Psychoacoustically Motivated Active Noise Control at Remote Locations

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Abstract— **Active noise control (ANC) is an effective way to cancel the low-frequency noise. The conventional ANC system creates the 'zone of quiet' by minimizing the mean square error (MSE) at the location of an error microphone. However, in practical applications, sometimes it is not possible to achieve the noise attenuation at the desired location due to physical constraints limiting locating the error microphone at certain points. Similarly, the performance of the conventional ANC system also compromises when the impression of audio sensation on human auditory does not match the numerical values of the system. It is because the human ear has complicated psychoacoustic properties. In this paper, we present a new psychoacoustically motivated ANC system for a remote location. Noise weighting filters are incorporated into remote ANC to improve the audio sensation of the residual noise. The performance of the purposed system is evaluated by computer simulation, and the perceptual loudness is selected as a performance criterion for the psychoacoustic assessment of residual noise.**

I. INTRODUCTION

Active noise control (ANC) is an efficient technique to reduce the low-frequency noises, where the traditional passive methods are bulky, costly, and ineffective. ANC is based on the principle of superposition of two sound fields and aims to cancel the acoustic noise by generating an 'anti-noise' that has an equal amplitude but opposite phase. Conventional ANC algorithms create a 'zone of quiet' by minimizing the acoustic pressure at the location of an error sensor. However, for practical applications, it can be unrealistic and ineffective by not achieving the noise attenuation at the desired location. Therefore, to introduce a more flexible positioning of the 'zone of quiet', several virtual sensing and remote sensing algorithms have been suggested in the literature [1][2]-[3].

Moreover, the ultimate objective of ANC is to minimize the noise disturbance on human auditory. The conventional ANC techniques aim to minimize the noise disturbance by using mean square error (MSE) criterion, which deals with the whole frequency range identically. While, research indicates that human hearing has selective sensitivity to different frequencies and it has a nonlinear frequency response, e.g. 60dB SPL (sound pressure level) at 1kHz is perceptually 20dBA louder than the 60dB SPL at 150Hz [4]. Psychoacoustic is another essential factor to be taken into account in the design of an ANC system. Psychoacoustic combines the study of acoustics and human auditory sensation and shows that the residual error of the ANC system has an uneven auditory sensation. Therefore, the objective of minimum psychoacoustic irritation brings up the need to improve the conventional ANC. Recently, several psychoacoustical motivated noise cancelling and masking algorithms have been published [5][6]-[7]. The main motivation of this paper is to study the psychoacoustic characteristics of our initially proposed remote ANC system [8]. The psychoacoustic model is incorporated with remote ANC system to improve the noise attenuation in perceptual perspective. In this case, the resulting 'zone of quiet' space is free from error microphone to utilise it effectively, and the energy of residual noise is minimized, which perceptually gives a softer and quieter result.

This paper is organised as follows. Section II describes the remote ANC algorithm. Section III introduce the proposed psychoacoustic model of remote ANC. Simulation results and discussion are in section IV. Finally, section V gives the concluding remarks.

II. REMOTE ANC SYSTEM

The traditional single channel ANC system comprises a reference microphone, error microphone, and a secondary source. The reference microphone picks the reference signal and feeds it to the adaptive controller, that estimates a control signal and generate the secondary noise. An adaptive algorithm is used to update the adaptive controller in accordance with the information provided by the reference microphone and an error microphone. The Filtered-x Least Mean Square (FxLMS) is considered as the fundamental algorithm in adaptive ANC system [9]. The FxLMS estimates the secondary noise called 'anti-noise' adaptively by utilizing the power of an error signal as cost function. Then the superposition of the primary and secondary noise minimize the acoustic pressure at the location of an error microphone and creates a 'zone of quiet' as a by-product. However, the resulting 'zone of quiet' in single channel ANC system is generally very small and is occupied by the error microphone itself. This problem can be addressed by virtual and remote sensing. Virtual and remote sensing techniques have been investigated in adaptive ANC systems, and several adaptive virtual sensing algorithms have been proposed to remove the error microphone from 'zone of quiet' [10][11].

