Active Room Compensation for 2.5D Sound Field Reproduction

Yitong Chen[✉] and Wen Zhang

Center of Intelligent Acoustics and Immersive Communications,
Northwestern Polytechnical University, Xi'an 710072, China
wen.zhang@nwpu.edu.cn
https://www.researchgate.net/profile/Wen_Zhang20

Abstract. Modelling secondary sources as 3D point sources to reproduce 2D desired sound field is named as 2.5D sound field reproduction, which has the intrinsic dimensionality mismatch problem. Existing methods for 2.5D reproduction have focused on solving the dimensionality mismatch problem and mostly considered free-field condition. However, in most cases, the reverberation caused by the listening room will degrade the reproduction performance. In this work, we propose an active room compensation strategy for 2.5D reproduction. Firstly, adopt sectorial mode matching algorithm to achieve 2.5D reproduction, the desired sound field and generated sound field are matched at the reproduction center. Secondly, the modal-domain algorithm is developed to estimate reverberant sound field and design the compensation signals of loudspeakers to compensate reverberant sound field. The proposed method is validated through simulation experiments to demonstrate its effectiveness against room reverberations.

Keywords: Sound field reproduction · 2.5D reproduction · Room compensation · Modal domain

1 Introduction

As psychoacoustics theory puts the fact that human ears are more sensitive to sounds from horizontal direction at the height of ears, it is more valuable to reproduce desired sound field on two-dimension field. Early studies regard it as 2D reproduction problem where secondary sources are modelled as vertical line sources [1]. However, due to the acoustic characteristics of the loudspeaker, modelling it as 3D point source is more reasonable in practice. Due to the intrinsic dimensionality mismatch, modelling secondary sources as 3D point sources to reproduce 2D sound field is named as 2.5D reproduction [2–5]. Most existing methods for 2.5D reproduction can make good performance in addressing dimensionality mismatch problem [2,6]. However, these approaches focus on free-field condition while the reverberation caused by time-varying room responses will impair the generated sound field [1]. Thus, room compensation (or room equalization) has been proposed to reduce or eliminate the reverberant effects.

As for room compensation, current proposed methods can be divided into two ways: passive compensation manners and active manners. Passive compensation methods reduce wall reflections by acoustic absorption materials. Nevertheless, it gets costly and impractical especially at low frequencies in many real-world application scenarios. Thus, researchers put forward compensation methods by active manners. In order to equalize the effects of reverberation, adding appropriate compensation signals on the loudspeaker arrays is the core. Currently, most proposed room compensation techniques are based on Multiple Input Multiple Output (MIMO) system [7,8], which can only reduce the effect of reverberation at discrete points and its adjacent region. Designing compensation signals using modal domain processing [9] achieves compensation within a continuous region. However, this has only been considered for 2D reproduction.

In this paper, we propose an active room compensation approach in 2.5D reproduction through modal domain processing. Section 2 reviews the 2.5D reproduction using sectorial mode matching algorithm, and we propose an active room compensation algorithm in Sect. 3. In Sect. 4, the proposed algorithm is simulated and evaluated by reproduction of narrowband and broadband signals under reverberant environment.

2   2.5D Reproduction

2.1   Problem Formulation

In modal-domain, we express the 2D desired sound field at any point $\boldsymbol{x} = \{r_x, \phi_x\}$ through the interior solution of wave equation [10]

$$P_d(\boldsymbol{x}, k) \approx \sum_{m=-M}^{M} \alpha_m(k) J_m(kr_x) e^{im\phi_x}, \tag{1}$$

where k represents the wave number, $\alpha_m(k)$ represents sound field coefficients, $J_m(\cdot)$ represents cylindrical Bessel function. The truncation order is determined as $M = \lceil ekR/2 \rceil$ [11] where R is the radius of control region.

Consider adopting circular loudspeaker array to generate the desired sound field exactly as (1) shows, where the array is implemented on horizontal plane, the number of loudspeakers has to satisfy $L \geq 2M + 1$ [12].

Thus, reproduced sound field in free-field is formulated as

$$P(\boldsymbol{x}, k) = \sum_{l=1}^{L} d_l(k) H_l(\boldsymbol{x}, k), \tag{2}$$

where $H_l(\boldsymbol{x}, k)$ is acoustic transfer function (ATF) of the lth loudspeaker and the observation spot x, $d_l(k)$ is the driving signal of lth loudspeaker.

