ISTANBUL TECHNICAL UNIVERSITY ★ GRADUATE SCHOOL OF ARTS AND SOCIAL SCIENCES

CONVOLUTION
AN APPROACH FOR PRE-AURALIZATION OF A PERFORMANCE SPACE

M.A. THESIS

Engin Gurur GELEN

Department of Music

Music Programme

SEPTEMBER 2019

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İSTANBUL TEKNİK ÜNİVERSİTESİ ★ SOSYAL BİLİMLERİ ENSTİTÜSÜ

KONVOLUSYON
PERFORMANS ALANININ ÖN DİNLEMESİ İÇİN YAKLAŞIM

YÜKSEK LİSANS TEZİ

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Müzik Anabilim Dalı

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Engin Gurur GELEN, a M.A. student of ITU Graduate School of Arts and Social Sciences student ID 409151105, successfully defended the thesis entitled "CONVOLUTION AN APPROACH FOR PRE-AURALIZATION OF A PERFORMANCE SPACE", which he prepared after fulfilling the requirements specified in the associated legislations, before the jury whose signatures are below.

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Date of Submission : **26 August 2019**
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To my family,

FOREWORD

In 2016, a multimedia piece I created for an exhibition translated so different in the room its played. This example among many other individual experiences I had over the years made me question the acoustical responses of an audio spectrum in depth. As being an audio engineer, as well as a music producer, I came across some problematic points over and over again in different settings and environments. Even though the mix is perfectly balanced why it sounds different in different rooms. By trying to find answer and a possible solution suitable for a better auditory experience I realized the subject has gaps that can be studied on.

In this thesis the problematic encounters I mentioned are explained and some possible solutions that might be followed to improve the overall experience of all is suggested. I believe this simple method could make an impact for the artists to translate their music better in the performance venues.

I would like to thank my advisor Dr. Gökhan Deneç for his patience and friendship on this road. Also I would like to thank MIAM Sound Engineering and Sonic Arts students and research assistants for their support and to make it possible to collect the impulse responses and participate in the tests.

September 2019 Engin Gurur GELEN

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ABBREVIATIONS

Hz	**:** Hertz
kHz	**:** Kilohertz
dB	**:** Decibel
SPL	**:** Sound Pressure Level
IR	**:** Impulse Response
DAW	**:** Digital Audio Workstation
SNR	**:** Signal To Noise Ratio
FFT	**:** Fast Fourier Transform
MLS	**:** Maximum-length Sequences
RT	**:** Reverberation Time

SYMBOLS

c	**:** Speed of sound
m/s	**:** Meters per Second
λ	**:** Wavelength
f	**:** Frequency
$\mathbf{t}$	**:** Time

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CONVOLUTION
AN APPROACH FOR PRE-AURALIZATION OF A PERFORMANCE SPACE

SUMMARY

Sound is a physical phenomena. An acoustic disturbance creates vibrations which travel through a medium such as gas, liquid or solid. Due to acoustic differences in rooms, the music made in one room sounds different in another room. This change is mainly because of the acoustical response of the room and the speaker set, mixer that is used. The overall spectrum balance changes. By reflections, absorptions and diffusions sounds that are travelling in the room interfere with each other constructively and destructively in complex ways. Therefore, the overall spectrum balance changes and reverberation occurs. This thesis investigates an optimal solution to minimize the spectral difference due to room effects and playback system effects for the artists.By using a free software called REW Room Acoustics, a frequency sweep of 20Hz - 20kHz played in the room 2 times 21.8 seconds in total to calculate the response of the room. Mean response for those sweeps calculated. A frequency response graph and waterfall diagram is generated as a result of the frequency sweeps. An impulse response audio file is generated to later use in a survey made with 7 artists and sound engineers. The participants are asked to make changes to their mix to translate better to the convolved version by using a 100% wet convolution reverb plugin with the gathered impulse response. Original material and the mix made through convolution are played back randomly in the concert room to the creators. The participants are asked to decide which version is preferred in a blind test manner. Every change they make on their mix, the preference and commentary on why they decided the way they do is noted.This thesis investigates if having an impulse response of a performance space beforehand could make a difference on the mixing stage and the feasibility of the changes made by using impulse response of the performance space. A blind test is conducted to answer the posed questions. Being able to monitor the sound of a musical piece beforehand like it's played in the performance space is a simple yet powerful concept.

KONVOLUSYON
PERFORMANS ALANININ ÖN DİNLEMESİ İÇİN YAKLAŞIM

ÖZET

Ses fiziksel bir olgudur. Akustik bir etkileşim gaz, sıvı veya katı gibi bir ortamdan geçen titreşimler yaratır. Odalardaki akustik farklılıklar nedeniyle, bir odada yapılan müzik diğer odada farklı duyulur. Bu farklılık, esas olarak odanın fiziksel özellikleri, akustik yanıtı ve kullanılan hoparlör, mikser gibi cihazların frekans yanıtlarından kaynaklanmaktadır. Bütün bunlar genel spektrum dengesini değiştirir. Yansıyan, soğurulan ve yayılan sesler karmaşık şekillerde yapıcı ve yıkıcı olarak birbirine karışır. Bu nedenle, genel spektrum dengesinde değişiklikler oluşur ve yankılanma meydana gelir. Bu tez, oda efektleri ve oynatma sistemi efektleri nedeniyle oluşan spektral farkı sanatçı açısından en aza indirmek için uygun bir çözüm araştırmaktadır. REW Oda Akustiği adlı ücretsiz bir yazılım kullanarak, odanın akustik tepkisini hesaplamak için 2 kez 20Hz - 20kHz aralğında olmak üzere toplamda 21.8 saniye süren bir frekans taraması yapıldı. Bu taramalar sonucunda oda özelinde ortalama frekans cevabı hesaplandı. Frekans taramalarının bir sonucu olarak bir frekans yanıt grafiği ve şelale diyagramı oluşturuldu. 7 sanatçı ve ses mühendisiyle yapılan bir ankette daha sonra kullanılmak üzere etki yanıtı içeren bir ses dosyası oluşturuldu. Katılımcılardan, etki yanıtını kullanılarak müzikleri üzerinde değişiklik yapmaları istendi. Müziğin orjinal versiyonu ve konvolüsyonlu dinleme yoluyla yapılan diğer versiyon konser odasında yaratıcılara rastgele çalınır. Katılımcılardan hangi versiyonu tercih ettikleri bilgisi toplanır. Müziklerinde yaptıkları her değişikliğin, versiyon tercihlerinin ve tercihlerini neden o yönde yaptıklarına dair yorumlar toplanır. Bu tez, performans öncesinde bir performans alanının etki yanıtına sahip olmanın miks aşamasında herhangi bir fark yaratabileceğini ve performans alanının etki yanıtını kullanılarak mikste yapılan değişikliklerin fizibilitesini araştırıyor. Bu sorulara cevap bulmak adına kapalı bir test yapıldı. Bir müzik parçasının performans mekanında çalınıyormuş gibi önceden dinleyebilmek ve buna göre önceden karar alabilmek basit ama güçlü bir konsepttir.