In this paper, adaptive remote FxLMS have been used to create a silent point 'zone of quiet' at the nearby remote location, while the error microphone remains at the actual location. Remote FxLMS is based on traditional FxLMS algorithm and utilizes typical single-channel architecture. Moreover, it does not require the preliminary identification from physical to a remote point, that makes it proficient for

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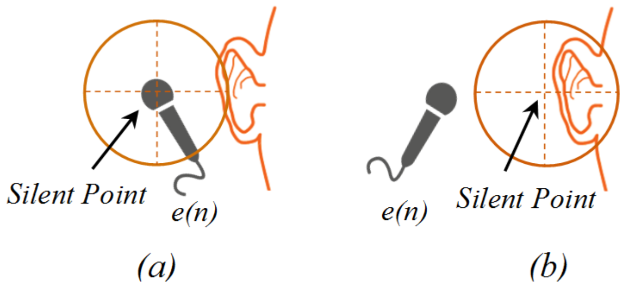

Fig. 1. Locations of 'zone of quiet' and an error microphone (a) Conventional ANC systems, (b) Virtual and remote ANC system

practical applications. Fig. 1. illustrate the difference between silent points in conventional ANC and remote ANC. As shown in Fig. 1. (a), the silent point in a conventional ANC system is at the location of an error microphone, and 'zone of quiet' is produced around this silent point as a by-product. While in Fig. 1. (b), the silent point in remote ANC system is shifted from the location of an error microphone to the nearby remote location. In this case, the 'zone of quiet' is independent of the error microphone location and can be utilized effectively.

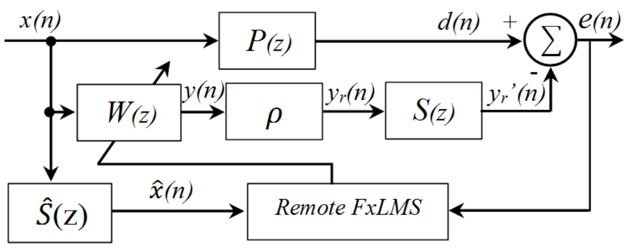

Fig. 2. Functional block diagram of single channel adaptive remote FxLMS

The functional block diagram of FxLMS based adaptive remote ANC system is shown in Fig. 2. It has two physical systems P and S called the primary and secondary path, respectively. Whereas $W(z)$ is the adaptive filter, and $\rho(z)$ is the remote controller. The remote controller transfer function is given by:

$$\rho(z) \; = \; K_r \; z^{-\Delta_r} \tag{1}$$

Where K_r is the static gain, and Δ_r is the time-delay of a remote controller. The complete derivation of K_r and Δ_r is presented in [8]. The standard FxLMS algorithm is modified to compensate the influence of a remote controller $\rho(z)$. The reference signal of the adaptive remote algorithm is the same, but the error signal in a remote ANC system is different from the error signal in the original FxLMS based system. The error signal of the standard FxLMS based ANC system in the z-domain can be expressed as:

$$E^*(z) = P(z)X(z) + W(Z)S(z)X(z) \tag{2}$$

Similarly, the error signal in remote ANC system can be expressed as:

$$E(z) = P(z)X(z) + W(Z)\rho(z)S(z)X(z) \tag{3}$$

so, the difference between error signals can be obtained by subtracting $Eq.$ 3 from $Eq.$ 2.

$$\Delta E(z) = E^*(z) - E(z) \tag{4}$$

$$\Delta E(z) = W(Z)S(z)X(z) - W(Z)\rho(z)S(z)X(z) \tag{5}$$

As shown in Fig. 2., $Eq.$ 5 can be simplified as

$$\Delta E(z) = S(z)Y(z) - Y_r(z) \tag{6}$$

so in time domain $\Delta E(z)$ can be expressed as

$$\Delta e(n) = s(n) * \{y(n) - y_r(n)\} \tag{7}$$

$\Delta e(n)$ is the difference between the remote ANC error signal and the desired error signal. So $\Delta e(n)$ should be added to $e(n)$ to update the error signal that can create a 'zone of quiet' at a nearby remote location. Then the updated remote FxLMS algorithm can be expressed as:

