USING ACOUSTIC HOLOGRAPHY FOR VIBRATION ANALYSIS

ANALÝZA VIBRACI POMOCÍ AKUSTICKÉ HOLOGRAFIE

ZKRÁCENÉ ZNĚNÍ DOKTORSKÉ PRÁCE

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1 INTRODUCTION

Measurement of noise and vibration is one of the basic areas of measurement of non-electrical quantities in technical practice [19]. Influence of noise and vibration on environment and humans and systematic pressure on their reducing or limiting causes demands for fast and accurate measurement of acoustic quantities and localization of sources of noise or vibration.

With classical methods of vibration measurement with accelerometers there are difficulties with placement and fixing of the sensor to the measured object which is obviously the problem of all contact measurement methods. Also the weight of the sensor plays its role and can cause large measurement error when the examined object is very small. And the measurement with contact sensors on moving parts (e.g. rotating machines) is more complicated and sometimes impossible [19].

Therefore the non-contact methods have been developed systematically during last decades. Their main advantage is basically in zero influence of measurement object without affecting the object surface or its acoustic parameters. There can be used many principles of measurement starting with pure acoustic, where the acoustic quantities is evaluated, and ending with methods for measurement surface displacement of the vibrating object based on laser interferometry [3, 18].

There are usually used intensity probes, calibrated microphones or microphone arrays during measurement of acoustic quantities. The microphone arrays are very useful for localization of source of vibration which produces noise and for creation of noise maps above vibrating objects. For processing of measured data from microphone arrays the acoustic holography methods are used. Acoustic holography is an experimental technique for processing the input data obtained by measurement in hologram area. Input information have to be processed effectively to obtain accurate values of acoustic quantities which describe generated sound field from vibrating surfaces of analyzed objects and consequently could describe the vibrating source itself (energy, location, etc.). For that reason the algorithms and methods for processing input holographic data are developed. With the rising demands on accuracy and speed of calculation of acoustic quantities the new algorithms have to be developed or existing algorithms need to be optimized for new applications or environments [17, 4].

This doctoral thesis thus deals with optimization of existing and development of new acoustic holography algorithms for areas where existing algorithms fails or they are less accurate (confined space, cabins, etc.). The algorithms published in last two decades were taken in account and the main orientation of this thesis was selected to focus on acoustic holography methods based on transformation into wavenumber domain and with measurement systems using novel double layer microphone array. Future research should be focused on methods using double layer microphone array where there could be applied iterative procedure of prediction of sound fields between layers to enlarge the originally measured hologram area to open the possibility of better measurement with “patch” to “patch” principle. Also future development of measurement systems based on MEMS sensors could be valuable.
2 STATE OF THE ART

The main objective of this thesis is a localization and characterization of sound sources and vibration analysis with non-contact methods, especially with near-field acoustic holography (NAH). In this chapter, the basic theory and state-of-the-art of the acoustic holography methods is described.

2.1 BASIC THEORY OF NEAR-FIELD ACOUSTIC HOLOGRAPHY

Near-field acoustic holography is an experimental technique for localization and quantification of sound sources. In general the holography is a technique carrying large amount of information recorded in a plane (holography plane) to reconstruct three-dimensional wave field and thus compose 3D picture of any observed quantity.

In acoustics, the time record of an acoustic quantity (usually sound pressure) in holography plane carries all information about 3D sound field near the plane. The 3D sound field can be expressed with pressure sound field, acoustic particle velocity vector field or field of sound intensity vectors. From these acoustic quantities the mechanical quantities of sound (vibration) source surface could be determined consequently (surface velocity, location and strength of driving mechanical force). The information in the acoustic hologram is an amplitude and phase of measured acoustic quantity (usually sound pressure) and the variation in time.

Acoustic holography also removes the wavelength resolution limit of reconstructed picture of classical holography, where the length of the radiating wave defines the overall resolution in the 3D space. This feature can be achieved in near field by including special part of the sound field in holography calculation.

For calculation of sound quantities near vibrating surface or near acoustic sound source determination of propagation of sound waves should be determined. The governing equation for wave propagation in near field is homogenous wave equation as described in Eq. (2.1). The pressure sound field in time domain $p(\mathbf{r}, t)$ near source is determined by this wave equation.

$$\nabla^2 p - \frac{1}{c^2} \frac{\partial^2 p}{\partial t^2} = 0, \quad (2.1)$$

where $c$ is speed of sound [$ms^{-1}$] and $\nabla^2$ is Laplace operator $\frac{\partial^2}{\partial x^2} + \frac{\partial^2}{\partial y^2} + \frac{\partial^2}{\partial z^2}$.

The sound field generated by the source as propagating in space can be expressed differently based on distance to the source. If the observation point is closer to the source surface than the half of the wavelength ($\lambda/2$) of the radiating wave then we talk about near-field. Otherwise the observation point is in far-field. The difference between these two domains is in the composition of wave field existing in the domains. The near-field domain contains two types of waves, the propagating waves and the evanescent waves. The propagating waves as title describes propagates from the source through the space without changing their amplitudes (neglecting the losses...
in the medium), but their phase is changing. On the other hand the phase of the evanescent waves remains constant, but the amplitude is changing.

Figure 2.1: Propagating and evanescent waves in \( k \)-space and holograms’ configuration in spatial domain [14].

Classical NAH is based on spatial transformation, where the most used is Fourier transformation. The transformation pair is described in Eq. (2.2) and Eq. (2.3).

\[
\begin{align*}
P(k_x, k_y, z, \omega) &= \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} p(x, y, z, t) e^{j(k_x x + k_y y - \omega t)} \, dx \, dy \, dt \quad (2.2) \\
p(x, y, z, t) &= \frac{1}{(2\pi)^3} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} P(k_x, k_y, z, \omega) e^{-j(k_x x + k_y y - \omega t)} \, dk_x \, dk_y \, d\omega \quad (2.3)
\end{align*}
\]

The usefulness of using transformation into the wavenumber domain is easier calculation of convolution in wavenumber domain. After the transformation, recalculation of sound pressure to any parallel plane near or far from the source can be processed by simple multiplication of the image pressure by the propagator.

