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JASA ARTICLE

An iterative approach to optimize loudspeaker placement for multi-zone sound field reproduction

Min Zhu[1,a)] and Sipei Zhao[2,b)]
[1]Tongda College of Nanjing University of Posts and Telecommunication, Nanjing, China
[2]Centre for Audio, Acoustics, and Vibration, Faculty of Engineering and IT, University of Technology Sydney, Sydney, Australia

ABSTRACT:

Various array patterns, such as circular, linear, and arc-shaped arrays, have been used in multi-zone sound field reproduction, but most of them are based on empirical rather than judicious selection. This article proposes an iterative optimization method to select the loudspeaker positions from a large set of candidate locations. Both the number and locations of the loudspeakers can be designed with superior performance. Both single-frequency and broadband simulations based on the acoustic contrast control method are performed to validate the proposed scheme, and the performance of the optimized array is compared with that of an arc-shaped array and that of an array optimized with an existing method. © *2021 Acoustical Society of America*. https://doi.org/10.1121/10.0005042

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I. INTRODUCTION

Multi-zone sound field reproduction, or personal audio system, has attracted significant research interest over the past decade due to its wide potential applications, such as personal audio devices (Chang *et al.*, 2009) and car cabins (Cheer *et al.*, 2013). Many control methods have been explored for multi-zone sound field reproduction, such as the acoustic contrast control (ACC) method (Choi and Kim, 2002), the least squares method (Kirkeby and Nelson, 1993), and a combination of them (Chang and Jacobsen, 2012).

The ACC method aims to maximize the acoustic potential energy ratio between the bright and dark zones and was first proposed by Choi and Kim (2002) with an X-shaped loudspeaker array and was later investigated for a circular array (Shin *et al.*, 2010), linear array (Zhao *et al.*, 2015), and arc-shaped array (Zhu *et al.*, 2017a). The least squares method was first used to reproduce multi-zone surround sound by Poletti (2008) based on a circular array, and its performance for the arc-shaped array (Zhu *et al.*, 2017b) and linear array has also been studied. The ACC and least squares methods have been combined to achieve a balance between the bright zone reproduction error and the inter-zone sound interference. Chang and Jacobsen (2012) combine the two methods with a simple weighting factor for a double-layer circular array, whereas Cai *et al.* (2014) regard the acoustic contrast (AC) as a constraint to the reproduction error for a linear array. Although various array patterns have been used for multi-zone sound field reproduction, all of the above-mentioned research is based on empirical rather than judicious considerations.

Recently, various methods have been explored to optimize the loudspeaker placement for sound field control (Koyama *et al.*, 2020). The first such method is the Lasso-based method, which is motivated by the compressive sampling theory (Lilis *et al.*, 2010). Other methods, such as singular value decomposition and constrained matching pursuit, have also been investigated for loudspeaker placement optimization, and it is found that the constrained matching pursuit method produces the least reproduction error (Khalilian *et al.*, 2016). The Lasso-based method was recently extended to multi-zone sound field reproduction in combination with the least squares method (Radmanesh *et al.*, 2016; Radmanesh and Burnett, 2013). However, the loudspeaker positions optimized in this approach depend heavily on the desired sound field, such as that from a few virtual sources (Radmanesh and Burnett, 2013).

In an alternative approach, this article proposes an iterative method to optimize loudspeaker placement for multi-zone sound field control based on the ACC method. A similar iterative method has been used to optimize the microphone array for acoustic beamforming design (Arcondoulis and Liu, 2019a, 2019b) but has not been explored for personal sound systems. The objective of the loudspeaker placement optimization is to select a desired number of loudspeaker positions from a large set of candidate locations. In this article, each loudspeaker at the given set of candidate locations is muted in sequence, the performance indices for the remaining loudspeakers are calculated, and then the loudspeaker position that shows the minimum performance index is removed to reduce the number of candidate locations. The process is repeated iteratively until the designed number of positions are selected out of the original set of candidate locations. The performance of the loudspeaker array designed with the proposed iterative approach is compared with that of an arc-shaped

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[a)] Also at: Telecommunication and Information Engineering College, Nanjing University of Posts and Telecommunication, Nanjing, China.

[b)] Electronic mail: sipei.zhao@uts.edu.au, ORCID: 0000-0002-6261-8864.

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array and an array optimized with the Gram–Schmidt orthogonalization (GSO) method (Asano *et al.*, 1999) in terms of AC and AE.

