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A Sound telescope: a control of zone of interest

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ABSTRACT

For hands-free audio communication, a beamforming has been widely used to focus a broadband signal. Algorithms of conventional beamformers are based on focusing received signals from sensor array toward the location of a target single source. However, conventional beamformers cannot be directly applied when we desire to focus on multiple sound sources inside a zone of interest. The design issue of our interest is how to control the beamformer's beamwidth, from which we can change zones of interest (bright zone and quiet zone) and listen to any possible sound source located inside the bright zone with small amount of distortions. For the case of multiple sources, integration of array gains within desired directions of interests can be applied, and it is a direct approach of using a conventional beamformers. The results are the moving average of a beam power in controlled regions. To extend the ideas of conventional beamforming, a concept of regional focusing is suggested. The algorithm is based on the inverse approach of acoustic contrast control which has been used for designing a desired sound field by using loud speakers. The performance of two, direct approach and inverse approach, are compared. To simulate proposed algorithms in a practical situation, an array of microphones placed in circular positions is used, and the performance of proposed algorithms is tested. In the case of direct approach, the solution basically has the same problems of the conventional beamforming technique. In the case of inverse approach, remarkable reduction of any sound came from quiet zone is expected.

INTRODUCTION

To magnify a desired sound signal and reduce an unwanted noise, an array of microphones has been used to focusing a single target sound source. To develop an algorithm to realize, various beamforming techniques has been used. [1] In general, previous algorithms focus the sound wave on a single target. Therefore, conventional point focusing algorithms cannot be directly applied when the number of sources of interest is more than two, especially when the exact locations of them are unknown. If the locations of sources are known inside the zone of interest, beam output signal can be easily derived by summation of beamforming output for point focusing on each sound source. In a practical situation, the locations of sources can be estimated via visual information or via beamforming to search source locations (find dominant peak-locations through beamforming by changing a direction of interest). However, if the exact locations of sources are unknown, for example, no visual information is provided or interested sound sources are including not only loud sounds but also all small incidents inside zone of interest, then dominant peak locations can not represent the sounds come from small incidents. In this case we cannot directly apply conventional beamforming techniques.

To focus the multiple sources, many algorithms were suggested to control the beamwidth of beamformer. [2-8] However, most of them do not generally achieve frequency-independent directions. Moreover, they do not suggest how to successfully reduce sounds came from certain directions (quiet directions) and how to control such beamwidth for reducing sound.

In this paper, extended methods of beamforming are introduced. A goal of proposed methods is to magnify a desired sound field all inside a zone of interest (bright zone), and to reduce unwanted sounds propagated from the outside the zone (quiet zone). The performance of proposed methods is compared with conventional beamforming technique, delay-and-sum beamforming. [9]

PROBLEM DEFINITION

Suppose that M microphones are circularly placed around a zone of interest to focus on a 2 dimensional wave. (**Figure 1**)

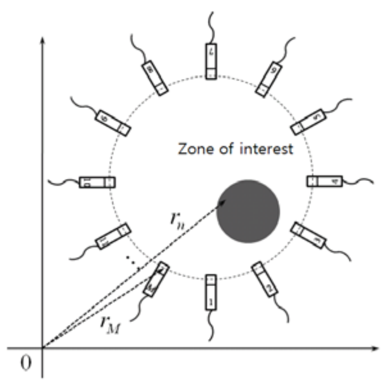

Figure 1. Definition of problem dealt in this paper. M microphones are circularly placed around a zone of interest. A gray circle indicates the zone of interest.

Further assume that the wave field of interest is in a free-field condition. Signals measured from m^th microphones can be defined as a summation of possible sounds propagated from the inside a zone of interest ($s(t)$) and other possible sounds propagated from the outside a zone of interest ($n(t)$),

$$x_m(t) = s_m(t) + n_m(t). \tag{1}$$

Assume that information of source characteristics, such as number of sound sources and physical size of sound source inside the zone, is unknown. The problem defined in this paper is how to extract and magnify an interested signal, $s(t)$, from measured signal, $x(t)$, by using microphone gains, h.

