PSYCHOACOUSTIC ACTIVE NOISE CONTROL BASED ON DELAYLESS SUBBAND ADAPTIVE FILTERING

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ABSTRACT

In this paper we present a new psychoacoustic active noise control system based on delayless subband adaptive filtering structure. Human hearing has selective sensitivity for different frequency bands. Motivated by this psychoacoustic fact, noise weighting is incorporated into active noise control system. In this new system, the noise weighting is decomposed into subband forms for computation efficiency and performance improvement. Loudness, one psychoacoustic measure, is used to evaluate the system performance. Simulation on multi-tone signal shows improvement in noise cancellation and reduction in computation complexity.

Index Terms— Psychoacoustic active noise control, Loudness, Delayless subband adaptive filtering, Noise weighting

1. INTRODUCTION

Active noise control (ANC) technology [1] aims to cancel out acoustic noise by generating an “anti-noise” with equal amplitude and opposite phase. They are preferred over passive methods due to their better capabilities for reducing the narrowband and wideband acoustic signal noises.

Many single-channel and multi-channel adaptive ANC algorithms have been developed for attenuating noise at a single point or a certain zone. Note that the ultimate goal of ANC is to minimize the annoying influence of noise on human hearing system. We should make it clear what is important is how human being perceives under the noisy environment. However, most of the current ANC algorithms ignore the non-uniform property of human hearing when designing the system. In conventional ANC system, the adaptive filtering algorithms aim to reduce the noise by minimizing the variance of the error between the primary noise and the anti-noise signals, i.e. using the mean square error (MSE) criterion, which treats different frequency components identically. Research on psychoacoustics [2] indicates that human hearing sensation has selective sensitivity to different frequencies. For example, we will not feel equally loud for two audio signals with the same amplitude but different frequencies: 3 kHz and 20 kHz.

The psychoacoustic analysis brings up the need to improve the conventional ANC system by taking into account the special human hearing characteristics as much as possible. Kuo [3] proposed a new ANC structure which features residual noise shaping. In [4] we initially introduced the idea of incorporating realistic noise weighting, e.g. A-weighting which reflects the frequency response of human hearing, based on Kuo’s Filtered-Error LMS (FELMS) structure.

To further improve the system efficiency, subband adaptive filtering (SAF) [5] is considered for adoption. Subband adaptive filtering based on multirate filtering technique [6] has been in widespread use for its computation complexity reduction. The downside of conventional subband adaptive filter is delay which is introduced by bandpass filters necessary for subband decomposition. Morgan and Thi [7] developed novel delayless subband adaptive filtering structure to solve the above problem. In this structure adaptive weights are updated in subbands and then transformed into fullband filter collectively. Signal path delay is avoided in this way while retaining the benefits of SAF structure.

In this paper, an efficient active noise control system based on delayless subband adaptive filtering structure is presented with incorporation of psychoacoustic factor. The computational complexity is greatly reduced by decomposing the noise weighting filter into subband forms. Loudness, instead of sound pressure level (SPL), is selected as performance criterion to evaluate the system from psychoacoustic perspective.

This paper is organized as follows. Section 2 describes the psychoacoustic active noise control with incorporation of noise weighting. The proposed system with SAF structure is introduced in Section 3. Simulation results shown in Section 4 demonstrate the effectiveness of the new system.

2. PSYCHOACOUSTIC ACTIVE NOISE CONTROL

Fig. 1 shows the diagram of Filtered-X LMS (FXLMS) [1] which is a commonly used structure for conventional ANC system. $P(z)$ represents transfer function of the primary path (the acoustic cavity) with a reference sensor measuring the input signal (the source noise signal) and an error sensor measuring the output signal or the attenuated signal at the point where noise cancellation occurs. Secondary path $S(z)$ represents system response for all the electrical and acoustical signal transmission paths from the cancellation speaker to the canceling zone.

