Management Of Networked IoT Wearables – Very Large Scale Demonstration of Cultural Societal Applications
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1 Executive Summary

This report documents the first steps of the development of the Adaptive Sound Field Control System (ASFCS), the local Quiet Zone System (QZS) and the Noise Monitoring for the MONICA-project, all with the purpose to mitigate noise annoyance with neighbors and non-participating visitors during outdoor musical events, as well as to enhance the sound quality and musical experience of the audience. The report present initial simulations and measurements in order to validate the chosen strategies. Moreover, signal chains and protocols, as well as necessary hardware have been identified and documented.

The two important prerequisites of ASFC to perform well in the practical situation are 1) obtaining the accurate sound propagation model of the venue and 2) providing the suitable controlling signal through the secondary sources. The former is enabled using various sensors and microphones and the latter is done by monitoring the whole PA system output. Monitoring of electro-acoustic channels is a straight-forward task, but it is a difficult problem in practice. The signal format (analog or digital) and protocol (AES/EBU, MADI) change as passing equalizer or amplifier, and resulting different EQ/Delay filters are applied to each step. For this reason, sensible integration of ASFCS to the venue's PA system is needed.

The sound zone signal processing and optimization algorithm in ASFCS is based on the combined solution of Pressure Matching and Acoustic Contrast Control (PM-ACC). The PM-ACC solution fits well for the objective of the project because the sound field control in the dark zone must not negatively affect the sound experience in the audience area. Aiming for a very large scale cultural event, there should be some careful adaptation of the technique. Control of low-frequency band would require a regularization technique that gives a robust solution that can work in the outdoor conditions. Furthermore, 'large-scale' condition will impose various physically constraint real-time issues, leading modification of the given optimization scheme.

It is shown through simulations that the ASFS can considerably increase the difference in sound pressure level between the audience area and the venues surroundings in ideal environments. However, a compromise has to be made between the size of the control region in the neighbourhood and the equipment and computational modelling effort needed to cover a larger area. Comparing an extension mode of the ASFCS (only secondary sources being controlled, leaving the main PA uncontrolled) with a full mode (controlling both secondary and main PA sources) we can conclude that for a small dark zone, the benefit of using the full mode is small, but for the large dark zone, this benefit is essential, with an increase of the contrast of 10-15 dB. Moreover, it has been shown that due to model uncertainty, using model updating/machine learning techniques updating the propagation model will be essential for having a good result; a benefit of about 10 dB was found in the low frequencies. Experimental verifications of these simulations are under way in the time this report was written.

Local quiet zones in loud environments can support safety such that they provide a place to communicate health or security issues or just a zone of limited noise exposure. These quiet zones are integrated into the MONICA architecture but they work also as a self-contained solution. The system is made of active electronics which cancel out the low frequencies and a wall which blocks the high frequencies. In order to achieve the highest possible attenuation in the zone without interfering the surrounding optimized sound-field the system has to be local. In that way the sound energy transmitted from the cancelling loudspeakers is kept within a dedicated region (as much as possible) and the passive barrier (the wall) can be smaller which reduces the size of the acoustic shadows.

Initial measurements of the Source Separation/Contribution techniques of the Noise Monitoring System have been successfully preformed.

In all, the initial validations of Adaptive Sound Field Control system as well as the Noise Monitoring configuration, using simulations and measurements, have shown positive results, but also indicating potential weaknesses to be aware of in the coming development.

2 Introduction

This chapter outlines the purpose, background and context of this deliverable as well as the structure of the remaining document.

2.1 Purpose, Context and Scope of this Deliverable

The objective of WP4 is to deploy components that can mitigate noise annoyance with neighbors and non-participating visitors during outdoor musical events. The main approach to do so is using an Adaptive Sound Field Control (ASFC or ASFCS)[1], consisting of loudspeaker arrays with an adaptive model updating system that adjusts for changes in climate and audience configuration. The ASFCS will be developed as a Sound Zone System (SZS) and will be integrated with the organizers Public Address (PA) system into the Acoustic Closed Loop System. The WP also contains a noise monitoring system. Data for the model updating of the ASFCS will be provided by stationary and wearable IoT sensors and apps. As such, WP4 is structured as follows:

- Task 4.1 Loudspeaker Array Configuration
- Task 4.2 Microphone Sensor Configuration
- Task 4.3 Sound Propagation Model Updating
- Task 4.4 Noise Annoyance Monitoring

This deliverable documents 'Validation of the Adaptive Sound Field Control (ASFC) for far and near field control, etc. (T4.1) and Noise Monitoring System Configuration'. The 'Noise Monitoring System Configuration' is located in Task 4.2 and 4.4; much of this task is however rather described in D4.4 'Precision IoT enabled Microphone Sensor 1'. The 'Sound Propagation Model Updating' of T4.3 is only briefly considered in this document and mainly left for the coming deliverables D4.2 and D4.3. The documentation of the task 4.1 will be based on its three subtasks:

- Subtask 4.1.1 Audience area loudspeaker array configuration – far field control
- Subtask 4.1.2 Quiet spots – near field control
- Subtask 4.1.3 Sound zone signal processing and optimization

2.2 Background

The WP will work on three physical scales: a) the external region – minimizing Annoyance; b) the audience area – maximizing Sound Quality[2]; and c) quiet spots within or close to the audience area – minimizing Loudness.

Most modern sound reinforcement systems are based on the line array principle, which allows for the control of directivity of the sound radiation of high and mid frequencies. However, the radiation of low frequencies cannot be as easily controlled, as sound waves at these frequencies are less attenuated by air and reflections from boundaries and are damped the least by the structures of residential buildings. Low frequencies are therefore the most critical frequencies in the noise problem of outdoor concerts. As controlling the sound field over large areas with a feasible number of loudspeakers is restricted to low frequencies, tackling the low frequency problem with this method is appropriate.

2.3 Tasks

The tasks in WP4 is here described shortly (mostly corresponding to the text in ANNEX 1 (part A)).

[1] The abbreviation ASFC is used denote the concept of the method. For the physically implemented system, ASFCS is used.

[2] Sound Quality here includes several perceptual dimensions, including Loudness, Directivity, Distortion, Echos etc. In this report we mainly focus on the Loudness and to some extent the Directivity aspect. In the coming deliveries D4.2 and D4.3, more perceptual dimensions will the considered.

2.3.1 Loudspeaker Array Configuration (Task 4.1)

As current PA systems are designed primarily with the coverage in the audience area in mind, improvements can be achieved not only by adding control sources, but also by reconfiguring the main PA design in light of the new performance requirement, as it will influence on the number of control sources needed and the achievable attenuation. Therefore, the MONICA ASFCS, consisting of the main PA and additional control sources, will be optimized as a complete system. At high frequencies (HF) the sound field control might be combined with passive solutions (absorbers and screens) – at least the quiet zones.

2.3.2 Audience area loudspeaker array configuration – far field control (Task 4.1.1)

This subtask addresses the ASFC of the external region as well as the audience area. In addition to the existing PA system loudspeakers, loudspeakers will be added to control the sound level in the external zone. The PA system input signals will be fed into the ASFCS for further use. At low and mid frequencies (LF/MF), an ASFCS will be deployed, using the concept of sound zones. The ratio of the acoustic energy in ensonified[3] zones to the acoustic energy in a quiet zone will be maximized, with maintained sound quality in the ensonified audience area. Also, the loudspeaker configuration and location will be optimized. The current meteorological conditions need to be taken into account in the control loop, described in T4.3.

2.3.3 Quiet spots – near field control (Task 4.1.2)

Within or very close to the audience area, smaller quiet spots will be created, intended for security personnel and for conversation spots. It is here crucial to not generate acoustic interference outside the quiet spot area, in order to preserve sound quality in the audience area. Therefore, the near field of smaller acoustic sources will be used: a) only a moderate acoustic power is needed to obtain an effective reduction, b) the interference effects far from the control region are minimized, and c) it will result in spatial stationarity, i.e. a quiet zone that is not moving depending on changing atmospheric conditions.

2.3.4 Sound zone signal processing and optimization (Task 4.1.3)

The ASFC task is an optimization problem. The overall cost function will be combinations of minimizing annoyance at external regions, maximizing sound quality (e.g., perceived directionality and intended loudness) in the audience area, and minimizing loudness in the surrounding areas. Good estimates of the transfer functions between source and receiver positions are essential, as developed in T4.3. The signal processing is a combination of Acoustic Contrast Control combined with Pressure Matching (*PM-ACC*). In order to go from present state technology in rooms to real outdoor event conditions, the underlying optimization problems should be developed to minimize the radiation to the surrounding. To have 'escape' for the produced sound power and to have more degrees of freedom for the optimization, radiation upwards is allowed, but not to the side (3D zones). The concept of robust optimization will be useful, taking into account uncertainties in sensor input and forward modelling.

2.3.5 Microphone Sensor Configuration (Task 4.2)

High quality and accurate microphones, which can withstand the weather, will be developed into IoT enabled microphone devices. To be consistent with the concept of IoT and to be able to exploit this device for other applications, it will be developed as a generic sound level meter connected to the Internet, discoverable online by applications, and being able to provide information on deliverable data and location. It is annotated with semantic information so that applications can select them on or off, both during the programming of applications and during run-time. In this way, administrators can set up different sound meters ad hoc and applications can find the exact location of these devices. As an example, it can be used for identifying and monitoring noise sources in public, as in T4.4. The IoT enabled microphone is mainly described in Deliverable D4.4 and D4.5.

2.3.6 Sound Propagation Model and Parameter Updating (Task 4.3)

The forward model in T4.1.3 needs to be updated to match the actual atmospheric/weather conditions, by use of adaptive filters and Bayesian statistical methods. The forward model is based on Nord2000 (Plovsing, 2006),

[3] With 'ensonified' is meant an area with an enhanced sound field, filled with sound. Compare with illuminated for light.

used for predicting outdoor noise propagation. It includes major mechanisms of attenuation for any terrain shapes including screens. It will be usable as is when estimating the Sound Pressure Level (SPL) at the affected neighbour's position, but for the transfer function estimate in the audience area it has to be adjusted. Information of the current situation will be provided using the inputs from the distributed IoT sensors, including distributed climate sensors (temperature, wind, humidity). For the audience area, the audience absorption and scattering is of high importance. To estimate these data, a mapping of the density of the spectators, and notification of significant changes in this, will be done using video cameras and image processing. A mapping of SPLs obtained from a dense network of high quality calibrated sensors supplemented with additional data from cheaper uncalibrated sensors from available wearables. Subjective feedback information from users regarding sound quality and annoyance, via mobile phone apps, will also be included in the dataset – mainly used for the decision support system, T6.3.

