akuTEK

www.akutek.info

PRESENTS

Sound pressure levels and sound source components for orchestra musicians

by H K Rydland

ABSTRACT

The presented study aimed to do investigations on the balance of the sound components found at the ears of orchestra musicians. This was interesting as orchestra musicians are exposed to much sound in their vocational life and one would like to find the primary sources inducing the sound. Measurements were performed on four instruments. The trumpet was chosen as a representative for wind instruments, the oboe for woodwind instruments, and the violin together with the cello represented the string instruments.

For the overall results, it was found that the direct sound of the self was likely to often be dominant or almost dominant at the ears of the trumpeter. The balance of the sound components shifted between dominance of the direct sound of the self-played instrument and the background, consisting of direct sound from others and the reverberant sound. The reverberant sound component was found to be less dominant and within 2-20% for the oboe, the trumpet and the violin. It was found that the balance of the sound components was the most consistent for the trumpet, although more varied musical contents ought to be further tested.

The found results of the short-term Foreground-to-Background Balance was statistically best described by the table below.

	Median	μ	σ	No. of observations
Cello	-7.1	-7.5	6.6	116
Oboe	-6.5	-5.4	7.6	111
Trumpet	-1.3	-0.4	4.6	119
Violin	-2.7	-2.7	6.4	259

akuTEK navigation:

Home
Papers
Title Index
akuTEK Research
Concert Hall Acoustics

NTNU – Trondheim
Norwegian University of
Science and Technology

Sound pressure levels and sound source components for orchestra musicians

Helena Kristine Rydland

Master of Science in Electronics
Submission date: July 2015
Supervisor: Peter Svensson, IET
Co-supervisor: Magne Skålevik, Brekke & Strand Akustikk AS

Norwegian University of Science and Technology
Department of Electronics and Telecommunications

Project text

For a symphony orchestra, and the individual musician, the acoustic conditions throughout the working process varies a great deal – from the individual solo rehearsal in a small rehearsal room to full orchestra rehearsals and concerts. The variation in acoustic conditions between rooms of rehearsal and spaces of performance make influence on the feedback received by the musician, as well as the sound exposure/wear on the ears. The sound exposure of a musician is a hot topic - what causes the most damage to the hearing ability? How much of the strain is caused by the musician's own instrument and how much is induced by the orchestra?

In the project sound pressure levels will be measured for some musicians in orchestra situation. By the use of microphones placed by the musician's ears and in the diffuse sound field, recordings of the same musical piece should be done together with the orchestra and the musician in solo. This makes it possible to analyze how much of the sound which is caused by direct sound from the musician's own instrument, by direct sound from others, and by the reverberance.

Abstract

The presented study aimed to do investigations on the balance of the sound components found at the ears of orchestra musicians. This was interesting as orchestra musicians are exposed to much sound in their vocational life and one would like to find the primary sources inducing the sound. Measurements were performed on four instruments. The trumpet was chosen as a representative for wind instruments, the oboe for woodwind instruments, and the violin together with the cello represented the string instruments.

For the overall results, it was found that the direct sound of the self was likely to often be dominant or almost dominant at the ears of the trumpeter. The balance of the sound components shifted between dominance of the direct sound of the self-played instrument and the background, consisting of direct sound from others and the reverberant sound. The reverberant sound component was found to be less dominant and within 2-20% for the oboe, the trumpet and the violin. It was found that the balance of the sound components was the most consistent for the trumpet, although more varied musical contents ought to be further tested.

The found results of the short-term Foreground-to-Background Balance was statistically best described by the table included.

	Median	μ	σ	No. of observations
Cello	-7.1	-7.5	6.6	116
Oboe	-6.5	-5.4	7.6	111
Trumpet	-1.3	-0.4	4.6	119
Violin	-2.7	-2.7	6.4	259

Sammendrag

I den denne masteroppgaven var målet å undersøke balansen av lydkomponenter på orkestermusikeres øre. Dette er et aktuelt tema ettersom orkestermusikere eksponeres for mye lyd gjennom sitt arbeid og for å gjøre effektive støytiltak må en finne de primære lydkilder. I undersøkelsen ble det gjort målinger med fire musikkinstrumenter. Blåseinstrumenter ble representert ved trompet, treblåsgruppen ved obo og fiolin samt cello representerte strykegruppen.

Lyden på øret byttet mellom å være dominert av egen direktelyd og bakgrunnen, som her besto av direktelyd fra resten av orkesteret og etterklangslyd. Komponenten av etterklangslyd ble funnet å være lite dominerende da den utgjorde 2-20% for oboen, trompeten og fiolinen. Fordelingen av lydkomponenter ble funnet å være mest konsistent for trompeten, men det må legges til at et mer variert musikkutvalg for denne burde undersøkes i videre arbeider.

Forgrunn-bakgrunn balansen ble statistisk sett best beskrevet ved tabellen inkludert under.

	Median	μ	σ	Antall observasjoner
Cello	-7.1	-7.5	6.6	116
Obo	-6.5	-5.4	7.6	111
Trompet	-1.3	-0.4	4.6	119
Fiolin	-2.7	-2.7	6.4	259

Aknowledgements

This Master's thesis concludes my time at the study programme of electronics at the Norwegian University of Science and Technology in Trondheim. I would like to thank the creative and sporty musicians in the symphony orchestra of Trondheim for their highly valuable participation to this project, their patience and enthusiasm. I would also like to thank Magne Skålevik for introducing such an interesting problem description, for his vocational support and guidance throughout the work process. A special thanks goes to Professor Peter Svensson who guided the project on my work location at the university site in Trondheim. I also thank Tim Cato Netland for his help with finding the appropriate measurement equipment.

Lastly, a special thanks is in order to Suleiman D. A. Hemidan, without whom I may never have carried out such a work as this if it were not for him providing me with inspiration and encouragement from an early age. I'm also grateful to the late Bjørn E. Meyer for taking pride in me and now, alas, for his silent cheers and motivation.

Helena Rydland

Helena Rydland

Contents

Abbreviations and acronyms

FBB	Foreground-Background-Balance
FG	Foreground
BG	Background
DF	Directivity Factor
T	Reverberation Time
LR difference	Left-ear-to-right-ear difference

Chapter 1

Introduction

The work to be presented in this Master's thesis is a contribution to an ongoing project by Magne Skålevik, Senior Consultant at Brekke & Strand akustikk AS, Oslo, Norway. The overall objective of the project is to find effective measures on how to preserve orchestra musicians hearing ability, and find out if there are parameters describing how loud musicians play in different environments - is it possible to find acoustic parameters that remain consistent between music rooms which are considered well suited for their purpose?

This report will draft a theme from the problem complex of «variation in acoustic conditions for an orchestra musician, from rehearsal to concert performance». In his paper *Consistency in music room acoustics*, Magne Skålevik suggested to do further studies with instruments and their performance between music rooms. [1] This contribution will look at one room situation from Skålevik's suggested topic, and do investigations on the components of sound experienced by orchestra musicians - which components dominate the sound as perceived by the orchestra musician?

1.1 State of the art and previous work

Previous work on the topic has mostly been done by Odeon simulations. Three papers by M.Skålevik, related to the problem at hand, are published on the open website «akutek.info». Among those is a pilot study of room acoustic variations between rehearsal spaces used by the orchestra musician. The pre-study, which includes measurements with a violinist, presents material on the balance of the sound elements at the musician's ear. Ian O'Brien did an extensive research with more than 4000 hours of dosimeter measurements, during three years. He has suggested that the sound at the musician's ear is dominated by the musician himself.

1.2 Problem description

A central problem is the question of what the sound at the musician's ear consists of, and what is it that decides the sound pressure level at the musician's ear? Is it dominated by the playing musician's own instrument, by direct sound from other intstruments in the orchestra, or by the reverberant sound?

Why is it interesting to obtain a deeper knowledge the properties of the sound received at the musician's ear? Among professional musicians it is fairly common to suffer from reduced hearing to a varying extent. They spend a lot of time rehearsing individually, in groups and with the orchestra, and the nature of their work make the hearing ability an important part of their vocational life. It is no secret that musicians expose themselves, and are exposed to, much sound. Although of varying intensity, the degree of exposure over a longer period of time is high enough to affect the hearing ability, especially in some instruments like the trumpet.[2]

Some investigations suggest that as much as 50 per cent of orchestra musicians suffer from grade 1 hearing loss, giving them a loss of 20 dB at some third octave band according to hearing tests. The music orchestra is interested in screening the change in hearing abilities among their staff, so the problem is on the agenda. A study concluded that musicians suffer from more hearing loss than the average population when compared to age and gender. [3]

The hearing ability works as a muscle and need occational rest in order to function properly. An important further question is therefore to find efficient measures on how to reduce the risk of hearing loss. If the wrong measures are put in action, it may actually make the situation worse. In order to find the good countermeasures it is of high importance to know the properties of the problem. That is to say, how is the nature of the sound received at the ears of the orchestra musicians?

1.3 Investigation points and objectives

This thesis will draft the outlined problem, and try to provide an answer on the basis of measurements performed with a symphony orchestra. Some specific parameters which describe the balance of the sound components of the soundscape will be investigated:

A) dry self,

B) dry others,

C) reverb all.

Dry self is the direct component of the sound produced by the musician himself, without other contributions. Dry others is the direct sound from the orchestra excluding the musician in question. Reverb all is the reverberant sound component found with the orchestra as a whole.

Objectives of the investigation

The aim of this work is to present an outline of what may be found in the vast sphere of orchestra music. The results may vary with composer, musical work, musical time periods, instrument composition in the orchestra and the way the various groups are placed in relation to each other. That is to say that this investigation merely scrapes on the surface of the problem at hand.

1.4 Report structure

A theoretical background of relevant themes is given in Chapter 2. The ideas for the investigation and perspectives on the method chosen is presented in Chapter 3. The measurement procedure and equipment is then described in Chapter 4. The main results are given in Chapter 5 and discussed in Chapter 6. The conclusion for the investigation is then given in Chapter 7. Some of the further aquired data and the background data is presented in the Appendix, and the data scripts and more details on the post-processing procedure can be found in its last chapter.

Chapter 2

Theory

This chapter presents a basic introduction to theory which may be needed in order to understand the contents of this report. In accordance with general calculations in acoustics, the logarithms presented have a base of 10 (log_{10}).

2.1 Root-mean-square pressure

A sound signal is physically a deviation in the atmospheric equilibrium pressure. When measuring this with a microphone, the signal will be represented with positive and negative impact values. In order to rectify this one can look at the root-mean-squared sound pressure (RMS). The RMS value for some time interval τ of measurements can be defined by the following formula,

$$p_{rms} = \sqrt{\frac{\int_0^\tau p^2(t)\mathrm{dt}}{\int_0^\tau \mathrm{dt}}}, \tag{2.1}$$

where p denotes the instantaneously measured pressure at some time t within the intervall τ. [4] Finding the root-mean-square value of the pressure means that the atmospheric pressure deviation is equalized in time, leaving one single value to represent the signal. A compromise however, is that the information on the variance of the signal, e.g. the peak values, are not represented.

2.2 Sound pressure level L_p

$$L_p = 20log\left(\frac{p}{p_{ref}}\right)[dB] \tag{2.2}$$

Sound pressure level is generally found by (2.2) where p_{ref} is the reference pressure, and is approximately the hearing threshold of the humain ear at $20\mu Pa$. Sound pressure is given in logarithmic scale with unit decibel (dB for short notation) due to the vast range it represents.

2.3 Statistical parameters: mean value and standard deviation

If X_n,with $n \in [1, N]$ represent a random sample of size N then the sample mean is defined by the formula

$$\bar{X} = \frac{1}{N}\sum_{i=1}^{N} X_i. \tag{2.3}$$

The same random sample will also have a sample variance defined by

$$S^2 = \frac{1}{N-1}\sum_{i=1}^{N}\left(X_i - \bar{X}\right)^2, \tag{2.4}$$

which gives it a sample standard deviation of $S = +\sqrt{S^2}$. [5]

2.3.1 Kolmogorov-Smirnov test

The Kolmogorov-Smirnov test sees if a normally distributed dataset is likely to come from a standard normal distribution, and returns the likelihood of this. The one-sample Kolmogorov-Smirnov test is used with a null hypothesis that the dataset at test comes from a standard normal distribution by comparing it to an alternative of not being of this distribution. The outcome is zero if the null hypothesis can not be rejected, and 1 if it is rejected, with the level of significance is in the latter case being at 5%. Larger datasets samples at test will make the outcome result more reliable. [6]

2.4 Sound propagation in space

The theoretically expected sound pressure level at some point of distance r from the emitting sound source can be found by the classic equation

$$L_p = L_W + 10log_{10}\left(\frac{DF}{4\pi r^2} + \frac{4}{A}\right)[dB], \tag{2.5}$$

where L_W denotes the sound power level of the source, DF the directivity factor of the sound source and A is the absorption factor of the room. Note that $L_W = 10log(W/10^{-12})$ where W denotes the power of the source.

2.4.1 Barron revised theory

$$L_p = L_W + 10log_{10}\left(\frac{DF}{4\pi r^2} + \frac{4}{A}e^{-2\delta r/c}\right)[dB] \tag{2.6}$$

Equation (2.6) is similar to the classic equation from (2.5) only with an added factor of $e^{-2\delta r/c}$. This comes from *Barrons revised theory* where the decay of the sound level in concert auditoriums were revised with regards to the distance relation to the sound source. [7]

$$L_W = 10log\left(\frac{I^2}{W_0}\right)[dB] \tag{2.7}$$

where $I^2 = \frac{p_{rms}^2}{\rho_0 c}$ and $W_0 = 10^{-12}W$. A typical value for c is 343.5 at 20°C, but it may also be calculated by $c = 331.4 + 0.6\theta$, where θ is the temperature of the air given i centigrades. [8, p. 7]

2.5 Frequency weighting, A-filtering

Fletcher and Munson did a thorough documentation on the humain hearing in 1933 with regards to the sensitivity, or perceived loudness level of the ear as according to pure tone frequencies. The results, and later updates, have later been used to form the basis of the equal-loudness curves, which show that the hearing ability is highest at approximately 3k-6kHz and weakest for low frequency content. These properties have lead to weighting curves in order to reflect the varying sensitivity of the aural system. [9]

Performing A-filtering on a signal sample is a form of making it more representative as to how the sound is perceived by the ear. It is essentially a band-pass filtering which supresses the low frequency content and the high frequency content outside of the most sensitive aural range. It also gives a small enhancement to the predominant frequencies. [4, p. 53] [10]

2.6 Directivity of musical instruments

Directivity is a way to describe the radiation characteristics of a sound source. It concideres the angle of prapagation and the frequency content of the emitted sound. Directivity is described by the directivity factor DF, which will vary with the sound source. In general it is described as

$$DF = \frac{I_\theta}{I_{average}} = \frac{p_\theta^2}{p_{average}^2}, \tag{2.8}$$

where I_θ is the intensity of the source in a certain direction, and $I_{average}$ is the intesity average of an omni-directional source emitting the same sound power as the one being measured. [11, p. 40]

Dealing with an omni-directional source (DF = 1) is often the simplest case. For musical instruments the directivity factor will vary much depending on both direction and frequency. The complexity of the DF properties mean that this will not be a main focus in this report, however it may be worth knowing about. Some information on the topic can be seen in the paper "Directivity of musical instruments" by Magne Skålevik or in the appendix of Jurgen Meyer's book, *Acoustics and the Performance of Music* [12] [13]

2.7 Room acoustic parameters

2.7.1 Reverberation time

The reverberation time is considered one of the more important room acoustical criterias. The classical formula for this is Sabine's equation.

$$T = 0.161\frac{V}{A} \tag{2.9}$$

T is the reverberation time, V denotes the room volume, and A is the equivalent absorption area of the room.

2.7.2 Critical distance

The critical distance is defined as the point where the energy densities of the direct and the reverberant sound components are equal.

$$r_c = \sqrt{\frac{A}{16\pi}} \approx 0.1\sqrt{\frac{V}{\pi T}}. \tag{2.10}$$

The second part of the formula is obtained by the use of (2.9). Equation (2.10) is an approximation for the critical distance of an omni-directional sound source. This is the simplest case and is for this special case called the room radius. For other cases where the sound source has a varying directivity for angle and/or frequency, the critical distance must account for the specific sound radiation pattern in question, and so the formula becomes

$$r_c = \sqrt{\frac{gA}{16\pi}} \approx 0.1\sqrt{\frac{gV}{\pi T}}. \tag{2.11}$$

where g is the directivity factor of the direction where the intensity is at its max. g can be found as

$$g = \frac{I_{max}}{P/4\pi r^2} = 4\pi c r^2 \frac{w_{max}}{P}. \tag{2.12}$$

If there are sound sources of strong directivity present during measurements, careful attention should be made to the critical distance if the objective is to be outside of this range.

