (8)$$

And $f(n)$ is used to update the filter coefficient of $\hat{S}(z)$:

$$\hat{s}(n+1) = s(n) + u_m f(n) v(n)$$
$$= \hat{s}(n) + u_m[d(n) - y'(n) + v'(n) - \hat{v}'(n)]v(n) \quad (9)$$

u_c and u_m represent the step size for two adaptive filter, respectively.

The equation can be further expressed as

$$\hat{s}(n+1) = \hat{s}(n) + u_m[v'(n) - \hat{v}'(n)]v(n) + u_m[d(n) - y'(n)]v(n) \quad (10)$$

In the ideal situation, $v'(n) - \hat{v}'(n)$ will become zero and the performance of the system will be degraded by component $u_m[d(n) - y'(n)]v(n)$, which will cause slow convergence and even divergence of the system. Furthermore, the fixed step size for two different adaptive filters can also be a drawback to the system, since it is difficult to determine the appropriate step sizes for both adaptive filters and the system will not be able to adjust itself according to the different input signal.

2.2 Zhang's method

The disadvantages of Eriksson's method are solved by adding a third adaptive filter to the ANC system. Among those approaches, Zhang's method acquires the best performance. A block diagram of ANC system based on Zhang's method [5] is shown in figure 4

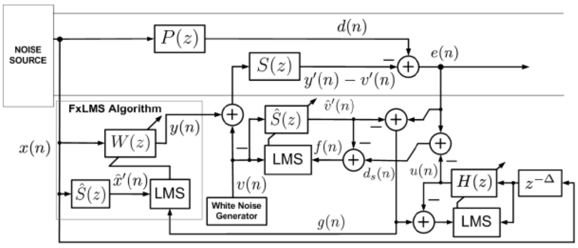

Fig. 3. Block diagram of Zhang's method for feedforward ANC system.

Zhang's method applied a third adaptive filter $H(z)$ to the system, which is a cross-updated filter to decrease the interference among the other filters. The error signal $g(n)$ used for the update of filter $W(z)$ in this method is given as

$$g(n) = d(n) - y'(n) + v'(n) - \hat{v}'(n) \quad (11)$$

While the error signal $f(n)$ used for the update of filter $\hat{S}(z)$ is given as

$$f(n) = d(n) - y'(n) + v'(n) - \hat{v}'(n) - u(n) \quad (12)$$

Where $u(n)$ is the output of the third adaptive filter $H(z)$. In the ideal condition where the $\hat{S}(z)$ converges, $v'(n) - \hat{v}'(n) = 0$, and $g(n) = d(n) - y'(n)$.

Then if the third filter converges, $u(n) = g(n) = d(n) - y'(n), d_s(n) = v'(n)$. Since $f(n) = d_s(n) - \hat{v}'(n), \hat{v}'(n) = v(n), f(n)$ will eventaully beocme zero. and therefore the improvement of system's performance is expected. The step size for each filter is determined from experiments.

2.3 Akhtar's and modified Akhtar's method

Although Zhang's method was considered as the best online secondary modeling method, the usage of three adaptive filters extensively increases the computation complexity of the system and makes it harder to define optima step size for each filter.

The Akhtar's method [2] was proposed in 2006 and achieved a better overall performance than zhang's methods by only using two adaptive filters. A block diagram of ANC system based on Akhatr's method is shown in figure 5

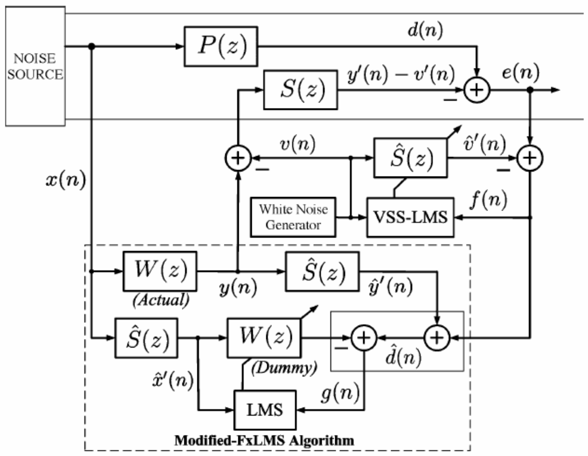

Fig. 5. Block diagram of Akhtar's method for feedforward ANC system based on MFxLMS.

