**Management Of Networked IoT Wearables – Very Large Scale Demonstration of Cultural Societal Applications**
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# D4.3 Validation of the ASFC and Noise Monitoring System Configuration and Model Updating 3

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# Index:

# 1 Executive Summary

This report documents the development of the Adaptive Sound Field Control (ASFC), the local Quiet Zone (QZ), the Model Updating 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 presents simulations and measurements (pre-tests and pilot tests) in order to validate the chosen strategies. The report is an updated version of the report D4.2, now covering also the findings and lessons learned from the third year of the project, including pilot tests in Torino Kappa Futur Festival (KFF2019) and Roskilde SOUND2019 (ROFH2019). 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 ASFC to the venue's PA system is needed. From the first year, we have applied DANTE (**D**igital **A**udio **N**etwork **T**hrough **E**thernet) protocol to ASFC to avoid conversion between various signal formats and to manage long-distance transmission with a simple and reliable connection using standard network infrastructure. Also in the last year (2018), to monitor different EQ/Delay filters and compression effect, we developed a high-power analog signal monitoring device called "Sniffer" which can monitor the end-signal from the PA amplifier.

The sound zone signal processing and optimization algorithm in ASFC 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. From the large-scale outdoor pre-test this year (2019), we found out that even when the controlling secondary array (a set of loudspeakers) is located proximate to the audience (within 10m), an audience cannot perceive the negative effect from the sound source. Therefore, we could select the PM-ACC solution to focus more on making the "contrast" effect.

Aiming for a very large scale cultural event, there should be some careful adaptation of the technique. Control of the low-frequency band would require a regularization technique that gives a robust solution that can work in the outdoor conditions. Furthermore, 'large-scale' conditions will impose various physically constraint real-time issues, leading modification of the given optimization scheme.

The ASFC can considerably increase the difference in sound pressure level between the audience area and the surroundings of the venue 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 ASFC (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.

Local quiet zones (QZ) in loud environments can support communication and minimize the noise exposure of staff. Phone calls in emergency situations are an example of how even issues of safety can be supported. The challenge is to provide a zone of quiet close to a loud event area. While sound energy needs to be inserted in order to cancel the unwanted noise, such a system has to take that the surrounding area is not affected by disturbance introduced by the quiet zone controller. The system is made of active electronics that cancel out the low frequencies and a wall that blocks the high frequencies. In order to achieve the highest possible attenuation in the zone without interfering with 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.

Validation of the Source Separation/Contribution techniques for the Noise Monitoring System has been successfully performed in the ROFH2019.

In all, the final validations of the ASFC as well as the Noise Monitoring configuration, using simulations and measurements, have shown positive results, indicating potential usages in the coming cultural event situations.

# 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. To achieve the objectives, the ASFC was developed as a Sound Field Control System and integrated with the organizer's Public Address (PA) system into the Acoustic Closed Loop System. The WP also contains a noise monitoring system. Initially, data for the model updating of the ASFCS was planned to be provided by stationary IoT SLM and regarding apps but due to the limitations of the number of IoT SLMs available, the data for the model was provided using ordinary microphones. However, the validation of the system was done by IoT SLMs and the MONICA COP. 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 updated parts of 'Validation of the Adaptive Sound Field Control (ASFC) for far and near field control, etc. (T4.1). The 'Noise Monitoring System Configuration' in relation with Task 4.2 and 4.4 is already delivered in D4.4 'Precision IoT enabled Microphone Sensor 1' and D4.5 'Precision IoT enabled Microphone Sensor 2'. The updated approach for 'Sound Propagation Model Updating' of T4.3 is discussed in section 5 in relation with the experimental results in Chapter 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 Zone – near field control
- Subtask 4.1.3 Sound zone signal processing and optimization

## 2.2 Background

The WP approached working on three physical scales: a) the external region – minimizing Annoyance; b) the audience area – maximizing Sound Quality[2], and c) quiet zones 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.

Furthermore, we could conclude that the negative effect on the sound quality of the audience area is negligible from the pre-test and experiment results (See Brunskog et al, 2019), so only the two approaches are discussed for this report.

[1] The abbreviation ASFC is used to 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, Echoes, etc. In this project, good sound quality in the audience area means not disturbing the sound made by sound engineers for the concert.

## 2.3 Tasks and approaches

The tasks in WP4 are here described shortly (mostly corresponding to the text in ANNEX 1 (part A)) and the approaches are briefly summarized.

### 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. The system only focuses on reducing the low-frequency component.

### 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 are added to control the sound level in the external zone. The PA system input signals are fed into the ASFCS for the real-time rendering of the sound. For controlling the low frequencies (LF), the ASFCS is 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 is maximized using various optimization process. Current meteorological conditions are taken into account in the control loop, described in T4.3.

### 2.3.3 Quiet zones – 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, which means that the quiet zone area is not moving nor degraded in its performance due to 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 Least Square and Pressure Matching solution for the purpose of robust optimization, 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 the reports 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. A candidate for the forward model is *Nord2000* (Plovsing,

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

2006), 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. During the first year, we investigated the adequateness of Nord2000 as a proper propagation model for outdoor sound field control, presenting severe limitations when modeling sound propagation at low frequencies. It implied the research and development of alternatives. During the second year a new model based on spherical harmonics was successfully tested under controlled conditions in the anechoic chamber for a scaled setup of 2 x 5 m (Caviedes et al. 2019). During 2019 the model was improved to incorporate these effects while keeping the simplicity needed for fast computations.

### 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 distributed 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. For detailed information, the report D4.7 provides complete summary of the tasks.

# 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 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 enables 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).

From the preliminary listening test taken in the large-scale outdoor experiments (Brunskog et al, 2019), it is known that even though the secondary array is placed very close to the audience area, the audibility of the secondary loudspeaker is negligible due to the masking effect of the human auditory system. This implies that the optimization problem only needs to consider the level in the dark zone. Therefore we defined a new performance indicator *Insertion Loss (IL)* which is the mean change in sound pressure level in the dark zone when comparing the sound field with active and deactivated control sources (see Chapter 5, Sound zone signal processing and optimization).

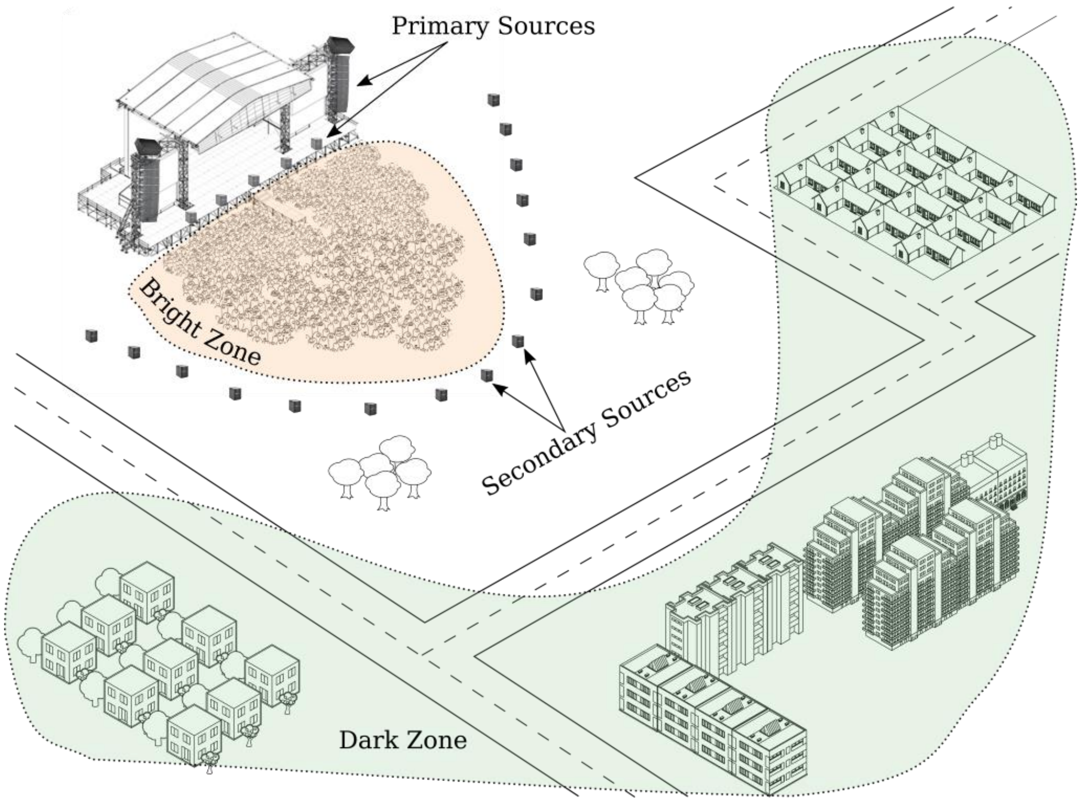


**Figure 1: Sketch of the adaptive sound field control system, which is optimized for a good audio experience in the bright zone and low sound pressure level in the dark zone**

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 approach in the development of the ASFCS because 1) fewer sources are controlled, and 2) sound engineers will be less skeptical because the main system is left untouched except for the monitoring purpose. Moreover, 3) 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 Insertion Loss 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.

| 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 |

**Table 1: Comparison of operation modes**

The ASFCS must be distinguished from generating *Quiet zones* (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. The ASFCS consists of specific hardware which needs 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 ASFC solution is related to the following use case:

**Sound Level Adjustment:**

- The ASFC provides an optimized sound field in the audience area and minimizes the impact on neighboring areas

## 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 propagation model behind the Sound Heat Map is described in Chapter 5 of the report D4.2).

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.

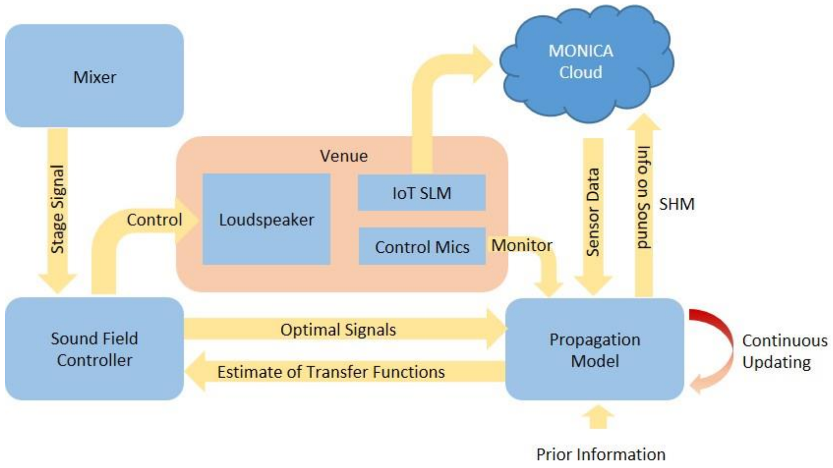


**Figure 2: Information flow in the ASFCS**

### 3.3.2 Infrastructure and deployment of the ASFCS

#### 3.3.2.1 Extension mode versus Full Mode

In practice, the ASFCS is 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. Optionally, 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.

In most of the pilots, accessing to the primary sources is very much restricted in the event situation, only partial monitoring of the PA channels is allowed. Therefore, the ASFCS was used as the extension mode throughout the MONICA project.

