НАЦІОНАЛЬНИЙ ТЕХНІЧНИЙ УНІВЕРСИТЕТ УКРАЇНИ
«КИЇВСЬКИЙ ПОЛІТЕХНІЧНИЙ ІНСТИТУТ
імені ІГОРЯ СІКОРСЬКОГО»
Факультет електроніки
Кафедра акустики та акустоелектроніки

«На правах рукопису»
УДК 534.75

«До захисту допущено»
Завідувач кафедри
_______ Дідковський В.С.
(підпис) (ініціали, прізвище)
"___"_____________2019 р.

Магістерська дисертація

зі спеціальності (спеціалізації) 171 Електроніка
(код і назва спеціальності)

на тему: Active Noise Control at low frequencies for Open Air events

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3. Об'єкт дослідження Активний контроль та оптимізація акустичного поля практично застосовані до гучномовців для контролю шуму в умовах Open Air подій.

4. Предмет дослідження (Вихідні дані – для магістерської дисертації за освітньо-професійною програмою) Існуючі методи та алгоритми активного контролю акустичного поля; методи створення акустичних зон; методи прямого та ітеративного розрахунку коефіцієнтів фільтру; оцінка впливу навколишнього середовища та атмосферних умов на стійкість та надійність розглянутих методів і алгоритмів.

5. Перелік завдань, які потрібно розробити

1. Дослідити існуючі методи активного контролю шуму та створення звукових зон.

2. Дослідити вплив навколишнього середовища та атмосферних умов на стійкість та надійність розглянутих методів і алгоритмів для контролю шуму в умовах Open Air подій.

3. Моделювання розглянутих методів і алгоритмів в середовищі МАТЛАБ. Розрахунок коефіцієнтів фільтру для гучномовців.

4. Порівняння розглянутих методів та вибір оптимального методу для заданої конфігурації та геометрії контрольованої області.

6. Перелік графічного (ілюстративного) матеріалу______________________

7. Дата видачі завдання ____

Календарний план

№ з/п	Назва етапів виконання магістерської дисертації	Строк виконання етапів магістерської дисертації	Примітка
1	Дослідити існуючі методи активного контролю шуму та створення звукових зон.		
2	Дослідити вплив навколишнього середовища та атмосферних умов на стійкість та надійність розглянутих методів і алгоритмів для контролю шуму в умовах Open Air подій.		
3	Моделювання розглянутих методів і алгоритмів в середовищі МАТЛАБ.		
4	Порівняння розглянутих методів та вибір оптимального методу для заданої конфігурації та геометрії контрольованої області.		

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ABSTRACT

Number of pages: 80, figures: 31, tables: 0, bibliography sources: 19

Every year more and more open-air events and large concerts are held in the city, near residential areas. The main problem of this is noise pollution in nearby areas. On the other hand, an integral part of such concerts is a high SPL. Creating sound zones is one possible solution to this problem.

In this paper, we examine and characterize the main methods of creating sound zones: Acoustic Contrast, Pressure Matching and combined method. The classical method of active noise control based on the LMS algorithm is also considered. For all methods, final expressions are derived for calculating the optimal complex volume velocities of the loudspeakers. The meaning of regularization parameters of these methods is described.

To calculate the sound zones, measured or modeled propagation transfer functions are used. The effect of the environment and atmospheric conditions is studied in this work and their impact is evaluated.

Simulations of all methods were performed at MATLAB. Comparison of the results was carried out on the calculated performance metrics and on the frequency response of the calculated optimal weights. The importance of the regulation parameters has been shown when simulating various methods. The optimal method was chosen for the system under study.

In general, this work aimed to make a comparison and search for the optimal method for creating sound zones, which is used to control large zones within an open air event; and also make an assessment of the possible influence of atmospheric conditions on the accuracy and robustness of these methods.

Key words: electroacoustic, sound zones, active noise control, sound field control, acoustic contrast, pressure matching, cost function, Lagrangian multipliers, audience, weather conditions, sound propagations, low frequencies, loudspeaker, open air events.

РЕФЕРАТ

Сторінок: 80, рисунків: 31, таблиць: 0, літературних джерел: 19

Щороку в місті, поблизу житлових районів, проводяться все більше і більше open air заходів та великих концертів. Головною проблемою цього є шумове забруднення прилежних районів. З іншого боку, невід'ємною частиною таких концертів є високий рівень звукового тиску. Створення звукових зон є одним з можливих шляхів вирішення цієї проблеми.

У даній роботі ми розглянемо та охарактеризуємо основні способи створення звукових зон: акустичний контраст, pressure matching та комбінований метод. Розглянуто також класичний метод активного контролю шуму на основі алгоритму найменших квадратів. Для всіх методів отримані кінцеві вирази для розрахунку оптимальних комплексних об'ємних швидкостей гучномовців. Описано значення параметрів регуляції цих методів.

Для обчислення звукових зон використовуються виміряні або змодельовані передатні характеристики розповсюдження звуку. У цій роботі вивчається вплив навколишнього середовища та атмосферних умов.

Моделювання всіх методів проводилося у середовищі MATLAB. Порівняння результатів проводилося за розрахунковими показниками продуктивності та за частотною характеристикою розрахованих оптимальних ваг. Важливість параметрів регулювання показана при моделюванні різних методів. Для досліджуваної системи обраний оптимальний метод.

В цілому метою цієї роботи є порівняння та пошук оптимального методу створення звукових зон, який використовується для управління великими зонами в межах open air події; а також зробити оцінку можливого впливу атмосферних умов на точність і надійність цих методів.

Ключові слова: електроакустика, звукові зони, активний контроль шуму, управління звуковим полем, акустичний контраст, узгодження тиску, функція витрат, Лагранжові множники, слухачі, погодні умови, поширення звуку, низькі частоти, гучномовець, open air події.

SUMMARY

USED ACCRONYMS AND ABBREVIATIONS

PA – Public address system

ANC – Active noise control

SPL – Sound pressure level

LMS – least mean square

PM – Pressure Matching

ACC – Acoustic Contrast Control

INTRODUCTION

Recently, more and more attention has been paid to issues of noise pollution and its negative impact on quality of life. Sometimes only daily life in a large metropolis can be a source of great noise pollution, but what about numerous outdoor events held in the city, the number of which has increased significantly over the past decade. These open-air events are important noise sources in urban spaces. Despite established noise limits in different countries, according to a recent report [1], there are many complaints from citizens due to large noise pollution from outdoor events. A striking example is one of the largest festivals in the world Ultra Music Festival, which is held annually for several days in the center of Miami. At the same time, a large number of complaints from the urban population recorded every year. [2]

When it comes to open air events the main source of noise pollution are low frequencies (up to 100-300 Hz) from concert venue's PA systems. Unlike high frequencies, due to low attenuation in the atmosphere and big wavelength, low frequencies can diffract over large obstacles and propagate over great distances. It is also worth mentioning that popular music and movie spectral balances have evolved in the last decades: the low frequency spectral components have increased. [3]

Within the framework of noise regulation, a problem often arises when reducing the sound level in a concert venue in order to reduce noise pollution to allowed limit, sound imbalance may occur and the audio experience can degrade, because high SPL is an essential part of the concert performance, especially for some sound genres. To avoid this problem and to reduce noise pollution, sound field control or sound zones principles and its modifications can be used. The main idea of these methods is to produce significant SPL differences between control areas, in our case concert venue with audience and residential area behind it, by using secondary sources (control sources) usually placed at the end of the venue, opposite to the main PA system near the stage.

Originally sound zone principle was developed and introduced for creation multiple small loud and quiet sound zones for multiple listeners within a room or comparable in size areas [4]. Control loudspeakers could be installed around the perimeter of the room or assembled into a single array. Required information for the algorithm that calculates complex volume velocities at each frequency is transfer functions from each loudspeaker to each sampled point in sound zones. In case of creating indoor sound zones, these transfer functions can be easily measured or modeled, since there is no great influence from changing atmospheric conditions over a time, propagation distance are relatively small and, in general, acoustic conditions are unchanged with time. There are only numerous reflections from the walls and, as a result, the frequency-dependent reverberation time, which in general can be nullify using active room compensation technics [5, 6].

When referring to creating sound zones in size of open-air events, propagation transfer functions not only hardly to measure (a large number of measurements, changing atmospheric conditions over time) but also difficult to model accurately: influence by the atmospheric conditions and their local change over time, the reflections from the ground, and the propagation of sound through the audience. Therefore, the issue of creating large sound zones within the framework of open-air events is an open and relevant issue requiring further research.

Nowadays there are two companies (that has been found) which have implemented and tested active noise control systems for open air events: MONICA project of EU in collaboration with Technical University of Denmark (DTU) [7-10] and Rocket Science company [11]. Project by Rocket Science is more closed and commercially sensitive, when MONICA project is more open and several articles about its system implementation had been published.

Active noise control, sound zones for open air events and environment effects with their influence on it have been studied partially in articles [7-9]. As for the influence of the audience, this effect has not been taken into account during the development of existing systems. Also at the moment there is no software product that would allow simulating ANC system or different methods of sound zones taking into account all the above factors. Moreover, many algorithms and methods must be tested to find the most appropriate one for a particular problem.

In view of the above and taking into account importance of the problem today, the idea of this master's thesis and its main goals were defined.

1. ANC AND SOUND ZONES TECHNICS OVERVIEW

1.1 Active control of acoustic radiation using multiple primary and control sources

The method presented below is a classical approach for ANC and it is based on LMS error minimization [12]. The main aim is to reduce noise only at N_e error sensing locations using control sources. Thus, we can only define Dark zone in which we want to reduce noise, which is not applicable for our case - open-air events where we want to define and control Bright zone - zone with listeners and maximal audio quality and reducing SPL in a Dark zone. However, this method is worth considering as it forms the basis for the more complex methods described below. This method has proven itself very well and is used in headphones with active noise cancellation, and to reduce the noise level inside an airplane cabin caused by its engines.

The main principle of active noise cancelling is based on superposition of two pressures with the same amplitude but opposite in phase (180 deg.). So the main idea is to create the same wave front of primary sources but opposite in phase by using control (secondary) sources to cancel out the noise at the error locations (dark zone). Figure 1 represents the basic propagation process described above.

The error location point is the point where cancellation is supposed to happen. The pressure $p_e(r_i)$ in this point can be expressed as [12, pp. 824-854]:

$$p_e(r_i) = \sum_{m=1}^{N_p} q_{p,m} z_{p,m}(r_i) + \sum_{n=1}^{N_c} q_{c,n} z_{c,n}(r_i) \tag{1}$$

Where $q_{p,m}$ are the volume velocities of the N_p primary sources, when $q_{c,n}$ and N_c are respectively the same for control (secondary) sources.

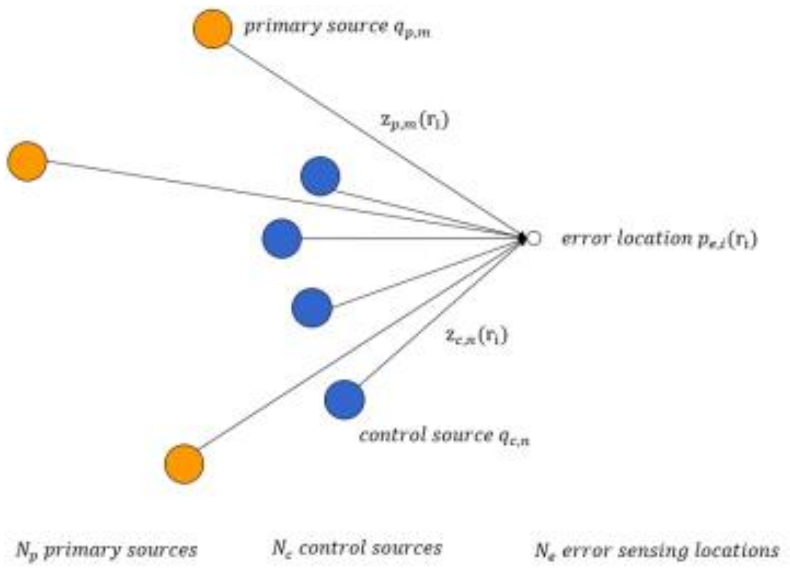


Figure 1. The sum of primary and secondary (control) sources in a point of a dark zone [9].

The propagation model based on monopole radiation, the transfer functions from primary sources to error location point is equal to:

$$z_{p,m}(r_i) = \frac{j\omega\rho_0 e^{jk|r_i - r_{p,m}|}}{4\pi|r_i - r_{p,m}|} \tag{2}$$

And respectively for transfer functions from control sources to the point in a dark zone $z_{c,n}(r_i)$. Equation (1) can be rewritten in a matrix form for all points inside the dark zone (all error location points):

$$p_e = Z_p q_p + Z_c q_c \tag{3}$$

Where, in this case, Z_p and Z_c are matrices, containing transfer function from each source to every point in a dark zone, and respectively q_p and q_c are the vectors of volume velocities.

Our aim in this task is to minimize the sound pressure at error sensing locations, or we can minimize the sum of the squared acoustic pressure amplitudes (in a matrix form):

$$\sum_{i=1}^{N_e} |p(\boldsymbol{r}_i)|^2 = \boldsymbol{p}_e^{\mathrm{H}} \boldsymbol{p}_e \tag{4}$$

Using eq. 3, this can be expanded as:

$$\boldsymbol{p}_e^{\mathrm{H}} \boldsymbol{p}_e = [\boldsymbol{Z}_p \boldsymbol{q}_p + \boldsymbol{Z}_c \boldsymbol{q}_c]^{\mathrm{H}} [\boldsymbol{Z}_p \boldsymbol{q}_p + \boldsymbol{Z}_c \boldsymbol{q}_c]$$
$$= \boldsymbol{q}_c^{\mathrm{H}} \boldsymbol{Z}_c^{\mathrm{H}} \boldsymbol{Z}_c \boldsymbol{q}_c + \boldsymbol{q}_c^{\mathrm{H}} \boldsymbol{Z}_c^{\mathrm{H}} \boldsymbol{Z}_p \boldsymbol{q}_p + \boldsymbol{q}_p^{\mathrm{H}} \boldsymbol{Z}_p^{\mathrm{H}} \boldsymbol{Z}_c \boldsymbol{q}_c + \boldsymbol{q}_p^{\mathrm{H}} \boldsymbol{Z}_p^{\mathrm{H}} \boldsymbol{Z}_p \boldsymbol{q}_p \tag{5}$$

The cost function (error criterion) which we want to minimize expressed as:

$$J_{prs} = \boldsymbol{p}_e^{\mathrm{H}} \boldsymbol{p}_e = \boldsymbol{q}_c^{\mathrm{H}} \boldsymbol{A}_{prs} \boldsymbol{q}_c + \boldsymbol{q}_c^{\mathrm{H}} \boldsymbol{b}_{prs} + \boldsymbol{b}_{prs}^{\mathrm{H}} \boldsymbol{q}_c + c \tag{6}$$

where

$$\boldsymbol{A}_{prs} = \boldsymbol{Z}_c^{\mathrm{H}} \boldsymbol{Z}_c$$

$$\boldsymbol{b}_{prs} = \boldsymbol{Z}_c^{\mathrm{H}} \boldsymbol{Z}_p \boldsymbol{q}_p$$

and

$$c = \boldsymbol{q}_p^{\mathrm{H}} \boldsymbol{Z}_p^{\mathrm{H}} \boldsymbol{Z}_p \boldsymbol{q}_p \tag{7}$$

The subscript *prs* denotes pressure minimization. In case of one error location, one primary and control sources plot of the cost function (squared acoustic pressure amplitude) as a function of the real and imaginary parts of the control source volume velocity generate a 'bowl' (Figure 2), where the optimum control source volume velocity that will minimize the squared acoustic pressure amplitude is the bottom of this bowl. When dealing with multiple error location and sources plot of the cost function J_{prs} will be a $(2N_c +1)$ – dimensional hyper-paraboloid. [12]

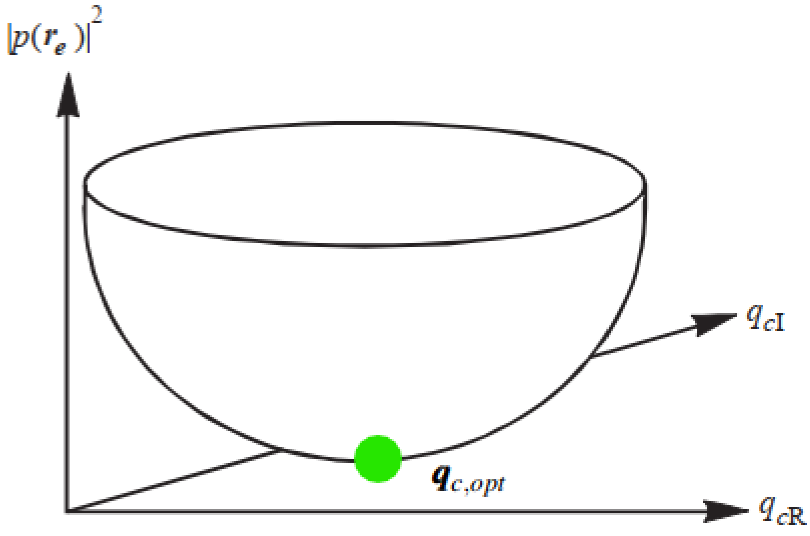


Figure 2. Plot of squared acoustic pressure amplitude as a function of the real and imaginary parts of control source volume velocity. Case for one error sensing location, one primary and control sources. [12, pp 824-854]

In order to find the vector of optimal control source strengths, which results in the smallest possible pressures in the error locations, the cost function J_{prs} is now derived with respect to the control source strengths and set to 0. Differentiation should be performed on the real and imaginary parts separately, which gives:

$$\frac{\partial J_{prs}}{\partial \boldsymbol{q}_{c_{\mathrm{R}}}} = 2\boldsymbol{A}_{prs_{\mathrm{R}}}\boldsymbol{q}_{c_{\mathrm{R}}} + 2\boldsymbol{b}_{prs_{\mathrm{R}}} = 0$$

and

$$\frac{\partial J_{prs}}{\partial \boldsymbol{q}_{c_{\mathrm{I}}}} = 2\boldsymbol{A}_{prs_{\mathrm{I}}}\boldsymbol{q}_{c_{\mathrm{I}}} + 2\boldsymbol{b}_{prs_{\mathrm{I}}} = 0 \tag{8}$$

Combining real and imaginary parts from eq.8 gives the following expression for the optimum vector of control source volume velocities:

$$\boldsymbol{q}_{c,opt} = -\boldsymbol{A}_{\boldsymbol{p}}^{-1}\boldsymbol{b}_{\boldsymbol{p}} \tag{9}$$

To calculate the total pressure in any point in space, the optimal source strengths $\boldsymbol{q}_{c,opt}$ from eq. 9 can now be inserted into eq. 3. It is important to note that since we only control error sensing locations, the total pressure from the primary and control sources in uncontrolled areas (points) can be significantly amplified due to interference.

To avoid the matrix $\boldsymbol{A}_{prs}$ being singular resulting in an infinite number of 'optimal' control source strength $\boldsymbol{q}_{c,opt}$, the number of error locations must be at least as many as there are control sources. Equal number of control sources and error locations will result in nulling the acoustic pressure at the error locations. In case of more error locations than control sources, acoustic pressure at the error locations will be reduced but not as much as in the case of equal number of control sources and error locations.

Modeling of this method presented in the following sections.

1.2 Multizone sound control

In general, problem of creating multiple sound zones can be formulated as follows: we want to reproduce different sound fields (quiet and loud zones) over Q sound zones at which M pressure controlling microphones are placed. In total we have QM controlling points (Figure 3).

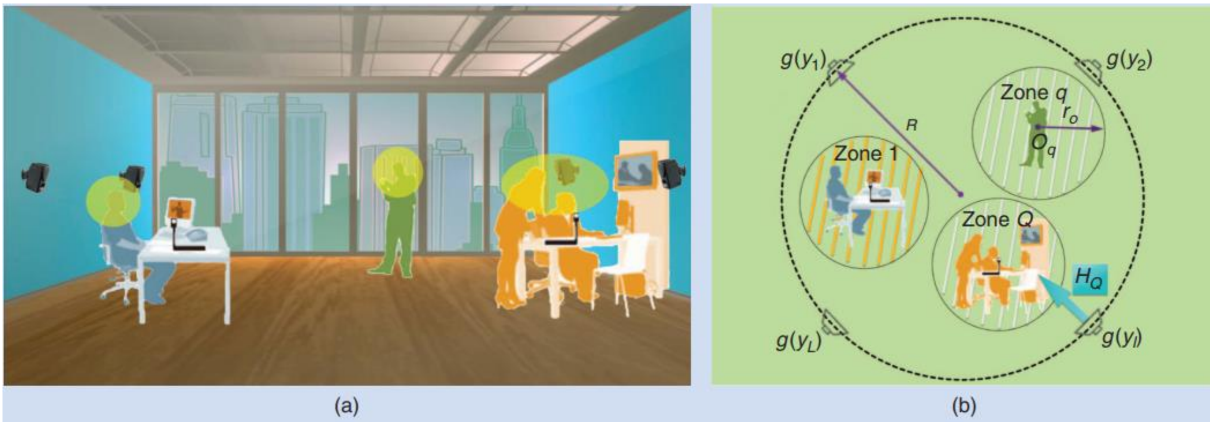


Figure 3. (a) - An illustration of personal sound zones in an entertainment room; (b) Sketch of sound zone problem formulation [5].

Measured sound pressures at the microphones positions in each zone q can be presented as a vector [5]:

$$\boldsymbol{p}_q = [p\,(\boldsymbol{x}_{q,1},\omega),\ldots,p\,(\boldsymbol{x}_{q,M},\omega)]^T \tag{10}$$

And calculated as:

$$\boldsymbol{p}_q = \boldsymbol{H}_q\boldsymbol{g} \tag{11}$$

Where

$$\boldsymbol{g} = [g\,(\boldsymbol{y}_1,\omega),\ldots,g\,(\boldsymbol{y}_L,\omega)]^T \tag{12}$$

Are the vector of loudspeaker volume velocities (or driving signal) at frequency ω which are used to create these sound zones and *Hq* is a matrix of acoustics transfer functions between loudspeakers and microphones in zone *q*.

Nowadays there are two main approach/methods of sound field control and sound zones reproductions: Acoustic Contrast Control (ACC) and Pressure Matching (PM). There are also their modifications and combinations that can be attributed to the third combined ACC – PM method. The following sections examine these methods in detail.

1.2.1 Acoustic contrast control

ACC was firstly formulated by Choi and Kim [4] in terms of creating two kinds of sound zones, the bright zone where we aim to create sound with high acoustic energy, and the dark zone, where low level of acoustic energy is desired. In other words, this method concentrate sound energy in a bright zone and at the same time reduce the sound energy in a dark zone, thus the acoustic contrast is maximized between these two zones. Usually the problem is set by having one bright zone and one or several dark zones.

The acoustic energy in the bright zone is defined from the sound pressures measured at the M matching points:

$$E_{\mathrm{b}} = \| \boldsymbol{p}_{\mathrm{b}} \|^2 = \| \boldsymbol{H}_{\mathrm{b}} \boldsymbol{g} \|^2 \tag{13}$$

With || . || denoting *l2* or Euclidean norm.

At the same time, the acoustic energy in the dark zone:

$$E_{\mathrm{d}} = \| \boldsymbol{p}_{\mathrm{d}} \|^2 = \| \boldsymbol{H}_{\mathrm{d}} \boldsymbol{g} \|^2 \tag{14}$$

Where *Hb* and *Hd* are matrixes of acoustics transfer functions between loudspeakers and microphones in bright and dark zone respectively.

Choi and Kim defined Acoustic contrast as a ratio between the average acoustic potential energy density produced in the bright zone to that in the dark zones, is maximized. [4]

Level in dB of Acoustic contrast can be calculated as follows:

$$AC = 10\log_{10}\left(\frac{M_d P_b^H P_b}{M_b P_d^H P_d}\right) \tag{15}$$

Where M_d/M_b is the ratio that normalizes the result for different numbers of microphones in the bright and dark zones and the superscript H denotes the Hermitian, complex conjugate, transpose.

The acoustic contrast maximizing method may perform well over the dark zones but may be unrobust to providing the desired maximum energy in the bright zone [5]. In this case, derived loudspeaker strength cannot be realized in the real world, for example big enormous or very small magnitude of filters. To avoid this problem and to ensure the sound energy within different zones are optimized simultaneously, the problem can be reformulated as maximizing the acoustic energy in the bright zone with the constraint that the energy in the dark zone is limited to a very small value D_0 [5].

Additionally another constrained is set to the loudspeaker power consumption, it is limited with value E_0. This is known as array effort.

These constraints ensure that sound leakage outside the Q zones not excessive and that realized loudspeaker weights are chosen to ensure the implementation is robust to driver positioning errors and changes in the acoustic environment. [5]

Finally, the ACC problem can be formulated as follows:

$$\begin{aligned} &\max_{g} \| \boldsymbol{H}_{\mathrm{b}} \boldsymbol{g} \|^2 \\ &\text{subject to} \| \boldsymbol{H}_{\mathrm{d}} \boldsymbol{g} \|^2 \leq D_0 \\ &\| \boldsymbol{g} \|^2 \leq E_0 . \end{aligned} \tag{16}$$

This constrained optimization problem can be solved “directly” by using MATLAB Optimization toolbox for which we need to put numerical values for D_0 and E_0 or can be solved in more analytical way by rewriting objective (1st expression in eq. 16) and constrains into a single objective function using the Lagrangian technics [5]:

$$\begin{aligned} &\max_{g} L_c(\boldsymbol{g}) = \| \boldsymbol{H}_{\mathrm{b}} \boldsymbol{g} \|^2 - \lambda_1 (\| \boldsymbol{H}_{\mathrm{d}} \boldsymbol{g} \|^2 - D_0) - \lambda_2 (\| \boldsymbol{g} \|^2 - E_0), \\ &\lambda_1, \lambda_2 \geq 0, \end{aligned} \tag{17}$$

The advantage of use Lagrangian multipliers is to adjust the importance of constrains without specifying different numerical values for constrained and solving the problem each time.

