A Subband Active Noise Control System with Automatic Tap Assignment in Consideration of Psychoacoustic Properties

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***Abstract*—This paper proposes a high-performance subband active noise control (ANC) system in consideration of psychoacoustic properties. In conventional subband ANC using a tap assignment algorithm, excessive tap lengths are assigned to certain subbands in the initial stage of noise control update, resulting in poor reduction performance. Another problem is that the audible impression of the sound actually attenuated by ANC becomes smaller than the numerical noise reduction. This is due to the complex interplay of psychoacoustic characteristics. In this study, we propose a new algorithm that takes into account the error signal power in the conventional algorithm to prevent excessive tap assignment and improve the attenuation performance of subband ANC. Furthermore, we construct a subband ANC considering psychoacoustic properties and verify its noise reduction performance in human hearing.**

I. Introduction

Active noise control (ANC) is an effective method to suppress the noise level at the noise control point by simulated sound [1]-[5]. In conventional ANC systems, the order of the required adaptive filters increases as the unknown system becomes higher order. The use of higher order adaptive filters increases the amount of computation required for filtering and updating the coefficients. Therefore, a subband ANC system was introduced to replace the high-order adaptive filters with several low-order adaptive filters that operate at a low rate by dividing the signal into several bands and operating an adaptive filter for each band [6].

Some advantages of the subband ANC system are listed below [7]-[8]. First, the decimation process increases the sampling interval and reduces the number of taps in each band, thereby improving the efficiency of computation. This effect is extremely pronounced when the amount of computation is proportional to the number of taps. This allows the hardware to operate in real time. Secondly, since the analysis can be performed over time and frequency, the processing can be changed for each band to reflect the different characteristics of the signal and system at each frequency. This allows us to further improve the efficiency and performance of the processing. Third, colored signals, which do not have a flat spectral shape when viewed over the entire frequency range, are divided into band signals with a near flat spectrum and processed. In other words, the input signal is whitened. This improves the convergence speed of adaptive algorithms for correlated signals such as speech.

In the case of non-stationary, colored speech signals, ordinary fullband ANC systems are not effective enough to suppress the noise. Therefore, subband ANC systems are considered to be more effective than fullband ANC systems for speech signals [9]. As an application of the subband ANC system, a subband ANC system with a masking function, which is one of the auditory characteristics, has also been studied [9]-[10]. At the beginning of the research, equal tap lengths were assigned to each subband adaptive filter of the subband ANC, but the appropriate tap lengths are different for each subband. Therefore, we considered introducing the automatic tap assignment algorithm proposed in [11] to the subband ANC system. However, the automatic tap assignment algorithm proposed in [11] has a problem that excessive taps are assigned to a particular subband. Therefore, in this paper, we propose a new appropriate automatic tap assignment algorithm to prevent excessive tap assignment and introduce it to subband ANC.

Normally, an ANC system works to minimize the mean square error of the error signal acquired at the error microphone point. However, even if noise is suppressed at the error microphone point, humans may not fully benefit from noise suppression due to their psychoacoustic characteristics. Therefore, Bao proposed an ANC system that takes psychoacoustic characteristics into account, and demonstrated its effectiveness using random signals synthesized with MR noise, which is stationary noise [12]. In [12], an ordinary ANC system with A-weighting and ITU-R 468 defined in IEC 61672:2003 [13] was proposed. The error signal and the reference signal are weighted by these noise weighting filters and used to update the noise control filters. Moreover, the ANC system with psychoacoustic characteristics has also been demonstrated to be effective against non-stationary acoustic noise [14]. Therefore, by introducing psycho-acoustic characteristics into the subband ANC system proposed in this paper, the effectiveness of the tap assignment considering human auditory characteristics is verified via computer simulation.

This paper is organized as follows. In Chapter 2, the subband ANC system and the automatic tap assignment algorithm are described. Chapter 3 describes the psychoacoustic characteristics. Chapter 4 compares and discusses the results of noise reduction simulations using the proposed automatic tap assignment algorithm and conventional algorithms in subband ANC. Furthermore, we verify the effectiveness of the proposed algorithm in subband ANC with psychoacoustic filters. Finally, conclusions are drawn.

II. Subband ANC System Using Appropriate Inter-Subband Tap Distribution

Subband signal processing, also called band division processing, is a typical example of multi-rate signal processing because it involves the conversion of sampling frequency. Subband signal processing divides a signal into several frequency bands and processes each band independently, so that the processing can be changed for each band to reflect differences in the characteristics of the signal and system in each band. The signals in each band have a flatter frequency response than the fullband signals. In other words, the signal is whitened, which has the advantage of speeding up the convergence of the adaptive filter.