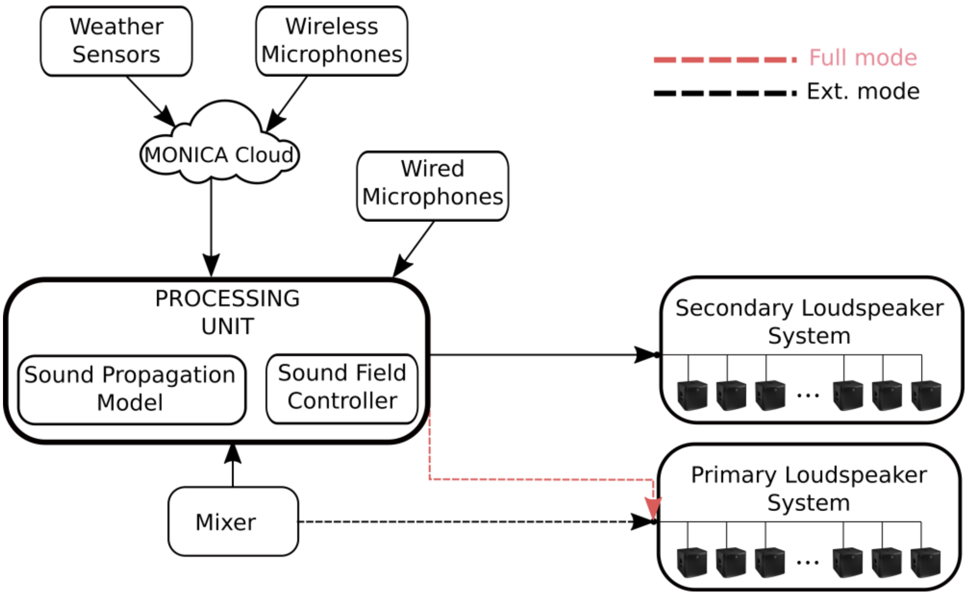


**Figure 3: Infrastructure and deployment of the ASFCS. In full mode the primary loudspeaker system is fed from the Processing Unit. In the Extension Mode the primary loudspeaker system is fed directly from the mixer**

#### 3.3.2.2 Infrastructure and signal flow of the ASFCS

The infrastructure and signal flow of the ASFCS is described in Figure 4. *ASFCS Core* is a processor that calculates optimized driving solution. 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. 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 red dotted line in Figure 4 represents audio analog signal flow over DANTE (Digital Audio Network Through Ethernet) protocol and the black 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). We selected the DANTE protocol for connecting ASFCS components because the distances between ASFCS components can be very long compared to the standard audio device setup. To ensure the reliable signal transmission over several hundreds of meters, signal routing over Ethernet cable is the most suitable approach. ASFCS requires monitoring a high number of PA channels, but this can be easily done when the DANTE protocol is used. Once the signals are on the DANTE network using any DANTE-enabled device, now they are ready to be controlled.

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 accurate monitoring, it is recommended to monitor the whole electro-acoustic signal from the last node (Monitoring Path 2 in Figure 4). In this case, all analog channels from the PA amplifier should be fed to Analog-to-Digital Converter (ADC) in ASFCS. In 2018, we have developed a device called *Sniffer*[4] which can monitor the analog output (The monitoring path 2) from the PA amplifier. The effect of using nonlinear amplifiers or transducers is shortly reviewed in (Lumpert, 2019). Instead, the monitoring can be done at the previous

[4] The Sniffer is a hardware devise developed by DTU for the MONICA project. It is placed in the signal line between the PA amplifier and the PA loudspeaker, and thus monitor the signal actually provided to the loudspeaker. This signal is sent to the DANTE interface for further processing, see Figure 4.

nodes (Monitoring Path 1) if the transfer function to the end node is known in advance. In this case, digital monitoring outputs are fed into any DANTE-enabled devices 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 through DANTE through the inverse path of monitoring.

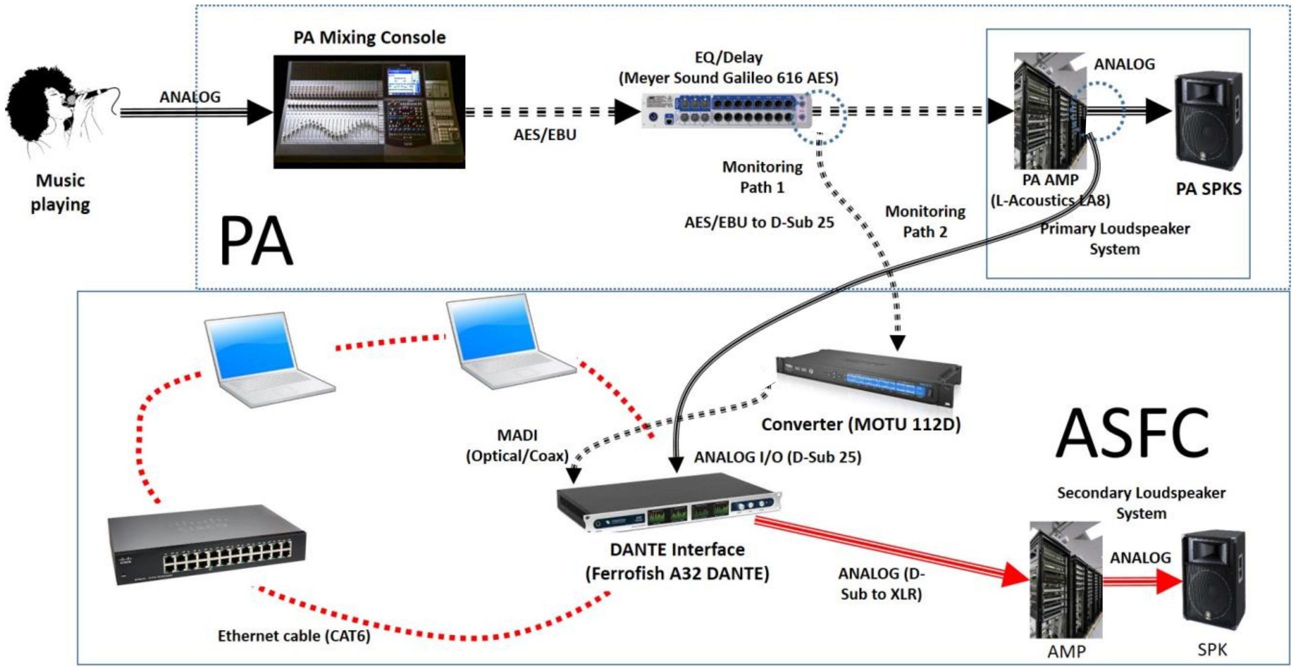


**Figure 4: Example of ASFCS connected with PA system of the pilot at Tivoli**

## 3.4 Simulations (Feasibility test)

Throughout the project, the sound field control principle was validated with many 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 results have been already published by the authors in (Heuchel and Caviedes, 2017) and also can be found in the report D4.2.

## 3.5 Measurements

In 2019, we have conducted many lab-measurements. But in here we will mainly address one large-scale outdoor test held at Roskilde (Section 3.5.1), and two pilot tests (Section 3.5.2 and 3.5.3).

### 3.5.1 Large scale outdoor experiment

In June 2019, another full scale test was conducted on the premises of the Roskilde Festival. The goal of this experiment was to investigate and test the sound field control system together with the sound propagation model (See 5.1.2 Sound Propagation Model) in realistic conditions. In comparison to previous tests, we used here the sound propagation model to estimate transfer-functions between loudspeakers and the control region. The sound propagation model was fitted using only 3 microphones in each zone.

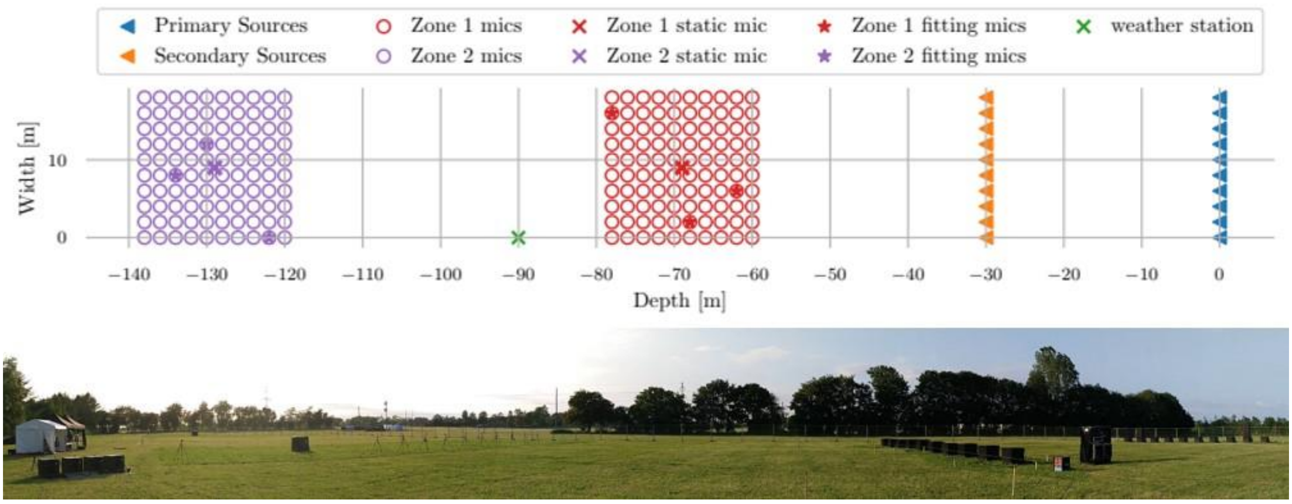


**Figure 5: Experimental setup of large scale experiment at Roskilde. Top: Plan of microphones and loudspeakers. Bottom: Photograph of setup.**

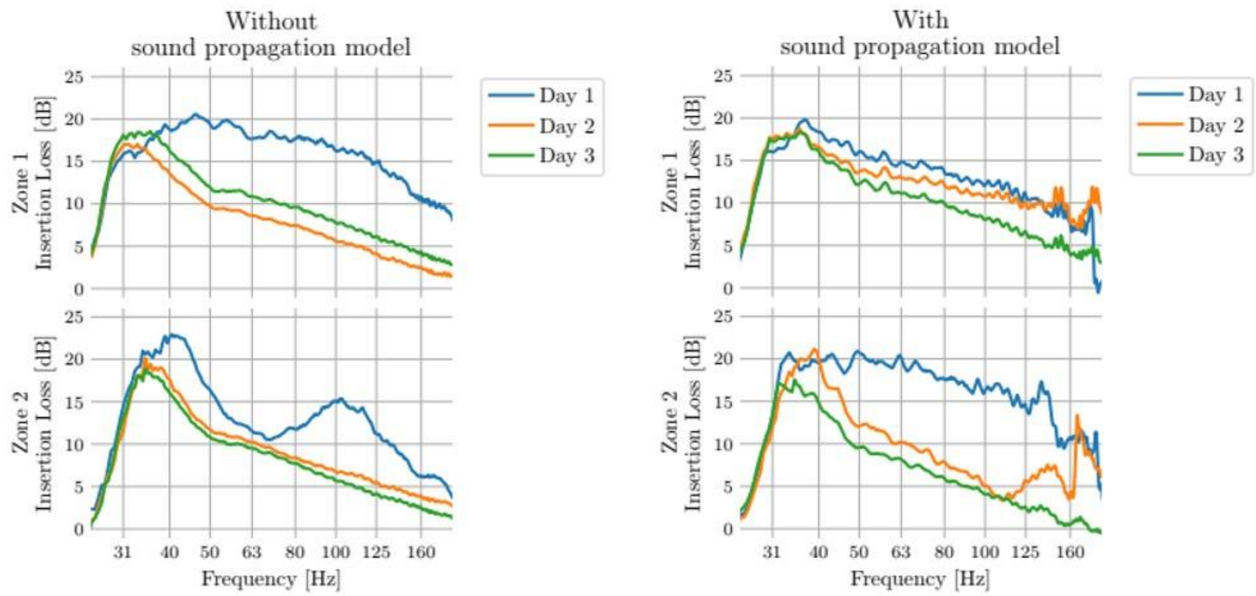


**Figure 6: Performance metrics at outdoor experiment.**
**(Top) insertion loss. (Bottom) primary to secondary ratio**

### 3.5.1.1 Experimental setup

Figure 5 shows a photo of and plan of the experimental setup. The primary sources comprised 10 subwoofers arranged in a line array with 2 m spacing. The secondary sources consisted of 10 subwoofers of the same type. The subwoofer model had an intrinsic cardioid radiation pattern and a nominal frequency range of 37 − 115 Hz (-5dB). We did not use a double layer secondary array here, as previous experiments showed control of the bright zone is not necessary. The cancellation of the sound field was tested in two separate zones (zone 1 and zone 2) to investigate the performance at two different distances. The sound fields were measured at 100 microphone points in each of the two zones.