The expression of $H_l(\boldsymbol{x}, k)$ is as follows

$$H_l(\boldsymbol{x}, k) = \frac{e^{-ik\|\boldsymbol{y}_l - \boldsymbol{x}\|}}{4\pi\|\boldsymbol{y}_l - \boldsymbol{x}\|}, \tag{3}$$

where $\boldsymbol{y}_l = \{r_l, \phi_l\}$ represents the loudspeaker location and $\| \cdot \|$ denotes the Euclidean distance of the vectors.

In modal domain, $H_l(\boldsymbol{x}, k)$ is formulated as

$$H_l(\boldsymbol{x}, k) \approx \sum_{m=-M}^{M} \sum_{n=|m|}^{M} \gamma_n^m(l, k) j_n(kr_x) Y_n^m(\frac{\pi}{2}, \phi_x), \tag{4}$$

where $\gamma_n^m(l, k)$ is the ATF coefficient of lth source. In free field condition,

$$\gamma_n^m(l, k) = -ikh_n^{(2)}(kr_l)\overline{Y_n^m(\frac{\pi}{2}, \phi_l)},$$

Substituting (4) into (2) gives the expression

$$P(\boldsymbol{x}, k) \approx \sum_{m=-M}^{M} \sum_{l=1}^{L} d_l(k) \sum_{n=|m|}^{M} \gamma_n^m(l, k) j_n(kr_x) Y_n^m e^{im\phi_x}, \tag{5}$$

where the spherical harmonic function at elevation $\theta = \pi/2$ is defined as $Y_n^m(\frac{\pi}{2}, \phi_x) = Y_n^m e^{im\phi_x}$, with $Y_n^m \triangleq A_n^m P_n^m(0)$, $A_n^m = \sqrt{\frac{2n+1}{4\pi} \frac{(n-|m|)!}{(n+|m|)!}}$ and $P_n^m(\cdot)$ represents associated Legendre function.

The loudspeaker driving signals are designed by equating (1) with (5). Using the orthogonality property of complex exponentials, it gives the following equation

$$\alpha_m(k) J_m(kr_x) = \sum_{l=1}^{L} d_l(k) \underbrace{\sum_{n=|m|}^{M} \gamma_n^m(l, k) j_n(kr_x) Y_n^m}_{h_m(l,k,r_x)}, \tag{6}$$

for $m = -M, ..., M$. From (6), it shows that the expansion is only over mode m and the Bessel function $J_m(kr_x)$ denotes the radial propagation in 2D sound field, and the modal expansion is over both n and m and the spherical Bessel function $j_n(kr_x)$ denotes the radial propagation in 3D sound field, which reflects the dimensionality mismatching. The solution of this problem is finding appropriate matching distance r_x.

2.2   Sectorial Mode Matching

It has been proved [2] that adopting the center of reproduction region as the matching spot $(r_x = 0)$ can reach a relatively less distortion due to dimensionality mismatch. Note that when $kr_x \to 0$, the summation over order n in (4) reduces to the single term $n = |m|$ [2,13] where $Y_{|m|}^m(\frac{\pi}{2}, \phi_x)$ is termed as the sectorial harmonics, and $h_m(l, k, r_x)$ in (6) is expressed as

$$h_m(l, k, r_x) = \gamma_{|m|}^m(l, k) j_{|m|}(kr_x) Y_{|m|}^m, \tag{7}$$

Thus, the driving signals $d_l(k)$ are derived by matrix

$$\mathbf{d} = \mathbf{H}^\dagger \mathbf{b}, \tag{8}$$

where

$$\mathbf{d} = \begin{bmatrix} d_{l_1}(k) \cdots d_{l_L}(k) \end{bmatrix}^T,$$

$$\mathbf{H}^\dagger = \begin{bmatrix} h_{-M}(l_1, k, r_x) & \dots & h_{-M}(l_L, k, r_x) \\ \vdots & \ddots & \vdots \\ h_M(l_1, k, r_x) & \dots & h_M(l_L, k, r_x) \end{bmatrix}^\dagger,$$

$$\mathbf{b} = \begin{bmatrix} \alpha_{-M}(k) J_{-M}(kr_x) \cdots \alpha_M(k) J_M(kr_x) \end{bmatrix}^T.$$

The dependence on wavenumber k is omitted for notation simplicity. Then, driving signals can be expressed simply as

$$\mathbf{d} = (\mathbf{H^d})^\dagger \mathbf{b}, \tag{9}$$

where $\mathbf{H^d}$ denotes the direct-path loudspeaker array ATF in modal domain.