A. Subband ANC System

Fig. 1 shows a block diagram of a subband ANC system, where $P(z)$ is the primary path, $S(z)$ is the secondary path, and $\hat{S}(z)$ is the model of the $S(z)$. In a subband ANC system, the reference signal and error signal are divided into multiple bands by a band-division filter bank, and the adaptive filter is updated for each band. When the filter bank consists of N filters, the update equation for the mth subband adaptive filter is

$$\mathbf{w}_m(n+1) = \mathbf{w}_m(n) + \mu \mathbf{r}_m(n) e_m(n), (m = 0, 1, \cdots N), \tag{1}$$

where $\mathbf{w}_m(n)$, $\mathbf{r}_m(n)$, $e_m(n)$ and μ are the coefficient vector, filtered reference signal vector, subband error signal, and step size parameter of the noise control filter $W_m(z)$ at time n, respectively, and

$$\begin{aligned} \mathbf{w}_m(n) &= [w_{m,0}(n)\ w_{m,1}(n)\ \cdots\ w_{m,M-1}(n)]^T, \\ \mathbf{r}_m(n) &= [r_m(n)\ r_m(n-1)\ \cdots\ r_m(n-M+1)]^T, \end{aligned} \tag{2}$$

where M is the number of taps in $W_m(z)$.

The signals processed by the adaptive filter in each subband are Fourier transformed by FFT, and the results are combined to form a fullband filter. The noise control filter $W(z)$ is then created by inverse Fourier transforming this filter. In this way, a subband adaptive filter with reduction effects in each band can be created.

B. Band Division Using Hadamard Transform

The Hadamard transform is a generalization of the Fourier transform. The Hadamard transform can be viewed as being constructed from a discrete Fourier transform of size 2, which is equivalent to a multidimensional discrete Fourier transform of size 2 to the power of 2. By using the Hadamard transform for the filtered reference signal and the error signal in the filter bank, the respective signals can be band divided. In this paper, the Hadamard transform is used because it is computationally efficient and does not require the design of a band division filter. The Hadamard transform is also used as a filter in the filter bank section of the subband ANC system shown in Fig. 1.

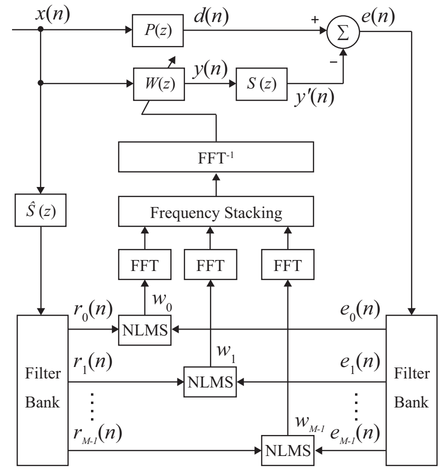


Fig. 1. Block diagram of a subband ANC.

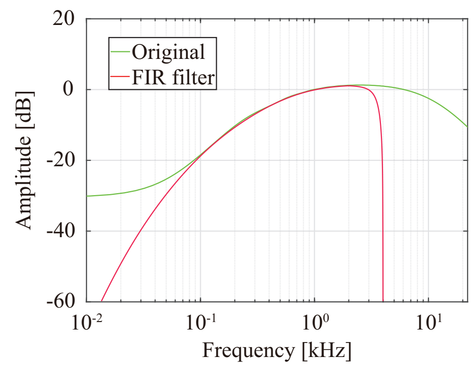


Fig. 2. Frequency responses of A-weighting filter and the corresponding FIR designed filter.

C. Appropriate Tap Assignment

In the conventional automatic tap assignment algorithm for subband ANC, excessive taps are assigned to the subband whose adaptive filter converges faster. As a result, the noise reduction performance of the ANC system was degraded due to excessive errors in that subband and reduced tap assignments to other subbands. To solve this problem, we propose a new algorithm which is an improvement over the conventional automatic tap assignment algorithm [11]. In the proposed algorithm, the energy of the error signal in each subband, which is the second term of γ, is added to prevent extreme tap assignment that occurs in the early stage of updating an ANC system. In the case of the M division of the proposed algorithm, the number of taps in the i-th subband is updated by