METHODS

Direct approach

To solve the abovementioned problem, an extend concept of point focusing of conventional beamformer can be a one possible solution. The solution of regional focusing is based on the assumption of unknown sound sources are distributed inside a zone of interest. Therefore, for direct approach, microphone gains for target position is all integrated in the considered area. For better understanding, let's assume that conventional delay-and-sum beamformer is introduced to focusing a target point, somewhere inside a zone of interest r_n as,

$$\hat{p}(\vec{r}_n, t) = h_1 x_1(t) + \cdots + h_M x_M(t), \tag{2}$$

where, $\hat{p}(r_n, t)$ is beamforming output by selecting microphone gain towards r_n position. For delay-and-sum beamforming, beamforming output signal can be calculated by time shifting of all output signals at a target point as,

$$\hat{p}(\vec{r}_n, t) = \sum_{m=1}^{M} l_m x_m(t - \Delta_{nm}), \tag{3}$$

where, Δ_{nm} is time delay to match each measured microphone signal to a desired point, and l_m is a magnitude gain. For a spherical wave, which is proper both in the near- and far-field conditions, acoustic pressure of a distal point source with unit magnitude can be wrote as,

$$p(\vec{r}, t) = \frac{1}{|\vec{r}|} e^{j(\vec{k}\cdot\vec{r} - \omega t)}, \tag{4}$$

where, r is distance between the location of point source and location of microphone position. Now, if we assume that an actual point source is located at r_n position, beamforming solution can be simplified by,

$$\begin{aligned}\hat{p}(\vec{r}_n, t) &= \sum_{m=1}^{M} l_m x_m(t - \Delta_{nm}) = \sum_{m=1}^{M} l_m x_m(t) e^{j\omega\Delta_{nm}} \\ &= \sum_{m=1}^{M} l_m x_m(t) e^{jk|\vec{r}_n - \vec{r}_m|}.\end{aligned} \tag{5}$$

If we match the magnitude of spherical wave, magnitude gain, l_m, simply is the distance between a microphone and a target position, r_n. Therefore eq. (5) will be simplified as,

$$\hat{p}(\vec{r}_n, t) = \sum_{m=1}^{M} |\vec{r}_n - \vec{r}_m| x_m(t) e^{jk|\vec{r}_n - \vec{r}_m|} = \sum_{m=1}^{M} \frac{1}{G(\vec{r}_m \mid \vec{r}_n)} x_m(t). \tag{6}$$

where, $G(r_m \mid r_n)$ is a free-field Green's function defined as follows,

$$G(\vec{r}_m \mid \vec{r}_n) = \frac{1}{|\vec{r}_n - \vec{r}_m|} e^{-jk|\vec{r}_n - \vec{r}_m|}. \tag{7}$$

To extend a conventional beamforming to a zone control, beamforming output signals are now integrated in a desired zone of interest.

$$\begin{aligned}\hat{y} &= \int_{V_b} W(V)\, \hat{p}(\vec{r}_n, t) dV \\ &= \int_{V_b} W(V) \sum_{m=1}^{M} \frac{1}{G(\vec{r}_m \mid \vec{r}_n)} x_m(t) dV \\ &= \sum_{m=1}^{M} \int_{V_b} W(V) \frac{1}{G(\vec{r}_m \mid \vec{r}_n)} x_m(t) dV \\ &= \sum_{m=1}^{M} x_m(t) \left(\int_{V_b} \frac{W(V)}{G(\vec{r}_m \mid \vec{r}_n)} dV \right),\end{aligned} \tag{8}$$

where, V_b is a volume of zone of interest, and W is a weighting function with respect to the location of zone of interest. For direct approach, the m^th microphone gain is simply,

$$\hat{h}_m = \int_{V_b} \frac{W(V)}{G(\vec{r}_m \mid \vec{r}_n)} dV. \tag{9}$$

By hearing an integrated beamforming output signals, $\hat{y}$, we can expect that any possible sound propagated from inside the volume, V_b, can be magnified. Moreover, by setting weighting function, W, we can relatively magnify or reduce a interesting regions inside a zone. When W is defined as a delta function, proposed algorithm is exactly equal to the conventional beamforming techniques.

