Design of a Reverb Plugin and Evaluation of the Quality of Convolution Reverb

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Abstract

Convolution reverb is the process used for reverberating a signal by an impulse response of an actual space. The product of this operation is a third signal, containing reverberation of the space where the impulse response is captured. This project is based on creating a convolution reverb plugin on MaxMSP and evaluating the perceptual quality of convolution reverb.

The real-time fast convolution reverb plugin was implemented using multiple frequency delay line non-uniform partitioned convolution method on MaxMSP. An experimental design methodology was introduced in order to evaluate the perceptual quality of convolution reverb. In order to realise, anechoic drum kit music samples were recorded and re-recorded in a chamber to form the control signal. Also, using same equipment and setup, an impulse response was captured in the same chamber to form the test sample. These samples were then used for subjective analysis in a pair-wise categorical preference type listening test. The samples were also analysed in their spectrogram views in order to analyse the quality objectively.

Listening tests were designed so that realism, quality and personal preference categories were present. Null hypothesis was proved, where the realism difference of the two samples resulted in 52% of the participants preferring chamber reverb (control signal). However personal preference category resulted in 61%of the subjects preferring the chamber reverb. This was justified by objective analysis, where the convolution reverb shown to be having a faster decay rate for high frequency bands, thus sounding unnatural.

Acknowledgements

I would like to thank to my family, Resat Shenhuy, Ilhamiye Shenhuy and Boran Shenhuy for their unconditional support throughout my entire life.

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1.0 INTRODUCTION

1.1 Problem definition

Convolution reverb is an effect used to reverberate a signal by convolving the signal with an impulse response of an actual space. Any physical space can be emulated by using this technique, thus it is a widely used artificial reverberation technique for film and audio applications. Although its wide use, lack of a freely available MaxMSP plugin is an issue. Another problem in this subject is that the deficiency of a reliable study comparing convolution reverb to a natural reverb such as a chamber reverb.

1.2 Scope

The scope of this project covers activities needed associated with the achievement of objectives stated in 1.5. These include research for implementing real-time, fast convolution and carefully designing of the listening tests to truly assess the perceived quality of convolution reverberation. The scope of this project does not cover the testing of the other auralisation approaches and research for the objective analysis tools of the artificial reverberation algorithms.

1.3 Rationale

The topic was chosen in order to provide a freely available MaxMSP plugin for convolution reverberation for digital media industries and to provide a reliable study on the evaluation the quality of convolution reverb compared to natural reflections in a physical space. Frequency analysis and listening tests will supply both subjective and objective analysis on the quality of convolution reverb. The product of this project can benefit a large scale of amateurs and professionals working in the digital media industry.

Convolution reverb can be widely used in many digital media industries like TV, film and games where the receiver should get an accurate perception of space when audio is convolved with the impulse response of the target space. This means impulse responses for convolution reverb must be recorded from the target for audio and games, or actual spaces for film industry.

Convolution also can be used for adding other effects to the signal rather than reverb. This project will design and conduct experiments for subjective evaluation of the perceptual quality of convolution as a reverb effect.

1.4 Aims

To design a fast convolution patch on Max/MSP and to evaluate quality of convolution reverb compared to recordings from the physical space from which impulse response recorded.

1.5 Objectives

- To recall the theories and processes behind real-time convolution and efficient collection of impulse responses.
- To design a Fast Convolution patch on Max/MSP
- To record IR for testing the patch
- To apply the patch on recorded IR and perform frequency analysis
- To create testing criteria for listening tests
- To perform listening tests
- To evaluate with respect to frequency analysis and listeners‘ opinions

1.6 Background information

Artificial reverberation has been a tool used in commercial music recording nearly a century, with the methods of chamber reverberators in early days, spring reverberators in late 1930's, plate reverberators in late 1950's and with the rise of the digital age and digital signal processing methodologies and tools, digital reverberators and finally convolution reverb for the last 30 years. However real-time convolution reverb is fairly new to the industry as opposed to the techniques stated above. The implementation of real-time, fast convolution reverb was viable with regards to high CPU power of recent years.

Testing the quality of artificial reverberators has been a common and important subject among the researchers through last four decades. However these studies are focused on the different techniques of artificial reverberators rather than convolution reverb. The latest papers on this subject were published in 2007. These studies are mentioned and analysed in the later sections of this report.

2.0 Review of Existing Knowledge

2.1 Introduction

This section combines the existing knowledge with the findings from broad research into areas of propagation of sound, reverberation, digital signal processing, artificial reverberation techniques and methods of evaluation of the quality of artificial reverberation that are crucial for understanding the concept and findings of this paper.

2.2 The propagation of sound

Sound is a mechanical wave motion that travels in air or other elastic media and can be perceived by humans or other animals (Everest and Pohlmann 2009). Propagation of sound is often described as air molecules moving back and forth forming pressure variations above and below atmospheric pressure called compressions (region of high pressure) and rarefactions (region of low pressure) (fig. 1.1).

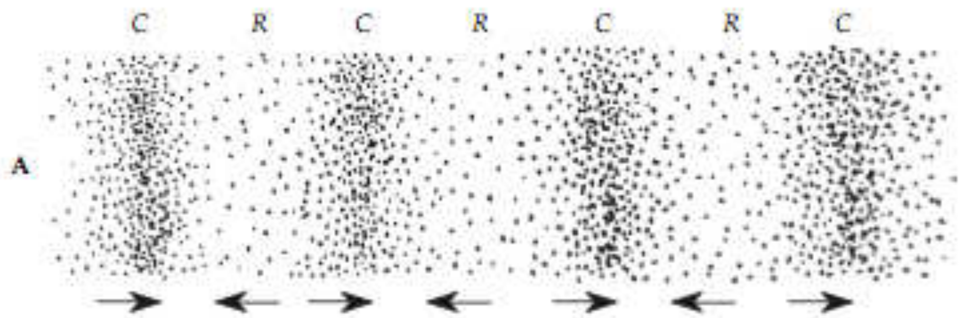

Fig. 1.1 Motion of air particles (taken from Master Handbook of Acoustics, 2009)

Speed of sound in air can be calculated by the following equation:

$$\boldsymbol{c_{gas}} = \sqrt{\frac{\boldsymbol{E_{gas}}}{\boldsymbol{P_{gas}}}} = \sqrt{\frac{\boldsymbol{\gamma P}}{(\frac{\boldsymbol{PM}}{\boldsymbol{RT}})}} = \sqrt{\frac{\boldsymbol{\gamma RT}}{\boldsymbol{M}}}$$

Where P = the pressure of the gas

E = the equivalent Young's Modulus of the gas

R = the gas constant (8.31 $J\ K^{-1}\ mole^{-1}$)

T = temperature in K

M = the molecular mass of the gas ($k\ mole^{-1}$)

γ = the heat capacity ratio of the gas (1.4 for air)

2.2.1 Sound Intensity

Howard and Angus (2009) states that 'sound from a point source is propagated spherically and uniformly in all directions'. As the area of a sphere is $4\pi r^2$, the area of a selected region on a sphere varies as the square of its radius (Everest and Pohlmann 2009). Therefore sound intensity diminishes as the square of the radius. This is called inverse square law. Figure 1.1.1 shows an example of an area on a hemi-spherical sound field formed by a loudspeaker. Everest and Pohlmann states that 'the intensity of a point-source sound in a free field is inversely proportional to the square of the distance from the source. Therefore sound intensity of a point source can be given by the following formula:

$$I = \frac{W}{4\pi r^2}$$

Where I = intensity of sound per unit area

W = power of source

r = distance from source (radius)

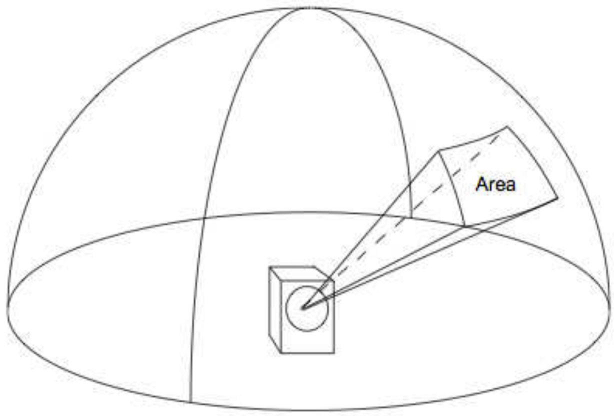

Fig. 1.1.1 Sound Intensity (taken from Acoustics and Psychoacoustics, 2009)

2.2.2 Sound Power

Sound power is a measure of the total power radiated from a point source in all directions (Howard and Angus 2009). It is given the abbreviation SWL. SWL is expressed in dB as the logarithm of a ratio of existent power level to a reference level of 1 picowatt.

$$SWL = 10log_{10}\left(\frac{W_{actual}}{W_{ref}}\right)$$

Where W_{actual} = the actual sound power level (in watts)
and W_{ref} = the reference sound power level ($10^{-12}\ W$)

2.2.3 Sound Pressure

Although sound intensity is a functional measure of describing the amplitude of a sound at a given point, it is not the usual measure for depicting amplitude of the sound. Instead, pressure is used (Howard and Angus 2009). Human perception capacity encompasses a large dynamic: a factor of 1:1000000, which extends from

20 μPa (threshold of hearing) to 20 Pa (threshold of pain) (Mommertz et al 2009). Sound pressure level is often expressed in logarithmic scale because the way humans perceive sound. It is given as a ratio of actual sound pressure to the human threshold of hearing at 1 kHz of 20 μPa (Howard and Angus 2009).

$$SPL = 20log_{10}\left(\frac{P_{actual}}{P_{ref}}\right)$$

Where P_{actual} = the actual pressure level (in Pa)
P_{ref} = the reference pressure level (20 μPa)

2.3 Reflection and sound in enclosed spaces

Howard and Angus (2009) describes free field as '… a free space that allows sound to travel without interference'. When sound wave hits a hard boundary some of its energy is absorbed and some is reflected depending on the size of the boundary compared to the wavelength of the sound and absorption coefficient of the materials (Rumsey & McCormick, 2006). Angle of reflection is equal to the angle of incidence if the reflecting surface is perfectly flat; this is called specular reflection (fig. 1.2 (a)). If the reflecting surface is not smooth, sound is scattered in all directions; this is called diffuse reflection (fig. 1.2 (b)) (Peters, 2013).

a) b)

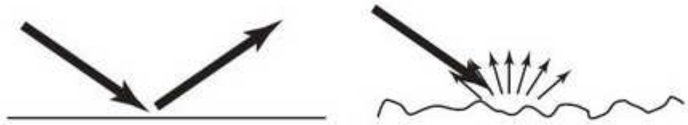

Fig. 1.2 Specular Reflections (a) and Diffuse Reflections (b) (taken from Acoustics and Noise Control, 2013)

When sound energy is radiated in a room, direct sound reaches the listener first followed by early reflections and reverberation. Perception of early reflections determines the acoustical size of the space and the position of the source in the space (Howard and Angus 2009). Reflections delayed more than 30 milliseconds are perceived as echoes (Haas effect) (Howard and Angus 2009).