3   Active Room Compensation

The previous section reviews sectorial mode matching algorithm for 2.5D reproduction. This section introduces the active room compensation algorithm for 2.5D reproduction in modal domain.

We use a circular microphone array of Q microphones to encircle the control region with the microphones uniformly placed. Figure 1 shows the system setup.
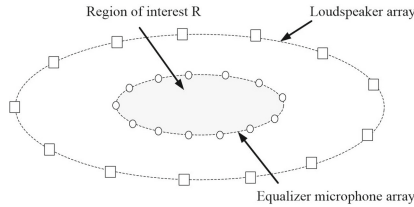

Fig. 1. Active room compensation system setup: the loudspeaker array and a plane control region circled by equalizer microphone array.

Note that the aim here is to compensate the reverberation caused by the loudspeaker signals. Based on Eq. (7) and given the fact that only the sectorial modes are controlled, derive the sound field coefficients measured by equalizer microphone array

$$\beta_m(k) = \frac{1}{Q j_{|m|}(kr_M) Y_{|m|}^m} \sum_{q=1}^{Q} P(\boldsymbol{x}_q, k) e^{-im\phi_q}, \qquad m = -M, \dots, M, \tag{10}$$

where r_M denotes the microphone array radius. In vector form, the measured sound field coefficients are constituted by direct-path modes and reverberant-path modes

$$\boldsymbol{\beta} = \boldsymbol{\beta}^{\mathbf{d}} + \boldsymbol{\beta}^{\mathbf{r}}, \tag{11}$$

The modal domain sound field coefficients can also be expressed as the loudspeaker driving signals and the corresponding ATF coefficients

$$\boldsymbol{\beta} = \boldsymbol{\Gamma}\mathbf{d} = \boldsymbol{\Gamma}^{\mathbf{d}}\mathbf{d} + \boldsymbol{\Gamma}^{\mathbf{r}}\mathbf{d}, \tag{12}$$

where $\boldsymbol{\Gamma} = [\boldsymbol{\gamma}_1, \dots, \boldsymbol{\gamma}_L]$, and $\boldsymbol{\gamma}_l = [\gamma_{|M|}^{-M}(l,k), \dots, \gamma_{|M|}^{M}(l,k)]^T$ which represent the ATF coefficients matrix. The ATF coefficients matrix is also decomposed into the direct path and reverberant path, that is $\boldsymbol{\Gamma} = \boldsymbol{\Gamma}^{\mathbf{d}} + \boldsymbol{\Gamma}^{\mathbf{r}}$.

Then, the compensation problem is formulated as introducing the compensation signals $\boldsymbol{\delta}\mathbf{d}$ at the loudspeakers to minimize the reverberant-path modes, i.e.,

$$\begin{aligned}
\min \|\boldsymbol{\beta}^{\mathbf{r}}\| &= \|\boldsymbol{\beta} - \boldsymbol{\beta}^{\mathbf{d}}\| \\
&= \|\boldsymbol{\Gamma}(\mathbf{d} + \boldsymbol{\delta}\mathbf{d}) - \boldsymbol{\Gamma}^{\mathbf{d}}\mathbf{d}\| \\
&= \|(\boldsymbol{\Gamma}^{\mathbf{d}} + \boldsymbol{\Gamma}^{\mathbf{r}})\boldsymbol{\delta}\mathbf{d} + \boldsymbol{\Gamma}^{\mathbf{r}}\mathbf{d}\|
\end{aligned} \tag{13}$$

Thus, by solving (13) in the least-squares approach, the compensation signals at the loudspeakers for eliminating reverberation is given by

$$\boldsymbol{\delta}\mathbf{d} = -(\boldsymbol{\Gamma}^{\mathbf{d}} + \boldsymbol{\Gamma}^{\mathbf{r}})^{\dagger}(\boldsymbol{\beta} - \boldsymbol{\Gamma}^{\mathbf{d}}\mathbf{d}), \tag{14}$$

where the measured sound field coefficients $\boldsymbol{\beta}$, the loudspeaker driving signals $\mathbf{d}$ for direct path (or free-field) propagation $\boldsymbol{\Gamma}^{\mathbf{d}}$ can be accessed straightforwardly. The reverberant path ATF coefficients are obtained in an adaptive manner as addressed in [9].

4 Evaluation

This section describes the simulation based experimental setup and performance of the proposed room compensation algorithm.