$$N_{i,mL} = N_{i,(m-1)L} - R + \mathrm{INT}\left[RM\left\{\gamma\frac{\sum_{p=(m-1)L+1}^{mL} v_{i,p}\,\bar{\mathbf{c}}_{i,p}^T\bar{\mathbf{c}}_{i,p}}{\sum_{p=(m-1)L+1}^{mL}\mathbf{v}_p^T\tilde{\mathbf{c}}_p} + (1-\gamma)\frac{\sum_{p=(m-1)L+1}^{mL}\bar{E}_{i,p}^2}{\sum_{p=(m-1)L+1}^{mL}\tilde{E}_p^2}\right\}\right], \quad (3)$$

where,

$$\tilde{\mathbf{c}}_k = [\bar{\mathbf{c}}_{1,k}^T\bar{\mathbf{c}}_{1,k}\ \ \bar{\mathbf{c}}_{2,k}^T\bar{\mathbf{c}}_{2,k}\cdots\ \bar{\mathbf{c}}_{M,k}^T\bar{\mathbf{c}}_{M,k}]^T, \quad (4)$$

$$\bar{\mathbf{c}}_{i,k} = [c_{i,1}\ c_{i,2}\ \cdots\ c_{i,k}]^T, \quad (5)$$

$$\mathbf{v}_k = [v_{1,k}\ \ v_{2,k}\cdots\ v_{M,k}]^T, \quad (6)$$

$$v_{i,k} = \mathbf{x}_{i,k}^T\mathbf{x}_{i,k}. \quad (7)$$

$N_{i,mL}$ is the number of taps in the i-th subband at the m-th tap redistribution, where L is a predetermined nonnegative integer. This means that a new number of taps for the i-th subband is calculated at every L coefficient adaptations. R is the tap redistribution step size where is the unit number of taps collected from each subbing at a single tap redistribution, INT$[\cdot]$ is an operator to take the nearest integer for its argument and p is the number of times the coefficient is updated. In the i-th subband at the k-th iteration, the filter coefficient power are given by (4) and (5) and the input signal power are given by (6) and (7). In addition, $\tilde{\mathbf{c}}_k$, $\mathbf{v}_k$ and $\tilde{E}_k^2$ denote the filter coefficient power, input signal power and error signal power for the entire subband, and $\bar{\mathbf{c}}_{i,k}$, $v_{i,k}$ and $\bar{E}_{i,p}^2$ are for the m-th subband.

The first term in $\{\}$ of equation (3) means the ratio of the product of the input signal power and the filter coefficient power in each subband to the total subband power, and the second term means the ratio of the error signal power in each subband to the total subband power. γ indicates the ratio of the weights of the first and second terms, and in (3), γ = 1 means the conventional tap assignment algorithm.

III. Subband ANC System in Consideration of Psychoacoustic Properties

A. Psychoacoustic Properties

When humans hear sounds, auditory sensors have properties such as masking, loudness, and pitch. In this paper, we focus on loudness. Loudness refers to the subjective loudness of sound as perceived by humans, and is an important indicator in noise evaluation. The relationship between the sensory quantity and the physical sound pressure level is shown by the equal-loudness-level-curve, which connects the sound pressure levels at which sounds of various frequencies sound the same loudness (loudness) to the senses. This characteristic means that even if the sound pressure level is the same, if the frequencies of the sounds are different, humans will perceive the sounds as different loudness.

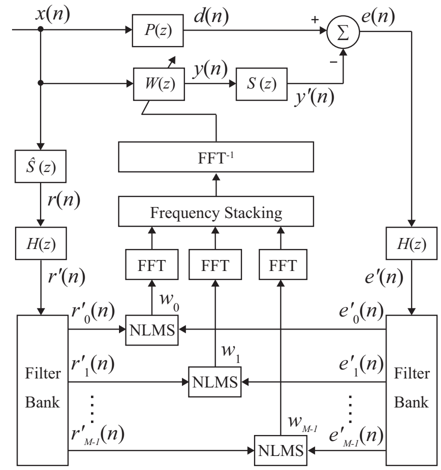
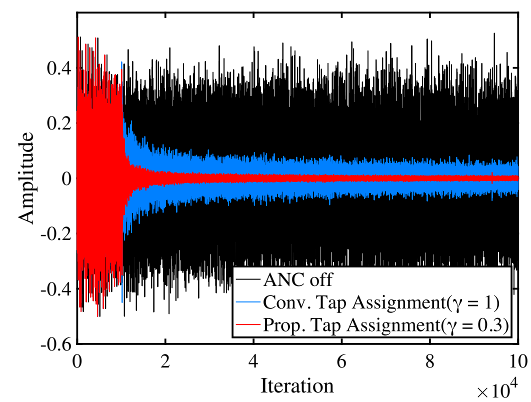
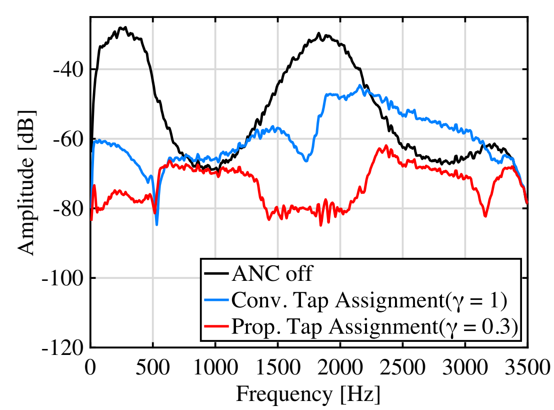