2.3.7 Noise Annoyance Monitoring (Task 4.4)

To monitor the neighbours' annoyance and the audience acoustic comfort, it is necessary to separate the different source contributions in the signal at a given location. Two general strategies will be used, depending on the distances between sensors and information available. For IoT devices correlation-based approaches can be used via access to the source signal, and depending on the location of the sensor combined with the propagation model in T4.3. In other cases, when only isolated microphones are available, machine learning techniques will be used. Noise annoyance metrics will be calculated for each separated source based on the outcome of the source separation techniques. The measured input data comes partly from IoT devices disturbed at the neighbours' position, and partly from the sensors developed in T4.2. Feedback information from users regarding sound quality and annoyance, via mobile phone apps T6.5, will also be included in the dataset. The subjective data obtained by the survey will be analysed by experts in T10.2.

3 Audience area loudspeaker array configuration – far field control (T4.1.1)

3.1 Technology overview

Traditional loudspeaker systems for outdoor sound reinforcement typically consist of two loudspeaker line arrays and a set of subwoofers arranged in a horizontal array or as two left-right clusters. In the ASFC, these systems (*primary sources*) are extended by the use of additional low frequency loudspeakers *(secondary sources*). These are placed behind the audience in between the primary sources and the neighboring region in which the sound from the event should be reduced (*dark zone*). The method of creating spatially separate acoustical zones is often called sound zoning (see e.g. the review by Betlehem et al, 2015).

A sketch of the setup is shown in Figure 1. The basic idea is to optimize the radiation from the secondary sources in such a way, that the sum of sound pressures from the primary and secondary sources effectively reduces the total sound pressure level in the dark zone. Use of additional loudspeakers to control the sound in the dark zone must not negatively impact the sound experience in the audience area, the *bright zone*. This restriction must be included in the loudspeaker configuration design by either using directive loudspeakers facing away from the bright zone for the secondary sources or in the formulation of the loudspeaker signal optimization problem (see Chapter 5, Sound zone signal processing and optimization).

A well performing ASFCS will enable a high sound pressure level in the bright zone relative to the sound pressure level in the dark zone. A performance indicator for this problem is the *acoustic contrast,* which is the difference of the mean SPL in the bright zone to the mean SPL in the dark zone (Choi and Kim, 2002).

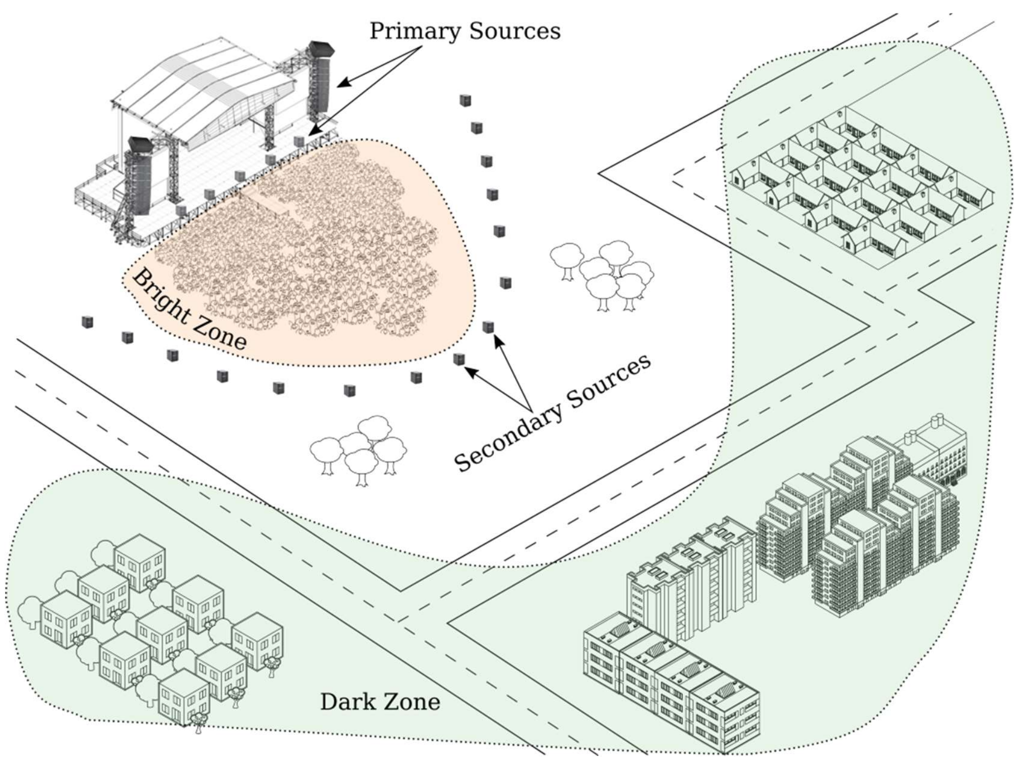


Figure 1: Sketch of the adaptive sound field control system, which is optimized for a good audio experience in the bright zone and low sound pressure level in the dark zone

The ASFCS can be implemented in two modes: in the *extension mode* described above, the ASFCS only controls the secondary sources. This mode of operation is a natural first step in the development of the ASFCS, because 1) less sources are controlled, and 2) sound engineers will be less skeptical because the main system is left untouched. Moreover, 3) in this way there is no need for a definition of a target sound field in the bright zone. Also, 4) in case of failure, the system can be easily shut off without interfering with the main system.

In the second mode of operation, *full mode,* both the primary and secondary sources are optimized by the ASFCS. Using the additional degrees of freedom (more loudspeakers) will enable a larger acoustic contrast and optimization of the sound field within the audience area. However, this poses the non-trivial question of a perceptually ideal sound field in the audience area. In case of failure, the ASFCS can be bypassed and the sound reinforcement system used in its default mode. Table 1 compares the two operation modes.

Table 1: Comparison of operation modes

Extension Mode	Full Mode
Only secondary sources are controlled	Both primary and secondary sources are controlled
SPL in dark zone is minimized	SPL in dark zone is minimized and sound field in audience area optimized
More easily accepted by sound engineers	Acceptation questionable
Shut off in case of failure	Bypassed in case of failure
Information on full system needed	Information on full system needed

The ASFCS must be distinguished from generating *Quite spots* (Task 4.1.2, see the section 4). The former technology controls sound in a large area and is based on sound zoning techniques, while the latter controls the sound in a very confined area and is based on adaptive filtering and active noise control.

3.2 Services enabled

The ASFC solution aims to provide an optimized sound field in the audience area and minimizes the impact on neighboring areas (Use case: Sound level adjustment). The ASFCS consists of specific hardware which need to be added to a venue's sound-system. The software (sound field optimization algorithm and sound propagation model) is developed and tuned specifically for each venue. The service is under development and will be described in deliverable D4.2 and D4.3.

3.3 Infrastructure and integration with the MONICA IoT platform

3.3.1 Information flow between ASFCS and MONICA IoT platform

The ASFCS interacts with the MONICA IoT platform in two ways:

1. The MONICA platform provides various collected sensor data (e.g. weather condition and sound pressure) to the ASFCS upon requests, which is used to update the sound propagation model and estimate the sound propagation in and around the venue.
2. The sound propagation model supplies information on the sound condition in and around the venue to the MONICA IoT Cloud in form of a Sound Heat Map (the Sound Heat Map is handled in Task 4.3, which is not covered in detail in this document, but will be included in the coming deliverables D4.2 and D4.3 – some aspects of the sound propagation model behind the Sound Heat Map is however described in Chapter 5).

Figure 2 shows a schematic of the information flow in the acoustic closed loop system. Compared to a traditional sound reinforcement chain, we insert a *Sound Field Controller,* which is a processing unit in between the mixer and the loudspeaker system, which computes the optimal loudspeaker signals. IoT enabled microphones and weather sensors distributed throughout the venue and the control areas continuously measure the sound pressure field created by the ASFCS and current weather conditions. This data is made available to the *Sound Propagation Module* via the MONICA IoT platform. The Sound Propagation Module uses this data to estimate the transfer-functions between sound sources and the control areas, which are needed by the Sound Field Controller's optimization routine.

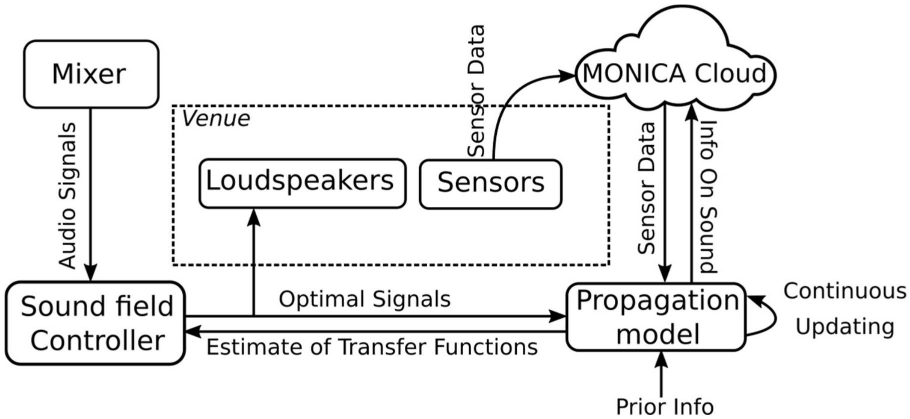


Figure 2: Information flow in the sound field control system

3.3.2 Infrastructure and deployment of the ASFCS

3.3.2.1 Extension mode versus Full Mode

In practice, the ASFCS will be deployed as shown in Figure 3. The Sound Field Controller and Propagation Model modules are running on a local processing unit called *ASFCS core* (high-performance computer with audio processors and interfaces), which also handles the communication to the MONICA cloud either over Ethernet or WIFI connection. Optional wired microphones can also be connected to the processing unit. In Full Mode, the mixing signals are directly fed to the processing unit and from there to the loudspeaker system. In Extension Mode, the signals from the mixing board are fed both the primary sound system and ASFCS processing unit. The ASFCS is set up, monitored and controlled directly through the processing unit.