Even with only two adaptive filters, the computation complexity of Akhtar's method is still higher than zhang's method. Therefore, author proposed a modification to his own method and it has been named as modified Akhtar's method [11]. The plot of modified method is shown in figure 6.

In section 2.2 Eriksson' method, the component $u_m[d(n) - y'(n)]v(n)$ that can degrade the performance of the system is mentioned. It is critical for us to realize that the cancelling noise $y'(n)$ is zero in the initial stage of the modeling process, and disturbance due to this component will decrease as $y'(n)$ gradually converge to $d(n)$. Therefore, in order to improve the convergence rate and maintain a stable system, we should apply a small step size $u_s$at the beginning of the modeling process and switch it to a large step size $u_L$in the later stage of the modeling process. Thus, the VSSLMS is proposed for the ANC system with the online modeling secondary path.

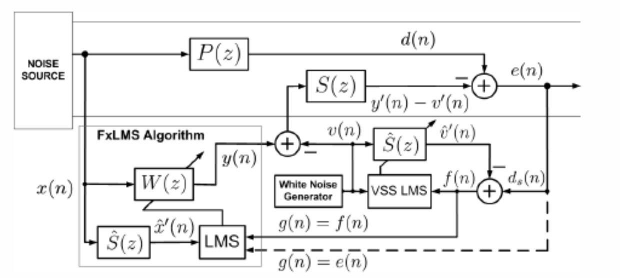

Fig. 6. Block diagram of Akhtar's modified method for feedforward ANC system based on MFxLMS.

First, we may define the power of each error signal as followed:

$$P_e(n) = \lambda P_e(n-1) + (1-\lambda)e^2(n) \qquad (13)$$

$$P_f(n) = \lambda P_f(n-1) + (1-\lambda)f^2(n) \qquad (14)$$

Where λ is the forgetting factor ($0.9 < \lambda < 1$). The variable step size can be obtained as

$$u(n) = p(n)u_{min} + \big(1 - p(n)\big)u_{max} \qquad (15)$$

where ratio $p(n)$ can be calculated as $p(n) = \frac{P_f(n)}{P_e(n)}$ and u_{min} and u_{max} are minimum and maximum step size boundaries set for the system. At the beginning of the modeling stage, $P_f(n) = P_e(n)$, and thus $u(n)$ starts with $p(n)$=1 and therefore minimum step size is achieved. Maximum step size is achieved as $p(n) = 0$ in the final stage. The error $e(n)$ used for the update if filter W(z) is also changed to the error signal $f(n)$ to reduce the interference of input audio signal to the modeling process of filter W(z).

Ahmed [6] argued that performance of the system may be degraded by using the variable step size proposed above when the step size reaches the maximum value and proposed a modified method

$$u(n) = \big(1 - p(n)\big)u(n) \qquad (16)$$

$$u(n) = \begin{cases} u_{min}; & \beta(n) < u_{min} \\ \beta \quad ; & u_{min} \leq \ \beta(n) \leq u_{max} \\ u_{max}; & \beta(n) < u_{min} \end{cases}$$

where β is determined by

$$\beta \ \ (n+1) = \alpha\beta(n) + (1-\alpha)f^2(n) \qquad (17)$$

Where α is the control parameter like λ. The step size for this method will reach its maximum value when the system converges and then reduces to its minimum value to keep the system stable.

The two methods for variable step size have their own advantages and disadvantages. In the paper, we have tested two methods and decided to use the method proposed by Akhtar, since Ahmed's method has a higher computation complexity and a slightly slower convergence rate than Akhtar's method.

3.AUDIO-INTEGRATED ANC SYSTEM

3.1 Basic theory

The ANC system is able to effectively reduce noise at low frequency. However, even after the convergence of the system, the residual noise can still be presence in the system, since the error signal can only become 0 in ideal condition. And in the environment with low frequency noise, residual noise could be extremely annoying. Therefore, it is necessary for us to mask the residual noise.

Since the error signal in Akhtar's modified method is given as

$$e(n) = d(n) - y'(n) + v'(n) \qquad (18)$$

By replacing the white noise $v(n)$ with the audio signal $a(n)$, error signal picked up at the error microphone will contain both audio signal and residual noise. The performance of the system will also not be degraded by the audio signal if the audio signal is uncorrelated with the cancelling signal.

3.2 Modification of audio signal

There has been barely any research regarding the choice of the audio signal for this system. Generally speaking, the choice of audio signal should be pleasant enough for human to hear, such as nature sound. And the selection of the audio signal is largely depending on the characteristic of residual noise and can possibly be guided by the psychoacoustic principles. Since the basic approach is to mask the residual noise using the audio signal.

