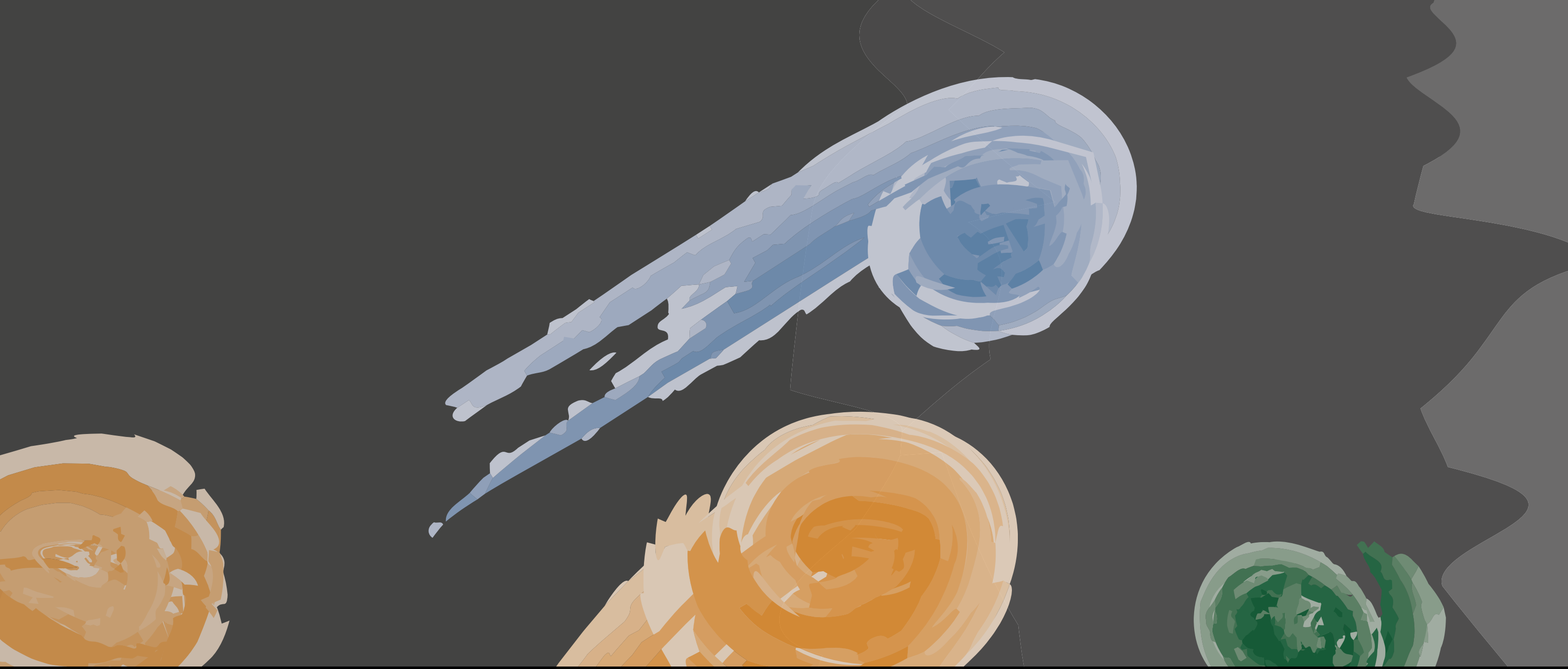


Concept development for an immersive audio mastering application.

Designing a tool for object-based audio mixing.

Master thesis of Stan Vugs
MSc. Integrated Product Design

TU Delft Delft University of Technology

d&b audiotechnik

COLOPHON

MASTER THESIS
Delft University of Technology
Faculty of Industrial Design Engineering
Master Integrated Product Design

d&b audiotechnik GmbH & Co. KG

AUTHOR
Stan J. M. Vugs
4359208

GRADUATION COMMITTEE

TU Delft	Chair	Dr. R. van Egmond
TU Delft	Mentor	Ir. M. Tassoul
d&b audiotechnik	Company mentor	Dipl. Ing. M. Renz

May 2020

PREFACE

Besides challenging, complex and educational, this project has especially been an incredibly inspiring one. To be able to work so very involved with an innovation of this magnitude has been a tremendous opportunity for me. The awareness of this, mainly came to me on the first day of the project, at which I experienced the possibilities of immersive audio.

It has served for me also as an introduction into the world of professional audio reproduction, one which has proved to have many facets. The vast amount of technical education that is required to be a part of the audio world does not always contribute to its appeal, rather more to the contrary. The hospitality, however, of its colleagues makes up for this deceitful image in abundance. At every turn in the project where I have sought out the help of experienced professionals, despite my novelty, I have received a warm welcome. Without their advice, guidance and knowledge, the outcomes of this project would not have been anywhere near their current state.

In particular I would like to thank my chair René van Egmond and mentor Marc Tassoul for their supportive, sharp and very suitable feedback at the appropriate moments during the project. My gratitude also goes out to Martin Renz, who has done similarly as my coaches from the university, but, even more important, supported me in applying my education in the context of one of the world's leading companies for audio equipment. This gratitude also reaches to all other colleagues at d&b audiotechnik with whom I have been fortunate enough to collaborate.

My further thanks goes out to all participants, my fellow students, of the idea generation at the Technical University of Delft. My deep appreciation goes to the participants of the observational studies and evaluation study from d&b audiotechnik, the Dutch National Opera & Ballet and KlevR sounddesign. At last I would like to thank everyone who has supported me in the writing of this thesis in any other way.

I reflect back on this project with much joy, and hope that you enjoy reading my thesis.

Stan

EXECUTIVE SUMMARY

Immersive audio is an upcoming innovation within the world of professional audio reinforcement. The amount of possibilities of enhancing the listening experience is enlarged drastically by setting up arrays of speakers in front of and around the audience. Technological advancements make it possible to render recorded audio channels for an immersive audio system during live performances. These advancements rely on spatial audio rendering techniques, which will be described partly in this thesis. The elaboration on these techniques serves to clarify how the created audio effects are generated.

Object-based audio (OBA) mixing allows the rendering of (pre)recorded audio channels for an immersive audio system. OBA mixing allows to be very close to the listening experience. In the project that is presented in this thesis, an in-depth look is taken at the use cases of immersive audio in the application of live performances.

The users of immersive audio systems, who carry the responsibility of developing and executing the content that is reproduced over such systems, are determined and involved in defining a number of common use cases. Through the ability of defining specific requirements, these use cases lay the foundation for the development of a concept of a mixing tool for immersive audio systems. These requirements are supported by a respective definition of quality.

Qualitative research is also done in terms of generating ideas. Furthermore, a look is taken at existing relevant software, with analogous functionalities.

The research is bundled in a concept for a mixing tool, which is worked out in a visual mock-up. This mock-up is used in an evaluation study, to create an insight into the desirability of the concept. The concept is also laid parallel to its requirements that have been set for it.

At last, elements of the concept, which are missing despite their relevance, are discussed. Further research is suggested for the development of these elements. Additionally, some use cases that arose during the project for the use of OBA mixing in the further future are presented.

CONTENTS

ASSIGNMENT

1.

1.1 | INTRODUCTION

1.1.1 | d&b audiotechnik
This project has been executed in collaboration with d&b audiotechnik GmbH & Co. KG. d&b audiotechnik is a manufacturer of high-end sound systems for professional sound reproduction. This entails that d&b audiotechnik produces loudspeakers and amplifiers, but also additional simulation tools and software modules. Their office is based in Backnang, Germany.

d&b audiotechnik has developed an immersive audio system, named Soundscape. The insights that are created in this project are aimed to guide the design of a mastering tool for immersive audio mixing in general. However, in this project, Soundscape will be used as the basis for the development of a concept of such a design.

The reason to use Soundscape as a basis is that immersive audio systems in their entirety can be rather complex. The usage of Soundscape's parameters enabled to concretely develop the concept. The concept that is presented in this report is not related to the software development of d&b audiotechnik.

Figure 1: an immersive loudspeaker setup with loudspeakers surrounding the audience, highlighted in orange.

1.1.2 | immersive audio
An immersive audio system, as Soundscape, differs from more traditional public address (PA) systems in the number of speaker positions that are being used. An array of speaker positions is set up in front, and possibly around the audience. Through acoustic effects that can be created with a high amount of speaker positions, the listener experiences the feeling of being immersed in the audio. Soundscape's immersive experience is facilitated by two different modules. Each of these modules relies on a different acoustical effect.

The first module of Soundscape is called En-Scene. It provides a realistic spatial localization of sound throughout and around the stage as well as the audience. In other words, sounds can be reproduced as if their source is positioned at any location on and around the stage and audience. These sounds can be moved around with a high resolution.

The second module of Soundscape is called En-Space. It allows the users to apply the acoustical signature of renowned performance venues to their own audio.

A deeper elaboration on the technical operation of these modules and the acoustical effects on which they rely is given later in chapter 3: Technical elaboration.

1.2 | PROJECT BRIEF

For every PA system, its operation relies on the one hand on how the loudspeakers are set up, and on the other hand on how the performance is mixed. What is completely new with the development of immersive audio systems is that these two aspects have become independent of each other. The DS-100, Soundscape's signal processor, enables this loudspeaker independence. It does this by live rendering up to 64 audio inputs, based upon object-based audio (OBA) mixing. In its rendering, it automatically processes the loudspeaker setup. Thus, where traditional audio-channel mixing always started with a consideration of the loudspeaker setup, OBA mixing is venue independent.

The relevant result of this independence is that mixing audio-channels for an immersive audio performance allows for a completely new workflow. This workflow starts with the consideration of the listening experience, bringing the mixer closer to his own end product.

To ultimately address this workflow, a new software tool is required for content creators to work on their audio-channel mix. Adjusting the spatial elements of a multi-channel production has been described as spatial mastering (Melchior, Michaelis, & Steffens, 2011). The brief for this project is to gain insights into the requirements of a tool for spatial mastering, and develop them into a prototype. The complete project brief which was signed at the start of this project can be found in appendix A: Project Brief.

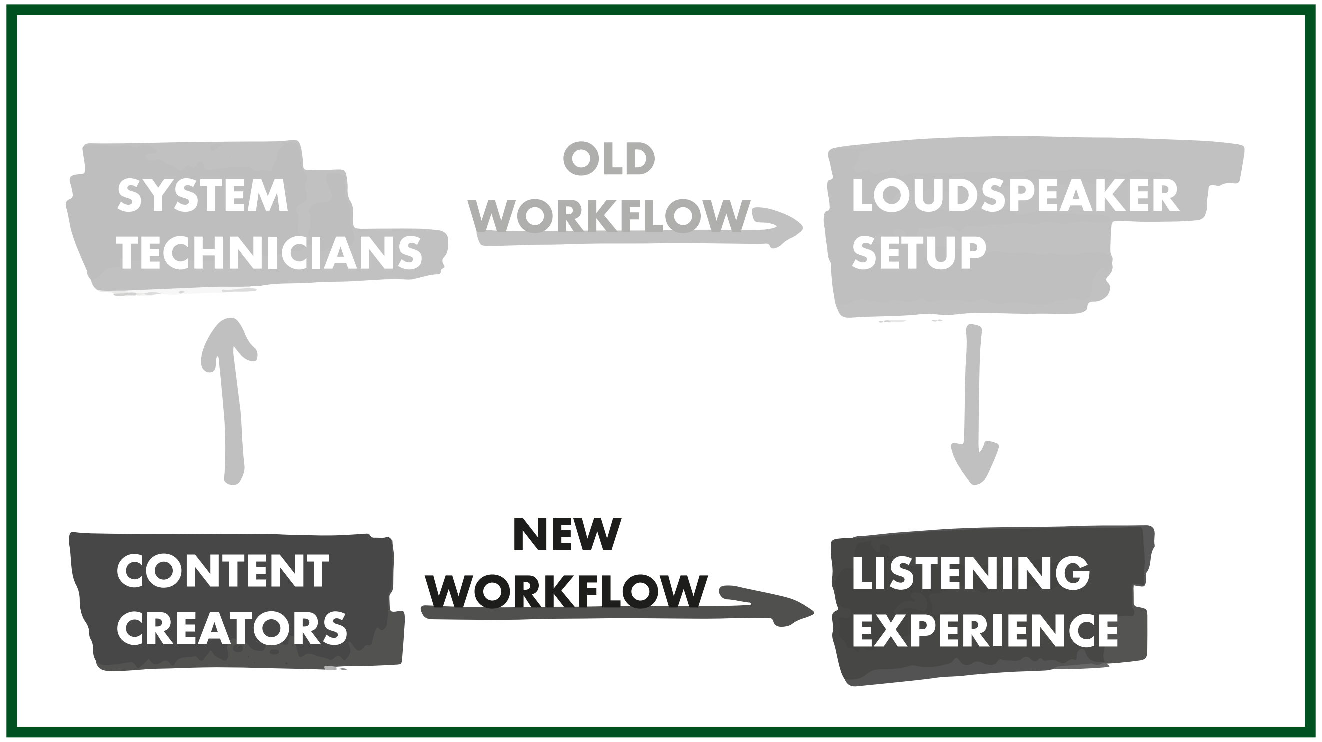


Figure 2: new workflow.

VISION & IMPACT

2.

2.1 | DESIGN VISION

During the project, a design vision has been developed in order to highlight important aspects of the concept. This was done to ensure a focus on these aspects throughout the duration of the concept development.

> **"A graphical software tool to control immersive audio mixing elements, that visualizes the listening experience and integrates into the content creator's workflow through adjustability of its controls and visualizations."**

2.1.1 | visualizing listening experience
The listening experience is at the core of the workflow. The focus on one side of the software tool is to provide its users with an insightful depiction of the actual listening experience that an immersive audio system offers. The basis of providing insight into the listening experience lies in an understandable visualization. The fact that the workflow allows the user to work on a project, without the access to a sound system, drastically increases the demand for an understandable visualization. On the other side of the software tool are its controls over the listening experience. Immersive audio systems rely on a complex technical operation. This operation is elaborated upon in chapter 3: Technical elaboration. The parameters that are provided to control Soundscape are directly related to this complex mechanism. Direct visualizations that relate to the listening experience when these parameters are adjusted, will make the controls understandable.

2.1.2 | adjustable controls and visualizations
As the scope of use cases is rather broad, the relevance of the envisioned tool's functions will differ. This also applies to specific visualization of the listening experience. To ensure that the visualization as well as the controls are kept clear, their presence should be adjustable. As some users, in various phases of their work, might need specific functions, adjustments to the presented controls should be available. The same reasoning works for visualizations of the listening experience. These adjustments will be completely separate from the user's work. Templates of controls can be defined so that the tool integrates well into different phases of the workflow of audio mixing.

2.2 | IMPACT

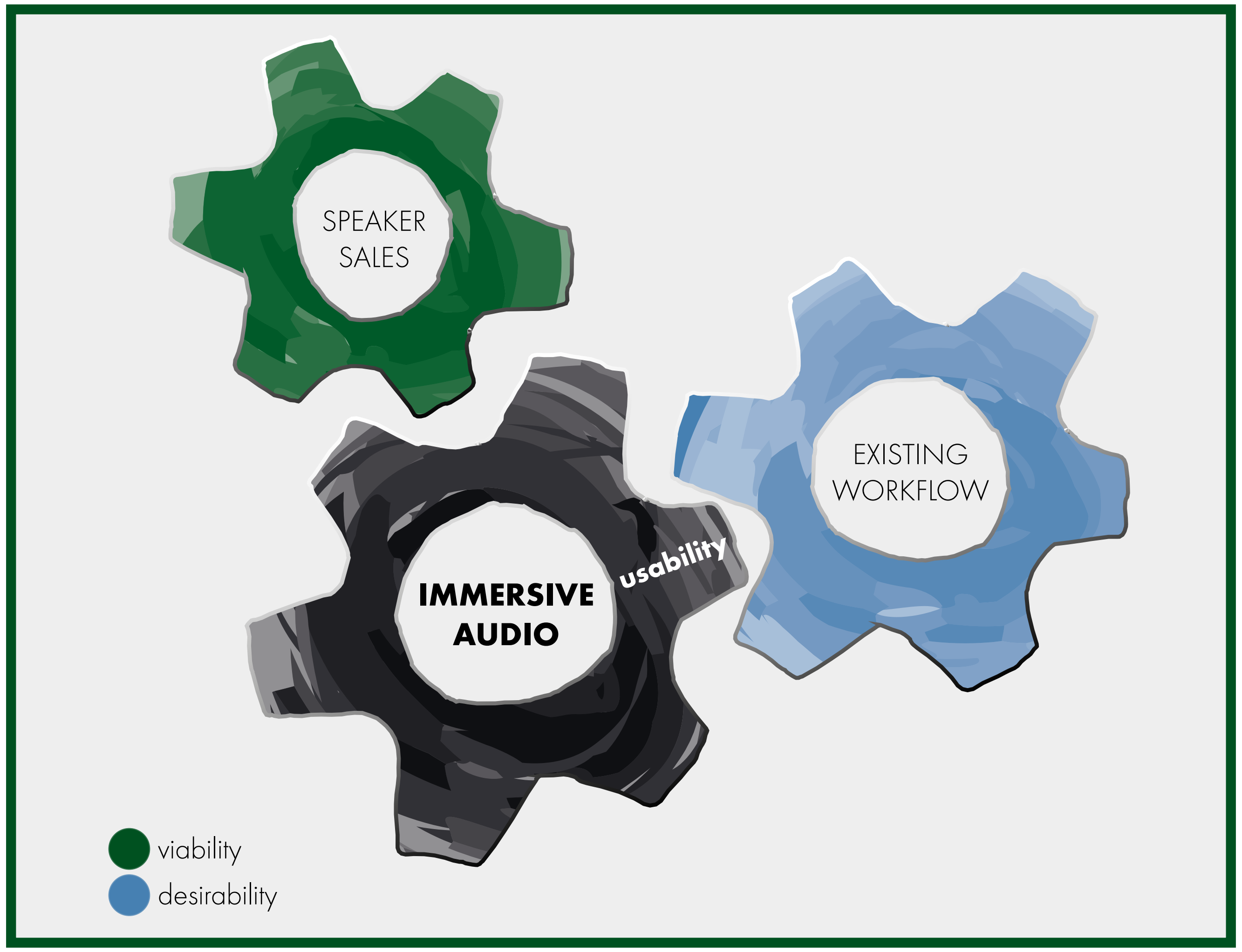


Figure 3: the importance of usability in relation to viability and desirability.

2.2.1 | viability
The viability of an immersive audio system relies on the sale of its loudspeakers, as immersive audio relies on the usage of a larger number of speakers. The number of speakers increases most when speakers are also positioned around the audience. Innovations are made in many different facets of immersive audio system, which have the common drive to develop immersive audio into the industry standard for live-sound reproduction (Naef, Staadt, & Gross, 2002). Some of these innovations aid immersive audio in enhancing the listening experience.

The development of the concept that is presented in this report, is part of this network of innovations in a different way. It aims to aid immersive audio by increasing the usability of its functions.

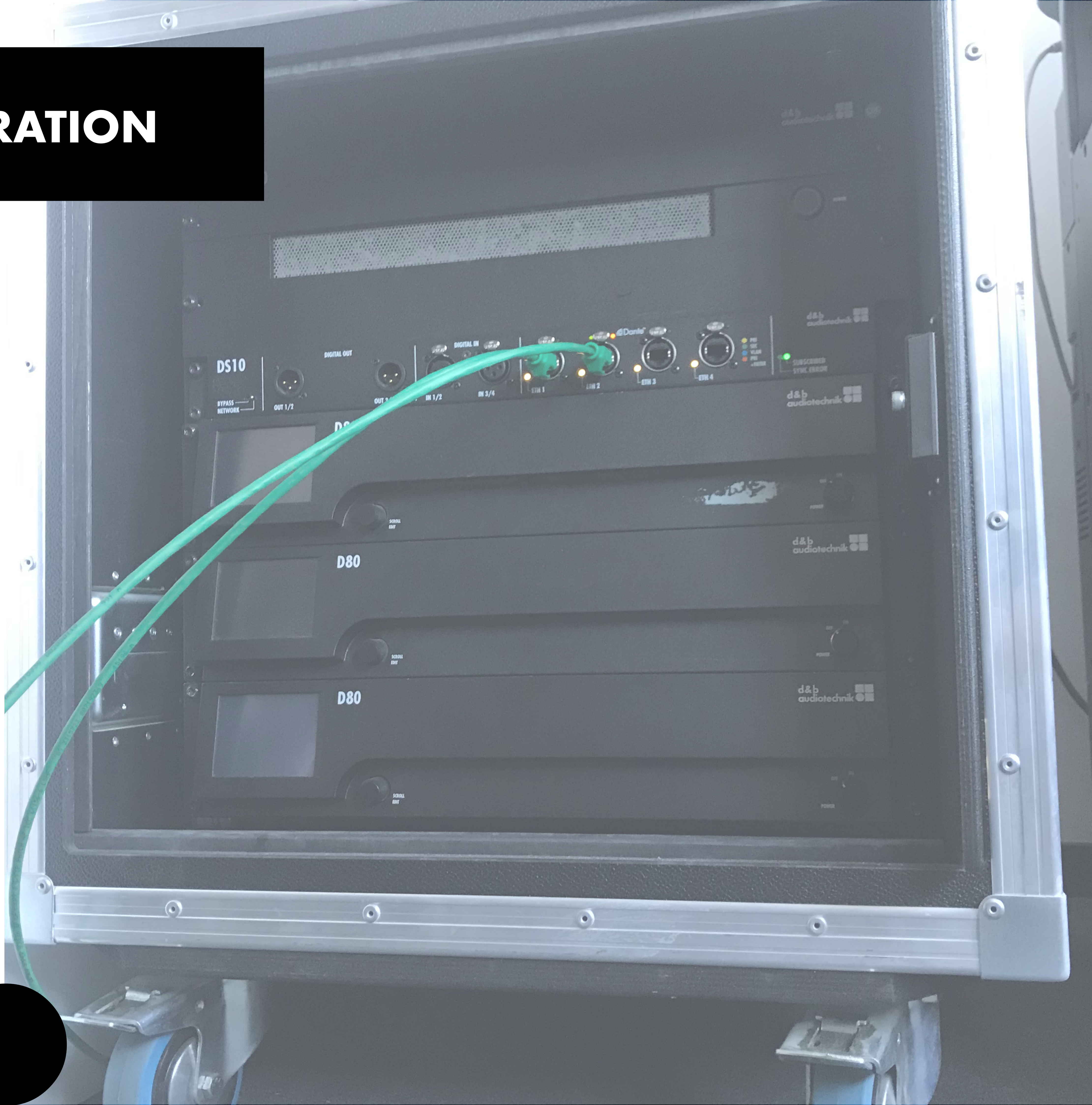
2.2.2 | desirability
The two aspects that are stated in the design vision both target the usability of immersive audio systems. The integration into an existing workflow and a more comprehensible depiction of the listening experience both increase the ease of working with such a system.

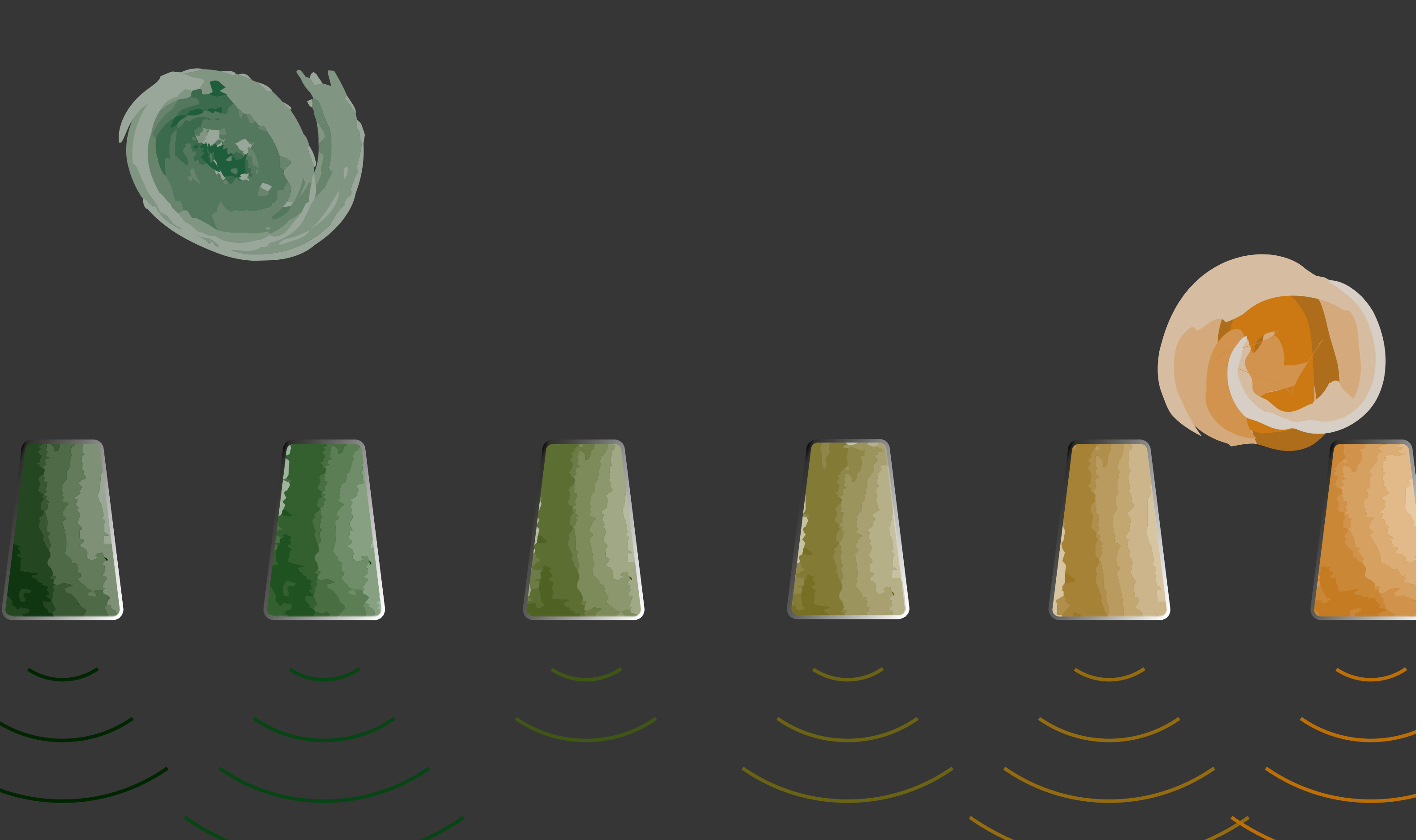
The importance of this usability lies in the desirability of the whole immersive audio system. And the desirability of the whole system is as vital to the network of developments as the increase in loudspeaker sales. Without useable functionalities, immersive audio is not desirable, and thus nor viable. This is depicted in figure 3.

The feasibility of this concept is addressed in the technical elaboration in chapter 3 and the elaboration of the concept in chapter 7.

TECHNICAL ELABORATION

3.





