1aSP6. Extension of perceived source width using sound field reproduction systems

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We propose a method to extend the perceived spatial width of a virtual sound source using an array of loudspeakers. In general, the control of perceived source width or apparent source width (ASW) has been attempted by decreasing the inter-aural correlation, i.e., the correlation between two signals on the left and right ears. For this purpose, numerous decorrelators have been proposed for stereo loudspeakers or headphones. However, most of these techniques are inadequate for the sound field reproduction system incorporating multiple loudspeakers.

For sound field reproduction, the extension of source width has to be realized with over a large listening area. In order to design a spatial decorrelator that can meet these requirements, we design a proper target sound field that has reduced correlation over a defined zone of interest. The performance of the proposed method is investigated by examining the inter-channel correlation coefficient of the reproduced sound field over a wide area.

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INTRODUCTION

Perceived width of a sound source is one of the important properties that determine the auditory scenery of the reproduced sound field. The perceived source width, however, is a broad terminology that includes all the micro- and macro scale sensations. For example, Morimoto and Maekawa [1] categorized the width sensation in terms of two different components: the auditory source width (ASW) and listener envelopment (LEV). The former is related to the attribute of a sound source that radiates sound, and the latter characterizes the sensation on the listening environment. More recently, Rumsey [2] classified the perceived width into three different types of sensations: the individual source width, ensemble width, and environment width. Among these, the individual source width (ISW) is more focused on the “lateral extent of a single source[2]”, which can be related to the locatedness describing “the degree to which an auditory event can be said to be clearly in a particular position[3]”.

The individual source width is a sensation produced by the combination of many source-related and room-related attributes, such as early reflections, spatial distribution of the radiating body, and its radiation patterns. It has been well known that lateral reflections are strongly related to the source broadening or increasing source width. For example, Barron and Marshall [4] found that the incidence angle of the early lateral reflections has to do with the apparent area of the perceived source. More fundamental aspect of the perceived source width has been investigated by examining the correlation between the left and right ear signals. Sayers and Cherry [5] showed through the headphone listening test that the lowered cross-correlation results in the subjective fusion of the perceived sound.

Other works by Plenge [6], Kurozumi and Ohgushi [7], and Kendall [8] also support this observation in case of two-channel loudspeakers. The similar trends between the inter-loudspeaker cross-correlation and perceived source width were also reported [10]. Currently, a widely accepted measure for describing the perceived (virtual) source width is the interaural cross-correlation coefficient (IACC), which is defined as the maximum value of the correlation function given by

\[ IACC = \max_{\tau \in [-1, 1]} \frac{\int_{-\infty}^{\infty} h_l(t)h_r(t+\tau)dt}{\sqrt{\int_{-\infty}^{\infty} h_l(t)^2 dt \int_{-\infty}^{\infty} h_r(t)^2 dt}} \]  

where \( h_l \) and \( h_r \) denote the left and right ear signals, respectively. The typical integration period for evaluating ASW is given as 0 to 80 msec.

Many techniques to control the perceived width utilize the source and room-related attributes. One popular way is to decorrelate left and right channel signals by additional early reflections; however, such decorrelation makes it difficult to assign the perceived source width as an attribute of a sound object. Since early reflections depend on the dimension and shape of the room, it requires the design of virtual environment rather than that of the virtual source.

Another approach for decorrelating ear signals makes use of the pseudo-stereo technique, which originally used for producing two stereo signals from the single mono sound. In the early works of pseudo-stereo [12-13], it was reported that the rapidly interchanging phase or panning direction for different frequencies lead to the reduction of IACC and increase of ASW. In order to generate two decorrelated signals, Kendall [8] proposed using FIR structure derived from the spectrum of flat magnitude but randomized phase. Later, this approach was extended to the sub-band structure [9], and more recently, a deterministic design method based on the periodic variation of the phase and interchannel time difference (ICTD) was proposed by Zotter et al. [10].

The pseudo-stereo technique is straightforward in that the energy ratio or phase difference between the loudspeakers is directly controlled by a single parameter. The technique is well-suited for the object-oriented reproduction scenario, since one can express and control the perceived width of a sound source solely in terms of a single source parameter, instead of wall reflections. However, main use of the pseudo-stereo technique is limited to the stereo or discrete multichannel system such as the 5.1 configuration. Since the size of sweet spot of the discrete multichannel system is small, the perception of the source direction or width can be easily degraded for the off-centered listener.

