Perceived Differences between Natural and Convolution Reverberation in 5.0 Surround Sound

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Tiivistelmä – Abstract

This Graduate Thesis investigates the perceived differences between natural and convolution reverberation in surround sound. Two spaces with distinct reverberation times were used for this study. Initially three musical excerpts from three instruments (Cello, Oboe, and Piano) were recorded in a dry studio environment. Then the Impulse Response (IR) of the spaces was captured using two methods: balloon burst and sine sweep. The dry excerpts were then recorded in the spaces to capture the natural reverberation pattern while the IRs were convolved with them to create the artificial reverberation excerpts. A listening test was then conducted using six perceptual scales to rate these 18 excerpts (3 reverberation types, 3 instruments and 2 spaces).

The main findings were that the "artificiality" in the artificial excerpts could be perceived by the listeners. The sine sweep reverberation type was perceived to be more similar to the natural reverberation of the space rather than the balloon burst method. The two artificial reverberation types were better suited to a longer reverberation time and a larger space while being best suited for the Oboe and least suited for the Piano. Though a few interesting insights into the perception of convolution reverberation have been provided, the main success of this thesis is the creation of a methodology that can be used for future perceptual studies in artificial reverberation which aim to investigate and improve its perception.

Asiasanat – Keywords natural & convolution reverberation, surround sound, perceived differences, listening test

Säilytyspaikka – Depository

Muita tietoja – Additional information

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1. Introduction

This thesis investigates perceived difference between natural and convolution reverberation in 5.0 surround sound. Reverberation adds a sense of space to music which has become essential to music listening. It can be argued that listening to music without reverberation makes the process unnatural and not enjoyable. This statement can be justified by the abundance of research conducted in the field of artificial reverberation. With the music studio recording environment being more or less acoustically dead, artificial reproduction of reverberation has been a very popular subject of interest since the advent of commercial music recording.

Since the humble beginnings of artificial reverberation using simple techniques, it has evolved into a significant research area. Reverberation was initially reproduced by placing microphones and loudspeakers in reflective rooms with tiled basements that increased reflections. Suspended glass plates were also used to create a variable reverb environment. The standard and popularity of studios were determined by the different rooms that were built to mimic different reverberation environments.

Following the discovery of analogue spring reverb, the demand for multiple studio reverberation environments was rendered obsolete. It was found that analogue sound signals passed through a spring with a transducer at the other end, added a sense of reverberation to the signal. However the first major artificial analogue reverberation invention was the reverb plate. Reverberation was added to the signal by passing it through an isolated and soundproofed vibrating metal plate. This however did not permit the reverberation to have spatial identity. Therefore plate reverb was used in tandem with tape delay, which solved this issue.

The advent of digital music technology meant that artificial reverb could be produced with the help of signal processing algorithms. This was the next generation of artificial reverberation techniques. This is essentially a top to down approach where the defined algorithm sets the rules to determine the optimum outcome, which in this case is reverberation. Multiple feedback delay circuits were used to create various types of reverb patterns. In as early as 1961, Schroeder discussed methods of producing artificial reverberation that comes very close to its natural counterpart. Other digital signal processing techniques, using various types of filters such as high-pass, low-pass and comb filters, were also applied to improve the quality of artificial reverberation.

Moorer (1979) reviewed artificial reverberation algorithms that were already in use. His review reported work that could be done towards improving the reverberation algorithms. He suggested the use of impulse responses (IRs) of rooms to produce better artificial reverberation by using methods such as multi- dimensional scaling and factor analysis to help decompose the IR. This led to the next generation of producing artificial reverberation using IRs called Convolution Reverberation. This is a bottom to top approach where an example end result is statistically analyzed and processed to produce other similar outcomes. Obtaining the impulse response of a space meant recording an impulse in that space, generally using a starter pistol or bursting a balloon. This impulse response essentially contains the reverberation pattern of the space. This IR of that space is then digitally processed or "convolved" with an input signal which results in an output signal. This resultant output signal is how the input signal would have sounded if recorded in that space. This way the reverb of any space could be captured and added to any audio signal. This method has become very popular since its invention.

An abundance of research in the field of artificial reverberation has resulted in such modern and exciting techniques such as convolution, which enables the capturing of reverberation of any space. Online IR libraries such as Audio Ease and Open Impulse Response Library make available a various range of IRs from famous acoustic halls around the world, such as the Berlin Philharmonie and Sydney Opera House, to the IR of even famous analogue plate reverb units such as EMT 140evolution.

Artificial reverberation has been studied extensively. This thesis is a perceptual study in the field of convolution reverberation in surround sound. There have been perceptual studies conducted in the field of artificial reverberation for stereo sound. A hybrid reverberation algorithm created by Browne (2001) uses both the Impulse Response (IR) of a space while using digital filters to reproduce other aspects of reverberation. A paper by Karjalainen (2001) discusses an algorithm that produces "perceptually late reverberation". Papers by Czyzewski(1990) and Sankiewicz

(1988) also study and evaluate the perceptual aspects of reverberation for stereo sound. Artificial reverberation has also been studied in the field of sound localization. Begault (1992) reported the effects of using artificial "spatial" reverberation on localization judgments. In surround sound, convolution reverberation has been studied closely in field of Ambiophonics (Farina, et.al, 2001). Despite all this research in the field of artificial reverberation, the perception of convolution reverberation has not been studied in surround sound, creating a research gap in this field.

This thesis studies the perceived differences between natural and artificial convolution reverberation types. A perceptual study was conducted where the listeners initially rated 18 excerpts of music according to six different perceptual scales. The excerpts consisted of three different instruments (Cello, Oboe & Piano) with three types of reverberation (one natural and two artificial) recorded in two separate spaces. The excerpts were rated according to 6 perceptual scales. Following this, the listeners were asked to order the excerpts in terms of preference. The goal was to find out the perceived differences between these natural and convolution reverberation types.

Part 2 initially gives background about Natural and Convolution Reverberation. Following this is a review of the literature concerning studies in this field. **Section 2.4** reviews perceptual studies in the field of Artificial Reverberation. In **Section 2.5** the review progresses unto research conducted in artificial reverberation in surround sound. Practical fields such as Ambiophonics, Sound Localization and Speech Intelligibility in which Reverberation and Surround Sound are studied, are discussed in **Section 2.5.2 and 2.5.3**. A short case study about the differences in perception of different surround sound microphone techniques is present in **Section 2.5.4**.

Section 3 details the empirical portion of the thesis. The methodology used for addressing the main research question is explained. The experimental design is first detailed in **Section 3.1**. Following this is the data analysis process, in **Section 4**, where the results are presented according to the type of analysis performed on the data to answer specific research questions. Different tables and figures have been used to clearly show and illustrate the results.

In **Section 5**, these results are discussed in terms of the main research question. Observations and inferences are made and explanations are given where possible for the results. The Conclusion, in **Section 6** follows this, where an overview of this thesis study is given, followed by recommendations for future studies, implications of this study and some closing thoughts.

2. Reverberation

2.1 Natural Reverberation

Reverberation is a natural phenomenon that occurs in an enclosed space due to the sound reflecting off the different boundaries of that space. Sound in an enclosed space passes through different stages:

2.1.1 Direct Sound

This is the sound which travels directly from the source to the listener unaffected by any boundaries in the room. This can be equated to a sound in a free field, which is an area with no reflections of any kind. At this stage in the propagation of sound is like being in an acoustically dead environment. The only delay between the sound generation and the listener hearing it is the distance between the listener and the sound source (Hall, 1991). The figure below illustrates direct sound:

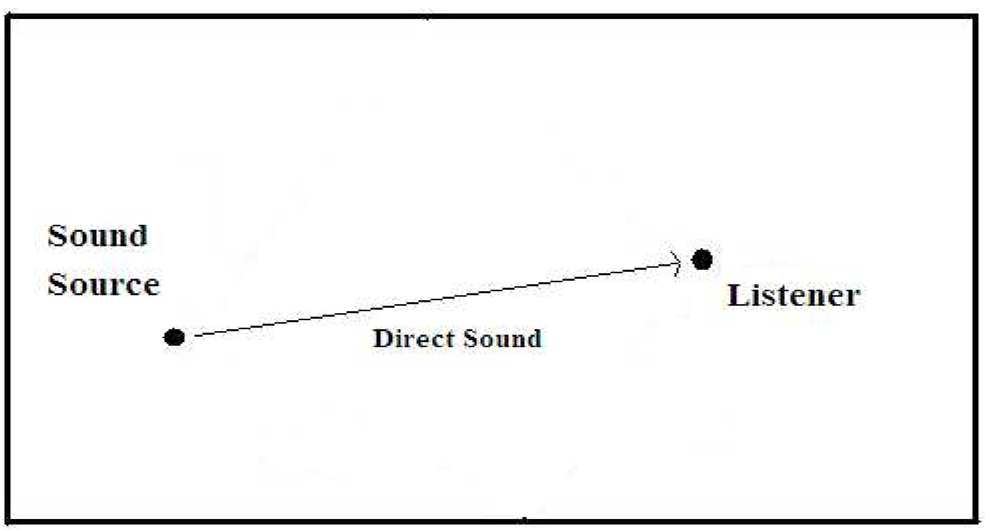


Figure 1: Direct Sound: Sound wave travels directly from the source to the listener's ears.

2.1.2 Early Reflections

Shortly after the direct sound is heard, early reflections are heard. Reflections heard between 5ms to 50ms are considered early reflections. These are reflections of the original sound reflected from the surfaces of the room such as the walls, the floor etc. They are separated both in time and direction from the original direct sound (Hall, 1991). Human hearing is designed to use these reflections as sound localization cues to obtain information about the space we are in. These reflections give us a fair idea about size and shape of room, as well as distance and direction of sound source (Cook, 2001). These reflections are weaker in intensity than the direct sound. This happens because the distance travelled is greater causing the energy to dissipate. Also surfaces absorb the energy. The figure below illustrates early reflections

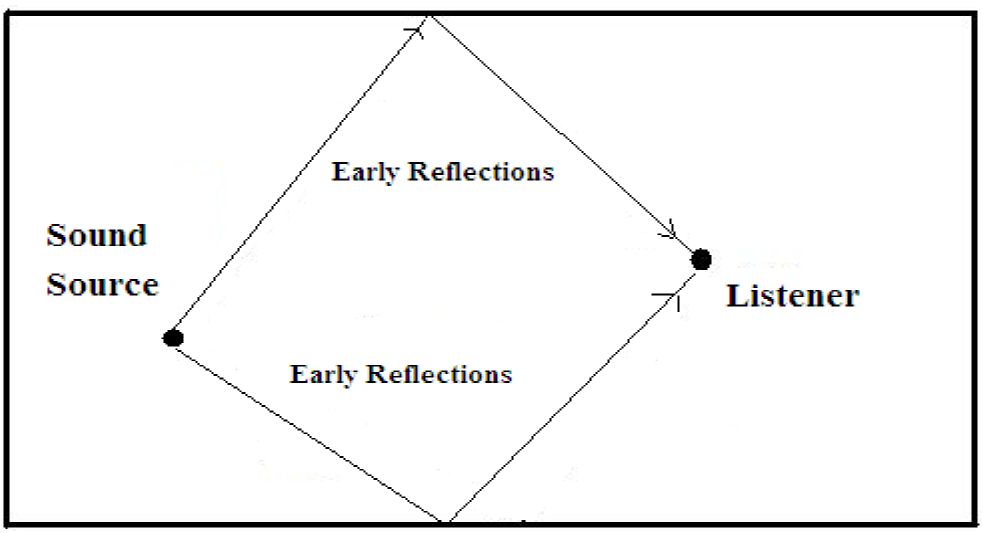


Figure 2: Early Reflections: Sound waves reflect of the surfaces of the room to reach the listener's ears.

2.1.3 Reverberant Sound

After the early reflections a denser set of reflections reaches the listener. Typically these reflections are heard by the ear 50ms after the direct sound is heard. This sound is reflected many times off all surfaces and in all directions (Hall, 1991). The figure below illustrates late reflections.

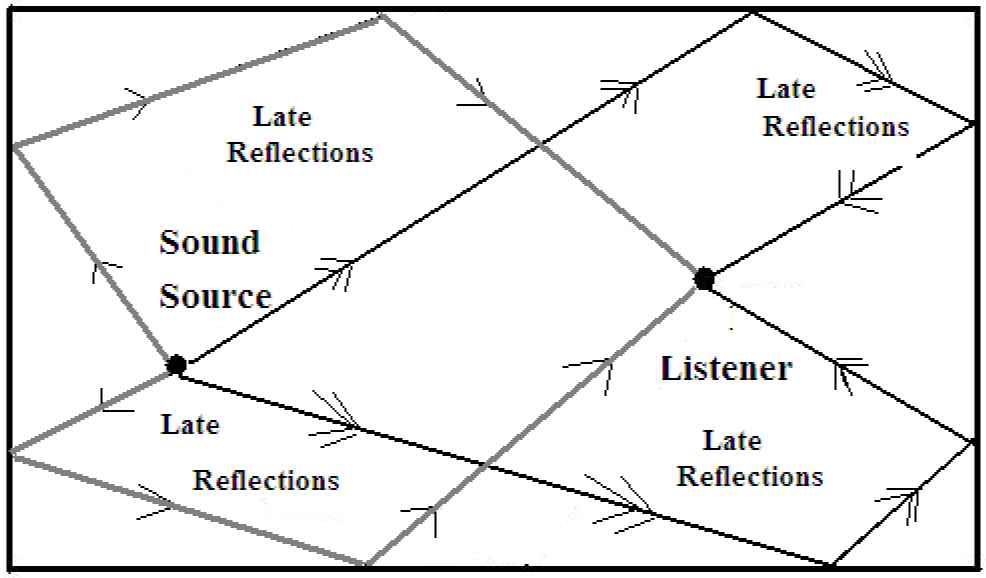


Figure 3: Reverberant Sound: Sound waves reflect of the surfaces of the room multiple times to reach the listener's ears.

The intensity of these reflections is further reduced than the early reflections.

The amount of time it takes for this sound to die away is called the reverberation time. It is referred to as RT60. This refers to the amount of time required for the sound field in a space to decay 60dB, or to one millionth of the original power (Hall, 1991). Figure 4 below shows the RT60 of a room.

