# A method steering a null toward a point in high reverberate non-minimum phase acoustic space

Kensaku Fujii, Takuto Yoshioka, Yusuke Kuwahara, Mitsuji Muneyasu, Masakazu Morimoto

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In this paper, we propose a method capable of steering a null toward a point in high reverberate non-minimum phase acoustic space. This method is characterized by applying the cascade connection of recursive and non-recursive adaptive filters to steering the null. Practically, the cascade connection can completely express the structure of acoustic paths from a sound source to microphones. However, as well known, the recursive adaptive filter has two problems. One is that the performance of the recursive adaptive filter is apt to become unstable. We solve this problem by applying the lattice structure to the adaptive filter. The stable performance is guaranteed by limiting the reflection coefficients to less than unity. Another is that the response of the recursive filter diverges when the minimum phase condition is not satisfied. We hence divide the recursive filter into the minimum and the non-minimum phase components and then apply the former to the cascade connection. The latter is transformed into a non-recursive filter, which is connected to the output of another microphone. The cascade connection can thereby steer a null with high stability and precision.

# 1 Introduction

Microphone array systems are classified into two types. One is *delay-and-sum array*, which reduces noises by steering a main lobe toward the incident direction of a target signal [1]. Another is *adaptive beam former* [2], which eliminates a noise by steering a null. The principle of adaptive filter is generally used for steering the null and the main lobe. The adaptive filter composes the replica of the desired signal detected by one microphone, using the output of another microphone as a reference signal. The adaptive filter estimates the coefficients of its own so as to minimize the difference between the replica and the desired signal; thereby a null is steered toward the incident direction of the target signal when the difference is minimized, and inversely the addition of the replica and the desired signal composes a main lobe.

The non-recursive type is generally applied to the adaptive filter composing the replica in order to guarantee the stable estimation of the coefficients. However, the target signal is practically assumed to arrive at the microphones through reverberate acoustic paths. In this case, the unknown path estimated by the adaptive filter is described by the cascade connection of the non-recursive and the recursive filters. It is accordingly desirable that the adaptive filter is also formed by the cascade connection. The recursive adaptive filter, however, has two problems.

One is that the recursive adaptive filter is apt to become unstable. The equation error method [3] is well known as an effective method in solving this problem. In this method, two non-recursive adaptive filters are connected to each output of the microphones; thereby the stable estimation is guaranteed. One of the non-recursive adaptive filters, however, must be finally transformed into the recursive type. The stable performance is not guaranteed to the transformed recursive filter. In this paper, we solve this problem by applying the lattice structure to the adaptive filter. Actually, the stable performance of the recursive adaptive lattice filter is guaranteed by limiting all the amplitudes of the reflection coefficients to less than unity.

Another problem is caused by the acoustic path not satisfying the minimum phase condition. In this case, the stable performance is not guaranteed even to the lattice filter. We solve this problem by dividing the non-recursive adaptive filter into non-minimum and minimum phase components. We moreover transform only the latter to a recursive filter and connect the former to the output of another microphone. A null can be thereby steered with high precision and stability even in the non-minimum phase acoustic space.

In this paper, we also present a method for dividing the non-recursive filter into non-minimum and minimum phase components and verify using computer simulations that a null can be steered toward the incident direction of the target signal by the division method.

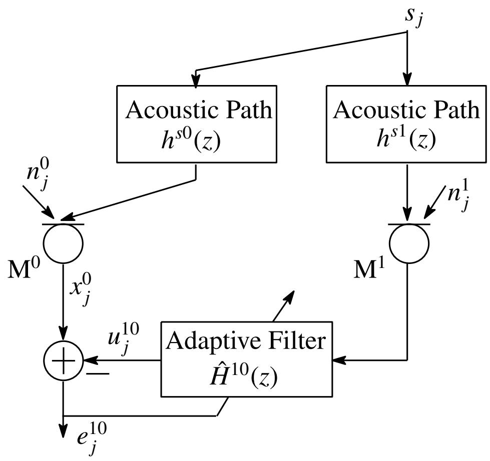


Figure 1: Basic configuration of microphone array system.

