PA Design and Optimization

A closer look to the factors and steps needed to achieve
a good sound.

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ABSTRACT

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Live sound reinforcement has become better through time as new technologies have emerged and new equipment have been designed. The theory behind the design and optimization of sound systems is the same but, as time goes by, the tools used for the job keep changing. In this thesis, the target is to analyze, study and examine the main techniques used when designing and optimizing a PA for a certain venue or space.

As the pre-production part of the events is so important for the design of the sound system, a brief description of how everything starts is included. This description will cover the main factors that might affect how one might choose the needed gear for the occasion.

This thesis will be focused on music events, defined as events where a singer or a band is performing. Also, since Finland is such a small country and the two largest companies use mostly the same brands and equipment, the main software used for describing and presenting examples will be narrowed down to d&b audiotechnik’s R1 and ArrayCalc for the design, Rational acoustic’s Smaart and d&d audiotechnik’s R1 for the Optimization.

This thesis also includes a description of the actual setup. During the setup one brings to life the design and, depending on many factors that I will mention during this thesis, this tends to be the moment for sudden changes. This is also the moment to ensure the design and the actual setup are a good match.

This thesis will not be an in-depth description of the technical aspects nor a scientific analysis of sound. This study is an attempt to provide future students interested in live reinforcement with an easy-to-read, yet informative, go-to paper to read in order to start working in this field.

Key words: PA, live sound, design, optimization, pre-production.

CONTENTS

GLOSSARY

PA	Public address system, is an electronic system that amplifies and distributes sound and it is used to address audiences.
Sound	"*A vibration or mechanical wave that is an oscillation of pressure (a vibration back and forth) transmitted through some medium (such as air), composed of frequencies within the range of hearing*" (McCarthy 2016.)
System	A set of interactive or interdependent components forming an integrated whole.
Optimization	A scientific process whose goal is to achieve the best results when given a variety of options.
Audio	Stream of data beginning and ending as sound.
Phase	Defined as the difference in time-vs amplitude between two audio sources.
Beamwidth	A characterization of speaker directional response over frequency.
Frequency Response	The charted output of an audio device.
Horizontal dispersion	The speaker's characteristic sound projection pattern, broad or narrow.
Spectrograph	A three-dimensional data plot, displayed in two dimensions with color representing the third dimension (or z-axis). The spectrograph is a topographical representation of the once-common waterfall display. (Smaart V8, Manual)

1 INTRODUCTION

Music has been a really big part of my life since I was 13. I have performed in many good sounding places and also in many places where, for some reason, I could not hear myself properly when performing, or the bands when being in the audience. Through time I developed an interest in sound, the theory behind it and its behaviour.

Before I started working in live events in Finland I tried to find some books related to this topic and I didn’t succeed, the books were too complicated theoretically or more dedicated to the science behind sound. This thesis is a guide on how to design and optimize a sound system without going too deep into the theoretical aspects, lets call it a 101 guide. It is 2017 and there are still sound technicians that believe that if you put a PA high enough and loud enough it should work. Let’s fix that. Sound matters. It is always better to make the sound pressure as even as you can throughout the venue when possible.

The practical part of my thesis was done during the summer of 2016. I spent time rigging PAs, setting up subs, rigging subs, being part of some of the biggest festival productions in Finland. I also took part in a seminar done by d&b audiotechnik in Akun Tehdas about system design and I helped the system designers with the optimization for Eppu Normaali’s 40 years old gig at Ratina stadium in Tampere.

2 HOW EVERYTHING STARTS

Before starting with the design, set up and optimization, I will describe what happens prior to that. In a perfect world one could decide the gear to be used in a concert but the reality is a bit different. There are certain factors that play a role when deciding the overall production and I will describe them next.

2.1 Pre-Production

In order to create a successful event sound-wise there are many things one have to take into consideration even before thinking about the design of your overall gig. There are many aspects to take into consideration when planning an event, these might be crucial when designing a sound system so, before I jump into describing the process of designing and optimizing a sound system, I will describe how to get the best possible situation to work in. In order to get more accurate data about the pre-production of an event I interviewed Olli Pörhölä, production manager at Akun Tehdas.

”I always look at things considering the whole production, not just sound” (Pörhölä, 2016.)

Sales

The first step when planning the overall event is to check what the salesperson has sold to the client. Everything works on a budget and the decisions on what gear will be used, is based on the negotiations between the salesman or production manager and the client. Sometimes the client has paid for a sound system that has coverage over the whole venue and, in this case, one could take decisions about the most suitable sound system for the given venue. Sometimes the client has bought a sound system where all listed equipment is already specified (Pörhölä, 2016.)

There are many possibilities when negotiating a gig. The best situation is when the production manager can be present before the deal is closed with the client, this way every single part of the event will be optimized, not just the PA but the crew, the equipment and the schedules among others.

Crew

The crew is an important part when organizing an event. *"Some people go with gear first, but in my opinion, a good crew of professionals is more important"* (Pörhölä, 2016)

One could set up the best sounding ever sound system but it means nothing unless one have the technicians that can get the most out of it, but if you do have a good crew of technicians they can make the best out of the worst set up and equipment. (Pörhölä, 2016.)

The size of the crew is also determined by the budget, sometimes it is determined in the contract with the client (Pörhölä, 2016). This determines the overall crew be it light crew, sound crew or video crew among others.

Venue

One important part of the pre-production is checking the venue. The main things to check out in the venue are its rigging possibilities, the measurements, the power limitations, the audience areas and the load in and load out routes (Pörhölä, 2016).

Schedule

Every production needs a schedule. Everybody is going to ask for it. It defines the amount of time one has for loading in, setting up, rigging a PA and optimizing. The schedule also shows which team does what and in what order (Pörhölä, 2016).

Equipment

The next step is getting the equipment, which might or not be determined in the contract. Normally one can get the equipment within the company one works in, but if the event is happening in more than one venue it might get more complicated. Sometimes one has to rent the equipment from different companies and organize its transportation and make sure everything arrives at the venue at the right time. At this point having a professional crew is vital and everyone in the production should be able to trust everybody else with the equipment. (Pörhölä, 2016.)

Other aspects

In the pre-production of the event the production manager also takes care of the transport and machinery, accommodation and meals. (Pörhölä, 2016.)

2.2 Observation

In order for an event to work sound-wise one needs to take into consideration that, at least in music, there are too many factors playing their part at the same time, be it the song, the arrangement, the acoustics, the sound crew, the light crew, the video crew, the quality of the equipment, the weather and the band gear among others, it is not just the sound system that counts for a good concert. Also, the organizers sometimes book bands in the wrong venues (it seems to be the common factor) and the result is that when a show doesn't go as expected the blame is on the sound *"Who do you think is going to take the fall for a famous band in a hall with a reputation for great acoustics?"* (McCarthy 2016). The audience will always have an opinion on the sound probably based on where they have been before and they are the ones one has to please.

"*Sound engineers are from Mars, acousticians are from Venus, audiences are from the Bronx* " (McCarthy 2016.)

2.3 Collect data

Once you know the place you are going to work in, and if the production manager didn't provide you with all the information, there is some data you can try to gather for your design like the area that needs to be covered, how high can the PA fly, how many people can fit in the venue, are they standing or sitting, do we need more than the main PA, does the place need a delay PA, side fills, front fills, what kind of sub configuration will you install, among others.

One nice tool to have while measuring a venue, if the design of the venue is not given, is a laser meter. This particular tool will allow you to measure the most difficult places and some will even show you the angle between measuring point and with simple trigonometry you could calculate almost any distance, for design or for assigning delays.

2.4 Team Work

One thing to remember while evaluating our sound system design for a small or a big venue is that in most cases sound is not the only enhancement a venue will get. The sound crew in most cases will be working along with the light crew, the video crew, security, the pyrotechnics crew, an acoustician and many others. In order for our event to be successful we don’t have to get along with everyone but, getting along and understanding each other’s needs, helps and smoothens the installation and set up of our system.

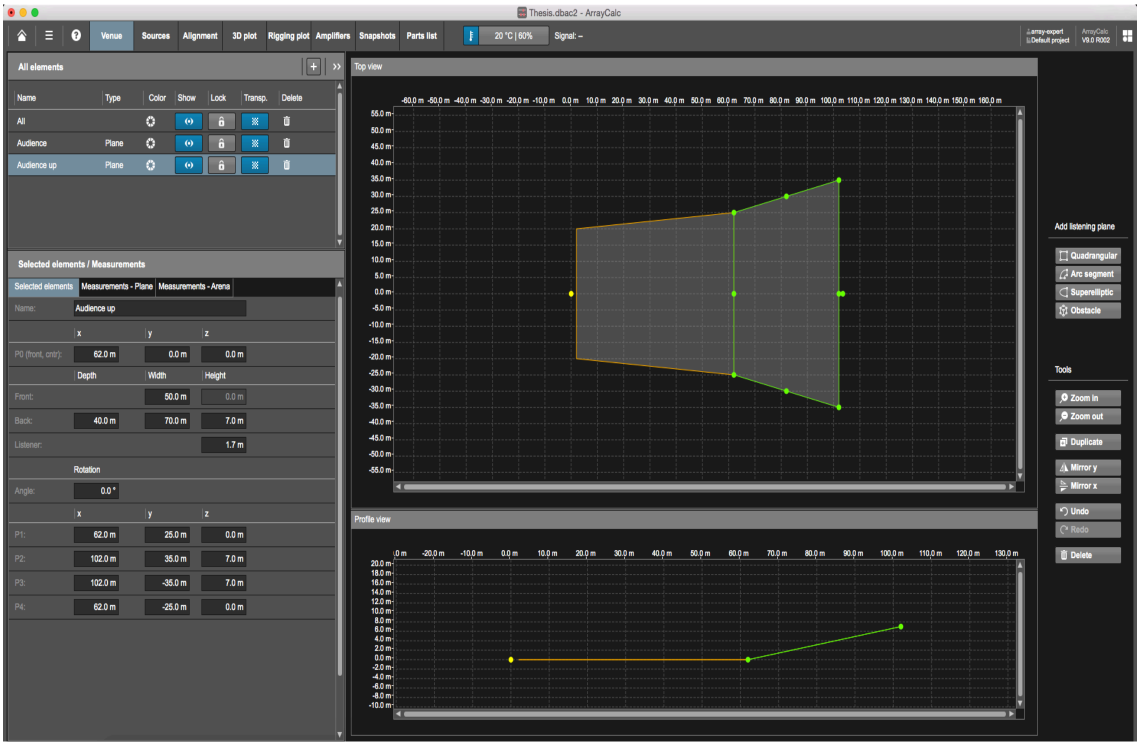
Depending on the occasion the sound might play a secondary role in the event, sometimes all that matters, for example in a TV production, is that the sound that goes to the TV is great and that the cables around the stage are somewhat invisible. In any case everything is decided beforehand and our role is to follow the common plan.

