The Acoustics of Stockholm Concert Hall and Artificial Reverberation Systems

EVALUATION OF STORA SALEN AND SIMULATION OF ITS ELECTRONIC REVERBERATION SYSTEM

CHRISTOFFER CARLSSON
The Acoustics of Stockholm Concert Hall and Artificial Reverberation Systems

Evaluation of Stora salen and simulation of its electronic reverberation system

Akustiken i Stockholms konserthus och artificiella efterklangssystem
Utvärdering av Stora salen och simulering av dess elektroniska efterklangssystem

CHRISTOFFER CARLSSON
chrc@kth.se

Master Thesis in Speech and Music Communication
Master of Science in Media Technology, Master in Media Technology
Provider: ACAD-International AB
Supervisor at CSC: Anders Friberg
Supervisor at ACAD: Joel Johansson
Examiner: Sten Ternström

Master of Science Thesis
Stockholm, Sweden, December 2015
Abstract

This master thesis examines the effects on the acoustical properties of a concert hall caused by an artificial reverberation system (ARS) and the possibility of simulating these properties. By examining the case of the Stockholm Concert Hall, which recently installed such a system, a greater understanding of the ARS will be gained and additional improvements of simulating such systems will be explored. This study comprises two parts: (1) objective data obtained through acoustical measurements are evaluated both internally and to other halls and (2) by computer simulation of the concert hall and its electronic reverberation system evaluate the acoustics of the hall.

The study shows that the effect of the ARS on the acoustical properties of Stockholm Concert Hall is not excessive but noticeable. An 0.3 second increase in reverberation time is a desirable outcome but comes at the cost of clarity, which sees a reduction of 0.7 decibels. Moreover, it is possible to simulate a concert hall, having an ARS installed, with fairly realistic results. However, in order to compile the simulated impulse response, a script had to be created –combining the transfer functions related to each component of the reverberation chain from source to receiver, including all the microphones and loudspeakers of the ARS.

Keywords: artificial reverberation system (ARS), acoustical measurements, concert hall acoustics, acoustical simulation
Referat

Akustiken i Stockholms konserthus och artificiella efterklangssystem
Utvärdering av Stora salen och simulering av dess elektroniska efterklangssystem

Det här examensarbetet undersöker påverkan på de akustiska egenskaperna hos en konserthall orsakad av ett artificiellt efterklangssystem. Likaså undersöks möjligheterna för att simulera dessa akustiska egenskaper. Genom att undersöka Stockholms konserthall, som nyligen installerade ett efterklangssystem, kommer en bättre förståelse för artificiella efterklangssystem skapas och ytterligare förbättringar för simulering kommer att möjliggöras. Den här studien genomförs i två delar: (1) objektiv data, inhämtad från akustiska mätningar, utvärderas både internt och mot andra konserthaller samt (2) genom datorsimulering av konserthallen och det elektroniska efterklangssystemet utvärderas de akustiska egenskaperna.

Studien visar att inverkan på de akustiska egenskaperna hos Stockholms konserthall orsakade av det artificiella efterklangssystemet inte är överdrivna men noterbara. En önskad ökning av efterklangstiden med 0.3 sekunder uppnås men detta på bekostnad av att ljudets klarhet minskar med 0.7 decibel. Vidare är det möjligt att simulera ljudutbredningen i en konserthall som har ett efterklangssystem installerat med ett tämligen realistiskt resultat. För att uppnå detta simuleringsresultat skapas ett skript vilket väger samman alla överföringsfunktioner mellan ljudkällan och mottagaren, inklusive de mellan efterklangssystemets mikrofoner och högtalare.
Acknowledgments

This master thesis is carried out at the acoustic consultant company ACAD-International AB (ACAD); one of the biggest independent consulting firms in its area, operating mostly in the Stockholm region. The company has for example been the acoustic designers of the Tele2-arena and has for long worked with the acoustics of the Stockholm Concert Hall.

I would like to start by thanking my supervisor at ACAD, Joel Johansson, for sharing your passion about concert hall acoustics with me and for the input of analyzing suggestions. Also, a thanks to Lennart Karlén at ACAD, for the opportunity to work with Stockholm Concert Hall and the inspiring conversations and life-stories concerning acoustics in some way.

Thank you Anders Friberg and Sten Ternström, my supervisor and my examiner at KTH, for the help of the thesis and the liberty to explore the room acoustics area further.

And finally, thanks to all of my friends and family who have helped and supported me in the process of creating this thesis.
Contents

I Introduction 1

1 Introduction 3
  1.1 Background .............................................................. 3
  1.2 Aim ................................................................. 4
  1.3 Objective ............................................................ 4
  1.4 Limitations and delimitations .......................................... 5
  1.5 Choice of methodology ................................................ 5
  1.6 Structure of the thesis ................................................ 5
      1.6.1 Intended reader ................................................ 5
      1.6.2 Outline .......................................................... 5

2 Theory 7
  2.1 Acoustical parameters of a concert hall ............................ 7
      2.1.1 Reverberation ................................................ 7
      2.1.2 Level .......................................................... 9
      2.1.3 Energy ratios .................................................. 9
      2.1.4 Spaciousness ................................................... 11
      2.1.5 Stage parameters ............................................. 12
  2.2 Related work regarding acoustical parameters .................... 12
      2.2.1 Concert Halls and Opera Houses ............................... 12
      2.2.2 Matching subjective perceptions and objective parameters ... 12
      2.2.3 Estimation of parameters ..................................... 13
  2.3 Related work regarding reverberation systems .................... 14
      2.3.1 Reverberation systems ....................................... 14
      2.3.2 Artificial reverberation systems, ARS ........................ 15
      2.3.3 Sustainability .................................................. 17
      2.3.4 The Stockholm Concert Hall .................................. 17

II Ranking the Stockholm Concert Hall 19

3 Theory of acoustical measurements in concert halls 21
  3.1 Transfer functions .................................................. 21
## List of Figures

2.1 The reverberation time derived from the sound level decay of the interrupted sound. ........................................... 8  
2.2 The sound energy field is affected by the MCR system. ......................... 16  
2.3 Schematics of an artificial reverberation system. .......................... 17  
3.1 Schematic illustration of a system. ........................................... 21  
3.2 Signals viewed in frequency domain. ........................................... 22  
3.3 The obtained impulse response viewed in time- and frequency domain. 23  
3.4 Two sinusoidal signals with the same frequency but different amplitude. 24  
3.5 Correlation charts between signals. ........................................... 24  
4.1 The generalized setup used in the acoustical measurements. ............... 26  
5.1 Energy magnitude spectrum of the calibration signal. .......................... 29  
5.2 The sound energy decay curve measured with different equipment setups. 30  
5.3 Correlation charts between the obtained impulse response functions. .......... 31  
5.4 Energy decay curve of obtained signal with the ARS inactive and active. .... 32  
5.5 The reverberation time of Stockholm Concert Hall compared to other halls. 34  
5.6 The early decay time of Stockholm Concert Hall compared to other halls. .... 35  
5.7 The strength factor in relation to the $EDT/V$ ratio of Stockholm Concert Hall  
compared to other halls. ............................................................. 36  
7.1 Formation of image sources. .................................................... 45  
7.2 The Ray trace method. ......................................................... 45  
8.1 The resulting reflection direction calculated by two reflection vectors. ....... 48  
9.1 The results of the simulated reverberation time. ............................. 52  
9.2 The results of the simulated early decay time .................................. 53  
9.3 The results of the simulated clarity parameter .................................. 54
List of Tables

2.1 Values of echo criteria derived by Dietsch and Kraak (Lø vstad 2003). . . . . . . 10
2.2 A listener’s subjective perceptions can be linked to objective measurements. . 13
2.3 Coefficients for calculating the reverberation time of an occupied hall when
    the unoccupied reverberation time is measured. . . . . . . . . . . . . . . . . . 14
5.1 Results of the ARS effect. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 33
Part I

Introduction
Chapter 1

Introduction

As the field of acoustics evolves, so does the music consumer awareness of the acoustics in various venues. With the help of artificial reverberation systems, the reverberation time of the venue—identified as an important parameter for the subjective quality of the sound—can be modified to fit the performance. This master thesis investigates the acoustical properties of the Grand Hall (in Swedish: Stora salen) of the Stockholm Concert Hall, in which recently an artificial reverberation system was installed. More specifically, the aim of the thesis is to examine the acoustical effects caused by such a system. This is accomplished through acoustical measurements and simulations.

Artificial reverberation systems have been developed since the middle of the last century and are now becoming a popular tool for acousticians. It is important for the acousticians to be in line with and even ahead of the development of new techniques for improving the sound quality of event venues. As cities expand, so does the need for event venues like concert halls. On the other hand, urban space limitation is forcing the constructors and designers to think outside the box; making the artificial reverberation system a great tool for meeting the acoustical demands.

1.1 Background

The event venues of today are often built for several types of events: from conferences to theater, sports events, pop music concerts and last but not least, classical music concerts. In these multi-purpose halls, there are high demands on well-adjustable acoustical properties. The properties are defined by acoustical parameters, and by tuning them, one can achieve a configuration that fits the particular event. In classical music concerts, for example, the preferred reverberation time is approximately 1.7 seconds, whereas organ music concerts require up to twice as long a reverberation time (Hoover and Ellison 2013). This way of designing event venues has not been present for long: several venues built during the last century are used for one type of event only. Such venues often lack the physical possibility to change the interior and thereby change its acoustical parameters, especially enhancing the reverberation time. One convenient solution to this problem is to install an artificial reverberation system, as done in the Stockholm Concert Hall (Barron 2009).
CHAPTER 1. INTRODUCTION

Even though the installation in Stockholm was done mainly in order to enhance carefully the acoustics of the classical music concerts, the use of artificial reverberation has great potential for changing the acoustical properties of a particular hall to match a variety of performance demands. The acoustic of the Stockholm Concert Hall is now proposed by senior acousticians at ACAD as being of world class, whereas the opinion of others are more contradictory (Karlén 2015). Therefore, an evaluation of the acoustical properties may be of use in order to rank the hall.

Throughout the years of construction of concert halls the knowledge of how the sound will be distributed over the audience has been gained from physical scaled models of the intended hall. The evolution of technology has now made it possible to create computer models of the sound propagation in a room. This has facilitated the design of the desired acoustical environment for acousticians. However, there is still a lack of knowledge of how the acoustical properties are affected by the introduction of an artificial reverberation system.

1.2 Aim

The aim of this thesis is to investigate how the objective acoustic parameters are affected by an artificial reverberation system. These parameters are compared with those of other concert halls—giving the author a possibility to rate the acoustics of the Stockholm Concert Hall. Also, a computer model of the same venue is created in order to investigate how such reverberation systems can be modeled and simulated. The results are evaluated by comparing the simulated acoustical parameters with the measured ones, in order to make a judgment of the realism of the simulation.

The stated questions will also generate additional reflections on, for example, measurement equipment, calibration aspects and software limitations etc. These answers will help ACAD to evolve their understanding of concert hall acoustics and artificial reverberation systems.

This master thesis may be considered to be successful if the results from the measurements and the simulations, even if they are not equal, can be explained and put into a reasonable context.

1.3 Objective

The present thesis addresses the following questions:

*How are the acoustic properties of Stockholm Concert Hall affected by the artificial reverberation system?*

*What are the possibilities of simulating the acoustics of the hall using computer software as used by professional acousticians?*
1.4 Limitations and Delimitations

Due to limitations in financial- and human resources, and accessibility to the hall, no additional measuring gear other than that present at ACAD is used. Furthermore, only two additional persons are available when conducting the measurements, making it difficult to change the state of the artificial reverberation system between every measuring position. Moreover, this thesis has been delimited in the following aspects:

- The measurements include only the airborne sound. No structure-born sound transmission is investigated.
- The evaluation of the objective data is not mainly based on statistical models. However, some instances of standard deviation and correlation are performed.
- Only the middle frequency octave bands, 500 Hz and 1000 Hz, are analyzed.
- No listening tests are performed.

The limitations and delimitations can be revised in future work in order to give more precise results.

1.5 Choice of Methodology

Observations of acoustical and geometrical properties of the hall must be done in order to rank the Stockholm Concert Hall by objective data. This is achieved by conducting acoustical measurements according to international standard procedures and record geometric quantities of the hall. Furthermore, the computer model will be used as an estimation of the physical hall. The environment of the hall’s interior can be controlled in the simulation software, in order to hear and visualize the acoustical properties of the hall. The input data of the model is retrieved from measurements, and the simulation result will objectively be compared with the measured parameters.

1.6 Structure of the Thesis

1.6.1 Intended Reader

The reader of this thesis should have some background knowledge of the physics of sound, e.g. the properties of a signal are frequency dependent and the frequency content of the signal can be analyzed if applying a Fourier transform to the signal and frequency segmentation into octave bands. This might help in understanding, as some things have been taken for granted by the author.

1.6.2 Outline

The work addresses two major objectives; evaluating the acoustical properties of the Stockholm Concert Hall and examining the possibility of simulating artificial reverberation
systems. The thesis is accordingly divided as follows. In Part I, the background information of this thesis is explained followed by a description of relevant theory used to address each of the objectives. Thereafter, in Part II, the focus is on the evaluation of the acoustics of Stockholm Concert Hall and on presenting some additional theories concerning acoustical measurements in concert halls. Later, the results of the measurements are presented, leading to a rank of the hall. The part ends with discussions about the first objective and possible improvements when performing acoustical measurements. Part III presents the second objective, that is, the simulation of the artificial reverberation system as well as complications regarding the modeling of a concert hall and acoustics simulations. Part IV proposes future work related to the findings. Finally, the conclusions of this master thesis are put forward.
Chapter 2

Theory

2.1 Acoustical parameters of a concert hall

The acoustical parameters of interest for this thesis, presented in the following sections, are briefly summarized from Beranek (2004), Barron (2005; 2009), Randal (1987), Brüel&Kjær (2015b) and the International Organization for Standardization (2009). First, the group of reverberation parameters will be explained followed by two sound level parameters. Secondly, energy level parameters of a sound signal is presented along with a few parameters regarding the perceived spaciousness. Finally, a stage parameter is introduced—significant mainly for the music performers. Although not all of the parameters presented here apply to the measurement part, the following sections provide comprehensive explanations of the parameters mentioned later in the thesis.