To increase the sweet spot size, one can use the sound field reproduction technique that aims at reproducing the physical wave of a target sound field using an array of loudspeakers. Then, how can we utilize the pseudo-stereo technique for the sound field reproduction? Since the sound field reproduction technique requires a pre-defined target field, we first need to design an appropriate target field that can provide the increased ASW. Although there have been a lot of studies focusing on the reproduction of early reflections from virtual boundaries, the use of pseudo-stereo technique for the sound field reproduction has not been addressed well. Recent examples by Ahrens and Spors [14] and Potard [11] showed the possibility to manipulate ASW in terms of the randomized mixture of
virtual sources or spherical harmonics, but there might be inherent difficulties with the control of randomized coefficients and evaluation.

In this paper, we propose a source extension technique based on the deterministic phase modulation of a sound field. The main focus of the proposed technique is on the design of a target field that can deliver the extended width perception of a sound source in a deterministic way. Accordingly, the proposed target field can be reproduced by various sound field reproduction techniques.

**Problem Statements**

We begin with the two-channel all-pass filter structures developed by Zotter et al.\[10\], which has the following form:

\[ H_{l,r}(\omega) = e^{j\tau \omega / \sin(\omega / \Delta \omega)} , \]  

where \( H_l \) and \( H_r \) express two channel filters for left and right ears in frequency domain, respectively. In this form, the phase of each filter oscillates in frequency domain with a period of \( \Delta \omega \) Hz, and the amplitude of oscillation is given by \( \tau \). The all-pass structure of Eq. (2) produces interchannel phase difference

\[ \phi_l - \phi_r = 2\tau \Delta \omega \sin(\omega / \Delta \omega) , \]  

which leads to the inter-channel time delay (ICTD) of

\[ ICTD(\omega) = \tau_l(\omega) - \tau_r(\omega) = \frac{\partial \phi_l}{\partial \omega} - \frac{\partial \phi_r}{\partial \omega} = 2\tau \sin(\omega / \Delta \omega) . \]  

Accordingly, the ICTD between the left and right channels alternates with respect to \( \omega \), and its magnitude is limited within \( \pm 2\tau \). This alternating ICTD between L/R channels results in the decrease of IACC and increase of the perceived source width\[10\].

However, there are some constraints to consider with the design of decorrelators. First, the coloration artifact can arise. The early reflections up to about 20 ms from the arrival of direct sound can cause the attenuation or amplification of the spectrum that result in the timbral change. The all-pass structure of Eq. (2) is thus desirable for suppressing the coloration artifact. Secondly, as pointed out by Zotter et al.\[10\], the localization shift can occur due to the change of ITD or ILD. Therefore, the oscillation of ITD or ILD should be rapid enough such that mean ITD or ILD change for each critical band can be minimized. In addition, the total length of the temporal response should be limited, because the response after echo threshold might be perceived as a separated event from the direct sound. Such a long decorrelator response is also not desirable from the view of computational complexity.

Under these constraints, we aim at producing the constant ICTD of Eq. (4) for all listener positions in the same distance from a radiator. For example, the listener 1 and 2 of Fig.1 should sense the same ICTD irrespective of the angular position of their head(\( \theta \)).

![FIGURE 1. Listeners at various locations from a radiator](image-url)
This objective can be accomplished by generating a sound field of constant interchannel phase difference at two ear positions. If such a target field can be designed, then we can utilize the sound field reproduction technique to reproduce the defined target field. A loudspeaker array controlled to reproduce the target field will provide the constant ICTD between two pressure signals measured at two ear positions \( \theta_{l,r} = \theta_0 \pm \Delta \theta \) irrespective of the angular position \( \theta_0 \).

Then, what kind of sound field can meet this objective? Since the ICTD between two different angular positions is nonzero, the target field should change in angular direction \( \theta \). To simplify the problem, suppose that the allowed listener position is limited to a plane at the same height(elevation angle \( \phi = \pi / 2 \)) as the radiator. Then, from the spherical harmonic expansion, any sound field from the radiator can be described as

\[
P_{l,r}(r \varphi, \theta, \omega) = \sum_{n=0}^{\infty} \sum_{m=-n}^{n} a_{nm} h^{(1)}_n(kr) P^m_n(\cos \varphi)e^{im\theta},
\]