Fig. 3. Block diagram of the subband ANC considering psychoacoustic property.



(a) Time waveform



(b) Frequency spectrum

Fig. 4. Comparison of tap assignment methods in case of changing γ.

B. Noise Weighting Filter

In noise evaluation, frequency correction characteristics are used to reflect the human hearing characteristics in the sound

TABLE I
SIMULATION CONDITIONS

Target noise	Band-limited white noise
Sampling frequency	8000 Hz
Update algorithm of filters	NLMS
Tap length of noise controll filter W	1024
Tap length of filters P, S and $\hat{S}$	250
Step size parameter α	0.05
Regularization parameter β	1.0×10^{-6}
Number of subbands M	8
Block length for tap control L	10000
Tap redistribution step size R	60
Ratio γ	0, 0.3, 0.5, 0.8, 1

pressure level. In this section, a filter with A characteristic is used. This filter corresponds to the equal loudness curve, and its frequency response is shown in Fig. 2. In Fig. 2, the original A-weighting filter is shown as a green line. However, in this paper, we use a filter that approximates the A-weighting filter with short taps FIR filter, which is a red line.

C. Subband ANC System in Consideration of Psychoacoustic Properties

Fig. 3 shows a block diagram of the proposed subband ANC system, where $P(z)$ is the primary path, $S(z)$ is the secondary path, and $\hat{S}(z)$ is the model of the $S(z)$, and $H(z)$ is the noise weighting filter, respectively. Moreover, $r(n)$ is the filtered reference signal; $r'(n)$ is the filtered reference signal through $H(z)$; $r'_0(n)$... $r'_{M-1}(n)$ are the filtered reference signals through $H(z)$ divided by filter bank; $e'(n)$ is the error signal through $H(z)$; $e'_0(n)$... $e'_{M-1}(n)$ are the error signals through $H(z)$ divided by the filter bank; M is the number of subbands.

IV. SIMULATION RESULTS

A. Simulation Conditions

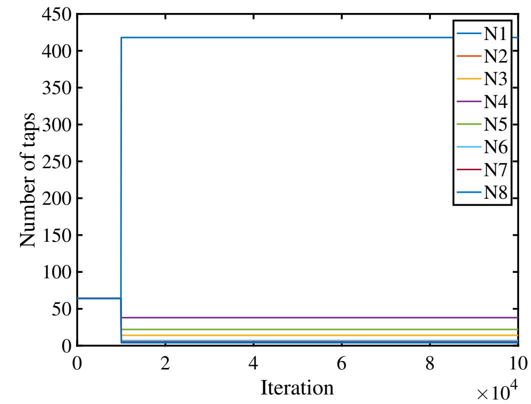
In this chapter, we verify the effectiveness of the proposed automatic tap assignment algorithm for subband ANC systems by comparing it with the conventional algorithm. The comparison of the noise reduction performance of the conventional algorithm (γ = 1) and the proposed algorithm is done by computer simulation with different values of γ. The conditions of the computer simulation are shown in Table I.

Next, we compare the noise reduction performance of the subband ANC with and without A-weighting filter at the value of γ that yields the highest noise reduction performance in the previous simulation. The A-weighting filter used in this simulation is implemented by a 1024-tap FIR filter designed using the window function method with a sampling frequency of 8000 Hz and 16384 FFT points.