# 2 Structures of acoustic paths

Figure 1 shows the basic configuration of the microphone array system steering a null toward the incident direction of sound source signal $s_j$, where $h^{s1}(z)$ and $h^{s0}(z)$ are the transfer functions of the acoustic paths from the sound source to microphones $M^1$ and $M^0$, $n_j^0$ and $n_j^1$ are environmental noises, and $j$ denotes sample time index, respectively. In this system, the transfer function of the adaptive filter, $\hat{H}^{10}(z)$, approximates to that of the unknown path from microphones $M^1$ to $M^0$,

$$h^{10}(z) = \frac{h^{s0}(z)}{h^{s1}(z)}, \tag{1}$$

with the decrease of the output error,

$$e_j^{10} = x_j^0 - u_j^{10}, \tag{2}$$

where $x_j^0$ and $u_j^{10}$ are the outputs of microphone $M^0$ and the adaptive filter, respectively. The sound source signal detected by microphone $M^0$ is canceled when the output error is minimized; consequently a null is steered toward the incident direction of the sound source signal.

The acoustic paths can be generally assumed to be linear time-invariant, whose transfer functions can be expressed by

$$h^{s0}(z) = \frac{a^{s0} z^{-K}\left\{1 + a^{s0}(1)z^{-1} + \cdots + a^{s0}(M^{s0})z^{-M^{s0}}\right\}}{1 + b^{s0}(1)z^{-1} + \cdots + b^{s0}(N^{s0})z^{-N^{s0}}}$$

$$= a^{s0} z^{-K} \frac{1 - h_F^{s0}(z)}{1 - h_I^{s0}(z)} \tag{3}$$

and

$$h^{s1}(z) = \frac{a^{s1}\left\{1 + a^{s1}(1)z^{-1} + \cdots + a^{s1}(M^{s1})z^{-M^{s1}}\right\}}{1 + b^{s1}(1)z^{-1} + \cdots + b^{s1}(N^{s1})z^{-N^{s1}}}$$
$$= a^{s1} \frac{1 - h_F^{s1}(z)}{1 - h_I^{s1}(z)} \tag{4}$$

respectively, where $a^{s0}$, $a^{s0}(1) \sim a^{s0}(M^{s0})$, $b^{s0}(1) \sim b^{s0}(N^{s0})$, $a^{s1}$, $a^{s1}(1) \sim a^{s1}(M^{s1})$ and $b^{s1}(1) \sim b^{s1}(N^{s1})$ are constant, $M^{s0}$, $N^{s0}$, $M^{s1}$ and $N^{s1}$ are integer,

$$1 - h_F^{s0}(z) = 1 + a^{s0}(1)z^{-1} + \cdots + a^{s0}(M^{s0})z^{-M^{s0}}, \tag{5}$$
$$1 - h_F^{s1}(z) = 1 + a^{s1}(1)z^{-1} + \cdots + a^{s1}(M^{s1})z^{-M^{s1}}, \tag{6}$$
$$1 - h_I^{s0}(z) = 1 - b^{s0}(1)z^{-1} + \cdots + b^{s0}(N^{s0})z^{-N^{s0}}, \tag{7}$$

and

$$1 - h_I^{s1}(z) = 1 - b^{s1}(1)z^{-1} + \cdots + b^{s1}(N^{s1})z^{-N^{s1}} \tag{8}$$

provide the zeros and the poles of the acoustic paths, and the sound source signal is assumed to arrive at microphone $M^0$ $K$ sample times later than at microphone $M^1$.