While using 2D Fourier transformation from spatial to wavenumber domain, common characteristics of the transformation should be taken in account. There are dependencies in sampling distance, aperture size and continuity of the signal in original (spatial) domain with spectral bandwidth, resolution and leakage in projection (wavenumber) domain. Some of the parameters of input sound pressure signal could not be neglected to obtain corresponding wavenumber spectrum and correctly predict the sound field.

To reduce the errors caused by Fourier transformation which creating unwanted high wavenumber components, the smooth periodicity of original measured data has to be fulfilled. This can be done by windowing of the original hologram aperture, usually with rectangular cosine window (Tukey window).

After inverse Fourier transformation of wavenumber spectrum back to the spatial domain due to the circular convolution in transformation, ghost images of the original aperture are present. To enhance the resolution in wavenumber domain and move ghost images faraway there can be used the treatment of the original hologram which includes enlargement of the hologram by zero padding in spatial domain.
2.1.1 NAH with Simple $k$-space Filter

The reconstruction of the acoustic quantities near examined source is provided by simple basic acoustic holography calculation, where the measured pressure or velocity field in the hologram position is multiplied with propagator. This multiplication is ill-posed problem because the propagator amplifies evanescent components (waves lying outside radiation circle) and can also amplify noise and all distortion caused by imperfections of the measurement path must be taken in consideration. Presence of noise signal in the measurement can’t be avoided thus some kind of filtration in wavenumber domain ($k$-space) should be realized. If there is no filtration in $k$-space the resulting sound field in prediction plane could be totally damaged and important information usually obtained with backward prediction will be lost. Filtration is applied to wavenumber spectrum of an acoustic quantity obtained in holography plane. The low-pass filter in $k$-space is usually realized with using Harris exponential two-dimensional window [17]. The setup of filter parameters depends on amount of noise presented in the measurement. This information could be obtained through an experiment or set with some general knowledge about the measurement system and their characteristics.

2.1.2 Iterative NAH with Recursive Filtration

Improved variant of the algorithm with $k$-space filter is iterative NAH algorithm. The principle of the algorithm is using forward prediction of the first estimation of the sound field from reconstruction plane $z_x$ to measurement (holography) plane $z_h$ and comparison of the forward prediction result with originally measured values. All prediction calculation is performed in wavenumber domain. The difference between the wavenumber spectrum of sound pressure field values calculated with forward prediction and the spectrum of true pressure values measured in holography plane represents the prediction error which is subtracted from original spectrum estimation of sound pressure field in reconstruction plane and the whole procedure is repeated [3, 1].

The first estimation of the reconstructed field is made with simple setting of wavenumber filter, where the cut-off frequency is selected to one half of the maximum frequency of measurement, equals to stepping in spatial domain (distance between microphones). This estimation is later used in iterative technique based on recursive Wiener filtering algorithm.

$$P_{zx} = P_{zx-1} + S \cdot (P_{zh} - G_p \cdot P_{zx-1})$$  \hspace{1cm} (2.4)

where $P_{zx}$ is a wavenumber spectrum of sound pressure field in prediction plane, $P_{zx-1}$ is a wavenumber spectrum of sound pressure field in prediction plane determined on previous iteration result or during the first step with initial estimation of sound field based on simple algorithm with k-space filter, $S$ is a feedback operator, $P_{zh}$ is a wavenumber spectrum of sound pressure field measured in the holography plane, $G_p$ is direct (forward) transformation function (Green’s function). As a feedback
operator $S$ there can be used Wiener optimal filter which oneself guarantees the convergence of the whole iterative procedure. The Wiener filter in wavenumber domain can be expressed in Eq. (2.5) [5].

$$ W(k_x, k_y, z_h - z_x) = \left( \frac{G^*_p}{|G_p|^2 + \varepsilon^2} \right) $$ (2.5)

where $G^*_p$ is a complex complement to the wavenumber spectrum of direct transformation function. Wiener filter incorporates regularization parameter $\varepsilon$ to adjust transfer function of the filter. This parameter reduces very high evanescent wave amplification in recursive Wiener filtering, thus eliminates influence of the noise and imperfections in the measurement (non-ideal transducers). The regularization coefficient in the transfer function of Wiener filter can be represented in Signal-to-Noise Ratio (SNR), describing the expected difference between measured values and overall noise in the measuring path.

2.1.3 Acoustic Holography with Direct Calculation in Spatial Domain

In this section there are presented the acoustic holography algorithms which predict the sound field near source with direct calculation of backward propagation in the spatial domain without using transformation into wavenumber domain. These methods are usually called “patch” methods because there are no disturbing effects on the hologram edges in compare with algorithms based on Fourier transformation.

Statistically Optimized Near-field Acoustic Holography (SONAH)

Statistically optimized near-field acoustic holography (SONAH) is one of the so-called patch holography algorithms which doesn’t use spatial transforms and thus avoid leakage. SONAH assumes that if there can be found the appropriate wave spectrum, it means sufficient set of elementary waves, which creates the sound field propagated from the source to the microphone positions, the coefficients which describes this elementary wave set, can also in similar way describe the pressure set which is in measured and examined points in the sound field.

The main challenge in using this algorithm is to determine the correct value of regularization parameter $\theta$ in the Tikhonov regularization in SONAH. For the calculation of autocorrelation and cross correlation matrices it is necessary to use numerical integration and the evaluation have to be done for all discrete points in the prediction plane where the knowledge of pressure field values is required. From the mathematical derivation it is straightforward that SONAH algorithm can as opposite to classical NAH (with discrete 2D Fourier transform) predict sound field values outside the $(x, y)$ positions in prediction plane $z_x$ where sound pressure is measured in holography plane $z_h$.