2.1.1 Reverberation

Reverberation time

The reverberation time is defined as the time in seconds it takes for the emitted sound level in a room to decrease by 60 dB from a sudden interruption, which is illustrated in Figure 2.1. There are many versions of the reverberation time parameter; $RT$, $T_{10}$, $T_{20}$, $T_{30}$, where $RT$ corresponds to the previous definition and the $T$-parameter’s subscript number denotes the decrease in decibel from $-5$ dB of its interrupted sound level, i.e. $-5$ dB to $-15$ dB, $-5$ dB to $-25$ dB and $-5$ dB to $-35$ dB respectively. The value of the latter parameters are extrapolated from the slope of the impulse response when viewed as the sound level decay in time, expressed by (2.1) as

$$T_{\Delta y} = \frac{60}{\Delta y} \Delta x \ [s].$$

(2.1)

where $x$ is the time it takes for the decay to reach the sound level $y$ below the interrupted level. The $T$-parameter is multiplied with a factor of $60/\Delta y$ in order to make all the reverberation measures comparable to $RT$. 

7
Chapter 2. Theory

Figure 2.1: The reverberation time derived from the sound level decay of the interrupted sound. Retrieved from (Barron 2009; p. 29).

One should remember that the reverberation time is frequency dependent. Therefore, no direct comparison of the $RT$ value can be done unless it is clear which frequency that has been measured. Often the 500 Hz and 1000 Hz octave band average constitutes the reverberation time for the whole room, denoted as $RT_{mid}$.

Early Decay Time

The early decay time, $EDT$ expresses the decay time from 0 dB to $-10$ dB attenuation of the interrupted sound level. In order to make reverberation time parameters comparable to $RT$, the $\Delta x$ is multiplied with a factor of six.

$$EDT = \frac{60}{\Delta y_{[0,-10]}} \Delta x \ [s].$$ (2.2)
2.1. ACOUSTICAL PARAMETERS OF A CONCERT HALL

Bass ratio

The bass ratio, \( BR \), is a ratio of the average reverberation time in low octave frequency bands, to those of middle octave frequency bands. The low octave frequency bands have the center frequency 125 Hz and 250 Hz denoted in subscript, i.e. \( RT_{125} \) and \( RT_{250} \) respectively. The middle frequency octave bands have the center frequency 500 Hz and 1000 Hz, correspondingly \( RT_{500} \) and \( RT_{1000} \). Unlike many other parameters, this is measured when the hall is fully occupied.

\[
BR = \frac{RT_{125} + RT_{250}}{RT_{500} + RT_{1000}} \quad [-].
\]  

(2.3)

2.1.2 Level

Sound strength

The sound strength, \( G \), is expressed in decibels. The logarithmic ratio is composed of the integration of the measured sound power at an audience seat in the concert hall, \( p^2 \), and the integration of the sound power emitted from the same source during calibration, \( p^2_A \).

\[
G = 10 \log \left( \frac{\int_0^\infty p^2(t) \, dt}{\int_0^\infty p^2_A(t) \, dt} \right) \quad [dB].
\]  

(2.4)

Signal to noise

The signal to noise parameter, \( SNR \), is the level difference between the power of the steady state background noise, \( p^2_{\text{noise}} \), and measured signal, \( p^2_{\text{signal}} \). It is expressed as

\[
SNR = 10 \log \left( \frac{p^2_{\text{signal}}}{p^2_{\text{noise}}} \right) \quad [dB].
\]  

(2.5)

2.1.3 Energy ratios

Center Time

The center time parameter, \( T_S \), is the ratio of early and late arriving sound energy, or the balance between clarity and reverberance. Low values of the parameter corresponds to a clear sound. The division in time of the arriving sound wave, is indicated by the integral limits in seconds. The sound pressure of the signal, which is proportional to the square root of the energy, is denoted \( p \) and the time is denoted \( t \).

\[
T_S = \frac{\int_0^{0.08} tp^2(t) \, dt}{\int_0^\infty p^2(t) \, dt} \quad [ms].
\]  

(2.6)

The \( T_S \) parameter is almost identical to the clarity parameter, \( C_{50} \). However, \( T_S \) yields a more human-like model of the sound arriving at the ear.
CHAPTER 2. THEORY

Clarity

The clarity of a sound, $C_{80}$, is measured as the ratio between the sound energy arriving at the receiver in the first 80 milliseconds, to that which are arriving after the first 80 milliseconds. The ratio is expressed in decibels.

$$C_{80} = \frac{\int_0^{0.08} p^2(t) \, dt}{\int_{0.08}^{\infty} p^2(t) \, dt} \text{ [ms]}. \tag{2.7}$$

Definition

The sound definition, $D_{50}$, is a parameter for measuring how much sound energy that arrives at a position during the first 50 milliseconds after the direct sound, in relation to the total sound energy. The ratio is expressed as a percentage.

$$C_{50} = 100 \frac{\int_0^{0.05} p^2(t) \, dt}{\int_{0.05}^{\infty} p^2(t) \, dt} \text{ [%]}. \tag{2.8}$$

Echo Criterion

The echo criterion, $EC$, is a measurement of how much of the reflected sound that is perceived as an echo. It uses the $T_S$ but the time frame, $\tau$, and exponent, $n$, are varying depending on the type of sound, i.e. music or speech, as noted in Table 2.1 below. To indicate the different approach to the center time parameter, it is here denoted as $T_{S,\tau,n}$.

$$EC = \max \frac{\Delta T_{S,\tau,n}}{\Delta \tau} \text{ [ms]}, \quad \text{where}$$

$$T_{S,\tau,n} = \frac{\int_0^\tau t p^n(t) \, dt}{\int_0^{\infty} p^n(t) \, dt} \text{ [ms]}. \tag{2.10}$$

Table 2.1 should be read as in the following example: If the $EC$ value exceeds 1.8 milliseconds when listening to music, more than 50% of the audience will perceive the sound as an echo – having a negative impact on the total musical perception.

<table>
<thead>
<tr>
<th>TYPE OF SOUND</th>
<th>$n$</th>
<th>$\tau$ [ms]</th>
<th>$EC_{10%}$</th>
<th>$EC_{50%}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>Speech</td>
<td>2/3</td>
<td>9</td>
<td>0.9</td>
<td>1.0</td>
</tr>
<tr>
<td>Music</td>
<td>1</td>
<td>14</td>
<td>1.5</td>
<td>1.8</td>
</tr>
</tbody>
</table>
2.1. ACOUSTICAL PARAMETERS OF A CONCERT HALL

2.1.4 Spaciousness

Initial-Time-Delay Gap

The initial-time-delay gap, ITDG, is the time between the direct sound wave and the first reflected sound wave to arrive at the receiver. This parameter can be derived from a reflectogram by measuring the time gap between the direct sound and the first reflection.

Lateral Energy Fraction

The lateral energy fraction, LF, describes the amount of total sound energy that arrives from the stage and from the side walls at the audience. The ratio consists of the sound pressure level measured from a figure-of-eight microphone, \( p_8(t) \), and an omnidirectional microphone, \( p(t) \), within certain time frames, indicated by the integral limits in seconds.

\[
LF = \frac{\int_{0.08}^{0.005} p_8^2(t) \, dt}{\int_{0}^{0.08} p^2(t) \, dt} \quad [-].
\] (2.11)

Interaural Cross-Correlation Coefficient

The interaural cross-correlation coefficient, IACC, is a parameter strongly connected to the perceived width of the source. The measurement requires two receivers i.e. a binaural microphone –like a dummy head having one microphone in each ear. The single number parameter is obtained from the interaural cross-correlation function, IACF, as

\[
IACC_t = \max |IACF_t(\tau)| \quad \text{for} \quad -1 < \tau < +1 \quad [-],
\] (2.12)

\[
IACF_t(\tau) = \frac{\int_{t_1}^{t_2} p_L(t) p_R(t + \tau) \, dt}{\left(\int_{t_1}^{t_2} p_L^2(t) \, dt \cdot \int_{t_1}^{t_2} p_R^2(t) \, dt\right)^{1/2}} \quad [-].
\] (2.13)

Here \( t_1 \) and \( t_2 \) indicates the start and stop time of investigation, subscript \( L \) and \( R \) declare the left and right microphone respectively and \( \tau \) corresponds to the time it takes for the sound to travel around the head to the other ear.

Binaural Quality Index

The binaural quality index parameter, BQI, is an indicator of the perceived spaciousness of the hall. As seen in (2.14), it uses the previously mentioned parameter, IACC. The subscript \( E \) and \( 3 \) denotes the time frame of interest –here the early sound from time 0 to 80 milliseconds, and the average over three octave bands; 500 Hz, 1000 Hz and 2000 Hz.

\[
BQI = (1 - IACC_{E3}) \quad [-].
\] (2.14)
2.1.5 Stage parameters

Support

The stage support parameter, $ST$, is expressing the degree of sound energy coming from the sides of the stage and that from the rest of the hall. The support is measured as the difference between the sound energy, $p$, apparent at a performer’s seat within the first 10 milliseconds, and that within the 20 to 100 millisecond time frame. This time frame is standardized, denoted as the subscript early.

$$ST_{\text{early}} = 10 \log \left( \frac{\int_{0.01}^{0.02} p^2(t) \, dt}{\int_{0}^{0.01} p^2(t) \, dt} \right) \text{[dB]}.$$  \hspace{1cm} (2.15)

2.2 Related work regarding acoustical parameters

2.2.1 Concert Halls and Opera Houses

One of the most influential and important persons in the field of concert hall acoustics is Leo Leroy Beranek (1914 - ). His work began in the 1930s when as a student at MIT he got in touch with many of the leading scientists in field of acoustics (Acoustical Society of America 2014). The passion for concert hall acoustics took off as Beranek conducted a survey regarding the subjective acoustic perception of concert halls. The survey inquired the world’s leading conductors, musicians and music critics at the time. The result was a ranking list –eventually being the foundation the book "Concert Halls and Opera Houses: Music, Acoustics, and Architecture", today revised in a second edition.

The main part of Beranek’s book is a presentation of 100 concert halls and opera houses from around the world. The presentation includes a summary of each hall’s objective acoustical measures, a brief description of its history and the subjective perceptions of its acoustics. In the latter part of the book, the halls are evaluated to form a ranking list of the subjective acoustical quality and the relation to objective measurements.

2.2.2 Matching subjective perceptions and objective parameters

Many scientists in the field have been inspired by Beranek’s work, leading to a major reformation in 1997 of the measuring standard for concert hall acoustics, the ISO 3382-1. Yet today, the standard is evolving so that the subjective perception can be measured objectively (Gade 2013). Scientists have agreed upon a set of parameters which links subjective perceptions to actual objective measurements (Barron 2009; Beranek 2004; International Organization for Standardization 2009). These are presented in Table 2.2.
2.2. RELATED WORK REGARDING ACOUSTICAL PARAMETERS

Table 2.2: A listener’s subjective perceptions can be linked to objective measurements.

<table>
<thead>
<tr>
<th>SUBJECTIVE PERCEPTION</th>
<th>OBJECTIVE PARAMETER</th>
</tr>
</thead>
<tbody>
<tr>
<td>Clarity</td>
<td>$C_{50}, D_{50} \text{ and } T_S$</td>
</tr>
<tr>
<td>Intimacy</td>
<td>ITDG and $G$</td>
</tr>
<tr>
<td>Listener envelopment</td>
<td>$LF_{late} \text{ and } G_{late}$</td>
</tr>
<tr>
<td>Loudness</td>
<td>$G$</td>
</tr>
<tr>
<td>Reverberance</td>
<td>EDT</td>
</tr>
<tr>
<td>Source broadening</td>
<td>$LF_{early}$</td>
</tr>
<tr>
<td>Warmth</td>
<td>$BR$</td>
</tr>
</tbody>
</table>

Many of the parameters are closely related to each other and therefore differences between the subjective and objective match prevail. According to Beranek (2008; 2004), Barron (2009) and Pätyinen et al. (2014), a good concert hall should have the following objective and subjective parametric values:

- The average reverberation time of the middle frequency octave bands average should be in the range of 1.7 to 2.1 seconds, or 0.1 seconds longer if the early decay time is measured. The values are stated for fully occupied halls, so if measured without audience a correction must be applied, described in subsection 2.2.3.

- There should be an intimate relation between the performed music and the audience, meaning that the hall should not be perceived as too big. The initial-time-delay gap should be at a maximum of 35 milliseconds.

- Sound strength influences the limitations of the dynamic range of the hall. With a satisfactory sound strength the performed music can be perceived as more dramatic and interesting. The value of the parameter should exceed 0 dB throughout the hall.

- The impression of surround sound is mainly affected by three objective parameters. First, the clarity parameter is controlled by the structure and texture of surfaces, like walls and ceiling. In a good concert hall it should be in the range of $-2 \text{ dB}$ to $2 \text{ dB}$. Second, the lateral fraction affects the perception of the surrounding sound but also the width of sound source. It should be in the range of 0.1 to 0.35. Third and final, the binaural quality index should be high in order to give the listener a good impression of surround sound.