The transfer function of the unknown path estimated by the adaptive filter is accordingly described by

$$h^{10}(z) = \frac{a^{s0} z^{-K}}{a^{s1}} \left\{1 - h_F^{s0}(z)\right\}\left\{1 - h_I^{s1}(z)\right\} \times \frac{1}{\left\{1 - h_F^{s1}(z)\right\}\left\{1 - h_I^{s0}(z)\right\}}. \tag{9}$$

Eq. (9) states that the unknown path is desired to be estimated by the cascade connection of the non-recursive and recursive adaptive filters, $\hat{H}_F^{10}(z) \times 1/\left\{1 - \hat{H}_I^{10}(z)\right\}$, where

$$\hat{H}_F^{10}(z) \approx \frac{a^{s0} z^{-K}}{a^{s1}} \left\{1 - h_F^{s0}(z)\right\}\left\{1 - h_I^{s1}(z)\right\} \tag{10}$$

and

$$\frac{1}{1 - \hat{H}_I^{10}(z)} \approx \frac{1}{\left\{1 - h_F^{s1}(z)\right\}\left\{1 - h_I^{s0}(z)\right\}}. \tag{11}$$

The problem is that the response of the recursive adaptive filter diverges when minimum phase components are involved in $1 - h_F^{s1}(z)$.

# 3 Proposed system

Figure 2 shows the configuration of the microphone array system proposed in this paper. In this configuration, the non-recursive adaptive filters, $\hat{H}_F^{10}(z)$ and $1 - \hat{H}_I^{10}(z)$, give the relation,

$$h^{s0}(z)\left\{1 - \hat{H}_I^{10}(z)\right\} \approx h^{s1}(z)\hat{H}_F^{10}(z), \tag{12}$$

when equation error $\hat{e}_j^{10}$ is minimized. This relation can be moreover rewritten to

$$\hat{H}_F^{10}(z) \times \frac{1}{1 - \hat{H}_I^{10}(z)} \approx \frac{h^{s0}(z)}{h^{s1}(z)} = h^{10}(z), \tag{13}$$

which states that the cascade connection of the non-recursive and the recursive adaptive filters, $\hat{H}_F^{10}(z)$ and $1/\{1 - \hat{H}_I^{10}(z)\}$,

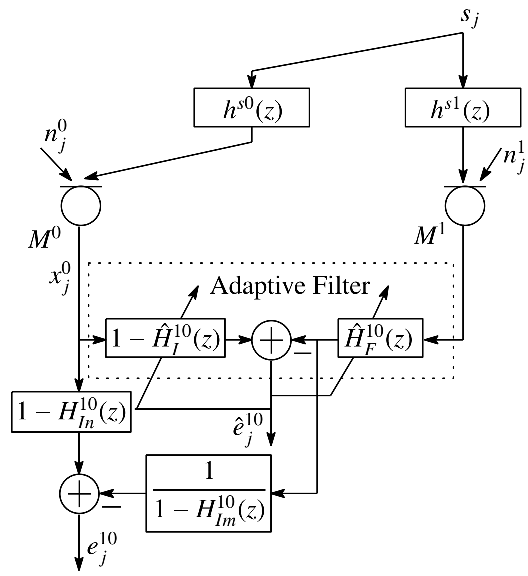


Figure 2: Configuration of proposed system.

is approximated to unknown path $h^{10}(z)$. A null can be thus steered toward the incident direction of the sound source signal when non-minimum phase components are not involved in acoustic path $h^{s1}(z)$.

To prevent the divergence caused by the non-minimum phase components, we divide the non-recursive adaptive filter, $1-\hat{H}_I^{10}(z)$, into the non-minimum and the minimum phase components, $1 - H_{In}^{10}(z)$ and $1 - H_{Im}^{10}(z)$, as

$$1 - \hat{H}_I^{10}(z) = \left\{1 - H_{In}^{10}(z)\right\}\left\{1 - H_{Im}^{10}(z)\right\}. \tag{14}$$

Similarly, $1-h_F^{s1}(z)$ can be also divided into the non-minimum and the minimum phase components denoted by $1 - h_{Fn}^{s1}(z)$ and $1 - h_{Fm}^{s1}(z)$, respectively, as

$$1 - h_F^{s1}(z) = \left\{1 - h_{Fn}^{s1}(z)\right\}\left\{1 - h_{Fm}^{s1}(z)\right\}. \tag{15}$$

Eq. (11) can be then rewritten to

$$\frac{1}{1-\hat{H}_I^{10}(z)} \approx \frac{1}{1-h_{Fn}^{s1}(z)} \times \frac{1}{\left\{1-h_{Fm}^{s1}(z)\right\}\left\{1-h_I^{s0}(z)\right\}}. \tag{16}$$

This equation states that that the proposed system shown in Figure 2 can steer a null toward the incident direction of the sound source signal when the relations,

$$1 - H_{In}^{10}(z) \approx 1 - h_{Fn}^{s1}(z) \tag{17}$$

and

$$1 - H_{Im}^{10}(z) \approx \{1 - h_{Fm}^{s1}(z)\}\{1 - h_I^{s0}(z)\} \tag{18}$$

are satisfied.