Several audio signals have been tested during the experiment, they are white noise, pink noise, the sound of running water, the sound of wind, and the sound of raining. Based on the characteristic of tested audio signal, the sound of running water is used as the audio signal for this paper.

However, the amplitude of the input signal could also strongly affect the performance of the system. Therefore, method [9] was proposed to adjust the input signal so that the system is able to achieve a faster convergence rate and become more stable. The audio signal $v(n)$ can be modified by the equation of

$$v(n) = \sqrt{(1-\rho(n))\sigma_{min}^2 + \rho(n))\sigma_{max}^2} \cdot v(n) \qquad (19)$$

Where $v(n)$ is controlled by the power ration $\rho(n)$ and maximum and minimum values of the variance of $v(n)$.

4.SIMLUATION

4.1 Data and Parameters

Like most of the ANC system, offline modeling of estimated secondary path $\hat{S}(z)$ is required, since if the operation of the ANC system begins with the null vector $\hat{S}(z)$**,** the performance of the system will be degraded and in worst case it will not converge. The offline modeling signal used for the system is the same audio signal that used as input audio signal for online modeling,

For this part, the zero-mean uniform white noise is used as the reference noise, which is the noise we would like to cancel. The system performance emulation will be based on the error signal and the modeling error, which is defined by the ratio of difference between secondary path and its estimation.

$$\Delta S(\text{dB}) = 10\log 10\left(\frac{[s(n) - \hat{s}(n)]^2}{s(n)^2}\right) \qquad (20)$$

The parameter used for each ANC system is shown below:

Eriksson's method: $u_w = 5 \times 10^{-3}, u_s = 1 \times 10^{-3}$

Zhang's method: $u_w = 5 \times 10^{-3}, u_s = 1 \times 10^{-2}$,

$$u_v = 1 \times 10^{-2}$$

Akhtar's method: $u_w = 5 \times 10^{-3}$, $u_{min} = 7.5 \times 10^{-3}$

$$u_{max} = 2.5 \times 10^{-2}$$

$$\sigma_{min}^2 = 0.001, \sigma_{max}^2 = 1.0$$

4.2 Results

The computer simulations of residual error for the three approaches mentioned above are shown below:

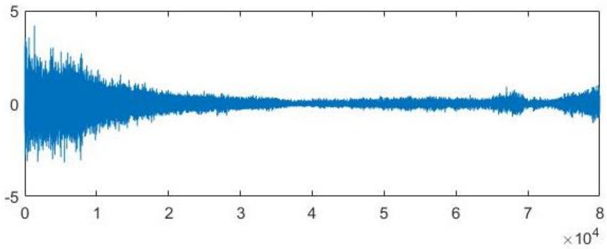

Fig. 7. Residual error of Eriksson's method

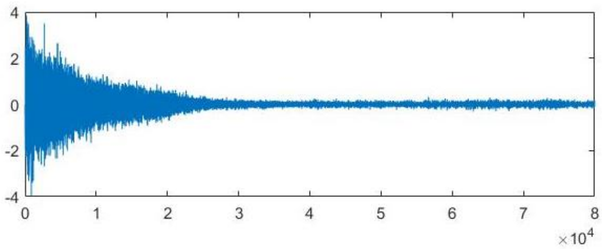

Fig. **8**. Residual error of zhang's method

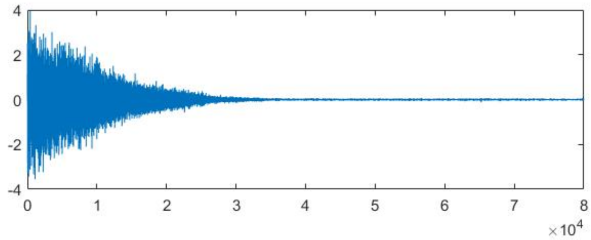

Fig. 9**.** Residual error of modified Akhtar's method using variable step size.

From the plots of residual error, we are clearly able to tell the difference between these three approaches. Eriksson' Method, with the input signal being an audio signal, converged in a relatively slow rate, and the system started to become unstable after 4×10^4 of iterations. This demonstrates that as the first ANC structure with the online secondary path modeling, Eriksson's method comes with some obvious disadvantages and spaces for further improvements.

