An Investigation into the Effectiveness of using Headphones with Integrated Microphones to Simulate Concert Hall Acoustics for Musicians in Small Acoustic Environments

by

Tim Hendrik ter Huurne

Thesis presented in partial fulfilment of the requirements for the degree of Master of Philosophy (Music Technology) in the Faculty of Music at Stellenbosch University

Supervisor: Dr. G.W. Roux

March 2018

Declaration

By submitting this thesis electronically, I declare that the entirety of the work contained therein is my own, original work, that I am the sole author thereof (save to the extent explicitly otherwise stated), that reproduction and publication thereof by Stellenbosch University will not infringe any third party rights and that I have not previously in its entirety or in part submitted it for obtaining any qualification.

Date: March 2018

Abstract

An Investigation into the Effectiveness of using Headphones with Integrated Microphones to Simulate Concert Hall Acoustics for Musicians in Small Acoustic Environments

T.H. ter Huurne

Department of Music,
University of Stellenbosch,
Private Bag X1, Matieland 7602, South Africa.

Thesis: MPhil (Music Technology)

December 2017

In this thesis fifty-one musicians participated in a structured interview testing the effectiveness of a headset simulating the acoustics of a concert hall known to the participants. A prototype headset was constructed by externally attaching two omnidirectional microphones to headphones. The microphone signal of the headset was processed by a convolution reverberation plugin and routed to the headphones to simulate the hall. Additionally, the filtered dry signal was reproduced over the headset to compensate for the headset's high frequency attenuation. Participants rated responses concerning: 1. the accuracy of the headset in reproducing the natural acoustic environment 2. the comfortability of the headset 3. the realism of the simulated concert hall. These ratings proved positive. While open-ended responses indicate that further refinement of the headset is required to eliminate the occlusion effect and further improve the headset's accuracy, forty-eight of the fifty-one participants answered positively to whether they could imagine practicing with a similar headset. The study concluded that headphones with integrated microphones can indeed be effective at simulating concert hall acoustics in small practice rooms.

Uittreksel

’n Ondersoek na die Doeltreffendheid van Kopfone met Geïntegreerde Mikrofone vir die Simulasie van Konsertsaalakoestiek in Klein Ruimtes

T.H. ter Huurne

Departement Musiek,
Universiteit van Stellenbosch,
Privaatsak X1, Matieland 7602, Suid Afrika.

Tesis: MPhil (Musiektegnologie)

Desember 2017

In hierdie tesis het een en vyftig musikante deelgeneem aan ’n gestruktureerde onderhoud waar die effektiwiteit van ’n kopstuk om die akoestiek van ’n bekende konsertsaal na te maak getoets is. ’n Prototipe kopstuk is gebou met twee alomgerigte mikrofone wat aan ’n stel kopfone vasgeheg is. Die seine van die mikrofone is verwerk deur ’n gesimuleerde konvolusie nagalm van die lokaal by te voeg en teruggestuur na die kopfone. Verder is daar ook gekompenseer vir die kopstuk se hoëfrekwensiedemping deur ’n vereffende droë sein terug te voer na die kopstuk. Deelnemers se gegradeerde terugvoer is ingesamel rakende: 1. die akkuraatheid van die kopstuk in die reproduksie van die akoestiese ruimte 2. die gemak daarvan 3. die realisme van die gesimuleerde saal Alhoewel respons op die oop vrae in die onderhoud daarop dui dat verdere verfyning nodig is om die afsluitingseffek te verminder en die akkuraatheid van die kopstuk te verbeter, het agt en veertig uit die een en vyftig deelnemers positief gereageer op die vraag of hulle met ’n soortgelyke kopstuk sou oefen. Die studie het bevind dat kopfone met geïntegreerde mikrofone wel daarin kan slaag om ’n konsertsaal se akoestiek te simuleer in kleiner oefenkamers.

Acknowledgements

I would like to express my sincere gratitude to the following people and organisations:

Gerhard Roux for supervising my thesis and for sharing his expertise in the field of Music Technology.

Marlene Vlok for both supporting me as well as always being there when I needed a break.

My Family for being supportive and not doubting that I will eventually complete my thesis.

Annelise and Jan Vlok for reminding me to work on my thesis and for your great hospitality.

Dedications

This thesis is dedicated to **Charlotte ter Huurne** for supporting me and taking care of me. The good food you cooked definitely contributed to the completion of this thesis.

Contents

List of Figures

List of Tables

Chapter 1

Introduction

1.1 BACKGROUND

Music students are required to practice many hours a day to master their instruments. Many of these hours, though dependent on the type of instrument, are spent in small practice rooms for solo practice. In their study, Phillips & Mace (2008:40) found the average number of practice hours for undergraduate music students, in their practice rooms, to be 2.3 hours per day. Another study, by Lamberty (1980:149), alternatively suggests that music students practice around 42 hours a week and that students' feelings towards their practice rooms can have an effect on the amount of hours they practice, as well as the benefits of this practice. According to Jorgensen (2014:4) very few studies have addressed issues regarding the influence of different institutional characteristics, including equipment and facilities, on music students. This emphasizes the need to consider the environment in which the students practice. This chapter will, therefore, briefly describe the acoustic properties of small rooms and discuss the effect of acoustics on both practicing as well as performance. This will bring to light problems of practice room acoustics and the need for variable acoustics for successful practice.

1.1.1 SMALL ROOM ACOUSTICS

Small rooms perform very differently to large rooms such as concert halls in their acoustic reproduction. Small rooms' dimensions are typically comparable to the wavelengths of the lower parts of the audible spectrum.[1] As a consequence, small rooms act as resonators in the lower frequency spectrum. According to, Everest *et al.* (2009:331), small rooms need to be considered as a resonant cavity below 300 Hz, as the air resonates in sympathy with the sound source. Above 300 Hz, however, the sound produced by the source can be considered as

[1] The audio spectrum ranges from approximately 20 Hz to 20 kHz, corresponding to wavelengths of approximately 17 m to 1.70 cm (Rumsey & McCormick, 2009:4; Szymanski, 2008:97)

rays (Everest *et al.*, 2009). Dickreiter *et al.* (2008:9) also state that reflection of sound at a barrier is analogous to the laws that govern the reflection of light, on the condition that the barrier is significantly large in comparison to the wavelength of the sound. Some of the acoustic parameters of small rooms will be discussed to understand the environment in which musicians are required to practice.

1.1.1.1 ROOM MODES

Resonances occur when sound waves of specific frequencies reflect back on themselves to form standing waves (Jones, 2008:127). At these specific frequencies the reflected waves will interfere with the direct wave in such a way as to create points of no displacement as well as points where the air vibrates between maximum positive and negative displacements. These resonances make for uneven sound pressure levels across the room at different frequencies (Rumsey & McCormick, 2009:25).

The room will resonate at different modes, very much like a string. The fundamental resonant mode between two parallel walls will occur at a frequency whose wavelength is double the length of the distance between the walls. The walls will further give rise to resonances at frequencies that are integer multiples of this 'fundamental' mode, also known as harmonics (Jones, 1990:56). The simplest and most dominant resonant modes will occur between two parallel boundaries in a room. Secondary modes are formed by reflections between more than two boundaries. Rumsey & McCormick (2009:24) claim that as a rule of thumb, only resonances below about 200 Hz are problematic, as the places of no displacement and maximum displacement are spaced far apart.

Most music students are required to use practice rooms that are significantly small in volume. The fundamental resonant modes of a typical practice room at the Stellenbosch University music department, to provide an example, were calculated using equation 1.1 and are provided in table 1.1 for each of the dimensions of the room.

$$f = \frac{c}{\lambda} \tag{1.1}$$

	Length	Height	Width
Dimensions	3.6 m	2.6 m	3.5 m
Fundamental Mode	42.2 Hz	65.4 Hz	48.6 Hz

Table 1.1: Fundamental Resonant Modes for the Respective Dimensions of a Typical Music Practice Room at the Stellenbosch University

Equation 1.1 is a commonly used formula, also presented by Rumsey & McCormick (2009:4). The symbol c, in this case, represents the speed of sound in air, which was chosen as 340 m/s.[2] f represent the frequency and λ the wavelength in meters, which in this case is double the distance between two parallel walls. The fundamental resonant modes that were determined can be problematic for instruments that extend to the low frequency audio spectrum, below around 70 Hz. According to the frequency range of instruments provided by Everest *et al.* (2009:81), there are numerous instruments that contain fundamental notes below this frequency, including: piano, cello, contrabass, tuba, harp and some bass woodwinds. The harmonics of these modes, although typically weaker, can be triggered by other instruments as well, including guitar and numerous woodwind instruments (Everest *et al.*, 2009:81). Musicians' instruments are, therefore, likely to be misrepresented in their frequency response in a small practice room.

1.1.1.2 EARLY REFLECTIONS

Small rooms contain many early reflections, which are reflections occurring within the first 50 ms of the direct sound (Rumsey & McCormick, 2009:26). These reflections have both timbral effects on the sound source and provide spaciousness without changing the apparent location of the source (Loy, 2006:211). Early reflections actually give information about the acoustic space, such as the size and location of the source within the space (Howard & Angus, 2009:281). The intensity level of early reflections are dependent on the surface from which they are reflected as well as the distance travelled before reaching the listener (Howard & Angus, 2009:281).

Early reflections affect the perception of the direct sound differently, depending on the delay time. According to Dickreiter *et al.* (2008:23) reflections will generally increase the loudness level of the direct sound, but reflections that come between 0.8 ms and 20 ms after the direct sound are unpleasant because of the timbral effect that constructive and destructive interferences have on the sound source. In small untreated practice rooms, with dimension similar to those of the Stellenbosch University Music Department, (dimensions were provided in table 1.1), early reflections will, therefore, contribute significantly to the timbre as well as the overall loudness of a sound source. For this reason it is advised that small practice rooms be treated with appropriate absorption panels or diffusers. Osman (2010:3) suggests placing this treatment on all three planes so that no untreated sections of the room face each other. This helps to reduce flutter echoes, which can, according to (Rumsey & McCormick, 2009:26), occur as a "ringing" sound between two parallel walls when an impulse is sounded.

[2] This is an approximate value, since the speed of sound in air is dependent on air temperature.

1.1.1.3 REVERBERATION

Reverberation is defined by Davis & Jones (1989:259) as a high density of reflections occurring within an enclosed space where individual reflections can not be distinguished. Reflections, therefore, arrive at the listener in very quick succession (Howard & Angus, 2009:287). The reverberation time of a room is, according to Everest *et al.* (2009:153), "a measurement of the rate of decay of sound." Larger rooms will naturally contain longer reverberation, as the sound is able to travel for a longer time period. Lewcock *et al.* (2001:76) suggests that rooms with longer reverberation times sacrifice clarity for loudness and sustain.

Reverberation times are typically measured using RT_{60}, which is both the oldest and most well known "room-acoustical quantity" (Ahnert & Tennhardt, 2008:148). It describes the time required for the ambient sound field, after a sound source has been stopped, to decrease by 60 dB in sound intensity level (Davis & Jones, 1989:260). Because reverberation times differ across the frequency spectrum, reverberation times of different rooms are typically compared at "mid-frequencies", with RT_{60} being averaged for measurements at 500 Hz and 1000 Hz (Lewcock *et al.*, 2001:76).

Concert halls traditionally have a reverberation time (RT_{60}) of about 0.8 s to 3.0 s (Ahnert & Tennhardt, 2008:149; Everest *et al.*, 2009:171). RT_{60} does not accurately describe the reverberation characteristic of small rooms, as the sound decay is significantly affected by room modes rather than diffuse reflections (Everest *et al.*, 2009:348). According to Jones (2008:135), true reverberation will only be approached in large rooms.

Still, educational standards provide recommendations for reverberation times in different purposed and sized venues. Osman (2010:3) tabulated these recommendations and a compacted list of the recommendations is provided in table 1.2.

Room	Volume (m^3)	AS2017, 2000	DfES, 2002	BB93, 2003
Teaching or Practice Room	14-30	0.7-0.9	0.3-0.6	< 0.8
Ensemble or Music Studio	38-150	0.7-0.9	0.5-1.0	0.6-1.2
Recital Room	150-400	1.1-1.3	1.0-1.5	1.0-1.5

Table 1.2: Recommended Reverberation Times for Small Rooms as suggested by Osman (2010:3).

1.1.1.4 LOUDNESS LEVELS

Since musicians practice many hours with instruments that are capable of generating high sound pressure levels, the levels that musicians are exposed to in small practice rooms can be of consequence. In a study done by Phillips & Mace

(2008), noise exposures in music practice rooms were measured as a percentage of daily dosage recommended by the National Institute of Occupational Safety and Health (NIOSH). Ten students from each of four instrument groups were selected, namely: brass, string, woodwind and percussion. The results were dangerously high for most of the instrument groups, especially the brass instruments. When seen in context of the many hours that some classical musicians practice per day, all measured instrument groups indicate potential for hearing damage.

In a similar study conducted by O'Brien *et al.* (2013:1746-1754) sound level exposures of professional musicians during solo practice sessions in a small room, having a volume of $54\,\mathrm{m}^3$, were measured. The results once again demonstrated that a large portion of the musicians exceeded daily recommended sound exposures when calculated over a 2.1 hour practice period. Because reflections add to the overall sound pressure levels in practice rooms, acoustic treatment in practice rooms is therefore also beneficial for reducing loudness.

1.1.1.5 ACOUSTIC PREFERENCES FOR MUSICIANS

When considering the acoustics of a music practice room, the preferences of musicians are also an important consideration. Reverberation, or what the musicians subjectively believe to be reverberation, is an important factor for musicians during practice. Christian *&* Gade (2015:233) suggest that musicians prefer to practice with less reverberation than what they enjoy for a concert. Practicing, therefore, likely requires different acoustics to performing.

Blankenship *et al.* (1955:775) tested musicians' subjective response to the acoustics of small practice rooms containing varying number of absorption panels. The practices rooms had a volume of around $12\,\mathrm{m}^3$ and the panels, made up of a wooden frame filled with fiberglass, had dimensions of about 2×1 meters. The results show that the musicians favoured rooms containing either one or two absorption panels. In terms of the reverberation preferences an untreated room was agreed to be too live, with a single panel or two panel room being satisfactory for most participants. Blankenship *et al.* (1955:775) concluded that a reverberation time of around 0.4 s to 0.5 s are, therefore, desired for practice rooms of these dimensions. These suggestions are similar to the reverberation recommendations by the Department for Education and Skills (DfES) provided for small practice rooms in table 1.2.

Lamberty (1980:149) conducted a study that required music students to provide judgement on certain criteria of practice rooms, including the preferred reverberation times, background noise tolerance and the importance of the physical space. When students were asked about their preference for practicing in either a dead room, a live room or a room midway between, it was found that 59 % preferred live rooms (0.9 s), 30 % midway (0.7 s) and 11 % dead rooms (0.5 s). Interestingly, students with a higher degree of application preferred practicing in dead rooms, in comparison to students with a lesser degree of application, who preferred practicing in fairly live rooms. Most students, however, agreed that variable acoustics would be ideal, allowing both dead conditions and then live conditions for a more pleasurable experience. Osman (2010:7) also recommends variable acoustics for musicians, as different instruments, too, require different reverberation times.

1.1.2 THE EFFECT OF ACOUSTICS ON PERFORMANCE

The acoustic environment in which a musician plays can have a significant effect on the performance attributes of the musician (Toole, 2008:30). Ando *&* Cariani (2009:172) mention that: "When performers on the stage play a musical instrument, the concert hall acts as a second instrument". The reverberation character of the space therefore influences the duration of the music performance. Meyer (2009:385) states that the tempo of a performance is an important means of musical expression and that performing at an appropriate tempo is an essential part of the interpretation of a work. He further states that a hall can have tonal effects on the performance and that the tempo of a performance, therefore, needs to be suited to the acoustical conditions of the hall. According to Christian *&* Gade (2015:233), a performer may, for example, play faster and use more legato in a dry space, while in a wet space use more staccato technique and play slower.

Ueno *&* Tachibana (2005:156) investigated, through interview questionnaires, how professional musicians react to concert hall acoustics and their cognition about concert hall acoustics. The results show that the professional musicians adjust their technique to match the acoustics of the concert hall. They, therefore, use the hall in conjunction with their instrument for musical expression. One of the musician's responses to making performance adjustments when performing in a concert hall was as follows:

> I play the instrument to match the sound to the hall by listening to the length of reverberation, timing of the hall response and tonal quality of harmonics.

The effect of reverberation time on performance has also been tested experimentally. In an experiment carried through by Kato *et al.* (2008) recordings of four different instruments were made under several different simulated acoustic environments. Significant differences were found in the length that the musicians held tones in these simulated acoustic environments. Kato *et al.* (2008:8118) found that out of the four instruments, violin, oboe and two flutists, the oboe and violin most clearly demonstrated the effect of shortening tone lengths in acoustic environments containing longer reverberation times.

Ueno *&* Tachibana (2005:158), however, believe that musicians are not consciously aware of the acoustic characteristics of a hall and that their response to a hall can be explained by a theory known as "tacit knowing". This theory explains the acquisition of a skill through repeated behaviour and responses. A musician repeatedly hears or senses the reaction of the hall, which will over time eventually influence the playing technique of the musician.

1.2 DISCUSSION AND PROBLEM STATEMENT

It is clear that the acoustic properties of small rooms are substantially different to those of concert halls. Although the dry acoustic properties of small rooms can be beneficial for certain aspects of practice, they do not well-prepare musicians for performance venues, where the interaction with the acoustics, especially the reverberation time, is an important skill. In addition, the dead acoustic environment of small practice rooms can be demotivating for musicians who spend lengthy periods in this environment. Musicians should, therefore, more frequently be provided with the opportunity to practice in acoustics that resemble those of performance venues.

Venues with these requirements are difficult for solo musicians to gain access to, due to their high demand for orchestra rehearsals and performances; or otherwise: their expensive hiring fee. An alternative solution is therefore required, where practice rooms can be made to better represent the performance venue. Obtaining reverberation characteristics in small rooms that are similar to those of concert halls would be impractical to implement structurally, as musicians would be exposed to excessive sound pressure levels.

1.3 RESEARCH OBJECTIVES

This thesis investigates the effectiveness of using an electro-acoustic system in small practice rooms to simulate the acoustics of a concert hall. This system needs to be affordable and easy to implement, as limited funding and equipment is available for this study as well as for future implementation of such a system. The thesis examines current technologies in electro-acoustics, including headphones and binaural technology. Previous literature and technology that have addressed the issues of practice room acoustics through electro-acoustic means are also evaluated. A system that can be implemented and tested on music students in practice rooms is then to be devised by the author. The effectiveness of the system is determined by the subjective responses of music students by means of interview questions.

1.4 THESIS STATEMENT

By installing an electro-acoustic system in small practice rooms, using headphones with integrated microphones, musicians will be able to practice in a virtual acoustic environment that accurately simulates the acoustics of a concert hall.

Chapter 2

Electro-Acoustic Enhancement Systems

2.1 BACKGROUND

An electro-acoustic enhancement system alters the sound field in a space through the use of microphones, loudspeakers and electronics (Lokki & Hiipakka, 2001:1). Such systems are considerably cheaper than variable acoustics, which rely on large physical structures to produce acoustic alterations (Rumsey & Kok, 2014:449). Electro-acoustic systems allow the acoustics of a venue to change at the press of a button (Rumsey & Kok, 2014:449). Variable acoustics, which require space, are not suitable for acoustic enhancements in small practice rooms and are not able to increase reverberation times significantly.