Practically, the transfer function of the reference signal path from the sound source signal to the output of the recursive filter can be rearranged as

$$h^{s1}(z)\hat{H}_F^{10}(z) \times \frac{1}{1 - H_{Im}^{10}(z)}$$
$$\approx a^{s1} \frac{1 - h_F^{s1}(z)}{1 - h_I^{s1}(z)} \times \frac{a^{s0} z^{-K}}{a^{s1}} \{1 - h_F^{s0}(z)\}\{1 - h_I^{s1}(z)\} \times \frac{1}{1 - H_{Im}^{10}(z)}$$

$$\begin{aligned}
&\approx \{1 - h^{s1}_{Fm}(z)\}\{1 - h^{s1}_{Fn}(z)\} \times a^{s0} z^{-K} \{1 - h^{s0}_{F}(z)\} \\
&\quad \times \frac{1}{\{1 - h^{s1}_{Fm}(z)\}\{1 - h^{s0}_{I}(z)\}} \\
&= \{1 - h^{s1}_{Fn}(z)\} \times h^{s0}(z). \qquad (19)
\end{aligned}$$

This relation states that the transfer function of the reference signal path is approximated to that of the desired signal path from the sound source signal to the input of the subtracter through microphone $M^0$; consequently a null is steered when the divided non-minimum phase component,

$$1 - H^{10}_{In}(z) \approx 1 - h^{s1}_{Fn}(z), \qquad (20)$$

is inserted as shown in Figure 2.

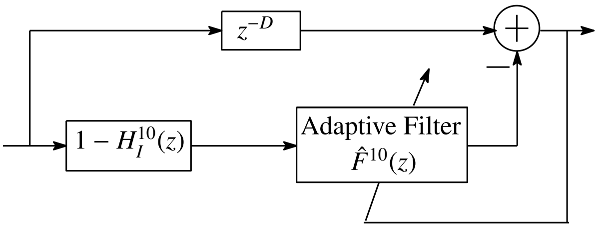


Figure 3: Transformation system into inverse filter.

# 4 Division method

In this paper, we propose a method for dividing the non-recursive adaptive filter, $1 - \hat{H}^{10}_{I}(z)$, into the non-minimum and the minimum phase components, $1 - H^{10}_{In}(z)$ and $1 - H^{10}_{Im}(z)$. The procedure of the division is as follows.

(1) Copy the coefficients of adaptive filter $1 - \hat{H}^{10}_{I}(z)$ to the non-recursive filter shown as $1 - H^{10}_{I}(z)$ in Figure 3 at appropriate intervals.

(2) Estimate the coefficients of non-recursive adaptive filter $\hat{F}^{10}(z)$ so as to minimize the remainder.

(3) Suspend the above estimation at a moderate time and then separate

$$\hat{\boldsymbol{F}}^{10}_{n} = \left[\ \hat{F}^{10}(0) \quad \hat{F}^{10}(1) \quad \cdots \quad \hat{F}^{10}(D)\ \right]^T \qquad (21)$$

from the coefficient vector of non-recursive adaptive filter $\hat{F}^{10}(z)$,

$$\hat{\boldsymbol{F}}^{10} = \left[\hat{F}^{10}(0) \ \cdots \ \hat{F}^{10}(D) \ \cdots \ \hat{F}^{10}(N-1)\right]^T. \qquad (22)$$

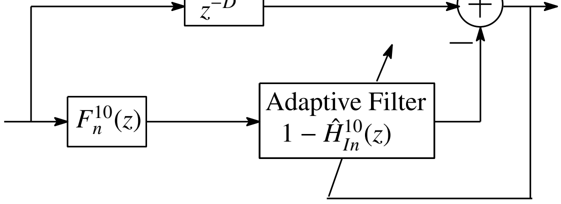


Figure 4: Transformation system of non-minimum phase component.