4.1 Simulation Results

Consider a 2D reverberant room of size $2\,\mathrm{m} \times 2\,\mathrm{m}$. Simulate reverberant environment by image source method [14]: the wall reflection is 0.8 (the ceiling and floor are perfectly-absorbing) and the image depth of 7. Control region circled by equalizer microphone array is centred at $(1\,\mathrm{m}, 1\,\mathrm{m})$ with the radius of $0.25\,\mathrm{m}$. The loudspeaker array is uniformly placed on the circle of $1\,\mathrm{m}$ radius. Note that the number of loudspeakers and microphones are both 39, which satisfies the mode truncation requirement $M = \lceil ekR/2 \rceil + 1$ [13].

We simulate reproduction of two kinds of narrowband sources, i.e., plane wave and cylindrical wave, and a broadband source. For the narrowband cases, the plane wave source is of 3 kHz frequency, incident from $\phi_v = \pi/3$ and the cylindrical wave source is of 3 kHz frequency, locating at $r_v = 1.5$ m, $\phi_v = \pi/3$ away from the system center. For the broadband cases, the cylindrical wave source is of 3 kHz bandwidth: frequencies 100 Hz to 3 kHz, and the location is as same as narrowband cylindrical wave source. Figure 2 and Fig. 3 show the compensation performance of narrowband sources, where the desired, reverberant and compensated sound fields are displayed in (a), (b), (c) respectively. Compared with the reverberant sound fields in Fig. 2(b) and Fig. 3(b), compensation is achieved within the whole reproduction region.
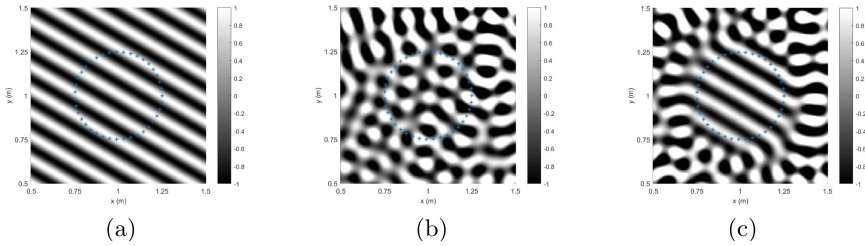

(a)   (b)   (c)

Fig. 2. Reproduction of a plane wave (frequency at 3 kHz, incident from $\phi_v = \pi/3$) in a 0.25 m control region circled by the asterisk

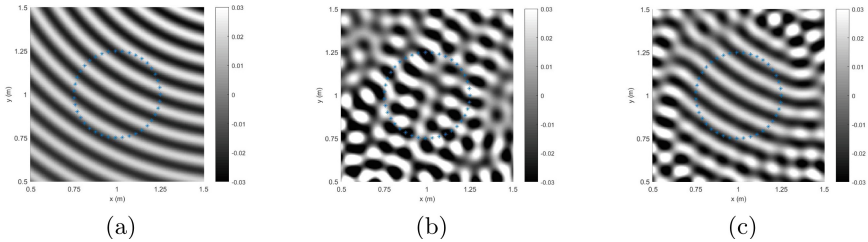(a)   (b)   (c)

Fig. 3. Reproduction of a cylindrical wave (frequency at 3 kHz, locating at $r_v = 1.5$ m, $\phi_v = \pi/3$ away from the system center) in a 0.25 m control region circled by the asterisk

4.2   Evaluation of Results

The compensation performance is measured by the normalized equalization error ε over the whole control region. The normalized equalization error is defined as follows,

$$\varepsilon(w) = 10 \log_{10} \frac{\int_R |P(\boldsymbol{x}, k) - P_d(\boldsymbol{x}, k)|^2 d\boldsymbol{x}}{\int_R |P_d(\boldsymbol{x}, k)|^2 d\boldsymbol{x}} \tag{15}$$

In this simulation, 40000 observation points are uniformly selected within the reproduction region to approximate the integral.

In broadband case, set the number of loudspeakers as 45, which is slightly more than the number of loudspeakers designed at the maximum frequency 3 kHz, other simulation settings are the same as in the examples of Fig. 2 and Fig. 3. In Fig. 4, it demonstrates the normalized equalization error over 3 kHz bandwidth, which shows that setting the number of the loudspeakers more than the minimum requirement of certain frequency can achieve satisfying results. As the frequency increases, the performance will gradually degrade.
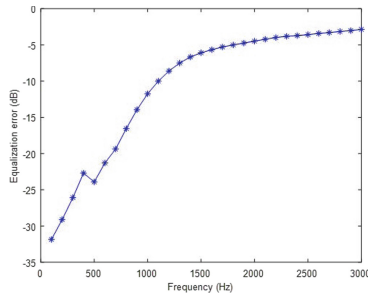

Fig. 4. The normalized equalization error of cylindrical wave reproduction over 3 kHz bandwidth frequency range.