3.1 | EN-SCENE

The En-Scene module of Soundscape allows audio-channels to be given positional metadata. Metadata is data that tells something about other data. In this case the metadata consists of the positions of a sound source, which is the original data. En-Scene then uses this metadata to calculate how the audio is distributed over the loudspeaker setup. For each loudspeaker, a difference in loudness (level) and time (delay) is calculated. Through this dispersion, the listener is able to recognize a simulated location of the auditory event's origin (Blauert, 1997). The ability to recognize a sound's origin is called spatial localization.

Audio-channels can be placed within and around the stage and audience with En-Scene.

In this chapter, a detailed description is given of the immersive audio system produced by d&b audiotechnik, Soundscape (d&b audiotechnik, 2019). The technical operation of Soundscape will serve as a concrete basis for the concept which is developed during this project. The focus of this elaboration is laid upon how the listening experience is realized by acoustic effects, as this knowledge is relevant for the concept development. As the loudspeaker setup is not relevant for the use of the concept, the details on the loudspeaker setup of immersive audio are not presented.

The developments of this concept are in no way related to the software development at d&b audiotechnik. The Soundscape system only servers as a basis to allow an easier understanding of how a concept for mastering immersive audio can be realized.

3.1.1 | wave field synthesis

Simulated spatial localization is realized by Wave Field Synthesis (WFS) (Berkhout, Vries, & Vogel, 1993). With this technique, sound waves at any desired location are recreated with a number of speakers at fixed positions. The sound source which is simulated to be placed at the desired location is called the phantom source. The operation of WFS relies on the precedence effect.

The precedence effect describes that when two sounds are separated shortly after another, within their echo threshold, the sound is perceived as one auditory event. Dependent on the amount of delay, a human will localize the auditory event somewhere in a space. The maximum delay between two arriving sounds depends on the auditory event's echo threshold. Complex sounds, as speech, (up to 40 ms) allow for a larger delay than a tight sound, as a click (1 to 5 ms) (1949, Wallach et al.).

The applied time difference is derived from the distance from the auditory event to the loudspeaker, as well as the speed of sound. The closer to a loudspeaker, the smaller the delay. In figure 4, the greenest loudspeakers have the least amount of delay.

The listener's position affects which loudspeaker he will hear first. But independent of the listener's position, this loudspeaker will be between the phantom source's location and the listener. This allows for spatial localization to be in effect correctly for every position. It also allows for the object's position to be independent of the loudspeaker's position.

In addition to a time difference, a relative difference in loudness is applied to the reproduced sound for each loudspeaker. Speakers with the least delay, will produce the most loudness. This is done to increase the accuracy of the spatial localization. If sound sources are placed on the front side of the loudspeaker, only a difference in loudness is used between the loudspeakers. This is done, because the delay would have to be below zero in this case. A playback of audio before it is recorded is impossible.

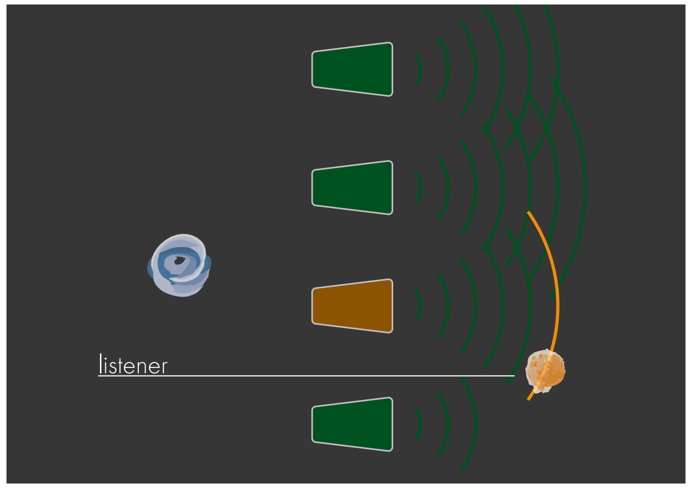
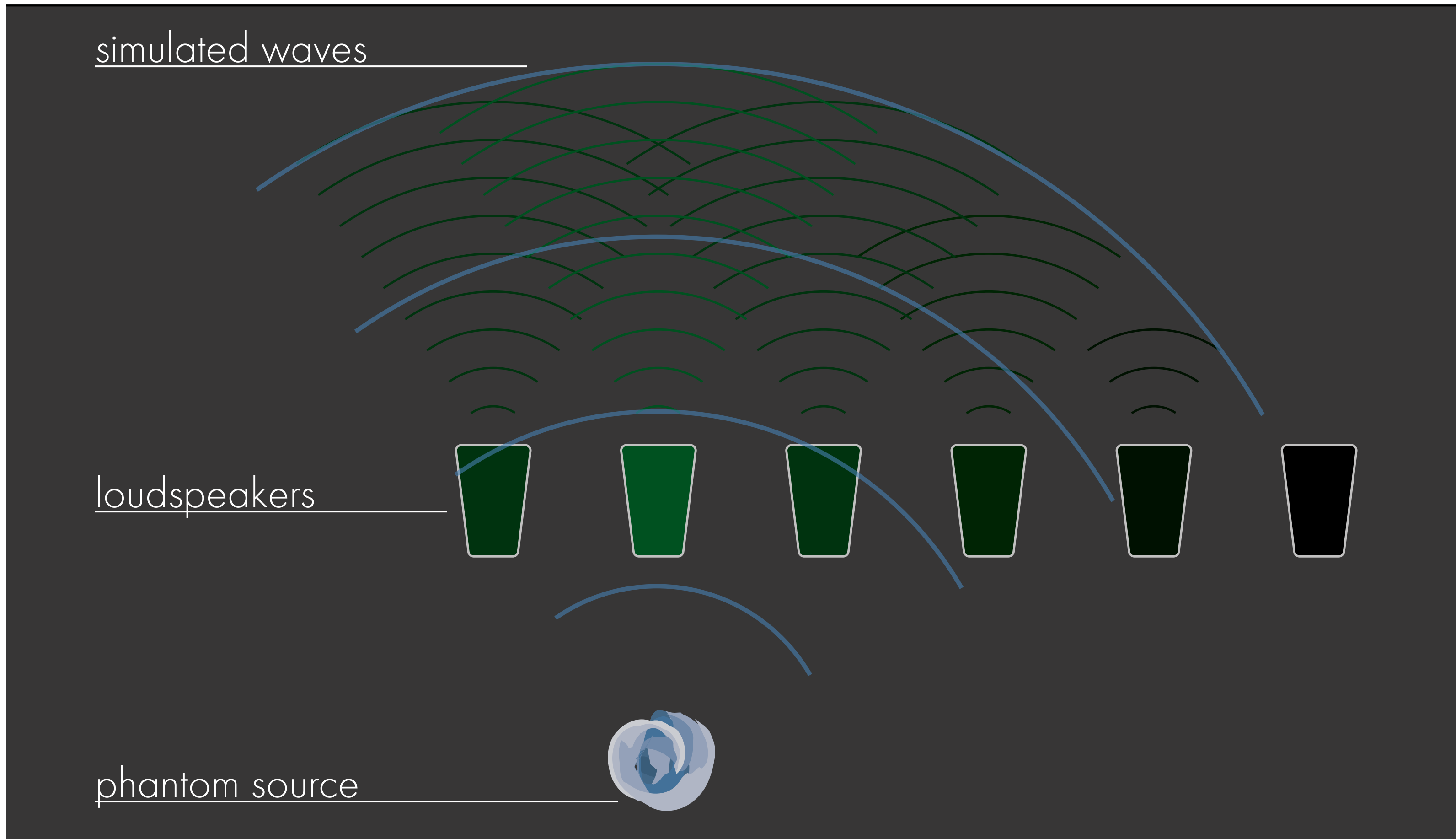


Figure 5: the first heard loudspeaker depends on the listener position.

Figure 4: wave field synthesis with a phantom source behind the loudspeakers.



3.1.2 | OBA mixing

To be able to control the positional meta-data of sound sources, Soundscape's signal processor relies on object-based audio mixing. When a sound source is given this positional data, it becomes a sound object. The position of the sound object is used to determine the sound's phantom location.

Sound objects also have a few more relevant properties, which determine their reproduction, than merely their position.

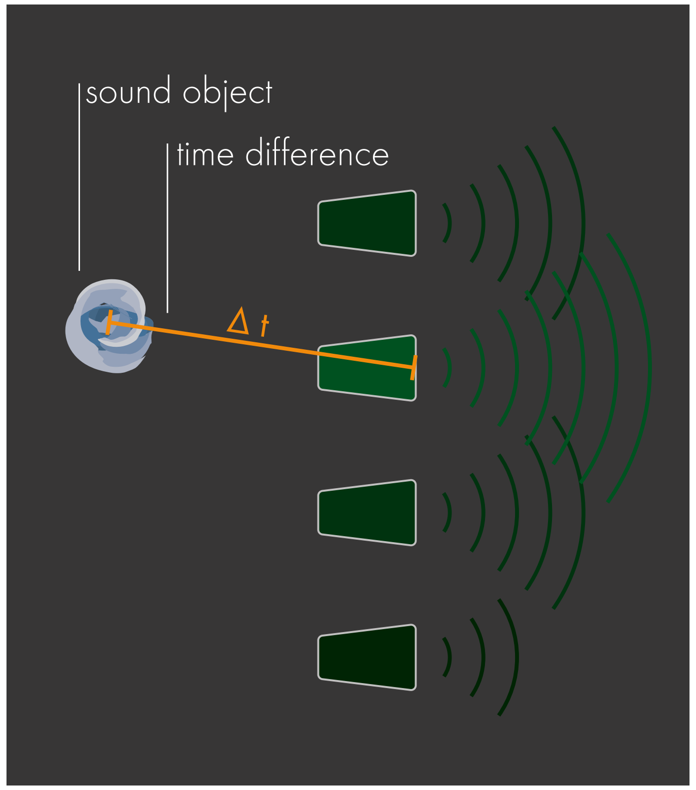


Figure 6: subtracted time difference for delay mode Tight.

3.1.3 | delay modes

For each object, different delay modes are available. The delay mode of an object defines what kind of time difference is applied to create a phantom source. An object's delay mode can be set to Off, Full and Tight.

When the delay mode is set to Full, a difference in loudness and in time is applied to a reproduction. For each loudspeaker, a delay of the sound reproduction is calculated and applied. This is useful for creating precise spatial localization and can be used for acoustic and amplified instruments.

When the delay mode is set to Off, only a difference in loudness is applied to a sound's reproduction. This means that all speakers reproduce the sound simultaneously, which results in a less precise spatial localization. It can be useful for fast moving objects, which would otherwise create signal artefacts: when objects move at very high speed, while having a certain delay for to create spatial localization, a distortion in the sound may occur.

When the delay mode is set to Tight, the loudness difference, once again, stays the same. The time difference is minimized, by subtracting the time difference between the object and the nearest loudspeaker. The relative delay between loudspeakers is kept, so that the spatial localization is unaffected. Tight mode is beneficial for the reproduction of electronic instruments and/or pre-recorded material, so that delays between channels are kept minimal. This is useful in these cases as the mix of these kind of sounds allows for less relative delay between them.

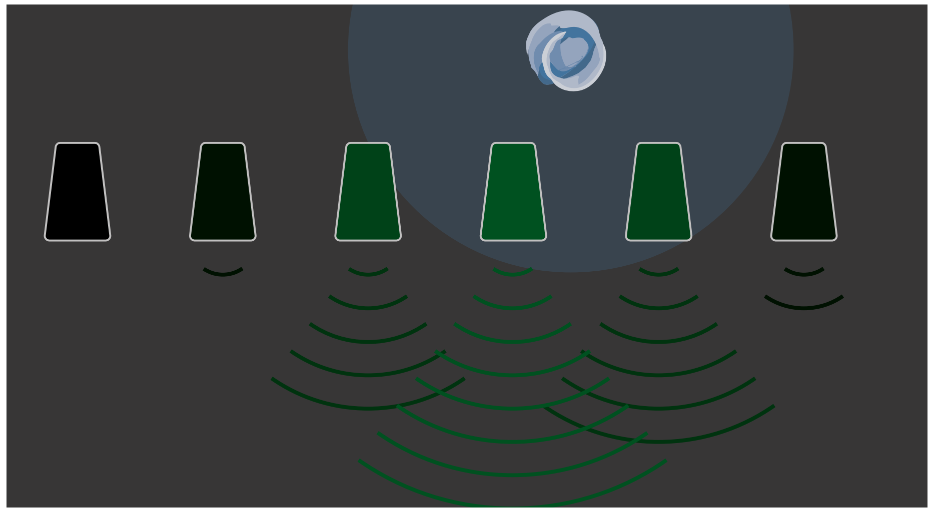


Figure 7: effect of a sound object's spread.

3.1.4 | spread

The spread of an object defines how focussed its sound is reproduced. A focussed object is reproduced by only the closest loudspeaker. A wide object is reproduced by more loudspeakers in the area. A focussed object delivers a sharp image, while a wide object delivers a more even coverage of the sound divided over the speakers. The spread does not influence the sum of the loudness of a sound object.

3.1.5 | 180 & 360

An important distinction between two modes within En-Scene is made. When sound objects only have to be placed on a stage, behind one array of loudspeakers, a 180 setup will suffice. Only the orange speakers in figure 8 are then placed.

When sound objects have to be placed in other locations, such as around and behind the audience, a 360 system is required. In this setup, the audience is surrounded by loudspeakers. Only the 360 setup allows En-Space virtual acoustics.

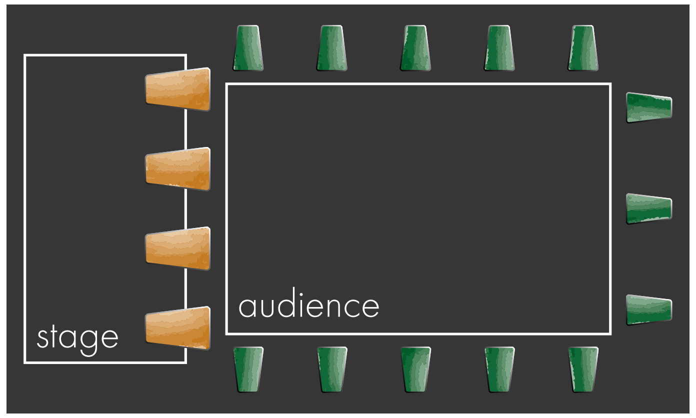
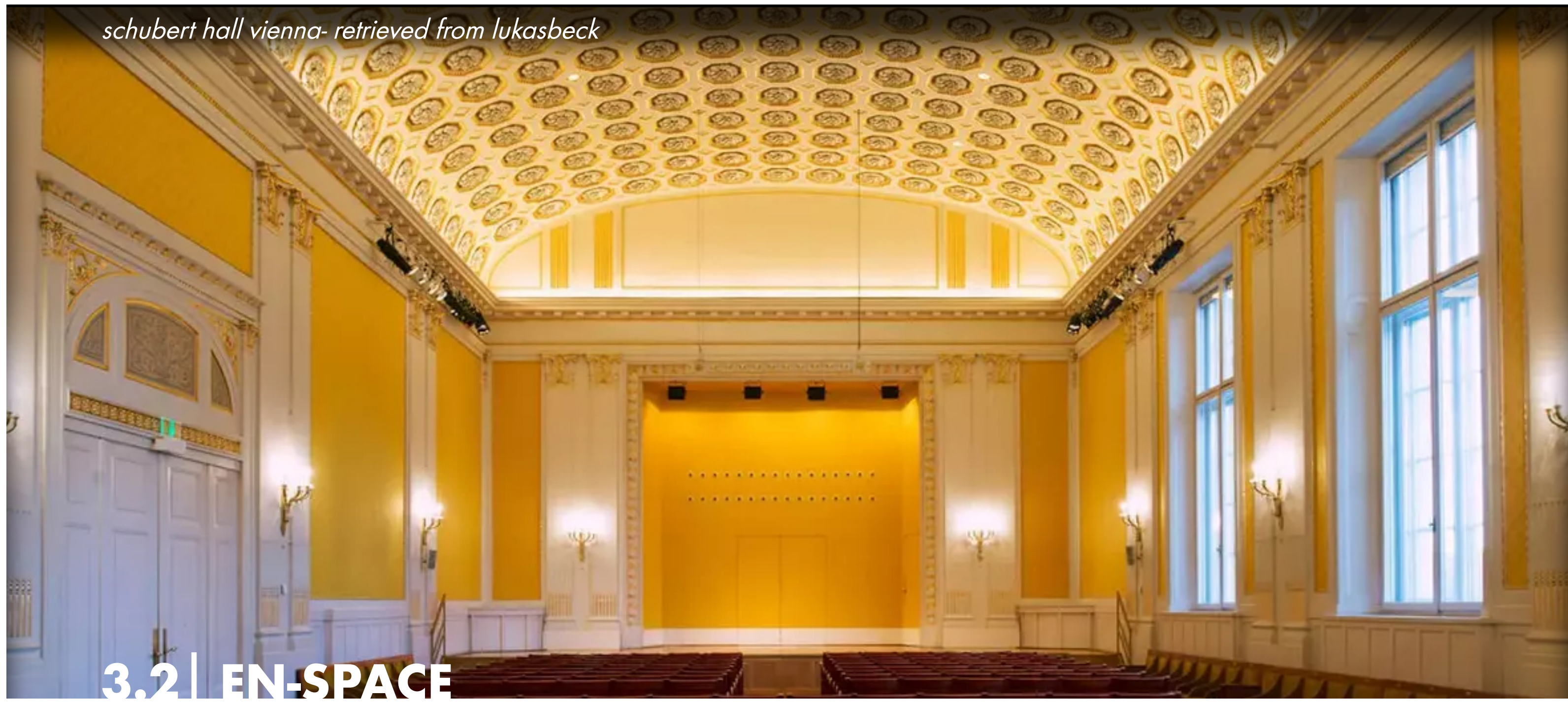


Figure 8: a 360 loudspeaker setup around the audience.



3.2 | EN-SPACE

The En-Space module of Soundscape adds the acoustical reverberant characteristics of pre-recorded sampled concert halls to reproduced sounds. Hereby it emulates the original acoustical space with virtual acoustics. It contains a set of seven sampled concert halls with different reverberant attributes.

The En-Space module only operates with a 360 Soundscape system, as the reverberation effects rely on surrounding speakers to match wall reflections.

3.2.1 | acoustic space

An acoustic space is the environment in which sounds are heard, which unavoidably affects the listening experience. These effects of the acoustic space are built by reverberations. A reverberation is the effect that occurs when a sound is reflected (multiple times) by surfaces and objects in the space (Roeser, Valente, & Hosford-Dunn, 2007). The accumulation of all reverberations in a room result in the room's acoustics. Surfaces of the acoustic space influence the amount of reflection and absorption of a reverberation.

3.2.2 | boundary plane emulation
En-Space virtually recreates the acoustical effects of seven sampled spaces. These samples are built up by a set of 144 boundary plane responses from 64 locations within the sampled space.

A boundary response sample contains the information of how sound is reverberated at one location. It is sampled with a known sound. Derived from this sample, is the acoustical responses at one boundary position. An example of a boundary plane response sample is shown in the figure 9 in which for each frequency, a reverberation time is given. The reverberation times shown in the graph are for a decay of 40 dB (RT40) (Mulcahy, 2019).

These locations are located at the walls of the venue and in front of the stage, which are the same locations at which the (up to) 64 speakers will be placed. Therefore, this method of sound field reproduction ensures the highest accuracy.

The sound field that is generated by all boundary plane responses of a space contains the **signature** that is unique to each acoustic space.

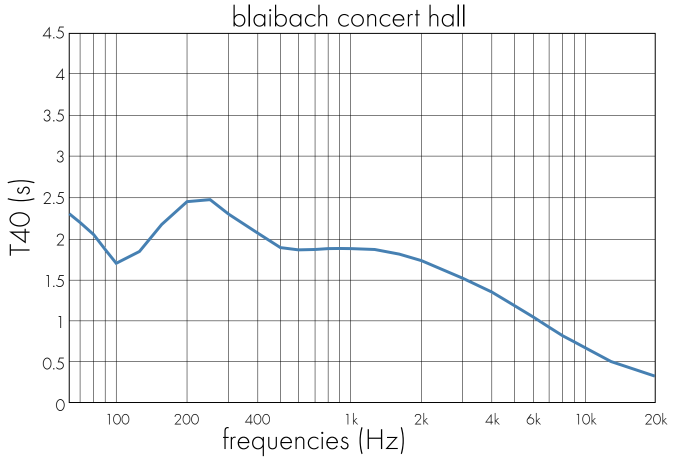


Figure 9: reverberation time plot of blaibach concert hall - original retrieved from d&b audiotechnik.

3.2.3 | loudspeaker independence
The sound field is generated by a setup with 64 positions. However, each speaker is provided with an individual boundary response, independent of the venue's size or shape. The sound field is automatically mapped to the actual loudspeakers position.

The reproduced acoustical response is always added to the reverberation of the actual space. This limits the audible effect when a sampled space is not much larger than the actual venue.

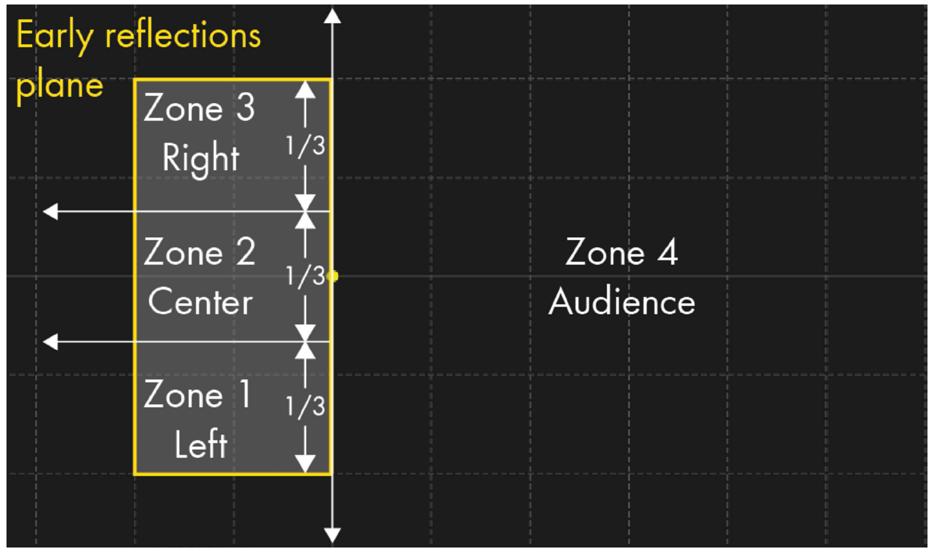
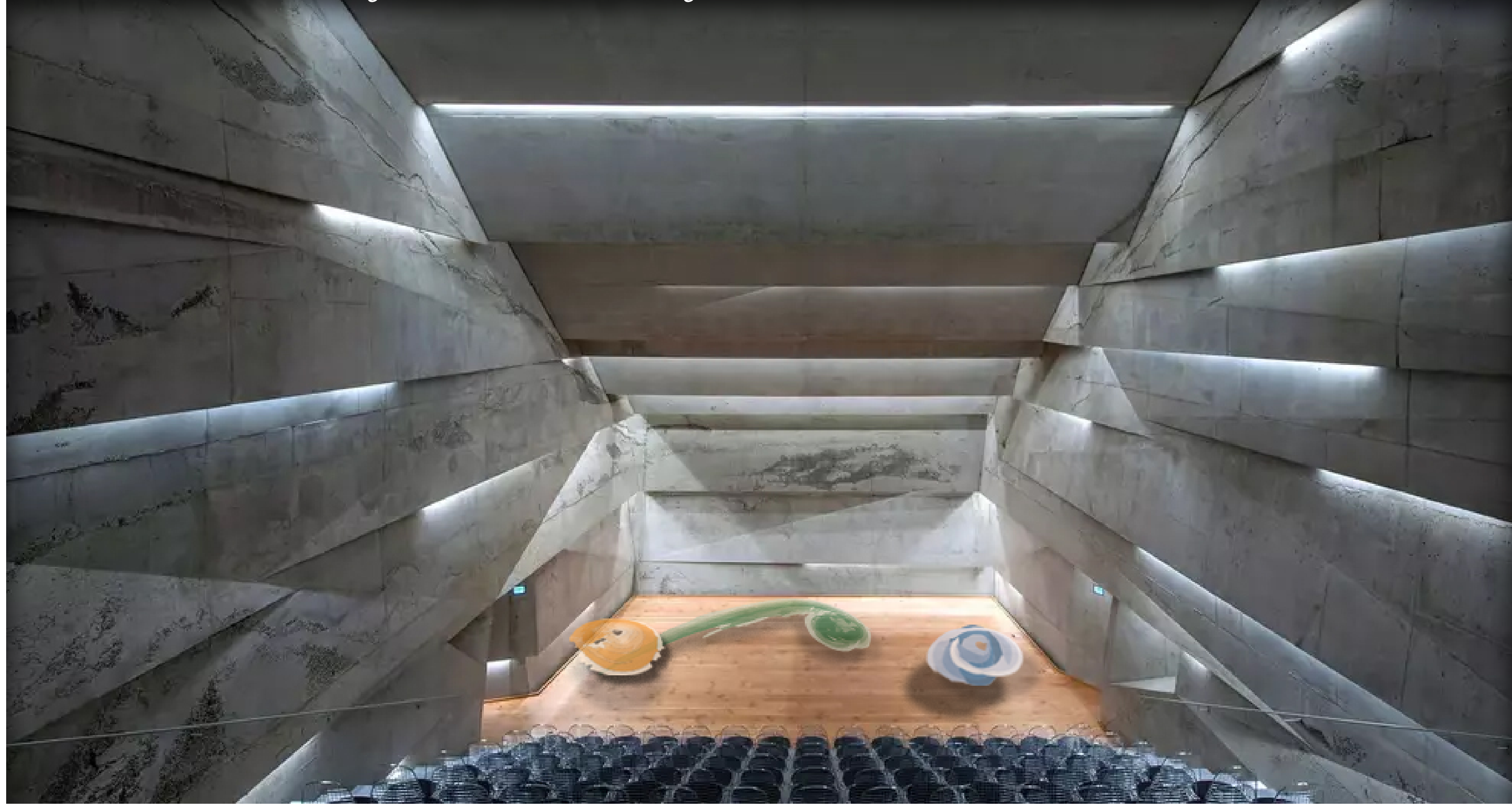


Figure 10: early refection zones - retrieved from d&b audiotechnik.

3.2.4 | early reflection zones
The venue in which the sound field is reproduced is divided into 4 zones. The first three zones are within the early reflections plane. This is often the stage. The early reflection plane is equally split into zone 1 to zone 3 from left to right.

For the loudspeaker positions, there are different boundary responses for the 4 different zones. The reverberation at a certain loudspeaker position will differ if the source is located in a different zone.

So, for each sound, its zone has to be specified. When En-Scene is used in parallel to En-space, the zone mixing is performed automatically dependant of the object positioning.



blaibach concert hall - original retrieved from HeidelbergCement

3.2.5 | pre-delay
Independent of the selected sampled space, its response can be modified to better match the loudspeaker setup. The first parameter to alter the virtual acoustics is its pre-delay.