2.7.3 Reverberant sound and the diffuse sound field

The reverberant sound is the sound that arrives late, i.e. after the direct sound impulse and early reflections. The reverberant sound consists of reflections arriving from all angles and with a marginal time gap. The sound rays have been scattered around the room. With a continuous sound source of constant emitted sound, the reverberant field will hit an equilibrium at a point where the rate of the energy supplied is equal to the rate of energy absorbed.

The diffuse sound field is found at distances larger than the critical radius. Here, most of the sound has been scattered around in the surroundings, meaning that the reverberant sound is most dominant.

2.7.4 Strength factor G

The strength factor G is a parameter for describing the properties of a sound source in an enclosure without descibing the sound source itself, thus describing merely the enclosure. It serves as an indicator as to how loud a sound source or performance is perceived in a room. [14]

$$G_d = 10log\left(\frac{w}{w_A}\right) = 10log\left(\frac{T}{V}\right) + 45dB \tag{2.13}$$

Equation (2.13) describes the strength factor as it would be encountered in a diffuse sound field.

2.8 Distinction of sound components

Imagine that the soundscape is created by an orchestra and the reverberant response of the surrounding performance space. It is wanted to find the balance between the components of the sound. The total sound of the orchestra can be split into components of the individual musician (Self) and other musicians (Others), in addition comes the reverberant room response. In order to find out how much of the sound at the ear which is caused by the musician him/herself it can be usefull to introduce measures on how to separate the different sound components.

$$All = Self + Others + Reverb \tag{2.14}$$

2.8.1 Dry and reverberant components, for solo and orchestra performances

The *Reverberant* sound component is found by having two reference microphones in the diffuse sound field. An approximation of the reverberant sound close to the orchestra musician is made with

$$p^2_{\text{reverb, far}}(r) = 10^{0.1(0.176\frac{r}{T})} p^2_{reverb}. \tag{2.15}$$

p^2_{reverb} is the component by the musician, and the second term is a distance correction which comes from Barron's revised theory. [15] This method is equal for finding the reverberant component of both individual and orchestra measurements.

$$p^2_{self} = p^2_{\text{dry self}} + p^2_{\text{reverb self}}, \tag{2.16}$$

where $p^2_{\text{reverb self}}$ is found from (2.15). If parallell measurements are made close to a musician playing in solo in a room along with measurements in the reverberant sound field, the *Dry Self*-component can be estimated using equation (2.16).

The total sound field found with the orchestra can be regarded as a parameter consisting of the parts *dry self*, *dry others* and *reverb all*. The *dry others*-component is then found by

$$p^2_{\text{dry others}} = p^2_{orchestra} - p^2_{\text{dry self}} - p^2_{\text{reverb all}}, \tag{2.17}$$

where $p^2_{orchestra}$ and $p^2_{\text{reverb all}}$ could be measured in the same manner the parameters of equation (2.16).

Normalized sound components

Normalized elements of the sound components are:

- S = Self/All
- O = Others/All
- R = Reverb/All

2.8.2 Foreground-Background-Balance

Foreground-Background-Balance (FBB) is a parameter that gives a measure on the received direct sound of a source compared to the received direct sound from others and the reverberant sound. The parameter is found as FBB = FG - BG, where FG is the foreground, being the direct sound produced by the musician in question, and BG is the background component which consists of direct sound from others and the reverberant sound. FBB has been used/presented by Magne Skålevik in his paper *Rehearsal room acoustics for the orchestra musician* in order to give a simple presentation of the balance of the music components between the foreground and the background. [16]

$$p^2_{\text{background}} = p^2_{\text{dry others}} + p^2_{\text{reverb all}} = p^2_{orchestra} - p^2_{\text{dry self}} \tag{2.18}$$

The background BG can be found as the logarithm of (2.18), and the foreground FG as the logarithm of (2.16).

$$FBB = Foreground - Background = 10log\left(\frac{p^2_{\text{dry self}}}{p^2_{\text{orchestra}} - p^2_{\text{dry self}}}\right)[dB] \qquad (2.19)$$

This means that the logarithmic FBB is found from (2.19).

Chapter 3

Experimental approach

This chapter presents the experimental approach chosen in order to solve the problem at hand. It also describes some properties in connection with the measurement sessions.

3.1 Background and idea (principle of the measurements)

The object of the investigation is to study which sound components that dominate the received sound at the ear of an orchestra musician. The sound scape consists of the direct component of the musician's own instrument, the direct component of others in the orchestra as a whole, and the reverberant sound of the room. In order to learn about this it is necessary to separate the sound components. In order to have the ability to do this one needs to know some of the properties of the conditions under which the musician plays. E.g. properties of the performance space and the sound level which is produced by the musician himself.

To learn about the sound received at the musician's ears, a pair of microphones are mounted on their ears. It is also useful, if not crutial, to have some reference as to the level in the diffuse sound field, therefore two microphones are used to measure this. The reference microphones reveal information of the reverberant sound pressure level.

The musicians wore microphones on their ears during the recording, and two microphones were placed in the diffuse sound field in order to capture a reference of the reverberant sound. Two microphones were chosen for the reverberation reference as that would provide two indices of the sound level in the diffuse sound field. They were placed at the same seat row in order to have roughly the same distance to the stage. In the post-processing stage the captured results of the reference microphones could be averaged, thus providing smaller uncertainty of the sound level as compared to only one reference. The data from these microphones would later provide crutial information for the separation of the components dry self, dry others and reverb all. 2.8

The reference microphones were placed freely at a heigth of approximately 130-160cm whereas the ear microphones provided results influenced by the musician's head. In order to make the sound levels comparable these data would have to be converted into free field values, something which can be approximated by correcting for the effect of the head. The correction value however is dependent on the type of microphone in use. [17]

The principal behind the two measurement forms, individual and in orchestra, is that professional musicians are able to repeat themselves with impressive similarity, almost like a machine (see [1]). The method of choise bases itself on is this ability to give a good repeatition of the performance. The individual performance of the musician serves as a reference to the performance with the orchestra, and thus it should be possible to distinguish between what is likely the musicians own influence and what comes due to the influence of other musicians, i.e. the orchestra ensemble in this case.

The consistency of the musician makes it possible to do recording several times and expect approximately the same performance from the musician. This allows to do an individual session and use it as a reference to the orchestra performance. By doing this it should be possible to distinguish the sound level influence of the musician in question, and the rest of the orchestra. Although some part of the sound captured will be due to reverberation of the room, this influence is attempted to be kept at a minimum as the recording sessions are both done in the same room/space. The individual measurement is expected to have relatively more reverberant sound (in percentage) compared to the orchestra performance because of the influence by Others.

3.2 Position of musicians relative to the orchestra

Measurement sessions were performed in co-operation with the symphony orchestra in Trondheim. Four musical instruments were chosen for investigation; the violin as a representative for string instruments, the cello for the bass string instruments, the trumpet as a candidate for wind instruments and the oboe for wood wind instruments. The violin was especially chosen as it constitutes one of the main instruments in a symphony orchestra, playing almost constantly in many forms of orchestral music.

Figure3.1 illustrates the approximate and relative positions of the music groups measured on in this investigation. An idea of the measurements was to have the representative positions of the musicians. If the musicians have what can be regarded

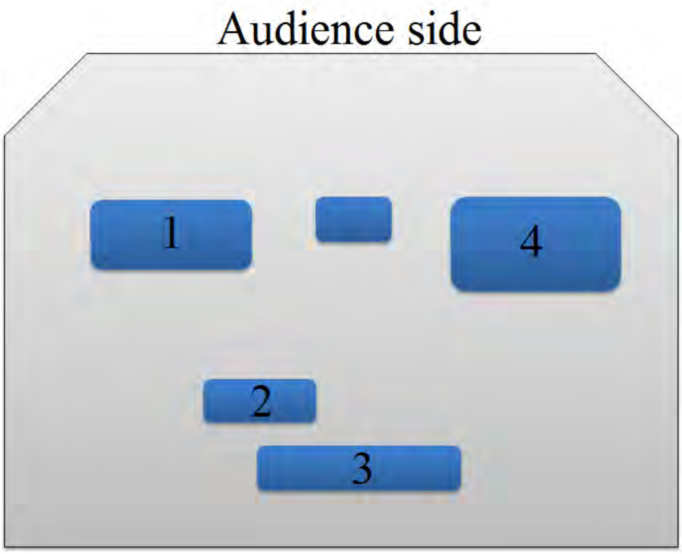


Figure 3.1: Positions of instrument groups in orchestra. Audience side on the top. 1 - Cellos, 2 - Oboes, 3 - Trumpets, and 4 - Violins, Conductor's stage left unmarked in the middle.

as an average position, the measurements will be more representative for a general outlook.

VIOLIN: The violinist was at a spot in the group with no extremes, at the second table with a left position, leaving her with one other violinist to her right. This means that the right ear microphone were supplemented with a source close to it instead of an empty room, thus it leaves us a properly average position and a representation closer to the averaged positioned violinist. If the violinist were at the edge of the orchestra,this would not be representative and some correction would have to be made for the right ear. All in all, we are left with something that can be regarded as sufficiently representative for the violinists. In the second recording session with violin, the musician was 1^{st} violinist and therefore had the seat closest to the conductor and at the audience facing edge of the orchestra.

OBOE: The oboist was seated a little to the left of the middle in the orchestra, being the rightmost musician playing the oboe. This means that the bass group was to the left, some wind instruments were behind, horn instruments to the right and violins located to the right some distance in front.

TRUMPET: The trumpeter was located at the last row all the way at the back of the stage, with two other trumpeters to the right and three trombones and one tuba to the left.

CELLO: The cellist was at the recorded occation seated at the last table on the conductors right hand side. She had one other cellist to her left-hand side and the

Figure 3.2: Mounting of an ear microphone demonstrated, the shown model is the Sennheiser MKE2 P-C.

contrabass group on her right-hand side.

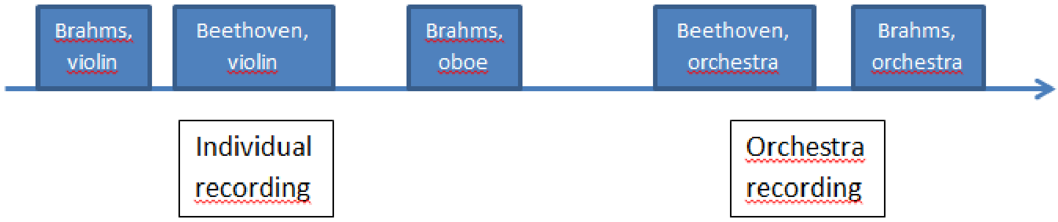
3.3 Instrument composition for recording

The measurements were performed as a recording session with the musicians. Two musicians were recorded in the same session, something which ensures that the music recorded of those intsruments were of similar character (time period, rythm, composition, etc.). Two reference microphones were placed (off the stage) in the diffuse sound field. Both musicians wore a pair of microphones on their ears, their mounting is illustrated in Figure3.2. All the microphones were connected to a 16 channel sound card and the software Adobe Audition CS6 was used for recording. This composition ensured a synchronized recording start for all microphones in use.

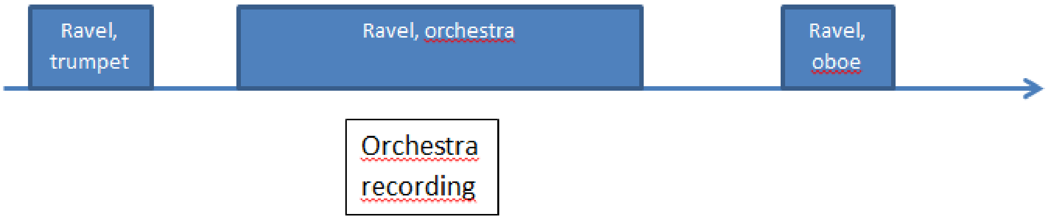
3.3.1 Musical pieces

Musical pieces used for the production of the results in this work is sections of *Egmont Overture* by L. Beethoven, *Symphony nr.1 in C-minor* by J. Brahms (violin and oboe). Pieces from M. Ravel's composition *Concerto pour la main gauche* was the contents of the second measurement session (oboe and trumpet). In the third session it was recorded on *Ungarische Tanze No.5* by J. Brahms, *The Barber of Seville* by G. Rossini, *The Nutcracker* by P. Tchaikovsky, *Battalia in D-major* by H.I.F Biber, and a known modern children's good night song in Norway by the name *Fantorangen: sove* [copyright Truls Waage].

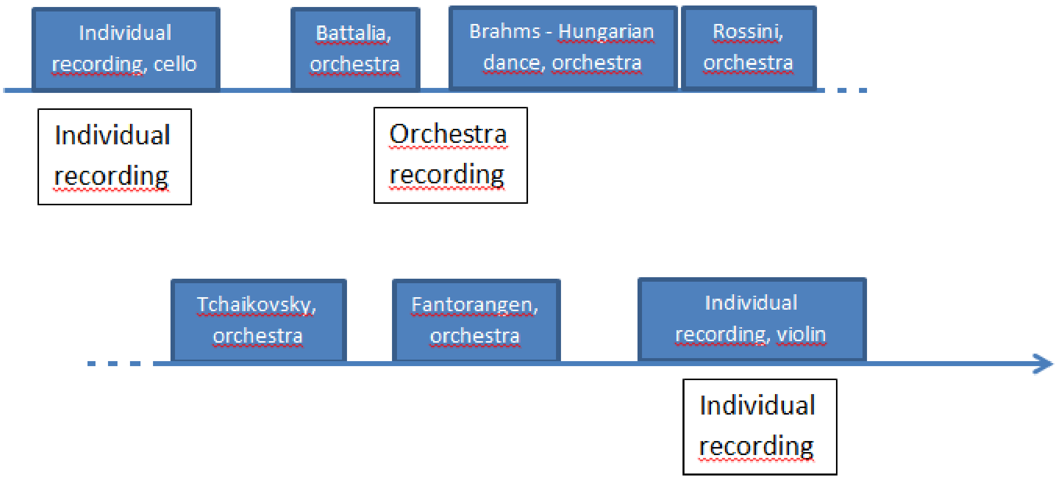
Figure3.3 shows the order in which the recording was done. It should be noted that the boxes labeled *Individual recording* in Figure3.3c are used as shorthand



(a) Measurement session 1.



(b) Measurement session 2.



(c) Measurement session 3.

Figure 3.3: Time sequential viewing of the measurement sessions. Time lapse going to the right-hand side.

notation for the individual recording of *all* the used compositions for the respective instrument.

Measurement session 1: Violin and oboe

Individual recording was performed within the hour before the start of the last concert rehearsal with both the violinist and the oboist.

Measurement session 2: Oboe and trumpet

Individual recording was performed an hour before the last rehearsal with the trumpetist. The oboist recording was for practical reasons, done shortly after the very same rehearsal.

Measurement session 3: Violin and cello

This session took place at an orchestra rehearsal the last day before the dress rehearsal. It was a show for kindergardens which included actors and sopran at some of the musical performances. Compositions which were influenced by this were Rossini, Fantorangen, and a little bit on the Tchaikovsky piece as the actors were running around on stage while making some sound (not singing or actually speaking). Individual recording with the cellist was performed during the main break, and with the violinist after the rehearsal ended.

Chapter 4

Measurement procedure and equipment

This chapter presents the method used for carrying out the measurements outlined in the previous chapter, and further the method for analysis of the data in order to obtain the results. The list of used equipment is also included here.

4.1 Procedure

4.1.1 Reverberation time

The reverberation time was measured using the standard procedure ISO3382 with six measurement positions for every source position, and using two source positions, which made a total of 12 measurements. [18],[19] The room was empty during the measuremens. The source signal used was a 20 second long sweep with frequencies from 50 Hz to 18kHz.

4.1.2 Room description

All measurements were performed in the concert hall of Olavshallen in Trondheim. This is a concert hall used for several purposes (multi usage). The hall is said to be intimate with its 31m from the stage to the farthest seat.