2.2.3 Estimation of parameters

One of the challenges when doing acoustical measurements in a concert hall is the absence of an audience. For achieving the most realistic values of the acoustical parameters, the hall should be filled with audience and performers along with their instruments. But such situations are often hard to achieve since the measurements require complete silence during many measurement sessions. Also, it can be harmful to be present in the room if safety equipment is not used. Instead, almost every acoustical measurement is conducted in empty halls and the results are thereafter corrected in order to simulate a fully occupied
CHAPTER 2. THEORY

The correction factor is based on experience from measurements in both unoccupied and occupied halls but also from measurements in laboratories and physical assumptions (Hidaka et al. 2001; Beranek 2004; Skålevik 2010).

The seats themselves are one of the most significant absorbents in a concert hall and therefore one of the most investigated (Beranek and Hidaka 1998; Hidaka et al. 2001; Beranek 2006; Rossell and Vicent 2002). Often seats are constructed in such way that the absorption coefficient will remain the same with or without audience present. Beranek and Hidaka have formulated absorption coefficients for three different types of seats: lightly-, medium-, and heavily upholstered. When the lightly upholstered seats are used, the absorption coefficient will significantly increase as audience is present, whereas for the heavily upholstered seats, this effect is less significant. A formula of calculating the occupied reverberation time, $RT_{occupied}$, from unoccupied values, $RT_{unoccupied}$, is presented by Hidaka et al. (2001) as

$$RT_{occupied} = a - be^{-RT_{unoccupied}} [s],$$

(2.16)

where $a$ and $b$ are regression coefficients obtained from measurements according to Table 2.3.

Table 2.3: Coefficients for calculating the reverberation time of an occupied hall when the unoccupied reverberation time is measured.

<table>
<thead>
<tr>
<th>OCTAVE BAND CENTER FREQUENCY [Hz]</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1000</th>
<th>2000</th>
<th>4000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Regression coefficient $a$</td>
<td>2.58</td>
<td>2.46</td>
<td>2.31</td>
<td>2.19</td>
<td>2.07</td>
<td>1.84</td>
</tr>
<tr>
<td>Regression coefficient $b$</td>
<td>4.83</td>
<td>4.50</td>
<td>4.26</td>
<td>3.91</td>
<td>3.48</td>
<td>2.45</td>
</tr>
</tbody>
</table>

2.3 Related work regarding reverberation systems

2.3.1 Reverberation systems

The aim of a reverberation system is to modify the reverberant space of a room. The systems can be divided into two categories; passive and active systems. An elementary thing to do in order to change the reverberant space is to add or remove absorptive or reflective materials in the room. If one desires a more reverberant sound, i.e. make the sound die out slowly, all soft materials should be replaced by harder ones. This is an example of a passive reverberation system. Another way to change the reverberation of a room is to add a series of microphones and loudspeakers to the room. The microphones pick up the sound in the room and emit the signal through the loudspeakers into the room again. This is a typical active reverberation system. If the emitted signal is modified, the system is referred to as an artificial reverberation system.

In concert halls, a passive system could for example be sheets of fabric that can be unfolded along the walls or in the ceiling. More permanently, reverberation chambers can be constructed and joined with the actual concert hall. In other words, the room’s physical properties are changed by adding or subtracting reflectors and absorbents, and
2.3. RELATED WORK REGARDING REVERBERATION SYSTEMS

by adjusting the volume of the room. For a long time, passive systems have been the only accepted methods of enhancing the acoustics of concert halls, while the artificial reverberation systems have not been as acknowledged. However, it is worth mentioning that the artificial reverberation systems are often more flexible when it comes to the amount of reverberation enhancement. That is because it uses different parts of the sound field to modify the reverberation (Kleiner 1990).

2.3.2 Artificial reverberation systems, ARS

The theory of artificial reverberation systems and the actual physical systems have been implemented and tested on several locations since the 1960s. The first ARS was introduced in the Royal Festival Hall in London in 1964 (Svensson 1994; Parkin and Morgan 1965 cited in). However, despite the rise of ARS installations, there is still no simple way of modeling the effect of these systems (Rouch and Schmich 2012). Svensson (1994) conveys that the effect to the sound energy, caused by an ARS, was first described by Franssen in 1968 in terms of mathematics. His expression was later revised by Philips Electroacoustic Division and de Koning in 1983. The sound energy of the room without an ARS, $w_{s, inactive}$, is expressed as

$$w_{s, inactive} = P_0(1 - \bar{\alpha}) \frac{4}{cA'}, \quad (2.17)$$

where $P_0$ is the power of the sound source, $\bar{\alpha}$ is the average absorption factor of the room, $c$ is the speed of sound and $A'$ is the total absorption area. When introducing an ARS of $n_L$ uncorrelated loudspeakers, the energy density in the room, $w_{s, active}$, is expressed as

$$w_{s, active} = P_0(1 - \bar{\alpha}) \frac{4}{cA'} \frac{1}{1 - n_L \frac{4\mu^2}{cA'}(1 - \bar{\alpha})}, \quad (2.18)$$

$$w_{s, active} = w_{s, inactive} \frac{1}{1 - n_L S^2}, \quad (2.19)$$

$$S^2 = \frac{4\mu^2}{cA'}(1 - \bar{\alpha}), \quad (2.20)$$

and where $\mu^2$ is the gain factor of each loudspeaker output. The reverberation time is proportional to the sound energy and is accordingly affected by the change of sound energy in the room as

$$RT_{active} = RT_{inactive} \frac{1}{1 - n_L S^2} \quad [s]. \quad (2.21)$$
CHAPTER 2. THEORY

Figure 2.2: The sound energy field is affected by the MCR system. The original reverberation field (left figure) is affected by the MCR-system (middle figure), resulting in a longer reverberation time (right figure).

It is now understood that an ARS increases the energy of a room. However, the approach is somewhat different between systems; The Acoustical Control System, ACS, places its microphones in the direct field near the source and delays the output of the loudspeakers; The Active Field Control, AFC, uses a more digital approach by emitting synthesized reflections to the audience; The Multi-Channel Reverberation, MCR, places its microphones in the diffuse field and emits an amplified signal, only effecting the later part of the sound energy, seen in Figure 2.2 (Svensson 1994; Kleiner 1990; Bakker and Yamaha Commercial Audio Systems Europe 2012; De Koning 1983)

The MCR is designed to regenerate a natural sound as if it had been reflected by a surface present in the hall. The microphones are positioned in the diffuse field and each microphone has a designated loudspeaker emitting the signal back into the hall. Due to the risk of acoustic feedback, this system causes a limitation of the signal amplification. The feedback can be seen as a never ending loop, lasting as long as the chain is not interrupted. In practice, this is controlled by level controls and equalizers along each microphone-loudspeaker chain. However, this causes the output sound level of each chain to be weak, therefore several chains have to be used in order to create a noticeable change of the reverberation time. Phillips continued to develop the MCR-system during the 1980s, yielding a maximum of 2 % increase of the reverberation time per number of chains without introducing artifacts. In reality however, acousticians expect about 1.25 % increase per chain. If the sound level of a chain is exceeded, but still not causing acoustic feedback, artifacts such as coloration and localization can occur (Mulder 2001; Kahle and Mulder 2015; De Koning 1983; Svensson 1994). Figure 2.3 displays the schematics of an MCR using two chains where each contribution-chain is marked by an individual color.

---

1The signal from the ARS interfere with the natural frequency content of the direct sound
2The sound level from the ARS’s loudspeakers is predominant in the receiver position
2.3 RELATED WORK REGARDING REVERBERATION SYSTEMS

Figure 2.3: Schematics of an artificial reverberation system. The colors indicate different sound paths corresponding to each ARS chain and the natural reflections of the room.

2.3.3 Sustainability

The implementation of an artificial reverberation system in a hall can transform it into a multi-purpose venue. This would not only be convenient for the audience, to only have one address to keep in mind, but also for the technicians as they can avoid heavy lifting etc. in order to change the acoustics of the hall to fit the performance. For example, the desired reverberation time for organ music is 2.1 to 4.2 seconds, early-classical music is 1.6 to 1.8 seconds, whereas amplified music such as rock or pop music desire 0.5 to 1.2 seconds (Hoover and Ellison 2013). The idea of multi-purpose venues can also be justified in the aspect of sustainability. This has been studied by Schwenke and Duty (2010). Their conclusions were that:

- the volume of the building can be reduced,
- the surface materials can be thin and light,
- there can be more efficient seating with deeper balconies,
- amounts of material used for changing the reverberation can be minimized.

The effects of these conclusions contribute to a larger extent of material recycling and reuse of existing buildings, less waste and consequently, a smaller footprint (Schwenke and Duty 2010).

2.3.4 The Stockholm Concert Hall

The Grand Hall of the Stockholm Concert Hall was opened in 1926. Since the opening, it has gone through major changes to improve its acoustical properties. It was in this very hall that one of the first artificial reverberation systems for enhancing the reverberation was
implemented. However, at that time, the electronic amplification met great resistance by the conservative philharmonic society (Dahlstedt 1974; Karlén 2015). Recently, acceptance has increased and another artificial reverberation enhancement system is used in the hall since 2014, yielding a 0.3 second longer reverberation time. The relatively short increase of the reverberation time is supposed to correct the sound energy loss caused by the semi-transparent ceiling that holds the lighting and some technical equipment in place (Kahle Acoustics 2014; Stockholms Konserthusstiftelse 2015).
Part II

Ranking the Stockholm Concert Hall
Chapter 3

Theory of acoustical measurements in concert halls

3.1 Transfer functions

The sound propagation in a room can be expressed as a second order differential equation, as explained in section 7.1. For engineering purposes, such equations can often be linearized and therefore handled as a system, expressed by Equation 3.1 and described in Figure 3.1 (Webb et al. 1999).

\[ y(t) = h(t) \ast x(t), \quad (3.1) \]

where \( y(t) \) is the output of the system, \( h(t) \) is the transfer function of the system and \( x(t) \) is the input of the system. The asterisk represents a convolution operation.

![Diagram of a system](image)

Figure 3.1: Schematic illustration of a system.

The sound that is emitted from a source is reflected in a variety of surfaces and a part of it is also absorbed, until it finally reaches the receiver. In a system approach, the emitted sound is the input of the system, \( x(t) \), and the sound reaching the receiver is the output of the system, \( y(t) \). The output signal is affected by the reflections and absorption of the room, described by the characteristic transfer function, \( h(t) \), of the room. In other words, the transfer function describes how the signal is changed on its way from a source to a receiver (Riederer 1996). The transfer function can be derived by using the Fourier transform or from impulse response measurements (Riederer 1996; Webb et al. 1999), described in the following section.
CHAPTER 3. THEORY OF ACOUSTICAL MEASUREMENTS IN CONCERT HALLS

3.2 Impulse response function

An impulse response function of a room is acquired by exciting the room with an impulse and recording the response. The impulse can be created from a pistol shoot, a distinct hand clap or by popping a balloon. Such impulse excite a broad range of frequencies and can mathematically be described as a Dirac delta function (Beerends 2003; Rossing et al. 2002). From the impulse response function of the room, all of the previous mentioned acoustical parameters can be derived.

An alternative method to acquire the impulse response function, \( h(t) \), is to convolute the response of the room, \( y(t) \), when excited by an exponential sinusoidal swipe signal, with a designed filter signal, \( f(t) \). This is expressed in time domain representation as

\[
\begin{align*}
    h(t) &= y(t) * f(t), \\
    \text{where } y(t) &= \sin[K(e^{L} - 1)], \\
    \text{and } K &= \frac{T\omega_{\text{start}}}{\ln(\frac{\omega_{\text{stop}}}{\omega_{\text{start}}})}, \quad L = \frac{T}{\ln(\frac{\omega_{\text{stop}}}{\omega_{\text{start}}})},
\end{align*}
\]

and \( \omega_{\text{start}} \) and \( \omega_{\text{stop}} \) is the start and stop frequency of the exponential sweep, denoted as angular frequencies and \( T \) is the signal duration in seconds.

The filter signal is constructed from the swipe signal as its time-reversal, illustrated in frequency domain in Figures 3.2a and 3.2b. When multiplying the output signal of the system with the filter signal, both represented in frequency domain, the outcome is the impulse response function of the room, later transformed into time domain as illustrated in Figure 3.3a.

The advantage of this method is that the signal-to-noise ratio is higher than the previous mentioned methods. This means that there can be substantial background noise present when measuring, without affecting the result. Another advantage is that harmonic distortion can be isolated from the resulting impulse response, since the distortion will occur ahead of the impulse itself. This can be seen in Figure 3.3b as vertical lines (like ghost impulses) before the actual impulse that occurs at the time 17 seconds (Farina 2007;
3.3. THE ARS CONTRIBUTION

Stan et al. 2002; Meng et al. 2008). The colors in Figure 3.3b represent the magnitude of the particular frequency, blue symbolizes a low magnitude whereas red colors symbolizes high magnitude. Figures 3.2a to 3.3b are retrieved from (Meng et al. 2008; p.3).

(a) Impulse response of a room, in time domain. (b) Impulse response of a room, in frequency domain, where blue and red color indicate low and high magnitude, respectively.

Figure 3.5: The obtained impulse response, h(t), viewed in (a) time- and (b) frequency domain, acquired using the sinusoidal sweep signal method.

3.3 The ARS contribution

If the microphone- and loudspeaker position are exactly the same during the measurements of the impulse response function, with the ARS activated and inactive respectively, then the ARS contribution, \( h_{ARS} \), can be extracted according to Svensson et al. (1992). This is done by subtracting the measured impulse response as the ARS was active, \( h_{TOT, ARS-active} \), with the same when inactive, \( h_{TOT, ARS-inactive} \), as expressed in the following equation:

\[
h_{ARS} = h_{TOT, ARS-active} - h_{TOT, ARS-inactive}.
\]

In order to use this equation successfully, a measurement of the position accuracy must be examine. This is done by studying the correlation between the amplitude of the two impulse responses. If the positions were unchanged, then the two signals should be in phase and only differ in amplitude. This is illustrated in the graphs below, where Figure 3.4a illustrates the signals in phase and Figure 3.4b when the signals are out of phase due to slightly shifted microphone or loudspeaker position.

