**Nearfield acoustic holography based on the equivalent source method and reproducing kernels**

13A1

1. The topic of the paper is of limited interest to ICASSP.

* The NAH problem is closely related to inverse filtering problems for audio arrays signal processing. To our knowledge, this inverse reconstruction problem has never been addressed in the literature from the perspective of kernel regression.

1. The regularization proposed for the NAH problem in the paper is common.

* Regularization methods are not the main focus of this paper. Instead, to perform optimization in a hybrid discrete vector space and a continuous function space is really what we aim for. Since a Hilbert space is an inner product space, the NAH formulation is more tractable with a least squares approach. This combined inversion and interpolation approach results in a smoother reconstructed source image.

1. There is no reason that the distribution of sources should follow the model as the Bessel function.

* We agree that the source distribution is not necessarily a spherical Bessel function, which is a solution of the acoustic wave equation. However, we are only using its localized property. Other localized functions, such as a Gaussian kernel, should also be applicable. We will add this comparison when we get a chance to revise.

1. The method should be tested on real data.

* We used simulated data is because the plate vibration has analytical solutions for us validate our proposed NAH method. Testing on real data is currently underway.

3B05

1. The absent of noise in the signals is a fundamental shortcoming.

* We will add microphone noise or position errors to simulation when we get a chance to revise.

42B3

1. Welcome to provide an intuition on that all methods tend to improve with the increase in frequency.

* This could be a result misinterpretation of the reviewer. In fact, the reconstruction error increases with frequency as shown in Fig. 6.

1. Can the authors comment on the fitness of isotropic kernels in anisotropic fields?

* Isotropic kernels are not applicable to anisotropic fields. However, the sound field and the thin plate considered herein are isotropic.

**Semi-Blind estimation of acoustic transfer functions with application to dereverberation using convolutive transfer functions**

4309

1. This paper has no noticeable original contribution presented.

* This paper has two original contributions that distinguish it from the existing ATF identification. First, we develop an iterative optimal filtering scheme that is completely different from the widely used adaptive eigenvalue decomposition method. Second, we extend the proposed method to address long ATFs based on the Convolutive Transfer Function (CTF). By Wiener filtering or recursive least squares, the RIRs can be efficiently obtained without knowing the source signal.

1. The details of getting the reference signal via delay and sum beamformer are omitted. The references are from 2010 or older.

* The references have been updated to include the latest baseline methods. The details of obtaining the reference signal via the DAS beamformer will be added to the paper when it is accepted.

430A

1. The proposed method is not fully blind.

* We agree with the reviewer that our method is not "fully" blind, since the source position must be known or located in advance. Therefore, we will change the title to "Semi-Blind..." if this paper is accepted.

1. The number of CTF coefficients L is not mentioned in the paper.

* We will add this information if this paper is accepted.

1. Several BSI methods for RIRs have been proposed in the literature, none of which has been cited or compared with.

* We will compare the BSI method suggested by the referee with our approach in the revised paper. From our experience, these BSI techniques suffer from an equalization issue that could degrade the subsequent dereverberation performance. Another problem with these algorithms is that they are effective only for the cases with T60 less than 0.1 s. In contrast, our proposed approach, which is based on the CTF signal model, is capable of handling cases with T60 up to 1.6 s.

070E

1. The latest paper cited for the dereverberation is from 2019.

* The current state-of-the-art dereverberation method for the conventional signal processing methods remains Weighted prediction error (WPE), with its extended work published in 2021 as follows: R. Ikeshita, K. Kinoshita, N. Kamo and T. Nakatani, "Online Speech Dereverberation Using Mixture of Multichannel Linear Prediction Models," in IEEE Signal Processing Letters, vol. 28, pp. 1580-1584, 2021, doi: 10.1109/LSP.2021.3099715, and we will update it to reference.