” Don’t be a jerk. Be inclusive and collaborative. This is a team sport. Don’t embarrass or humiliate anyone.” (McCarthy 2016)

3 SYSTEM DESIGN

As I mentioned before I will be explaining most of my examples using d&b audiotechnik’s ArrayCalc prediction software.

Once we know the measurements and have visited the venue, we can start designing our PA. This is the point where we make a virtual simulation on how the sound will behave with any given sound system of our choice. I will go through the overall possible operations in ArrayCalc for a simple sound system set up.



PICTURE 1: The basic layout when opening ArrayCalc by d&b audiotechnik (©d&b audiotechnik 2016)

3.1 ArrayCalc

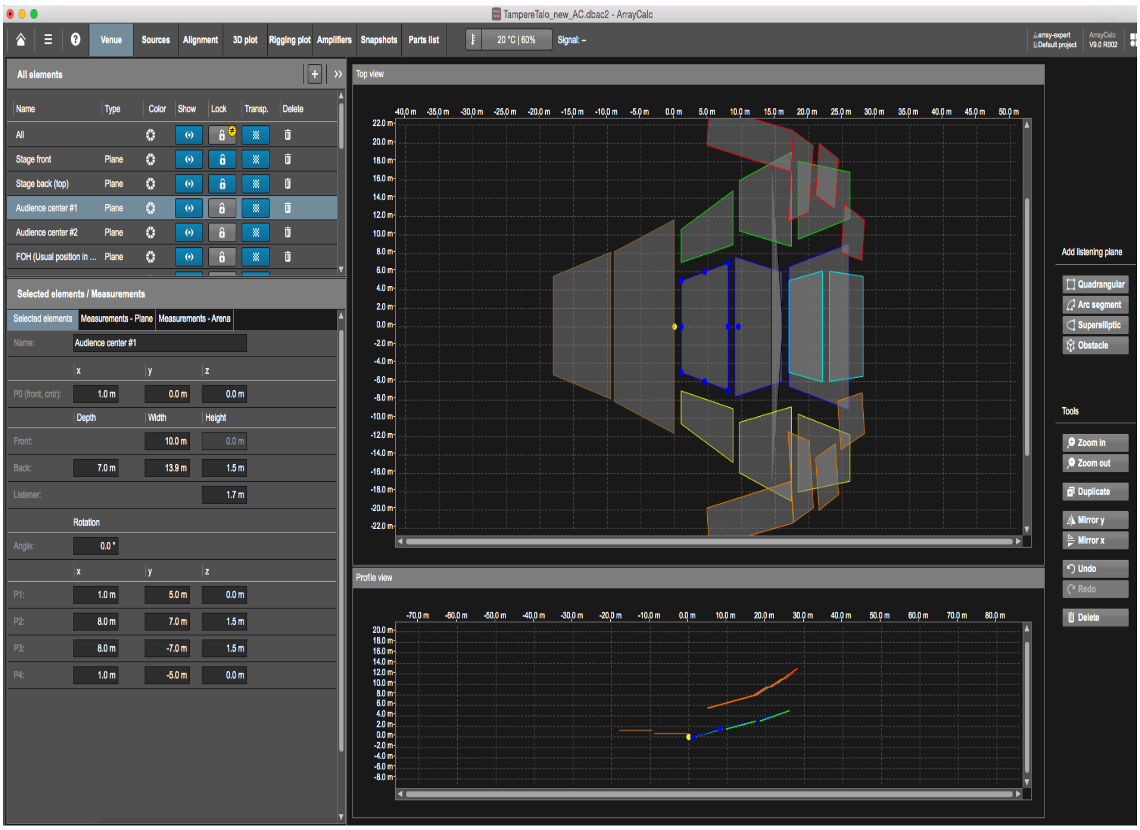
This is the most common program you will run into when designing the sound system for any particular venue in Finland. It is a good program; it is effective and it is free. Picture 1 shows the basic Layout. Now lets take some time going through its basic features since it is a big part of the system design.

3.1.1 The Venue Editor Page

As shown in picture 1 this is the place where we design the overall venue. In the venue page one can assign the size of the venue and its elements. The program itself calls them listening planes, defined as the different parts within a venue; front seats, balconies, side wings, among others. (d&b audiotechnik 2016.)

The Venue editor offers 2 useful views, the top view and the profile view. The first offers an overview of the length and width of the venue and the second allow us to see the height and the inclination of the planes. Each plane is fully adjustable.

A couple of months ago I went to a venue in Tampere, my hometown, called Tampere Talo to check the design of their venue and to check their sound system design. I spent some time measuring the place and designing the venue in ArrayCalc. Picture 2 is the result of that process and it shows clearly the naming of the planes and how the different views interact with each other.

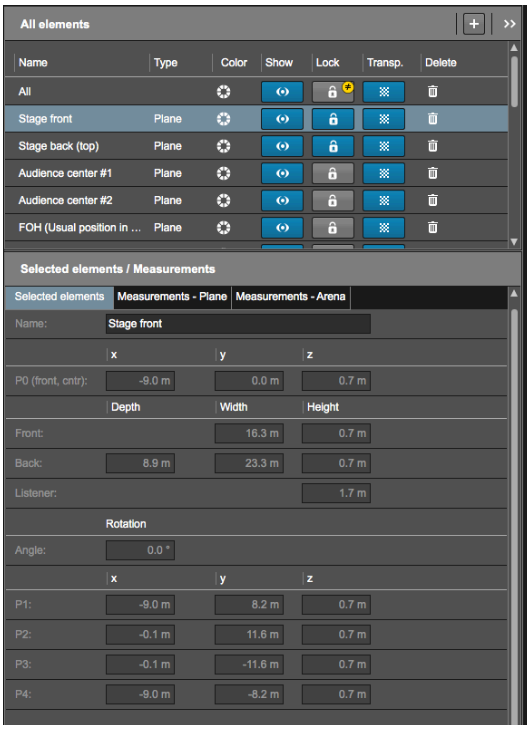


PICTURE 2: Venue design made in ArrayCalc (©d&b audiotechnik 2016)

Other options while editing your venue at the venue editor in ArrayCalc are:

- The possibility of setting up the listener's height. It is a common practice to set up a height of 1,7 meters for a standing audience and a height of 1,2 for a sitting audience.
- One can assign any plane to be transparent or not. If the option transparent is switched on the plane will not absorb the sound, which means that the beam will go "through" to the place this particular plane is shadowing. On the other hand, if the plane is not switched to transparent the plane will absorb the sound. This last option is good when simulating the places under a balcony or behind a wall.
- In case the venue has many sitting and standing areas there is the possibility of naming and colour coding the different planes

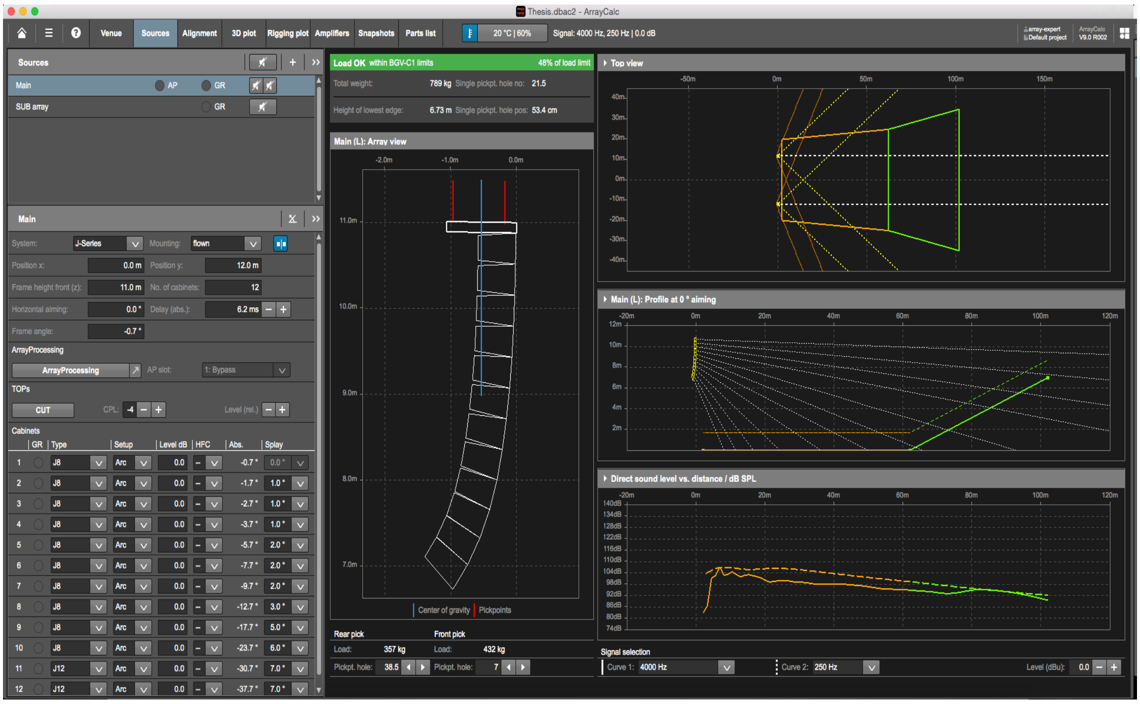
All these options are really useful and play an important role when designing a sound system in any given venue. Picture 3 shows how these options are represented in ArrayCalc's venue editor.