The correlation coefficient, \( r \), can serve as an indicator of the accuracy. If \( r \) takes the value of 1.0, this will indicate that the two signals are exactly the same, but if \( r \) approaches a value of 0, the opposite can be said about the signals – indicating an extreme change of the equipment position (Clark 2013; Lund Research Ltd 2013). For the signals in phase, shown in Figure 3.4a, the correlation chart looks like Figure 3.5a, where \( r \) is 1.0. For the signals out of phase, shown in Figure 3.4b, the correlation chart looks like Figure 3.5b, where \( r \) is 0.86.
CHAPTER 3. THEORY OF ACOUSTICAL MEASUREMENTS IN CONCERT HALLS

(a) The signals are in phase.  
(b) The signals are out of phase.

Figure 3.4: Two sinusoidal signals with the same frequency but different amplitude.

(a) Signals in phase.  
(b) Signals out of phase.

Figure 3.5: Correlation charts between signals.
Chapter 4

Method

4.1 Implementation method

4.1.1 Calibration method

In order to acquire correct measurements of the acoustical parameters of a concert hall, the measurements must be planned with care. If the obtained data is to be compared with other data, standard procedures should be followed. In this case the international standard ISO 3382-1:2009 is used with some additional measurements. ISO 3382-1:2009 explains two ways of how to calibrate the sound source. The most convenient is to make a free-field calibration in the concert hall itself. Thus no extra venue is required. The alternative is a diffuse-field calibration made in a reverberation room. The drawback to the free-field calibration is the inaccuracy in low frequencies. In order to get correct measurements of the acoustical strength parameter, $G$, calibration must be done (International Organization for Standardization 2009; Brüel & Kjær 2015a).

In this thesis, the diffuse-field calibration was done using Dirac (version 6)\(^1\) and the same equipment which was used during the measuring occasions, see subsection 4.1.1. The calibration took place in the Reverberation room at Marcus Wallenberg Laboratory, MWL, at Royal Institute of Technology, KTH. The room has a volume of 247 m\(^3\) and the surface material inside the room is hard and reflective (Bodén 2011), resulting in a flat frequency response of the room itself. The frequency response of the room is measured as the loudspeaker emits the exponential sinusoidal sweep signal. Since the frequency response of the room is flat, the measured sound is the loudspeaker’s frequency response. This is repeated for several microphone and loudspeaker positions according to the calibration process (Brüel & Kjær 2015a).

4.1.2 Equipment setup

Measurements were performed on the 4\(^{th}\) of May 2015. The equipment used can be seen in Table A.1 in Appendix A. The setup was straight forward, illustrated in Figure 4.1. The

\(^1\)A computer software used to calculate acoustical parameters from impulse response measurements. More information can be found at http://www.acoustics-engineering.com/html/dirac.html
computer runs Dirac which is set to both send and receive a signal. The output signal goes from an external sound card to the amplifier and thereafter to the loudspeaker. The omnidirectional microphone, connected to the input of the external sound card, picks up the signal emitted by the loudspeaker. The signal was an exponential sinusoidal sweep from 0 Hz to 22050 Hz, taking 11.9 seconds to complete.

\[ \text{Figure 4.1: The generalized setup used in the acoustical measurements.} \]

In order to determine the LF parameter, another measurement was performed on the 11th of May 2015. The setup was similar to the former, but the receiver was replaced with a sound field microphone. The microphone is made out of four tilted figure-of-eight microphones close to each other, giving the possibility to determine the directivity of the recorded sound. The sound card was also replaced and the recording software used was IRIS (version 1.0)\(^2\). The signal used was again an exponential sinusoidal sweep from 20 Hz to 20000 Hz, taking 30 seconds to complete. The entire equipment list is found in Table A.2 in Appendix A.

### 4.1.3 Procedure

The measurement in Stockholm Concert Hall consisted of 14 different receiver positions distributed over the parquet (5 positions), the stage (1 position), the choir balcony (1 position), the first balcony (4 positions) and the second balcony (3 positions). The height of the microphone was approximately 1.2 meter relative to the floor in each position. According to ISO 3382-1:2009 the minimum number of receiver positions is a function of the auditorium size. It requires at least ten microphone positions for a hall of 2000 seats. Stockholm Concert Hall has about 1800 seats. In line with the standard, the source was altered in three different positions on the stage for each receiver position. Finally, the artificial reverberation system was altered between being turned off and on for every

\(^2\)Computer software for capturing and analyzing room impulse responses in 3D. More information can be found at http://www.iris.co.nz/
4.2. ANALYSIS METHOD

combination of receiver and source position. In total, 84 measurements were conducted during each of the two measurement occasions. The source and receiver positions are marked on the blueprint of the hall, presented in Figure A.1 in Appendix A.

4.2 Analysis method

4.2.1 Reliability of measurements

The reliability of the measurements was briefly assessed by comparing the data from the two measurement occasions (the Dirac-system and the IRIS-system). It is done by comparing the frequency content of the two signals, measured in the same position but with the two measuring equipment respectively. Also, a comparison of the decay of sound energy will show that the measurements are reliable if they give the same, or almost the same, results.

The deviations between the two sets of data were also evaluated against the just-noticeable difference (JND). The JND is a value that corresponds to the smallest audible change in the quantity of a parameter. These values can be fixed or variable depending on the parameter examined (Bradley and Wang 2007; International Organization for Standardization 2009).

4.2.2 Correlation

To determine the accuracy between the performed measurements, the correlation between the two signals can be used. If the microphone- or loudspeaker position was changed between the measurements of the impulse response function, as the ARS was switched between active and inactive, this would be evident in the correlation calculations.

4.2.3 Comparison of acoustical parameters

The values of the acoustical parameters obtained from the measurements with the ARS active and inactive, respectively, are compared against each other to see the effect of the reverberation system itself. Again, the JND is used as a reference for how large the acoustical difference, caused by the ARS, is. Finally, the resulting parameter values, corresponding to the ARS-active measurements, are compared with those of other concert halls which are presented in Beranek’s work.
Chapter 5

Results

5.1 Calibration results

The calibration shows that the loudspeaker has a relatively flat frequency response from the octave band with the center frequency of 500 Hz to 4000 Hz, as shown in Figure 5.1. For frequencies below 500 Hz, the loudspeaker output is amplified, and for frequencies higher than 4000 Hz, the magnitude of the sound level drops rapidly. The data from the measurements of the Stockholm Concert Hall is calibrated with the average of these calibration measurements to compensate for the loudspeaker’s frequency response.

Figure 5.1: Energy magnitude spectrum of the calibration signal, obtained in the Reverberation room at MWL, shows the frequency response of the loudspeaker used during the measurements.
5.2 Comparison of the two sets of equipment

The two setups of measurement equipment show a high resemblance when looking at the sound energy decay curve of the same position, as shown in Figure 5.2. The comparison of the acoustical parameters show small deviations between the two measuring setups. The majority of the parameter deviations lay within the JND-limits.

![Figure 5.2: The sound energy decay curve of the 500 Hz octave band in the same source-receiver position but measured with different equipment setups; green curve represents the Dirac-system and the blue represents the IRIS-system.](image)

5.3 The effects of the ARS

The measurements of the impulse response function, when the ARS was inactive and active, were analyzed. Examples of the correlation between the signals in the same source-receiver position can be seen in Figures B.23a and B.23b. Here, \( r = 0.93 \) for the signals obtained in the source-receiver position S1R13. The signals obtained in the source-receiver position S2R3 have the correlation coefficient value 0.37.
5.3. THE EFFECTS OF THE ARS

(a) Source-receiver position S1R13.

(b) Source-receiver position S2R3.

Figure 5.3: Correlation charts between the obtained impulse response functions, when the ARS was inactive and active.
As mentioned in subsection 2.3.2, only an increase of the sound energy in the latter part of the signal was expected. This would be equivalent to a less steep slope of the energy decay curve. Figure 5.4 displays that the sound energy in source-receiver position S1R13 dissipates more slowly when the ARS is active, as expected.

![Energy decay of the 500 Hz octave band of two signals obtained in source-receiver position S1R13; the green represents the signal with the ARS inactive, and the blue represents the signal with the ARS active.](image)

In Table 5.1, a summary of the acoustical parameter values are shown, measured with the ARS inactive and active, respectively. It can be seen, in the column "DIFFERENCE", that the reverberation time is extended by about 0.3 seconds, which corresponds to approximately three times the just-noticeable difference, explained by column "DIFF. IN NUMBEERS OF JND". Moreover, the sound strength, $G$, is amplified with a half of JND, and the clarity, $C_{80}$, is decreased with the same amount of JND. In Appendix B, complete tables for groups of receiver positions are shown, e.g. receivers 1 to 4 represent the average of the parquet, and receivers 12 to 14 represents the 2nd balcony.
5.4. COMPARISON WITH OTHER HALLS

Table 5.1: Results of the ARS effect. Single number frequency averaging of acoustical parameters measured in Stockholm Concert Hall with the artificial reverberation system inactive (ARS-INACTIVE) and active (ARS-ACTIVE).

<table>
<thead>
<tr>
<th>SINGLE NUMBER FREQUENCY AVG. OF PARAMETER</th>
<th>AVG. ALL RECEIVER POS. ARS-INACTIVE</th>
<th>AVG. ALL RECEIVER POS. ARS-ACTIVE</th>
<th>DIFFERENCE</th>
<th>1 JND. BASED ON NUMBERS OF JND</th>
<th>DIFF. IN NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>1.68</td>
<td>1.94</td>
<td>0.26</td>
<td>0.08</td>
<td>3.1</td>
</tr>
<tr>
<td>$T_{30}$ (s)</td>
<td>1.75</td>
<td>2.03</td>
<td>0.27</td>
<td>0.09</td>
<td>3.1</td>
</tr>
<tr>
<td>$T_{50}$ (s)</td>
<td>1.75</td>
<td>2.03</td>
<td>0.28</td>
<td>0.09</td>
<td>3.2</td>
</tr>
<tr>
<td>BR (-)</td>
<td>1.01</td>
<td>1.10</td>
<td>0.09</td>
<td>-1.00</td>
<td>-0.3</td>
</tr>
<tr>
<td>$G$ (dB)</td>
<td>5.03</td>
<td>5.32</td>
<td>0.29</td>
<td>1.00</td>
<td>0.3</td>
</tr>
<tr>
<td>$G$ [0, 80] (dB)</td>
<td>2.41</td>
<td>2.38</td>
<td>-0.04</td>
<td>1.00</td>
<td>0.0</td>
</tr>
<tr>
<td>$G$ [80, ∞] (dB)</td>
<td>1.42</td>
<td>2.09</td>
<td>0.66</td>
<td>1.00</td>
<td>0.7</td>
</tr>
<tr>
<td>$G_{125}$ (dB)</td>
<td>5.42</td>
<td>6.04</td>
<td>0.62</td>
<td>1.00</td>
<td>0.6</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.24</td>
<td>0.24</td>
<td>0.00</td>
<td>0.05</td>
<td>0.1</td>
</tr>
<tr>
<td>$L_{F_{med}}$ (-)</td>
<td>0.27</td>
<td>0.26</td>
<td>-0.01</td>
<td>0.05</td>
<td>0.1</td>
</tr>
<tr>
<td>$L_{F_{C}}$ (-)</td>
<td>0.30</td>
<td>0.30</td>
<td>0.00</td>
<td>0.05</td>
<td>0.0</td>
</tr>
<tr>
<td>$G_{LL}$ (dB)</td>
<td>-2.53</td>
<td>-1.71</td>
<td>0.82</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>$C_{60}$ (dB)</td>
<td>0.99</td>
<td>0.29</td>
<td>-0.70</td>
<td>1.00</td>
<td>0.7</td>
</tr>
<tr>
<td>$C_{50}$ (dB)</td>
<td>-2.25</td>
<td>-2.75</td>
<td>-0.50</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>$D_{50}$ (%)</td>
<td>0.40</td>
<td>0.38</td>
<td>-0.03</td>
<td>0.05</td>
<td>0.6</td>
</tr>
<tr>
<td>$T_{S}$ (ms)</td>
<td>109.17</td>
<td>125.14</td>
<td>15.96</td>
<td>10.00</td>
<td>1.6</td>
</tr>
<tr>
<td>$E_{C_{music}}$ (-)</td>
<td>1.05</td>
<td>0.93</td>
<td>-0.12</td>
<td>-</td>
<td>-</td>
</tr>
</tbody>
</table>

5.4 Comparison with other halls

The acoustical parameters derived from the measurements in the Stockholm Concert Hall, as the ARS was active, were compared to corresponding parameters of other concert halls, presented by Beranek (2004). The following Figures 5.5 to 5.7 are reproduced from Beranek’s work. The results from the Stockholm Concert Hall are superimposed on the figures, marked with a red dashed line.

Reverberation Time

The reverberation time of the hall, when fully occupied, was estimated to 1.7 seconds, according to subsection 2.2.3. This was based on the measured average reverberation time of the middle octave frequency bands when the hall was empty. In Figure 5.5, the subjectively best rated hall is found in the left hand side of the horizontal axis. Stockholm Concert Hall value is based only on the objective parameter value.

Early Decay Time

The early decay time of the measured hall is averaged to 1.9 seconds. Once again, Figure 5.6 reveals that this value of the measured EDT results in a middle-ranking for the Stockholm Concert Hall.
Sound strength

The strength of the sound in the hall is measured to be 5.3 dB on an average. If $G$ is displayed in relation to the ratio of $EDT$ and hall volume, $V$, then the Stockholm Concert Hall is placed on the right hand side of the line representing a 3 dB slope per doubling of the $EDT$ per $V$ ratio, as seen in Figure 5.7.