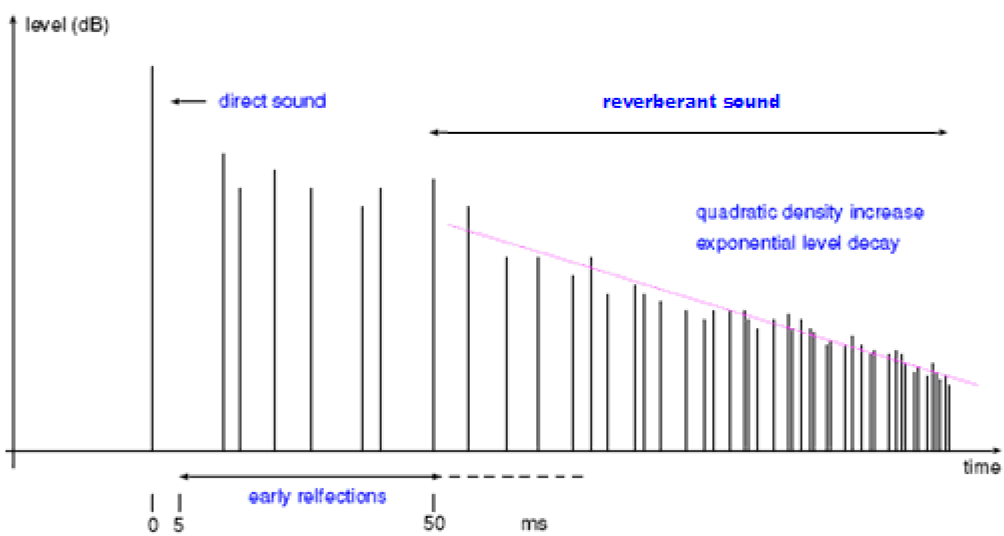


Figure 4: Typical natural reverberation pattern or RT60 of a room (Adriaensen, 2006).

2.1.4 Impulse Response

An impulse response of a room refers to how an impulse decays in the room. Recording an impulse in a room will essentially capture the reverberation pattern of the room. These IRs can be utilized by convolving them to any piece of audio in turn applying these acoustic characteristics to it. An ideal impulse is a mathematical construct containing all the audible frequencies in an infinitesimally narrow time. This cannot exist in reality and therefore starter pistols and balloons are mostly used, among other techniques. The convolution process and methods that could be used to record an IR are discussed in **Section 2.3**.

Another relevant concept in reverberation is the critical distance. The intensity of the reverberant sound throughout a space is roughly the same. The intensity of direct sound however varies according to the distance from the source with which it has an inverse relation. Therefore there is a critical distance for every room where the intensity of the reverberant sound exceeds the intensity of the direct sound. This has implications for clarity and intelligibility in music and speech, which are discussed in **Section2.5.3**.

2.2 Artificial Reverberation

Since the beginning of studio recording technology artificial reverberation has become a topic of interest. Acoustically dead studios meant that the recordings were extremely dry with almost negligible amount of reverberation. Therefore in the early days of recording, reverberation was to be recorded by natural means as post processing was not possible. This was achieved by placing microphones and loudspeakers in reflective rooms which had tiled basements. Suspended glass plates were also used to create a variable reverb environment. The standard and popularity of studios were determined by the different rooms that were built to mimic different reverberation environments. However the advent of artificial reverberation rendered obsolete the need for multiple reverberation environments (White, 2006).

2.2.1 Analogue Reverb

One of the first artificial reverb units were analogue spring reverb units. It was found that analogue sound signals passed through a spring with a transducer at the other end, added a sense of reverberation to the signal. The first major artificial reverberation invention however was the reverb plate. Reverberation was added to the signal by passing it through an isolated and soundproofed vibrating metal plate. The vibration of this metal was picked up by a transducer. This reverberation produced very dense sound with no distinguishable early reflections. This did not permit the reverberation to have spatial identity. Therefore plate reverb was used in tandem with tape delay which solved this issue. The tape delay controlled the injection of early reflections making it possible to control the size of the room in which the artificial reverberation was produced (White, 2006). The popularity of plate reverb has resulted in digital algorithms being written to emulate it. These digital algorithms were the next breed of artificial reverb units.

2.2.2 Digital Reverb

The advent of digital music technology meant that artificial reverb could be produced with the help of signal processing algorithms. This is essentially a top to down approach where the defined algorithm sets the rules to determine the optimum outcome, which in this case is reverberation. Multiple feedback delay circuits are used to create various types of reverb patterns. Gated reverb units where the RT60 could be cut off and reverse reverb units where reflected sounds get louder with time, were also possible to be created (Lehman, 2009).

A study conducted by Czyzewski and Sankiewicz (1988) discusses the subjective test methods for assessing and comparing the quality of different artificial reverberation programs. They try to better understand objective parameters such as frequency band, initial time gap, reverberation time which directly influence subjective ratings and assessments. Czyzewski (1990) followed this study with a paper researching the different methods for testing the quality of artificial reverberation. In this paper he proposes new cross-correlation criteria for testing the quality of artificial reverberation. A more recent study by Marui (2007) evaluates three artificial reverberation algorithms designed for headphones. It considers various perceptual categories of artificial reverberation for headphone reproduction of music.

2.3 Convolution Reverberation

Convolution reverberation was the next phase of artificial reverberation. Convolution is an important digital signal processing technique which is essentially a mathematical way to combine two signals to form a third signal (Smith, 1997). The two signals that are convolved are the input signal and an impulse response of a system. An impulse response contains the information as to how an impulse decays in a specific system or space. By convolving it with an input signal, an output signal is obtained which is how the input signal would have decayed in that specific system or space.

2.3.1 Convolution for Audio

Convolution is a mathematical operation. The convolution process involves the input signal being decomposed into impulses which are multiplied with the impulse response and then added up. In the case of an audio signal, the impulse response (IR) contains the information about how an impulse decomposes in a particular space which is essentially the reverberation pattern of that space. Each sample point of an input audio signal is multiplied with the impulse response, which shapes how the sample point would have decayed in that particular space. Successive sample points after being multiplied with the IR are then added together to give the resultant output signal. This outcome is how the input signal would have sounded or decayed if recorded in that space. Figure 5 below illustrates this point:

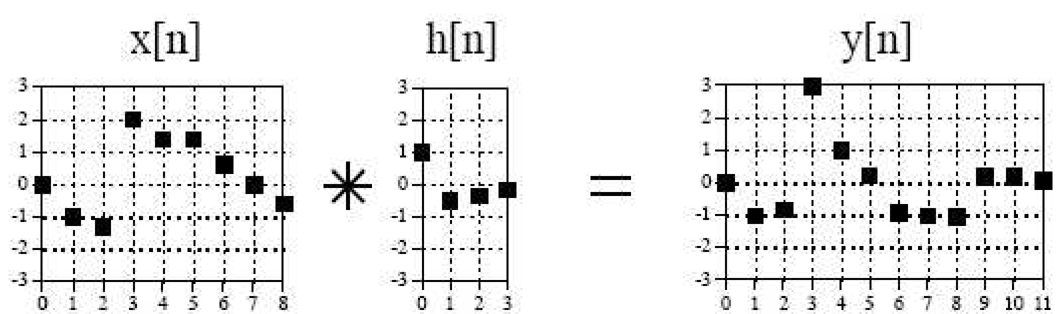


Figure 5: Convolution between two signals. (Smith, 1997).

In the figure above, x[n] is the input signal with 9 sample points being convolved with an IR with four sample points h[n] which results in y[n] which has three more sample points than the input signal (Smith, 1997 & Carson, et.al, 2009). The mathematical equation used to calculate y[n] is:

$$\sum_{k=-\infty}^{+\infty} x[k]h[n-k] \qquad where \begin{cases} x[n]\ is\ the\ input\ signal \\ h[n] is\ the\ IR\ of\ the\ system \end{cases}$$

2.3.2 Capturing an Impulse Response

There are two significant and fundamentally different methods used to record the IR of spaces. First is a simpler transient method while the second is the sine sweep method which is computationally more exhaustive. Both have their advantages and disadvantages which are discussed below:

i) **Transient Method**: This method uses a starter pistol or a balloon burst which is recorded in the particular space whose reverberation pattern is to be captured. This is a simple and straightforward recording process. The impulse produced is ideally supposed to have energy in all frequencies. The impulse can be produced where the sound source in the room would have originated from while the recording set up can be placed at the ideal listening point in the room, where the listener can perceive the most reflections. This recorded IR is ready to be convolved with any audio signal which results in the reverberation pattern of the room to be embedded into that signal. Even though this process is quick, it has its own disadvantages. Due to the loud and harsh nature of the starter pistol or the balloon burst, distortion is produced at low frequencies which tend to affect the recording. Also there is not much low and high frequency content in these impulses therefore resulting in an inadequate translation of the space's reverberation qualities at those frequencies (Apple Inc., 2007). Figure 6 is illustrates an IR.

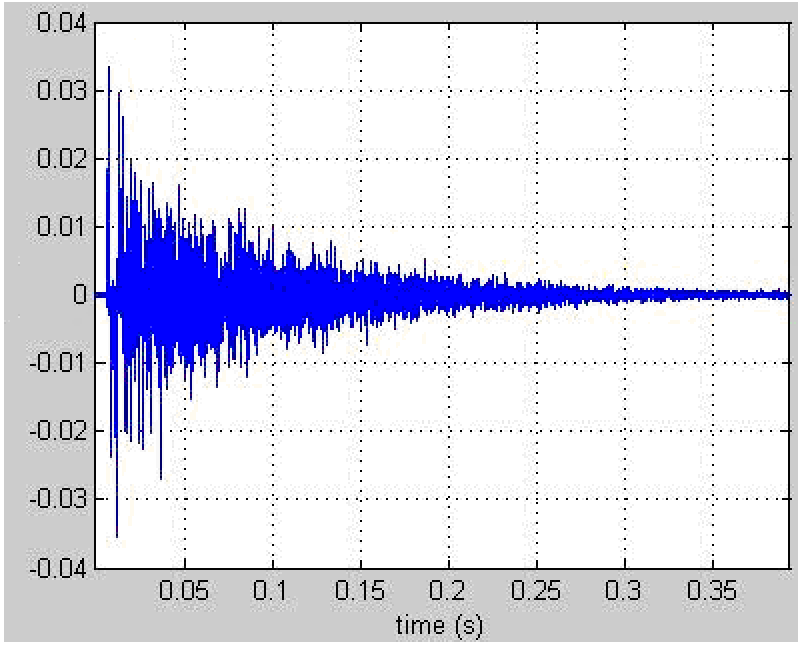


Figure 6: Recorded IR of a particular space. Y-axis represents amplitude. (Lehmann, 2009).

ii) <u>**Sine Sweep Method:**</u> For this method, instead of an impulse, a sine wave sweep of all the audible frequencies is recorded in the particular space. The response of the room to all the frequencies is stretched out over the length of the sine sweep. This method is considered more comprehensive and generally preferred to the transient method. This however after having been recorded, does not constitute the IR. A more computationally exhaustive process known as *deconvolution* is required. Deconvolution is the inverse of convolution. The convolution process for audio involves a dry signal being convolved with an IR to result in a wet signal. In the case of deconvolution, the recorded sine sweep (wet signal) is deconvolved with the sine sweep (dry signal) to result in an IR. Therefore using this mathematical process, the recorded data (sine sweep) is time compressed and level compressed into an IR. This IR which is *deconvolved* is now ready to be convolved with any audio signal (Apple Inc., 2007, Sheaffer & Elyashiv, 2005).

2.4 Studies in Artificial Reverberation

The improvement of artificial reverberation has been a steady progress from the time of analogue reverberation units to digital algorithmic reverberation methods and now to the latest convolution reverberation technique. This latest Convolution reverberation technique has gained popularity over the last decade. Impulse Response libraries such as Audio Ease and Open Impulse Response Library are available online which offer various range of impulse responses from famous acoustic halls around the world, such as the Berlin Philharmonie and Sydney Opera House, to the impulse response of famous analogue plate reverb units such as EMT 140.

Despite such progress of artificial reverberation, the majority of studies have been technical and objective, by trying to improve the digital signal processing methods and algorithms, and creating virtual simulations of rooms and spaces. As this current thesis looks into the perceived differences in natural and artificial reverberation types, the perceptual studies that have been conducted in this field have been reviewed below. The review progresses from perceptual studies in artificial reverberation to artificial reverberation studies in surround sound.

2.4.1 Perceptual Studies in Algorithmic and Convolution Reverberation

Karjalainen and Järveläinen in 2001 proposed a digital reverberation design motivated by the results from conducting listening experiments based on the perceptual aspects of reverberation. Their main focus was on what they term "perceptually good late reverberation". Their aim was to create an algorithm with ideal late reverberation. They achieved this by conducting listening tests, using auditory modelling approach and experimentation on simple digital filter models for modal responses. Two sets of listening tests were conducted. The aim of the first listening test was to estimate JND (Just Noticeable Difference) in reverberation time. This data was then used to better understand the underlying phenomena of how the predictions of the auditory models they created would turn out. To achieve this, four subjects participated in two separate listening tests where JNDs for reverberation time and the effects of reduced modal density were manipulated. Based on the results of the first test, a second formal listening test was conducted. The listeners were exposed to 6 stimuli which were speech and impulses with 3 fixed reverb times (RT60= 0.5 s, 1.0 s, and 2.0 s.). The listeners had to grade the quality of each stimulus from a scale of 1 to 5 comparing them to a test tone. The perceptual terms associated with the scale were 'Very metallic', 'Rather metallic', 'Somewhat metallic', 'Very little metallic', and 'Not at all metallic' from 1 to 5 respectively. From the result analysis they suggest an artificial reverberation algorithm based on parallel filter structures. However the main drawback of their algorithm is its computationally exhaustive nature which can only be useful in specific cases such as reverberation experimentation and special audio effects.

Browne (2001) proposed a hybrid algorithm which took the IR of space into account while using digital filters to model the reverb tail (RT). This method eases the computational demands of a regular convolution process. This holds consistent with what Adrianensen (2006) concluded in his paper which elaborates a design of an application that optimizes the convolution process for reverb. In this paper he observes that early reflections (ER) are not heard as a separate sound and that they rather merge with the direct sound. He states that ER need to be as consistent as possible to with the positioning of the sound source of the space as it is these reflections that enable the detection of direction and distance of the sound as well as the size and nature of the acoustical environment. He concludes that the RT is perceived as separate sound and RTs of different spaces cannot be differentiated in general. This gives more credence to Browne's study (2001) where digital filters are used to model the RT.

Browne's hybrid algorithm uses a truncated IR to artificially reproduce the ER and "rich energy of the space" while using digital filters to model the late echoes and reverberation tail. The computational demands are reduced by truncating the impulse response which results in decreased amount of computation that is necessary. However he aims to maintain the same level of quality by using digital filters to compensate for the missing portion of the IR. Following this, he conducted listening tests with the aim to find out if there is a perceived difference between the artificial reverberation produced by his hybrid algorithm and the ideal IR. IRs of a medium sized hall and the Bergamo Cathedral in Italy were used. Four input signals; a male voice speaking, clip from Symphony #9 in C minor, drum set loop and dry electric guitar were used. To these artificial reverberation was added using the ideal IR and the hybrid algorithm. The 20 test subjects were made to listen to the four input signals to which the artificial reverberation was added randomly. Following this initial listening process, the listeners were made to listen to each of the four extracts to which reverberation from both the techniques were added. They had to then identify which reverberation type was the one the listened to initially. 75% of the times the test subjects were able to identify the reverberation type. However Browne after having investigated standard deviation across the 20 subjects, found that there were considerable amount of deviations. Therefore he stated that he could not conclude that the hybrid algorithm output could not be distinguished from the linear convolution, giving credibility to this hybrid algorithm. Another noteworthy observation is that this does not reveal the quality of the hybrid algorithm, which is yet to be tested.