Electro-acoustic systems are able to increase reverberation times in rooms that do not contain sufficient reverberation times or can alternatively place a listener into a virtual acoustic environment that is entirely different from the room in which the listener is located. In cases where there is a lack of reverberation Lewcock *et al.* (2001:77) state that artificial reverberation can be applied by the distribution of microphones and loudspeakers around the room, including the walls, ceiling and floor. In the case of creating a virtual acoustic environment, Woszczyk (2011:381) states that reflections, reverberation and other acoustic properties may be artificially reproduced to immerse a listener in a simulated room that "coexists with the actual room".

One of two methods can be used, according to Rumsey & Kok (2014:449), to artificially alter the reverberation times and reflections of a space: regeneration techniques or artificial reverberation. Regeneration involves feeding reverberation, that is picked up by microphones in the reverberant field, back into the acoustic environment via loudspeakers. Woszczyk (2011:381) refers to this type of system as a *non-in-line* system. Systems of this type are only suitable for application in large venues or concert halls that contain a diffuse reverberant field.

Electro-acoustic systems using artificial reverberators have microphones placed close to the source. This method is described by Woszczyk (2011:381) as an *in-line system* and requires a large number of loudspeakers to disperse the artificial acoustics into the natural environment.

Acoustic enhancements through artificial reverberation typically make use of an artificial reverberator, similar to those used in studio effects, which generate digital early reflections and reverberation (Rumsey & Kok, 2014:450). Other methods of creating artificial reverberation exist, and date back as far as the 1920s in the form of either reverberation chambers or electromechanical reverberation devices such as springs and plates (Valimaki *et al.*, 2012:1421-1422). According to Poletti (2011:12), electro-acoustic enhancement systems were already being tested in the 1960s, using magnetic tape and acoustic tube delays.

Modern systems, however, rely principally on digital technology for the production of artificial reverberation. Valimaki *et al.* (2012:1422) and Case (2011:202) suggest that different methods of realising these digital implementations exist, generally falling into one of the following categories:

Delay Networks In this method the input signal is delayed numerous times and sent through various digital filters including comb filters[1] and all pass filters.[2] The density of these delays and the use of filters can be varied to bring about a desired response. Parameters for the reverberation characteristics are usually provided to the user, allowing the user to make desired changes to the reverberation response.

Convolution In this method the input signal (in its digital form) is imprinted with the acoustics of a different, either real or virtual, acoustic environment. In the case of a real space, an impulse response is required from the acoustic environment using a certain measurement technique. This is most accurately acquired by means of a sinusoid sweep, which involves playing a "logarithmic sweep of constant amplitude" into a room with loudspeakers and recording it (Valimaki *et al.*, 2012:1434). A time-reversed version of the sweep is combined with the recorded sweep, which leaves the impulse response. The impulse response contains the information of the reflections that define the unique character of the acoustic space that was measured (Case, 2011:202). Convolution reverbs are, according to Valimaki *et al.* (2012:1429), used especially for virtual reality systems.

[1] A filter that superimposes a signal with a delayed signal of itself. This leads to constructive and destructive interference. The frequency response graph of these superimposed signals resembles that of a comb, as a consequence of the consistently spaced notches (caused by the destructive interference).

[2] A filter that changes the phase of a signal relative to frequency while maintaining a flat frequency response.

Valimaki *et al.* (2012:1422) state that artificial reverberation techniques can, "in extreme cases", be used to convert a small room into having the reverberation qualities of a concert hall and further suggests that this is useful for rehearsal purposes. Ahnert & Tennhardt (2008:189) warn, however, that when long reverberation times are produced in small rooms through electronic enhancement systems, listening experience may be negatively affected as the perceived acoustics deviate significantly from the visual impression of the room. Rumsey & Kok (2014:452) also state that acoustic consultants generally stay away from creating acoustic environments that are significantly different from the actual environment of the room, to not impair the congruity between the visual and aural qualities of the space.

A diffuse reverberant field in a natural setting involves an infinite number of reflections that originate from all directions. An electro-acoustic system that intends to artificially recreate this, is restricted when using a limited number of loudspeakers. Woszczyk *et al.* (2013:2) state that with limited loudspeakers, which are widely spaced, audible 'holes' may exist in the reverberation field as the density of the reverberation is insufficient. A large number of loudspeakers is therefore required to increase the reverberation density and to prevent the localization of speakers (Poletti, 2011:15).

Acoustic feedback is another problem in electro-acoustic enhancement systems, especially in small rooms where loudspeakers and microphones are placed in close proximity (Lokki & Hiipakka, 2001:1). *In-line systems*, with microphone placed close to the source are less at risk of feedback, but as Poletti (2011) states, "at sufficiently high loop gains, any active system can become unstable." Lokki & Hiipakka (2001:1) define the term *gain before instability* (GBI) as the maximum gain before the system becomes unstable as a consequence of feedback. Simple considerations in terms of microphone placement and type, as well as loudspeaker type, and increasing the amount of independent channels can decrease the GBI (Griesinger, 1990:2). Independent channels can be obtained by limiting the different channels to certain frequency bandwidths (Rumsey & Kok, 2014:450)

In many cases more complicated feedback reduction techniques are required to allow for a sufficient GBI. Rumsey & Kok (2014:451) state that some decorrelation techniques, especially those using time-varying algorithms, can increase the GBI of a system by more than 10 dB. Time-varying algorithms work by means of changing the signal continuously so as to impede any build-up of energy at specific frequencies. Time variations can be applied to the channels by means of changes in amplitude, delay time, phase or frequency of the signal (Lokki & Hiipakka, 2001:1).

2.2 COMMERCIAL SYSTEMS

There are a diverse range of commercial systems available, which utilize different techniques and technology. Those that use only regeneration techniques will not be focused on greatly, as they are not suitable for small venues that, as mentioned by Jones (2008:135), do not contain a true reverberant field. The commercial systems differ largely in their use of feedback reduction techniques. Some of the most commonly used systems will be looked at below. This will provide an understanding of the potential of such systems.

2.2.1 CARMEN

This system is of a regenerative type that is compiled of a number of "cells", each consisting of a microphone, an electronic filter, a power amplifier and a loudspeaker (Hardiman, 2009:96). The cells are placed around the venue to enclose the audience with virtual walls, that can be extended by means of delays. Artificial reflections are therefore produced by each cell and the overall characteristic of these reflections are controlled by a computer (Poletti, 2011:14). The microphones and loudspeakers are placed approximately one meter apart, using the help of directive microphones as well as a form of echo cancellation to reduce feedback (Ahnert & Tennhardt, 2008:196). Typically a number of 16-40 cells have been used for previous installations, with some installations allowing acoustic presets for different music requirements (Hardiman, 2009:96). Ahnert & Tennhardt (2008:196) mention that although Carmen has been installed in numerous concert halls the system seems to be especially effective in theaters.

2.2.2 MEYER SOUND CONSTELLATION

The Meyer Sound Constellation has been installed in numerous large venues, including the Zellerbach hall in California, seating 2014 people (Hardiman, 2009:93). An average of around 40 channels are used in Constellation installations, with the number of microphones and loudspeakers being similar (Ahnert & Tennhardt, 2008:195). Microphones are placed both close to the stage as well as in the audience area. The closely placed microphones are signaled through what Poletti (2011:14) refers to as an "early reflection generator", or according to Ahnert & Tennhardt (2008:196) a "unitary delay system", which uses appropriate delays to simulate early reflections.

The microphones placed further from the stage are routed to a room processor responsible for simulating late reverberation (Ahnert & Tennhardt, 2008:195). Constellation therefore uses a combination of an in-line system, with closely placed microphones, as well as a non-in-line system, where microphones are distributed in the audience area (Hardiman, 2009:93). The loudspeakers surround the audience, with the loudspeakers closer to the stage reproducing the early reflections; and the loudspeakers further into the hall reproducing the late reverberation energy.

2.2.3 ACOUSTIC CONTROL SYSTEM

Acoustic Control System (ACS) uses 12-36 microphones that are placed close to the stage. ACS is not designed for maximum stability, according to Griesinger

(1990:4), as it does not use any time-variation methods to reduce feedback. The large number of loudspeakers, however, does increase the stability of the system somewhat, as each channel only reproduces a small part of the total reproduced signal. Each channel contains a digital processor that, when combined with the other channels, simulates the reflections of a desired acoustic environment through appropriate delays (Ahnert & Tennhardt, 2008:194). The loudspeakers are typically placed in the audience area of the hall, away from the microphones, where the loudspeakers are carefully set to produce both early reflections as well as late reverberation. Loudspeakers can also be set up around the stage area to enhance the acoustics for the musicians, who are required to hear one another well.

2.2.4 SYSTEM FOR IMPROVED ACOUSTIC PERFORMANCE

The System for Improved Acoustic Performance (SIAP) is a system designed in the Netherlands. It uses a small number of microphones in the stage area and a larger number of loudspeakers in the audience to create a diffuse acoustic field. According to Hardiman (2009:95) SIAP use a multichannel approach to prevent feedback (32 or 64 loudspeakers) and Poletti (2011:13) suggests that they occasionally make use of time-varying techniques. The SIAP system uses convolution processing, with each channel having a slightly different impulse response so that the loudspeakers reproduce unique signals (Ahnert & Tennhardt, 2008:194). The system is designed to be natural and aims to maintain a realistic balance between the aural and visual perceptions of the environment (Poletti, 2011:13).

2.2.5 LEXICON ACOUSTIC REINFORCEMENT AND ENHANCEMENT SYSTEM

The Lexicon Acoustic Reinforcement and Enhancement System (LARES) system, already developed in the 1980s, is according to Hardiman (2009:89), the most utilized acoustical enhancement system. LARES uses a small number of microphones, placed close to the stage, and a large number of loudspeakers in the walls and ceiling in order to achieve well distributed sound energy (Poletti, 2011:13). The microphone signals are routed to numerous independent time-varying reverberation devices, which each feed the loudspeakers (Ahnert & Tennhardt, 2008:194). The LARES system's success is, according to Hardiman (2009:89), partly due its effective time varying feedback reduction technique. LARES has been installed in numerous venues in Europe and the United States, both indoors and outdoors.

2.2.6 WENGER CORPORATION

The Wenger Corporation[3] uses the technology of LARES to produce virtual acoustic environments for rehearsal and practice rooms. They have developed

[3] https://www.wengercorp.com/

the SoundLok Sound-Isolation Rooms, which as the name suggests, are pre-configured isolated rooms that have been specifically constructed for musicians' practice. These rooms are also designed to include their Virtual Acoustic Environment (VAE) technology, which allow the musicians to practice in any of ten virtual acoustic environments, selectable by means of a control panel. The environments include a practice room, large recital hall and a cathedral.

The Wenger corporation also offers the possibility of installing their Studio VAE System in ordinary practice rooms. This is according to the company, ideal for installation in small teaching studios or teaching offices for private lessons (Wenger, 2013). This setup uses two microphones situated on either side of the room, as well as four loudspeaker boxes, each containing 2 speakers, placed in each corner of the room. The author enquired about the costs to implement the system in a practice room of the Stellenbosch Music Department and was quoted around 6200 US Dollars.

2.3 ELECTRO-ACOUSTIC SYSTEMS IN PRACTICE

This section looks into implementations of electro-acoustic systems in smaller venues and the subjective responses to these systems. Most commercial systems, discussed, have been designed for the purpose of larger venues, where installations involve a large number of microphones and loudspeakers. The implementation of these systems in smaller venues, such as practice rooms, therefore, require modification. The Wenger Corporation's systems, which are intended for small practice rooms, are expensive and it is unclear whether these systems are indeed effective at simulating the acoustics of concert halls.

2.3.1 VIRTUAL HAYDN PROJECT

The technique of creating a virtual acoustic environment for musicians was used for the *Virtual Haydn Project*, a recording project, where the keyboard sonatas of the well-known composer, Joseph Haydn (1793-1809), were performed in nine different virtual acoustic environments (Woszczyk *et al.*, 2009:4). These virtual rooms were reproductions of real rooms, that Haydn composed for or performed in, including his own study in Eisenstadt, Austria.

The impulse responses of the different venues were obtained by exciting the space with a "logarithmic swept sine wave", reproduced by means of 10-14 loudspeakers distributed around the stage area of the venue. Three microphone arrays, of 8 microphones each, were used to record the signal from three different distances respectively. The microphone signals were then sent through a convolution processor, capable of obtaining the necessary data to reconstruct the acoustic properties of the space.

The rehearsal and recording of the performer took place in an acoustically dead environment with a reverberation time of around 0.3 seconds. 24 loudspeakers surrounded the musician in the shape of a half sphere, to reproduce the response of the different acoustic environments (Woszczyk *et al.*, 2009:3).

The virtual acoustics were ultimately, however, reproduced over headphones for the performer.

The subjective response of the performer was positive. The performer suggested that both the loudspeaker and headphones portrayed the acoustics realistically and that the acoustics influenced his performance: "Perception of a room, through headphones or through loudspeakers, became an essential factor in my recorded performances" (Woszczyk *et al.*, 2009:6). The headphones, however, were favourable to the musician, as they allowed him to be more critical and focused.

2.3.2 A VIRTUAL WALL

Lokki *et al.* (2000) developed a prototype of an electro-acoustic system to address the acoustic problems of orchestral rehearsal rooms. The prototype hoped to allow the orchestra to better interpret the performance by simulating concert hall acoustics (Lokki *et al.*, 2000:1). The system was tested on orchestral musicians in a music rehearsal space, with dimensions $20\,\mathrm{m} \times 15\,\mathrm{m} \times 7\,\mathrm{m}$. A total of 28 loudspeakers were placed against a single wall, which had simultaneously been treated with curtains so as to be acoustically dead. The speakers were made to reproduce late reverberation, with the early reflections coming from the three walls of the natural rehearsal space. This setup, therefore, aimed to replicate the stage area of a concert hall, where late reverberation returns to the musicians from the audience area and early reflections come from the surrounding walls. The subjective evaluations of the musicians were positive and the acoustics in the room were claimed to have sounded like those of a concert hall (Lokki *&* Hiipakka, 2001:5).

2.3.3 VIRTUAL ACOUSTICS IN PRACTICE ROOMS

Pätynen (2007) addressed the same problem as that of this thesis in his Master's thesis titled *Virtual Acoustics in Practice Rooms*. He implemented a virtual acoustic system in three different sized rehearsal rooms, including a smaller sized practice room. In addition to increasing reverberation times, Pätynen (2007:i) aimed to maintain the same sound pressure level (SPL) in the practice rooms as that encountered without the electro-acoustic system.

A time-varying reverberation algorithm was to used to reproduce mainly the late reverberation energy (Pätynen, 2007:41). Although a significant amount of absorption was installed in the practice rooms to control the interaction of the reproduced reverberation and the room, Pätynen (2007:35) relied on the room itself for early reflections. He mentions that, when the rehearsal room is of a similar size to the stage of a concert hall it is best to recreate only the late reverberation electro-acoustically.

The electro-acoustic system implemented in each of the rooms was similar, although the number of speakers and microphones were varied according to the size of the room. The biggest room was a performance venue and will, therefore, not be addressed. The electro-acoustic system of the smaller two rooms, with volumes of around 40 m^3 and 103 m^3 respectively, both made use of two microphones placed at a height of 1.75 m and spaced apart. The smaller room used four full-range speakers, one in each corner, and the larger one six. The walls of each of the two rooms were treated with curtains reaching just over 2 m in height. The acoustic treatment significantly reduced the SPL readings in the rooms. The SPL readings with the system activated were still less than the original SPL readings (Pätynen, 2007:52-71).

During the experiment, two different reverberation times were reproduced to the participants, each at a soft and loud setting. A short reverberation time of 1.5 s and a longer setting of 2.4 s were used for the middle sized room, with the smallest room's long reverberation setting being adapted to 2.0 s. For the smallest venue subjective impressions of reverberation enhancements of 1.5 seconds were positive. A reverberation enhancement setting of 2.4 s became disturbing for listeners, with visual and aural incongruence being an issue (Pätynen, 2007:65-66). For the middle sizes room, impression were both negative and positive. This study, however, did not have a large number of participants, with only ten people participating for the middle sized room and two for the smallest room. Pätynen (2007:81) mentions that additional research is required for electro-acoustic installations in small rooms.

2.4 CONCLUSION

This chapter found that electro-acoustic systems have the potential to be implemented for acoustic enhancement solutions of practice room acoustics, by loudspeakers. The accuracy of these systems have, however, not been well evaluated. The studies, which were described in section 2.3, did not receive sufficient subjective responses for a virtual acoustic simulation in a small practice room. The few responses that they did obtain were, however, positive. This, therefore, supports the author's belief that electro-acoustic systems can potentially benefit musicians' practice.

Lokki *et al.* (2000) as well as Pätynen (2007), in their studies, both relied on the early reflection of the natural acoustic space. For implementation of an accurate virtual acoustic environment in a small practice room environment, the author believes, however, that the early reflections should also be simulated, since the early reflections of a practice room are significantly different to those of a concert hall.

Through literature it was found that an accurate simulation of concert hall acoustics requires a large number of loudspeakers, which is impractical for implementation in small practice rooms. The current systems also require musicians to stand in a given position in their practice room for the most accurate simulation. This is not practical for multi-usage practice rooms where different musicians will be situated in different parts of the room.[4] For a more accurate simulation of concert hall acoustics, the author, therefore, believes that an alternative system should be considered and tested. This system should also be cheaper to implement than existing systems.

[4] Most practice rooms contain a piano placed against a side wall. Pianists will therefore be located more to the side of the room, while other musicians will in all probability practice more centrally or towards the opposite side of the room.

CHAPTER 3

Headphones and Augmented Reality Audio

3.1 BACKGROUND

Headphones are an obvious alternative to loudspeakers for the reproduction of a virtual acoustic environment. According to Dickreiter *et al.* (2008:177), high quality headphones can be constructed with little effort, implying that good quality audio can be produced for less cost than loudspeakers. In addition to this, headphones have the advantage of being more mobile than loudspeakers. The quality of the reproduced audio is not compromised by the position or movement of the listener. Despite that, headphones are known to reproduce inaccurate stereo images and create an impression of the sound originating from inside the head (Self *et al.*, 2009:732). This chapter will look at the uses and potential uses of headphones for monitoring and acoustic simulation. Headphone types will also be discussed, as this largely influences the quality of the reproduced audio. Current headphone technology, including new developments in binaural technology as well as augmented reality audio will be discussed. This chapter aims to help the author determine how best to construct a headphone system that can be used to accurately simulate a performance environment for musicians.

3.1.1 MONITORING TECHNIQUES OF THE RECORDING STUDIO

Headphone monitoring is standard in many recording studios and is becoming more and more common in live band performances. Since most recordings today use overdubbing techniques, the musicians are required to perform along with pre-recorded instruments, which are usually reproduced over headphones (Huber, 2009:87). Alternatively, in live amplified performances, musicians use headphones to isolate themselves acoustically and allow a controlled monitor mix. In order for the musicians to perform accurately the reproduced monitor mix needs to be of sufficient quality. Many professional recording engineers stress the importance of this monitor mix or 'cue' mix for the performance of the musician; and have, through experience, determined successful techniques to best present musicians with a simulated environment that suit the musicians' needs (Clark, 2011:51-60). Similarly, in live performances, musicians need to hear clearly to optimize their performance.

The objective in the monitoring techniques of the studio and in live sound is different to that of this thesis. The studio, especially, does not typically intend to accurately reproduce an acoustic environment, whereas the success of the system designed for this thesis is dependent on its accuracy in reproducing a concert hall environment. Still, very much the same technology, as that which is used in both the studio and live setup, will be used for the system of this thesis. For this reason, the uses of some of this technology, in the studio, will be described.