II. METHOD

A. ACC

A multi-zone sound field reproduction system with a circular and an arc-shaped loudspeaker array is illustrated in Figs. 1(a) and 1(b), respectively. The aim of the ACC method is to maximize the ratio of the averaged acoustic potential energy density between the bright and dark zones (Choi and Kim, 2002); hence, the cost function can be written as Eq. (1) following the indirect Lagrangian formulation (Elliott *et al.*, 2012)

$$J(\mathbf{q}) = \mathbf{q}^{\mathrm{H}}\mathbf{G}_{\mathrm{D}}{}^{\mathrm{H}}\mathbf{G}_{\mathrm{D}}\mathbf{q} + \lambda_1\left(\mathbf{q}^{\mathrm{H}}\mathbf{G}_{\mathrm{B}}{}^{\mathrm{H}}\mathbf{G}_{\mathrm{B}}\mathbf{q} - B\right) + \lambda_2\left(\mathbf{q}^{\mathrm{H}}\mathbf{q} - E\right), \quad (1)$$

where the superscript H denotes the Hermitian transpose, $\mathbf{q} = [q_1, q_2, \ldots, q_L]^T$ denotes the source weights of the L loudspeakers to be optimized, $\mathbf{G}_{\mathrm{B}}$ ($\mathbf{G}_{\mathrm{D}}$) denotes the $M \times L$ transfer matrix from the L loudspeakers to the M control points in the bright (dark) zone, λ_1 and λ_2 are the Lagrange multipliers, and B and E are the constraints on the sound pressure level (SPL) in the bright zone and the AE, respectively.

The optimal solution to minimize the cost function in Eq. (1) is (Elliott *et al.*, 2012)

$$\mathbf{q}_{\mathrm{opt}} \propto \Phi\left[(\mathbf{R}_{\mathrm{D}} + \lambda_2\mathbf{I})^{-1}\mathbf{R}_{\mathrm{B}}\right], \quad (2)$$

where $\Phi[*]$ denotes the eigenvector corresponding to the maximum eigenvalue of a matrix, $\mathbf{R}_{\mathrm{B}} = \mathbf{G}_{\mathrm{B}}{}^{\mathrm{H}}\mathbf{G}_{\mathrm{B}}$, $\mathbf{R}_{\mathrm{D}} = \mathbf{G}_{\mathrm{D}}{}^{\mathrm{H}}\mathbf{G}_{\mathrm{D}}$, $\mathbf{I}$ denotes the identity matrix with the same dimension as $\mathbf{R}_{\mathrm{D}}$, and the superscript -1 denotes the matrix inversion operation. In the simulations, the regularization parameter λ_2 is carefully chosen to meet the constraint on the AE (Coleman *et al.*, 2014). The detailed procedure is as follows: (1) initialize λ_2 in Eq. (2) as $10^{-10}D_{\max}$ ($D_{\max}$ is the maximum eigenvalue of $\mathbf{R}_{\mathrm{D}}$) so that the condition number of the matrix ($\mathbf{R}_{\mathrm{D}} + \lambda_2\mathbf{I}$) is less than 10^{10}; (2) calculate $\mathbf{q}_{\mathrm{opt}}$ from Eq. (2) and scale $\mathbf{q}_{\mathrm{opt}}$ to ensure the constraint on the SPL in the bright zone is met, i.e., $\mathbf{q}^{\mathrm{H}}\mathbf{G}_{\mathrm{B}}{}^{\mathrm{H}}\mathbf{G}_{\mathrm{B}}\mathbf{q} = B$; (3) if $\mathbf{q}^{\mathrm{H}}\mathbf{q} \le E$, output $\mathbf{q}_{\mathrm{opt}}$ as the solution; otherwise, increase the value of λ_2 and repeat steps (2) and (3) until both the constraints on the SPL and the AE are met.

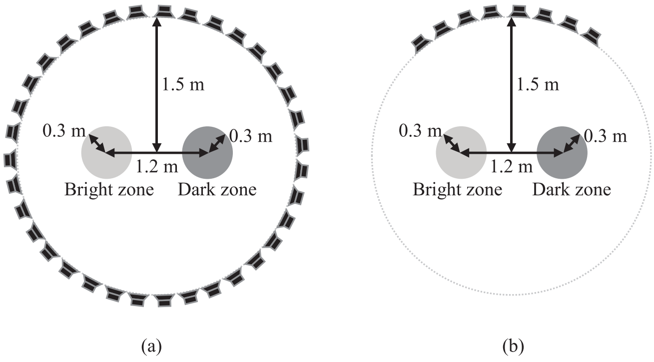


FIG. 1. (Color online) Diagram of the circular loudspeaker array with the bright and dark zones.

