XVIII IMEKO WORLD CONGRESS
Metrology for a Sustainable Development
September, 17 – 22, 2006, Rio de Janeiro, Brazil

VERIFICATION OF THE DOUBLE LAYER HOLOGRAPHIC ARRAY FOR EXTRACTION OF SOUND FIELDS IN REVERBERANT CONDITIONS

Zdenek Havranek [1], *Ludvik Bejcek*[2]

[1]Department of Control and Instrumentation, Brno University of Technology, Brno, Czech Republic, havranekz@phd.feec.vutbr.cz
[2]Department of Control and Instrumentation, Brno University of Technology, Brno, Czech Republic, bejcek@feec.vutbr.cz

Abstract: This paper describes methods and presents verification results of improvement of algorithms for visualization of sound fields based on near-field acoustical holography (NAH) by the double layer holografic array (DLA) measurement and iterative technique to remove disturbing sound fields which are not in the examined area under measurement planes (holograms).

Keywords: acoustical holography, sound characterization, vibration analyses.

1. INTRODUCTION

The near-field acoustical holography (NAH) technique was developed in 1980s [1] and there has been many improvements by this time. Planar NAH has many disadvantages in comparison with other methods which have been developed, but its strength is in simplicity and high calculation speed. The prerequisite for single layer NAH is that there is no other sound source behind the hologram. It means that there is only one sound source producing the sound into free-field in half-space.

DLA measurement with iterative technique can go beyond this limit. Better reconstruction accuracy with iterative algorithm has already been verified in single layer configuration [3], but it has never been adapted for NAH calculation with double layer microphone array.

2. PURPOSE OF THE PAPER

The main objective of this paper is to verify the expected strenght of using DLA measurement for reducing the sound pressure prediction error near examined vibrating surface in reverberant conditions. The reduction of the total reconstruction error for particle velocity and active sound intensity is also important. Original DLA calculation, which has already been presented in [2], hasn't been tested for more complicated sound field, which can be produced by vibrating thin plate. This paper also shows the results of calculation of sound field near sources with DLA technique and reconstruction accuracy improvement by recursive Wiener filtering algorithm.

In this paper, two models of reverberant conditions are used. Both models try to produce quite complicated sound field, thus simulate realistic measurement conditions in real life, because there are some other sources or reflective surfaces in most areas, where the sound source localization and characterization is carried out. The first one is an existence of another sound source on the opposite side of the first sound source surface (behind hologram planes), which disturbs the measured sound field in the holograms' area. In the figure 1 there are four layers, where the nearest one represents the sound pressure field very near the examined vibrating surface, the second and the third one are measurement layers (holograms – matrix microphone arrays) and the last one is the pressure field near the disturbing sound source.

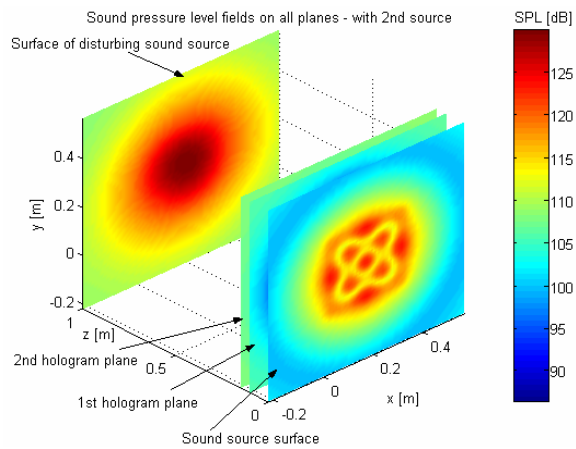

Fig. 1. The simulation setup for DLA with disturbing sound source

The examined vibrating surface is represented by a thin plate in infinite rigid baffle driven by harmonic force with defined frequency, strength and acting point. The similar model is used for the simulation of disturbing sound source. In this case, the reflection from the second sound source is not taken in account, due to the simplification of the calculation of the sound field between the sources. From this point of view, the second sound source is modeled as the set of small point sources assembled in matrix, with no reflecting surface. This case was for some simulations reduced to only one point source situated in the centre of the virtual source surface.