In the classical algorithms the limitation is based on 2D Fourier transform where there have to be constant interval between positions in measurement plane and consequently the same $(x, y)$ positions in prediction plane [16]. The SONAH algorithm
has been initially developed for planar acoustic holography, but after successful application and evaluation of this principle it has been used after certain extension to other geometries, like cylindrical. The comprehensive description of the theory and properties of SONAH algorithm has been published recently [6].

**Helmholtz Equation Least Squares (HELS)**

The HELS method assumes that radiated sound pressure can be expressed by formulation with extended basis function [18]. The equation is very similar to that one used in SONAH algorithm and it is also based on assumption that sound pressure in the field point can be defined with the sum of contribution of each elementary function defining the radiation from the vibrating object. In the HELS algorithm the elementary function is defined as $\Psi_j$ and they are obtained by Gram-Schmidt orthonormalization process of particular solution of Helmholtz integral. The particular solution can be easily expressed in spherical coordinates where there are known spherical Hankel and continuous Legendre functions [18].

The main advantage of the HELS method is its mathematical simplicity, effectiveness of the calculation and adaptability to the specific application. The method is working also in the field where there are reflecting surfaces present and requires low number of measurement points than planar NAH or inverse Boundary Element Method in the same measurement conditions [18].

### 2.1.4 Acoustic Holography Based on Boundary Element Method

For reconstruction of acoustic quantities radiated by object with arbitrary surface geometry there can be used the theory of Helmholtz integral which defines relation between sound pressure field above vibrating object and normal component of surface velocity on the sound source surface. This theory is valid for outer and also for inner space. For arbitrary geometry there is not clear solution of Helmholtz integral. From that reason it was necessary to find numerical solution of the integral. Most common method for numerical solution of Helmholtz integral is Boundary Element Method (BEM) which divides the arbitrary surface of sound source to small elements and afterwards calculates the acoustic quantities at specified discrete points on those elements [15, 13].

### 2.2 REGULARIZATION PROCEDURES FOR NEAR-FIELD ACOUSTIC HOLOGRAPHY ALGORITHMS

Ill-conditioned transfer matrices of the NAH algorithms (especially in algorithms which calculate predictions directly in spatial domain) are usually present and for inverse calculation, some kind of regularization should be used. Basically there are two mostly used methods for regularization and two methods for determination of regularization parameter.
As the methods for regularization of the inverse solution, the Tikhonov regularization and Truncated SVD are often used in connection with treatment of inverse matrixes of near-field acoustic holography algorithms.

There are also two automated regularization parameter identification methods that are usually used in connection with acoustic holography. They are “L-curve” analysis/criterion and Generalized Cross Validation (GCV).

### 2.3 MEASUREMENT SYSTEMS FOR NEAR-FIELD ACOUSTIC HOLOGRAPHY

For measurement of acoustic field quantities (acoustic pressure, particle velocity, sound intensity) there can be selected many different type of sensors working on different principles. The main usual measurement procedures should be:

- Measurement of sound pressure - microphones
- Measurement of sound intensity - intensity $p - p$ probe (microphone pair),
- Measurement of particle velocity - intensity $p - u$ probe (Microflown sensor).

#### 2.3.1 Microphones

For measurement of sound pressure field values at discrete points above the vibrating object (sound source) there are often used classical microphones with capacitive transducers. The important parameters of the microphones are amplitude and phase frequency response and self noise (including potential amplifier). These parameters should be very similar for all sensors used in the measurement (it there is more than one) because the acoustic holography methods are very sensitive to amplitude and phase mismatch of the microphones. For this reason the sensors (microphones) are usually calibrated and the frequency response is stored in data file or directly (as TEDS - Transducer Electronic Data Sheet) in the smart sensor (in non-volatile memory). The phase matching of typical array microphone (B&K Type 4958) is $< \pm 3^\circ$ at a frequency range from 100 Hz to 3 kHz.

#### 2.3.2 Sound Intensity Probes

The intensity probes are devices (sensors) for vector measurement in compare with microphones which measure scalar sound pressure. They can give the value of the sound intensity including the direction of the intensity vector. The sound intensity could be calculated based on measurement with two microphones ($p - p$ probe) or with one microphone and one particle velocity sensor ($p - u$ probe).

#### 2.3.3 Measurement Procedure

The measurement of the field acoustic quantities could be done using one sensor with scanning robot or with line array of sensors also with scanning robot or with matrix array where all the field points are measure simultaneously. The first two methods
require using of the reference sensor to obtain the information about phase shift between subsequent measurements. These two methods can be used only for stationary sound field, where the sound radiating from the vibrating source is not changing its frequency and energy (amplitude and phase of sound intensity vectors) and also its position. If there is more than one sound source present in the measured field, the measurement requires multiple reference signals to determine proper input pressure signal at microphone position for each measurement step.

3 GOALS OF THE THESIS

Analysis of vibration and visualization of sound fields with acoustic holography is very interesting for technical diagnostics due to high amount of information that can be acquired and also for the nature of non-contact measurement. The algorithms for acoustic holography calculation has been developed for more than 25 years and recently the algorithms for prediction of acoustic quantities to describe structural vibration of analyzed surfaces are growing. It can be allowed with improvement of computer systems for processing measure data, where there is a possibility to acquire input signal very fast with many measurement channels and process it numerically. Current research in the field of acoustic holography is focused on development of new or optimization of existing algorithms. Sustained requirement of more accurate prediction and characterization of sound fields near the surface of vibrating object is rising the complexity of measurement system to obtain sufficient amount of input data, but such systems are less applicable in technical practice. Analysis of vibration of large scale structures (larger than measurement array) is also required. Researchers try to solve this task with the new numerical methods which in most cases avoiding spatial transformation which has been usually used in classical algorithms. Involving spatial transformation into calculation procedures yields to the problems with preparation of input data for transformation (windowing function). On the other hand, the calculation procedure with spatial transformation is very fast.

Applicability of classical methods (with transformations) for near-field acoustic holography calculation is still relevant and their optimization and detailed characterization of each possible parameter in relation to prediction accuracy of sound field near vibrating surface have not been sufficiently performed yet. There can be some future improvements in prediction based on combination of different approaches and their optimization.