The mean free path (MFP) of a room gives the average distance a sound wave travels between interactions of surfaces (Everest and Pohlmann 2009). An average MFP of a rectangular room can be calculated by using the equation below:

$$MFP = \frac{4V}{S}$$

Where MFP = the mean free path (in m)

V = Volume of the room (in m^3)

S = Surface area of the room (in m^2)

Therefore an average length of time takes a sound wave to travel between successive reflections can be predicted by dividing the MFP by the speed of sound:

$$T = \frac{4V}{Sc}$$

Where T = the time between reflections (in s)

and c = the speed of sound (in ms^{-1})

2.4 Reverberation

As stated above sound field in an enclosed space is divided in three regions: direct sound, early reflections and reverberation. After the direct sound and early reflections reaching the ear, a dense sound field is formed by very closely spaced reflections from all directions. This region of the sound is called reverberation (Howard and Angus 2009). Mommertz (2009) states that the reverberation of a room is the most significant acoustic quality feature.

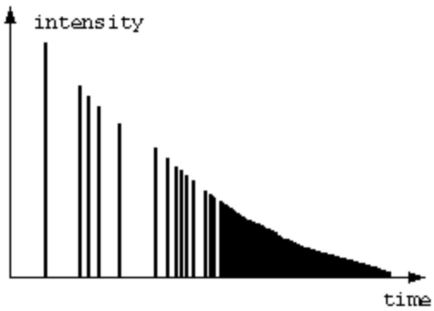

Fig. 1.3 Individual reflections as impulses on intensity over time graph

Figure 1.3 shows intensity levels of multiple reflections as impulses occurring over time. Direct sound can be seen as the most intense single impulse on the graph followed by early reflections spaced over time with relatively low intensities and reverberant field of dense, less intense reflections decaying exponentially to inaudibility.

2.4.1 Reverberation time

The time that it takes sound to decay into inaudibility is called the reverberation time (Everest and Pohlmann 2009). It is often given the abbreviation RT_{60}, which accounts for pressure level decreasing by 60 dB, into inaudibility. According to Peters (2013), W. C. Sabine’s investigation into room acoustics proved that the reverberation time, RT, of a room is dependent on its volume and amount of absorption of its surface area. Sabine’s formula is the simplest to predict the reverberation time of a space.

$$RT_{60} = \frac{O.16V}{A}$$

Where RT_{60}= reverberation time (in s)

V = volume of the room

A = total absorption of the room (in metric Sabins)

Using this equation one can obtain RT_{60} of a room but the following assumptions and limitations should be considered when dealing with large halls:

- Air absorption is neglected
- Audience absorption is neglected
- Quantity A is the total absorption of all the surfaces of the room and it is given by $A = \sum S_i\alpha_i$, where S_i is surface area of the material and α_i is the absorption coefficient of the material. In most cases doors, windows, seats etc. are not taken into account (Everest and Pohlmann 2009).

2.4.2 Artificial reverberation techniques

Everest and Pohlmann points out that music recorded in a 'dry' room often miss richness of the room sound. In the early days of studio recording, reverberation was added to the record by natural means. The process included, placing a loudspeaker and a microphone in a reverberant chamber and recording the room's response. This signal was then added to the mix. The reverberation characteristics of a room were altered by placement of glass plates around the room. Small reverberation rooms are often dictated by room modes, which cause timbral defects. On the other hand large halls were often expensive to hire for reverberation purposes. This yielded a search for practical solutions for reverberating a signal (White, 2006; Everest and Pohlmann, 2009).

2.4.3 Analogue reverb

First approach was spring reverb, where a loudspeaker was coupled at one end of a spring and a pickup on the other end. When an audio signal is playing, the waves travel along the spring and reach the microphone, but a part of the waves energy is reflected and stays in the spring, which creates the sense of reverberation over sound (Lehmann, 1996).

Next approach was plate reverberation. In the late 1950's Elektro-Mass-Technik (EMT) introduced EMT 140, a large plate of sheet metal is vibrated by a loudspeaker at one side and vibrations picked up by a pickup (two pickups for stereo use). This technique gave more control over the reverberation time by a damping pad made from framed acoustic tiles (Dennis, 2001). Plate reverberation was widely used in commercial recordings; the biggest advantage over spring reverb was its brightness (Robjohns, 1997). While this technique was widely used, it was not capable of simulating true room acoustics (Bateman, 2002).

2.4.4 Digital reverb

With rise of digital signal processing, digital reverb algorithms elevated. The earliest approach was a theorised by Schroeder in 1961, where he proposed using a combination of allpass and comb filters to create digital delay networks that will form early and late reflections that naturally occur in a space (Schroeder, 1961, 1962). Later on Moorer, in his 1979 Computer Music Journal article, developed Schroeder's approach but again this method was not capable of emulating a satisfactory true room reverberation (Bateman, 2002).

2.4.5 Convolution reverb

Convolution reverberation uses one of the most important digital signal processing fundamentals, convolution: an input signal is combined with an impulse response to output a third signal. In this method impulse response of a linear time invariant system, in this case a room, is convolved with an input signal to truly emulate the systems response to an impulse, decay over time.

2.5 Digital audio

Manolakis and Ingle (2011) states that '… a signal is defined as any physical quantity that varies as a function of time, space, or any other variable or variables'. Transforming sound waves into electrical currents by a microphone forms audio signals. A signal can be decomposed into series of sine and cosine waves called Fourier series (Smith, 1997). Jean Baptiste Joseph Fourier theorized that combining enough sine waves one could model any waveform (Howard and Angus 2009). This approach has simplified the analysis of any complex waveforms (Howard and Angus 2009).

2.5.1 DFT and FFT

DFT refers to discrete Fourier transform: calculating frequency domain data from a given time domain signal (Smith, 1997). DFT works with discrete and periodic signals; therefore it is crucial for understanding the digital signal processing.

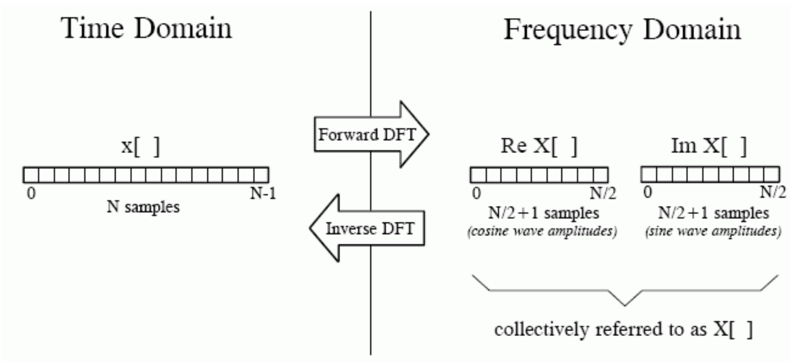

Fig. 1.4 DFT Terminology (taken from The Scientist and Engineer's Guide to Digital Signal Processing, 1997)

Figure 1.4 shows the application and terminology of DFT. Time domain input signal is translated into frequency domain, outputting real $Re\ X$ [ ] and imaginary $Im\ X$ [ ] parts. This technique is very important when performing convolution because it is computationally less expensive to perform convolution in frequency domain rather than in time domain (Smith, 1997; Lyons, 2004).

While there are several ways to compute DFT, the most efficient algorithm is the Fast Fourier Transform (FFT). Smith (1997) states that FFT reduces computation time by hundreds. He further describes the efficiency as 'flying in a jet aircraft versus walking' (Smith, 1997). It operates by decomposing an *N* point signal into *N* signals each containing a single point by bit reversal sorting algorithm. After this step each point contains frequency spectra of their selves. Last step is to combine *N* frequency in the exact reverse order (Smith, 1997).

2.5.2 Digital filters

Lyons (2004) states that '… filtering is the processing of a time-domain signal resulting in some change in that signal's original spectral contents'. In this case a room can be described as a filter, altering the spectra of an impulse giving an impulse response. There are two types of digital filters: the Finite Impulse Response (FIR) and the Infinite Impulse Response (IIR) (Marven and Ewers 1996). An IIR filter uses output samples to feed the input (recursive) therefore its output is infinite. In contrast an FIR filter, also known as non-recursive filter, only use current and past input samples (Kirk and Hunt 1999). Block diagrams of FIR and IIR filters can be seen in fig. 1.5. For this projects purposes FIR filters will be studied.