B. Iterative loudspeaker placement optimization

The circular array in Fig. 1(a) can usually obtain superior performance over the linear and arc-shaped arrays in multi-zone sound field reproduction in terms of AC because more loudspeakers are used. However, a large number of loudspeakers are needed to form the circular array, many of which may generate low sound pressure and have little contribution to the final reproduced sound field. By contrast, the arc-shaped array [Fig. 1(b)] has been employed for multi-zone sound field reproduction with fewer loudspeakers (Zhu *et al.*, 2017b), but the arc-shaped array is empirical, and its performance is not optimized. Therefore, an optimization strategy is needed to place the lesser number of loudspeakers judiciously without dramatic degradation in performance.

The goal of the iterative approach is to find the optimal loudspeaker placement from a set of candidate locations to achieve as good performance as possible with a smaller number of loudspeakers. The idea of the iterative method is simple and intuitive. In the given set of L candidate loudspeaker locations, each loudspeaker is muted in sequence, and the optimal source weights and the performance index for the remaining ($L - 1$) loudspeakers are calculated. The loudspeaker that has the lowest impact on performance is removed from the configuration in each iteration. The process is repeated iteratively until the desired number of loudspeaker positions are selected out of the original set of candidate locations.

The performance index for comparison is critical and will determine the final results. For the ACC method described in Sec. II A, two evaluation metrics are usually used to assess its performance, i.e., AC and array effort (AE), which are defined in Eqs. (3) and (4), respectively. A higher AC means less inter-zone interference, while the higher the AE, the poorer the acoustical efficiency of the system,

$$\mathrm{AC} = 10\log_{10}\left(\frac{\mathbf{q}^H\mathbf{R}_{\mathrm{B}}\mathbf{q}}{\mathbf{q}^H\mathbf{R}_{\mathrm{D}}\mathbf{q}}\right), \quad (3)$$

$$\mathrm{AE} = 10\log_{10}\left(\frac{\mathbf{q}^{\mathrm{H}}\mathbf{q}}{|q_{\mathrm{r}}|^2}\right), \quad (4)$$

where q_{r} is the source strength of the single reference source that generates the same sound energy in the bright zone as the array, which can be calculated as $|q_{\mathrm{r}}|^2 = B/(\mathbf{G}_{\mathrm{r}}{}^{\mathrm{H}}\mathbf{G}_{\mathrm{r}})$, where $\mathbf{G}_{\mathrm{r}}$ is the $M_{\mathrm{B}} \times 1$ transfer vector from the reference source to the M_{B} control points in the bright zone. In the simulations, the first sound source in the original circular array is chosen as the reference source, so $\mathbf{G}_{\mathrm{r}}$ is the first column of $\mathbf{G}_\mathrm{B}$, i.e., $\mathbf{G}_\mathrm{r} = \mathbf{G}_\mathrm{B}(:, 1)$. It should be noted that both AC in Eq. (3) and AE in Eq. (4) are defined at a single frequency.

An ideal multi-zone sound field reproduction system should possess a high AC with a low AE. Therefore, a single performance index P_f is defined as the AC per unit AE, which can be written as the difference between the AC and AE in log scale, i.e.,

$$P_f = \mathrm{AC} - \mathrm{AE}. \tag{5}$$

The performance index P_f can be physically interpreted as AC per unit AE. To facilitate more flexibility for the optimization, different weighting coefficients can be applied to AC and AE to emphasize the relative importance. This will be investigated in the future. While P_f can be used to optimize the loudspeaker placement for each frequency, in practical applications, broadband control of sound field is required; thus, the average performance index P_avg is used for broadband optimization of the loudspeaker placement, i.e.,

$$P_\mathrm{avg} = \frac{1}{K}\sum_{k=1}^{K} P_{f_k}, \tag{6}$$

where K is the number of frequency bins. In each iteration, the loudspeaker that produces the maximum performance index (P_f for single-frequency and P_avg for broadband) is removed from the configuration to reduce the number of candidate loudspeaker locations. The complete description of the iterative optimization algorithm is summarized in Table I.