The zhang's method, by using three adaptive, achieved a much better convergence rate and also obtained a more stable system than the system in Eriksson's method. However, due to the complexity of the system, the computation complexity is higher than that of modified Akhtar' method. Step sizes for three adaptive filters in zhang's method are fixed, which also make the system difficult to adjust itself to the input signal.

The modified Akhtar' method is able to achieve the most stable system and has the best overall performance due to the use of variable step size. The rate of convergence was comparable to the rate in zhang's method.

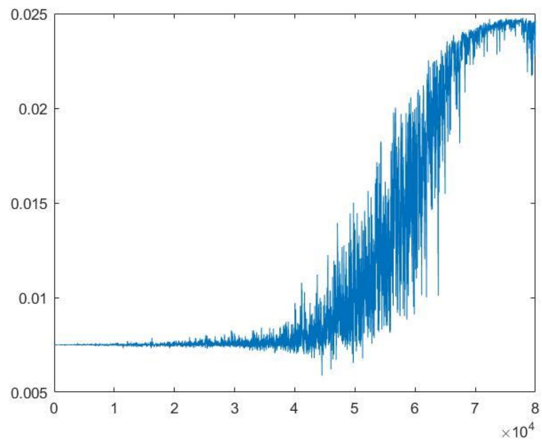

Fig. 10**.** Variable step size in modified Akhtar' method

The plot of variable step size used in modified Akhtar' method is shown in the figure 10. The step size is first initiated with the minimum step size defined previously, as the system becomes more and more stable, the step size begins to approach the maximum value to speed up the rate of convergence.

5. AN AUDIO INTEGRATED ANC HEADPHONE

The function of audio integrated ANC system is not limited to the masking of residual noise. The audio signal injected into the system can also act as a main audio signal, such as music, which results in an audio integrated ANC headphone. [4]

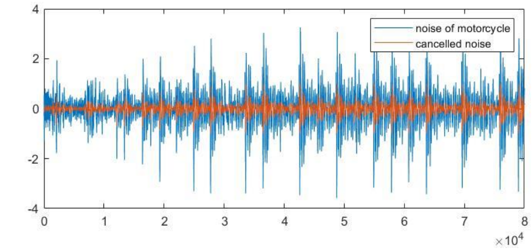

Fig. 8. The simulation of ideal cancellation in MATLAB

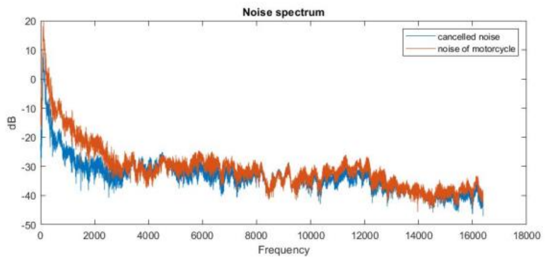

Fig. 9. Noise Spectrum of ideal cancellation in MATLAB

The simulation was conducted within the MATLAB. The sound of motorcycle was selected as the representative of the low frequency noise and daily noise. The input audio signal was selected randomly, since it would not have any effect on the performance of the system and is the desired audio signal. The figure 9 shows a clear 5 to 10 Db noise reduction below 1000Hz, which is mainly the frequency range for motorcycle and other noises in low frequency

6. CONCLUSION

In the paper, we did a thorough review on the well-known existing methods for online secondary path modeling of ANC system. Three approaches were compared, analyzed and tested. The modified Akhtar' method achieved the best overall performance in convergence rate and the stability of the system. The approach was used to design an audio integrated ANC system which is able to mask the residual noise generated by the normal ANC system. The method was also used to design a simulation of audio integrated ANC headphone system which can cancel the environmental noise and other low frequency noises, such as motorcycle's noise, while playing desired music signal. The system has been proved effective based on the simulation result.

7. FUTURE WORK

The simulation process and data collection process were not strict enough. The plots of residual noises of all approaches were printed based on one attempt each. During the testing process, the result varied from time to time, and Eriksson's method failed to converge 10% of the time. Therefore, we will conduct a more thorough and detailed data collection for the simulation part. The convergence rate of the system by using the audio signal as the modeling signal and input signal is still not ideal. More research will be conducted on the improvement of the convergence rate and the stability of the overall ANC system. The selection of audio signal in this paper is based on my personal preference. The cancelling effect using different input audio signal will also vary according to different people's personal preference, therefore, more research and experiment should be done on the selection of audio signal in a more psychoacoustic way.

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