The pre-delay factor adds a delay to all the reverberation responses. This changes the listening experience such that it **scales** the reverberation of the sampled venue.

Reverberation is reproduced earlier or later compared to the initial mapping of the sound field. This parameter also makes use of the effect that it takes time for sound to travel from its source to the listener.

When an actual venue is larger than a sampled space, the system avoids that reverberation is produced earlier than the direct sound.

3.2.6 | rear level
The generated sound field is often mostly balanced towards the front of the venue, namely the stage. As this is the location where sounds are produced, this will also be the location where most reverberations take place.

A distribution with relatively louder reverberations in the back of the venue can be desirable in different cases. The **balance** between the loudness of the reverberations in the front and the back of the venue can be altered with En-Space's rear level. The balance of the level of the reverberations can be moved towards the back of the room.

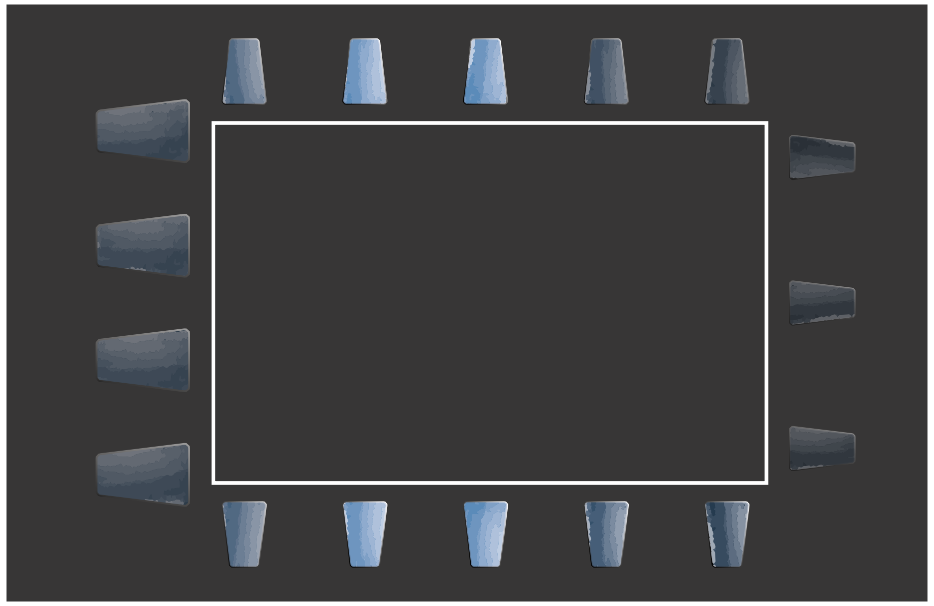


Figure 11: the balance of the reverberation adjusted by the rear level.

3.2.7 | en-space send

The reverberation only comes forth from the original sound sources. For each sound source, the amount of reverberation can be set individually. For this amount, each sound source has the En-Space send value.

If En-Scene is used concurrently to En-Space, sound objects will have the En-Space send value.

3.3| DS100

The DS100 is the signal engine on which Soundscape relies. It is a 64 x 64 digital audio channel matrix. The in- and outputs provide signal processing and matrix cross points control level and delay. It is equipped with a Dante interface. The DS100 is the host for the En-Scene and/or En-space modules, which direct all related matrix functions.

For the DS100 to be able to operate, additional software is used.

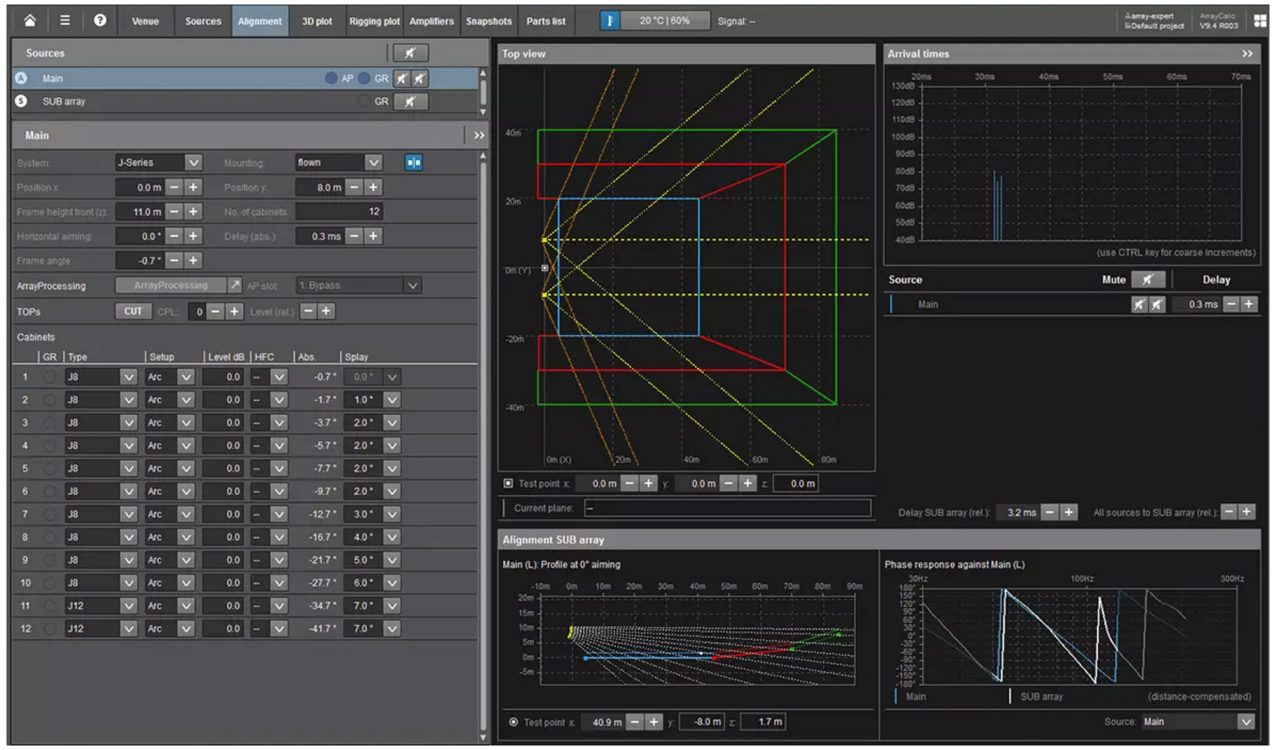


Figure 12: drawing the loudspeaker setup in ArrayCalc - retrieved from d&b audiotechnik.

3.3.1 | ArrayCalc

ArrayCalc is the software by d&b audiotechnik in which loudspeaker setups are drawn. This creates an image of which loudspeaker is at which position. The audience and stage plane are also defined in this software. This data is used by the DS100 to let the En-Scene and En-Space module work properly.

3.3.2 | R1

R1 is the software that is used to control the loudspeakers in d&b systems. It is designed for traditional sound mixing, which is dependent of the loudspeaker setup.

It is currently also used in order to control the mixing elements of a Soundscape setup, as the En-Space library and the En-Space object properties, including positions. For the use with Soundscape, it entails a positional overview with the ability to place and move sound objects.

3.4| PROCESS FLOW

In the following block diagram is depicted how audio channels are processed within the DS100. The audio channels move from left to right, and they are altered by the En-Scene and En-Space modules. Four sorts of inputs are defined, which control the DS100.

The loudspeaker setup is drawn in ArrayCalc, and is depicted in green. The other inputs, namely object positions, manual control and the spaces library is currently controlled with R1. However, the concept that is presented in this report is envisioned to take over two of these inputs. The object positioning and the spaces library will be controlled with the envisioned concept and are depicted in blue. The manual control over the level and delay matrix will not be integrated into the envisioned application and only remain controllable in R1. This is depicted in orange.

- concept GUI
- R1
- ArrayCalc

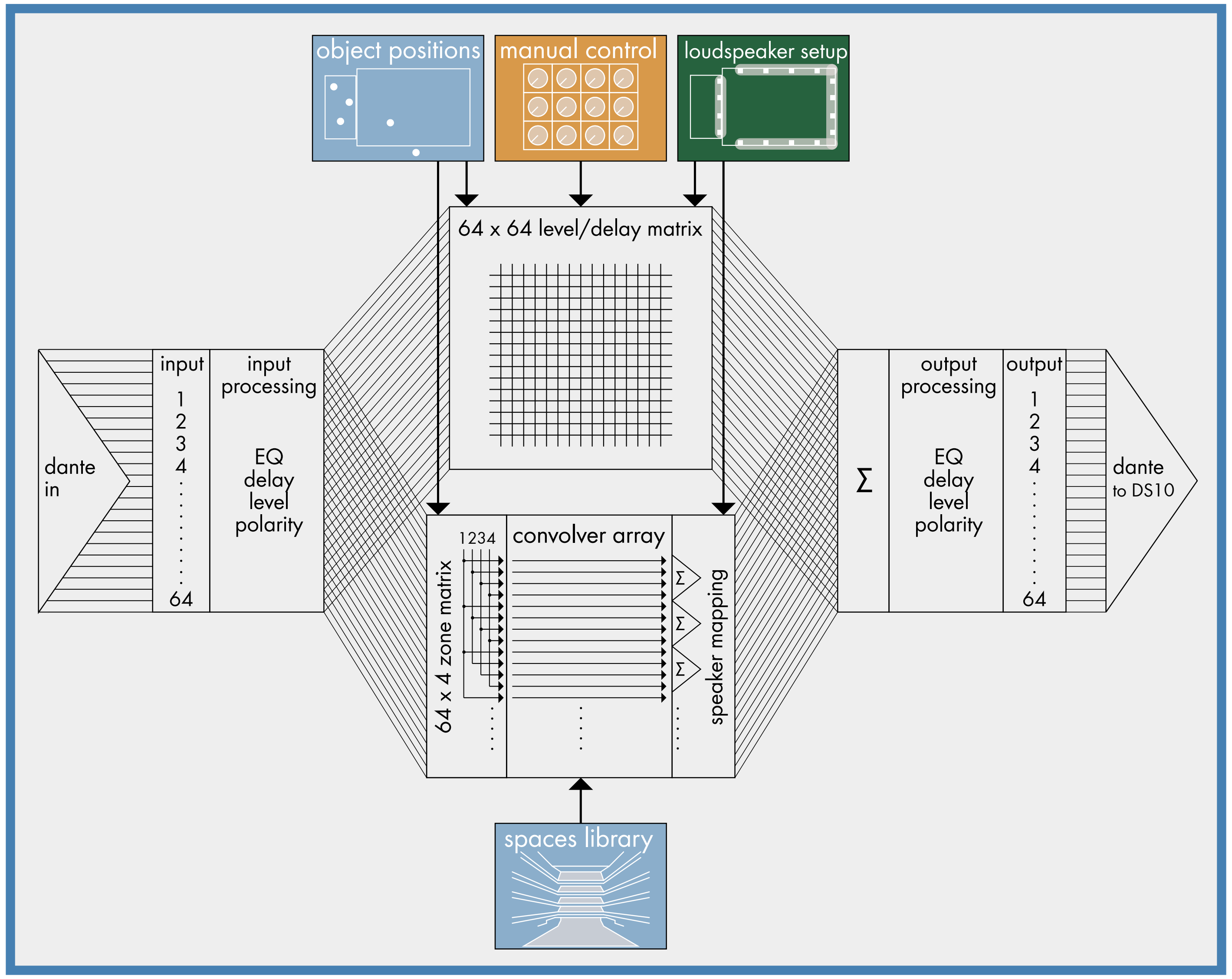
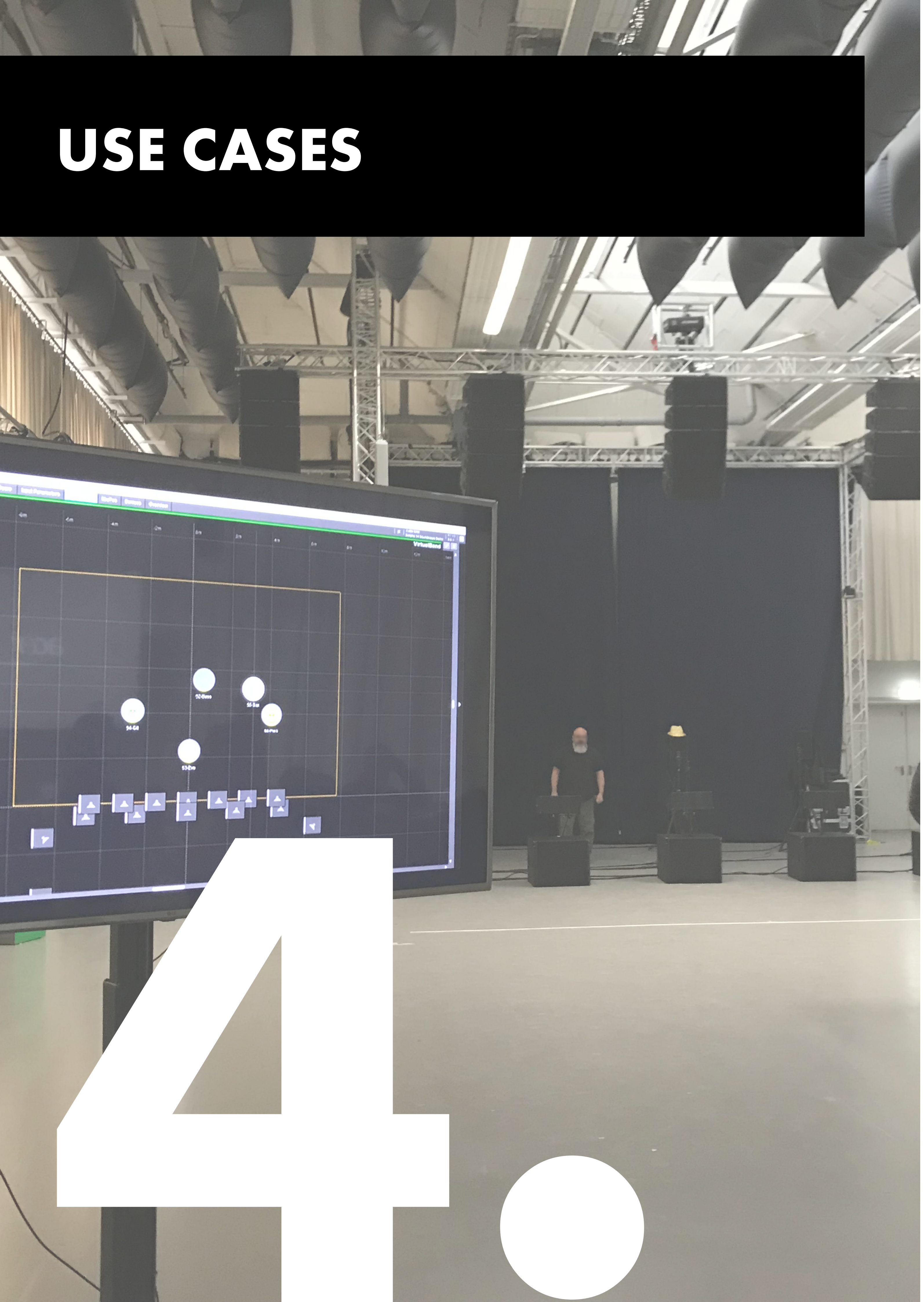


Figure 13: process flow within the DS100 - original retrieved from d&b audiotechnik.

USE CASES



Through the techniques and tools that were presented in the previous chapter, immersive audio makes an introduction into live applications.

The exact focus within live applications, which is chosen for in this project, is elaborated upon in this chapter. A few examples of these applications are given in the embodiment of three use cases. These use cases serve to substantiate design choices later on in the process.

4.1 | APPLICATION FOCUS

A large variety of live performances can have a benefit from immersive audio reinforcement. Mastering immersive elements for audio reinforcement is dependant on multiple facets of the scenario.

4.1.1 | installed

A clear distinction is found in scenarios in which immersive audio systems are being used. On one hand, an immersive audio system can be installed for permanent use in a certain venue, such as a theatre. In other cases, the audio system is build up for one performance specifically. This is done in locations where there is not a permanent use for an immersive audio system.

4.1.2 | genres

The focus of this project does not go specifically into one genre of music. This means that the varying considerations for mixing of different genres will be explored and taken into account into the concept development. In this project, the distinction between genres is generalized to whether the genre is more classical or more modern.

4.1.3 | dynamic and static

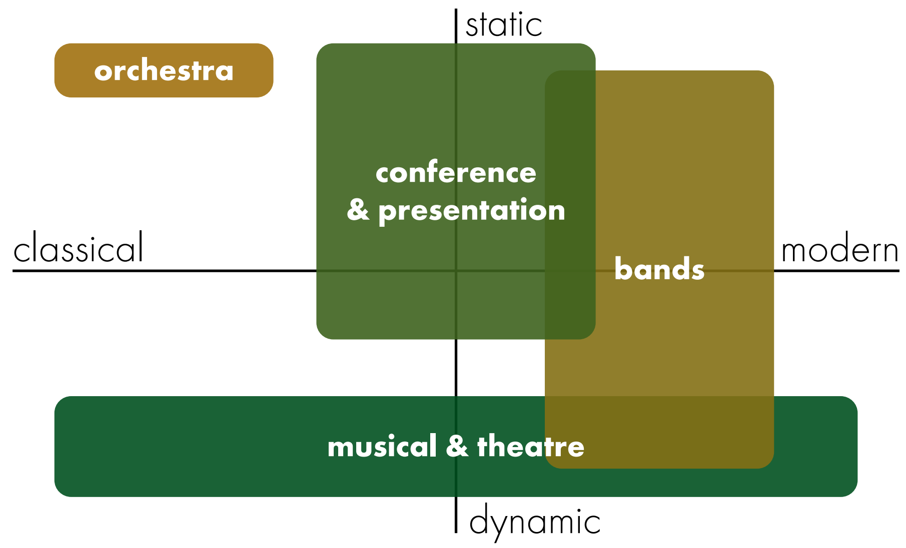
The complexity of mastering the immersive elements of a performance is partly dependant on the amount of moving sound sources. As more movements are required, more movements have to be programmed and monitored by an audio engineer. Examples with moving sound sources are present for both ends of the genre spectrum. Scenarios such as a theatre play or a musical can have multiple moving actors on a stage, which can be accompanied by stationary instruments. These scenarios make for the most complex use cases, and thus have the highest requirements for a concept to master immersive audio elements.

The reinforcement of an orchestra requires much less dynamic elements. An orchestra is usually set up before the performance and remains in position during the performance.

Performances with dynamic as well as static elements fall under the focus of this project. As performances with many dynamic sources are more complex, these scenarios are more useful for evaluating the concept.

4.1.4 | DJ performances

To narrow the focus of the project, the performances of diskjockeys (DJs) have been taken out of the focus. For performances of a DJ, sound sources often do not have an actual position on stage. Therefore, it was expected that these scenarios require different movements compared to reinforced live instruments.



In figure 14 a few scenarios are depicted depending on their genre and amount of dynamic elements. It shows that for use cases within one sort of scenario, different properties may apply.

Figure 14: scenarios mapped on their genre and amount of dynamic sources.

4.2| USE CASES

To capture the application focus of the project in a concrete way, three use cases have been created. The utility of these use cases is also to substantiate design choices for the concept development.

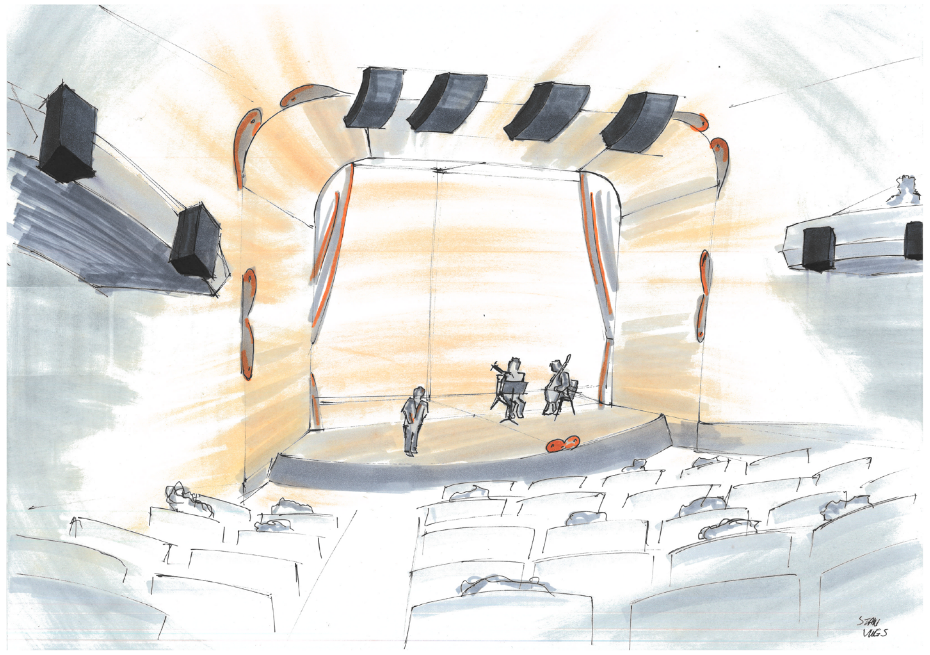


Figure 15: use case: musical theatre.

4.2.1 | musical theatre

Theatrical pieces are performed in a classical venue. The immersive audio system is installed for permanent use in the venue.

Actors move on stage along a script. Their voices are accompanied by live classical music. As actors also speak quietly, everything is reinforced over the PA system.

The reinforcement of the music strives to provide the audience with the most natural sound of the musicians. The sound should be as close to what is actually produced.

The venue is quite small and thus room emulation is often used to add more reverberation to the reinforced sound.

4.2.2 | touring band

For the presentation of their new album, a band has decided to use an immersive audio system with their tour. Their crew brings along and builds the loudspeaker system in all places where they perform. The setup may differ each performance, as it depends on the possibilities that the venue offers.

They have worked out the immersive effects together with their Front of House technician in a studio beforehand.

As the movements of the lead singer are not fixed, they decided to use a tracker which monitors his location on stage. They also use prerecorded material in their sound.

The mix of the performance strives to sound similar to how their new album sounds, with the addition of the immersive effects.

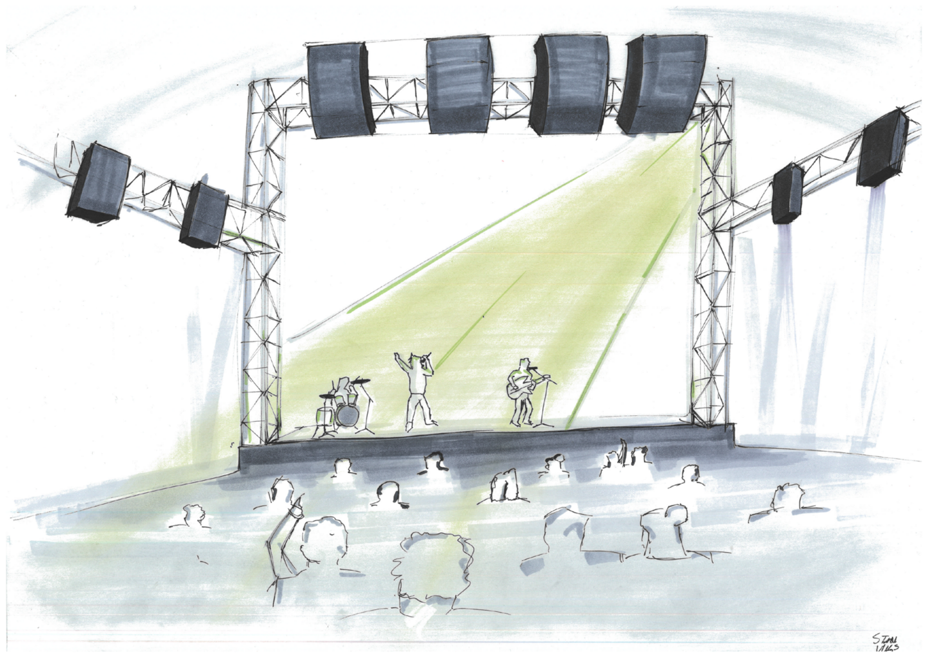


Figure 16: use case: touring band.

4.2.3 | corporate presentation

For the presentation of a new product, a company has hired a venue with an installed immersive audio system. They produce their show without a studio, and only are able to try out their production once they enter the rented venue.

They use immersive effects in the sound of their presentational video and music during the presentation.

They also make use of the lighting system which is integrated into the venue. The lighting effects are synchronized with the show. The video is the leading element in the presentation.

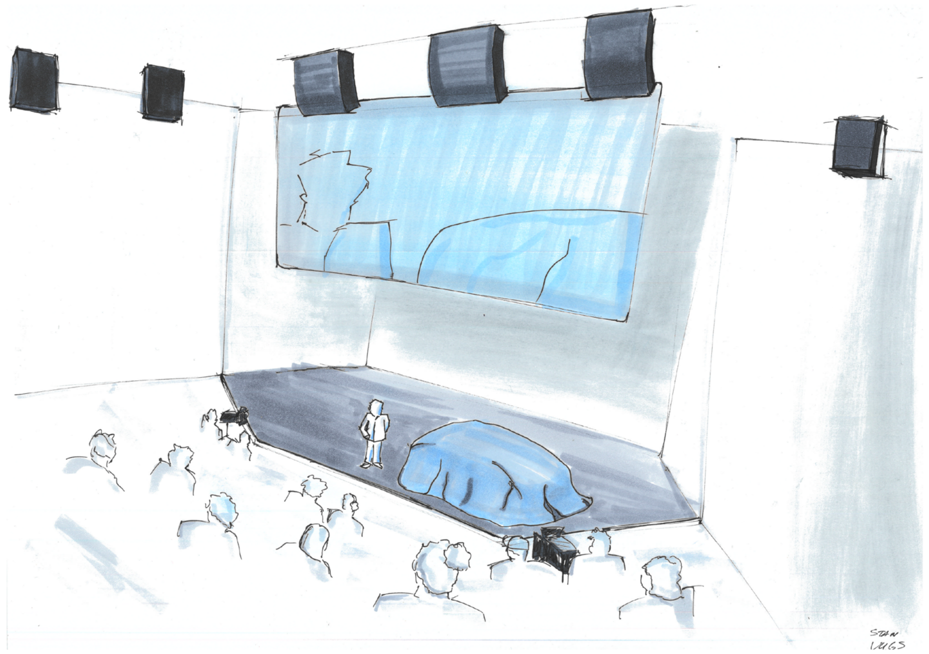


Figure 17: use case: corporate presentation.

4.3| CONTENT DESIGNERS

The targeted user group within the project has been defined as content designers. This definition targets all people who carry a responsibility in the production and execution of the immersive audio elements in a mix. A clear exclusion is made of people who carry a responsibility over setting up the immersive audio system.

Within the group of content designer are still many various users. To gain insight into the different perspectives within the group of content designers, a set of involved personas has been defined.

An important note is that not in every scenario, all personas are embodied by one individual. The presence of these personas during a real scenario is dependant of the performance. In some cases, multiple personas can be embodied by one individual. In other cases, some personas may not be present at all.