III. SIMULATIONS AND DISCUSSIONS

To validate the efficacy of the proposed method, simulations are performed to reduce the number of loudspeakers from a circular array with 60 loudspeakers uniformly distributed along a circle with a radius of 1.5 m [Fig. 1(a)]. In the simulations, the bright and dark zones are separated 1.2 m apart, both with a radius of 0.3 m, as depicted in Fig. 1(a). The control and evaluation points are both uniformly distributed in the bright and dark zones with an interval distance of 5 and 3 cm, respectively.

The performance of the array optimized by the proposed iterative approach is compared to that of the common arc-shaped array [Fig. 1(b)] and the array optimized by the method based on the GSO (Asano *et al.*, 1999) with the same number of loudspeakers. Most of the methods mentioned in Sec. I are developed for the least squares method, which requires the desired sound field and hence cannot be readily applied to the ACC method used in this article. By contrast, the GSO method determines a certain number of loudspeakers with the most linearly independent transfer impedance vectors, which can be applied to the ACC method here. The detailed procedure for the GSO method can be found in Asano *et al.* (1999) and is not shown here for the sake of brevity. It should be noted that the GSO method was stated to have poor performance for loudspeaker placement in pressure matching-based sound field reproduction because it does not take the desired sound field into account when selecting the loudspeaker locations (Khalilian *et al.*, 2016). However, this conclusion may not hold for the ACC method investigated in this article because no desired sound field is involved in the ACC method.

In each iteration of both the proposed iterative approach and the GSO method, simulations are performed in two stages. The first is the design stage to calculate the optimal sound source weights of the loudspeaker array, while the second is the reproduction stage to calculate the sound field and the performance index with the optimal sound source weights from the design stage.

In the design stage, the loudspeakers are assumed to be monopole point sources in a free field environment, i.e., the transfer function from the lth loudspeaker to the mth control microphone is $G_{m,l} = -\mathrm{j}\rho\omega \mathrm{e}^{\mathrm{j}kR_{m,l}}/4\pi R_{m,l}$, where j is the imaginary unit, ρ is the air density, ω is the angular frequency, $k = \omega/c$ (c is the speed of sound) is the wavenumber, and $R_{m,l}$ is the distance between the lth loudspeaker and the mth control microphone. In the simulations, the averaged SPL in the bright zone is constrained to 76 dB, which was found to be a comfortable level for subjective listening evaluation of sound zone interference (Francombe *et al.*, 2012). The SPL constraint is applied by setting the value of B in the cost functions in Eq. (1) according to $L_p = 10\ \log_{10}(B/M_\mathrm{B}/p_\mathrm{r}^2) = 76\ \mathrm{dB}$ (M_B is the number of control points in the bright zone) with the reference sound pressure $p_\mathrm{r} = 20\ \mu\mathrm{Pa}$. In the simulations, the AE is limited to be smaller than 0 dB, i.e., $10\ \log_{10}(\mathbf{q}^\mathrm{H}\mathbf{q}/|q_\mathrm{r}|^2) \le 10\ \log_{10}(E/|q_\mathrm{r}|^2) = 0$, hence $E = |q_\mathrm{r}|^2$.

TABLE I. Iterative optimization algorithm: select N out of L loudspeakers.

Step 1. For frequency f, mute the lth loudspeaker from the L loudspeakers and calculate the optimal source weights $\mathbf{q}_\mathrm{opt}(l)$ for the remaining $(L-1)$ loudspeakers based on Eq. (2);

Step 2. Calculate the performance index $P_f(l)$ by substituting $\mathbf{q}_\mathrm{opt}(l)$ obtained from step 1 to Eqs. (3)–(5);

Step 3. For broadband optimization, repeat steps 1 and 2 to calculate $P_f(l)$ for all the frequency bins and calculation the average performance index $P_\mathrm{avg}(l)$ based on Eq. (6);

Step 4. For single-frequency optimization, repeat steps 1 and 2 for $l = 1, 2, \ldots, L$, and find the l_0th configuration with the maximum performance index, i.e., $l_0 = \arg\max[P_f(l)]$;

For broadband optimization, repeat steps 1–3 for $l = 1, 2, \ldots, L$, and find the l_0th configuration with the maximum average performance index, i.e., $l_0 = \arg\max[P_\mathrm{avg}(l)]$;

Step 5. Remove the l_0th loudspeaker to select the remaining $(L-1)$ loudspeakers from the original L loudspeakers;