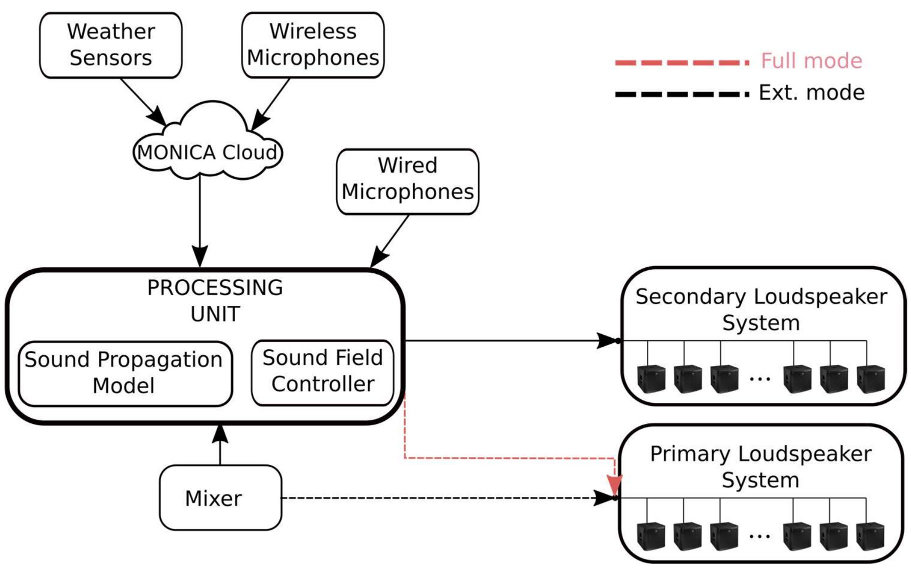


Figure 3: Infrastructure and deployment of the ASFCS. In full mode the primary loudspeaker system is fed from the Processing Unit. In the Extension Mode the primary loudspeaker system is fed directly from the mixer

3.3.2.2 Infrastructure and signal flow of the ASFCS

The infrastructure and signal flow of the ASFCS is described in Figure 4 (with pilot Tivoli as an example). *ASFCS Core* is a processor that generates Sound Heat Map and calculates optimized driving solution from MONICA cloud. When the core also performs as a signal renderer (signal filtering), it should be connected to the signal hub (*Audio Interface*) and receive the input signal. If separate DSP is used, then the Core is only used as a solution generator. The Audio Interface is the hub of the ASFCS, receives monitored signal from router or preamp (for closed-loop control) and finally delivers the signal to Core or DAC. Mic Preamp/ADC or Router are used as the gateway of ASFCS, receives a signal from the PA system or microphones. These gateways also deliver the control signal to the main PA system in the use of Full mode. Note that the *Secondary Loudspeaker System* needs to deliver comparable energy level with the main PA system.

The solid line in Figure 4 represents analog signal flow and the dotted line is for the digital signal flow using AES/EBU or MADI protocols. The triple compound line denotes multiple parallel cables are required to connect between two nodes. On the other hand, the simple line represents a simple delivery of the multichannel signals through a single cable (e.g. MADI).

In Figure 2, the Sound Field Controller requires monitoring the audio signal (Music playing in Figure 4) from the mixer in order to generate the optimal signal. It is important to note that the signal output from the Primary Loudspeaker System can be different from the PA Mixing Console output. Generally, after the music signal passes the mixing console, there are EQ/delay filters applied afterward which are selected to fit the specific venue. For the accurate monitoring, it is recommended to monitor the whole electro-acoustic signal from the last node (Monitoring Path 3 in Figure 4). In this case, all analog channels from the PA amplifier should be fed to ADC in ASFCS. Instead, the monitoring can be done at the previous nodes (Monitoring Path 1 or 2) if the transfer function to the end node is known in advance. In this case, digital monitoring outputs are fed into MADI router in ASFCS. If there is no possibility of change in transfer function during the event, the latter approach can be practical but if not, the final node after the PA amplifier should be monitored. For the Full mode, the control signals to Primary Loudspeaker System is delivered from the Router of ASFCS through the inverse path of monitoring, but if the Monitoring path 3 is used, then Full mode cannot be used.

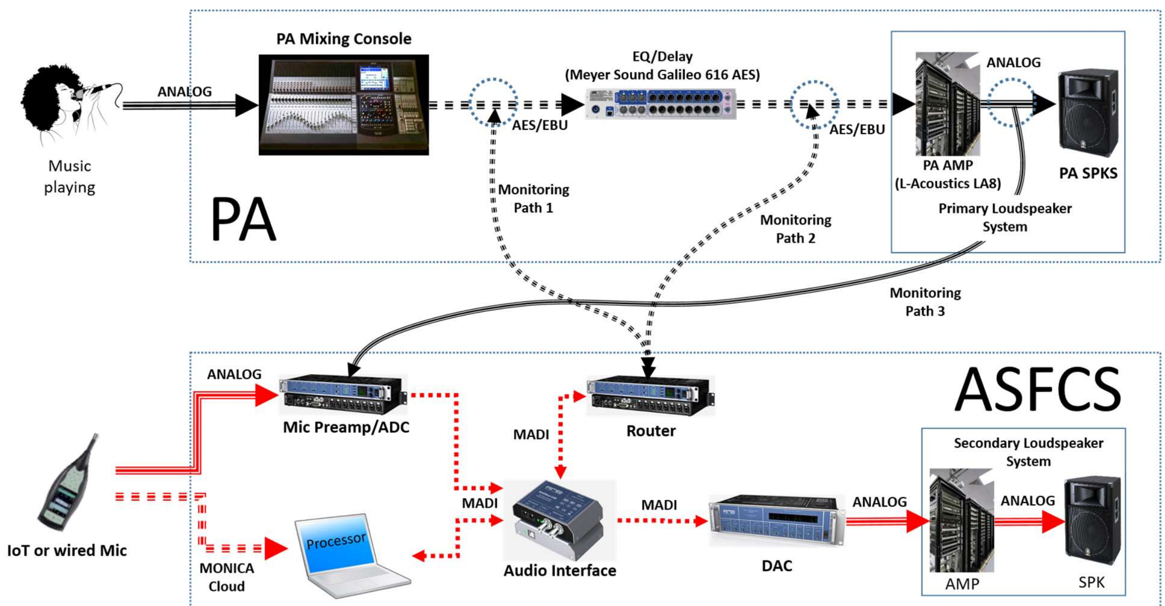


Figure 4: Example of ASFCS connected with PA system of the pilot at Tivoli

3.4 Simulations

The sound field control principle is validated here with two sets of simulations for a simple case resembling a small, open air concert setup in which the radiation of sound to a sensitive area 1) around the concert or 2) behind the concert is to be mitigated. These two cases are useful in illustrating the differences between operation in extension mode and full mode.
Similar results have been already published by the authors in (Heuchel and Caviedes, 2017).

3.4.1 Assumptions

The simulations shown here make the following assumptions:

- The transfer-functions between control sources and control areas are known precisely. However, the sound propagation model will be responsible to compute accurate enough estimates of these transfer-functions in practice.
- The conditions do not change and thus, the transfer-functions are constant over time. In practice, changing conditions like weather or size of audience will change over time and the sound propagation model has to adapt continuously to these changes.
- The loudspeakers can be modelled as omnidirectional monopole sources. Real loudspeakers have complex radiation patterns, which can be modelled in the future by using a complex directivity point source model (Feistel, 2014).
- The geometry of the venue consists only of plane grounds. Real venues have a high geometrical complexity which will be modelled in the future by the sound propagation model.

3.4.2 Setup

The simulation setup is shown in Figure 5. The *surrounding dark zone* resembles the problem of Figure 1 and the *small dark zone* illustrates a case where there is only a specific, local area with a noise issue. The primary sources resembling the conventional sound reinforcement system are modelled by a horizontal array with 6 loudspeakers. The 12 secondary sources are placed below the bright zone in form of two lines with each 6 loudspeakers. The double layer array can act as a combination of monopole and dipole sources that mainly direct their sound energy away from the bright zone.

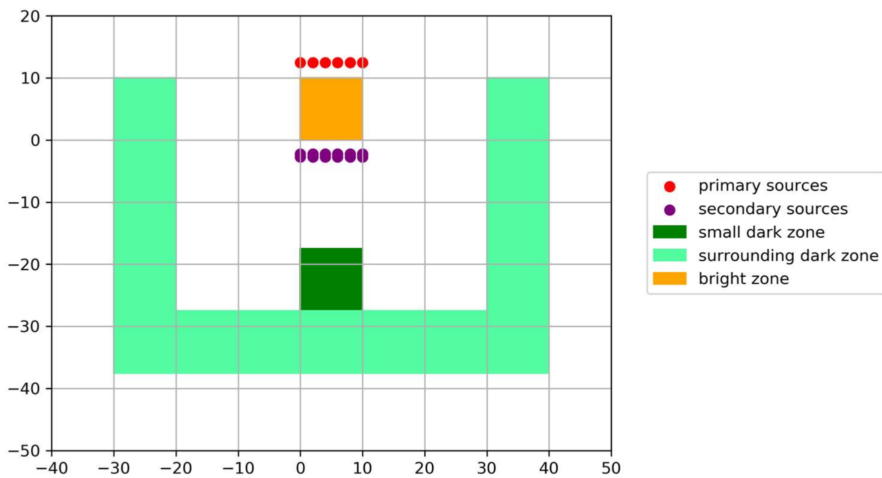


Figure 5: Sketch of the adaptive sound field control system, which is optimized for a good audio experience in the bright zone and low sound pressure level in the dark zone

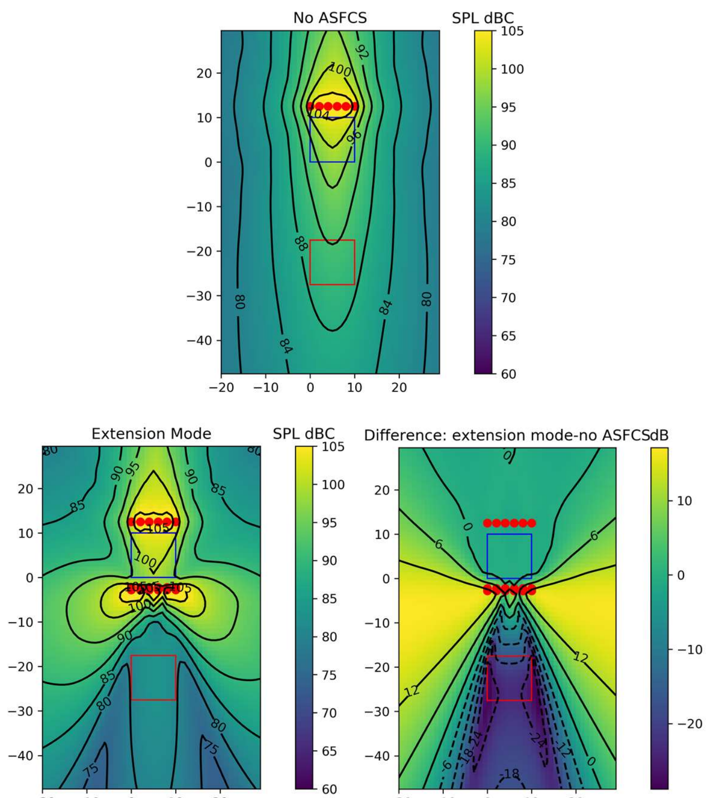
The frequencies of interest are the bass frequencies ranging from 20 to 250 Hz – in the simulations below the frequency range is however limited to 20 to 150 Hz. The sources are placed at a height of 0.4m while the bright and dark zones are sampled at a height of 1.6m. The ground is considered compacted park area (Impedance Class E according to Nordtest) with flow resistance σ = 700 kPa · m/s. The speed of sound is c = 343 m/s and air density ρ = 1.2 kg/m³. The loudspeakers are modelled as monopoles with constant magnitude response.
To optimize the loudspeaker driving signals, one needs to know how the driving signals are related to the sound pressure in the control zones. This relationship is described by the *transfer-function*. In this simulation this transfer-function is obtained by sampling the control zones at 2.5 points per maximum wavelength in each direction and computation of the transfer-function between sources and the sample positions using the Nord2000 propagation model (Plovsing, 2006), see further in section 5.1.2.