In this case λ_1 and λ_2 adjust the relative importance of each constraint. The solution that maximizes the cost function is obtained at the same way as in section 1.1: the derivative is taken with respect to q (real and imaginary parts separately) and equating it to zero. After these steps the solution recognized as a generalized eigenvector problem:

$$\lambda_1 [H_{\mathrm{d}}^H H_{\mathrm{d}} + \frac{\lambda_2}{\lambda_1} I] g = [H_{\mathrm{b}}^H H_{\mathrm{b}}] g \tag{18}$$

The optimum source strength vector q_c is set as the eigenvector corresponding to the maximum eigenvalue of the matrix:

$$[H_d^H H_d + (\lambda_2/\lambda_1)I]^{-1}[H_b^H H_b] \tag{19}$$

The ratio of Lagrange multipliers λ_2 / λ_1 determines the tradeoff between the performance and array effort and must be chosen iteratively for the constraint on the array effort to be satisfied. [5]

1.2.2 Pressure matching method

The PM method aims to reproduce a desired (target) sound field in the bright zone, while producing silence in other zones.

The first difference from ACC that we introduce a desired target sound field in a bright zone, which we want to match (to reproduce). Then PM formulation can be written using the same *l2* norm for the objective and the same constraints:

$$\begin{aligned} &\min_g \| H_b g - p_{des} \|^2 \\ &\text{subject to} \| H_d g \|^2 \le D_0 \\ &\| g \|^2 \le E_0. \end{aligned} \tag{20}$$

Then the problem can be written as a Lagrangian cost function:

$$\begin{aligned} &\min_g L_p(g) = \| H_b g - p_{des} \|^2 + \lambda_1 (\| H_d g \|^2 - D_0) + \lambda_2 (\| g \|^2 - E_0) \\ &\lambda_1, \lambda_2 \ge 0. \end{aligned} \tag{21}$$

In this case we are interested to minimize the difference between reproduced and desired fields, so the solution that minimizes cost function is obtained by taking derivative of real and imaginary parts of *Lp* with respect to *q* and setting it to zero we can get the solution:

$$[H_b^H H_b + \lambda_1 H_d^H H_d + \lambda_2 I]g = H_b^H p_{des} \tag{22}$$

Here the meaning of Lagrangian multipliers are the same. When λ_1 =1 we apply equal effort to matching the pressure in the bright zone and minimizing the energy in the dark zone and the solution becomes:

$$g_p = [H_b^H H_b + H_d^H H_d + \lambda_2 I]^{-1} H_b^H p_{des} \tag{23}$$

The PM approach gives an explicit solution to obtain the loudspeaker driving signals and does not require solving an eigenvector problem, as is required in the case of acoustic contrast optimization. PM is especially suitable for the situation that different constraints are imposed on each sound zone when the listeners require different quality of listening experiences. However a series of Lagrange multipliers need to be determined, and a generalized eigenvalue solution is no longer possible [5].

1.2.3 Combined PM-ACC method

As we seen above both ACC and PM methods control all the sources. When it comes to open air events usually for practical reasons it is better not to change a radiation from primary array. For this purpose a modification and combination of ACC and PM methods has been introduced in [8]. There, the primary weights w_p are set as constant and the target (desired) transfer function in a bright zone set as the transfer-function of the primary system:

$$\mathbf{h}^t := \mathbf{H}_B^p \mathbf{w}^p \tag{24}$$

The cost function is written as follows:

$$\min_{\mathbf{w}^s} \kappa \|\mathbf{H}_B^s \mathbf{w}^s\|^2 + (1-\kappa) \left\|\mathbf{H}_D^s \mathbf{w}^s + \mathbf{h}_D^p\right\|^2 + \lambda \|\mathbf{w}\|^2 \tag{25}$$

Where a Tikhonov regularization term with parameter λ has been added.

And

$$\mathbf{h}_D^p := \mathbf{H}_D^p \mathbf{w}^p \tag{26}$$

Here the superscripts p and s are the primary and secondary sources respectively, and subscripts D and B mean Dark and Bright zones.

This cost function minimizes the radiation of the secondary sources into the bright zone and the total sound energy in the dark zone. The optimization problem can be efficiently solved by rewriting it as a linear least squares problem [8, 18]:

$$\min_{\mathbf{w}^s} \|\mathbf{A}\mathbf{w}^s - \mathbf{b}\|^2 \tag{27}$$

Where

$$\mathbf{A} = \begin{bmatrix} \sqrt{\kappa}\mathbf{H}_B^s \\ \sqrt{1-\kappa}\mathbf{H}_D^s \\ \sqrt{\lambda}\mathbf{I} \end{bmatrix} \quad \text{and} \quad \mathbf{b} = \begin{bmatrix} 0 \\ -\sqrt{1-k}\mathbf{h}_D^p \\ 0 \end{bmatrix} \tag{28}$$

The purpose of the regularization term is two-fold: firstly, it enables us to solve the possibly ill-posed inverse problem by making the solution robust against noise in the measured transfer-functions. Secondly, it smoothly distributes the array effort over the control loudspeakers and limits the magnitude of the resulting control gains. This way realizable solutions, i.e. secondary sources play in their linear range, can be found by tuning λ. As a recommendation, λ should be chosen such that the maximum gain of the filters is approximately 0 dB and there is no large deviation of the filter's magnitude over the frequency range.

Once any of above optimization problem is solved for all relevant frequencies, we get the complex gain (volume velocities) of the loudspeakers as a function of frequency. The discrete Fourier transform of the frequency domain filters is a set of FIR filters. The loudspeaker array driving signals may be obtained by real time convolution of the audio signal with the corresponding FIR filters.

1.3 Calculation methods: Direct and Iterative.

This section explaines ways to calculate optimal source strength and difference between feedforward and feedback systems.

There are two fundamental ways in ANC for calculating and solving problem: Direct and Iterative calculations.

Under the direct calculation all the required transfer functions are simulated or measured before calculation and then having a set of input data for algorithm the optimal source strength are calculated.

Iterative calculations means performing adaptive calculations and updating calculated source strength in real time. For large controlling areas (sound zones) this requires a lot of microphones, DSP channels and processing large amount of data which significantly increased complexity and unrobustness of the system. In the other hand real time implementation for a small set of control points and controlling only dark zone has main advantage in measuring and taking into account the real environment and atmospheric conditions. In the frame of open-air events attempt of creating a local small sound zone has been made [9].

The ANC systems may be divided in two categories: feedforward and feedback. A schematic diagram of a typical implementation of each of these controller types is shown in Figure 4. Each acts to suppress the noise generated by some source, referred to here as the primary source [12].

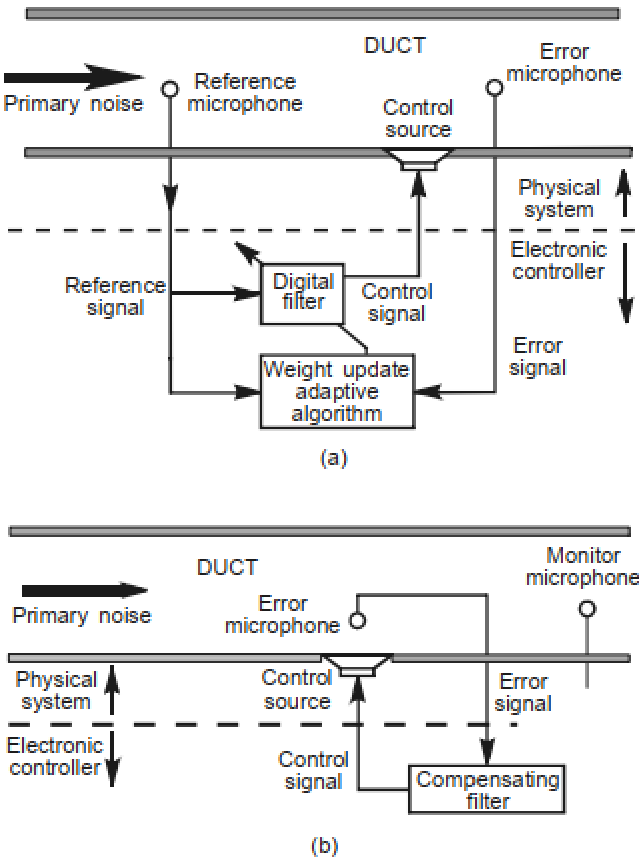


Figure 3. Schemes for active control of plane waves propagating in ducts: (a) feedforward with a microphone signal for the controller reference input; (b)feedback [12]

Feedforward controllers, require a reference signal, which is a measure of the incoming disturbance (noise or vibration). Reference signal must be received by the controller in sufficient time for the required control signal to be generated and output to the control source when the disturbance (from which the reference signal was generated) arrives.

Feedback control systems differ from feedforward systems in the manner in which the control signal is derived. Whereas feedforward systems rely on some predictive measure of the incoming disturbance to generate an appropriate ‘cancelling’ disturbance, feedback systems aim to attenuate the residual effects of the disturbance after it has passed. Feedback controller derives a control signal by filtering an error signal, not by filtering a reference signal as is done by a feedforward controller.

For free-field, random sound sources, there are no known physical mechanisms that would allow global control (either feedback or feedforward) using acoustic sources, although it is possible to achieve local zones of cancellation which are generally at the expense of increased levels elsewhere [12].

Based on the foregoing, a more realistic and suitable solution to the problem of reducing the noise level in terms of open events is the use of a direct calculation method and one of the existing algorithms (ACC, PM or ACC-PM) with found optimal regulation parameters (Lagrangian multipliers). However, as mentioned above, the main problem is to accurately model or measure the propagation transfer functions and take into account the influence of the environment and atmospheric conditions, possible effect of which is estimated below.

2. ASSESSMENT OF ENVIRONMENTAL AND ATMOSPHERIC CONDITIONS INFLUENCES

In this section, the influence of environment and atmospheric condition on sound propagation is examined. In view of the influence from environment means the influence of the presence of listeners on the propagation of sound. This effect is not particularly studied when dealing with creation of sound zones, although it can significantly affect the robustness and performance of the method, as it directly contributes to the propagation transfer functions. In addition, the reflection from the ground and its influence will be studied. But firstly the system under study is defined.

2.1 System under study

The system under study (Figure 4) is an open air venue with audience of 20 m wide and 70-75 m long, behind which there is a residential area at a distance of 150 m. By using an array of 11 primary sources - subwoofers (array of circles near the stage) and 11 secondary ones (right array of circles) it is necessary to create a dark zone (red area) at residential area and meanwhile ensure a uniform sound field at the bright zone – audience (green) area.

The distance between subwoofers is 1.5 m. The frequency range of interest is about 30 - 100 Hz. The used subwoofers are the V-SUB cardioid subwoofers from d&b audiotechnik, directivity plots are presented at the Figure 5.

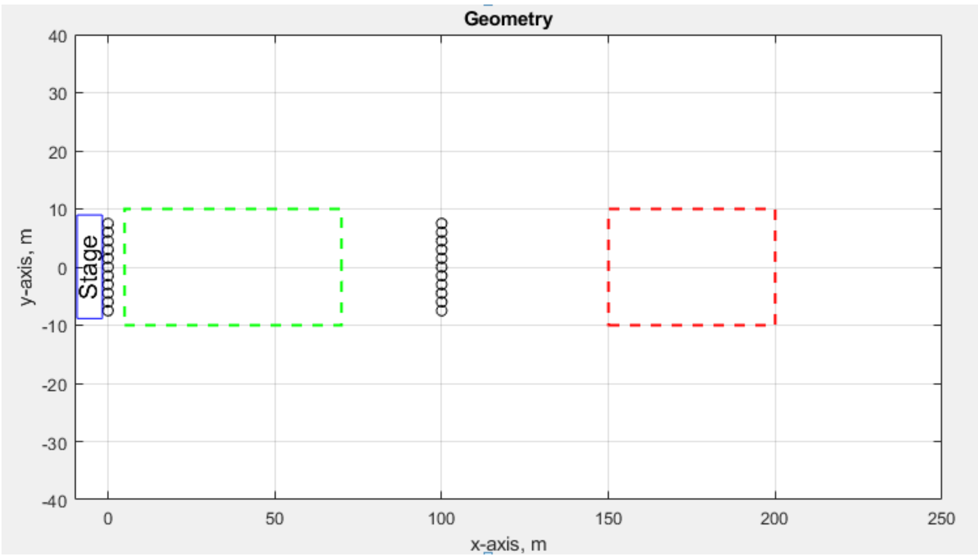


Figure 4. Geometry of the system under study

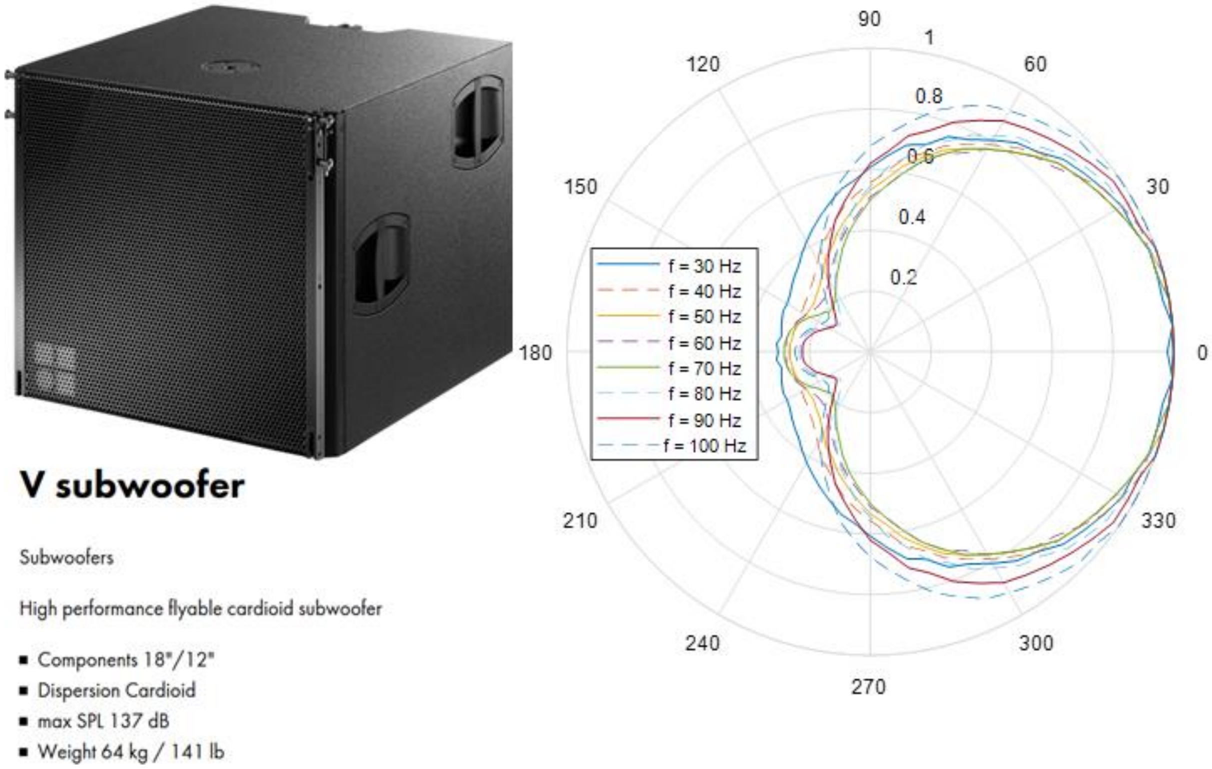


Figure 5. V-SUB measured horizontal directivity plots

2.2 Crowd effect or propagation of sound through the audience

2.2.1 A one-dimensional model calculation

A one-dimensional model of an audience has been presented in [13]. The main idea of the model that the audience represented as a homogeneous medium with a complex speed of sound and complex density. The audience is modeled as infinitely long hard cylinders (Figure 6.), where acoustic impedance can be estimated using a porous medium model with the concentration of the audience as an input parameter. The model is described in detail in its original source.

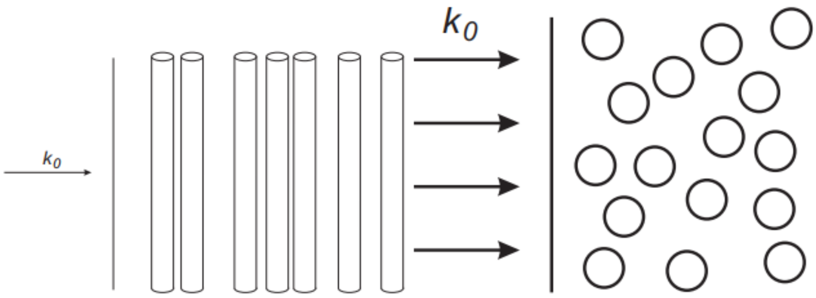


Figure 6. Plane wave propagation through a set of cylinders [13]

Equation 28 represents the speed of sound and density inside the medium – crowd with concertation (number of people per m^2).

$$\rho_v = \rho_0\left(\frac{\xi}{h} - i\frac{\sigma}{\omega\rho_0}\right)$$
$$c_v = \frac{c_0}{\sqrt{\xi - i\dfrac{\sigma h}{\omega\rho_0}}}, \qquad (28)$$

where ξ, h, σ are respectively the structure factor, porosity and flow resistivity [13]. In this model, the average radius of a human body assumed 25 cm and it used to calculate model parameters. Figure 7 represents real and imaginary parts of the speed of sound depending on the concentration *n*.

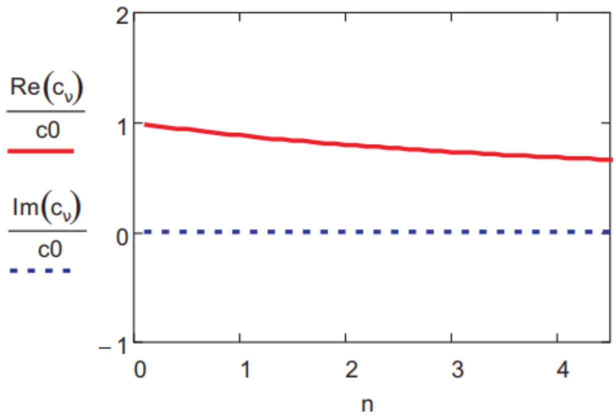


Figure 7. Real and imaginary parts of the speed of sound depending on the concentration *n* [13].

In the real case, if we consider a big concert venue, quite often the largest concentration of people is near the stage and it gradually decreases towards the edges and the back of the stage (Figure 8). It is obvious that such a placement of crowd is not always repeated and can be different, but for further simulations this particular distribution of crowd will be considered.

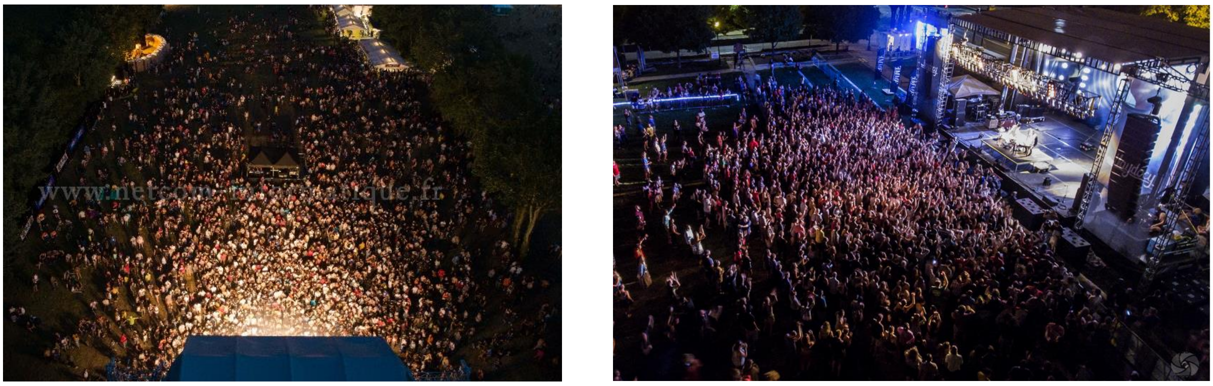


Figure 8. Distribution of crowd at the concert venue

A gradual change in the concentration leads to a smooth change in the wave impedance and the absence of reflections, which is greatly simplify calculations and reduce the possible error. According to [13] reasonable values of the concentration of people near the stage is around 2 people per m^2, which was taken for further simulations.

Figure 9 represents modeled distribution (changing concentration) of crowd at the concert venue similar to our system under study (to increase the calculation time, the size of a bright zone was reduced, that will not affect the evaluation of the process and its ideas) and calculated speed of sound according to this concentration.

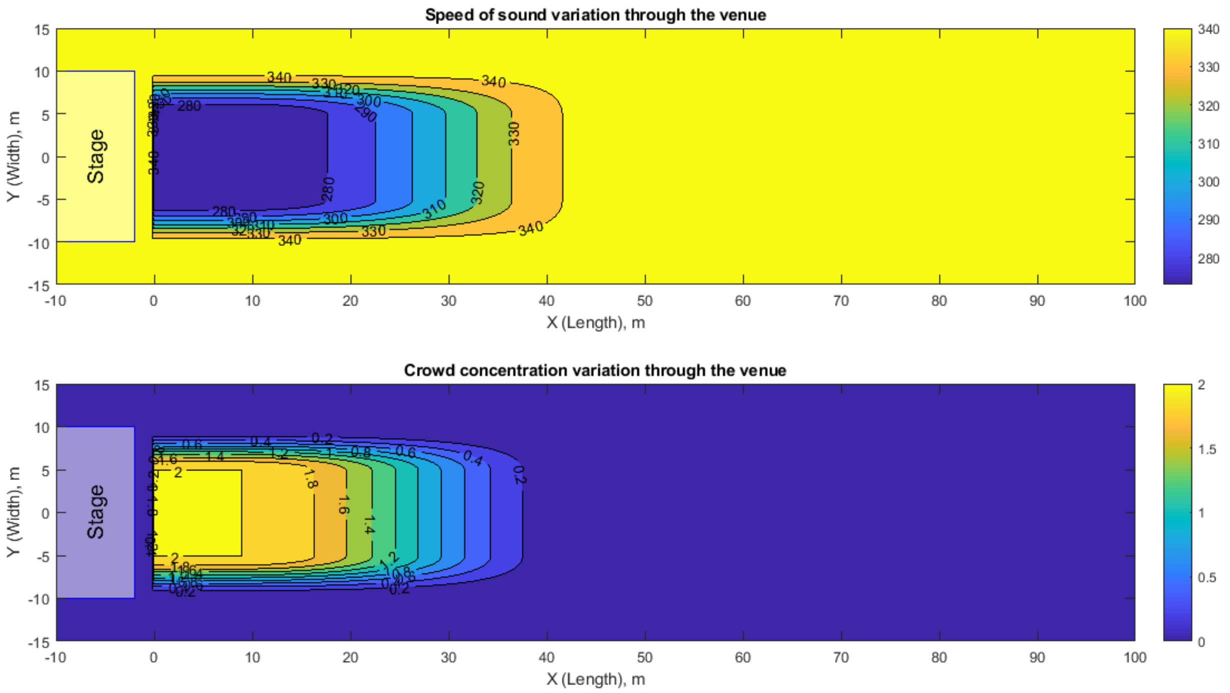


Figure 9. Modeled distribution of crowd and calculated speed of sound

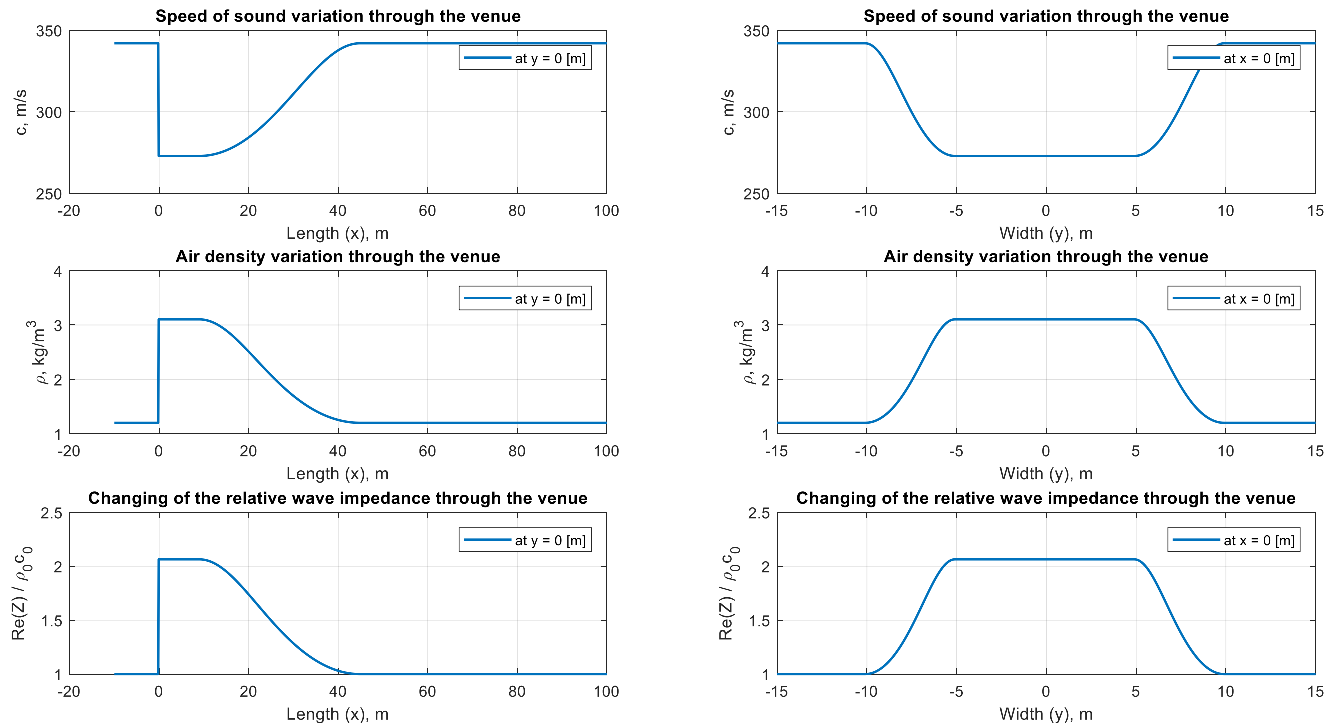


Figure 10. Speed of sound, density and wave impedance variation through the venue

As can be seen from figure 10, the higher the concentration, the lower the speed of sound, but opposite statement for density is noticed. Due to the small discretization of the calculation (0.1 m), the change in the speed of sound and density are accurate to a tenth. The change in impedance is smooth, which can be seen from the shape of the curves, but near the stage at $x = 0$ m, a sharp jump in the concentration value occurs, which leads to a sharp jump in wave impedance. In this project, we assume that the array of primary sources, which is built of cardioid subwoofers, is located very close to a crowd of people and taking into account a large wavelength (at frequencies up to 100 Hz), reflections due to impedance mismatch are negligible and not taken into account.