Inverse approach of acoustic contrast control

Another solution of areal focusing can be found by closely looking at a solution of acoustic contrast control in the field of sound manipulation. [10] The goal of acoustic contrast control is find proper input gains of multi-channel speakers to make a designed sound field. As an inverse approach of acoustic contrast control, we can derive same problem to find a proper microphone gains to hear sounds came from inside the zone of interest. In the case of inverse approach, because we already know a condition of wave field in a free-field condition, we assume that we know the sound propagation of acoustic point source as eq. (4). If a unit point source is located at target position, a beamforming output signals can be rewritten as,

$$\hat{p}(\vec{r}_n, t) = h_1 x_1(t) + \cdots + h_M x_M(t). \tag{10}$$

Beamforming output signal can be divided by time and spatial parts as,

$$\hat{p}(\vec{r}_n, t) = \hat{p}(\vec{r}_n) e^{-j\omega t}. \tag{11}$$

where, $\hat{p}(\vec{r}_n)$ is defined by

$$\hat{p}(\vec{r}_n) = \begin{bmatrix} G(\vec{r}_1 | \vec{r}_n) & \cdots & G(\vec{r}_M | \vec{r}_n) \end{bmatrix} \begin{bmatrix} h_1 \\ \vdots \\ h_M \end{bmatrix} = \mathbf{G}(\vec{r}_m | \vec{r}_n) \mathbf{h}. \tag{12}$$

From the beamforming output signals for target point, we can define a power of output signal, e, inside a zone of interest and outside a zone of interest, respectively.

$$e_b = \frac{1}{V_b}\int_{V_b} \hat{p}(\vec{r}_n)^H \, \hat{p}(\vec{r}_n)\, dV = \mathbf{h}^H \left(\frac{1}{V_b}\int_{V_b} \mathbf{G}(\vec{r}_m|\vec{r}_n)^H \, \mathbf{G}(\vec{r}_m|\vec{r}_n)\, dV \right)\mathbf{h} \tag{13}$$

$$e_q = \frac{1}{V_q}\int_{V_q} \hat{p}(\vec{r}_n)^H \, \hat{p}(\vec{r}_n)\, dV = \mathbf{h}^H \left(\frac{1}{V_q}\int_{V_q} \mathbf{G}(\vec{r}_m|\vec{r}_n)^H \, \mathbf{G}(\vec{r}_m|\vec{r}_n)\, dV \right)\mathbf{h} \tag{14}$$

Now, we define a cost function J as a ratio of signal power inside and outside zone as,

$$J \equiv \frac{e_b}{e_q} = \frac{\mathbf{h}^H \mathbf{R_b} \mathbf{h}}{\mathbf{h}^H \mathbf{R_q} \mathbf{h}}, \tag{15}$$

where, $\mathbf{R} = \frac{1}{V}\int_V \mathbf{G}(\vec{r}_m|\vec{r}_n)^H \, \mathbf{G}(\vec{r}_m|\vec{r}_n)\, dV.$

The unknown parameter, microphone gain $(\mathbf{h})$, can be analytically calculated by maximizing eq. (15). The physical meaning of solving optimization problem is that, by choosing microphone gain, the resultant beamforming signal will maximize sound propagated from the inside a zone of interest and minimize sound propagated from the outside. The problem of maximizing J can be replaced, without loss of generality, the problem of maximizing μ as follows,

$$\therefore \mu = \mathbf{h}^H \mathbf{R_b} \mathbf{h} + J\left(I - \mathbf{h}^H \mathbf{R_q} \mathbf{h}\right) \tag{16}$$

The solution of eq. (16) is well known to find a first dominant eigen vector of matrix, $\mathbf{R_q}^{-1}\mathbf{R_b}$. [11, 12]

$$\therefore \left(\mathbf{R_q}^{-1}\mathbf{R_b}\right)\mathbf{h} = J\mathbf{h}. \tag{17}$$

COMPARISON OF TWO METHODS

Consider a microphone array with 180 identical omnidirectional elements and about 10.47 cm inter-element spacing (2 degrees angular difference) with the circular position. Assume that there are two zones of interest: bight zone and quiet zone. The radius of each zone is 1m, and the locations of them are different. (**Figure 2**) From the 180 channel microphone signals, sounds propagating inside the bright zone desired to be magnified and sounds come inside the quiet zone desired to be minimized though the exact locations of sound sources are unknown.