3.4.2.1 Small dark zone

Figure 6 shows plots of the total SPL around the simulated venue for the three cases

1. Without ASFCS: the primary sources are playing with constant gain. This is the equivalent to a conventional sound reinforcement system
2. Extension Mode: only the secondary sources are controlled by the ASFCS, while the primary sources are driven as in case 1
3. Full Mode: all sources are controlled by the ASFCS. The target field in the audience area is the sound field of case 1

Together with SPL comparison between case 1 and cases 2 and 3. Both full and extension mode create a similar SPL map with an area of destructive interference close to the dark zone, while at the same time increasing the SPL in other areas, which are not controlled. This is a classic result of sound power interaction of coherent sources. In free-field, a control source has to be closer than half a wave-length to a noise source to be able to reduce the total emitted sound power effectively, see e.g. (Jacobsen and Juhl, 2013). The secondary source array will thus not work as an active absorber, but rather create destructive interference in some area at the expense of higher sound pressure levels at other positions. Care must be taken in designing the loudspeaker arrays and optimization problem, such that the reduction of noise levels in the dark zone does not lead to new noise problems in other areas.



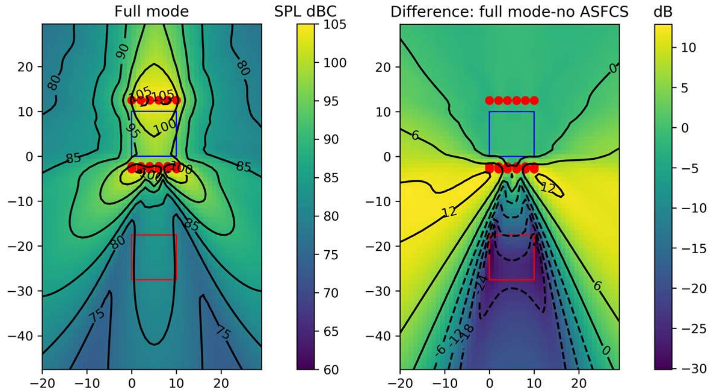


Figure 6: Comparison of total sound pressure level in the vicinity of the simulated venue, small dark zone.[4]

Figure 7 shows a comparison of the achieved acoustic contrast between bright and dark zones over frequency. The ASFC increases the acoustic contrast considerably over the whole frequency range. Extension and full modes give very similar results.

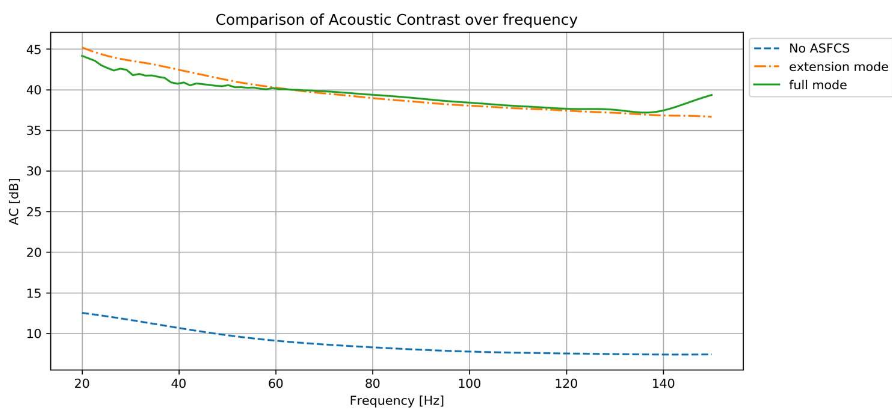


Figure 7: Comparison of total sound pressure level in the vicinity of the simulated venue for the small dark zone case

3.4.2.2 Surrounding dark zone

Figure 8 shows plots of the total SPL around the simulated venue for the three cases "No ASFCS", "Extension Mode", "Full Mode", analogous to figure 5. In extension mode, the reduction of SPL is focused on the area below the secondary sources, while a full control enables the reduction in most of the vicinity due to a stronger directivity of the primary array on to the bright zone.

[4] Note that dBC is used in Figure 6 and 8 as it better corresponds to the perception of loudness for loud sound events. The dB difference figures are however independent of the frequency weighing.

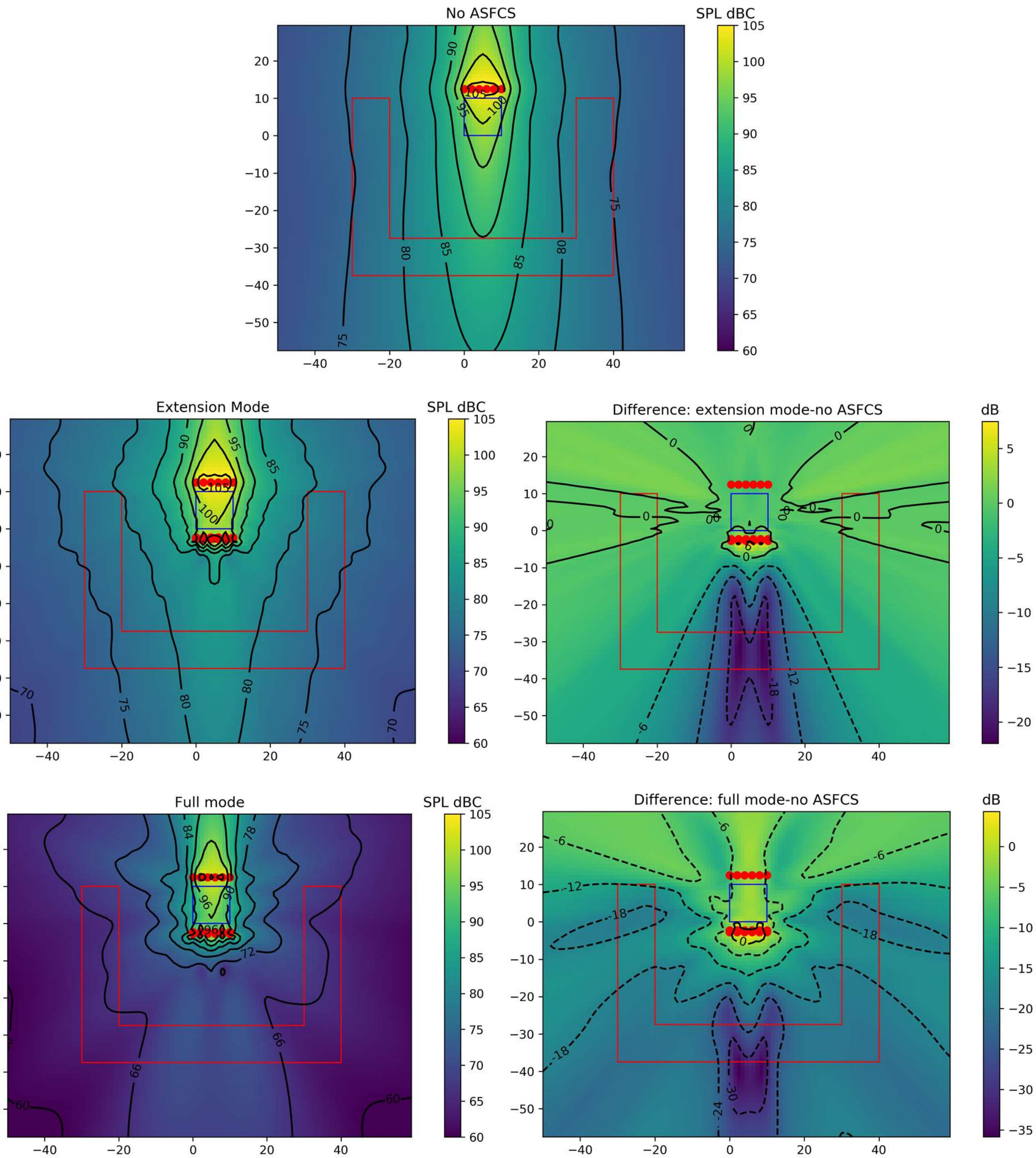


Figure 8: Comparison of total sound pressure level in the vicinity of the simulated venue, surrounding dark zone

Figure 9 shows a comparison of the achieved acoustic contrast between bright and dark zones over frequency. The ASFC in both full and extension mode increases the acoustic contrast over the whole frequency range, with the full mode leading to a larger mean contrast, because the SPL is reduced in a much larger area.

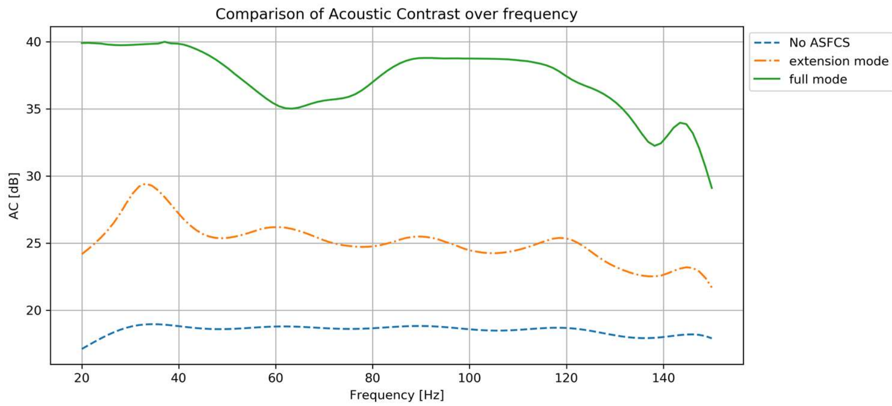


Figure 9: Comparison of total sound pressure level in the vicinity of the simulated venue for the surrounding dark zone case

3.4.3 Summary

The above simulations show that the ASFCS can considerably increase the acoustic contrast between audience area and venue's surroundings. Larger dark zones and control over more sources are helpful in reducing the SPL spill into neighbouring regions. However, this comes at the cost of increased complexity. A larger dark zone needs more microphones to observe the sound field and more sources lead to a higher computational effort in the optimization problem. While a larger (e.g., surrounding) dark zone will always lead to a larger area of reduced SPL, control over more sources does not always lead to an increased contrast, as was seen in Figure 7. Moreover, Comparing Figure 7 and 9, we can conclude that for a small dark zone, the benefit of using the full mode is small, but for the surrounding dark zone, this benefit is essential, with an increase of the contrast of 10-15 dB.

3.5 Measurements

An experimental validation of the ASFCS accompanying the simulations above is currently under way, and the main results of these will be presented in deliverable D4.2. The main goal is the test of the hypothesis that sound field control and sound zoning can be based on transfer-function estimates from sound propagation models instead of their direct measurement. The tests will be realized as a scaled down variant of the *small dark zone* simulation example. Until now, only preparation measurements for the characterization of loudspeakers have been completed.