PICTURE 3: Options for any given plane within the venue editor in ArrayCalc (©d&b audiotechnik 2016)

3.1.2 The Sources Editor Page

The sources editor page is without a doubt the most important part of this designing process. The coverage of your sound system depends significantly on the decisions you take while configuring the sources. As shown in picture 4 there are many variables that play an important role when deciding on how many cabinets/speakers you will use.



PICTURE 4. The Sources editor page (©d&b audiotechnik 2016)

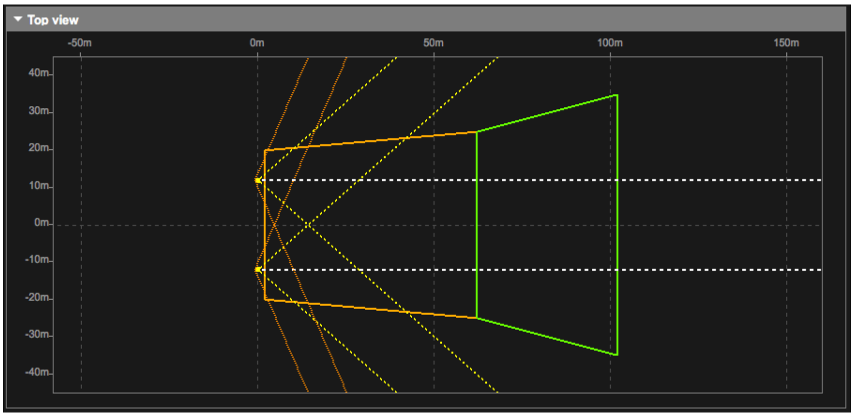
I will go through the most important options you will find while navigating the sources editor page in ArrayCalc.

When editing the main PA, you can decide here where your sound system array will be located, how many cabinets you will have, how much distance there is between the main arrays, at what height they are from the ground, which kind of cabinet will be used and the horizontal aiming among others. Also while adding more cabinets the program shows you the total weight of the array and the load limit, this way it prevents you from designing an array too heavy for your rigging system to carry. There are four graphics that represent the overall info you need to know:

- The main Array view: It shows the cabinets rigged from a profile view. It also gives the information about how to rig the frame that holds the sound system, its center of gravity and the overall mechanical set up.

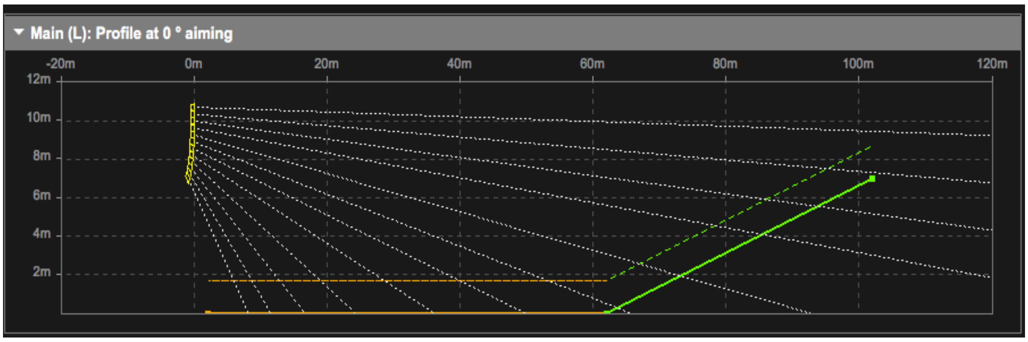
PICTURE 5: Main Array View (©d&b audiotechnik 2016)

- The top view: It displays the horizontal dispersion of the sound coming from the selected array of cabinets in this case the main PA. The white dotted line represents the main axis, the yellow dotted line the top cabinets and the orange line the bottom cabinets. In this particular example it was chosen to work with d&b J-series speakers and they have 2 versions the J8s and the J12s. The main difference between them is the horizontal dispersion which is 80° when working with J8s and 120° when working with J12s. This is the main reason why the graphic we get from the top view while using the same series of speakers differs between top cabinets and bottom cabinets. Normally the top cabinets have the narrower horizontal dispersion so the sound is more directional and the bottom ones are wider to cover more of the first rows. (d&b audiotechnik 2016.)



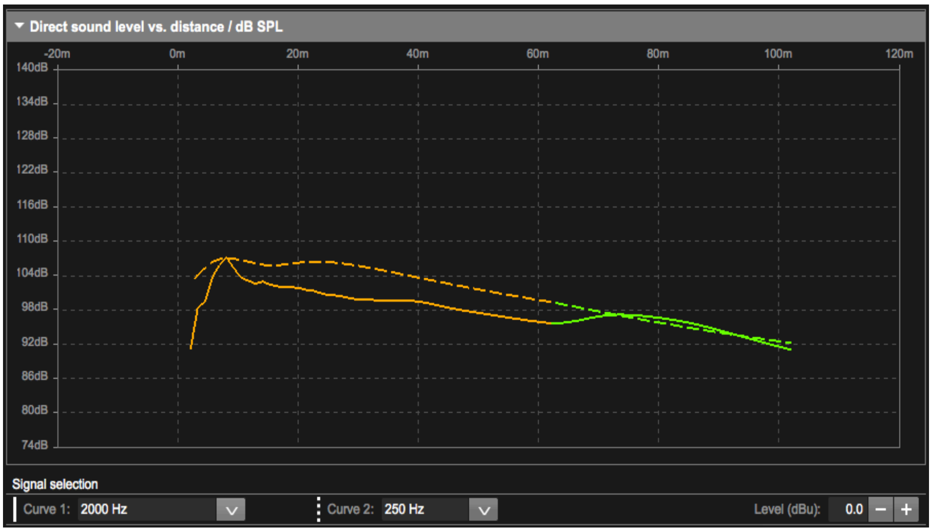
PICTURE 6: source editor’s Top View (©d&b audiotechnik 2016)

- The main profile at 0° aiming: It gives a graphic of the coverage of every particular cabinet’s main axis over distance. This is very important when figuring out if the designed sound system will be able to cover the needed distance. The second line in the graphic indicates the listeners ear height.



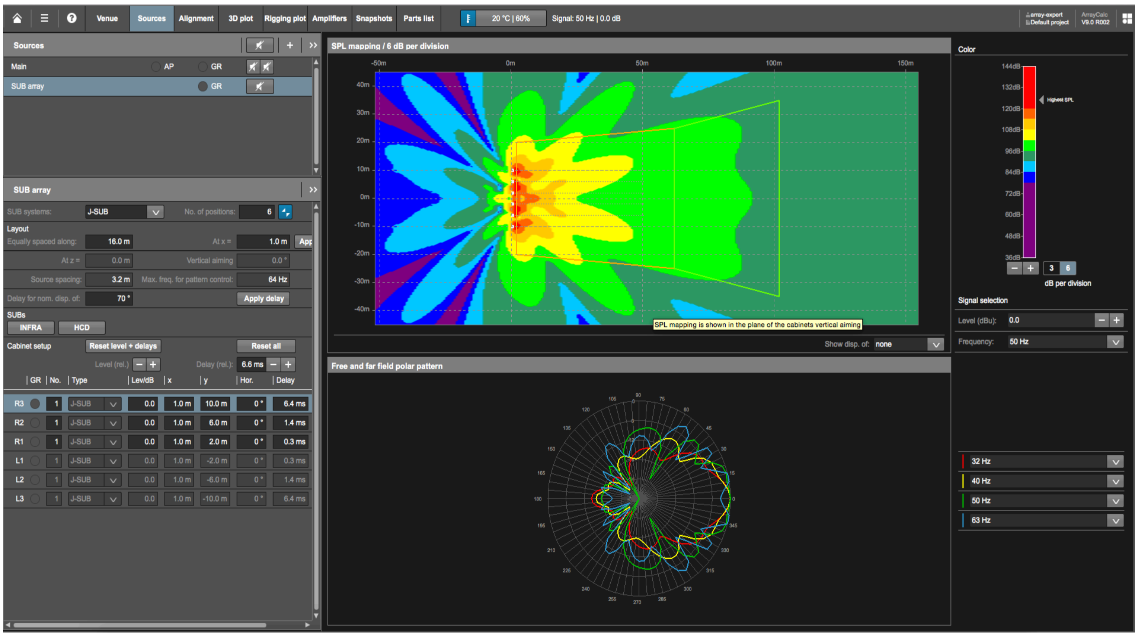
PICTURE 7: Source editor’s Profile at 0° aiming (©d&b audiotechnik 2016)

- The direct sound level vs. distance/dB SPL: it shows how the direct sound level reacts at a given distance for two frequency bands or a signal. The main goal is to keep those curves smooth and as close to each other as possible. This curve varies as you change the splay angles between cabinets. If you reduce the angles between cabinets, the high/mid-level in their target area will increase and the other way around too. (d&b audiotechnik 2016.)



PICTURE 8: Direct sound vs. Distance graph (©d&b audiotechnik 2016)

The selection of cabinets, set up levels and splay is also done on the page. Additionally, the Subwoofer configuration can be done here.