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\{e(n) + \Delta e(n)\}\hat{\mathbf{x}}(n) \tag{8}$$

Where μ is the adaption step-size and $\mathbf{w}$ is the weight vector of length L to represent the impulse response coefficient W.

$$\mathbf{w}(n) = \begin{bmatrix} w_0(n) \; w_1(n) & \dots & w_{L-1}(n) \end{bmatrix}^T \tag{9}$$

and

$$\hat{x}(n) = \hat{s}(n) \; * \; x(n) \tag{10}$$

$\hat{x}(n)$ is the filtered reference signal of the input $x(n)$. Where $\hat{s}(n)$ is the estimated impulse response of the secondary path S, and $\mathbf{x}(n)$ is the input signal vector defined as:

$$\mathbf{x}(n) = \begin{bmatrix} x_0(n) \; x_{n-1}(n) & \dots & x_{n-L+1}(n) \end{bmatrix}^T \tag{11}$$

Consequently, the remote FxLMS algorithm updates the adaptive weight vector W by $Eq.$ 8 and computes a control signal $y(n)$ as:

$$y(n) = \mathbf{w}^T(n)\mathbf{x}(n) \tag{12}$$

The secondary source generates the control signal $y(n)$ in the acoustic domain. Then, the superposition of primary and secondary sound fields creates a 'zone of quiet' at the nearby remote location rather than the location of an error microphone.

III. Psychoacoustic Remote ANC

Considering the importance of psychoacoustic, human perception of sound has been taken into account to design the remote ANC system. Noise weighting has been suggested as a means of quantifying the hearing sensitivity with respect to frequency. In order to minimize psychoacoustic irritation caused by the residual error, psychoacoustic parameters should be used as performance evaluation. Several noise weighting standards (A, B, C, and D) have been used according to the nature of noise [12][13]. Among all four standards, A-weighing filters are most commonly used. A-weighting noise filters approximate the property of the human ear based on the equal-loudness contour, and it comes from the listening experiment of pure tone. The frequency response of the A-weighting filter is shown in Fig. 3.

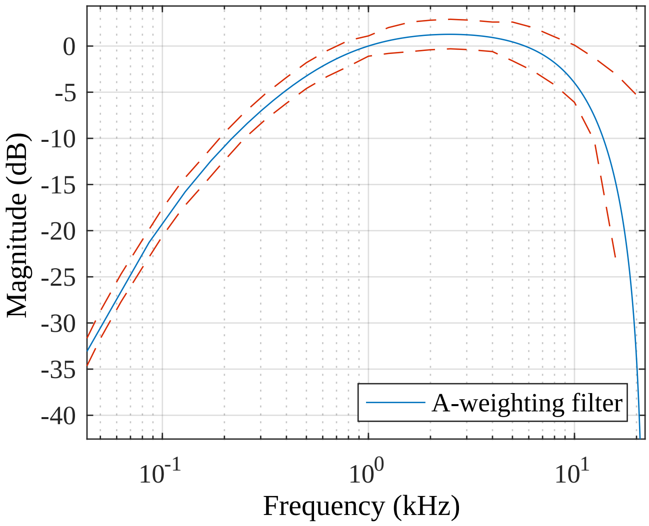

Fig. 3. Frequency response of A-weighting filter

Considering human auditory characteristic, A-weighting filters are incorporated with remote ANC system to improve the noise reduction in terms of acoustics impression. The functional block diagram of psychoacoustic motivated remote ANC is shown in Fig. 4.

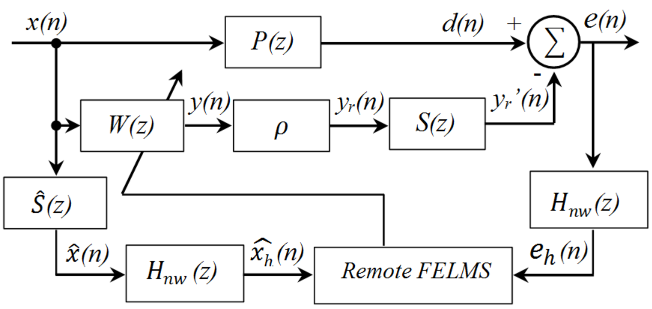

Fig. 4. Functional block diagram of noise weighting based psychoacoustically motivated remote ANC