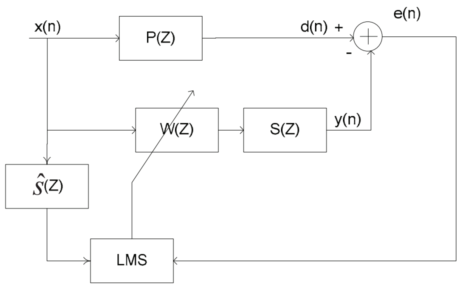

Fig. 1 Conventional ANC system with FXLMS structure

Considering human hearing characteristics, the conventional FXLMS structure has to be modified to compensate for the non-uniform property in frequency domain. One method is to shape the spectrum of the residual noise. FELMS structure is utilized to achieve this goal. In order to incorporate psychoacoustic factor, noise weighting is adopted in the modified ANC system as show in Fig. 2. The original residual noise is filter by noise weighting filter. To make the system stable, another copy of this filter is added sequentially following the estimated version of secondary path transfer function.

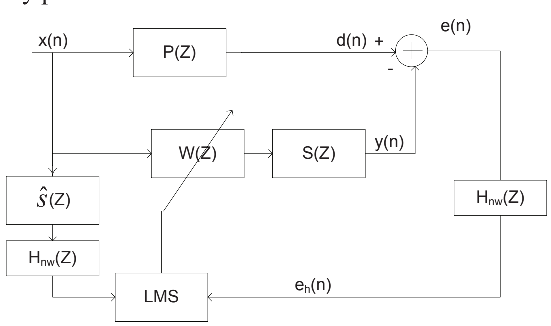

Fig. 2 ANC system with noise weighting

The coefficient adapting algorithm associated with the system in Fig. 2 can be expressed as

$$\mathbf{w}(n+1) = \mathbf{w}(n) + 2\mu e_h(n)[x(n) * \hat{s}(n) * h_{nw}(n)] \quad (1)$$

$$e_h(n) = e(n) * h_{nw}(n) \quad (2)$$

where, $\mathbf{w}(n)$ is the coefficient vector of the adaptive filter, $e_h(n)$ is the output of the noise weighting filter, $\hat{s}(n)$ is the estimation of the secondary path impulse response, $h_{nw}(n)$ is the impulse response of the noise weighting filter. The selected noise weighting should reflect human auditory system.

3. DELAYLESS SUBBAND ADAPTIVE FILTERING FOR PSYCHOACOUSTIC ANC

Incorporation of noise weighting increases the computation burden for the system. In order to model the human hearing more precisely, long taps of FIR filter is required which makes the additional computation cost significant. To solve this problem, the noise weighting is decomposed into a set of subband filters as shown in Fig. 3. Instead of generating the filtered residual noise and reference signal through the entire band of interest, they are achieved in each subband.

The polyphase FFT technique [8] can be used to decompose the signals into subband forms. This technique realizes M continuous single-sideband bandpass filters which produce M complex subband signals by down-sampling the output with decimation rate $D = M/2$.

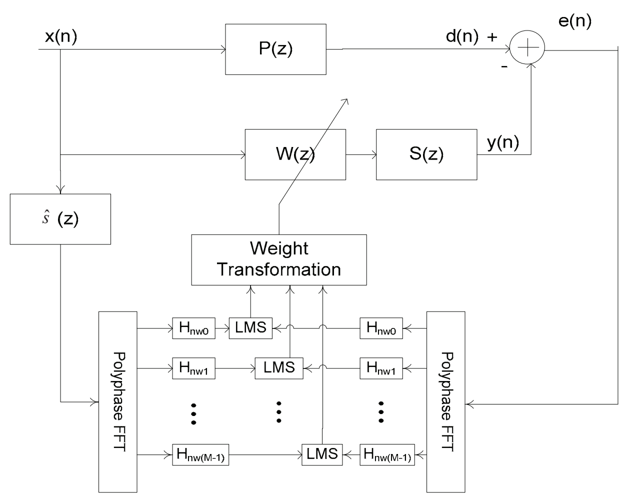

Fig. 3 Delayless subband ANC system with subband-decomposed noise weighting

Since the signals and filter coefficients in subbands are complex number, the complex NLMS algorithm is employed as follows:

$$\mathbf{w}_i(k+1) = \mathbf{w}_i(k) + \frac{\mu \mathbf{u}_i^*(k)}{\left\|\mathbf{u}_i(k)\right\|^2} e_i(k) \quad (3)$$

where k is the iteration index, $\mathbf{w}_i(k)$ is the filter coefficient vector in i^{th} subband, $\mathbf{u}_i(k)$ and $e_i(k)$ are reference signal and error signal filtered by noise weighting in i^{th} subband.