Since now the speed of sound and density depend on x and y coordinates, it is necessary to take into account in the calculations that sound wave propagate through different layers with different speed of sound, even if the changes are very gradual. Figure 11 and equation 29 represent approximated calculation of this 2D problem. The calculation is done for each point (microphone) in the discretized grid from every source. For example, for the first point, using the Bresenham's line algorithm [9], the nearest value of the wave vector is found at each discretized moment of wave propagation (for example, $k_1(x, y)$). The distance r is a discretization step, which is calculated for each point, while R is the distance from the source to the receiver.

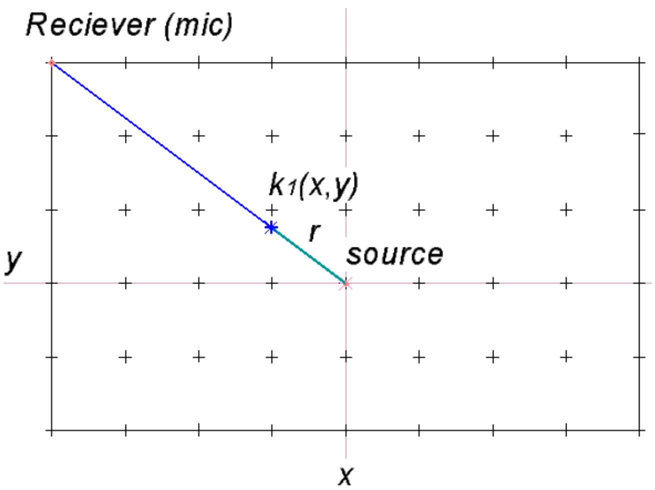


Figure 11. Calculation method.

$$\text{Propagator} = \frac{\exp(-i \cdot r \cdot (k_1 + k_2 + ...k_n))}{4\pi R}$$
$$\text{P}_{reciever} = i\omega\rho(x, y) \cdot \text{Propagator} \tag{29}$$

The results of calculations and their comparison are presented in the figure 12. In this case, the simulations were carried out for a single source located close to the crowd of people (at *x=0, y=0*). To verify the calculations, matrices with the values of c_0 and ρ_0 (340 m/s and 1.2 kg/m^3) were used, as can be seen this method of calculation gives the same result (model validation curve) as the calculation for free-field by using common monopole formula. For audience case sound pressure level near the stage center more by around 7 dB than without it. This can be explained by the fact that density at the point with high crowd concentration is greater by more than 2 times than value for ρ_0. Further, the trend is observed until the value of ρ (audience) equals the value of ρ_0.

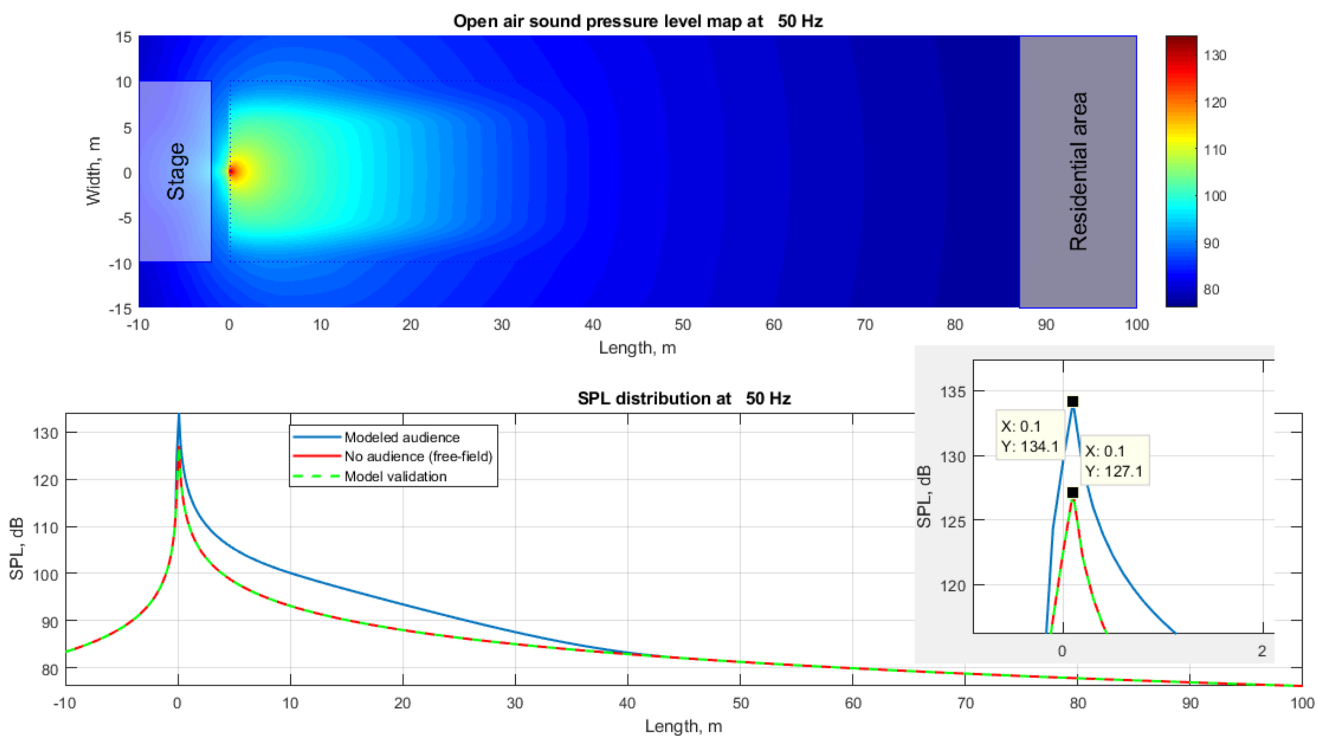
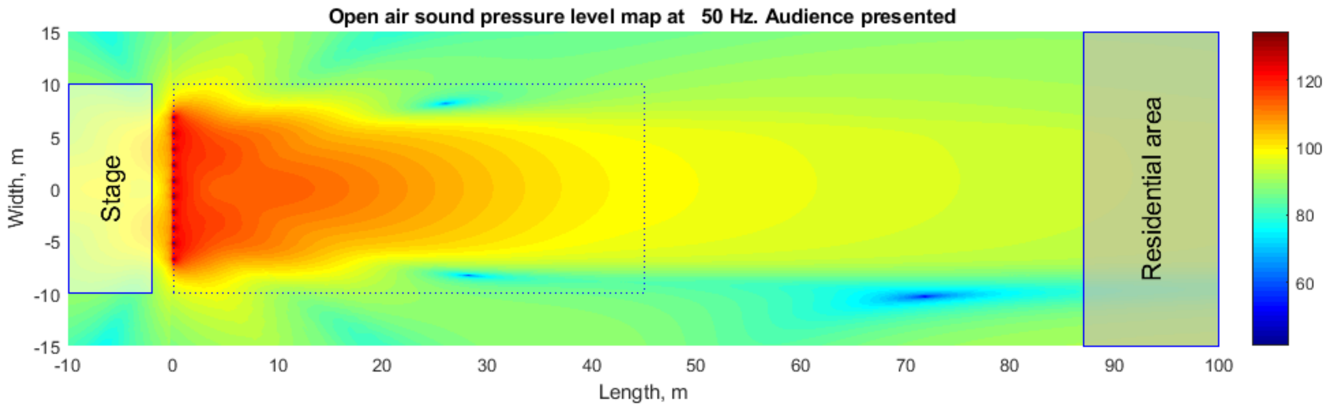


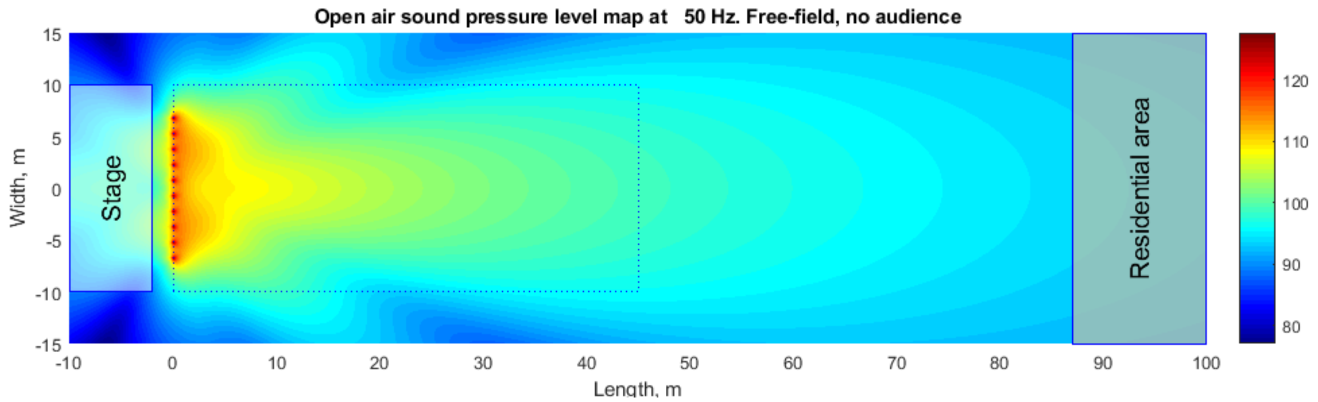
Figure 12. Sound field calculation for one source with audience.

In general, it can be noted that the shape of the field changes significantly and acquires a form that is different from the propagation of a spherical wave. These calculations are preliminary (taking into account approximations and assumptions) and should be checked and compared with the full, realistic model of a crowd, for example, in COMSOL. It is worth noting that the model takes into account the audience as infinitely long hard cylinders, which is an approximation and introduces an error.

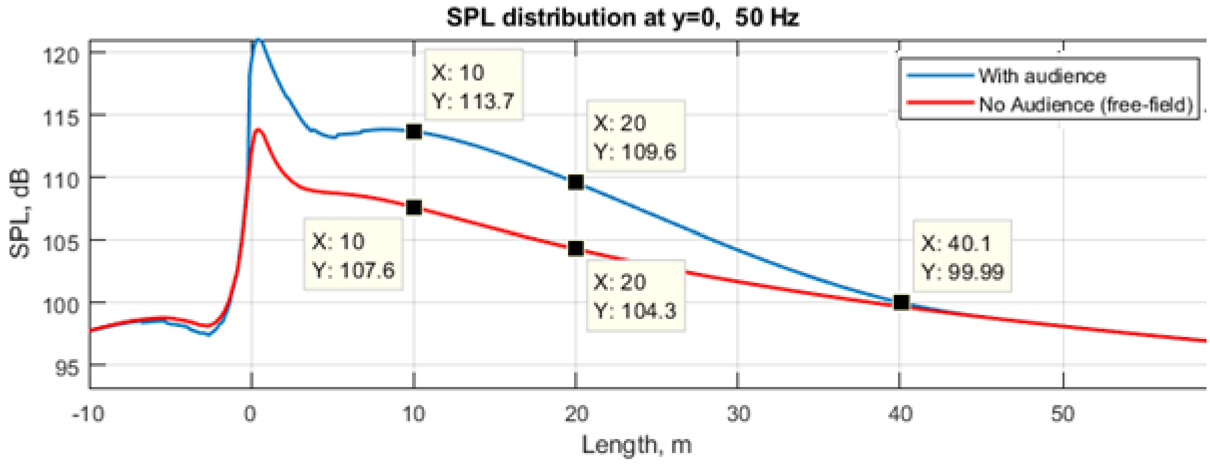
Sound field calculation for a line array of 10 subwoofers is presented in the figure 13. As expected for the line array, the *1/r* law does not work as for a monopole because of the different geometry of the near field/far field. For the calculation with the crowd, the SPL change is faster due to the influence of variable density in crowd area.



a) Calculation with audience



b) Free-field case



c) SPL comparison at y=0

Figure 13. Sound field calculation for a line array of 10 subwoofers V-SUB in different scenarios.

2.2.2 Finite element analysis In COMSOL

The problem is considered in 3D with the Pressure Acoustics, Frequency Domain Physics Interface with Frequency Domain study. The frequencies of interest up to 100 Hz.

Description of the model creation is presented as it has a significant role on the results and the possibility of correct comparison in the future.

2.2.2.1 Geometry

Audience was modeled as a set of finite cylinders of 1.7 m height (average human height) and the radius of 25 cm (average radius of human body) [13]. This set of cylinders is divided in to 7 regions with different concentration: the highest concentration near the stage (0, 0) and the lowest at the end of the venue. The values of maximum concentration near the stage were close to the real situation. In this case, only the change in concentration along the x axis is considered. The geometry was created using cad tools of COMSOL *Cylinder* and *Array* feature (Figure 14).

The presented geometry of the bright zone's open-air venue is reduced, which allows to significantly speed up the calculation time and the possibility of their launch on a laptop.

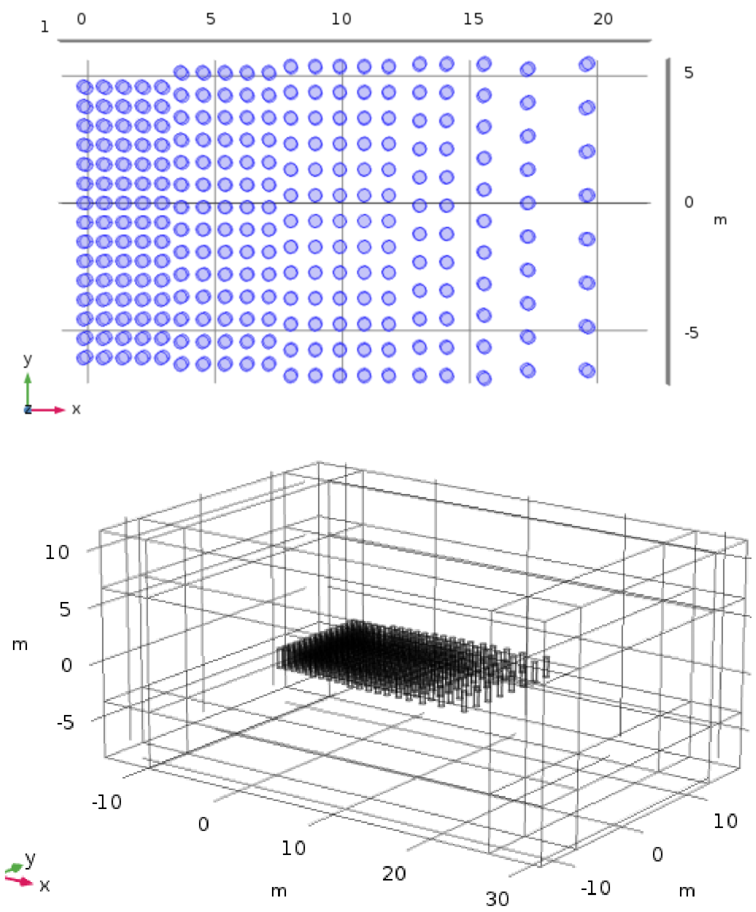


Figure 14. Geometry. Set of cylinders with different concentration and PML geometry.

Next step in geometry was creating a big block and putting the cylinders inside it. The purpose of this block is to limit the domain of calculation. The block has layers of each side (thickness is 5 m) which are used in *Perfectly Matched Layer* (PML) definition. The physical purpose of PML is to absorb the energy of all outgoing waves without any impedance mismatch (thus avoiding reflections at the boundary). There is an important point in PML settings, the type of geometry should be chosen wisely taking into account both the geometry of the problem and the type of the wave considering. In our case most suitable is Cartesian type of geometry. The thickness of PML also plays an important role and there is a rule, that number of meshed PML layers in case of swept mesh (will be discussed later) should be at least 6 in order to obtain reasonable results at high frequencies. "Efficiency" of the PML can be checked with sound pressure level plot where we can see attenuation inside PML.

2.2.2.2 Materials

The built in material *Air* was chose for all domains (except set of cylinders) since we are interesting to calculate sound propagation in air. The material for cylinders can be set as built in *Skin* with physical parameters of human skin.

2.2.2.3 The Pressure Acoustics, Frequency Domain Interface

This branch contains all the nodes which are needed to calculate the problem according to selected physics. The *impedance node* was added in order to simulate the impedance of human body. The *psychological model* of impedance in COMSOL allows to choose *human skin* for simulations. The impedance node was applied only for set of cylinders. The important point: the impedance node can recognize all cylinders only if we do *difference* operation for block and cylinders (since the cylinders are inside) otherwise they are not applicable.

Added *monopole point source node* was set for the point in front of the audience at the height of 0.5 m. This point represents a subwoofer. It is also possible to use a *line source*, but to see better the influence of audience a single omnidirectional subwoofer was used.

No boundaries were set for *Initial values node* and *Sound Hard Boundary* (at first step we do not want to see reflections from the ground). To take into account reflections from the ground appropriate surface can be set to *Sound Hard Boundary* or it is possible also to specify ground's impedance. The *Pressure Acoustics node* was set only for one domain – air inside a block.

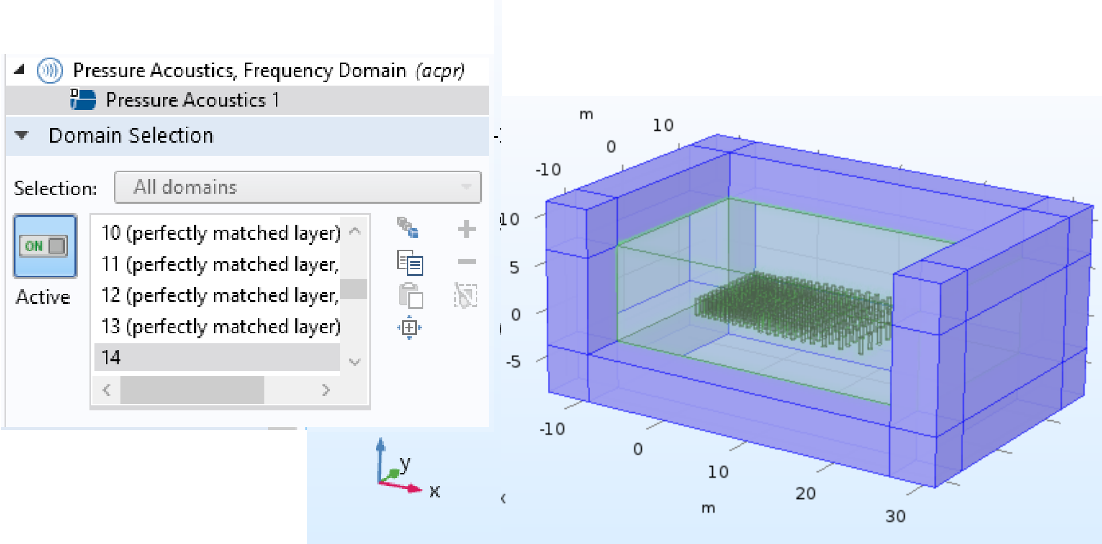


Figure 15. Physics. Pressure Acoustics node

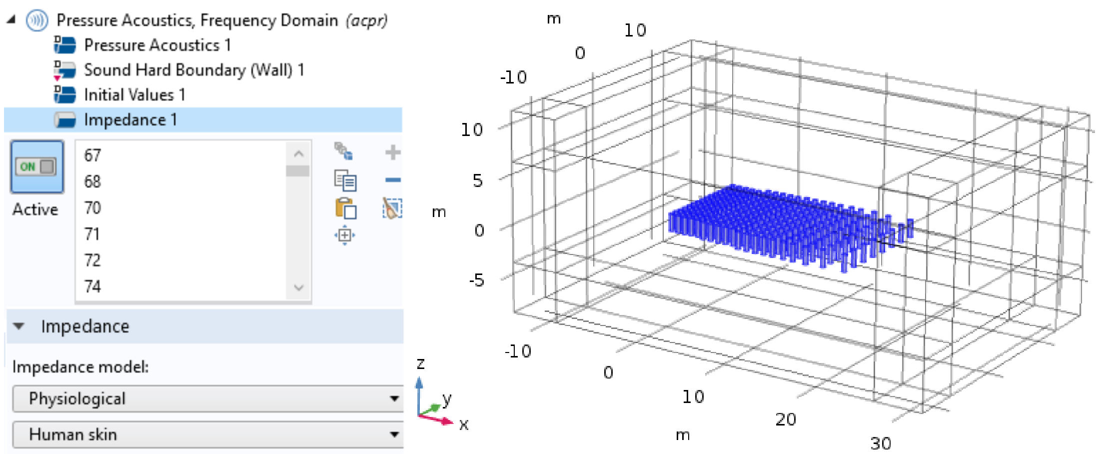


Figure 16. Physics. The Impedance node for cylinders

2.2.2.4 Mesh

The basis of accurate, time-optimal and resource-consuming results is a Mesh. By default mesh is set to physics controlled sequence with selectable element size. The mesh and the size can be changed. There are different types of meshes available. In our case we use *free tetrahedral* mesh only for domain 14 (inside a block) and *swept mesh* for PML domain (layers of the block). The swept mesh intended and accordingly works best for regular shapes such as cube, cuboid. The distribution parameter should be chosen according to geometry, this can also affect PML calculation performance and accuracy. Concerning the size of mesh in acoustics there is a general rule that the maximum element size should be 6 times smaller than wavelength.

Once geometry is meshed it is possible to analyze quality of meshing by *Statistics*, which gives information about quality of the meshing. For 2D case desirable average element quality value is around 0.7-0.75 (and more), when for 3D case it is more difficult to mesh properly everything in geometry and average element quality value can be lower, but in our case its 0.78 which is very good. Histogram below displays a histogram plot of the mesh element quality.

As can be seen from figure 4, average element quality is higher when using combination of Swept + free tetrahedral mesh (as in our case) than using only free tetrahedral mesh everywhere (in this case meshing time is also increases).

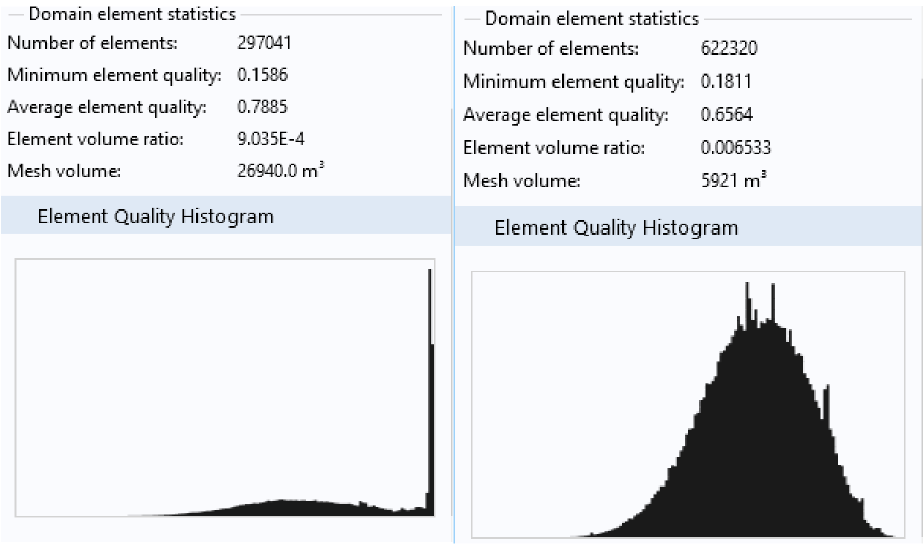


Figure 17. Mesh statistics. Swept + free tetrahedral (left), only free tetrahedral (right)

In order to “see” the quality of the mesh in 3D we can plot a mesh and by applying the filter to see the worst quality elements (in this case factorization is 0.05). We can see that almost everything is meshed properly except of area near cylinders, but still according to average quality value we have got a good mesh.