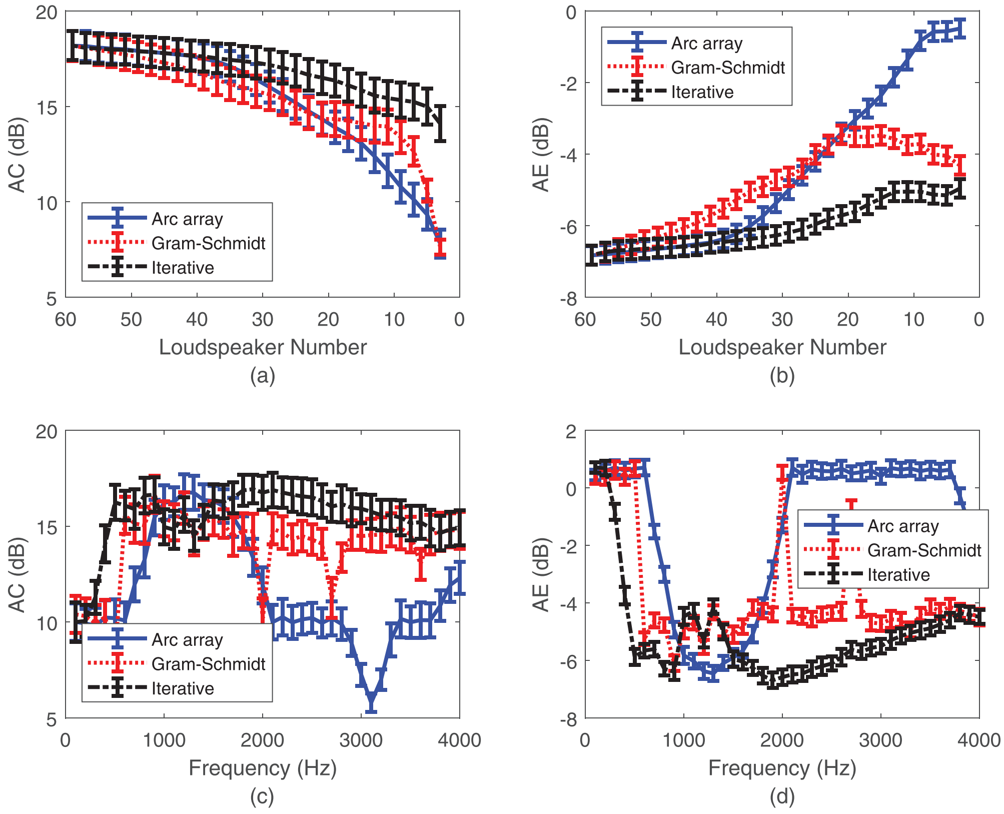
Step 6. Repeat steps 1–5 to remove more loudspeakers until the desired number of N loudspeakers are selected from the original L loudspeakers.

In the reproduction stage, perturbations in the transfer functions that are inevitable in practical applications are taken into account, which are assumed to be additive errors, i.e., $\widehat{G}_{m,l} = G_{m,l} + a_{m,l}e^{j\phi_{m,l}}$ with $a_{m,l}$ and $\phi_{m,l}$ being the amplitude and phase of the spatial errors, respectively. In the simulations, the error amplitude $a_{m,l}$ is assumed to have a Gaussian distribution with a mean value of zero and a variance of 1.41, while the error phase is uniformly distributed between $-10°$ and $10°$. Both $a_{m,l}$ and $\phi_{m,l}$ are drawn from Monte-Carlo simulations with 100 trials for each simulation.

A. Single-frequency performance

For the single-frequency simulations, a certain number of loudspeakers are selected from the candidate sets for each frequency based on the performance index P_f in Eq. (5), and the specific loudspeaker locations might be different for different frequencies. The simulation results are shown in Fig. 2, where the vertical bars indicate the standard deviation. Figures 2(a) and 2(b) present the AC and AE averaged over the frequency band from 100 to 4000 Hz varying with the number of loudspeakers, which show that the AC decreases while the AE increases with the reduction of the number of loudspeakers. Although the GSO method does not improve or even degrades the performance when more than 25 loudspeakers are selected, its performance outperforms the arc array when the number of loudspeakers is less than 25. In contrast, the proposed iterative approach shows better performance than both the arc array and the GSO method when fewer than 30 loudspeakers are selected from the original 60 candidate loudspeakers.