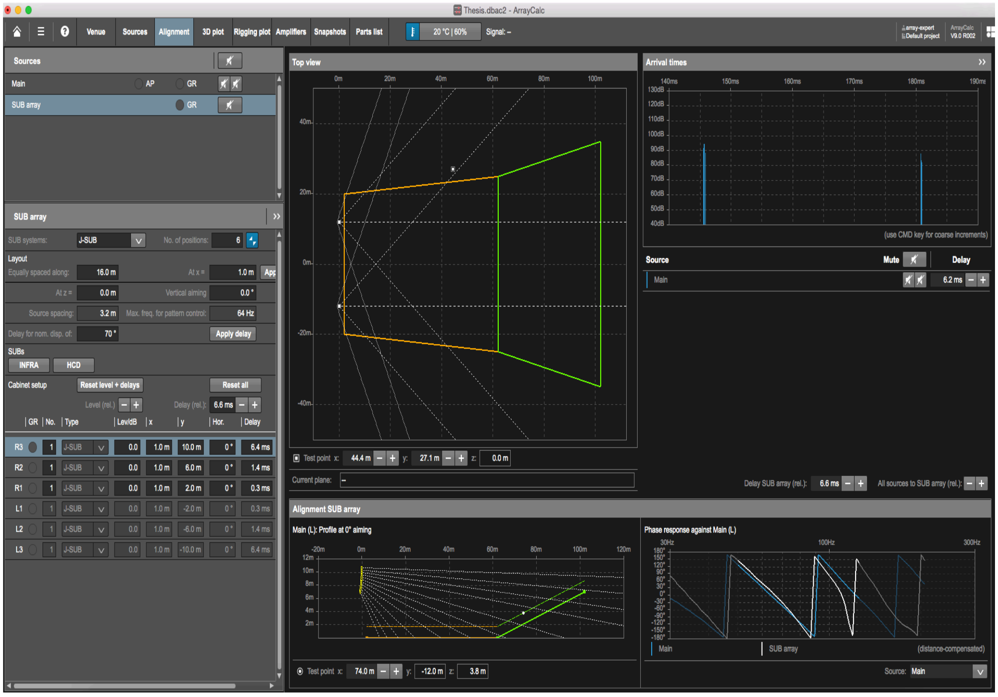


PICTURE 9. Subwoofer configuration (©d&b audiotechnik 2016)

In most cases, for designing a sound system that works it would be enough to work on the last 2 pages "venue" and "sources". I will go briefly through the next windows starting with the Alignment editor page

3.1.3 The Alignment Editor Page

When the configuration of the sound system being designed is more complex than just a Stereo Left and Right and you have multiple sources it is essential to do time alignment and phase alignment (d&b audiotechnik. Manual). During the optimization part of this thesis I will explain how this can be done using Rational Acoustics Smaart.



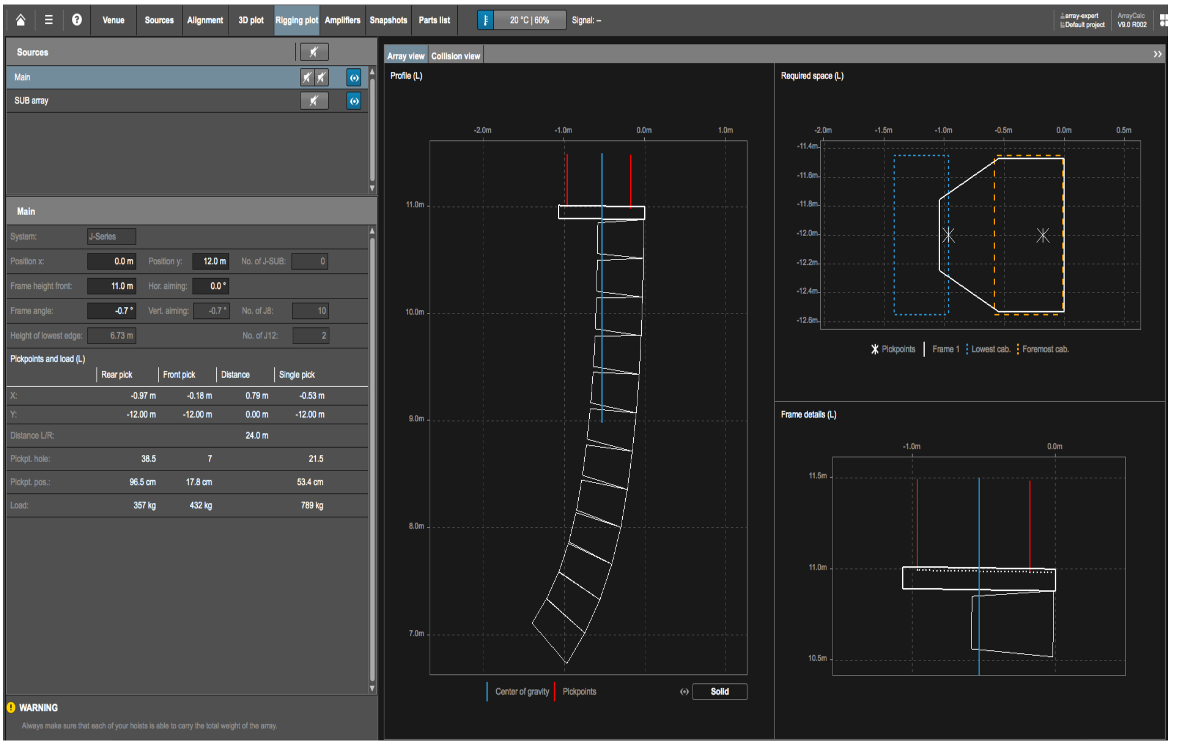
PICTURE 10: The Alignment editor page a tool to do phase and time alignment (©d&b audiotechnik 2016)

3.1.4 The 3D plot Editor Page

It is also possible in the program to calculate an SPL mapping at listener's height with the possibility to set a fully selectable frequency band. The 3d model can be zoomed and it is completely rotatable. (d&b audiotechnik, 2016.)

3.1.5 The Rigging plot Editor page

As I mentioned before one of the most important things is team work. This Rigging Plot Editor Page has a lot of information that needs to be given beforehand to the stage crew and to the rigging crew. The information provided here includes the space taken by the array, the weight of the array and its inclination and the frame angle. There are three graphic representations of the Array



PICTURE 11: Information for riggers and stage crew in the Rigging plot editor (©d&b audiotechnik 2016)

3.1.6 The Amplifiers Editor page

The Amplifiers Editor page automatically generates a configuration for our amps, it assigns IDs and channels and it also recognizes the delays. Once you get used to the D&B amplifiers you don't need to use this page to choose the number of amps needed.

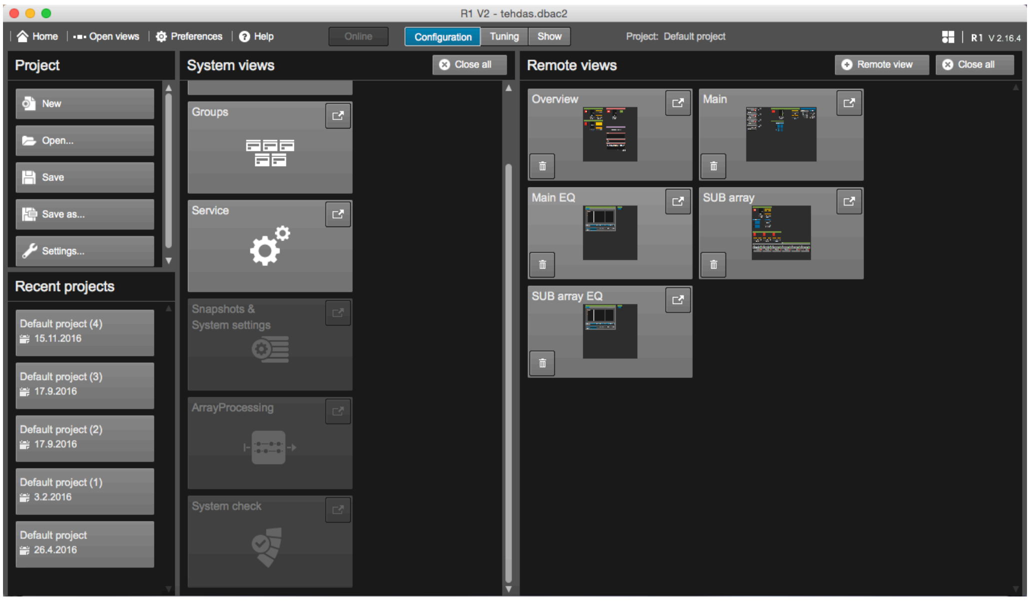
3.1.7 The “Part list” Editor page

This page gives you a count on how many amplifiers you need and their model, how many flying frames, how many hoist connectors and safety chains you will need for the job. One could print this page as a packing list, but when planning a sound system, it is better to make your list in Excel and have full control.