Figure 5.5: The reverberation time in occupied state of each hall denoted along the horizontal axis, sorted by descending subjective ranking. The Stockholm Concert Hall is represented by the red dashed line.
5.4. COMPARISON WITH OTHER HALLS

Figure 5.6: The measured early decay time for thirty-six concert halls, sorted by descending subjective ranking, along with the result of measurements of the Stockholm Concert Hall—marked by the red dashed line.
Figure 5.7: The strength factor in relation to the EDT/V ratio. Stockholm Concert Hall can be found in the right hand side of the diagonal line.
Chapter 6

Discussion

6.1 Calibration discussion

The non-flat frequency response of the loudspeaker that was discovered during the calibration was not expected. The strong dip in the higher frequencies sets limitations on the useful frequency range of the measurements. The ideal would be if the loudspeaker had a flat frequency response throughout the human hearing range, i.e. roughly 20 Hz to 20 kHz. The analysis considered only the middle frequency octave bands due to this finding. Earlier studies by Beranek (2003) and Svensson (1991) etc. did also only cover these octave bands.

6.2 Equipment and implementation discussion

As showed in subsection 4.1.2, there were two measurement occasions with different equipment used. The results in section 5.2, show that it is unnecessary to use both sets of equipment. The Dirac-system can be seen as a more flexible setup due to the minor complexity of the required equipment. However, the benefit of the IRIS-system is that parameters which require omnidirectional and figure-of-eight directional microphones can be measured. The drawback of the IRIS-system is that the microphone heads cannot be calibrated with a hand-held calibrator; only a pre-calibration value obtained by the manufacturer is available. This may cause incorrect measurement levels, if, for some reason, the microphone response curve has changed.

The distance between the source and the receiver should be sufficient in order for the receiver not to pick up the direct sound only. The distance is dependent on the volume of the hall and the reverberation time (Bodén 2011; Christensen and Koutsouris 2015). For Stockholm Concert Hall, the minimum source-receiver distance is calculated to about 12 meter. This would imply that the receiver position 5, 6 and 9 are sometimes too close to the source, thus yielding inaccurate values of the measured acoustical parameters, as the direct sound is prominent in that particular combination of positions.
6.3 Discussion about ARS effect

The correlation coefficient between the two measurements in the same position, with ARS active and inactive respectively, should be close to one (1.0), hence the ARS-active signal is only an amplification of the ARS-inactive signal. Still, the correlation between most of the measurements was not close to this. This can be explained by recalling the wavelengths of sound waves: signals containing frequencies of 1000 Hz have the wavelength of about 0.3 meters, which means that the pressure maximum only occur each 15 centimeters. Therefore, a slight move of the source or the receiver would easily affect the result. During the performed measurements, the loudspeaker and microphone were unfortunately moved several times before the ARS state was changed. One would have preferred if the ARS state was altered before any source and receiver positions were changed. The effects of mismatching positions can be seen in the correlation charts in Appendix B.

The reverberation system affects the acoustical properties of the hall. However, the effect is not excessive or audible for all the measured parameters. The ARS effect is displayed in numbers of JNDs which is a suitable measure for the audibility of the effect. It appears that the audibly affected parameters are those measuring the reverberation time and the center time. However, the sound clarity and sound strength are also affected by the ARS, yet not to the point of becoming audibly. These results imply that the ARS affects the sound the way the designers wanted –yielding as few negative consequences as possible. Furthermore, it is discussed if the subjective impressions of the ARS effects are more prominent when listening to a single impulse response than if listening to a classical music concert. The author believes that auditory masking and other psychoacoustic effects play an important role and therefore the measured ARS effect would be hard to perceive during an actual concert.

The analysis of the ARS effect was partly done in the frequency domain; comparing the energy magnitude spectrum of the signals respectively. This led to the discussion of how the ARS, and the MCR in particular, works. The ARS uses the natural reverberation of the room which entails that imperfections of the reverberation spectrum of the room will be present also when the ARS is active. Thus, using such a system does not improve the frequency content of the room, as some would expect.

6.4 Discussion of ranking

The ranking list found in Beranek’s work is initially formed out of subjective perceptions of the acoustics of the concert hall. Thereafter, attempts have been made to match those perceptions with objective measurements of corresponding acoustical parameters. In this master thesis, only objective measurements has been taken into consideration when doing the rating of Stockholm Concert Hall.

The values of the measured acoustical parameters shows that the Stockholm Concert Hall is similar to many other halls. This leads to a reasonable suggestion from the author to rank it as a middle-class concert hall, only based on objective data. The fact that the strength parameter measured in Stockholm Concert Hall was fairly high would imply
6.4. DISCUSSION OF RANKING

that it is a bit too loud for a hall of that size. The rise of the $G$ due to the ARS was not subjectively noticeable but if an even longer reverberation time was desired it would have caused problems with the sound strength.
Part III

Simulating an artificial reverberation system
Chapter 7

Theory of acoustics simulation

7.1 Modeling and simulation

To predict the acoustical properties of a new concert hall before construction has always been important for acousticians and architects. During the 20th century this was done by physical scaled models of the intended hall. Nowadays, it is more convenient to use a computer model to achieve the same result. These models can also be used when acoustical renovations of an existing hall is planned and designed (Barron 2009; Beranek 2011; Bork 2000).

The room acoustical simulation software of today can predict any of the acoustical parameters mentioned earlier in section 2.1. The accuracy of the software is important to acousticians. In an article in The Journal of the Acoustical Society of America (2013; p. 1203), Vorländer state that

"...[computer simulations] are indeed powerful tools, but their reliability depends on the skill of the operator..."

The uncertainties of the simulated acoustical parameters are still an issue but nowadays it is not due to the lack of computational power. Instead the uncertainties are due to the lack of knowledge of material properties used in the model and how it affects the sound propagation (Vorländer 2013; Foteinou and Murphy 2012). Clearly, with great skill and time, a satisfactory result can be obtained. Case studies by Brüel & Kjaer show that the results from the simulation software Odeon are not completely accurate but show the overall expected result (Brüel & Kjaer 2009; 2011).

Looking at it scientifically, sound is waves of pressure variations propagating through a medium. When a loudspeaker moves the air particles around it, a slight concentration of particles builds up a wave front that propagates away from the loudspeaker (Bodén 2011). It is difficult to explain the correct physics of a propagating sound wave since different frequencies may result in different types of propagation. The mathematical expression used to describe the propagation is the wave equation—a second order differential equation. Often, a propagating sound wave is approximated to have similar properties as a ray of light, especially when it comes to simulations. Unlike light, sound waves have great
variation when it comes to the wavelength, \( \lambda \). For visible light the wavelength is of the order \( 10^{-7} \) m whereas for sound it is of the order \( 10^2 \) m, derived from (7.1).

\[
\lambda = \frac{c}{f} \ [\text{m}],
\]

where \( c \) is the speed of sound in air and \( f \) is the frequency in hertz. This means that all light, when having the wavelength of sound as a reference, can be approximated as having the same order of wavelength and therefore the same properties. This cannot be assumed when dealing with sound. Therefore several models for the different wavelengths of the sound are needed (Röber et al. 2007).

The most realistic and physically correct model of sound propagation is the wave-based approach with methods like Finite Element Method (FEM) or waveguides. These methods use a mesh of the medium and describes how the different elements of it interact when excited by an external force. Despite its physically correct approach, the methods are only accurate in the low frequency range.

Another approach to simulate sound wave propagation is by using methods based on geometrical models where the sound wave propagation is approximated as a ray. The ray of sound interacts with the scenery as if it was a ray of light. The background research on the geometrical modeling approach comes from the visualization and computer graphics field where this approach is dominant (Röber et al. 2007; Svensson and Kristiansen 2002; Alpkocak and Sis 2010; Vorländer 2013; Välimäki et al. 2012). There are two geometrical methods used in Odeon; one for the direct sound that reaches the receiver and one for the sound that has been reflected in the surfaces surrounding the scenery before reaching the receiver (Christensen and Koutsouris 2015; Foteinou and Murphy 2012). These are the Image source method and the Ray trace method which are described in more detail in the next sections.

Many scientists are currently researching the development of room acoustic simulations. One suggestion is to use several models in order to achieve a more realistic result throughout the audible frequency range (Välimäki et al. 2012). Moreover, new methods approaching sound energy propagation to be similar to heat transfer have been proposed (Navarro and Escolano 2015).

### 7.2 Image source method

The Image source method deals not only with the direct sound but it also handles the first two reflections as the direct sound from a virtual source from outside the room, as illustrated in Figure 7.1. These are often called "early rays". The amplitude of the virtual direct sound wave is recalculated accounting for the acoustical properties of the surface through which the wave passes on its way to the receiver. However, the roughness of the surface cannot be handled with this method since only specular reflections are concerned. Therefore another model must be used to handle the roughness of the reflecting surface, namely the Ray trace method (Duraiswami 2006; Kufner 2008; Christensen and Koutsouris 2015; Svensson and Kristiansen 2002).
7.3 RAY TRACE METHOD

Figure 7.1: The image source method creates virtual sources for the first reflections of the sound wave—making the reflection appear as direct sound from the virtual source.

7.3 Ray trace method

The Ray trace method is used to calculate the diffuse reflections of acoustical waves which is caused by the roughness of the reflective surfaces. These are referred to as the "late" rays. In the reflection points on the surfaces, small secondary sources are generated. The power and directivity pattern of these sources are both recalculated with regard for the structure of the surfaces (Kufner 2008; Svensson and Kristiansen 2002; Christensen and Koutsouris 2015).

Figure 7.2: In the Ray trace method reflections are scattered due to the roughness of the reflective surface, resulting in both specular reflections (solid line) and diffuse reflections (dashed lines). Retrieved from (Svensson and Kristiansen 2002; p. 6).
Chapter 8

Method

In order to acquire a realistic simulation of the acoustics of a venue having an artificial reverberation system, there are three steps to go through. First, a three dimensional computer model of the venue must be constructed. Second, the simulation of the sound propagation of the hall must be done, in order to obtain each transfer function of the source-receiver path, including those of the ARS. Third, all transfer functions that are involved to compile the resulting impulse response in a specific position are added together. These three steps to a realistic result are further described in the following sections.

8.1 Modeling

The model created in this project was done in SketchUp (version 15.3.331)\(^1\). It was based on blueprints, photos and actual measurements of Stockholm Concert Hall. When making a computer model of a venue, for the purpose of doing acoustical simulations, one must approximate the reality in some aspects. For example, the details of surfaces in a concert hall, e.g. the ceiling and seats, are simplified in order to make the transfer to the simulation software possible. This will of course affect the accuracy of the result. However, it can be compensated for by editing the surface properties in the simulation program (Christensen and Koutsouris 2015).

When the model is transferred and opened in Odeon (version 13)\(^2\), the properties of each surface must be set. The materials can be chosen from a material list which is based on actual acoustical measurements of the absorption coefficient of the material. If no predefined material matches the actual material of the examined surface, an approximation is done either by choosing a similar material or setting a possible absorption coefficient. Each surface must also be assigned a scattering coefficient, describing its roughness. The coefficient can be measured in a laboratory environment but is often hard to achieve when making models of an existing concert hall, thus it is often approximated. The coefficient is

\(^1\)Computer-assisted design software for making technical drawings in 3D. More information can be found at http://www.sketchup.com/

\(^2\)A room acoustics software for measurement, simulation and auralisation. More information can be found at http://www.odeon.dk/
a single value between zero and one, 0 to 1.0, where zero is an ideal flat surface, yielding only specular reflections (Embrechts et al. 2001; Zeng et al. 2006; Wang and Rathsam 2008; Christensen and Koutsouris 2015).

### 8.2 Simulation

In Odeon, the reflection of the sound wave against a surface is based on the roughness of the reflective surface. The roughness is controlled by the earlier mentioned scattering coefficient, $s$. When simulating a reflection, Odeon calculates the final direction of the sound ray as the resultant of two reflection vectors. These vectors are based on the specular reflection and the diffuse reflection, respectively. The specular vector is described by Snell’s law and is weighted by the value of $1-s$. The diffuse vector is described by a random direction according to Lambert’s specular reflection distribution. This vector is weighted by the value of $s$, as illustrated in Figure 8.1 (Christensen and Koutsouris 2015).

![Figure 8.1: The resulting reflection direction calculated by two reflection vectors based on the specular- and the diffuse reflection. Retrieved from (Christensen and Koutsouris 2015; p. 71).](image)

The settings in Odeon have to be adjusted with care in order to achieve a realistic result. These are the settings for this model: the impulse response length was set to 2700 ms; the number of early rays used in the model was 6846 and number of late rays was 3423; and the transition order was set to 2 as recommended by Odeon.

### 8.3 MATLAB-script

The transfer functions for each combination of source-receiver position and the ARS components are obtained from Odeon. In order to get a realistic result of the total impulse response in a specific position, when the reverberation system is active, all transfer functions related to that position must be combined. This is done in MATLAB (version
8.4. ANALYSIS METHOD

R2014b\(^3\), by using (8.1) and (8.2) obtained from related works by Svensson (1991; 1994).

\[ H_TOT(\omega) = H_{SR}(\omega) + H_{RES}(\omega), \]  

\[ H_{RES}(\omega) = H_{SM}(\omega) \frac{G_{ML}}{1 - G_{ML}(\omega)H_{LM}(\omega)}, \]  

and \( H_{TOT} \) denotes the frequency response function of a particular source-receiver position when the ARS is active; \( H_{SR} \) refers to the transfer function between the source and receiver including all natural reflections of the hall; \( H_{ARS} \) refers to the total transfer function caused by the ARS alone; \( H_{SM} \) refers to the transfer function between the source and the ARS microphone; \( G_{ML} \) refers to the electronically transfer function between the ARS microphone and loudspeaker; and finally, \( H_{LM} \) refers to the transfer function between the ARS loudspeaker and microphone. The MATLAB-script can be found in Appendix C.