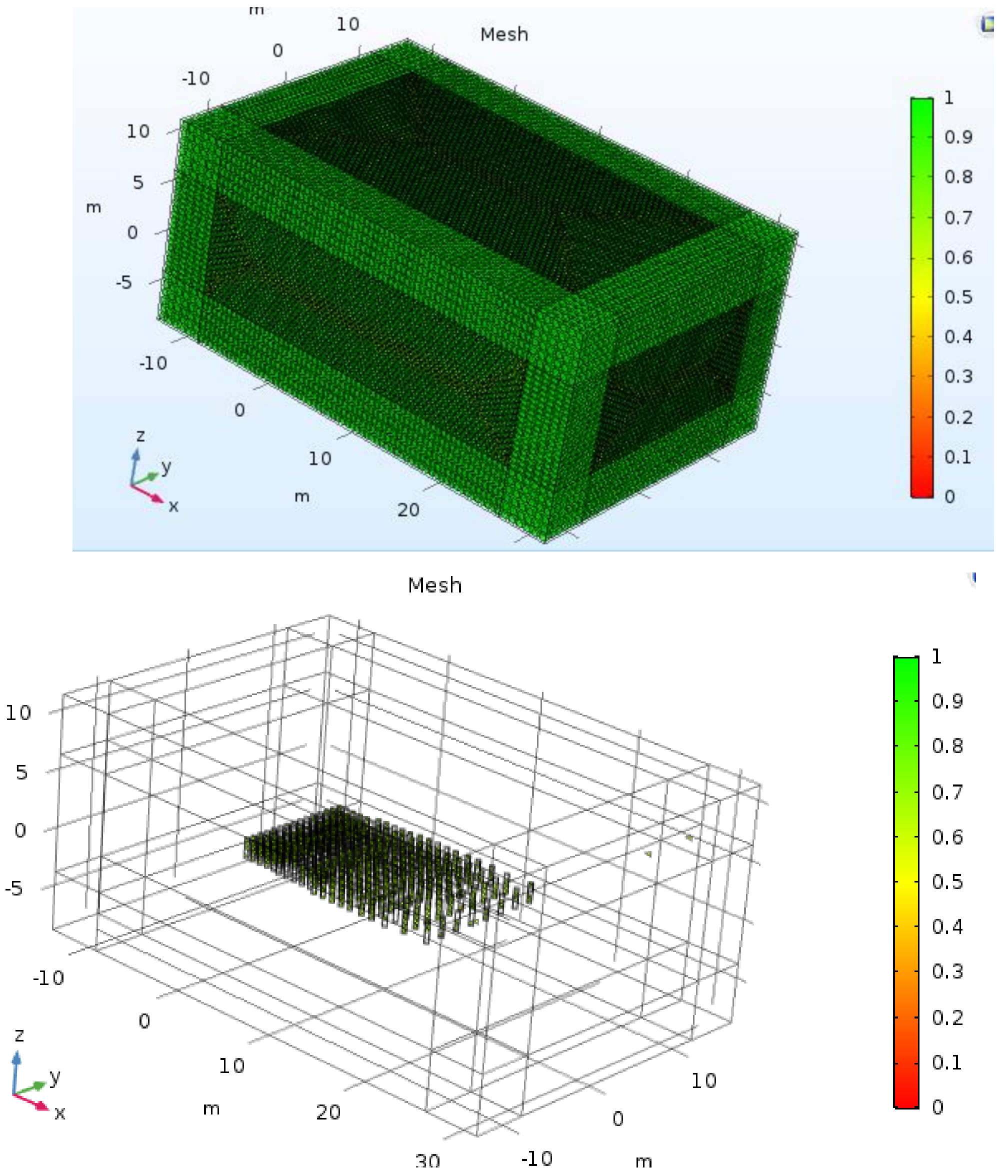
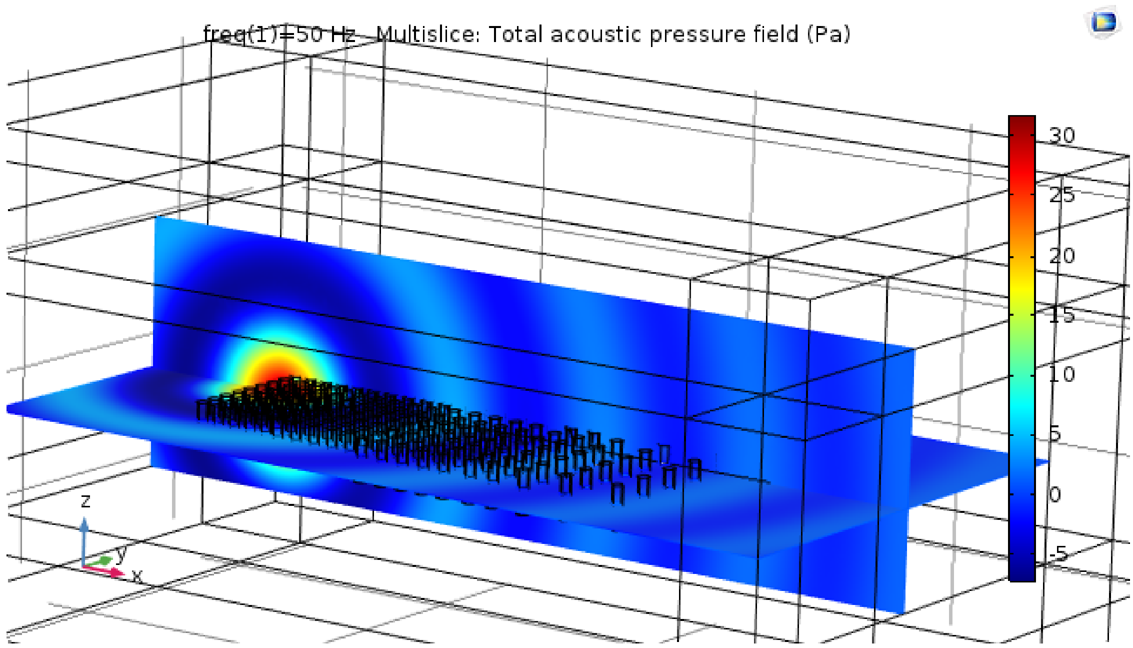


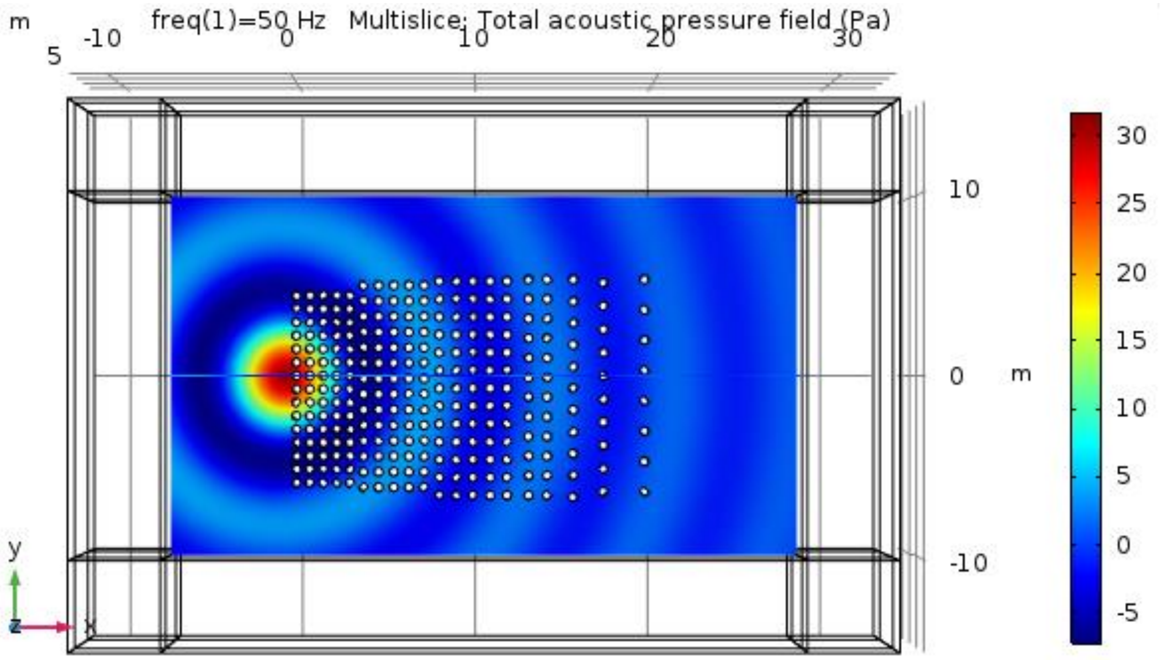
Figure 18. Mesh quality plot. Mesh plot with worst quality filter at the bottom

2.2.2.5 COMSOL simulation results

The study in COMSOL was done for 50, 70 and 100 Hz. The following figures represent multislice 3D plots for total acoustic pressure field.



a)



b)

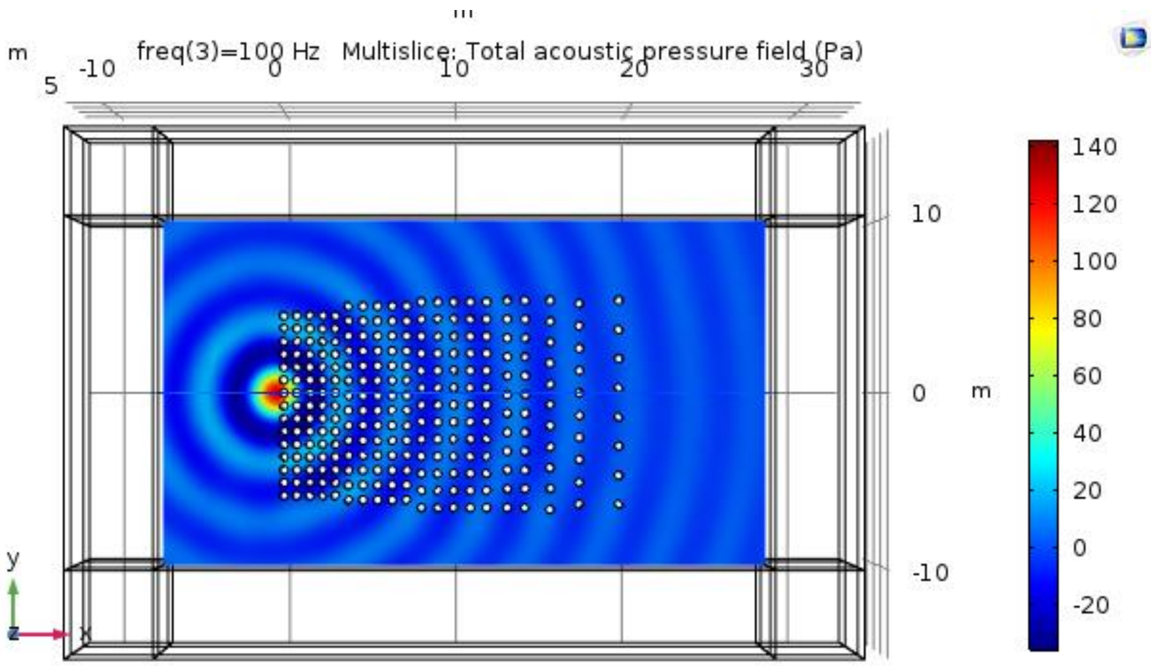


Figure 19. Calculated total acoustic pressure field at 50 and 100 Hz

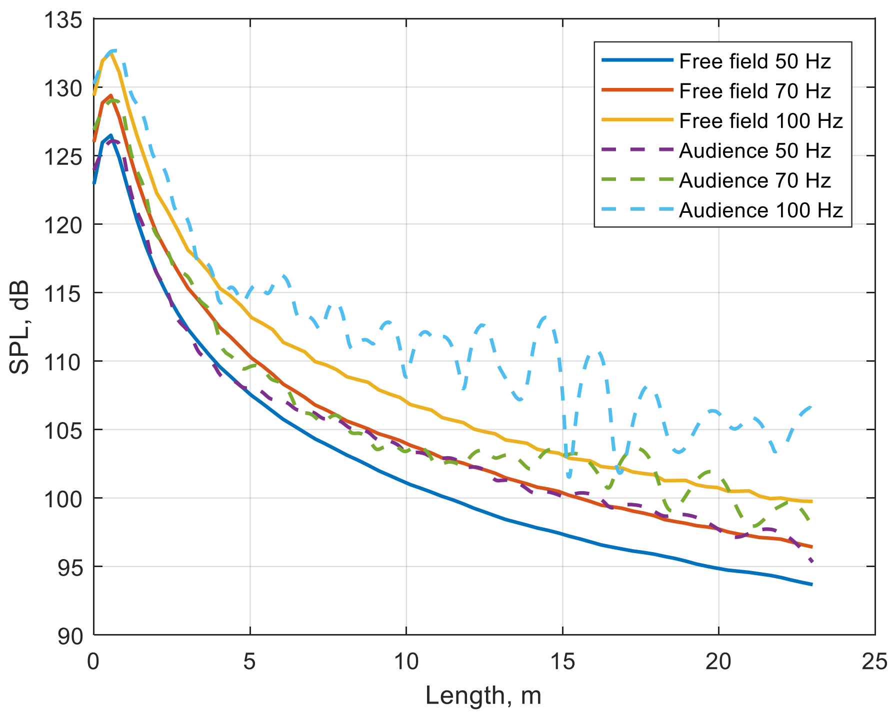


Figure 20. SPL comparison

From Figure 20 (on-axis, 1.6 m height SPL response) can be seen that in case of audience presence there is a slight increase in pressure. With increasing distance concentration decreases and significant fluctuations of sound pressure are observed, especially with increasing frequency (the distance between the cylinders approaches the wavelength), which is caused by reflections and the possible appearance of modes.

From Figure 21 we can see how the waves attenuate in PML at different frequencies. In general attenuation is about (or even more) 60 dB.

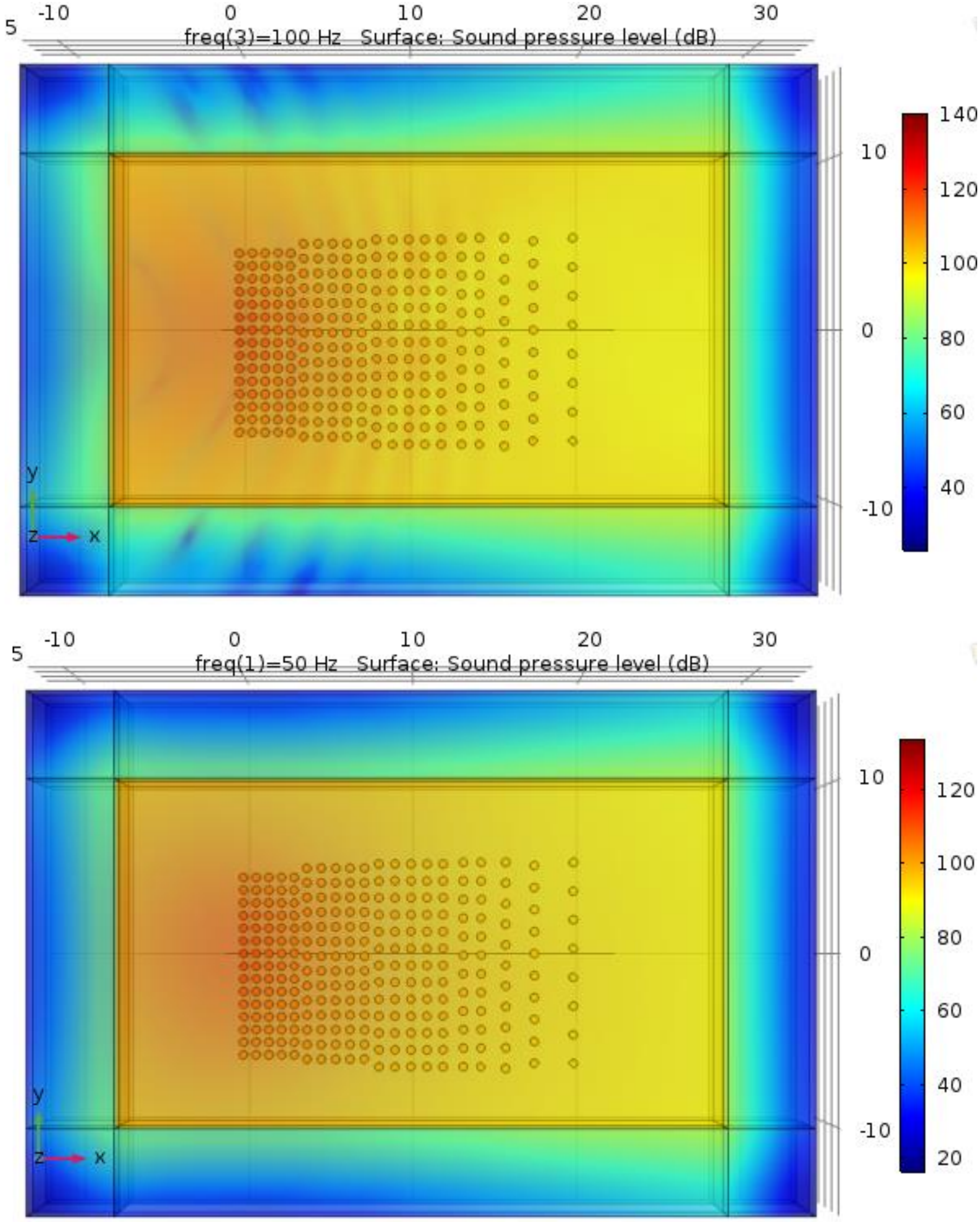


Figure 21. SPL at different frequencies, PML attenuation.

2.2.2.6 Comparison and conclusion

When building a model, important parameters are the definition of the type and size of the mesh, correct definition of all physics nodes and materials.

In this case, reflections from the ground were not taken into account, although a simple one parameter model of ground is present in the COMSOL.

Comparing the results with the calculations of the simplified model in MATLAB, it can be concluded that, in general, there is a tendency in increasing pressure, but not as significantly as in the case of the simplified model. The simplified model also does not take into account the presence of reflections and possible modes, because the change of concentration in MATLAB is modeled as continuous, which is an approximation. To obtain more accurate results, the geometry of the model should be reconstructed more closely to the actual placement of people on the venue.

2.3 Influence of atmospheric conditions on sound wave propagation

One of the most famous and well-studied effects of atmospheric conditions is atmospheric absorption of sound waves. A dissipative processes in the atmosphere causes a sound wave energy loss [15]. In addition to reducing amplitude, atmospheric absorption also affects the phase of the sound wave. This process is frequency-dependent, so sound waves of different frequencies can propagate at different speeds. In general, atmospheric absorption also depends on air humidity.

According to [15] atmospheric absorption at low frequencies and small distances (less than 500 m) does not make sense to take into account because of the small absorption coefficient (around 0.1-0.2 dB/km).

More complex nature has the effect of atmospheric refraction - when a sound wave is refracted toward regions where the sound speed is low or varying which is due to spatial variations of the temperature and the wind velocity [15]. With a flat, homogeneous ground surface, approximation can be done that the temperature, the wind velocity, and the effective sound speed are functions of height z only:

$$c_{eff}(z) = c(z) + u(z) \, , \tag{30}$$

where c is the adiabatic sound speed and u is the component of the wind velocity in the direction of sound propagation.

However, in reality, all three parameters depend on *x,y,z*. A consequence of refraction is that higher levels are generated by source in downwind directions than in upwind directions. From the same [15], atmospheric refraction should be taken into account for distances of the order of 100 m or more and when source and receiver are close to the ground (few meters or less). However, at the same time, the effect is less pronounced at low frequencies. In our case the distance is about 80-100m, but the distance between the sources (primary and secondary) is about 40-50 m. It must be assumed that the effect of refraction can influence in our case.

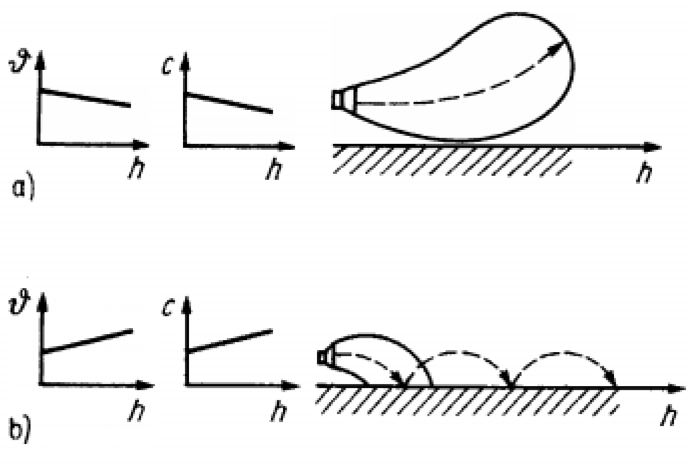


Figure 22. Atmospheric refraction effect depending on temperature and the wind velocity gradients

There are also rapid fluctuations of wind and temperature (i.e. atmospheric turbulence) which have a considerable effect on atmospheric sound propagation, especially it causes significant and local fluctuations of the sound pressure [15]. With account of turbulence, the effective speed of sound can be expressed:

$$c_{eff} = c_0\sqrt{T/T_0} + u\,, \tag{31}$$

where T is the turbulent fluctuations of the temperature; u – wind velocity component. In this case c_{eff} is averaged effective sound speed.

The above-described atmospheric conditions to a greater extent affect and cause large errors for the feedforward active noise cancellation system, since all these phenomena should be taken into account with relative accuracy by model in the radiation transfer-functions. The ANC feedback system is less affected, because it tracks changes in the radiation transfer-functions in real time, but only local changes are recorded (at the microphone locations) and not at all concert venue. However, the main problem is that the change of wind velocity, temperature, etc. happen continuously and secondly there is a great difficulty in measuring these values and obtaining consistent values for the model.

When it comes to direct calculation when the transfer function are measured or modeled the problem arise: measuring transfer function at many points of big sound zone requires a lot of time and during this time atmospheric conditions may change many times. It is also difficult to update these measurements during the open-air show. In the other hand, when the transfer functions are modeled, the modal should be updated and be very accurate, taking into account as much information about atmospheric condition (local changes) as possible, which can complicate the system and computation time, what is crucial. A more detailed information about measurements of transfer function and atmospheric condition influences in terms of open-air events can be found in [7, 8].

2.4 Ground effect

Another effect that can significantly influence the sound field and change it (in terms of amplitude and phase) is the ground effect (i.e. reflections from the ground). In this case the total pressure at each point is the sum of the pressure generated by each source and its reflections from the ground. There are several outdoor propagation models, but Nord2000 is one of the most common. This model is based on the image source method and on geometrical ray theory, ground surface is assumed flat and homogeneous [15].

Taking into account ground reflections, the monopole radiation transfer - function (eq. 2) can be rewritten:

$$H = \frac{j\omega\rho_0}{4\pi}\left(\frac{\exp(-jkR_1)}{R_1} + R\frac{\exp(-jkR_2)}{R_2}\right), \tag{32}$$

where R_1 and R_2 are the direct and indirect paths (figure 23); R is the reflection coefficient.

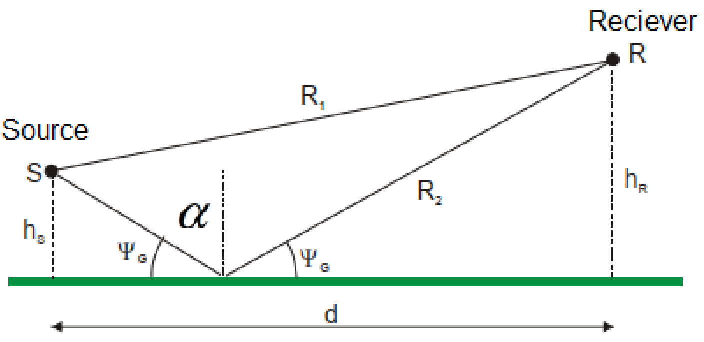


Figure 23. Geometry. Propagation over flat ground

In the frame of this project for simplicity reason, plane-wave reflection coefficient is used:

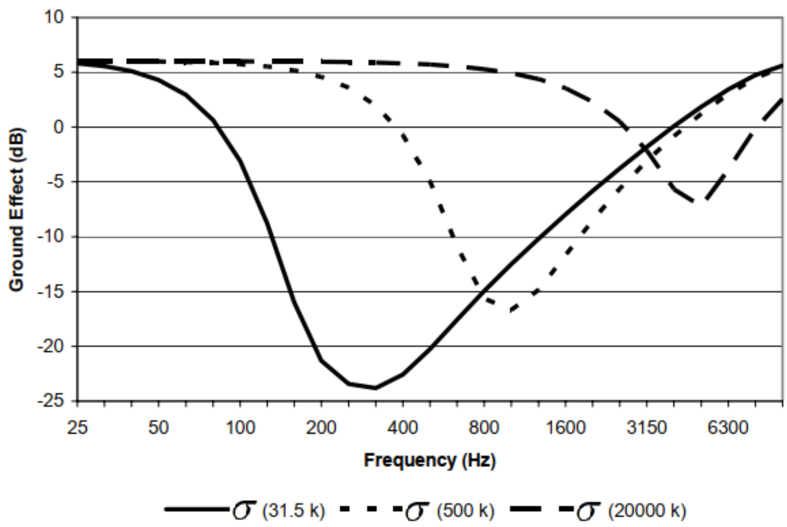
$$R = \frac{Z\cos(\alpha) - \rho_0 c_0}{Z\cos(\alpha) + \rho_0 c_0}, \tag{33}$$

where $\alpha = 90° - \psi_G$ is reflection angle; Z is the normalized ground impedance, which characterizes the ground surface acoustically and depends on the frequency and type, the structure of the ground.