Every time measurements were taken for the purpose of this present thesis, the hall was always in orchestra configuration. The current orchestra configuration was first made in 2001. This was an alteration from the original configuration that was made at the opening in 1989 and which can be found described in the eminent Leo Beranek's book *Concert Halls and Opera Houses*. [20, p. 437] The author of this thesis has it from one of the acoustic consultants who worked on the original configuration and also with the later alteration, that the new orchestra cabinet was placed at approximately the same place as the original. This should mean that the data found in Beranek's book should be more or less correct also at the present day. One value which is used for the calculations in the processing is the volume of the concert hall. This is given to be $V = 13,000m^3$.

4.1.3 Microphone positions

The co-operating musicians wore a pair of small microphones which were mounted on their ears. This was done in order to capture the sound as heard by the musician, which in turn would be important to know for the separation of the direct and reverberant components of the produced sound. Figure3.2 illustrates the mounting of the earmicrophones in use.

In order to have a reference of the produced sound and capture the reverberance, a microphone would be requiered present in the diffuse sound field. The usage of two reference microphones in the far field, however, enables to make a more accurate estimation of the sound level in the diffuse sound field. This was thus the aim when choosing their positions for the measurement sessions.

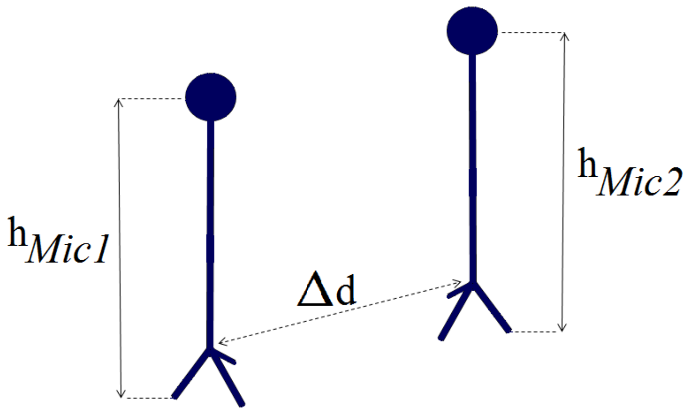


Figure 4.1: Reference microphones relative position to each other.

Table 4.1: Description of the location of the reference microphones relative to each other. Δd denotes the distance between the microphones and h the heigth of the respective microphone.

	Δd	$h_{Mic1}[m]$	$h_{Mic2}[m]$
Session 1	2.93	1.54	1.33
Session 2	6.48	1.54	1.34
Session 3	2.58	1.60	1.62

The recorings of a composition, both individual and with orchestra, were performed on the same day. The order of the recordings are shown in Figure3.3. The

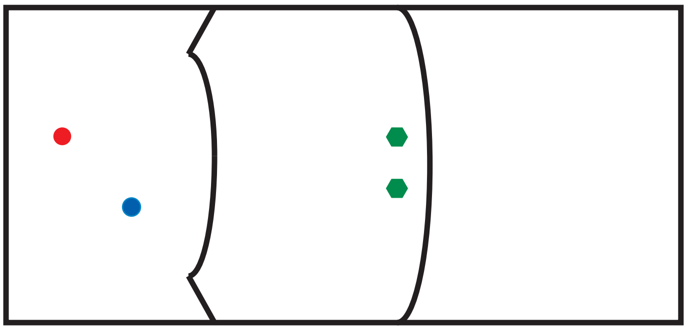


Figure 4.2: Microphone positions of the first measurement session, made together with the violinist and the oboist.

Table 4.2: Distance from the reference microphones to the musicians.

	Distance Mic1 [m]	Distance Mic2 [m]
Cello		
Session 3	11.89	12.84
Violin		
Session 1	9.94	9.50
Session 3	9.95	9.43

microphone positions remained the same for the whole duration of the measurement session. The relative positions of the reference microphones are given measures in Table 4.1 and illustrated in Figure4.1. The distance to the closest musician recorded is shown in Table 4.2. Due to many obstacles on the stage, the distance from the reference microphones to the oboist and the trumpeter was not measured, but estimated by measuring the distance to their respective podium. The estimate gave 18m distance to the oboist and 20m to the trumpeter. This was later estimated from the delay between the reference microphones and the ear microphones and found to be 18.4m and 20.2m respectively.

The green hexagons in Figure4.2 marks the reference microphones, the red dot (to the left) marks the position of the oboist and the blue dot the position of the violinist during the recording sessions. The reference microphones were roughly in the same position relative to the stage in all the measurement sessions. The musicians measured on would be present on the stage as according to Figure3.1.

4.1.4 Calibration and settings

The calibration of the microphones were performed in two steps. Due to different sizes of the microphone caps the process was done by measuring a sweep signal from a loudspeaker in an anechoic chamber and comparing the results of these to a reference microphone. The response from the reference microphone was then corrected with a measurement of a calibrator signal at 1000Hz at 94dB.

1. The microphones were placed directly onto the grid in front of the loudspeaker element. The measurements were performed consecutively with various gain settings on the recording equipment (equivalent to those used during recording sessions with the orchestra musicians). This process was then repeated with every microphone. All the calibration measurements were performed in the same session and at the same and optimal position on the loudspeaker.

2. The microphones used as far-field reference microphones were $^1/_2$-inch in size and thus fitted in the standard format of the B&K-calibrator. Reference microphone 2 is the one which served as reference for all the other microphones. The recorded signal of this was therefore used as a correction factor in the post-processing stage.

3. In the post-processing stage the impulse response signals were transformed to the frequency domain. A scaling factor was then found based on the difference in the range 100Hz-1000Hz between the microphone response and that of the microphone chosen as a reference, $^1/_2$-inch B&K-microphone 2. One single scaling factor was considered to be adequate due to a fairly flat frequency response of the microphones. The scaling factors were saved to a separate file (format *.mat*) and later used to correct every used signal when they were loaded into Matlab for processing.

4. Correction for different gain settings during recording sessions were done by using the most frequent gain setting of the microphone in question as a reference, then a correction factor was produced for the less used gain factors. The gain settings were corrected for in the Matlab-script when the recordings were read and cut into the used excerpts.

Table 4.3: Gain levels used on microphones during recording sessions as according to instruments.

Microphone	Sennheiser MKE 2-PC		Sennheiser MZA 900P		B&K 4165 $^1/_2$-inch	
	Left	Right	Left	Right	Mic 1	Mic 2
Cello	30/35	30/35	-	-	20/30	20/30
Oboe	30	30	-	-	30	30
	-	-	35	35	20/30	20/30
Trumpet	25/30	25/30	-	-	20/30	20/30
Violin	-	-	25/35	35	30	30
	-	-	30/35	35	20/30/40	20/30/40

Table 4.3 show the gain settings used during the recording. The gain of the Sennheiser earsized microphones was kept at zero at the microphone and it was

only adjusted on the sound card. For the B&K 4165 1/2-inch reference microphones the case was the opposite, as the gain was adjusted at the separate Norsonic Front end microphone amplifier, and kept at zero on the sound card.

4.2 Post-processing

The recordings were saved to wav-files of 32-bit quantization, which is the same as the resolution used by the recording programme Adobe Audition CS6. [21]

The music files used were cut in such a way that breaks were mostly cut out. For those files where the test musician played several sections in a composition, the composition was split into pieces where this was natural due to the breaks. If the musician had two or three beats to skip, then that was overlooked and put into the same take. The co-ordination of the individual takes to the similar orchestra takes was mostly done with the help of the music sheets of the composition where this was available. This helped the accuracy and sped up the process of making the approximate cuts of the recorded material, compared to not having the music sheet available. It helped as it was then possible to both to listen to and to read where the musician played.

The post-processing was then done in the computer program Matlab by Mathworks. The mono channel sound files were loaded into a temporary variable which combined the channels used in that recording. This variable had the left ear microphone in column 1, the right ear microphone in column 2, then the reference microphones in column 3 and 4 respectively. The channels were calibrated for their respective properties and then sent to a selection file for the respective instrument in question. This was done in order to simplify the next processing stages.

A script was produced in Matlab in order to save the selected data of the script `openSoundFile.m`, and the matching sound file for the individual recordings and the orchestra recordings. After selection these were adjusted in the cases where the gain settings of the recording differed from the reference gain setting of the calibration. The selected sound clip was then A-weighted with the function `afilter.m` and the RMS-equivalent level was found. The results of the microphones were then stored on to a variable for later use. In the end this variable was used in order to calculate the decibel values. The procedure described was used for each recorded instrument.

The short-term values and statistics were found by partitioning the selected parts of the sound files into parts of 1 second for the orchestra take and a little longer or shorter for the individual take according to its length. Statistics were then made on the short-term levels in order to get more detailed data of the music than what could be found for the procedure described in the section above. Because of the unevenness in excerpt length and the wish for whole second analysis, the last time slot of the signal had a data length different to the previous time slots. Because of this, the last value of return from the function `timeLP.m` was let out from the further analysis.

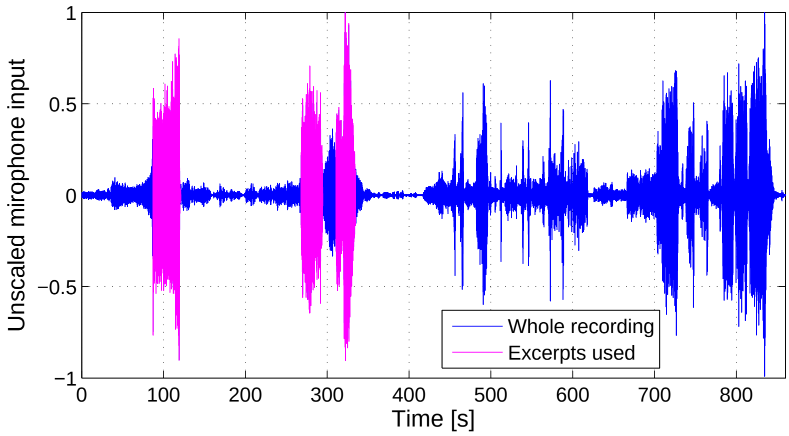


Figure 4.3: Orchestra recording of the oboe playing Ravel, with corresponding sections to individual recording. Mono channel viewing from the left ear microphone.

Figure4.3 shows an example of excerpts used from an orchestra recording. The blue line is the orchestra recording chosen and the magenta colored sections are corresponding musical pieces to the material gathered from individual recordings with the musician. The example is from the oboe takes of Ravel, take 6 to take 8.

4.3 Equipment

Device	Manufacturer & Model	Serial Number	Units
Calibrator	Bruel & Kjær, type 4231 (1kHz, 94dB)	NC-2021	1
Microphone($^1/_2$ inch)	Bruel & Kjær, 4165	2068937	2
		2068936	
Microphone body	Norsonic, type 1201	30651	2
		23824	
Microphone amplifier	Norsonic FRONT END, type 336	20579/	1
for B&K-microphones	(with 9V battery or power cord)	CB-2089	
Microphone earsized	Sennheiser MKE2 P-C	BC-4098/	2
		BC-4099	
	Sennheiser MZA 900P	No. 2045/	2
		No. 2047	
Loudspeaker	Genelec Active monitor, model 1029A	BA-2044	1
Sound card	Roland Studio Capture, model UA-1610	Z1D1698	1
Recording programme	Adobe Audition CS6		1
Sound generator	Computer with Winmls		1
Post processing	Computer with Matlab		1
Microphone rack			2
Microphone rack holder	(Small type for Bruel & Kjær mics)		2
XLR cables		35m	
Wires for microphones	Norsonic wires for Bruel & Kjær mics	25m	
BNC cable	BNC cable with XLR output	1m	2
Jack cable	One jack-to-jack and one jack-to-XLR cable		2
Microphone cap	Spherical cap for Bruel & Kjær mics		2
Distance measurement eq.	Bosch, PLR 30	RL-4001	1
Headset	Beyerdynamic DT990, professional 2x250Ω		1

Chapter 5

Results

This chapter presents the results of the investigation. First room acoustic findings are presented, then equivalent levels of whole recordings and the peak values found. The analysis presented is the separation of the sound components and the Foreground-Background Balance. In places where only one reference microphone value is found, this is because the two microphones were averaged.

5.1 Room acoustic parameters

This section presents some background material on the room acoustics found at the measurement location, Olavshallen.

5.1.1 Reverberation time

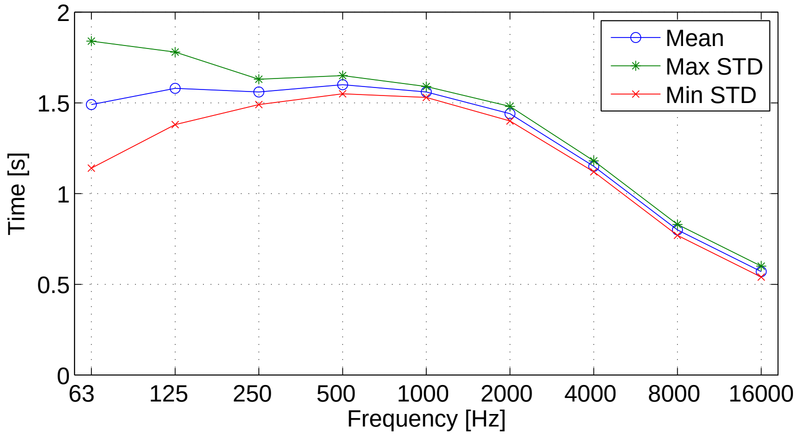


Figure 5.1: Reverberation time in Olavshallen.

Figure 5.1 show the measured reverberation time in the grand concert hall at Olavshallen where all the measurements were performed. For calculation purposes, the reverberation time has been set to 1.55 seconds as the main frequency content of the music would be in the mid frequency area, said to be 250Hz-1000Hz for this purpose.

5.1.2 Critical distance

The critical distance is much dependant on the directivity factor of the source. In this research little is known about the directivity of the investigated instruments since there are many unknown factors which would play a role in the measurement situations, with e.g. many sources at the stage, varying frequency content of the repertoar, etc. The number of sources being large make the source (all instruments, the stage as a whole) more similar to an omni-directional source as the sources are unequal (of musical instrument groups) and also have a large spread in frequency. Therefore it is possible to assume omni-directionality of the orchestra when viewed as one source, however the author realizes that this may be a bold assumption which may not be valid in all situations, e.g. solo play.

Table 5.1: Influence of the directivity factor on the critical distance.

Directivity factor, DF	0.5	1.0	1.5	2.0	2.5	3.0
Critical distance, r_c [m]	3.6	5.1	6.3	7.2	8.1	8.8

A schematic of the DF influence on the critical distance is presented in Table 5.1. These have been calculated with equation (2.11) from the Theory chapter. DF = 1 means that the source is omni-directional, larger values denoted a stronger directionality in the direction. Here the most likely situation is that the most directional instruments in the orchestra (trumpets, trombones, etc.), play in the direction of the audience area where the reference microphones were placed. The trumpets and the trombones were in all measurement sessions seated at the backmost places in the orchestra, some 2-3 meters in front of the back wall. This wall was measured at approximately 23 meters from the positions of the reference microphones.

5.1.3 Strength factor G

Table 5.2: Calculation of the strength factor G in Olavshallen, presented in octave bands with the corresponding measured reverberation time, T_{60}.

Frequency [Hz]	63	125	250	500	1000	2000	4000	8000	16000
T_{60} [s]	1.49	1.58	1.56	1.60	1.56	1.44	1.15	0.80	0.57
G [dB]	5.6	5.9	5.8	5.9	5.8	5.4	4.5	2.9	1.4

Strength factor G is presented in Table 5.2. It has been calculated by the use of equation (2.13), with the reverberation time as given in the table and with an approximated room volume of $13,000 m^3$ (as mentioned in subsection 4.1.2).

5.1.4 Background noise

Table 5.3: Background noise levels detected at individual measurements. $L_{A,short}$ presented.

		Duration [s]	L [dB]	R [dB]	$\overline{Refmics}$ [dB]
Cello	Session 3	2.5	41	41	33
Oboe	Session 1	5.0	43	43	31
Oboe	Session 2	12.0	37	36	29
Trumpet	Session 2	2.0	32	32	25
Violin	Session 1	3.0	39	36	29
Violin	Session 1	1.0	40	37	30
Violin	Session 3	1.4	43	43	34
Mean			39	38	30
Std			3.9	4.0	3.1

The background noise in the concert hall was found to vary between the measurement sessions and with the location of the musician present on the stage. The results are presented in Table 5.3 and they show that the background noise was roughly 8-9dB higher at the stage than at the reference microphone positions. The presented results were found from the individual recording sessions as these would not be subjected to talk or whispering from the other orchestra musicians.