The electronic transfer function has a limitation due to the instability of the acoustical feedback of the system. As seen in (8.2), the product of \( G_{ML} \) and \( H_{LM} \) cannot be equal to one, since the denominator is then zero, leading to an undefined quotient. In this thesis, the solution is to construct \( G_{ML} \) from the \( H_{LM} \) values as

\[ G_{ML}(\omega) = \frac{\max |H_{LM}(\omega)|}{g \cdot d}, \]  

where \( g \) and \( d \) are constants. In the calculations leading to the presented results, \( g \) and \( d \) have the value of 0.9 and 12, respectively.

8.4 Analysis method

To analyze the validity of the results from the simulation, a comparison between the simulated and the measured acoustical parameters was done. In order to determine if the simulation results correspond to reality, the difference between the simulated and the measured data was evaluated in relation to the just-noticeable difference. With the intention of avoiding simulation errors in the low frequency range, due to the shortages of the simulation method, only the 500 Hz and 1000 Hz octave bands were analyzed.
Chapter 9

Results

In the following subsections, results of the simulated acoustical parameters, as the ARS is active, will be presented. Figures 9.1 to 9.3 show the parametric value for each receiver position as the average of the 500 Hz and 1000 Hz octave band and three source positions. The blue bars represent the measured parameter value and the yellow bars represent the corresponding simulated values. The red circles indicate the absolute difference between the measured and simulated data, and the green dashed line is the individual limit for the JND of each receiver position. In Appendix B, the complete table and graphs of the simulated data can be found.

9.1 Reverberation Time

The results of the simulated reverberation time, $T_{30}$, were accurate, when compared to the measured data throughout the concert hall. As seen in Figure 9.1, the difference between the simulated data and the measured data exceeds the JND in 3 out of 14 receiver positions.

9.2 Early Decay Time

The simulated early decay time agrees well with the measured data in 6 out of 14 receiver positions, if the JND is referred to as the separator. At the same time, some simulations of the parameter were overestimated, namely for receiver number 4, 5, 6, 9 and 10.

9.3 Clarity

The results of the simulated clarity parameter shows significant deviations in several of the receiver positions. The simulated clarity was in some receiver positions by no means in line with the measured data. As seen in Figure 9.3, the receiver positions 2, 3 and 5 have the opposite positive or negative value of the measured parameter. Moreover, in receiver position 9, the difference is excessive. However, in the remaining positions, the difference between the measured and simulated data does not, or just slightly, exceed the JND.
Figure 9.1: The simulated reverberation time, $T_{30}$, is close to the measured data for 3 out of 14 receiver positions.
9.3. CLARITY

Figure 9.2: The early decay time measured and simulated in each receiver position. The difference is shown in red and the corresponding JND is the dashed green line.
Figure 9.3: The measured and simulated values of the clarity parameter in each receiver position. In some receiver positions, the simulated value was not accurately estimated.
Chapter 10

Discussion

10.1 Methodological limitations

The simulated total impulse response function could not instantly be obtained in Odeon, which would be possible for simulation of a venue without an ARS. Instead, each transfer function had to be obtained individually and then combined through a MATLAB-script. This is due to the fact that the ARS uses the natural reverberation of the hall, i.e. the ARS microphones pick up the sound present in the diffuse sound field. Consequently, much of the Odeon evaluation functionalities, such as Grid-maps and Auralisation, cannot be used in such a simulation. Therefore, the evaluation of the simulated parameters had to be done one by one, for all receiver positions. This extra step is time consuming and therefore a drawback to the simulation possibilities.

During the work, it was found that Odeon has a default setting of applying a head related transfer function (HRTF) on the simulated impulse response. This is normally done in order to simulate what a human would hear if he or she was in that particular position; the result would be more realistic. However, since the measured acoustical properties of the Stockholm Concert Hall were obtained with a single microphone instead of a dummy-head, adding such a HRTF to the simulated data would not correspond to the measured data. In earlier stages of the work, this was not corrected which resulted in discrepancies in relation to the reality.

Furthermore, the created electronic transfer function, $G_{LM}$, included two constants, $g$ and $d$, in order to overcome the acoustical feedback of the system; as presented in section 8.3. The constants are the result of trial and error until satisfactory values of the acoustical parameters were obtained. However, it is suggested by the author that these constants would correspond to physical properties of the ARS. The constant $g$ would correspond to a resistance in the system due to sound energy attenuation, and the constant $d$ would correspond to the number of microphone-loudspeaker chains used in the ARS.
10.2 The simulation outcome

As mentioned in section 6.2, some of the source-receiver distances were too short, resulting in incorrect measured values of the acoustical parameters. Similarly, this is the case for the simulated values of the acoustical parameters, hence the receiver number 5, 6 and 9 should be neglected.

Even though the model of the Stockholm Concert Hall was simplified in many aspects, the simulation yields a relatively realistic result, especially when it comes to the reverberation time, $T_{30}$, with the ARS in active state. This is probably due to the settings of the surface properties in the model. The overall absorption coefficient is close to the actual one, but some surfaces are more and some less absorptive than in reality. Moreover, the scattering coefficient of the surfaces are not completely correct set, hence the poorly estimation of the clarity parameter in some of the receiver positions. The most erroneous estimation of $C_{80}$ was the one corresponding to receiver position number 2 (without taking into consideration the receivers to be neglected, as stated in the paragraph above). This is probably due to the property of the seats, which in reality causes a more spread sound, i.e. having a higher scattering coefficient. Due to the lack of time, this has not been investigated further, yet it would be easy to do since the absorption and scattering coefficients can be controlled in Odeon.
Part IV

Further work and Conclusion
Chapter 11

Further work

11.1 Electronic transfer function

The construction of the electronic transfer function, $G_{ML}$, used in the MATLAB-script, was derived from the values of the measured transfer function, $H_{LM}$, between the ARS loudspeakers and microphones. In this case, the ARS was already installed in the investigated hall. Nevertheless, if such a system is about to be installed in another venue, such a transfer function cannot be measured and used in advance of the installation. As a continuation of this work, a deeper understanding of the construction of $G_{ML}$ could be investigated.

11.2 Stage vibrations

In parallel to this study, there has been a discussion about the sound energy radiated from vibrations in the stage floor. The stage of the Stockholm Concert Hall is divided into several parts which are hydraulically operable in height. The floor material of some parts have been modified in a way that vibrations in the structure increase or decrease. It is suggested by Askenfelt and Guettler (2013) that the vibrations will help the musicians to perform on their instrument as they get feedback from the structure, but this will also mean that the airborne sound energy will be compromised; yielding a shorter reverberation time. In connection to the measurements performed in this study, some additional stage vibration measurements were done. At the end of the summer of 2015, all the parts of the stage were changed in favor of the musicians. It would be interesting to measure the effect on the reverberation time and other acoustical parameters investigated in this thesis, caused by the change of stage floor material.

11.3 Other simulation methods

As noticed, when conducting this thesis, there are many other methods of how to model the sound propagation. The models used in this work were limited to only geometrical models that give good results in middle and high frequency range. The lack of estimating the lower frequency range is intriguing and further work using other methods, such as
CHAPTER 11. FURTHER WORK

FEM, could be possible. The desirable outcome would be to combine several different simulation methods in order to achieve an even more realistic result, covering the whole frequency range.

Finally, a further study would be to achieve real time simulations where the total impulse response and acoustical parameters are instantly calculated and viewed as walking around in the model. It would also be convenient to avoid running any additional script to get the total impulse response of a room with an artificial reverberation system present. This could be a topic for another bachelor- or master thesis.
Chapter 12

Conclusion

This thesis has investigated the acoustical properties of the Stockholm Concert Hall which has an artificial reverberation system installed. The reverberation system is in general activated during the events in the hall, compensating for the sound energy loss of the semitransparent ceiling over the stage and parquet.

It has been seen that the artificial reverberation system in the Stockholm Concert Hall is both helping and hindering the acoustical properties of the hall. As the reverberation time of the hall is increased, the clarity of the sound is decreased. However, the negative outcome is right below the just-noticeable difference, whereas the positive impact of reverberation time is approximately three times the JND, i.e. it should be audible. The acoustical properties of the Stockholm Concert Hall, when the reverberation system is active are: $T_{30,\text{unoccupied}}=2.0\text{ s}$, $T_{30,\text{occupied}}=1.7\text{ s (estimated)}$, $EDT=1.9\text{ second}$, $C_{80}=0.29\text{ dB}$ and $G=5.3\text{ dB}$. These parametric values indicate that the investigated hall should be ranked as a middle class concert hall –not entirely reaching up to the standard of the best ones in the world. The measurement data and a summary of the findings are now sent to Beranek, in hope of extending the next edition of his book, "Concert Halls and Opera Houses: Music, Acoustics, and Architecture", with the Stockholm Concert Hall.

The second objective of the thesis was to investigate the possibilities of simulating the artificial reverberation system in Odeon. It is possible to achieve an almost realistic simulation, if evaluating the simulation results based on the just-noticeable difference. However, the impulse response cannot be derived from the software directly, thus a MATLAB-script was constructed. The script adds up each transfer function of the source to receiver path, along with those of the reverberation system, all obtained from Odeon. Still there is much to learn in this area, especially when it comes to the understanding of the electronic transfer function between the microphones and loudspeakers of the reverberation system, hindering the acoustical feedback. However, in this thesis it was handled with non-physical constants yielding satisfactory results.
Bibliography


Brüel & Kjær (2015a). Dirac 6, calibration.

Brüel & Kjær (2015b). Dirac 6, energy ratios.


Appendix A

Measurement notes
### APPENDIX A. MEASUREMENT NOTES

#### Table A.1: Measurement equipment used when measuring the Stockholm Concert Hall on the 4th of May 2015, with the Dirac system.

<table>
<thead>
<tr>
<th>INSTRUMENT</th>
<th>MANUFACTURER</th>
<th>TYPE NO.</th>
<th>SERIAL NO.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Loudspeaker</td>
<td>Norsonic</td>
<td>P270H</td>
<td>30779</td>
</tr>
<tr>
<td>Amplifier</td>
<td>Norsonic</td>
<td>P280</td>
<td>2804215</td>
</tr>
<tr>
<td>Sound pressure level meter</td>
<td>Brüel &amp; Kjær</td>
<td>2250</td>
<td>3005972</td>
</tr>
<tr>
<td>Microphone</td>
<td>Brüel &amp; Kjær</td>
<td>4189</td>
<td>2851042</td>
</tr>
<tr>
<td>Calibrator</td>
<td>Brüel &amp; Kjær</td>
<td>4231</td>
<td>2605907</td>
</tr>
<tr>
<td>Laser distance meter</td>
<td>BOSCH</td>
<td>DLE 40 Professional</td>
<td>101926548</td>
</tr>
<tr>
<td>Computer</td>
<td>HP</td>
<td>HP EliteBook 840 G1 Notebook PC</td>
<td>101926548</td>
</tr>
<tr>
<td>Software</td>
<td>Acoustics Engineering</td>
<td>Dirac</td>
<td>v.6.0.5510.2034</td>
</tr>
<tr>
<td>Soundcard</td>
<td>Brüel &amp; Kjær</td>
<td>ZE 0948</td>
<td>02006</td>
</tr>
<tr>
<td>Microphone cable</td>
<td>Brüel &amp; Kjær</td>
<td>MiniXLR 6-pin</td>
<td>PPA 8020</td>
</tr>
<tr>
<td>Cable; 3.5 mm phone connection</td>
<td>Brüel &amp; Kjær</td>
<td>2260</td>
<td>0586-D-150</td>
</tr>
<tr>
<td>Cable; phone – XLR connection</td>
<td>Brüel &amp; Kjær</td>
<td>PPX 8020</td>
<td>30779</td>
</tr>
<tr>
<td>Loudspeaker cable</td>
<td>Speakon</td>
<td>2 pcs. Total 15 meters</td>
<td></td>
</tr>
<tr>
<td>Adapter; 3.5 mm phone to RCA</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Tripods</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### Table A.2: Measurement equipment used when measuring the Stockholm Concert Hall on the 11th of May 2015, with the IRIS system.

<table>
<thead>
<tr>
<th>INSTRUMENT</th>
<th>MANUFACTURER</th>
<th>TYPE NO.</th>
<th>SERIAL NO.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Loudspeaker</td>
<td>Norsonic</td>
<td>P270H</td>
<td>30779</td>
</tr>
<tr>
<td>Amplifier</td>
<td>Norsonic</td>
<td>P280</td>
<td>2804215</td>
</tr>
<tr>
<td>Microphone</td>
<td>Core Sound</td>
<td>2294</td>
<td></td>
</tr>
<tr>
<td>Laser distance meter</td>
<td>BOSCH</td>
<td>DLE 40 Professional</td>
<td>101926548</td>
</tr>
<tr>
<td>Computer</td>
<td>HP</td>
<td>HP EliteBook 840 G1 Notebook PC</td>
<td>101926548</td>
</tr>
<tr>
<td>Software</td>
<td>Marshall Day Acoustics</td>
<td>IRIS</td>
<td>V1.0</td>
</tr>
<tr>
<td>Soundcard</td>
<td>MOTU</td>
<td>NO</td>
<td></td>
</tr>
<tr>
<td>MiniXLR to CAT5e transmitter</td>
<td>Core Sound</td>
<td>PPA 8020</td>
<td></td>
</tr>
<tr>
<td>XLR to CAT5e receiver</td>
<td>Core Sound</td>
<td>CBM 8020</td>
<td></td>
</tr>
<tr>
<td>Microphone cable</td>
<td>Core Sound</td>
<td>MiniXLR 6-pin</td>
<td>PPA 8020</td>
</tr>
<tr>
<td>Cable; EtherCon CAT5e</td>
<td>Brüel &amp; Kjær</td>
<td>30 meters</td>
<td></td>
</tr>
<tr>
<td>Cable; 3.5 mm phone connection</td>
<td>Brüel &amp; Kjær</td>
<td>12-08-2-217</td>
<td></td>
</tr>
<tr>
<td>Cable; phone – XLR connection</td>
<td>Brüel &amp; Kjær</td>
<td>12-08-2-216</td>
<td></td>
</tr>
<tr>
<td>Loudspeaker cable</td>
<td>Speakon</td>
<td>2 pcs. Total 15 meters</td>
<td></td>
</tr>
<tr>
<td>Tripods</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Figure A.1: Source and receiver position during measurements in Stockholm Concert Hall.
Appendix B

Results in tables and graphs
### Table B.1: Comparison of the ARS effect from the measurement data, average over all receiver positions.