We measured all transfer-functions on three consecutive days (Day 1, Day 2, Day 3). On the afternoon of the third day, we measured the complete sound field control system using the following procedure: first, the sound propagation model was fitted using only three microphones in either of the zones. Then, the sound propagation model estimated the transfer function on the remaining microphones. Using this estimate, we computed control filters for secondary sources. The insertion loss was then computed by comparing the measured sound fields of the system with secondary sources turned on and off.

#### 3.5.1.2 Results and Discussion

Figure 6 compares the measured insertion losses of the sound field control system in zones 1 and 2 when using no model and the input from 100 microphones (left figure) and when using the model based on only 3 microphones (right figure). In both cases, large insertion losses above 10 dB can be achieved in most of the frequency range of the loudspeakers. However, the result varies strongly with the transfer function from different days.

#### 3.5.1.3 Conclusions

We could successfully validate the use of the sound propagation model for sound field control in realistic outdoor conditions. Reductions of average sound pressure levels were as high as 20 dB. However, changes in atmospheric conditions can lead to large performance variations. The adaptive method of section 5.1.2 (Also see Figure 2) is tackling this issue.

### 3.5.2 Pilot Test: Kappa Futur Festival 2019 (KFF2019)

The sound field control system was again put to test during the Kappa Futur Festival 2019 (Figure 7). Compared to Kappa Futur 2018 (See the report: D4.2 Validation of the ASFC and Noise Monitoring System Configuration and Model Updating 2) we used a much finer grid of microphones that were concentrated to the churchyard only (no microphones on the roof). Bad weather caused a delay in the production schedule with meant very limited measurement time for the system and therefore only the non-adaptive static system was only tested.

**Figure 7: Kappa Futur Festival**

#### 3.5.2.1 Experimental Setup

The experimental setup can be seen in Figure 9. As in the year before, we placed the dark zone in a nearby church yard and the secondary sources at the border of the festival area. Note that these are nearly the same positions as in the year before, but shown here mirrored such that the positons can be identified in the panoramic photo (See Figure 8. We measured transfer functions from the stage system and the secondary sources the day before the festival. For that we used 120 microphone positions (red in the figure). We did not yet use any sound propagation model at this pilot. The resulting insertion loss was measured during day one of the festival with filters computed from the day before.

**Figure 8: Control subwoofer at Kappa Futur Festival. The festival stage can be seen in the back.**

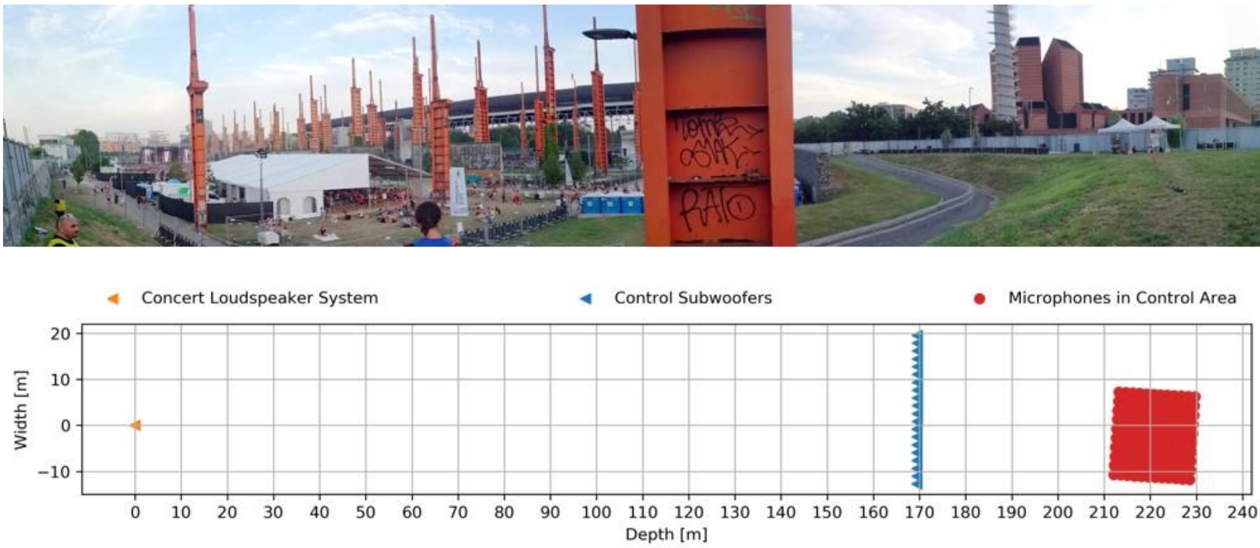


**Figure 9: Setup at Kappa Futur Festival 2019. Top: photo with stage at the very left and church with dark zone at very right. Bottom: Layout showing distances.**

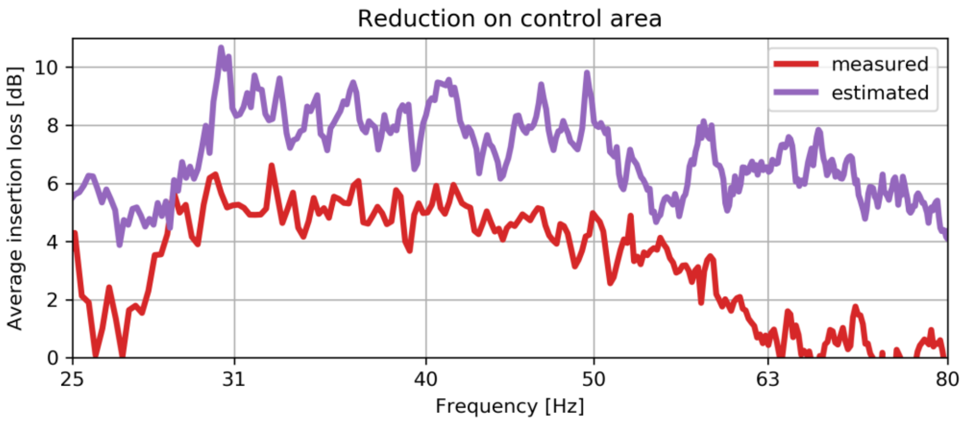


**Figure 10: Estimated and measured insertion loss in the dark zone at Kappa Futur Festival 2019**

#### 3.5.2.2 Results and Discussion

shows the estimated and measured IL by comparing the transfer functions to the dark zone with secondary sources turned off and on. The estimated IL is calculated from the measured transfer function and is clearly higher than the actual measured value. We expect this difference to come from a change in atmospheric conditions between the two days.

#### 3.5.2.3 Conclusion

In this pilot, the sound field control system achieved similar insertion losses compared to Kappa Futur Festival 2018. However, an effective IL is limited to very low frequencies below 63 Hz. We could see that practical limitations on the measurement process in both events (KFF2018 and KFF2019) have greatly limited the performance of the ASFC. First, the dark zone in the churchyard was completely isolated from KFF2019 event area by a tram railway without electricity or means of information connection (WiFi or cellular). Due to this environmental limitation, a simultaneous data recording/monitoring and synchronization (using a wired connection) was not possible. In 2018, the LAN-XI solution provided by B&K was used to overcome this limitation and the GPS-synchronization using IRIG-B signal method was tried in 2019. However, at least 30 minutes of a time delay between transferring, processing and testing of the solution was inevitable, which made difficult to validate the calculated solution in the time-varying environment.

From the lesson learnt from the KFF2019, we decided to test the adaptive solution of the ASFC at Roskilde SOUND2019, in order to guarantee sufficient measurement time for the validation process and to ensure that simultaneous data recording/monitoring and synchronization is possible.

### 3.5.3 Pilot Test: Roskilde SOUND2019 (ROFH2019)

A sound propagation model based, adaptive version of the sound field control system (ASFCS) was tested at the Roskilde SOUND2019 (ROFH2019), which was organized be the Roskilde Højskole on November 30th .

Figure 11: Scenes from Roskilde SOUND2019

#### 3.5.3.1 Experimental Setup

A layout and photo of the experimental setup for this pilot is shown in Figure 12. As in many of the other experiments, we used 10 primary sources and 10 secondary sources. In the dark zone, we placed 7 control (fitting) microphones and 32 validation microphones. The ASFCS continuously recorded the sound field with the control microphones, fitted the sound propagation model with that data, estimated transfer functions on a fine grid of virtual microphones and updated the control filters based on the estimated transfer functions.

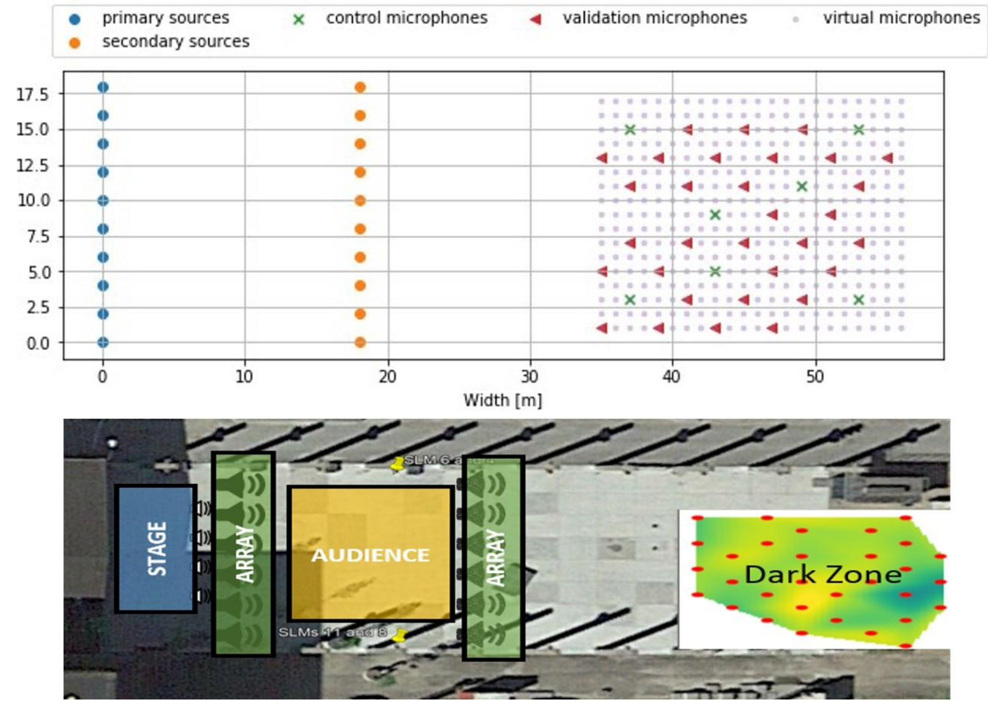


**Figure 12: Layout and photo of sound field control setup at Sound Summit 2019**

### 3.5.3.2 Result 1: Propagation modelling based approach

The insertion loss created by the ASFCS during the pilot concert in a 15-minute measurement window is shown in Figure 13. After a very quick convergence period, the average insertion loss stays constantly above 10 dB. The average insertion loss at the control and validation microphones is approximately equal, showing that the sound propagation model fits the sound field in the whole dark zone equally well.