5.2 Long-term equivalent levels

Table 5.4 presents the recorded material regardless of where the active/measured musician was playing, the values are equivalent levels for the whole composition. The exception is the recording of Brahms with the oboist, as he removed the microphones due to discomfort shortly after finishing the parts which were recorded in the individual session. This piece was therefore reduced in length. The removal was also audiable on the recorded sound file within 5 seconds after his part finished at 5min and 21s.

Table 5.5 show the the peak levels detected in the data of Table 5.4. Alas, some of the recorded material was subjected to overload. This means that the long-term equivalent level would have been somewhat higher (minor influence), and that this can not be corrected for as the information was lost. It also means that the true peak level is equal or higher than the one presented. The instances where this happened are marked in red, whereas the gray instance marks a case of a recorded channel left with an undocumented change of the gain settings. The shown value is therefore one which has been corrected for an *assumed* gain setting of 10dB higher than what was noted in the measurement journal (documented gain setting).

Table 5.4: Measurements with orchestra. The two reference microphones have been averaged. Presented material as $L_{A,eq}$ long-term equivelent levels.

	Musical piece	Duration [mm:ss]	Left ear [dB]	Right ear [dB]	$\overline{Refmics}$ [dB]
Cello					
Take 1	Brahms (Hungarian dance No5)	02:31	88	90	82
Take 2	Tchaikovsky	02:25	82	82	79
Take 3	Fantorangen	02:59	84	85	78
Oboe					
Take 1 - 5	Brahms	05:21	94	93	80
Take 6 - 8	Ravel	14:20	93	96	82
Trumpet					
Take 1 - 6	Ravel	14:20	98	99	82
Violin					
Take 1 - 2	Brahms	17:01	103	97	80
Take 3	Beethoven	08:53	100	94	82
Take 5	Brahms (Hungarian dance No5)	02:31	101	95	82
Take 6 - 7	Rossini	05:08	95	90	81
Take 8	Battalia	01:36	96	90	73
Take 9	Fantorangen	02:59	95	88	78

The gain settings change from the recording of Beethoven to Brahms of the left ear microphone channel with the violin were inadvertently left undocumented during the orchestra recording session. It seems likely however that the correction should be +10 dB as that is coherent with the other results of the violin. The results were therefore corrected with the factor found likely, and used as though they were this all along. An argument for the correction is that the uncorrected LR-difference would be -4.1dB for Take 1 and -3.4dB for Take 2 in the orchestra session. This would correspond badly with the individual recordings which had an LR-difference of respectidely 7.0dB and 7.1dB for Take 1 and Take 2. Also, with the assumed correction the results are coherent with the LR-difference results of the orchestra measurements of Take 5 - 8, the corrected data therefore make more sense. Corrected results of the LR-difference are shown in Table 5.6.

5.3 Short-term data $L_{A,1s/2s}$

In order to have comparable results between the reference microphones and the ear microphones at the musician's head, the data of the reference microphones were therefore converted to a non-free field value for calculation purposes. The correction made for this was an added 3dB for the influence of the head. This value was an estimate found for the type of microphone used from an article written by Kuhn. [17]

Table 5.5: $L_{p,peak}$ found in recordings of whole compositions played with the orchestra. All values in dB.

	Musical piece	Left ear	Right ear	Mic 1	Mic 2
Cello					
Take 1	Brahms (Hungarian dance No5)	116	116	108	107
Take 2	Tchaikovsky	108	107	106	107
Take 3	Fantorangen	107	108	101	100
Oboe					
Take 1 - 5	Brahms	119	115	105	104
Take 6 - 8	Ravel	122	121	109	109
Trumpet					
Take 1 - 6	Ravel	125	125	109	109
Violin					
Take 1 - 2	Brahms	131	121	106	107
Take 3	Beethoven	123	120	108	107
Take 5	Brahms (Hungarian dance No5)	123	120	108	107
Take 6 - 7	Rossini	120	116	109	109
Take 8	Battalia	118	116	95	95
Take 9	Fantorangen	117	110	101	100

It was found for the trumpet that the synchronization between the orchestra recording and the individual recording was so good that a realization of $L_{A,1s}$ could be done. For the cello, oboe and the violin, the synchronization at some of the takes were somewhat poorer, therefore it was decided that all the takes would be analyzed with a resolution of $L_{A,2s}$.

Figure 5.2 is an excerpt of the recorded music (whole take) cut into short-term sections, and the L_{Aeq}-levels were then calculated for each of those small fractions. The duration of the orchestra take was allways chosen as the reference length due to somewhat varying length of the individual data and the orchestra data. This means that the individual takes were sometimes analyzed in sections smaller than the reference length and at other times these sections were somewhat larger. Note that in the figure, some parts of the individual recording show to be of a higher sound pressure level than the orchestra recording at the corresponding time.

The script performing the sectioning was subjected to an energy test and it was found that the split signal only differed from the original, unsplit signal at the seventh decimal. This is reasonably good and well within engineering precision. Test data shortly seen in the Appendix, part D.3.

Figure 5.3 shows the short-term sectioning of the oboe take 5. In this take it is evident that a small break was included as the SPL of the individual measure-

Table 5.6: Left-ear-to-right-ear difference of the violinist, for individual recordings and orchestra recordings.

	ΔLR [dB]	
	Individual session	Orchestra session
Take 1	7.0	6.1
Take 2	7.1	6.7
Take 3	7.3	3.6
Take 4	6.6	5.7
Take 5	5.4	6.0
Take 6	5.6	5.7
Take 7	6.1	6.2
Take 8	6.9	5.2
Mean	6.5	5.6
Std	0.7	0.9

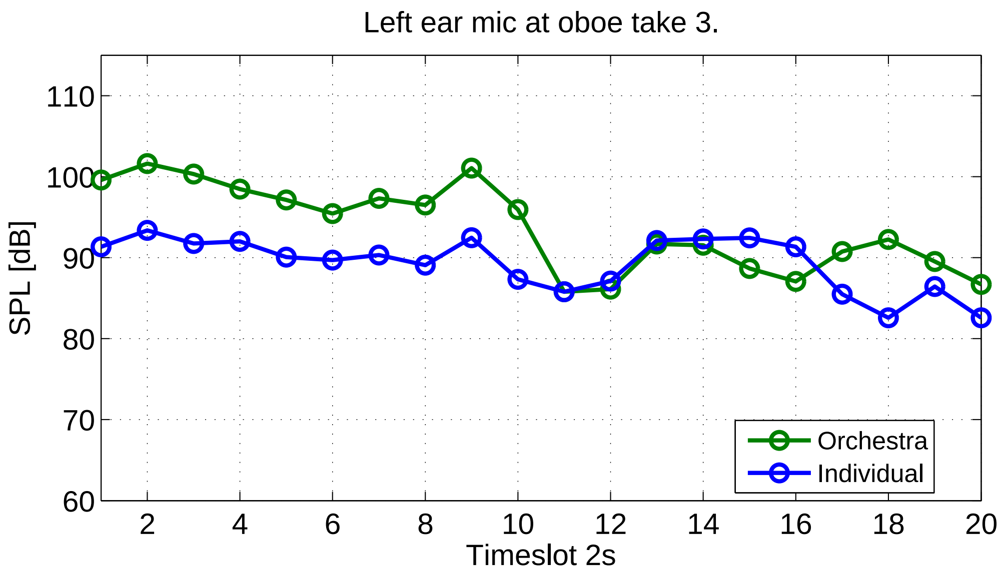


Figure 5.2: Take 3 of the oboe, left ear microphone. The orchestra recording was analyzed at intervals of 2.00s and the individual one at 1.81s.

ment decreases towards the background noise level at one point. The outcome data from the short-term sectioning of the takes are further analyzed in the next sections.

5.4 Sound components

Figure 5.4 examplifies two cases found with short-term sectioning of two separate takes. The data presented is the energy of the sound components as percentage of the total sound measured. The method of calculation was presented in the Theory,

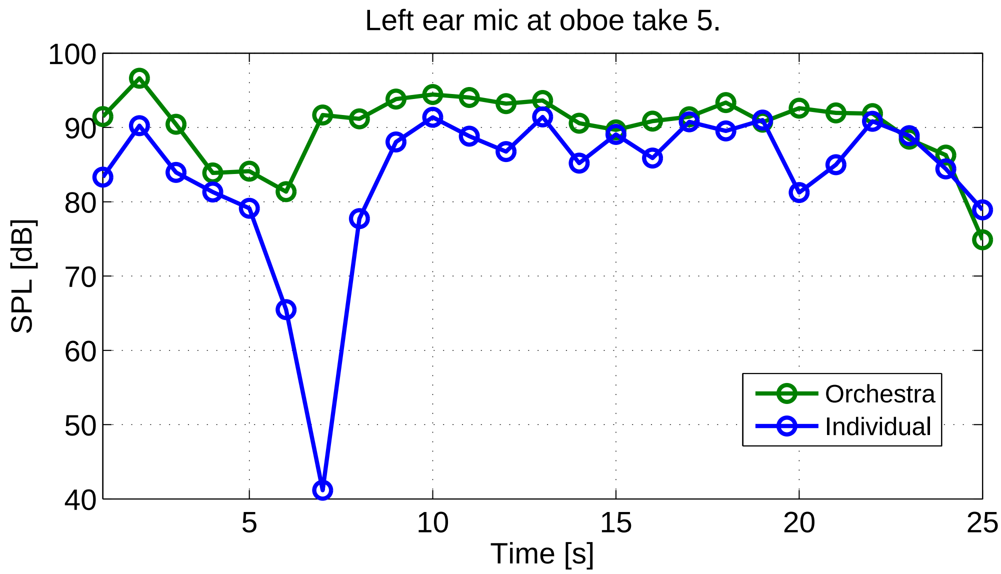


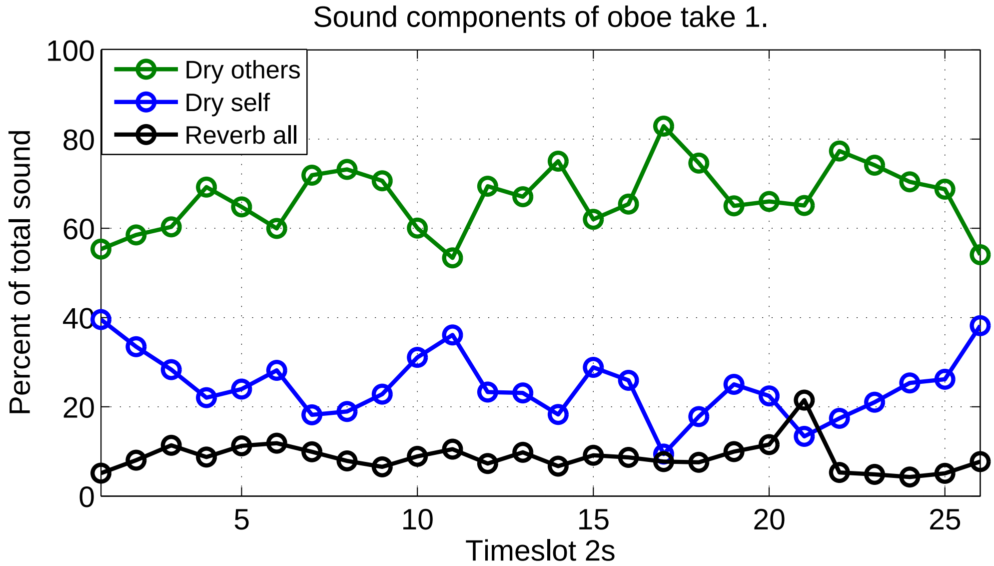
Figure 5.3: Take 5 of the oboe, left ear microphone. The orchestra recording was analyzed at intervals of 2.00s and the individual one at 1.86s.

part 2.8. From these is can be seen that the balance of the sound components shifts within the duration of a take. Other examples can be seen in the Appendix, part B.3.

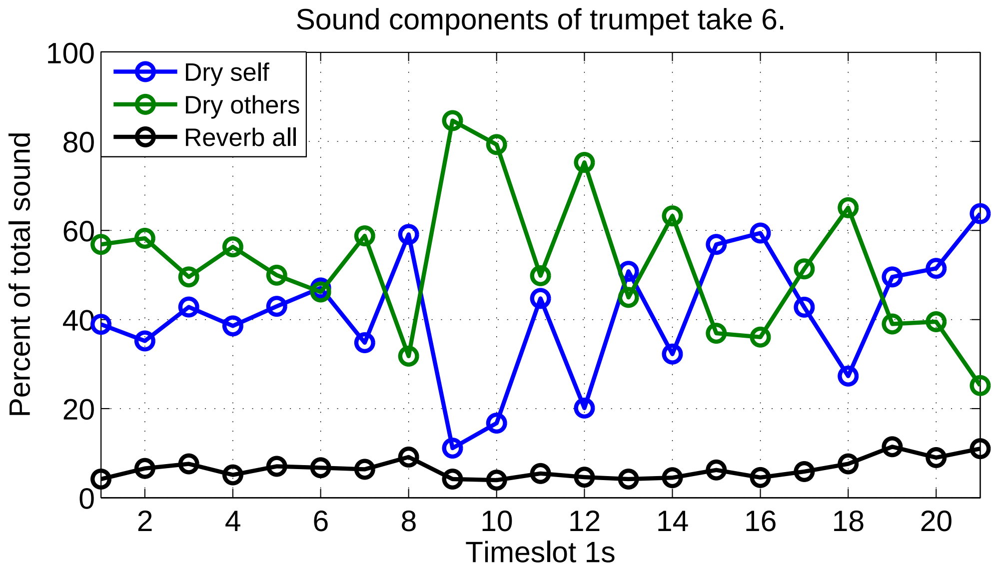
Alas, some odd occurrences found in some of the short-term sectioned takes made it strange to use the separation of components technique. This, as the *dry others*-component would sometimes appear profoundly negative. It was therefore found best to calculate the separation of the sound components from the equivalent sound levels in order to discontinue the problems with the short-term data. However, it can be mentioned that the *reverberant all*-component remained in a believable interval for almost all of the takes, even where the *dry*-components were out of bounds.

Figure 5.5 show the sound components of the takes. These were found from the L_{eq} of the whole takes due to the reasons given above. The results are spread over a wide range for the dry-components, but less for the reverberant component, and especially dense for that of the trumpet. The reverberant component of the cello results have a wide spread, which is much due to the variety of the musical contents. This opposes somewhat to the reverberant component of the other instruments, which is for all cases held in the intervall between 2-20%.

The spread in the results of the oboe and the violin resemble each other. The trumpet results stand out as the one which is the least spread in percentage. Details of the illustrated data are given in the Appendix, Table C.1.



(a) Sound components found in oboe take 1, data in percent.



(b) Trumpet take 6.

Figure 5.4: Composition of the sound components found for short-term sectioning.

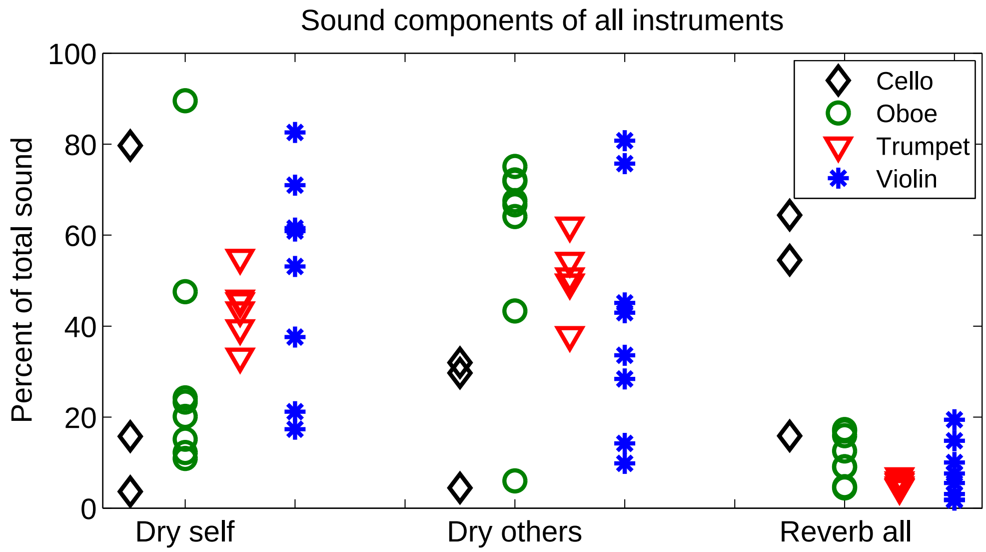


Figure 5.5: Sound components found in all takes and instruments.