The second reverberant condition model is defined as a space with reflecting surface (plane) normal to the vibrating surface in an appropriate distance. The model of this example is made by the mirror image of the original vibrating surface with the double distance from the original. The arrangement of this case is in the figure 2.

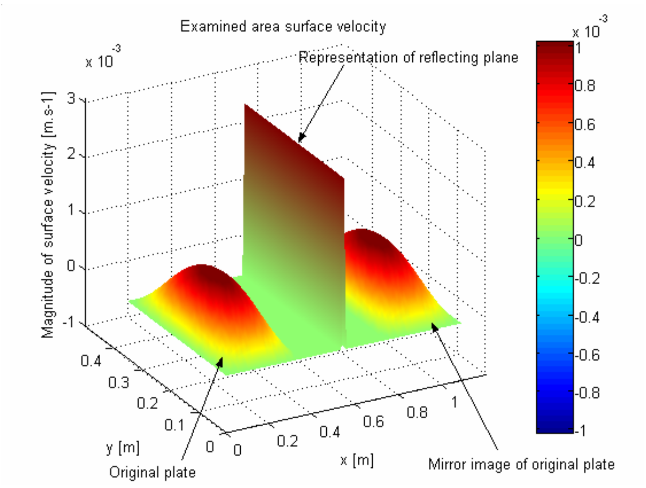

Fig. 2. The simulation setup for DLA with reflecting wall

On these two models the verification of the DLA technique has been carried out. All the results in the verification procedure are made by simulations in discrete points of the space.

3. METHODS

The two methods for reduction the sound pressure prediction error are used. The first one is the above mentioned DLA measurement technique with two parallel microphone arrays.

For the separation of sound field on the "non-measurement" side of the holograms by DLA technique one can use this equation [2]:

$$P_{zh}(k_x, k_y) = \frac{P_{zh1}(k_x, k_y) - P_{zh2}(k_x, k_y) e^{ik_z D}}{1 - e^{i2k_z D}} \tag{1}$$

The result of the equation 1 is the image of the pressure field P_{zh} (in wavenumber domain) in the first hologram plane which comes only from examined surface under the first hologram. Other sound sources are removed by the difference of pressure field images in holograms (P_{zh1}, P_{zh2}), with distance D, and a special backward propagator in the denominator. Due to this propagator, the calculation is singular in the discrete points identical with discrete spacing in wavenumber domain. For this reason, the special treatment for reducing the error caused by this singularity has been used. It has been derived from general calculation of Rayleigh integral [4], which has been often used for reduction of singularity in NAH calculation with input data measured by velocity probes.

The second method involved in calculation is iterational method with Wiener filtering [3,5], which has been used only for single layer array measurement yet. This method reduces the pressure reconstruction error and the core of this method uses recursive principle to minimize the difference between the forward propagated near-surface pressure, calculated with NAH method, and the measured pressure field (input values from microphone array). The equation 2 represents the iterative technique:

$$P_{zx} = P_{zx-1} + S \cdot \left(P_{zh} - G_p \cdot P_{zx-1}\right) \tag{2}$$

In the equation 2, the image of the pressure field near surface in the last step, P_{zx-1}, is adjusted by the difference of an actual image of measured pressure field P_{zh} and the forward propagated near-surface pressure field. The propagator G_p is the Green's function for free field condition and S is the Wiener filter transfer function:

$$W(k_x, k_y, z_{h1} - z_x) = \frac{G_p^*}{\left|G_p\right|^2 + \varepsilon^2} \tag{3}$$

The parameter ε is the regularization parameter and ensures the convergention of the whole calculation [5]. The setting of the ε parameter is dependent on the amount of background noise in the measured pressure data and on the imperfections in signal path. The most important part of the signal path is the pressure sensor (microphone) itself, where amplitude and phase mismatch can cause small error in measured data, but huge error in reconstructed calculated pressure field near examined sound source surface.

These two methods were combined together and the simulations have been performed with the above mentioned reverberant condition models.