3.2 R1

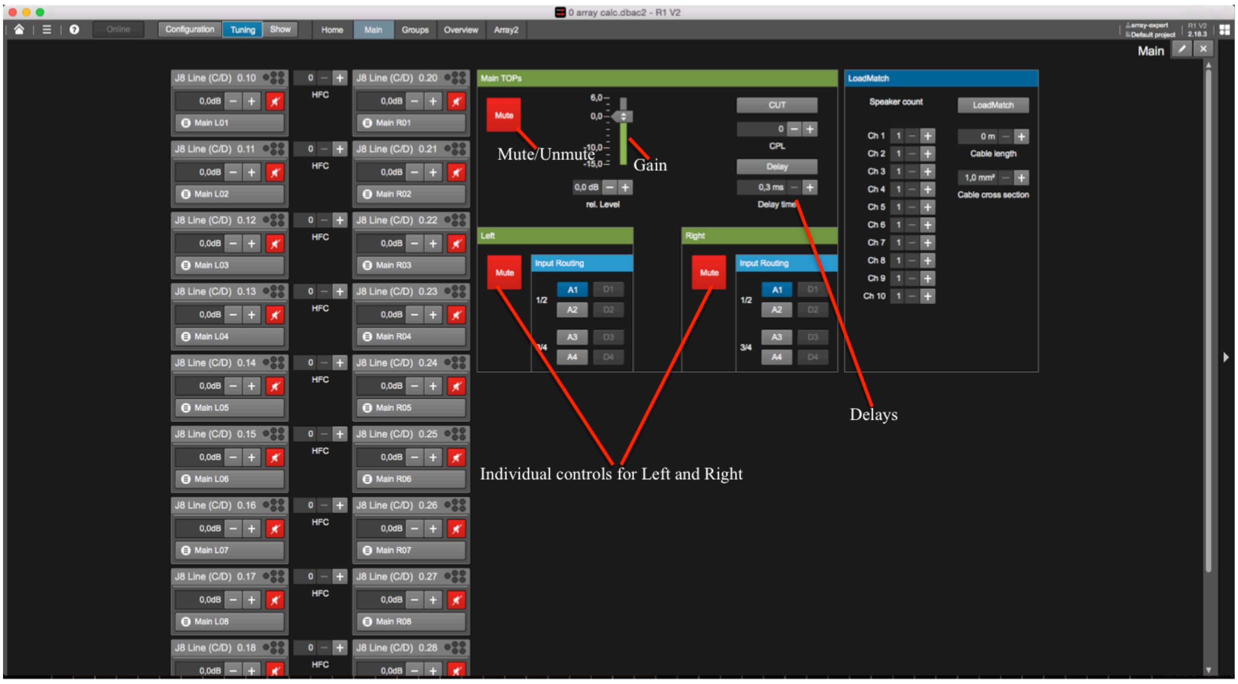
R1 is the d&b remote network control software. It provides very simplified features for setting up, controlling, testing and monitoring d&b using CAN-bus or Ethernet technologies (d&d audiotechnik. Manual R1 V1.8:X).

This particular tool can be used for scanning all connected devices, creating a graphical interface to fully operate the system, checking the system in general and to create user definable parameters to control each amplifier channel. Some of the parameters one can control with R1 are gains, loudspeaker selections, routing, equalizing and delays. Once one have the design ready in ArrayCalc one can import the file into R1 and it will automatically generate a fully operational and customizable set of parameters that can be controlled by using the software. (d&b audiotechnik. Manual V1.8:X.)



PICTURE 12: R1 Configuration page (© d&b audiotechnik)

At the optimization phase, all needed delays and equalizations can be done by using R1. Picture 13 shows an example of some of the controls that one can access from R1.



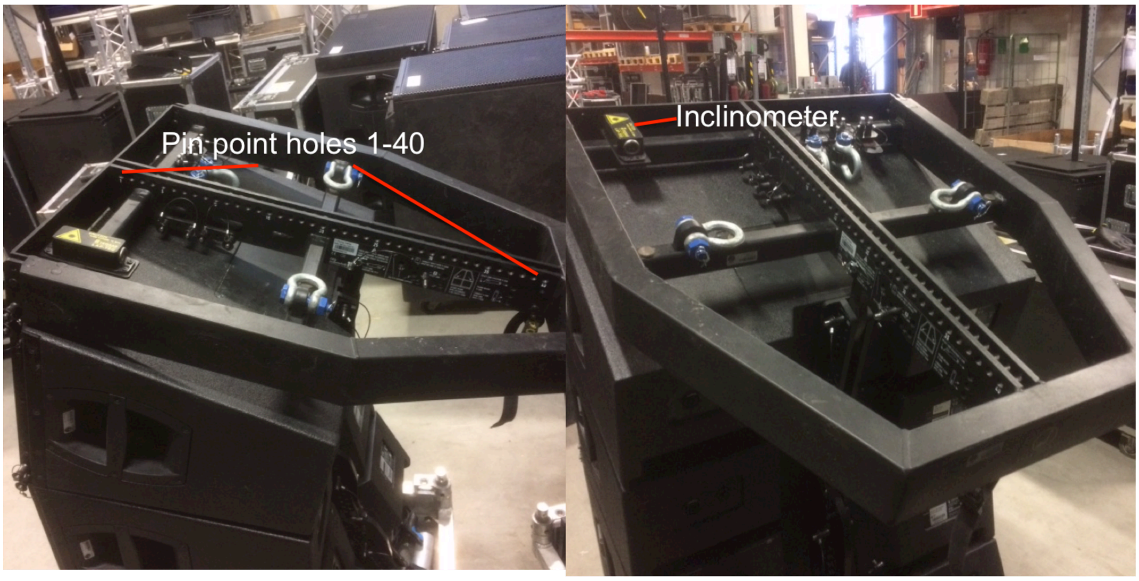
PICTURE 13: The Mains page and its controls in R1 (© d&b audiotechnik)

4 SET UP

After the design is done and while being at the venue one can start the process of setting up the equipment. Every piece of data introduced in ArrayCalc is now brought to life while setting up, for example, the pin points for the frame of our PA, the splay angles for the cabinets, the amount of cabinets, the height of the PA. Picture 14 shows the splay angles set up in the V series by d&b audiotechnik and how the Array looks when attached to the frame (d&b audiotechnik).

PICTURE 14: A closer look to the splay angles behind a V8 cabinet by d&b audiotechnik and a mini array with a set up frame (Moreno, 2016)

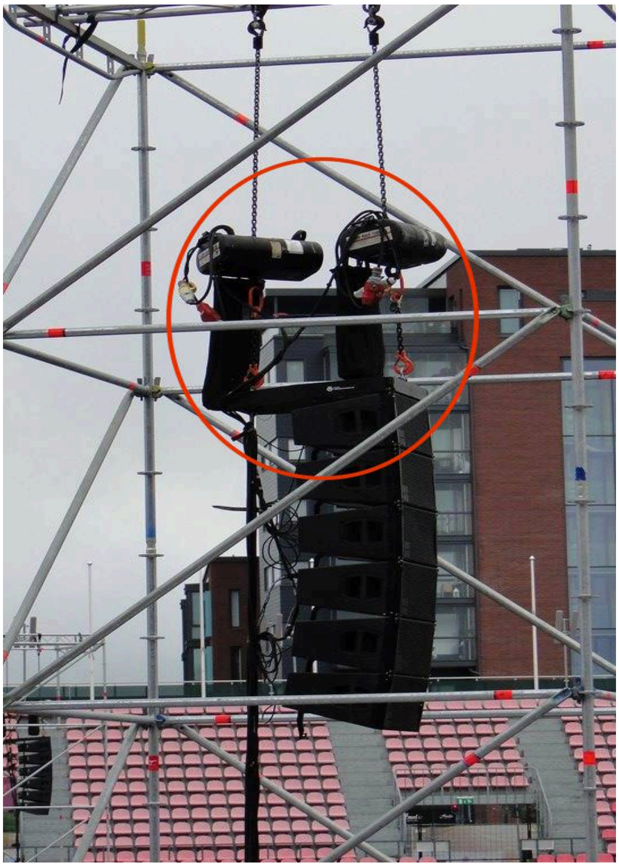
The design made in ArrayCalc provides us with practical and useful information like, for example, the pin point holes and the frame angle. The frame angle is calculated connecting a reader to the inclinometer located at the top left of the frame. Picture 15 shows where this can be located in the frame. The frame is mainly used when the PA is flown. When the PA is Stacked, if our array consists of subs and tops an adaptor is placed on top of the subs and, the top part of the array is installed on top of the adaptor (d&b audiotechnik 2016).



PICTURE 15: Pin point holes and inclinometer in the frame (Moreno, 2016)

As mentioned before there are two ways of setting up a PA, it can be stacked and it can be flown (d&b audiotechnik). In both cases the process is the same with the difference that when the PA is flown, before you bring it to its height there are a couple of things to check:

- All cabinets should work properly. They should react to their respective amplifier/ amplifier channel
- The cabinets that are being rigged should be safely locked
- The cables that are going up with our sound system should be a bit longer than the height from the highest cabinet to the ground. Depending on the location of our amplifiers we might need more or less cable. Also, there should be enough cable for a strain relief.
- The frame attached to the cabinets should be locked to the cabinets using front links and it should be also attached to the motors using the hoist connector chains.



PICTURE 16: delay tower at Eppu Normaali's 40 years' anniversary concert in Ratina's Stadium in Tampere (Moreno, 2016)

At the set up phase teamwork is the most important part. Most of the equipment is heavy and having a second person helping or what we refer to as a helping hand is sometimes indispensable. In bigger productions where there is a video crew, a light crew and a sound crew the set up can be tricky. Everyone needs space, cables can take a lot of space and the rigging of the light equipment and video equipment can happen in the same place that the sound system needs to be rigged, in this case taking turns is common practice. This makes it so important to fly your sound system when it is totally ready.

Sometimes the planned design changes at this point due to miscalculated distances, a normal case is that the PA cannot fly as high as planned in our design due to many different factors. For example, the chain that drives the motor hits its limit, which normally causes the motor to stop its upward motion.

PICTURE 17: Main PA left side at Eppu Normaali's 40 years anniversary concert (Moreno, 2016)

As an example, picture 17 shows a PA rigged during a big production and it shows how close it is from the trusses that surround it. Under the subwoofer's array you can see a truss with lights rigged. At this point if something would happen to our sub-array and we would have to bring it down for maintenance or fixing, the light crew would have to bring their truss down and, in most cases, that would not be good news. In a big production there are always fairly tight schedules. So the preparation of the equipment before rigging it is imperative.

It is always good in the set up day to talk to the organizer, the producer or the stage manager for the event. These people always have a good insight on where the equipment can be placed, where the empty cases can be stored, where the cables can be ran through. Most of the decisions on the placing equipment should be taken while planning the event or in the designing phase (Pörhöla, 2016). Sometimes when the bands send their tech rider they forget to send the updated version and it results in some changes to be made, sometimes simple changes, sometimes not so simple.