3.1.1.1 ARTIFICIAL REVERBERATION

Artificial reverberation is commonly used in the studio for both the recording process as well as the mixing process. Case (2011:203-222) describes several uses of reverberation effects in the mixing process, which can be applied to live monitoring as well, if the processing power of the system is sufficient. Reverberation can be used to compensate for the close placement of microphones in both live sound and the studio. Alternatively, reverberation is used for creative effects. To compensate for dry recordings convolution reverberation plugins allow the recreation of real acoustic spaces (Case, 2011:202). Clarke *et al.* (1999:4.8) suggest that reverberation should be used for both the end-listener and the musician. He further mentions that artificial early reflections can help musicians pitch better.

3.1.1.2 MICROPHONE TECHNIQUES

The microphones in both live and studio setups are typically placed in a configuration which does not consider the musician's perspective, but rather takes on the perspective of an audience member. When creating a monitor mix, the same microphone signals used for the recording are sent to the musician. This signal is not necessarily an accurate representation of the musician's instrument or voice. If the microphone placement is a significant distance away from the source an acoustic delay can occur, which can degrade the monitoring quality (refer to section 3.3.3 for Latency).[1] In order to create an accurate monitoring experience for the musician the microphones should, therefore, be placed close to the ears of the musician. A binaural microphone technique can potentially create a more accurate monitoring experience than conventional studio microphone techniques and will therefore be investigated.

---

1 Since the speed of sound is around 340 $\mathrm{ms^{-1}}$, a delay of 1 ms is introduced for around every 0.34 m. In the imperial measuring system a foot accounts for about a 1 ms delay.

3.1.2 BINAURAL TECHNOLOGY

Binaural spatial sound impressions are achieved by providing the listener with a similar signal as that which would have been present at the ears of the listener in the source environment. Binaural recordings therefore involve placing microphones inside or as close to the listener's or a dummy head's ears and reproducing this signal over headphones (Algazi *&* Duda, 2011:34). Many of the cues needed for accurate spatial perception are thus contained in binaural reproduction (Rumsey, 2001:65).

Dickreiter *et al.* (2008) suggest that with true binaural reproductions the listener is able to perceive the following attributes of direction and room impressions:

- Horizontal directional cues for all directions around the head
- Elevation
- Distance
- Depth
- Accurate Room Impressions
- Enveloping Sound

The spectral cues that humans use for localization of a sound source, known as head-related transfer functions (HRTF), are according to Schonstein *et al.* (2008:8440) a result of the structure of our outer ear, head and torso. Our body filters sound differently depending from which direction it originates. HRTF can be accurately measured by placing small microphones, once again, close to the eardrums of either a dummy head or human. These measurements can be used to apply appropriate filters to a dry sound signal, theoretically allowing placement of a virtual source anywhere around the listener's head (Zhang, 2008:58). Binaural impressions are, therefore, possible for both recordings and virtual sources.

Augmented reality audio (ARA) systems have proven quite successful in reproducing the real sound environment, also referred to as the pseudoacoustic environment, while simultaneously superimposing a virtual auditory environment (Härmä *et al.*, 2004:635). According to Rämö *&* Välimäki (2012:1) ARA can be implemented using either bone conduction headphones or a headset that contains headphones with integrated binaural microphones, where the microphones feed the headphones in real-time to reconstruct the natural surrounding space. ARA systems could therefore be used to accurately represent both the natural sound of the a musician's instrument as well as the simulated acoustics of a concert hall.

(a) Closed-back Circumaural Headphones

(b) Open-back Circumaural Headphones

(c) Supra-aural Headphones

(d) Intra-aural Headphones

Figure 3.1: Examples of Different Headphone Types. Courtesy Sennheiser

3.2 HEADPHONE TYPES

The type of headphones used will both impact the effectiveness of binaural localization as well as the amount of attenuation of the direct sound. Headphones can primarily be divided into the circumaural, supra-aural and intra-aural types. Bone-conduction headsets are also discussed, as well as active-noise canceling technology.

3.2.1 CIRCUMAURAL AND SUPRA-AURAL HEADPHONES

The circumaural design, or pressure type, has the ear enclosed by the earpiece. This design, if well sealed, allows the ear to be pressure coupled with the diaphragm at lower frequencies, giving a more linear response (Self *et al.*, 2009:742). The supra-aural design, or velocity type, involves the cushion of the earpiece resting on the surface of the ear. Supra-aural types have limited potential for attenuation of outside noise, with only about 10 dB of isolation at around 5 kHz. Circumaural types have potential for high isolation, with about 5 dB of attenuation at 100 Hz and 40 dB at 10 kHz (Dickreiter *et al.*, 2008:178).

3.2.1.1 OPEN-BACK AND CLOSED-BACK

Both types can be further separated into open-back or closed-back types, depending on whether the rear surface of the diaphragm is enclosed or not (Borwick, 1999:2.64). This further determines the potential sound isolation, with open-back headphones producing "virtually no isolation from the outside noise" (Newell, 2008:613).[2]

Open-back headphones are generally more favourable, as musicians "feel less cut off from their environment" (Newell, 2008:613). According to a study conducted by Schonstein *et al.* (2008) open-back headphones also faired well in localization tests compared to closed-back and bone conduction headphones (refer to section 3.2.4). This suggests that open-back headphones are better suited for accurately reconstructing an acoustic environment. Closed-back headphones, however, are useful for reducing overall sound exposure and help prevent closely located microphones from picking up the reproduced sounds (Newell, 2009:623).

3.2.2 INTRA-AURAL HEADPHONES

Another type of headphones are intra-aural headphones, or earphones, where the sound is signalled directly to the ear canal (Self *et al.*, 2009:743). Earphones are practical for their small size but should be custom made when used professionally (Dickreiter *et al.*, 2008:178). Attenuation properties of earphones differ substantially, depending on the type of design. Two models' attenuation properties were measured by Härmä *et al.* (2004:625). One model, containing earplugs, has a measured attenuation of 10 dB–30 dB over the frequency range of 100 Hz to 10 kHz. The other model, with its earpiece placed at the entrance of the ear, only measured attenuation of 1 dB–5 dB over this frequency range.

3.2.3 ACTIVE NOISE-CANCELING HEADPHONES

Active noise-cancelling (ANC) headphones intend to increase the isolation properties of headphones by means of cancelling the ambient noise that enters the ear of the listener. Microphones placed inside or outside the headphones pick up the ambient noise and a loudspeaker in the earcup then reproduces a signal, termed an antinoise signal by Valimaki *et al.* (2015:93), that is of

---

[2] In certain sources the terms 'open' or 'closed' headphones refer to 'open-back' or 'closed-back' headphones. In other sources 'open' and 'closed' headphones refer to supra-aural and circumaural headphones respectively. The terms 'open' and 'closed' headphones will therefore be avoided in this work to prevent any ambiguity

opposite polarity to the noise leaked into the headphones, in turn cancelling the leaked sound.

ANC equipped headphones are more successful at removing low frequencies than conventional headphone types using passive isolation techniques (Kuo *et al.*, 2006:331). ANC technology has been implemented in commercial headphones of circumaural, supra-aural and intra-aural type. Bose produce a circumaural ANC design, the Quietcomfort series, and have also recently introduced their intra-aural QuietControl 30. The QuietControl 30 contain controllable noise-cancellation. The product contains ambient microphones that allows the users to monitor their surroundings when required. Similarly, the Plantronics Backbeat Pro+, which is of a wireless circumaural design, allows the user to control the amount of noise-cancellation. This can be controlled from an application on a mobile device.

3.2.4 BONE-CONDUCTION HEADSETS

Bone-conduction devices conduct sound to the inner ear via the bones of the skull. Lindeman *et al.* (2007:173) suggests that recent developments in this technology have made it possible to use these devices at a "consumer-grade" level. Several commercial products exist, including products from Aftershokz[3], shown in figure 3.2. According to Walker & Stanley (2005:218), most designs have the transducer attached to the mastoid: "the raised portion of the temporal bone located behind the ear". Other designs, however, have the transducer located at the front of the ear on the cheekbone, including the Aftershokz products (Lindeman *et al.*, 2007:173).

Figure 3.2: Aftershokz Trekz Titanium wireless bone-conduction headset. Courtesy AfterShokz

Bone-conduction devices' main advantage over typical headphones is that they do not impair the perception of the natural surrounding environment, as the ear remains open (Mcbride *et al.*, 2010:1). This makes them potentially

[3] https://aftershokz.com/pages/technology

effective for ARA applications. Kondo *et al.* (2013) concluded in their study, which tested the effectiveness of using a bone-conduction headset to reproduce speech signals in the presence of noise, that bone-conduction headsets are applicable for ARA applications (Kondo *et al.*, 2013:6).

According to Walker *et al.* (2005:1615), in the field of psychoacoustics, little research has been done on sound conducted through the skull in comparison to air-conducted sound. Separation between left and right signals for a stereo bone-conduction headset is not well understood. It was originally believed that stereo separation between left and right signals are not at all possible, as the vibrations on each side of the head reach the opposite ear with nearly the same intensity. Studies by both Walker *et al.* (2005) as well as MacDonald *et al.* (2006) show, however, that spatial separation is achievable to quite a high degree of accuracy. Nevertheless, Albrecht *et al.* (2011:7-8) mention in their study of an ARA system, that bone-conduction "headphones" are typically of inferior quality to standard headphones.

Owing to the fact that the frequency response of bone-conduction headsets are not readily available and the accuracy of bone-conduction headsets for complex binaural reproductions is not well known, the author believes that standard headphones are better suited for the accurate simulation of a concert hall.

3.2.5 THE OCCLUSION EFFECT

Sound can also be conducted to the ear drum via the jaw and skull, rather than through the open ear. This sound affects the perception of one's own voice or instrument excited by the mouth (woodwind and brass instruments). The *occlusion effect* is a phenomena where the sound pressure at the eardrum increases as a result of the ear canal being blocked, as is the case with conventional earplugs (Niquette, 2006:55). The sound vibrations that reach the eardrum via the bones have less means of escaping when the ears are occluded and will therefore be trapped in the air between the end of the earplug and the eardrum. According to Ross (2004:3-5) this can lead to an increase in sound pressure in the low frequencies (about 500 Hz and lower) of up to 20 dB. A deeply sealed earplug is necessary to prevent the occlusion effect from occurring in intra-aural headphone types (Niquette, 2006:55). Circumaural headphone types are also prone to the occlusion effect, as is demonstrated in the results of the tests conducted by Vorlaender (2000:2087). The open-back headphones are, however, least prone to the occlusion effect and therefore, once again seem the better choice for the system of this thesis, which requires an accurate frequency response.

3.3 AN INTEGRATED HEADSET

The integration of microphones on headphones for the purpose of monitoring the ambient environment is not new technology. Mueller & Karau (2002), for instance, attached high-quality microphones to headphones in a binaural manner to produce "augmented audio". The microphones were connected to a computer and routed to the headphones in realtime, allowing alterations of the source. Mueller & Karau (2002) suggest several applications of this technology, albeit not for the simulation of acoustic environments. Sigismondi (2008:1425) describes a recent technology, called *active ambient earphones*, which are intra-aural headphones containing small microphones that allow a musician to independently mix in the surrounding ambience while monitoring.

3.3.1 COMMERCIAL PRODUCTS

Several commercial products exist that integrate binaural microphones into headphones. One example is the Roland CS-10EM[4], which combines intra-aural headphones with binaural microphones and allows real-time monitoring. These are wired headphones that provide separate 3.5 mm stereo mini connectors for the headphones and microphones respectively. The microphone signal can be manipulated by the user and sent back through the headphones, making it suitable for virtual acoustic applications.

The latest technologies in intra-aural headphones, include products by Doppler Labs and Apple. Doppler Labs have developed the Here One[5], shown in Figure 3.3. This product contains wireless earbuds with multiple integrated microphones, which are used for both ANC functionality as well as binaural monitoring. A software application, which can be installed on a mobile device, allows the user to manipulate surrounding sounds by use of level adjustments, equalization and reverberation effects. This technology can, therefore, potentially be used by musicians to play in simulated concert hall acoustics. Doppler labs does not, however, market the product for this application.

Apple's AirPods, like the Here One, contain multiple microphones in each earpiece, although the apple headphones do not allow the user as much control as the Here One in terms of manipulation of the surrounding environment. The microphones are not accessible by the user, but are controlled fully by Apple's IOS operating system and are currently only used for ANC as well as calling functionality. The technology is, however, very much the same as the Here One, which shows that the necessary technology for a system which simulates a concert hall for musicians is readily available. This suggests that future implementation of a system using headphones with integrated microphones in order to simulate concert hall acoustics will not be excessively expensive.

Although Kondo *et al.* (2013) compared speech intelligibility of a reproduced speech signal represented by a Roland CS-10EM and bone conduction headphones in the presence of noise, the use of this technology for creating virtual acoustics for musicians has not been well investigated. Integrated microphones on headsets in ARA are primarily used for the purpose of reconstructing the real

4 http://www.rolandus.com/products/cs-10em/
5 https://www.hereplus.me/

Figure 3.3: Doppler Labs: Here One. Courtesy Doppler Labs

environment. This study will, however, use the microphones on a headphone microphones prototype to simultaneously trigger a convolution reverberation plugin, which will provide the listener with a virtual concert hall environment.

3.3.2 REPRESENTING THE NATURAL ENVIRONMENT

One of the most important considerations with a system using headphones, is that the musicians are able to accurately hear their natural instruments. As Newell (2008:259) proposes: "many musicians play off their own tone". This suggests that small changes in the perception of their instruments can make musicians adapt their playing technique. The designed system should, therefore, accurately reconstruct the natural response of the musicians' instruments.

According to Härmä *et al.* (2004:618) integrated microphones whose signals are routed to the headphones expose a user to a pseudo-acoustic representation of the real environment. This modified real acoustic environment, or pseudo-acoustic environment, should be as similar to the real acoustic environment as possible.

If a nearly identical representation of the real environment is to be attained, the isolation properties of the headphones need to be considered, since the sound leaking through the headphones will interact with the reproduced sound at the listener's ears. Ranjan & Gan (2015:1991-1992) measured the effect of four different headphone types on direct sound emanating from a source at different locations around a dummy head. The results of these measurements show the significant influence that headphones have on the spectral cues of the direct sound, especially above 1.5 kHz. External microphones should, therefore, be used to compensate for the attenuation properties of the headphones.

Ranjan & Gan (2015:1991) recommend the signal from the external microphones be modified with appropriate equalization filters. Valimaki *et al.* (2015:96) state that high-pass filters are typically applied to the signal, as low frequencies are attenuated less by the headphones than high frequencies.

Bone conduction headphones do not require any modification, as they leave the direct sound relatively unmodified, and according to Valimaki *et al.* (2015:96) open-back headphones, which leave the direct sound relatively unmodified, also do not require any filters. According to the graphed results by Ranjan & Gan (2015:1990), open-back circumaural headphones do, however, attenuate the direct sound by up to approximately 15 dB at frequencies above around 3 kHz. Filters will, consequently, also need to be considered for open-back headphones. Since digital filters allow for more accurately controllable parameters, the prototype used in the study of this thesis will prefer to use a digital system rather than an analogue one.

3.3.3 LATENCY

All digital audio systems introduce some delay, known as latency. It is important to consider acceptable latency values for an integrated headphone and microphones system as it can lead to quality degradation.

The process of converting an analogue electrical signal to a digital signal, via an Analogue to Digital Converter (ADC), and back again via a Digital to Analogue Converter (DAC), contribute to a short, inevitable latency. According to Inglis (2007:1) this delay will typically be considerably less than 5 ms. The delay caused by the ADC and DAC will be the minimal latency that a digital system can have, with further delay being introduced by digital filters as well as the buffer of the computer (Inglis, 2007:1).

When a musician is presented with both a direct signal, which leaks through the headphones, as well as a delayed signal, the latency can be perceived as either a comb filtering effect or, in extreme cases, an echo (Lester & Boley, 2007:1). For this reason Rämö & Välimäki (2012:9) suggest that latency values in an ARA system should be very small.

Rämö & Välimäki (2012) developed an ARA system using a digital signal processor (DSP) with a total latency of only 1.4 ms. Still, this delay has the potential to cause comb-filtering in the higher frequencies, especially when the leaked and reproduced signals are of similar levels. The headphones used for this experiment, however, had high attenuation properties at these frequencies, making the comb-filtering less significant. Valimaki *et al.* (2015:96) state for this reason that: “a colourless hear-though system is easiest to implement for headphones that attenuate outside sounds well”.

Lester & Boley (2007) tested the subjective response of musicians to different latency values when monitoring their instruments over headphones, with reference to an analogue system containing no latency. The experiment found that for musicians wearing intra-aural headphones, delays of less than 3 ms already seemed to produce comb-filtering artifacts for some musicians. The results, however, depended significantly on the type of instrument played, with keyboardists still feeling content to play with 35 ms latency.

Rämö & Välimäki (2012:11-12) furthermore conducted a listening test (the results are shown in figure 3.4), to determine at what condition a participant could determine a quality difference in both a speech and music sample, as a result of comb-filtering. The participants were repeatedly asked whether the sound quality of a sample, a reference sample with a superimposed delay of various lengths and amplitude, were as good as the pure reference sample played prior to the modified sample. A high "detection rate" implied that participants could not distinguish the quality of the two samples. The results demonstrate that the attenuation of the signal has a larger influence on the quality of the sample, than the delay value. For a delay of only 1 ms, attenuated by 6 dB, all of the participants registered a quality difference.

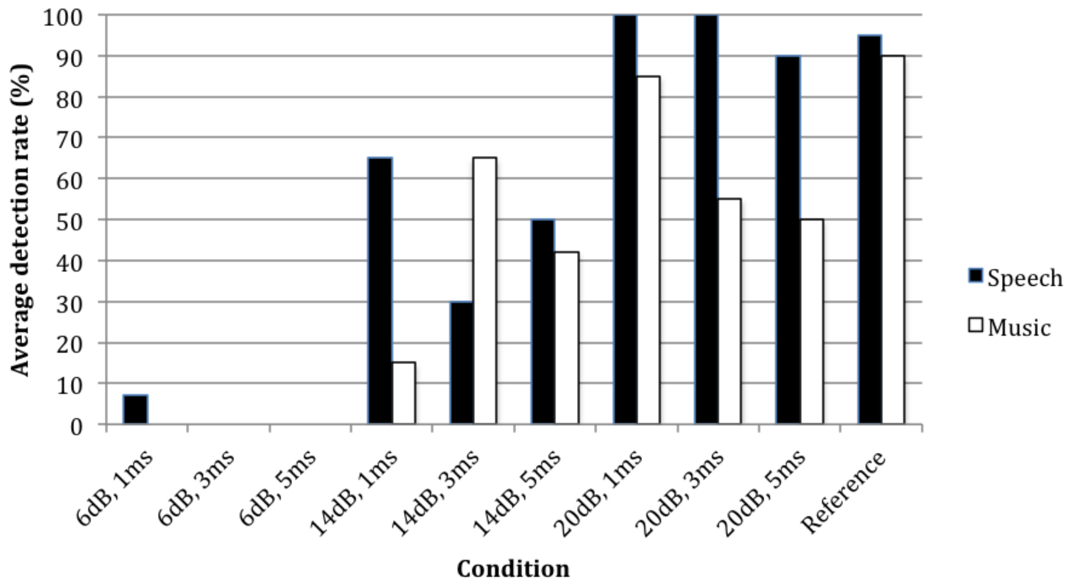

Figure 3.4: Results of the Listening Test Conducted by Rämö & Välimäki (2012)

In both of the listening tests conducted by Rämö & Välimäki (2012) as well as Lester & Boley (2007) it was found that some listeners noted comb-filtering artefacts at very short delays, which are comparable to that of a powerful digital signal processor (DSP). It is important to note, however, that according to Dove (2008:991), this is largely owing to the fact that the listeners are provided with a reference. It is likely that without a reference, listeners would be content to play with larger delays.