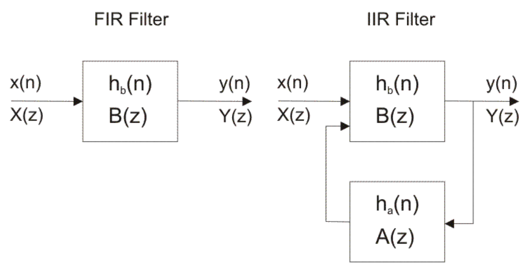

Fig. 1.5 Block diagrams of FIR and IIR filters (taken from Digital Filter Design by Zoran Milivojević)

In fig. 1.5, block diagram of an FIR filter shows input $x(n)$ passing through a system depicted as $h(n)$ giving the output $y(n)$. In digital signal processing, this is described as: $y(n)$ equals to the convolution of $h(n)$ and $y(n)$ (Lyons, 2004).

2.5.3 Convolution

Smith (2003) states that 'an impulse is a signal composed of all zeros except a single non-zero point'. Delta function is a normalized impulse, where sample number zero has a value of one and all other samples have a value of zero. Impulse response is a name given to a product signal from a system where delta function is the input (Smith, 2003). See fig. 1.6 where $\delta[x]$ is the delta function and $h[n]$ is the impulse response of a linear system.

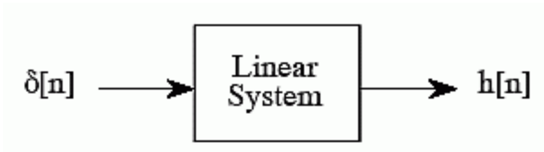

Fig. 1.6 Delta Function & Impulse Response (taken from Digital Signal Processing)

Convolution is a mathematical operation just like addition and subtraction that is used in digital signal processing for describing linear time-invariant systems and designing filters. In figure 1.7 convolution is abbreviated as '$*$' which refers to convolution operation given in the formula below called convolution integral.

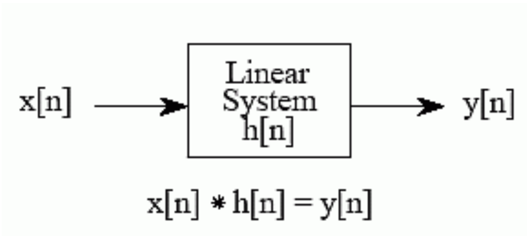

Fig. 1.7 Convolution operation (taken from Digital Signal Processing, 2003)

$$y(t) = \int_{-\infty}^{\infty} h(\tau)x(t-\tau)d\tau = h(t) * x(t)$$

The length of the output sample $y\ (n)$ is equal to the length of input sample $x(n)$ plus impulse response $h(n)$ minus one. For example if the input sample $x\ (n)$ consisting of 4 points is convolved with a 3-point impulse response $h\ (n)$ outputs a 6-point signal $y\ (n)$ (Lynn and Fuerst 1994).

2.5.4 Convolution machine

Understanding how convolution works is the single most important part of achieving the objectives of this project. Steven Smith (2003) suggests studying convolution from the viewpoint of output side algorithm in order to understand the practical side of this operation. This viewpoint analyses each output sample by finding the contributing input points. Figure 1.8 shows a convolution machine, a flow diagram of input $x\ (n)$ contributes to output $y\ (n)$.

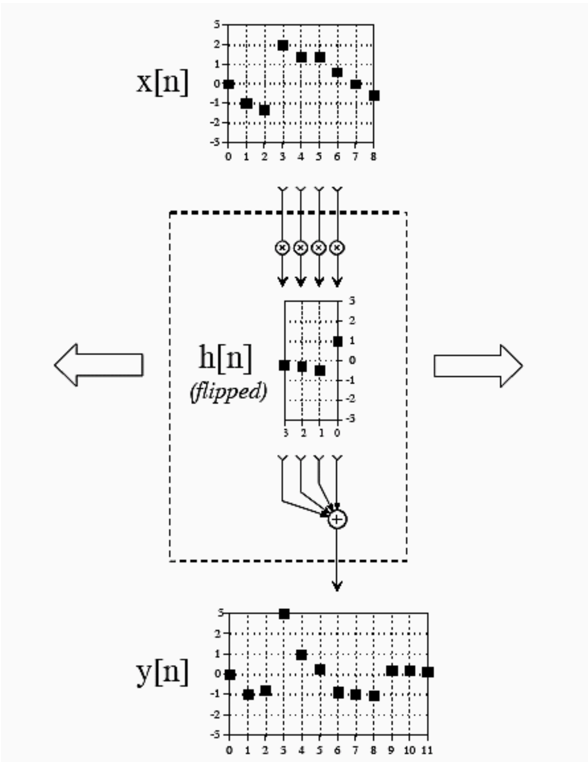

Fig. 1.8 The convolution machine (taken from S. Smith (2007)).

In figure 1.8, output sample $y\ (6)$ is obtained by multiplying $x\ (6)$ by $h\ (0)$, $x\ (5)$ by $h\ (1)$, $x\ (4)$ by $h\ (2)$ and $x\ (3)$ by $h\ (3)$, then by summing up all the values. This machine is free to move from left to right or right to left. An important part to remember is that impulse response $h\ (n)$ is flipped. In order to calculate output sample $y\ (0)$, flipped $h\ (n)$ is shifted so that sample $h(0)$ is in correspondence with $x\ (0)$. But this leaves samples $h\ (1), h\ (2)$ and $h\ (3)$ in line with the non-existent samples; this problem is overcome by simply adding samples each having a value of zero. This is called padding (Smith, 2003).

2.5.5 Overlap-add method and FFT convolution

In order to operate convolution for real-time purposes, N point input signal is segmented into overlapping partitions $(N_1 + N_2 + \ldots)$. In this case convolution is applied simultaneously and each overlapping product is summed to form the output signal. This method is called overlap-add (Lockhart and Barry 1989; Lynn and Fuerst, 1994).

For example if a 300-point signal is to be filtered by a 101-point impulse response, the output signal will be 400 samples. To implement fast convolution, input sequence is broken into 3 segments that are running from 0 to 99, 100 to 199 and 200 to 299. The end of the each segment is padded with zero's to allow overlapping. Each input segment is filtered by the 101-point impulse response resulting in a 200-point output segment. In order to form output signal, each overlapping output segment is summed. For example, to form samples running from 100 -199 output segment 1 (0-199) is added to output segment 2 (100-299) (Smith, 2003).

Smith (1997) states that after the development of FFT algorithms convolution in frequency domain became more efficient than in time domain. FFT Convolution uses overlap-add method; only difference is when the segments are transformed in frequency domain, calculations are carried out such that real and imaginary parts of frequency response and input segments are calculated and then transformed back into time domain.

2.6 Real-time convolution algorithms

As stated earlier room impulse response (RIR) contains information of the response of a reverberant space to an impulse. These RIR's are often longer than 3 seconds, which makes them almost impossible to convolve with an incoming signal for real-time purposes (Muller-Tomfelde, 1999). In order to realize this operation room impulse response is divided into segments where convolution for each block can be achieved in parallel with appropriate frequency-domain delay lines (FDL) (Battenberg and Avizienis 2011). This method is called uniform partitioned convolution where each segment of impulse response is partitioned in uniform block sizes by FFT convolution. This method has reduced time required for the operations however it is not ideal for real-time purposes. In order to accomplish enough reduction in latency, Gardner (1995) suggested using blocks of different lengths. This approach uses smaller partitions in the beginning of an IR and exponentially increasing blocks towards the end (Battenberg and Avizienis 2011). This further improvement in the partitioning scheme has reduced the latency at a stage that real-time convolution is now achievable.

2.7 Earlier studies on evaluation of the quality of artificial reverberation techniques (subjective methods)

A number of studies have been carried on the evaluation of artificial reverberation techniques. Only a small percentage of these studies assessed the perceptual differences of artificial reverberation techniques. Czyzewski (1990) in his article on Journal of Audio Engineering Society (JAES) suggests the efficiency of paired comparison tests for the subjective assessment of reverberation techniques. His method assessed the subjective preference of the individuals on a two level ''better/worse'' scale. On his earlier article (1988) with Sankiewicz, they highlight two testing methods, parametric and non-parametric methods. Parametric methods include assessing the predetermined parameters of sound. Disadvantage of this method is that the participants have to have significant background in the subject area. Non-parametric methods include comparison test methods stated earlier.

Another study carried out by Extra et al. (2006), employed a non-parametric method, Two Alternative Forced Choice (2AFC) listening test, similar to the method stated above, where participants are expected to compare the same signal reverberated with two different algorithms. This study included only one comparison category that was which signal sounds more natural. Their study concluded that objective measures couldn't replace listening tests. Another outcome was that drum samples are a good starting point for further tests.

In 2007, two more articles were published in the subject area. Marui (2007) has created most in-depth and exhaustive listening test, comprising of 225 sound stimuli. He asked the participants to relate each of the samples to three parameters depending on the amount of reverberation present on each signal. He included three test samples: a noise sample, a small orchestral piece and drums music. He also points out the importance of recording these test samples in an anechoic environment.

Another interesting study carried out by Haines et al. (2007) assessed participants' preference on three different reverberation techniques. These techniques included convolving the test audio samples with mono impulse responses, placed impulse responses, where impulse responses for each instrument depending on their position on a stage and a panned dry recording with a centre impulse response. They used three preference categories, realism, quality and personal preference to obtain subjective results. A paired comparison test was their test strategy to minimise the concerns due to participants' ability to distinguish between the samples. Their study concluded that location-based convolution reverberation is a promising technique having a high realism over other techniques. They also stated that due to preference ratings being high on location-based reverberation, this technique has a real viability in the commercial realm.

3.0 Methodology

3.1 Hardware

In this section available hardware apparatus has been analysed in order to detect and use the most appropriate ones. Important data have been tabulated in order to show in a more convenient style.