2.5 Studies in Artificial Reverberation and Surround Sound

Any speaker set up that enables three dimensional human hearing is considered surround sound. Surround sound is not as popular a commercial format as it could be due its tedious set up. However in a professional environment such as music studios and cinema theatres it is a preferred format to stereo sound. It is also slowly but steadily gaining popularity as a comprehensive home audio system over the stereo set up. Surround sound naturally adds to the sense of spaciousness, depth and ambience to the recording due to its multi speaker set up. Although this nature of surround sound makes studying reverberation in surround an interesting task, there have not been many studies in artificial reverberation conducted that directly relate to surround sound. Artificial reverberation algorithms that have been developed for surround are discussed here. Ambiophonics, discussed in **Section 2.5.2,** is another area of study in which artificial reproduction of reverberation is an important aspect. Sound localization is another area of study along with speech intelligibility where reverberation has to be considered. There is a dearth of perceptual studies in surround sound, making it a difficult process to review such studies. However a perceptual study in this field is reviewed followed by a perceptual study conducted in surround sound microphone techniques.

2.5.1 Artificial Reverberation Algorithms in Surround Sound

Producing artificial reverberation for surround is more complicated than for stereo. Beltran and Beltran (2000) presented a reverberation system for a multi-loudspeaker configuration. As the sense of space is more apparent in a surround sound environment, creating artificial reverberation becomes a complex affair. The reverberation pattern reproduced from each of the loudspeakers has to be different and accurate at the same time. To give the correct perception of the distance of a sound source they take into account sound spatialization techniques. Their algorithm takes into account early reflections from different parts of the room and treats them as virtual sound sources. The ER patterns are obtained by creating IRs for each of those isolated virtual sources by what is known as source image method which is an efficient algorithm to simulate the early response of a virtual room. The reverb tail pattern is added using an artificial reverberation algorithm. The final output signal for each of the loudspeakers is derived from processing a given input signal with a different impulse response and the common reverb tail pattern. This algorithm is similar to that of Browne's discussed earlier. However in this case the IR's are not recorded in a space but reproduced using a separate method. Beltran's study also shares similar principle views with Murphy's (2001) paper where complete de-correlation of the reverberation pattern from each of the speaker output's of the surround system is suggested. Murphy's (2001) study reports on the possibility of applying digital waveguide mesh models for creating room IRs for convolution based surround-sound reverberation.

2.5.2 Ambiophonics and Reverberation in Surround Sound

Ambiophonics is a method used in sound reproduction or used both in sound recording and reproduction that employs digital signal processing (DSP) and physical principles to improve reproduction of stereophonic and 5.1 surround sound for music, movies, and games in home theatres and other relevant applications. The aim of Ambiophonics is to create a realistic 3D reproduction of recorded sound by giving the listener a sense of perceived reality in the recording.

Reverberation is a concept that is closely related with studies in Ambiophonics. This is apparent in the paper by Farina (Farina, et.al, 2001) where Ambiophonic principles for the recording and reproduction of surround sound for music are investigated. This paper states that IRs of spaces is carefully studied when recreating a realistic reproduction of sound recorded in that space. The authors suggest as regular recording have very limited "3D surround" information, suitable room impulse responses are processed to recreate the missing information. Therefore room reverberation and room IRs are carefully studied in this field. This is confirmed in the paper by Farina and Torger (2001) where real time partitioned convolution algorithm is presented for Ambiophonics in surround sound. They state that real-time Ambiophonics systems requires multi-channel convolution with very long impulse responses for creation of believable reverberation. To ease the burden of such intensive computational demands a partitioned convolution algorithm is suggested, which requires less processing power.

2.5.3 Sound Localization, Speech Intelligibility in Surround Sound

Reverberation has been studied in relation to various fields such as sound localization cues and speech intelligibility. Sound localization is the ability of the human ear to locate the sound source. Human hearing is designed to use reverberation in sound as sound localization cues to obtain information about the space. The reverberant sound produced by a space give us a fair idea about size and shape of a room, as well as distance and direction of the sound source (Cook, 2001). Studies in these areas generally help achieve better understanding about the nature of reverberation which contributes towards its artificial reproduction.

Begault (1992) reported in his paper the effects of using artificial reverberation on localizations judgments of his 5 subjects. He investigated the perceptual effects of synthetic reverberation on three- dimensional audio systems. He conducted an experiment where his five participants were subjected to listen to a speech via headphones. The speech stimuli were presented with and without artificial "spatial" reverberation. Sound source localization judgments were asked to be made by the test subjects. They were asked to identify the location using three cues: azimuth, which was defined as 0 to 180° left or right with 0° being right in front; elevation which was defined as 0 to 90° up or down where 0° was at the ear level; distance judgments were made in inches with 0 inches being directly at the centre of the head. From the results Begault concluded that using spatial reverberation increased the amount of azimuth and elevation localization errors while minimizing intracranially heard stimuli. He suggests the results are applicable to artificial reverberation processors that seek to increase the auditory sensation of "spaciousness".

Another perceptual study conducted by Hodgson (2002) was looking into the effects of acoustic quality of class rooms towards speech intelligibility thus effecting student learning. In his study, Hodgson rates and ranks 279 university classrooms according to their acoustical qualities and speech intelligibility. The classrooms are 70% occupied for four different instructor voice levels, After conducting tests, 81% of the 279 classrooms were found have ''good,'' ''very good,'' or ''excellent'' acoustical quality with a ''typical'' average-male instructor while 18% of the classrooms had ''fair'' or ''poor'' quality. He suggests that as reverberation plays a role in speech intelligibility, classrooms designed have to make it a priority to take into consideration the controlling of this phenomenon to improve speech intelligibility. This study highlights the perceptual terms associated with reverberation and acoustical qualities of rooms.

2.5.4 Perceptual study in Surround Sound

Hietala (2007) studied the perceptual difference between four different 5.0 surround microphone recording techniques. He recorded a music piece played through a single speaker, which acted as the sound source, using four pre-existing microphone techniques. Two different spaces were also used for the recordings. The musical piece used for the recording was an excerpt from a previously recorded American traditional song "I Am a Pilgrim" which is a mandolin and acoustic guitar duet. The 16 participants were asked to rate the perceived differences in the recording from a scale of 1 to 5 according to four perceptual terms; *width, depth, naturalness* and *ambience*. His results reported on the perceived differences in the four chosen microphone techniques and discussed some general characteristics about each of the four microphone array techniques. Employing a similar rating scheme by using perceptual terms for listeners to rate seems favourable to this thesis study. The perceptual terms such as naturalness and ambience used for surround ratings can also be applied to reverberation in surround. This possibility is further discussed in **section 3.6**. However a main drawback of Hietala's study is the lack of justification given for using these perceptual terms. Also the reasons for using that particular musical piece are not discussed.

2.6 Summary

Above, perceptual studies in algorithmic and convolution reverberation and studies conducted in artificial reverberation and surround sound have been discussed. The papers by Karjalainen (2001) and Browne (2001) have a perceptual aspect to their study. Karjalainen applies the results from the perceptual study to develop an artificial reverberation algorithm whereas Browne uses the study to add credence to his hybrid algorithm. The papers by Czyzewski(1990) and Sankiewicz (1988) study the perceptual aspects of reverberation and its evaluation. Marui (2007) as well evaluate artificial reverberation algorithms designed for headphones using perceptual categories. All these perceptual studies have been conducted in stereo sound thereby giving a good understanding in the perception of reverberation in stereo. However studies in the perception of artificial reverberation in surround sound have been very limited. There have been artificial reverberation algorithms created for surround sound (Beltran & Beltran, 2000 and Newton & Howard 2001) but its perception has not been studied. Therefore perceptual studies in this field are a viable option having its own further implications.

Even though the perception of artificial reverberation in surround sound has not been a popular area of research, there has been significant amount of non-perceptual studies conducted on surround sound involving artificial reverberation. Ambiophonics is a new area of research where reproduction of reverberation is carefully considered (Farina, et.al, 2001). There have also been hybrid convolution algorithms created (Farina & Torger, 2001) for this field of study. Sound localization (Begault, 1992) and speech intelligibility (Hodgson, 2002) are other research areas where artificial reverberation has been used. Begault (1992) studied the perceptual effects of artificial algorithmic reverberation on sound localizations cues. However the perception of artificial reverberation has not been studied. The study proposed for this thesis has implications in all these fields of research. Therefore studying the perceived differences in natural and convolution reverberation types adds to the body of research that has been conducted in this field of study while having implications in other fields discussed above.

3 Research Methods

3.1 Experimental Design

The experimental design for this thesis does not lend itself directly to previous studies. Although artificial reverberation has been researched extensively, studies investigating its perception have been scarce with its perception in surround sound being even rarer. The perceptual studies that were found in this field, discussed in **Section 2.4.1**, did not focus on the perceptual aspect making it difficult to map out a similar strategy based on such previous studies. As the difference in perception of natural and artificial convolution reverberation types was being investigated, a listener rating process of the different reverberation types using perceptual scales was required. This listener rating process used was similar to that of Hietela (2007), whose thesis was discussed in **Section 2.5.4**. Perceptual scales used for the rating process were similar and are explained further in **Section 3.6**. Short musical excerpts which contained the different IR types were used. This required recording a music excerpt in an extremely dry studio environment, to which the IRs could be convolved with.

To begin with, three different instruments were recorded in a dry studio environment. These dry excerpts were then re-recorded, using 5.0 surround recording techniques, in two different spaces to capture the natural reverberation of those spaces. The IR of those two spaces was further captured using two different techniques. These IRs were then convolved with the dry musical excerpts to give the artificial convolution reverberation excerpts. With there being three instruments used in tandem with three reverberation types, one natural and two artificial, for two separate spaces, a total of 18 excerpts were prepared to be rated. Each excerpt was then rated by 27 participants according to six perceptual scales. The preference rating of these excerpts was also given by the participants. These steps together contain the data collection process. Each of these steps are explained below in detail.

3.2 Dry Excerpt Recordings

Though the primary focus was on investigating the differences in reverberation types, it was interesting to find out if and how an instrument could affect the quality in the perception of reverberation. Oboe, Cello and Piano were the three different instruments chosen as each of these instruments have a distinct quality and timbre. With Oboe being a woodwind instrument, Cello a bowed string instrument and Piano a struck string instrument, their frequency spectrum and harmonic content are different from each other which resulted in them being chosen.

The main music studio in the Music Department of the University of Jyväskylä was used for recording the excerpts. The excerpts were played by students of the department who had professional training in their instruments. All the three instruments were recorded in the live studio room which is meant for vocal recording. The room had negligible amount of reverberation, with RT60 being below 0.2 seconds, with all the surfaces covered with absorbents which damped the sound, causing almost no reflections. The aim was to get as dry a recording as possible. Microtech Gefell M300 condenser microphone was used for the recording. It has a cardiod polar pattern which allows for more direct sound to be picked up when pointed towards the body of the instrument. This in turn does not pick up the reflections off the surfaces facing opposite to the instrument making the recording as dry as possible. The instrumental pieces recorded were:

i) Cello – *J.S.Bach – Suite No.1. Prelude*.
ii) Oboe – *C.Saint-Saens – Sonate*
iii) Piano – *J.S.Bach – Italia Concerto 1st Movement*

All the pieces were recorded at the sample rate of 48 kHz and bit resolution of 32. The pieces were edited in the music sequencing software Logic Pro to about 30 seconds in length.

3.3 Natural Reverberation Recordings

The natural reverberation recordings were done in two separate spaces with distinct RT60 times. Different spaces were used as it would be useful to find out if a marked difference in the RT60 time would cause a change in the perception between the natural and artificial reverberation. The first space was the Taulumäki Church in Jyväskylä. With it being a space which could seat around 700 people and having a maximum height of 30m, it had a RT60 time of over 2.5 seconds.

After having secured a room with a considerably long RT60 time, lecture room C1 in the C-Building of the University of Jyväskylä with a short RT60 time of below 1 second was used.

The natural reverberation of both these spaces was captured by recording the dry excerpts in the spaces. Using Logic Pro, the dry excerpts were played through a single Genelec 8030 A monitor speaker, which acted as the sound source. Live reproduction of the recorded musical excerpts was not conducted as the physical playing of the piece would have differed from that of the studio recording. This would have affected the perceptual ratings given by the participants. By reproducing it via a speaker resulted in the same dry excerpt being used for all the reverberation types. This Genelec monitor speaker has a frequency response of 60Hz to 20 kHz. In both the spaces, the speaker was placed where the instrument or the speaker would have been placed when giving a live performance, which was at the centre of the stage facing the audience.

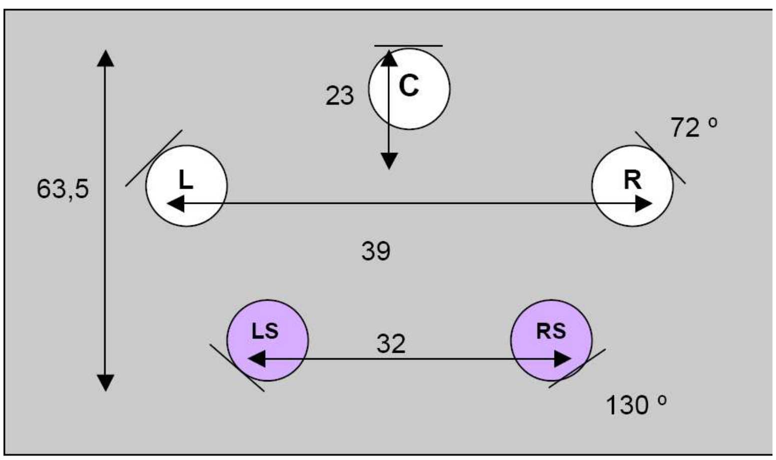


Figure 7: The Michael Williams 72 surround microphone setup (Hietala, 2007). Cms is the unit of distance. (L - Left, R - Right, C - Centre, Ls - Left Surround, Rs - Right Surround)

The microphones used for the surround recording were three AKG-C414 (L, C &R) and two Audio Technica AT4033 (Ls & Rs) condenser microphones. The polar pattern for these microphones was set to cardiod, allowing more reflections to be picked up only in the direction they were facing. The microphones were connected to an Apogee Ensemble soundcard which was connected to an Apple Mac. Logic Pro was used for the surround recordings. The surround sound microphone recording technique used was the *Williams 72,* which is illustrated in the figure 7. This was based on Hietala's (2007) thesis, which reported that this technique was more sensitive to space than the other such recording techniques suggested in his thesis. Also this technique was perceived to be more ambient and natural. With the completion of this process, 6 excerpts (3 instruments x 2 spaces) were ready for rating, which were all of the natural reverberation type. The following sections will detail how the artificial reverberation excerpts were prepared.