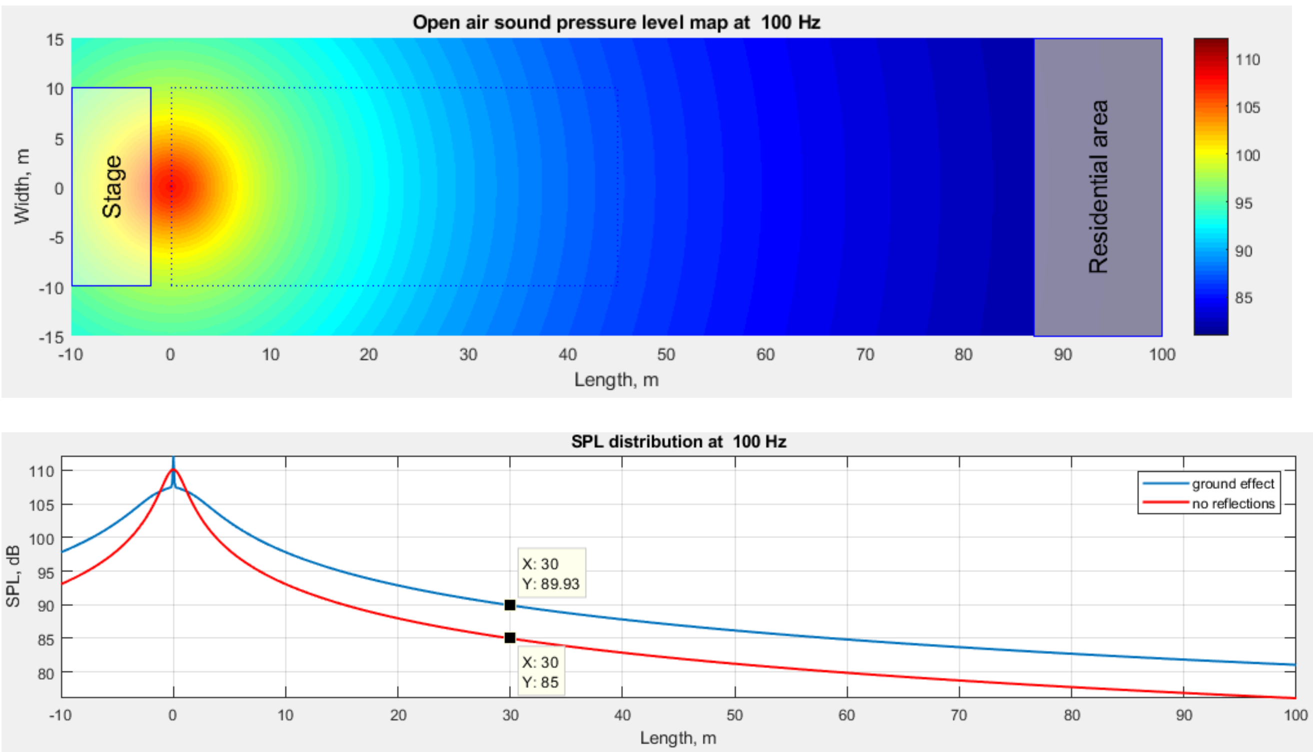
Various models exist for the normalized ground impedance, but Nord2000 uses the empirical one-parameter model of Delany-Bazley [15]. This model also recommended as the default model for predicting outdoor ground impedance in the HARMONOISE prediction scheme [16]. The main advantage of the model is that it requires only one input parameter – flow resistivity σ, but this model cannot be used for porous type of ground, like porous asphalt and two or three (i.e., including layer depth) parameter models should be used [16]. In the project simulations, the Miki model of ground impedance (improved model based on the experimental data of Delany and Bazley, [7]) is used:

$$Z = \rho_0 c_0 \left(1 + 5.51\left(\frac{1000f}{\sigma}\right)^{-0.632} - j8.42\left(\frac{1000f}{\sigma}\right)^{-0.632}\right), \tag{34}$$

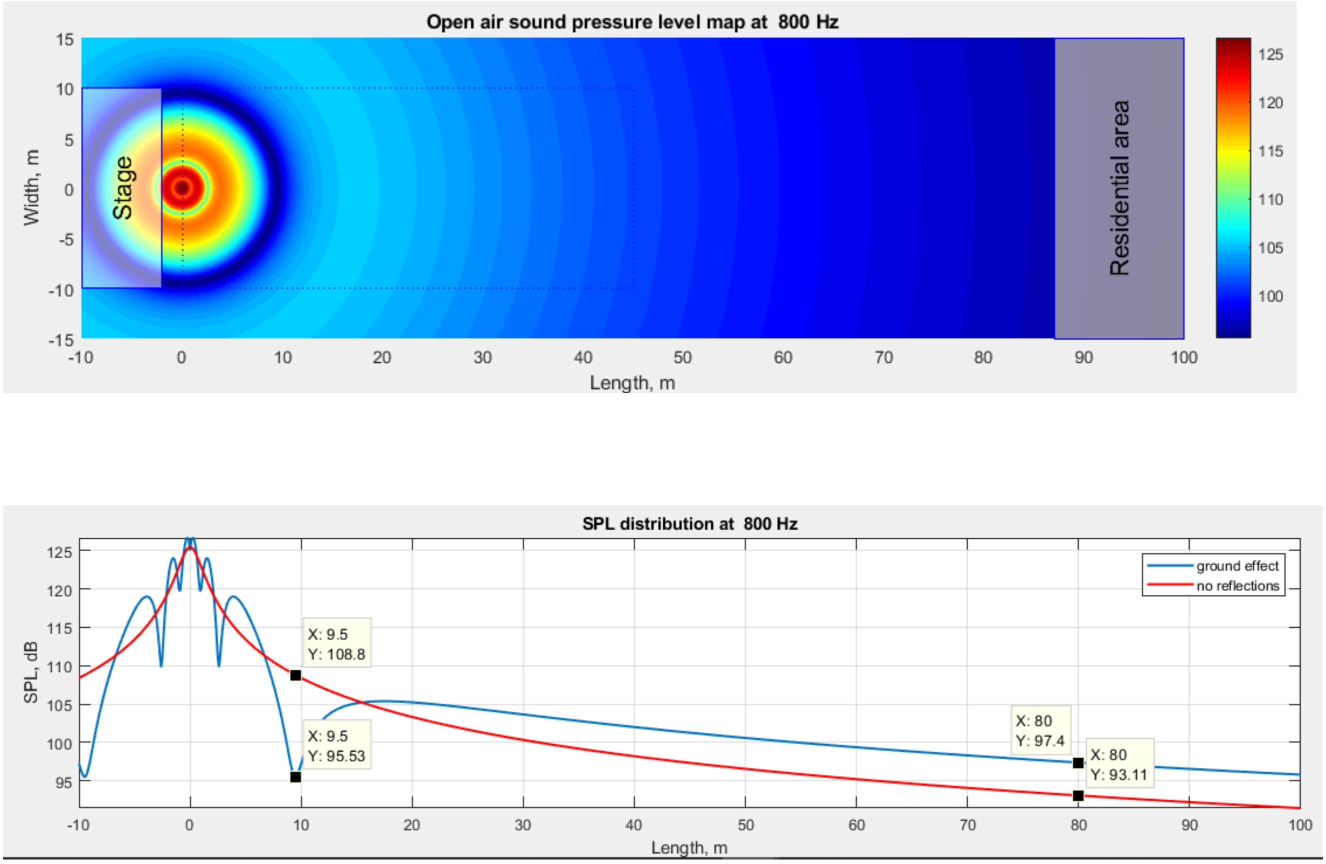
where $\sigma\ \left[Nsm^{-4}\right]$. For primary simulations, as ground, the compacted lawns, park area with flow resistivity of $\sigma = 500\ \left[kNsm^{-4}\right]$ was used [15].



a) theoretical curves [11]



b) simulated, 100 Hz



c) simulated, 800 Hz

Figure 24. Simulations of Ground effect

The influence of the ground reflections (Figure 24) can be described as follows: at very low frequencies, due to small phase difference between direct and reflected waves, the total sound pressure is doubled relatively to the sound pressure with infinite baffle case, so the ground effect is +6 dB. At a higher frequencies, the phase difference increases and two waves can be out of phase (at certain frequencies, destructive interference effect appears), but due to small difference in amplitude (waves travel different distances) the sound field is not totally cancelled. Further, with increasing the frequency, the situation repeats and constructive and destructive interference is observed (the maximum increase in the level of the resulting field is up to 6 dB).

Based on this (Figure 24), it can be concluded that at low frequencies (up to 100 Hz at the range of our interest) the ground effect does not negatively influence (does not cause destructive interference, does not change field homogeneity) on the resulting sound field and only increases the sound level by 6 dB. However, in the real case, the ground on the open air event may have irregularities (ground is no more flat homogeneous surface, as assumed by the model), which can cause significant phase differences and even at low frequencies change the sound field.

3. SIMULATIONS OF ACTIVE NOISE CONTROL AND SOUND ZONES METHODS

This section contains simulations of the methods described above for the geometry under study. To find the most optimal method and tune it for open-air case, radiation transfer functions were modeled without taking into account the influences of the surrounding environment and atmospheric conditions, that is, free field radiation conditions.

The following performance metrics will be used in order to easily interpret the results and compare different method:

1. Acoustic contrast (mentioned in detailed above) – ratio of the average acoustic potential energy density in the listening (Bright) zone to that in the quiet zone, expressed in dB [17]

$$AC = 10\log_{10}\left(\frac{M_d P_b^H P_b}{M_b P_d^H P_d}\right)$$

2. Array (control) effort – is the energy that the loudspeaker array requires in order to achieve the reproduced sound field, expressed in dB [17]:

$$\text{Effort} = 10\log_{10}\left(\frac{\mathbf{q}^H\mathbf{q}}{|q_r|^2}\right) \quad (35)$$

For example, a high control effort implies poor acoustical efficiency, with high sound pressure levels emitted in to the uncontrolled regions. Sometimes there is an existing solution (mathematically), but due to the limitation imposed by the ability of the loudspeaker array physically reproduce the required signals, and the electrical requirements necessary for such reproduction, this solution cannot be realizable in a real world. Control effort is defined as the total array energy relative to a single reference source q_r producing the same pressure in the bright zone. Using a reference source ensures that the effort performance is physically useful: A score of 0 dB means that the array requires the same energy as that source to reproduce the target sound pressure, with negative scores improving upon this. [8]

3. The insertion loss metrics – represents the decrease in sound energy in the dark zone due to the control sources. Large IL indicating a strong reduction.[8]

$$\mathrm{IL} = 10\log\left(\frac{1}{N_B}\left\|\mathbf{h}_D^p\right\|^2\right) - 10\log\left(\frac{1}{N_D}\left\|\mathbf{H}_D^s\mathbf{w}^s + \mathbf{h}_D^p\right\|^2\right) \tag{36}$$

4. The primary to secondary ratio in the bright zone quantifies the ratio of sound energies coming from the primary and the secondary sources. A large PSR value means that the sound from the primary sources dominates the sound field in the bright zone.[8]

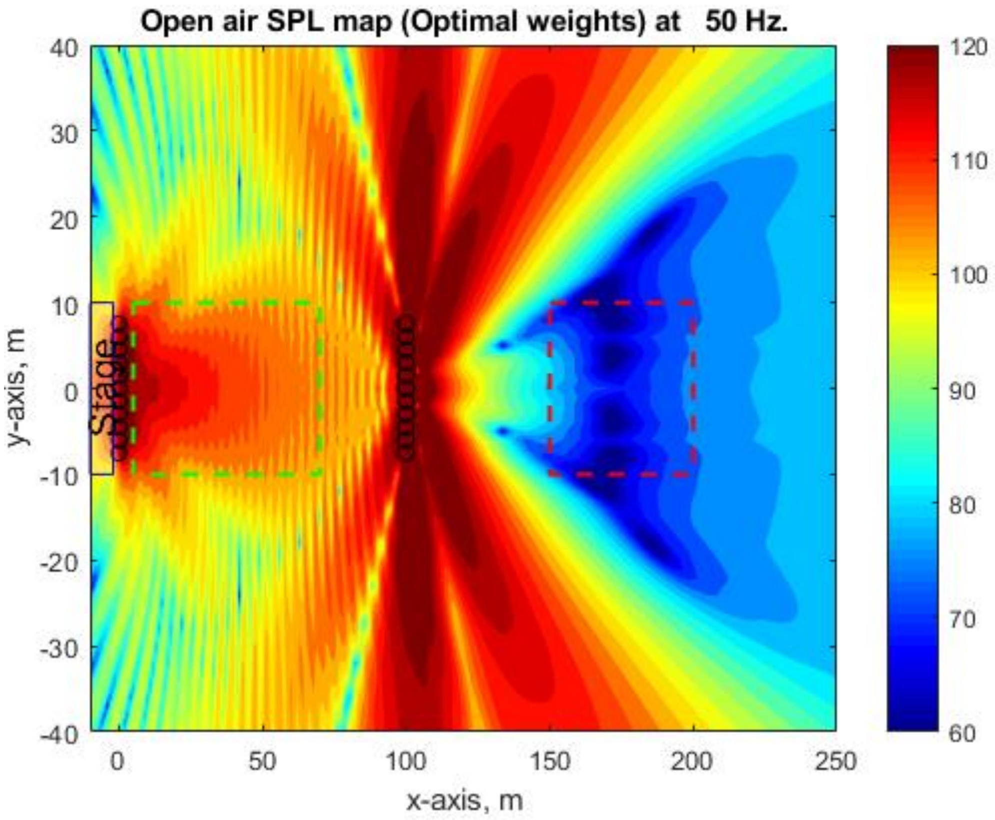
$$\mathrm{PSR} = 10\log\left\|\mathbf{H}_B^p\mathbf{w}^p\right\|^2 - 10\log\left\|\mathbf{H}_B^s\mathbf{w}^s\right\|^2 \tag{37}$$

3.1 Simulation of the LMS error minimization based method

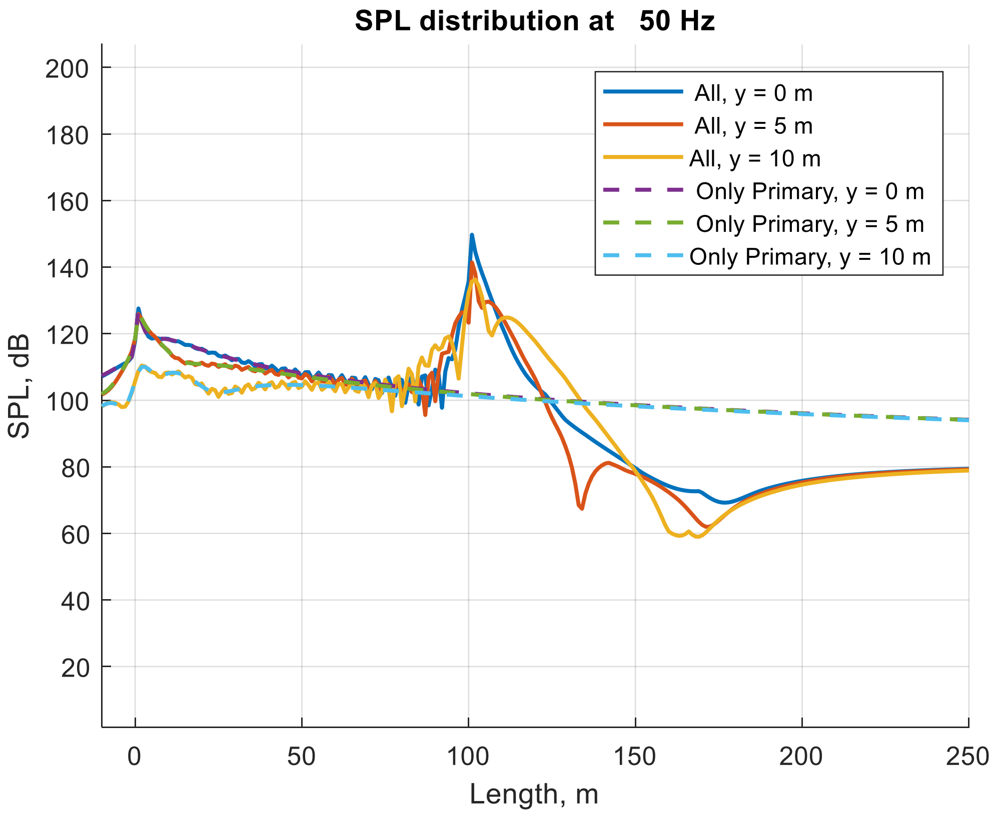
This method explained in section 1.1 of this thesis and final expression for the optimum vector of control source volume velocities presented by eq. 9:

$$\boldsymbol{q}_{c,opt} = -\boldsymbol{A}_{\boldsymbol{p}}^{-1}\boldsymbol{b}_{\boldsymbol{p}}$$

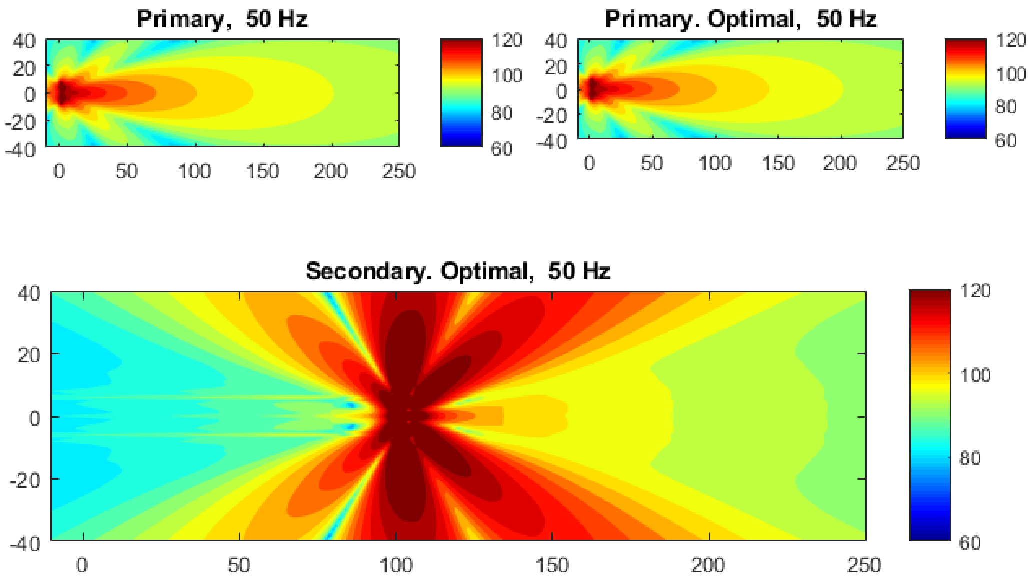
The results of simulations presented at Figure 25.



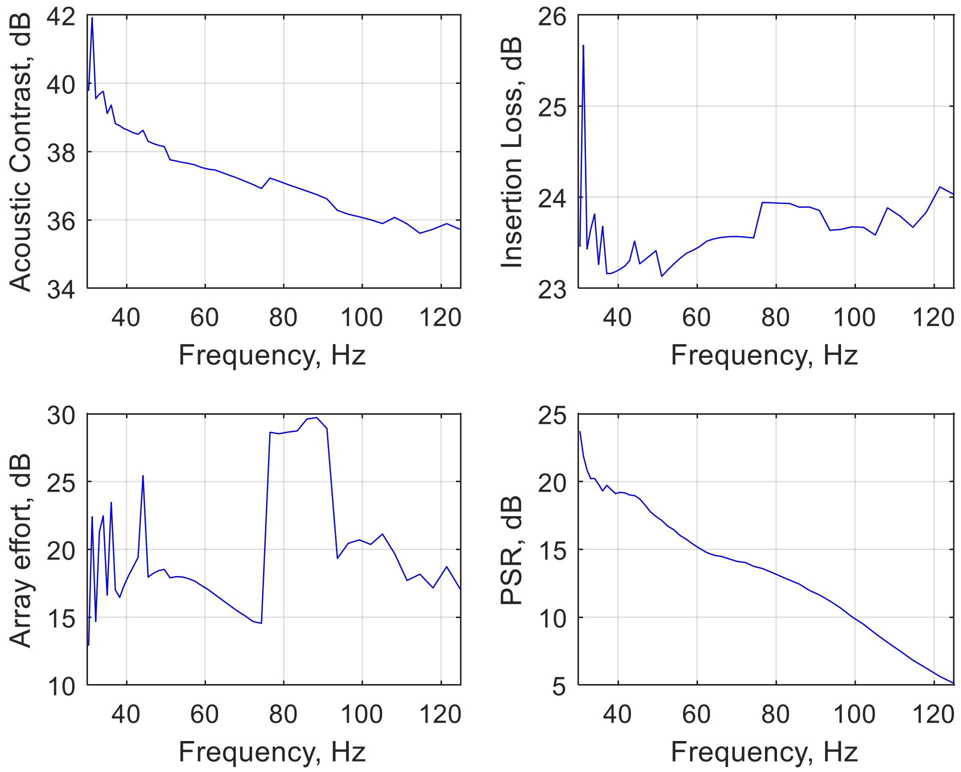
a) SPL map with calculated optimal volume velocities for control sources



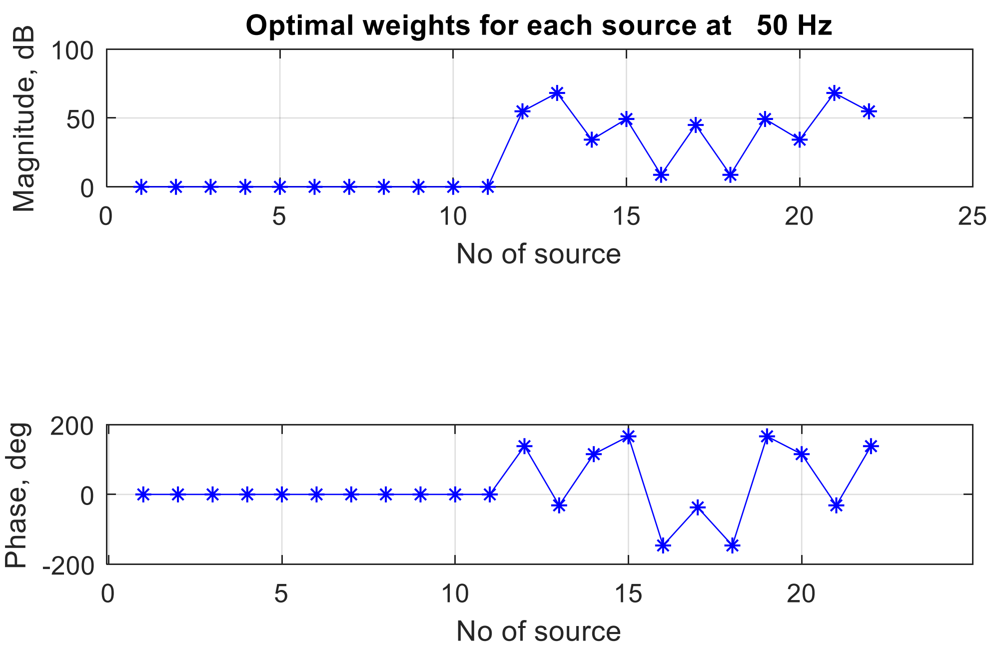
b) SPL distribution at different y-axis level



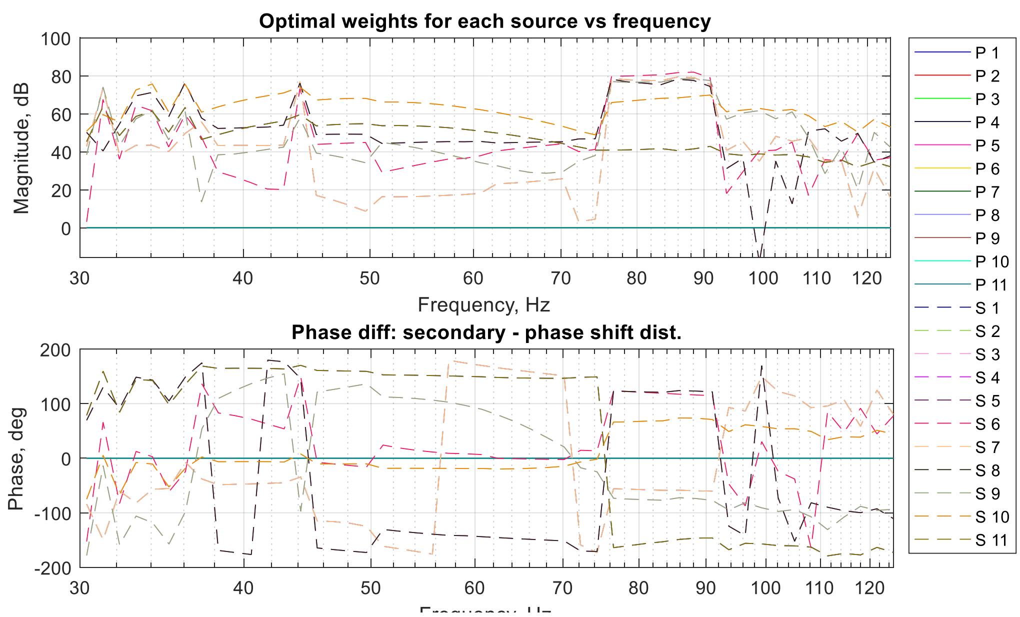
c) SPL map comparison for primary and secondary (control) arrays



d) Performance metrics



e) Magnitude and Phase for each loudspeaker at a given frequency



f) Magnitude and Phase for all loudspeaker over the frequency range

Figure 25. Simulations of the LMS error minimization based method for geometry under study.

As can be seen, the calculated filter coefficients have an excessively large magnitude, which in reality is impossible to implement. However, from the SPL map and the plots of the Acoustic Contrast and Insertion Loss, we can see that the algorithm works and minimizes the pressure amplitude in the dark zone. It turns out that a mathematical solution exists, but physically it is not realizable. It is also worth noting the degradation of the sound pressure in the bright zone and a significant increase in SPL outside the control area, on the sides. However, since the cost function only minimizes the sound pressure at error sensing locations (dark zone) and there are no imposed constraints and control on the bright zone, array effort, the obtained results are quite consistent. Another reason is that the above-described requirements for the number and location of sources, depending on the size of the dark zone (the number of error sensing locations) are not met, which also significantly affects the result. As noted above, this method is the simplest one, not suitable in our case, but it works well for other applications.