5 OPTIMIZATION

After the system is designed and set up in the venue we can start the process of optimizing our sound system. During this process we time align and phase align our tops and subs, our main PA and the sidefills, frontfills and delay towers or any source of sound that plays a roll sound-wise at our event. As mentioned before I will use Rational Acoustics Smaart for doing the optimization.

There are many different ways to approach the process of optimizing a sound system. For this thesis I will use one of those approaches. *"In general, we want the overall measured response to be flat and smooth, but there are practical limits on this"* (Thurmond, 2016)

"The goal is straightforward: Same everywhere. Same level, frequency response, clarity and sonic image location. The applicable principles are the same for single speakers and complex arrays, for a full black-box theatre or a theater full of black boxes" (McCarthy, 2016)

To a certain extent the optimization can be done by ear, but when we are looking for the best sound system, a more advanced optimization is achieved with the use of dedicated equipment. Then again one should always listen to the system. *"If the system doesn't sound great, it's not a great system" (*Gibson, 2011)

PICTURE 18: Bob McCarthy, Sound Engineer. (http://sounddesignlive.com/sound-system-design-optimization-bob-mccarthy/)

The basic set needed to fully optimize a sound system would consist of:

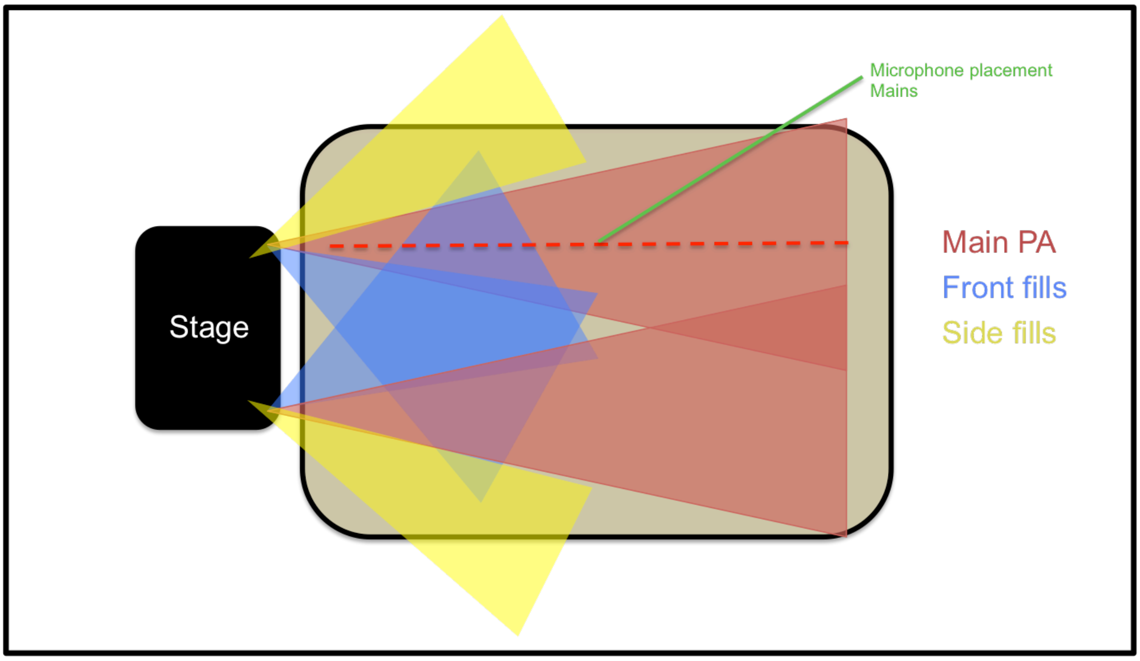
- A measurement microphone, usually small diaphragm condenser microphones with a flat response. This microphone should be omnidirectional

PICTURE 19: Beyerdynamic's MM1 measuring microphone (http://north-america.beyerdynamic.com)

- A sound interface with at least 2 inputs.
- An analysis software, in this thesis Smaart.
- A program or device that allows us to control our system.

"Place microphones strategically. We're looking for specific answers to specific questions. A mic in the middle (ONAX) does EQ. Mics at the edges tell us if it's aimed right (top and bottom for vertical and side for horizontal). A mic between two subsystems tells us if splay angle or spacing is right." (McCarthy, 2016.) One example of the microphone placement can be seeing in picture 20.

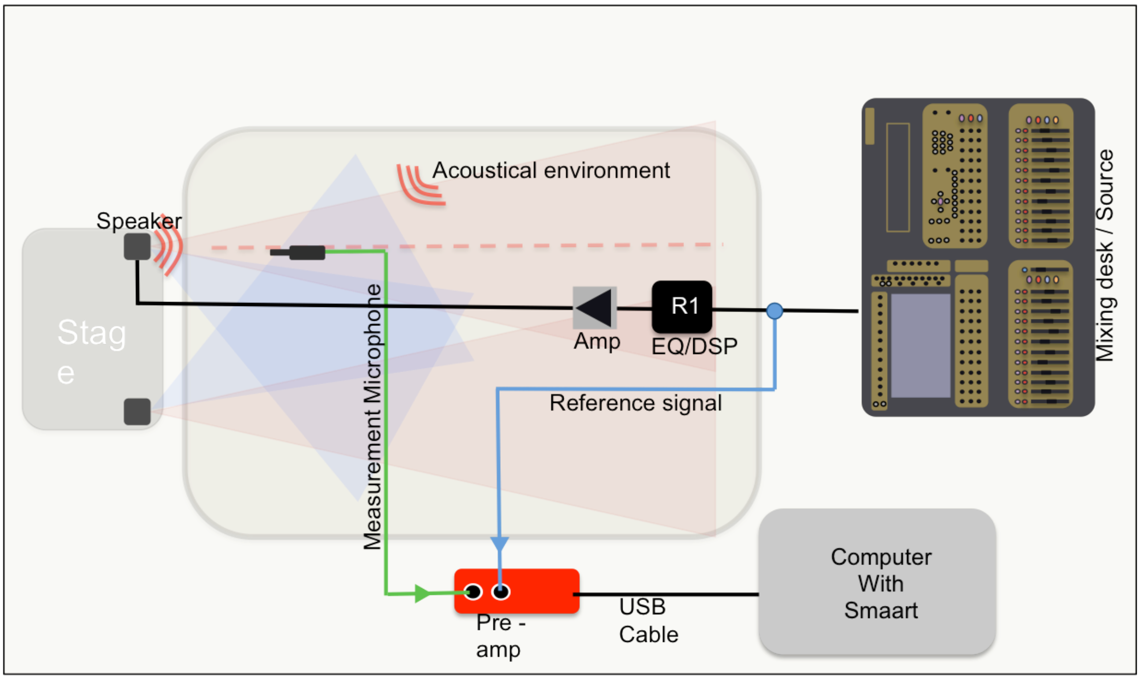
One way to start the process of optimization would be to place the measurement microphones in front of the main speakers. The number of microphones and the distance between them depends on the size of the venue. Picture 20 shows the horizontal dispersion (not in scale) for the mains, front fills and side fills as well as a red line showing where to place the microphones.



PICTURE 20: Microphone placement ONAX for measuring the Mains (Moreno, 2016)

"In its simplest sense, Smaart v.7 Di is a two-channel analyzer – it takes in two audio signals and allows the user to examine the frequency content (spectrum) of those signals, and to compare those two signals (transfer function) in order to view the difference between them. Smaart v.7 Di therefore comprises three measurement engines; two Spectrum engines – one for each input – and one Transfer Function engine that calculates the difference. In standard practice, the two signals you feed into Smaart Di are the input (Reference) and output (Measurement) signals of a system, and so the Transfer Function between the two (output vs. input / Measurement vs. Reference) represents the response of that system in frequency (Magnitude & Phase) and the time (Impulse response). " (Smaart v.7Di. Manual.)

The signal chain for Smaart to work is shown in picture 21 and it consists of a two or more channels I/O, one or several measurement microphones, USB or Firewire cable, XLR or plug cables and a computer that can host the software. Smaart has a signal generator that provides us with different signals we can use, normally pink noise.



PICTURE 21: Functional diagram for Smaart to work (Moreno, 2016)

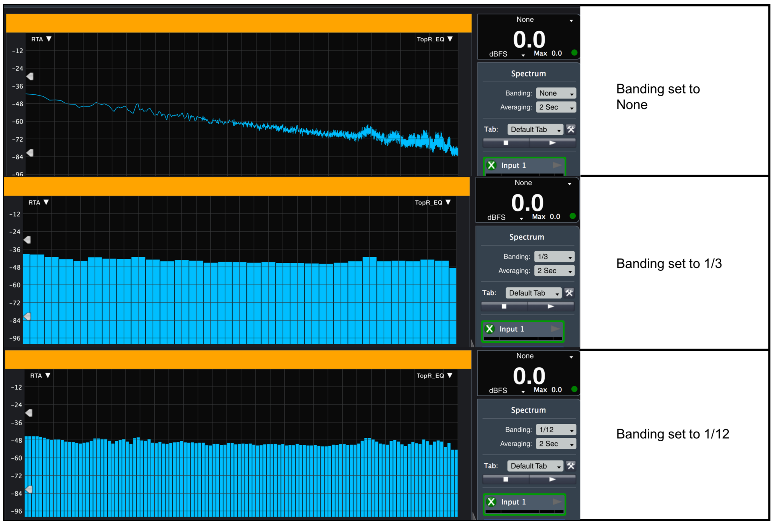
Once the setup is done we can take a look at the interface and start measuring our sound and optimizing our sound system.