The main objective of this thesis is development, implementation and optimization of near-field acoustic holography algorithms based on spatial transformations with respect to increasing of prediction accuracy in real measurement conditions. The procedure with spatial transformations has been selected with assumption that these algorithms can calculated the predictions rapidly while using 2D Fourier transformation. There are also many improvements already published in last decade which have not been combined and could bring new ideas into future research. The promising procedure is measurement with double layer microphone array (in two plan-parallel planes
with small distance between layers) for prediction of sound fields in complex environments, because from the theory of holography, amount of information in two parallel planes should be larger than with only single plane measurement. This procedure certainly has an advantage similar to measurement with intensity probes, where also the direction of the propagation of waves in sound field can be examined.

The reconstruction of whole sound field above vibrating surface with more than one measurement step is also prospective research task and has been studied within this research. The commercial research in this field is focused mainly to numerical methods. The classical approaches could be also successful if there will be some treatment of input data to obtain better prepared pre-processed data before spatial transformation.

Development of new methods leads to construction of new improved measurement systems. This is also scope where the research can be done and is also described in this thesis. The procedure with double layer microphone array requires twice the number of acoustic sensors (microphones) to measure the sound field. With commonly used special array sensors (very precise and stable), these methods will be very expensive. The research in this field is focused on using common sensors (electret microphones, commercially available high volume sensors - MEMS microphones) incorporated with some additional information (e.g. stored frequency response of the sensor) which can be used during the measurement to correct the input data acquired with such sensor. Also the manufacturing precision of high volumes of acoustic sensors (e.g. microphones for cell phones) is increasing and these sensors can be used now for holography measurement where strict requirements are applied.

Finally, experimental validation of acoustic holography method for non-contact vibration analysis is expected to confirm applicability of the method for such vibration analysis.

4 ADVANCED ACOUSTIC HOLOGRAPHY METHODS

This chapter describes development and enhancement of planar near-field acoustic holography algorithms which has carried out as main goals of this thesis. The presented algorithms try to minimize prediction errors caused mainly by real measurement conditions (non-ideal transducers, presence of noise in measurement path, non-free field conditions, etc.).

4.1 SIMULATION MODEL OF VIBRATING PANEL

Based of literature study there were designed simple model of vibrating surface in Matlab environment. The basic idea of this approach is the modelling of the vibrating surface by matrix of elementary point sources which are placed on the surface. Displacement $w$ of the surface of that designed vibrating panel in points of fictitious point sources and driven with harmonic force $F$ can be calculated. For development and optimization of acoustic holography algorithms it is necessary to know the true sound pressure field values at certain distances from the vibrating surface. To the specified
spatial points there we can place virtual microphones and then calculate the contribution of each fictitious vibrating point of the surface to the total sound pressure at defined spatial point. The sound pressure values at the spatial points \( r \) where there are virtual microphones placed above vibrating surface can be calculated [3].

### 4.2 SIMULATION TEST CASES FOR EVALUATION OF ACOUSTIC HOLOGRAPHY ALGORITHMS

The simulation test case is outlined in Fig. 4.1 (configuration for double layer array processing). The examined source is a thin steel panel with dimensions 0.35 m x 0.35 m and a thickness of 3 mm mounted in an infinite baffle. The panel is driven by a harmonic point force near one of the corners to produce high number of natural modes. In this simulation, the plate was driven with force of strength in units of Newton (from 1 N to max. 10 N), at the point 0.05 m x 0.05 m far from one of the corners. The resulting surface velocity is calculated as a conventional modal sum. The number of modes included in the sum corresponds to modal indices from 1 to 15 in both the \( x \)- and \( y \)-direction. The sound pressure at the microphone positions is calculated from a numerical approximation to Rayleigh’s first integral [17].

![Figure 4.1: Outline of the simulation case with sources on both sides of the microphone array (left) and position of microphone array and vibrating panel (right).](image)

Two error norms can be applied to evaluate the effectiveness of the algorithms under test. The core of the both error representation is the same, but there is a difference in scaling. Both norms 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 logarithmic error norm calculates relative average error level and can be more transparent when there is a big difference between minimal and maximal error of compared algorithms and in the selected frequency range [7].
\[ L_{err_p} = 10 \cdot \log_{10} \left( \frac{\sum |p_{i}^{true} - p_{i}|^2}{\sum |p_{j}^{true}|^2} \right) \]  \hspace{1cm} (4.1)

4.3 ADVANCED METHODS FOR SINGLE LAYER ARRAY NAH

Evaluation and testing of new acoustic holography algorithms with focus on algorithms based on transformation to wavenumber domain is one of the goals of the thesis. Within this scope the improvement of accuracy of existing algorithms, which have been published in the last two decades, and expansion of their usability and applicability into new areas are also the objectives of this thesis. The section about single layer acoustic holography algorithms (algorithms using input data from measurement in one plane parallel to examined source) mainly describes some important improvements and evaluates the properties of the algorithms.

4.3.1 SONAH with Lower Computational Cost

One new computer implementation of the SONAH algorithm [7] has been proposed in the scope of this thesis. The new implementation uses, as compared with recent classical calculation with numerical integration, simplified but faster calculation of infinite integrals. Classical implementation of SONAH algorithm is based on original derivation by Hald, where infinite integrals are solved by numerical integration with Gauss-Laguerre and Gauss-Legendre quadrature.

Proposed new implementation of SONAH is derived from the basic infinite integral equations with respect to the original idea of the SONAH algorithm, where the limited number of the elementary waves is considered. The original integrals can be represented in polar coordinates and then transformed to the finite sum of the elementary waves. The optimal set of elementary waves used for SONAH calculation (number of waves, maximal wave-number) can be determined from frequency range of used microphone array and its configuration. The simplification of the calculation procedure has also one useful advantage, which can speed up NAH calculation in many measurement cases, of only one-time calculation of all Bessel functions for selected measurement set-up - spacing between microphones and stand-off distance of microphone array from sound source surface. This means that the first-time calculated Bessel functions can be used for any other frequency components of future NAH calculation with the same set-up.