3.1.1.Microphones

The two measurement microphones at TEE studios have been analysed for their frequency response in order to capture a linear IR of the corridor. In frequency response charts, Audix TM1 shows flatter response. Therefore, all the recordings including drum audio sample has been captured withTM1.

Table 1 Available measurement microphones in TEE studios

Available Measurement Microphones	Frequency Response	Polar Pattern
Audix TM1	20 Hz - 25 kHz +/-2dB	Omni-directional
Behringer ECM8000	20 Hz - 20 kHz +/-5dB	Omni-directional

3.1.2 Loudspeakers

Most of the available monitor loudspeakers at TEE studios have been analysed for their frequency responses in order to reproduce flat playback of the recorded drum sample and sine sweeps for collection of impulse responses. Through in-depth research Tannoy has been the most suitable loudspeaker for the application. As well as its near-flat frequency response, it is capable of reproducing efficient playback of low-end frequencies as low as 33 Hz, where the competitors HS5 and Rokit 5'' are capable of 54 Hz and 45 Hz respectively. This low frequency response is extremely important as the designed audio test sample being a drum kit containing a kick drum.

Table 2 Available monitor loudspeakers in TEE studios

Available Monitor Loudspeakers	Frequency Response
Tannoy Ellipse 10 iDP	33 Hz – 50 kHz +/- 1.5 dB
Yamaha HS5	54 Hz – 30 kHz +/- 10 dB
KRK Rokit 5''	45 Hz – 35 kHz +/- 10 dB

3.1.3 Audio interfaces

Portable audio interfaces available in TEE studios have been analysed. Most important specifications are detailed below. Input and output frequency responses are as important as microphones and loudspeakers. In this category Saffire Pro has shown the least effect on responses. M-box mini 2 has failed the test as it only accommodates only a headphone output. Signal to noise ratio (SNR) is important in terms of capability of reproducing and recording clear signals. Saffire Pro has a far better SNR than the other competitor, Fast Track Pro. The last test category, digital conversion specifications, has made it obvious that Saffire Pro is the most appropriate interface to for recording and reproduction purposes.

Table 3 Available portable audio interfaces

Available Portable Interfaces	Input Frequency Response	Output Frequency Response	A/D – D/A Conversion	Multiple Outputs ?	Signal to Noise Ratio
M-Audio Fast Track Pro	20 Hz – 20 kHz, +/- 0.2 dB	20 Hz – 20 kHz, +/- 0.2 dB	16 Bit 48 kHz	Yes	-102 dB
M-Audio M-Box Mini 2	20 Hz – 20 kHz, +/- 0.1 dB	20 Hz – 20 kHz, +/- 0.25 dB	24 Bit 44.1, 48 kHz	No	Not specified
Focusrite Saffire Pro	20 Hz – 20 kHz, +/- 0.1 dB	20 Hz – 20 kHz, +/- 0.1 dB	24 Bit 96 kHz	Yes	-105.3 dB

3.2 Software

This section provides information and comparison about alternative software for programming, capturing IR's and analysis uses. Each software have been analysed for their advantages and limitations in order to detect most applicable ones. Justifications have been made towards the usefulness, user-friendliness and material costs in order to save time, effort and money.

3.2.1 Programming

Although programming software platform has been defined for this project as Max/MSP in the project aims, a research has been made to find alternative ones to justify the selection. A range of audio synthesis software is given below. First category for comparison deals with the cost of the software. In this sense, Reaktor has been eliminated in the first stage due to budget constraints where it is unattainable neither on TEE computers nor on author's personal computer. Max/MSP being reachable on TEE computers eliminates the need for a purchase. Another concern is the native platform of the software. Byond, being only available for Windows computers makes it almost impossible to use as the author's personal computer running Mac OSX and permission and stability of installing software on TEE computers is not known. The last and the most important criteria is the programming language used by the software. The language required is a real issue where, if the author is not familiar with it, it will consume much time and much effort. In this case, Faust and SuperCollider requiring functional and dynamic programming respectively are eliminated due to author's theoretical and practical knowledge constraints. The remaining software Pure Data and Max/MSP are the most convenient ones to use. In retrospect, Max/MSP has been the fundamental software used in Sound Synthesis UG2 and Digital Audio Effects UG3 modules. This makes Max/MSP a more advantageous environment compared to Pure Data.

Table 4 possible programming software comparisons

Software	Cost	Language	Platform
Faust	Free	Functional programming	Linux, Mac OSX, Windows, Unix
Pure Data	Free	Visual programming	Cross-Platform
Max/MSP v.5	$400, available on TEE computers	Visual programming	Windows, Mac OSX
SuperCollider	Free	Dynamic programming	Linux, Mac OSX, Windows, FreeBSD
Reaktor	$399	Conventional programming	Windows, Mac OSX
Byond	Free	C#	Windows

3.2.2 IR capturing

In the table 5 some of the alternative impulse response capturing software are included. IR Utility is included in the Logic Pro software package, which is licensed on the author's personal computer. Room EQ Wizard is a great tool for acoustical analysis of spaces. It includes sine sweep method for capturing impulse responses but does not involve a deconvolution function. FuzzMeasure Pro 2 is free to use but it does not involve a deconvolve function in this version rather it is included in Pro 3 version where it is priced for $90 for educational use. Therefore most applicable software is IR Utility. The advantage of sine sweep method is justified later on in this chapter.

Table 5 Possible impulse response capturing software

Software	Platform	Method(s)	Cost	Deconvolve?	File type
IR Utility	Mac OSX	Sine Sweep	Included in Logic Pro	Yes	.aiff, .wav
Room EQ Wizard	Mac OSX, Windows	Sine Sweep	Free	No	-
FuzzMeasure	Mac OSX	Sine Sweep	Pro 2 Version: Free Pro 3 Version Educational Use: $90	Not in Pro 2 version	.aiff, .wav

3.2.3 Spectrum Analysis

A range of spectrum analysis software have been analysed in order to detect most appropriate ones. Through analysis Sonic Visualiser has been found to be the most appropriate software offering flexible settings to view the spectrogram of the signals. Another advantage of this software is that, it is free to download and it offers versions for Mac OSX. The ease of use of this software was another advantage saving the time needed to learn how to use.

3.3 Plugin Design

Initial design procedure for the creation of plugin has started with analysing available max objects and concepts towards creating a convolution reverb plugin. As described earlier target plugin will use non-uniform partitioned convolution method in order to achieve near zero latency convolution in frequency domain. Therefore transformation of a signal from time domain to frequency domain is the first goal. In order to achieve this, typical max objects are analysed. Max/MSP's pfft~ object is a tool designed for spectral audio processing that uses Fast Fourier Transform (FFT) to transform time domain signals into frequency domain. The object works as a sub-patch containing fftin~ and ifftout~ objects which determine the number of inputs and outputs of its mother-patch. Using these objects one can process any signal in its frequency domain. The object fft~ outputs a stream of real and imaginary numbers where ifft~ expects these. These values are Cartesian coordinates where vertical axis is the real part and horizontal axis is the imaginary part. In order to work with signals amplitude and phase components there are two ways to compute. Those Cartesian coordinates are transformed into polar coordinates by cartopol~ objects (poltocar~ for polar coordinates into Cartesian coordinates). Another way to compute phase and amplitude is to use complex multiplications, which is a method that takes less CPU power.

Figure 3.4.1 below shows the first prototypes created using this method. Prototype on the right shows a method for using less CPU power. First inlet (fftin~ 1) into the sub-patch loads the input signal and the second inlet (fftin~ 2) loads the impulse response.

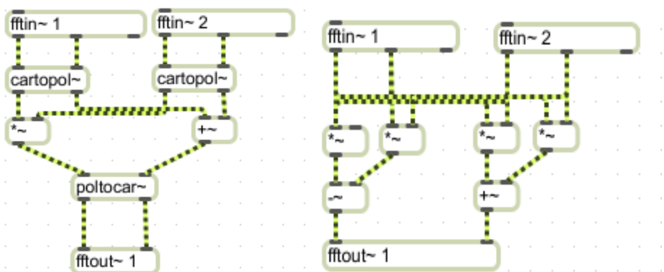

Fig. 3.4.1 First Prototypes for convolution

Using the prototypes shown above one can convolve a signal with another signal however; these methods are not practical for a real time convolution. Instead alternative external objects are needed.

The most efficient externals found through research are part of Alex Harker Max/MSP externals (2011). The two external objects named partconvolve~ and timeconvolve~ copies the impulse response from a buffer and performs partitioned convolution to an incoming signal. Real-time convolution can be achieved when partconvolve~ is used in combination with timeconvolve~ with correct attributes. The attributes needed to state are fftsize, offset, length and chan. These attributes are needed in order to set the partitioning scheme.

Figure 3.4.2 shows convolution sub-patch for left channel that is used inside the final patch named Corridor Reverb. Inlet 1 loads the input signal and inlet 2 loads the impulse response where they are sent to following partconvolve~ and timeconvolve~ objects. The object timeconvolve~ is used for the first partition where the block size is 128 samples thus, convolution in time domain is faster. Series of partconvolve~ objects are attributed in exponentially increasing block sizes as Gardner (1995) suggests. The outlets of these objects are sent to series of signal summation objects in order add overlapping blocks. In the final stage, summed signal is sent to the outlet of the sub-patch into the mother-patch Corridor Reverb for mixing stage.