To further investigate the performance of the proposed method at different frequencies, the AC and AE as a function of frequency are depicted in Figs. 2(c) and 2(d) when 11 loudspeakers are used. The placement of the arc-shaped array is illustrated in Fig. 1(b), and 11 loudspeakers span 60° with the interval angle of 6°. It can be observed that the arc array shows good performance in the frequency range between 800 and 1600 Hz but deteriorates at other frequencies where the AE reaches the maximum limit 0 dB [Fig. 2(d)]. The GSO method improves the AC and reduces the AE above 2000 Hz compared to the arc array. The proposed iterative approach further improves the performance in the whole frequency range except that between 1000 and 1500 Hz, where performance is slightly worse than that of the arc-shaped array. This may be because the algorithms have fallen into a local minimum and cannot achieve satisfactory performance at these frequencies. Theoretically, the arc-shaped array should be a special case of the solution sets, and the algorithms should stop at the arc-shaped array if a better solution cannot be found. However, the problem is non-convex, and the iterative algorithms are not ergodic, so it may not even reach the special case of the arc-shaped array, not to mention the globally optimal solution, if it is stuck in a local minimum. It is noteworthy that the AE in Fig. 2(d) is slightly larger than the upper limit value of 0 dB at some frequencies. This is because perturbations were



(a) (b) (c) (d)

FIG. 2. (Color online) The average (a) AC and (b) AE as a function of the number of loudspeakers; the (c) AC and (d) AE as a function of frequency when 11 loudspeakers are selected based on single-frequency optimization.

TABLE II. Comparison of the average AC and AE between the optimized arrays and the circular and arc-shaped arrays when 11 loudspeakers are selected out of 60.

Loudspeaker number	$L=60$	$N=11$		
Performance	Circular array	Arc array	GSO method	Proposed iterative approach
Single frequency				
AC (dB)	18.1	11.6	13.9	15.4
AE (dB)	−6.8	−1.3	−3.7	−5.1
Broadband				
AC (dB)	18.1	11.8	11.8	14.5
AE (dB)	−6.8	−1.4	−3.3	−4.1

added in the transfer functions in the reproduction stage but not in the design stage, as described above.

The overall performance of the array optimized with the proposed iterative approach is better than that of the arc array and the GSO method optimized array. This can be clearly observed from Table II, where the average AC and AE over the whole frequency range are summarized in the third and fourth rows. When 11 loudspeakers are used, the average AC and AE are 11.6 and −1.3 dB, respectively, for the arc array, which are improved to 13.9 and −3.7 dB by the GSO method. With the proposed iterative approach, the AC is further improved to 15.4 dB, and the AE is reduced to −5.1 dB. The above results demonstrate that the proposed iterative approach outperforms both the GSO method and the empirical arc array when the placement is optimized at each frequency.

B. Broadband performance

To further investigate the performance of the proposed approach and its feasibility for broadband applications, the array placement is optimized based on the average performance index P_{avg} in Eq. (6). The broadband simulation results are shown in Fig. 3, where Figs. 3(a) and 3(b) depict the average AC and AE with the number of loudspeakers and Figs. 3(c) and 3(d) present the AC and AE as a function of frequency when 11 loudspeakers are selected. Figures 3(a) and 3(b) show that the performance of arc array is the same for broadband and single-frequency control, as expected, and the performance of the proposed iterative approach for broadband optimization is similar to that for the single-frequency scenario, demonstrating its robustness and effectiveness for practical applications. In contrast, the GSO method for broadband control degrades significantly compared to the single-frequency case, and its overall performance is worse than that of the arc array, especially when the selected number of loudspeakers is between 20 and 40.

When 11 loudspeakers are selected, Figs. 3(c) and 3(d) show that the proposed iterative is superior over both the arc array and the GSO method in terms of AC and AE when the frequency is higher than 1500 Hz. By comparing the broadband results in Figs. 3(c) and 2(c), it is noted that the