1. INTRODUCTION

Different rooms with different playback systems create different frequency responses. The room dimensions, material of the room, objects in the room and the playback system are all affect the sound spectrum. It's possible to get clean results in the acoustically well treated mixing room with the equipment that we are used to. How about playing the music in a place that is reverberant and not treated acoustically. Those poorly treated acoustic spaces are part of our lives considering exhibition halls, many art spaces and even music clubs. Some of them are excessively reverberant some of them have unbalanced frequency responses, some of them have both issues.

In this thesis, I am going to explain some problematic encounters I had, give some examples and try to suggest some possible solutions that might be followed to improve the overall experience. I believe this study will be beneficial for the sound artists to get more insight beforehand about the performance space. In their work environment, the artists should be able to monitor how their piece will translate in the performance space. By using this information, they might use these acoustic unbalances artistically or technically for their own good. But first, the right questions needs to be directed. The change in space creates a big difference on the overall experience of an audio. To put it better, It can be considered as an equalizer applied to the audio itself along with a spectral reverb which the decay time differentiate throughout the audio spectrum. By keeping that in mind, can we suggest a solution to minimize the spectral difference? Could it be possible to monitor music as its played in the space that will be performed in? By pre-monitoring, can we optimize the piece to translate better in the peformance space beforehand? In order to understand the points I'm mentioning here, some of the concepts on room acoustics and human auditory system should be understood well.

1.1 Physics of Sound

First of all, it is crucial to understand the basics of sound physics. Sound is an acoustic disturbance transmitted through a medium such as gas liquid or solid and propagates

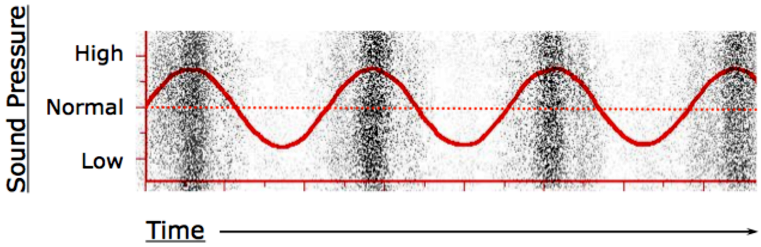

Figure 1.1 : Sound wave over time.

away from its source. The sound source causes a vibration and disturbs the particles in the surrounding medium and the energy becomes less intense when it gets further (Rossing, 2007). The sound waves travel by the interchange between kinetic and potential energy (compression and rarefaction) of the air molecules. This acoustic disturbance creates a longitudinal alternating pressure in air. Even though the sound waves are quite complex they can be simplified to pressure and time. Fig.1.1

While acoustic wave travels through a medium, it causes the pressure to vary periodically. The word "period" has a significance. Period is the amount of time it takes to complete one cycle of a wave. The standard unit for the wave period is seconds. The real world sound waves are much more complex than a sine wave although any periodic impulse can be represented mathematically by its components as sine/cosine waves by a process known as Fourier decomposition. To simplify the concepts, sound waves are often simplified as sine waves. The sine waves are characterized by its properties as frequency,f also known as pitch, wavelength, λ and amplitude, A. The frequency is the inverse of the period that mentioned earlier and it is defined as the number of oscillations occur per second.

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

By using the speed of sound in the medium its traveling through ($c \cong 340\ m/s$ in dry air at 20°C) we can calculate the wavelength. Eq.1.1 Wavelength carries a big importance when dealing with low frequencies. In the frequency range of 50 Hz to 150 Hz, wavelength sizes are similar to room dimensions which leads to acoustic resonances, discussed in later chapters Tab.1.1.

Table 1.1 : Wavelengths of Frequencies.

Frequency	Hz	50	70	90	110	130	150
Wavelength	*m*	6.8	4.85	3.77	3.09	2.61	2.26

1.2 Human Hearing

The auditory system is a really sensitive and dynamic device. The pressure hits the eardrum and creates a periodic motion. Human ear as a transducer converts the mechanical information into electrical nerve impulses. These impulses then transmitted to the brain and decoded as its period and amplitude. The weakest sounds that we perceive (20μPa) and the sounds that are on the upper discomfort limit (200 Pa) have the ratio of 10^{12} (Rossing, 2007). Compared to lowest and highest frequency ratio of vision, ear is 9 times greater. In general terms the auditory frequency perception ratio of upper and lower limit is 10^3. It immediately and precisely decodes pitch, timbre and the direction of a sound source. The most generalized threshold of human frequency hearing is 20 Hz to 20000 Hz but it always varies from person to person due to age, occupation, environment etc. Hearing 20 kHz might be true in case for a newborn but a fair upper limit for the most of the people is lower than 18kHz. Similarly, for the lower threshold below 20 Hz human cannot hear phenomenon is incorrect. With high enough pressures humans can hear down to 1 Hz (Jürgen Altmann, 2001). Although, the pitch perception is lost below 16-18 Hz (Leventhall, 2003). The frequencies above 20kHz are called ultrasound and the frequencies lower than 20 Hz is called infrasound.