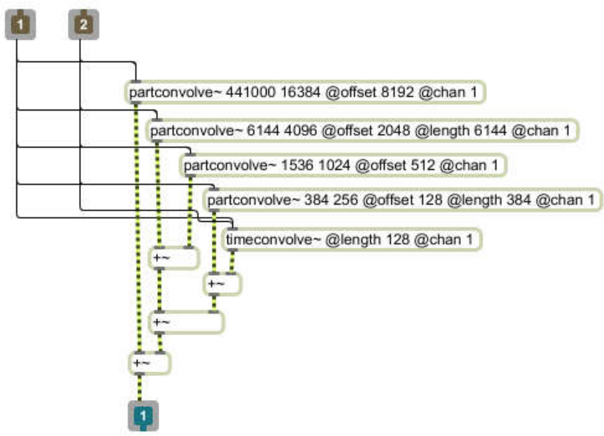

Fig. 3.4.2 Convolution sub-patch for the left channel

The patching mode of final patch Corridor Reverb is shown in the fig. below. The object sfplay~ 2 loads the user-selected input signal buffered by open object into the patch. The object buffer~ contains the impulse response that has been recorded in the corridor leading to semi-anechoic chamber in Millenium Point building of Birmingham City University. Mixing stage of the patch contains a live.dial object for setting the amount of the wet signal to a line~ object where it sends a control signal whose amplitude is set by 'Wetness' to be multiplied by the wet signal and the dry signal.

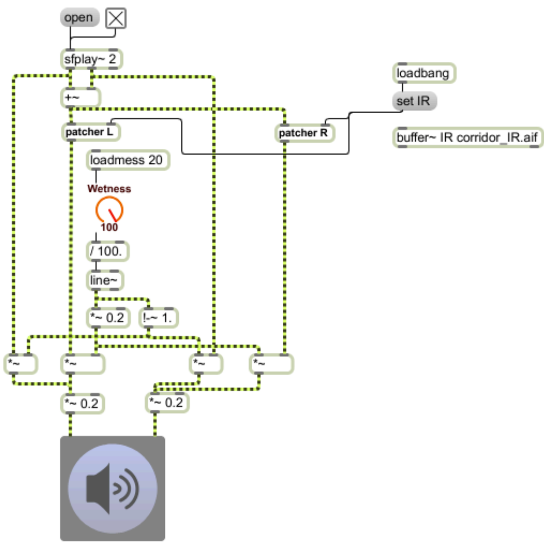

Fig. 3.4.3 Working patch with the mixing stage present

3.4 Impulse response capture methodology

Exponential sine sweep method, as stated in the literature review section, has been accommodated for capturing of the impulse responses due to the method having greatest signal to noise ratio. The most important element of this activity has been ensuring dry audio sample was also recorded with every source-receiver position of the recording of the impulse response. The importance of this element is due to offering smallest variation for test purposes.

Prior to the recordings, the software has been calibrated so that hardware has least effect on the recorded impulse response. Another important precaution was using acoustically absorbent stands for microphone and speaker. This is due to vibrations travelling from loudspeaker stand reach to the microphone stand before the pressure waves reaching the diaphragm wherefore sound waves travelling at a faster rate in denser materials.

As stated earlier IR Utility was employed for the auralisation of the corridor reverb. Tannoy loudspeaker has been used as the source, positioned in one end of the corridor, facing the other end with equal distances from the sidewalls. The microphone, Audix TM1, was positioned in various distances from the loudspeaker pointing upwards.

3.5 Anechoic chamber recordings

Audio test sample was recorded in BCU's semi-anechoic chamber in order to capture a signal that has the least reverberation. Instrument chosen for this recording was a drum kit comprising of a kick drum, a snare and a ride cymbal. The instrument has two advantages in terms of testing. Each element of the drum kit has their spectrum in different spectral bands. The bass drum covers low-end and low mid, snare covers low mid and high mid and ride cymbal covers high mid and high-end spectral bands. This coverage is important in terms of assessing the reverberation for a wide range spectrum. Another advantage is that the instrument is percussive which will ease the perception of reflections for assessing the characteristics of the reverberation. An alternative instrument could be a piano where it is still a percussive instrument that can cover a wide range spectrum. Another advantage of drum kit opposed to a piano is its ease of transportation.

Microphone employed for this recording session was the Audix TM1. This is due to the microphone having least coloration on the recorded sound as stated earlier and also its polar pattern being omni-directional. Position for the microphone was set by trial. The most appropriate position for the microphone was discovered by examination of the spectrum via real-time analyser while drummer was playing the beat. Equal magnitude over the spectrum was the most important principle for this test. After several trials, the most appropriate position was found to be 30 cm above the bass drum in between the snare and the ride cymbal.

The beat composed for the recording was in 4/4 time signatures, playing each element of the kit on separate times to ensure each hit can be perceived distinctly.

3.6 Experimental design

The initial research on the assessment methods of perceptual quality of artificial reverberation in 2.7 showed that most reliable way to assess these qualities is the careful design of listening tests. Another reason for the need for subjective listening tests is that, as Extra et al. (2006) stated, objective measures for perceptual qualities being less effective on reflecting healthy results. Thus, the experimental design of this study shall fulfil careful design of testing criteria allowing participants to assess the qualities of techniques in predetermined objective metrics. These objective metrics should be intuitive to allow the administration to be able to use naïve participants as well as semi-professionals in the field. The importance of selecting the subject pool is another aspect. Although Haines et al. (2007) points out that demographics of the subjects has no expected effect on the results, noting the personal backgrounds of participants is a convenience when analysing the results to show possible correlations. Another important precaution mentioned by Marui (2007) that there were no known auditory health issues of their subjects.

The author believes that the gender of the participants has lesser effect on the results than the age. This is due to the fact that hearing spectrum range of the humans become narrower with respect to age; thus, subject pool of ages in between18 and 26 is selected.

Dry sound samples chosen for these tests are justified above in 3.6, anechoic drum music, should be generated so that a control signal was paired with the convolved signal. In order to generate this control signal, the anechoic drum sample has been played from the Tannoy loudspeaker in the corridor and recorded in the same setup and source-receiver positions for a fair control signal. This signal has been named as

Chamber Reverb and given the symbol ‘B’. The test signal has been given the name Convolution Reverb and given the symbol ‘A’ hereafter. Both signals have been equalised for 0 dB peak amplitudes in order to minimise the effects of processing.

Test methodology chosen for the test is the categorical pair-wise comparison test. Three preference categories were identified similar to the strategy followed by Haines et al. (2007). These categories were: realism, defined as the likeliness to actual reverberations that take place in a real space, quality and personal preference, defined as ‘which technique do you would prefer to use for reverberating a signal’. Personal preference category allowed participants to state the reason for their preference optionally. These statements may help the administration to identify the perceptual differences in an informal way.

A questionnaire has been prepared for the listening test with naïve subjects in mind. This was realised by stating the definitions of reverberant sound field and reverberation from Audio Engineering Society’s (AES) Acoustical Terms Dictionary. Tabulation was designed for intuitive recording of the responses. Also each category was defined prior to the test on the questionnaire to ensure each subject understood the parameters. Instructions were written on the questionnaire to minimise the administration needs and to ensure each subject received the same information ensuring a fair test for all subjects. The questionnaire can be seen in the appendix section.

The most appropriate environment for a test of this nature is the semi-anechoic chamber in Millennium Point. However access to this facility is restricted to the staff where students are expected to access with the studio manager. Another issue was the TEE’s procedure for non-students entering the faculty. Non-students are expected to be informed to security staff three days prior to the entry for recording purposes. These limitations made the use of semi-anechoic chamber for listening tests impractical. The solution found was using a portable high quality sound interface card with flat frequency response headphones for the realisation of listening tests. The solution eliminated the need for an anechoic environment due to subjects listening the material on headphones.

In order to avoid bias, listening tests have been carried out in single blind fashion that is subjects were informed that they were comparing different reverberation techniques without knowing the nature of techniques. Subjects were told to read the questionnaire first, then to set the volume to a comfortable level that would stay the same through the test. In order to avoid perspective loss, subjects were told to listen to the samples as minimum times as they would need to.

4.0 Results

4.1 Spectral tests' results

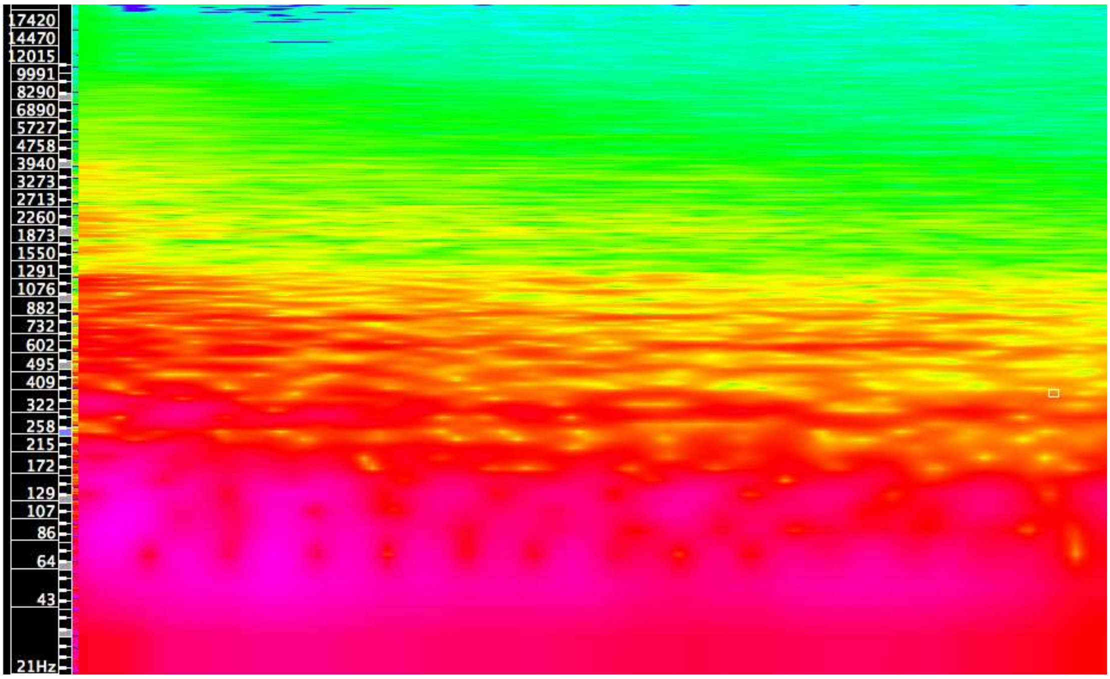

Fig. 4.1.1 Spectrogram view of the bass drum (chamber recording)