4.3.1 | learning curve

For each persona within the group of content designer, it is assumed that their job requires quite a lot of experience. This entails that they are to some degree aware of the technical operation of an immersive audio system. This also allows for the concept to have some room for a learning curve. People with more experience in operating similar software, will require less time to overcome such a learning curve (Carver, Shull, & Basili, 2006).

Experimental artist: Gives the performance, but also has an interest in how immersive effects are used in his performance. Has a desire to controls various immersive elements during the performance. This is mostly done with a custom 3rd party device.

Front of House (FoH) technician: Besides the immersive effects, he is responsible for all other elements that make up the mix for the audience. He monitors and controls all aspects of this mix. His job is mostly done during the performance.

Monitor technician: He creates a mix for artists and performers on stage. The purpose of this mix is to give each performer a sound so that he is able to play well.

Show control operator: His job is to ensure that all aspects of the show are running correctly and simultaneously. Therefore, he can use timecode or cues during the show.

Sound designer: The loudspeaker setup is his responsibility. If necessary, he closes the bridge between the loudspeaker setup and the content.

Studio engineer: Preconfigures a mix in a studio. He controls all elements of the mix, including the immersive elements. He has more time than a FoH technician, as his work is not done during the performance.

QUALITATIVE RESEARCH

5.

In this chapter the research that has been done, before the concept development started, is presented. Relevant software is discussed and compared to derive takeaways from existing techniques. An idea generation has been conducted to gain additional insights into the specific topic of animating data before a performance. And several observation studies are conducted to gain an insight into the use contexts and the users.

These insights are processed into a set of personas and use cases which serve to substantiate design choices of the concept development.

5.1 | RELEVANT EXISTING SOFTWARE

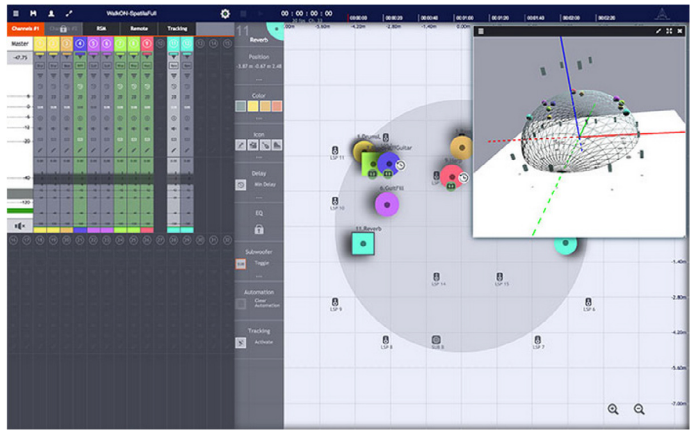


Figure 18: astroaudio - retrieved from Astroaudio.

Figure 19: wave performer - retrieved from Sonic Emotion.

A broad look was taken at software with comparable functions to gain inspiration for the concept development.

5.1.1 | object-based audio software

Existing software tools for object-based audio mixing were found and showed that the control of object positions happens in a 2-dimensional overview.

A 3-dimensional view is plotted in which objects move. In some applications, the objects are unable to be moved in a third dimension. In other programmes, the objects can also move in a third dimension, namely the height (Dolby Laboratories, 2013). This additional dimension would only change the reproduced sound if the loudspeaker setup includes speakers at different heights. As Dolby Atmos is used for audio production in movie theatres, the placement of sound objects in height is very valuable. Immersive audio systems for live audio reinforcement often do not have loudspeakers placed at different heights.

Preprogrammed animations of values, including positions are possible in existing software (Astro Spatial Audio, 2019).

The research also showed that other programs have the possibility to group objects. The last function that was found was the possibility to route sources to objects (Sonic Emotion, 2015).

5.1.2 | adobe creative suite

Inspiration was taken from two programs of the Adobe Creative Suite. Adobe Illustrator is a vector-based drawing program for 2-dimensional illustrations. It contains many functionalities for transforming, scaling, rotating and drawing in a 2-dimensional plane. Some of these functionalities can prove useful for object-based audio mixing (Adobe Systems, 2020).

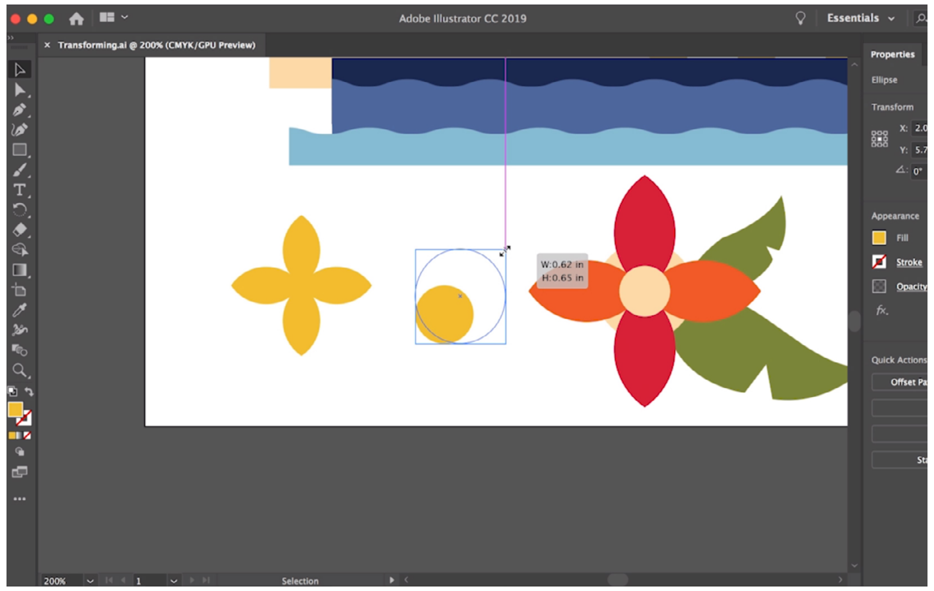


Figure 20: adobe illustrator - retrieved from Adobe.

The other inspiration of the Adobe Creative Suite is After Effects. This is a software in which animations are made. The animation in Adobe After Effects are displayed upon a timeline. The software's tools to alter the animation on the timeline, and the timeline itself can be translated for the automation functionality of the concept (Adobe Systems, 2020).

5.1.3 | cue control

When the use of cues came up in the project, Qlab was studied. This is a program in which the user is given easy control during a performance by having all elements setup in cues that merely have to be started at the right moment (Qlab, 2019).

Before the performance, the cues can be programmed with in-depth functionalities.

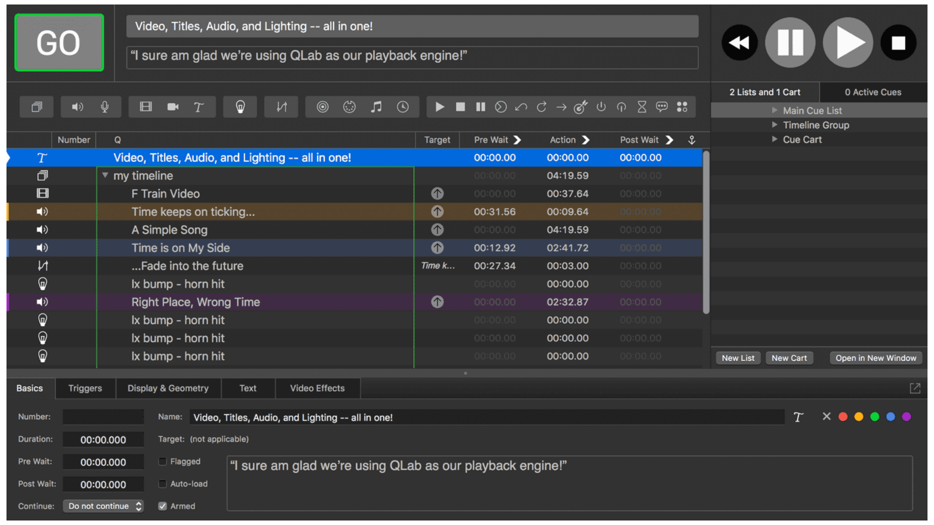


Figure 21: qlab- retrieved from Qlab.

A look was also taken at programs for sequencing show elements and lighting.

A complete overview of all insights is summarized in appendix B: Comparable Software.

5.1.4 | takeaways

The research into relevant software that is already being used has several takeaways. Programs that are used for object-based audio mixing showed multiple important functionalities that should be considered.

The Adobe Creative Suite gives an insights in how to give the user of a digital application a clear graphical overview and the ability to alter that overview, even with the dimension of time through a timeline.

The research into Qlab was very valuable for presenting the functionalities that are important for controlling cues.

It applies to all programs that they show what the existing method of working is for professionals. It was seen as an important step to take these existing methods into account. Introducing new ways of operating a digital interface also requires the 'unlearning' of the previous way, which may lead to failure of the concept (Becker, 2010).

5.2| IDEA GENERATION

In order to address the wide variety of possibilities for the automation functions, a brainstorm session has been conducted.

The complete description of the setup and execution of the idea generation can be found in appendix C1: Idea generation - setup and execution.

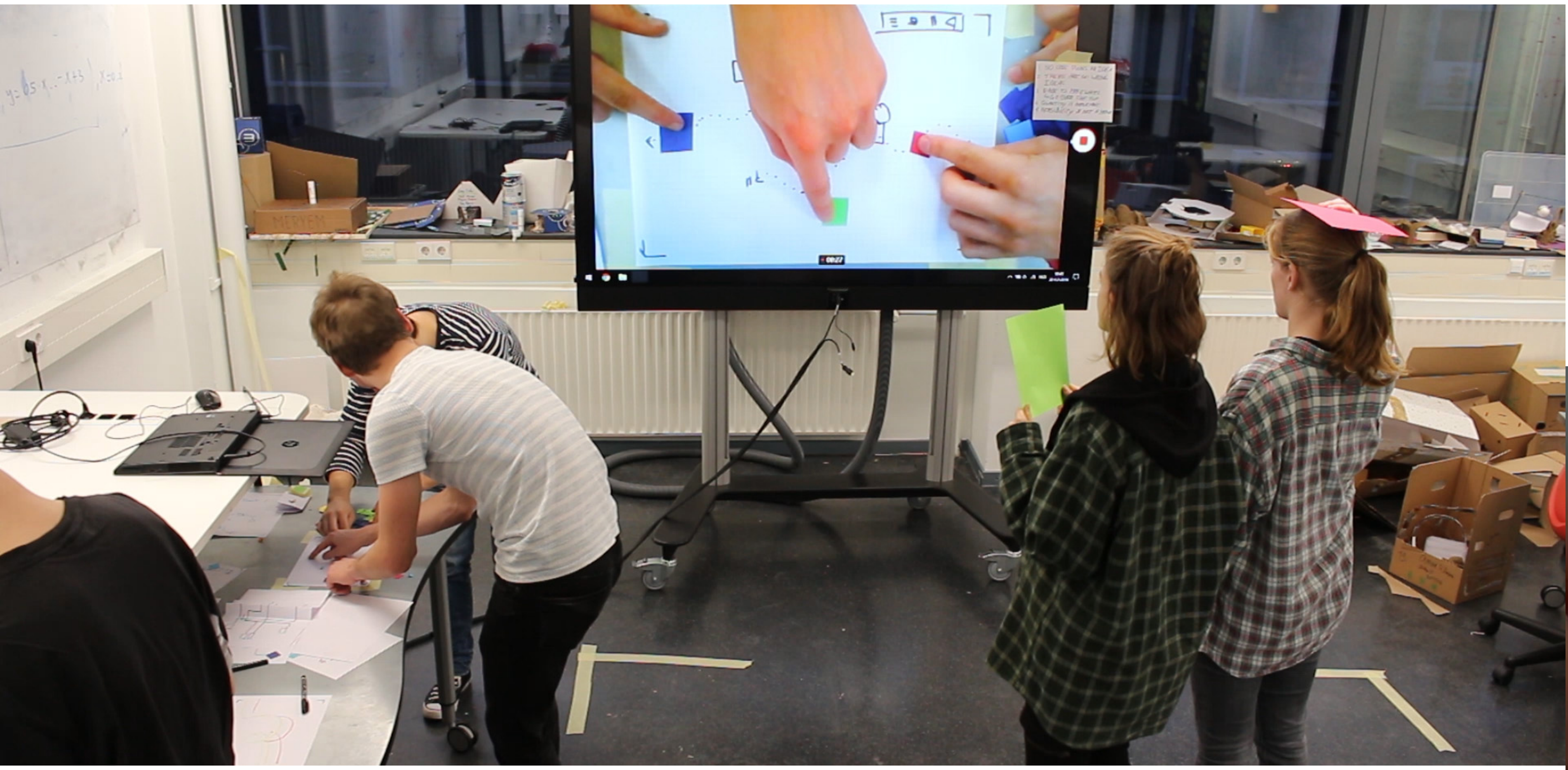


Figure 22: idea generation session.

5.2.1 | setup

Basic rules to ensure an open environment for creative problem solving were explained and kept present during the idea generation (Tassoul, 2009).

The setup of the brainstorm was made so that the generated ideas are within the domain of graphical user interfaces. The ideas had to be drawn on A4 papers, which could be placed under a camera. The image of the drawn idea was then projected on a large display in the room. Small coloured pieces of paper could be placed upon the drawing in order to display changing values within an idea. An idea could then be presented to the rest of the group. This allowed for ideas to immediately be picked-up by the rest of the group.

A translation was made for each of the three different types of data that ideas were searched for. This was done so that generated ideas could easily be explained and discussed. The translation for each type of data was made so that one participant controls the other participants within a defined space on the floor. This resulted in the following three types of data.

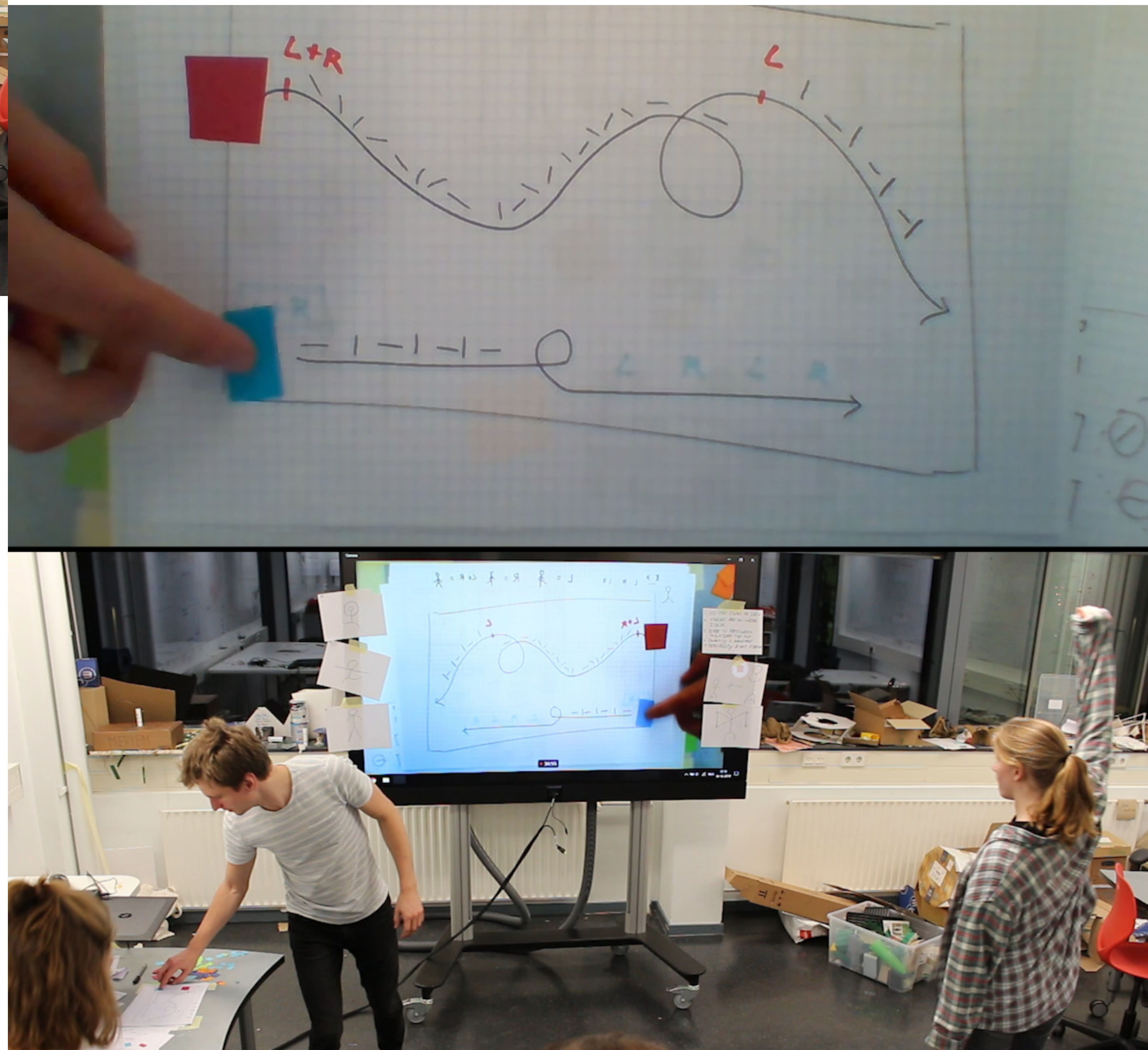
Figure 23: recording setup of the idea generation.

original data type	translation
positional data	participant's positions
discrete data	fixed participant's postures
continuous data	continuous participant's postures

Besides the animation of all types of data, insights also cover how an overview of automations could look.

The filmed ideas that were presented on the big screen were recorded. The participants enacting the ideas were also filmed.

Figure 24: recordings of ideas next to the execution.



5.2.2 | insights
Insights of the brainstorm are separated into two categories. In the first category are ideas into how the types of data can be animated. The second category entails ideas into how animations can be overviewed and arranged.
A complete overview of insights of the idea generation can be found in appendix C2: Idea generation - insights.

5.2.2.1 | ANIMATING

positional
For animating positions, drawing in the positional overview by hand or mouse came up. An addition to drawing by hand, is drawing with predefined geometrical shapes. Drawing positions does not necessarily cover the duration of an animation. It can be derived from the duration of a drawing. A more precise method is to manually specify the duration. If necessary, speed within the animation could be adjusted through a graph.

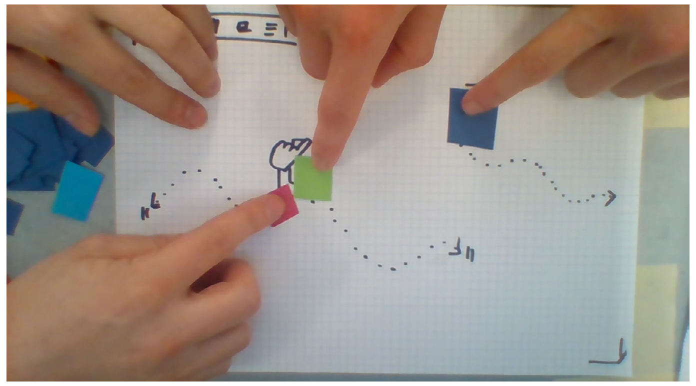


Figure 25: animating positional data.

continuous
Sliders allow the adjustment and thus animation of continuous data. Animation is then again realized with keyframes or scenes. Gradients are possible between these moments.
A simpler version of animating continuous data can be realized through choosing preset (looping) sequences.

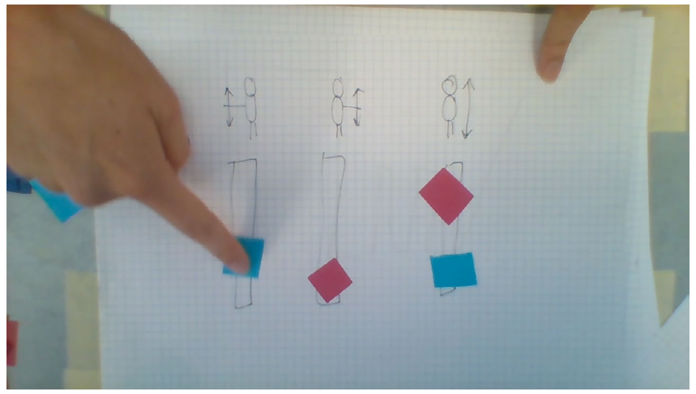


Figure 26: animating continuous data.

discrete
Setting discrete data can be done through simple selection of the discrete value from a selection window. Animation is done either with keyframes or cues. A preview can be graphed, although solely through a block-graph.

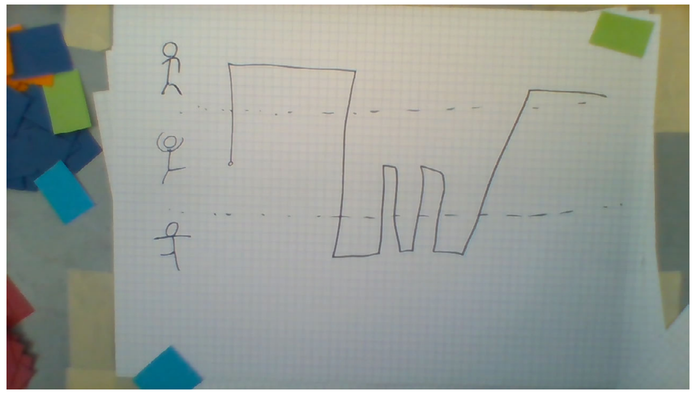


Figure 27: animating discrete data.

5.2.2.2 | ARRANGING ANIMATIONS

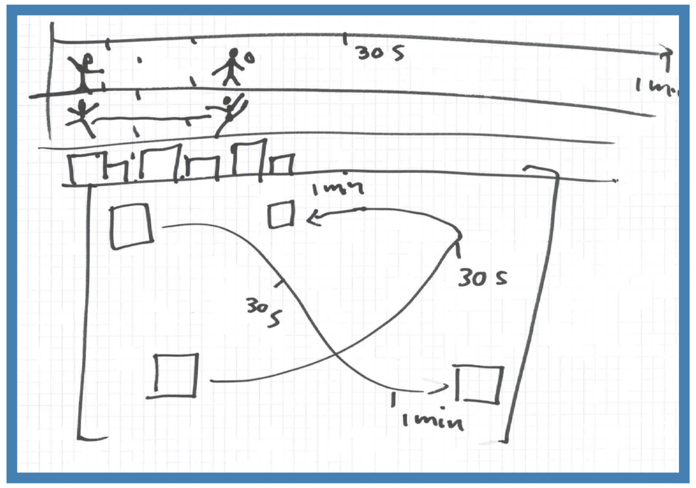


Figure 28: a timeline on which animations are altered.

timeline
An often-reoccurring element in the ideas is a timeline that is able to depict an overview of all different types of animations next to each other. This allows adjustments within the dimension of time of the animations. It also shows the present moment within a mix of animations.

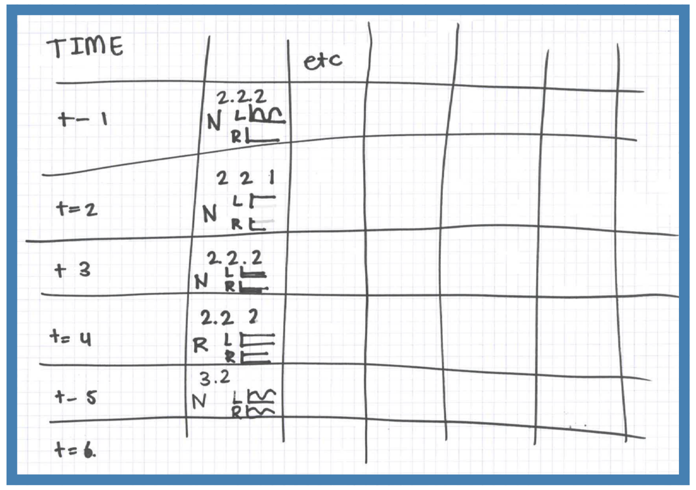


Figure 29: cues to trigger animations.

cues
Another way of arranging all animations is by setting them into cues. In a cue, the current state of all objects and their values are saved. Animations can occur within and during the transition between cues. The advantage of cues is that it allows easy control during a live performance.

Cues could be integrated into a timeline as a keyframe that overrides all other data.

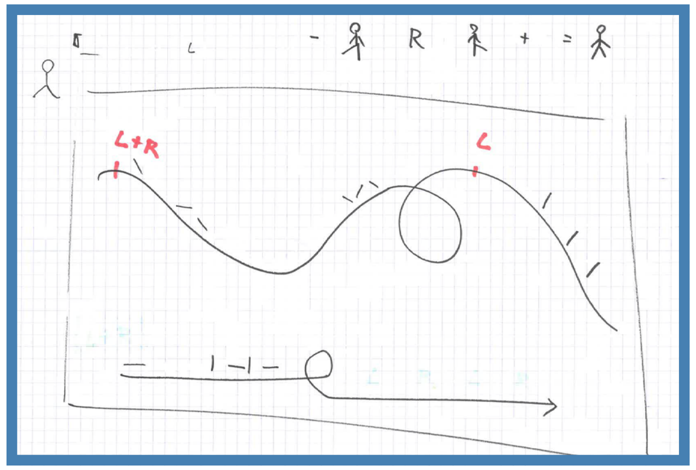


Figure 30: previews in icons of animations.

icon previews
For both continuous as well as discrete data, ideas were generated for icons to display the changes in the live positional overview. As the aim is for an objects data to be shown in the positional overview, changes through animation will automatically be shown as well. Whether all parameters will be shown depends on the previews.

looping
For all types of animations, looping can be applied. Within looping, there are different sequences that can be followed. A new loop can start from its original starting point or first return there by an inversed animation.
Loops can continue infinitely long or be set to a certain amount of iterations.

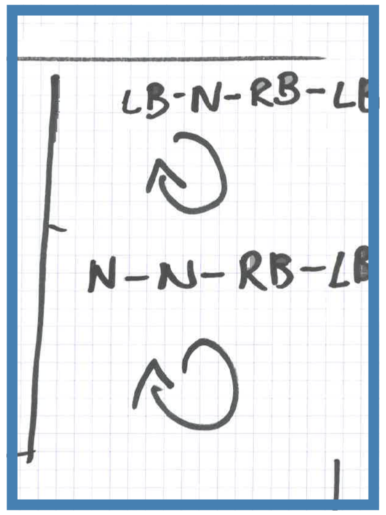


Figure 31: looping animations.

Figure 32: mixing studio at KlevR.

5.3| OBSERVATIONAL STUDIES

Three different user observational studies were executed. The observations were combined with interviews, to gain a better understanding of the participant's motivations (Boeijen, et al., 2013). The aim of the observations and interviews was to see in what way immersive audio mastering will fit onto the responsibilities of its users. Audio mixers of different disciplines were approached for the observational studies. This was done to ensure a varying input into the study.

The observations were held with the following persons.

- **Serge Gräfe**, who is the Front of House engineer of Kraftwerk. He has been involved with Soundscape for a long period.
- **Wibo Vermeulen**, who is an audio engineer at the Dutch National Opera & Ballet. He had no prior experience with immersive audio
- **Alan van Ramshorst**, who is a sound designer at KlevR, a sound design studio for films. He has experience with mixing for surround setups without OBA mixing.