The sound propagation model uses a complex directivity point sources (CDPS, (Feistel, 2014)) as simplified models for the loudspeakers. A measurement procedure has been developed, that finds the best fit of a loudspeaker to the CDPS model, such that it can be used in the sound propagation model.
A setup of the procedure, deployed in the large anechoic chamber at DTU, is shown in Figure 10. The loudspeaker under test is a small test loudspeaker with 4-inch driver. It is mounted on a turntable, such that its frequency-response can be simultaneously measured over different radiation angles at three microphone positions using the swept-sine technique (Farina, 2007).

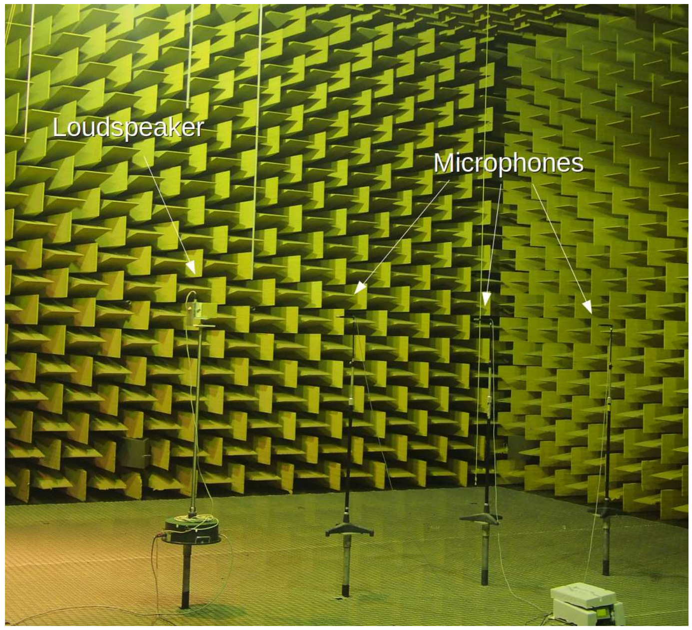


Figure 10: Measurement setup for loudspeaker characterization

The loudspeaker's directivity dependent frequency-response is measured and fit to a CDPS model. A comparison between measured and modelled response is shown in Figure 11. One can see that the absolute pressure and phase of the loudspeaker can be reproduced by the fitted CDPS model with a relative error of up to 25%, equivalent to 2 dB. While this shows that the source can be modelled as a CDPS at all frequencies and directions, there is still a considerable error considering that the test is done under ideal conditions.

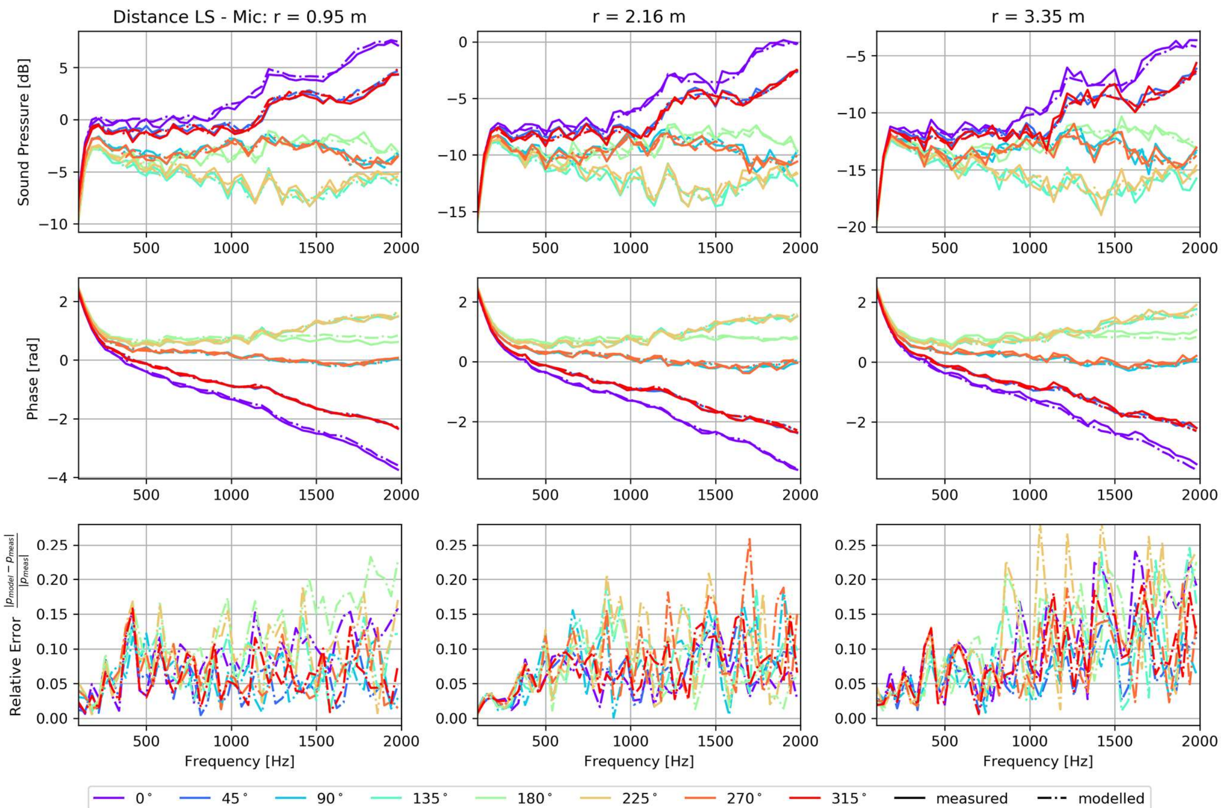


Figure 11: Modelled and measured response of the loudspeaker under test at the microphone positions together with the relative error. Note that the phase is back-propagated in time and unwrapped for a clearer presentation and readability.

3.6 Future work

The measurement series described above will soon be completed. The scaled sound zoning system is going to be tested in anechoic laboratory conditions with a single ground reflection as well as in an outdoor environment. The gained insights from these small-scale tests are planned to be transferred to a 1:1-scale test in spring 2018.

For the deployment of the ASFCS in real, changing environments, we will investigate the use of robust optimization and regularization techniques.

4 Quiet spots – near field control (T4.1.2)

4.1 Technology overview

The quiet zone system is a noise barrier which makes use of active elements to cancel out low frequencies and passive elements to block high frequency noise. The goal is here to obtain the highest possible attenuation of noise across the whole listening spectrum. The system is integrated into the ASFC environment and provides local quiet zones within or close to the optimized sound-field of the ASFC system without interference.

The active part of the QZ system consists of arrays of loudspeakers which synthesize the cancelling sound-field to cancel out the noise-field in the desired zone. Such multichannel active noise cancelling methods in freefields are described in (Kuo, 1996) and (Hansen and Snyder, 2002). The active noise cancelling principle is nowadays well known from appropriate headphones which have entered the consumer markets. Of special challenge in the MONICA project is that we have to cope with distributed noise sources which reach the quiet zone areas from different angles and are transmitted through varying paths due to changing conditions (wind, temperature, moving people). The task is to cancel out sound under free-field conditions, with sparse, random and broadband noise sources (the music signal coming from the many distributed loudspeakers of the venue. Figure 12 shows an open air concert area with ASFC and quiet zones.

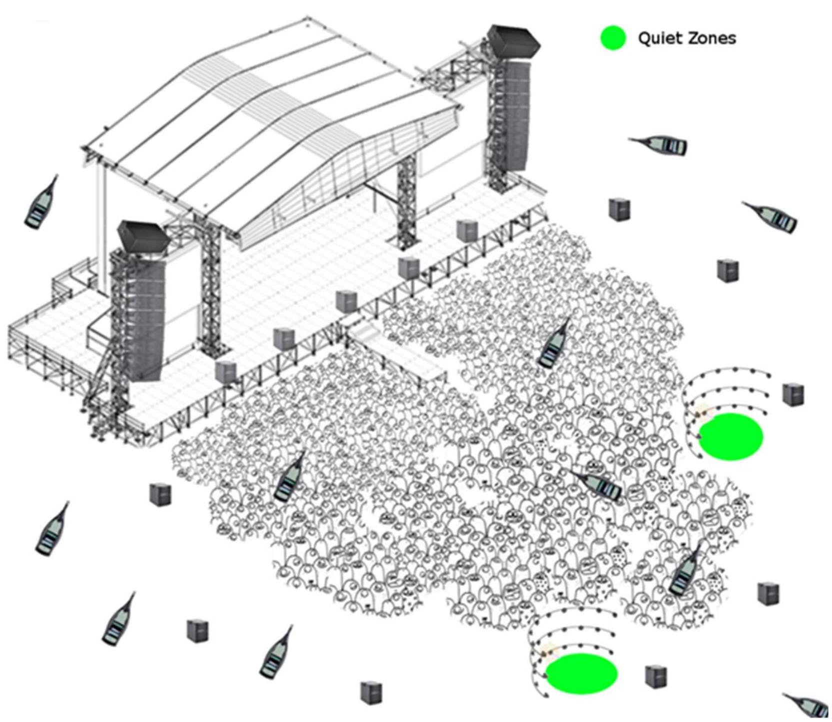


Figure 12: A possible use case of quiet zone systems in a loud outdoor venue

Figure 13 shows a prototype sketch of a quiet zone system made of a semi-transparent wall as passive element and arrays of loudspeakers to actively produce a three-dimensional quiet zone. The system here has a diameter of 2 m and a height of 2.5 m and should be able to accommodate at least two persons trying to communicate.

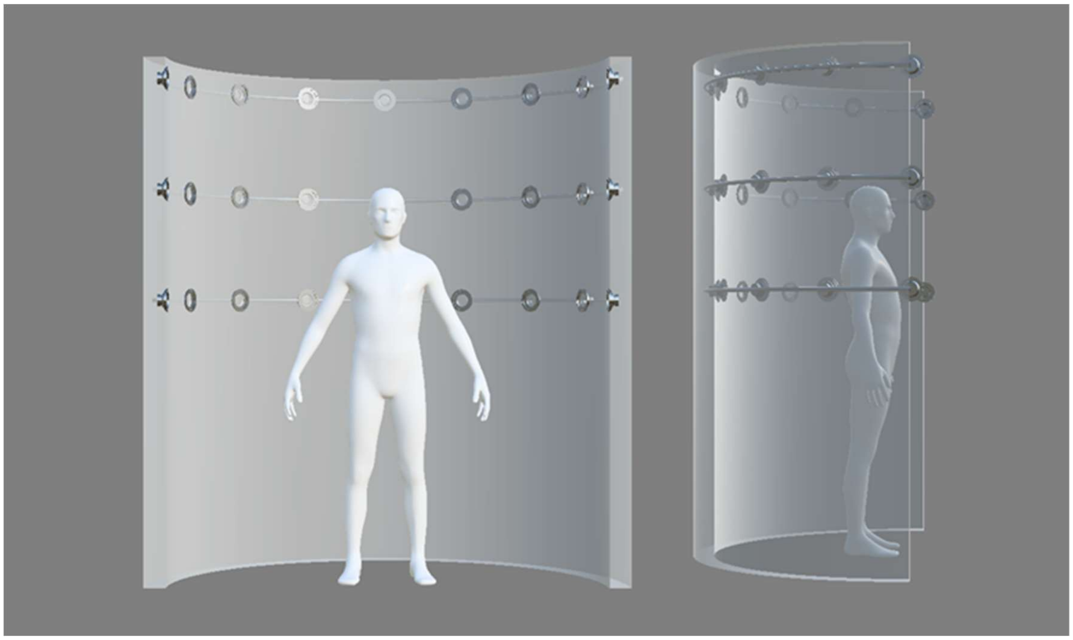


Figure 13: A possible prototype of a local quiet zone system. Loudspeakers are used to actively attenuate low frequencies, a wall is used to passively reduce high frequencies.