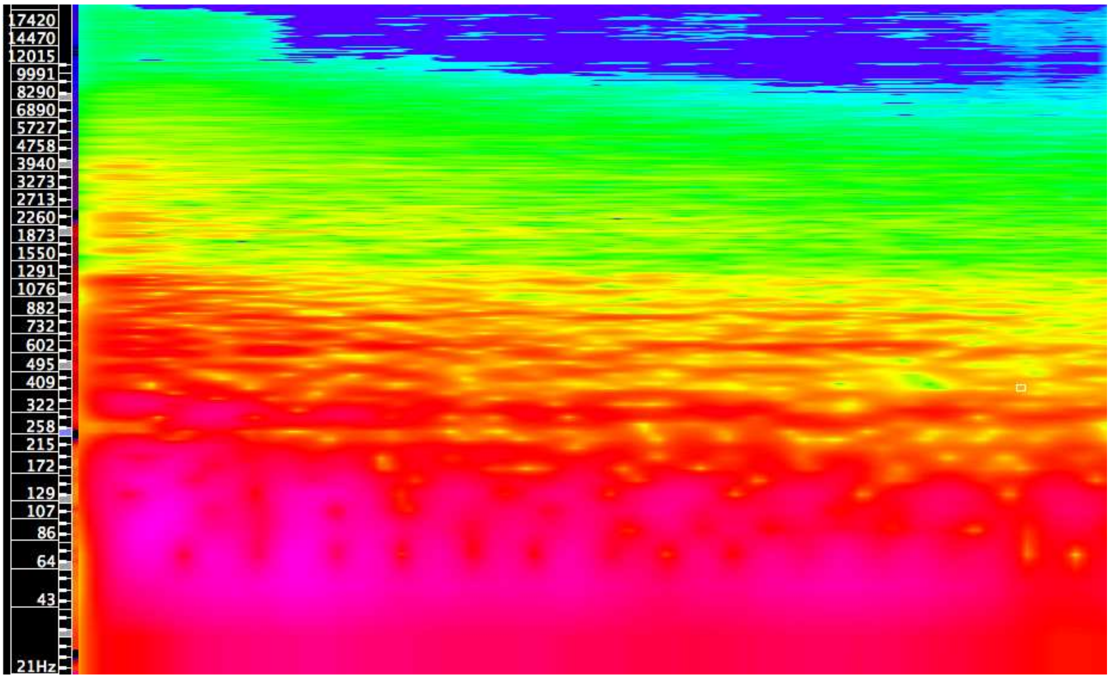

Fig. 2.1.2 Spectrogram view of the bass drum (convolved signal)

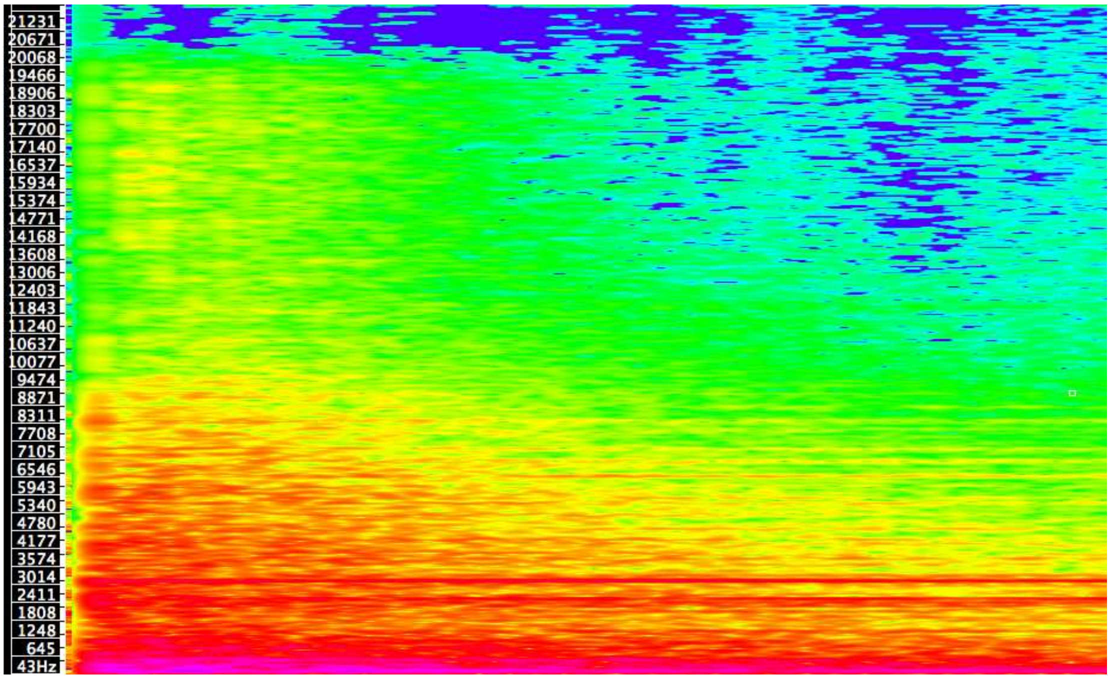

Fig. 4.1.3 Spectrogram view of the snare (chamber recording)

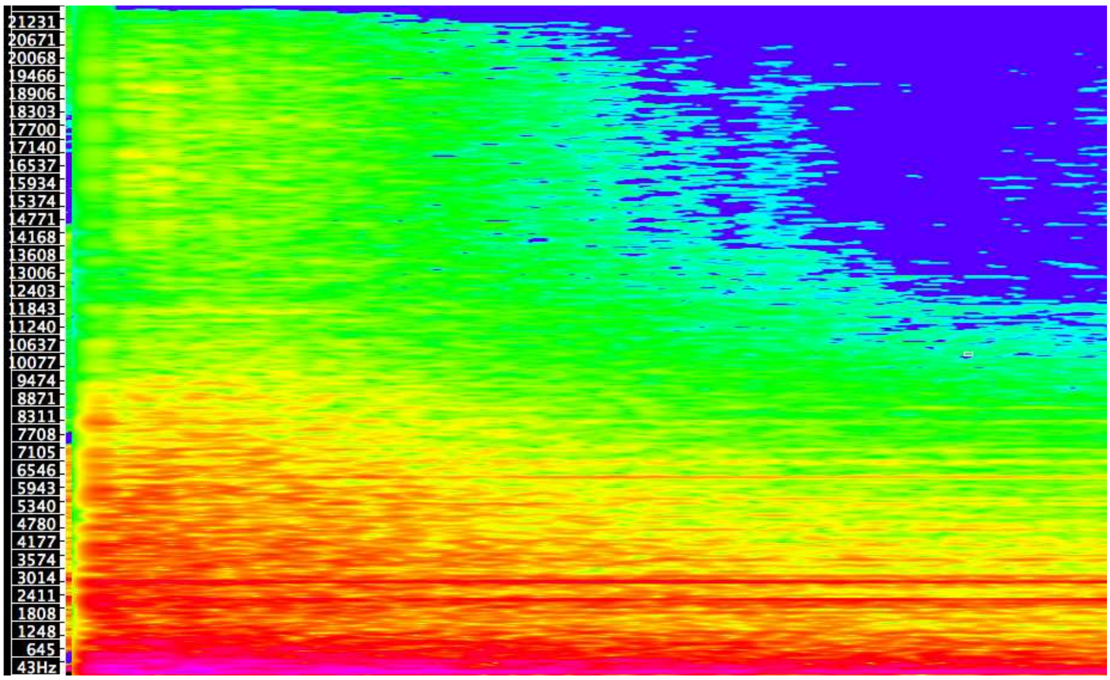

Fig. 4.1.4 Spectrogram view of the snare (convolved signal)

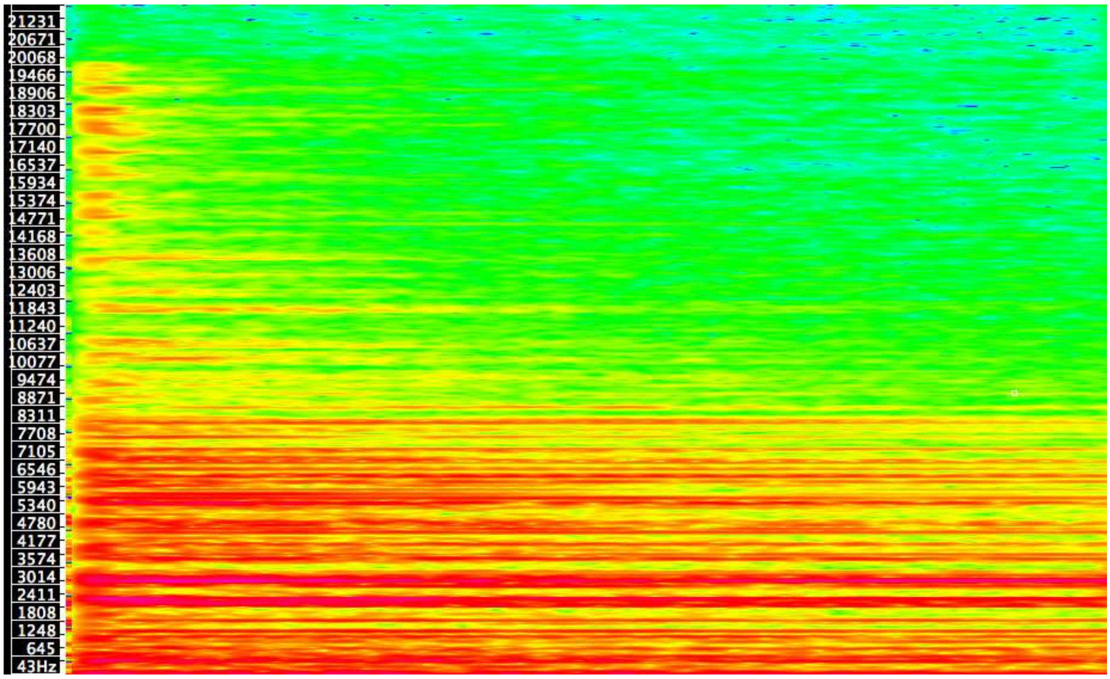

Fig. 4.1.5 Spectrogram view of the ride cymbal (chamber recording)