5   Conclusion

In this paper, we propose an active compensation algorithm for 2.5D sound field reproduction. Firstly, the sectorial mode matching method is used to derive the loudspeaker driving signals assuming direct-path propagation between the loudspeaker array and reproduction region. Then, based on sectorial mode, an active compensation algorithm is introduced to compensate room reverberation. From simulation results, it can proved that the proposed algorithm can achieve effective room compensation over the whole reproduction region.

References

1. Betlehem, T., Abhayapala, T.D.: Theory and design of sound field reproduction in reverberant rooms. J. Acoust. Soc. Am. **117**(4), 2100–2111 (2005). https://doi.org/10.1121/1.1863032
2. Zhang, W., Abhayapala, T.D.: 2.5D sound field reproduction in higher order Ambisonics. In: 14th IEEE International Workshop on Acoustic Signal Enhancement, pp. 342–346. IEEE Press, Juan-les-Pins (2014). https://doi.org/10.1109/iwaenc.2014.6954315

3. Zhang, W., Zhang, J., Abhayapala, T.D., Zhang, L.: 2.5D multizone reproduction using weighted mode matching. In: 2018 IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 476–480. IEEE Press, Calgary (2018). https://doi.org/10.1109/ICASSP.2018.8462511

4. Okamoto, T.: 2.5D higher order ambisonics for a sound field described by angular spectrum coefficients. In: 2016 IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 326–330. IEEE Press, Shanghai (2016). https://doi.org/10.1109/ICASSP.2016.7471690

5. Okamoto, T.: Horizontal 3D sound field recording and 2.5D synthesis with omni-directional circular arrays. In: 2019 IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 960–964. IEEE Press, Brighton (2019). https://doi.org/10.1109/ICASSP.2019.8683009

6. Wang, W., Jia, M., Bao, C., Zhang, J.: 2.5D interior/exterior sound field reproduction and its extension to narrowband speech signals. In: 2016 IEEE International Conference on Audio, Language and Image Processing, pp. 7–12. IEEE Press, Shanghai (2016). https://doi.org/10.1109/ICALIP.2016.7846548

7. Brännmark, L.: Robust audio precompensation with probabilistic modeling of transfer function variability. In: 2009 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, pp. 193–196. IEEE Press, New Paltz (2009). https://doi.org/10.1109/ASPAA.2009.5346533

8. Brännmark, L., Bahne, A., Ahlén, A.: Compensation of loudspeaker-room responses in a robust MIMO control framework. IEEE-ACM Trans. Audio Speech Lang. **21**(6), 1201–1216 (2013). https://doi.org/10.1109/TASL.2013.2245650

9. Talagala, D.S., Zhang, W., Abhayapala, T.D.: Efficient multi-channel adaptive room compensation for spatial soundfield reproduction using a modal decomposition. IEEE-ACM Trans. Audio Speech Lang. **22**(10), 1522–1532 (2014). https://doi.org/10.1109/TASLP.2014.2339195

10. Williams, E.G.: Fourier Acoustics: Sound Radiation and Nearfield Acoustical Holography. Academic, New York (1999)

11. Kennedy, R.A., Sadeghi, P., Abhayapala, T.D., Jones, H.M.: Intrinsic limits of dimensionality and richness in random multipath fields. IEEE Trans. Signal Process. **55**(6), 2542–2556 (2007). https://doi.org/10.1109/TSP.2007.893738

12. Ward, D.B., Abhayapala, T.D.: Reproduction of a plane-wave sound field using an array of loudspeakers. IEEE-ACM Trans. Audio Speech Lang. **9**(6), 697–707 (2001). https://doi.org/10.1109/89.943347

13. Poletti, M.A., Betlehem, T., Abhayapala, T.D.: Analysis of 2D sound reproduction with fixed-directivity loudspeakers. In: 2012 IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 377–380. IEEE Press, Kyoto (2012). https://doi.org/10.1109/ICASSP.2012.6287895

14. Allen, J.B., Berkley, D.A.: Image method for efficiently simulating small-room acoustics. J. Acoust. Soc. Am. **65**(4), 943–950 (1979). https://doi.org/10.1121/1.2003643