We can apply delay-and-sum beamforming technique in the each point inside the bright zone and integrate all beamforming outputs (direct approach). For direct approach, consideration of quiet zone is missing, and the performance of magnification is depends on inter-microphone spacing, and aperture size. The microphone gain is calculated from the eq. (9) based on the location and size of bright zone. We use 0.1m inter-spacing grid points, total 317 points, for bright zone.

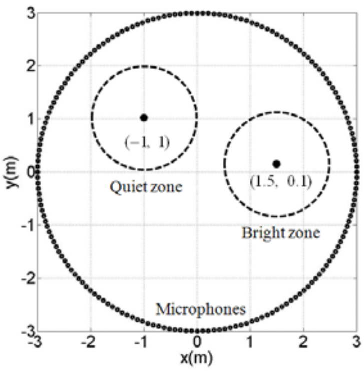

Figure 2. Schematic drawing of problem to solve. 180 identical microphones, indicated as small circles, are located about 10.47 cm inter-element spacing (2 degrees angular difference) with circular positions. Two zones of interest are depicted as dotted line: quiet zone (centered at $(-1,\ 1)$) and bright zone (centered at $(1.5,\ 0.1)$)

Beamforming outputs, with respect to the location of an actual acoustic point source, can be calculated from the eq. (8). The results are depicted from **Figure 3** to **Figure 5**. In this case, we choose a constant weighting function, $W(V_b)=1$, and all figures are normalized as beamforming output signal at $(1.5,\ 0.1)$ position is 0 dB.

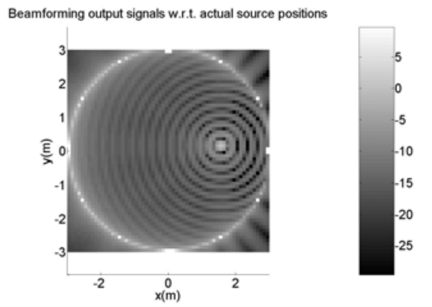

Figure 3. Beamforming output signals with respect to actual source positions (direct approach). Result represents magnitude of output signal with frequency of 600 Hz, and results are depicted in a dB scale. A weighting function set to be $W(V_b)=1$, and the result is normalized that $(1.5,\ 0.1)$ position is set to be 0 dB.

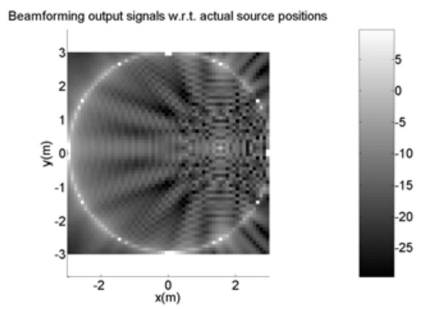

Figure 4. Result represents magnitude of output signal with frequency of 1000 Hz (direct approach). Detailed descriptions are noted in the caption of **Figure 3**.

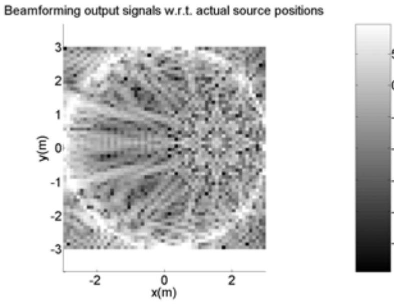

Figure 5. Result represents magnitude of output signal with frequency of 2500 Hz (direct approach). Detailed descriptions are noted in the caption of **Figure 3**.