3.4 IR Recordings

The IR of the two spaces was recorded before convolving them with the dry musical excerpts. Two different methods, sine sweep and transient, which were discussed in **Section 2.3.2** were used to record the IR. Impulse Response Utility (IR Utility) was the software used for the recordings. The microphones used, were the same as mentioned in **Section 3.3**. The microphones and speakers set up were according to the IR Utility manual.

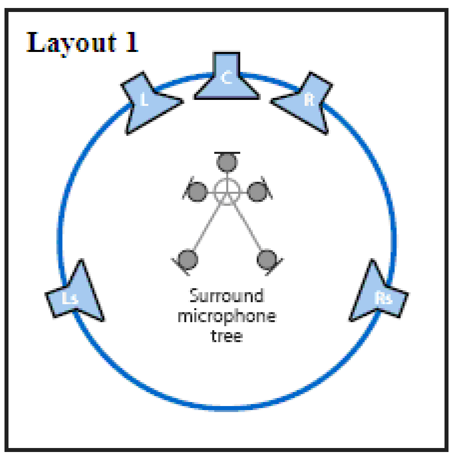


Figure 8: Loudspeaker and microphone set up to capture 5.0 IR (Apple Inc., 2007)
(L - Left, R - Right, C - Centre, Ls - Left Surround, Rs - Right Surround)

The monitor speakers which generated the sine sweep were placed as shown in the figure above. The default 10sec sine sweep, of the audible frequencies, in the IR utility software was used for the recording. The monitor speakers used were Genelec 8030 A.

For the transient method a regular set of balloons were used. Five balloons were used, which were burst from the same points where the speakers for the sine sweep method were placed. Five balloons were required which resulted in five IRs. As the reverberation of the space was being captured for surround reproduction, each of these five IRs per method was to be convolved with the dry excerpts for reproduction from each of the five surround speakers. The size of the balloons was made sure to be the same for the recordings in both the spaces. Also the Sound Pressure Level (SPL) of the balloon burst was calculated to match with the SPL level used for the sine sweep, to make sure there are no discrepancies in the recordings.

Williams 72 was the microphone technique used. To capture all the reflections evenly, the five microphones were placed in the centre of the spaces while making sure they were pointing towards the loudspeakers and the balloons.

3.5 Convolving IRs and Dry Excerpts

After having recorded the IRs, they were to be edited before convolving them with the dry excerpts. The IRs of the spaces recorded using the transient method were edited in IR Utility by removing the silences at the beginning and the end of the recordings while the sine sweep recordings were *deconvolved*. After the editing process, these four surround IRs were then exported to Logic Pro where Space Designer reverberation patch was used to convolve them with the dry musical excerpts. The RT60 of the Church was *2.6 sec* while for C1 it was *0.9sec*. With there being four surround IRs (two spaces and two different methods) and three dry musical excerpts, a total of 12 artificial excerpts were prepared for rating.

3.6 Listening Experiment

The 18 excerpts, with 6 of them containing natural reverberation and the other 12 containing artificial reverberation, were then rated according to 6 perceptual scales by 27 participants. All the participants were students of the university, ranging from ages 14 to 45, with 18 of them having been music students. However none of them were experts in the field of Reverberation.

There were six perceptual rating scales used. Each scale was associated with a perceptual term. The terms used were:

Naturalness: It is an appropriate term for the ratings as the quality of natural and artificial reverberation types is being studied.

Spaciousness: It is appropriate as it has been associated with reverberation in a previous study by Begault (1992), which was discussed in **Section 2.5.3**.

Ambience: It is defined as something "which surrounds or encompasses". As this study was being conducted in surround sound, which adds to the sense of depth and ambience to recordings **(Section 2.5**), Ambience was used as a perceptual term.

Distance: It is an appropriate term as the amount of reverberation in a sound acts as a sound localization cue which helps with distance perception (Cook, 2001).

Having participated in the ISSSM (International Summer School for Systematic Musicology) in 2010 and interacting with professors in this field, it was suggested that "*Roughness"* and "*Density"* were appropriate terms that could be used for the perceptual ratings.

It was possible that the participants could have interpreted these terms differently, which could adversely affect the results. Therefore brief definitions of these scales were provided to them. A discrete scale from 1 to 7 was used, instead of 1 to 5, as it was felt that it could give more room for participants to commit to a certain half of the scale.
The brief definitions of the scales given to them were:

i) *Naturalness*: Perceived naturalness of the sound
ii) *Spaciousness*: Perceived space or area in which sound source is present
iii) *Ambience*: Perceived quality (dryness and wetness) of the sound
iv) *Distance*: Perceived distance from sound source
v) *Roughness*: Perceived roughness and smoothness of the sound
vi) *Density*: Perceived compactness and scatteredness of the sound in the space

The listening tests were conducted in main studio control room of the University of Jyväskylä. Genelec 1037 surround monitoring system was used which was laid out according the ITU-R BS.775, a music industry standard. Listeners were instructed to pay attention to the overall sound. They were instructed to "try not to pay attention to the melody, quality of playing and the background noise". The 18 excerpts were presented in a random order to each of the participants with the use of Max/MSP. Figure 9 shows the user interface.

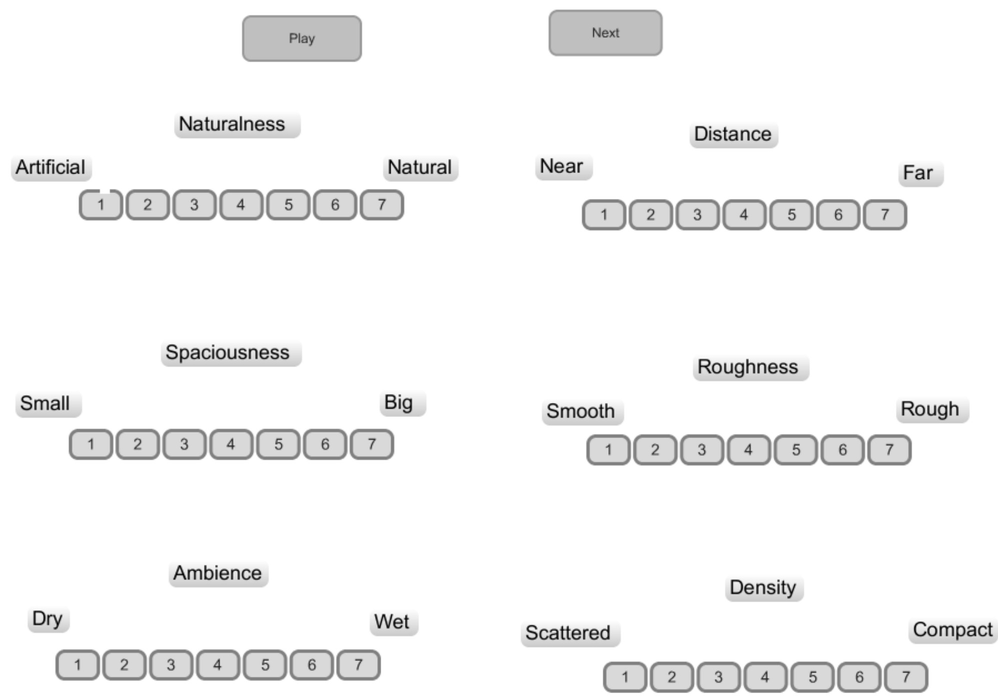


Figure 9: User interface for the perceptual ratings

Following the perceptual ratings, the participants were asked to give their preference of the excerpts. The excerpts for this task were grouped in a different manner before being presented randomly to the participants using Max/MSP. The 18 excerpts were divided into six groups of three. Each group consisted of each of the three reverberation types (one natural and two artificial) for a particular instrument in a particular space. For example, the Cello excerpts of the Church were grouped together. Therefore this group consisted of one natural reverberation recording of the dry Cello excerpt in the Church and its two artificial counterparts. The artificial reverberation excerpts were produced from convolving the two artificial IR recording methods with the dry Cello excerpt. In this manner, the excerpts in the groups were ranked by the participants in their order of preference. Figure 10 shows the user interface. These results of the preference ratings would reveal which reverberation type was more preferred.

Figure 10: User interface for the preference ratings

To sum up the data collection process, two different spaces (Church & C1) with distinct reverberant qualities were used, whose reverberant pattern were captured both naturally and artificially (Transient & Sine Sweep). Three different instruments (Cello, Oboe and Piano) were used to effectively bring out the differences in perception of these reverberation patterns or IRs. These differences were rated by 27 participants according to six perceptual scales. The participants also rated their preference in reverberation type for each of the instruments in a particular space. The analysis of the collected data is discussed in the next section.

4 Results

The data collected, which consisted of 6 perceptual ratings of 18 excerpts for 27 participants, was analyzed using Matlab. Firstly the outliers were removed. Following that a correlation between the 6 perceptual scales was calculated. A three-way ANOVA was conducted which was followed up by a one-way ANOVA for the reverberation types. Post-hoc tests were conducted for significant results. Also the preference ratings were summed up for analysis. The motivation for all these steps and the results obtained are detailed in this section.

4.1 Outlier Removal

Outlier removal was carried out separately for each of the six perceptual scales. Firstly, as the perception of reverberation was rated for spaces with distinct reverberation times, the data for the two spaces was isolated. This resulted in twelve 27 by 9 matrices. For each matrix the inter-subject correlation was calculated, to give a 27 by 27 matrix. The correlations for each participant were averaged out and all participants with negatively correlations were removed. The same process was repeated and this time around the mean inter-subject correlations and standard deviations were calculated. All participants who were three times the standard deviation less than the mean inter-subject correlation were removed. This process was conducted for each of the twelve 27 by 9 matrices. On average there were 4 participants who were outliers per perceptual scale per space except for the Naturalness scale in the Church which had 10 outliers. After having removed the outliers, the Cronbach Alpha was calculated to find out the consistency of the ratings. The mean and the Cronbach Alpha (α) of the 6 scales in both the spaces are given in the tables below:

	Naturalness α = 0.68	Spaciousness α = 0.84	Ambience α = 0.73	Distance α = 0.92	Roughness α = 0.87	Density α = 0.86
Cello – Natural	5.21	5.81	4.06	3.52	4.05	3.29
Cello – Balloon Burst	4.79	5.91	4.65	3.14	3.55	2.95
Cello – Sine Sweep	4.07	5.06	4.82	3.23	4.15	3.0
Oboe – Natural	4.64	5.86	5.29	3.95	2.8	4.10
Oboe – Balloon Burst	3.79	5.59	4.82	3.90	2.2	4.05
Oboe – Sine Sweep	4.86	5.72	5.47	3.95	2.6	3.86
Piano – Natural	5.07	5.68	4.53	3.90	3.0	3.76
Piano – Balloon Burst	4.0	6.04	4.94	2.85	3.5	2.62
Piano – Sine Sweep	5.14	4.23	3.71	5.00	3.0	4.95

Table 1: The Means and Cronbach alpha values of Church Excerpts

	Naturalness α = 0.90	Spaciousness α = 0.85	Ambience α = 0.85	Distance α = 0.87	Roughness α = 0.78	Density α = 0.90
Cello – Natural	5.58	2.86	3.23	2.67	4.55	5.11
Cello – Balloon Burst	4.91	4.29	4.64	3.83	3.90	4.05
Cello – Sine Sweep	4.96	3.71	3.91	3.63	4.23	3.79
Oboe – Natural	5.54	2.95	3.00	2.25	3.45	5.47
Oboe – Balloon Burst	4.67	3.00	3.00	2.79	3.05	4.74
Oboe – Sine Sweep	4.04	3.43	3.00	2.92	3.73	5.16
Piano – Natural	5.46	2.71	3.5	2.46	3.59	5.58
Piano – Balloon Burst	3.96	3.90	4.18	3.33	3.77	3.63
Piano – Sine Sweep	2.67	4.43	4.68	3.63	4.64	2.84

Table 2: The Means and Cronbach Alpha values of C1 Excerpts

The Cronbach Alpha for all the scales in both the spaces is above 0.70 except for the *Naturalness* scale in the Church which is 0.68. This confirms that all the data is reliable and consistent and conclusions drawn from them are valid.

4.2 Collinearity of Rating Scales

Having used six perceptual scales for the ratings, it was necessary to find out how each of the scales correlated. Therefore the correlation between each of the scales in the two spaces was calculated. The six by six correlation matrices for the two spaces can be found below in tables 3 and 4.

	Naturalness	Spaciousness	Ambience	Distance	Roughness	Density
Naturalness	1	-0.175	-0.466	-0.33	0.1224	0.358
Spaciousness	-0.175	1	0.587	0.617	-0.004	-0.61
Ambience	-0.466	0.587	1	0.35	-0.331	-0.268
Distance	-0.330	0.617	0.35	1	0.126	-0.582
Roughness	0.122	-0.004	-0.331	0.126	1	-0.655
Density	0.358	-0.609	-0.268	-0.581	-0.654	1

Table 3: Correlation table of the 6 perceptual scales for the Church. * for $p<0.05$, ** for $p<0.01$, *** for $p<0.001$

There were no correlations between the scales for the Church space at $p<0.05$.

	Naturalness	Spaciousness	Ambience	Distance	Roughness	Density
Naturalness	1	-0.66	-0.424	-0.399	-0.217	0.661
Spaciousness	-0.66	1	0.881***	0.922***	0.46	-0.894***
Ambience	-0.424	0.881***	1	0.854**	0.531	-0.842**
Distance	-0.399	0.922***	0.854**	1	0.477	-0.873**
Roughness	-0.217	0.46	0.531	0.477	1	-0.522
Density	0.661	-0.894***	-0.842**	-0.873**	-0.522	1

Table 4: Correlation table of the 6 perceptual scales for C1. * for $p<0.05$, ** for $p<0.01$, *** for $p<0.001$

In the space C1, four of the scales, Spaciousness, Ambience, Distance and Density, correlate highly with each other. While the first three scales correlate positively with each other, Density correlates negatively with them.

4.3 Perceived Differences between Reverberation Types

The aim of this thesis study was to find out if there were differences in the perception of the natural and convolution reverberation. With that being the focus of this section of analysis, a three-way ANOVA was conducted for each of the six perceptual scales in Matlab. The three main factors in the ANOVA were the reverberation types, the spaces and the instruments. The results of the first Main effect that of the reverberation types are reported in the table 5. The results show whether there are significant perceptual differences between the reverberation types irrespective of the instrument type or the space.

	Main Effect 1: Reverberation Type (Natural ,Balloon Burst, Sine Sweep)	
	F Value	Significance Value
Naturalness	9.83	**0.0001**
Spaciousness	5.18	**0.0061**
Ambience	3.21	**0.0415**
Distance	5.83	**0.0032**
Roughness	2.9	0.0563
Density	13.51	**0**

Table 5: F value and Significance results for the three reverberation types for all the six perceptual scales. $p< 0.05$ level results are highlighted.