The complete description of the setup and execution of the observational studies can be found in appendix D1: Observational studies - setup and execution.

Figure 33: guided tour at the Dutch National Opera & Ballet.

5.3.1 | setup

For each observation, a custom checklist and one general observation sheet was prepared in order to guide the sessions.

Each checklist was set up to guide the observations and discussions in a semi-structured way. Discussions and observations alternate in the setup, ensuring as little discussion during observations. This was done to limit the involvement of the researcher, so that an open perspective is not lost (Singh, 2019).

Topics of discussion are set up in a way that starts with the current way of operating, on the basis of the observations. The topic moves then to past experiences, in order to later discuss the future needs (Sanders, 2001). This setup concurrently ensures that discussions on technical aspects are avoided in the beginning. This is important as the goal of the study is focussed on the end results of mixing, namely the listening experience, rather than the technical operations that it relies on.

5.3.2 | insights

From all interviews, interesting quotes were gathered, and an interpretation was added. The interpretation serves to assure that the intention of the quote is still emphasized when the quote is removed out of the context of the interview. When the prepared observation sheet was used, interesting insights were also processed with an interpretation. In total 59 insights with interpretation were gathered.

5.3.3 | mapping

All 59 insights were first grouped based upon their subject. The connections between the groups were sought and the quotes were laid at their relevant position. This process created a map of all insights that were found. Each quote was numbered and given a colour that represents speaker.

All numbered quotes can be found in appendix D2: Observational studies - quotes. The complete transcriptions of the interviews can be found in appendix D3: Observational studies - transcriptions.

The map provides an insight into the responsibilities of an audio engineer who might use immersive audio, and how the use of that fits into the tasks of mixing. While the responsibilities stay fairly similar, the purpose of the responsibilities differ for each show. This makes the tasks that are carried out for a certain responsibility dependant of the content of the performance.

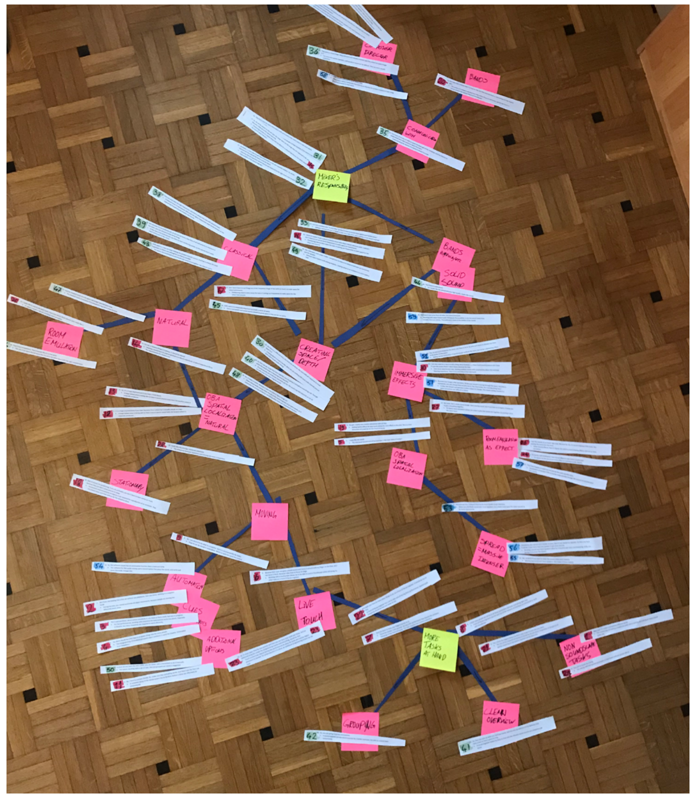


Figure 34: mapping of quotes and insights.

equal listening experience
The mixer has the responsibility to give all listeners the same listening experience. An immersive audio system allows OBA mixing while not nullifying this responsibility, as was seen in chapter 3.1.1: Wave field synthesis.

"...that the same information ends up in every chair, every place in the room."

---

creating space & depth
The most obvious responsibility, which can be aided by Soundscape, is creating space and depth in a mix. In the case of a classical performance, the aim of creating depth is naturalness and to stay as close to the real performance.

"For me, in theatre things, it is always, it should be natural."

While for bands, the necessary aim of depth is to create a solid sound with immersive effects.

"That there is solid sound. That people can let themselves be carried away."

Insights showed that OBA mixing as well as room emulation effects contribute to both these tasks.

---

multiple tasks at hand
There are numerous other mixing tasks that a mixer takes care of during a performance, besides controlling Soundscape. Therefore, its controls should be easy in use. For live control, touch-control was suggested, and for automated control, cues were preferred over snapshots. For programming automations, time-efficiency is not so relevant, as this process is less time restrained. Its controls can therefore be rather extensive, as long as the end result is quick to control.

"You have so many other things to do, as a Front of House engineer."

It was also named that a mixer keeps a clear overview over his controls by customization. This customization should also be allowed for the controls over Soundscape elements, to ensure a clear overview of OBA mixing elements.

"So, then you pick up what you need per scene."

---

communication with others
Independent of the performance, the mixer has the responsibility to process certain wishes of other stakeholders of the performance into the mix. These can be the performers, a director or the composer. The mixer should be able to speak their language to communicate well.

The clear overview of all Soundscape elements can improve this communication through visual feedback. A clear overview and communication can then also encourage these content creators to take OBA into their writing, as is already the case in film. This could serve very well for the development of immersive audio systems into the industry standard.

"So, write sound localization in your theatre play."

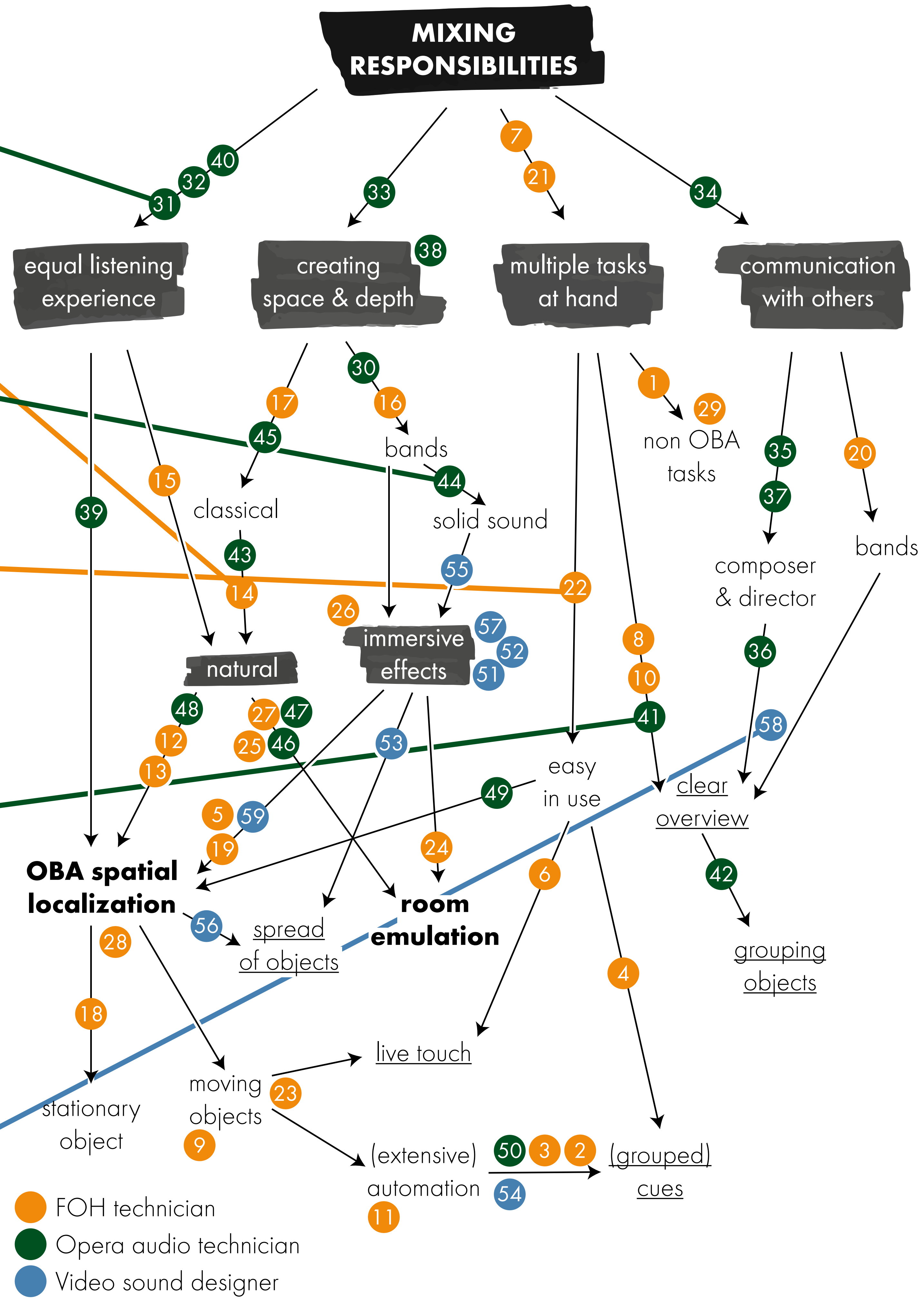


Figure 35: map of all insights and quotes.

REQUIREMENTS

6.

In this chapter, the defined requirements for the concept are stated. The largest aspect of these requirements is the result of a function analysis. Integrated into this analysis are the definitions of quality for each of these functions.

Furthermore, an elaboration is made about the applications of live immersive audio which is focussed upon.

6.1 | FUNCTIONS

At the start of the project, a function analysis was made to determine the scope of the concept. During the project, the function list was kept updated. The functionalities were divided into 7 different categories.

An important notice about the envisioned application is that it creates no completely new functionalities. All functionalities are already accessible in one way or another. Objects can be placed and moved around in R1. Automations of movements and other values can be set through a plug-in. Without the introduction of new functions, the added value of the application lies in the increase of usability.

6.1.1 | definition of quality

A definition of quality was included for each functionality. This was done to be able to determine whether the functionality was incorporated correctly. A reference to this function analysis can then later be made to evaluate the concept.

The complete list of functions can be found in appendix G: Function analysis.

1. **live mastering en-scene & en-space**
2. **object configurations**
3. **automation**
4. **automation live control**
5. **scaling**
6. **external linking**
7. **additional**

An obvious attribute of all functions becomes apparent. On one side, functions to set up a mix allow for more time to use the application and require more **extensive functionalities**.

On the other hand, functions that are being used during a performance have to be quick in use. Increased usability plays a key role in saving time of use (Goodwin, 1987). The goal for the concept is to achieve usability by **targeted reduction** of superfluous functionalities. To aim this reduction, an understanding of the **use context** is required.

Figure 36: amount of functionality versus increased usability.

CONCEPT ELABORATION

7.

In this chapter, a detailed description is given of the concept that has been developed during the project. The choices that have led to the concept are elaborated.

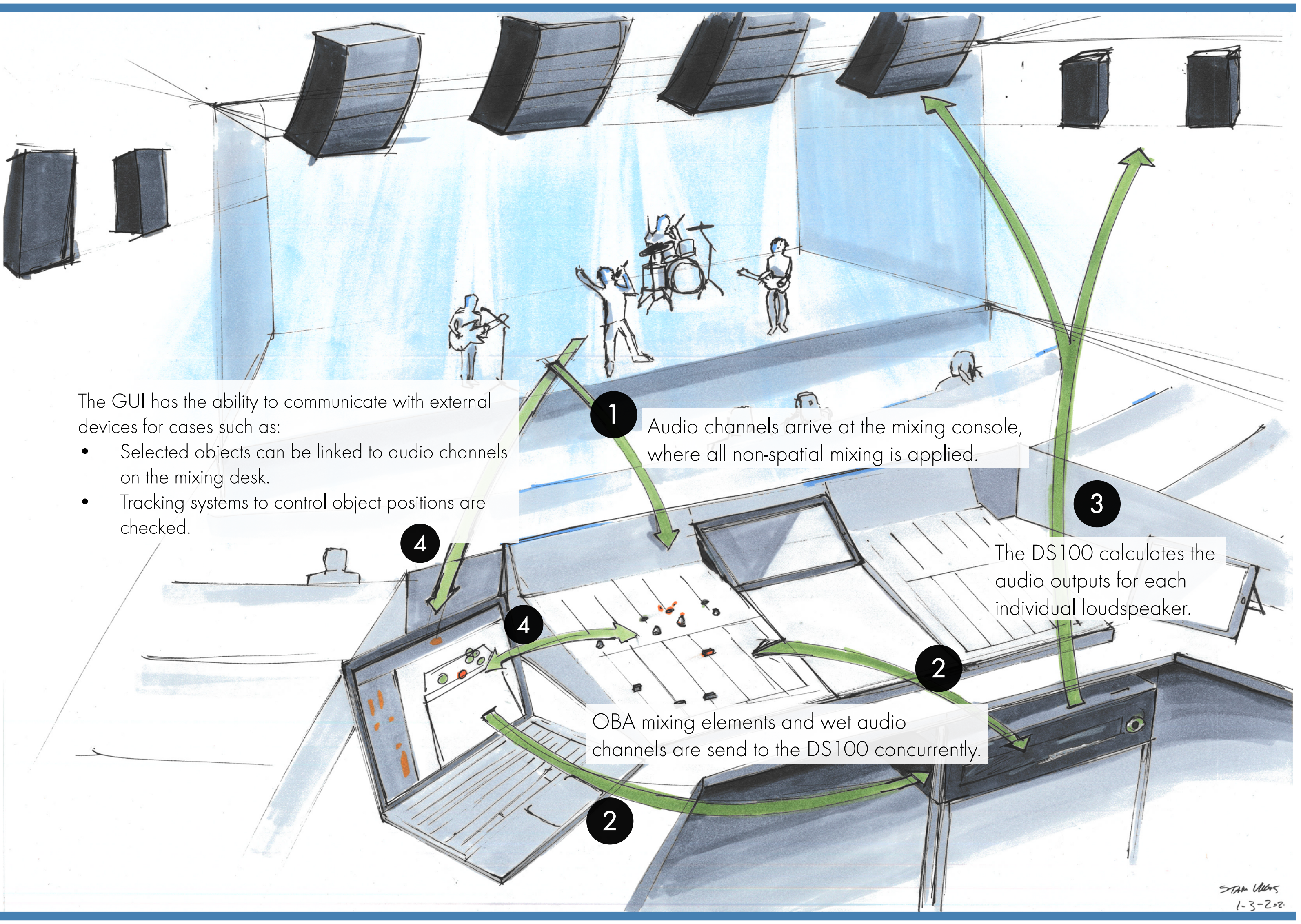


Figure 37: integration of concept GUI in the audio mixing workflow.

7.1 | CONCEPT USE

The use of the concept graphical user interface (GUI) is intended to be integrated into the current workflow of an audio engineer.

1. This entails that first the audio channels come in from where they are recorded. This can also be pre-recorded material that is sent from a digital audio workstation (DAW).

2. Then the spatial aspects in the GUI are executed concurrently with other mixing tasks. The altered audio channels, wet audio channels, are sent at the same moment to the DS100.

3. The individual audio channels for all the loudspeakers are then calculated and send to the speakers.

4. In some cases, external communication with the GUI is desired. External devices can be able to control elements within the GUI. An example is a tracking device that sends positional data towards the GUI. The GUI should be able to read this data and let the device control an object's position. The GUI should also be able to send data, such as timecode or source selection.

7.2 | OBJECT OVERVIEW

As described in the design vision, the listening experience is at the core of the new workflow that the GUI addresses. It was therefore chosen to work with a 2-dimensional object-based overview. In this overview, sound sources become sound objects, which are represented by a coloured circle. The overview serves as the depiction of the current listening experience. It also serves as the control over sound object positions. Object can be dragged around to adjust their position. A selected object is coloured orange.

As the listening experience is the core, it was chosen that the overview is always prominently visible.

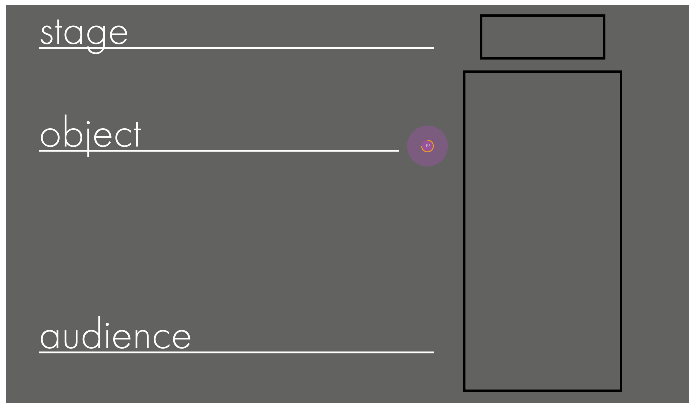


Figure 39: items in overview.

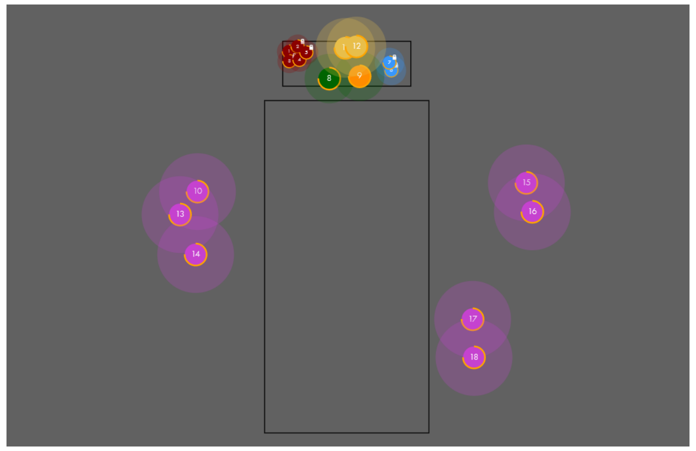


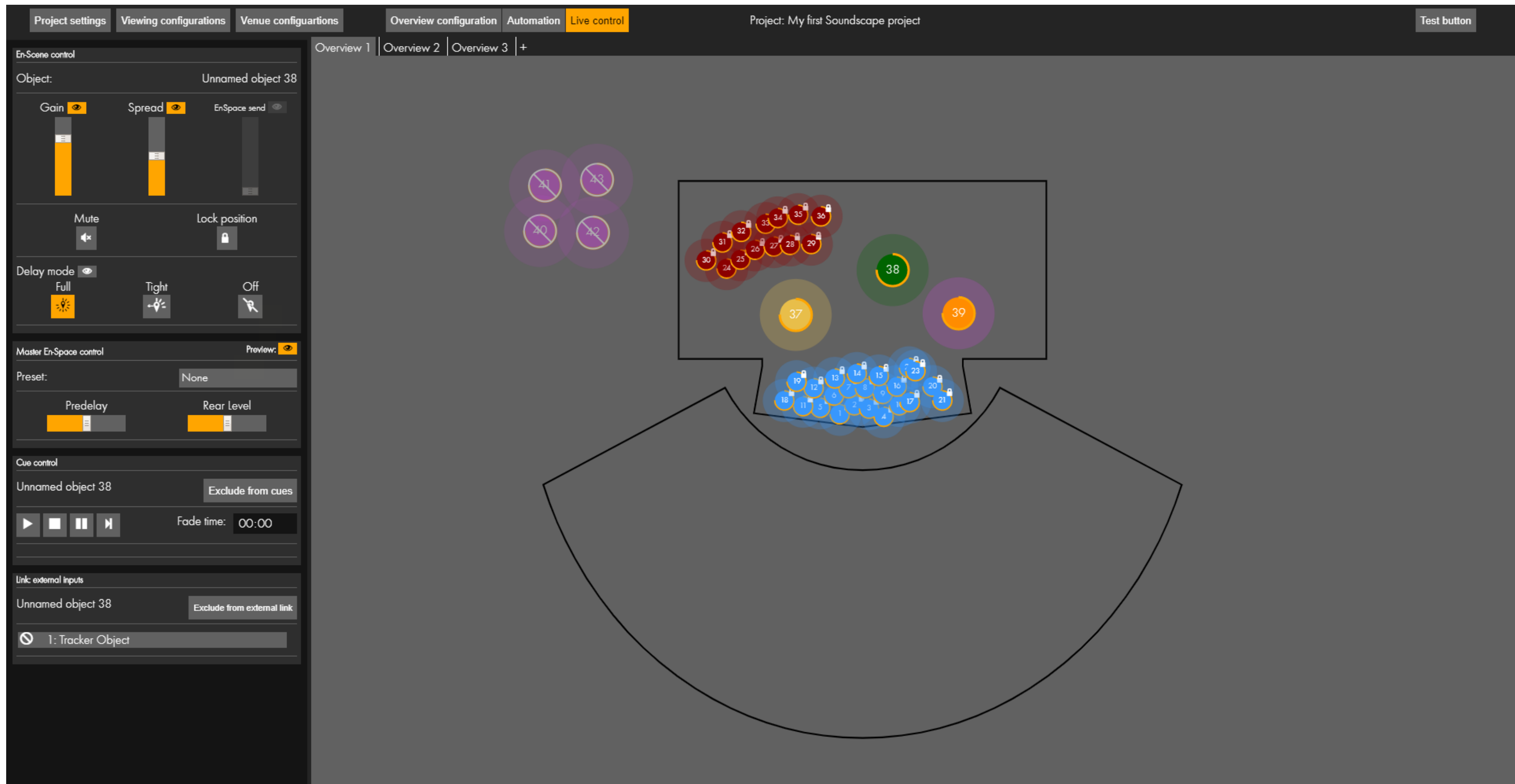
Figure 38: overview excerpt.

7.2.1 | reference

The venue's stage and audience borders are depicted in the overview as a frame of reference for object positions. These borders were chosen as they are always present and provide information about multiple aspects. It shows the position of the audience within the listening experience. It also gives an indication of in what positions sound objects can be set.

The stage and audience reference can be chosen from a set of examples or can be imported from ArrayCalc. More information regarding the setting of the venue's layout can be found later in chapter 7.3.1: Layout.

Figure 40: overview with live object controls.



7.2.2 | object groups

Multiple objects can be selected by pressing down the shift-key when another object is selected, or by using a selection box that is opened by pressed the alt-key and dragged in the overview over the objects. Multiple objects can also be placed in a group with the key combination ctrl + G. A group of objects serves as a multiselect of all objects within the group.

Multiple object positions can be moved in more ways than single objects, as they have positions relative to each other. Three kinds of transformation of multiple objects have been defined. Inspiration for these functionalities has been taken from Adobe Illustrator, see chapter 5.1.2: Adobe creative suite.

To allow multiple sorts of transformation, a box is drawn around the object positions.

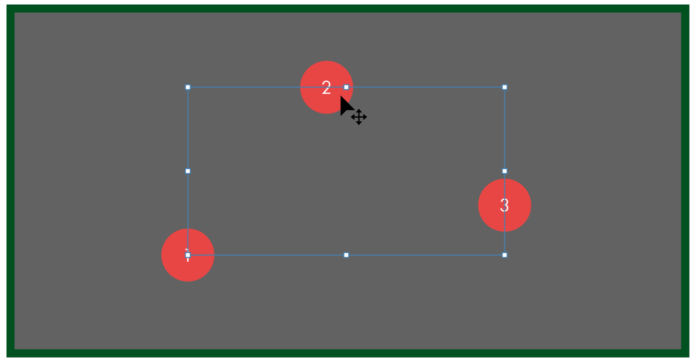


Figure 41: transformation box: translate.

The first transformation is translating. This is similar to how one object is moved around within the overview. The whole group is moved and relative distances with the group remain the same. It is initiated by dragging one of the objects.

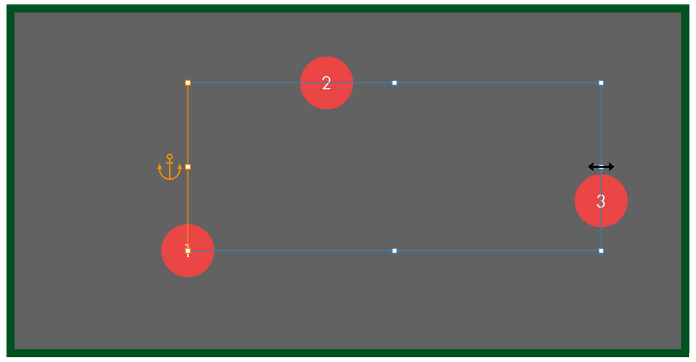


Figure 42: transformation box: scale.

The second transformation is scaling. It is initiated by dragging one of the outer edges or corners of the transformation box. The opposite edge or corner will serve as an anchor of the scaling. The anchor is depicted with an orange line. Scaling can also be done by having the middle as an anchor. When the box then scales towards the left, it also scales to the right in the same amount. This can be initiated by scaling with the alt-key pressed.

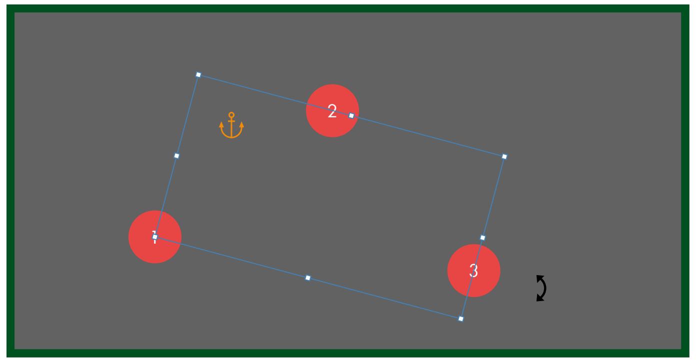


Figure 43: transformation box: rotate.

The third transformation is rotating. A rotation requires an anchor point around which the objects are rotated. This is normally set in the middle. As the transformation box is drawn, the anchor point can be reset. During rotation, relative distances remain the same, but the angle between objects changes.

7.3 | PROJECT SETTINGS

Upon initiating a new project, the program displays a window to set the project's settings. These settings are required for starting a project, and are expected to be known at the start of the project. However, they are not definite and can still be changed later on.

The settings involve the project's name. Whether En-Scene and/or En-Space is available. If En-Scene is used in 180 or 360 mode. And the amount of DS100 signal processors. This knowledge is required for the application to determine how many input sources are available for the project. As one DS100 has a capacity for 64 sources, 64 input sources are available to be made into sound objects per DS100.

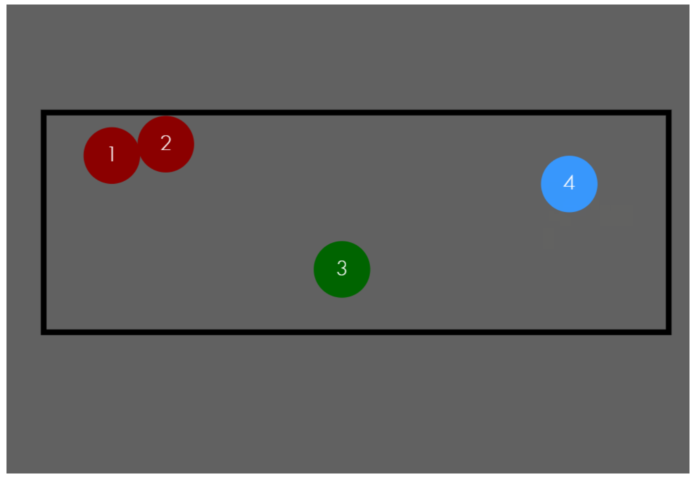


Figure 45: template of a 180 setup.