The calculation time of new proposed implementation of SONAH is more than 20 times faster for small array and 5 times faster for large array than old implementation with numerical integration. The main reduction of computation cost of new algorithm is in one-time calculation of Bessel functions, while old implementation has to calculate all Bessel functions independently (dependent variables in the function parameters). The reduction of computation cost in compare with recent implementation
with numerical integration is evident and the pressure prediction error in near-field remains in the same level as with recent version.

4.3.2 Novel Method for Determination of Regularization Parameter

In this section, two different NAH algorithms are used together to obtain correct value of the regularization parameter for any near-field acoustic holography algorithm. The main idea of the novel method is in comparison of results of pressure prediction of two different acoustic holography algorithms. If the selected algorithms are using different calculation procedure for pressure prediction and involves different procedure for regularization then different influence of regularization parameter to prediction accuracy could be expected. From this pre-requisition we can assume if proper amount of regularization is applied to both algorithms the highest accuracy of pressure prediction could be achieved and difference between pressure prediction could be minimal. If the regularization parameter has not been selected ideally, then difference between predictions obtained with both algorithms will be larger. For this purpose we can combine results (predicted pressure values) from two selected algorithms (SONAH and classical NAH), while using same amount of regularization for them.

Due to different calculation method of both algorithms, the difference between pressure values is minimal when proper amount of regularization is used. This feature can be explained in the procedure of using regularization in both algorithms.

While in SONAH algorithm, the regularization reduces singularities in autocorrelation matrix, where each element corresponds to one elementary plane wave, thus regularization influence all plane waves (propagating and evanescent). On the contrary in classical algorithm, regularization in Wiener filter causes filtering of only elementary waves with higher wavenumber only (mostly evanescents). This different application of regularization causes different impact into processing of input noisy data and produces different predicted pressure field near sound source surface, if there is not ideal amount of regularization used. An ideal amount of regularization produces the lowest possible pressure prediction error of all NAH methods.

The proposed method can be run once at the beginning (for one significant frequency component), before total sound field mapping (for all significant frequencies), calculating predicted pressure differences $DIF$ for small region in the centre of predicted plane (approx. 50% of whole predicted plane/microphone array) for all possible SNR assumed in measured data. While minimal value of $DIF$ corresponds to the signal-to-noise ratio in the input data and the right value of regularization parameter can be successfully adjusted. After right amount of regularization has been found, for all other frequency components, same regularization parameter can be used.

In the Table 4.1, summary of pressure prediction errors for ideal, determined and no regularization in SONAH algorithm is presented [9].

As shown in Table 4.1, the added error caused by non-ideal determination of regularization parameter is very low, thus validation of proposed method has been done.
Table 4.1: Pressure prediction errors with the SONAH algorithm and with different amount of regularization.

<table>
<thead>
<tr>
<th>Added amplitude mismatch [dB]</th>
<th>Prediction error of pressure field calculated with SONAH [%]</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Zero regularization</td>
</tr>
<tr>
<td>0</td>
<td>5.0</td>
</tr>
<tr>
<td>0.01</td>
<td>6.2</td>
</tr>
<tr>
<td>0.02</td>
<td>8.7</td>
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<tr>
<td>0.05</td>
<td>19.4</td>
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<td>0.5</td>
<td>148</td>
</tr>
<tr>
<td>1.0</td>
<td>303</td>
</tr>
</tbody>
</table>

In compare with huge prediction errors with zero regularization, proposed method estimates regularization very close to ideal and reduces prediction error successfully.

### 4.4 DOUBLE LAYER ARRAY NAH

In conventional planar acoustic holography sources on the “wrong” side of the measurement plane (including image sources due to reflections) contaminate the sound pressure in the measurement plane and give rise to errors. By contrast, the double layer technique can, at least in principle, separate the two contributions from each other [2].

Basic planar acoustic holography theory expects that all sound sources are only on one side of the hologram (measurement array) and there are no other sources. If there is a situation with other sources presented on the other side (mostly behind the hologram), the reconstruction of sound field near examined sources are very difficult and creating huge errors. The other source on the opposite side causes errors in measurement of basic acoustical quantity (acoustic pressure) at the microphone positions. The second source (behind the hologram) can be introduced as mirror image of this source on the same size as examined source. The result of this configuration leads in most cases to underestimation of original sound field in examined plane.

The main idea of measurement with two parallel planes in the environment where there are sources on both sides of the measurement planes is described in Fig. 4.2.

This arrangement introduced four parallel planes where there are two of them called “source planes” and the rest are “hologram (measurement) planes” placed in parallel between the source planes. The separation of sound fields coming from different sides of the hologram plane can be expressed in Eq. (4.2). This equation expects two measurement layers with microphones assembled in matrix. Acoustic quantity is measured in both layers and the separation of incoming sound waves from opposite directions can be achieved.
Figure 4.2: Composition of sources and planes in double layer holography [2].

\[ P_{1A}(k_x, k_y) = \frac{P_1(k_x, k_y) - P_2(k_x, k_y)e^{jkzd}}{1 - e^{jkzd}} \]  

(4.2)

This equation is generally ill-posed due to the expression in the denominator. The only possibility of to avoid the difficulty connected with the \( k \)-space aliasing (ill-posedness) is to smoothen the transformation function (separation term) \( \frac{1}{1 - e^{jkzd}} \) at discrete points in wave-number domain. The solution can be simplified for only smoothen \( k_z \). Averaged \( k_z \) can be expressed in Eq. (4.3) [10].

\[ \overline{k_z} = \frac{(k^2 - k_{1z}^2)^{\frac{3}{2}} - (k^2 - k_{2z}^2)^{\frac{3}{2}}}{3k_0\Delta k}, \]  

(4.3)

where \( k \) is the wavenumber of analyzed (calculated) sound field.