3.3.4 PLACEMENT OF THE MICROPHONES

The placement of the microphones in an integrated headset relative to the ears are dependent on the type of headphones used. With circumaural and supra-aural headphones the microphones can not be placed as closely to the ears as with inter-aural headphones. Ranjan *&* Gan (2015:1992) compared the frequency response of microphones placed outside of a circumaural headphone design to those of reference microphones recorded at the ears of a dummy head. The frequency responses proved quite similar aside from the notches associated with the pinnae. Important binaural cues are, therefore, lost when microphones are placed externally.

Since inter-aural headphones allow for a closer placement of the microphones to the ear canal, the microphones are able to register accurate binaural cues (Härmä *et al.*, 2004:624). Härmä *et al.* (2004:620) suggests, however, that listeners can adapt to a modified binaural representation within a reasonable time and the connection of a sound source to a visible real-world object also improves the perceived external localization of the source.

Chapter 4

Design of Instrumentation

Having looked at different possibilities of creating virtual acoustics for musicians in small acoustic environments, the author has decided to construct a prototype of an integrated headphone and microphone combo, similar to previous prototypes that have been used for the purpose of augmented reality audio (ARA) applications. The first section of the chapter will describe this prototype and the considerations that the author had to make to best use the resources that were made available to him. The process of obtaining the impulse response, using a sine-sweep test, of the Endler concert hall in the Stellenbosch Music Department will also be discussed.

Furthermore, the chapter will describe how the author went about with setting up the session, explaining the signal flow of the system. The chapter also discusses the optimization of this prototype, recounting the pre-experiments conducted by the author. These pre-experiments include; determining, through listening tests, the appropriate equalization needed to compensate for the imperfections of the headset, as well as finding the correct balance at which musicians hear the recreated space. Lastly, the acoustic space in which the research study was conducted will be described. The limitations of the system will be discussed throughout the chapter and the author will report how specific problems were resolved.

4.1 THE HEADSET

The choice of headset was determined partly by the limitation of available equipment but also by what was believed to best reproduce the pseudo-acoustic environment and accurately simulate a virtual acoustic environment. Since the implementation of a virtual acoustic headset in music school or universities would require musicians to share headsets, a circumaural design was thought to be appropriate for hygienic reasons and comfortability. Nevertheless, the author recommends that the effectiveness of intra-aural headphones with integrated microphones, specifically commercial designs, for simulating concert hall acoustics should be tested in future studies. Intra-aural headphones that have

Figure 4.1: The Headset Prototype Designed for the Research Study

been moulded to the ears of the musicians could prove an effective choice for accuracy and comfortability. Bone-conducting headphones were not available and were concluded in the previous chapter (section 3.2.4) to be unsuitable for application in this thesis. The design of the specific headset of this thesis is discussed below and figure 4.1 shows the prototype.

4.1.1 HEADPHONES

The author decided to use open-back headphones for the research study. The headphones that were used, were the Sennheiser HD600 (figure 3.1b), which are professional quality open-back headphones. Sennheiser HD380-Pro (figure 3.1a) closed-back headphones were made available to the author as well, but were, through experimentation, judged significantly less effective than the open-back headphones. The closed-back headphones gave rise to the occlusion effect, discussed in section 3.2.5, which would be unfavourable to any singer, woodwind or brass player. The open-back headset allowed the musicians to hear the natural environment without much attenuation from the headphones.

4.1.2 MICROPHONES

Karma K-micro capacitor microphones, which have omnidirectional polar patterns (figure 4.2), were installed on the headphones. These microphones were the smallest omnidirectional microphones available to the author, but future implementations of such a system should use smaller microphones. Frequency response diagrams of the microphones were difficult to find. The article by White (2011), however, suggests that the low-end frequency response is relatively accurate, with the high frequencies rolling off above around 15 kHz. The frequency response is further suggested to have a dip at around 2 kHz and a "presence hump" at around 6.5 kHz, which can be compensated for with an appropriate equalization.

Figure 4.2: Karma K-micro Microphone

4.1.3 INTEGRATION

Microphones were integrated with the headphones externally by means of velcro. The diaphragms of the microphones were placed in line with the ears to allow for the most accurate binaural cues. Unfortunately, the microphone configuration is not truly binaural as it does not include the spectral cues of the ears, and owing to the dimensions of the headphones the microphones are spaced wider than our ears. Discussed in the previous chapter (3.3.4), Härmä *et al.* (2004:620) suggest, however, that listeners are able to adapt to this modified binaural representation, and the visual aid of the sound source can help with the required binaural cues.

The diaphragms of the microphones were aimed downwards, allowing the cable to extend upwards, where it least disturbed the musicians. According to Rumsey & McCormick (2009:54), since the microphones are of omnidirectional polar pattern and contain diaphragms of small diameter, the off-axis response of the microphones will have slightly attenuated high frequencies. Instruments that propagate most of their sound energy off-axis to the diaphragm will, therefore, be registered with less energy at frequencies of around 5 kHz and higher. Few of the instruments tested, however, propagated their sound energy off-axis to the microphones' diaphragms. Lightweight microphone cables were used for the research study and were suspended from a boom above the musicians.

4.2 OBTAINING THE IMPULSE RESPONSE

The Waves IR1 plugin was used to recreate the acoustics of a concert hall for the musicians. This is a convolution reverberation processing plugin that offers parametric controls of the reverberation, such as the reverberation time, as well as the energy of both the early and late reflections. The plugin provides an impulse response (IR) library and more importantly the potential to import individually recorded impulse responses.

For this dissertation the author chose to recreate the Endler Hall (figure 4.3), a concert hall located in the Stellenbosch Music Department that is familiar to most musicians studying in and around Stellenbosch. The impulse response (IR) of the Endler Hall was obtained through a sine sweep test, as recommended by the IR1 manual. This test entails reproducing the sweep file with a chosen number of loudspeakers into the hall and recording the resulting response of the hall with one or more microphones.

Figure 4.3: The Endler Concert Hall

4.2.1 THE SINE SWEEP

The sine-sweep audio sample made available by Waves on their website, is 25 seconds in duration, the first 15 seconds containing the sweep of pure frequencies through the entire audible frequency range, and the final 10 seconds being silence. Valimaki *et al.* (2012:1434) mention that sinusoidal sweeps have become the standard method for high-quality room impulse responses (RIR). They suggest that sinusoidal sweeps are more successful than traditional impulse test, such as a pistol or balloon, as they contain more energy in the low frequencies.

4.2.2 THE LOUDSPEAKER

The sample was reproduced by a single high-quality loudspeaker, a Meyer Sound CAL 32, that was suspended above the stage. The CAL 32 loudspeaker contains eight 4 inch cone drivers for the low frequency reproduction and 24 20 mm tweeters for the high frequency reproduction. The coverage of the loudspeaker is stated to be 120 degrees, therefore, allowing the hall to be well excited with acoustic energy across the audible frequency range. The suggested operating frequency is, however, stated to be from 100 Hz to 16 kHz (MeyerSound, 2015:31). For a more accurate sweep response for the very low frequencies an additional subwoofer should, therefore, be used. The author, however, did not have access to one.

The setup aimed to replicate the performance situation of a solo performer where most of the sound energy is produced from the stage area with the late reverberation coming mainly from the audience (figure 4.4). To obtain a high quality IR, the sweep test was reproduced at a high sound pressure level, therefore, allowing the recording to pick up all the detailed reflections of the hall. The author made sure that the speaker was free of audible distortion.

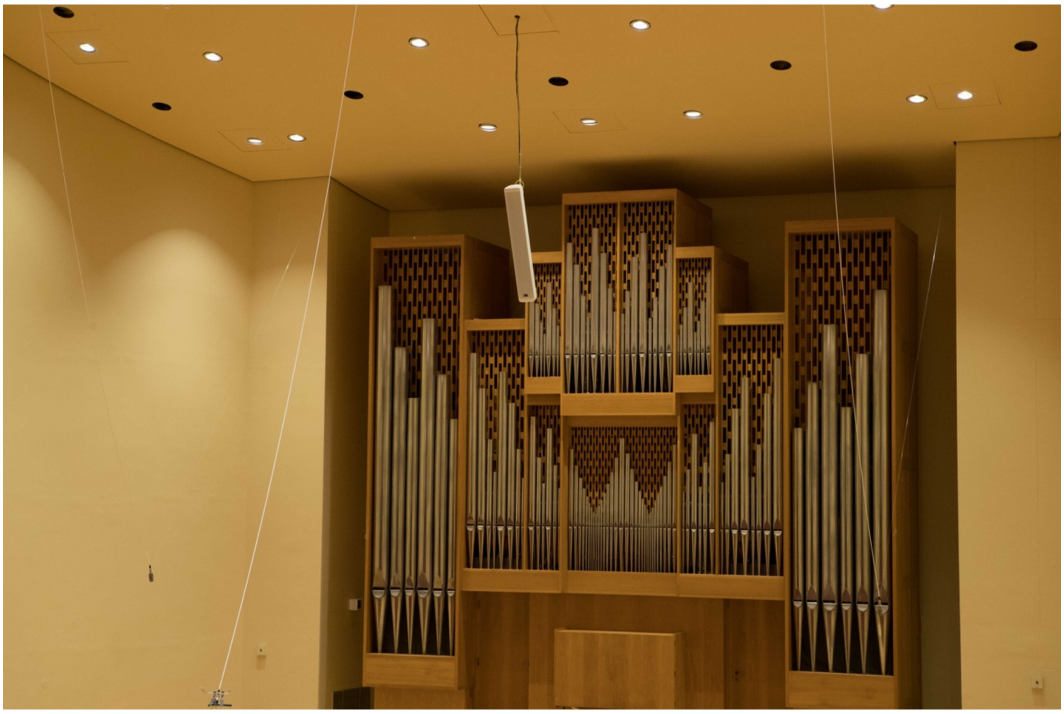

Figure 4.4: The Position of the Loudspeaker used for the Sine Sweep Test

4.2.3 THE BINAURAL MICROPHONE TECHNIQUE

The impulse response was recorded using a binaural microphone technique, since the virtual environment would be reproduced over headphones. A dummy head was improvised by the author using a small speaker cabinet, resembling the size of a human head, to separate the two microphones. For a more accurate binaural impulse response, a dedicated professional dummy head should be used, which includes the spectral cues of the ears. The binaural microphone configuration was placed at the front of the stage, similarly to where a musician would stand.

4.2.4 MICROPHONES

Two AKG C414EB microphones were used for the binaural impulse response. These are high quality professional capacitor microphones. The polar patterns of the microphones were set to omnidirectional. The frequency response curve, provided by AKG, shows a consistent response, with a maximum change in response of around 3 dB, from 30 Hz up to 20 kHz. The microphones are suggested to have a self-noise of 17 dBA, which is according to Rumsey & McCormick (2009:64) similar to a typical high-quality capacitor microphone.

4.3 SYSTEM SETUP

The headset only forms part of the system and in order for the headset to be accurate in simulating a concert hall the rest of the system needs to considered, including the equipment, software and the acoustic qualities of the natural room. The research experiment made use of equipment, software and the control room of the *Stellenbosch University Studio*. The entire system will be described, including the equipment, the setup of the recording software and the acoustic space in which the research study was conducted. In order for the headset to accurately simulate the Endler Concert Hall the frequency balance as well as the levels of the instrument needed to be considered. Pre-experiments via listening test were conducted by the author to improve the accuracy and realism of the headset in simulating the Endler Hall.

4.3.1 THE EQUIPMENT

The *Stellenbosch University Studio* is equipped with a Pro Tools HD system that runs on an Apple computer with digital signal processing (DSP) cards that allow faster processing capabilities and lower latency numbers (refer to section 3.3.3). Furthermore, the system made use of an Audient ASP 8 Channel preamplifier and a Samson S-phone headphone amplifier.

4.3.2 WAVES IR1-EFFICIENT SETUP

Consideration needed to be given to the setup of the Waves IR1 reverberation plugin to best reproduce the virtual acoustic environment. The IR1-Efficient

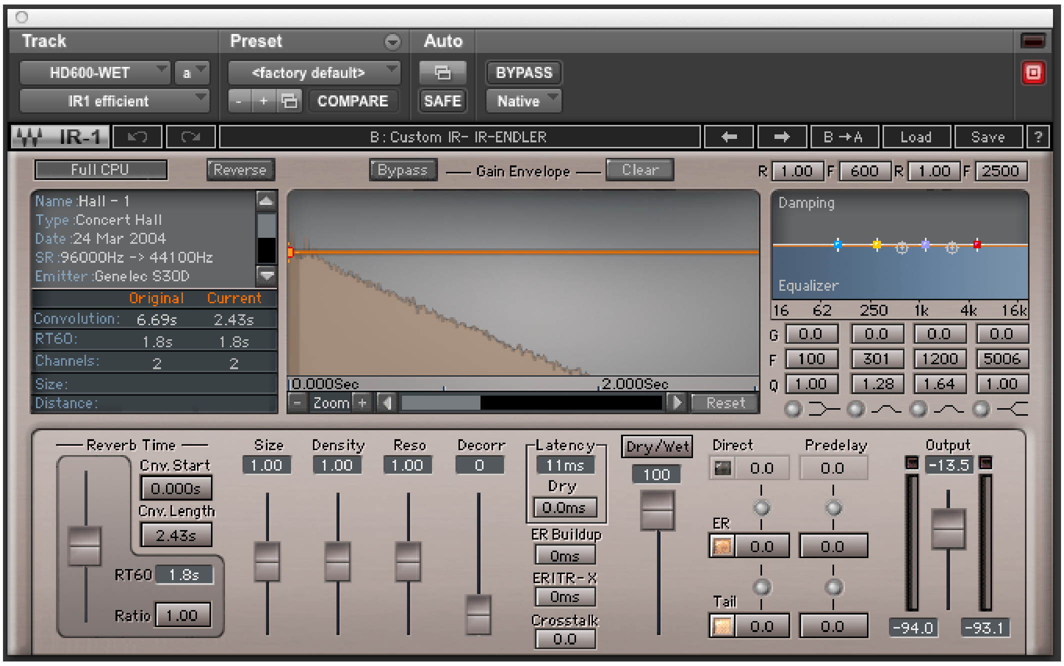

Figure 4.5: Screenshot of the Waves IR1-efficient Plugin

plugin, shown in figure 4.5, was set up on an auxiliary channel, named HD600-WET. Once the stereo sweep signal was imported, the plugin provided numerous potentially variable reverberation parameters. For the research experiment, these were left unchanged to best represent the original acoustic environment. The ratio of dry to wet ratio was set to 100 %, implying that the auxiliary channel only provides the reverberation signal. The dry signal was routed to the HD600-WET via a send from a dedicated stereo audio channel, named HD600-DRY, which received the raw microphone signal. The send was set to pre-fade to keep the reverberation level independent of the channel fader of the audio channels. The send was set to mono, with both left and right pan knobs being centered. This was done to replicate the sweep response, which used a mono source, the single loudspeaker placed in the centre of the stage.

Unfortunately, the Waves IR1 plugin was not available as a DSP plugin for the author and, therefore, had to be processed by the computer's inbuilt processor (CPU). Since this processor is not as powerful, there was a significant latency between the direct sound and that of the reverberation. This does not pose a problem for the late reverberation, where a pre-delay of several milliseconds is not disturbing. The early reflections, which are meant to provide cues to the acoustic environment, however, lose their accuracy as a consequence of the latency. The convolution, however, still sounded accurate to the author. For future implementations of a such a system, the convolution plugin should, nevertheless, be processed by DSPs that contain minimal latency values.

Figure 4.6: Screenshot of the Pro Tools Session Setup

4.3.3 EQUALIZATION

Equalization needed to be considered for the compensation of both the frequency response of the microphones as well as the attenuation properties of the headphones. Listening tests were conducted by the author to select appropriate filters.

For both listening tests, a reference pink noise signal was reproduced by a pair of Bowers and Wilkins 805 Diamond loudspeakers. This signal was reproduced at 65 dBA. This was measured using a Radioshack sound level meter, set to a slow response time, at the listening position. In order to determine whether the headsets' frequency response was accurate, the pink noise signal was recorded by the headset's microphones and reproduced over the headphones at a subjectively similar level to that of the reference signal. The author alternated between playing the reference signal over the speakers and the recorded sample over the headset. It was decided that the frequency

Figure 4.7: High-pass Compensation Filter for High Frequency Attenuation of Headset

response of the headset was remarkably similar to that of the reference signal and, therefore, no equalization was applied for the purpose of addressing any inaccuracies of the headset's frequency response.

The second listening test was conducted to determine whether or not a high-pass-filtered dry signal should be used to compensate for the high-frequency attenuation of the headphones. In this test the author alternated between listening to the reference pink noise signal with and without the headset. At this stage the headset was not in any way active. The attenuation properties of the headphones were realized as a clearly audible high-frequency loss in the reference signal. A high-pass filtered binaural signal from the headset's microphones was then reproduced over the headset and once again compared to the reference pink noise signal. With some modifications of the filter, the author managed to compensate for the loss in high frequencies quite accurately. The microphones did, however, add some noise. A standard DSP based Pro Tools equalizer plugin was used with a high-pass filter of slope 12 dB per Octave at 2.60 kHz (figure 4.7). The author made sure that Pro Tools's delay compensation was inactive. This allows the filtered dry signal to feed the headphones with minimal latency. When delay compensation is active, the computer delays the dry signal by the same time the computer takes to process the wet channel containing the highly demanding convolution plugin.

4.3.4 LEVELS AND BALANCE

The levels of the systems had to be set in a way to accommodate the diverse instruments that would be tested in the research study. The gain was, therefore, set rather low to prevent distortion. During pre-experiments the author noted that the IR1 plugin strengthened the input signal significantly, which caused the input stage of the HD600-WET channel to be remarkably higher in level than the HD600-DRY channel. The output fader of the plugin, therefore, needed to be reduced substantially to prevent the input stage of the channel fader from distorting.

Since the headset used open-back headphones the author was careful that the system would not introduce feedback. Through multiple tests of the system the author was confident that no feedback would occur during the research study.

In order to establish the correct balance between the direct signal and that of the virtual acoustic environment, a reference recording was conducted in the Endler hall. An excerpt of a classical guitar piece was performed and recorded in the hall by the author while wearing the headset. The author performed the identical excerpt in the test environment and adjusted the level of the reproduced virtual acoustics until the balance matched, as closely as possible, that of the recorded sample.

4.3.5 THE NATURAL ACOUSTIC SPACE

The research experiment was conducted in one of the recording rooms of the Stellenbosch University Studios. This room is decoupled from the rest of the building by floating structures, providing high sound isolation (Newell, 2008:34). This room is larger than the typical practice room of the music department with a volume of $156\,\mathrm{m}^3$ (figure 4.1). The room was chosen out of practicality, since an isolated environment was required and the chosen equipment was only available for this room. The recording room has connectivity to the studio control room, where the equipment required for the experiment was situated.