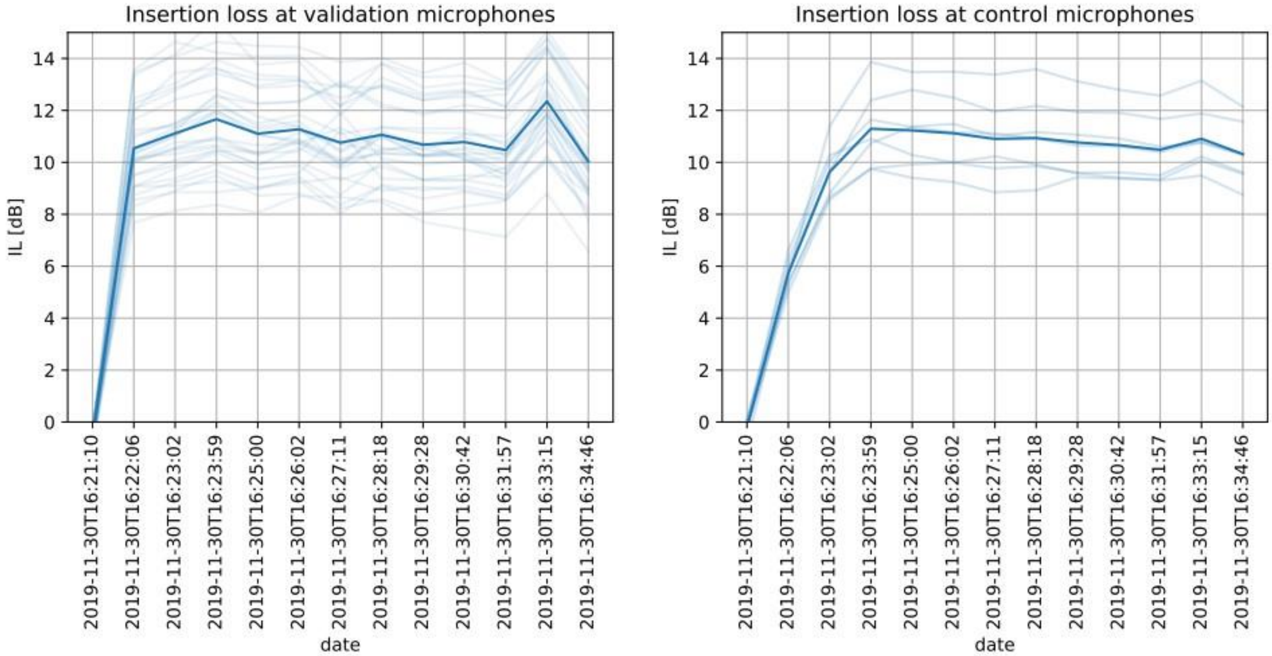


**Figure 13: Insertion loss over time averaged over the frequency range 37-110 Hz at validation (left) and control (right) microphones during a 15 minute measurement window at the Sound Summit pilot. Mean over microphones in thick line.**

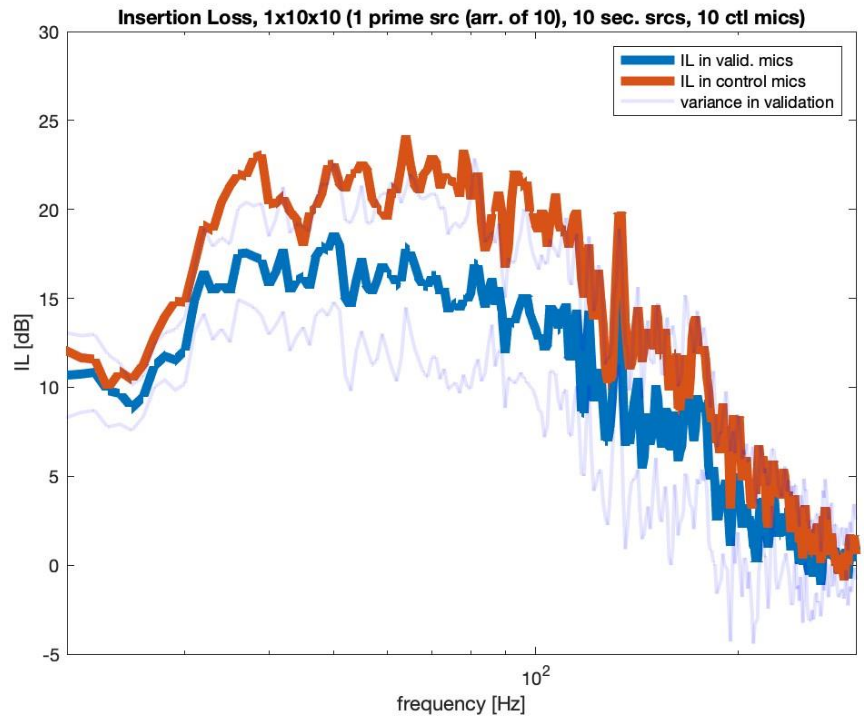


**Figure 14: The performance of ASFCS when driven by adaptive QZ algorithm using 10 control microphones**

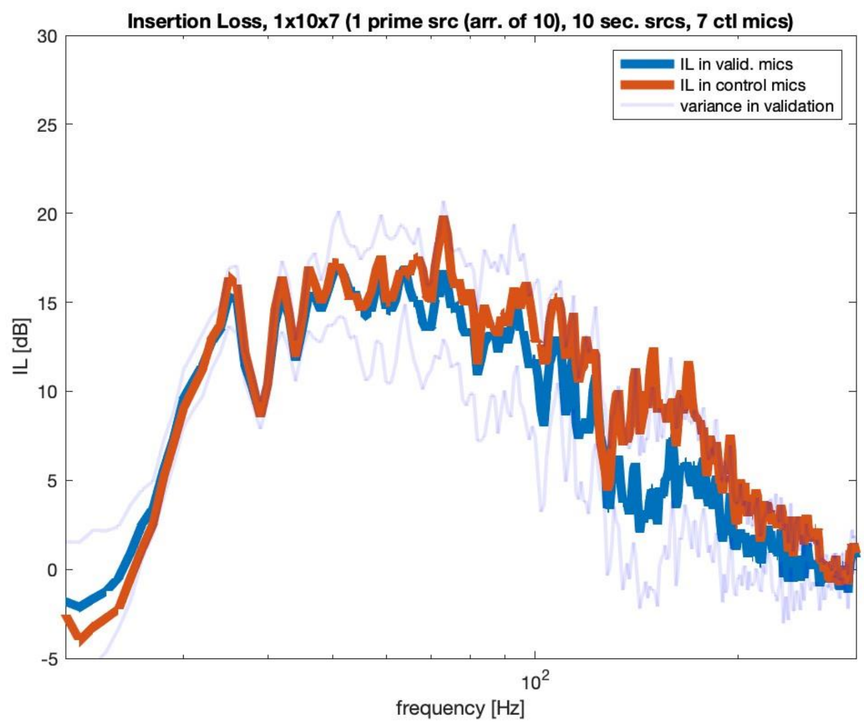


**Figure 15: The performance of ASFCS when driven by adaptive QZ algorithm using 7 control microphones**

#### 3.5.3.3 Result 2: Based on the QZ algorithms

We also applied the quiet zone system algorithms based on adaptive filtering methods as a solution to the far-field sound field control problem. See chapter 4 for details about the adaptive filtering approach. The physical hardware setup was the same as described before. Two sets of control microphones were tested:

1. The ten closest control microphones to the secondary sources
2. The same 7 control microphones that have been used for the model update approach.

The performance was validated with the remaining 28 and 31 validation microphones in the zone.

The adaptive filters converged within approximately one minute, which means that they found their optimum value and settled at a steady state. Figure 14 shows the results of the configuration with 10 control microphones,  shows the results for 7 control microphones. It should be noted that the QZ algorithm does not include a sound propagation model and therefore it requires that the microphone are placed inside the control zone, whereas the control scheme based on the sound propagation model allows for more flexible placement of the control microphones.

#### 3.5.3.4 Conclusion

We successfully demonstrated the adaptive, model-based sound field control system leading to stable average sound pressure level reductions above 10 dB in the frequency range of the subwoofers.

## 3.6 Conclusions

The adaptive sound field control system was implemented and tested successfully, demonstrating at least 10 dB of attenuation in the dark zone. The model based approach makes it possible to use relative few microphones and allows for a flexible placement of error microphones. As an alternative method the adaptive filter algorithm was shown to give slightly better results, but this method is less flexible with respect to microphone placement. The minimum number of microphones required is closely related to the “openness” of the venue, and the venue of the ROFH2019 was quite open space but we are expecting that applying various strategies in predicting the accurate propagation model in space would delimit this open-ness constraint. The use of a few control microphones makes the ASFC more usable in many real outdoor events. However, deploying the system will require additional time for preparing and tuning the system, which adds to the total rental cost of the PA system. This could limit the usability of the system because it requires the PA system to be installed in advance (a few days) for the validation and tuning. Practical non-scientific/non-technical aspects are limiting, therefore a good collaboration with sound engineers, venue organizers and local authorities is critical.

# 4 Quiet Zones – near field control (T4.1.2)

## 4.1 Technology overview

The Quiet Zone (QZ) system is a noise barrier that makes use of active electronics 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 active control part is based on (traditional) active noise control and applies multichannel adaptive filters.

The active part of the QZ system consists of arrays of loudspeakers that synthesize the canceling sound-field to cancel out the noise-field in the desired zone. Such multichannel active noise canceling methods in free-field are described (Kuo, 1996) and (Hansen and Snyder, 2002). The active noise canceling 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 that 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 and broadband noise sources (the music signal coming from the many distributed loudspeakers of the venue. Figure 16 shows an open-air concert area with ASFC and quiet zones.

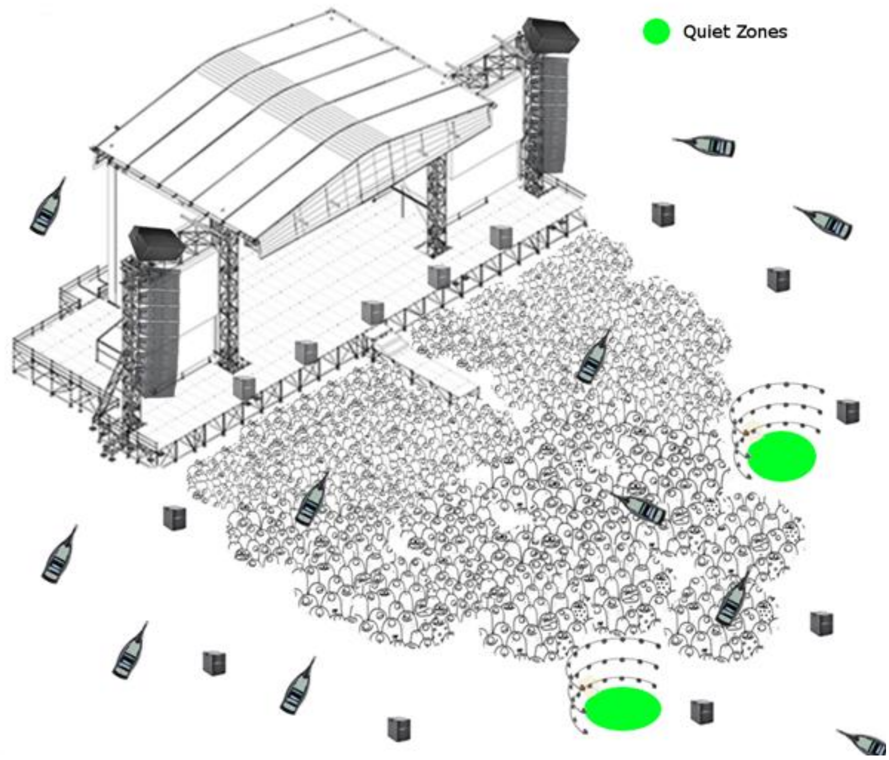


**Figure 16: A possible use case of quiet zone systems in a loud outdoor venue**

Figure 17 shows the first prototype of the quiet zone system. It has been constructed and tested in the *Fredagsrock* Concert at the pilot venue Tivoli. The quiet zone area is about 1.5 m x 1 m x 0.5 m (w/d/h) in order to accommodate at least two persons for easy verbal communication.

**Figure 17: Prototype of the Quiet Zone System tested at the Tivoli pilot site. The controller and the error microphones are not shown here.**

The active part of the system makes use of a multichannel filtered reference signal least mean square algorithm (MC FxLMS). Based on parameter studies in simulation a practical number of 4 microphones and 4 secondary loudspeakers have been chosen. The microphones measure the sound pressure in their individual positions and feed the algorithm, which finds the optimal set of filters, that are needed to cancel the sound in the error microphone positions as best as possible. Changes in the electro-acoustical conditions are taken into account, the system adapts automatically to such changes by adjusting the filter coefficients continuously. Figure 18 shows such a multichannel controller architecture.