Human hearing response for frequencies is not flat. First research to determine equal loudness contours were conducted by Fletcher-Munson in 1933. By conducting empirical experiments they asked for the listeners to compare a large set of sine tones against a pure tone in 1 kHz. They asked the subjects to match the level for the tone they're investigating to the reference tone in 1 kHz. As a result first equal loudness contours developed. The contours revised 2 times by Robinson and Dadson in 1956 and by collaborative studies of researchers from Japan, Germany, Denmark, UK and USA in 2003 which became the basis of ISO226:2003. Shown in Fig.1.2.

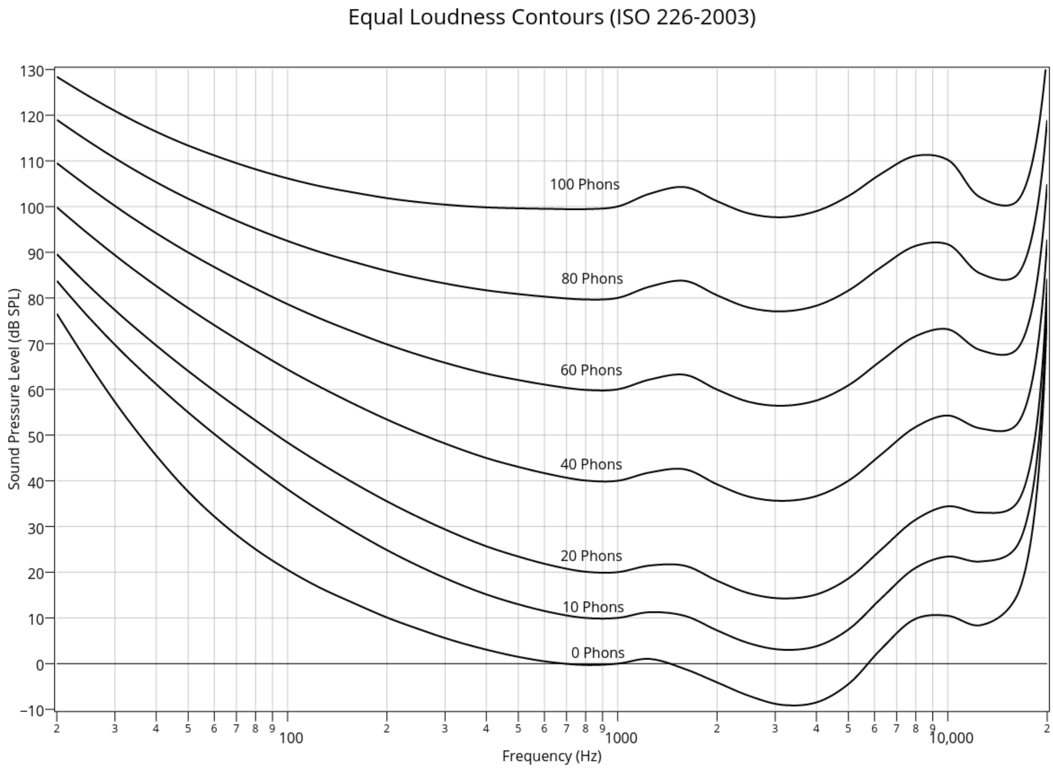

Figure 1.2 : Equal Loudness Contours (ISO 226:2003).

As can be seen from the Fig.1.2 human response is not flat along the frequency spectrum. 60 dB of 1000 Hz tone perceived equally loud as 80 dB of 100 Hz and 110 dB of 20 Hz. Low frequency sounds needs more pressure to perceived as loud as high frequency sounds. 50 dB difference is a big difference in fact, it corresponds to an intensity ratio of 100 000:1. The 2 kHz - 5 kHz band is the one that humans are most sensitive to. It makes total sense since the intelligibility of human speech mostly lies in that frequency range. Also, it can be seen in the Fig.1.2 when the frequency gets lower the perceivable dynamic range compared to 1kHz decreases. For 1 kHz tone the dynamic range is 100 dB whereas the same SPL range for 20 Hz is 50 dB.

1.3 Room Acoustics

In different rooms, the same sound source played by the same equipment sounds different. The sound source takes the character of the room it is played in. Sound waves travels in all directions, gets reflected, diffracted and scattered by the surfaces and objects. All these properties change the overall frequency distribution that reaches the ear.

In room acoustics, wavelength plays a big role. The behavior of sound is room dimension dependent. Schroeder divides the room in two parts as Resonator and Reflector (1954). The resonator part deals with the low frequency sounds, in small rooms this cross over frequency is close to 200 Hz. Below the Schroeder frequency the room acts like a resonator (wave energy) and above this frequency it is acts like a reflector (ray energy). Eq.1.2 shows how to calculate the Schroeder frequency. It can be calculated by dividing the room's reverberation time (T) in seconds by the volume of your room in cubic meters (V') and multiplying the square root of the result by 2000.

$$f_c = 2000\sqrt{\frac{T}{V'}} \tag{1.2}$$

The overall frequency balance (timbre) changes due to reflections in the room. This change in frequency can be constructive or destructive. The effects are more audible in lower frequencies due to room modes. The acoustical sum of a sound source and its reflections creates the overall sound we hear in a specific location in the room. By considering a single frequency which might be a Fourier component of a complex sound, it can be easier to understand the change in frequency domain caused by reflections in a steady-state event. In Fig.1.3 an idealized interference is shown for a single sine tone interfering with delayed version of itself with same amplitude. The result of the interference is doubled amplitude in the constructive interference and perfect cancellation in destructive interference. An abrupt transient sound would interfere in a different way. The event itself could be in decay while the reflected version arrives. In that case the interference would be different. In a real time event, the sound source is much more complex and the delayed sound is weaker than the original sound source. This means that all the frequencies will interfere with each other in many different ways to produce the sound as we hear it in the room.