where \( a_{nm} \) denotes the arbitrary constant representing the contribution of each modal function \( h^{(1)}_n(kr)P^m_n(\cos \varphi)e^{im\theta} \). At the far-field from the radiator \( kr >> 1 \), the pressure field can be approximated to

\[
P(r,0,\theta,\omega) \approx \frac{e^{i\omega \rho}}{kr} \sum_{n=0}^{\infty} \sum_{m=-n}^{n} (-i)^{n+1} A_{nm} P^m_n(0)e^{im\theta},
\]

\[
= \frac{e^{i\omega \rho}}{kr} \sum_{n=0}^{\infty} \left[ \sum_{m=-n}^{n} (-i)^{n+1} A_{nm} P^m_n(0) \right] e^{im\theta}
\]

\[
= \frac{e^{i\omega \rho}}{r} \sum_{n=0}^{\infty} A_n(\theta)e^{in\theta},
\]

and can be described in terms of the angle dependent radiation pattern \( H(\theta, \omega) \) and the distance dependency \( e^{i\omega \rho} / r \).

\[
P(r,0,\theta,\omega) \approx \frac{e^{i\omega \rho}}{r} H(\theta, \omega),
\]

**SOLUTION METHOD**

In order to have different phase at the left and right ear positions, the radiation pattern cannot be constant across \( \theta \), and hence, the harmonics \( e^{im\theta} \) with non-zero order \( m \) is necessary. Moreover, the frequency dependency of the ICTD should not be affected by the listener movement in angular direction. In view of all these, the intuitive modification of the all-pass structure of Eq. (2) may be written as

\[
H(\theta, \omega) = e^{i\omega \rho \sin(\alpha \theta_0 + \omega / \Delta \theta)},
\]

where \( \alpha \) is a tuning parameter controlling the degree of angular variation in \( H \). The radiation pattern at the listener’s two ear positions can be written as

\[
H_{l,r}(\omega) = H(\theta_0 \pm \Delta \theta / 2, \omega).
\]

The corresponding ICTD of Eq. (9) is given by

\[
ICTD(\Delta \theta, \omega) = i \tau_0 \left[ \cos(\alpha(\theta_0 + \Delta \theta) + \omega / \Delta \theta) - \cos(\alpha(\theta_0 - \Delta \theta) + \omega / \Delta \theta) \right]
\]

\[
= 2 \tau_0 \sin(\alpha \theta_0 + \omega / \Delta \theta) \sin(\alpha \Delta \theta)\sin(\omega / \Delta \theta).
\]

The ICTD of Eq. (10) shows the same sinusoidal oscillation in frequency domain as Eq. (4), and only its initial phase changes in accordance with the listener’s head location \( \theta_0 \). More importantly, the amplitude of the oscillation depends on the angular distance \( \Delta \theta \). This \( \Delta \theta \) dependency of ICTD amplitude well coincides with what happens in reality; the perceived source width increases as the listener moves closer to the finite-sized virtual source. Since the angular distance between the ear positions \( 2\Delta \theta \) will be large for the listener in the vicinity the radiator(listener 3 of Fig. 1) than for those who at far-field(listener 1 and 2), the amplitude of ICTD variation \( 2 \tau_0 \sin(\alpha \Delta \theta) \) should increase as the listener moves closer to the virtual source. However, this statement is only valid when \( \sin(\alpha \Delta \theta) \) of Eq. (10) monotonically increases with increasing \( \alpha \Delta \theta \). Therefore, we have the following constraint on \( \alpha : \alpha \Delta \theta \leq \pi / 2 \).
Since the angular distance $\Delta \theta$ can increase to $\pi / 2$ when the listener is at the location of the virtual source, the tuning parameter $\alpha$ has to be
\[ \alpha \leq 1, \]
for obtaining the monotonically increasing ICTD as the listener approaches to the virtual source. However, when the listener is at far-field from the virtual source, then the possible range of $\Delta \theta$ will be very small. In such a case, one can use larger $\alpha$ without restriction of Eq. (11).