Length	Width	Height	Volume
8.25 m	4.85 m	3.90 m	$156\,\mathrm{m}^3$

Table 4.1: Dimensions of the Room used for the Research Study

The room, although containing several absorption panels, had audible acoustics of its own. Additional absorption panels were, for this reason, used to surround the musicians and minimize the number of reflections. The top was left open allowing some natural reflections (figure 4.8). The length and width of the smaller acoustic space in which the participants were required to play were approximately $2.5\,\mathrm{m} \times 1.5\,\mathrm{m}$ respectively. The panels were mostly around 2.1 m

Figure 4.8: The Natural Acoustic Space in which the Study took Place

high and about 100 mm thick. Newell (2008:99) and Szymanski (2008:103) both show porous absorption panels with 100 mm thickness to be effective at absorbing frequencies above 125 Hz.

The author believed that by minimizing the effect of the room, the virtual acoustic environment would seem more realistic to the participants. The dryer acoustics were also important in reducing the sound pressure levels that the musicians would be exposed to, since the additional virtual acoustic environment adds to the overall loudness of the room.

It is suggested by the author that in future implementation of the system in practice rooms, the acoustics of the natural space should, as in this study, be treated in order to minimize loudness levels and the character of the natural space. A few early reflections can, however, help in allowing the microphones to pick up a well balanced instrument.

Chapter 5

The Method

In order to test the effectiveness of the system that was designed to simulate the acoustics of the Endler concert hall, the author chose to use a survey-based research design. Scheduled structured interviews on a one-to-one, face-to-face basis were conducted with the research participants during the progression of the experiment, which required the musicians to perform with their instruments while wearing the prototype headset. The effectiveness of the system was, therefore, determined by the subjective responses of the research participants. The specific method used to obtain the necessary data to determine the effectiveness of the system will be discussed in this chapter, including the choice of research design, the sampling techniques as well as the choice of data analysis.

5.1 RESEARCH DESIGN

A survey-based research design was chosen for the study of this dissertation, since the effectivity of the system is believed, by the author, to be strongly dependent on the subjective responses of musicians. The implementation of virtual acoustic environments for musicians in small practice rooms with headsets would only be feasible if musicians found the system helpful in preparing for performances. The chosen design, therefore, allowed the research participants to express their opinion of the system, through a structured interview on a face-to-face basis with the author.

The survey-based research design is an example of a pre-experimental design, which, according to Bless *et al.* (2013:137), has the least scientific rigour for quantitative research, but is commonly used for qualitative research. More specifically, the design is an example of a one-shot case study, which involves measuring the dependent variable after an event or intervention. The dependent variable, for this research study, was the attitudes of the musicians towards the effectivity of the system. Bless *et al.* (2013:137) suggest that for quantitative research this design is difficult to interpret, since the effect of the intervention can not be compared to an initial state. With a questionnaire being more qualitative in nature, this is less of a problem as the participants provide feedback on the experience of the intervention. The author was, therefore, not aiming to measure the effect of the intervention objectively.

As a comparison, Kato *et al.* (2008), in their study, measured the effect of different virtual acoustic environments on the playing technique of musicians through objective measurements and signal analysis. A similar approach for this research study, would, however, not prove the effectivity of the system. Rather, it would, once again, only prove or disprove that virtual acoustics influence the performance attributes of the musicians. In order to objectively measure the effectivity of the system, the musicians' improvements in their concert performances would have to be measured, which is not practical.

As with this research study, Lamberty (1980:149) too relied on questionnaires to determine musicians' preferences of practice room acoustics. Lamberty (1980:149) suggests that with ever more advanced measuring techniques and computers it is easy to forget that our ears are the most sophisticated equipment for acoustic design purposes. Christian *&* Gade (2015:232) similarly stresses the importance of musicians' preferences for acoustics when designing performance or rehearsal rooms. Musicians' subjective responses on the effectivity of a virtual acoustic environment are, therefore, confirmed to be valuable data.

This research study did not provide the research participants with a reference to which they could compare the effectiveness of the system. The research study did not intend to show the effectiveness of the system relative to any other previously designed virtual acoustic system, since this does not determine whether musicians would indeed be willing to use or be advantaged by the use of the specific system. The author, therefore, assumed that the musicians were able to determine the effectiveness of the system through previous experiences in either practice rooms or performance halls.

5.2 THE SAMPLE

The research study was conducted during the 14$^{\text{th}}$ annual Stellenbosch International Chamber Music Festival (SICMF), an event where classical musicians from all around South Africa as well as overseas come to participate in classical music performances over a ten day period. The musicians taking part range from scholars to internationally renowned classical musicians. The festival, as the research study, took place in the Stellenbosch Music Department, with daily performances and rehearsals taking place in the Endler Hall. The festival, therefore, proved the perfect opportunity to conduct the research study. The study took place over a three day period during the festival.

5.2.1 SAMPLING TECHNIQUE

Quota sampling was used for the research study. Since different instruments were expected to respond differently to the effectiveness of the system, the sample was divided into seven instrument groups. The SICMF is a busy time for the musicians taking part and, therefore, the author selected participants by means of convenience. The author approached the musicians in person, since emails and posters were believed not to gain many participants. The selected musicians ranged from first year music students to professional musicians. The experience levels were, therefore, diverse. A total of 51 musicians participated in the research study. Scholars, that were still in school, were not included in the study, as their experience in practicing in small practice rooms and performing in concert halls were believed to be restricted.

5.2.1.1 INSTRUMENT GROUPS

The research aimed to test the effectiveness of the system for diverse instrument groups. In order to get an understanding of how different instruments responded to the survey, the author aimed to have at least five participants for each instrument group that was tested. The author chose to include instruments that had different means of sound production, as this could potentially determine the response of the musicians. Instruments were also included on a basis of availability.

The following instruments, or instrument groups, were selected by the author:

Violin and Viola These two string instruments, both forming part of the bowed string instruments, have, according to Adler (2002:8), similar playing techniques and acoustical properties. Although the viola is larger than the violin, it is likewise held on the left shoulder and supported by the chin (Adler, 2002:51-65). The responses of musicians playing these instruments were expected to be similar and, for this reason, the instruments were placed in a single instrument group.

Cello This bowed string instrument has a unique position relative to the body, owing to its still larger size in comparison to the violin and viola. It is supported between the knees, with the neck of the instrument extending over the left shoulder (Adler, 2002:75). Because the headset could potentially be obstructive to musicians playing this instrument the author chose to include the cello in the study.

Clarinet and Oboe These instruments fall into the woodwind category of instrument types and the mouthpieces utilize reeds to produce sound. The oboe uses a double reed and the clarinet a single reed construction. Hosken (2011:9), however, suggests that the means of sound production are similar. The author, furthermore, deduced from Sevsay (2013:78-80) and Adler (2002:193-209) that the playing techniques are, too, rather similar. Since availability of musicians for each instrument was limited the author placed the two instruments into one instrument group.

Flute This is another woodwind instrument. The flute, however, is a non-reed instrument and, therefore, involves a significantly different sound production technique, as is illustrated by Hosken (2011:9). The flute is also played transversely, making it a unique instrument to include in the research study.

Trumpet and Trombone These two instruments are both brass instruments and, as suggested by Sevsay (2013:88), form part of their own instrument family. Sevsay (2013:88) furthermore indicates similarity in the mouthpiece of the two instruments, placing them both under the classification of a "shallow cup mouthpiece". Hosken (2011:11) shows that in all brass instruments sound is produces by means of "lip buzzing". The author, therefore, expected uniform responses from participants playing brass instruments. Since the trumpet and trombone were expected to be most accessible they were included in the study as one instrument group.

Piano This instrument was included because of its popularity as a solo instrument. An upright piano was used for the research study. Since the instrument does not get in the way of the headset, the author expected positive feedback with regards to the comfortability of the headset from the participants.

Classical Guitar This instrument is a plucked string instrument employing nylon strings (Fletcher *&* Rossing, 1998:239). The instrument is, for a few reasons, unique in comparison to the other instruments included in the research study. Its sound production is achieved by means of the fingernails or fingers of the right hand plucking the strings. The classical guitar, as suggested by Adler (2002:102), is quite soft, and is most suitable for chamber works or solo performances. The instrument is placed in front of the musician, typically supported between the legs. Participants playing the guitar were, therefore, not expected to have any issues with wearing the headset.

The selected instruments were believed to give a good indication of the effectivity of the system for all musicians. Singers were, unfortunately, not tested because there were not enough singers at the festival to take part. The author completed the study once seven, or in some cases, eight musicians had been selected for each instrument group.

5.3 ORDER OF PROCEDURES

In order to make the research study accessible to all the musicians, the author ensured that the questionnaires and study were kept short, as suggested by Bless *et al.* (2013:155). In most cases the study took less than ten minutes, which aided in obtaining the necessary number of participants. The author made his best effort to be accommodating and ensured that the participants felt comfortable throughout the study. The author was in the room with the participants throughout the study. A laptop connected to the control room's computer via an ethernet connection was used to make any necessary changes on the software during the progression of the study.

The author created a a step by step 'order of procedures' document in order to conduct the research study as smoothly as possible. This document is included in Appendix A. The author did one test-run of the research study after which he felt confident with the procedures of the study. For the first few participants the 'order of procedures' helped in keeping the study organized. After the first participants, however, the author conducted the study with ease and did not have need of it anymore.

5.3.1 CONSENT OF PARTICIPANTS

Before commencing with the study the author explained the purpose of the study and what would be required of the participant. The consent form was then given to the participant, which provided further details about the research study. This was also summarized verbally to the participant by the author. Once the participants were satisfied with what was expected of them in the research study the author asked for their signature. The consent form is included in Appendix C.

5.3.2 EXPERIENCE LEVEL OF THE MUSICIANS

Other than the instrument type that the musicians played, the author noted the musician's experience level by asking the participants whether or not they are professional; and if not which year of studies they are currently in. This information was noted in order to later determine whether this has a significant effect on the response of the participants. As agreed upon in the consent form, the author did not require any personal information from the participants.

5.3.3 THE STUDY

The participants were asked to be seated during the study to ensure that all the participants were exposed to similar acoustics. This also prevented the need to adjust the cable length for different participants. The cables were suspended from a boom above the musicians in such as way as to allow enough manoeuvrability without the cables getting in the way.

The study required the participants to play a short musical excerpt or anything that gave the musicians an impression of the acoustic space. The author suggested that this should be around 30 seconds in length. This excerpt needed to be repeated on three occasions during the study. The author was not strict on the length of the excerpt and whether or not the musician repeated the same excerpt each time. The instructions only served as a guideline. The author used his discretion to judge when a participant had played for long enough.

On the first instance of playing the excerpt the participants played without the headset. This allowed the participants to familiarize themselves with the acoustics of the natural acoustic space. The author communicated to the participant that the purpose of this exercise was to get a feel for the acoustics of the space. The familiarization of the acoustic space served as reference to the participants before putting on the headset.

The author then helped the participants in putting on the headset, adjusting its size to make it fit comfortably. The headset was at this stage of the study merely meant to reproduce the natural environment as accurately as possible. As described in the previous chapter (4.3.3), the headset reproduced only high frequencies with the rest of the natural acoustic environment leaking through headphones. The participants were again asked to play a musical excerpt, which allowed them to get an impression of the natural acoustic environment while wearing the headset. The author asked the participants to pay attention to whether the room and their instruments sounded the same as it did before. After this, the author asked the first two questions, which concerned the accuracy of the headset in reproducing the natural surrounding environment as well as the headset's comfortability.

For the final stage of the study, the simulated Endler hall was activated and the participants once again played a musical excerpt. On completion of the excerpt the author deactivated the simulated hall and asked the final questions. The participants remained seated for these final questions and the headset was kept on until the questionnaire was complete. This allowed the participants to answer the final questions without delay, which could have potentially influenced their reaction. The questions concerned the realism of the simulated hall and whether or not the participants could imagine practicing with a similar headset. The author then helped to remove the headset.

5.4 THE QUESTIONNAIRE

The interview was in a structured format and for that reason made use of a questionnaire. The questionnaire contained only four subjective questions. Different types of questions were used for the study. The first three questions required the participants to rate their response along an ordinal scale from one to seven; and the final question required a yes/no response. Participants were allowed to provide reasoning for all their answers in an open-ended question format, to allow a better understanding of the participants' opinions. The questionnaire aimed to determine the effectivity and accuracy of the headset. This section will discuss and justify the questions used in the research study.

5.4.1 CLOSED-ENDED QUESTIONS

The author's main goal was to obtain answers, which could be analyzed with quantitative techniques, as this was believed to provide the best indication of musicians' subjective response towards the effectiveness of the system. Closed-ended questions, containing fixed responses, therefore, formed the core part of the questionnaire (Ruane, 2005:131). A rating scale was used to determine the effectiveness of the headset.

5.4.1.1 THE RATING SCALE

Iarossi (2006:59) suggests that the number of categories in the rating scale should be chosen according to the specific demands of the study, but between five and nine categories are typically recommended. The author chose to use an uneven number of categories in order to allow the participants to choose a middle value.

Recommendations for labeling numbered categories differ among different authors. Iarossi (2006:63) suggests that category labels should be applied only to the extreme values and the middle values, whereas Schaeffer *&* Presser (2003:78) suggest that more reliable measurements are attainable when all intermediate categories are labeled. Sapsford (2007:223) describes a specific measuring scale, semantic differentials, where participants place their response between two opposed adjectives on a number scale. This type of scale, therefore, only contains two extreme labels. Sapsford (2007:224) stresses the importance of the labels and that participants should themselves see the adjectives as opposites.

For the questionnaire of this research study, which was conducted in an interview format, the author chose only to verbally communicate labels for the extreme values, one and seven, respectively. The same number of categories were used for the three questions, with the labels changing in accordance to what the question was measuring.

5.4.2 OPEN-ENDED QUESTIONS

Open-ended questions are those questions which do not provide any answer categories. The participants are, therefore, free to express such an answer in their own words. Ballou (2011:2-3) suggests different uses of the open-ended question type, including for an explanation of a previous answer. This was the function of the open-ended questions in the questionnaire of the research study. The open-ended questions allowed participants to elaborate or reason their answer to the closed-ended questions. Since the author has not undergone any interviewer training, as is advised by Ballou (2011:4), the open-ended questions were not intended to be the main type of data collection. Ballou (2011:4) suggests that with open-ended questions in an interview format the interviewer should record the answers of the participants verbatim (in exactly the same words). This, however, was thought to be impractical for the research study as it would have disrupted the proceedings of the study. The author/interviewer, therefore, noted the responses in a summarized form, which was less time consuming. The summarized answer was read to the participant to ensure that the author did not misinterpret the response. In cases where participants were struggling to find the correct term to describe their response, the author suggested terms which seemed most suitable to what the participant was saying. When participants used terms that were contradictory to their overall response, the author suggested words which were more suitable.

5.4.3 QUESTION 1

The first question of the interview questionnaire was intended to test the effectiveness of the headset in reproducing the pseudo-acoustic environment (refer to section 3.3.2). Ideally the musicians should not hear any differences between playing with and without the headset, since any distortions in the sound of the natural acoustic environment, especially the sound of their instruments, could potentially affect their playing. This was, therefore, an important question to ask. Also, by first testing the effectiveness of the headset in reproducing the natural acoustic environment, before testing its effectiveness in simulating a concert hall, the author was able to isolate any problems in the construction of the headset. The question was phrased as follows:

How accurately can you hear your surrounding environment?

1 = completely different to not wearing headphones

7 = identical to not wearing headphones

The phrasing of the question was believed to be concise but easy to understand. The labels of the extreme categories were intended to further help the participant in their choice of rating, as these labels were specific to the study. These labels were thought to be more helpful than using labels which merely described the degree of 'accuracy' such as this:

1 = very inaccurately

7 = very accurately

The author was concerned that participants might rate the question according to how well they hear their instrument, rather than how accurately they hear it. The labels 'completely different to not wearing headphones ' and 'identical to not wearing headphones', therefore, ensured that the participants rate the question according to how similar the pseudo-acoustic environment is to the natural acoustic environment. To provide a reference of how the natural acoustic environment sounded without the headset, the participants were first asked to play their instruments without the headset and then with the headset.

5.4.4 QUESTION 2

The second question tested the comfortability of the headset. This question would help determine whether a design similar to the prototype developed for this study would be feasible for future implementation. This question was rather simple and, therefore, did not require much additional explanation from the interviewer. The question was phrased as follows:

How comfortable is the Headset?

1 = very uncomfortable

7 = very comfortable

The results of the question were predicted to vary according to the type of instrument being played. The violin/viola, cello and trombones were predicted to have the lowest ratings, since the headset could potentially be obstructive to these instruments. Overall the ratings were, however, expected to be high, as the specific headphones are believed to be less obtrusive to the ear than other headphone types discussed in section 3.2.

5.4.5 QUESTION 3

This question concerned the effectiveness of the headset in simulating a concert hall. The participants were asked to rate the realism of the concert hall. The 'realism' of the hall was believed to well describe what the author intended to determine. The question was phrased as follows:

How realistic does the concert hall sound?

1 = very unrealistic

7 = very realistic

The interviewer communicated to the participants that the headset was specifically simulating the Endler hall. Since the Endler hall was in the same building as the research study, all the participants had either performed, rehearsed or attended a concert in the Endler hall. Participants were, therefore, able to use their previous experiences of the hall as a reference to answer the question. The author predicted that both the instrument as well as the level of experience would influence the results. The open-ended questions were believed to help with identifying the shortcomings of the headset.

5.4.6 QUESTION 4

The final question was a yes/no question phrased as follows:

Could you imagine practising in a virtual acoustic environment with a similar headset?

This question aimed to determine whether future implementation of such a system is viable. This was placed last in the questionnaire so that participants could use their overall experience of the system as a reference. The open-ended question, which asked for the participants' reasoning, was intended to provide insight into the advantages and disadvantages of the system according to the participants.

5.5 ANALYSIS

The questionnaire collected both quantitative as well as qualitative data, in the form of numbered ratings and words respectively (Denscombe, 2010:237). The numbers obtained from the ordinal rating scales form the core of the analysis. The numbered responses were tabulated and illustrated with cumulative frequency graphs to better present the data. For the rated responses a measure of central tendency was used to show the average response of participants as a whole and in different subgroups. This was used to determine whether the system was positively received by the musicians and whether responses differed between instrument types and experience levels of the musicians.

As suggested by Denscombe (2010:248), the scale of measurement used for research determine the type of statistical analysis that can be used. Since the scale of measurement used for the numbered question was an ordinal scale, there were some restrictions to the type of analysis used. Some authors, including Denscombe (2010:248) and Ruane (2005:183), state that the mean as a measure of central tendency should be used only for interval and ratio level data. Bless *et al.* (2013:256), however, does not restrict the mean measure to interval and ratio data and considers it suitable for opinionated numbered responses. The author believed the mean to give the best impression of the responses. The mean was, therefore, also used to compare the responses of the different instrument groups. The author did not use any inferential statistics to compare the mean scores, since the mean scores were only intended to give an impression to the reader and inferential statistics were also not thought to be suitable for this thesis as the responses were purely subjective.

For the analysis of the open-ended responses the author summarized the varying responses by means of coding, as suggested by Denscombe (2010:282). The author took time to understand whether there were any recurring themes in the responses and then chose to code these responses by a term that was believed to best encompass the similar responses. The coded terms that occurred most frequently were discussed. The author also used his discretion to discuss any rare responses that were believed to be important.