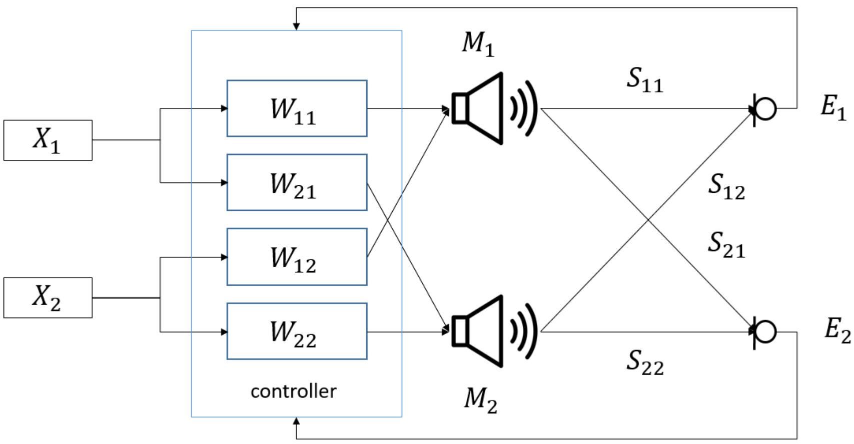


**Figure 18: Multichannel Controller Architecture. X: Reference signal (from mixing engineers console), $W_{ii}$ FIR filters, $M_i$ Secondary loudspeakers, $S_{ii}$ sound propagation paths, $E_i$: Error measurement positions.**

The passive element has two purposes. It acts as a noise barrier and blocks high frequencies from the quiet zone on the one hand, on the other hand, it is the cabinet for the four subwoofers and provides the appropriate volume, which is needed to keep the resonance frequency of the loudspeaker low in order to get a good performance of sound radiation down to ca. 35 Hz.

The overall technical system consists of standard professional audio equipment, which was slightly changed in-between the individual pilot tests. See Figure 19 with a general setup configuration.

- 4 one channel power amplifiers (1KW)
- 4 subwoofers

- 1 Multichannel Audio Interface
- 4 B&K free-field microphones + signal conditioner
- 1 Computer running the ANC software in MATLAB

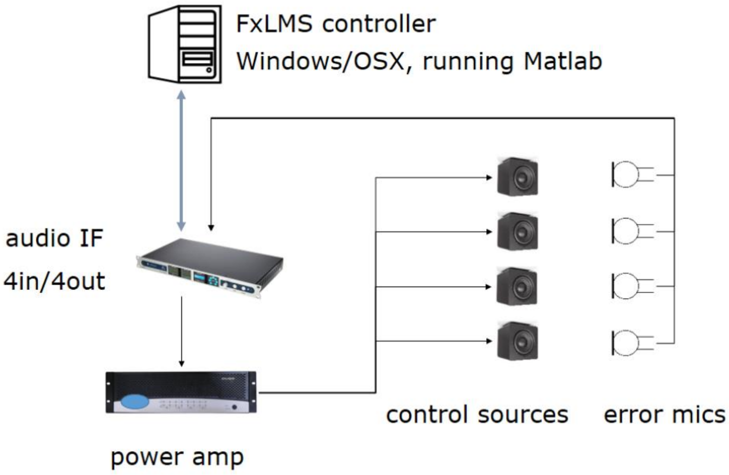


**Figure 19: Technical Architecture of the Quiet Zone system (active part).**

## 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

The position of the quiet zone systems will be displayed in the COP (Common Operational Picture); this was implemented in the 2018 Tivoli test, the Kappa Future Festival in Torino and the Tivoli test in 2019.

### 4.3.1 Audio Signals from the Venue

The quiet zone system receives audio signals from the venue's console or from behind the PA's power amplifiers. The latter would be the primary choice because the standard PA system incorporates signal processing units into the power amplification stage. For example, devices such as limiters cause nonlinear components which will decrease the performance of the active noise controller. These audio signals, that are needed to feed to an active noise control algorithm, will be called the reference signals in the following. In all pilot events were reference signals obtained from the console (mixer) and from signal sniffer devices that were inserted in-between chosen power amplifier and speaker channels. Both versions finally led to good results.

### 4.3.2 Operational Status

The operation status of the system could only be evaluated indirectly. The active noise controller is running under heavy computational cost and robust performance is needed to avoid audible dropout. The calculation of the performance measure insertion loss (IL) in real-time and on the same machine would have added the risk of dropouts. The performance could be otherwise roughly evaluated through the real-time metering of the sound pressure levels in the error microphones. This is a rough estimate because the sound meters show a broadband signal, while the active noise controller acts on low frequencies only. A quantitative estimate of the performance in real-time is the insertion loss suggested. IL is directly related to the amount of sound pressure per frequency that is attenuated by the system. Such a measure could be estimated on a parallel computer in future applications and presented as time-varying frequency spectrum on a screen or sent to the COP.

## 4.4 Simulation

Simulations have been carried out to estimate the optimal performance under practical constraints. These simulations led to the configuration of 4 control sources and error microphones, which has been used in measurements to validate the theory and was finally tested at the Tivoli pilot site. The configuration has been changed for the Kappa Future Festival. Figure 20 shows the results of simulations using a band-passed pink noise signal (100-300 Hz) as a reference signal (which represents the music from the venue's PA). What we see is that the closer we come to the microphone array, the larger is the insertion loss, which makes sense because the optimization routine tries to reduce the pressure in the error microphone positions. What also can be seen is that there is still promising attenuation in the close vicinity of the error microphones – the quiet zone area. So the quiet zone area can be seen as the result of an acoustic shadow effect behind an active barrier.

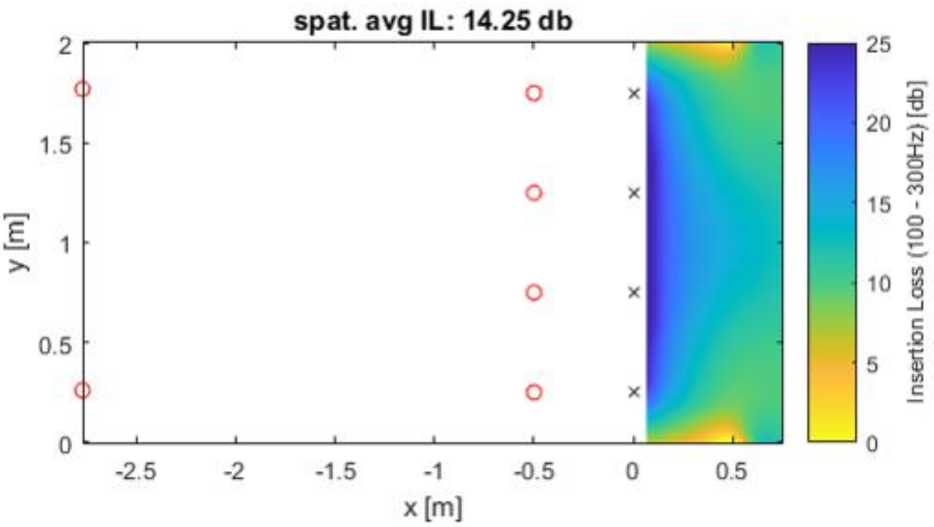


**Figure 20: Results from Simulations. This plot shows the Insertion loss in the quiet zone (colored area). Red circles on the left: the venue loudspeakers, red circles in the middle: control speakers, black crosses: error microphones. The insertion los averaged over the whole zone is 14.25 dB.**

## 4.5 Measurements

The results from simulations have been validated by measurements in an anechoic chamber and the three pilot tests. The setup has been arranged according to the arrangement in the simulations. Figure 21 shows the results measured with a 60 channel microphone from B&K in the quiet zone. This array was moved 4 times to measure 240 positions in total. It can be seen that the measured insertion loss is quite congruent with the simulations. Furthermore, the vertical extension of the zone has been measured and it can be seen that up to 30 cm below and above the microphone array is still a reduction of about 10 dB for frequencies up to 300 Hz, see Figure 22. Finally, we can say that with 4 secondary sources and microphones it is possible to produce a volume of silence with ca. 13 dB average attenuation in the dimensions 2m x 0.75m x 0.6m (width/depth/height).

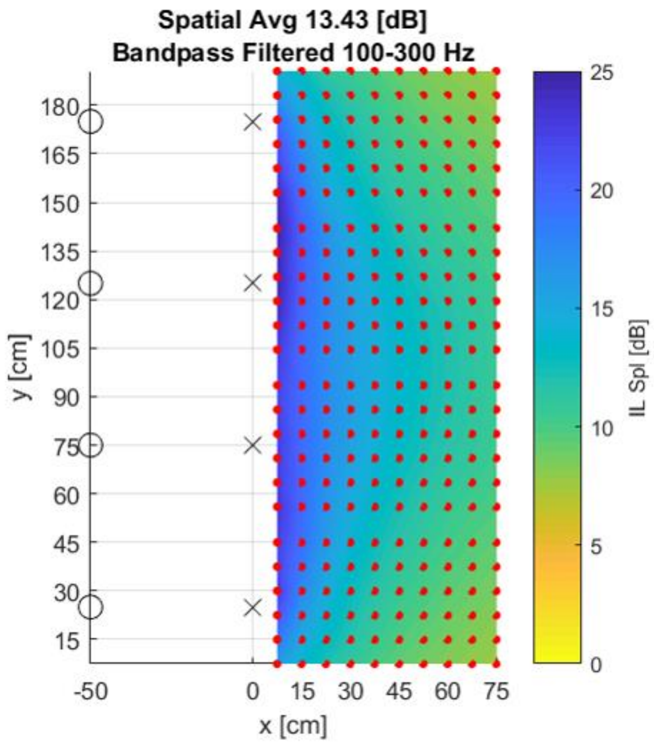


**Figure 21: Measured Insertion Loss in Quiet Zone, red dots: measurement positions, black circles: control sources, black crosses: error microphones**

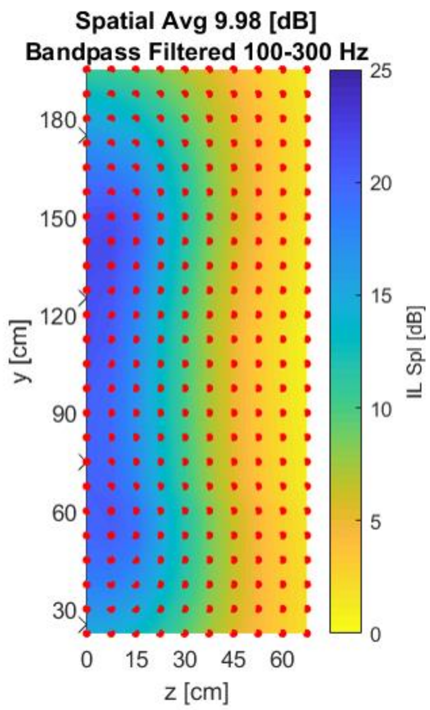


**Figure 22: Vertically measured Insertion Loss, up to 10 dB attenuation 30cm above the microphone array**

## 4.6 Pilot Test at Tivoli 2018

Finally, we constructed the complete hybrid system (active + passive) and tested it at the Tivoli pilot site in 2018 and 2019.   We tested in both cases the full hybrid system with similar configuration but slightly different algorithms. This chapter presents the results from the first test in Tivoli 2018. Due to practical limitations was the following performance measure chosen:

$$\text{SPL (with Quiet Zone)} = \frac{ErrorMic}{Reference}$$

$$\text{SPL (without Quiet Zone System)} = \frac{RefMic}{Reference}$$

In order properly measure the performance of a passive noise barrier it is needed to measure the transfer function from the noise sources to the point of interest, here the error microphones. Such measurement could not be done due to noise regularisations. Instead, was a reference microphone placed close to the quiet zone system but outside to estimate the contribution of noise without passive noise barrier. This reference microphone (Ref Mic) has been placed about 3 m aside the quiet zone. The error microphone (Error Mic), which was used to estimate the full hybrid system performance was at the front edge of the quiet zone, off-centered with 25 cm, which is a spot where the active and the passive element are expected to have high performance. Figure 23 shows the two curves gained from 20 seconds average.