Two error representations can be applied to evaluate the effectivness of presented algorithm. The first norm compare difference between true acoustic quantity values (analytical calculation of Rayleigh's integral) and values calculated by indirect NAH algorithm, and total true square values at all discrete points on the examined plane. The calculation can be also limited only for discrete points which lie directly above the source. In this modification, the error caused by finite aperture calculation is reduced and results are better comparable. When this procedure is applied, range of indexes i and j in equation 4 are reduced and they represent only points which are strictly above the source region. This error method calculates the global the mean square average error in percent and can't handle big errors in few points.

$$MSE = \sqrt{\frac{\sum_i \left|p_i^{true} - p_i\right|^2}{\sum_j \left|p_j^{true}\right|^2}} \cdot 100 \ [\%] \tag{4}$$

For more precise evaluation of error, there could be also used another error representation, which uses direct acoustic quantity values on the diagonal of the calculated plane or only above the vibrating plate, and a difference between true and calculated values are shown directly on the figure. This representation can detect big errors in isolated points, but it is limited only to the diagonal, but usually this region determines the errors for the whole plane.

4. RESULTS

In both cases (models), the examined sound source is simulated as thin steel plate mounted in infinite baffle with dimensions 0.35 m x 0.35 m and thickness of 3 mm. The plate is driven by harmonic force near to the one of the corners to produce high number of natural modes. By the modal composition theory, the complete surface vibration velocity field is calculated as a sum of all considered modes.

The matrix microphone array is used to measure acoustic pressure in the discrete points on the plane. The dimensions of the array are 0.75 m x 0.75 m, and there are 16x16 microphones. The distance between microphones is 0.05 m. The first layer (microphone array) is 10 cm far from examined sound source surface. The calculation plane is 1 cm above the source surface. If there are two layers used, distance between layers is also 0.05 m. For detailed understanding see figure 3.

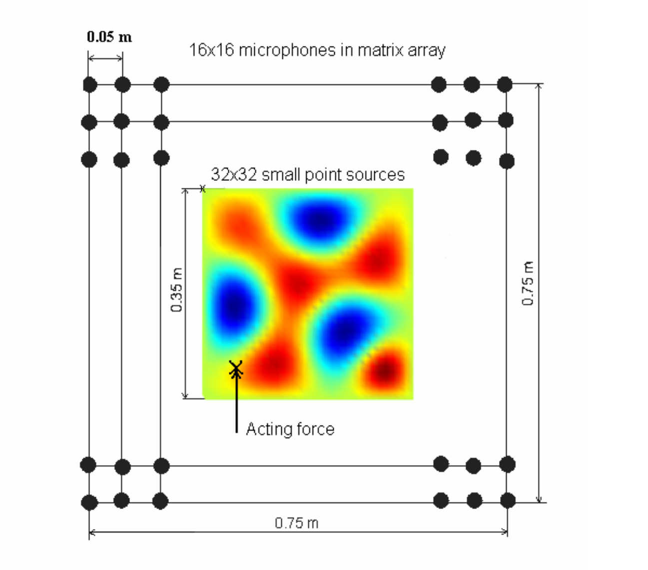

Fig. 3. Position of the microphone array and the panel represents the first sound source

The windowing effect of finite aperture size (limited dimensions of the microphone array) is reduced by 16-point Tukey window and the basic regularization (wavenumber space filtering) was performed with tapered cosine window with a cut-off frequency of 0.3 times the Nyquist frequency and value of parameter α (steepness of the window edges) was set to 0.2.