(4) Copy the coefficient vector, $\hat{\boldsymbol{F}}^{10}_{n}$, to the non-recursive filter shown as $F^{10}_{n}(z)$ in Figure 4.

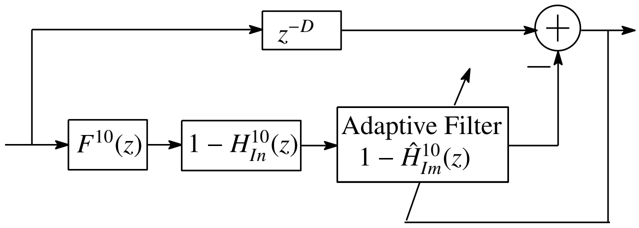


Figure 5: Extraction system of minimum phase component.

(5) Estimate the coefficients of non-recursive adaptive filter $1 - \hat{H}^{10}_{In}(z)$ so as to minimize the remainder.

(6) Suspend the above estimation at a moderate time and then copy the coefficients of the adaptive filters, $\hat{F}^{10}(z)$ and $1 - \hat{H}^{10}_{In}(z)$, to the non-recursive filter shown as $F^{10}(z)$ in Figure 5 and two non-recursive filters denoted as $1 - H^{10}_{In}(z)$ in Figures 2 and 5, respectively.

(7) Estimate the coefficients of the non-recursive adaptive filter shown as $1 - \hat{H}^{10}_{Im}(z)$ in Figure 5 so as to minimize the remainder.

(8) Suspend the above estimation at a moderate time and then copy the coefficients of $1 - \hat{H}^{10}_{Im}(z)$ to $1/\{1 - H^{10}_{Im}(z)\}$ shown in Figure 2.

In the extraction system shown in Figure 5, the cascade connection of $F^{10}(z)$ and $1 - H^{10}_{In}(z)$ can be approximated to the inverse filter of the minimum phase component of $1 - H^{10}_{I}(z)$; consequently the non-recursive adaptive filter $1 - \hat{H}^{10}_{Im}(z)$ converges on that of $1 - H^{10}_{I}(z)$ with the decrease of the remainder. We then form the adaptive filter $1 - \hat{H}^{10}_{Im}(z)$ with the lattice type and limit the amplitudes of the reflection coefficients to less than unity. The stable performance of the recursive filter $1/\left\{1 - H^{10}_{Im}(z)\right\}$ is thereby guaranteed.

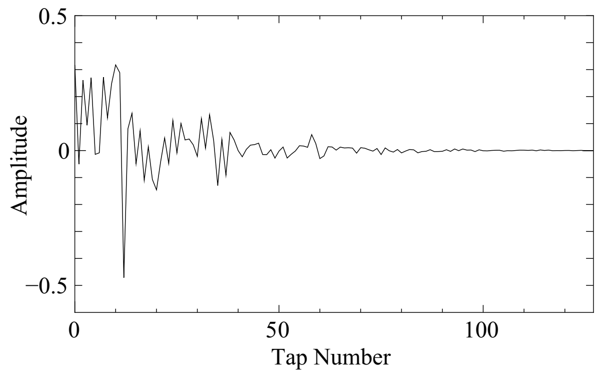


Figure 6: Impulse response of acoustic path $h^{s1}(z)$.

# 5 Computer simulations

We finally verified using computer simulations that the proposed system can successfully steer a null toward a point in high reverberate non-minimum phase acoustic space. The simulation conditions are as follows.

- Impulse response samples of acoustic paths, $h^{s1}(z)$ and $h^{s0}(z)$: Exponentially decayed normal numbers shown in Figures 6 and 7, respectively.