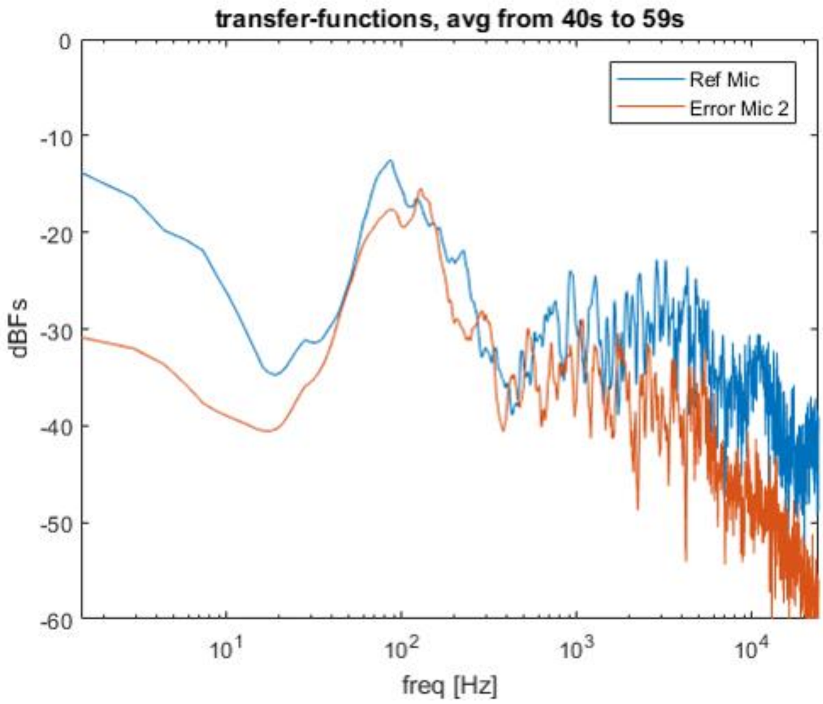


**Figure 23: Extracted transfer functions from an Error Microphone and the Reference Microphone.**

It can be seen that there is no attenuation for frequencies in the low end (30-300 Hz) and increasing attenuation for higher frequencies. The active controller did not work as expected, the passive as expected on the other hand. The evaluation of the system failure has later been analysed and it turned out that small change in the secondary path caused the algorithm to diverge. These small changes occurred because the secondary path was measured in the morning while they were used in the algorithms in the evening. This problem could be reproduced in simulations. The solution was to run the system with much smaller samples rate. Low frequencies are much more tolerant to uncertainties in phase.

## 4.7 Pilot Test: Kappa Futur Festival (KFF2019)

At the Kappa future Festival was the focus on the active control element. Here we used an array of the same subwoofers that were used by the stage. These subwoofers had a cardioid directivity pattern which forced us to set them farther apart to prevent complicated interference patterns in the close environment. The overall space that could be used for the system construction was limited because of safety regulations, which led to a reduction of components. The dimensions are:

Distance Error Mics - Control Sources: 1.5m

Distance in between Error Mics: 1.3m

Distance in between control sources: ca. 1.3m

The big distance between the subwoofer control sources was used because d&b recommends a minimum of 60 cm between the edges of each cabinet. So the distance is approximately the result of 60cm + 2x 1/2 width of the cabinet.

Finally, a low range (only subwoofers) 2x3x3 configuration was tested and a full range (including top speakers) 2x2x2 configuration. The configuration signature is: #reference signals x #secondary speakers x #error mics (# denotes "the number of"). Figure 24 shows the active quiet zone system on-site in close-up, Figure 25 shows the QZS to the stage.

**Figure 24: The active QZ system in close-up**

**Figure 25: Location of QZ in relation to the stage**

### 4.7.1 Results

The performance of the controller at the Kappa future Festival showed good performance close to the expected performance in simulations. Figure 26 shows the performance of the system in the error microphones and a set of validation microphones 50 cm behind the error microphone array. Further away from the optimal error zone could not be measured due to the limitations of space. But even behind the fence in ca. 2m distance could randomly passing people experience significant drops of sound pressure.

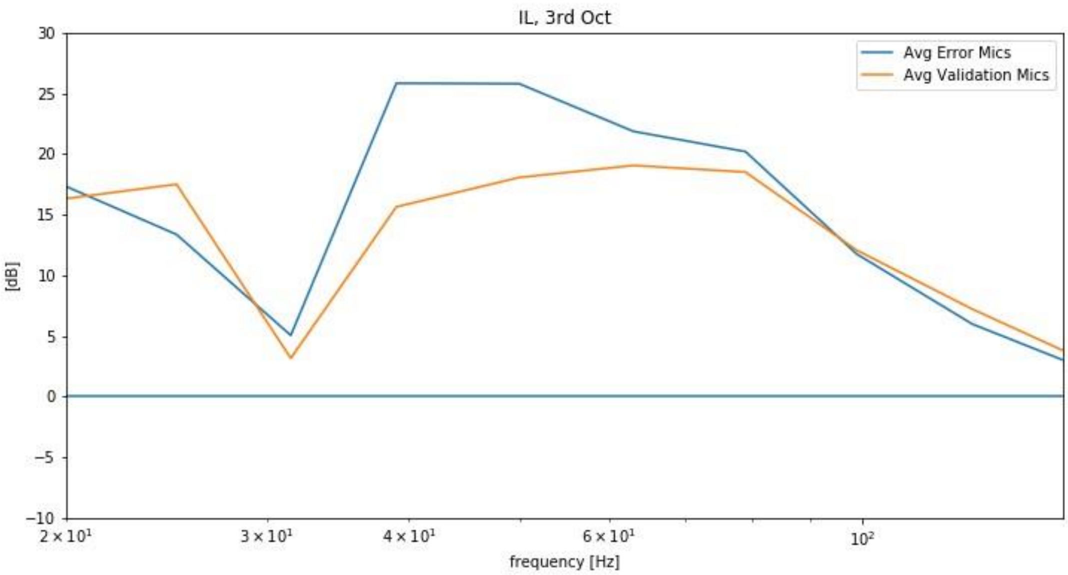


**Figure 26: The performance of the QZ system in KFF2019**

## 4.8 Pilot Test: Fredagsrock at Tivoli (Tivoli2019)

In the last test at Tivoli in 2019 was again the full hybrid system tested. In comparison to the Kappa Future configuration, the major differences are the distance of error microphones to the control loudspeakers and the distances in between the components themselves:

Distance Error Mics - Control Sources: 1.0m

Distance in between Error Mics: 0.5m

Distance in between control sources: ca. 0.5m

Because the sound pressure close to sources drops much faster than further away, is expected that the quiet zone area will be of smaller size. The shadow effect of the active barrier is reduced that way. This way is the quiet zone even more compact and the contamination of the surrounding sound field because of secondary source emissions is reduced as much as possible.

### 4.8.1 Results

Figure 27 shows the same two curves presented in the Kappa Future Pilot section, the IL in the error microphones and the validation mics. The frequency range here goes beyond 10kHz so that the effect of the passive element if visible. What we see is an overall good performance in the error microphones, whereas we see a gap of approximately 20dB around 150Hz in the validation mics. This is due to the decreasing performance at higher frequencies further away from the optimization points. (error mics).

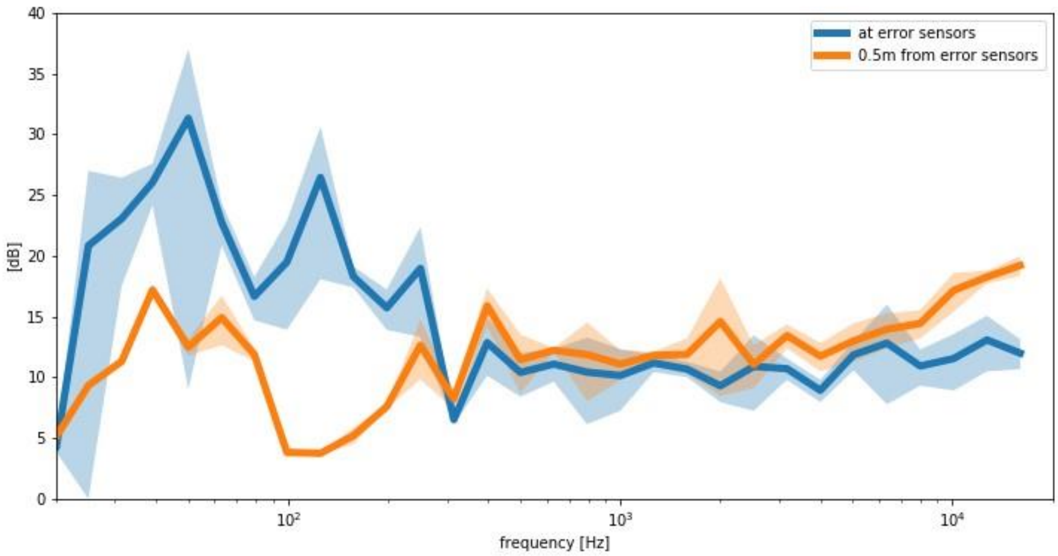


**Figure 27: The performance of the QZ system in Tivoli2019**

## 4.9 Conclusions

The full hybrid quiet zone system has been finally and successfully tested at Tivoli pilot event. The overall performance has been ok in the accessible region of the system (the quiet zone region, outside the error microphones). The gap in the frequency spectrum around 150 Hz could be treated with bigger distances of the error positions to the control speakers (see Kappa). Finally, it can be seen that relying on the shadow effect of active control barriers is very limited. Applying virtual and remote microphone methods to the control algorithms, to move the optimal zone of control into the quiet zone area is recommended.

# 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.

In parallel with the large-scale outdoor experiment (section 3.5.2), we have conducted an informal listening test to see the effect of a secondary array to the sound quality in the listening area. Total 21 subjects spread out in the listening area and tested whether they can discriminate the sound difference when the secondary array is turned on. However, all 21 subjects failed to discriminate the sound difference and only possible to perceive the difference when the subject is very close to the secondary array less than two meters. It means that we can ignore the sound quality degradation due to the secondary array. Therefore, PM-ACC solution that we use is modified in order to maximize the Acoustic Contrast and the pressure field error is neglected.

Figure 28 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*[6] in the bright zone are 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 a 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.

[6] The target field is the desired sound filed in the audience area. For example, this can be the propagating wavefield from sources in the position of the PA loudspeakers, without disturbing reflections and artefacts.