### 4.4.1 Improvements in DLA Technique

Some possible improvements of the original algorithm for sound field separation in wavenumber domain could be used in connection with double layer processing. The methods are derived from single layer calculations. The filtration of evanescent waves in \( k \)-space should be also assumed in case of double layer array measurements. The double number of microphones involves higher noise in the measurement and also numerical errors in processing procedure. The equation for the separation of front and back sound field supplemented with partial filtering of evanescent waves of the second sound source (unwanted background noise) in wavenumber domain.

To obtain the similar reconstruction error as with single layer measurement technique, the recursive iterative method can be used too [1]. The Wiener filtration is applied to the resulting sound field coming only from the first (examined) source after sound field separation. The procedure is similar to that one used in single layer configuration.
In double layer array processing, there could be also used error reduction procedure which enlarges measurement hologram aperture by zero padding. There were few simulation studies and the hologram has been enhanced to twice and four times the original dimensions. Due to the windowing before discrete Fourier transformation, the both measurement planes must be enlarged. The advanced method is adaptive enlargement, which uses iterative procedure with direct and inverse transformation (from spatial to wavenumber domain) of filtered wavenumber spectrum in \( k \)-space. For double layer holography the iterative procedure can be written in Eq. (4.4).

\[
p_{zh1,2}^{\text{n}(4M \times 4N)} = F_{x,y}^{-1} \left[ F_{x,y} \left[ p_{zh1,2}^{\text{n}-1(4M \times 4N)} \right] K_f \right] \leftarrow p_{zh1,2}^{\text{orig}(M \times N)}
\]  

(4.4)

Restored pressure data in spatial domain can then be fully used into transformation to \( k \)-space, where the kernel of the double layer NAH calculation is applied.

### 4.4.2 Simulation Results with Double Layer Technique

In the presented test case the measurement of original sound field is disturbed by noise from the other side of the measurement plane. The second source is similar to the primary source, but the point force driving the secondary panel is ten times larger than the primary force and acting at another position. Thus the disturbing source is stronger than the primary source, but much further away from the measurement plane(s). Not surprisingly the double layer method performs significantly better than the single layer method in this case, as can be seen from Fig. 4.3. However, the double layer method is also disturbed by the extraneous noise and the general error level is increased.

![Figure 4.3](image.png)

Figure 4.3: Error in pressure reconstruction with disturbing background noise from the wrong side of the array, regularization in Wiener filter \( \varepsilon = 0.001 \).

The comparison of sound field maps obtained with single and double layer holography and compared with analytical “true” solution is in Fig. 4.4. From the reconstructed
maps, it is evident that DLA perform much better than SLA in case of presence background disturbing sound field.

![Figure 4.4: Reconstructed active intensity directly above the vibrating plate (in central part of the microphone array) for frequency 1 kHz; upper - analytical solution; bottom left - single layer reconstruction; bottom right - double layer reconstruction with disturbing background noise from the wrong side of the array, regularization in Wiener filter $\varepsilon = 0.001$.](image)

**4.5 ENTERING INTENSITY MEASUREMENT**

Most of the recent NAH algorithms progressively developed in last three decades could be used in such conditions which are very close to free-field. Thus, no other disturbing sound sources than identified source under microphone array are present. In real measurement conditions, unwanted sound sources located behind the array are usually present and they are generated by real sources (vibrating panels) and reflections from the walls opposite to the source surface under examination. All these disturbances (not free-field conditions) can be found in cabins of the automotive vehicles, aircrafts, small rooms etc. This means it is necessary to handle all these conditions and extract only sound field components which are related to the examined sound source. These components can be expressed with one sound field quantity, as entering intensity - sound intensity of the source (vibrating panels) in front of the measurement array [8]. The disturbing sound field coming from behind, defined as incoming filed, is scattered on the examined surface with defined acoustic surface absorption or admittance and propagated to the array - scattered field. Both of these sound field components coming from there examined source direction (entering and scattered field)
are present at measurement positions and can be entitled - outgoing field expressed as sound intensity or sound pressure. Total sound field present at measurement point contains both incoming and outgoing sound field components. Entering sound field at calculation points (very close to the surface, but not on the surface) could be determined with Eq. (4.5).

\[ p_{\text{ent}}(z_0) = F^{-1}_{x,y} \left[ F_{x,y} \left[ p_{\text{out}}(z) \right] K_f G_p^{-1} - \frac{1-a}{1+a} F_{x,y} \left[ p_{\text{in}}(z) \right] G_p \right] \] (4.5)

Calculation of the first term of Eq. (4.5) should be also done iteratively, similarly to procedure using Wiener filter to reduce reconstruction errors.

![Figure 4.5: Predicted active sound intensity field maps with double layer NAH (known true values - odd column, calculated with DLA-NAH - even column), background disturbing sound field (plane wave) present.](image)

### 4.6 METHODS FOR PREDICTION OF SOUND FIELDS BASED ON PATCH MEASUREMENT

The portions of pressure field measured with microphone array could be combined into one large area hologram basically, in the case where the NAH algorithm with spatial transformation is used, by two approaches.
Assemble the Patches in Spatial Domain

The first approach combines the portions of predicted pressures (patches) in spatial domain, where one portion of predicted near-field pressure data is appended with neighbour portion to cover whole examined area above vibrating surface under test. In this case the NAH calculation is separate for each patch, and the combination of the patches has been done when all the reconstructed (predicted) sound maps are present. The patches can be obtained with these attributes:

1. No overlapping areas (exact match of all patches),
2. Small overlap on the edges (patches slightly shifted),
3. Row/column overlapping (intentional).