To avoid aliasing, under about 1630 Hz signal is required for conventional delay-and-sum beamformer (half of wavelength). For **Figure 3** and **Figure 4**, satisfying in this condition, sounds propagated from the center of the bright zone can be maximally magnified, and the sounds came from the other space of the bright zone can be magnified or reduced in some cases because of the array patterns. This result is due to the moving averaging of beam outputs in the zone of interest. By taking moving averaging of beam outputs, some area can not be magnified because of side-lobe interferences. In high frequencies, over about 1630 Hz, no meaningful magnification is obtained because of moving averaging of beam outputs with aliasing and side-lobe interferences (**Figure 5**).

We can apply inverse approach for the problem defined in **Figure 2**. In this case, both bright and quiet zone are considered, and microphone gain can be calculated based on the eq. (17). The results are depicted from **Figure 6** to **Figure 8**.

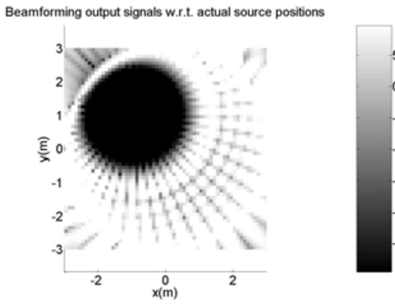

Figure 6. Beamforming output signals with respect to actual source positions (inverse approach). Result represents magnitude of output signal with frequency of 600 Hz, and results are depicted in a dB scale. The result is normalized that $(1.5,\ 0.1)$ position is set to be 0 dB.

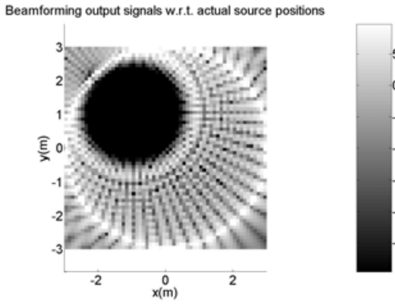

Figure 7. Result represents magnitude of output signal with frequency of 1000 Hz (inverse approach). Detailed descriptions are noted in the caption of **Figure 6**.

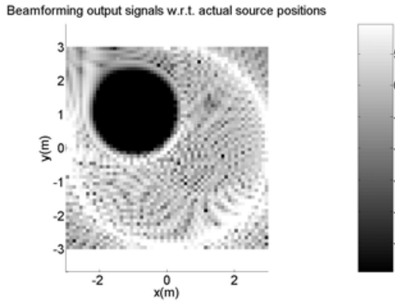

Figure 8. Result represents magnitude of output signal with frequency of 2500 Hz (inverse approach). Detailed descriptions are noted in the caption of **Figure 6**.

By comparing the results of direct approach, sounds came from the bright zone can not be magnified for inverse approach. However, powerful reduction of sound came from the quiet zone can be expected. Reduction of sound propagated inside the quiet zone is obtained even in the frequency over the aliased frequency, which is not reported for conventional beamformers.

CONCLUSIONS

To extend the concept of point focusing to areal focusing, performance of two different methods (direct approach and inverse approach) are compared. For direct approach, different microphone gains with respect to a target location inside a zone of interest are integrated, and the result is a moving average of all gains. Finally calculated microphone gains are strongly depends on the method of used conventional beamformer, so the solution basically has the same problems of the conventional beamforming technique. For example, the performance of magnification is seems to depend on what kind of beamformer is used for moving average. In this paper, conventional delay-and-sum beamformer was used, and magnification is depends on high-frequency cut-off. Over the high frequency cut-off, because aliasing is occurred for conventional beamformer, final microphone gain is blended and no meaningful magnification is expected. Even though the controlled frequency is under the high frequency cut-off, some area can not be magnified inside the zone of interest because of side-lobe interferences.

For inverse approach, microphone gain is calculated by optimization of maximizing relative sound power between bright zone and quiet zone. The solution is seems to minimize the possible sound propagated inside the quiet zone rather maximize the possible sound power inside the bright zone. The reason of this can be explained by looking at the cost function, eq. (15). If we find the microphone gains which set zero sound power inside the quiet zone, cost function will go to infinity. That is a solution to maximize the cost function. From the simulations, control of minimizing sound power inside quiet zone is successful. The proposed inverse approach can provide powerful solution to extract out the unwanted sound in quiet zone. However, regional magnification inside zone of interest can not be expected.

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