All perceptual scales, except the *Roughness* Scale, had significant differences in their perception. This meant that the reverberation types for each of those scales were perceived to be significantly different irrespective of the instrument or space. Tukey-Kramer multiple comparison post-hoc tests were conducted to find out how each these reverberation types differed for each of those scales. Below, figure 11 illustrates the differences between the means of the natural reverberation excerpts and artificial Balloon Burst IR and Sine Sweep IR reverberation excerpts.

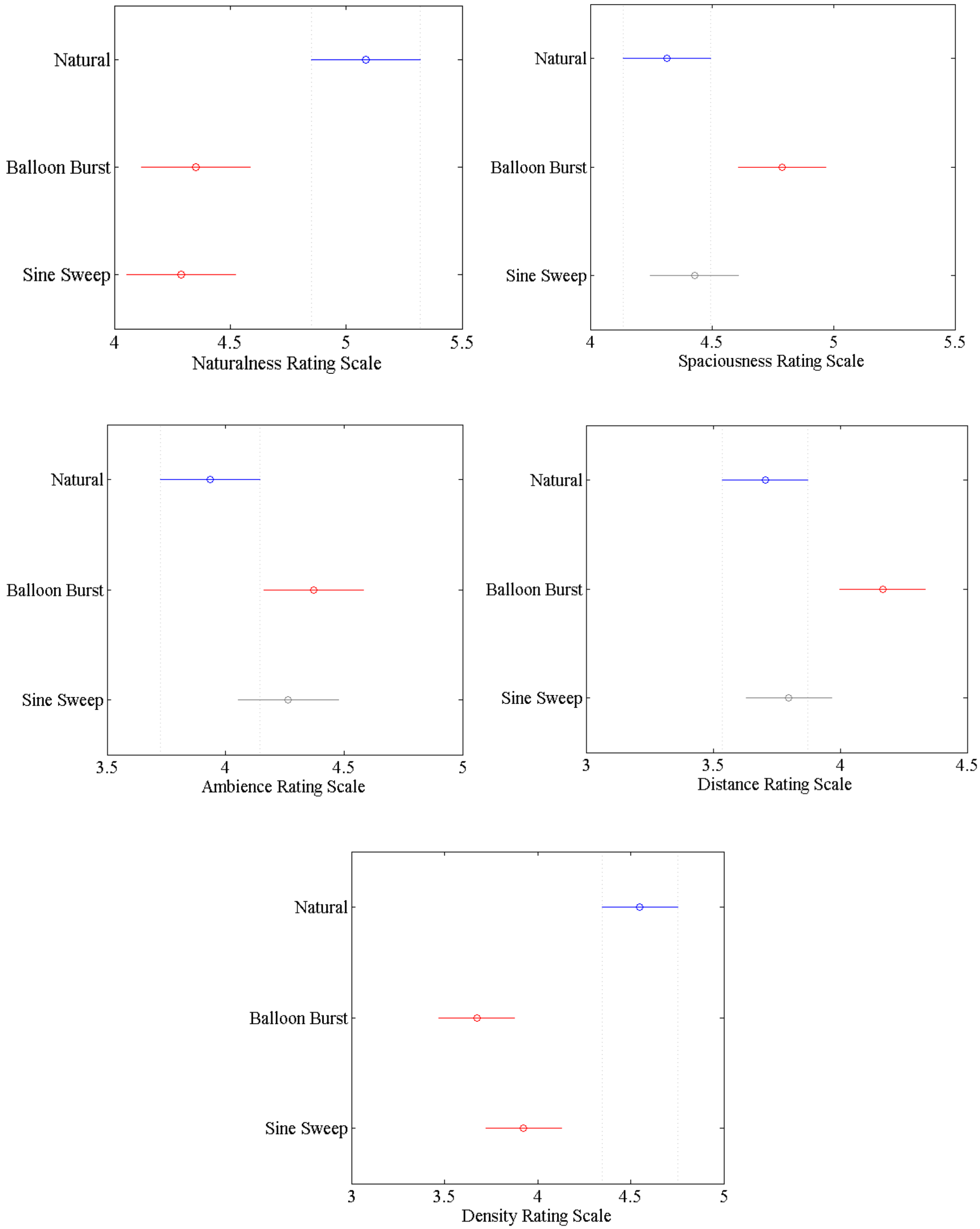


Figure 11: Illustrated are the Means and the Confidence Intervals of the three reverberation excerpts

The post-hoc tests revealed how the perception of the reverberation types varied according to the perceptual scales irrespective of which instrument was being heard in either of the spaces. The results for each of the scales are summed up below:

i) *Naturalness scale*: The mean of the natural reverberation excerpts was found to be significantly higher than the other two types of artificial excerpts. Therefore the natural reverberation excerpts were perceived to be significantly more "natural" than the artificial excerpts.

ii) *Spaciousness Scale:* The mean of the artificial excerpts using the balloon burst IR were significantly higher than the natural reverberation excerpts. Therefore these artificial excerpts were perceived to be significantly more "spacious" than the natural excerpts.

iii) *Ambience Scale:* Once again the mean of the artificial excerpts using the balloon burst IR were significantly higher than the natural reverberation excerpts. Therefore, in this case, these artificial excerpts were perceived to be significantly more "ambient" and "wet" than the natural excerpts.

iv) *Distance Scale:* In this scale as well the mean of the artificial excerpts using the balloon burst IR were significantly higher than the natural reverberation excerpts. Therefore these artificial excerpts were perceived to be significantly more "distant" from them than the natural excerpts.

v) *Density Scale:* This scale was similar to that of the Naturalness scale with the mean of the natural reverberation excerpts being significantly higher than the other two types of artificial excerpts. Therefore the natural reverberation excerpts were perceived to be significantly more "dense" and "compact" than the other artificial excerpts.

The remaining results of the three-way ANOVA demonstrated there were significant differences between the perception of the excerpts according to space and instrument type. The Church excerpts had a significantly higher mean than the C1 excerpts for *Spaciousness, Ambience* and *Distance* scales. For the *Roughness* and *Density* scales C1 excerpts had a significantly higher mean than the Church excerpts. For instrument type, the Cello excerpts had a significantly higher mean than the other two instrument types for the *Naturalness* and *Roughness* scale. The Piano excerpts had a significantly higher mean than the Oboe excerpts for the *Roughness* scale. For the

Density scale, Oboe had a significantly higher mean than the other two instrument types. There were also significant interactions between the three factor levels. These results allow for interesting deductions which are not the focus of this thesis study. Therefore the results of the other two main effects and variable interactions are reported in **Appendix 1** along with the illustrated Tukey-Kramer multiple comparison post-hoc tests on the results with p<0.05.

4.4 Perceived Differences between Reverberation Types by Space and Instrument Type

Though the results of the three-way ANOVA provided an insight into the differences in perception of the three reverberation types, no insight was gained regarding how the other two factors, instrument type and space, affected their perception. Therefore 6 One-Way ANOVAs were carried out for each of the six perceptual scales by isolating the reverberation types according to space and instrument type. The three levels in the one-way ANOVAs were the reverberation types. With their being six perceptual scales, 3 instruments ad two spaces, a total of 36 one-way ANOVAs between the natural and two artificial reverberation types were conducted. The results of all the ANOVAs are reported below in table 6.

Church	**Naturalness**	**Spaciousness**	**Ambience**	**Distance**	**Roughness**	**Density**
Cello Excerpts	2.47 / 0.098	3.25 / 0.051	1.34 / 0.27	0.31 / 0.73	1.3 / 0.28	0.52 / 0.6
Oboe Excerpts	2.69 / 0.08	0.44 / 0.645	1.34 / 0.27	0.61 / 0.55	1.54 / 0.22	0.17 / 0.84
Piano Excerpts	2.42 / 0.10	**13.42 / 0.00**	3.14 / 0.05	**36.17 / 0.0**	1.12 / 0.33	**12.23 / 0.0**

C1	**Naturalness**	**Spaciousness**	**Ambience**	**Distance**	**Roughness**	**Density**
Cello Excerpts	2.01 / 0.14	**7.31 / 0.001**	**6.21/0.003**	**7.26/0.001**	1.24 / 0.296	**7.33/ 0.002**
Oboe Excerpts	0.91 / 0.41	0.72 / 0.49	0 / 1	2.36 / 0.102	1.25 / 0.294	1.42 / 0.251
Piano Excerpts	**23.05 / 0.0**	**9.24 / 0.00**	**3.3.5/0.041**	**6.17/0.003**	**3.37/ 0.041**	**17.03/ 0.00**

Table 6: *F values* and *Significance* results of the One-way ANOVAs. The p<.05 level results are highlighted.

In the Church, there were perceived to be no significant differences between the three reverberation types at p<.05 level for Cello and the Oboe excerpts in any of the perceptual scales. Only three scales for the Piano had significant differences perceived at p<.05 level. In C1, the Oboe excerpts had no significant differences perceived between the reverberation types at p<.05 level, similar to that of the Church. For the Cello excerpts, 4 scales had significant differences perceived between the three reverberation types at p<.05 level, while that was the case for all the scales for the Piano Excerpts.

Tukey-Kramer multiple comparison post-hoc tests were conducted on the p<.05 level results to find out how the means of the reverberation type excerpts differed. The figures 12, 13 and 14 below illustrate the differences between the natural reverberation excerpts, artificial reverberation excerpts using Balloon Burst IR and Sine Sweep IR.

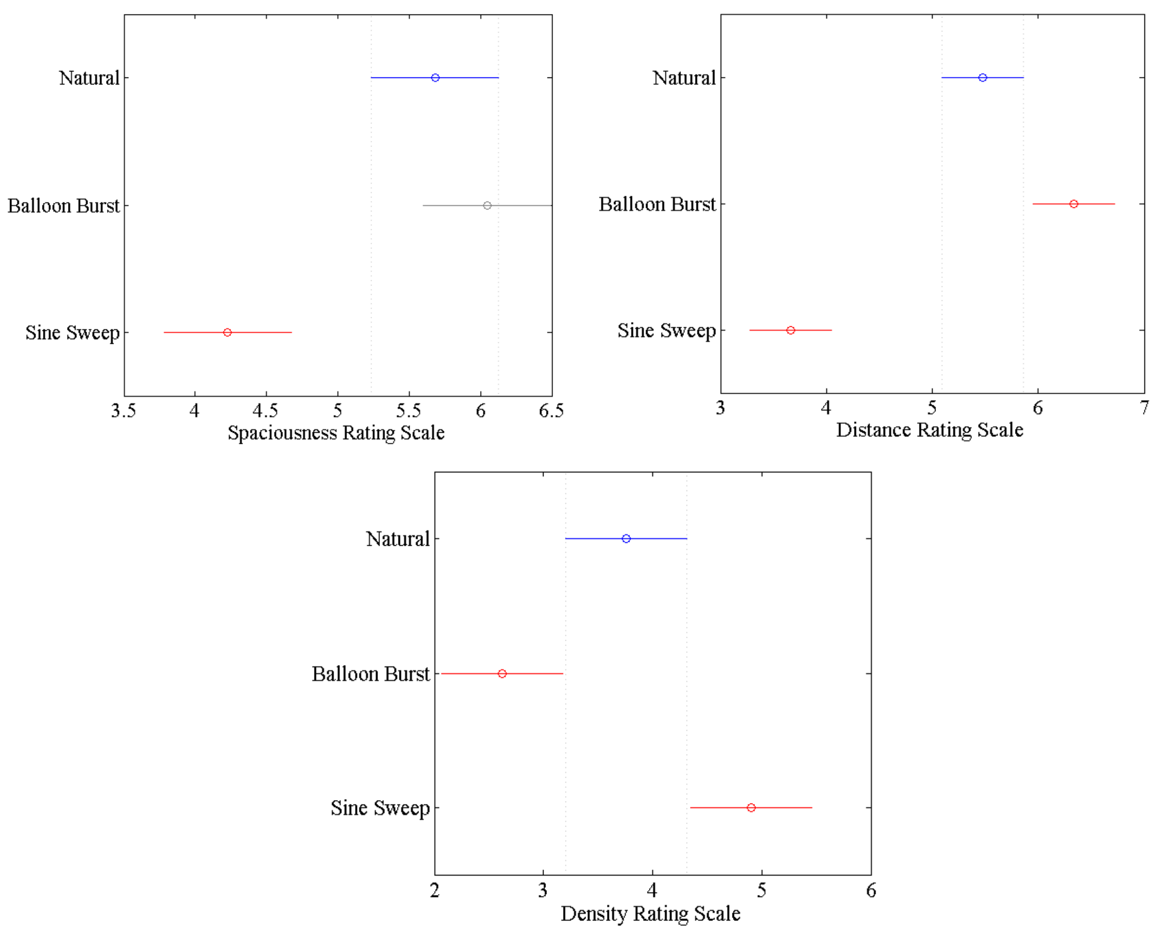


Figure 12: Illustrated are the Means and the Confidence Intervals of the three reverberation excerpts for the *Piano* in the *Church*

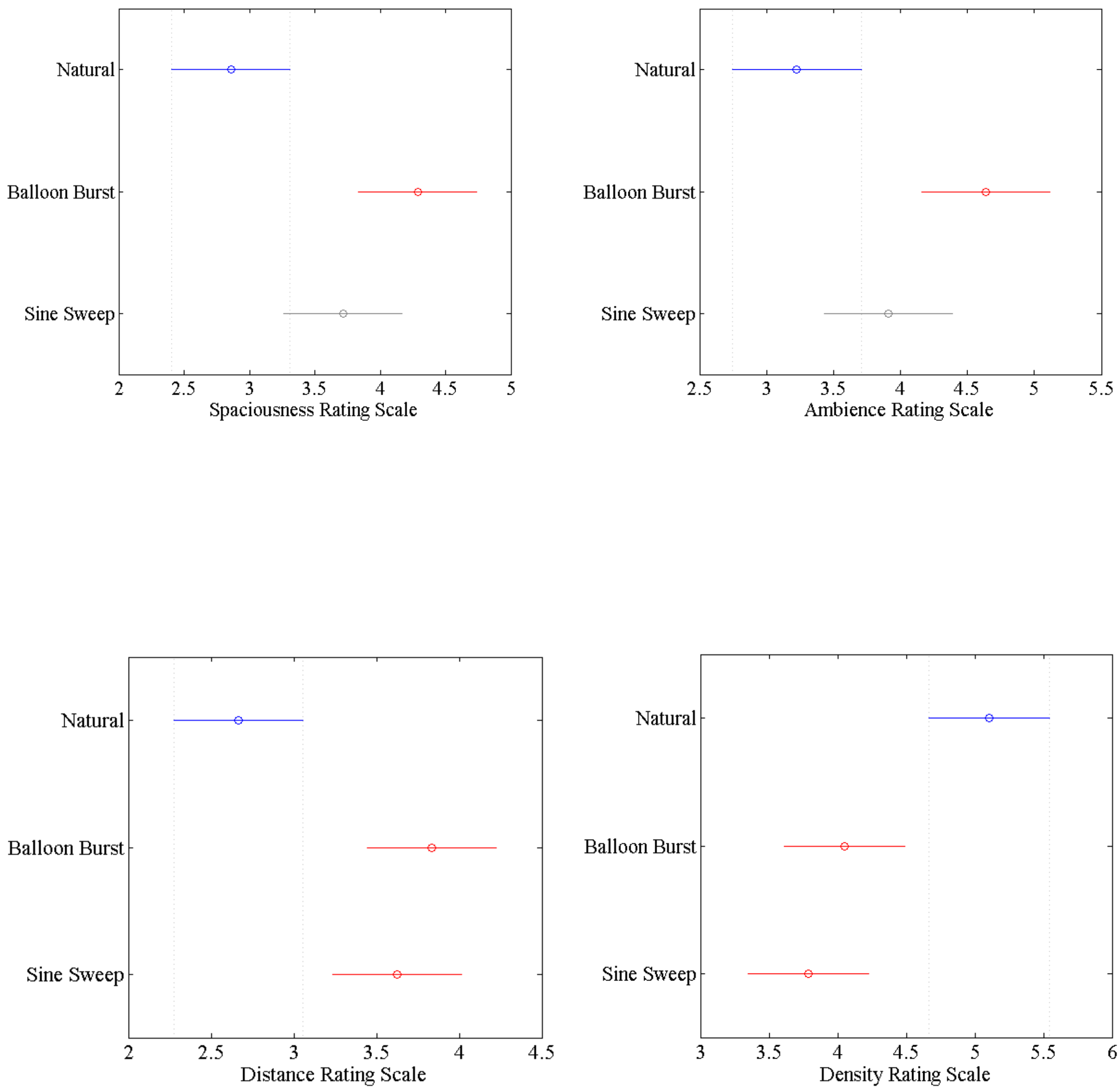


Figure 13: Illustrated are the Means and the Confidence Intervals of the three reverberation excerpts for the *Cello* in *C1*