The overlapping of the one or two rows or columns of the matrix, which represents the predicted sound field near source surface, will smooth the solution on the contact points between the patches. Thus, for covering the same area as with the previous two methods, in this case, one more patch should be acquired. Eq. (4.6) represents the combination of the patches with two rows and columns from each patch (edge data) left out [11].

\[
\begin{bmatrix}
    p_{x1,1}^{(M-1)x(N-1)} & p_{x1,2}^{(M-1)x(N-2)} & p_{x1,3}^{(M-1)x(N-1)} \\
    p_{x2,1}^{(M-2)x(N-1)} & p_{x2,2}^{(M-2)x(N-2)} & p_{x2,3}^{(M-2)x(N-1)} \\
    p_{x3,1}^{(M-1)x(N-1)} & p_{x3,2}^{(M-1)x(N-2)} & p_{x3,3}^{(M-1)x(N-1)} \\
\end{bmatrix}
\] (4.6)

Figure 4.6: Measured, “true” and calculated sound field maps with overlapping of one edge row/column, for the force frequency of 500 Hz and strength 10 N.
Combination of Patches in Wavenumber Domain

The patches are combined after adaptive aperture enlargement, where field pressure image of the patch has been calculated based on adaptive hologram aperture enlargement. The number of hologram points in wave-number domain is four times higher than in original hologram, thus providing better wave-number resolution. The wavenumber patches are then combined on basis of adjustment of added resolution points. The resulting pressure image in wave-number domain contains information from all measured patches, where patches transformed into wave-number domain has been assembled into large one and in iterative procedure, as a part of the NAH algorithm with spatial transformation and adaptive hologram aperture enlargement, restoration of the complete large sound field map could be done.

\[
P_{zh}^{n(4Mx4N)} = F_{x,y} \left[ F_{x,y}^{-1} \left[ P_{zh1}^{n-1(MxN)} K_f \right] + F_{x,y}^{-1} \left[ P_{zh2}^{n-2(MxN)} K_f \right] 
+ F_{x,y}^{-1} \left[ P_{zh3}^{n-2(MxN)} K_f \right] + F_{x,y}^{-1} \left[ P_{zh4}^{n-2(MxN)} K_f \right] \right]
\]

Figure 4.7: Measured, “true” and calculated sound field maps with assembly of patches in wave-number domain, with four times adaptive enlargement of the hologram, the force frequency is 500 Hz and strength 10 N.

5 MEASUREMENT SYSTEM FOR ACOUSTIC HOLOGRAPHY

The important part of each acoustic holography method is also the procedure of measurement of sound field information near the examined vibrating object. These information act as an input data for the holography algorithm and their quality is crucial.
for successful prediction (or reconstruction) of sound field close to the source surface (the most usual case of prediction). The commonly used sensors and measurement systems have been mentioned previously, but in most cases the whole measurement systems with such sensors (precision array microphones, $p-u$ probes, $p-p$ probes) are very expensive thus limit the applicability of such systems in practice.

The main reason for design and hardware construction of novel measurement hardware is to lower the expenses for purchase such system while maintaining the important parameters of the system which is required for the data processing algorithms.

Within the scope of this thesis, the linear scanning system with smart microphones has been designed and constructed. The measurement array has 16 microphones in one column and can cover the distance of approx. 1 m. The spacing between microphones is 5 cm and movement of the array is provided with stepper motor controlled from the PXI system. Also matrix measurement system has been designed and constructed as a result of this research. The matrix array contains 64 digital MEMS microphones from Knowles Acoustics in the 8x8 configuration with equal spacing of 3 cm in both directions. The microphone array is connected directly to FPGA card in PXI system from National Instruments. Decimation and filtration of all digital signals are processed in real time, thus visualization of measured and potentially also predicted sound field can be done in real time.

![Measurement system with linear microphone array and smart microphones](image1.jpg) ![Matrix microphone array](image2.jpg)

Figure 5.1: Measurement system with linear microphone array and smart microphones and with matrix microphone array and digital MEMS microphones.

### 6 EXPERIMENTAL VIBRATION ANALYSIS WITH NAH METHOD

The experimental verification of using acoustic holography for vibration analysis has been done on laboratory model of vibrating object which is used for examination of natural frequencies of the vibrating body. Measurement of sound field radiated by the vibrating beam has been done for few natural modes of the steel beam mounted...
on the shaker and excited on the supposed frequencies. For the processing the measured pressure signal in the hologram plane, there were selected the algorithm with Fourier transformation and implemented with zero padding of the original hologram and with filtration of high evanescent waves in wavenumber domain with iterative procedure based on Wiener filter. Results of experimental non-contact modal analysis are in Fig. 6.1.

![Sound field near beam surface](image)

**Figure 6.1:** Sound field very near (1 cm) from beam surface, vibrating at 625 Hz, classical algorithm [12].

There can be found only peaks in sound pressure amplitude of all antinodes (with positive or negative phase) together. From the modal theory of the steel beam, the number of antinodes has to be seven and also pressure field near the beam surface has seven peaks (one peak is not so significant, but it is still there - between the second and third).

The method with the linear microphone array is suitable for visualization of sound fields produced by constant stable vibrations of non-moving objects. There can be obtained single sound pressure or intensity map time after time only, but no real-time visualization is available. For the on-line real-time visualization of sound fields, the matrix microphone array with optimized algorithms is needed. Application of matrix array to similar modal analysis is planned for future research.

## 7 CONCLUSION AND FUTURE PROSPECTS

The presented thesis deals with development, optimization and evaluation of near-field acoustic holography algorithms used for vibration analysis. From the huge amount of possible algorithms, the procedures which involve spatial transformation have been selected due to their short computation time with 2D Fourier transformation. After detailed literature study about these holography methods and their improvements, few promising calculation procedures have been selected for further research and application. The tests of each method give an idea about possible applicability in real
measurement condition. The final set of the promising methods contains iterative Wiener filtering, adaptive enlargement of the hologram and certainly most simple filtration in $k$-space. Combination of these methods has been tested on simulations and from evaluation of prediction accuracy the new combined method has been developed.