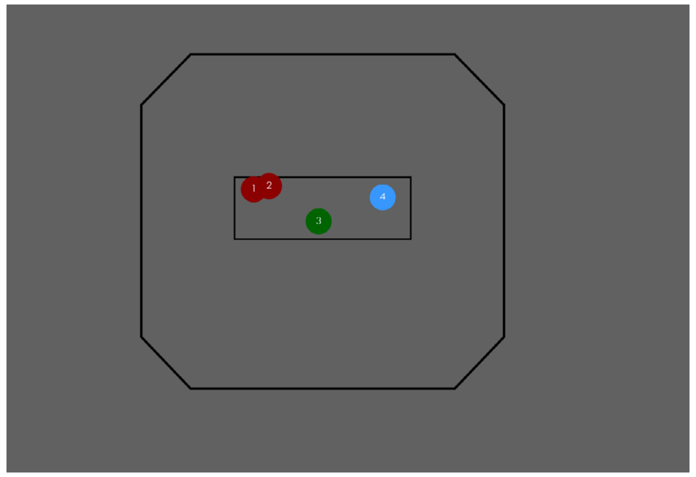


Figure 46: template of an arena layout.

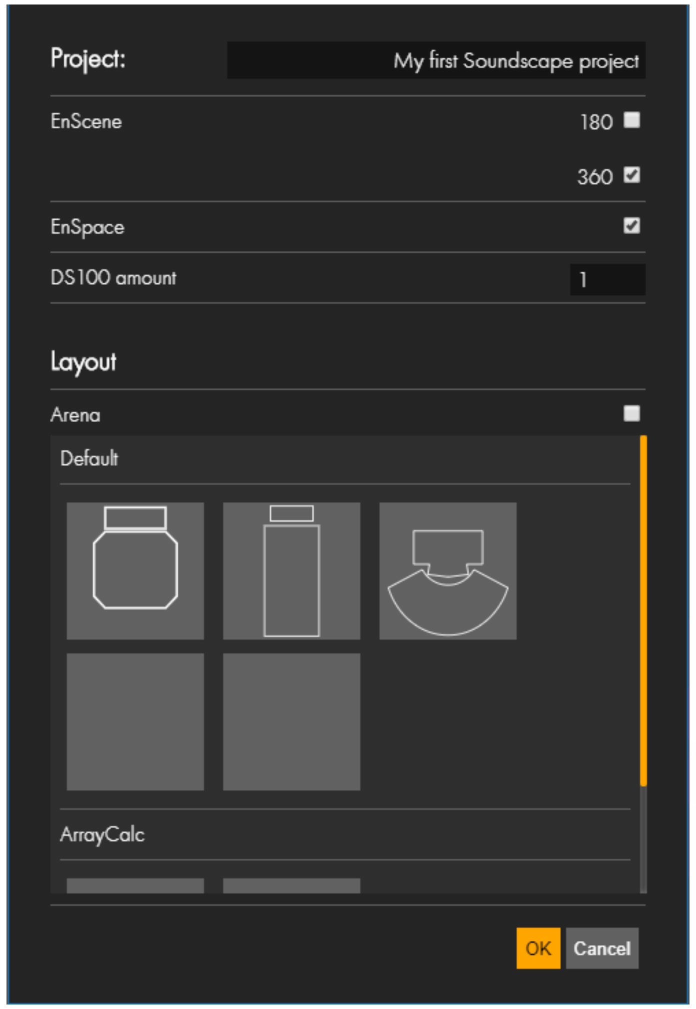


Figure 44: project settings.

7.3.1 | layout

Furthermore, a layout for the stage and audience can be set. Such a layout is either an arena or not. An arena layout is one where the stage is surrounded by the audience. This means that the stage is confined within the borders of its audience area. This always requires a 360 Soundscape setup.

The layout can be chosen from a template. This is useful when there is no drawing yet of the loudspeaker setup, or when the performance is given in multiple different venues. Templates should be available of different venues with an audience for 360 mode and without the audience for 180 mode. Templates for arena layouts should also be available.

If there is already a drawing made available in ArrayCalc, the top view of the drawing can be imported into the overview and be used as a reference for the objects. The audience area and listening planes that are drawn in ArrayCalc are then displayed in the overview.

7.4| OVERVIEW CONFIGURATIONS

Three different sets of controls have been defined to control the overview, namely **overview configurations**, **live control** and **automation**. Dependant of the control mode, different controls appear next to the overview.

The first set of controls that is presented besides the overview are the **overview configurations**. These controls are used to set up the overview.

Sources can be selected and dragged into the overview. When this is done, they automatically become an object. Sources that are already included in the overview will be marked with a checkmark. Sources can be placed multiple times as different objects. The sound source is then reproduced from different locations.

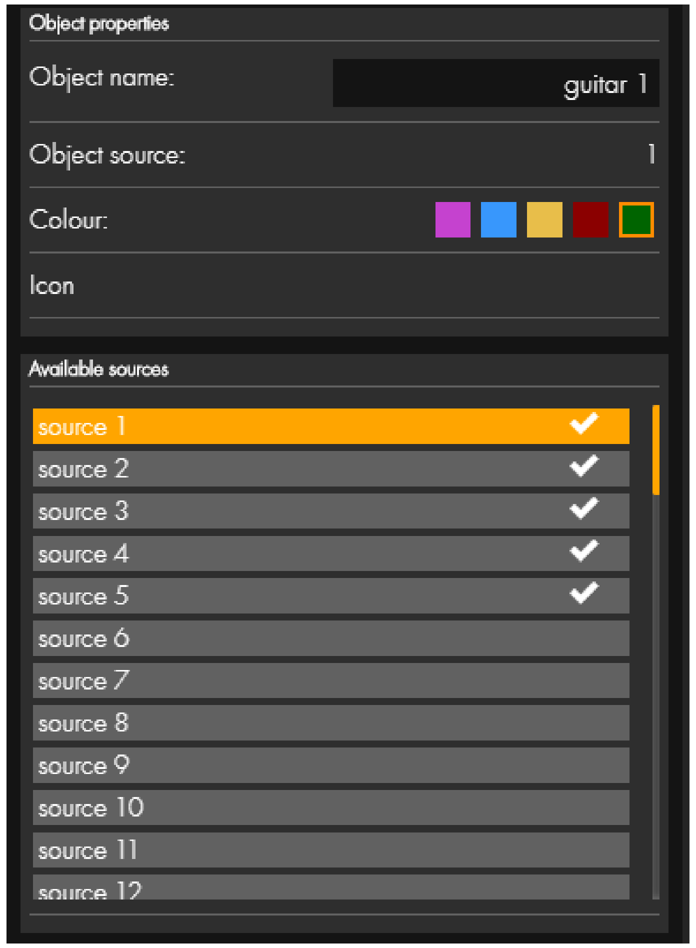


Figure 47: overview configurations.

7.4.1 | object recognizability

When more and more objects are placed into the overview, the recognizability of each individual object becomes less. To enhance the recognizability of objects and object groups, their appearance can be customized. A desire for such customizations also became apparent during the observational studies in chapter 5.3: Observational studies.

Within each object, its source number is displayed. This source number is always present, so that the object can always be traced back easily to its source.

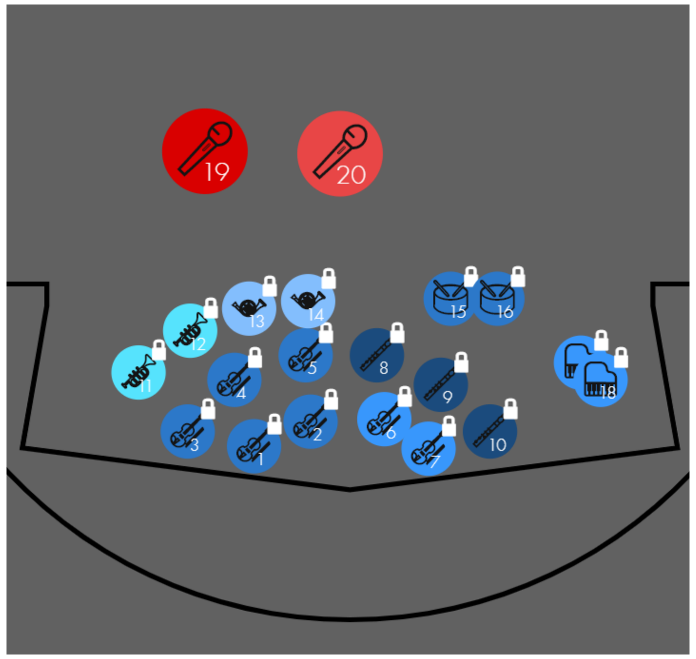


Figure 48: a blue coloured orchestra with two red coloured singers.

The colour of the object can be changed. This is the most recognizable feature of an object. The colour can be picked from a set of predefined colours. This ensures that the colours are distinguishable from each other and from the orange colour of a selected object.

The available colours are arranged in five sets of shades of distinctive colours. This allows for analogous instruments to be given analogous colours. For example, all instruments of an ensemble could be blue. Different instruments within this group would have different shades.

An icon for each object can be chosen from a set of icons. Icons that are included are icons of instruments to label the input source with the corresponding instrument. The source number will then be displayed smaller next to the icon.

7.5| LIVE CONTROL

In the **live control** mode, the control windows which are used during a performance, are present.

7.5.1 | en-scene

If En-Scene is used in the project, the En-Scene control window is presented. In this window, the properties of objects can be set.

If multiple objects are selected, all their properties will be set.

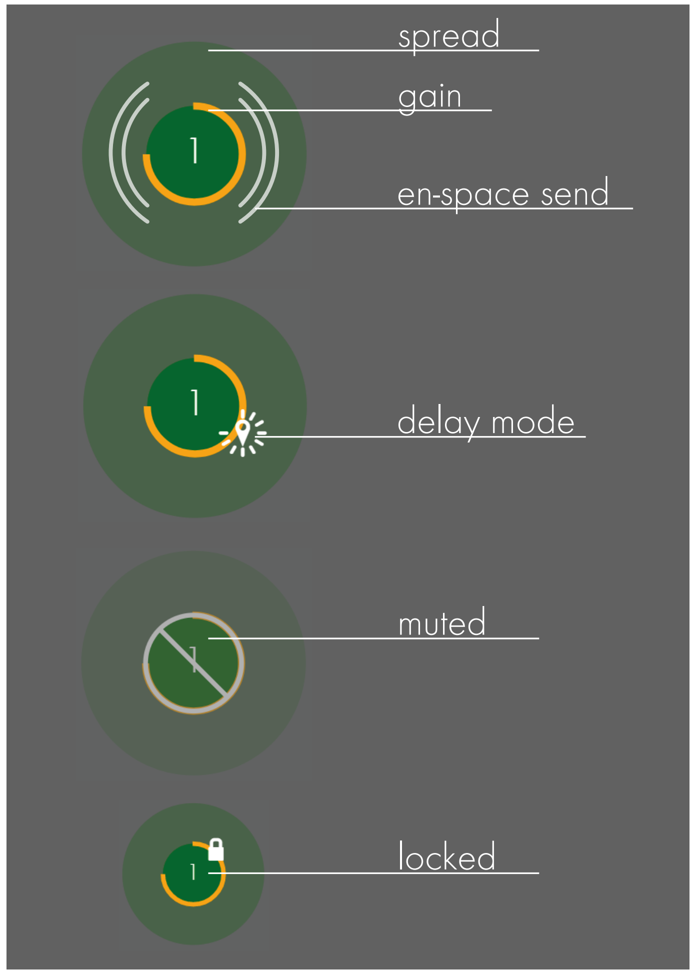


Figure 50: object parameter previews in overview.

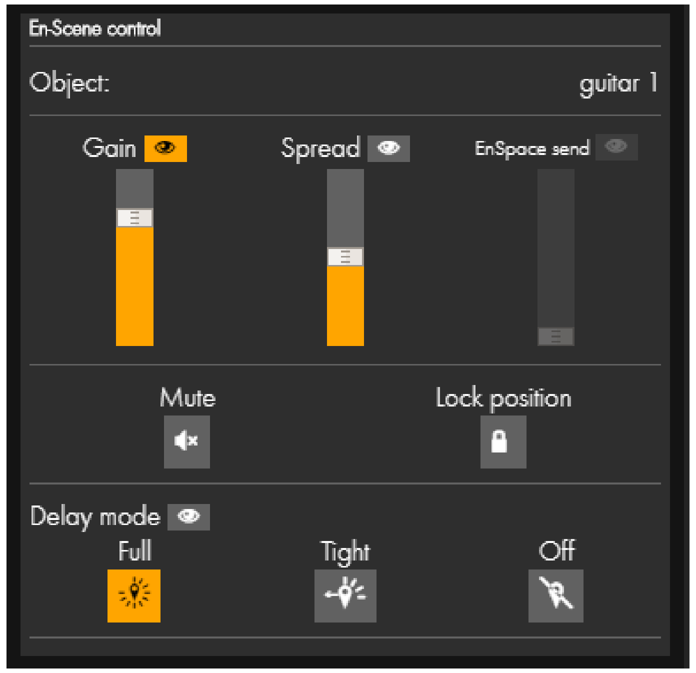


Figure 49: en-scene live control.

The properties are also previewed in the overview around each object.

In the image on the right, the changed attributes of the object relating to the object's parameters are shown.

It has been chosen not to depict possible changes before they are made. A reason to integrate these could be to show beforehand how the listening experience changes. However, it was argued that these depictions can be distracting. In addition, these controls are always the same in every project and can thus be assumed to be part of the learning curve of the software, as the preview does change upon the usage of the controls.

7.5.2 | en-space

If En-Space is used in the project, the En-Space control window is presented. This window contains a list of all sampled concert halls that can be used. Some information about the hall is given as well as a picture. This gives the mixer an idea of how the reverberation signature of the room will sound before that he chooses a sampled room.

In the window, the pre-delay and the rear level of the sampled room can be altered.

Once a room is selected, a preview of the room emulation is also given in the overview.

The concert hall is projected upon the layout of the project. The size of the room will change as the pre-delay is changed. The balance of the orange colour gradient within the projected concert hall represents the rear level.

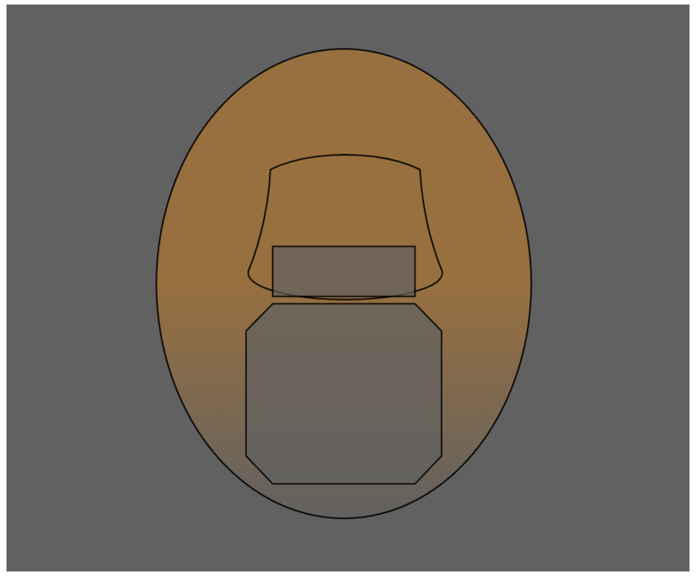


Figure 52: en-space room preview in overview.

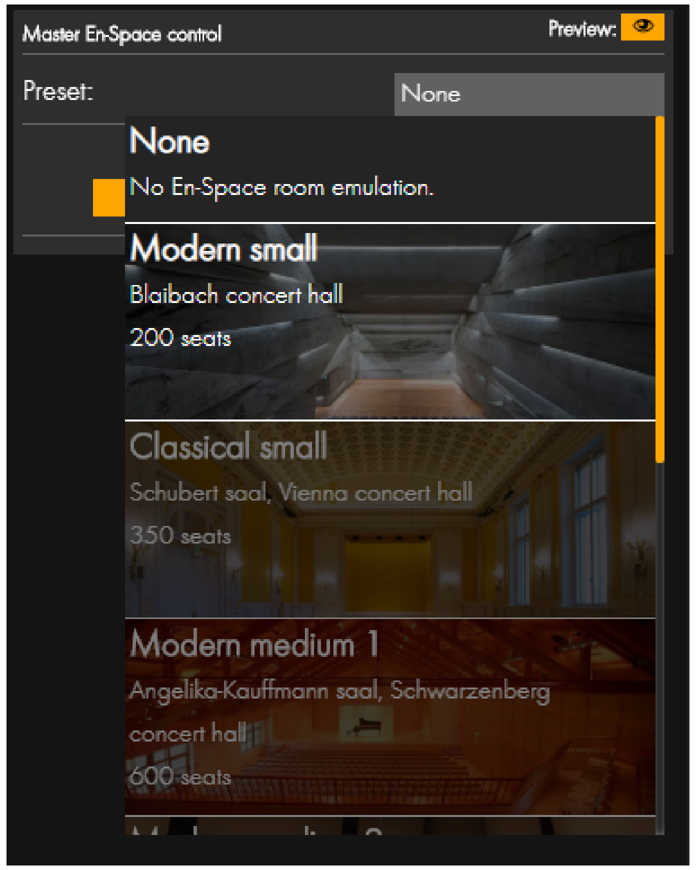


Figure 51: preview of room en-space room selection.

7.6 | AUTOMATION

In the **automation mode**, windows to configure cues are presented.

7.6.1 | cues

In the GUI, automations are programmed within cues. This choice was made, as it became apparent from the observational studies that during a live performance, not all movements happen exactly on the same point in time. An audio engineer should be able to start the right automations at his desired moment. Cues can be triggered at any time from the live control mode.

Their functionalities can be programmed before the performance in automation mode. In a cue, all controls that are available in live control can be saved. This goes for controls over En-Scene object properties as well as En-Space room emulation properties.

"As soon as you start the next scene, you can either start a movement or do it manually." - Wibo Vermeulen on triggering cues

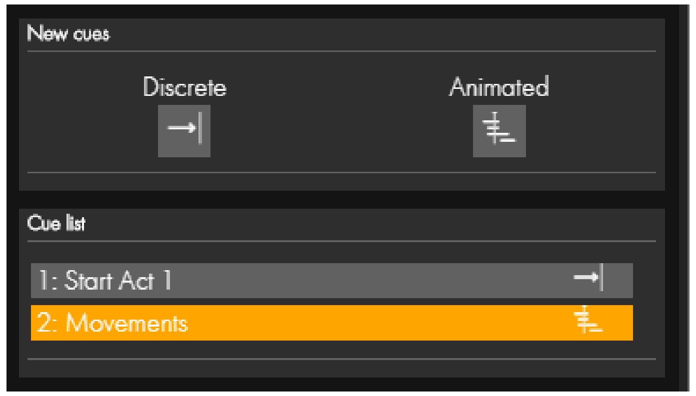


Figure 53: list of cues with buttons for creating new cues.

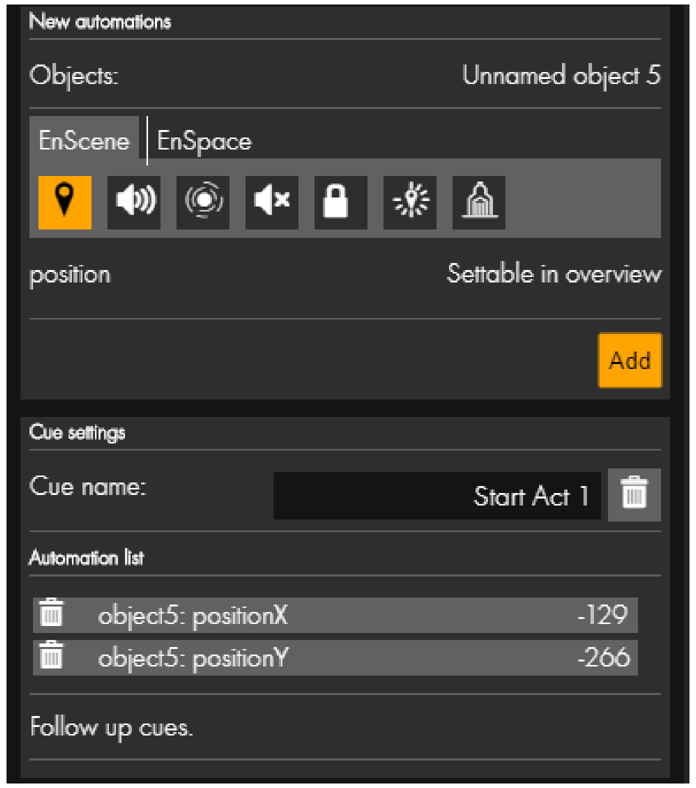


Figure 54: controls to adjust parameters in cues.

7.6.2 | discrete

There is a distinction between two types of cues. Discrete cues only contain one value of each property that is added. If the cue is fired, the properties will directly become the value stored within the cue.

Properties can be added upon selecting a cue, and in the case of En-Scene properties, an object should be selected as well. Dependent on the sort of property, a slider or button appears. The value can be set by the slider, button or in the overview and added to the cue.

7.6.3 | animated

The other type of cue is animated. For each value that is stored, a time indication is given. This creates the possibility to save movements and gradual changes within the cue. These movements and changes are depicted upon a timeline. On the timeline, the keyframes can be adjusted as well.

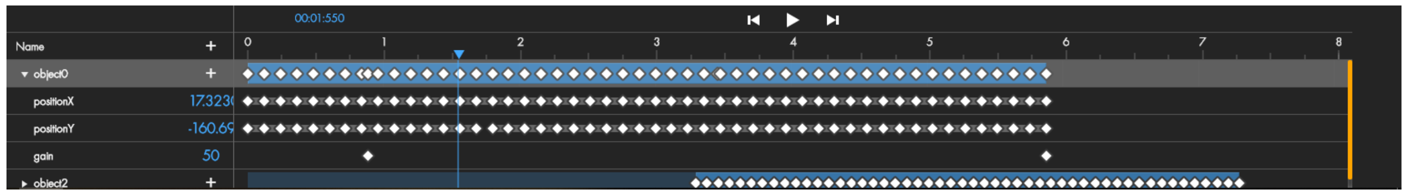


Figure 55: timeline to adjust the course within animated cues.

For all properties, keyframes are added to the timeline at the time at the currently selected time. For movements, the option to draw an object's path is added. This option was added to keep the programming of automations as close to live control as possible. Movements can then also be followed during a rehearsal. This is one of the ideas for animating data that came forth from the idea generation discussed in chapter 5.2.2.1: Animating.

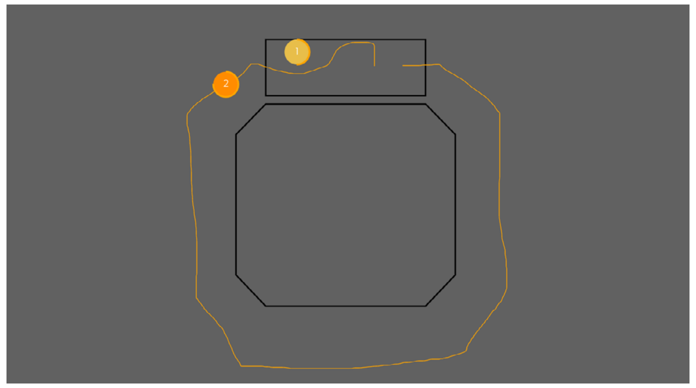


Figure 57: animating positional data by drawing.

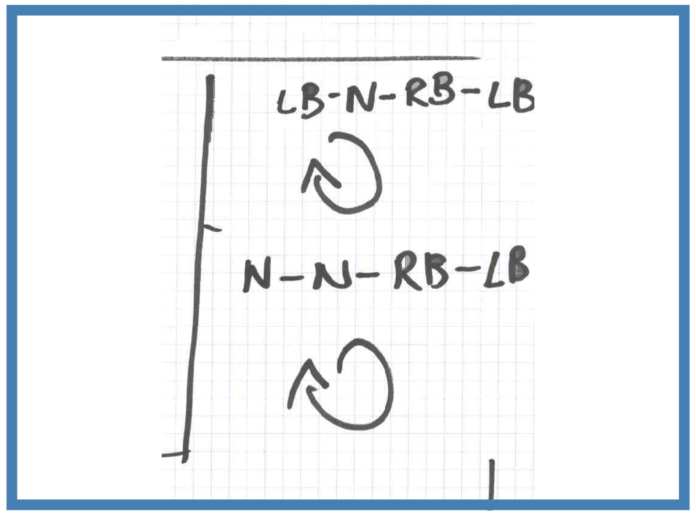


Figure 58: continuous or reversed looping of cues from idea generation..

7.6.6 | cue previews

Cues are fired in the live control mode and programmed in automation mode. This creates the situation that the mixer can create his own controls tailored for the performance. As cues are always different, these controls have no possibility to completely fall under the GUI's learning curve. It was chosen to display a semi-transparent preview of the selected cue. This lets the user know that he is certain of what he will adjust in the listening experience before it is adjusted.

"Also, for your own security. That you know if it's all right. (...) Yes, that is very important. You know, I'm going to do it right." - Paul Lardenoije on cue previews

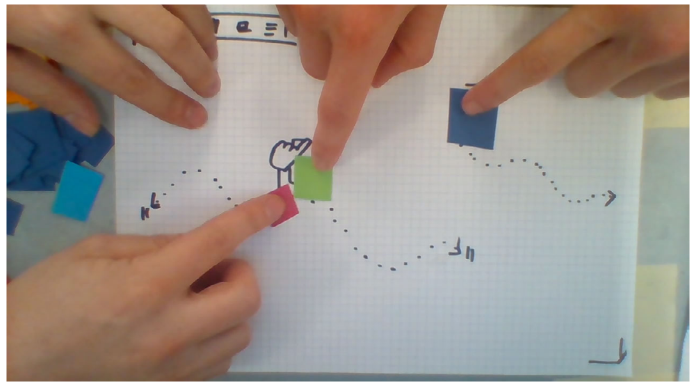


Figure 56: animating positional data from idea generation.

7.6.4 | cue settings

For each cue, settings can be altered. These settings include the cue's name, the amount of iteration a cue is looped, and whether looping is done reversed or continuous.

In the settings window, a list of all automations is shown and automations can be removed or adjusted.

Additionally, other cues can be selected to automatically start once this cue has finished. These are called follow-up cues.

7.6.5 | fade-in time

A fade-in time can be set for a cue. A cue can be played instantaneously or faded. If it is played instantaneously, the values within the cue are immediately activated. When a cue is faded, the actual values gradually move towards the starting values of the cue. The time that it takes is defined by the fade-in time.