In addition to this constraint, the target field has to be physically realizable, and hence, it should be the solution of the wave equation. To find the feasible value of $\alpha$, let us consider the Fourier Bessel expansion of Eq. (8):
\[ H(\theta, \omega) = e^{i\alpha r \sin(\theta - \omega t / 2)} \sum_{m=-\infty}^{\infty} \left[ J_n(\tau_0 \Delta f) e^{im \omega t / 2} \right] e^{im \omega \theta}. \]  
Equation (13) shows that the given target field is a sum of angular harmonics $e^{im \theta}$ with harmonic coefficients $A_m$ given by
\[ A_m(\omega) = J_n(\tau_0 \Delta f) e^{im \omega t / 2}. \]  
To satisfy the wave equation, the order of angular harmonics ($an$) should be an integer. Regarding the condition of Eq. (12), we can conclude that $\alpha = 1$ is the feasible integer value. Accordingly, the target function can be written as
\[ H(\theta, \omega) = e^{i\alpha r \sin(\theta - \omega t / 2)}, \]  
and the corresponding inter-channel phase difference (ICPD) is given by
\[ \text{ICPD}(2 \Delta \theta, \omega) = 2 \tau_0 \Delta f \cos(\theta_0 + \omega / \Delta f) \sin(\Delta \theta). \]  
Substituting Eq. (15) into (7) yields the frequency domain representation of the target sound field:
\[ P_r(r, \theta, \omega) = \frac{e^{2i\omega t}}{r} H(\theta, \omega) = \frac{e^{2i\omega t}}{r} e^{i\alpha r \sin(\theta - \omega t / 2)} \text{ for } \omega \geq 0. \]  
On the other hand, the temporal response of Eq. (13) with $\alpha = 1$ can be derived by taking inverse Fourier transform.
\[ \hat{h}(\theta, t) = \sum_{m=-\infty}^{\infty} J_n(\tau_0 \Delta f) e^{im \omega t / 2} \frac{1}{2\pi} \int_{-\infty}^{\infty} e^{im \omega t / 2} e^{-im \omega \theta} d\omega \]
\[ = \sum_{m=-\infty}^{\infty} J_n(\tau_0 \Delta f) e^{im \theta} \delta(t - m / \Delta f). \]  
It is noteworthy that the temporal response $\hat{h}(\theta, t)$ of Eq. (18) is a complex signal, and its frequency response does not satisfy the conjugate-even property ($H(\theta, -\omega) \neq H(\theta, \omega)^*$) of a real-valued pressure signal. The real-valued signal $h(\theta, t)$ that gives the identical frequency response to Eq. (13) for $\omega > 0$ has the following relation to its frequency domain expression $H(\theta, \omega)$:
\[ h(\theta, t) = 2 \text{Re} \left[ \mathcal{F}^{-1} \left[ U(\omega) H(\theta, \omega) \right] \right]. \]  
Here, $U(\omega)$ denotes the unit step function in frequency domain. The real operator, defined as $\text{Re}[y] = (y + y^*) / 2$, produces the negative frequency components $Y(\theta, -\omega) = Y(\theta, \omega)^*$ when applied to a complex time signal $y(\theta, t)$ of which frequency response is single-sided ($Y(\theta, \omega) = U(\omega) H(\theta, \omega)$).

In time domain, Eq. (19) can be written in terms of the temporal convolution of $u(t)$ and $\hat{h}(\theta, t)$:
\[ h(\theta, t) = 2 \text{Re} \left[ u(t) * \hat{h}(\theta, t) \right] \]
\[ = 2 \text{Re} \left[ \int_{-\infty}^{\infty} \left( \delta(t) - \frac{i}{\pi t} \right) \hat{h}(\theta, t) dt \right] \]
\[ = \text{Re} \left[ \left( \delta(t) + \mathcal{H} \left[ \delta(t) \right] \right) * \hat{h}(\theta, t) \right] \]
\[ = \left( \hat{h}(\theta, t) - \mathcal{H} \left[ \hat{h}(\theta, t) \right] \right). \]
where $\mathcal{H}$ represents the Hilbert transform operator, and $\hat{h}, \hat{h}_r$ denote real and imaginary parts of $\hat{h}(\theta, t) = \hat{h}_r + i\hat{h}_i$, respectively. Substituting Eq. (18) into Eq. (20) gives

$$h(\theta, t) = \sum_{m} J_m(\tau_{m}) \left[ \cos(m\theta) \delta(t) - \sin(m\theta) \mathcal{H}[\delta(t)] \right] * \delta(t - m / \Delta\tau) \tag{21}$$

If the audio signal to reproduce ($s(t)$) is fed into the transfer function, then the target pressure field at $(r, \theta)$ can be described as

$$p_r(r, \theta, t) = \frac{\delta(t - r / c)}{r} \left( h(\theta, t) * s(t) \right)$$

$$= \frac{\delta(t - r / c)}{r} \left( \sum_{m} J_m(\tau_{m}) \left[ \cos(m\theta)s(t) - \sin(m\theta)\mathcal{H}[s(t)] \right] * \delta(t - m / \Delta\tau) \right) \tag{22}$$