The adaptive filter coefficients of the full band filter $W(z)$ derive from the coefficients in each subband by weight transformation. We adopt the weight stacking method with procedures as follows:

1. For $l \in [0, N/2-1]$, $W(l) = W_k(m)$, where $W(l)$ and $W_k(m)$ are the FFT coefficients of the full band filter and the *k*th subband filter, respectively. $m = \lfloor lM/N \rfloor$, where $\lfloor \bullet \rfloor$ denotes rounding to the nearest integer; and $m = (l)_{2N/M}$, where $(x)_y$ represents x modulus y.
2. For $l = N/2$, $W(l) = 0$
3. For $l \in [N/2+1, N-1]$, $W(l) = W(N-l)^*$, where the superscript $*$ denotes complex conjugation.

In the above procedures, N is the filter length in the fullband, M is the subband number.

As for the noise weighting, we choose A-weighting [9] which matches human hearing response approximately. A-weighting is calculated by (4) and (5):

$$R_a(f) = \frac{12200^2 \cdot f^4}{(f^2+20.6^2)(f^2+12200^2)(f^2+107.7^2)^{0.5}(f^2+737.9^2)^{0.5}} \quad (4)$$

$$A = 2.0 + 20\log(R_a(f)) \quad (5)$$

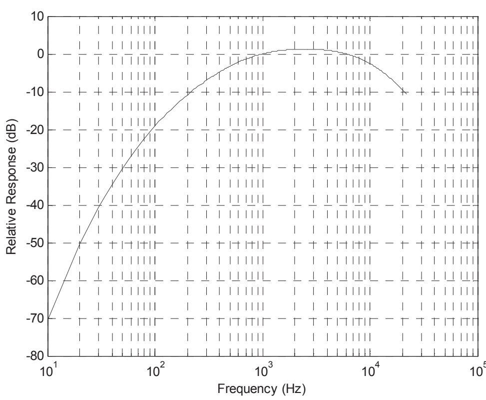

Fig. 4 Frequency response of A-weighting filter

Loudness is selected as performance criterion which measures the intensity sensation of human being. It can be calculated quantitatively with the following function:

$$L = \int_0^{24\,Bark} N' dz \quad (6)$$

where L is the loudness, N' is the "specific loudness", i.e. the loudness in a specific critical band, *Bark* is the unit which defines the scale of 24 critical bands of hearing.

4. SIMULATION RESULTS

In our simulation, we choose primary transfer function $P(z)$ and secondary transfer function $S(z)$ with frequency responses shown in Fig. 5 and Fig. 6, which come from the measurement in our test-bed as shown in Fig. 7. It consists of a half cylinder made of transparent acrylic mounted on a plywood base. The dimensions of the structure are length 1.5 m (5 feet) and diameter 0.76 m (2.5 feet). The lengths of primary and secondary path impulse response are truncated to 300.