<table>
<thead>
<tr>
<th>SINGLE NUMBER FREQUENCY AVG. OF PARAMETER</th>
<th>AVG. ALL RECEIVER POS. ARS-INACTIVE</th>
<th>AVG. ALL RECEIVER POS. ARS-ACTIVE</th>
<th>DIFFERENCE</th>
<th>1 JND, BASED ON NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>1.68</td>
<td>1.94</td>
<td>0.26</td>
<td>0.08</td>
</tr>
<tr>
<td>$T_{20}$ (s)</td>
<td>1.75</td>
<td>2.03</td>
<td>0.27</td>
<td>0.09</td>
</tr>
<tr>
<td>$T_{30}$ (s)</td>
<td>1.75</td>
<td>2.03</td>
<td>0.28</td>
<td>0.09</td>
</tr>
<tr>
<td>BF (-)</td>
<td>1.01</td>
<td>1.10</td>
<td>0.09</td>
<td>-</td>
</tr>
<tr>
<td>$G$ (dB)</td>
<td>5.03</td>
<td>5.32</td>
<td>0.29</td>
<td>1.00</td>
</tr>
<tr>
<td>$G_{[0, 80]}$ (dB)</td>
<td>2.41</td>
<td>2.38</td>
<td>-0.04</td>
<td>1.00</td>
</tr>
<tr>
<td>$G_{[80, \infty]}$ (dB)</td>
<td>1.42</td>
<td>2.09</td>
<td>0.66</td>
<td>1.00</td>
</tr>
<tr>
<td>$G_{125}$ (dB)</td>
<td>5.42</td>
<td>6.04</td>
<td>0.62</td>
<td>1.00</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.24</td>
<td>0.24</td>
<td>0.00</td>
<td>0.05</td>
</tr>
<tr>
<td>$L_{F\text{med}}$ (-)</td>
<td>0.27</td>
<td>0.26</td>
<td>-0.01</td>
<td>0.05</td>
</tr>
<tr>
<td>$L_{FC}$ (-)</td>
<td>0.30</td>
<td>0.30</td>
<td>0.00</td>
<td>0.05</td>
</tr>
<tr>
<td>GLL (dB)</td>
<td>-2.53</td>
<td>-1.71</td>
<td>0.82</td>
<td>-</td>
</tr>
<tr>
<td>$C_0$ (dB)</td>
<td>0.99</td>
<td>0.29</td>
<td>-0.70</td>
<td>1.00</td>
</tr>
<tr>
<td>$C_0$ (dB)</td>
<td>-2.25</td>
<td>-2.75</td>
<td>-0.50</td>
<td>-</td>
</tr>
<tr>
<td>$D_{50}$ (%)</td>
<td>0.40</td>
<td>0.38</td>
<td>-0.03</td>
<td>0.05</td>
</tr>
<tr>
<td>$T_{S}$ (ms)</td>
<td>109.17</td>
<td>125.14</td>
<td>15.96</td>
<td>10.00</td>
</tr>
<tr>
<td>$E_{C\text{music}}$ (-)</td>
<td>1.05</td>
<td>0.93</td>
<td>-0.12</td>
<td>-</td>
</tr>
</tbody>
</table>

### Table B.2: Comparison of the ARS effect from the measurement data, average over parquet receiver positions.

<table>
<thead>
<tr>
<th>SINGLE NUMBER FREQUENCY AVG. OF PARAMETER</th>
<th>AVG. OVER PARQUET (R1:R4) ARS-INACTIVE</th>
<th>AVG. OVER PARQUET (R1:R4) ARS-ACTIVE</th>
<th>DIFFERENCE</th>
<th>1 JND, BASED ON NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>1.65</td>
<td>1.90</td>
<td>0.25</td>
<td>0.08</td>
</tr>
<tr>
<td>$T_{20}$ (s)</td>
<td>1.75</td>
<td>2.01</td>
<td>0.27</td>
<td>0.09</td>
</tr>
<tr>
<td>$T_{30}$ (s)</td>
<td>1.75</td>
<td>2.03</td>
<td>0.28</td>
<td>0.09</td>
</tr>
<tr>
<td>BF (-)</td>
<td>1.00</td>
<td>1.11</td>
<td>0.11</td>
<td>-</td>
</tr>
<tr>
<td>$G$ (dB)</td>
<td>5.71</td>
<td>6.04</td>
<td>0.33</td>
<td>1.00</td>
</tr>
<tr>
<td>$G_{[0, 80]}$ (dB)</td>
<td>2.79</td>
<td>2.79</td>
<td>0.00</td>
<td>1.00</td>
</tr>
<tr>
<td>$G_{[80, \infty]}$ (dB)</td>
<td>2.58</td>
<td>3.23</td>
<td>0.64</td>
<td>1.00</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.29</td>
<td>0.28</td>
<td>0.00</td>
<td>0.05</td>
</tr>
<tr>
<td>$L_{F\text{med}}$ (-)</td>
<td>0.33</td>
<td>0.30</td>
<td>-0.03</td>
<td>0.05</td>
</tr>
<tr>
<td>$L_{FC}$ (-)</td>
<td>0.33</td>
<td>0.33</td>
<td>0.00</td>
<td>0.05</td>
</tr>
<tr>
<td>GLL (dB)</td>
<td>-1.60</td>
<td>-0.89</td>
<td>0.71</td>
<td>-</td>
</tr>
<tr>
<td>$C_0$ (dB)</td>
<td>-0.20</td>
<td>-0.44</td>
<td>-0.64</td>
<td>1.00</td>
</tr>
<tr>
<td>$C_0$ (dB)</td>
<td>-3.22</td>
<td>-3.83</td>
<td>-0.61</td>
<td>0.00</td>
</tr>
<tr>
<td>$D_{50}$ (%)</td>
<td>0.33</td>
<td>0.31</td>
<td>-0.03</td>
<td>0.05</td>
</tr>
<tr>
<td>$T_{S}$ (ms)</td>
<td>116.67</td>
<td>132.16</td>
<td>15.49</td>
<td>10.00</td>
</tr>
<tr>
<td>$E_{C\text{music}}$ (-)</td>
<td>0.90</td>
<td>0.87</td>
<td>-0.03</td>
<td>-</td>
</tr>
<tr>
<td>$G_{125}$ (dB)</td>
<td>6.09</td>
<td>6.66</td>
<td>0.57</td>
<td>1.00</td>
</tr>
</tbody>
</table>
### Table B.3: Comparison of the ARS effect from the measurement data, average over 1st balcony receiver positions.

<table>
<thead>
<tr>
<th>SINGLE NUMBER</th>
<th>AVG. OVER 1ST BALCONY (R7:8, R10:11)</th>
<th>AVG. OVER 1ST BALCONY (R7:8, R10:11)</th>
<th>DIFFERENCE</th>
<th>1 JND OF JND</th>
<th>DIFF. IN NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>1.63</td>
<td>1.91</td>
<td>0.28</td>
<td>0.08</td>
<td>3.4</td>
</tr>
<tr>
<td>TR (s)</td>
<td>1.76</td>
<td>2.03</td>
<td>0.27</td>
<td>0.09</td>
<td>3.1</td>
</tr>
<tr>
<td>TR (s)</td>
<td>1.75</td>
<td>2.03</td>
<td>0.28</td>
<td>0.09</td>
<td>3.2</td>
</tr>
<tr>
<td>BR (-)</td>
<td>1.02</td>
<td>1.10</td>
<td>0.08</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>G (dB)</td>
<td>3.38</td>
<td>3.59</td>
<td>0.21</td>
<td>1.00</td>
<td>0.2</td>
</tr>
<tr>
<td>G [0, 80] (dB)</td>
<td>0.49</td>
<td>0.20</td>
<td>0.69</td>
<td>1.00</td>
<td>0.7</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.24</td>
<td>0.25</td>
<td>0.01</td>
<td>0.05</td>
<td>0.2</td>
</tr>
<tr>
<td>LFmid (-)</td>
<td>0.27</td>
<td>0.28</td>
<td>0.01</td>
<td>0.05</td>
<td>0.3</td>
</tr>
<tr>
<td>GLL (dB)</td>
<td>1.50</td>
<td>0.65</td>
<td>-0.85</td>
<td>1.00</td>
<td>-</td>
</tr>
<tr>
<td>C80 (dB)</td>
<td>-1.73</td>
<td>-2.04</td>
<td>-0.31</td>
<td>0.00</td>
<td>-</td>
</tr>
<tr>
<td>D50 (%)</td>
<td>0.43</td>
<td>0.40</td>
<td>-0.03</td>
<td>0.05</td>
<td>0.6</td>
</tr>
<tr>
<td>TS (ms)</td>
<td>104.32</td>
<td>121.57</td>
<td>17.26</td>
<td>10.00</td>
<td>1.7</td>
</tr>
<tr>
<td>ECmusic (-)</td>
<td>1.00</td>
<td>0.81</td>
<td>-0.19</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>G125 (dB)</td>
<td>2.88</td>
<td>3.57</td>
<td>0.69</td>
<td>1.00</td>
<td>0.7</td>
</tr>
</tbody>
</table>

### Table B.4: Comparison of the ARS effect from the measurement data, average over 2nd balcony receiver positions.

<table>
<thead>
<tr>
<th>SINGLE NUMBER</th>
<th>AVG. OVER 2ND BALCONY (R12:R14)</th>
<th>AVG. OVER 2ND BALCONY (R12:R14)</th>
<th>DIFFERENCE</th>
<th>1 JND OF JND</th>
<th>DIFF. IN NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>1.81</td>
<td>2.06</td>
<td>0.25</td>
<td>0.09</td>
<td>2.8</td>
</tr>
<tr>
<td>TR (s)</td>
<td>1.77</td>
<td>2.05</td>
<td>0.28</td>
<td>0.09</td>
<td>3.2</td>
</tr>
<tr>
<td>TR (s)</td>
<td>1.76</td>
<td>2.05</td>
<td>0.28</td>
<td>0.09</td>
<td>3.2</td>
</tr>
<tr>
<td>BR (-)</td>
<td>1.03</td>
<td>1.14</td>
<td>0.11</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>G (dB)</td>
<td>3.02</td>
<td>3.54</td>
<td>0.52</td>
<td>1.00</td>
<td>0.5</td>
</tr>
<tr>
<td>G [0, 80] (dB)</td>
<td>-0.15</td>
<td>-0.02</td>
<td>0.13</td>
<td>1.00</td>
<td>0.1</td>
</tr>
<tr>
<td>G [80, ∞] (dB)</td>
<td>0.05</td>
<td>0.91</td>
<td>0.85</td>
<td>1.00</td>
<td>0.9</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.22</td>
<td>0.22</td>
<td>0.00</td>
<td>0.05</td>
<td>0.0</td>
</tr>
<tr>
<td>LFmid (-)</td>
<td>0.25</td>
<td>0.25</td>
<td>0.00</td>
<td>0.05</td>
<td>0.0</td>
</tr>
<tr>
<td>GLL (dB)</td>
<td>-3.57</td>
<td>-2.63</td>
<td>1.15</td>
<td>0.00</td>
<td>-</td>
</tr>
<tr>
<td>C80 (dB)</td>
<td>-0.20</td>
<td>-0.93</td>
<td>-0.73</td>
<td>1.00</td>
<td>0.7</td>
</tr>
<tr>
<td>C50 (dB)</td>
<td>-3.53</td>
<td>-4.27</td>
<td>-0.74</td>
<td>0.00</td>
<td>-</td>
</tr>
<tr>
<td>D50 (%)</td>
<td>0.35</td>
<td>0.31</td>
<td>-0.03</td>
<td>0.05</td>
<td>0.6</td>
</tr>
<tr>
<td>TS (ms)</td>
<td>126.13</td>
<td>144.24</td>
<td>18.12</td>
<td>10.00</td>
<td>1.8</td>
</tr>
<tr>
<td>ECmusic (-)</td>
<td>0.94</td>
<td>0.82</td>
<td>-0.13</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>G125 (dB)</td>
<td>3.97</td>
<td>4.82</td>
<td>0.85</td>
<td>1.00</td>
<td>0.9</td>
</tr>
</tbody>
</table>
Figure B.3: Correlation charts for source-receiver position S1R1-S1R6.
Figure B.6: Correlation charts for source-receiver position S1R7-S1R12.
APPENDIX B. RESULTS IN TABLES AND GRAPHS

Figure B.7: Correlation charts for source-receiver position S1R13-S1R14.

Figure B.9: Correlation charts for source-receiver position S2R1-S2R4.
Figure B.12: Correlation charts for source-receiver position S2R5-S2R10.
Figure B.14: Correlation charts for source-receiver position S2R11-S2R14.