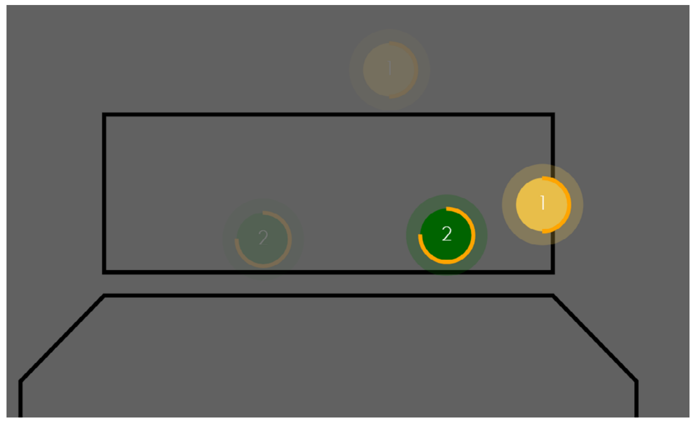


Figure 59: semi-transparent preview of cue upon selection.

7.7 | LINKING

To increase an integration of the GUI into the workflow of an audio mixer, the GUI is able to communicate with other devices. These communications happen in multiple ways.

7.7.1 | external control

All the GUI's controls can be read and written with the OSC protocol by external devices. As an external device writes values for certain properties, the external device will be listed within the GUI. The mixer is then also able to block a device from this list. This is useful in the case that an external device malfunction.

An example of the use of such an external device can be found in the second use case in chapter 4.2.2: Touring band. A tracking device monitors the lead singer's position, and transfers the data with the OSC protocol to the GUI as x and y coordinates.

Additionally, what makes the use of the GUI more integrated into the workflow, is that object's in the GUI are easily relatable to the sound sources on a mixing desk. Selecting objects is therefore seen as a functionality of the application which also can be read and written with the OSC protocol.

"I think many people are afraid of all these connections between everything right now." - Serge Gräfe on operation between devices

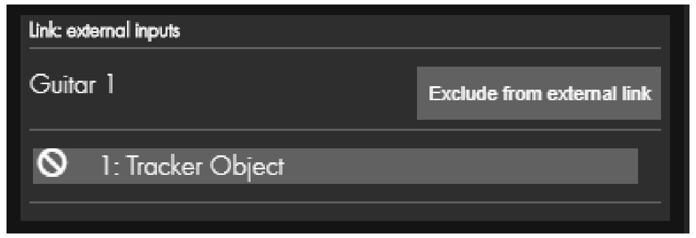


Figure 60: list of tracker object with blocking option.

7.7.2 | timecode

To allow a synchronization between immersive elements and other elements of a performance, the GUI has the ability to read and write timecode. The operation of timecode is integrated into cues.

When the GUI operates as a slave, it reads an external timecode. Cues can be set to start at a certain timecode, and will then be fired by the external timecode.

If the GUI operates as a master, an animated cue will run a timecode. External devices can be coupled to this timecode. The timecode will start once the animated cue is fired. Timecodes with which the GUI works are MTC, Midi timecode, and LTC, linear timecode.

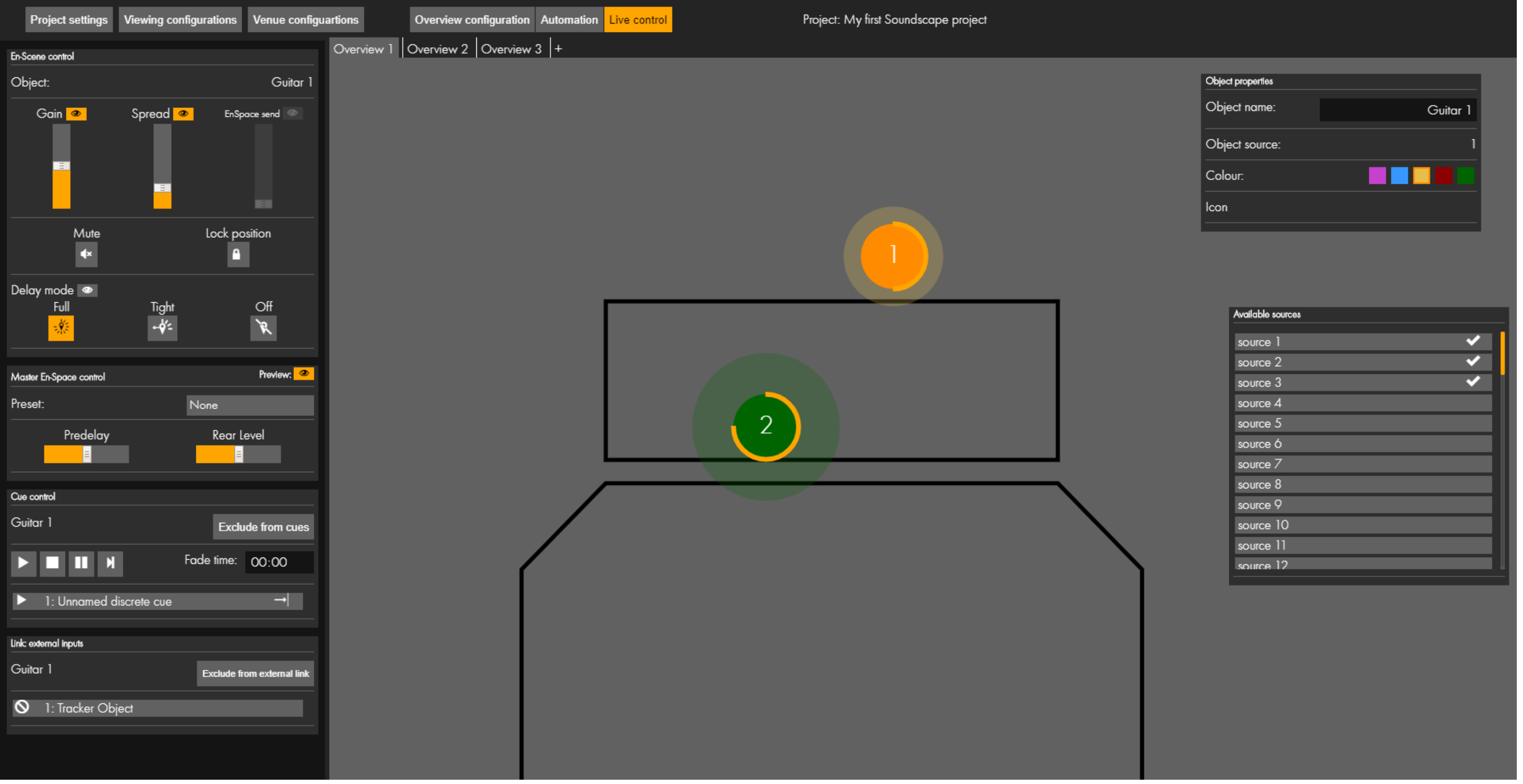


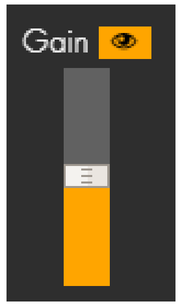
Figure 61: floatable windows of the overview configurations in the live control mode.

7.8 | CUSTOMIZABILITY

As the scope of use cases is quite broad, the relevance of previews and controls will sometimes differ from the default control modes. To ensure that the overview and controls are kept relevant to 100% of all use cases, customizations can be made.

7.8.1 | previews

As all properties are displayed in the overview, this can become cluttered and superfluous. The user can hide the preview of certain properties with the use of a small button next to the property's name. This option is available for all properties that can be irrelevant.



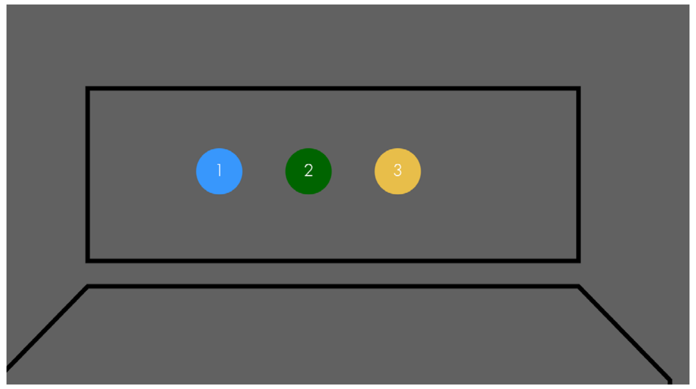


Figure 61: objects without any previews.

7.8.2 | floatable windows

In some cases, controls of different control modes might be required besides each other. The GUI therefore works with floatable control windows. Control windows can then be placed and rearranged for every desired scenario.

This also allows users to rearrange the positions of all windows. In the case that users are left-handed, it is more desirable to have the control windows on the right side of the screen. Otherwise he will constantly block the overview when operating the controls with a touchscreen.

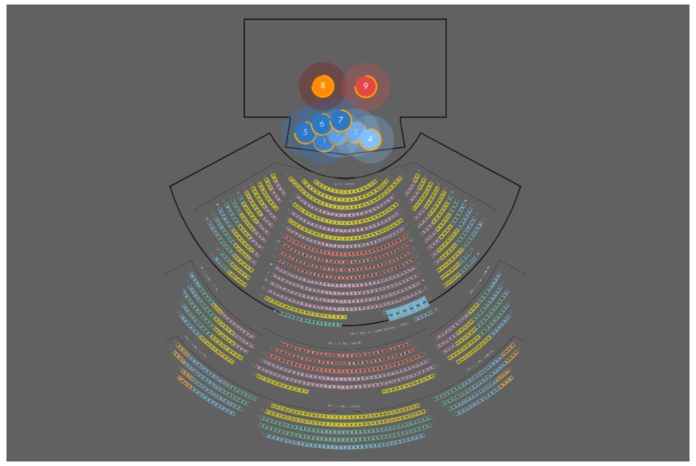


Figure 62: imported background to display audience - original retrieved from Nationale Opera & Ballet.

7.9 | ADDITIONAL FUNCTIONS

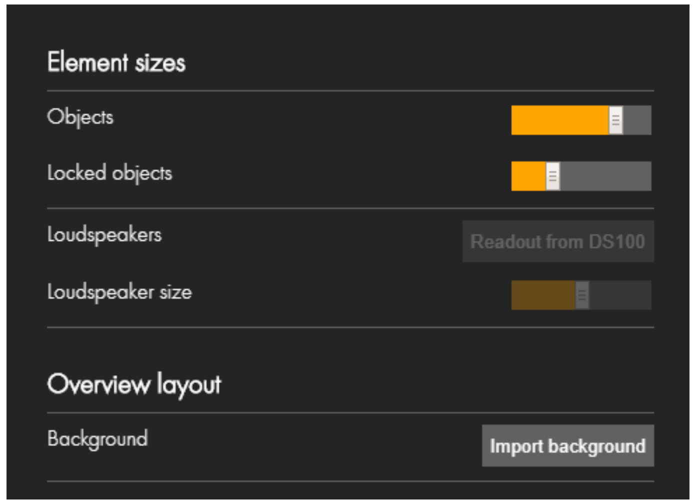


Figure 63: viewing configurations.

7.9.1 | viewing configurations

Some extra viewing configurations can be set in the viewing configurations. The size of the objects and locked objects can be set.

In the case the application is connected to a DS100, the loudspeakers of the loudspeaker setup can be previewed. This option is only available as a readout from a connected DS100. This was chosen as loudspeaker positions are otherwise a distraction from the listening experience. The philosophy is that the instruments and their positions are part of the listening experience, rather than the loudspeakers.

Lastly, a background can be placed behind the overview. This can be useful for example when sound objects have to be positioned at the musicians of an orchestra.

7.9.2 | scaling functions

An important aspect of object-based audio mixing is the ability to seamlessly adapt projects for different PA systems. The necessity of this has been depicted in the second use case in chapter 4.2.2: Touring band. The independence of the loudspeaker setup reaches beyond the preproduction in a studio. However, at some point, the production has to be scaled to varying loudspeaker setups. Not only positions, but also movements of positions should be able to be scaled.

An algorithm has been defined and tested over multiple iterations to scale positions in relation to the audience and stage to another audience and stage. The details of the development of this algorithm can be found in appendix E1: Scaling algorithm.

This scaling algorithm will work almost entirely automatic. The only input of a user defines the amount of logarithmic scaling for distanced objects. Besides, it comes with two requirements in order to be able to function.

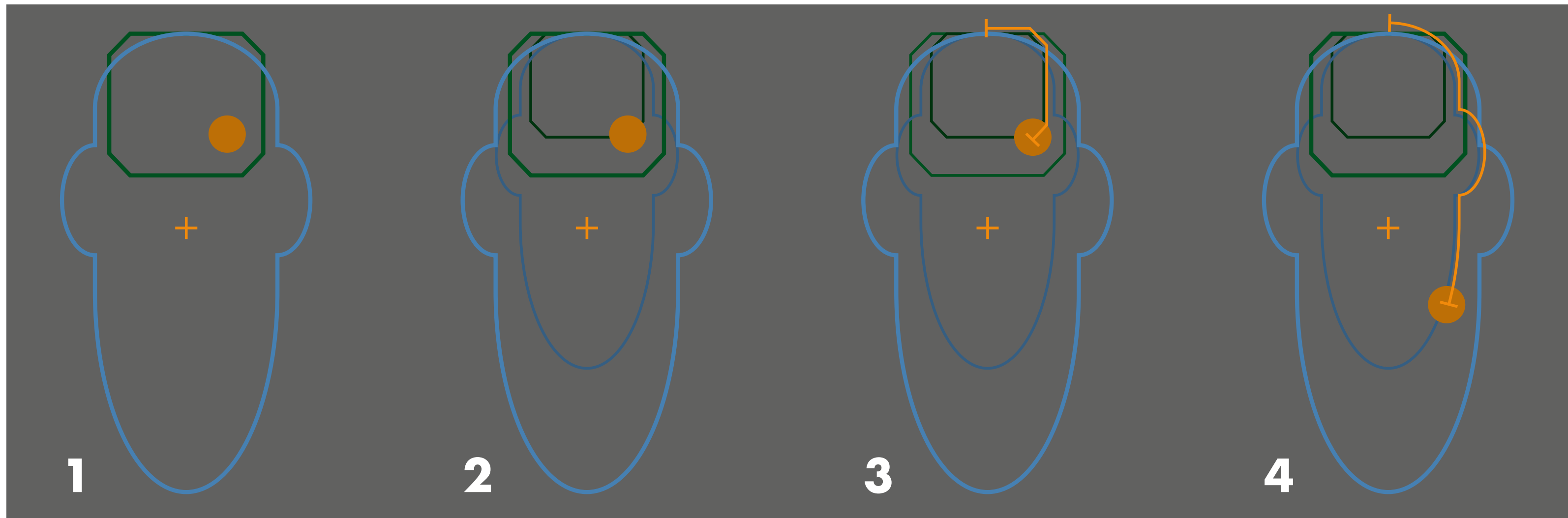
- The venue must be symmetrical.
- The ArrayCalc file must present a 2-dimensional image from the boundary edges of the venue's speaker layout.

In addition, another remark on the algorithm is the large amount of automation in itself. It has been noticed in the observational studies that mixers sometimes prefer to control everything themselves.

"So, it's nice for me, because I like that nerdy stuff. It's not out of the box." – Serge Gräfe on being in control over his mix

To tackle the exemptions, that do not fulfil both requirements, or in the case a mixer prefers to scale the production, manual scaling should still be available in addition to the automatic scaling algorithm. How manual scaling would operate is defined in appendix E2: Manual scaling.

Figure 64: operation of the scaling algorithm.



7.9.3 | source and object matrix

In the third use case that was presented in chapter 4.2.3: Corporate presentation, the show is preconfigured without access to the venue. As the loudspeaker setup is installed permanently, there is no room for adjustments in the loudspeaker setup. The case might appear in which objects are arranged to the wrong source.

In this case, the rearrangement of input sources to objects should be possible. A matrix has been included in the GUI that rearranges the DS100's input sources to the GUI's objects. This functionality was also seen in Wave Performer, the OBA mixing software by Sonic Emotion (Sonic Emotion, 2015).

This rearrangement is saveable in a configuration file. The configuration file can then be set one time by the system integrator and passed on towards the audio engineer.

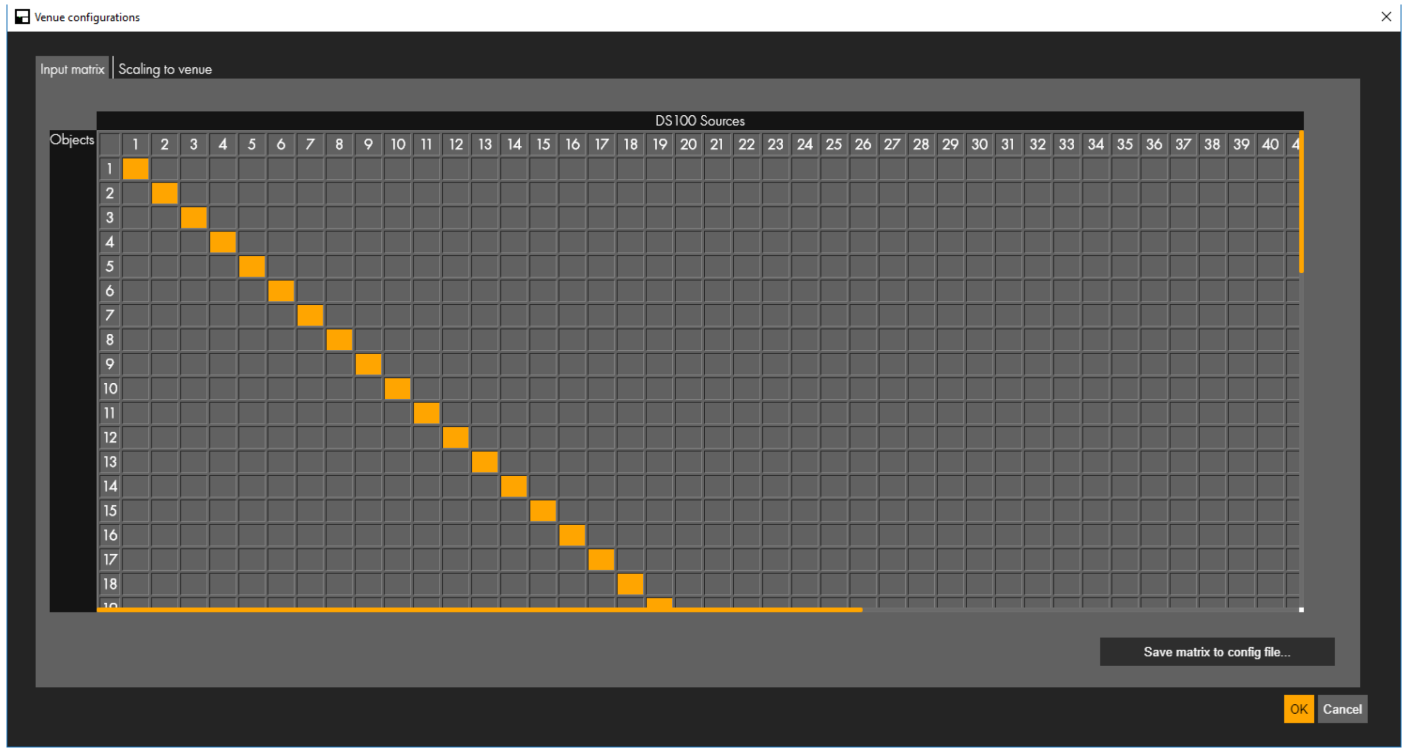


Figure 65: sound source to sound object reconfiguration matrix.

7.10 | TOUCH CONTROL

"Because then I just can grab a sound source and move it around with my finger on the iPad, and I can leave it there. Because I also have to focus on stage." – Serge Gräfe

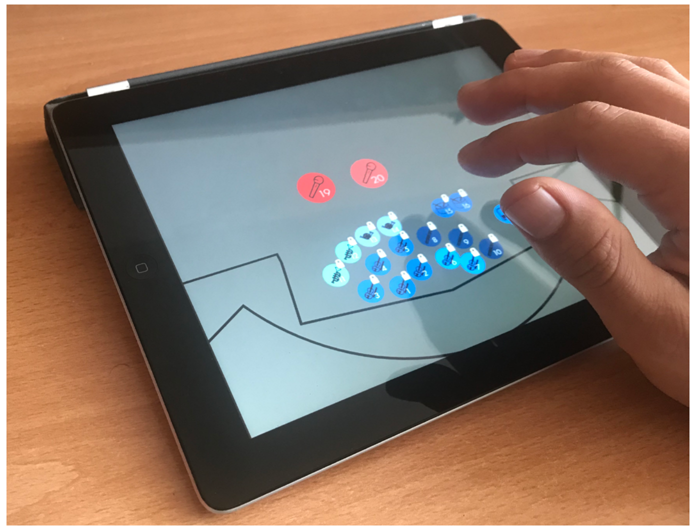


Figure 66: touchable control over object positioning.

As audio mixers have multiple tasks at hand, touch control for live object positioning is favoured. Positioning objects with touch is quick and precise.

The object positioning controls have been designed so that they are also available with touch control. This has not specifically been taken into account for other controls. As the added value of touch only applies to object positioning, only this feature is available for extensions of the application in touch environments.

Multi-selection is possible through object groups. The transformation box around multiple objects allow these transformations to be accessible through touch just as well as with a cursor.

7.11 | GUI WORKFLOW

The choice was made to separate the controls over objects in three modes. These modes were based upon different phases within an audio engineer's workflow.

- In the first mode, **overview configuration**, objects are placed in the overview and made recognizable to their linked sound source.
- In the last mode, **live control**, objects and their properties are controlled. The room emulation is also set and controlled.
- The **automation mode** falls in between these two modes. For some performance, no automations are required. Automations are set up before the performance, and triggered in live control mode.

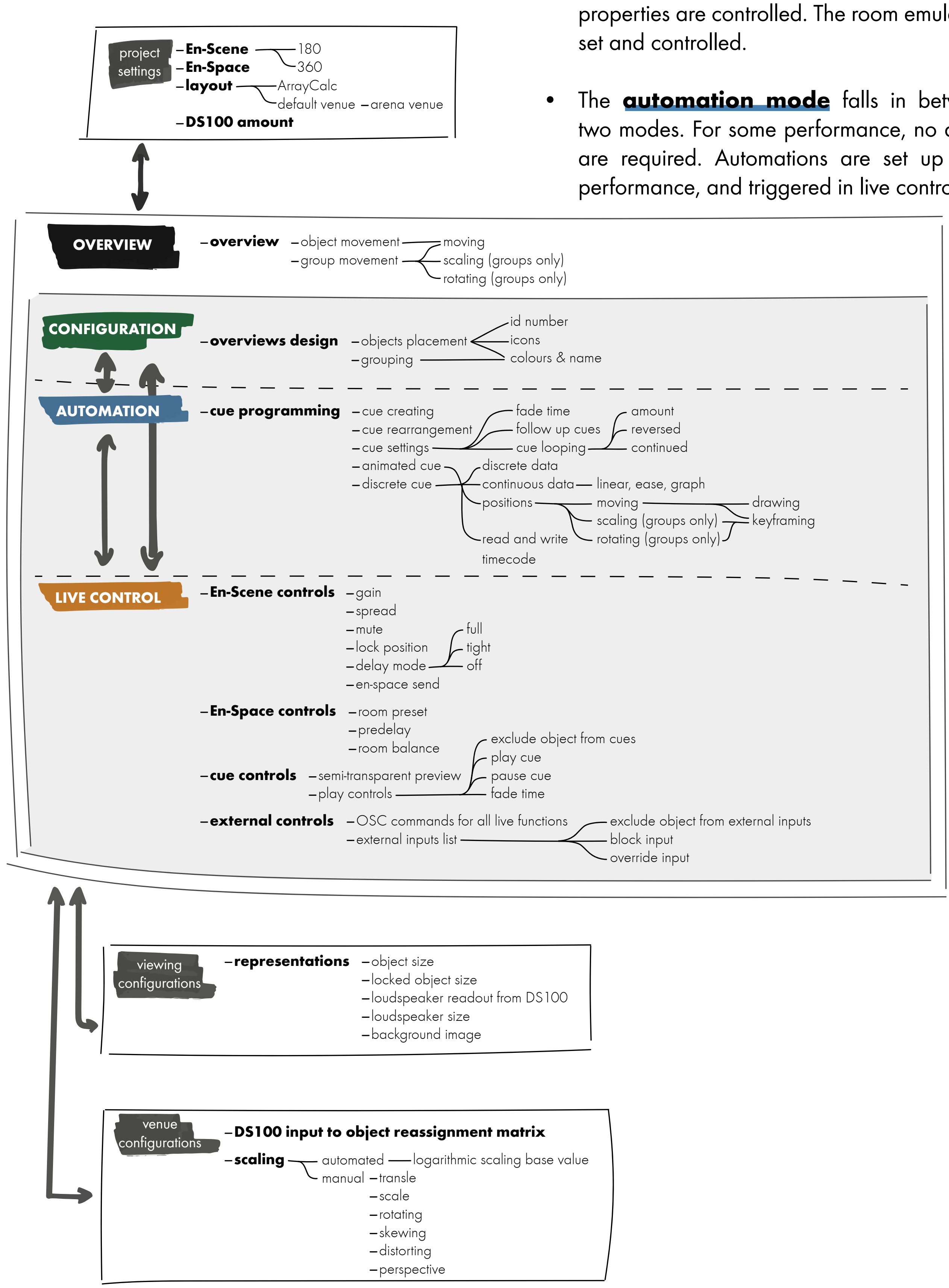
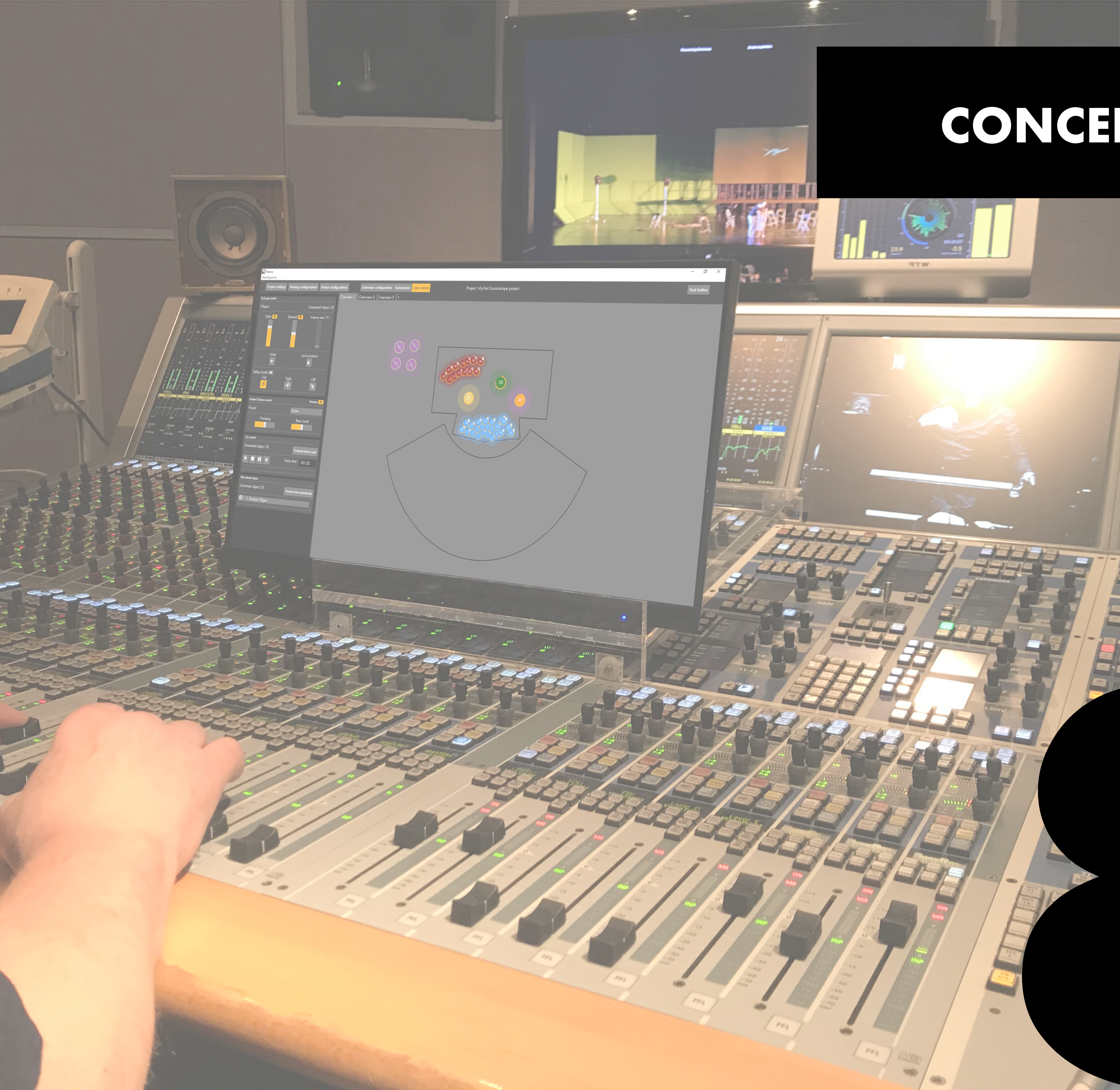


Figure 67: workflow between controls within the GUI.