In order to compensate the non-uniform property of the frequency domain, the remote FxLMS algorithm has been modified by utilizing filtered-E least mean square (FELMS) structure. FELMS is the method to shape the residual noise spectrum proposed by Kuo and Tsai [14]. The original reference signal and error signal are filtered through noise weighting filters to update the noise control filter $W(z)$. The updated equation of remote ANC in consideration of the hearing characteristic can be expressed as:

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\{e_h(n) + \Delta e_h(n)\}\hat{\mathbf{x}}_h(n) \quad (13)$$

Where $\mathbf{w}$ is the coefficient vector of noise control filter, μ is the adaption step size. $\hat{x}_h(n)$ is the convolution of filtered reference signal $\hat{x}(n)$ and $h_{nw}(n)$. The $h_{nw}(n)$ is the impulse response of the noise weighting filter H_{nw}.

$$\hat{x}_h(n) = \hat{x}(n) \, * \, h_{nw}(n) \quad (14)$$

$$\hat{x}(n) = \hat{s}(n) \, * \, x(n) \quad (15)$$

Where $\hat{s}(n)$ is the estimated impulse response of the secondary path S, and $x(n)$ is the input signal.

$$e_h(n) = e(n) \, * \, h_{nw}(n) \quad (16)$$

$h_{nw}(n)$ is considered as the model of the human ear and $e_h(n)$ is the perceived error signal obtained by the convolution of the error signal $e(n)$ and $h_{nw}(n)$. Similarly, $\Delta e(n)$ can be obtained to compensate the difference of remote error signal.

IV. SIMULATION RESULTS AND DISCUSSION

In this paper, we have evaluated the psychoacoustic characteristics of remote ANC. Loudness is one of the major psychoacoustic parameters of perceptual analysis. Therefore, loudness is considered as the performance criterion to evaluate the performance of the proposed system.

Loudness is a subjective quality, and a subjective test is a direct approach to measure the psychoacoustic characteristics. But the constraints such as inconsistent evaluation, time and cost of testing and training of listeners, restrict its usage. Therefore the objective model is used to estimate the subjective evaluation. Loudness, the perceptual intensity of sound depends on, sound pressure level, frequency, and amplitude envelope. Psychologically, loudness can be ordered on a scale from quiet to loud. It can be calculated as:

$$L = \int_{0}^{24Bark} N^{'} dz \quad (17)$$

Where, L is the overall loudness, and $N^{'}$ is the loudness in a specific critical band. Each critical band represents one bark and respective central bark frequency can be calculated as:

$$Bark = 13 \arctan\left(0.00076 f\right) + 3.5 \arctan\left(\left(\frac{f}{7500}\right)^2\right) \quad (18)$$

$$Critical\,band = rate\left(Bark\right) = \left[\frac{(26.81 f)}{1960 + f}\right] - 0.53 \quad (19)$$

$$Critical\ bandwidth(Hz) = \frac{52548}{(z^2 - 52.56z + 690.39)} \quad (20)$$

According to the definition, the sound pressure level (SPL) 40dB at 1kHz is equal to 1 sone. Sone is the unit of loudness. Based on the experiments, the combination of sound pressure levels and frequencies which perceived as equally loud by the listener are standardised in the ISO $226-2003$ and called Equal loudness contour [4]. The ISO $226-2003$ based equal-loudness contour with the targeted central bark frequencies is shown in Fig. 5. MATLAB simulation experiments were carried out to evaluate the proposed ANC model. As shown in Fig. 4, the ANC model has two unknown modules $P(z)$ and $S(z)$. Adaptive FIR filters were used to model the impulse response of these systems. The simulation parameters were chosen as follow. The input signals are pure tones with sampling frequency $44100Hz$. The tap length of $P(z)$, $S(z)$ and $W(z)$ is 500. The updating algorithm is LMS, and the step size μ is empirically chosen as 0.05. To measure the loudness, the excitation level of each bark