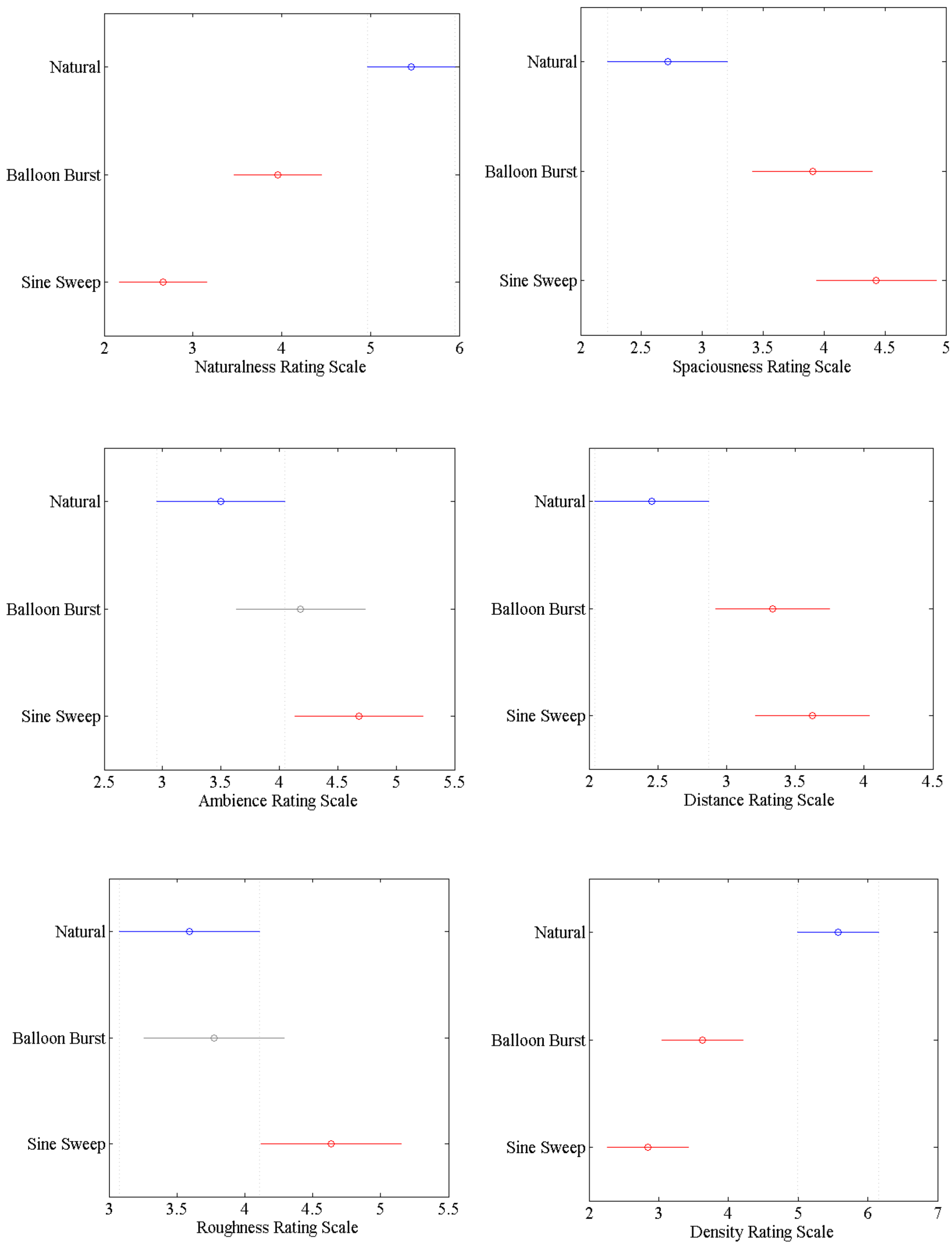


Figure 14: Illustrated are the Means and the Confidence Intervals of the three reverberation excerpts for the *Piano* in *C1*

The post-hoc tests revealed how the perception of the reverberation types varied according to the perceptual scales with respect to instrument type and space. The results are summed up below:

i) *Piano excerpts in the Church*: For the *Spaciousness* scale the sine sweep IR reverberation type was rated to be significantly less "spacious" than the other two reverberation types. For the *Distance* scale the balloon burst IR reverberation type had significantly higher mean than the other two reverberation types and the natural reverberation type had a significantly higher mean than the sine sweep IR reverberation type. For the *Density* Scale the sine sweep IR reverberation type had significantly higher mean than the other two reverberation types and the natural reverberation type had a significantly higher mean than the balloon burst IR reverberation type.

ii) *Cello Excerpts in C1:* For the *Spaciousness* scale, the balloon burst IR reverberation type had significantly higher mean than the natural reverberation type. For the Ambience scale in C1, the balloon burst IR reverberation type had significantly higher mean than the natural reverberation type. For the *Distance* scale, the natural reverberation type had a significantly lower mean than the other two artificial reverberation types while for the Density scale the natural reverberation type had a significantly higher mean than the other two artificial reverberation types.

iii) *Piano Excerpts in C1:* For the *Naturalness* scale, the natural reverberation type had a significantly higher mean than the other reverberation types and the balloon burst IR reverberation type had a significantly higher mean than sine sweep IR reverberation type. For the *Spaciousness* scale, the natural reverberation type had a significantly lower mean than the two other reverberation types. For the *Ambience* scale, the sine sweep IR reverberation type had significantly higher mean than the natural reverberation type. For the *Distance* scale the natural reverberation type had a significantly lower mean than the other reverberation types. For the *Roughness* Scale, the sine sweep IR reverberation type had significantly higher mean than the natural reverberation type. For the *Density* scale, the natural reverberation type had a significantly higher mean than the other reverberation types.

4.5 Preference Ratings

The preference rating for the reverberation types for each of the instruments in both the spaces was given by the participants. They were asked to rank them in order, from the most preferred to least preferred. In the tables below: *pref 1:* Most preferred, *pref 3:* Least preferred

CELLO	**Natural**	**Balloon**	**Sine Sweep**	**OBOE**	**Natural**	**Balloon**	**Sine Sweep**
Pref 1	11	9	7	**Pref 1**	9	5	13
Pref 2	13	5	9	**Pref 2**	5	11	11
Pref 3	3	13	11	**Pref 3**	13	11	13
PIANO	**Natural**	**Balloon**	**Sine Sweep**				
Pref 1	3	3	21				
Pref 2	22	1	4				
Pref 3	2	23	2				

Table 7: Preference ratings count of the participants for the Church Excerpts

CELLO	**Natural**	**Balloon**	**Sine Sweep**	**OBOE**	**Natural**	**Balloon**	**Sine Sweep**
Pref 1	16	6	5	**Pref 1**	15	6	6
Pref 2	9	3	15	**Pref 2**	5	14	8
Pref 3	2	18	7	**Pref 3**	7	7	13
PIANO	**Natural**	**Balloon**	**Sine Sweep**				
Pref 1	22	3	2				
Pref 2	3	16	8				
Pref 3	2	8	17				

Table 8: The participant's preference ratings count for the C1 Excerpts

The results above are summed up below:

i) For the Church excerpts the artificial sine sweep reverberation type was preferred most (41 times) with the artificial balloon burst reverberation type was preferred least (47 times).

ii) For the C1 excerpts the natural reverberation type was preferred most (53 times) while both the artificial excerpts were preferred least (Balloon Burst reverberation type: 33 times, sine sweep reverberation type 37 times).

iii) For both the spaces put together Sine Sweep reverberation type (54 times) has been preferred more than the Balloon Burst reverberation type (31 times).

iv) The level of agreement between the participants was higher in the C1

v) The Piano had a higher level of agreement than the other instruments for both the spaces.

5. Discussion

The main aim of this thesis study was to investigate the perceived differences between natural and convolution reverberation. A listening experiment was conducted where participants had to rate various musical excerpts. For creating the excerpts, there were three instruments (Cello, Oboe and Piano) used which were recorded in a dry studio. Two different spaces (Church and C1) were used whose reverberation pattern was captured using two different methods apart from recording its natural version. After having convolved the IRs of those spaces with the dry musical excerpts, the 18 musical excerpts were rated by 27 participants according to six perceptual scales. The preference ratings of the three reverberation excerpts were also taken after having grouped them according to space and instrument type. The data collected was then analyzed to find out the collinearity between the scales. Significant differences between the perceptions of the three reverberation types (natural, balloon burst, sine sweep) irrespective of the space or instrument type were calculated using a three-way ANOVA. One-way ANOVAs was also conducted to find out how the spaces and instruments affected the perception of the three reverberation types. Inferences and observations from the analyzed data are discussed in this section. Explanations for the results are given where possible.

5.1 Collinearity of Rating Scales

Six perceptual scales were used for the ratings. The correlation between the six scales was calculated and reported in **Section 4.2.** It was found that correlations varied according to scale type and space.

There are no significant correlations between the scales for the Church space, whereas for C1 four scales correlated significantly. This suggests that all the six scales in the Church have their own individual perceptual relevance, with each of them being perceived differently by the ear. Therefore the perceptual scales used are more effective in the Church, a larger space with a higher reverberation time, than C1. This suggests that the six perceptual scales used have more relevance and are more effective in a bigger space rather than a small one, and in a longer RT60 rather than a shorter one.

In C1, the Spaciousness and Distance scales have the highest correlation of 0.922. This could be attributed to the fact that both these scales deal with the sense of space and distance, which have the same unit of measurement. Also with C1 being a small space, the perceived distance from the sound source (Distance Scale) and the perceived space within which it is present (Spaciousness Scale) could be similar as the ear is dealing with smaller distances. Therefore this overlapping of small distances could have caused the significant correlation.

Further, both these scales negatively correlate with the Density Scale. This means that the participants perceived a sound source in a large space and further away from them, to be less dense and more scattered. This could be explained as more energy dissipates as a space gets larger, in turn causing these scales to negatively correlate.

Ambience Scale also correlates positively with the Distance and Spaciousness Scales. Therefore the participants perceived an excerpt in smaller spaces, and situated in smaller distances to them were less ambient and drier. This fits well as smaller spaces are drier with shorter RT60.

Using similar reasoning, it could be said that larger spaces are wetter and have longer RT60, which should in turn result in Ambience, Distance and Spaciousness Scales to correlate positively. This was the case with these scales in the Church, however the correlations were not at $p<0.05$ level. It still remains to be seen why significant correlations occurred in C1 (a smaller space with a RT60<1 sec) while this was not the case with the Church (larger space with RT60>2.5sec).

5.2 Perceived Differences between Reverberation Types

The priority of this thesis study was to investigate the perceived differences between the three reverberation types. With there being three primary factors, reverberation type, space and instrument type, a three-way ANOVA was performed for each of the six perceptual scales taking all the data into consideration. The main focus of this analysis was to investigate the difference in perception of the reverberation types. Therefore the results of the first Main effect, that of reverberation types were reported in **Section 4.3**, followed up by the results of the post-hoc tests.

The post-hoc result for the Naturalness scale suggests that the "artificiality" in both the artificial excerpts could be detected by the participants. This suggests that more research has to be conducted and better ways of recording IRs need to be found to improve the perception of "*naturalness*" for the artificial reverberation types.

In three of the scales (*Spaciousness, Ambience, Distance*), the artificial balloon IR excerpts were significantly different from the natural excerpts. Assuming the artificial reverberation methods would like to mimic as closely as possible the natural reverberation of a space, the sine sweep method to acquire the IR is better than the balloon method as its excerpts were perceived consistently less different and much closer to the natural excerpts than the artificial excerpts using the balloon IRs. Also as two of the scales (*Spaciousness and Distance*) deal with a sense of space and distance, it suggests that the balloon method for recording IR doesn't translate a sense of space and distance as effectively as the Sine Sweep method.

5.3 Perceived Differences between Reverberation Types by Space and Instrument Type

Though the three-way ANOVA showed the differences in perception of the three reverberation types, it did not explain how the other factors affected their perception. To gain more insight into how the other two factors, space and instrument type, affected the perception of the reverberation types, the one-way ANOVAs were performed and reported in **Section 4.4.**

There were more significant differences perceived in C1 rather than the Church. This suggests that the two artificial reverberation types are better suited to longer RT60 and larger spaces as less significant perceptual differences can be heard.

No significant perceptual differences could be in heard for the Oboe excerpts in either of the spaces. This suggests that the artificial reverberation types used are better suited for Oboe rather than Piano or Cello.

For the Cello excerpts, four scales had significant perceptual differences between the reverberation types for C1 whereas there were none in the Church. This suggests that the artificial reverberation types used are better suited for Cello in smaller spaces than larger spaces, having a shorter RT60.