Not shown here is the technical periphery, like the quiet zone controller, an array of microphones and the power amplifiers for the loudspeakers, see Figure 14. The microphones will be used as error sensors by the adaptive quiet zone controller. The error is any deviation from a possible minimum of sound pressure at a specific location, within or close to the quiet zone. This measured error will be fed into the pressure minimization algorithm, which will adapt the minimization processes. The music signal from the PA speakers is still needed because it is a reference input which can increase the stability of such a system and support methods for disturbance rejection (Ophem and Berkhoff, 2013) The microphones will surround the quiet zone area because they can't be placed within, due to the existence of people here. Virtual sensing methods need to be applied in order to move the actual error locations into the quiet zone (Moreau et al, 2008). Additionally, it could be helpful to make use of a second microphone array in order to separate the noise of interest from disturbances – like chatting and screaming people. The processing unit can be close to the quiet zone or remote but in any case, multichannel audio connections need to be established in between the controller, the quiet zone system and the venue's PA. The processing unit can be close to the quiet zone or remote but in any case, multichannel audio connections need to be established in between the controller, the quiet zone system and the venue's PA.

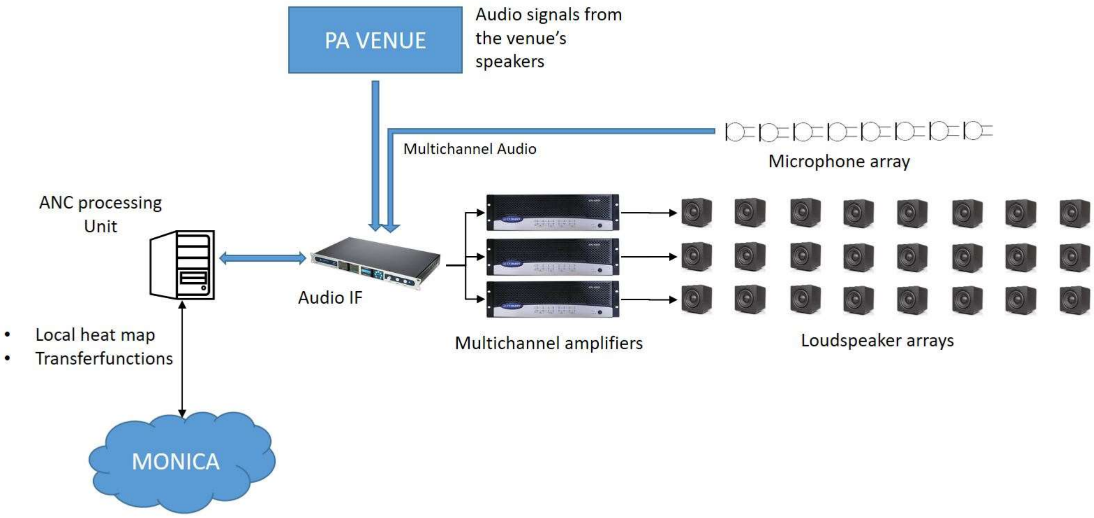


Figure 14: The components of the active part of the quiet zone system

The technical system consists of standard professional audio equipment:

- 1x per Array: Multichannel amplifiers: 1x QSC CX108V
- 24x loudspeakers: the type and the final amount needs to be verified in measurements.
- 1x Multichannel Audio Interface: Ferrofish A32 Dante with 32 channel I/O
- 8x Microphones: 8x Behringer C2
- 1x Computer running the ANC software

4.2 Services enabled

The quiet zones are related to the use case "Sound Level Adjustment" and the solution "Sound Control".

4.3 Infrastructure and integration with the MONICA IoT platform

4.3.1 Audio Signals from the Venue

The quiet zone systems receive the audio signals from the venue's console or even from behind the PA's power amplifiers. The latter case is more complicated to realize but provides better reference signals for the active noise cancelling algorithms. The same infrastructure could be used here like shown in Figure 4. So the quiet zone systems use the same audio signals as the Sound Field Controller.

4.3.2 Operational Status

The operational status is communicated to the Current Operational Picture (COP). The status comprises:

1. Status: On/Off
2. The maximum sound pressure level in the quiet zone

Figure 15 shows a link to the propagation model unit. This link is used as a "through" port to the Monica cloud only but provides the possibility for further optimizations of the ASFC and QZ system, such that the local and far-field transfer-functions could be exchanged for example.

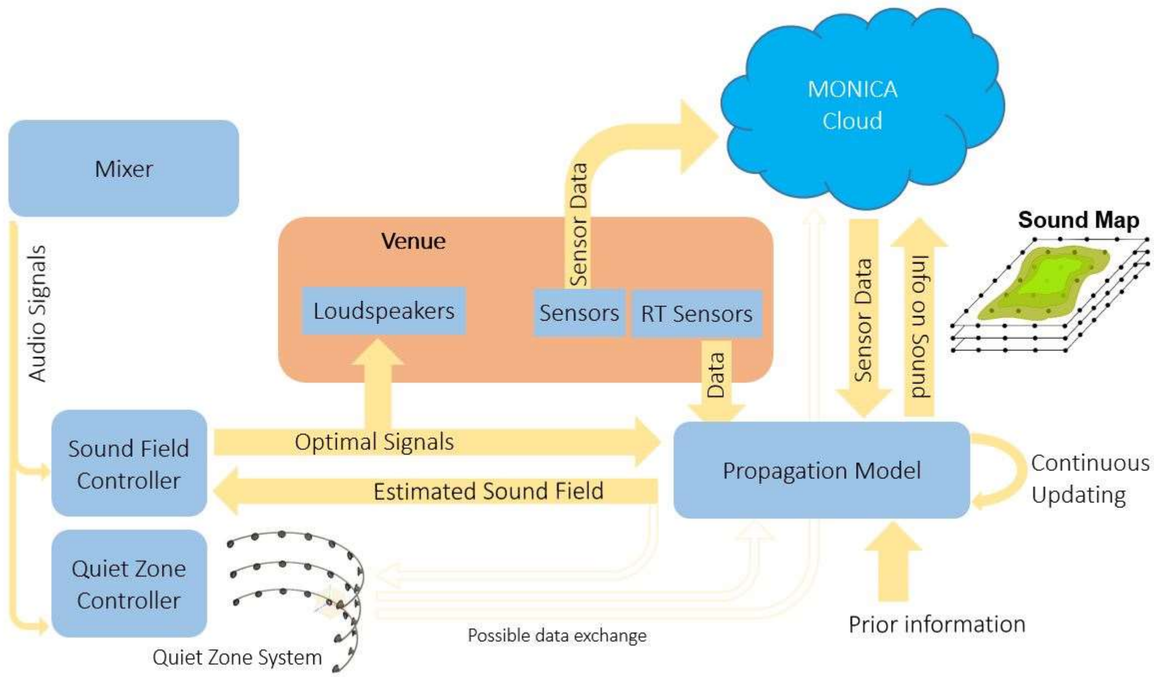


Figure 15: Information flow chart, ASFC including the Quiet Zone.

4.4 Simulation

Two sets of simulations have been executed so far in MATLAB (The MathWorks, Inc., Natick, Massachusetts, United States).

4.4.1 Simulation under free-field conditions

The first set of simulations have been executed in order to investigate if a limited set of reference points in the quiet zone is sufficient to generate a quiet zone of extended size and what is the effect of changing the distance in between the control sources and the reference points. In this section all the sources are implemented as monopoles and a multichannel minimization approach based on (Hansen and Snyder, 2012) was used. The momentarily used algorithm minimizes the pressures in these reference points but does not take the space in between into account. In the real-world implementation of the quiet zone controller there will be microphones to measure the minimized pressure in the reference spots. Because any deviation from the minimum is assumed to be an error, these microphones are called error microphones. Note that the simulated sound field is simplified and do not correspond to the prototype in Figure 13. The following preliminary results have been obtained:

1. In Figure 16 it can be seen that in the space in between the reference points the SPL is actually attenuated; this depends on the frequency of the noise source and on the distance between the reference points. In Figure 16, microphone distance is smaller than a quarter of wavelength.
2. In Figure 17 it can be see that with increasing distance between the control source array and the reference point array the minimal amount of attenuation in the line between the two outer reference points gets worse and that the sound energy of the control sources disturbs the surrounding area much more – note the 'beam' in the y-direction out from the control array. Further investigations have to be done here in order find the optimal distance, etc.

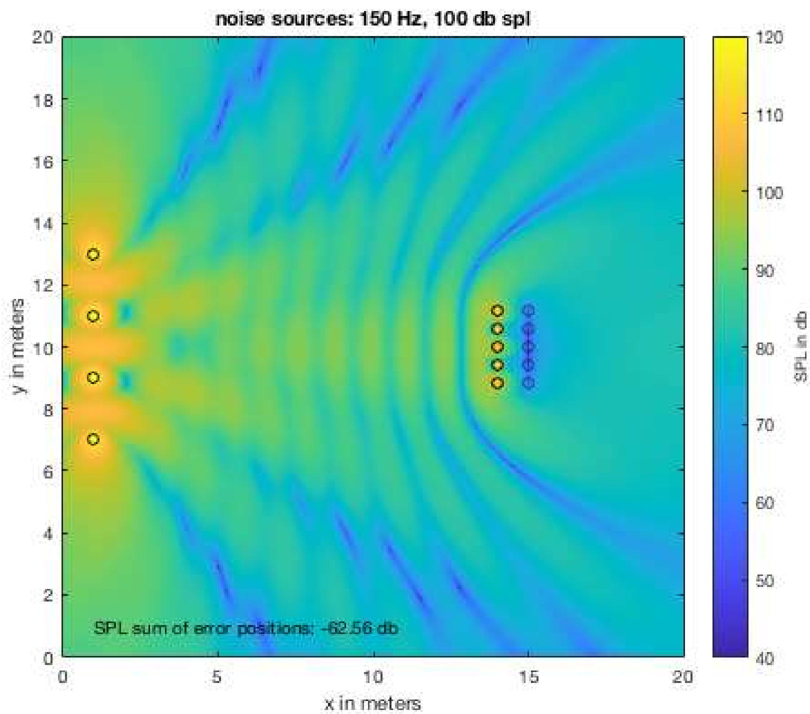


Figure 16: Error Mic positions close to the control sources with a distance of 0.5m in between. Note that at 150 Hz this distance is smaller than a quarter of a wavelength.