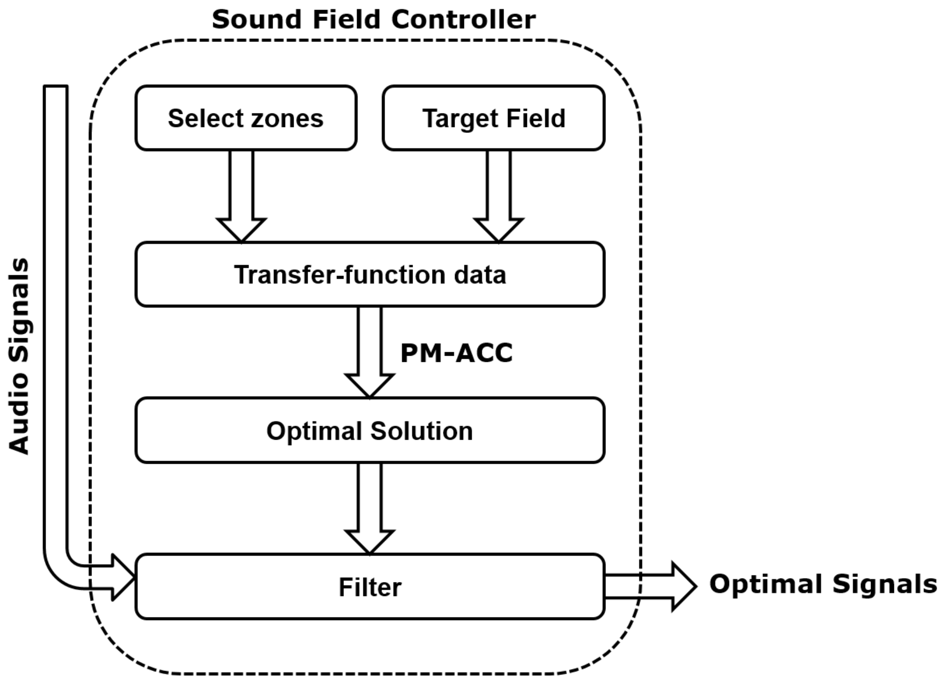


**Figure 28: Signal processing flow of the Sound Field Controller, which is a module shown in Figure 2.**

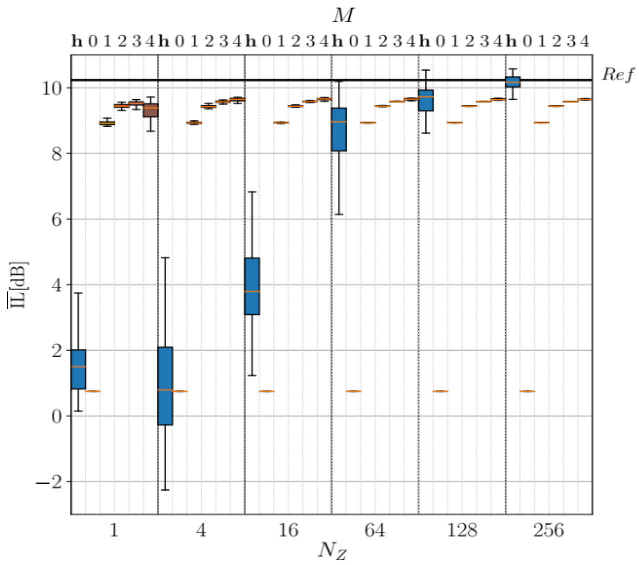


**Figure 29: Insertion loss averaged from 100 to 1000 Hz. The box plots show the median in orange and the boxes extend from first to third quartile. The whiskers represent the extreme cases (max and min). Bottom x-axis: Number of measurement positions. Top x-axis: Truncation number in the spherical harmonics basis. h: No model used. *Ref*: Insertion loss using half of the dense grid of measured transfer functions .**

### 5.1.2 Sound Propagation Model

Sound field control in outdoor concerts requires accurate estimates of the transfer functions between sources and receivers. Feedforward 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 for propagation models to characterize the sound field in such large areas and provide the ASFCS the proper transfer functions. During the first year, we investigated the adequateness of *Nord2000* as a proper propagation model for outdoor sound field control, presenting severe limitations when modeling sound propagation at low frequencies. It implied the research and development of alternatives. During the second year, a new model based on spherical harmonics was successfully tested under controlled conditions in the anechoic chamber for a scaled setup of 2 x 5 m (Caviedes et al. 2019). The acoustic contrast produced when using this model is very similar to the one we would get using a dense grid of microphones. Figure 29 shows a comparison of the insertion loss for such setup between measuring transfer functions at 1400 microphone positions and measuring only a few sparse microphone positions using the model (See Figure 30 for the setup).

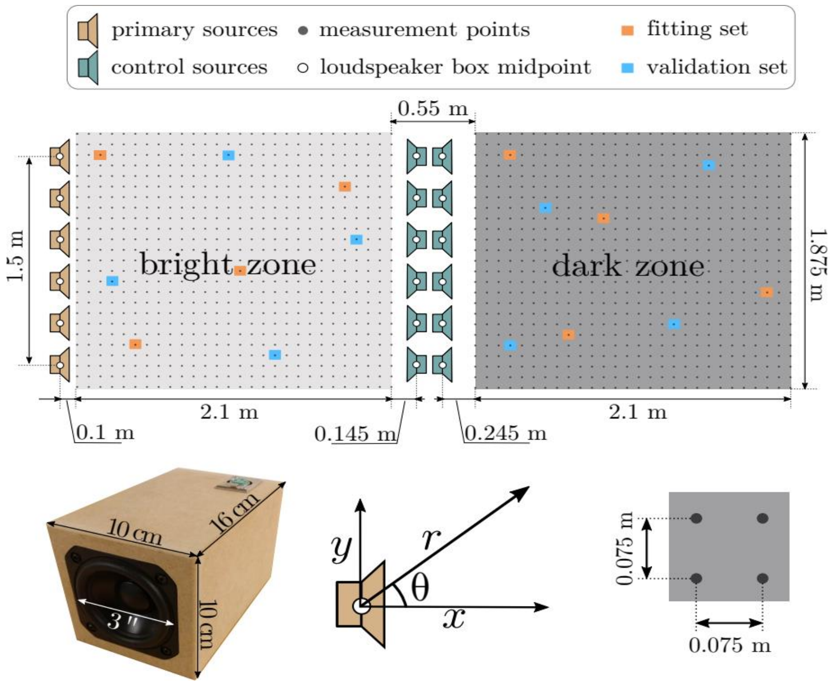


**Figure 30: Setup with 1400 microphones (dots) and 18 loudspeakers. Four microphones in blue were used to fit the model and reconstruct the transfer functions.**

Machine learning techniques are used to fit the parameters of the model according to the sparse measured data and deal with the uncertainties present in the parameters, leading to a good performance of the sound field control strategy. In (Caviedes and Brunskog 2018) we showed how machine learning techniques can be used in fitting models for sound field control problems.

These uncertainties come from different causes such as atmospheric variations, error in the measurements and noise. During 2019 the model was improved to incorporate these effects while keeping the simplicity needed for fast computations. This way it was possible to successfully apply this solution in two controlled large scale tests in Roskilde.

Finally, the adaptive function was also implemented in 2019. The model is constantly fitting to ongoing measurements during the concert, finding optimal parameters for the given conditions at that precise moment. To do so, the model was tweaked such that it is possible to de-multiplex the sound created from each different loudspeakers based on music recordings from the concert.

## 5.2 Services enabled

Signal processing and optimization algorithm is embedded in ASFCS computational core. Further information is introduced 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 Section 3.3.

## 5.4 Simulations and Measurements

The simulation and measurement results about the sound propagation model updating with machine learning and used together with the ASFCS is specified in (Heuchel and Caviedes, 2017) and can be seen in the report D4.2.

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

## 6.1 Technology overview

### 6.1.1 Noise Monitoring System

The Noise Monitoring System consists of the Source Separation/Contribution techniques (T4.4), the Annoyance measures (T4.4), the Noise Heat Map (Sound 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 the reports D4.4 and D4.5 and is thus not further covered here. For the implementation of the Sound Heat Map (See Figure 32), a simplified forward sound propagation model developed in T4.3 was used. For the detailed explanation about the Sound Heat Map, see the report D4.7, section 3.2.3.

The Noise Monitoring System has been tested during different pilot tests in 2019.

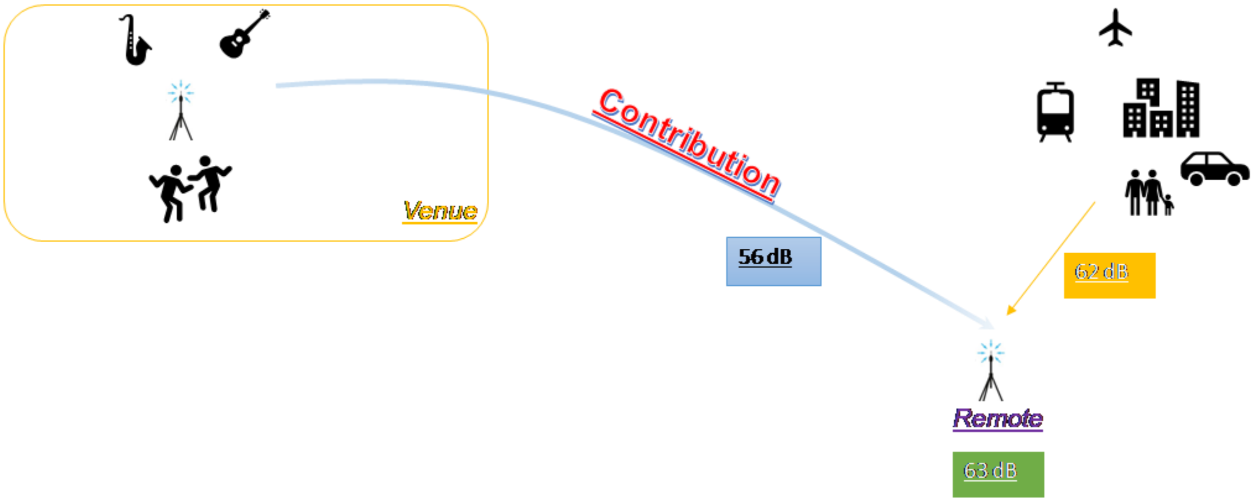


**Figure 31: Contribution analysis using the Coherence method, using one SLM in the venue and one SLM at a remote location, where we want to get the contribution from the venue only. In this example, the overall level measured at the remote location is 63 dB. This level is the sum of contributions from the venue in one hand, and from other noise sources (traffic noise, human activities…) on the other hand. The task of the Contribution analysis is to extract the contribution from the venue only, here 56 dB.**

The approach that has been selected is named the Coherence method in the document. 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 ('remote' signal).

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. The basic elements and the hardware developed here have been used in different 2019 pilot tests: Kappa Futur Festival and Roskilde SOUND2019 (ROFH2019). However, the Coherence method requires to transfer audio recordings to the cloud to perform the Contribution analysis. This necessitates a stable network that can transfer a relatively high rate of data (1 Mbit/sec was required). We did not have a dedicated network at the ASFCS location during the Kappa pilot test: it was not possible to get audio data recordings transferred to the SLM gateway. But it was possible to get a good and stable network during Roskilde SOUND2019: the audio data were transferred without problem to the cloud. It was then possible to test the Contribution analysis during this pilot test.

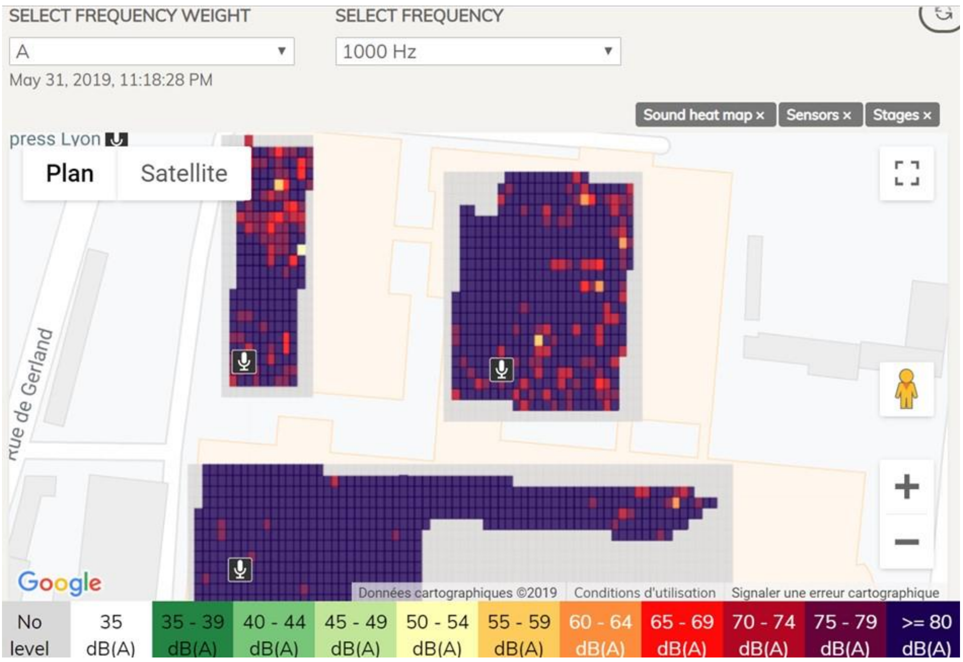


**Figure 32: Sound Heat Map example (Computed during Nuits Sonores 2019)**

### 6.1.2 Noise Annoyance Monitoring

The purpose of the Annoyance index is to give a more accurate estimate of noise annoyance, based on subjective perception data.