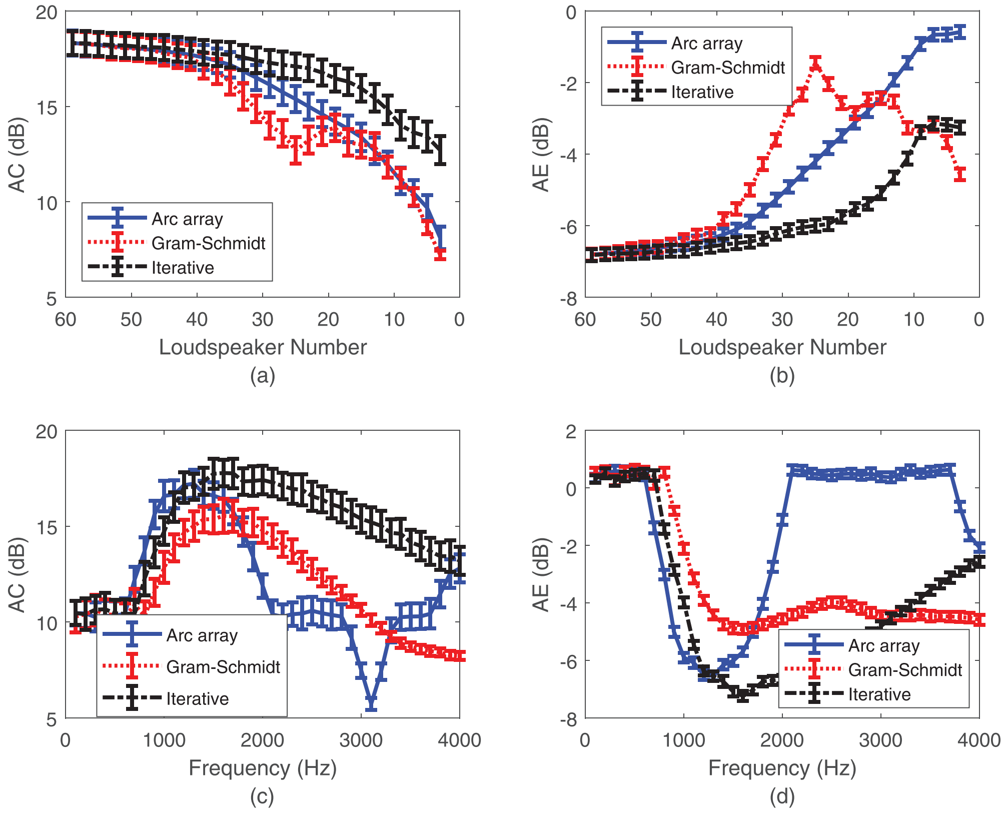


FIG. 3. (Color online) The average (a) AC and (b) AE as a function of the number of loudspeakers; the (c) AC and (d) AE as a function of frequency when 11 loudspeakers are selected based on broadband optimization.

broadband results are better than the single-frequency results at around 1300 kHz. This may be because the single-frequency optimization is stuck at a local minimum as discussed above. In the broadband optimization, the performance index is averaged over the whole frequency range, which reduces the probability of being stuck in a local minimum because it is less likely for the optimization procedure to fall into a local minimum at all of the frequencies at the same time.

The overall AC and AE averaged over the whole frequency band are compared in the last two rows of Table II, which show that compared to the single-frequency scenario, the proposed iterative approach achieves a slightly lower AC for broadband optimization with a slightly higher AE. By contrast, the performance of the GSO method for broadband control degrades compared to the single-frequency case. This is because when the performance index is averaged over a broad frequency band, the transfer impedance vectors selected by the GSO method may not be independent at some frequencies (Asano *et al.*, 1999).

The spatial distribution of the SPL is illustrated in Fig. 4 for 11 loudspeakers, where blue crosses denote the loudspeaker locations and the green solid and red dashed circles indicate the bright and dark zones, respectively. It can be seen that the proposed iterative approach achieves a higher SPL in the bright zone and a lower SPL in the dark zone, leading to a higher AC. In addition, the proposed iterative approach possesses the lowest average SPL outside the control zones, indicating a lower AE. These observations are consistent with the results in Figs. 3(a) and 3(b) and Table II.

The above results for both single-frequency and broadband signals demonstrate the superiority of the array designed with the proposed iterative approach for loudspeaker placement over the conventional arc-shaped array and that optimized with the GSO method. The shortcoming of the proposed approach is its relatively higher computation load than the GSO method due to the iterative scheme, especially when a small number of loudspeakers are selected from a large candidate set. However, this should not be a concern because the loudspeaker displacement needs to be designed only once beforehand.

It should be noted that both the proposed approach and the GSO method have no prerequisite for the geometry of the candidate locations (e.g., symmetric or not) because its optimization procedure does not rely on the array geometry.