5.5 Foreground-Background Balance

In the post-processing, some imaginary numbers occurred in the FBB results. These data were excluded from further processing as they are illogical. Reasons for this will be explained in the discussion, part 6.7.

5.5.1 Boxplot

Boxplot is a Matlab function which plots the statistics of an input matrix. On the x-axis one finds separate groups and on the y-axis is the range of values for the data of the groups. The box is limited at the 25th and 75th percentiles, and the whiskers show observations made outside of this, up to $\pm 2.7\sigma$. Points considered as extremal observations are placed outside of the whiskers and are shown as separate, red points. [22]

Figure 5.6 shows a summary of the FBB observations made of each measured instrument. One negative extremal point of both the oboe and the violin was left out of the shown window in order to better the interpretation. The figure demonstrates the main range of the observations. Note that there are more extremal observations found for the oboe and the trumpet than for the cello and the violin.

Table 5.7 shows the statistical variables found from the data background of Figure 5.6. (This also goes for the histograms presented in the next section, 5.5.2.) The statistics reveal that the noise exposure of the musicians is less due to the self produced sound than that of the background, as the magnitude of the observations are negative for all of the measured instruments. However, for the trumpet this is

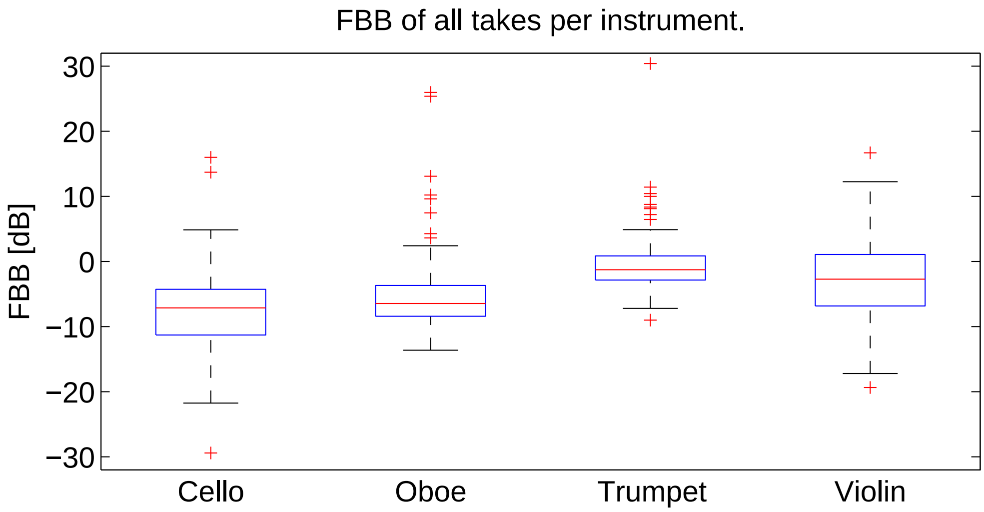


Figure 5.6: Boxplot of the total amount of FBB observations per instruments. One negative extremal observation of the oboe and the violin is left out of the viewed window (but included in the statistics).

Table 5.7: Satistical parameters found from FBB observations. The median is found from the boxplot.

	Median	μ	σ	No. of observations
Cello	-7.1	-7.5	6.6	116
Oboe	-6.5	-5.4	7.6	111
Trumpet	-1.3	-0.4	4.6	119
Violin	-2.7	-2.7	6.4	259

nearly not the case. Boxplots of the separate takes can be found included in the Appendix, part B.2.

5.5.2 Statistical populations

Extremal observations made with the cello were -29dB low and 16dB high, where the lower one was left out of the viewed window of the histogram in Figure 5.7. The histogram is fairly dense around the mean value at -7.5dB. A total of 116 observations were made.

For the oboe, a total of 111 observations were made. The histogram is given in Figure 5.8. One extremal observation was made at -49dB in take 5, however it was previously seen (Figure 5.3) that this take contained a break which lasted some seconds, so that in the individual take, the short-term values came towards the level of the background noise in that time. The data point thus came out with such a

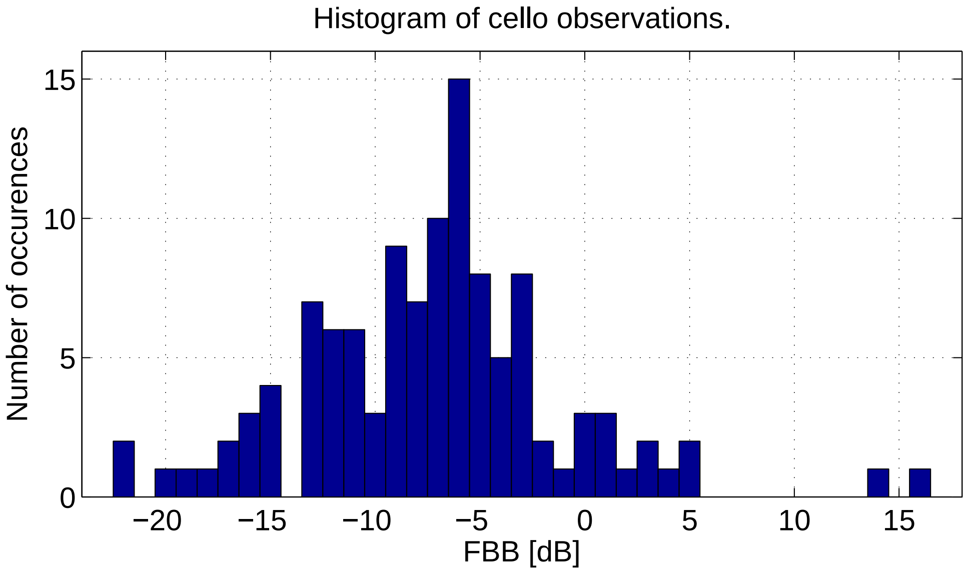


Figure 5.7: Histogram with 1dB bins over FBB observations from all cello takes.

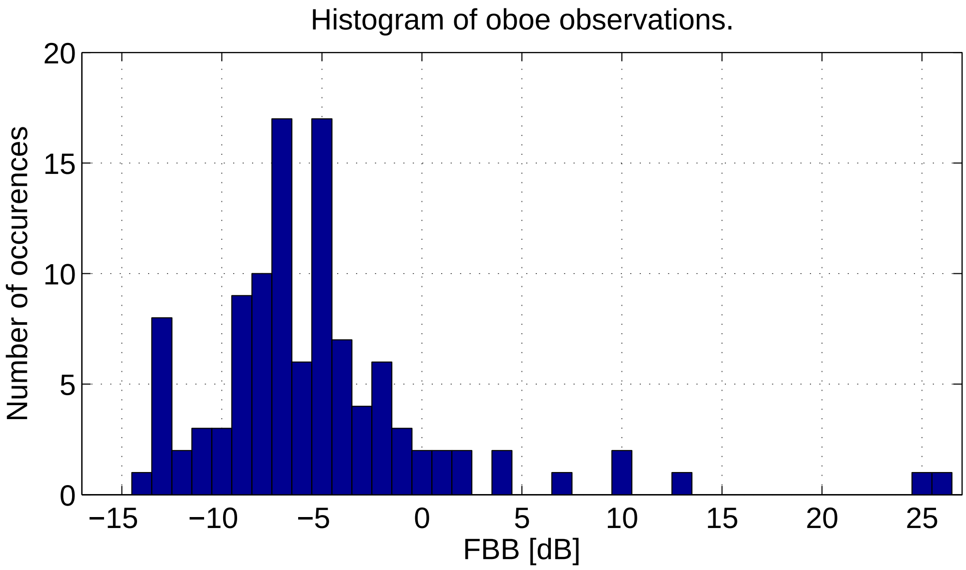


Figure 5.8: Histogram with 1dB bins over FBB observations from all oboe takes. One observation at FBB =-49dB is left out of the window.

negative FBB as the orchestra kept playing during that break of the oboe. The highest value of the oboe at FBB = 26dB was found in the oboe solo (take 4), this was also recognized in take 3 but then with 25dB.

Figure 5.9 shows a statistic view on how the FBB-results turn out for measurements made with the trumpeter. Observations of the FBB were made between -9dB to 30dB, with the mean value being hardly much negative at $-0.4dB$. This was found from a total of 119 observations.

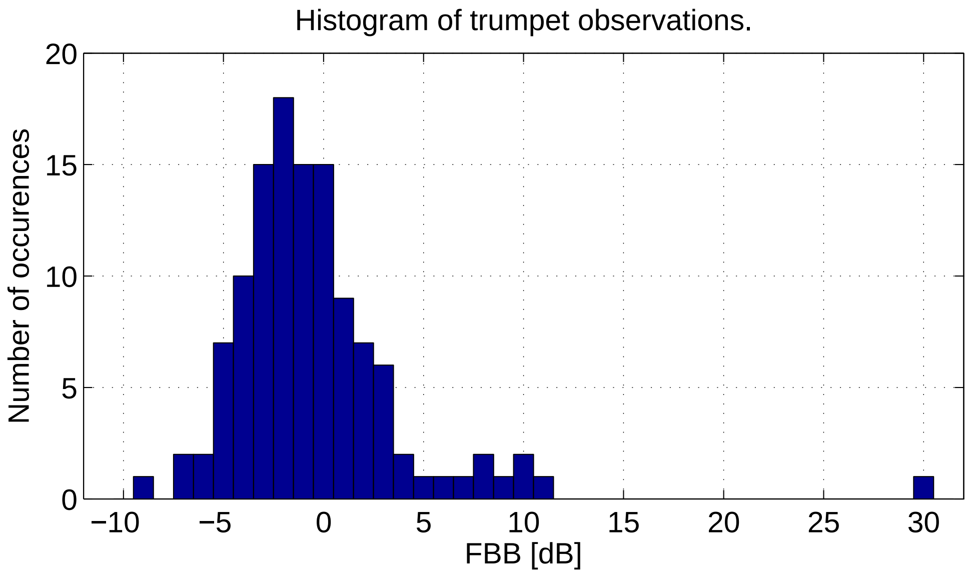


Figure 5.9: Histogram with 1dB bins over FBB observations from all trumpet takes.

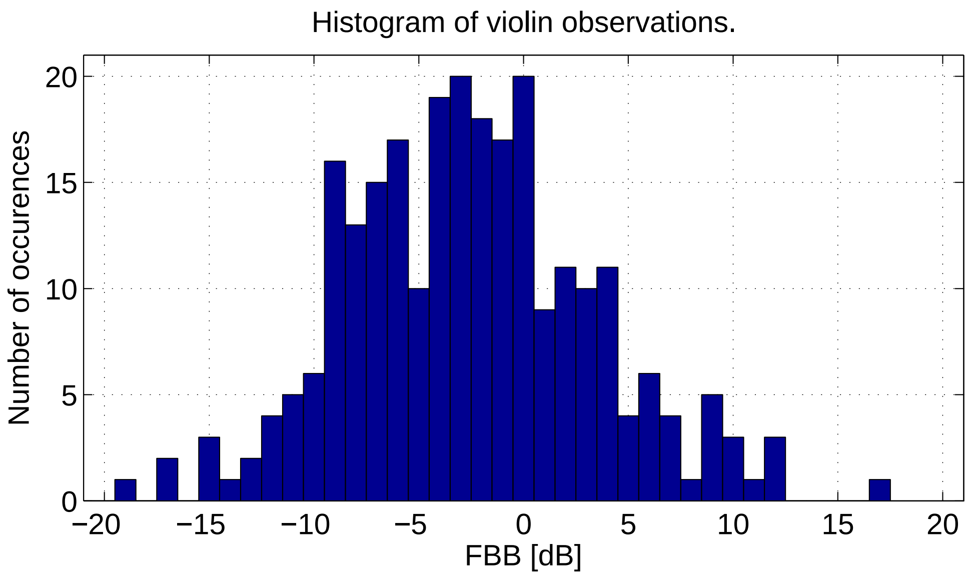


Figure 5.10: Histogram with 1dB bins over FBB observations from all violin takes. One observation at FBB = -45dB is left out of the view.

Figure 5.10 shows the histogram of the FBB observations made with the violin. The minimum level was found at FBB = -45dB, and the max at 17dB. The mean was found at $-2.7dB$ out of the 259 observations made.

Table 5.8: Kolmogorov-Smirnov test. Likelihood of dataset to make part in a normal distribution, where H=1 rejects the stand.

	H	p	No. of observations
Cello	0	0.2467	116
Oboe	1	0.0036	111
Trumpet	1	0.0108	119
Violin	0	0.4625	259

5.5.3 Kolmogorov-Smirnov test

Table 5.8 shows the outcome of the Kolmogorov-Smirnov test when the total of the observations made of FBB for the measured instruments were tested. Upon testing the data had to be temporarily adapted into a standard normal distribution. This was done in the manner of equation 5.1 where X were the observations and Z the approximated standard normalized data of the test.

$$Z = \frac{X - \mu}{\sigma} \sim \mathcal{N}(0, 1) \tag{5.1}$$

From the collected data, it is demonstrated that it can not be rejected that the observations made of the cello and the violin make part in a normal distribution. This is demonstrated well within the margin of the rejection criterion at $p \geq 0.05$. On the other hand, for the oboe and the trumpet the hypothesis of such a distribution is clearly rejected. If the extremal observation of -49dB of the oboe is left out of it's statistics, the likelihood of it being of a normal distribution even decreases to a p-value of 0.0026. This means that the inclusion of the break to the dataset rather increases the likelihood of such a distribution.

Chapter 6

Discussion

6.1 Critical distance

Michael Barron found in his paper that most full-size concert halls have a critical distance around 5 meters. Table 5.1 was calculated using the specific parameters found for Olavshallen. It is demonstrated that the Hall radius of DF = 1, is $5.1m$ and thereby close to the rule of thumb from Barron's paper. [7] With this background it seems likely that the reference microphones were in fact in the diffuse sound field in general. Exceptions from this can still have occurred for solo and maybe also semi-solo parts of instrument groups with strong directivity, e.g. trumpets and trombones. Yet, these instruments were present only at the backmost part of the stage, with an approximate distance of 18m from the location of the reference microphones. Also, pure solo play was not noticed from the instruments mentioned during the measurement sessions. From tables in the appendix of J. Meyer's book, the highest directivity factor was found to be 6.6 at 15kHz.[13, p. 413] In Olavshallen this would give a critical distance of 13.3m. It therefore seems argumented that the microphones were in fact in the diffuse field.

The one time where this may not have been the case is for the third measurement session when there were singing actors present on the stage. The critical distance would be calculated to 7.3m using DF =2 (again found in J. Meyer's book, [13, p. 414]), while the radius between the edge of the stage and the reference microphones were measured around 9m. Yet, this may actually have been the most critical point with regards to sound attenuation as the singers would amplify the sound level in between the measuring points, with more directionality towards the reference microphones than the musicians. Singing was performed during the opera number *Rossini*, and less strong singing was present at *Fantorangen*.

6.2 Absorption on stage during individual performance and orchestra performance

A qualified opinion says that the difference in room absorption between the individual recording and the orchestra recording, is unsignificant as there were no highly reflecting surfaces nearby the playing musician. The solo musician was alone on the stage during individual recordings. The nearby surroundings were thus not people but chairs. These chairs were polstered and should have an absorption factor at about 0.5, therefore the reflection from these were not high, and the absence of other musicians should be negligible within the uncertainty of this investigation.

6.3 Consistency in SPL for instruments

The observed SPL results for the oboe and trumpet are fairly constant, whereas the results of the violin and the cello deviate more. It should be noted that the number of different musical compositions measured play a role here. Within one composition, the style and the method of playing may be very similar, while these can be very different from one composition to another. This is as the musical composition style, and to some extent, the playing methods and the instruments themselves have developed throughout history.

The observations of the trumpet was made from one composition, for the oboe this amounted to two different compositions, for the cello it was three and for the violin the results came from six musical compositions. For an investigation with large default uncertainties, the data amount collected is critical for the accuracy of the results.