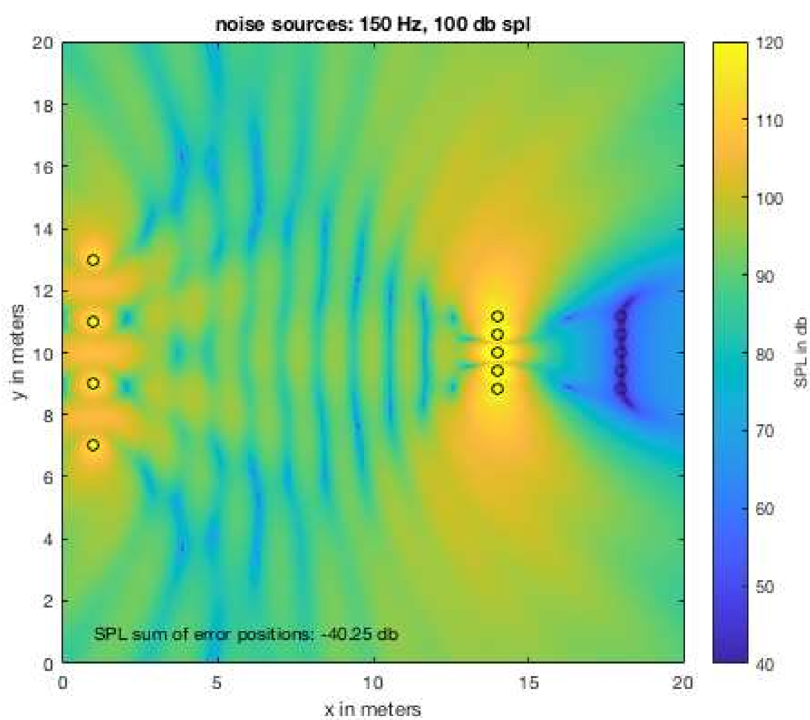


Figure 17: Error Mic positions further away from the control sources. It is obvious how the control sources contaminate the surrounding.

4.4.2 Simulations of real-time adaptive filtering algorithms

Because of changing acoustical conditions in between the venue's PA and the quiet zone an adaptive filtering approach has been investigated in a real-time scenario. The FxLMS is a widely used algorithm for active noise canceling purposes and was chosen for this set of simulations. (Tabatabaei and Ardekani, 2011) It is supposed to be stable and robust against disturbances. The simulations were executed in Matlab with the following routine:

1. Definition of an arbitrary true primary path impulse response
2. Definition of an arbitrary true secondary path impulse response
3. Producing an estimate of the secondary path IR (Sec Path + Error)
4. Convolving the reference signal with the primary path IR and adding some noise
5. Providing the secondary path estimate to the FxLMS algorithm and application

The simulation runs in real time and the attenuation is audible and visible in its spectrum. Figure 18 shows the steady state of the filter after convergence.

Good results have been achieved so far for the attenuation of low frequencies of a music signal by utilizing one noise source and one control source speaker. In Figure 18 we see such a simulation where low frequency noise is strongly attenuated (10-50 dB up to one kHz). To achieve up to 50db attenuation for low frequencies would be very sufficient but we have to keep in mind that we get this result for a single measurement point only. Multichannel active noise cancelling adaptive filters will be investigated next as well as the power of state space models which can be specifically designed for multichannel purposes and optimized for the sake of stability and disturbance and measurement noise rejection. As seen in simulation set 1: multichannel applications can extend the dimensions of the quiet zones.

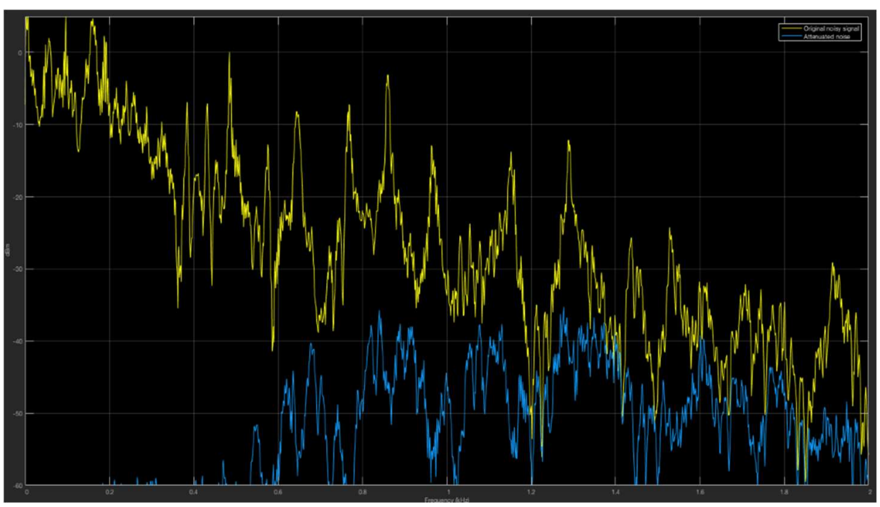


Figure 17: Noise attenuation at one Error Mic position. Yellow: the music signal before active control, blue: the music signal with active control.

4.5 Future work

During the first month of 2018 the adaptive algorithms will be optimized and measurements will be executed under laboratory and outdoor conditions. In May 2018 a 1:1 scaled test series is planned where the quiet zone system is tested under real conditions, together with ASFCS – results will be presented in deliverable D4.2.

This will take place at one of MONICA's pilots, the amusement park Tivoli in Copenhagen, and at an external location.

5 Sound zone signal processing and optimization (T4.1.3 and T4.3)

5.1 Technology Overview

5.1.1 Sound Zone Signal Processing

Sound zone signal processing and optimization scheme in ASFCS is based on the combined solution of Pressure Matching and Acoustic Contrast Control (PM-ACC) proposed in (Chang and Jacobsen, 2012). PM-ACC is a technique that can select a hybrid solution between the pure contrast solution (ACC) (Choi and Kim, 2002) and the pure pressure matching solution (PM). The solution aims to minimize the pressure field error in the bright zone while keeping the acoustical potential energy low in the dark zone (the neighboring region). PM-ACC solution fits well for the objective of the project because the sound field control in the dark zone must not negatively affect the sound quality and music experience in the audience area.

Figure 19 shows the process of obtaining the solution. The zones (bright and dark) are selected considering the condition of venue and the neighboring region. Transfer-function data and desired target field[5] in the bright zone is needed for calculating the optimal solution. The optimal solution applies to FIR filter coefficients for the input signals to secondary sources. The input signal to each secondary source is coherent with the monitored input signal that fed into the primary sources.

The optimal solution of PM-ACC is explicitly given in (Chang and Jacobsen, 2013). Since the solution is given explicitly, no additional optimization process is needed for the calculation. However, the solution needs matrix inversion process which requires regularization methods for obtaining a robust solution. The matrix involved in the inversion process is a positive semidefinite matrix, where the inversion of the matrix can make the system sensitive to the transfer-function error estimated from the Sound Propagation Model. To avoid this situation, various regularization techniques or additional constraints can be applied. In this case, the solution may require a different optimization process.

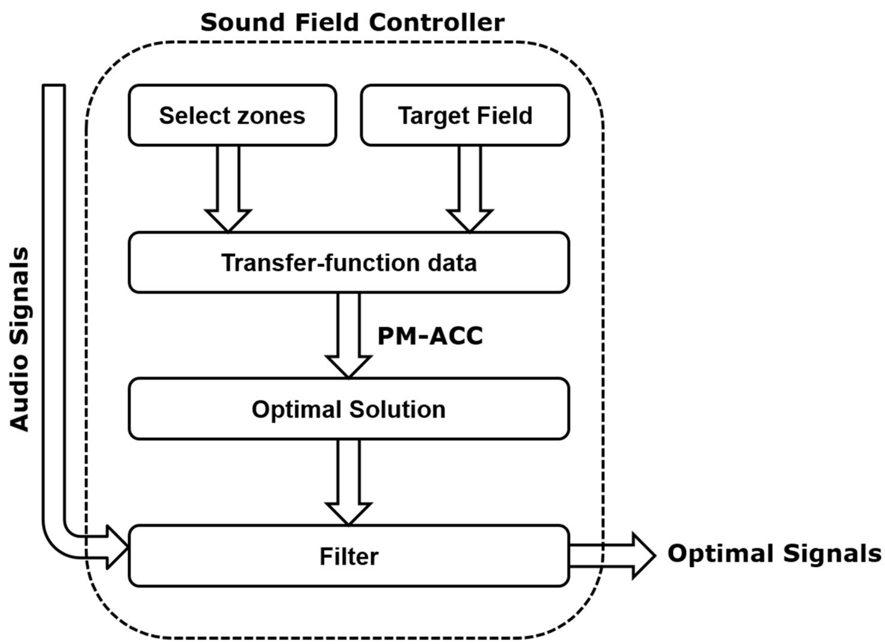


Figure 18: Signal processing flow of the Sound Field Controller, which is a module shown in Figure 2.

[5] The target field is the desired sound filed in the audience area. This can for example be the propagating wave field from sources in the position of the PA loudspeakers, without disturbing reflections and artefacts.

5.1.2 Sound Propagation Model

Sound field control in outdoor concerts requires accurate estimates of the transfer functions between sources and receivers. Feed forward approaches are based on direct measurements of the transfer functions in a dense grid of points. This makes them intractable for large scale situations like the ones present in this project, showing the need of propagation models in order to characterize the sound field in such large areas and provide the ASFCS the proper transfer functions. Nord2000 specifies a calculation method for the prediction of the attenuation of sound during propagation outdoors. It can be used as a method for the calculation of the EU noise indicators Lden and Lnight, as well as for the production of noise maps. The method has been successfully validated with several test cases (Plovsing, 2006). This model is taken as our baseline.

There are some drawbacks in order to use this model as a valid tool for the estimation of the transfer functions that ASFCS needs. Uncertainty in the parameters introduced in the propagation model, such as meteorological, acoustical and geometrical ones, lead to inaccurate estimates of the transfer functions and therefore to a poor performance of the sound field control strategy. Moving from simulated scenarios to real measurements implies that most of the parameters involved in a sound propagation model such as Nord2000 are only known with some uncertainty, causing wrong propagation predictions.

Machine learning techniques are methods that find better estimates of these parameters on the basis of data, improving the accuracy. In our case it implies that the approach could work thanks to the IoT network deployment present in MONICA. Data from weather and acoustic sensors can be continuously stored and used, and the propagation model can be continuously updated accounting for weather conditions, change in the crowd distribution and other causes that may affect how the sound propagates during a concert.

The underlying idea presented in (Heuchel and Caviedes, 2017) states that a forward prediction of the sound pressure generated by a source on a venue can be recurrently enhanced thanks to measuring continuously the sound pressure at a few positions and correcting the input values of the model so the measurements and the predictions get closer.