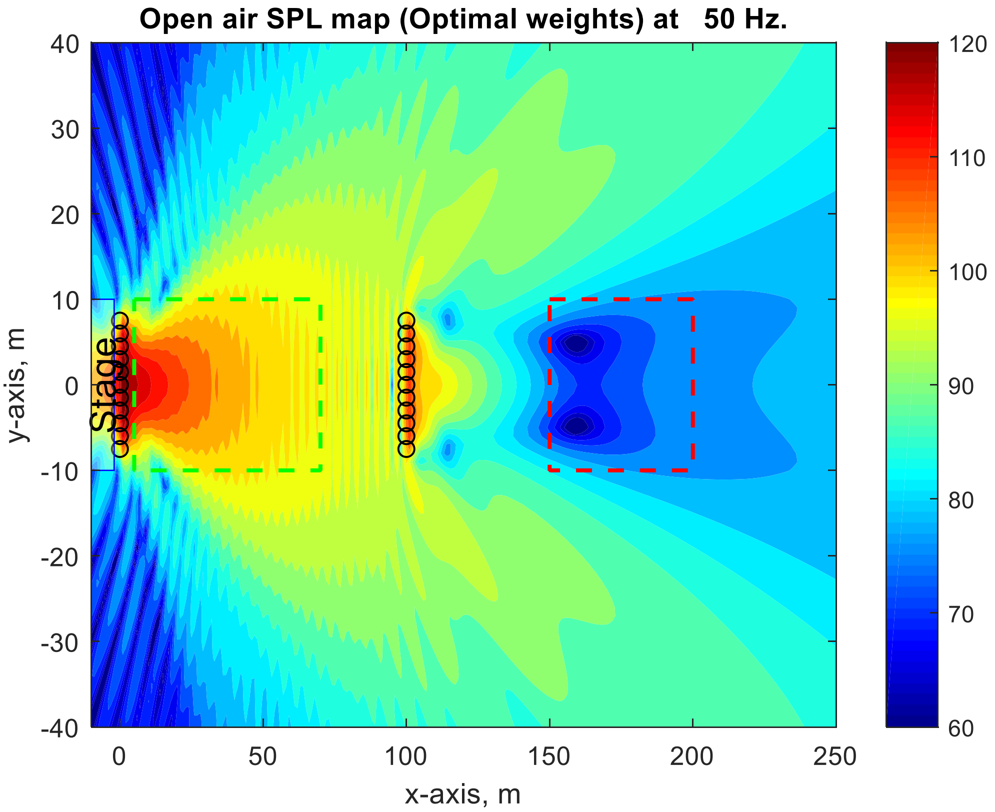
3.2 Simulation of the Acoustic contrast control method

As mentioned above, optimal source volume velocities are set as the eigenvector corresponding to the maximum eigenvalue of the matrix:

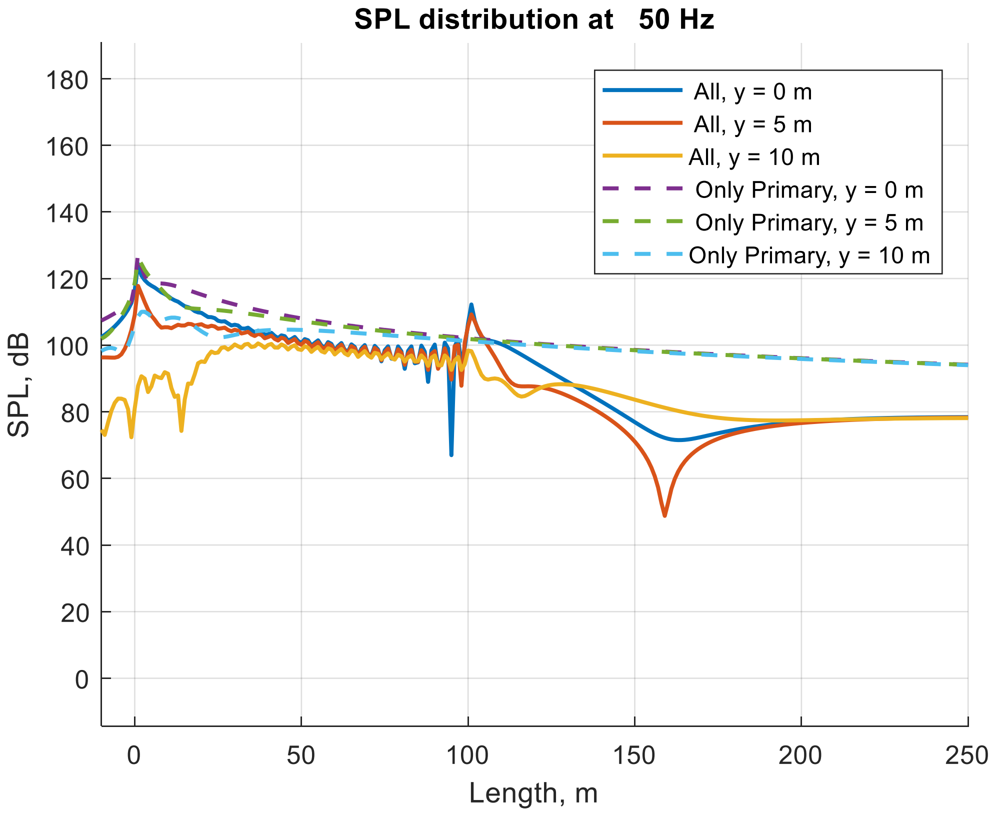
$$[\boldsymbol{H}_{\mathrm{d}}^{H}\boldsymbol{H}_{\mathrm{d}} + (\lambda_2/\lambda_1)\boldsymbol{I}]^{-1}[\boldsymbol{H}_{\mathrm{b}}^{H}\boldsymbol{H}_{\mathrm{b}}]$$

The eigenvalue problem was calculated in MATLAB. Results of simulation are depicted at Figure 26.

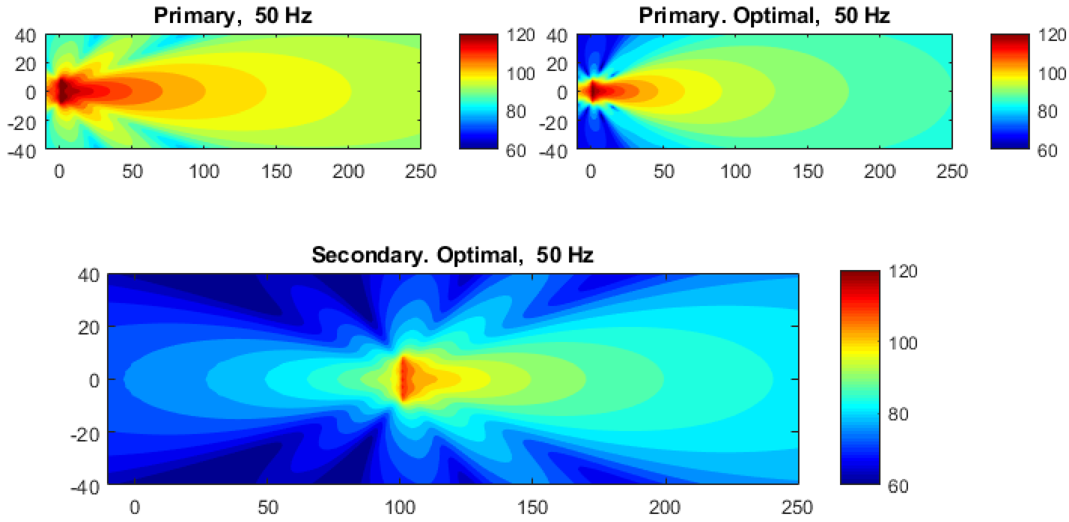
The ratio of Lagrange multipliers that determines the tradeoff between the performance and array effort has been set as 100. This value gives the most optimal results, but still some problems remain (to be discussed below).



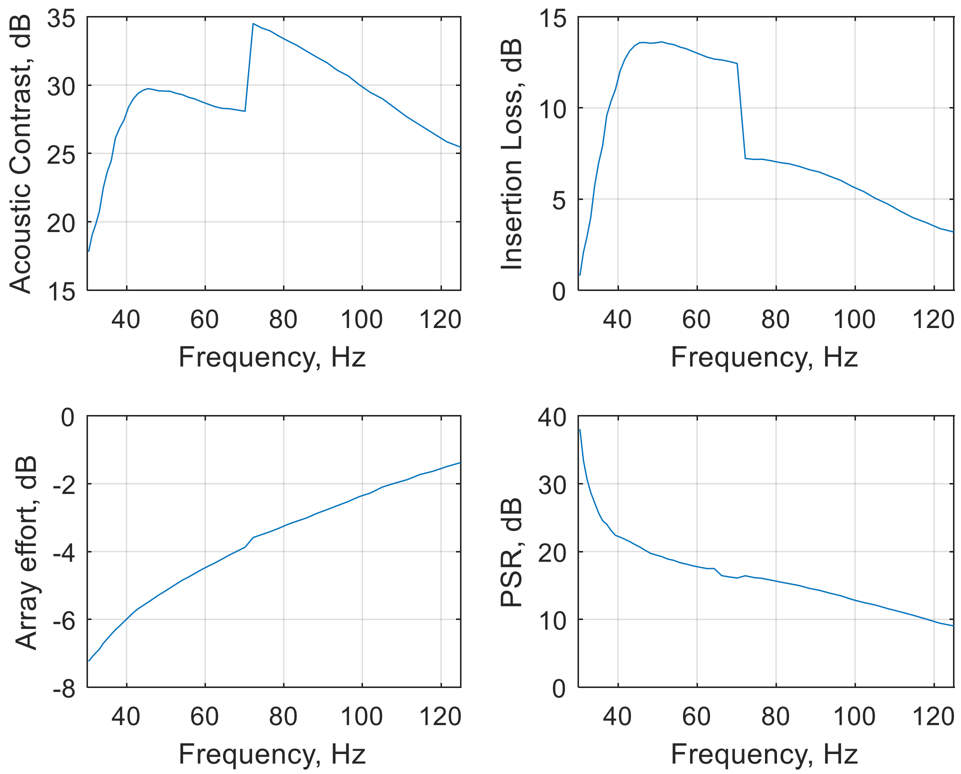
a) SPL map with calculated optimal volume velocities for control sources



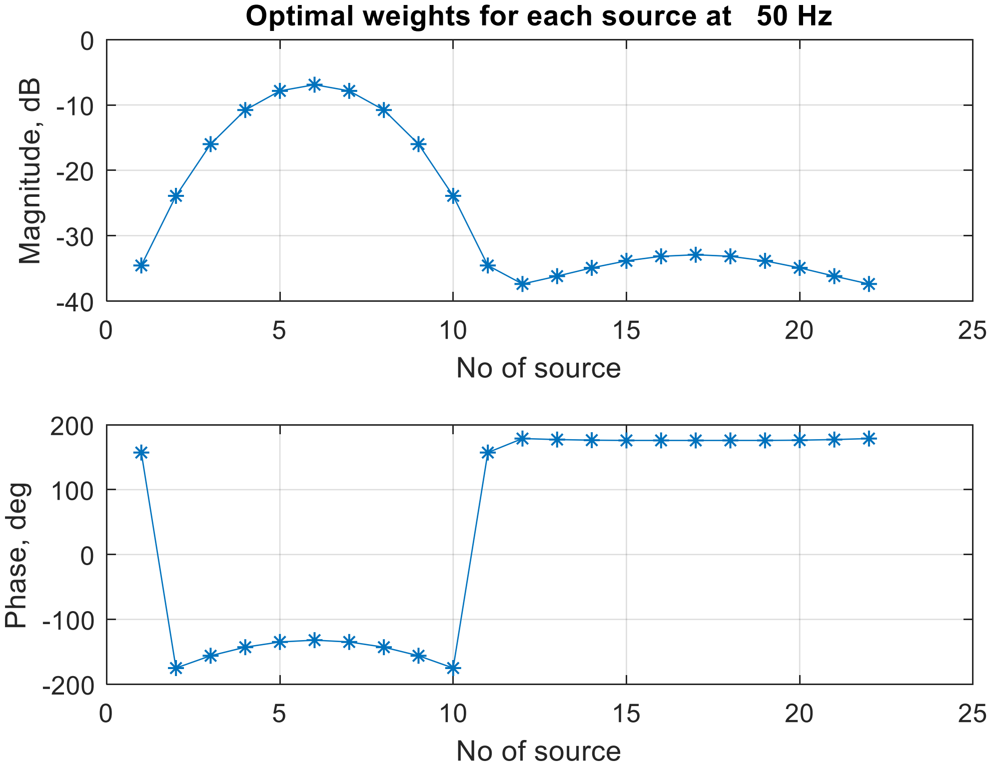
b) SPL distribution at different y-axis level



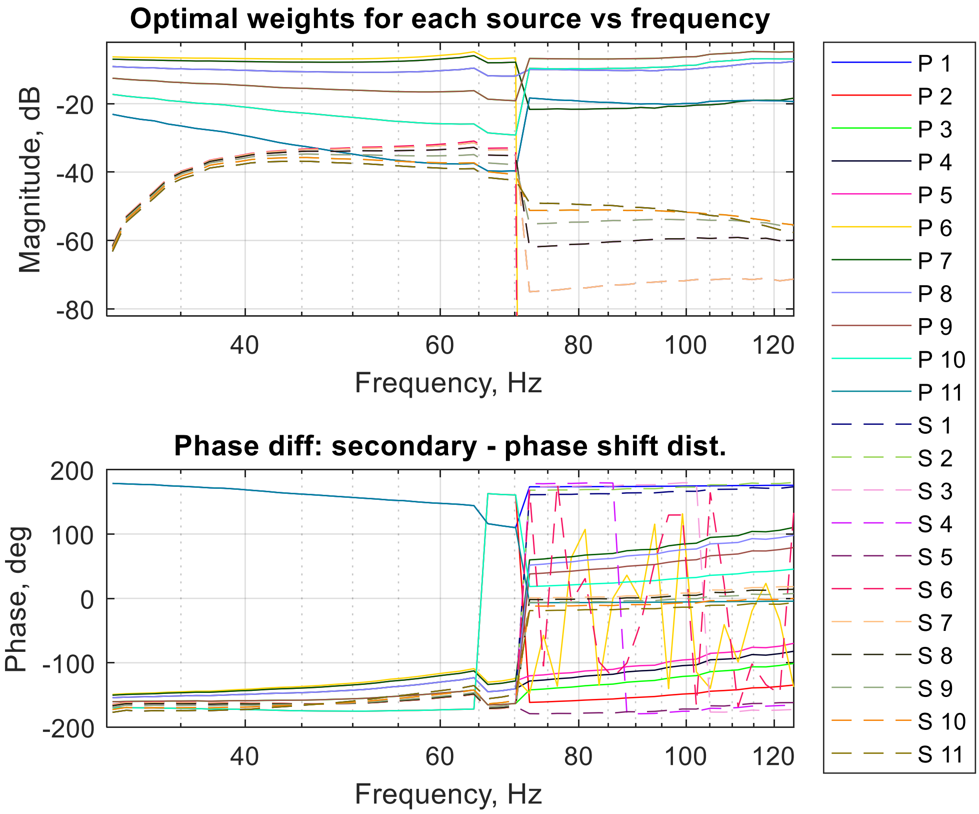
c) SPL map comparison for primary and secondary (control) arrays



d) Performance metrics



e) Magnitude and Phase for each loudspeaker at a given frequency



f) Magnitude and Phase for all loudspeaker over the frequency range

Figure 26. Simulations of the ACC method for geometry under study.

As in the previous case, the resulting solution cannot be physically feasible but at this time due to the low values of the volume velocities magnitude and their non-uniformity in the frequency range. In relation to the central element of the primary array, which at frequency of 50 Hz has a level of about -6 dB, the rest are physically turned off for the real system, although a mathematical solution has been found. Under the conditions of open-air events and specifically for our system under study, the physical result is not optimal since already in the bright (listening) zone there is a significant degradation, unevenness of the acoustic field and loss system headroom. However, unlike the previous case, the constraints for loudspeaker power consumption and for energy in the dark zone were introduced into the cost function. Therefore, for example, the values of array effort metrics is less and more consistent for case of ACC method. In addition, this method tries to limit the acoustic energy only in the dark zone, and not everywhere as in the previous case. On this basis, we can observe less energy amplification outside the controlled areas. Since finding the optimal loudspeaker strength comes down to solving eigenvalue problem, the solution is very sensitive to the possible numerical errors of inverting matrices and their bad conditioning.

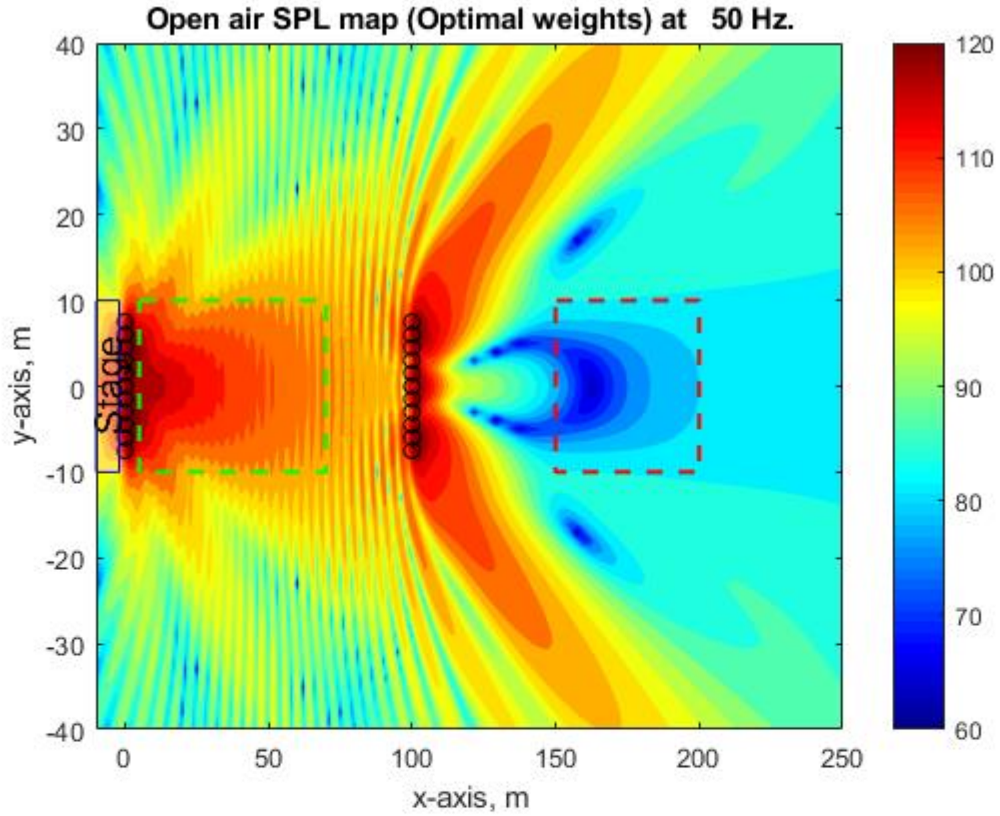
3.3 Simulation of the Pressure Mathing method

The solution of PM method, already discussed above, written as:

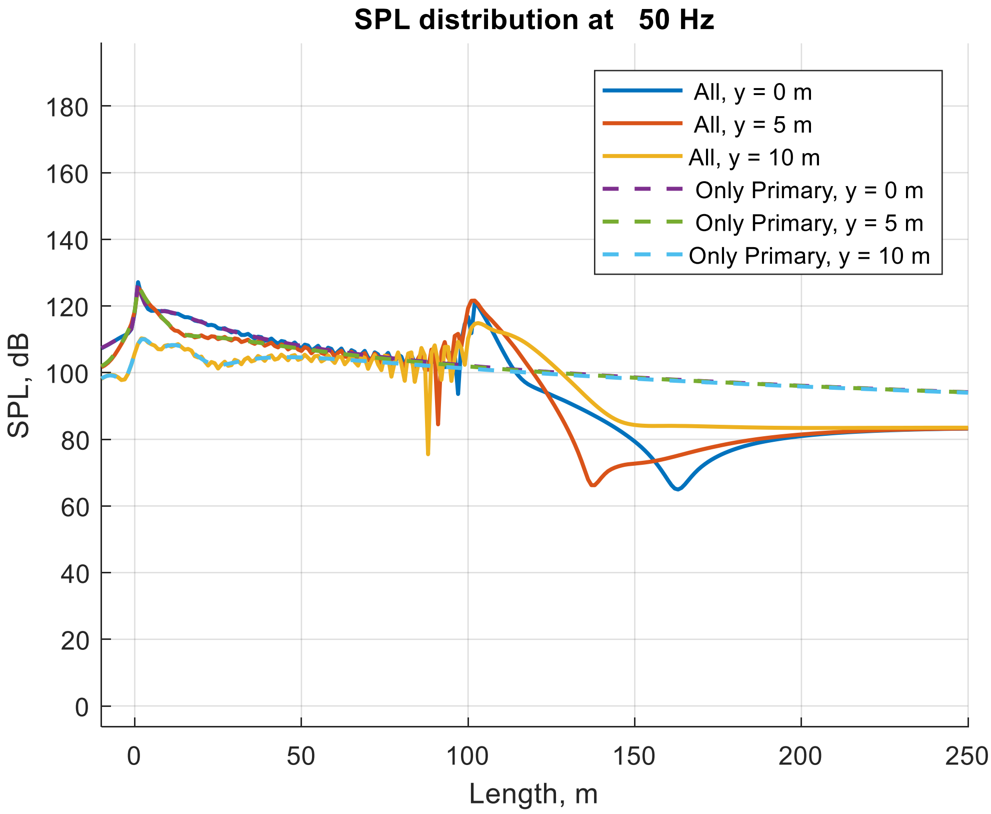
$$[H_{\mathrm{b}}^{H} H_{\mathrm{b}} + \lambda_1 H_{\mathrm{d}}^{H} H_{\mathrm{d}} + \lambda_2 I] g = H_{\mathrm{b}}^{H} p_{\mathrm{des}}$$

Here the Lagrange multipliers λ_1 and λ_2 regulate performance and array effort. For example, when λ_1 =1 means applying equal effort to matching the pressure in the bright zone and minimizing the energy in the dark zone. Decreasing λ_1 leads to applying effort for matching the pressure in the bright zone, when increasing λ_1 – more effort to minimizing the energy in the dark zone, so we get more acoustic contrast and IL but it affects also array effort which is mainly regulated by λ_2.

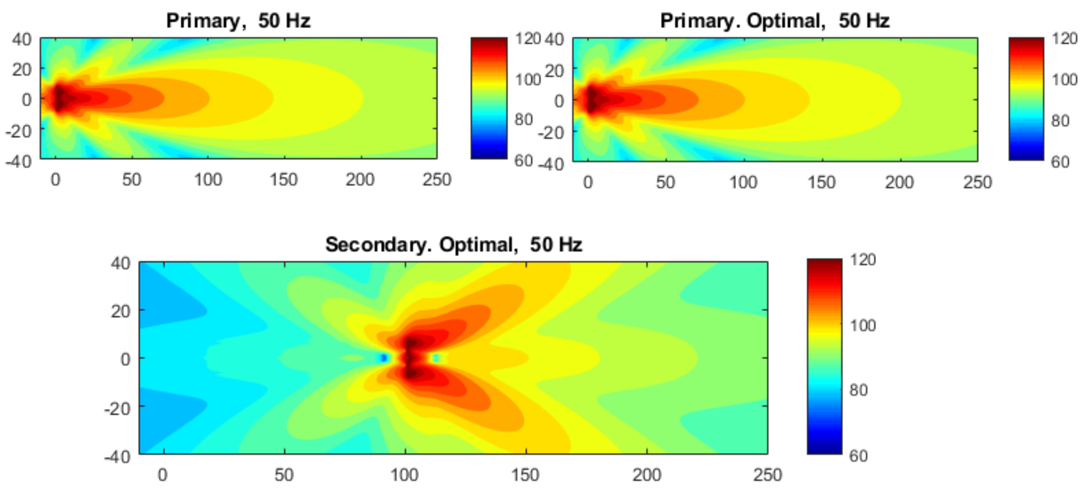
In order to find optimal solutions we will vary both λ_1 and λ_2. The obtained results presented at the figures below.



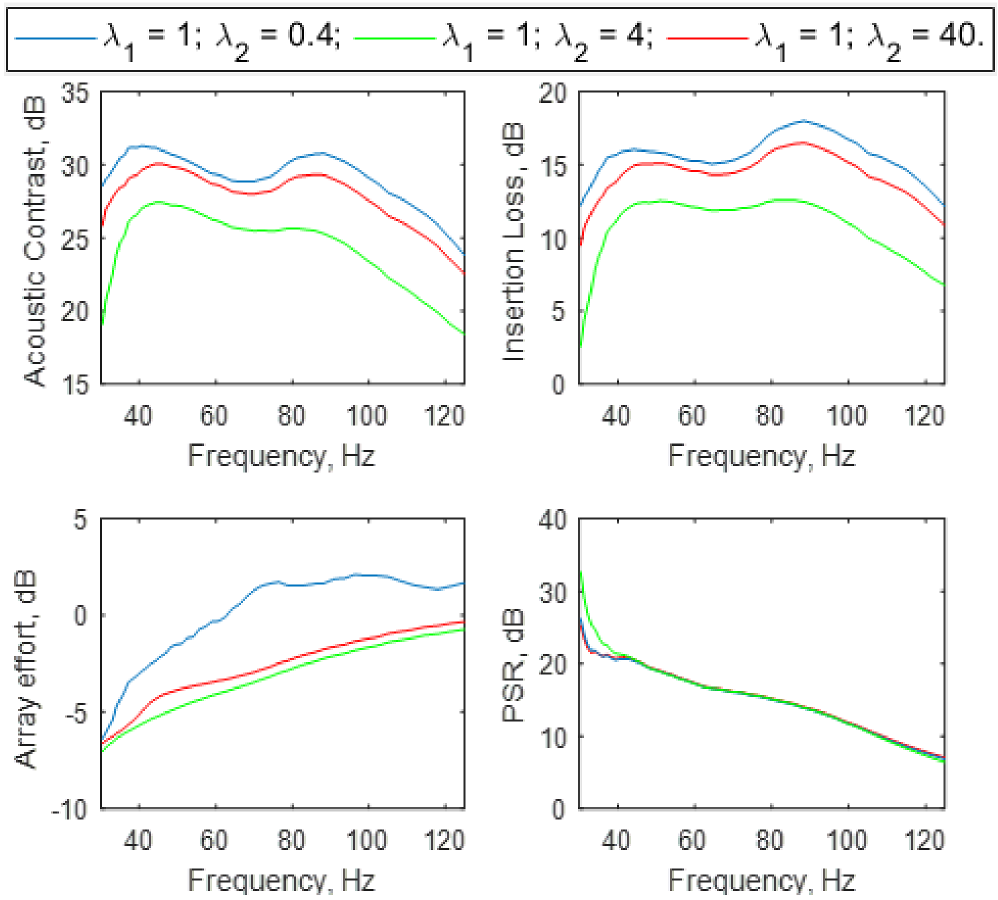
a) SPL map with calculated optimal volume velocities for control sources for λ_1 =1, λ_2 =4.