5.6 ETHICAL CONSIDERATIONS

The author obtained ethical approval before the commencement of the research study section (appendix D), since the study involved human participants. As discussed in 5.3.1, a consent form was provided to the participants in order for them to fully understand the ethical considerations. The ethical considerations are summarized in the consent form (appendix C).

CHAPTER 6

Results and Discussion

This chapter shows tabulated results of the responses to the rated questions of the research study and discusses the responses to the open-ended questions. Appendix B contains a summarized table of the individual participants' information and responses.

The mean ratings were calculated for both the total as well as the separate instrument groups for each of the first three questions. All in all, the results of the research study show positive responses from the musicians towards the headset. The means for all three rated questions fall above the the middle point of the 7-point scale, therefore demonstrating a positive trend.

The least positively rated question was question one, which concerned the accuracy of the headset in reproducing the pseudo-acoustic environment. The comfortability of the headset was rated considerably positive, which indicates that most of the musicians are not seriously disturbed by the headset and are able to play their instrument without obstruction. The realism of the simulated concert hall, specifically the Endler Hall, was also rated positively.

48 of the 51 participants answered yes to the final question, which asked whether the musicians can imagine practicing in a virtual acoustic environment with a similar headset.

The responses of the musicians to the open-ended questions of the questionnaire provide a better understanding of the participants' opinions towards the system. There were a number similarities in the responses to the open-ended questions, which will be discussed. Any specific comments by the musicians that the author believed to be important are also discussed.

6.1 ACCURACY OF THE HEADSET

The rated responses of the participants to question 1, concerning the accuracy of the headset in reproducing the pseudo-acoustic environment, are shown in table 6.1. The mean rating for all the 51 musicians was 4.9, with the most frequently selected score, the mode, being 5. The frequency distribution of the musicians' responses are visually illustrated in figure 6.1.

Instrument Group	**Rating**							**Total**	$\bar{x}$
	1	**2**	**3**	**4**	**5**	**6**	**7**		
Violin/Viola	0	0	2	1	4	0	0	**7**	4.3
Cello	0	1	1	0	3	2	0	**7**	4.6
Clarinet/Oboe	0	1	2	1	2	2	0	**8**	4.3
Flute	0	0	1	2	3	0	1	**7**	4.7
Trumpet/Trombone	0	1	0	0	4	3	0	**8**	5.0
Piano	0	0	0	1	2	3	1	**7**	5.6
Guitar	0	0	0	1	2	2	2	**7**	5.7
Total	**0**	**3**	**6**	**6**	**20**	**12**	**4**	**51**	**4.9**

Table 6.1: Participant Ratings for the Accuracy of the Headset in Reproducing the Surrounding Environment

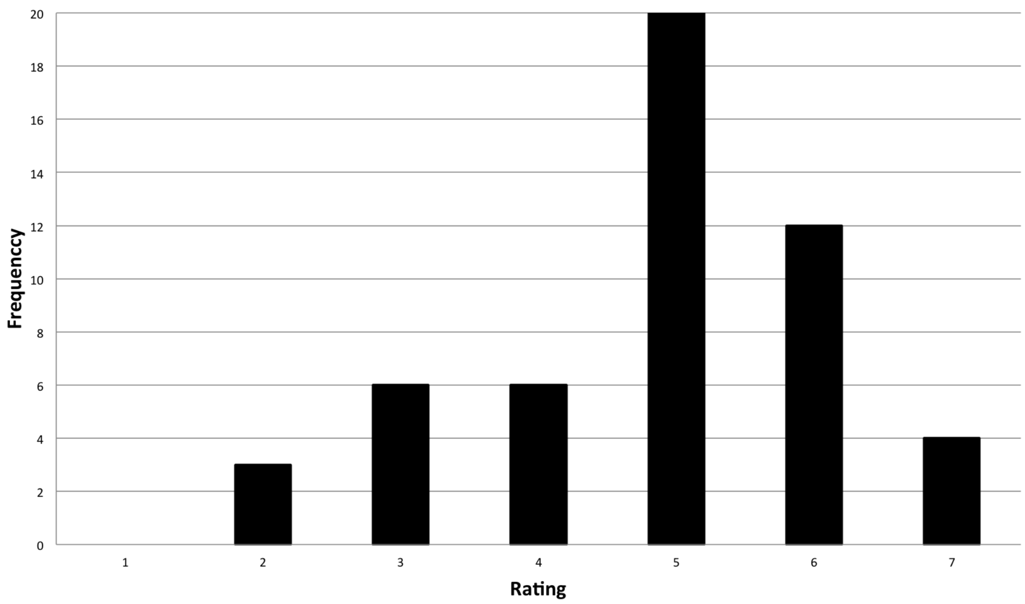

Figure 6.1: Frequency Distribution of Participant Ratings for the Accuracy of the Headset in Reproducing the Surrounding Environment

6.1.1 RATINGS FOR DIFFERENT INSTRUMENT GROUPS

The mean response of the different instrument groups ranged from 4.3, for the Clarinet/Oboe and Violin/Viola groups, to 5.7 for the Guitar group.

The two instrument groups, Piano and Guitar, have appreciably higher mean ratings than the other groups, with 5.6 and 5.7 respectively. This can potentially be attributed to the fact that the instruments are, in comparison to the other instruments, further away from the headset. This would explain the lower ratings for the Violin/Viola group, where the left microphone is close to the soundboard of the instrument, likely altering the natural sound of the instrument.

The lower mean rating for the Clarinet/Oboe group can, on the other hand, be as a result of the complicated resonances that occur in the head, which, with obstruction from the headset, can change the perception of the instrument. This, however, does not explain the higher mean ratings of 5.0 for the Trumpet/Trombone group.

6.1.2 OPEN-ENDED RESPONSES

The comments allowed the musicians to clarify perceived differences between the natural acoustic environment with and without headphones. Only four out of the 51 participants noted a hiss over the headset, a consequence of the noise of the equipment, especially the headphone amplifier.

The participants used different terms to describe similar changes in the surrounding environment. The most frequently perceived change, according to the comments, was a change in high-frequency perception. The following terms used by the participants are believed, by the author, to refer to a change in high-frequency perception:

- Presence
- Clarity
- Perception of Overtones
- Muffled
- Deader
- Duller
- Muted

The term *presence* will be used to encompass all the above terms, with more *presence* implying increased high-frequency perception. Although there were inconsistencies in the responses, a significant number of musicians stated that the *presence* of the surrounding environment, including the musicians' instruments, was less when wearing the headset. This was especially noticed by participants from the Flute and Violin/Viola groups. Five out of seven flutists suggested that while wearing the headset the natural environment was less *present*.

The participants from the Clarinet/Oboe group had unique responses in comparison to the other instrument groups. Three of the participants from the Clarinet/Oboe group mentioned that when wearing the headset the resonances of the instrument were more pronounced. The *presence* of the surrounding environment was also said to be less by three participants from the Clarinet/Oboe group.

The participants from the Piano and Guitar did not, according to the comments, perceive much difference in the surrounding environment while wearing the headset. This is supported by the ratings, with the Piano and Guitar groups scoring the highest for the accuracy of the headset in reproducing the natural environment. The Cello group, although having perceived differences, did not provide consistent responses. It is worth noting, however, that two of the cellists found the surrounding environment to be more *present*, including a professional musician. The responses from the Trumpet/Trombone group were inconsistent in terms of *presence* as well as loudness. One participant perceived the headset as being softer, whereas another participant believed it to be louder. The same was the case with *presence*.

6.1.3 CONCLUSION AND DISCUSSION

Although positive ratings were provided in most cases, musicians perceived audible differences between wearing or not wearing the headset. This was to be expected, since the attenuation properties of the headphones are complicated, varying from different directions. The *presence* or high-frequency perception of musicians while wearing the the headset seems to be less than without the headset. There were, however, inconsistencies in this matter, with some participants finding the headset more *present*. This can be as a consequence of the equalization, which was not perfectly refined to compensate for the attenuation properties of the headset. Future headsets should, therefore, aim to more accurately compensate for the specific attenuation properties of the headset.

Specifically the participants from the Clarinet/Oboe group perceived increased resonances when wearing the headset. A future design of a headset, such as the one used, should, therefore, be constructed in such a way as to minimize the occlusion effect (refer to section 3.2.5). The author recommends open-back headphones with refined ear-cups, which have a minimal effect on the resonances of the skull and jaw.

6.2 COMFORTABILITY OF THE HEADSET

The ratings of the different instrument groups to Question 2, the comfortability of the headset, are shown in table 6.2 and a visual representation of the overall ratings are shown in figure 6.2. The comfortability of the headset was rated remarkably positively, with the mean rating for all the participants being 5.8 and the most frequently selected score, the mode, being 7 on the 7-point numbered scale. Only six of the 51 participants rated the comfortability less than 5, with all these six participants rating the comfortability 4.

Instrument Group	**Rating**							**Total**	$\bar{x}$
	1	**2**	**3**	**4**	**5**	**6**	**7**		
Violin/Viola	0	0	0	1	0	2	4	**7**	6.3
Cello	0	0	0	0	5	2	0	**7**	5.3
Clarinet/Oboe	0	0	0	2	1	2	3	**8**	5.8
Flute	0	0	0	0	2	3	2	**7**	6.0
Trumpet/Trombone	0	0	0	1	1	1	5	**8**	6.3
Piano	0	0	0	2	3	1	1	**7**	5.1
Guitar	0	0	0	0	1	4	2	**7**	6.1
Total	**0**	**0**	**0**	**6**	**13**	**15**	**17**	**51**	**5.8**

Table 6.2: Participant Ratings for the Comfortability of the Headset

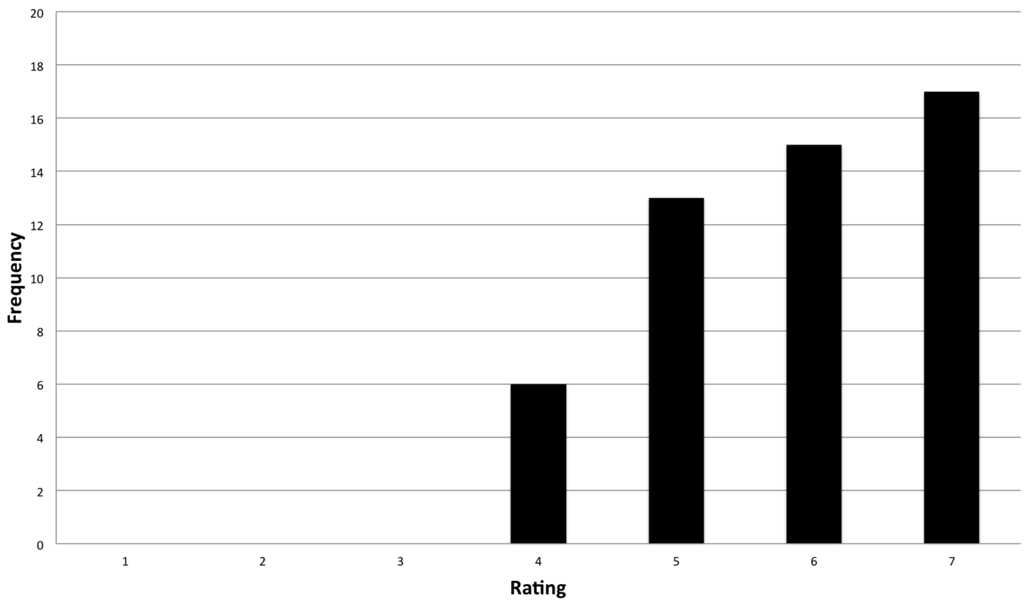

Figure 6.2: Frequency Distribution of Participant Ratings for the Comfortability of the Headset

6.2.1 RATINGS FOR DIFFERENT INSTRUMENT GROUPS

The mean ratings for the different instrument groups show that the Piano group gave the lowest comfortability rating for the headset, with a mean rating of 5.1. This is an unexpected finding, since the headset did not seem to be in any way obstructive to the playing of the pianists. Cellists provided a mean rating of 5.3 for the comfortability of the headset, which also falls slightly below the mean rating of the rest of the instrument groups. This was to expected, since the neck of the instrument comes very close to the headset. The Violin/Viola and Trumpet/Trombone groups have the highest mean rating of 6.3. This is surprisingly high, since violins, violas and trombones all have the instrument situated relatively close to the headset.

6.2.2 OPEN-ENDED RESPONSES

Many of the participants did not provide any comments as to the comfortability of the headset. A total of only three musicians, two cellists and one violinist, mentioned that the headset was obstructive. This is a positive finding, since the author expected more remarks with regards to the obstruction of the headset, especially from cellists and trombonists. All three trombonists mentioned in their comments that they can adjust the instrument pipes so as to avoid obstruction. Overall, seven participants suggested that the headphones felt a little heavy and that they could feel the added weight. One participant mentioned that with time the headset would feel uncomfortable.

6.2.3 CONCLUSION

The results for this question indicate that musicians are not obstructed much by the headset and can potentially practice with a similar headset for lengthy time periods. Instrument groups that were expected to be obstructed by the headset still rated the headset's comfortability highly. The differences between the mean ratings of the instruments are not of strong consequence, since those instruments which were expected to have lower ratings did not. Future implementations of such a headset could, therefore, be of a similar shape and size as the one used in the study. If the integrated microphones are more compact in future designs, the headset can, however, be even less obstructive than the prototype.

6.3 THE REALISM OF THE SIMULATED CONCERT HALL

The realism of the simulated concert hall was rated positively by the participants, with a mean rating of 5.6 on the 7-point numbered scale (table 6.3). The overall responses are visually illustrated in figure 6.3. The most frequently selected rating, the mode, was 6, with 20 of the 51 participants selecting this rating. Nine of the participants rated the realism a 7. According to this rating, the participants found the simulated concert hall to be very realistic.

6.3.1 RATINGS FOR DIFFERENT INSTRUMENT GROUPS

The rating were highest for the Flute and Guitar groups, both having a mean rating of 6.1. The lowest ratings were from the Piano group, with a mean rating of 5.0. It should be noted, that the pianists gave the highest ratings for the accuracy of the headset in reproducing the natural acoustic environment (section 6.1), whereas the flutists provided low ratings for the accuracy. This suggests that the two questions are not correlated.

Instrument Group	**Rating**							**Total**	$\bar{x}$
	1	**2**	**3**	**4**	**5**	**6**	**7**		
Violin/Viola	0	0	0	1	2	2	2	**7**	5.7
Cello	0	0	0	1	2	3	1	**7**	5.6
Clarinet/Oboe	0	0	1	0	3	4	0	**8**	5.3
Flute	0	0	0	0	1	4	2	**7**	6.1
Trumpet/Trombone	0	0	0	0	5	1	2	**8**	5.6
Piano	0	0	0	2	3	2	0	**7**	5.0
Guitar	0	0	0	0	1	4	2	**7**	6.1
Total	**0**	**0**	**1**	**4**	**17**	**20**	**9**	**51**	**5.6**

Table 6.3: Participant Ratings for the Realism of the Simulated Concert Hall

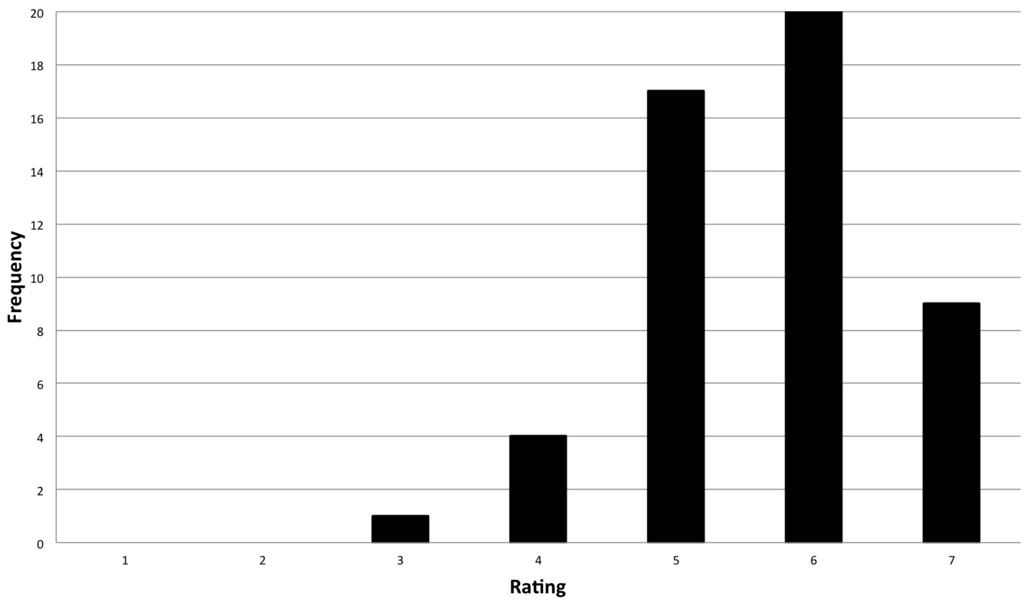

Figure 6.3: Frequency Distribution of Participant Ratings for the Realism of the Simulated Concert Hall

6.3.2 OPEN-ENDED RESPONSES

Although the rated responses were positive the comments on the realism of the simulated concert hall gave insight into the shortcomings of the headset in simulating the concert hall. Most of the participants provided feedback to this question. By far the most common response concerned the intensity of the reverberation or reflections. The participants once again used different terms to describe what the author believes to be the intensity of the simulated virtual acoustic space. These are the most used terms used by the participants:

- Wetness
- Echo
- Liveliness
- Intensity

The author will use the term *wetness* to encompass all the above terms. 17 participants suggested that the simulated concert hall was more *wet* than what they would expect from the actual hall and one participant stated that the simulated hall was louder than expected. Furthermore, three participants suggested that the simulated hall sounded like an empty Endler Hall. Many of the musician's comments, therefore, imply a potentially too *wet* virtual hall. Further comments concerning the exaggerated *wetness* of the hall include that the perception of the hall is more that of an audience member than a performer and that breath sounds, from a clarinetist, brought about a too intense response from the virtual hall.

Additional responses include a perceived exaggerated resonance, suggested by three participants, two of which were cellists. A slight delay in the responses of the virtual hall was noted by two participants, who were both professional musicians. The rest of the responses were less consistent and were deemed less significant by the author.

6.3.3 CONCLUSION AND DISCUSSION

Although the rated responses show positive results, the open-ended questions seem to indicate that the balance of wet/dry in the system was too high across most of the instrument groups. Since the aim of the virtual acoustic headset is to prepare musicians for performances, the balance should be adjusted to recreate the balance of the real venue as accurately as possible, preferably of a full concert hall, as some of the participants of this study noted that the simulated hall sounded like an empty concert hall.

The balance will need to best fit all instrument groups, seeing that the balance between perceived direct and reverberant sound energy in a real hall will differ for different instrument groups. In a real hall, the closer musicians get to their instruments, the stronger the proportion of direct to reverberant energy will be perceived. With the virtual acoustic headset, however, the balance between wet/dry remains constant since the dry signal 'feeds' the wet signal.

The short delay that was noted by two participants can be solved by implementing a more powerful processor for the convolution of the virtual hall.

6.4 PRACTISING WITH A VIRTUAL ACOUSTIC HEADSET

48 of the 51 participants answered yes to question 4, which asked whether musicians can imagine practicing in a virtual acoustic environment with a

similar headset to the one used in the research study. This on its own suggests, as proposed by the author, that a virtual acoustic headset would be beneficial for musician's practice.

The responses to the open-ended question provided further understanding into what purpose and benefits such a virtual headset would have for musicians in their practice rooms. Many of the responses validate arguments provided by the author in chapter 1. The author will use his discretion to discuss the most important and common responses provided by the musicians. The responses are discussed under headings that are believed to best categorize the responses.