Materials in a room change how the sound is being reflected and absorbed. Hard flat surfaces reflect the sound more than a randomly textured soft surface. The reverberation is caused by the sound source gets reflected by surfaces and objects in the room. When the sound source stops radiating energy, the reverberant sound field starts to decay. It is easier to hear the reverberation by using a transient sound such as a starter gun or a balloon pop. Reverberation time noted as RT_{60} is the time it takes for reverberant sound field attenuate by a factor of 60 dB. There are only two ways to change the reverberation time. One is changing the room dimensions and other is using

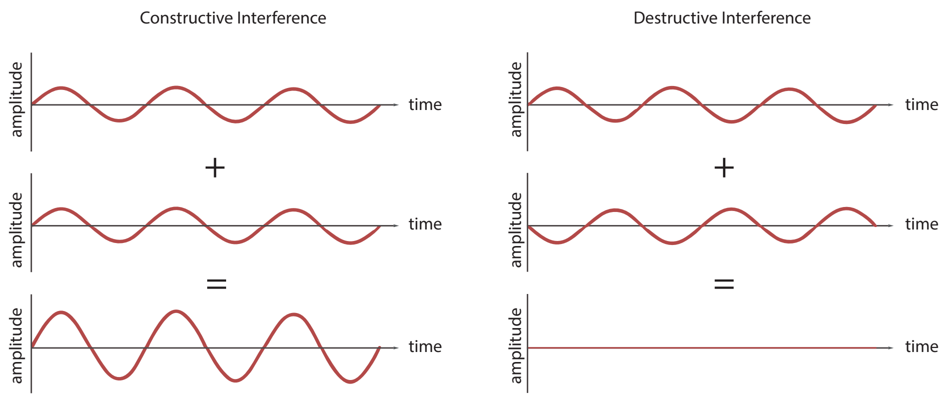

Figure 1.3 : Constructive and destructive interference of a pure tone.

absorbent materials and objects in the room. When there are absorbent materials and diffusers in the room such as acoustic foam or acoustic fabrics, reverberation changes due to interaction with those materials.

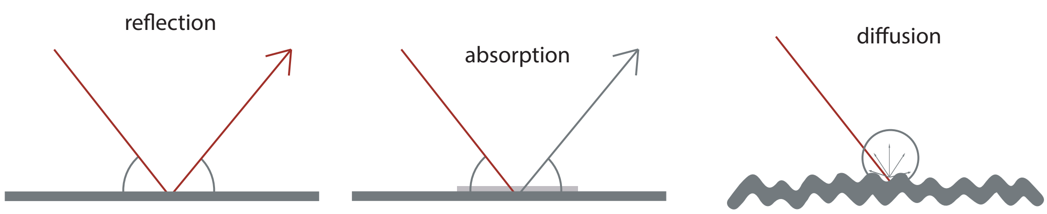

Figure 1.4 : Reflection, absorption and diffusion of a sound.

In a room with variety of objects and materials all these properties takes action in a complex way. In addition to the sound source reflected sounds fill up the space, interfere with each other constructively and destructively in different ways. In acoustical terms this whole event can be conceptualized as geometric acoustics and ray acoustics for the upper region of Schroeder crossover frequency.

Lower region of the Schroeder frequency has a different behavior. For these low frequencies all the rooms of all shapes and sizes are dominated by resonances (room modes, eigenfrequencies) and standing waves (Toole, 2008). The room modes are the resonant frequencies in relation to room dimensions. The standing waves can be easily heard by walking in the room while speakers play a pure first order resonant low frequency tone.

In acoustics, nonrectangular rooms are more complex to calculate. However for the purpose of this thesis we only need to know the room modes in relation to parallel surfaces. In Table.1.2 some room modes in relation to parallel surface length can be seen.

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Table 1.2 : Room modes in relation to room dimension.

Harmonics	Number of Nodes	Number of Antinodes	Length Relationship
1^{st}	1	2	$L = 0.5 * \lambda$
2^{nd}	2	3	$L = 1.0 * \lambda$
3^{rd}	3	4	$L = 1.5 * \lambda$
4^{th}	4	5	$L = 2.0 * \lambda$
5^{th}	5	6	$L = 2.5 * \lambda$
6^{th}	6	7	$L = 3.0 * \lambda$
n^{th}	n	$n + 1$	$L = (n * \lambda)/2$

The Eq.1.3 shows the room modes for small rectangular rooms with dimensions x, y and z where l is directional length, c speed of sound. n_x, n_y and n_z are integers applied to each dimension to identify the mode. For example, $f_{1,0,0}$ is the first order mode for x dimension, $f_{0,2,0}$ is the second order mode for y dimension, $f_{1,0,1}$ is a tangential mode for x and z dimensions and $f_{2,4,3}$ is an oblique mode for x, y and z dimensions involving all three dimensions.

$$f_{n_x n_y n_z} = \frac{c}{2}\sqrt{\left(\frac{n_x}{l_x}\right)^2 + \left(\frac{n_y}{l_y}\right)^2 + \left(\frac{n_z}{l_z}\right)^2} \tag{1.3}$$

Due to energy fall for every reflection, the frequencies that can complete their cycle with fewest reflections are the most powerful. Therefore the axial modes are the most powerful followed by tangential and oblique ones. Axial modes are the usual reason for bass imbalances in small rooms.

1.4 Convolution

Convolution in maths is multiplication of two different functions to form a third new function. The mathematical definition of the convolution of two finite sequences as follows:

$$a[n] * b[n] = output[k] = \sum_{n=0}^{N-1} a[n] \times b[k-n] \tag{1.4}$$

The Eq.1.4 shows an input sequence $a[n]$ convolved with another sequence $b[n]$. This can be considered as sequence b being the music to be convolved where sequence a is sample sequence of the impulse response signal. N is the sequence length in a and k is the sequence length in b. The equation works similar to a nested loop in computer programming . Every individual sample in a is multiplied by every sample in b then gets shifted in time domain. The resultant function has total samples of one less than summed sample sizes of a and b. In Fig.1.5 a diagram shows how convolution equation works.