PICTURE 22: Smaart transfer function screen. Measuring the main tops and their average shown in pink. The coherence is shown with a red line (©Rational Acoustics, 2016)

The program gives many possibilities on how to analyse sound. RTA, Spectrograph, phase, magnitude, coherence and transfer function among others. The most used for optimization is the transfer function. Picture 23 shows the right-hand side data bar and its possibilities. This varies depending on which windows you are in. Next I will show the options. First we will take a look at the right-side data bar for RTA-RTA, RTA-Spectrograph and Spectrograph-Spectrograph

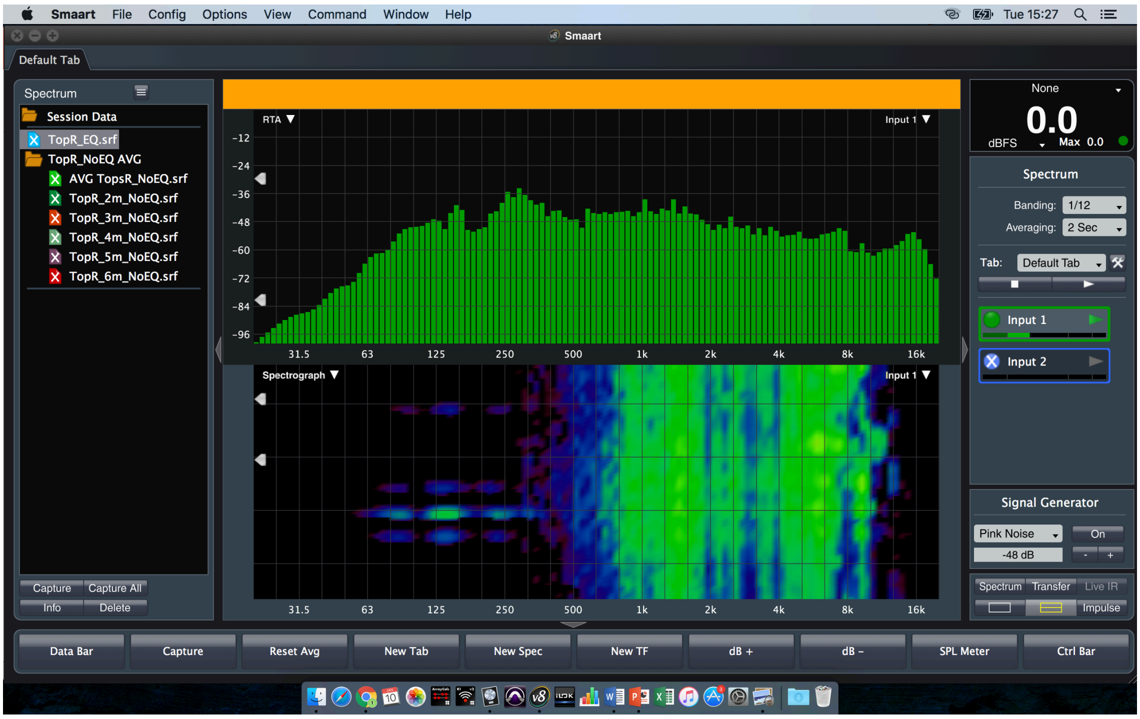
PICTURE 23: Right hand side data bars in Smaart V8 (©Rational Acoustics, 2016)

At the top of the first data bar there is the dB meter. Once you assign an input it will show in real time the SPL in dBFS (this can be changed in the options) Then there are the spectrum specifications where we find two options banding and averaging. The banding defines the way data is shown in the RTA as shown in picture 24. The averaging controls how fast the graphic will change when sudden changes in volume and frequency happen. This means that if our microphone reads a signal of any type that is not constant, and the averaging is set to 2 seconds, the graph might not register it. This is useful when optimizing a system in an outdoor venue or a noisy place.



PICTURE 24: Options when choosing banding in Smaart (©Rational acoustics, 2016)

After the banding and average you'll find the inputs. Picture 23 shows just two inputs since I was working with a two inputs interface. The coloured circle with the "X" means that this particular input is not going to be shown in the spectrogram and the triangle acts as an on/off bottom for activating or deactivating our input signal. After you assign the input, activate the input signal and allow it to show in the spectrograph Smaart will generate a graph as shown in picture 25



PICTURE 25: RTA and Spectrograph (©Rational Acoustics, 2016)

Last in our right-side data bar we find the signal generator. The signal generator provides us with the signal that we will use for optimizing our system. Among the signals you can choose from you will find pink noise, pink sweep, sine, dual sine and file. Pink noise is the most common option for calibrating and optimizing sound.

When you choose the transfer function, its right-side data bar looks a bit different as shown in picture 22 and it acts differently. The main difference in the data bar is that now you can control the smoothing of the phase and magnitude as well as the overall average.

5.1 Optimizing a System.

In this section I will describe how a system is optimized. There are, as mentioned before, many different ways and opinions about how to optimize a system. For this Thesis, I will take a general approach as to what to do and how to do it. One particular thing that every sound engineer I interviewed agreed on was the importance of proceeding in the following order:

1. Equalization of the system.
2. Finding the level and level matching our systems.
3. Timing our sound systems.

Every particular part of our sound system goes through the same procedure, be it tops, subs, front fills, side fills or delay towers (Köykkä, 2017) The main PA is the most important part since everything will be matched to it.

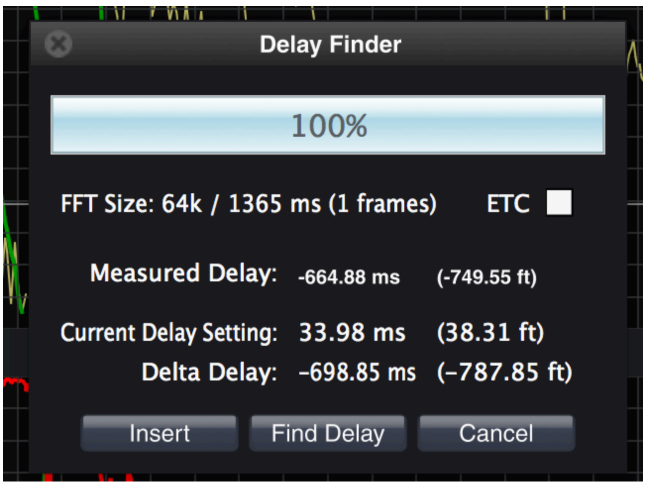
5.1.1 Main Tops

Once the program is configured, the measuring microphone is set up and you see both signals, the measuring microphone signal and the reference signal coming in, it is time to place the microphone in from of the Main PA aiming at its center (ONAX).

We position the microphone in position 1, normally close to the first expected raw of audience, put on the pink noise and measure.

PICTURE 26: A single measurement with no inserted delay. Impulse response if offset (©Rational Acoustics, 2016)

As shown in picture 26 the measurement shows impulse response, phase, magnitude and a coherence line. In order to get this properly working you need to find the delay in a way that the impulse response is in “0”. In order to do that the program offers a command called find delay easily accessible by pressing the letter “L” in your keyboard.

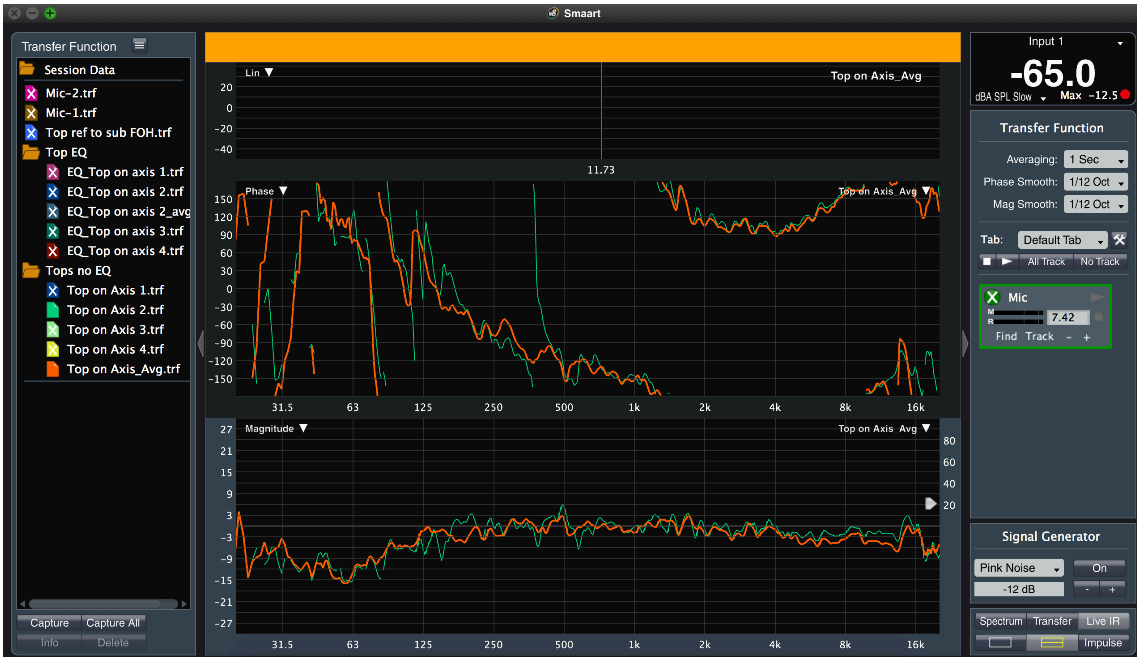


PICTURE 27: The insert delay function (©Rational Acoustics, 2016)

Once you find the delay you can store the measurement by pressing the space bar. This will allow you to name the measurement and assign a colour to it. After that you can mark the place where you took the measurement, move the measuring microphone further back and repeat the procedure for a couple of times more. After several measurements are taken you can analyse them and find an average in order to decide on the equalization. This procedure can be shorter and easier with a multi-microphone set up using a sound card with more inputs. In that case one could just place the microphones in their position and trace their signals in Smaart and do a live average that can be used for the overall Equalization of the system, but for this Thesis I did it with one microphone.