6.4.1 PERFORMANCE PRACTICE

Although most of the participants responded positively to using a virtual acoustic headset, it is important to note that several of these participants mentioned they would not always use the headset. The headset would, therefore, mostly be used for 'performance practice' or rehearsal purposes by these participants. Small room acoustics were suggested to be preferable for other aspects of practice, such as "accuracy" and "tone quality".

6.4.2 ACCESSIBILITY

An important response, which was brought up by eight of the participants, concerned the accessibility of concert halls. It was suggested that musicians do not always have access to a concert hall. One participant commented that with a virtual acoustic headset one can practice in a performance venue more often than would be viable in the actual performance venue. Another participant commented that it is not always convenient to book a performance venue.

6.4.3 PREPARATION

The leading theme of the responses to the open-ended questions concerned the headset's potential in helping the musicians prepare for the acoustics of a concert hall. The participants consistently suggested that the headset can help in adapting to the real venue, since playing techniques need to change in different acoustic environments. One professional pianist, for instance, stated that the headset would help the pianist decide on when to use the pedal. Other participants mentioned that their articulation changes with the headset as it would in the real concert hall, or that the headset helps the musician decide on musical aspects of a composition. Two of the participants also suggested that when playing with the virtual acoustic headset they are in "performance-mode". One participant mentioned that this helps to focus the musician more on mistakes.

6.4.4 ENJOYMENT

Another commonly occurring theme in the responses of the musicians was that practicing with the headset would be more enjoyable than practicing in an ordinary practice room. A total of nine participants provided responses that the author believed to be related to enjoyability. The terms used by the musicians to describe practicing with a virtual acoustic headset include:

- Encouraging
- Inspiring
- Increased Self-Esteem
- Increased Confidence
- Nicer
- Less of a Burden

Furthermore, two of the participants stated that they would practice more when wearing a virtual acoustic headset. It should be noted that the participants commenting on increased enjoyment were of varying experience levels, including three professional musicians and two first year students. Increased enjoyment is, therefore, believed to be important for all levels of musicians. One participants, however, suggested that playing with the headset is too enjoyable and would, therefore, prefer to not practice with a virtual acoustic headset.

6.4.5 NEGATIVE RESPONSES AND SUGGESTIONS

The two other participants that responded negatively to potentially using a virtual acoustic headset both criticized the comfortability of the headset. Furthermore, the one participant stated that the acoustics are not similar enough to the real hall and that the headset does not provide a visual impression of the hall. One of the professional musicians also similarly suggested that, apart from acoustics, there are more factors to consider in recreating a concert hall, including the visual impression. Nevertheless, this participant still reacted positively to using a virtual acoustic headset.

A few participants suggested some improvements to the specific headset used for the research study. One musician remarked that the headset should be wireless to reduce obstruction and another commented that it should be smaller. Although this relates to question 2, the comfortability of the headset, it is noted here since these impressions might influence whether or not the participant would use a virtual acoustic headset. Additionally, one of the participants mentioned that it would be useful to have different virtual venues to choose from. Another participant suggested that the settings of the virtual acoustic venue should be adjustable.

6.5 EXPERIENCE LEVELS

The mean ratings for participant sub-groups of different experience levels were calculated and are shown in table 6.4. The mean ratings provided by the students and professional show to be similar, with the professionals providing mean ratings for the three questions of 5.0, 5.8 and 5.7 respectively, and the student giving mean ratings of 4.8, 5.9 and 5.6.

Experience Level	**Participants**	**Question**		
		1	2	3
Professional	11	5.0	5.8	5.7
Students	40	4.8	5.9	5.6
Total	**51**	**4.9**	**5.8**	**5.6**

Table 6.4: Mean Ratings of Participants to the Questionnaire According to Experience Level

6.5.1 CONCLUSION AND DISCUSSION

The author had expected the more experienced musicians to provide lower ratings. The calculated mean ratings do not, however, suggest this trend, since the professional musicians provided very similar ratings to the students overall. This implies that a virtual acoustic headset can potentially be utilized for beneficial use by both music students as well as professional musicians.

CHAPTER 7

Conclusion

7.1 SUMMARY OF FINDINGS

In this thesis the author investigated methods of simulating the acoustics of a performance venue, specifically a concert hall, in small acoustic environments to help musicians better prepare for performances. The problems of small room acoustics were discussed and the acoustic properties were found to be considerably different to those of a typical performance venue, such as a concert hall, and consequently affect performance attributes of musicians. Electro-acoustic enhancement systems, which have been developed for reverberation enhancement in acoustically dry venues were examined but found to be primarily applicable for installation in larger venues, with the exception of the Wenger Corporation, who successfully developed a system that simulates larger acoustic environments in the practice room with microphones, a reverberation processor and loudspeakers. Implementation of this system was, however, found to be expensive. For this reason, as well as other limitations of such a system the author chose to investigate the potential of headphones for accurately simulating larger acoustic environments. The author found little to no previous research that investigates the effectiveness of such a system for musicians' practice.

For the most realistic acoustic simulation it was decided that microphones should be integrated with the headphones. The author found that integrated headsets have been designed and tested for applications in augmented reality audio ARA, where the surrounding environment or pseudo-acoustic environment is reproduced and, additionally, virtual audio components can be superimposed. The author used a similar approach to design a prototype headset that attempted to both: accurately reproduce the pseudo-acoustic environment; and simulate a virtual acoustic environment. In order to accurately reproduce the pseudo-acoustic environment the attenuation properties of the headphones needed to be considered and compensated for by means of sending a filtered microphone signal into the headphones. The simulated concert hall was triggered using convolution reverberation software, which required the acquisition of a sine-sweep response obtained in the Endler hall of the Stellenbosch Music Department.

The prototype was tested on musicians in order to determine the effectiveness of the headset in simulating the acoustics of a concert hall. A survey-based research design was used, which relied on participants responses to both closed-ended questions, rated along a 7-point numbered scale, and open-ended questions in an interview format. The sample group consisted of classical musicians, both students and professionals, participating in the Stellenbosch International Chamber Music Festival (SICMF) and was subdivided into instrument groups. The questions covered four aspects of the headset: its accuracy in reproducing the pseudo-acoustic environment, its comfortability, the realism of the simulated concert hall and whether musicians could imagine practicing with a similar headset. The rated responses were positive, with all the mean ratings to the three closed-ended questions being above the middle value, and 48 of the 51 participants having responded positively to practicing with a similar virtual acoustic headset.

Through the open-ended responses it was found that musicians noted a decreased perception of *presence* while wearing the headset. An increased perception of resonances was also perceived, specifically by the clarinet/oboe instrument group, which was believed to be a consequence of the occlusion effect. Open-ended responses to the realism of the simulated concert hall suggested that the level of the virtual concert hall was too high. This can, however, easily be attenuated in future implementations. The responses to the open-ended questions of whether the musicians could imagine using a similar virtual acoustic headset proved very positive and provided further insight into the benefits of such a system for musicians' practice. The headset was believed by many musicians to help in preparation for performances in concert halls and others responded positively by suggesting that practicing is more enjoyable this way.

7.2 CONCLUSIONS

The author predicted, before the onset of the research study that a headset with integrated microphones could accurately simulate the acoustics of a concert hall. The effectiveness and accuracy of the system was believed to be dependent on the subjective responses of the musicians. The author concludes from the positive responses of the musicians that were tested, that a virtual acoustic headset can indeed be used to effectively simulate a concert hall for musicians in practice rooms.

Although the simulated concert hall was believed to be too live by a number of the participants, the realism was still rated notably positive by the participants. The results therefore indicate, not only the potential for implementation of such a system, to which the musicians reacted very positively, but that the virtual acoustic environment reproduced over the headset is perceived to be similar to a real concert hall. The positive responses of the professional musicians, who have experience in playing in concert halls, supports the conclusion that a virtual acoustic headset has benefits for musicians practicing in small acoustic environments. Despite the results, suggesting that the headset's accuracy is partly dependent on the type of instrument being used, the author believes that with refinement of the headset, a virtual acoustic headset can be effectively used for all instrument types.

The specific headset designed for the research study is, to a certain extent, believed to have been effective due to the open-back circumaural headphone type that was used, which allowed the musicians to accurately perceive their instrument with minimal attenuation compensation required. The open-back design is also less prone, than other headphones types, to the occlusion effect. Since the comfortability ratings for the headset were also high, the author is of the opinion that open-backed circumaural headphones, similar to the ones used in the study, are an appropriate choice for a virtual acoustic headset. Further research can, however, be done to test the relative effectivity of different headphone types for the purpose of simulating concert hall acoustics. Regardless of the headphones type, high quality drivers are believed to be necessary to ensure an accurate reproduction of both the dry signal as well as the simulated concert hall.

Further improvement to the headset design is, however, necessary to reduce the remaining occlusion effect, which has a negative effect on the accuracy of the headset in reproducing the pseudo-acoustic environment for any musicians who rely on their mouth for sound production. This was perceived especially by the clarinetists. The pseudo-acoustic environment can further be improved by refinement of the compensation filter, since musicians noted differences in their instruments' tonal quality.

7.3 LIMITATIONS

7.3.1 LIMITATIONS OF THE HEADSET

The binaural sweep test was limited in its microphone technique, since a true binaural dummy head was not available. Future systems should, therefore, make use of high quality binaural dummy heads. For the application of circumaural headsets, such as the one used in this thesis, further filters could also be implemented to allow for the most accurate binaural representation of the virtual hall. For the purpose of this thesis, which was one of the first to test the potential of a virtual acoustic headset for musicians, the techniques used to obtain the impulse response of the hall were sufficient.

The processing power of the convolution plugin used was, however, limited and for the most accurate representation of the concert hall, the latency should be minimal. The simulation of the concert hall acoustics can, therefore, be improved by means of more powerful specialized processors.

The headset design has some limitations that prevent an absolutely realistic acoustic simulation of a concert hall. For the virtual acoustic headset, the balance between the dry signal and the simulated virtual acoustic space remained constant, since a set dry/wet ratio was used. In a real concert hall, however, the musicians' perception of this balance are dependent on how far away they are from their respective instruments. Therefore, in a real hall, the further the musicians are from their instruments, the more they will perceive the acoustics of the hall. The headset's balance of dry/wet, for this reason, needs to be set to approximate the average musician's perception of the hall's response. In order to improve the accuracy of the hall's response, the headset could include different balance settings that approximate specific instruments.

Another limitation is the fact that the microphones only pick up a limited frequency response of the instrument, since they are placed closely to the instrument. The artificial reflections of the convolution reverberation are, therefore, by a limited frequency bandwidth of the instruments. In a real concert hall, however, the reflections are indicative of the overall frequency response of the instrument.

It should also be noted that the headset lacks accuracy in that the virtual acoustic environment moves with the user. The user, therefore, always faces the virtual audience. For the application of this thesis, this is not necessarily a problem, as the author wished to allow mobility of the user and not restrict the user to a certain location in the room. For a more accurate virtual concert hall environment, the author, however, suggests further research in head-tracking technology for the purpose of concert hall simulation for musicians.

7.3.2 LIMITATIONS OF THE STUDY

The data obtained from the study is purely subjective and can not surely determine the effectiveness of the system. The fact that the musicians could not directly compare the actual hall to the virtual hall means that the responses are not as reliable. It is, therefore, important that when determining the correct reverberation balance of the system, that additional accurate listening tests are conducted.

The number of participants per instrument group, being seven to eight people, is limited to make accurate comparisons of rankings between the different instrument groups and to deduce any generalizations about musicians's responses. In comparison to previous research the total sample group size was, however, large.

The interview format that was used was conducted by the author who has not undergone interview training. This can potentially have biased the results, although the author made his best effort to not bias the responses of the participants.

7.4 RECOMMENDATIONS FOR IMPLEMENTATION

The study found that musicians would mainly use the headset for performance practice and that it is not beneficial for all types of practice. For this reason, the author recommends the implementation of the headset in only limited number of dedicated practice rooms. This should help to force students into using the headset for beneficial purposes, rather than just for enjoyment. Of course the number of headsets and dedicated rooms should be sufficient to be accessible for musicians when they have need of it. Ordinary practice rooms that have been optimized acoustically for the respective size, should therefore still be used for ordinary practice routines. It is believed that with strict implementation rules the headset can become an ordinary part of a music student's or musician's preparations for a performance.

As recommended previously in this thesis, the natural acoustic environment in which a headset is installed should be rather dry as the natural reflections should not interfere with those simulated by the headset. To optimize the accuracy of the simulated concert hall and to reduce the potential risk of hearing damage, there should be criteria in place for the natural room's acoustic loudness levels.

The headset should not contain wires as this would be obstructive to the musician. A wireless design could be achieved by having a fully integrated headset that contains a built-in preamplifier, convolution processor and headphone amplifier. This is only plausible for a circumaural headset, such as the one used in this research study. Alternatively a separate processing unit could be installed in the practice room, which wirelessly interacts with the headset. This would, however, require both wireless microphones technology as well as wireless headphones.

The headset should be preloaded with different impulse responses so that the musicians can choose in which virtual venue to practice. Sine sweep responses, using strict criteria, will need to be obtained from all these venues. The balance of the virtual acoustic hall relative to the dry signal of the natural room will need to be determined and set on the device. No parametric control should be given to the user, since the purpose of the headset is to accurately simulate an acoustic space rather than purely make practicing more enjoyable.

Appendix A

Order of Procedures

1. First I will provide an overview of the procedures to the participant, making sure he/she understands what is expected of him/her in the research study. This will include reading the information sheet.

2. I will then obtain the necessary information from the participant, such as their instrument, whether or not they are professional. If they are a student I will ask in which year they are and whether or not they are specializing in performance.

3. The participant will then be asked to sign the consent form.

4. I will then ask the participant to play a musical excerpt without headphones while paying attention to the acoustic characteristics of the natural environment.

5. I will then help the participant to put on the headset. The headset will contain no reverberation, only the compensation filter. This will be communicated to the participant. I will then ask the participant to once again perform the excerpt.

6. The participant will then need to answer the following questions. I will allow the participant to elaborate and provide reason for their answer.

   How accurately can you hear your surrounding environment? 1= completely different to not wearing headphones 7= identical to not wearing headphones

1	2	3	4	5	6	7

   How comfortable is the Headset? 1= very uncomfortable 7= very comfortable

1	2	3	4	5	6	7

7. I will then switch on the reverberation plugin which simulates the acoustics of the Endler Hall. The participant will once again be asked to play the excerpt.

8. The participant will then need to answer the following question and provide reasoning if desired.

   How realistic does the concert hall sound? 1= very unrealistic 7= very realistic

1	2	3	4	5	6	7

9. I will help the participant to take off the headset.
10. I will then ask the following question:

    Could you imagine practicing in a virtual acoustic environment with a similar headset? Clarify that future headset will contain smaller microphones and possibly wireless functionality.

YES	NO

Provide your Reasoning:

____

____

____

Appendix B

Questionnaire Results

Participant	Group	Instrument	Experience	Year	Specialized	Question 1	Question 2	Question 3	Question 4
1	Violin/Viola	Violin	Student	2		3	7	5	yes
2	Violin/Viola	Violin	Student	2		4	7	7	yes
3	Violin/Viola	Violin	Student	6	yes	5	7	4	yes
4	Violin/Viola	Violin	Student	5	no	5	4	5	yes
5	Violin/Viola	Violin	Student	1		5	7	7	yes
6	Violin/Viola	Violin	Professional			3	6	6	yes
7	Violin/Viola	Viola	Professional			5	6	6	yes
8	Cello	Cello	Student	1		5	5	6	yes
9	Cello	Cello	Student	2		3	5	5	yes
10	Cello	Cello	Student	5	no	2	5	6	yes
11	Cello	Cello	Student	1		5	6	6	yes
12	Cello	Cello	Student	4	yes	6	5	7	yes
13	Cello	Cello	Student	4	yes	5	5	4	yes
14	Cello	Cello	Professional			6	6	5	yes
15	Clarinet/Oboe	Clarinet	Professional			2	6	6	yes
16	Clarinet/Oboe	Clarinet	Professional			5	5	6	yes
17	Clarinet/Oboe	Clarinet	Student	3	yes	3	7	5	yes
18	Clarinet/Oboe	Clarinet	Student	2		3	4	3	no
19	Clarinet/Oboe	Clarinet	Student	4	no	4	6	5	yes
20	Clarinet/Oboe	Clarinet	Professional			6	7	6	yes
21	Clarinet/Oboe	Oboe	Student	6	yes	6	4	5	yes
22	Clarinet/Oboe	Oboe	Student	3	yes	5	7	6	yes
23	Flute	Flute	Student	3	yes	4	5	7	yes
24	Flute	Flute	Student	3	no	5	7	5	yes
25	Flute	Flute	Student	5	yes	7	6	6	yes

26	Flute	Flute	Professional			4	7	6	yes
27	Flute	Flute	Student	3	yes	3	5	6	yes
28	Flute	Flute	Professional			5	6	7	yes
29	Flute	Flute	Student	4	yes	5	6	6	no
30	Trumpet/Trombone	Trumpet	Student	4	yes	5	7	5	yes
31	Trumpet/Trombone	Trumpet	Professional			6	6	5	yes
32	Trumpet/Trombone	Trumpet	Student	3	yes	5	7	7	yes
33	Trumpet/Trombone	Trumpet	Student	1		2	7	5	yes
34	Trumpet/Trombone	Trumpet	Student	1		5	7	5	yes
35	Trumpet/Trombone	Trombone	Student	2		6	7	7	yes
36	Trumpet/Trombone	Trombone	Student	4	yes	6	4	5	yes
37	Trumpet/Trombone	Trombone	Student	2		5	5	6	yes
38	Piano	Piano	Student	2		5	4	5	yes
39	Piano	Piano	Student	1		5	6	6	yes
40	Piano	Piano	Student	5	yes	6	5	5	yes
41	Piano	Piano	Student	5	yes	4	7	5	yes
42	Piano	Piano	Professional			6	4	4	yes
43	Piano	Piano	Professional			7	5	6	yes
44	Piano	Piano	Student	4	yes	6	5	4	yes
45	Guitar	Guitar	Student	3	yes	4	6	6	yes
46	Guitar	Guitar	Student	4	yes	5	7	7	yes
47	Guitar	Guitar	Student	1		7	6	6	yes
48	Guitar	Guitar	Student	4	no	6	5	5	no
49	Guitar	Guitar	Student	1		5	6	7	yes
50	Guitar	Guitar	Student	3	no	6	7	6	yes
51	Guitar	Guitar	Student	4	no	7	6	6	yes

Appendix C

Consent Form

UNIVERSITEIT•STELLENBOSCH•UNIVERSITY
jou kennisvennoot • your knowledge partner

STELLENBOSCH UNIVERSITY
CONSENT TO PARTICIPATE IN RESEARCH

An Investigation into the Effectiveness of using Headphones with Integrated Microphones to Simulate Concert Hall Acoustics for Musicians in Small Acoustic Environments

You are asked to participate in a research study conducted by Tim ter Huurne (currently enrolled in a Master of Philosophy in Music Technology) at the Music Department of Stellenbosch University.
The results obtained from this study will be contributed to the thesis of the aforementioned investigator. You were selected as a possible participant in this study as you either a music student or a professional musician that plays an instrument fulfilling the requirement of participating in this research study.

1. PURPOSE OF THE STUDY

The study aims to assess the effectiveness of simulating concert hall acoustics in a small practice room using headphones that contain integrated microphones.

