Implementation of Virtual Sound Source
Considering Acoustic Path

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Abstract—Development of acoustic systems is a necessitate which require research over audio and speech processing. A novel and interesting concept in the real of acoustic systems is controlling room acoustic path. It means to orient the sound locally in a room to a desired person without annoying others and without using headphone. Room acoustic path enable us to create personal sound space in a specific location of the room. In our former research, we convert two input signals into three output signals. The output signals finally conclude two separated personal sound spaces by using a crosstalk cancellation pre-filter. Our research aimed to reach observed signals at each of two points independent from the other one. Up to the time, only room acoustic attenuation and phase shift was considered between speaker and observer. However, the acoustic impulse response was ignored. So, we considered the acoustic impulse response in this system which does not require the setting state of the speaker. In our new research we realize the above system. The research has two parts, hardware implementation and practical investigation of the realized system.

Keywords-component; Virtual sound source system; Acoustic path; Inverse filter; Equation considering convolution

I. INTRODUCTION

The research about acoustics is a necessary theme for sound processing and development of the audio system. Currently, the sound system that does not trouble another person by using an earphone and the headphones that carrying around is becoming the mainstream. Reproducing the acoustic signals that would have been produced by a specific source of sound located at a particular position relative to the listener at his ears is the main requirement in a lot of applications. As an example, while family members sit at a room, and a person wants to hear a sound source which may be unpleasant for other ones. The virtual sound system provides such facility that each person enjoy his favorite sound source without interfering the others.

The virtual sound system directs and separates sounds to desired positions. It enables each person to listen to his own sound source without hearing the others. This separation and direction is not by the mean of headphone which makes a person quite isolated from his surroundings. Gardner used a preprocessing technique adopted by compensation for the effect of acoustic crosstalk path [1]. Takeuchi et al, have employed a 2-channel multiway sound control system to improve the robustness performance of the system [2]. To improve the performance over the 2-channel case, Yang et al, proposed a 3-channel system for virtual sound imaging [3].

In this paper, the acoustic path has been also considered in virtual sound system. Controlling room acoustic path lead to more improvement of the system.

The organization of the paper is as follows. Section 2, explains the research goal in a virtual sound system. Section 3 describes how the inverse filtering reproduces sound into three speakers. In section 4 room acoustic path has been considered in the design of inverse filters. Section 5 presents the simulation results. Finally section 6 concludes the paper.

II. VIRTUAL SOUND SYSTEM

Virtual sound system creates personal sound space in a specific location of a room. It aims this goal by converting two input signal $D_1, D_2$ into three output signals using crosstalk cancellation filter. These two points are observed as the output signals at two identified points. If observed signal at each spot will be independent from the other spot, then, we could create the personal sound space [3-5]. Up to now, only room acoustic attenuation and phase shift has been considered between speaker and observer (Listener or Microphone). However, the acoustic impulse response was ignored. So, here, we consider the acoustic impulse response in this system.

The acoustic path consists of several reflection coefficients which are delayed after sound signal reflected from different obstacles in the room [6,7].

Fig.1 shows three speakers sound system to reproduce the desired sound localization. $D_m (m=1,2)$ or source signals to be heard at observation locations. Also, $V_n = (1,2,3)$ are output of the inverse filter, $H$, to be propagated from three speakers. $A$ is a $2 \times 3$ transform matrix that transfers input speakers signals, $V_n$, into signals, $D_m$, observed signals, $D_m'$. The components of matrix $A$ are

$$a_{mn} = e^{-jk \frac{r_{mn}}{r_{mn}}}$$

where, $k$, is frequency and $r_{mn}$ is the distance between $m^{th}$ observer and $n^{th}$ speaker.
Using symmetric properties of geometrical locations we can write
\( r_{11} = r_{23}, r_{12} = r_{22}, r_{13} = r_{21} \). Therefore, \( A \) becomes
\[
A = \begin{bmatrix}
   e^{-jkr_{11}} / r_{11} & e^{-jkr_{21}} / r_{21} & e^{-jkr_{31}} / r_{31} \\
   e^{-jkr_{12}} / r_{12} & e^{-jkr_{22}} / r_{22} & e^{-jkr_{32}} / r_{32}
\end{bmatrix}
\]
(1)
and input signals \( D_1, D_2 \) become
\[
D = \begin{bmatrix} D_1 \\ D_2 \end{bmatrix}.
\]
(2)

If we assume that \( V = HD \), then, the error signal will be
\[
e = D - AV
\]
(3)

The inverse filter, \( H \), should be designed such that the mean squared error, \( J \), will be minimized \( J = e^H e \) (\( H \) is Hermitian). Then, the inverse filter will be obtained as follows;
\[
H = A^H (AA^H)^{-1}
\]
(4)

Section Inverse filter design considering acoustic path.