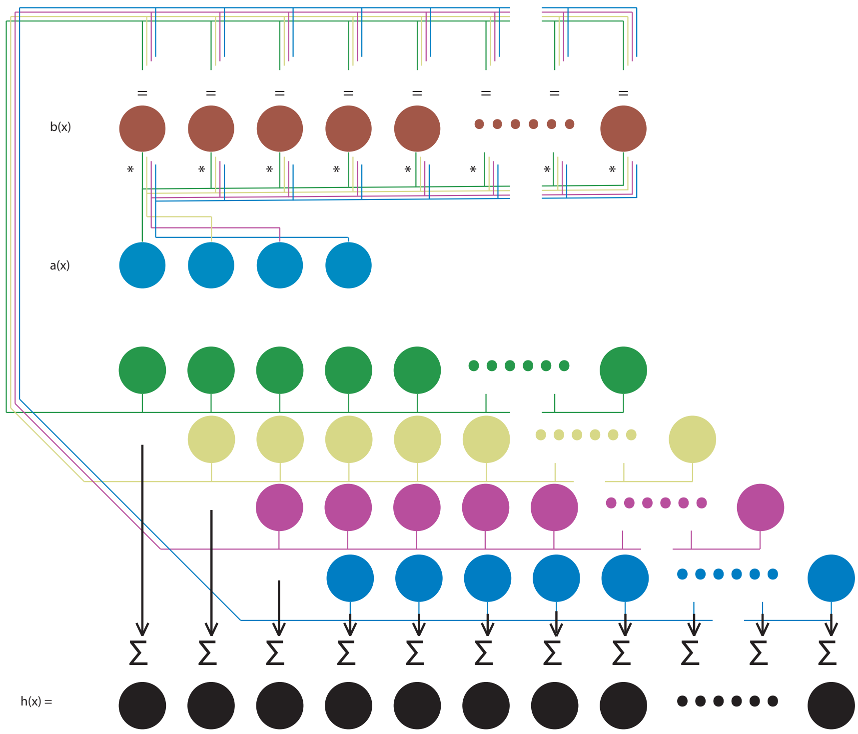

Figure 1.5 : Convolution calculation diagram.

1.5 Convolution Reverb

A convolution reverb combines two signals. First signal is the input signal, the music in our case and the second signal is the impulse response of the space that is replicated. The output signal is the convolution of any input signal and an impulse response.

Impulse response is the reaction of the space when it is excited externally (Kleiner, 2011). Although, theoretically any sound file can be used as an impulse response for convolution in digital domain. Convolving sound files with sources that are not intended to be used as IR may lead to get some interesting results. An impulse response of an acoustic space can be created by various methods. One widely used method is bursting a balloon, using a starter gun or clapping hands to create an instant short broadband noise impulse. A drawback for this method is that there is not enough energy in the impulse to excite broad enough frequencies in the room. Especially, when considering low frequencies, this method does generate enough energy to excite the room. Also, the energy distribution of the frequencies varies throughout. Such as hand clapping generates a peak resonance in 3 kHz - 500 Hz region relative to the rest of the spectrum. Bursting a balloon method has the drawback of repeatability. The balloon must have the same amount of air and they should be at least 40 cm in diameter to have a uniform directionality and to have enough energy on the low frequencies (Pätynen, 2011). As knowing the drawbacks of these methods one might ask why they are used anyway. The beneficial point of these methods is the convenience. It is easy to apply in places where there are no recording and playback equipment to use or in places where it is not possible to use them such as caves or mountains (Välimäki, 2012). Another problem with this abrupt noise methods are the low signal to noise ratio (SNR). The pop needs to be loud enough to capture the room behavior properly afterwards. This means a loud initial hit followed by a quiet room reverberation. Therefore, the microphone needs to be able to handle the dynamic range of extreme loud pop and quiet room reverberations.

Another known technique used to create impulse responses is MLS technique. It is based on exciting the room with periodic pseudo-random signal similar to white noise. The main problem with this method is distortion peaks due to speaker and measurement system. It is possible to reduce these distortion peaks by changing the signal amplitude. When the signal amplitude rises more distortion peaks occur while attenuation of the signal lowers the signal to noise ratio (SNR) (Stan, 2002).

Another way to create high quality impulse response is by using sine sweeps. This method is the standard method to retain high quality impulse responses. By this method much higher SNR can be achieved. The SNR is determined by the amplitude of the sweep itself and the background noise of the recording location. These sine sweeps are constant in amplitude throughout the frequency band, starts from the low frequencies and gradually rises in a logarithmic scale. The sweeps are played through loudspeakers and recorded back by using a microphone in the space with all the acoustic properties, reverberations and spectral characteristics. The resulting recording then convolved with the time-reversed version of the original sweep that is used to excite the room (Farina, 2000). The result is the impulse response of the room all through the given frequency range in given amplitude. To put it easier, a space gets excited by given frequency range and recorded back, the resulting recording of the space is then processed to replicate the behavior of the space and the result can be used on an entirely different signal to monitor how it would sound like if it would be played in that space. The result is the replication of the room behavior at a certain point (the microphone position) in a certain amplitude.

1.6 Hypothesis

This thesis focuses on simulating the effect of a rooms acoustic properties on audio spectrum by suggesting a method. In this thesis the feasibility of using impulse response of a performance room as convolution reverb to create a perspective in mixing stage for the artists without being in the actual room. Considering aims of this thesis, three step methodology is specified.

1. Creating impulse response of a performance space.
2. Efficiency of using convolution to monitor music beforehand like it is played in the venue.
3. Conducting a survey along with a blind test among 7 artists to see the feasibility of the changes made by using impulse response convolution.