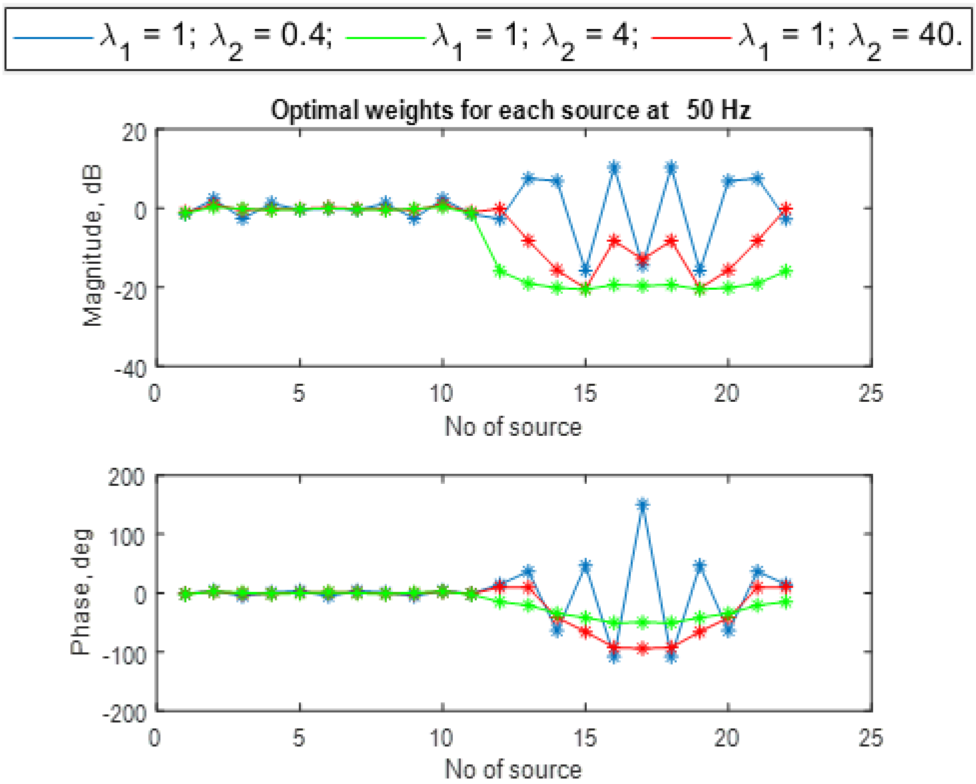
b) SPL distribution at different y-axis level for $\lambda_1 = 1$, $\lambda_2 = 4$.



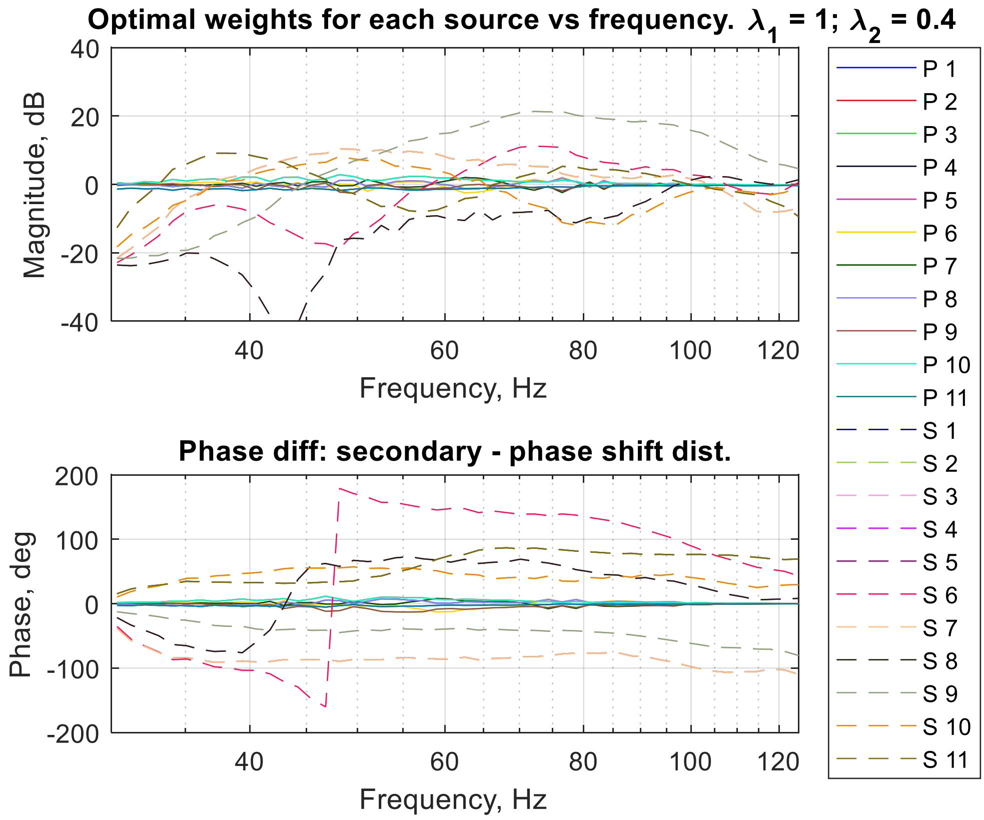
c) SPL map comparison for primary and secondary (control) arrays for $\lambda_1 = 1$, $\lambda_2 = 4$.



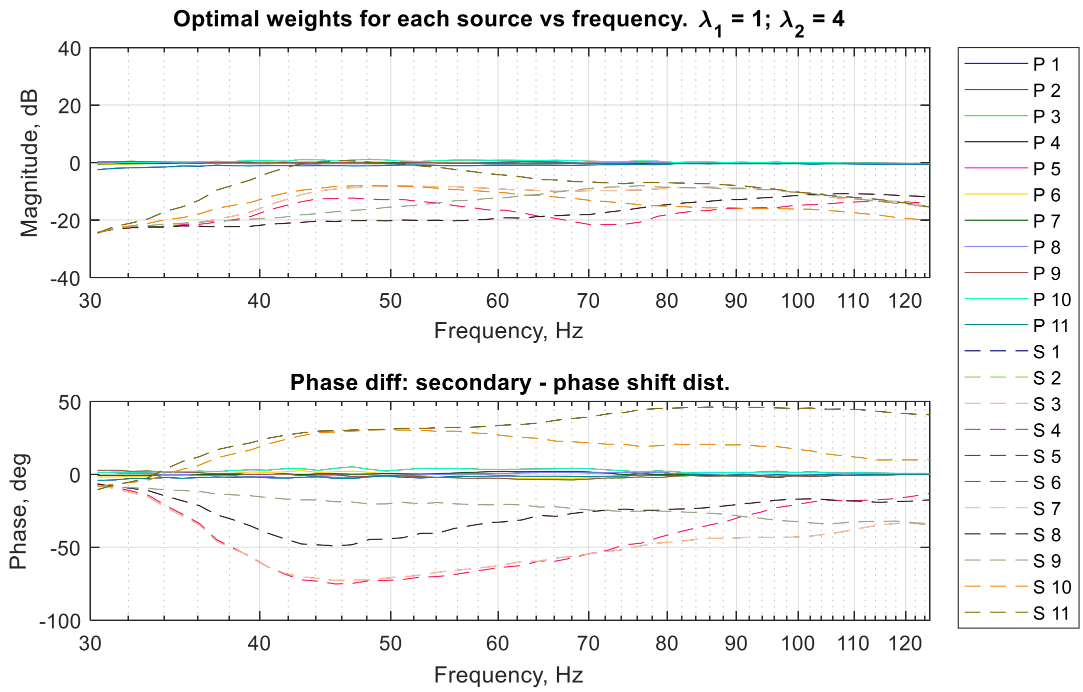
d) Performance metrics for different values of Lagrangians for λ_1 and λ_2



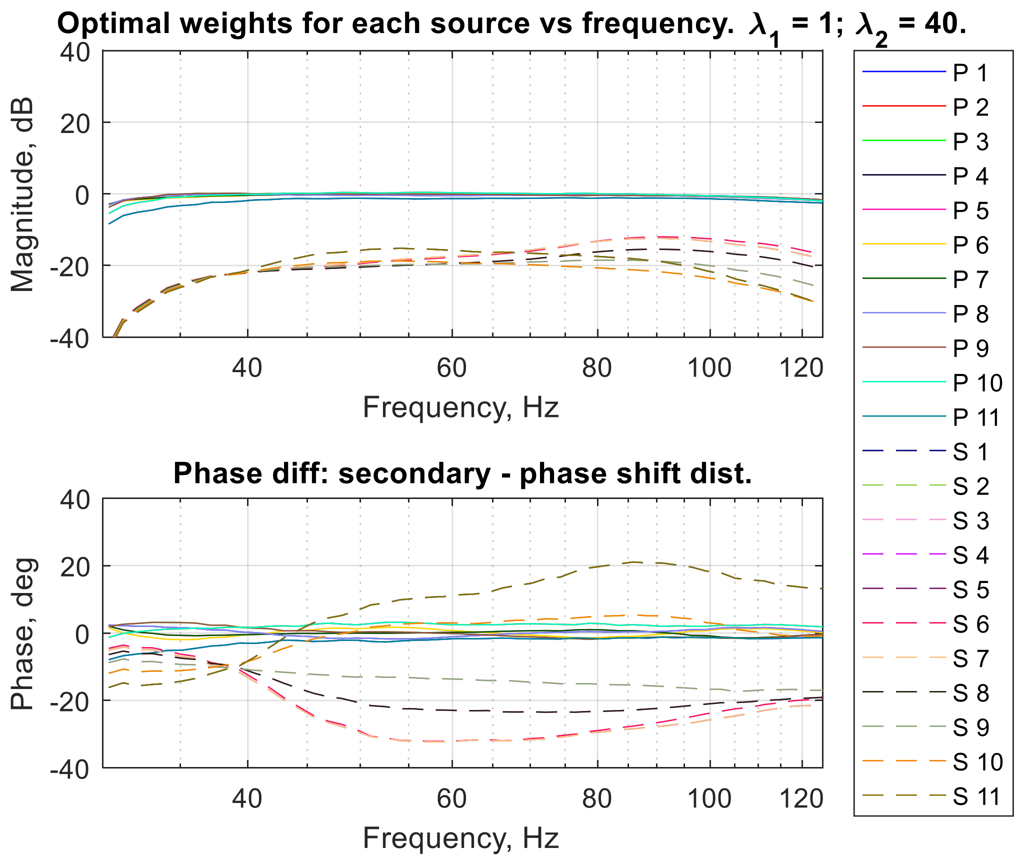
e) Magnitude and Phase for each loudspeaker at a given frequency



Optimal weights for each source vs frequency. λ_1 = 1; λ_2 = 0.4
Magnitude, dB
Frequency, Hz
Phase diff: secondary - phase shift dist.
Phase, deg
Frequency, Hz
P 1
P 2
P 3
P 4
P 5
P 6
P 7
P 8
P 9
P 10
P 11
S 1
S 2
S 3
S 4
S 5
S 6
S 7
S 8
S 9
S 10
S 11



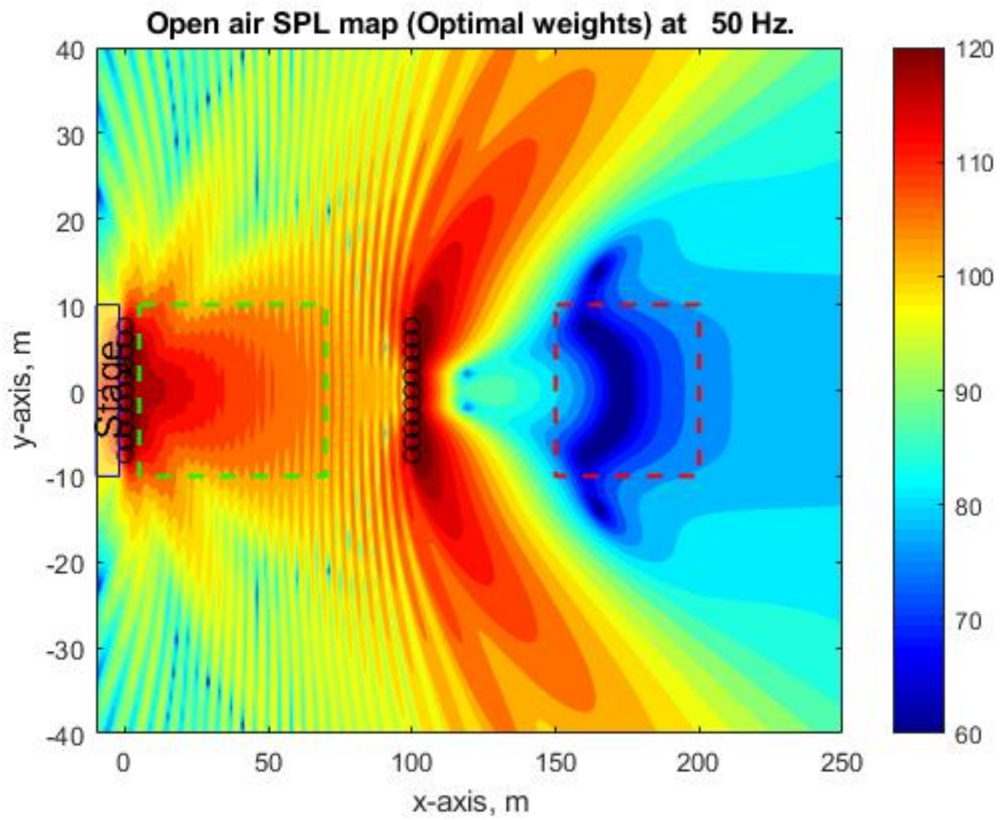
Optimal weights for each source vs frequency. λ_1 = 1; λ_2 = 4
Magnitude, dB
Frequency, Hz
Phase diff: secondary - phase shift dist.
Phase, deg
Frequency, Hz
P 1
P 2
P 3
P 4
P 5
P 6
P 7
P 8
P 9
P 10
P 11
S 1
S 2
S 3
S 4
S 5
S 6
S 7
S 8
S 9
S 10
S 11



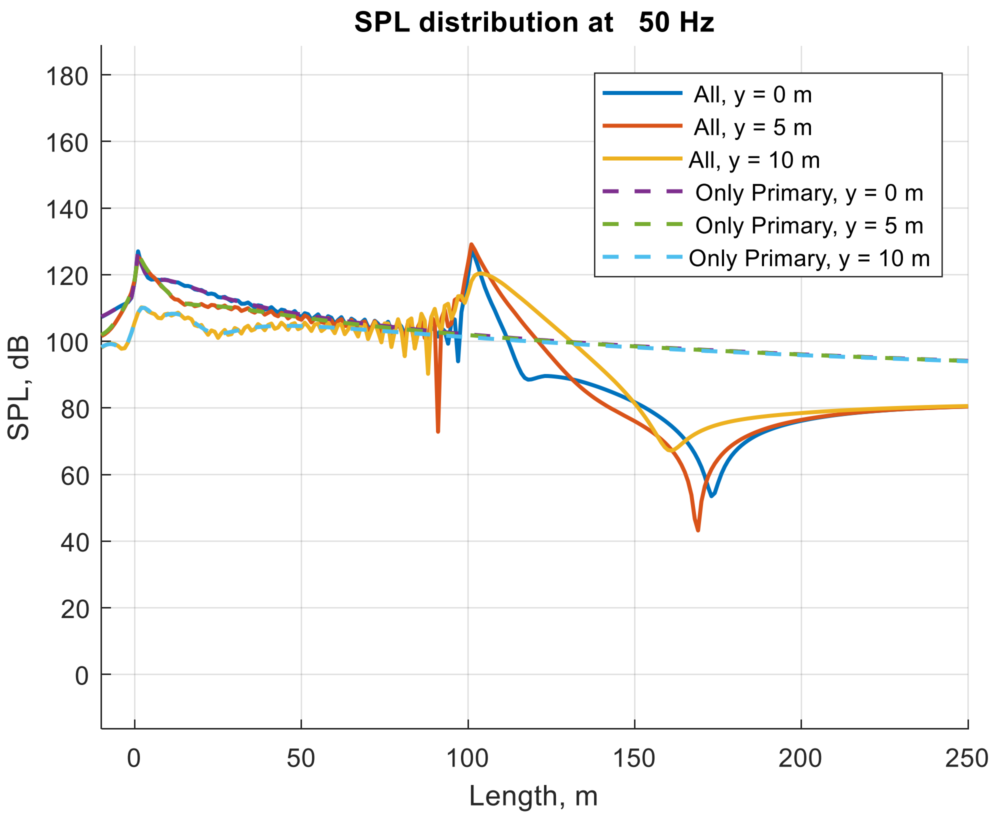
f) Magnitude and Phase for all loudspeaker over the frequency range for different values of Lagrangians for λ_1 and λ_2

Figure 27. Simulations of the PM method for geometry under study. Varying λ_2.

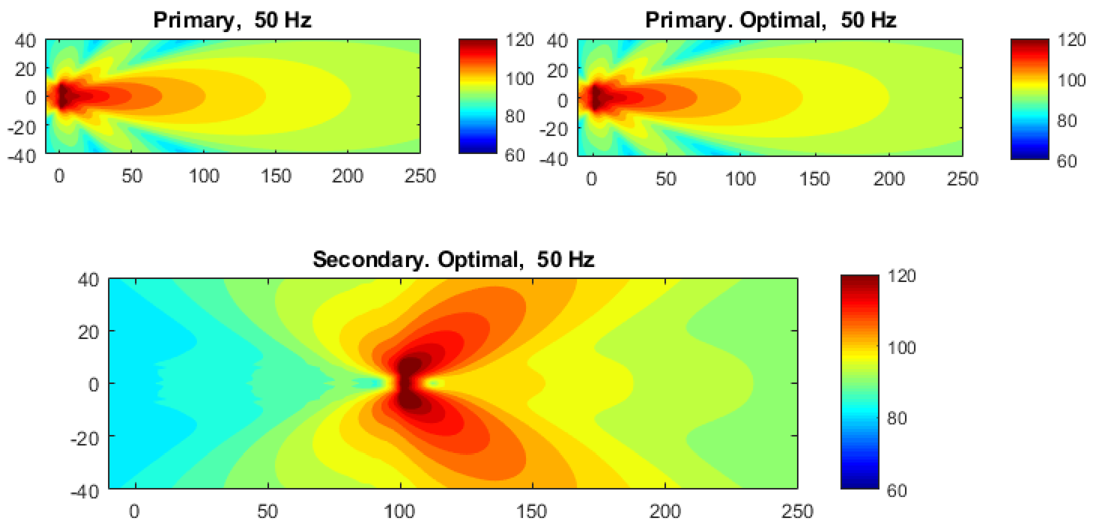
The following set of figures represents solution with varying λ_1 regularization parameter.



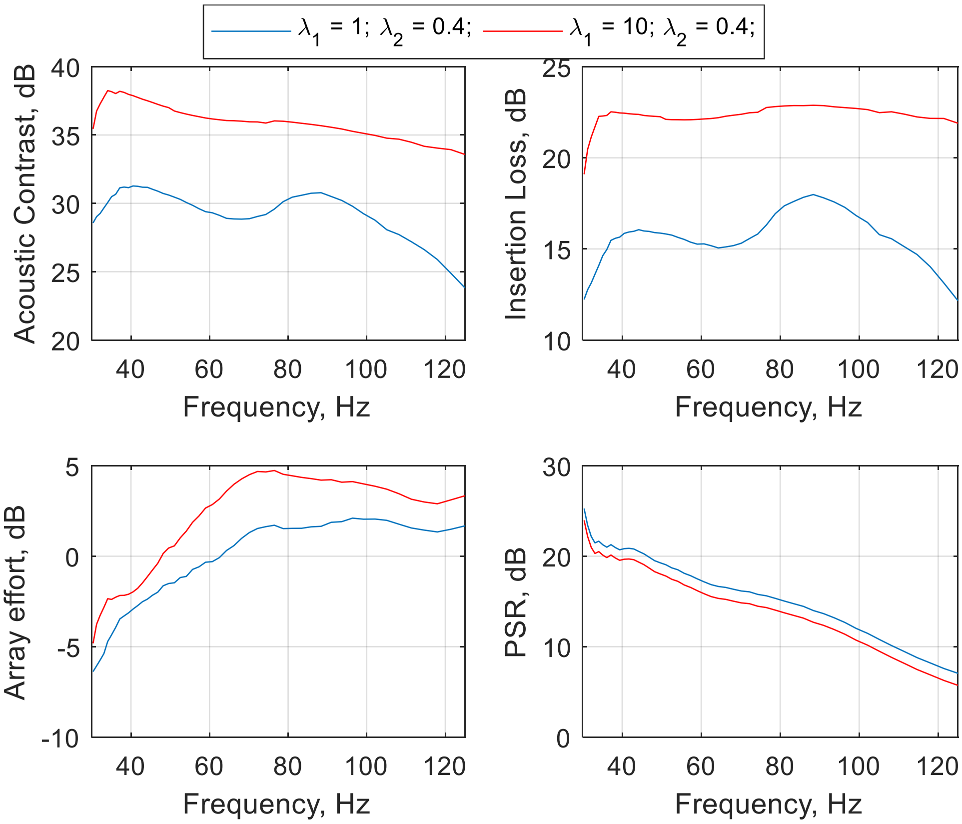
a) SPL map with calculated optimal volume velocities for control sources for λ_1 =10, λ_2=0.4.



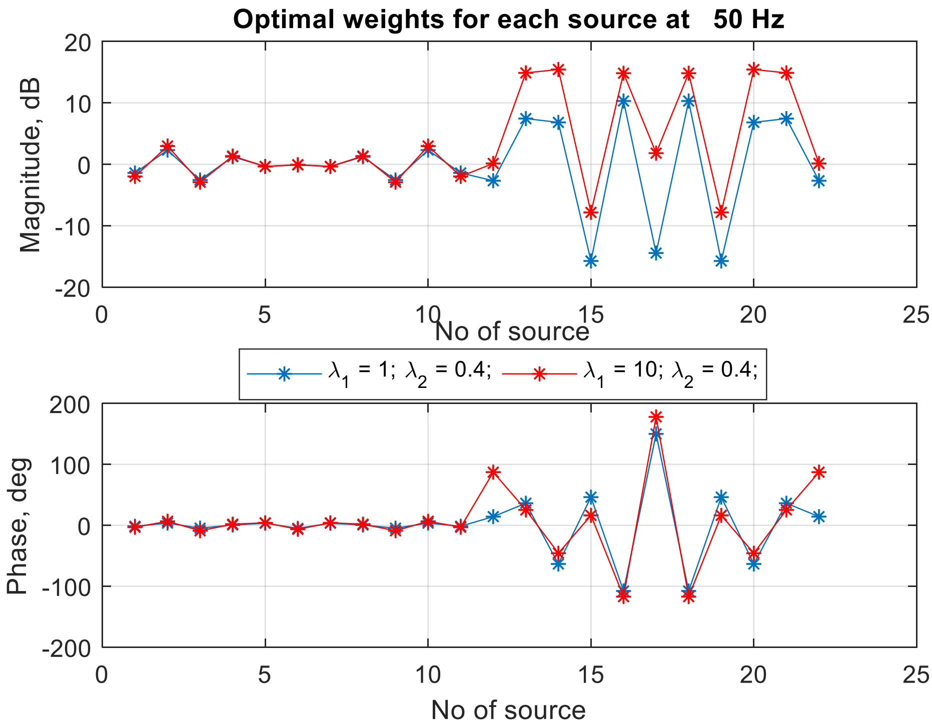
b) SPL distribution at different y-axis level for λ_1 =10, λ_2=0.4.