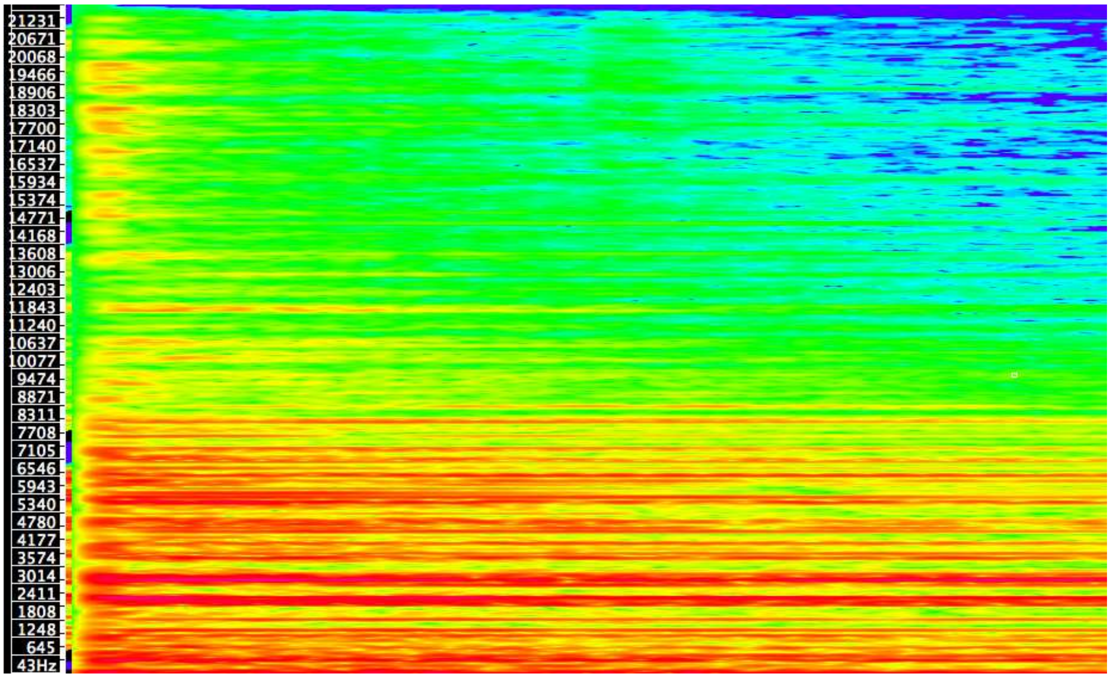

Fig. 4.1.6 Spectrogram view of the ride cymbal (convolved signal)

4.2 Listening tests

This section presents results from subjective listening tests. There are 23 subjects participated in the tests. As stated earlier 'A' stands for the convolution reverb and 'B' stands for chamber reverb (chamber reverb). Also in the table *** convolution reverb is referred sample 1 and chamber reverb is referred to as sample 2.

Table 6 Individual listening test responses with ID numbering

Subject ID	Realism	Quality	Preference
1	A	A	A
2	B	B	B
3	A	A	A
4	B	B	B
5	B	B	B
6	B	B	B
7	A	B	B
8	A	B	B
9	B	B	B
10	B	A	B
11	A	B	B
12	B	B	B
13	A	A	A
14	B	B	A
15	B	B	B
16	A	B	B
17	A	A	A
18	A	B	B
19	B	B	B
20	B	B	A
21	A	A	A

22	A	A	A
23	A	A	A

The table above shows the individual choices of participants for each category. Subjects are given numbers ID numbers.

Table 7 Participants' comments on their personal preference choices

Subject ID	Preference
1	Sample 1 sounds more nonsynthetic
2	Sample 2 fits the personal preference for drum reverb
3	Sample 1, better mixture of separate elements
4	Sample 2, no big difference.
5	Sample 2, no big difference
6	Sample 2,No big difference
7	Sample 2 sounds brighter
8	Sample 2 sounds cleaner, more distinguishable
9	Sample 2, no big difference
10	Sample 2 feels more roomy
11	Sample 2, more distinct cymbal sound
12	Sample 2, sounds brighter
13	Sample 1, sounds more powerful
14	Sample 1, better distinguishable sound
15	Sample 2, sounds more authentic
16	Sample 2, sounds more lively in the low-end and high-end is clearer
17	Sample 1, sounds warmer
18	Sample 2, sounds brighter
19	Sample 2, sounds brighter
20	Sample 1, better sounding
21	Sample 1, sounds more synchronizing
22	Sample 1 (no reasoning)
23	Sample 1 (no reasoning)

As stated earlier in the 3.7 subjects are given the option to state the reasons of their personal preferences. All the participants responded to this optional request except two subjects. They are labelled as '(no reasoning)'. ID numbers in this table are listed in order with respect to the previous table.

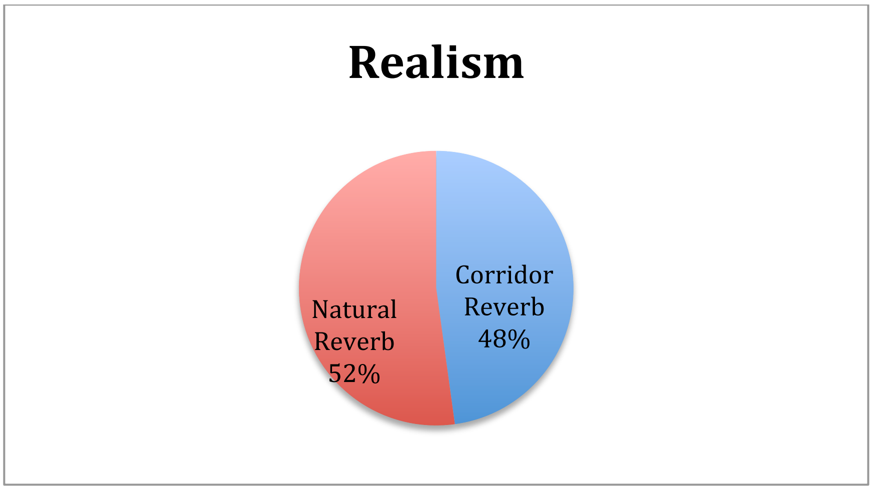

Fig. 4.2.1 Realism category responses shown as frequency on pie chart

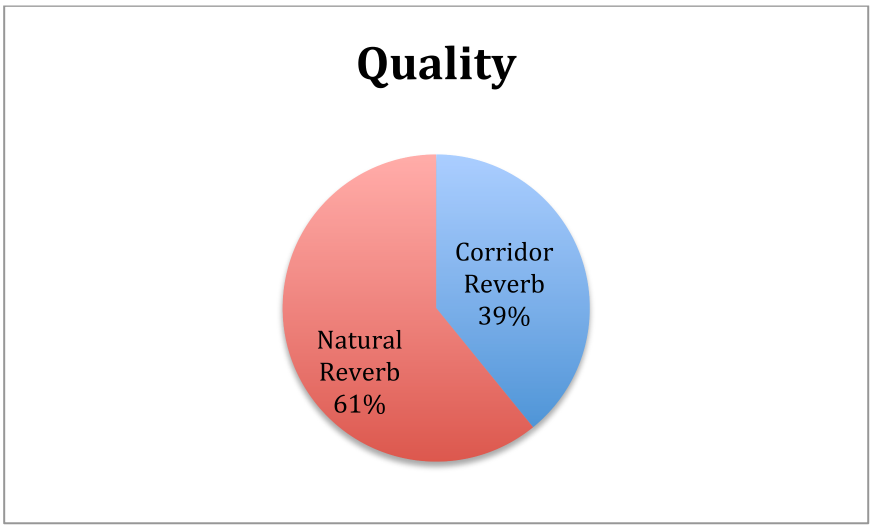

Fig. 4.2.2 Quality category responses shown as frequency on pie chart

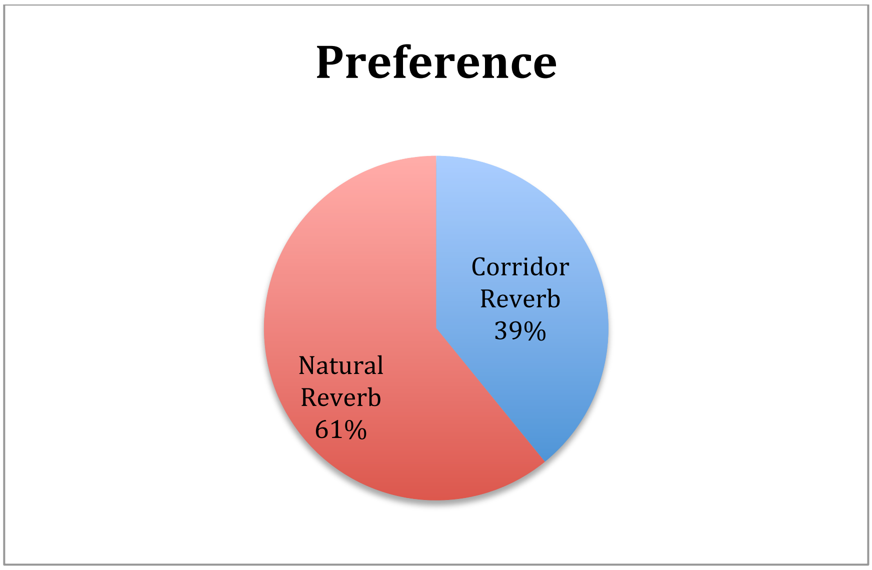

Fig. 4.2.3 Personal preference category shown as frequency on pie chart

5.0 Discussions and analysis

The main aim of this project was to build a convolution reverb plugin and to investigate the quality of convolution reverb by spectrum analysis and listening tests. This was done by collection of an impulse response from a chamber to convolve with a musical excerpt. For the test purposes same anechoic musical excerpt was recorded in the chamber by using the same equipment and setup. For objective measure purposes both signals' spectrograms were plotted using the software Sonic Visualiser with same settings for each pair to compare the responses fairly. For subjective assessment purposes carefully designed and administered listening tests were realised. The results of spectrogram analysis and listening tests were shown in 4.1 and 4.2 respectively. These results are analysed and discussed in the following sections.