Experiment setups have been developed to verify the posed questions. All the equipment are set up as the way it normally would in a performance or listening situation. The microphone is set up in a listening position where the direct sound from each speaker can reach the microphone at the same time. In addition to that, to be able to get the best stereo image, an equilateral triangle placement of microphone between speakers is used. By using the sine sweep method, impulse response of the room is generated. The participants are asked to make changes to their mix to translate better to the convolved version by using a 100% wet convolution reverb plugin with the gathered impulse response. Original material and the mix made through convolution are played back randomly in the concert room to the creators. The participants are asked to decide which version is preferred in a blind test manner. Every change they make on their mix, the preference and commentary on why they decided the way they do is noted.

1.7 Literature Survey

In impulse response measurement field there are many research made regarding measurement techniques and their comparisons (Suzuki 1995; Farina, 2000; Müller 2001; Holters, 2009; Pätynen, 2011; Guidorzi, 2015). Prof. Angelo Farina is one of the leading researchers in the field and written many papers in the impulse response measurement field. Also, he is the pioneer of sine sweep technique (Farina, 2000). Among wide range of subjects in acoustics, he worked on room correction by using inverse filtering convolution to for improving car audio playback systems. Many measurement techniques and measurement processing softwares use his reasearch (Mulcahny 2004; Adriaensen, 2006). Another study taking use of convolution and impulse responses is conducted specifically for binaural applications (Jeub, 2009). The study took place in different kinds of spaces and measurements made by the use of dummy heads to evaluate dereverberation and speech enhancement algorithms.

Convolution has wide range of applications in digital signal processing. Convolution reverbs are used for both artistically in music production and for film and game audio field. While shooting a movie on site impulse responses are gathered for later use in the studio to match the sonic characteristics of the scene location in case of any overdubbing. Some other well known musical applications for convolution is modelling guitar cabinets and modelling pro audio hardware. In his paper, Curtis Roads explains how convolution works and overviews it's musical uses (Roads, 1993).

In room measurements, the recording for measuring the room response is only valid for the exact location of the microphone. This is one of the drawbacks of the room measurement and room correction. Ajdler suggested a moving microphone technique for recording large sets of impulse responses that might solve the issue (Ajdler, 2007).

The musical use of room behavior is also a field many artists worked on. One of the best known works by Alvin Lucier, "I'm sitting in a room" (1969) makes use of room resonances. His recorded narrative played back in the room and re-recorded and played back repeatedly until his voice loses it's speech properties and turns into resonances. This can be considered as convolution. Similar result might be re-produced by convolving an audio recording the result and convolving it again with the same impulse response file repeatedly. Mark Bain in his site specific installation “Live Room” (1998) approached the waves in a physical way. Using acoustic-intensifying equipment mounted directly onto the walls of the building, he induced acoustical vibrations in the building. These vibrations created sounds directly in relation to the building itself. In this work the building became the speaker.

1.8 Methodology

The performance space (Usmanbaş Concert Space in ITU, MIAM) is set up as the way it normally would in a concert situation. Two loudspeakers, a subwoofer and the audio mixer to feed them is set. Due to varying perceptive loudness levels (see Fig.1.2), the system and the room is set up to 90 dB SPL on 1 kHz tone with a distance to create an equilateral triangle ($d = 6m$). Calibration microphone and audio interface are set up as shown in Fig.1.6. To make a measurement a logarithmic sine sweep is sent to the sound sources (loudspeakers and subwoofers). In the logarithmic sweep, it takes same amount of time raising frequency from 100 Hz to 1 kHz and 1 kHz to 10 kHz. The sweep starts from the low frequencies (20 Hz) and gradually rises until hitting upper frequency limit (20 kHz). What makes the sweep logarithmic is the rate at which the frequency changes, it takes a fixed time to double. For example, the time for the sweep to go from 20 Hz to 40 Hz is the same as the time to go from 1 kHz to 2 kHz. The microphone picks up the audio that travels through the air and hits directly to the microphone as well as the sounds that hit the surfaces in the room and bounces back.

The recorded signal is processed by using Fast Fourier Transform (FFT) to create the transfer function. The transfer function is the frequency based comparison of the original source frequency that is played by the speakers versus the one that the

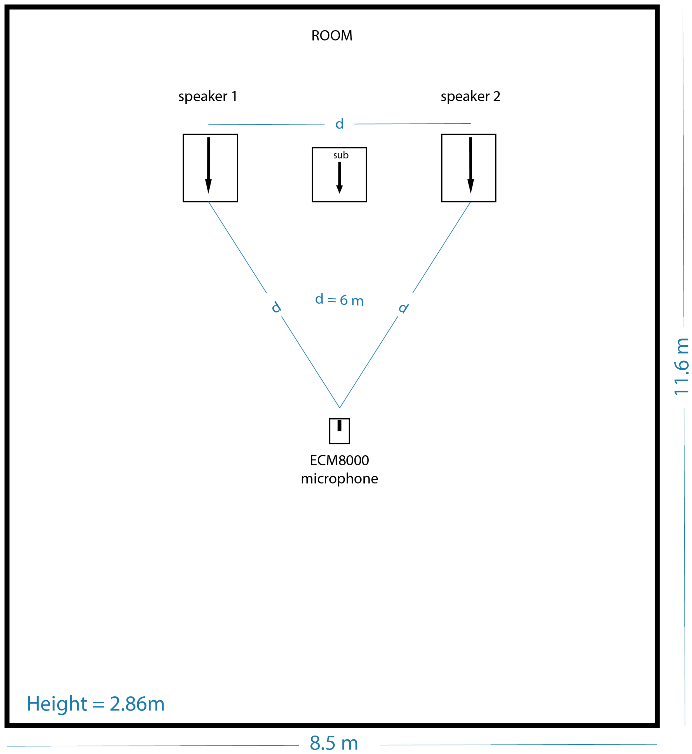

Figure 1.6 : Speaker and microphone setup to play and record the sine sweeps.

microphone picked up. While the frequency response of the listening location is not flat every point in the room, the measurements are valid for the point that is chosen as shown in Fig. 1.6. The listening (recording) position is set from the beginning and remained unchanged throughout the measurement process. After determining the transfer function, inverse FFT is applied to the frequency and phase information by the software to create the time based function called impulse response.