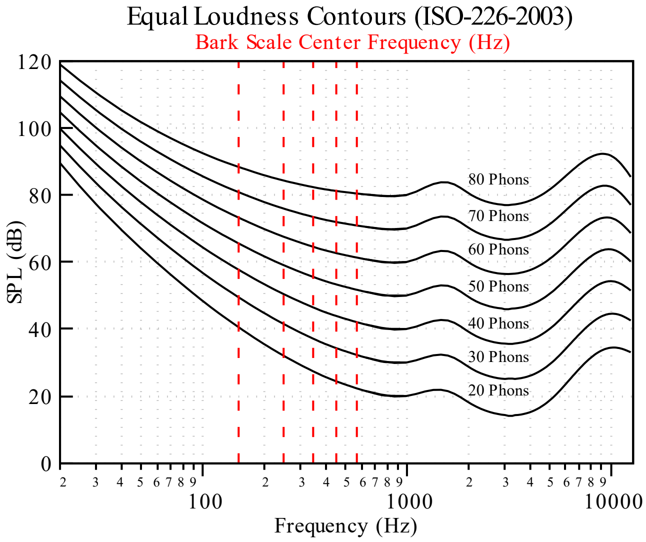

Fig. 5. ISO 226-2003 Equal loudness contours, and targeted center bark frequencies

is calculated from residual error signal and considered the auditory threshold $20\mu Pa$ as a reference sound pressure. For pure tones, loudness is measured in unit phon. Phon is a logarithmic scale 'dB', and one sone is equal to 40 Phon. The conversion of sone to phon, above one sone is calculated as:

$$Loudness\ N\ (sone) = N = 2^{\frac{LN-40}{10}} \tag{21}$$

$$Loudness\ level\ LN\ (phon) = LN = 40 + 10\log_2 N \tag{22}$$

The data is recorded to measure the sound pressure level SPL and loudness of the proposed system at the set of center bark frequencies (150Hz, 250Hz, 350Hz, 450Hz, & 570Hz). The comparison of remote ANC and noise weighting remote ANC is presented in TABLE I.

TABLE I

Comparison of residual error of remote ANC and psychoacoustic motivated remote ANC, in terms of Sound pressure level SPL and Loudness

No.	Tone Hz	Sound Pressure Level dB-SPL			Loudness A-weighted Phon		
		ANC OFF	Remote ANC	Proposed Re-ANC	ANC OFF	Remote ANC	Proposed Re-ANC
1	150	90.96	71.83	71.53	79.58	58.36	55.15
2	250	90.96	70.66	69.34	84.29	62.19	57.60
3	350	90.96	69.78	68.42	86.35	64.66	58.42
4	450	90.96	69.09	67.10	88.02	66.96	59.45
5	570	90.96	68.81	65.77	89.51	68.21	59.81

The result shows that in noise weighting incorporated remote ANC, the sound pressure level SPL at lower frequencies is almost the same as remote ANC, but it marginally reduces at higher frequencies of the targeted frequency band. On the other hand, the loudness is reduced significantly from 3dBA -9dBA across all targeted frequencies.

V. CONCLUSIONS

The traditional ANC system minimises the acoustic pressure at the location of an error microphone and adopts the mean square error as a cost function. However, in some cases, traditional ANC is ineffective to achieve the noise attenuation at the desired location. Besides, those models do not consider the attributes of the human auditory system.

In this paper, noise-weighting filters are incorporated with a remote ANC modelling to improve the effectiveness of the system and implicating the perceptual perspective. The proposed model shifts the silent point from the location of the error microphone to a remote location and shapes the residual noise according to the A-weighting, which imitates the human ear response. Therefore, the resultant 'zone of quiet' is independent of the error microphone location and perceptually quieter. The computer simulations show the effectiveness of the proposed model on a set of bark frequencies. The perceptual measurement, loudness, is selected as a performance criterion, and the results show that the noise reduction has improved considerably.

As a next step, the proposed model has to be tested further on other psychoacoustic parameters such as pleasantness, sharpness and roughness. Moreover, further work will be carried out to extend the model scope to multi-tonal and bandlimited noises.

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