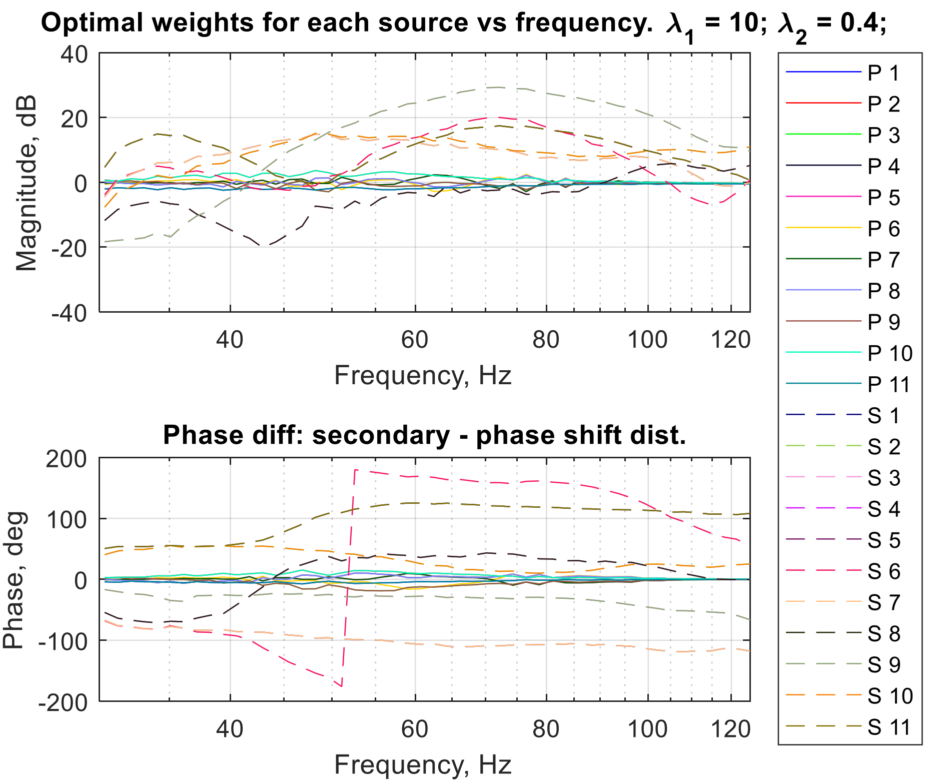
c) SPL map comparison for primary and secondary (control) arrays for $\lambda_1 = 10$, $\lambda_2 = 0.4$.



d) Performance metrics for different values of Lagrangians for λ_1 and λ_2



e) Magnitude and Phase for each loudspeaker at a given frequency

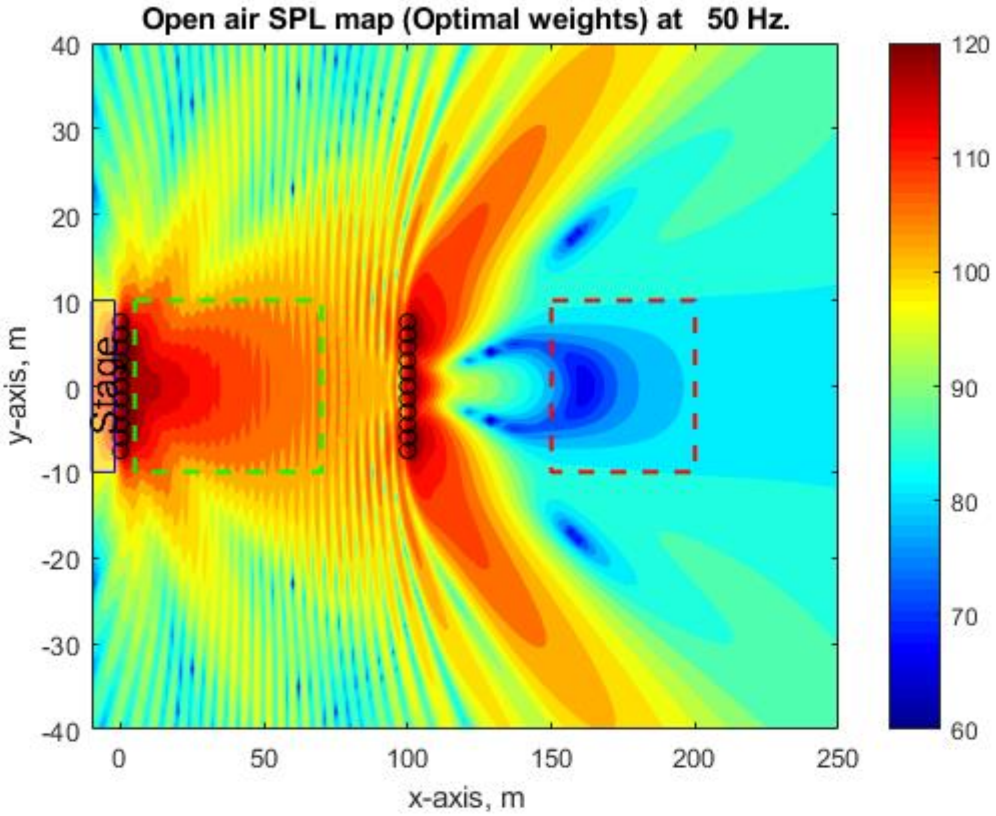


f) Magnitude and Phase for all loudspeaker over the frequency range for different values of Lagrangians for λ_1 and λ_2

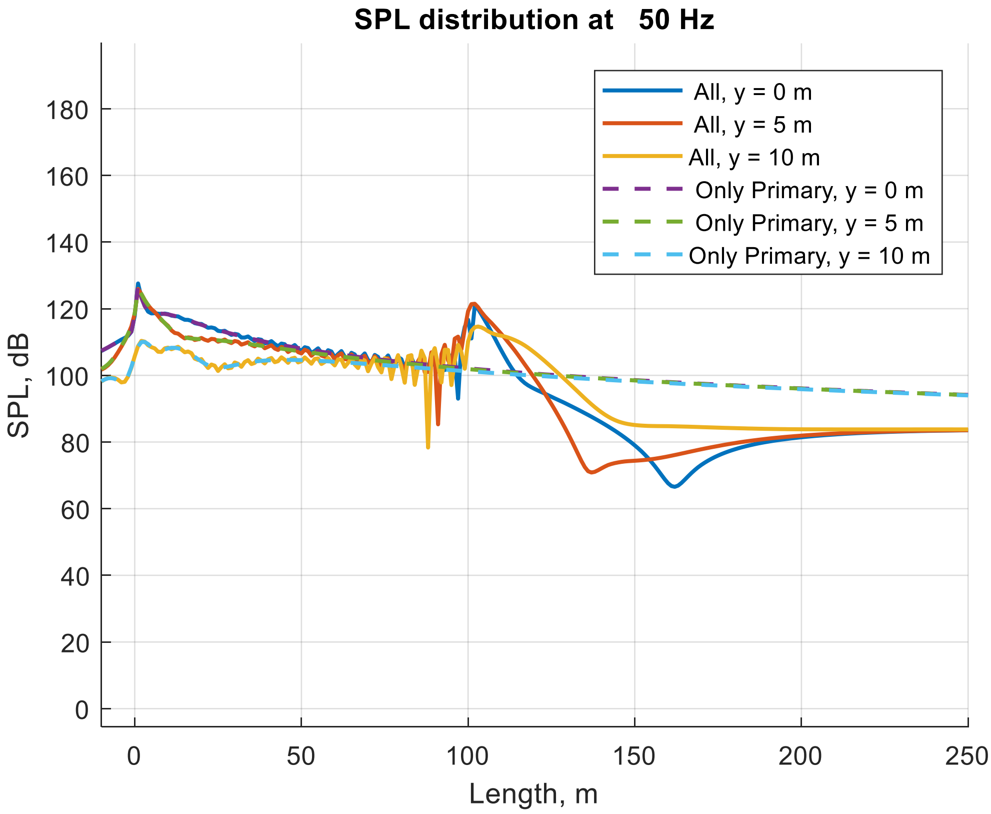
Figure 29. Simulations of the PM method for geometry under study. Varying λ_1.

3.4 Simulation of the combined PM-ACC method

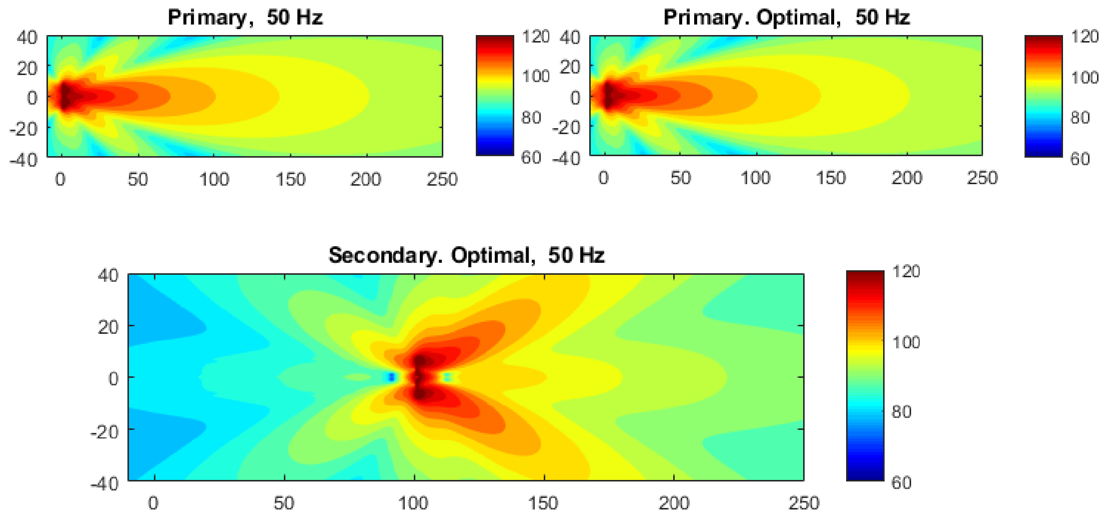
The results for combined PM-ACC method are presented below. In this modification, as mentioned at the beginning, only secondary (control) loudspeaker array is controlled, while primary stay untouched. There are two regularization parameters: k – parameter that weights the reproduction error in the bright zone relative to the energy in the dark zone and Tikhonov regularization term λ, which controls array effort.



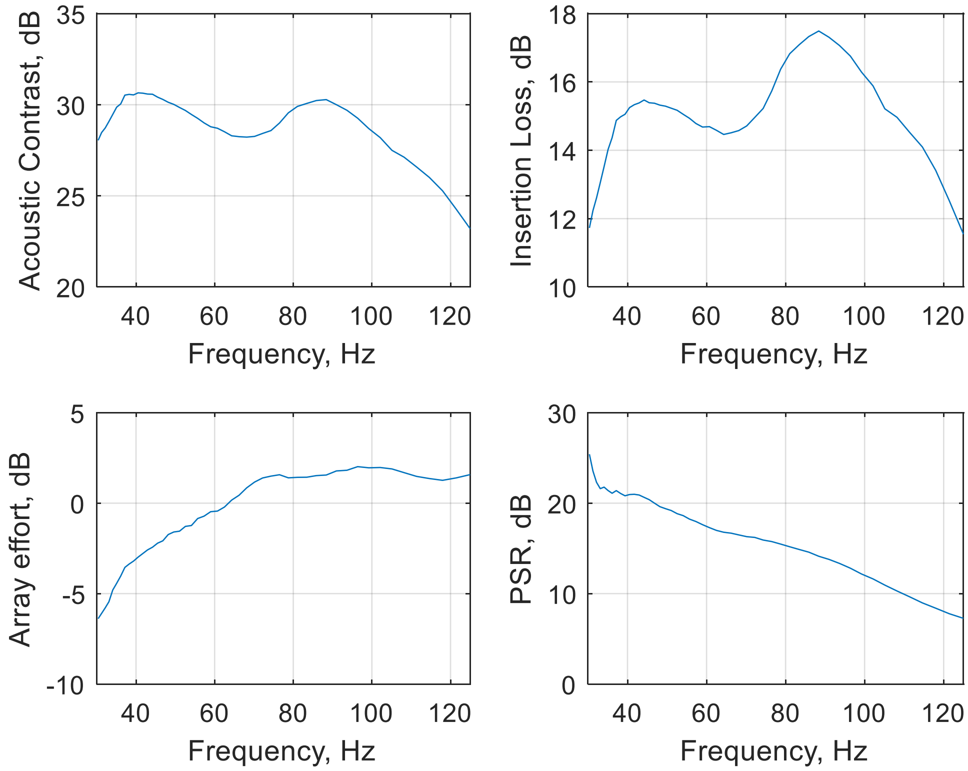
a) SPL map with calculated optimal volume velocities for control sources, k = 10 and λ = 4



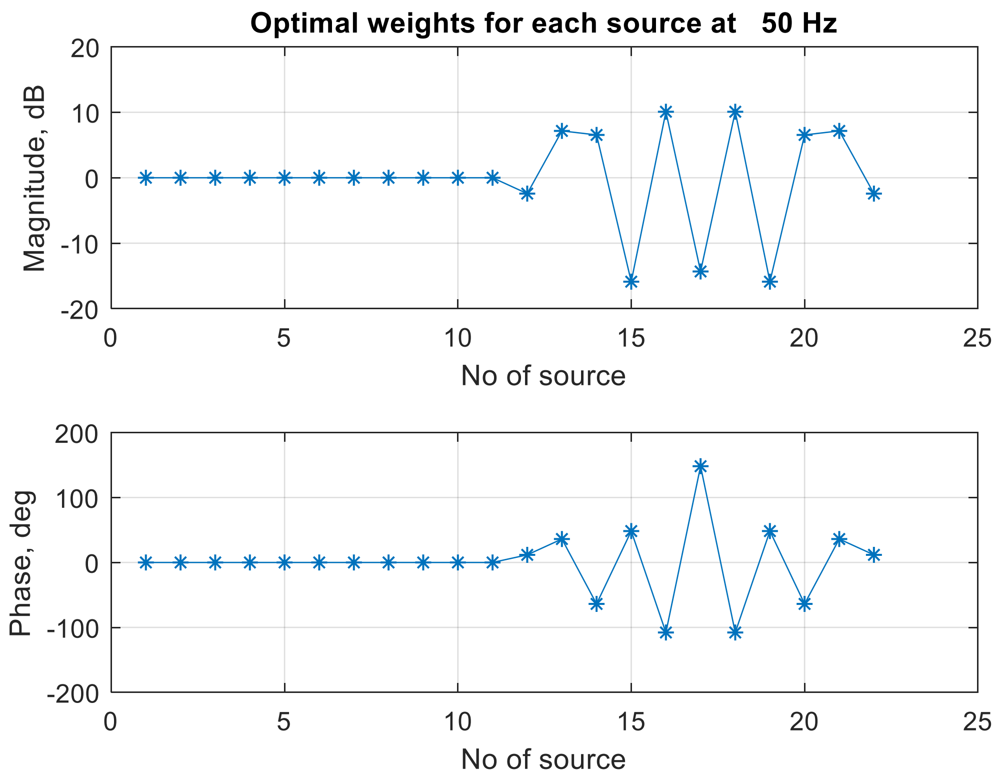
b) SPL distribution at different y-axis level, k = 10 and λ = 4



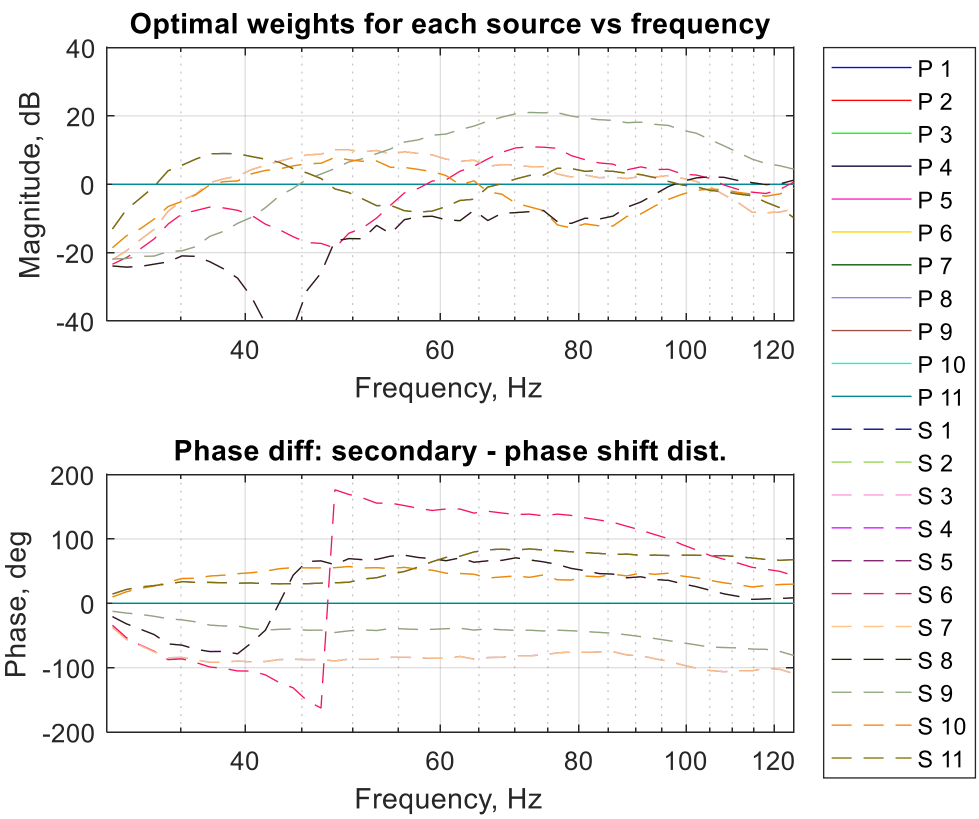
c) SPL map comparison for primary and secondary (control) arrays, k = 10 and λ = 4



d) Performance metrics, k = 10 and λ = 4



e) Magnitude and Phase for each loudspeaker at a given frequency, k = 10 and λ = 4

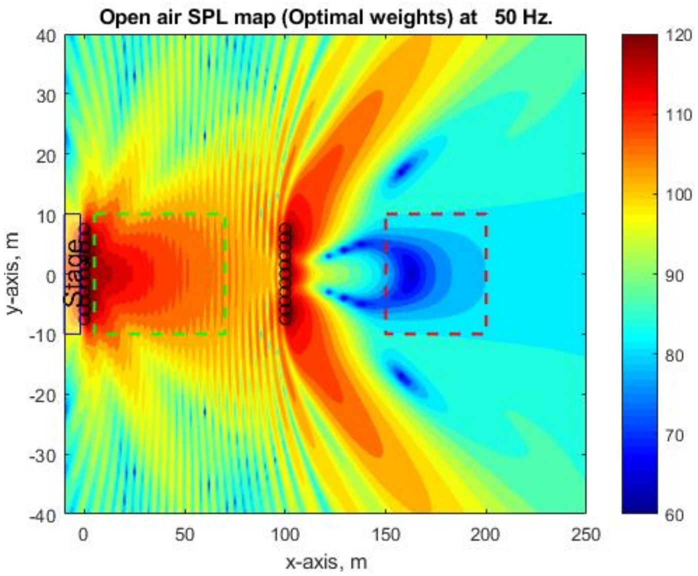


f) Magnitude and Phase for all loudspeaker over the frequency range, $k = 10$ and $\lambda = 4$

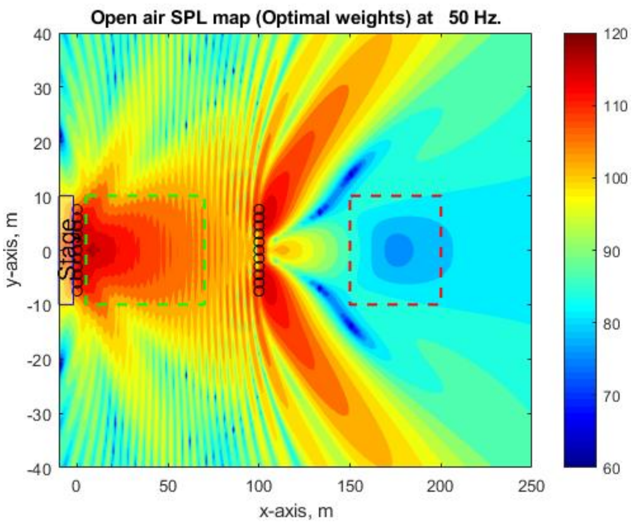
Figure 30. Simulations of the combined PM-ACC method for geometry under study.

3.5 Optimal Sound field calculation at different elevation

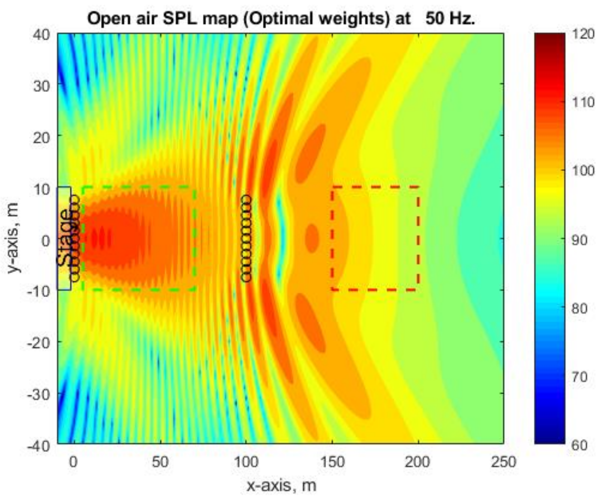
At the beginning all calculations were done for a 2D case: propagation transfer function matrices are modeled at one height (listening plane height), cost function and constraints were formulated for 2D problem. Therefore, it is necessary to take into account the change in the sound field at different heights, the following figures represents this simulation.



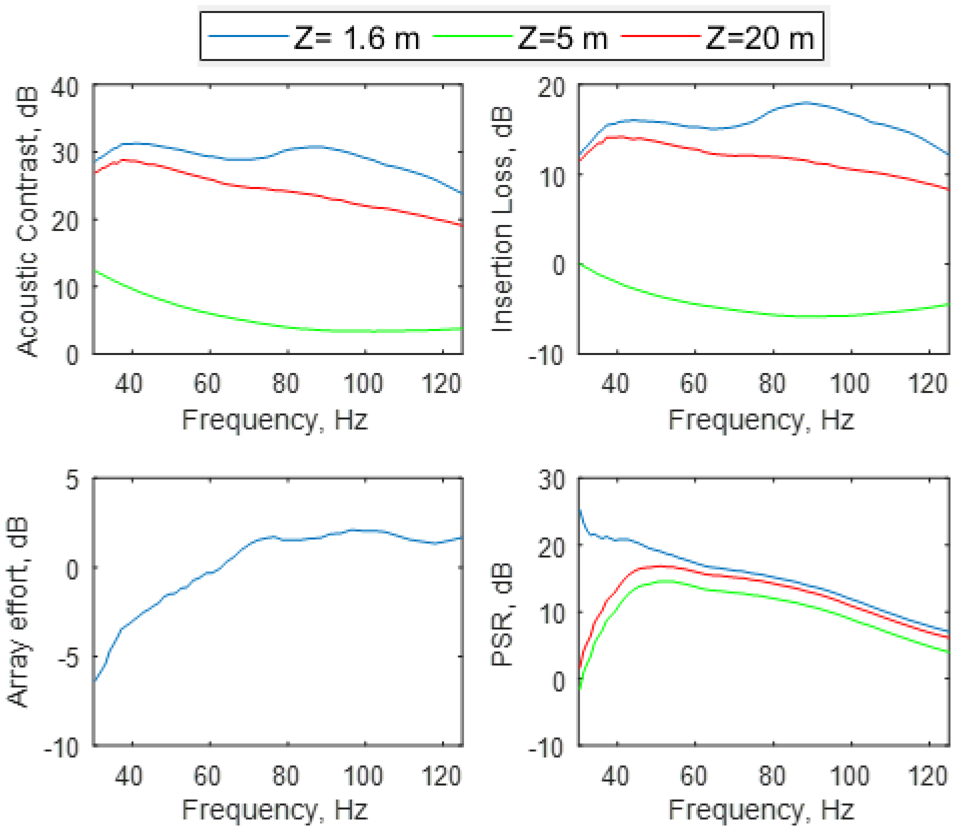
a) Z=1.6 m



b) Z=5 m



c) Z = 20 m



d) Performance metrics

Figure 31. Sound field at different elevation. The optimum source weights calculated with PM method at the listening plane height (1.6 m), $\lambda_1 = 1$; $\lambda_2 = 0.4$.

3.6 Evaluation and conclusion on the PM simulation results

As can be seen, the PM method and its modifications allow obtaining the most optimal results for the existing problem and outstanding the methods discussed above. First, changing the regularization parameters (Lagrangians [19]) one can achieve the desired results. However, ideally, in order to get the best control, the regulation parameters should be frequency-dependent and chosen more precisely.

Secondly, the optimally chosen regulation parameters allow us to obtain physically realizable filters within the amplitude and small fluctuations in the frequency range.

Thirdly, all performance metrics show good results: the insertion loss is about 15 dB in a wide frequency range, while the array efforts in the range -5 to 3 dB, and there is a high value of acoustic contrast. The difference between the desired and reproduced sound fields is minimal. It is also worth noting the amplification of the sound field on the sides, but as in the previous case, these zones are not controlled by the formulated cost function. The same applies in changing the sound field at different heights. In our case (2D calculation) with increasing height, the dark zone degrades and the sound field amplifies. This must be taken into account in order to avoid a situation when reducing noise for some group of people will not increase it even more for others. Ideally, by defining additional constraints for cost function, it is possible to both avoid pressure amplifies in non-controlled areas or with increasing height. However, this may require an increase in the number of loudspeakers and a transition from 2D radiation to 3D, which will generally complicate the system and increase errors.

CONCLUSION

When it comes to Active Noise Control for Open Air events it is important not only to reduce the noise in one area but also to preserve the high quality of sound for the listeners. Therefore, in this case, the methods of creating sound zones can be applied.

To calculate sound zones, it is necessary to simulate or measure the propagation transfer functions. The accuracy of the model or measurement has a key influence on the stability and robustness of the method and its results. The main problem with measurements is a large number of measured points (especially for open air events) and a change in atmospheric conditions during measurements. It is also necessary to update the measured information over time as the atmospheric conditions are variable. In the case of modeling the propagation transfer functions, the model should take into account various atmospheric conditions as much as possible, but here too there is a question about updating the model.

The main source of errors with a significant negative impact on the results is precisely atmospheric conditions, since local changes in temperature, wind speed and its direction are possible, which change with time. The influence of the environment, such as ground reflections and the presence of the audience (propagation trough the audience), may not be taken into account in the first approximation, as they have (/ do not have) effects at low frequencies or the effect is significantly less than atmospheric conditions. Neural network technologies, Bayesian interface can more fully take into account the influence of atmospheric conditions.

Based on the performed simulations of several methods, the PM method with optimal regularization parameters gives good and consistent results, as evidenced by the performance metrics. The PM method also takes into account the phase while minimizing the difference between the reproduced and the desired field. While AC provides more contrast between zones but sound artifacts are possible within the controlled area.

The main problem of all methods is that by choosing non-optimal regularization parameters a mathematical solution can be found, but practically not feasible due to an excessively large magnitude of complex volume velocity or a large gain difference between neighboring loudspeakers.

Controlling and reducing noise in one area can lead to significant gain in others or at different levels of height. Therefore, for each particular case, ideally, the cost function should be introduced using additional constraints.

Speaking about the actual use of sound zone methods on open air events, it is worth noting that only by 6-10 dB one was able to reduce noise in the dark zone [8], and taking into account the fact that noise reduction issues are becoming more relevant, this topic requires further research.

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