6.4 Long-term equivalent levels

The long-term equivalent levels were found regardless of where the musician in question played.

The results give equal SPL at the reference microphones matching musical pieces, as expected since they were the same recordings. The incident where this is more interesting is the recording of Brahms for the oboe and the violin, as the recorded material of the oboe was cut short for all the microphones in the post-processing stage due to removal of the ear microphones. The matching result of the reference microphones then reveal that the equivalent SPL after this time was similar to the part previous of the ear microphone removal. This means that an increase in the dynamic of the music was not detected for the latter parts compared to the earlier parts. Still, the ear microphones reveal a difference in SPL between the violinist and the oboist of 11dB on the left ear and 4dB on the right ear. This means that the violinist was subjected to more noise than the oboist.

The left-to-right-ear difference detected for the oboist at the Ravel measurement may be due to the positioning of the musicians as the oboist had the timpani to the left hand side and the wind instruments group to his right hand side.

From the takes which are similar between the cello and the violin, it is interesting to see that the violinist is exposed to much more stress than the cellist. In the *Hungarian dance* the violinist was subjected to a SPL of 13dB more on the right ear and 5dB more on the left ear. For the piece *Fantorangen* the difference was reduced to 11dB at the left ear and 3dB at the right ear. These differences reveal that the violinist was present in a part of the orchestra with much more sound than the cellist. It may be mentioned here that their positions in the orchestra were very different and that this, in combination with the instrument directivity, is an explanation for the large difference between the left ear and right ear SPL.

6.4.1 $L_{A,eq,long}$ in an HSE perspective

From the violin measurements, it is apparent that the musician's left ear is exposed to 5-7dB more than the right ear. The higher exposure on the left ear follows naturally due to the way the instrument is held and played. The violinist also reported that she had tinnitus on the left ear. This in combination with the measured results, reveal that it may be advantageous if something can be done in order to reduce the noise exposure on the ears, and especially on the left ear.

The long-term SPL does overall show that the highest noise exposure is found for the violinist in this investigation. In a more thorough research on symphony orchestras made by O'Brien, it was seen that the highest equivalent sound levels would be found with the trumpets, then a moderate level for the violins. A more acceptable exposure giving low risk of hearing loss would be present with the oboes and the cellos. [2] The equivalent levels presented in Table 5.4 show the same tendencies as found by O'Brien, only with the reversed order for the violinist and the trumpetist. The SPL found with the violinist is in general very high.

The collected data in this investigation do not reveal a lot about the strain of the ears in a long-term perspective or in dosimetry of a working day, but show examples of stress exposure for the musicians ears.

6.4.2 $L_{p,peak}$ in an HSE perspective

Sound in the perspective of Health, Safety and Environment, is most critical where the sound pressue level is high. This means that the peak level is critical when speaking in terms of work environment and safety for the hearing ability. The stress level that ears are allowed to be exposed to during a working day is in Norway set at 85dB equivalent level during an exposure time of 8 hours. Equally, the C-weighted peak level has a limit of 130dB. This means that if values higher than this are found, the work place/employer has to find a way of reducing the noise which the employes are exposed to, otherwise they can be punished by law. [23]

A peak level of 130dB at low frequency may be found pleasant by the listener while the same peak level would be very unpleasant, if not to say great risk for the hearing ability, at frequencies in the range $[3-6]$kHz. This is due to properties of the aural system as briefly described in the Theory, part 2.5.

Although the measurements were sometimes found badly suited for this kind of detection, they still give a look into the dynamic level of the music performed in the measurement sessions of this particular investigation. The peak levels are interesting to know as impact sound can kill the hearing ability for certain frequencies. High levels over time will wear down the hearing threshold but impact sound may do off with it instantaneously. [23]

6.5 Short-term sound pressure level, $L_{A,1s/2s}$

The musical contents of the takes correspond well for the computation of equivalent levels. A short-term analysis was wanted in order to see how the contents change over the course of the measurement. However, while it would be nice to have much data samples, the samples also had to be representative and of a certain quality. That is to say that the energy contents of the recorded material should match in order for the underlying materia in orderl to be comparable between the orchestra takes and the individual takes. High level of accuracy in synchronization would make it possible to use high resolution and small sections of analysis, and the opposite made it necessary to use lower resolution. This is as smaller intervals of analysis make the timing of the musical content become increasingly more crutial.

Properties of the synchronization between the orchestra recording and the individual recording was found to induce a problem for the short-term sectioning of the recorded material. It was found that for the trumpet takes it was possible to use one second intervals as these were very precise. In listening tests it was even found pleasant to simultaneously listen to the orchestra take and individual take. For the oboe, some of the takes were found equally good as for the trumpet, however others may have been up to 0.6s out of sync. Therefore it was considered better to perform analysis in two second intervals. This was to ensure that most of the energy would be analyzed at the corresponding section between the measurement forms. This was also done for the cello and the violin takes as they sometimes suffered from the same problem as the oboe. The reason why the synchronization became an issue in the first place was that the lengths of the matching recordings were of unequal lengths, as can be seen in the Appendix, Table C.2.

6.5.1 Possible reasons for problems with short-term sectioning

The reason why the short-term data was considered to be unsuited for the analysis of sound components was that the *dry others*-component appeared to be strongly negative at some instances. In a physical perspective this should not be possible and after some effort to solve the problems it was considered best to not carry on with the analysis in this manner.

It was believed that the *dry others*-component became negative at certain points where the short-term SPL was found to be higher for the individual measurement than with the orchestra. These occurring instances contradict the assumption for the chosen measurement setup. However there could be many reasons why it happened. For one, the short-term data show very detailed what went on in the recording, so small fluctuations became more significant. It may be due to a combination of systematic or random causes. For instance, the conductor may have wished for the orchestra to have a softer performance than what was previously rehearsed. It may also be due to a bias of the musician, as they are not used to performing in a grand space like Olavshallen all by themselves. A systematic or unsystematic effort for compensation of the lack of response from the surroundings. It can also be due to a flickering performance of the musician, which may be unsystematic or systematic in occurrence.

An energy test of the function performing the short-term analysis showed that the energy of the split signal did not deviate significantly from the original, unsplit signal. Also, from the plots of the sectioned recordings, no obvious asymmetry in time was found between the two measurement forms. Yet, synchronization problems may have occurred due to small differences in the content of the recordings.

6.6 Sound components

It was sought to find the short-term components of the recorded material. Alas, such efforts did not succeed. A hypothesis that it could be due to problems of synchronization was proved wrong as the resolution was increased, but little about the problem was changed. The odd occurrences could therefore not be caused by energy at uncorresponding sections.

At those takes where the short-term sectioning was found to be reasonable, however, the sound components *dry self* and *dry others* behaved almost like opposites.

This seem reasonable from the equations of the Theory-part 2.8, where *dry others* is shown to depend on the other components. *Reveberant all* is in calculation not dependant on the other components but relates to the sound level of the whole orchestra. In percentage it is fairly stable during the performance, and therefore, the *dry others*-component depends mostly on the *dry self*-component. This was demonstrated in Figure 5.4.

6.6.1 Separated components from $L_{A,eq}$ data

The musical content of the cello recordings varied from a pizzicato piece to what was almost a cello ensemble solo. The variance within these takes are sadly not reflected since they were calculated from the equivalent sound level. Yet, the deviation of outcome make sense given the content of the music.

For the separate takes of one instrument, the sound components dry self and dry others appear to behave almost like opposites. This should be expected given the reasonably stable content of the reverberant component and the calculation method. What it means though, is that from the reportoar screened, the main source of the sound exposure shifts back and forth between the musician himself and the other musicians.

It is interestingly found for the trumpet takes, that the balance of the sound components shifted little, and that a very stable percentage was found for the reverberant sound. The dry others component was found to have an average of 51%, the dry self component at 44% and reverb all at 5%, the latter with a deviation of 1%. This is coherent with results from the FBB analysis where the trumpet was found to have a mean slightly less than zero. Since all of the takes were excerpts from the same musical composition, it would in further work be interesting to see if the results from the trumpet remained as stable if more varied contents were studied.

The reverberant all component is found within the interval 2-20% for all instruments excluding the cello. The mean value of reverberant level is found to be 5%, 10% and 12%. For the cello this is not included as there were few samples and a high deviation between them.

The relatively similar values found for the reverberant component means that it is not the room, but the contents of the music which shifts the balance perceived. This may be expected since the room remains the same but the sound sources shift with the excerted music. Knowing that the contents of the takes for the cello varied much, it seems indicated that further screening, collecting more data, ought to contain a variety in repertoar and playing technique included. This could have been better reflected also in this investigation, had it not been for the faliure of the effort to perform the short-term analysis.

6.7 Foreground-Background Balance

The appearance of imaginary numbers in the data would mean that either the foreground or the background energy was negative, something which does not give meaning from a physical prespective. The occurrence of imaginary numbers however, states that the belonging energy of the background components, *dry others* and *reverb all*, were less than that of the foreground, *dry self*.

A reason for a soundwise weaker play of the orchestra can be that the synchronization of the recorded material came out wrong, so that the content was in fact not corresponding. On the other hand, if the synchronization was good enough, it may be due to an increase in the absorption of the orchestra excerpt from the individual excerpt. Another reason can be that the conductor wanted a softer performance than that which was known by the musician beforehand of the individual rehearsal recorded. In the case of the oboist it can be added here that the oboes are sometimes told not to play at musical sections where the orchestra play particularly loud. This, as the oboe gives a relatively weak sound level compared to many instruments, and that its contribution therefore would not be perceived in the total sound.

In the case of the FBB data, it was simpler to find a way to work around the issues of the short-term sectioning than for the separation of the sound components. This was made possible by excluding the imaginary occurrences from the datasets used in the analysis.

6.7.1 Boxplot and statistical populations

The boxplot which summarizes the observations made for the instruments, reveal that the FBB varies a lot. The limit of the boxes show the 25% and 75% limits of the observations, and the red points are the observations that are considered to be outside of the distribution. The boxplot then reveals that the trumpet is likely to be of an FBB just below zero. This is similar for the violin but with an expected value at about 2dB less and a larger variance.

The number of observations considered as extremal points are larger for the oboe and the trumpet than for the cello and the violin. Also, those extremal points are often on the positive side of the FBB. From the histograms it may be noted that their tale to the right hand side is longer than that to the left hand side. For the cello and the violin these tails are reasonably symmetric. Maybe with a slight skewedness to the left for the cello.

From these observations it seem evident that the sound level at the musician's ears will vary between dominance of the foreground, which is the self produced sound, and that of the background, consisting of sound from others and reverberant sound. The cello is the instrument in the investigation which is found to be the weakest in terms of the FBB. The oboe is found have the second weakest Foreground-Background Balance, and its variance is the largest out of all the instrument samples made. Knowing that the music contents of the oboe data varied from a solo part to an inclusion of a break, this is not startling. By the collection of more data, such events would be less dominant.

The trumpet histogram from Figure 5.9 showed signs of having a tale to the right hand side, and otherwise a concentration about an FBB of -2dB to -1dB. These results also had the smallest variance out of the observed instruments.

The FBB results of the violin had a variance which was pretty close to that found for the cello. The mean and the median value were almost the exact same. From the histogram it is shown that the foreground of the violin often would be less dominant, but also fairly often dominant at the ear of the violinist. However such positive FBB extremal observations as found in the oboe and the trumpet were not found in the violin results. In fact, only three out of 259 observations were found to be extremal points.

6.7.2 Kolmogorov-Smirnov test

All the histograms of the recorded instruments show fairly dense distributions centered about their mean. Some extremal points are found but that must be expected.

The cello and the violin observations make it possible to have seen normally distributed FBB-results, while this have been rejected for the case of the oboe and the trumpet results. The number of datapoints to the violin dataset makes it a fairly solid statistic with the highest number of observations out of the measured instruments. Its confidence of p=0.46 also beats that of the cello at p=0.25.

6.7.3 Negative FBB

An occurrence of negative FBB means than the musician heard less of his/her own performance than of the background, *dry others* and *reverb all*. A very negative FBB mean that the musician heard little or even nothing of himself, but could also mean that he did not play at that instant. This was found at the oboe take 5, Figure 5.3, where a break was small enough not to have been cut out, but also large enough to be clearly detected. This illustrates why it may be advantageous to skip longer parts in the recorded material where the recorded musician is inactive. Yet, as parts of the musical contents, shorter breaks will occur every now and then, therefore they should not be neglected entierly. The best thing is possibly to be aware of their influence on the data.

It can be kept in mind that a difference of 10dB is found to give a sound level difference so large that the musician would struggle to perceive his/her own performance at all. In orchestra music it is necessary to have a balance between the perceived sound of one self and the other musicians since it is an ensemble. If an attenuation of one's own performance is demanded, then the musician could use a screen shaped as a cut half circle around the head, having it at ear level, sit back into it and experience an attenuation of the self-produced sound. This gives a reference for the musician as to his/her own performance. From a health perspective however, it is not beneficial as the sound is attenuated positively and not reduced, meaning that the SPL is increased rather than reduced.

The very negative mean of the cello and the oboe leaves questions as to how the musicians are able to know what they are playing.

6.7.4 Summary of the FBB results

From the results, it is evident that the trumpeter experienced a Foreground-Background Balance which was hardly on the negative side. The violinist was a little more on the negative side with an average of -2.7dB. The oboist made an average of -5.4dB. For all of these instruments it was found a variance in the collected data which is large enough to indicate that the FBB shifts between negative and positive values. This means that the musician will sometimes hear himself more than the others, but most of the time the his own instrument will be less dominant than that of the others.

6.8 Aspects on relative performance

The position of the musicians relative to the orchestra may give influence to the results. This would especially influence on the cello measurements, as the cellist was seated on the backmost table at the right side of the stage. This was not the cellist groups normal position, and it means that the cellist had no sound sources behind her, and just one other musician to her right hand side. It would be recommended to avoid such outerpoints of observation. Yet this property of the position leaves a question as to how the balance of the sound components would differ if the relative position of the musician had been at another location in the orchestra. Is it then possible that the reverberant component would be less prevailing? Is there a possibility that the dry self component would be even weaker as there would be more sources surrounding the cellist? Are these musicians formidable at separating the own sound from noise, given such a weak balance in favour of their self produced sound?

The musical content in a performance is however seldom static over longer periods of time. Can it be that the variance in content leaves these musicians just enough of a reference to the direct self sound so that they manage to know what they are performing? After all, they have rehearsed the music well before the performance together with the orchestra, and are also experienced with the performance in orchestra situation. Yet, the question arises: Do they really know how they perform/sound?

6.9 Further work

From the study carried out in this work, it was found that some improvements could be made, and suggestions to relevant, interesting topics.

- More measurements on each instrument (in order to have better data), different music styles (era). This especially goes for the trumpet as it in this investigation were made measurements on just one composition. This ment that the musical expression was similar between the takes of the trumpet. A consequence is that the true variance of the instrument is not wel reflected in the gathered data material of this particular investigation.

- Measurements of different positions for the instrument, possibly with several musicians, could be done in order to study the effect of positioning in orchestra. Also, to see how the results are affected by the influence of the relative position measured within an instrument group, e.g. 1st violin and 3rd violin, edge position vs middle position.

- The FBB results in this report were made out of an average of the left and the right ear of the musicians. In a further study, and especially if theinfluence of the relative position of the musician is to be investigated, it may be useful to also consider the difference in sound exposure per ear. Because of its directivity properties this would be very relevant for the violin.

- The synchronization between the orchestra and individual musical excerpts could benefit from doing it in a more sophisticated manner. This could improve the accuracy of the synchronization and in turn the possible resolution of the short-term SPL analysis. This would provide more data from the measurements, and give it better quality for the statistical analysis. A suggestion in further work on this is to see if the energy is placed in the right time intervals for the sectioning. A possible way to investigate this could be to perform a frequency analysis of the contents.

- In order to investigate a little on the reproducibility of the musician, individual music samples with the musician could be made on two different times. It could be interesting to see the similarity of the two takes in order to have some comparison on how much the performance of the music differs from repetitions. The musical piece would have to be well rehearsed beforehand for having a relevance, so a suggestion could be to make individual measurements of a small selection of pieces (one or two) both before and after the last concert rehearsal, and then compare them in the analysis.

- A suggestion for further work is to investigate the effect the human head has on the captured SPL in a symphony orchestra situation.
- In further work it can be of relevance to carry out an analysis to see if separate takes form an undergroup to of a certain significance to the overall statistics of the instrument.