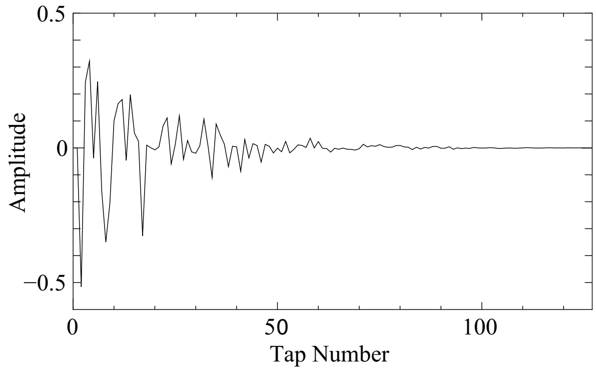


Figure 7: Impulse response of acoustic path $h^{s0}(z)$.

- Arriving time lag of sound source signal: $K = 2$.
- Numbers of taps of adaptive filters, $\hat{H}_F^{10}(z)$ and $1 - \hat{H}_I^{10}(z)$: 256, respectively.
- Number of taps of adaptive filter $\hat{H}^{10}(z)$ shown in Figure 1: 2048.
- Adaptive algorithm estimating coefficients of $\hat{H}_F^{10}(z)$, $1 - \hat{H}_I^{10}(z)$ and $\hat{H}^{10}(z)$: Normalized least mean square (NLMS) algorithm whose step size is 0.1.
- Sound source signal and environmental noises: White noise.
- Power ratio of sound source signal to environmental noise: 30 dB.
- Number of taps of adaptive filter $\hat{F}^{10}(z)$: 2048.
- Inserted delay $z^{-D}$: $D = 512$.
- Numbers of taps of non-recursive filters, $F_n^{10}(z)$ and $1 - \hat{H}_{In}^{10}(z)$: 512, respectively.
- Number of stages of non-recursive adaptive lattice filter $1 - \hat{H}_{Im}^{10}(z)$: 512.
- Adaptive algorithm estimating reflection coefficients of $1 - \hat{H}_{Im}^{10}(z)$:

$$\boldsymbol{\gamma}_{j+1} = \boldsymbol{\gamma}_j + \mu \frac{\tilde{e}_j^{10} \boldsymbol{b}_j}{\boldsymbol{b}_j^T \boldsymbol{b}_j} \tag{23}$$

proposed in [4], where

$$\boldsymbol{\gamma}_j = \begin{bmatrix} \gamma_j(1) & \gamma_j(2) & \cdots & \gamma_j(I-1) \end{bmatrix}^T. \tag{24}$$

is the forward reflection coefficient vector, whose elements are copied to the backward that, $\mu$ is a constant called step size, $\tilde{e}_j$ is the difference between the outputs of delay $z^{-D}$ and adaptive filter $1 - \hat{H}_{Im}^{10}(z)$ shown in Figure 5,

$$\boldsymbol{b}_j = \begin{bmatrix} b_j(1) & b_j(2) & \cdots & b_j(I-1) \end{bmatrix}^T \tag{25}$$

is the backward prediction error vector, and $I$ denotes the number of stages.

- Interval of copying coefficient vector of $1 - \hat{H}_I^{10}(z)$ shown in Figure 2 to $1 - H_I^{10}(z)$ shown in Figure 3: 100,000 sample times.
- Adaptive algorithm estimating coefficients of $\hat{F}^{10}(z)$, $1 - \hat{H}_{In}^{10}(z)$ and $1 - \hat{H}_{Im}^{10}(z)$: NLMS algorithm whose step size is 1.0.
- Number of times of above estimation: 100,000.

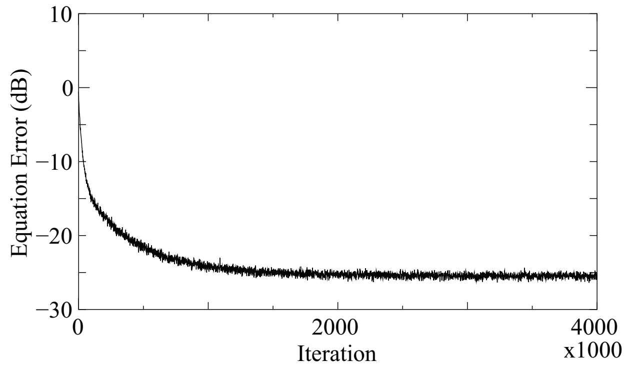


Figure 8: Decreasing property of equation error obtained by proposed method.