As reviewed in the literature review section, none of the earlier studies concentrated on the perceived quality of convolution reverb compared to chamber reverb, the experiments were rather focusing on digital artificial reverberation techniques. Thus expectations of the results of the experiments were not conclusive. This also resulted in a deficiency to relate the results obtained to a certain reason. However the same reason, lack of similar studies, was a great motivation for the completion of this project.

5.1 Analysis of spectrogram results

Spectrogram views are healthy representations of the magnitude, shown over a time vs. frequency plot. These views of the frequency domain of a signal are promising representations to analyse the differences of the responses of systems for the input signal. The results of these tests can be used to identify the perceptual differences that effected participants' preferences.

Results from the chamber recording show denser magnitude dispersion in the low frequency region in the bass drum tests. It also shows higher magnitude response in the same region. Another interesting result is that energy decay in the chamber recording is rather smooth in the region below 80 Hz. This decay forms earlier for the higher frequency bands where above the 4kHz range, the magnitude response becomes rather scarce. Interestingly in the mid range, 1.5 kHz – 4 kHz, both the magnitude response and the energy decay is more present in the convolved signal.

In the snare spectrograms, the dense dispersion of the magnitude in the 3 kHz – 9 kHz range is intense in the chamber recording. Energy decay differences can be clearly seen between the two spectrograms. In the 14 kHz - 19 kHz region, there is obvious magnitude differences. The decay of energy in the same region is similar until a point where, in the convolved signal, there is a quite rate of decay. However this decay is not present in the range below 10 kHz.

Cymbal recording spectrograms show a obvious difference of the dispersion of energy. The dense energy lines, which are the harmonics of the fundamental frequency of the ride cymbal, are clearer for the convolved signal. This shows that magnitude dispersion is weaker as opposed to the chamber recording. There is also an obvious difference of magnitude response for the frequency bands over 8 kHz. In this region, energy decay is rather at a faster rate in the spectrogram of the convolved signal.

Observations over the spectrogram views show poor energy decays of the convolution reverb plugin, especially for the higher frequency bands. Although early energy decays for the low frequencies match the opposing natural reverb characteristics, response to the high mid frequencies in this time region is rather weak.

5.2 Analysis of subjective test results

Prior to analysis of the listening test results, it should be mentioned that due to small population of participants, statistical significance tests were not carried out. Also due to tests being performed only once for each person, there is no need for a confidence interval or binomial distribution computation.

Hypothesis for comparison of realism of two reverberation techniques was that convolution reverb could emulate true characteristics of the reverberation chamber. This was due to the fact that an impulse response contains realistic reflection of a linear time-invariant system's response to any input signal. Subjects preferring convolution reverb for this category form 48% of the total population with the size of 11 out of 23 total responses. Results from the subjective listening tests for this category show that hypothesis is proved, which means subjects were unable to distinguish the quality of realism from either sample.

Quality category asked the participants to state their preference of the reverberation techniques proposed. A frequency of 61% of the participants stated that sample B (chamber reverb) has higher quality than sample A (corridor reverb). A reason for this result can be the plugin's poor response for the high mid frequency bands. Another reason for this result can be the plugin's faster energy decay for the high frequency bands, which may be perceived as lower quality in terms of reflecting higher frequency bands of reflections.

Personal preference category asked the participants to state their personal choices of reverberation techniques. This resulted in 61% of the population preferring chamber reverb. An interesting fact is that, 56.5% from 61% total preferences for chamber reverb also preferred the same technique in the quality category. This shows an obvious relation of quality to personal preference on the choices. The relation shows that perceived quality of reverberation has a great influence on personal preference. Alternative or rather main reasoning for this relation can be drawn from faster decay rates of high frequency of convolution reverb plugin.

Optional request for reasons of the participants' personal preferences has been yielded a range of answers that helped to identify the nature of their personal preference. Responses to the question by individuals with ID numbers: 7, 11, 12, 16,18 and19, shown that 35.7% of the personal preferences were chamber reverb because it sounded brighter. This obviously shows that results of spectrogram tests, plugins' weak response to high frequency and faster decay rate at high mid frequencies are perceivable. Another outcome of these responses was that 17.4% of the participants stated that they perceived very little or no differences.

6.0 Conclusions

In conclusion, the project aimed to cover the design of a convolution reverb plugin on MaxMSP and to evaluate the quality of convolution reverb with respect to spectrogram analysis and analysis of listening tests that has been carried out. The plugin was designed using non-uniform partitioned convolution method for real-time purposes. The designed plugin has been successful, convolving the input signal with an impulse response recorded from a chamber in Millennium Point campus of Birmingham City University with near zero latency. In order to fulfil the second part of the aim of this report, an anechoic drum sample has been recorded in the semi-anechoic chamber. This signal was then recorded in the chamber using the same equipment and setup, in order to create a testing material for the comparison of the different methods for testing the quality of the convolution reverb purposes.

In order to achieve the first objective, extensive research has been carried out in the areas of digital signal processing, artificial reverberation techniques, real-time convolution methods and methods for testing the quality of artificial reverberation. The second objective was to design a fast convolution plugin on MaxMSP, which has been achieved by implementing the multiple frequency delay line non uniform partitioned convolution method that has been researched. In order fulfil the next objective; exponential swept sine wave method has been used to capture the impulse response of the chamber. After this stage recorded impulse response has been applied to the plugin with the anechoic signal recorded in semi-anechoic chamber to test the plugin. After the succession of this objective extensive research has been carried out in the subject of perceptual quality evaluation of artificial reverberation techniques by identifying the earlier studies and observing the methods tested. After this stage, testing methodology has been created for listening tests. Listening tests were carried out with over 23 participants and results of these tests are analysed. In order to measure the quality objectively, spectrogram views of the signals have been visualised.

Results of the objective tests show that the convolution reverberation had shortcomings in the higher frequency spectrum range that was, faster decay of the energy than the chamber reverb technique which was sometimes referred to as original reverb. The reason for this result has not been identified clearly, but the possible justifications were made towards the impulse response capturing method. However testing of the alternative impulse response capturing methods was not the scope of this project, further studies should resonate in these tests.

Results of the subjective tests showed a clear relation to the shortcoming of the convolution reverb discussed earlier. That is, 35.7% of the participants who preferred chamber reverb technique justified that this technique was brighter sounding than the other technique. Interestingly realism category has verified null hypothesis, that they were unable to define which technique has a higher realism. This result showed that the convolution reverb could realistically emulate a space, which was the hypothesis at the initial stages of the project.

Overall, the aim of this project has been achieved with activities planned for the achievement of the demanding objectives. Recommendations for further studies are presented in 7.0.

7.0 Recommendations

This project supplies a firm starting point for further investigations, as the specified scope was rather limiting. Recommendations for further research are:

- Investigations into alternative impulse response capturing methods are needed to provide further auralisation approaches in order to test the quality of convolution with the new methods.
- Further investigations into perceptual audio quality testing in order to design more reliable subjective testing criteria for realisation of the qualities associated with perception of the reverberation.
- Identifying and using better hardware for more realistic auralisation approaches.
- Additionally the scope can be extended, covering the recommendations stated above to re-design and repeat the activities for identification of the problems raised in this project

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Appendix: A Listening Test Questionnaire

Comparison of Different Reverberation Techniques

This listening test is designed to compare two different techniques of reverberating an audio signal. Comparison categories are defined below. There are two different samples in the audio file provided named respectively Sample No: 1 and Sample No: 2.

Reverberant sound field: The sound in an enclosed or partially enclosed space that has been reflected repeatedly or continuously from the boundaries.

Reverberation: The persistence of sound in an enclosed or partially enclosed space after the source of sound has stopped; by extension, in some contexts, the sound that so persists.

*Definitions are taken from Audio Engineering Society' s Acoustical Term's Dictionary.

There are three comparison categories:

- ***Realism***: Which reverberation technique sounds more realistic?
- ***Quality***: Which reverberation technique has a higher quality?
- ***Personal Preference***: Which reverberation technique do you prefer?

Instructions:

I. Listen to the audio file provided**.**
II. Compare the samples with respect to category definitions and state your answer by simply adding 'X' into the relevant cell.

Comparison Categories	Sample No: 1	Sample: No: 2

III. (Optional) State why do you prefer your selection in the personal preference category.

Realism			
Quality			
Personal Preference	A: B:	I prefer Sample No: 1 I prefer Sample No: 2	

Appendix: B Digital Files

Digital files are uploaded to the Moodle, as a single .zip file