5.2 Services enabled

Signal processing and optimization algorithm is embedded in ASFCS computational core. See further in section 3.2

5.3 Infrastructure and integration with the MONICA IoT platform

The sound zone signal processing and sound propagation model are running on the processing unit as described in the section 3.3.

5.4 Simulations

In this section simulations are present on how the sound propagation model is updated with machine learning and used together with the ASFCS. (See the section 3.4 for ASFCS detailed simulations.) A simplified version of the simulations in the section 4.4 was specified in (Heuchel and Caviedes, 2017) and can be seen in Figure 20. The bright zone is bounded by two loudspeaker arrays with 5 and 10 loudspeakers, respectively. The target field in the bright zone is a 100dB SPL plane wave travelling in negative y-direction. The dark zone is placed 17.5m away from the bright zone. Both zones are sampled at 2.5 evaluation points per maximum wavelength in each direction. The frequency range of interest is from 20 to 250Hz with a frequency resolution of 15.33Hz. Both evaluation points and loudspeakers are positioned 1.6m above the ground. The ground is again considered compacted park area (Impedance Class E according to Nordtest) with σ = 700 kPa · m/s. The speed of sound is c = 343 m/s and air density $\rho = 1.2\ kg/m^3$. The loudspeakers are modelled as monopoles with constant magnitude response. The dark zone mean square pressure is bounded to be less than 60 dB SPL. The regularization parameter was chosen $E0 = 50m^6s^{-3}$, as it leads reasonable solutions for this problem.

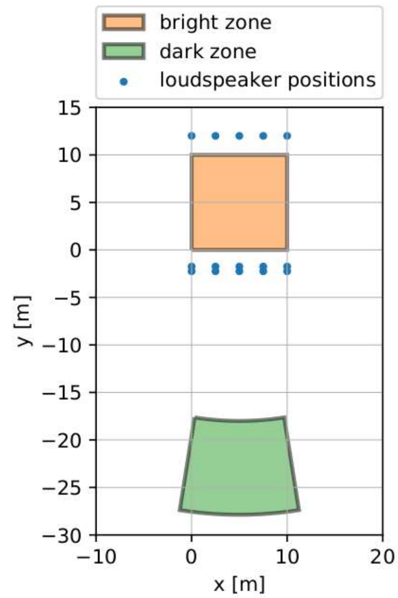


Figure 19: Setup of bright zone, dark zone and fixed loudspeaker positions in the simulation.

In order to resemble real conditions in the simulations, measurement noise and uncertainties need to be generated for the desired parameters. Measured coordinates of both sources and receivers are affected by 15 cm of random deviation. Measured pressure is distorted with 30dB SNR for all the frequencies. The ground is considered park area, so the flow resistivity is set as the representative value of the Impedance Class E in Nordtest (500 kPa · m/s). A summary of the parameters is presented in the following table.

Table 2: Nord2000 model parameters

Parameter		Value
D_0	Max. Mean squared pressure in DZ	20 μPa x $10^{60/10}$
E_0	Regularization parameter	50 m^6s^{-3}
SNR	Signal to Noise Ratio	30 dB
-	Coordinates Error	15 cm
σ	True Flow Resistivity	700 kPa · m/s
σ'	Forward Flow Resistivity	500 kPa · m/s

Figure 21 shows the Acoustic Contrast (AC) with and without ASFCS in the cases of true parameters, forward parameters and optimized parameters. No sound field control describes the case where only the upper loudspeaker array is active with constant source strengths. That contrast is only due to the distance of the zones to the sources (what cases like *Kappa Futur Festival* and *TIVOLI FREDAGSROCK* are experiencing). Applying the sound field control system improves the acoustic contrast by 17 dB-35 dB in the ideal case of true parameters. If the forward parameters are used without model updating/machine learning techniques, the acoustic contrast drops down to 10 dB, even though the uncertainty in the positions is small compared to the wavelength. This is the case of using directly an engineering sound propagation model such as Nord2000. Using model updating/machine learning to optimize, the contrast enhances the improvement again, especially at low frequencies; a benefit of about 10 dB is seen in Figure 21 for the low frequencies.

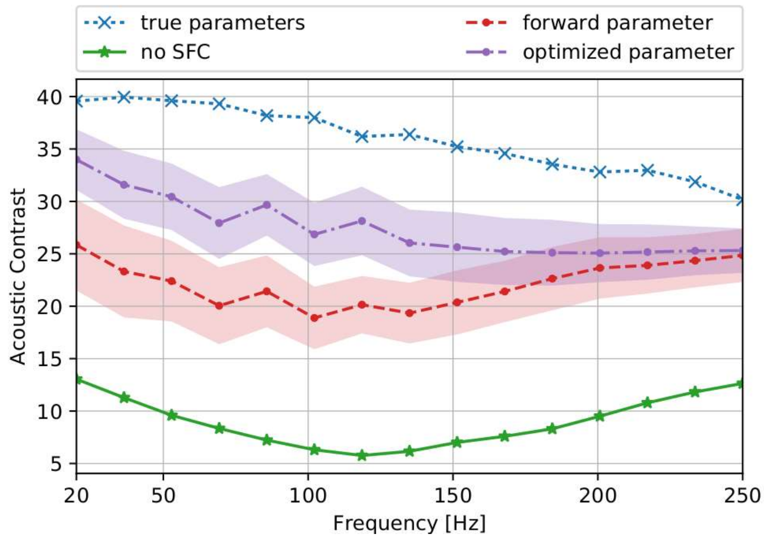


Figure 20: Comparison of acoustic contrast when using true parameters (blue), forward parameters (red), optimized parameter with Bayesian inference (purple) and no SFC (green).

5.5 Future work

In regard to the sound zone signal processing, robust optimization techniques based on regularization techniques will be investigated to make the synthesized sound field as robust as possible to changes in the environment and errors in the sound propagation module.

The solutions to the sound zoning problems have so far been computed with frequency domain methods (Chang and Jacobsen, 2012). Solving of these problems in time domain has been shown to be an interesting alternative (Simon Galvez et al., 2015; Møller and Olsen, 2016) and will be investigated for the application in the ASFCS.

6 Noise Monitoring System Configuration (T4.2 and T4.4)

6.1 Technology overview

The Noise Monitoring System consists of the Source Separation/Contribution techniques (T4.4), the Annoyance measures (T4.4), the Noise Heat Map (T4.3&4) and the Sound Level Meter (T4.2); the latter, including the first version prototype of the Sound Level Meter as well as its interface with the MONICA Cloud through the Sound Level Meter Gateway, is in detail described in deliverable D4.4, and is thus not further covered here.
Regarding the Source Separation/Contribution techniques, the aim is to estimate the amount of noise contribution (in sound pressure level in dB over time, or similar) that originates from the actual concert in the presence of background noise originating from other noise sources (such as traffic noise, people talking, etc.). Thus, it should answer the question: Is the noise coming from the concert or form other sources?
Two basic approaches are under investigation: the Coherence method and the Pattern recognition approach. In the Coherence method synchronized signal data is obtained from sensors close to the source ('reference' signal), and at locations where people are potentially annoyed, see Figure 22 for an early setup used in the B&K measurement campaign at Tivoli during the summer 2017. The goal is to find the coherent part of the reference signal in the other recordings, and from that to estimate the contribution of the concert. In the Pattern recognition approach the goal is to recognize music in the recordings, using machine learning approach, and in this way to estimate the noise contribution. Here no 'reference' signal is needed. On the other hand, the machine learning algorithms will need to categorize sound samples to learn from. This technique will also be used for event detection (fight, gun shot, etc.) in WP6.

Figure 21: Setup using front-ends for synchronous data, including a GPS antenna for synchronization.

The purpose of the Annoyance index is to give a more accurate estimate of noise annoyance, based on subjective perception data. This will require an app that can provide subjective feedback, which is still under development.
The Noise Heat Map will give an estimate of the SPL at other positions than the one being measured by the Sound Level Meter. This will be done using the forward sound propagation model developed in T4.3, based on the existing sound propagation model Nord2000, see section 5.1.2.

6.2 Services enabled

The Noise Monitoring system is related to the use case Monitor Sound Level.

6.3 Infrastructure and integration with the MONICA IoT platform

We here refer to delivery D4.4, which includes a description of the interface of the Sound Level Meter with the MONICA Cloud through the Sound Level Meter Gateway.

6.4 Measurements

A measurement campaign was performed by B&K and Acoucite during the summer 2017, at and around TIVOLI FREDAGSROCK in Copenhagen (see Figure 23) as well as in Lyon during Nuits Sonores. Two kinds of measurements were performed:

- Synchronized measurements (only at Tivoli): This is done using front-ends (PC & sound card) with GPS synchronization, see Figure 22. One of the devices is the reference (located at the top of the Tivoli Concert Hall). This measurement is done to give input for the Coherence method algorithm development.
- Non-synchronized measurements (Tivoli and Lyon): This is done using Sound Level Meters (SLM) at different places around Tivoli and Lyon to perform audio recordings (wav-files). These are used in the pattern recognition approach, using machine learning algorithms trained by the labeled recorded signals.

Figure 22: Measurements outside Tivoli using a front-end (left) and a SLM (right)

6.5 Future work

Measurements at Pilot sites will continue in the coming years. The IoT enabled SLM, described in D4.4, will then be used. Moreover, the developed algorithms will be implemented in the Sound Level Meter Gateway; this will be presented in D4.5.

7 Conclusion

We have shown that the ASFCS can considerably increase the acoustic contrast between audience area and the dark zones in the neighbourhood under ideal conditions. Experimental validations and the inclusion of robust and adaptive optimization techniques will be investigated in the future to make this technology ready for real word deployment.

Comparing an extension mode of the ASFCS (only secondary sources being controlled, leaving the main PA uncontrolled) with a full mode (controlling both secondary and main PA sources) we can conclude that for a small dark zone, the benefit of using the full mode is small, but for the large dark zone, this benefit is essential, with an increase of the contrast of 10-15 dB.

Applying the sound field control system improves the acoustic contrast by 17 dB-35 dB in the ideal case of true parameters. If the forward parameters are used without model updating/machine learning techniques, the acoustic contrast drops down to 10 dB, even though the uncertainty in the positions is small compared to the wavelength. This is the case of using directly an engineering sound propagation model such as Nord2000. Using model updating/machine learning to optimize, the contrast enhances the improvement again, especially at low frequencies.

We show that the use of a sound propagation model in combination with machine learning for estimation of the transfer-functions can improve the acoustic contrast created by the sound field control system in comparison to using traditional propagation modelling. This is feasible through the connection to the MONICA Cloud and IoT infrastructure that provides continuous data to update the sound propagation model.
The infrastructure of ASFCS consists of signal gateway/router (signal monitoring and control of main PA system), hub (Audio Interface), Core (DSP), and loudspeaker system (multichannel amplifier and loudspeakers). The ASFCS core calculates optimal solution based on sound propagation model. Multichannel signal rendering (filtering) may be processed using general PC or laptop but can be developed as stand-alone DSP for efficient calculation.

8 List of Figures and Tables

8.1 Figures

8.2 Tables

9 References

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