Figure 8 shows the decreasing property of the equation error obtained by the proposed system, where that is calculated using

$$Oa_n = 10 \log_{10} \sum_{j=nJ+1}^{(n+1)J} (\hat{s}_j^{10})^2 \Bigg/ \sum_{j=nJ+1}^{(n+1)J} (s_j^0)^2. \tag{26}$$

In Eq. (26), $J = 1,000$ is given in order to reduce the fluctuation of the equation error, and $s_j^0$ and $\hat{s}_j^{10}$ are the sound source signal components involved in the output of microphone $\mathrm{M}^0$ and equation error $\hat{e}_j^{10}$, respectively. As seen from the result, the proposed system reduces the sound source signal to about –25 dB. This result shows that the proposed system can successfully steer a null toward the incident direction of the sound source signal.

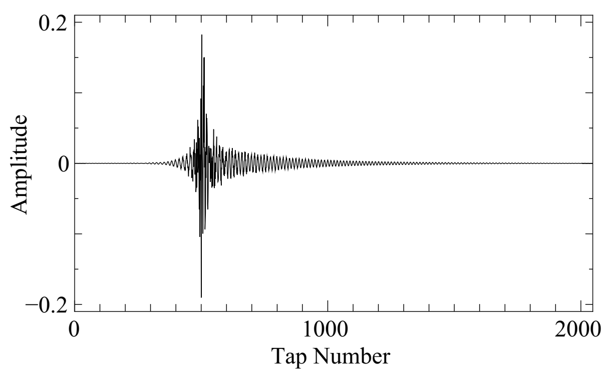


Figure 9: Coefficients of inverse filter $F^{10}(z)$.

In the proposed system, the coefficients of adaptive filter $1 - \hat{H}_I^{10}(z)$ are copied to non-recursive filter $1 - H_I^{10}(z)$ composing the transformation system shown in Figure 3. Figure 9 shows the coefficients of inverse filter $F^{10}(z)$ obtained by the transformation system. In this result, the elements of coefficient vector $\hat{\boldsymbol{F}}_n^{10}$ is approximated to the impulse response samples of the inverse filter of non-minimum phase component $1 - H_{In}^{10}(z)$. Actually, the elements of coefficient vector $\hat{\boldsymbol{F}}_n^{10}$ converges on zero when non-minimum phase components are not involved in non-recursive filter $1 - H_I^{10}(z)$.

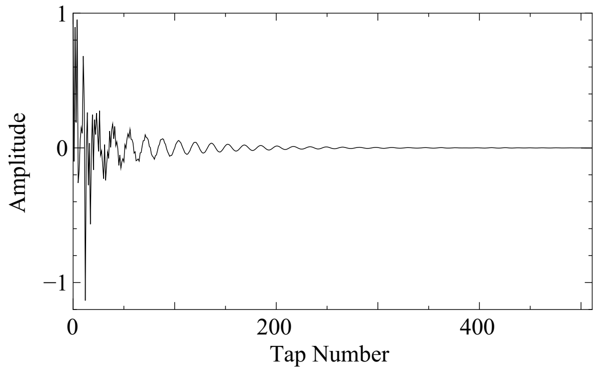


Figure 10: Coefficients of non-minimum phase component filter shown as $1 - H_{In}^{10}(z)$ in Figures 2 and 5.

We hence copy the separated coefficient vector, $\hat{\boldsymbol{F}}_n^{10}$, to non-recursive filter $F_n^{10}(z)$ composing the transformation system shown in Figure 4 and then estimate the coefficients of adaptive filter $1 - \hat{H}_{In}^{10}(z)$ so as to minimize the remainder. The adaptive filter, $1 - \hat{H}_{In}^{10}(z)$, thereby approximates to the non-minimum phase component of $1 - H_I^{10}(z)$. Figures 10 shows the coefficients of the non-minimum phase component estimated by using the transformation system shown in Figure 4, which are copied to two non-recursive filters shown as $1 - H_{In}^{10}(z)$ in Figures 2 and 5, respectively.