As a result from this study there has been proposed the new procedure for determination of regularization parameter for NAH algorithms. The regularization significantly influences prediction accuracy of sound field near examined source surface if there is noise in the measurement or if the acoustic sensors are not ideal (amplitude and phase mismatch between sensors). They are not ideal as usual. The method has been successfully evaluated on large set of simulated measurement setups and can estimate signal-to-noise ratio of the input signal in the measured input data. With this information, the setup of subsequent procedures of filtration of unwanted part of the sound spectra (original evanescent waves are affected with the noise and transducer mismatch) is easy and can provide good reconstruction results and accuracy. The procedure involves two different NAH algorithms while each uses different procedure of filtration of the unwanted sound field components. It means that same amount of regularization for both algorithms result in different performance of sound field reconstruction near source surface. If the regularization is ideal (produces lowest prediction error) both algorithms predict same sound field. For this purpose classical NAH algorithm and SONAH algorithm have been used.

To speed up processing with SONAH algorithm which calculates the prediction of sound field directly in spatial domain (involving large matrix multiplication and calculation of Bessel functions) new computer implementation of that algorithm has been developed and later used for determination of proper regularization for NAH. With new implementation, one frequency component is calculated slightly faster than with original implementation, but for other frequencies the calculation is more than 5 times faster. The difference in prediction accuracy between original and new proposed implementation has been also evaluated and is negligible.

The third result of this research was implementation of classical holography algorithm with double layer array measurement and its optimization with similar approaches used in single layer holography to predict sound field near examined source if there is another disturbing source which affects the measured sound field, thus making prediction with single layer impossible. The procedure separates sound field coming from different sides of the measurement area (measurement planes) and using only partial field for reconstruction. This partial field is related only to the examined structure. The algorithm has been optimized for handling noisy data and simulated measurement certify this procedure for separation of sound field and applicability for reconstruction in reverberant fields. This method has been lately extended to measure entering sound field components. It means when the acoustic parameters of the examined source surface is not known, the method tries to estimate these parameters. After successful estimation, separation of sound field can be processed and calculation of entering sound field (sound energy produced with the examined source) can be done.
For large scale measurement, the procedures for combination of each subsequently measured hologram area ("patch") into large one have been also proposed. One procedure combine patches in spatial domain, another one in wavenumber domain. Both methods have been tested on simulated measurement with good results.

Finally, the measurement system for near-field acoustic holography with linear array and with smart, but not expensive, acoustic sensors has been designed and constructed. It uses common electret microphones, built-in preamplifier and electronic memory for storage of TEDS. The system can correct measured values based on information stored in TEDS (for each sensor), thus reduces imperfections of the transducers and amplifiers, as it is strictly required with acoustic holography methods. Also development of matrix microphone array has been successfully preformed. In this case, digital MEMS microphones have been used as a important part of the measurement path. Their main advantages are small size, sufficient sensitivity and digital interface with possibility of sharing one transmission line with two microphones simultaneously.

With similar system with linear array as proposed with this thesis (the older one), the acoustic holography method has been experimentally certified as a possible vibration analysis tool with its main advantages of fast and non-contact measurement. The experimental modal analysis of steel beam mounted on the shaker has been carried out with good results, where positions of nodes and antinodes on the beam surface can be easily determined.

REFERENCES


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ABSTRACT

The main aim of the thesis is application of near-field acoustic holography for non-contact vibration analysis. Near-field acoustic holography is an experimental technique for reconstruction of sound field close to the surface of the vibrating object based on measurement of sound pressure or acoustic particle velocity in certain distance from the examined object. Practical realization of this method depends on used calculation procedure.

The thesis is focused on analysis of acoustic holography algorithms with transformation into wavenumber domain (spatial transformation) where the reconstruction of the sound field near vibrating object is calculated. The introductory part of the thesis describes the theory of near-field acoustic holography with general characteristics and with analysis of most common algorithms used for localization and characterization of sound source and consequent vibration analysis. Principal part of the thesis deals with advanced processing methods where these methods try to optimize the accuracy of prediction of sound field near vibrating object in real environment. In this study, real measurement conditions represent the measurement system with non-ideal acoustic sensors and also areas with reverberant sound field. Based on literature study, there has been optimized and verified the new method which uses double layer microphone array to separate incoming and outgoing sound field, thus allows successful measurement in confined space e.g. cabins of cars and airplanes. Part of the thesis has been also focused on optimization, extension and successive experimental validation of selected classical algorithms published in last decade for possible measurement in real conditions and with common acoustic sensors.
ABSTRAKT

Disertační práce se zabývá bezkontaktní analýzou vibrací pomocí metod akustické holografie v blízkém poli. Akustická holografie v blízkém poli je experimentální metoda, která rekonstruuje akustické pole v těsné blízkosti povrchu vibrujícího předmětu na základě měření akustického tlaku nebo akustické rychlosti v určité vzdálenosti od zkoumaného předmětu. Konkrétní realizace této metody závisí na použitím výpočetního algoritmu.

Vlastní práce je zaměřena zejména na rozbor algoritmů, které využívají k rekonstrukci zvukového pole v blízkosti vibrujícího objektu transformaci do domény vlnových čísel (prostorová transformace), kde probíhá vlastní výpočet. V úvodu práce je vysvětlena základní teorie metody akustické holografie v blízkém poli s popisem základních vlastností a dále rozborem konkrétních nejčastěji používaných algoritmů pro lokalizaci a charakterizaci zdroje zvuku a pro následnou vibrační analýzu. Stěžejní část práce se věnuje pokročilým metodám zpracování, které se snaží určitým způsobem optimalizovat přesnost predikce zvukového pole v blízkosti vibrujícího předmětu v reálných podmínkách. Jde zejména o problematiku použitího měřicího systému s akustickými snímači, které nejsou ideální, a dále o možnost měření v prostorách s difúzním charakterem zvukového pole. Pro tento případ byla na základě literárního průzkumu optimalizována a ověřena metoda využívající dvouvrstvé mikrofonní pole, která umožňuje oddělení zvukových polí přicházejících z různých stran a tedy úspěšné měření v uzavřených prostorách např. kabin automobilů a letadel. Součástí práce byla také optimalizace, rozšíření a následné experimentální ověření algoritmů publikovaných v posledních letech pro měření v reálných podmínkách a za použití běžně dostupných akustických snímačů.