Therefore, the target field can be constructed in terms of multiple directional wavefronts each of which radiation pattern is given by $\cos(m\theta)$ or $\sin(m\theta)$. The amplitude of each wavefront is weighted by $J_m(\tau_{m})$ and the input signal to the virtual source is delayed by $m / \Delta\tau$ second. For a discretized input signal, the non-integer input delay ($mF / \Delta\tau$ samples) can be implemented by simple fractional delay filters. The delayed input signal is then directly fed to the wavefront with cosine radiation pattern ($\cos(m\theta)$), whereas the input to the sine radiation pattern ($\sin(m\theta)$) is the Hilbert transform of the original input signal. Figure 2 illustrates the procedure to generate the proposed target field.

Each signal processed for a specific order $m$ is then used as the input signal to the directional virtual source. In principle, any reproduction technique can be used at this stage for generating a sound field with the cosine or sine radiation pattern. For example, Warusfel et al.[15] and Corteel[16] formulated such driving function for wave field synthesis(WFS). When the virtual source is in between the loudspeaker and the listener, i.e., for a focused source, an accurate formula based on the multipole expansion[17] can be applied.

However, special care must be taken in realizing the Hilbert transform of Eq. (22) for discrete sequences, because the Hilbert transform filter of finite length can induce distortions in both magnitude and phase responses. In this work, however, we are interested in finding an accurate target field to reproduce, and hence, assume an idealistic response of the Hilbert transform filter that is given by the infinite length. For a discrete time sequence at $t = n\Delta\tau = 2\pi n / \omega_0$, the Hilbert transform can be written as[18]

$$\text{hil}[n] = \mathcal{H}[\delta(n)] = \begin{cases} 0 & \text{for } n: \text{even} \\ \frac{1 - \cos(n\pi)}{n\pi} & \text{for } n: \text{odd} \end{cases} = \sin\left(\frac{n\pi}{2}\right) \sin\left(\frac{n\pi}{2}\right) \tag{23}$$

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**FIGURE 2.** Overview of the structure for the generation of the decorrelated sound field.
Using Eq. (23) and the property of the delta function in discrete time domain, which is given by
\[ \delta[n] = \text{sinc}(n\pi) = \cos\left(\frac{n\pi}{2}\right) \text{sinc}\left(\frac{n\pi}{2}\right), \] (24)

Eq. (21) can be written as
\[ h(\theta)[n] = \sum_{m=-\infty}^{\infty} J_m(\tau_m \Delta f) \left\{ \cos\left(\frac{\pi}{2} (n - m\beta)\right) + \sin\left(\frac{\pi}{2} (n - m\beta)\right) \right\} \text{sinc}\left(\frac{\pi}{2} (n - m\beta)\right). \] (25)

Here, \( m\beta = mF_s / \Delta f \) denotes the amount of fractional time delay in samples, with \( F_s \) being the sampling frequency of the discrete time signal.

The target field of Eq. (25) is depicted along the angular position \( \theta \) (Fig. 3). For this example, finite number of circular harmonics \( M = 3 \) \( m = (-M, \cdots, M) \) are superposed, and the amplitude and period of the phase oscillation are set to \( \tau = 4 \) ms and \( \Delta f = 300 \) Hz. The sampling frequency was 22050Hz, thus the noninteger delay is proportional to \( \beta = 73.5 \) samples, approximately.

The impulse response presented in Fig. 3(a) has a maximum at \( t = 0 \), irrespective of the listener position. The amplitude of pre- and post- echoes produced by non-zero \( m \) s decrease with increasing \( m \), because the amplitude of each echo is proportional to the Bessel function of order \( m \). Therefore, the infinite summation of Eq. (25) with respect to the order \( m \) can be replaced with the finite summation. Even with the summation up to \( M = 3 \), the frequency response is almost flat in amplitude(Fig. 3(b)), which is beneficial to the suppression of coloration artifact. The phase(Fig. 3(c)) variation of the frequency response alternates in frequency \( f \) and angular direction \( \theta \) and is identical to what is intended(Eq. (17)).

![FIGURE 3](image)

FIGURE 3. Spatio-temporal responses of the proposed target field. (a) Impulse response (b) Magnitude (c) Phase of the frequency response.