Figure B.15: Correlation charts for source-receiver position S3R1-S2R2.
Figure B.18: Correlation charts for source-receiver position S3R3-S2R8.
Figure B.21: Correlation charts for source-receiver position S3R9-S3R14.
Table B.5: Comparison of the measured and simulated data when ARS was active, average over all receiver positions.

<table>
<thead>
<tr>
<th>SINGLE NUMBER FREQUENCY AVG. OF PARAMETER</th>
<th>AVG. ALL RECEIVER POS. MEASURED</th>
<th>AVG. ALL RECEIVER POS. SIMULATED</th>
<th>DIFFERENCE</th>
<th>1 JND BASED ON NUMBERS OF JND</th>
<th>DIFF. IN NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>1.94</td>
<td>2.22</td>
<td>-0.28</td>
<td>0.10</td>
<td>2.9</td>
</tr>
<tr>
<td>T50 (s)</td>
<td>2.03</td>
<td>2.03</td>
<td>-0.01</td>
<td>0.10</td>
<td>0.1</td>
</tr>
<tr>
<td>T90 (s)</td>
<td>2.03</td>
<td>1.98</td>
<td>0.05</td>
<td>0.10</td>
<td>0.5</td>
</tr>
<tr>
<td>BR (-)</td>
<td>1.10</td>
<td>1.12</td>
<td>-0.02</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>G (dB)</td>
<td>5.32</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
<td>-</td>
</tr>
<tr>
<td>G [80, ∞] (dB)</td>
<td>2.98</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
<td>-</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.26</td>
<td>-</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>LFmid (-)</td>
<td>0.30</td>
<td>-</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>GLL (dB)</td>
<td>-1.71</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>C100 (dB)</td>
<td>0.29</td>
<td>0.27</td>
<td>0.02</td>
<td>1.00</td>
<td>0.0</td>
</tr>
<tr>
<td>D100 (%)</td>
<td>-2.75</td>
<td>-1.99</td>
<td>-0.77</td>
<td>0.00</td>
<td>-</td>
</tr>
<tr>
<td>T5 (ms)</td>
<td>125.14</td>
<td>133.65</td>
<td>-8.52</td>
<td>10.00</td>
<td>0.9</td>
</tr>
<tr>
<td>ECmusic (-)</td>
<td>0.93</td>
<td>0.81</td>
<td>0.12</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>G125 (dB)</td>
<td>6.04</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
<td>-</td>
</tr>
</tbody>
</table>

Table B.6: Comparison of the measured and simulated data when ARS was active, average over parquet receiver positions.

<table>
<thead>
<tr>
<th>SINGLE NUMBER FREQUENCY AVG. OF PARAMETER</th>
<th>AVG. OVER PARQUET (R1-R4) MEASURED</th>
<th>AVG. OVER PARQUET (R1-R4) SIMULATED</th>
<th>DIFFERENCE</th>
<th>1 JND BASED ON MEASURED</th>
<th>DIFF. IN NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>1.90</td>
<td>2.03</td>
<td>-0.13</td>
<td>0.09</td>
<td>1.4</td>
</tr>
<tr>
<td>T50 (s)</td>
<td>2.01</td>
<td>2.00</td>
<td>0.02</td>
<td>0.10</td>
<td>0.2</td>
</tr>
<tr>
<td>T90 (s)</td>
<td>2.03</td>
<td>2.72</td>
<td>-0.69</td>
<td>0.10</td>
<td>6.8</td>
</tr>
<tr>
<td>BR (-)</td>
<td>1.11</td>
<td>1.12</td>
<td>-0.01</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>G (dB)</td>
<td>6.04</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
<td>-</td>
</tr>
<tr>
<td>G [0, 80] (dB)</td>
<td>2.79</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
<td>-</td>
</tr>
<tr>
<td>G [80, ∞] (dB)</td>
<td>3.23</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
<td>-</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.28</td>
<td>-</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>LFmid (-)</td>
<td>0.30</td>
<td>-</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>LFC (-)</td>
<td>0.33</td>
<td>-</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>GLL (dB)</td>
<td>-0.89</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>C100 (dB)</td>
<td>-0.44</td>
<td>0.01</td>
<td>-0.44</td>
<td>1.00</td>
<td>0.4</td>
</tr>
<tr>
<td>C100 (dB)</td>
<td>-3.83</td>
<td>-2.88</td>
<td>-0.96</td>
<td>0.00</td>
<td>-</td>
</tr>
<tr>
<td>D100 (%)</td>
<td>0.31</td>
<td>0.34</td>
<td>-0.04</td>
<td>0.05</td>
<td>0.7</td>
</tr>
<tr>
<td>T5 (ms)</td>
<td>132.16</td>
<td>132.80</td>
<td>-0.64</td>
<td>10.00</td>
<td>0.1</td>
</tr>
<tr>
<td>ECmusic (-)</td>
<td>0.87</td>
<td>0.81</td>
<td>0.06</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>G125 (dB)</td>
<td>6.66</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
<td>-</td>
</tr>
</tbody>
</table>
APPENDIX B. RESULTS IN TABLES AND GRAPHS

Table B.7: Comparison of the measured and simulated data when ARS was active, average over 1st balcony receiver positions.

<table>
<thead>
<tr>
<th>SINGLE NUMBER FREQUENCY AVG. OF PARAMETER</th>
<th>AVG. OVER 1ST BALCONY (R7:8, R10:11) MEASURED</th>
<th>AVG. OVER 1ST BALCONY (R7:8, R10:11) SIMULATED</th>
<th>DIFFERENCE</th>
<th>1 JND BASED ON NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>1.91</td>
<td>2.01</td>
<td>-0.11</td>
<td>0.10</td>
</tr>
<tr>
<td>T20 (s)</td>
<td>2.03</td>
<td>2.06</td>
<td>-0.03</td>
<td>0.10</td>
</tr>
<tr>
<td>T30 (s)</td>
<td>2.03</td>
<td>1.98</td>
<td>0.05</td>
<td>0.10</td>
</tr>
<tr>
<td>BR (-)</td>
<td>1.10</td>
<td>1.12</td>
<td>-0.02</td>
<td>-</td>
</tr>
<tr>
<td>G (dB)</td>
<td>3.59</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
</tr>
<tr>
<td>G [0, 80] (dB)</td>
<td>0.85</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
</tr>
<tr>
<td>G [80, ∞] (dB)</td>
<td>0.20</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.25</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>LF_mid (-)</td>
<td>0.28</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>LFC (-)</td>
<td>0.30</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>GLL (dB)</td>
<td>-3.47</td>
<td>-</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>C80 (dB)</td>
<td>0.65</td>
<td>1.00</td>
<td>-0.35</td>
<td>1.00</td>
</tr>
<tr>
<td>C800 (dB)</td>
<td>-2.04</td>
<td>1.03</td>
<td>-3.07</td>
<td>0.00</td>
</tr>
<tr>
<td>D80 (%)</td>
<td>0.40</td>
<td>0.00</td>
<td>0.40</td>
<td>0.05</td>
</tr>
<tr>
<td>TS (ms)</td>
<td>121.57</td>
<td>142.59</td>
<td>-21.02</td>
<td>10.00</td>
</tr>
<tr>
<td>EC_music (-)</td>
<td>0.81</td>
<td>0.01</td>
<td>0.80</td>
<td>-</td>
</tr>
<tr>
<td>G125 (dB)</td>
<td>3.57</td>
<td>3.57</td>
<td>1.00</td>
<td>3.6</td>
</tr>
</tbody>
</table>

Table B.8: Comparison of the measured and simulated data when ARS was active, average over 2nd balcony receiver positions.

<table>
<thead>
<tr>
<th>SINGLE NUMBER FREQUENCY AVG. OF PARAMETER</th>
<th>AVG. OVER 2ND BALCONY (R12:R14) MEASURED</th>
<th>AVG. OVER 2ND BALCONY (R12:R14) SIMULATED</th>
<th>DIFFERENCE</th>
<th>1 JND BASED ON NUMBERS OF JND</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDT (s)</td>
<td>2.06</td>
<td>1.42</td>
<td>0.64</td>
<td>0.10</td>
</tr>
<tr>
<td>T20 (s)</td>
<td>2.05</td>
<td>1.46</td>
<td>0.59</td>
<td>0.10</td>
</tr>
<tr>
<td>T30 (s)</td>
<td>2.05</td>
<td>1.44</td>
<td>0.61</td>
<td>0.10</td>
</tr>
<tr>
<td>BR (-)</td>
<td>1.14</td>
<td>0.88</td>
<td>0.27</td>
<td>-</td>
</tr>
<tr>
<td>G (dB)</td>
<td>3.54</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
</tr>
<tr>
<td>G [0, 80] (dB)</td>
<td>-0.02</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
</tr>
<tr>
<td>G [80, ∞] (dB)</td>
<td>0.91</td>
<td>-</td>
<td>-</td>
<td>1.00</td>
</tr>
<tr>
<td>LF (-)</td>
<td>0.22</td>
<td>-</td>
<td>-</td>
<td>0.05</td>
</tr>
<tr>
<td>LF_mid (-)</td>
<td>0.25</td>
<td>-</td>
<td>-</td>
<td>0.05</td>
</tr>
<tr>
<td>LFC (-)</td>
<td>0.30</td>
<td>-</td>
<td>0.05</td>
<td>-</td>
</tr>
<tr>
<td>GLL (dB)</td>
<td>-2.43</td>
<td>-</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>C80 (dB)</td>
<td>-0.93</td>
<td>0.07</td>
<td>-1.00</td>
<td>1.00</td>
</tr>
<tr>
<td>C800 (dB)</td>
<td>-4.27</td>
<td>-1.64</td>
<td>-2.63</td>
<td>0.00</td>
</tr>
<tr>
<td>D80 (%)</td>
<td>0.31</td>
<td>0.29</td>
<td>0.03</td>
<td>0.05</td>
</tr>
<tr>
<td>TS (ms)</td>
<td>144.24</td>
<td>93.40</td>
<td>50.85</td>
<td>10.00</td>
</tr>
<tr>
<td>EC_music (-)</td>
<td>0.82</td>
<td>0.54</td>
<td>0.27</td>
<td>-</td>
</tr>
<tr>
<td>G125 (dB)</td>
<td>4.82</td>
<td>-</td>
<td>1.00</td>
<td>-</td>
</tr>
</tbody>
</table>
Figure B.22: Comparison of measured and simulated early decay time, EDT, and reverberation time, $T_{10}$. 

85
Figure B.23: Comparison of measured and simulated reverberation time, $T_{20}$ and $T_{30}$. 
Figure B.24: Comparison of measured and simulated clarity, $C_{80}$, and definition, $D_{50}$. 

87
Figure B.25: Comparison of measured and simulated center time, $T_{S}$, and echo criterion, $EC_{\text{music}}$. 

APPENDIX B. RESULTS IN TABLES AND GRAPHS
Appendix C

MATLAB-script for calculating the total impulse response
APPENDIX C. MATLAB-SCRIPT FOR CALCULATING THE TOTAL IMPULSE RESPONSE

```matlab
function y = calculate_impulse_response(input, impulse_response)
    n = length(input);
    m = length(impulse_response);
    y = zeros(1, n + m - 1);
    for k = 1:n
        for m = k:m+n-1
            y(m) = y(m) + input(k) * impulse_response(m-k+1);
        end
    end
end
```
function (b, n, m) => add_transfer_functions(source, receiver, n, m)

// source['string'] = name of the source of interest
// receiver['string'] = name of the receiver of interest
// n = number of loudspeakers in the reverberation system
// m = number of microphones in the reverberation system

display('create job list start');

if (n > 1000)
  console.log('Too many loudspeakers');
if (m > 1000)
  console.log('Too many microphones');

source = 'stage';

display('create source list start');

for (i = 0; i < n; i++)
  display('create source list ' + i);

for (i = 0; i < m; i++)
  display('create receiver list ' + i);

end

% create Loudspeaker-Receiver list
for (i = 0; i < n; i++)
  display('create job list ' + i);

end

% create source-Microphone list
for (i = 0; i < m; i++)
  display('create job list ' + i);

end

% display all jobs in ascending order, necessary to get results for the S and M of interest
job = 'job';

for (i = 0; i < n; i++)
  display('...done making list ' + i);

end

% import transfer functions (audio files) from ODEN
b = 'b.wav'

for i = 1
  b = b + '_' + (i - 1);
  b = b + '.wav';

  display('...done importing b.wav');

end
APPENDIX C. MATLAB-SCRIPT FOR CALCULATING THE TOTAL IMPULSE RESPONSE

```matlab
function [h] = total_impulse_response(num, den, t)
    % Calculates the total impulse response of a system
    % Inputs:
    %   num: numerator coefficients of the system
    %   den: denominator coefficients of the system
    %   t: time vector
    % Output:
    %   h: impulse response vector

    % Calculate the impulse response using convolution
    h = conv(num, impulse_response(den, t));

    % Remove the zero-frequency component
    h = h(2:end);
end

function [h] = impulse_response(den, t)
    % Calculates the impulse response of a system
    % Inputs:
    %   den: denominator coefficients of the system
    %   t: time vector
    % Output:
    %   h: impulse response vector

    % Impulse is a vector of ones
    impulse = ones(1, length(t));

    % Calculate the impulse response
    h = conv(impulse, conv(1, den));

    % Remove the zero-frequency component
    h = h(2:end);
end
```
results of simulating hi ast units, JNGS(1), x1); 
plot(fx,abs(x_r);, x1,abs(x_r);, x1,abs(x_r);); 
label("Frequency (Hz)"); ylabel("Magnitude"); 
print("... done ploting result"); 

% display(...all done, do not forget to save h_tot!); 
display(" create Job list done ");