Chapter 7

Conclusion

The ability to repeat a performance with exactly the same presision was found to be an assumption with some modifications. For a practical approach of the problem at hand, it is an assumption which still opens some possibilities with regards to the problem at hand for this Master's thesis.

It has been argumented that the reference microphones were outside of the critical distance and in the diffuse sound field for all occurred cases in the measurement sessions. This was an important argument as the separation of the sound components would be of low validity if they were impacted by direct sound.

Short-term analysis were performed in sections of one second for the trumpet takes, and in two second intervals for the cello, the oboe and the violin due to somewhat erronous synchronization between the orchestral and individual recordings. It was found that the short-term results were unsiuted for telling the sound components apart as a general rule, since the direct sound component of *other* musicians, *dry others*, became negative at some instances. This gave illogic results for the sound component separation of *dry self* and *dry others*, while the reverberant component mostly remained in order. This forced the analysis of sound components to be performed from the $L_{A,eq}$-data, i.e. equivalent levels of the musical excerpts. Solutions was however worked out for the Foreground-Background Balance.

The analysis of the sound components revealed that the reverberant part of the sound was mostly found within 2-20% in the measurements with the oboist, the trumpeter and the violinist. With the trumpeter, the direct-self part was found at an average of 44%, the *others*-component at 51% and the reverberant part at 5%. The standard deviance was found to be 7%, 8% and 1% respectively. This was the most consistent result from the takes made. However a weakness of this result is that it was found from the data of only one musical composition. The composition of sound components for the oboe and the violin was found to vary more. These were also found from more varied musical material. The most varied musical contents was found with the cello, as outerpoints of playing methods were recorded. This is reflected in the results by high deviance.

From the investigations on the Foreground-Background Balance it was demonstrated that all the instruments were likely to vary between a positive and a negative self dominance at the musician's ear. However for the whole statistics it was found that it would mostly be at the negative side. It was found that the cello and the oboe are less loud instruments themselves, that the violin is moderatly loud in comparison and that the trumpet is the loudest and most likely to dominate in the soundscape out of the measured instruments. This as the median values of the FBB were found at -1.3dB for the trumpet, -2.7dB for the violin, -6.5dB of the oboe and -7.1dB for the cello.

From a distribution test was found that the FBB results of the cello, and especially the violin, may be of normal distribution, while this was rejected for the results of the oboe and the trumpet.

The observed tendencies reveal that the musical content is of high importance for the balance of the sound components, and therefore also who or what is responsable for the noise exposure to the musician's ears in the orchestra situation. Further work should aim to be made out from more varied musical contents as this investigation sometimes fell a little short in that aspect.

The investigation leads to a conclusion that the reverberant sound at the ears seems fairly similar for different instruments, but that it can possibly vary with position within the orchestra. The dominance of the direct sound from the musician's own instrument is most dependant to that of the direct sound from other musicians. The balance between these components were demonstrated to shift back and forth.

Appendix A

Pictures from measurement sessions

Figure A.1: Orchestra illustration in measurement session 1; with violin and oboe.

Figure A.2: Orchestra illustration in measurement session 2; with oboe and trumpet.

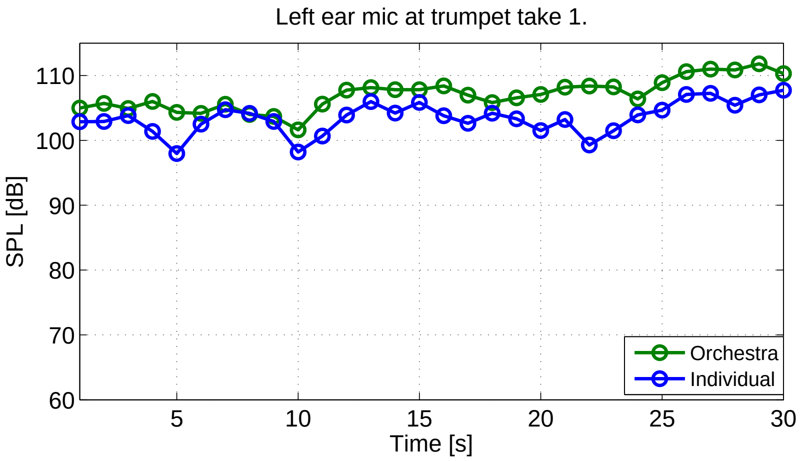
Figure A.3: Orchestra illustration in measurement session 3; with violin and cello.

Appendix B

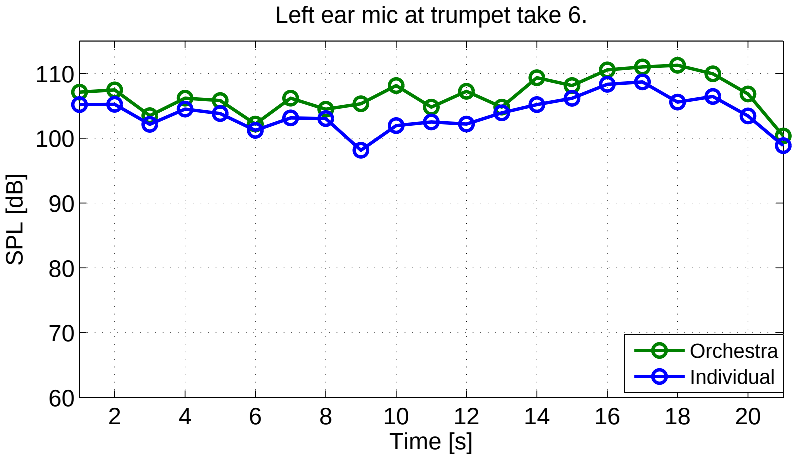
Additional figures

B.1 Content of recorded material presented as $L_{A,short}$

This section presents some examples on the short-term sectioned data material. These were background data for those examples which were presented in the report.



(a) Trumpet take 1.



(b) Trumpet take 6.

Figure B.1: $L_{A,1s}$ found at trumpet. The illustrations are of the left ear microphone signal.

B.2 Boxplot, all takes included

The figures in this section show the FBB data found for all the takes of the instruments. However, the statistics per take should not be emphasized too much as some of them were made from few observations. This goes especially for some of the shorter takes of the oboe and the trumpet.

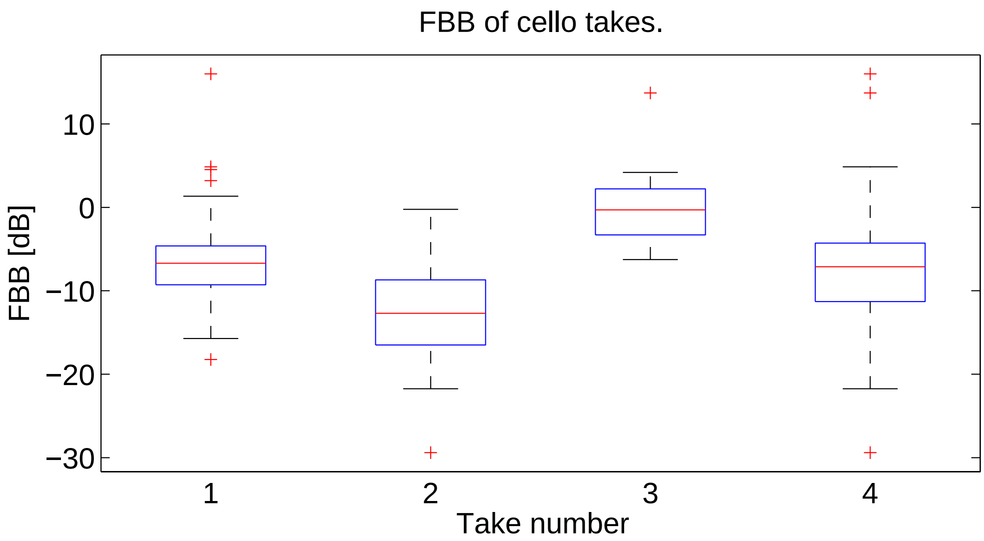


Figure B.2: FBB results of all the cello takes. Number 4 is an added statistic made out of all the observations combined.

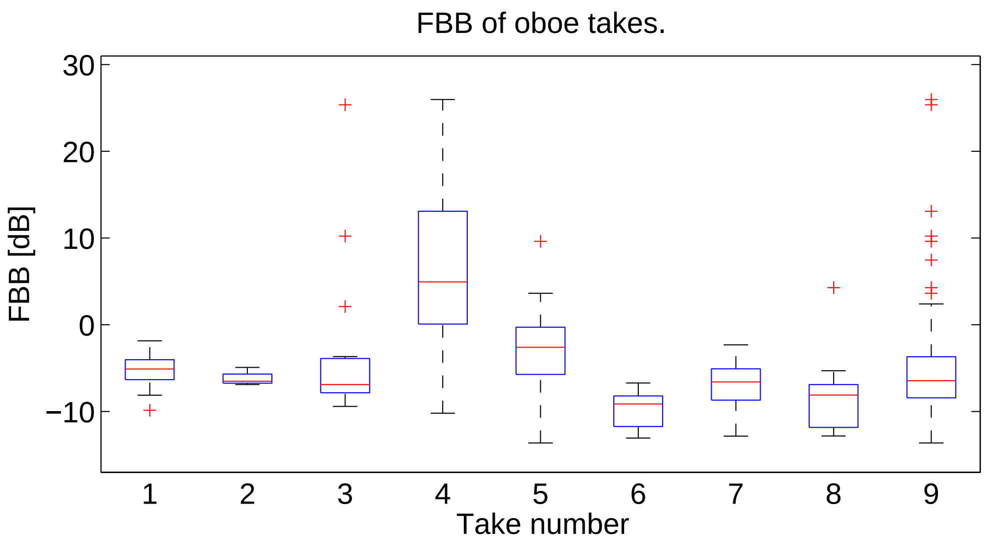


Figure B.3: FBB results of all the oboe takes. Number 9 is an added statistic made out of all the observations combined.

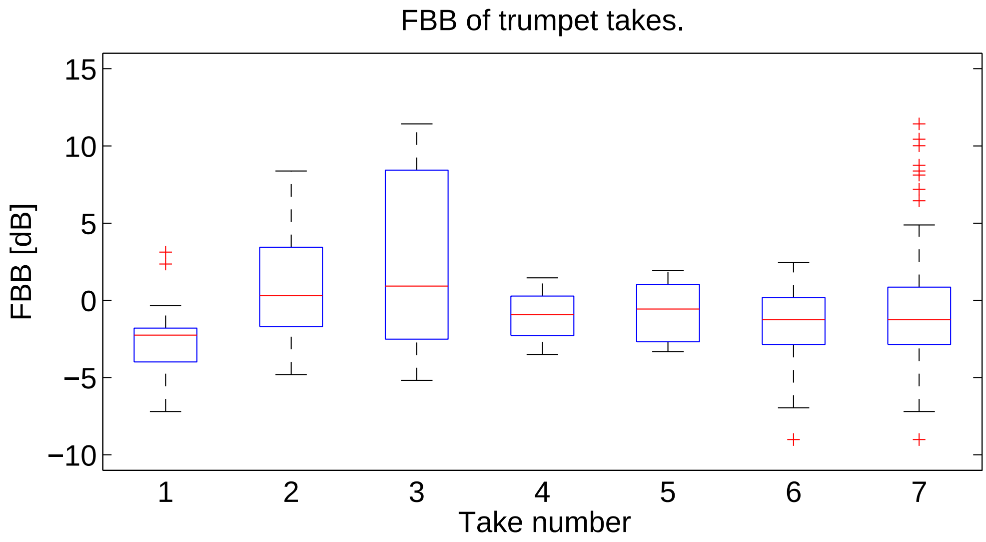


Figure B.4: FBB results of all the trumpet takes. Number 7 is an added statistic made out of all the observations combined.

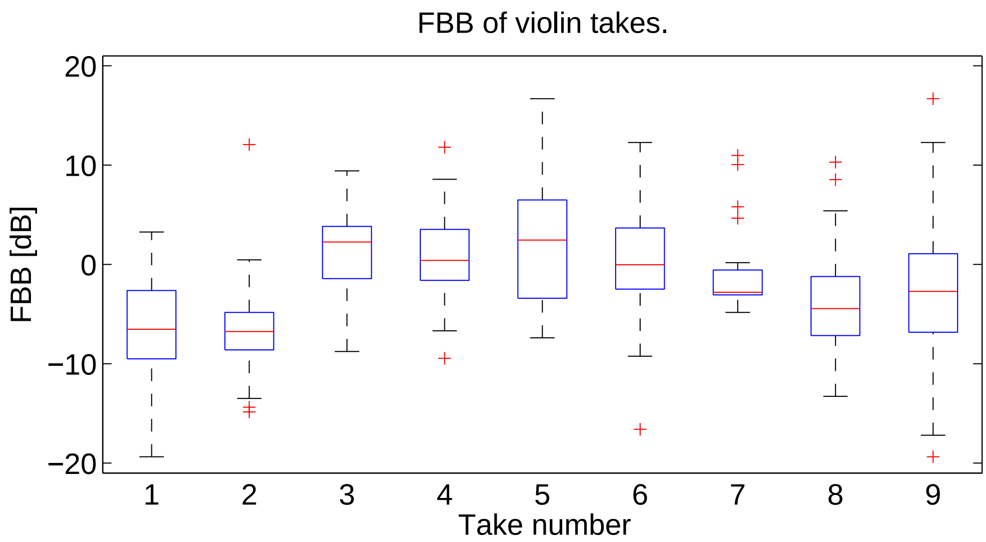
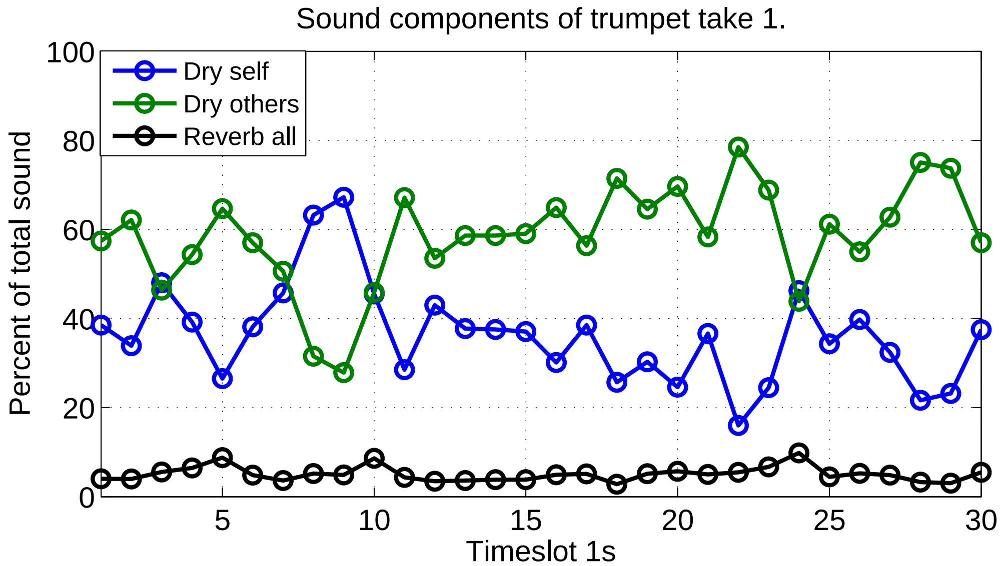
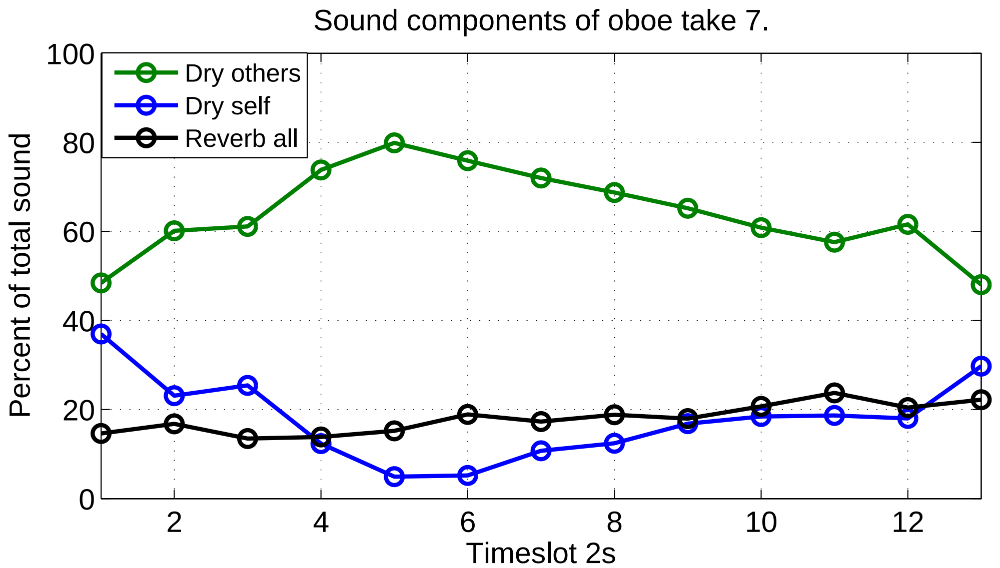


Figure B.5: FBB results of all the violin takes. Number 9 is an added statistic made out of all the observations combined.