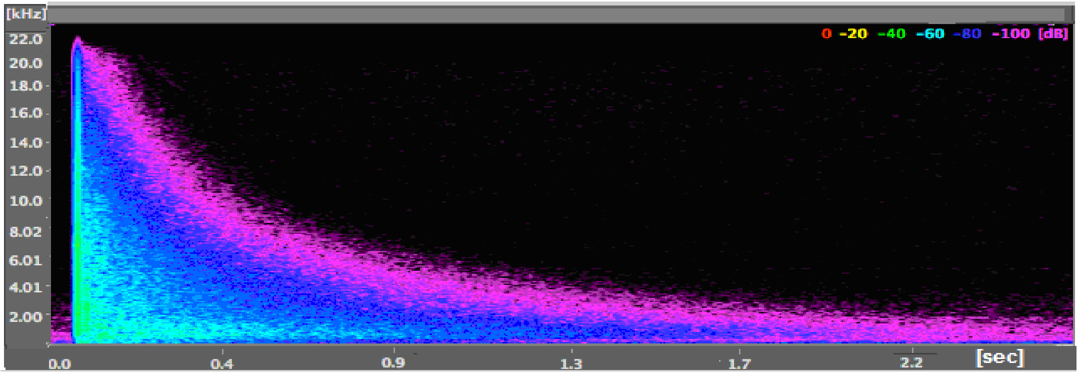
The Piano excerpts had more significant perceptual differences in both the spaces than the Cello or Oboe. This suggests that the artificial reverberation types used are not as suited for Piano as they are for the other two instruments. The timbral qualities of these instruments could have influenced this outcome. The attack/decay times of these instruments could have also played a role. Piano has a more dense spectrum as multiple tones can be played at the same time creating more energy and frequency content whereas for the Oboe only a single tone can be played. This translates to the fact that more lower fundamental frequencies are being produced by the Piano. A study regarding audibility of inharmonicity in stringed instrument tones by Jarvelainen (2001) claims that inharmonicity is more clearly audible for lower fundamental frequencies than other frequencies. Therefore that claim applied to this case, results in the Piano having a higher inharmonicity than the Oboe. This added to the presence of more partials and higher harmonic content could have caused more differences to be perceived by the participants.

5.4 Preference Ratings

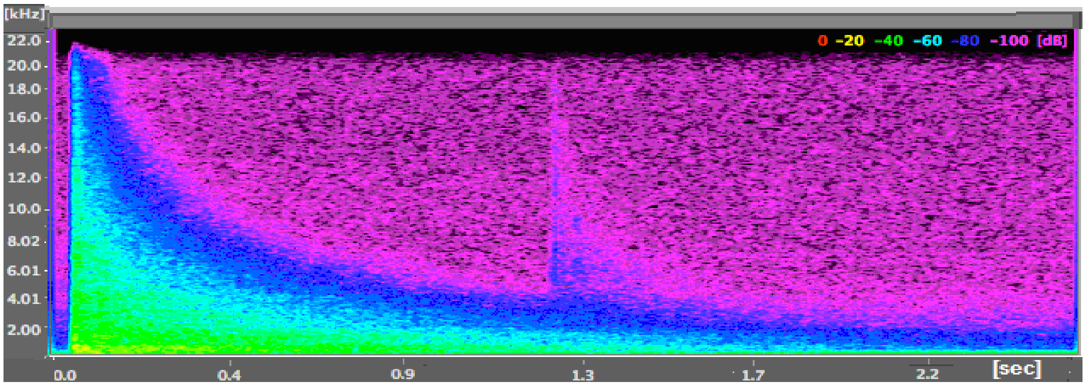
In the Church, the Sine Sweep reverberation type excerpts were preferred most which suggests that listeners preferred it to its natural counterpart. However in C1, the Natural excerpts were preferred most. This suggests that acquiring an IR of a space using the Sine Sweep method is more suitable for larger spaces with longer RT60 rather than a space with RT60 below one second.

Spectrograms and Energy Decay waveforms of IRs acquired using the sine sweep and balloon methods are in Figure 16 and 17.

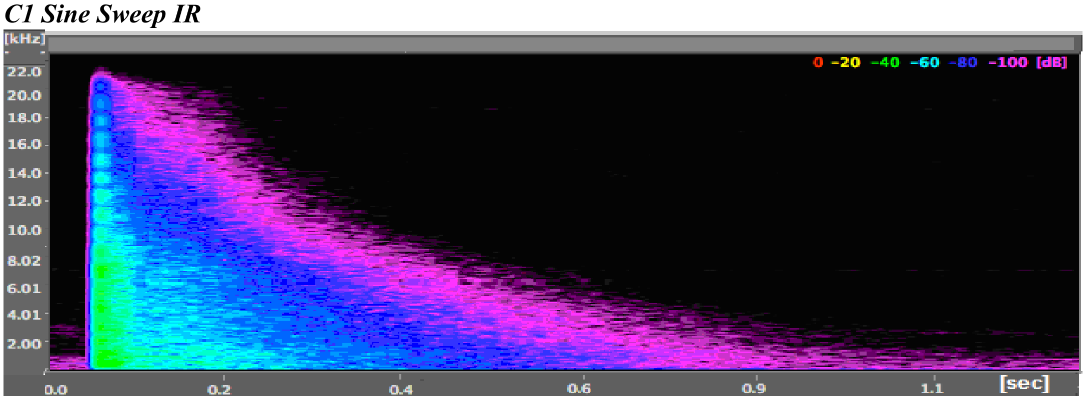
Church Sine Sweep IR



Church Balloon Burst IR



C1 Sine Sweep IR



C1 Balloon Burst IR

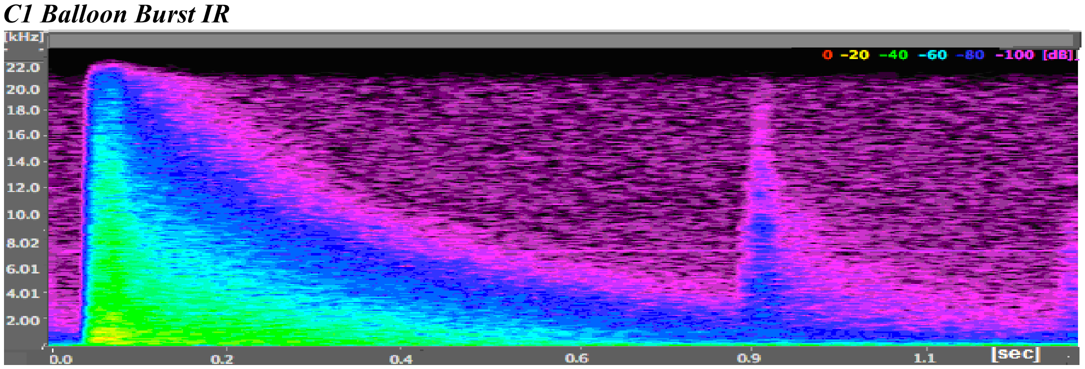
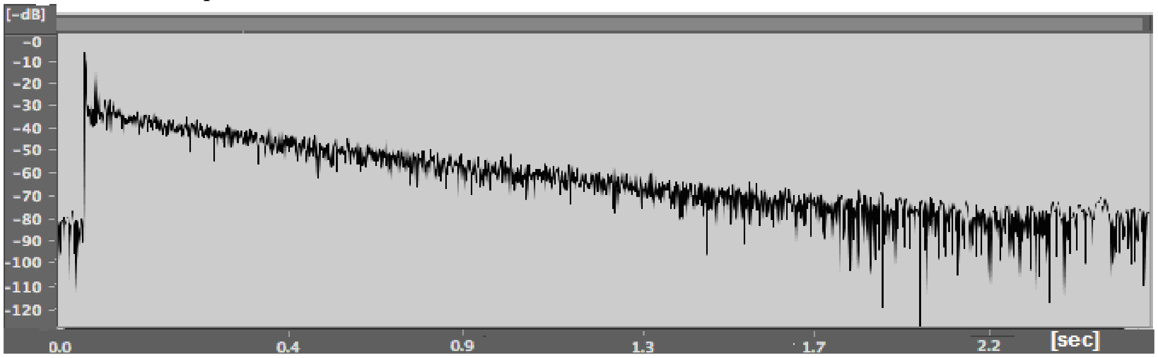
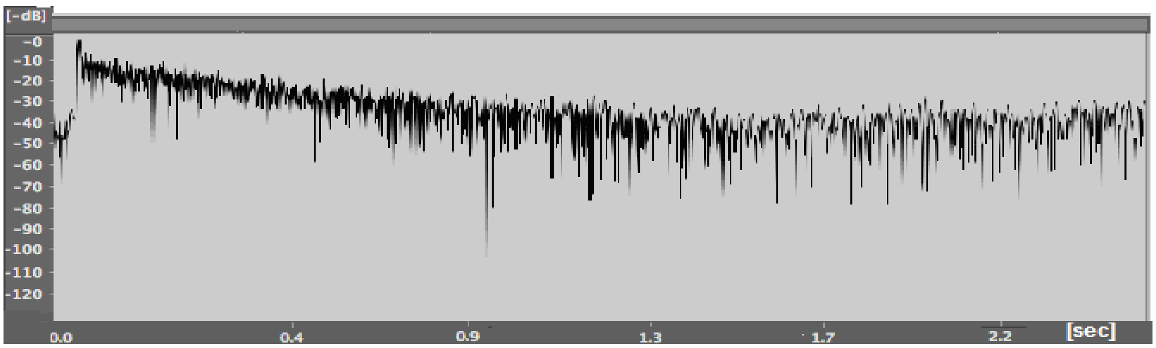


Figure 15: Spectrograms of the IRs

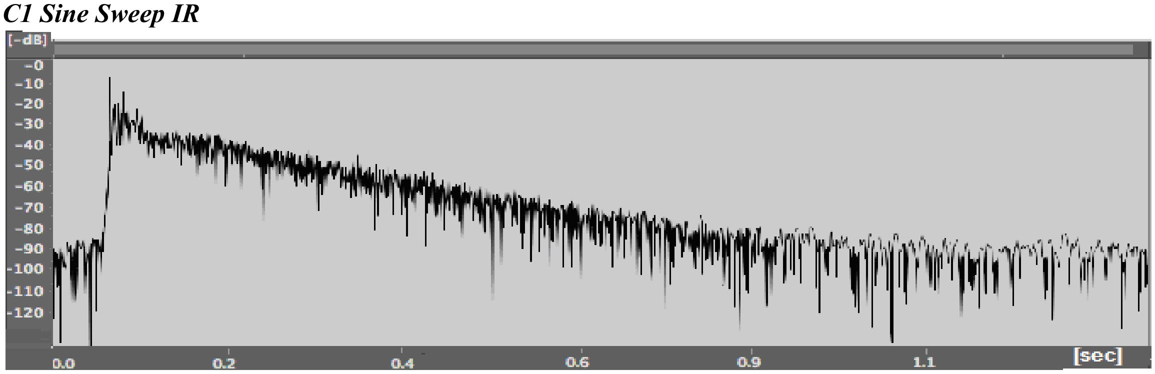
Church Sine Sweep IR



Church Balloon Burst IR



C1 Sine Sweep IR



C1 Balloon Burst IR

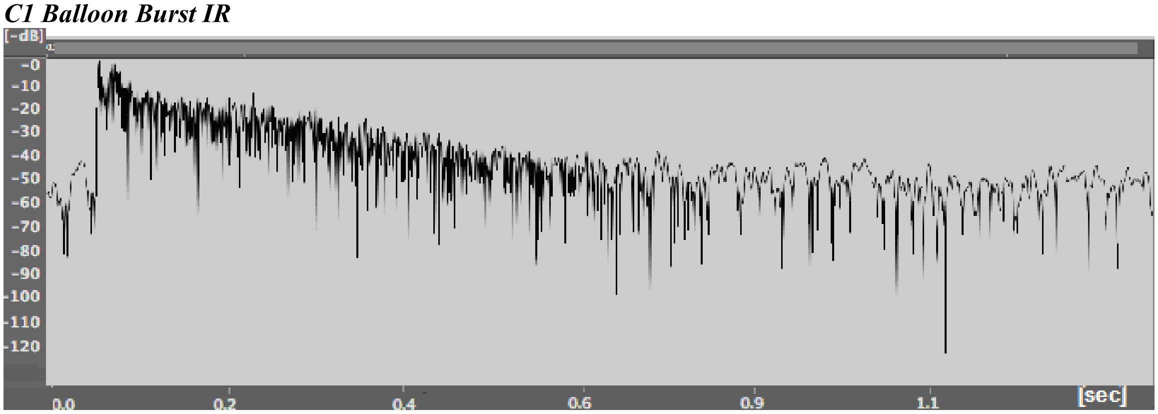


Figure 16: Energy Decay Waveform

It can be observed for the Balloon Burst IR in both spaces that there is a sudden unexpected spike of frequencies half way through the spectrograms. This was the microphone picking up the impact of the busted balloon falling unto the floor. The maximum amplitude of these boosted frequencies is -60dB with most of them at -80 dB. Therefore these frequencies even though present are only faintly audible when listening to the IR by itself. However by convolving those with the dry excerpts could have resulted in some abnormalities that could have been perceived up by the listeners. In addition to this, there is a more uniform decay of energy for the Sine Sweep IRs in the energy decay waveforms. The amplitude of the Sine Sweep IRs fades out to around -80dB for both the spaces. For the Balloon Burst IRs it fades out to a relatively much higher amplitude level at -30dB in the Church and -40dB in C1. The culmination of both these peculiarities could have resulted in the Sine Sweep excerpts being the most preferred at 54 times and the Balloon Burst IR excerpts being most preferred at 32 times and least preferred at 80 times.

6. Conclusion

6.1 Overview of Thesis

This thesis investigated the perceptual differences in natural and convolution reverberation types in 5.0 surround sound. The topic of reverberation was introduced and its artificial development and progress was discussed. In the literature review, pertinent studies regarding reverberation were discussed. Perceptual studies in algorithmic and convolution reverberation, artificial reverberation algorithms in surround sound were detailed. Also, its application in practical fields such as Ambiophonics, Speech Intelligibility was discussed.

After having created a research gap in the field of convolution reverberation perception in surround sound, an empirical study was outlined and presented. With the focus of the study being the differences in perception of natural and artificial reverberation types, a listening experiment in surround sound was designed to effectively highlight these reverberation types using different dry musical excerpts of three different instruments (Cello, Oboe, and Piano). The process of using two spaces (Church and C1) with different RT60s and further capturing its reverberation pattern (IR) in three different ways (natural, balloon burst IR and Sine Sweep IR) and convolving them with the dry musical excerpts was detailed. The use of six perceptual scales for the listening experiment was explained and the different ratings processes mentioned.

The several methods used to analyze the collected data from 27 participants were detailed. After having explained the outlier removal process, the collinearity of the six perceptual scales was investigated by calculating the correlation between the scales. The perceived differences between the three reverberation types for each of the scales were investigated using a three-way ANOVA. Also several one-way ANOVAs were conducted by isolating the factor levels to investigate how the spaces and instrument type affected the perception of the three reverberation types. The Tukey-Kramer post-hoc tests provided a clearer understanding of the interaction between the reverberation types according to the instrument type and the space. These differences were illustrated efficiently. A summary of the preference results for the 18 excerpts was also presented.