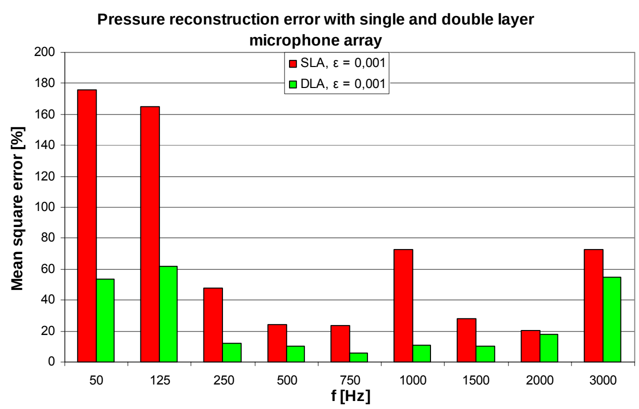

Fig. 4. Sound pressure reconstruction error for single layer array (SLA) and DLA, with second source presented and low regularization parameter ε (low amount of background noise)

In the figure 4, the advantage of DLA calculation in comparison with single layer can be seen from the difference between total mean square pressure reconstruction error with DLA and SLA. The reduction is more evident at low frequencies and in the mid-range, around 1 kHz.

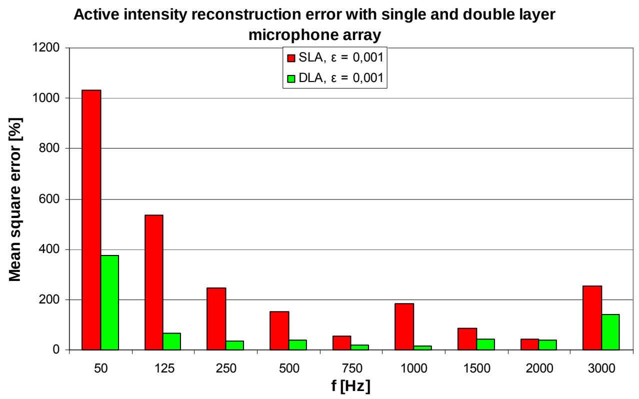

Fig. 5. Active intesity reconstruction error for single layer array (SLA) and DLA, with second source presented and low regularization parameter ε (low amount of background noise)

Similar reduction of reconstruction error can be achieved in calculation of active sound intensity near source surface, where mean square error (difference between true analytical active intensity and backward calculation of measured pressure data) is reduced three times in compare with single layer configuration.

There is also presented the contribution of the proposed calculation method to reduction the pressure prediction error near sound source surface on the diagonal of the plate. These results are shown below and they are calculated for both simulation cases.

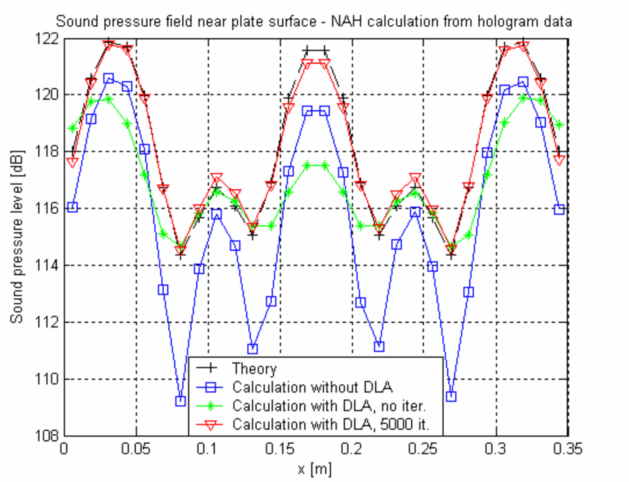

Fig. 6. Sound pressure level near the vibrating surface (*f* = 500 Hz) using analytical, NAH without DLA and NAH with DLA calculation (different number of iterations), with disturbing source (the same *f*)

The figure 6 shows the different results of calculation of the pressure field near examined surface with disturbing the measured (hologram) sound field with the second sound source as shown in the figure 1 by analytical calculation, calculation without DLA (the best setting of iterations) and with DLA method optimized by the stated number of iterations.