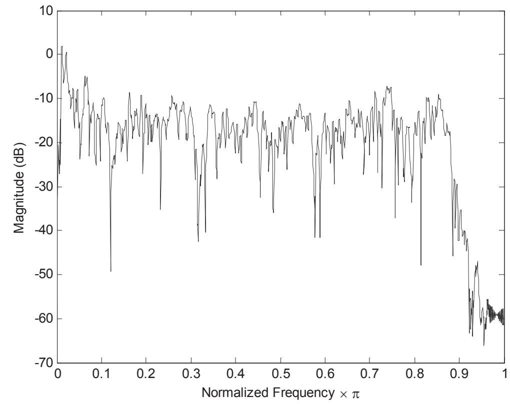

Fig. 5 Frequency response of the primary path used in the simulation

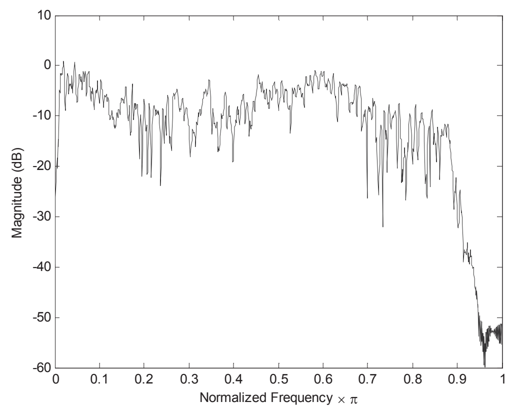

Fig. 6 Frequency response of the secondary path used in the simulation

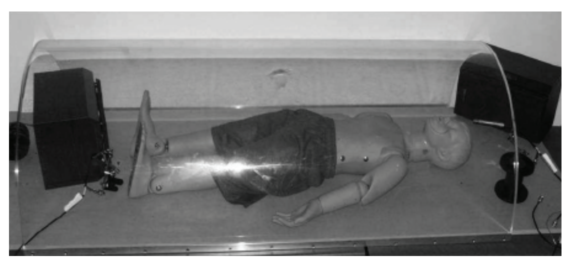

Fig.7 Test-bed for measurement of *P*(z), *S*(z)

Our simulation is conducted on multi-tone signal which consists of these frequency components: 0.2, 0.3, 1, 2, 3, 10, 20 kHz.

The order of full-band FIR filter is chosen as 512. The number of sub-bands is 16. The A-weighting filter in fullband has length of 128 which is decomposed into 16 taps in each subband. The prototype filter has 128 taps.

In Table 1, we compare the residual noise of conventional ANC ($e_0(n)$), psychoacoustic ANC with fullband ($e_1(n)$)

and subband ($e_2(n)$) structures in terms of SPL and Loudness. The data is recorded for the beginning period and after system convergence.

Table 1. Comparison of conventional ANC, psychoacoustic ANC with fullband and subband structures

	SPL(dB) Beginning	SPL (dB) End	Loudness(sone) Beginning	Loudness(sone) End
$e_0(n)$	91.5171	73.7746	34.0300	1.9480
$e_1(n)$	91.5171	74.0865	34.0300	1.2710
$e_2(n)$	91.5171	77.3458	34.0300	1.0510

Results show that incorporation of noise weighting and subband structure increases the SPL value of residual noise. But the loudness is reduced significantly. We can notice that the residual noise in the subband psychoacoustic ANC system is reduced by 46% in term of loudness compared to conventional ANC system.

The computational complexity is analyzed and compared for the fullband and subband version of psychoacoustic ANC. The operations can be separated into following categories:

1. Subband filtering;
2. Adaptive weight update;
3. Weight transformation from subband to fullband;
4. Convolution with noise weighting;
5. Convolution with estimated secondary path impulse response.

For convenience, only the number of real multiplications per input sample is compared.

Let us assume: N is the order of fullband, M is the number of subbands, K is the length of prototype filter, L_{nw} is the length of noise weighting filter in fullband, L_s is the length of estimated secondary path impulse response.

Table 2 Computational complexity in terms of the number of real multiplications per input sample: (A) Psychoacoustic ANC in fullband structure shown in Fig. 2; (B) Psychoacoustic ANC in subband structure shown in Fig. 3

Operation	(A)	(B)
Subband filtering	0	$\frac{4K}{M}+4\log_2 M$
Weight update	N	$\frac{8N}{M}+\frac{16N}{M^2}$
Weight transformation	0	$2\log_2\frac{2N}{M}+\frac{4}{M}\log_2\frac{2N}{M}+\log_2 N$
Convolution with noise weighting for both input signal and residual noise	$2L_{nw}$	$\frac{16L_{nw}}{M}+\frac{32L_{nw}}{M^2}$
Convolution with estimated secondary path impulse response	L_s	L_s

By replacing the parameters with conditions in our simulation, we get 1068 and 802.5 real multiplications per sample for fullband and subband structure, respectively. We can see the computation complexity is reduced by approximately 25%.

5. CONCLUSION

Conventional ANC system ignores the non-uniform characteristics in frequency response of human hearing system. In this paper, a new active noise control system motivated by psychoacoustic factor has been described. By incorporating noise weighting, residual noise is shaped according to A-weighing, which reflects human ear response.

In order to compensate for the additional computation cost brought by noise weighting, delayless subband adaptive filtering structure is utilized by virtue of its high computation efficiency. Corresponding coefficient update and weight transformation are implemented in the new system.

Simulation on multi-tone signal indicates significant performance improvement in terms of loudness measure. Computational complexity is analyzed in terms of real multiplications per sample and about 25% computation reduction has been achieved in our simulation case.

6. REFERENCES

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