With there being many variables (three reverberation types, three instruments and two spaces), concrete and conclusive inferences encompassing all the variables was a difficult process. After having reduced the variables by isolating the space and instrument type the findings proved to be very insightful and rather interesting. However few of the results were quite difficult to interpret and could not be easily understood or applicable. Though all the reasons and justifications for the results could not be provided, the findings pave the way for more detailed and specific avenues of research. A summary of the key findings is offered below:

- The "artificiality" in the artificial excerpts could be perceived by the participants
- The sine sweep reverberation type is consistently closer and more similar to the natural reverberation of the space rather than the balloon method
- The two artificial reverberation types are better suited to longer RT60s and larger spaces
- The two artificial reverberation types are best suited for Oboe and least suited for Piano
- The six perceptual scales used have more relevance and are more effective in bigger spaces with longer RT60s rather than a smaller spaces with shorter RT60s
- Sine sweep reverberation type was preferred more than the natural reverberation type for the larger space with the longer RT60

6.2 Implications and Future Directions

Though various instruments and spaces have been considered for the investigation into the natural and convolution reverberation types, there is still a considerable amount of other variables to consider. The analysis of these variables could involve using advanced analytical techniques, which would require more time.

Firstly, frequency analysis into the three pieces could give more insight into why they were perceived and rated in the manner they were. This can be achieved by analyzing the three pieces when dry and after having been modified to add the three reverberation types. The MIR Toolbox, developed at the University of Jyväskylä (Lartillot, Eerola & Toiviainen, 2007) could be used for this purpose. Similarly the 4 IRs could be analyzed using the same toolbox to gain more insight into how they are different from each other, which would help better explain the listener ratings.

Secondly the collected data could not be analyzed completely. With the focus being on the three natural and convolution reverberation types, interaction between these variables and spaces and instruments was not completely investigated. Completely analyzing the three-way ANOVA, considering the interactions between the three factors is a possible future plan. Significant results yielded in this analysis could mean performing different one-way ANOVAs to the ones presented in this thesis. These could give more insight into how the reverberation types, instruments and spaces were perceived differently from each other.

Thirdly, other variables to consider are the equipment used. The frequency response of the different microphones could be analyzed as they affect the outcome of the ratings. Different microphones were used for recording the dry excerpts and the IRs. Also the frequency response of the surround speakers used for the listening experiment and the monitors used for the dry excerpts and sine sweep playback in the spaces can be analyzed.

The results have shown that the instrument type affects the perception of the artificial reverberation type. This could imply that the IRs acquired from a space could be modified digitally with the use of various filters and other digital processing techniques to better suit the instrument type. Using multiple instruments for the same musical piece can also be considered. The sine sweep method was preferred more than the natural excerpts for large spaces. This would imply that this method seems to be working and more research into this field could prove useful. Further, the sine sweep method could be improved for smaller spaces with RT60s less than a second. Research could also be conducted into improving the IR acquired from the balloon burst method by studying more closely the sine sweep method.

The results of the post-hoc tests for all the scales for the one-way ANOVAs proved to be hard to interpret. It provided information about how the perception of each of the scales significantly differed for the reverberation types in a particular space and for a specific instrument. However, they could function as a useful guide for attempting to make artificial reverberation to be perceived more natural according to these perceptual dimensions. If, for instance, an artificial reverberation algorithm can be designed to make physical adjustments according to perceptual parameters, then the 'Spaciousness', 'Ambience', 'Roughness', 'Density' and 'Distance' dimensions can be adjusted to affect the artificial reverberation and hence make it sound more similar to natural reverberation. Such a study would be complimentary to the study conducted by Czyzewski and Sankiewicz (1988), which was briefly mentioned in **Section 2.2.2**, which discusses objective parameters such as frequency band, initial time gap, reverberation time which directly influence subjective ratings and assessments.

There is more analysis to be conducted to comprehensively take into account all the variables and state the perceived differences for the reverberation types in surround sound for those spaces and instruments. However the work done as of now still gives significantly new, fresh and thorough insights which add to the body of research that has been conducted in this field. Also, using musical excerpts to compare the perception of various reverberation types and IRs is a simple, effective and elegant method. To conclude, it can be asserted that the thesis has successfully established a methodology that can be used for future perceptual studies in artificial reverberation which can aim to thoroughly investigate and improve its perception.

6.3 Closing Thoughts

Music listening is the backbone for music production. It could be said from an origins of music point of view that originally reverberation did not play a big role in music listening as it was mainly produced in open spaces, which meant there was no significant reverberation. However music listening has evolved with the evolution of technology. Music listening taking place indoors meant that reverberation became an essential factor. This factor has evolved into reverberation becoming necessary for music listening. With almost all music recordings being conducted in small dry spaces, the use of artificial reverberation has become exceedingly important. This evolution of the necessity for artificial reverberation for an optimum music listening experience is an interesting study in itself.

The appeal of convolution reverberation lies in the fact that the feel of a room can be captured and reproduced and added to any audio. This technique is already used for movies and cinema production where IR of the location of filming is captured. This IR is later convolved with the dialogue recorded in a dry studio to give an authentic feel to the audio output. In this age of consumerism, hype and exaggeration, famous music halls and auditoriums across the world are gaining popularity and the sound and feel of those spaces could become a marketable product.

This put together with fast growing use and development of convolution reverberation results in more research being conducted in this field. Therefore this thesis study has examined a relevant and current subject which will help gain better understanding of the differences perceived between convolution and natural reverberation types in 5.0 surround sound.

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Appendix 1

	Main Effect 1: Reverberation Type (N,B,S)		**Main Effect 2: Space (Church,C1)**		**Main Effect 3: Instrument(C,O,P)**	
	F Value	Significance Value	F Value	Significance Value	F Value	Significance Value
Naturalness	9.83	**0.0001**	0.28	0.59	4.53	**0.0115**
Spaciousness	5.18	**0.0061**	270.71	**0**	0.67	0.5105
Ambience	3.21	**0.0415**	48.11	**0**	0.42	0.6559
Distance	5.83	**0.0032**	203.47	**0**	7.27	**0.0008**
Roughness	2.9	0.0563	24.94	**0**	22.32	**0**
Density	13.51	**0**	37.77	**0**	13.64	**0**

Interactions	**Main Effect: 1 vs 2**		**Main Effect: 1 vs 3**		**Main Effect: 1vs2vs3**	
	F Value	Significance Value	F Value	Significance Value	F Value	Significance Value
Naturalness	4.85	**0.01**	4.37	0.11	4.4	**0.002**
Spaciousness	17.18	**0**	5.05	**0.03**	3.88	**0.004**
Ambience	1.58	0.21	16.52	0.07	3.2	**0.014**
Distance	17.12	**0**	6.31	**0.002**	6.92	**0**
Roughness	1.07	0.35	1.82	0.16	1.33	0.26
Density	11.5	**0**	4.77	0**.05**	7	**0**

Table 9: F value and Significance results for the three-way ANOVA for all the variables for all the six perceptual scales. p<0.05 are highlighted.

Tukey-Kramer multiple comparison post-hoc tests were conducted on results with p<0.05 results for Main Effect 2 and 3. Below, figure 17 illustrates the significant perceptual differences between the means of the Church and C1 excerpts according to the six perceptual scales irrespective of reverberation or instrument type. Figure 18 illustrates the significant perceptual differences between the means of the Cello, Oboe and Piano excerpts according to the six perceptual scales irrespective of reverberation type or space.

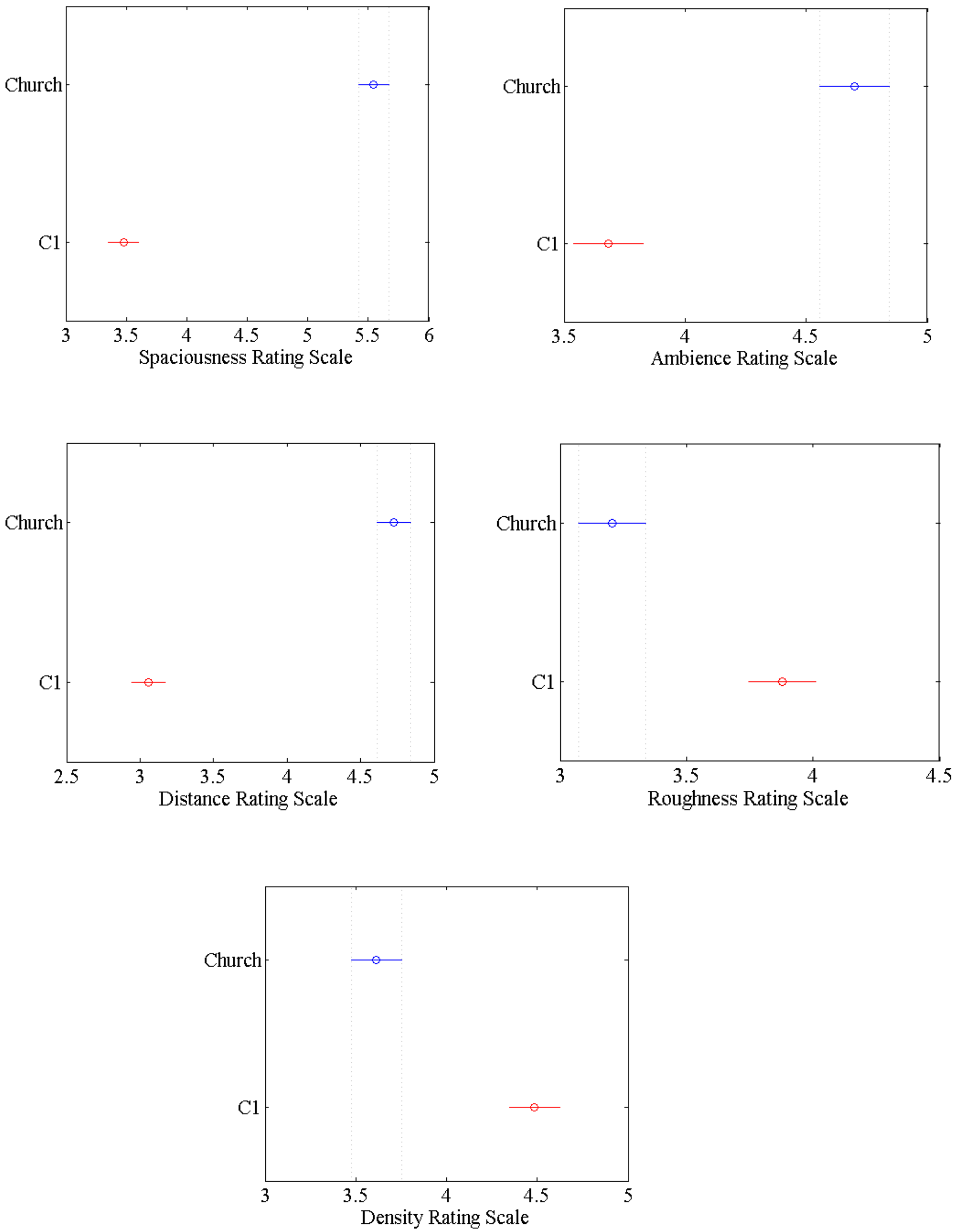

Figure 17: Illustrated are the Means and Confidence Intervals of the Church and C1 excerpts

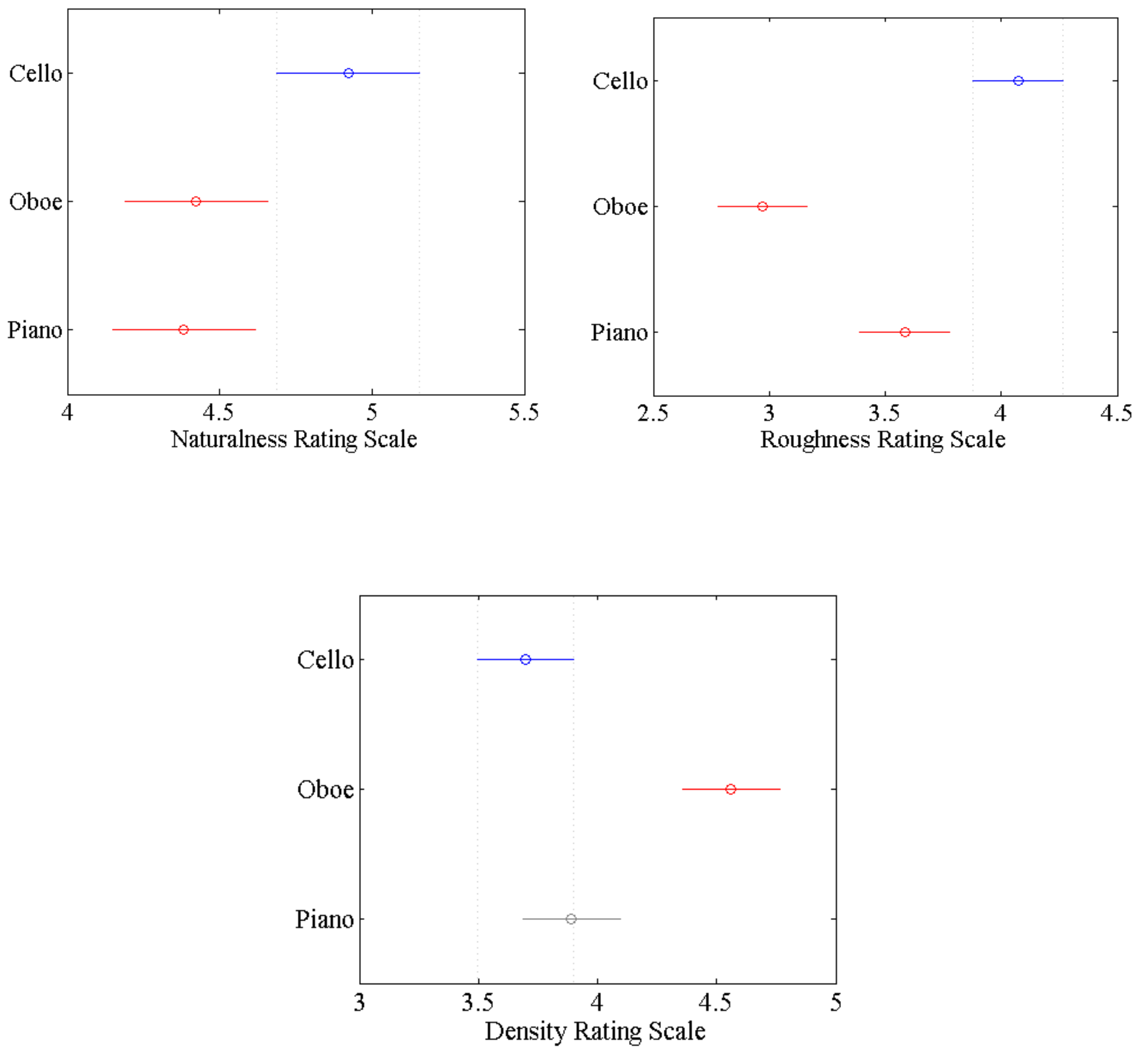

Figure 18: Illustrated are the Means and Confidence Intervals of the Cello, Oboe and Piano excerpts