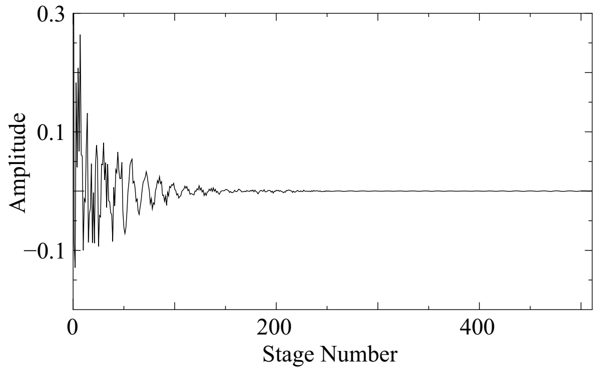


Figure 11: Reflection coefficients of minimum phase component filter $1 - H_{Im}^{10}(z)$.

Figure 11 shows the reflection coefficients of minimum phase component $1 - H_{Im}^{10}(z)$ obtained by the extraction system shown in Figure 5. As seen from the result, all the amplitudes of the reflection coefficients are less than unity, which states that recursive filter $1/\left\{1 - H_{Im}^{10}(z)\right\}$ stably works.

Figure 12 shows the decreasing properties of the output errors obtained by the proposed and the conventional systems shown in Figures 1 and 2, where those are calculated using

$$Ob_n = 10\log_{10} \sum_{j=nJ+1}^{(n+1)J} (s_j^{10})^2 \Bigg/ \sum_{j=nJ+1}^{(n+1)J} (s_j^{0})^2, \tag{27}$$

where $J = 1,000$, and $s_j^{10}$ is the sound source signal component involved in output error $e_j^{10}$ shown in Figures 1 and 2, respectively. As seen from the result, the proposed system reduces the sound source signal to about –25dB, whereas the conventional system can hardly decreases that.

In this result, we can see that the output error sharply decreases whenever the coefficients of the non-recursive adaptive filters, $1 - \hat{H}_{In}^{10}(z)$ and $1 - \hat{H}_{Im}^{10}(z)$, obtained by the transformation and the extraction systems shown in Figures 4 and 5 are copied to the non-recursive and recursive filters, $1 - H_{In}^{10}(z)$ and $1 - H_{Im}^{10}(z)$, shown in Figures 1 and 2. The fluctuation of the output error, however, decreases with the convergence of the coefficients of the adaptive filters, $\hat{H}_F^{10}(z)$ and $1 - \hat{H}_I^{10}(z)$.

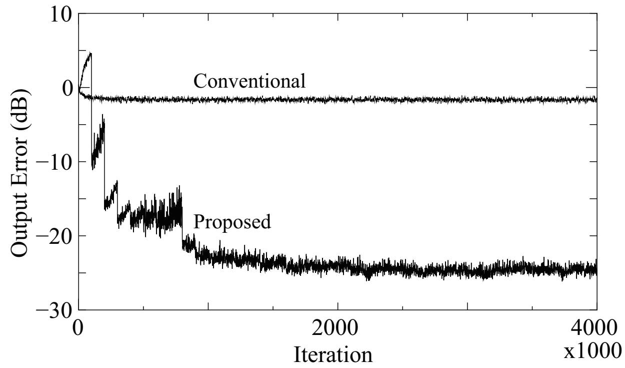


Figure 12: Decreasing property of output error obtained by proposed system.

# 6 Conclusion

In this paper, we have proposed a microphone array system capable of steering a null toward a point in high reverberate non-minimum phase acoustic space and have presented a method for dividing a filter into the non-minimum and the minimum phase components. In addition, we have verified using computer simulations that the proposed system and presented method successfully work. The division method can be applied to various fields, such as active noise control, sound reproduction. On the other hand, the environmental noise degrades performance of the proposed system. In the near future, we will study on the methods reducing the bias error caused by the environmental noise [5].

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