PICTURE 28: Four different measurements named and averaged (©Rational Acoustics, 2016)

As shown in picture 28 the measurements can be quite different and they can have different boosts here and there due to acoustical factors. The average, shown in pink, represents the overall curve for our Tops. Since we are not working in a multi-microphone set up and we want to see how the equalization we are about to do changes our graph, one thing we could do is to compare the results from all our measurements with the average. The closest measure to the average will be used for our equalization purposes.

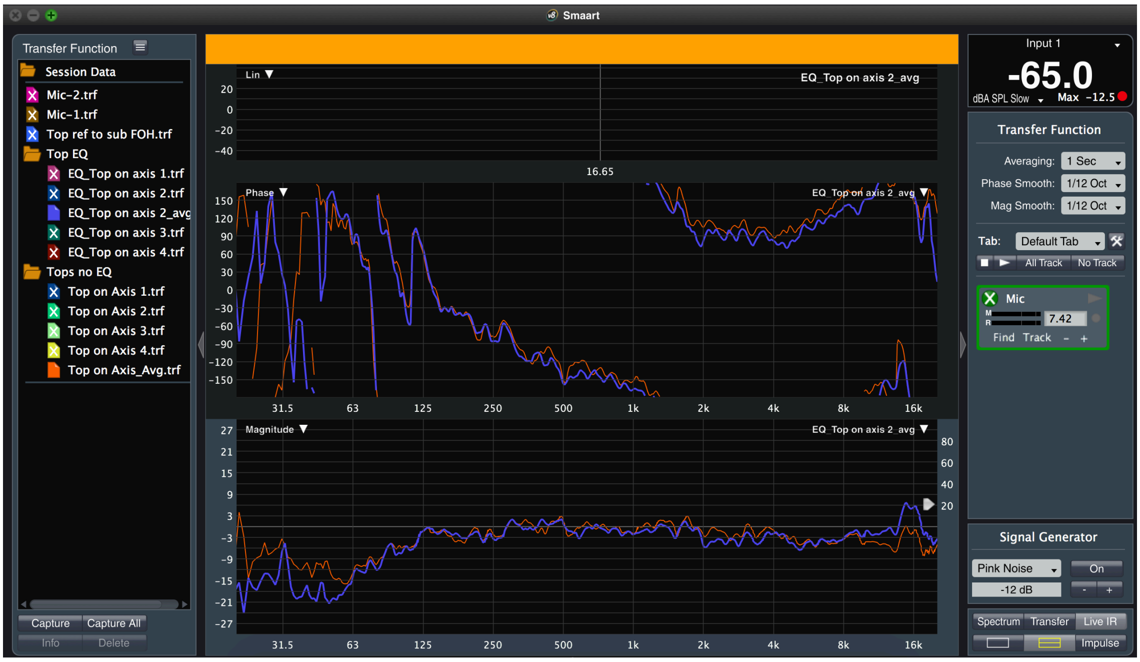


PICTURE 29: Average measurement compared to a single measurement in a given position (©Rational Acoustics, 2016)

The second measurement we took, is really close to the average as shown in picture 29 so if we place the microphone at that particular position we can do the equalization for the Tops based on the graphic that we see. This is the moment when one has to take a decision on how the system should respond. Do we want a flat response? do we want to cut/boost anything? Once the decision is taken we can start equalizing. The system response can also be considered in the designing phase while deciding one the type of cabinet and the angles between them, the position of the array and its aiming.

Once we have done our equalization we could repeat the whole process for every single measure, but now with the equalizer on. After the measurements are taken we can do an average again and then we can see properly how our equalization is changing the overall response of the tops.

Picture 30 shows the relationship between the average for the tops with no equalization (shown in orange) and the average for the tops with equalization (shown in blue). It affected mostly the 2K to 4k section of the frequency spectrum and we can see our frequency response is not totally flat but instead is going down smoothly.



PICTURE 30: Comparison between the frequency response of two different averages (©Rational Acoustics, 2016)

After we are satisfied with the sound of our tops we could set up the mic for example at the front of the house position and continue with our subs.

5.1.2 Subs

As mentioned before, in order to optimize any part of the PA we need to first find our equalization, then the level and then the time. In order to start we place the measuring microphone at the front of the house position. The main reason for this is that the FOH is the person who is going to mix the show and it is a good practice to get the best sounding low end at that position so, when mixing the show, the mixing engineer don't over boost any frequencies.

Once we are ready to start measuring our subs we send pink noise to our subs and read the signal with the microphone. Once we see our frequency response we can decide on the equalization and applied it if needed.

After we have equalized our subs we can mute them, send the pink noise to the tops, find the delay and once we have our frequency response and impulse response set we can save this measurement. Now it is time to send again the pink noise to the subs and compare.

PICTURE 31: Subs in reference to the tops at FOH (©Rational Acoustics, 2016)

The green curve in picture 31 represents the live tracking/frequency response of the subs and the brown curve the frequency response of the tops. Also, it shows a point of interest, the acoustical crossover point where the two signals intersect each other. Depending on the genre or the occasion one has to decide on the relationship between subs and tops.

At this point we have the right impulse response for the tops and we are tracking the subs so it is time to phase align them (Time alignment does not work properly here since the impulse response window does not show any information, the frequencies are so low that there is almost no attack). Here is when we apply the delay on the subs. This can be done in the amplifier, in the console or if you are using a program like R1 by d&b audiotechnik you can use their interface to control the amplifiers and their functions. Picture 13 shows R1s interface and where to apply the delay. What we are trying to do is overlap the phase curves as close to each other as possible. Picture 32 shows how the brown line representing the frequency response of the subs falls on top of the blue line that represents the frequency response of the tops.

PICTURE 32: phase alignment tops and subs (©Rational Acoustics, 2016)

At this point one could put music coming from the Main tops and subs and check around the venue the overall sound of the PA. In a small venue concert this is more than enough. If the venue is a bigger place with balconies, side seats or just big enough so the main PA cannot cover the whole place one needs to put more speakers and optimize them

5.1.3 Front fills, Side PA, Delay towers

In order to optimize this extra set of speakers one has to follow the very same rules; first Equalize, then level it to the main PA and last but not least timing it to the main PA.

I will demonstrate how to do it for the front fills. First of all, the sound of the front fills shouldn't be louder than the main tops, it shouldn't be brighter nor darker, it should be pretty much the same at the same level. While walking through the venue or the location where the stage is set up, one should not be able to localize the sound coming from one speaker or other, it should be theoretically the same.

In order to start with the front fills, we send our pink noise to the tops, find the delay and store our measurement. Then we mute the tops and send the signal to our front fills and see the frequency response. Now with a bit of equalization we can try and match the front fills frequency response to the tops. This can be done in the amplifier, in the console or in R1 if using d&b audiotechnik. Picture 33 shows a comparison between the main tops and the front fills after equalization.

PICTURE 33: Main tops in blue and live trace for the front fills after equalization in green (©Rational Acoustics, 2016)

After this we can place the measuring microphone in front of the front fills where the first row of audience will be and play pink noise from the tops and trace how loud it is. Next we can mute the main tops and play the pink noise through our FF and match the level to the one measured before.

Now we can send pink noise to both the tops and the front fills and find the place where they sound equally loud and at that point we set up the measuring mic to find the delay.

Now finally we have to time align the front fills. We start in the very same way send signal to our tops, find the delay, store the measurement and then mute the tops and send the signal to the front fills. Then we find the impulse response for the FF, that way we can see where it is in time in relation to the main tops. The time difference between the impulse response from top and front fills is the amount of delay needed to get them in the same time at the measurement position (Köykkä, 2017). Picture 34 shows where to find the offset delay for the front fills and how the frequency responses interact with each other.

PICTURE 34: Tops and subs in green and front fills in pink (©Rational Acoustics, 2016)

At this point our sound system is ready to be tested. One could start by playing a reference song that one knows and walk around the venue to hear how well spread the sound is and if the PA covers the wanted areas.

6 DISCUSSION

Live events have become much more than just a concert, there are many factors playing important roles in the entertainment business nowadays. Regarding the sound we are living in an era where technology is getting us closer and closer to better sound. Sound systems have become more powerful and more customizable, which also means that there are always new techniques to learn as all new devices have a new way of operating them.

During the process of writing this thesis I have had the opportunity of discussing with a lot of sound engineers and technicians about sound, about its importance, about ways of making the best out of any sound system.

Nowadays, the role of a sound engineer has become a lot more sophisticated and difficult as sound have become much more than just sound. In order to fully understand and be able to operate a professional sound system you have to know something about the theory behind sound, something about networks and their behaviour, some protocols. Also, you need to know how to set up a sound system, which involves knowing about motors, speaker set up and configurations, how to operate mixing consoles, microphone positioning and response, rigging, virtual design and configuration, among others. I am not trying to say that this is indispensable for any sound system set up, but if you are working in a large scale sound system set up you need to be able to get the best out of it.

During the process of learning about sound systems design and optimization and live sound in general I realised how important teamwork is. A “perfect” sound system set up is nothing without a good mixing engineer to operate it. The system engineer is not always the same person that mixes a concert from front of the house (FOH) and in this particular case it is always a good thing to know that if you do your work right the person operating your design will do the best with it.

An important part of the whole process is the pre-production. Every single concert is planned somehow and it is really important to make sure during this process that the sound part of the negotiation is enough.

I think that the companies dealing with live events need more people interested in sound and able to work with it, and it is in their best interest that people familiarize themselves with these techniques properly.

A couple of weeks after I started writing this thesis I realised that, even when I am really enthusiastic about this subject, I could not really go too deep into the techniques and theories behind sound system design and optimization since this study would become way too long. After writing a lot of info I had to cut down some parts and decided to try to make a simple yet well written guide on how to do this. In this study I am not trying to say this is how it is done, instead I tried to think about what a person that wants to start doing this needs to know and I went from there.

I'm convinced that there should be more information about this matter and it would be a great development if studying these techniques would be possible. Learning about sound systems design and optimization is just possible if you work in the field and spend time asking questions and being present. One should be able to get education on this topic.

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