**Relation to the Perceived Source Width**

We then need to verify if this target field truly leads to the extension of the perceived source width. The perceived source width can be quantified from the subjective test or by evaluating IACC of the dummy head measurement. For brevity, however, we evaluated the correlation between two free-field responses at the two ear locations. This free-field correlation can be regarded as the interchannel correlation that is acquired from two
microphones separated by the distance between two ears. Evidently, this does not reflect the scattering of the human head, and hence, may or may not be follow the true IACC. Nevertheless, in the low frequency region where the head scattering effect is negligible, the free-field correlation and IACC can be related to each other.

The free-field correlation at two microphone positions separated by \(2\Delta \theta\) can be derived as

\[
corr(2\Delta \theta,t) = h(\theta_0 + \Delta \theta,t) * h(\theta_0 - \Delta \theta,t)
\]

\[
= 2 \text{Re} \left[ \mathcal{F}^{-1} \left( U(\omega) \mathcal{H}(\theta_0 + \Delta \theta,\omega) \mathcal{H}(\theta_0 - \Delta \theta,\omega) \right) \right].
\]  

(26)

From Eq. (15), the multiplication of two transfer functions in frequency domain is given by

\[
Corr(2\Delta \theta,\omega) = H(\theta_0 + \Delta \theta,\omega)H(\theta_0 - \Delta \theta,\omega) = e^{i (2\pi \Delta \theta \sin(\Delta \theta)/\Delta \omega)}.
\]

(27)

In the same manner as Eqs. (20) and (21), the time domain correlation function can be rewritten as

\[
corr(2\Delta \theta,t) = 2 \text{Re} \left[ u(t) * \mathcal{F}^{-1} \left[ Corr(2\Delta \theta,\omega) \right] \right] = \sum_{m=-\infty}^{\infty} J_n(2\tau_0 \Delta f \sin(\Delta \theta)) \left\{ \cos(m(\theta_0 + \frac{\pi}{2}) \delta(t) - \sin(m(\theta_0 + \frac{\pi}{2})) H(\delta(t)) \right\} \delta(t - m / \Delta f)
\]

(28)

The discrete-time sequence of the correlation function also can be obtained using Eq. (25). That is,

\[
corr(2\Delta \theta)[n] = \sum_{m=-\infty}^{\infty} J_n(2\tau_0 \Delta f \sin(\Delta \theta)) \cos \left( m(\theta_0 + \frac{\pi}{2}) - \frac{\pi}{2} (n - m \beta) \right) \text{sinc} \left( \frac{\pi}{2} (n - m \beta) \right).
\]

(29)

Note that the correlation function of Eq. (26) is calculated over infinite time duration. This calculation is different from the convolution over finite time duration, which is common in calculating IACC (e.g., 0 ~ 80 ms for IACC_x). However, the decay rates of impulse responses \(h(\theta_0 + \Delta \theta,t), h(\theta_0 - \Delta \theta,t)\) are rapid enough, and Eq. (26) gives the similar result to the convolution over finite duration.

The correlation function of Eq. (26) evaluated for \(\tau_0 = 2\) ms, \(\Delta \theta = 300\) Hz and \(\Delta \theta = \pi / 6\) are shown in Fig. 4(a-c). The maximum value of the correlation within the interval \(t = [-1, 1]\) ms can be related to the perceived source width. In Fig. 4(d), it can be observed that the increasing amplitude of group delay oscillation (\(\tau_0\)) yields the decrease of maximum correlation value, which is expected to increase the perceived source width. Figure 5 depicts the variation of maximum correlation value with respect to \(\Delta \theta\) and \(\tau_0\). Since \(\Delta \theta\) depends on the location of the listener, \(\tau_0\) and \(\Delta \theta\) becomes the control parameter of the perceived source width. However, in order to obtain the accurate relation between the control parameter and source width, further investigation using the subject test or the evaluation of IACC is required.

![FIGURE 4](image-url)
SUMMARY

We propose a method for designing a target sound field that can provide the increased perceived source width when reproduced by the sound field reproduction technique. The proposed design is based on the two channel decorrelation technique that utilizes frequency-dependent ICTD panning. The conventional theory is extended so as to form a sound field consisting of high order cosine and sine patterns in angular direction. The proposed method shows that the correlation between two acoustic signals measured at ear positions can be decreased without perturbation on the spectrum magnitude.

REFERENCES