A. Introduction of acoustic path

From the previous equation, we can obtain the inverse filter, \( H \), as follows
\[
H = \begin{bmatrix}
   h_1 & h_3 \\
   h_2 & h_2 \\
   h_3 & h_1
\end{bmatrix}
\]
(5)

\[
h_1 = \frac{a_1 a_2 + a_2^2 + a_3^2}{(a_1 - a_3)[2a_2^2 + (a_1 + a_2)^2]}
\]
(6)

\[
h_2 = \frac{a_2^2}{2a_2^2 + (a_1 + a_3)^2}
\]
(7)

\[
h_3 = \frac{(a_1 a_2 + a_2^2 + a_3^2)}{(a_1 - a_3)[2a_2^2 + (a_1 + a_3)^2]}
\]
(8)

The transfer functions \( a_m \) are given by Eq.(1). However in this paper, we substitute them by acoustic impulse responses.

Unlike the transfer functions, we consider \( a_m \) as a vector of the following form
\[
a_m = [a_{m1} a_{m2} a_{m3} \cdots a_{mi}],
\]
(9)
where \( i \) is the length of samples in acoustic path.

The components of inverse filter, \( h_i \), are also represented in vector form as;
\[
h_m = [\beta_{i1} \beta_{i2} \beta_{i3} \cdots \beta_{ji}],
\]
(10)

Section Inverse filter design considering acoustic path.

B. Derivation of equation considering convolution

The speakers output signals are propagated into acoustic path to reach to observers. This process is done using
convolution. Therefore, the error signal will be obtained by using convolution;

\[
e = D - A \ast A^H \otimes (A \ast A^H)^{-1} \ast D
\]

\[
e = D - A \ast \left[ H \ast D \right]
\]

Eq.(11) was obtained using associative law in convolution. The observation signals are provided such that the error signal will be minimized. In addition, the multiplication of \( AA^H \) will be performed in convolution style as follows;

\[
A \ast A^H = \begin{bmatrix}
 b_1 & b_2 \\
 b_3 & b_4 \\
\end{bmatrix}
\]

\[
b_1 = a_1 \ast a_1 + a_2 \ast a_2 + a_3 \ast a_3
\]

\[
b_2 = a_1 \ast a_2 + a_2 \ast a_2 + a_3 \ast a_1
\]

From these, filter \( H \) is

\[
H = A^H \otimes (A \ast A^H)^{-1}
\]

\[
H = \begin{bmatrix}
 h_1 & h_2 \\
 h_3 & h_4 \\
\end{bmatrix}
\]

\[
h_1 = \frac{a_1 \ast b_1 + a_3 \ast b_2}{b_1^2 - b_2^2}
\]

\[
h_2 = \frac{a_2 \ast b_1 + a_4 \ast b_2}{b_1^2 - b_2^2}
\]

\[
h_3 = \frac{a_3 \ast b_1 + a_2 \ast b_2}{b_1^2 - b_2^2}
\]

III. SIMULATION

A. Simulation method

To show the effectiveness of the proposed inverse filter, we have performed simulation. We assume that the acoustic paths between installed speakers and observation points are symmetric. We have simulated the sound path \( a_m \) by:

\[
a_m(i) = \text{Rand} \left[ \exp(-8i/M) \right],
\]

with this expression, the exponential decaying shape will be decreased to -60 dB after \( M \) samples. That is, the sound generated from speakers will be diminished after several reflections from the wall and other furniture in the room. Therefore, selected acoustic path has exponential decaying shape as shown in Fig.3.

The input to \( D_1 \) is a voice signal and the one to \( D_2 \) is a music with 10,000 samples for this simulation.

B. Simulation result

In Fig.3, and Fig.4, the simulation results are depicted for input signals \( D \), observation signals \( V = A \ast H^* \ast D \), and the error signals \( e = D - A \ast H \ast D \) in upper row, center row, and bottom row respectively. As we can see the input and observation signals are very similar with small error overshoot. In this simulation the error might be the resulting or rounding error in the computations.

However, if we change the positions of observation without changing acoustic path, the error signals will be increased, that is not perfect separation is done.

IV. CONCLUSION

In this paper, the acoustic path has been considered in virtual sound source system. The system with two different input signals are excited to generate three signals for three speakers. At observation point, we could separate the two input signals, then virtually each input signal could be observed at assigned location. The error from other signal is very small. Therefore, a high quality source separation was obtained. As a future work, we plan to use the proposed algorithm in real experiment using real acoustic path.

B. G. thanks”. Put sponsor acknowledgments in the unnumbered footnote on the first page.

REFERENCES


Figure 4. Simulation Results