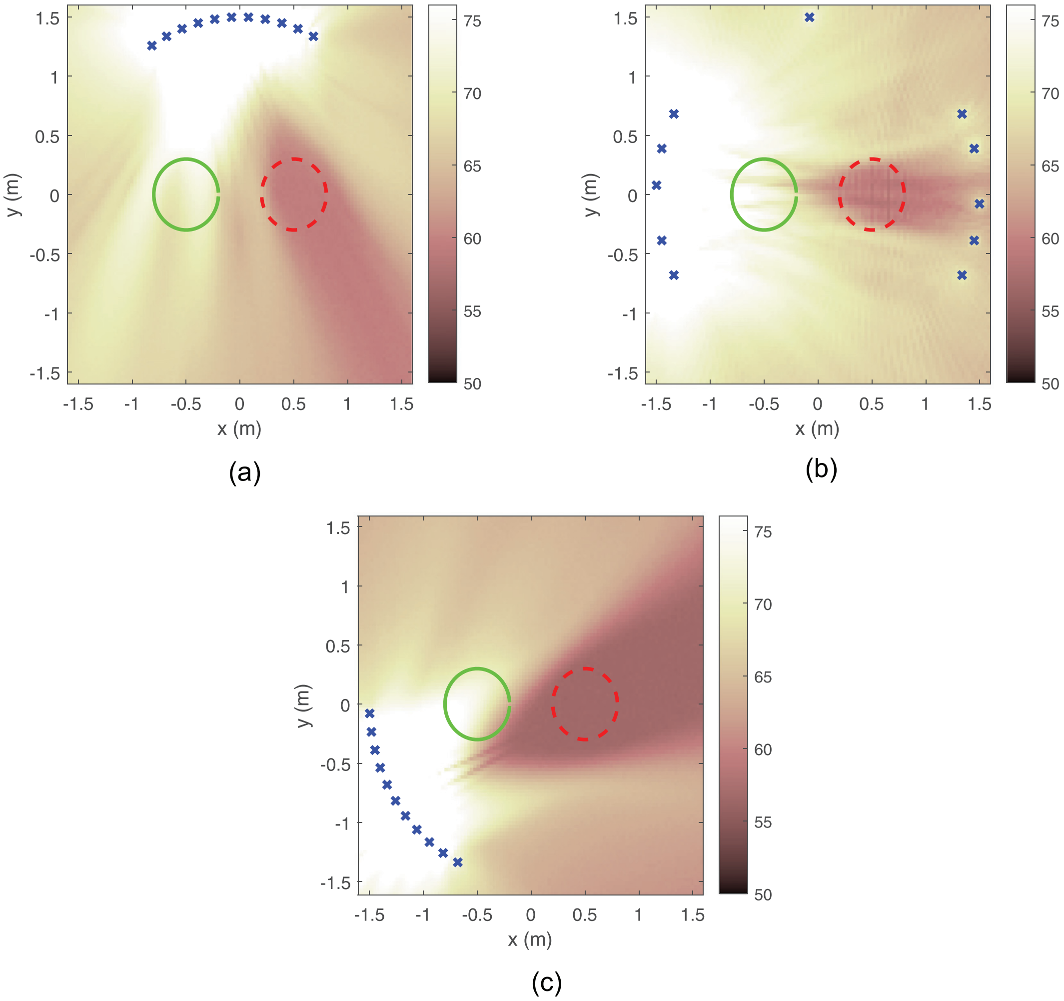


FIG. 4. (Color online) Spatial distribution of SPL for (a) arc array, (b) array optimized with the GSO method, and (c) array optimized with the proposed iterative approach when 11 loudspeakers are selected.

However, the optimized loudspeaker positions depend on the locations of the bright and dark zones, i.e., the optimized loudspeaker positions are expected to change with the locations of the bright and dark zones, although this does not affect the efficacy of the proposed approach. For some specific setup of the candidate locations and sound zones, the algorithm may fall into a local minimum at some frequencies and cannot achieve satisfactory performance, as discussed above. This needs to be analyzed case by case.

IV. CONCLUSIONS

This article investigates the loudspeaker placement optimization for multi-zone sound field control and proposes an iterative scheme to select a desired number of loudspeaker positions from a given set of candidate locations. In each iteration, one loudspeaker is muted in sequence, and the remaining loudspeakers are used to calculate the performance index, i.e., the difference between the AC and the AE. The loudspeaker that has the minimum effect on the performance index is removed from the candidate locations in each iteration. This process repeats iteratively until the desired number of positions are selected. Both single-frequency and broadband simulations demonstrate that the optimized array designed with the proposed iterative method is superior to the conventional arc-shaped array and the array optimized with the GSO method with the same number of loudspeakers in terms of AC and AE. Future work will carry out experiments to validate the proposed approach.

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