CONCEPT EVALUATION

8.

8.1 | MOCK-UP

A mock-up of the concept GUI has been developed with most functionalities as they have been described in chapter 7: Concept elaboration. The mock-up only works visually and does not interact with any immersive audio system. Its sole function is to present, test and discuss the integrated functionalities with stakeholders.

8.2 | EVALUATION STUDY

An evaluation study of the concept GUI was executed to gain insights on how well the design vision would be realized through the worked-out concept. The study contained an observation of the current working process of an audio technician combined with an interview. It also entailed the showing of the mock-up which was followed with a discussion.

The evaluation study was done with **Paul Lardenoije**, who is an audio engineer at the Dutch National Opera & Ballet.

The complete setup and execution of the concept evaluation study can be found in appendix F1: Evaluation study - setup and execution.
The complete transcription of the interview can be found in appendix F2: Evaluation study - Transcription.

8.2.1 | setup

The study was set up to be executed during an opera rehearsal in the theatre of the Dutch National Opera & Ballet. During the rehearsal, the audio engineer's responsibility was to prepare and rehearse the mix for the recording of the performance.

The choice to execute the evaluation study during an opera was stimulated by the complexity of such a performance. An opera contains audio sources of the orchestra that come from the orchestra pit. On stage there are moving actors, which are recorded with wireless microphones. The combination of these audio sources creates a complexity in which object-based audio mixing can offer an easy alternative. Similar to the argumentation in chapter 4.1.3: Dynamic and static, the complexity of the performance was expected to make the evaluation stricter, as then the concept is evaluated based upon the requirements of a more demanding performance.

Figure 68: mixing studio at the Dutch National Opera & Ballet.

8.2.2 | integration of control modes

The workflow of the audio engineer in this use-case was researched and laid out into the figure 69. How the control modes of the concept would be able to be used in different phases has also been added.

The rehearsal of the performance starts without an orchestra. The audio engineer then follows the actors on stage and makes notes on his music score. In this phase, the required audio sources are known and can be placed as objects. First automations can be programmed, but are not always definitive.

During later rehearsals, when the orchestra joins, the mix is finetuned. Moving elements are then set definitively and rehearsed. Thus, object automations can be programmed. This is a very logical progress, as predefined movements always have to be rehearsed by the actors, there will always be a rehearsal period which can be used by the audio engineer to program the automations. During the latest rehearsals, the performance is played completely, and its mix can be controlled in live control mode.

During the performance, the live control mode can be used to control the mix as the performance is given.

It became apparent that the workflow is a gradual process and thus would sometimes require the switching between control modes.

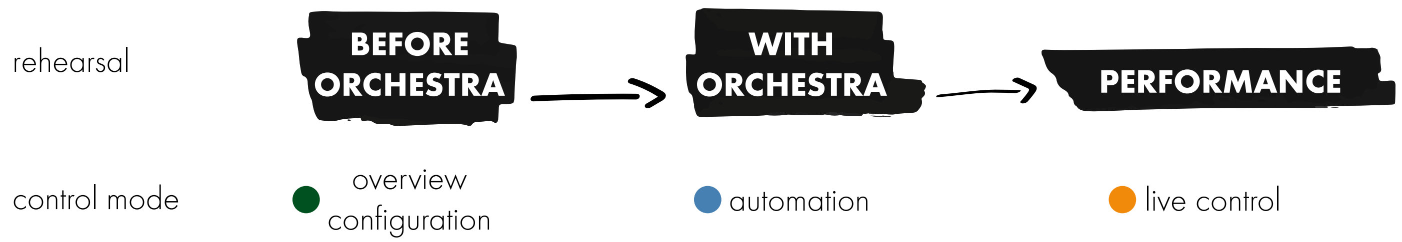


Figure 69: process of rehearsals with matching control modes.

8.2.3 | integration of functions

To determine if the added functionalities of the GUI can be integrated into all other tasks that the audio engineer executes, it was evaluated what tasks would be replaced.

Figure 70: digital score for the mixer to follow the rehearsal.

"or in a mono mix. We are talking about live. And stereo doesn't work in big venues." - Serge Gräfe

panning

Without immersive audio, space would be created in a stereo mix with the use of a panning function. (Audio Engineering Society, 2020). The panning for each sound source is replaced by the placement of sound objects in the x-direction within the application's overview.

However, as was discussed in the earlier observational studies, panning is not always used, as mixes are also made in mono. Panning between the left and right loudspeaker is then not applicable, thus not replaceable.

delay
To give sound sources the correct spaciousness in a mix, a delay is also added to each sound source.

This is done by measuring the distance to each microphone, from which the compensational delay is derived. This is a very lengthy process to precisely set the details of the mix. This is replaced by correctly placing the sound objects in the y-direction in the overview.

"I have not yet delayed anything in the orchestra. (...) I have such a rangefinder. In a little bit, I will measure a little bit. And then see how far they are." - Paul Lardenoije

reverb
Adding a reverb over all the audio sources is replaced by the use of En-Space room emulation.

"Of course, I also do the reverb." - Paul Lardenoije

Thus, the placement of objects in the overview replaces the delaying of sound sources and, dependant of the size of the mix, possibly also their panning. The use of En-Space room emulation replaces the use of other reverberation effects over all audio sources. As previous tasks are replaced, this makes room for the GUI's improved interaction.

"If it immediately compensates for delay and everything. That is great." - Paul Lardenoije

The mock-up was very liked in this test and seemed to provide the right functionalities for OBA mixing. Adjusting the workflow within the concept to the limited amount of time available to an audio technician makes the concept fruitful.

8.2.2 | cues
The use of cues within the concept was found to match the current workflow. The use of cues is possible in the mixing desk that was used. Discrete cues that have no time dimension are called snapshots. Animated cues that do have a time dimension are called scenes.

As these functionalities are already existing in the workflow of an audio engineer, they have a better chance to integrate into this existing workflow (Becker, 2010).

Figure 71: installing microphones above the stage at the Dutch National Opera & Ballet.

8.3 | FUNCTION ANALYSIS

From the function analysis, a list of functions was derived, which included a definition of quality for each function. In this chapter, these definitions of quality are compared to how the concept addresses these qualities.

The complete comparison to the function analysis with the definitions of quality can be found in appendix H: Function analysis comparison.

FUNCTIONALITY	REALIZATION
1. live mastering en-scene & en-space	Object-based overview that is always **present**, independent of the controls. Includes the option to hide and show all object's parameters. En-Space effects are represented **before and during** selection of a sampled space.
2. object configurations	The appearance of an object is customizable through its **source number, colour and an icon**. Selectable colours are arranged in five sets of **shades**, so that grouped objects have analogous colours.
3. automation	All En-Scene and En-Space parameters are adjustable beforehand with the **same way of input** as in live control mode. Cues are arrangeable at all times. Starting external devices happens with timecode. Cues can be started by **external triggers**. Complete scenes can be faded to with the **fade-in time** of a cue.
4. automation live control	Semi-transparent **preview** of cue upon selecting. Cues can be played and paused at the press of one button.
5. scaling	Takes into account project settings to guide scaling. Uses an **algorithm** to scale objects. **Extensive transform options** are integrated for exceptions in which the algorithm does not work.
6. external linking	A **list** of external inputs is shown. Inputs can be blocked and objects can be excluded from external inputs. Playing cues is accessible as a function through OSC and animated cues play a timecode that can serve as master.
7. additional	Input sources to object matrix allows **reassignment**. Loudspeaker setup can only be **readout** from the DS100, so that it does not obtrude during the production of a performance. Ability to import and transform background image.

RECOMMENDATIONS

9.

9.1 | WHAT IS MISSING NOW?

The concept that has been presented in this thesis provides insights into how a digital tool for mixing audio for immersive audio systems might look. In the research that was done to develop the concept, other topics were found that are relevant to address in the development of such a tool. These topics are not addressed by the concept that was developed, but are discussed here.

9.1.1 | application crash

In the case that the application crashes, the objects will be in a certain position without the application knowing where. If the application is restarted, the objects should not immediately jump to the initial positions in which the application starts.

An online and offline mode is used in R1 (d&b audiotechnik, 2011) to counter this problem. The values can be read out in the offline mode. After that, the application is set in an online mode to change the values.

As this option is already being used in R1, it is advantageous to adopt this function in a similar way in the concept. This would avoid the necessity of relearning a different approach for experienced users, thus increase the chance of success of the concept (Becker, 2010).

Figure 72: online mode in the R1 toolbar - original retrieved from d&b audiotechnik.

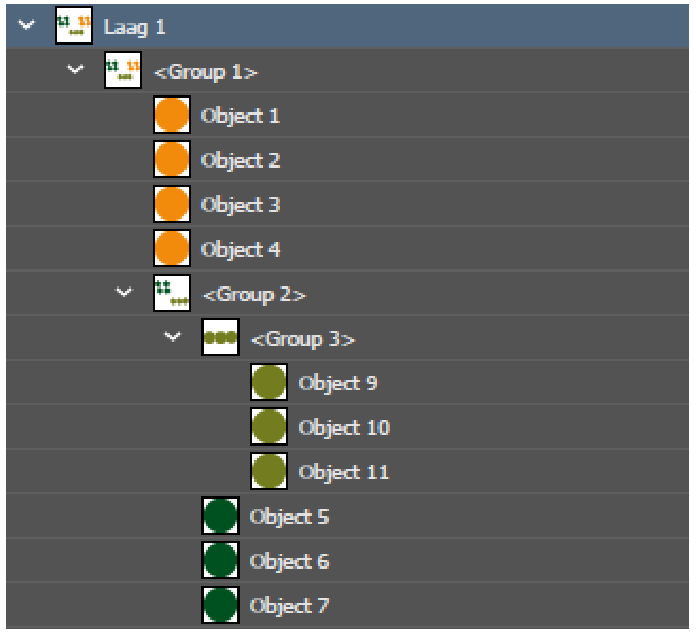


Figure 73: layers of groups within each other as listed in Adobe Illustrator - original retrieved from Adobe Illustrator.

9.1.2 | object group layers

Objects can be grouped in the overview. The additional value of grouping objects, is that all objects are multi-selected at once. What has not been researched in this project is the necessity for being able to create multiple layers of groups. As the amount of layers increases, the complexity of a group will increase.

Whether there is a necessity for a high amount of layers within object groups, is leading in determining the amount of layers. A study among audio content designers in different disciplines among the application focus can determine this necessity.

What is also valuable knowledge in determining the amount of layers that is available, is how the complexity increases when multiple layers are used. Potential solutions to decrease the complexity should be researched.

In Adobe Illustrator, drawings can be grouped. To deduct complexity, the user is given a list of the layers and groups that he has created. Drawings and groups can also be selected from the list.

9.2 | FURTHER FUTURE

Some use cases have come up during the project, that are not addressed in this project. As OBA mixing is still in a very early phase, the priority of the concept lies at the functionalities that have been described in this report. However, the use cases that are not addressed now, will be described here to guide the development of an immersive audio mastering tool in the further future.

9.2.1 | spatial mixing effects

It was concluded that the placement of objects serves to replace the panning and delaying function of individual sound sources. However, an audio engineer has much more functions at his command to alter a mix. Examples of such functions are compression, equalizers and limiting (Owsinski, 1999).

During the concept evaluation at the Dutch National Opera & Ballet, a discussion was held about potential use cases in which the spatial placement of these additional mixing functions could be useful.

Paul Lardenoije gave an example of the placement of an equalizer in a spatial overview. The walls besides the stage at the building where he often works are very high, and create a lot of reflections of the sound when actors approach it. These reflections are only in the higher frequencies. He then uses an equalizer to reduce the high frequencies in these cases. Placing this equalizer in the spatial plane, would automate the addition of this function. This would be extra useful for live mixing, as the walls are stationary. This function would then take away the necessity to monitor the actors and manually add the equalization.

9.2.2 | 3-dimensional plot

As was discussed in chapter 5.1.1: Object-based audio software, other OBA mixing applications are able to plot a three dimensional overview of the venue with the objects in it. A difference in reproduction will only be heard if loudspeakers are set up at multiple heights.

In the case that a loudspeaker setup allows this, a three dimensional plot of the venue creates a better insight into the listening experience in the venue.

As the DS100 requires that the loudspeaker setup is drawn in ArrayCalc, this drawing can be imported in the GUI. A plot can then be made in the GUI by the use of this drawing.

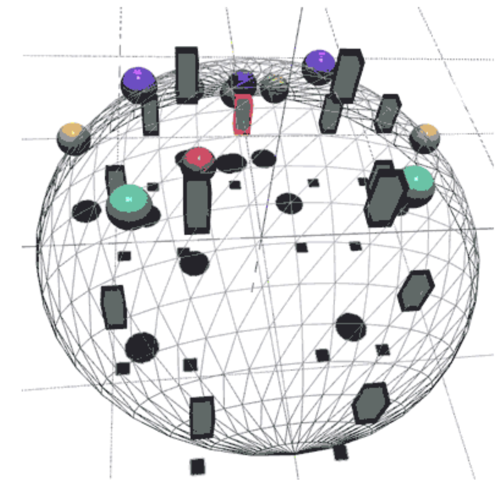


Figure 74: 3-dimensional plot of objects and loudspeakers - original retrieved from Astro Spatial Audio.

9.2.3 | loudspeaker groups

As described in chapter 4.3: Content designers, if we look at the monitor technician, his responsibility is to create a mix for the artists on stage. Often, he uses a set of monitor loudspeakers for this. In other cases, the artists will receive earbuds. To independently create a mix for the artists, it would be valuable for the monitor technician to exclude all other speakers from his mix. The mix for the audience can then also exclude the monitor loudspeakers.

The functionality to address a set of loudspeakers per set of objects would allow the monitor technician to easily create a monitor mix independently from the FoH technician.

10. PERSONAL REFLECTION

If I reflect on this project, I can say with much certainty that it has allowed me to develop myself in multiple different ways.

Maybe the most obvious difference between this project and other projects in my portfolio, is that the concept entails a digital tool. Despite that this is not completely new to me, the vast majority of work I have done in the previous part of my study entailed physical product development. I have been fascinated by both the differences as well as the similarities between developing a digital product and developing a physical product.

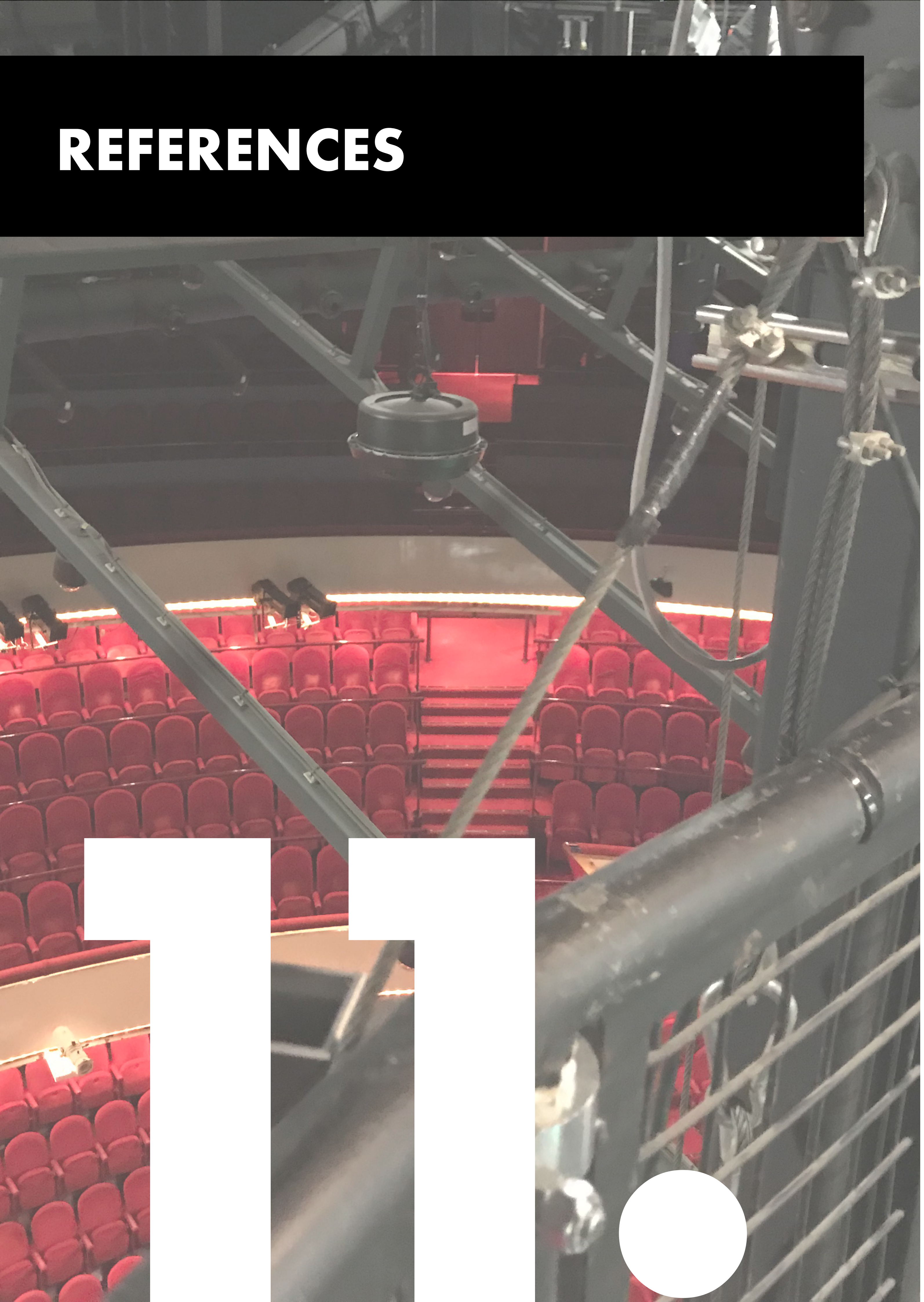
A difference that I had trouble to adapt to, in the beginning, is the large difference between the amount of functionalities that the end product provides. For the development of a physical product, the amount of functionalities that a product has, is often rather small. The ease at which more and more functions come up and are added to the product in a digital environment, require a continuous very sharp focus of the developers. To decide whether additional functionalities steer towards or away from the predetermined intention of the end product is then a much larger task, than in physical product design.

Another fact of the development within this project that surprised me, is that the project does not lead to new functionalities. All functionalities of the concept that have been presented and discussed, are already possible in a certain way. The goal of the concept is to bundle all functionalities and present them in an analogous way to its users.

The prototyping in a digital environment demonstrates a prompting analogy between digital and physical product development. When a prototype is made of an algorithm, the goal is to make a set of calculations, which is completely defined. The algorithm can not cope with exceptions, as it then fails to give the right result. Exceptions can only be allowed, if they are also defined beforehand. These exceptions then do not make use of the prototyped algorithm. The clarity of the importance of defining a prototype and the relevant exceptions completely, sometimes lacks in physical product development. To be able to make a proper physical prototype, also requires its definition in the highest completeness as possible. Working in a digital environment made me realize this importance.

What I have enjoyed very much during the whole duration of this project is that I was fortunate to work on a cutting-edge technology in the field of my interest. Not only was this astonishing, it also taught me hands-on, how I can apply my education in a company, and make it directly useful. It showed me the value of what I am able to work on for the company, but also for the end users that were involved in the process.

All in all, I can look back on the project with much satisfaction, as I am proud of the end result. It has served me very well as the last project in my study.



11. REFERENCES

1. Adobe. (n.d.). Use camera animations. Retrieved 2019, from https://helpx.adobe.com/illustrator/how-to/edit-artwork-basics.html

2. Adobe Systems. (2020). Adobe After Effects 17.0.6. Mountain View, CA.

3. Adobe Systems. (2020). Adobe Illustrator 24.0.1. Mountain View, CA.

4. Audio Engineering Society. (n.d.). Pro Audio Reference (P). Retrieved March 23, 2020, from http://www.aes.org/par/p/#pan

5. Astro Spatial Audio. (n.d.). Retrieved 2020, from https://www.astroaudio.eu/products/concept

6. Astro Spatial Audio. (n.d.). Retrieved 2019, from https://www.astroaudio.eu/products/software/production

7. Beck, L. (n.d.). Concert Hall, Vienna – Schubert Hall. Retrieved from https://www.dbsoundscape.com/global/en/system-profile/en-space/

8. Becker, K. (2010). Facilitating unlearning during implementation of new technology. Journal of Organizational Change Management, 23(3), 251–268. doi: 10.1108/09534811011049590

9. Berkhout, A. J., Vries, D. D., & Vogel, P. (1993). Acoustic control by wave field synthesis. The Journal of the Acoustical Society of America, 93(5), 2764–2778. doi: 10.1121/1.405852

10. Blauert, J. (1997). Spatial hearing: the psychophysics of human sound localization. London: MIT Press.

11. Boeijen, A. van, Daalhuizen, J., Zijlstra, J., Schoor-Rombouts, R. van der., Zijlstra, Y., & Kuntonen, J. (2013). Delft design guide: design methods, Delft University of Technoloy, faculty of industrial design engineering. Amsterdam: BIS Publishers.

12. Carver, J., Shull, F., & Basili, V. (2006). Can observational techniques help novices overcome the software inspection learning curve? An empirical investigation. Empirical Software Engineering, 11(4), 523–539. doi: 10.1007/s10664-006-9021-5

13. d&b audiotechnik. (2019). TI 501 d&b Soundscape 1.6. Retrieved from https://www.dbsoundscape.com/assets/products/downloads/ti/dbaudio-technical-information-ti501-dbsoundscape-1.6-en.pdf

14. d&b audiotechnik. (2020, January 21). The brand-new d&b Soundscape – More art. Less noise. Retrieved from https://www.dbsoundscape.com/global/en/system-profile/en-space/

15. d&b audiotechnik. (2020, April 16). R1 Remote control software. Retrieved from https://www.dbaudio.com/global/en/products/software/r1/#tab-tutorials

16. d&b audiotechnik. (2011). Retrieved from https://www.dbaudio.com/assets/products/downloads/out-of-production/dbaudio-manual-r10-v2.0.x-en.pdf

17. Dolby Laboratories. (2013). Authoring for Dolby Atmos Cinema Sound Manual. Retrieved from https://www.dolby.com/uploadedFiles/Assets/US/Doc/Professional/Authoring_for_Dolby_Atmos_Cinema_Sound_Manual(1).pdf

18. Goodwin, N. C. (1987). Functionality and usability. Communications of the ACM, 30(3), 229–233. doi: 10.1145/214748.214758

19. HeidelbergCement. (n.d.). Concert Hall Blaibach, Germany. Retrieved from https://www.dbsoundscape.com/global/en/system-profile/en-space/

20. Melchior, F., Michaelis, U., & Steffens, R. (2011). Spatial Mastering - a new concept for spatial sound design in object-based audio scenes. Retrieved from https://www.researchgate.net/publication/267840563_Spatial_Mastering_-_a_new_concept_for_spatial_sound_design_in_object-based_audio_scenes

21. Mulcahy, J. (2019). RT60 Graph. Retrieved March 31, 2020, from https://www.roomeqwizard.com/help/help_en-GB/html/graph_rt60.html

22. Naef, M., Staadt, O., & Gross, M. (2002). Spatialized audio rendering for immersive virtual environments. Proceedings of the ACM Symposium on Virtual Reality Software and Technology - VRST 02. doi: 10.1145/585740.585752

23. Nationale Opera & Ballet. (2019, September 3). Zaalplattegrond. Retrieved from https://www.operaballet.nl/nl/plan-uw-bezoek/tickets/zaalplattegrond

24. Owsinski, B. (1999). The mixing engineers handbook. Vallejo, CA: MixBooks.

25. Qlab. (n.d.). QLab Overview. Retrieved 2019, from https://qlab.app/overview/

26. Roeser, R. J., Valente, M., & Hosford-Dunn, H. (2007). Audiology. New York: Thieme.

27. Sanders, E. B.-N. (2001). Virtuosos of the Experience Domain. Retrieved from http://echo.iat.sfu.ca/library/sanders_01_virtuosos.pdf

28. Singh, A. (2019). Participant Observation: IDE Academy Course.

29. Sonic Emotion. (2015). WaveISuite UserManual v4.2. Retrieved from http://admin.stagetec.com.cn/resource/WaveISuite_UserManual_v4.2.pdf

30. Surround sound. (2020, January 30). Retrieved from https://en.wikipedia.org/wiki/Surround_sound

31. Tassoul, M. (2009). Creative facilitation. Delft: VSSD.

32. Wallach, H., Newman, E. B., & Rosenzweig, M. R. (1949). "The precedence effect in sound localization," The American Journal of Psychology, 62, 315–336.

references from appendices

33. Cockos Incorporated. (2020). Reaper 6.08. New York, NY.

34. IanniX Association. (2019). IanniX 0.9.20.

35. Imimot. (n.d.). Vezér. Retrieved 2019, from https://imimot.com/vezer/

36. Lightjams inc. (2019). Lightjams. Montreal

37. Madrix Lighting Control. (2019). Madrix 5.2.

Concept development for an immersive audio mastering application.
Designing a tool for object-based audio mixing.

Master thesis | Stan Vugs