By using the software called REW Room Acoustics, a frequency sweep of 20 Hz - 20 kHz played in the room 2 times to calculate the response of the room. The software averages the sequential measurements to create a better mean result. In user's manual it is suggested to use single long sine sweeps rather than using short multiple sweeps (Mulcahny, 2004). To create a better mean result long sweep averaged 2 times is used. A frequency response graph and waterfall diagram is generated as a result of the frequency sweeps. Filter coefficients are calculated and a wave audio file as an Impulse Response is created. The Impulse Response file is used in a convolution reverb plugin on the master out channel of the DAW. The sound output gets the character of the room and can be monitored as the way in a concert situation.

2. EQUIPMENT DESCRIPTIONS

In this chapter the equipments used in this project is defined. The reasons behind choosing the equipment accordingly is explained.

2.1 Microphone

Measurement microphones are omni-directional. They are produced without a modification on directionality. These microphones are generally used for measurement of Sound Pressure Levels and measurement of sounds coming from any direction in an acoustic space (Mathew, 2019). This allows to pick up and treat equally directional sounds occurs due to reflections. Also, when doing measurements, the frequency response of the microphone becomes crucial to prevent any coloration. The frequency response should be flat in order to get the frequency response of the room. For measurement purposes, small diaphragm microphones are used due to their extended frequency response. Due to their small diaphragm the internal resonance frequencies are higher than human-hearing range (Winer, 2019). Therefore, these microphones have the least color and suitable for this project. The microphone used in this project for impulse response recording is a Behringer ECM8000. As can be seen in Fig.2.1 the frequency response of these microphones are considerably flat (20 Hz – 20 kHz, $\pm$ 1.0 dB), they have small diaphragm, they are omni-directional and suitable for this project (Behringer Spezielle Studiotechnik GmbH, 2013). The SNR level specified is 72 dB.

2.2 Audio Interface

For measurement purposes, Audio Interface has a great importance. The output should be flat in frequency domain without any distortion as well as the input. The microphone preamps should be matched and linking the Left and Right channel should be possible to theoretically get the best balance out of the room. The recording of the impulse response is made in 96kHz, 24-bit resolution. The audio interface that is used for this project is Universal Audio, Apollo Twin. The device has two inputs with linking

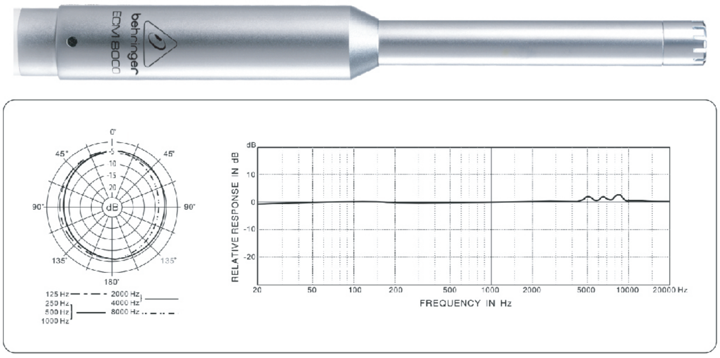

Figure 2.1 : Behringer ECM8000 small diaphragm, omni-directional, condenser microphone, frequency response and polar pattern.

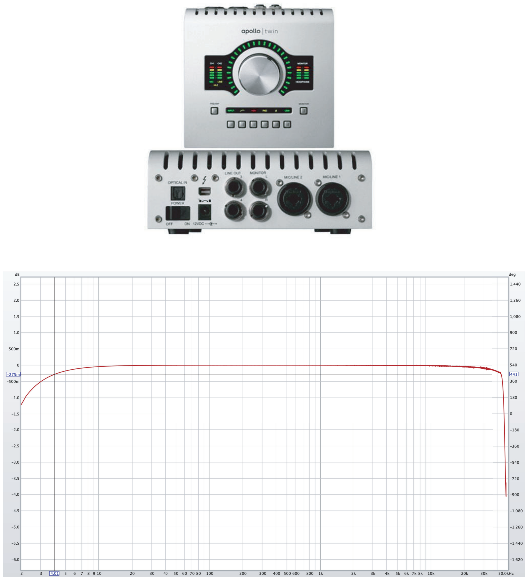

Figure 2.2 : Universal Audio, Apollo Twin Interface, Frequency response.

setting. The loop back test applied to the interface shown in Fig.2.2. As can be seen, it has flat in frequency domain (20 Hz – 20 kHz, ± 0.1 dB) with low signal to noise ratio (118 dB (A–weighting)). This interface is suitable for this project (Universal Audio Inc., 2017).

2.3 SOFTWARE OVERVIEW

REW (Room EQ Wizard) is an application written in java programming language by John Mulcahy first in 2004. The current version used for this thesis is version 5.19 (2018). It is used to measure room responses, modal resonances and reverberation time distribution of frequency spectrum. It includes tools for generating test signals; measuring SPL; measuring frequency and impulse responses; generating phase, group delay, spectral decay plots, waterfalls, spectrograms and energy-time curves; generating real time analyzer (RTA) plots; calculating reverberation times; displaying equalizer responses and automatically adjusting the settings of parametric equalizers to counter the effects of room modes and adjust responses to match a target. (Mulcahy, 2004) REW uses logarithmical sine sweeps for its measurements (Farina, 2000)(Müller, 2001). This is a reliable and accurate method as mentioned in convolution section. The sine sweep method has advantages over MLS method such as having a better signal to noise ratio (SNR) and having a robust non-linearity rejection (Guidorzi, 2015). Also, the needed high dynamic range, in excess of 90 dB, required to allow an analysis of frequency response is unattainable with MLS or noise measurements.