Acoucité investigated this topic further in 2018. This work allowed bettering defining the scope of Noise Annoyance Monitoring.

The analysis of acoustic measurements carried on outdoor festivals (Nuits Sonores, Kappa Futur Festival, Festival of Lights and Woodstower) showed that sound emissions at these events have a big amount of energy within the low-frequency range (i.e.: from 30 Hz to 250 Hz). In addition to that, results from annoyance survey carried out for the Kappa Futur Festival in July 2017 revealed that:

- Annoyance is associated with noise for more than half of the respondents.
- Low frequencies are predominant in the emergence of this discomfort.

Thus, both subjective and physical assessments conclude on the importance of low frequencies in the expressed noise annoyance of outdoor large scale events.

Following these observations, a review of the state-of-the-art was conducted mainly focused on annoyance and low-frequency noise. The main findings of this review are:

- Noise annoyance depends on multiple factors, the acoustical dimension being one of them.
- High inter-individual variability is observed on the expressed annoyance of subjects under the same acoustic stimulus.
- Assessment of annoyance based on acoustical measurement can bring to light the main tendencies of community annoyance, but it cannot be used as a predictor of individual annoyance.
- Conventional methods of assessing annoyance, typically based on A-weighted equivalent level, are inadequate for low-frequency noise and lead to incorrect decisions by regulatory authorities.
- Weighted level underestimates the effects of low-frequency noises.
- The annoyance of low frequencies increases rapidly with level.
- Low-frequency noise specific criteria have been introduced in some countries, but do not deal adequately with fluctuations.
- Loudness, and particularly loudness percentile N5 (loudness which is exceeded 5% of the time of observation) is preferred for describing time-varying sounds, take into account fluctuations.

To take into account the aspects previously described, it was decided to investigate loudness of sound recordings of festivals and its correlation with A-weighted and C-weighted sound pressure levels recorded every second. While A-weighted sound pressure levels showed a poor correlation with loudness, C-weighted correlates well with loudness ($R^2>0.9$). Even if loudness can provide a most precise description of the acoustical phenomenon in terms of sensorial response, C-weighted values are less time consuming and can be found on most of the sound level meters.

An annoyance index was thus proposed, built on three main rules:

- Easy to understand: linear scale from 0 to 10 (0: no annoyance; 10: maximum annoyance likelihood)
- Based on C-weighted sound pressure levels
- Comparison between sound levels with and without the event

The algorithm for the proposed annoyance index has been implemented in the SLM gateway. The proposed annoyance index was tested with measurement during the Nuits Sonores 2018, Kappa Futur and Woodstower festivals with coherent results. However, as noticed in the state-of-the-art, noise annoyance is highly subject to inter-individual variability, so the annoyance index could preferably be renamed as “Annoyance Likelihood Index”.

## 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 and D4.5, 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

The Sound monitoring system has been in use during different pilot tests in 2019: Rhein in Flammen, Kappa Futur Festival, Nuits Sonores, Woodstower, Fête des Lumières, Gadefest på Istedgade event (Copenhagen, for about 3 weeks continuous monitoring) and Roskilde SOUND2019. For most of them, the Acoustic Level monitoring (LAeq, LCeq, Spectra) was the primary purpose of the Noise Monitoring System. These levels were presented in the COP. A low number of data losses has been noticed during these events. One SLM is also continuously monitoring for the MOVIDA pilot test, since October 2018.

As mentioned above, the Contribution Analysis has been tested during the event SOUND2019 at Roskilde.

The setup for the Contribution Analysis is presented in Figure 33. As shown, the SLMs were doubled for each location: one was used for sound monitoring (so transferring low rate data: LAeq, LCeq, and spectra visible on the COP), and the other transferring time-stamped audio recordings for Contribution Analysis.

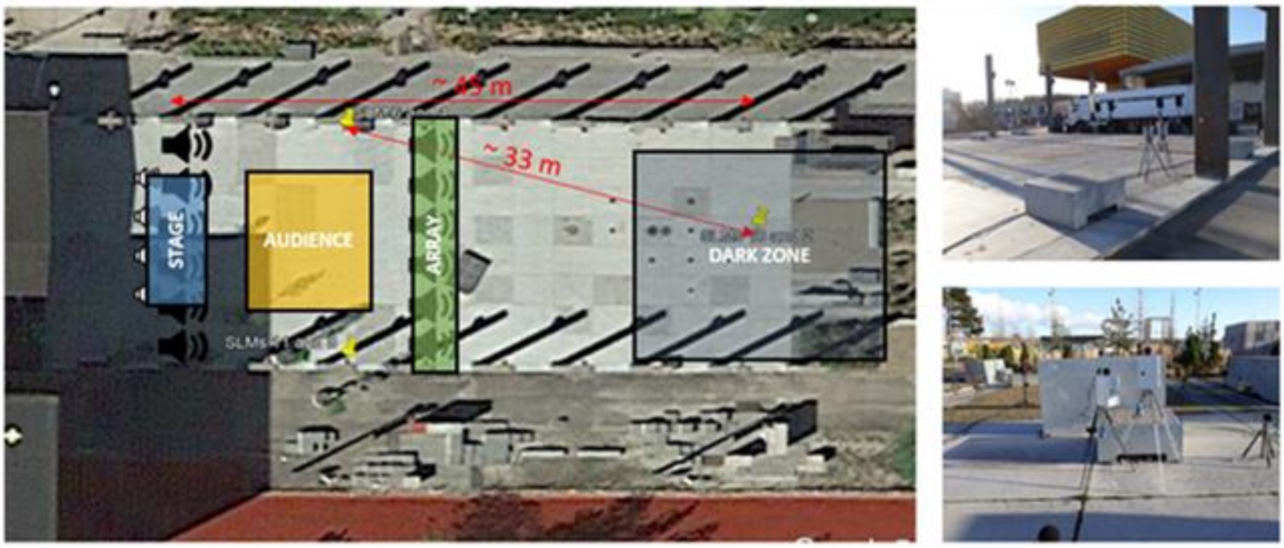


**Figure 33: The Reference SLM is located near the Audience, the remote SLM is placed in the Dark Zone. The top right image shows the reference SLM, Bottom right shows the remote SLM. The SLMs were doubled: one was used for only levels monitoring, the other for audio recordings.**

**Figure 34: Loudspeaker nearby the remote SLM, adding an acoustic contribution to the recording.**

To test the Contribution analysis, we have introduced different additional and significant acoustic contributions to the remote SLM, using a loudspeaker, very close to the sensor, as indicated in Figure 34. To check the validity of the Contribution analysis, a first recording is performed without added background noise: this define the reference levels ('true contribution'). In a second phase, a contribution is added (traffic noise), where the Contribution analysis is set up and tested. In a final phase, another contribution is added (traffic noise and human activities: loud talks), where the Contribution analysis is validated against the reference levels.

To check the validity of the Contribution Analysis, we compare the LAeq levels obtained with both contributions (from the venue and from the side loudspeaker), the LAeq levels from the reference case (side loudspeaker switched off), and the LAeq levels extracted from the Contribution Analysis (See Figure 35).

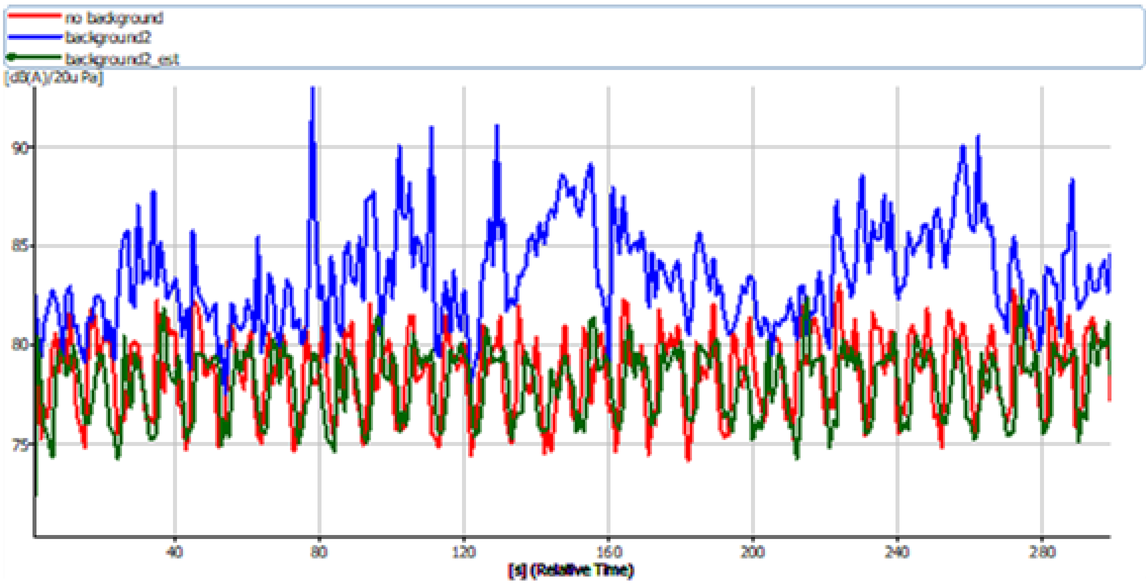


**Figure 35: Validation of Contribution analysis over a 5 minutes window. The blue curve shows the 1 second-LAeq levels as measured by the remote SLM, containing both contributions: from the venue and from the side loudspeaker. The red curve shows the reference LAeq levels, which represent the true contribution that we want to estimate. The green curve shows the LAeq level estimated by the Contribution analysis algorithm.**

The green curve, showing the Laeq levels estimated by the Contribution analysis algorithm, aligns pretty well with the true contribution shown as the red curve. This validates the Contribution analysis implementation.

# 7 Conclusions

From the ROFH 2019 results, we could see that the ASFC can run with few microphones and achieve good performance. The system can considerably increase the Insertion Loss in the dark zones in the neighbourhood up to 14dB.

Two adaptive algorithms were tested and both worked well. The minimum number of microphones required is closely related to the “openness” of the venue, and the venue of the ROFH2019 was quite open space but we are expecting that applying various strategies in predicting the accurate propagation model in space would delimit this “openness” constraint. The use of a few control microphones makes the ASFC more usable in many real outdoor events.

It is noteworthy that for the consecutive pilots (KFF2018 and KFF2019) held in the identical venue, the sound field control system achieved similar insertion losses but the effective IL was very limited only to very low frequencies below 63 Hz. We could see that environmental limitation and also the practical constraints on the measurement process in both events have greatly limited the performance of the ASFC. First, the dark zone in the churchyard was completely isolated from KFF2019 event area by a tram railway without electricity or any means of information connection (WiFi or cellular). Due to this environmental limitation, a simultaneous data recording/monitoring and synchronization (using a wired connection) which is required for the adaptive control was not possible. At the Roskilde SOUND2019, where simultaneous data recording/monitoring and synchronization is possible, the adaptive system worked well and achieved good performance.

Furthermore, we have shown that the local Quiet Zone system is capable to significantly reduce noise in a local area under real-world conditions. The latest Tivoli test has shown that insertion loss of greater 10dB could be achieved in the whole listening spectrum in the position of the control microphones. In the accessible quiet zone area we have also achieved an overall performance of greater 10dB but with a drop below 10dB in an octave band around ca. 130Hz. It is the area of transition from active control at low frequencies to passive control of high frequencies that is critical in the current configuration. The application of virtual microphone methods is recommended to move the region of optimal active control performance out of the physical control microphones into the accessible quiet zone. This way is it possible to close the gap in the insertion loss. If the stability of the active controller can still be guaranteed has to be evaluated in further studies.

The Noise Monitoring system has been in use during multiple pilot tests in 2019. Generally, the SLMs have performed very well in sending acoustic levels (LAeq, LCeq, and spectra) to the COP. The Contribution analysis has been also tested during the last year. The requirements for this part is demanding: synchronization between SLMs and high data rate requiring a high volume and stable communication network. This was obtained during the Roskilde SOUND2019 event, where the Contribution analysis was tested and successfully validated.

# 8 List of Figures and Tables

## 8.1 Figures

## 8.2 Tables

# 9 References

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