B.3 Sound components



(a) Sound components found in trumpet take 1, data in percent.



(b) Oboe take 7.

Figure B.6: Composition of the sound components found for short-term sectioning of trumpet takes.

Appendix C

Tables

The background data for plots and other relevant data can be found in this appendix.

Table C.1: Sound components with distribution in percent.

	Dry Self %	Dry Others %	Reverb All %
Cello			
Take1	16	30	55
Take2	4	32	64
Take3	80	4	16
Mean	33	22	45
Std	41	15	26
Oboe			
Take1	24	67	9
Take2	20	75	5
Take3	23	64	13
Take4	90	6	4
Take5	48	43	9
Take6	11	72	17
Take7	15	68	17
Take8	12	72	16
Mean	30	58	11
Std	27	23	5
Trumpet			
Take1	33	62	5
Take2	45	50	5
Take3	55	38	7
Take4	43	51	6
Take5	46	50	4
Take6	40	54	6
Mean	44	51	5
Std	7	8	1
Violin			
Take1	21	76	3
Take2	17	81	2
Take3	83	10	8
Take4	61	34	6
Take5	62	28	10
Take6	71	14	15
Take7	53	45	2
Take8	38	43	19
Mean	51	41	8
Std	23	26	6

Table C.2: Duration of used takes in seconds.

	Individual [s]	Orchestra [s]	Session no.	Composer
Cello				
Take1	148.50	146.50	3	Brahms Hungarian dance
Take2	63.50	62.50	3	Tchaikovsky
Take3	47.00	51.20	3	Fantorangen
Total	259.00	260.20		
Oboe				
Take1	42.20	52.44	1	Brahms
Take2	8.80	9.90	1	Brahms
Take3	35.80	39.10	1	Brahms
Take4	18.50	19.70	1	Brahms
Take5	46.40	49.20	1	Brahms
Take6	31.82	33.30	2	Ravel
Take7	24.60	26.30	2	Ravel
Take8	22.10	23.41	2	Ravel
Total	230.22	253.35		
Trumpet				
Take1	30.80	30.00	2	Ravel
Take2	29.70	26.30	2	Ravel
Take3	26.50	26.60	2	Ravel
Take4	13.82	12.20	2	Ravel
Take5	13.80	12.43	2	Ravel
Take6	23.20	21.46	2	Ravel
Total	137.82	128.99		
Violin				
Take1	120.03	140.40	1	Brahms
Take2	79.45	79.00	1	Brahms
Take3	59.70	66.80	1	Beethoven
Take4	141.80	145.90	3	Brahms Hungarian dance
Take5	23.40	25.40	3	Rossini
Take6	76.90	83.50	3	Rossini
Take7	49.60	51.15	3	Bieber Battalia
Take8	91.80	104.00	3	Fantorangen
Total	642.68	696.15		

Table C.3: Measurement and calculations for SPL of the cello. All values in dB.

	Situation	Measured L	Measured R	Computed Reverb	Computed Dry Self	Computed Dry Others	FBB
Cello							
Take 1	Ind	81	82	64			
	Orch	88	90	87	81	82	-7
Take 2	Ind	65	66	52			
	Orch	79	80	78	65	73	-14
Take 3	Ind	77	79	62			
	Orch	79	79	72	78	64	6
Mean					74.8	73.2	-5.3
Std					8.4	9.2	10.1

Table C.4: Measurement and calculations of the oboe. All values in dB.

	Situation	Measured L	Measured R	Computed Reverb	Computed Dry Self	Computed Dry Others	FBB
Oboe							
Take 1	Ind	91	88	75			
	Orch	96	96	85	90	94	-5
Take 2	Ind	81	77	64			
	Orch	88	84	73	80	85	-6
Take 3	Ind	90	88	75			
	Orch	96	94	86	89	93	-5
Take 4	Ind	89	86	72			
	Orch	90	86	75	88	74	10
Take 5	Ind	88	88	73			
	Orch	92	90	81	88	87	0
Take 6	Ind	89	94	77			
	Orch	100	102	94	92	100	-9
Take 7	Ind	91	92	74			
	Orch	97	101	92	92	98	-7
Take 8	Ind	91	93	75			
	Orch	100	102	93	92	100	-9
Mean					88.6	91.5	-3.9
Std					4.0	8.8	6.4

Table C.5: Measurement and calculations of the trumpet. All values in dB.

	Situation	Measured L	R	Computed Reverb	Dry Self	Dry Others	FBB
Trumpet							
Take 1	Ind	104	102	84			
	Orch	108	108	94	103	106	-3
Take 2	Ind	103	101	84			
	Orch	105	106	92	102	102	-1
Take 3	Ind	103	101	83			
	Orch	105	104	93	102	100	1
Take 4	Ind	102	101	84			
	Orch	106	105	92	102	102	-1
Take 5	Ind	104	102	83			
	Orch	106	107	93	103	103	-1
Take 6	Ind	104	103	86			
	Orch	107	108	95	104	105	-2
Mean					102.6	103.2	-1.0
Std					0.9	2.1	1.3

Table C.6: Measurement and calculations of the trumpet. All values in dB.

	Situation	Measured L	R	Computed Reverb	Dry Self	Dry Others	FBB
Violin							
Take 1	Ind	94	87	66			
	Orch	101	95	85	92	98	-6
Take 2	Ind	98	91	70			
	Orch	106	99	87	96	103	-7
Take 3	Ind	99	91	70			
	Orch	99	95	87	96	86	7
Take 4	Ind	99	92	70			
	Orch	101	95	87	97	94	2
Take 5	Ind	92	87	64			
	Orch	95	89	84	91	87	2
Take 6	Ind	92	86	63			
	Orch	93	88	84	90	82	4
Take 7	Ind	95	88	67			
	Orch	97	91	79	92	92	1
Take 8	Ind	92	85	63			
	Orch	96	91	88	90	90	-2
Mean					93.0	91.3	0.1
Std					3.0	6.8	4.7

Appendix D

Measurements on equipment

D.1 Calibration measurements (results)

FigureD.1 show the magnitudes of the microphones relative to that of the microphone chosen as reference, which was the B&K-microphone 2. The calibration constants were chosen to be calculated in the frequency area 100 to 1000Hz as this was flat and highly relevant for the frequency content of the music to be recorded. An important consideration of the calibration process was that the microphone pairs were found to be fairly flat compared with each other, that is to say that the relative difference between the measured sound pressure level was independent of frequency.

FigureD.2 shows the fequency response adjusted for microphone (and channel) sensitivity for all microphones. The levels measured and the deviation from the reference microphone can be seen in Table D.1. The highest deviation of 0.12dBA is acceptable in accordance with the precision of the data given.

Table D.1: Sound pressure levels found with all calibrated microphones, equal sound source level.

Channel number	CH1 MKE L	CH2 MKE R	CH3 MZA L	CH4 MZA R	CH5 B&K 1	CH6 B&K 2
SPL [dB]	92.52	92.50	92.54	92.52	92.40	92.42
Deviation from reference [dB]	0.10	0.09	0.12	0.10	-0.01	0.00

D.2 Sound card channel test

The influence on the results were looked at and the results of the test is shown in Table D.2. The test was measured with the B&K-microphone used as reference microphone for the calibration results. The method used was a sweep signal on the Genelec loudspeaker, the microphone was placed in the grid of this in front of the loudspeaker element, and kept in the same position during all measurements.

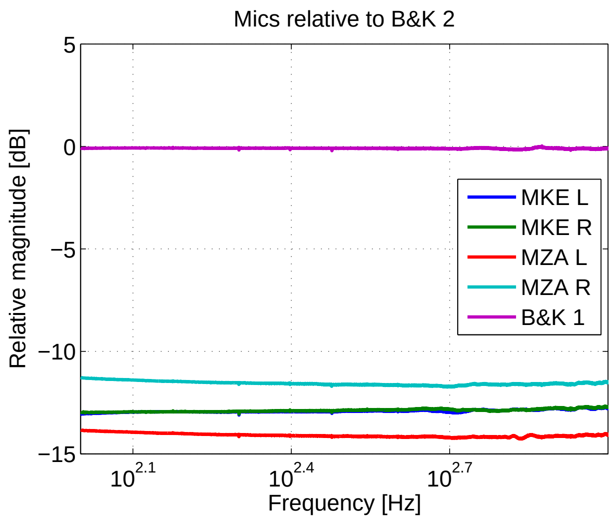


Figure D.1: Magnitude of microphones relative to reference microphone 2 (B&K 2).

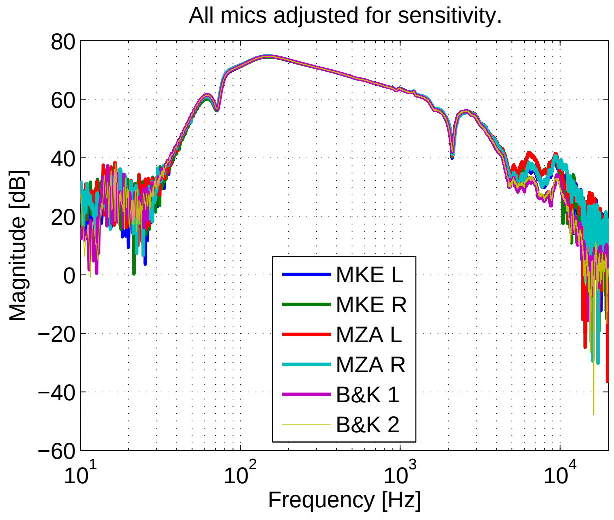


Figure D.2: Frequency response of calibrated signal, all microphones.

Table D.2: Measured electric input on sound card.

Channel number	CH1	CH2	CH3	CH4
Microphone	MKE L	MKE R	MZA L	MZA R
Measured level [dB]	92.84	92.90	97.92	98.13
L-R difference [dB]	-0.06		-0.020	

After measuring one channel, the cable was plugged to the next channel, starting from channel 1 to channel 4. The measurements then underwent the same analalysis in Matlab as the calibration measurements, by choosing the relevant sample space of equal lengths for all the measurement, and performing a Discrete Time Fourier Transform. The LR-difference then presents the difference of the electrical input for the channels for the respective microphone pair. The gain settings on the sound card was in this test the same as the most frequently used gain setting during the measurements, respectively 30 for channels 1 and 2, and 35 dB for channels 3 and 4. Hence the dB-level for channel 3 and 4 is expected to be +5dB compared to the result if channel 1 and 2.

D.3 Energy relation as test for short-term sectioning

Energy relation between equivalent takes and summed short-term pieces.

Table D.3: Energy relation test with example trumpet take 6.

	Individual take	Orchestra take
Whole piece	12373479.54	22825565.41
Sum oft he pieces	12373479.54	22825565.41

Like Table D.3 shows, there is no particular error which occur in the data of the short-term sectioned takes. The error only starts at the seventh decimal and is therefore considered small enough to be acceptable.

Appendix E

MATLAB-code

E.1 Script order

The Matlab-scripts used in this thesis can be given upon request to *helena.rydland@gmail.com*.

E.2 Short-term sectioning

```
function [Lpcurve, rms2curve, tvec, sigSeg] = timeLP(sig, fs, finenessFactor)
% This function will output a 1D-matrix with Lp-levels as they shift
% through the duration of a signal.
% Input 'Sig' is the signal of which the time shifting Lp-level is wanted.
% FinenessFactor is the resolution wanted, e.g. one value per second is
% found with input '1', 2 values per second is found with an input of '2',
% etc.

L = length(sig);
tmax = length(sig)/fs/finenessFactor;   % Finds the last element needed in the ↩
    output
Lpcurve = zeros(1,ceil(tmax));
rms2curve = zeros(1,ceil(tmax));
tvec = zeros(1,ceil(tmax));
sigSeg = zeros(1,ceil(tmax));

for n = 1:ceil(tmax)
    istart = round((n-1)*fs*finenessFactor+1);
    iend = round(n*fs*finenessFactor);
    if iend <= L
        valSignalpiece = rms(sig(istart:iend)); %dBA value for the fraction
        tvec(n) = length(istart:iend); % number of samples of the segment
        sigSeg(n) = sum(sig(istart:iend).^2);%energy summation of segment
    else %The same as above if at the last interval
        valSignalpiece = rms(sig(istart:L));
        tvec(n) = length(istart:L);
        sigSeg(n) = sum(sig(istart:L).^2);
    end
        Lpcurve(n) =  20*log10(valSignalpiece/2e-5);
        rms2curve(n) = valSignalpiece.^2;

end
end
```

Bibliography

[1] M. Skålevik, "Consistency in music room acoustics," *AKUTEK, Forum Acusticum 2014*, 2014.

[2] B. A. Ian O'Brien, Tim Driscoll, "Sound exposure of professional orchestral musicians during solidary practice," *Journal of the Acousical Society of America*, October 2013.

[3] E. J. M. Jansen, H. W. Helleman, W. A. Dreschler, and J. A. P. M. de Laat, "Noise induced hearing loss and other hearing complaints among musicians of symphony orchestras," 2009.

[4] D. R. Raichel, *The science and applications of acoustics, second edition.* Springer Science+Business Media, Inc., 2006.

[5] M. Walpole, Myers and Ye, *Probability & statistics for engineers & scientists, eighth edition.* Pearson Education International, 2007.

[6] "Description of kolmogorov-smirnov test, mathworks page," 26th of June 2015.

[7] M. Barron and L.-J. Lee, "Energy relations in concert auditoriums. i," *Journal of the Acoustical Society of America*, 1988.

[8] H. Kuttruff, *Room acoustics, fifth edition.* Spon Press, 2009.

[9] Rossing and M. . Wheeler, *The Science of Sound, third edition.* Addison Wesley, 2002.

[10] L. E. Kinsler, A. R. Frey, A. B. Coppens, and J. V. Sanders, *Fundamentals of Acoustics, fourth edition,.* John Wiley & Sons,Inc., 2000.

[11] R. Peters, B. Smith, and M. Hollins, *Acoustics and Noise Control, third edition.* Pearson Education Limited, 2011.

[12] M. Skålevik, "Directivity of musical instruments," *AKUTEK*, 2012.

[13] J. Meyer, *Acoustics and the Performance of Music, fifth edition.* Springer, 2009.

[14] G. Müller and M. Möser, *Handbook Of Engineering Acoustics.* Springer-Verlag, 2013.

[15] M. Barron, *Auditorium acoustics and architectural design.* Spoon press, 1993.

[16] M. Skålevik, "Rehearsal room acoustics for the orchestra musician," *AKUTEK, BNAM 2014*, 2014.

[17] G. F. Kuhn, "The pressure transformation from a diffuse sound field to the external ear and to the body and head surface," *The journal of acoustics society of America*, April 1979.

[18] T. C. ISO, "Measurement of room acoustic parameters - part 1: Performance spaces," Tech. Rep. ISO 3382-1:2009(E), Standard Norge, Oktober 2009.

[19] J. S. Bradley, "Using iso 3382 measures, and their extensions, to evaluate acoustical conditions in concert halls," *Acoustical Science and Technology*, pp. 170–178, 2005.

[20] L. Beranek, *Concert Halls and Opera Houses - Music, Acoustics ans Architecture, 2nd edition.* Springer-Verlag New York,Inc., 2004.

[21] "Documentation on adobe audition cs6," May 2015.

[22] "Description of boxplot, mathworks page," 3rd of July 2015.

[23] "Law of work environment, 'arbeidsmiljøloven, forskrift om vern mot støy på arbeidsplassen'," May 2006.