2. PROCEDURES

If you volunteer to participate in this study, you will be asked to do the following things:

You will need to play a small excerpt of a music piece during the experiment on three occasions. This excerpt should not be too technically demanding and should be approximately 30 seconds long. It is preferable that this excerpt is memorized.
You will be required to wear a pair of headphones during the progression of the study. The headphones will contain two microphones that have been attached to the headphones externally. The investigator will be operating his laptop and sound equipment, which is needed to control that, which is being reproduced over the headphones. The research study will not take more than 15 minutes of your time.
First you will be asked to play your prepared excerpt without the use of headphones. The investigator will ask you to pay attention to specific properties of the acoustics of the room.
You will then need to perform the excerpt again while wearing the headset. After having performed the excerpt you will be asked questions, about the effectiveness of the system. These questions will require a subjective rating along a scale from 1-7. At the end of the study you will be offered the chance to give your opinion on anything else with regards to the system.

3. POTENTIAL RISKS AND DISCOMFORTS

Pre-experiments done by the investigator will ensure that you will not be at risk or discomfort from the levels reproduced by the headphones. Pre-experiments will also minimize the risk of feedback occurring between the microphones and the speakers of the headphones.

The investigator will allow enough time before your scheduled time slot to ensure that the equipment and instrumentation is setup and ready, so as not to waste your time. Time intervals between successive participants will therefore be of sufficient length. You are at a very low risk of psychological harm and the interview requires no personal information. The investigator will be accommodating and make an effort to ensure you feel comfortable throughout the study.

4. POTENTIAL BENEFITS TO SUBJECTS AND/OR TO SOCIETY

You will most likely not benefit directly from the research study. The study does, however, have potential benefits for musicians in the future, as it investigates electro-acoustic means of improving musicians' practicing.

5. PAYMENT FOR PARTICIPATION

You will not receive any payment for your participation in the research study.

6. CONFIDENTIALITY

Any information that is obtained in connection with this study and that can be identified with you will remain confidential and will be disclosed only with your permission or as required by law. Confidentiality will be maintained by means of not including any personal information in the paper. The investigator will also use his discretion and not communicate your individual results with others. The results of the study will, however, be used for the investigator's thesis and potentially be published in an academic journal.

7. PARTICIPATION AND WITHDRAWAL

You can choose whether to be in this study or not. If you volunteer to be in this study, you may withdraw at any time without consequences of any kind. You may also refuse to answer any questions you do not want to answer and still remain in the study. The investigator may withdraw you from this research if circumstances arise which warrant doing so.

8. IDENTIFICATION OF INVESTIGATORS

If there are any questions or concerns about the research, please feel free to contact Tim ter Huurne [timterhuurne1@gmail.com; +27 74 124 8389] or his supervisor Gerhard Roux [mail@gerhardroux.com].

9. RIGHTS OF RESEARCH SUBJECTS

You may withdraw your consent at any time and discontinue participation without penalty. You are no waiving any legal claims, rights or remedies because of your participation in this research study. If there are any questions regarding your rights as a research subject, contact Ms Maléne Fouché [mfouche@sun.ac.za; 021 808 4622] at the Division for Research Development.

SIGNATURE OF RESEARCH SUBJECT OR LEGAL REPRESENTATIVE

The information above was described to me by Tim ter Huurne in [*Afrikaans/English/Xhosa/other]* and I am in command of this language or it was satisfactorily translated to me. I was given the opportunity to ask questions and these questions were answered to my satisfaction.

I hereby consent voluntarily to participate in this study. I have been given a copy of this form.

____

Name of Subject/Participant

____

Name of Legal Representative (if applicable)

______________________________________ ______________

Signature of Subject/Participant or Legal Representative **Date**

SIGNATURE OF INVESTIGATOR

I declare that I explained the information given in this document to ___________________. He/she was encouraged and given ample time to ask me any questions. This conversation was conducted in English and no translator was used.

______________________________________ ______________

Signature of Investigator **Date**

Appendix D

Ethical Approval

UNIVERSITEIT·STELLENBOSCH·UNIVERSITY
jou kennisvennoot • your knowledge partner

Approval Notice
New Application

27-Mar-2017
Ter Huurne, Tim TH

Proposal #: SU-HSD-004127

Title: **An Investigation into the Effectiveness of using Headphones with Integrated Microphones to Simulate Concert Hall Acoustics for Musicians in Small Acoustic Environments**

Dear Mr Tim Ter Huurne,

Your **New Application** received on **24-Feb-2017**, was reviewed
Please note the following information about your approved research proposal:

Proposal Approval Period: **17-Mar-2017 -16-Mar-2020**

Please take note of the general Investigator Responsibilities attached to this letter. You may commence with your research after complying fully with these guidelines.

Please remember to use your **proposal number** (**SU-HSD-004127**) on any documents or correspondence with the REC concerning your research proposal.

Please note that the REC has the prerogative and authority to ask further questions, seek additional information, require further modifications, or monitor the conduct of your research and the consent process.

Also note that a progress report should be submitted to the Committee before the approval period has expired if a continuation is required. The Committee will then consider the continuation of the project for a further year (if necessary).

This committee abides by the ethical norms and principles for research, established by the Declaration of Helsinki and the Guidelines for Ethical Research: Principles Structures and Processes 2004 (Department of Health). Annually a number of projects may be selected randomly for an external audit.

National Health Research Ethics Committee (NHREC) registration number REC-050411-032.

We wish you the best as you conduct your research.

If you have any questions or need further help, please contact the REC office at 218089183.

Included Documents:

REC: Humanities New Application

Sincerely,

Clarissa Graham
REC Coordinator
Research Ethics Committee: Human Research (Humanities)

List of References

Adler, S. 2002. *The Study of Orchestration.* 3rd edition. New York, USA: W. W. Norton & Company, Inc.

Ahnert, W. *&* Tennhardt, H.-P. 2008. Acoustics for auditoriums and concert halls. In G. Ballou (ed.), *Handbook for Sound Engineers*, 4th edition, chapter 7, pages 145–200. MA, USA: Taylor & Francis.

Albrecht, R., Lokki, T. *&* Savioja, L. 2011. A mobile augmented reality audio system with binaural microphones. In *Proceedings of Interacting with Sound Workshop: Exploring Context-Aware, Local and Social Audio Applications*, volume 978, pages 7–11. Stockholm, Sweden: ACM.

Algazi, V.R. *&* Duda, R.O. 2011. Headphone-based spatial sound: Exploiting head motion for immersive communication. *IEEE Signal Processing Magazine*, 28(1):33–42.

Ando, Y. *&* Cariani, P. 2009. *Auditory and Visual Sensations.* New York: Springer.

Ballou, J. 2011. Open-ended question.

Blankenship, J., Fitzgerald, R.B. *&* Lane, R. 1955. Comparison of objective and subjective observations on music rooms. *The Journal of the Acoustical Society of America*, 27(4):774–780.

Bless, C., Higson-Smith, C. *&* Sithole, S.L. 2013. *Fundamentals of Social Research Methods: An African Perspective.* 5th edition. Cape Town: South Africa: Juta and Company LTD.

Borwick, J. 1999. Acoustics and acoustic devices. In M. Talbot-Smith (ed.), *Audio engineer's reference book*, 2nd edition, chapter 2. Oxford, UK: Taylor & Francis.

Case, A.U. 2011. *Mix Smart: Pro Audio Tips For Your Multitrack Mix.* Waltham: Focal Press.

Christian, A. *&* Gade, S. 2015. Classical musicians' perception of room acoustic conditions. *Psychomusicology: Music, Mind, and Brain*, 25(3):232–235.

Clark, R. 2011. *Mixing, Recording, and Producing techniques of the Pros*. 2nd edition. Boston, MA: Course Technology PTR.

Clarke, T., Mellor, D. & Rumsey, F. 1999. Digital equipment. In M. Talbot-Smith (ed.), *Audio engineer's reference book*, 2nd edition, chapter 4. Oxford, UK: Taylor & Francis.

Davis, G. & Jones, R. 1989. *The Sound Reinforcement Handbook*. Milwaukee: Hal Leonard Corporation.

Denscombe, M. 2010. *The Good Research Guide for Small-Scale Social Research Projects*. 4th edition. Berkshire, England: The McGraw-Hill Companies.

Dickreiter, M., Dittel, V., Hoeg, W. & Wöhr, M. 2008. *Handbuch der Tonstudiotechnik*. Walter de Gruyter.

Dove, S. 2008. Consoles and computers. In G. Ballou (ed.), *Handbook for Sound Engineers*, 4th edition, chapter 25, pages 817–994. MA, USA: Taylor & Francis.

Everest, F.A., Pohlmann, K.C. & Books, T. 2009. *The Master Handbook of Acoustics*, volume 5. New York: McGraw-Hill.

Fletcher, N.H. & Rossing, T.D. 1998. *The Physics of Musical Instruments*. 2nd edition. New York, USA: Springer.

Griesinger, D. 1990. Improving room acoustics through time variant synthetic reverberation. In *Audio Engineering Society Convention 90*. Audio Engineering Society, Waltham: Lexicon Inc.

Hardiman, A. 2009. Electronic acoustic enhancement systems: Part one. *Lighting and Sound America*, pages 88–96.

Härmä, A., Jakka, J., Tikander, M., Karjalainen, M., Lokki, T., Hiipakka, J. & Lorho, G. 2004. Augmented reality audio for mobile and wearable appliances. *Journal of the Audio Engineering Society*, 52(6):618–639.

Hosken, D. 2011. *An Introduction to Music Technology*. New York, USA: Taylor & Francis.

Howard, D.M. & Angus, J. 2009. *Acoustics and Psychoacoustics*. Oxford: Taylor & Francis.

Huber, D.M. 2009. *From Demo To Delivery: The Process of Production*, chapter The Recording Session, pages 73–106. Burlington, USA: Focal Press, USA.

Iarossi, G. 2006. *The Power of Survey Design: A User's Guide for Managing Surveys, Interpreting Results, and Influencing Respondents*. Washington DC, USA: The World Bank.

Inglis, S. 2007. Living with latency.
Available at: `http://www.soundonsound.com/techniques/living-latency`

Jones, D. 2008. Small room acoustics. In G. Ballou (ed.), *Handbook for Sound Engineers*, 4th edition, chapter 6, pages 126–144. MA, USA: Taylor & Francis.

Jones, G. 1990. *Sound Reinforcement Handbook*. California: Yamaha Corporation.

Jorgensen, H. 2014. Western classical music studies in universities and conservatoires. In I. Papageorgi & G. Welch (eds.), *Advanced Musical Performance: Investigations in Higher Education Learning*, chapter 1, pages 3–20. Surrey: Ashgate Publishing Limited.

Kato, K., Ueno, K. & Kawai, K. 2008. Musicians' adjustment of performance to room acoustics, part iii: Understanding the variations in musical expressions. *Journal of the Acoustical Society of America*, 123(5):3610.

Kondo, K., Anazawa, N. & Kobayashi, Y. 2013. Characteristics comparison of two audio output devices for augmented audio reality. In *Signal and Information Processing Association Annual Summit and Conference (APSIPA), 2013 Asia-Pacific*, pages 1–6. IEEE.

Kuo, S.M., Mitra, S. & Gan, W.-S. 2006. Active noise control system for headphone applications. *IEEE Transactions on Control Systems Technology*, 14(2):331–335.

Lamberty, D. 1980. Music practice rooms. *Journal of Sound and Vibration*, 69(1):149–155.

Lester, M. & Boley, J. 2007. The effects of latency on live sound monitoring. In *Audio Engineering Society Convention 123*. Audio Engineering Society.

Lewcock, R., Pirn, R. & Meyer, J. 2001. Acoustics: I. room acoustics. In S. Sadie & T. John (eds.), *The New Grove Dictionary of Music and Musicians*, volume 1, chapter 1, pages 70—96. London: Macmillan Publisher Limited.

Lindeman, R.W., Noma, H. & de Barros, P.G. 2007. Hear-through and mic-through augmented reality: Using bone conduction to display spatialized audio. In *Mixed and Augmented Reality, 2007. ISMAR 2007. 6th IEEE and ACM International Symposium on*, pages 173–176. IEEE.

Lokki, T. & Hiipakka, J. 2001. A time-variant reverberation algorithm for reverberation enhancement systems. In *Proceedings of COST G-6 Conference on Digital Audio Effects (DAFX-01), Limerick, Ireland*, pages 28–32.

Lokki, T., Nummela, J. & Lahti, T. 2000. An electro-acoustic enhancement system for rehearsal rooms. In *Proc. EAA Symposium on Architectural Acoustics*, pages 16–20.

Loy, G. 2006. *Musimathics: The Mathematical Foundations of Music*, volume 2. Cambridge: The MIT Press.

MacDonald, J.A., Henry, P.P. & Letowski, T.R. 2006. Spatial audio through a bone conduction interface: Audición espacial a través de una interfase de conducción ósea. *International journal of audiology*, 45(10):595–599.

Mcbride, M., Patrick, R., Letowski, T.R. *&* Tran, P. 2010. Bone conduction intelligibility: Headset comparison study. In A. Johnson *&* J. Miller (eds.), *Proceedings of the 2010 Industrial Engineering Research Conference*, pages 1–7.

Meyer, J. 2009. *Acoustics and the Performance of Music: Manual for Acousticians, Audio Engineers, Musicians, Architects and Musical Instrument Makers.* Springer Science & Business Media.

MeyerSound. 2015. *Operating Instructions: CAL Column Array Louspeaker.* Meyer Sound, Berkeley, USA.

Mueller, F. *&* Karau, M. 2002. Transparent hearing. In *CHI'02 Extended Abstracts on Human Factors in Computing Systems*, pages 730–731. ACM.

Newell, P.R. 2008. *Recording studio design.* 2nd edition. Oxford, UK: Taylor & Francis.

Newell, P.R. 2009. The studio environment. In D. Self (ed.), *Audio Engineering Explained*, chapter 21, pages 615–628. MA, USA: Taylor & Francis.

Niquette, P. 2006. Hearing protection for musicians. *Hearing Review*, 13(3):52.

O'Brien, I., Driscoll, T. *&* Ackermann, B. 2013. Sound exposure of professional orchestral musicians during solitary practice. *Journal of the Acoustical Society of America*, 134(4):2748–2754.

Osman, R. 2010. Designing small music practice rooms for sound quality. In *Proceedings of 20th International Congress on Acoustics, Sydney, Australia.*

Pätynen, J. 2007. Virtual acoustics in practice rooms. Ph.D. thesis, Helsinki University of Technology.

Phillips, S.L. *&* Mace, S. 2008. Sound level measurements in music practice rooms. *Music Performance Research*, 2(1):36–47.

Poletti, M. 2011. Active acoustic systems for the control of room acoustics. *Building acoustics*, 18(3-4):237–258.

Rämö, J. *&* Välimäki, V. 2012. Digital augmented reality audio headset. *Journal of Electrical and Computer Engineering*, 2012(457374).

Ranjan, R. *&* Gan, W.-S. 2015. Natural listening over headphones in augmented reality using adaptive filtering techniques. *Audio, Speech, and Language Processing, IEEE/ACM Transactions on*, 23(11):1988–2002.

Ross, M. 2004. The occlusion effect–what it is, and what to do about it. Available at: `http://www.hearingresearch.org/ross/hearing_loss/the_occlusion_effect.php`

Ruane, J.M. 2005. *Essentials of Research Methods: A Guide to Social Science Research.* Oxford, UK: Blackwell Publishing Ltd.

Rumsey, F. 2001. *Spatial Audio*. Oxford, UK: Taylor & Francis.

Rumsey, F. *&* Kok, B. 2014. Acoustic enhancement systems. *Journal of the Audio Engineering Society*, 62(6):449–453.

Rumsey, F. *&* McCormick, T. 2009. *Sound and Recording*. 6th edition. Oxford, UK: Taylor & Francis.

Sapsford, R. 2007. *Survey Research*. 2nd edition. London, UK: SAGE Publications.

Schaeffer, N.C. *&* Presser, S. 2003. The science of asking questions. *Annual Review of Sociology*, 29:65–88.

Schonstein, D., Ferré, L. *&* Katz, B.F. 2008. Comparison of headphones and equalization for virtual auditory source localization. *Journal of the Acoustical Society of America*, 123(5):3724.

Self, D., Duncan, B., Sinclair, I., Brice, R., Hood, J.L., Singmin, A., Davis, D., Patronis, E. *&* Watkinson, J. 2009. *Audio Engineering: Know It All: Know It All*, volume 1. Newnes.

Sevsay, E. 2013. *The Cambridge Guide to Orchestration*. New York, USA: Cambridge University Press.

Sigismondi, G. 2008. Personal monitor systems. In G. Ballou (ed.), *Handbook for Sound Engineers*, 4th edition, chapter 37, pages 1413–1437. MA, USA: Taylor & Francis.

Szymanski, J. 2008. Small room acoustics. In G. Ballou (ed.), *Handbook for Sound Engineers*, 4th edition, chapter 5, pages 96–124. MA, USA: Taylor & Francis.

Toole, Floyd, E. 2008. *Sound Reproduction: The Acoustics and Psychoacoustics of Loudspeakers and Rooms*. Focal Press, USA.

Ueno, K. *&* Tachibana, H. 2005. Cognitive modeling of musician's perception in concert halls. *Acoustical science and technology*, 26(2):156–161.

Valimaki, V., Franck, A., Ramo, J., Gamper, H. *&* Savioja, L. 2015. Assisted listening using a headset: Enhancing audio perception in real, augmented, and virtual environments. *Signal Processing Magazine, IEEE*, 32(2):92–99.

Valimaki, V., Parker, J.D., Savioja, L., Smith, J.O. *&* Abel, J.S. 2012. Fifty years of artificial reverberation. *IEEE Transactions on Audio, Speech, and Language Processing*, 20(5):1421–1448.

Vorlaender, M. 2000. Acoustic load on the ear caused by headphones. *The Journal of the Acoustical Society of America*, 107(4):2082–2088.

Walker, B.N. *&* Stanley, R.M. 2005. Thresholds of audibility for bone-conduction headsets. In *Proceedings of ICAD 05-Eleventh Meeting of the International Conference on Auditory Display*, pages 218–222. Limirick, Ireland.

Walker, B.N., Stanley, R.M., Iyer, N., Simpson, B.D. & Brungart, D.S. 2005. Evaluation of bone-conduction headsets for use in multitalker communication environments. In *Proceedings of the Human Factors and Ergonomics Society Annual Meeting*, volume 49, pages 1615–1619. SAGE Publications.

Wenger. 2013. *Studio VAE® System: Technical Sheet*. Wenger Corporation, Owatonna, USA.

White, P. 2011. Karma k-micro.
Available at: `http://www.soundonsound.com/reviews/karma-k-micro-silver-bullet`

Woszczyk, W. 2011. Active acoustics in concert halls -a new approach. *Archives of Acoustics*, 36(2):379–393.

Woszczyk, W., Beghin, T., de Francisco, M. & Ko, D. 2009. Recording multichannel sound within virtual acoustics. In *Audio Engineering Society Convention 127*. Audio Engineering Society.

Woszczyk, W., Ko, D. & Jonathan, H. 2013. Architectural acoustics: Session 3aaaa: Virtual concert hall acoustics i. In *Proceedings of Meetings on Acoustics*, volume 19, page 1. Montreal, Canada: Acoustical Society of America.

Zhang, P.X. 2008. Psychoacoustics. In G. Ballou (ed.), *Handbook for Sound Engineers*, 4th edition, chapter 3, pages 41–63. MA, USA: Taylor & Francis.