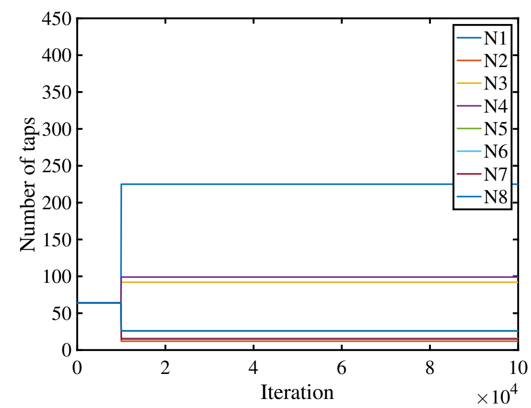
B. Simulation Results of Appropriate Tap Assignment for the Subband ANC System.

We changed γ = 0, 0.3, 0.5, 0.8 and 1.0 and then compared the noise reduction performance. As a result, the noise reduction performance of the subband where the sound energy is concentrated is greatly improved when γ value is close to 0. This is because an appropriate tap length is assigned to the subband where the sound energy is concentrated. The most effective noise reduction is achieved when γ = 0.3 and γ = 0.

Fig. 4 shows a comparison of the simulation results of the conventional algorithm (γ = 1) and the proposed algorithm (γ = 0.3). In the time waveforms shown in Fig. 4 (a), we



(a) Conventional method



(b) Proposed method

Fig. 5. Transitions of the number of taps.

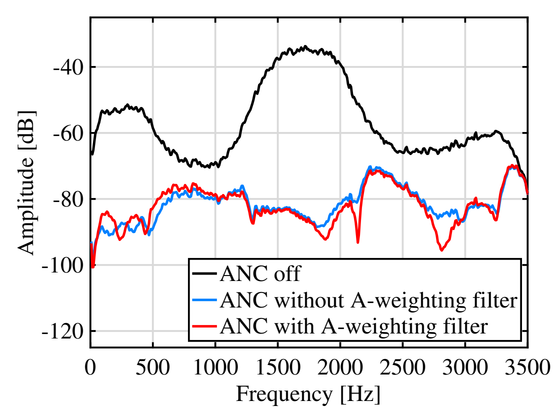


Fig. 6. Comparison with and without A-weighting filter (γ = 0.3).

can see that the noise reduction performance of the proposed algorithm is improved. In addition, the frequency spectrum in Fig. 4 (b) shows that the noise reduction performance of the proposed algorithm is up to 30 dB from 0 Hz to 500 Hz and 1400 Hz to 3300.

Fig. 5 shows the number of taps in each subband for each number of updates assigned by the algorithm. For example, if the sampling frequency is 8000 Hz and the number of subbands is 8, the number of taps for 0-500 Hz is N1, for 500-1000 Hz is N2, and for 1000-1500 Hz is N3. In the conventional algorithm of Fig. 5 (a), when tthe number of update is 10000, 420 taps are intensively assigned to N1. In contrast, in the proposed algorithm in Fig. 5 (b), 230 taps are assigned to N1, 100 taps to N4 (1500-2000 Hz), and 90 taps to N3. Thus, it can be seen that the proposed algorithm prevents excessive tap assignment in the initial stage of N1 update and assigns taps in a balanced manner.

C. Simulation Results of Subband ANC System in Consideration of Psychoacoustic Properties.

Fig. 6 shows a comparison of the simulation results with and without the A-weighting filter for subband ANC with γ = 0.3. Note that this simulation is performed with the noise in Chapter B changed. It can be seen that there is 5 dB degradation in reduction performance from 0 to 500 Hz, but 10 dB improvement in reduction performance from 2000 to 3000 Hz. This is thought to be the effect of the A-weighting filter. However, it is necessary to verify the effectiveness of the A-weighting filter by conducting subjective evaluation experiments to determine whether the filter is actually effective in making humans feel less uncomfortable.

V. Conclusion

In this paper, we proposed a new automatic tap assignment algorithm considering the error signal power of each subband and a subband ANC system considering the psychoacoustic characteristics. First, the conventional algorithm performs extreme tap assignment in the initial stage of noise control update. The proposed algorithm improves the noise reduction performance of the subband ANC by adding the error signal power to achieve a more appropriate tap assignment than the conventional method. Next, an A-weighting filter is introduced into the proposed subband ANC to take into account the human hearing characteristics. Although the A-weighting filter showed a reduction effect, we have not been able to confirm whether or not it actually makes humans feel less uncomfortable. In the future, we need to conduct subjective evaluation experiments to verify the effect of introducing the A-weighting filter into the subband ANC.

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