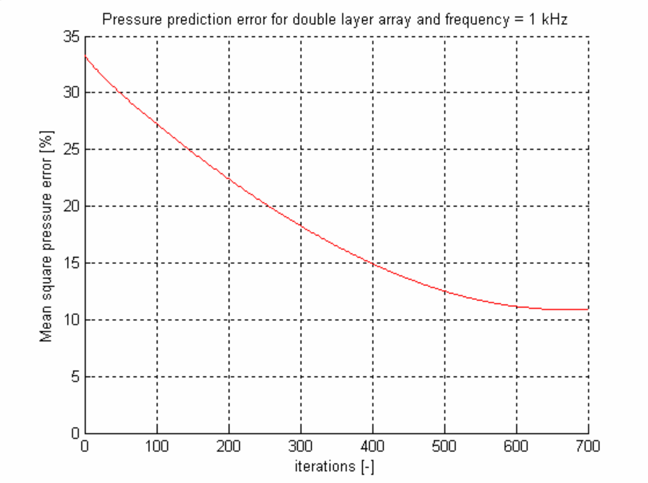

Fig. 7. Sound pressure reconstruction error for DLA processing algorithm and frequency 1 kHz with different number of iterations

The pressure reconstruction error dependency can be seen from figure 7, where the mean square error calculated on the whole area above the vibrating plate surface is reduced with higher number of iterations in recursive Wiener filtering algorithm. The optimal setting for number of iterations is 0.7 times the sound source frequency (driving force frequency). It means that for frequency of 1 kHz, the optimal setting of iteration count is 700. The total error reduction for frequency of 1 kHz can be about 25 percent, where there is 35 percent of error without using iterative improvement.

With the second simulation case, reflection from the plane (wall) normal to measurement and source surface plane, better reconstruction of pressure data near sound source can be reached.

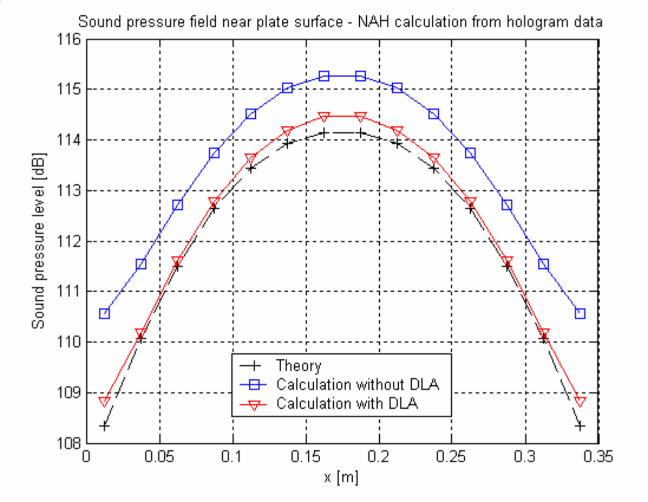

Fig. 8. Sound pressure level near the vibrating plate surface (*f* = 50 Hz) using analytical, NAH without DLA and NAH with DLA calculation, with reflections (mirror image)

The figure 8 represents results for the second simulation example with reflecting surface. The difference between theoretic (analytic) calculation of pressure data in the discrete points near source surface and DLA calculation is very small. The SLA technique produces overestimation of pressure field near source, if there is some reflection surface near sound source. This atribute of DLA technique isn't evident from the theoretical derivation of equation 1, where there is only sound field separation impact. But all double layers have, it is from basics, twice amount of sensors (microphones), so the input data for NAH calculation is doubled and of cource they carry more information about incomming sound field, even if it comes from one front side.

5. DISCUSSION

In comparison of this combined technique with that one presented in [2], better reduction of disturbing sound fields on examined pressure field is achieved, but also the computational time increases. Original DLA method produced huge errors at some frequencies due to the singularity in equation 1, which has been removed or minimized by integration of k_z in the area near the edge of the radiation circle.

The verification has been done also for another type of disturbing source – reflection from surface (mirror image of original) with good results.

6. CONCLUSION

From numerical simulations the advantage of using DLA technique with iterative improvement is evident. In both simulation cases the prediction error is reduced, with another disturbing source roughly by two times for pressure reconstruction and three times for active sound intensity, in the environment with reflections by 40 percent. The effect of the DLA technique is more evident at low and mid-range frequencies.

ACKNOWLEDGMENTS

This work was supported by Grant Agency of the Czech Republic, Grant No. 102/06/1617 and Czech Ministry of Education, grant FRVS: G1/3265.

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