3. ANALYSIS

The placement shown in Fig.1.6 is set. By using REW Room Acoustics software with the settings shown in Fig.3.1 frequency sweeps in range of 20Hz - 20kHz are generated 2 times. The mean values of these measurements are made automatically. The resulting graph in Fig.3.2 is the resulting frequency response of the room room with sweeps calibrated to 90dB SPL in 1kHz. Relevant filter settings shown in Fig.3.3. Resultant filters that might be applied to create a flatter frequency response in the room shown in Fig.3.4 along with the predicted room response after filter application.

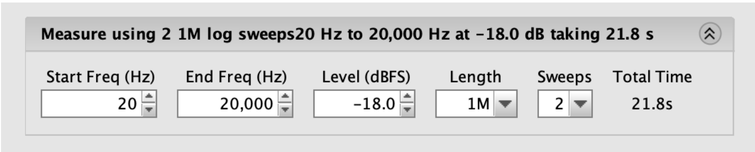

Figure 3.1 : Frequency sweep settings.

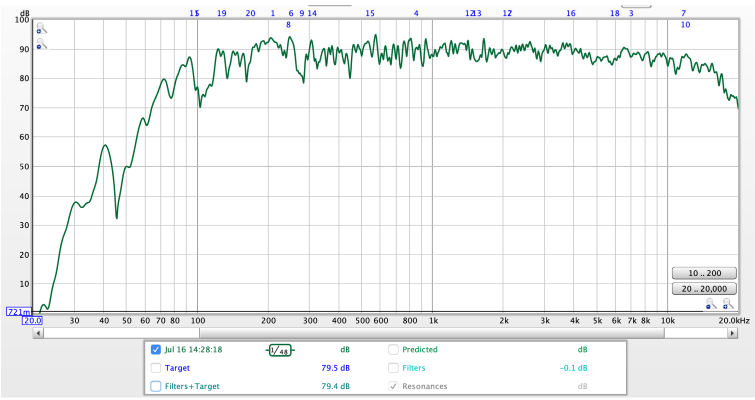

Figure 3.2 : Measurement result.

The waterfall diagram of the impulse response shown in Fig.3.5. As can be seen in Fig.3.5 the reverberation decays slowly in lower frequencies than higher frequencies. Between 100 Hz and 10 kHz Reverberation times lie between 480 ms and 640 ms where as in lower frequencies reverberation times are much higher. This makes the sounds build up more in the lower frequencies. Even though the artist use artificial reverb in their pieces this result may create some unwanted reverberation and unwanted build up while their pieces played in the performance hall.

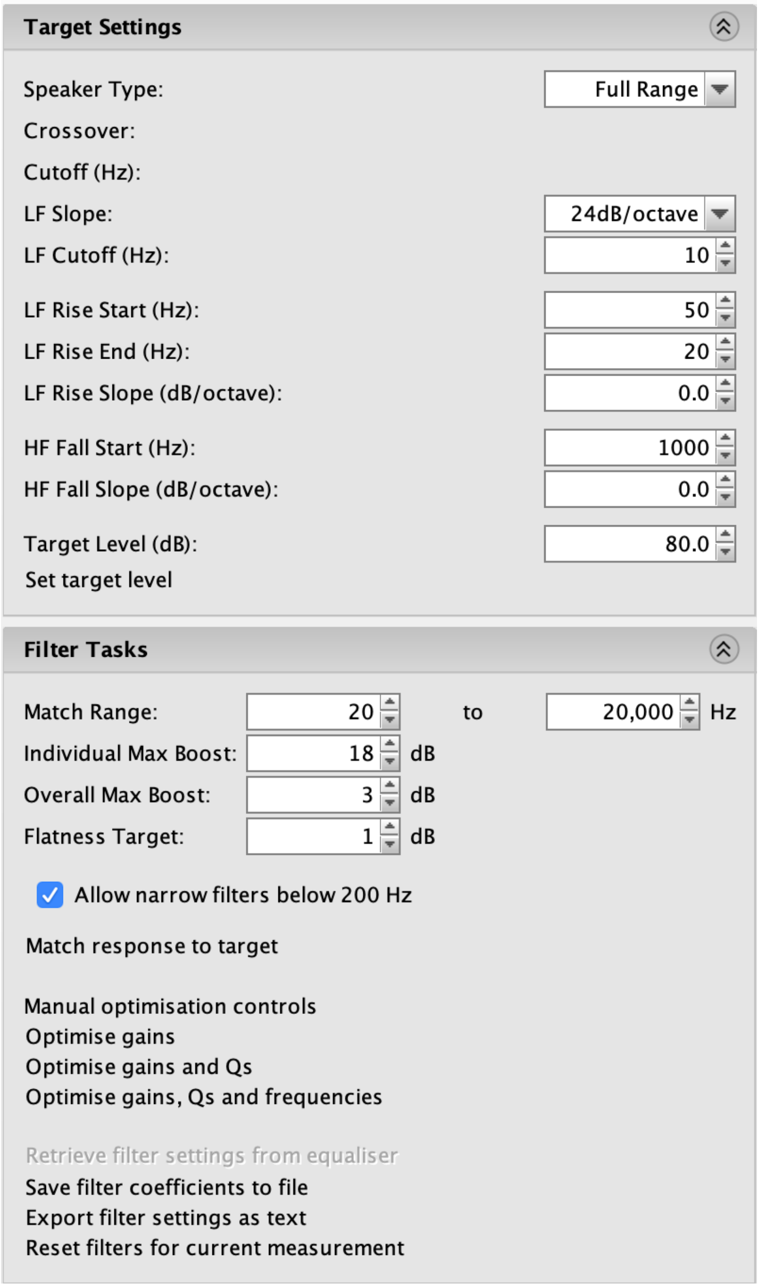

Figure 3.3 : Filter generation settings.

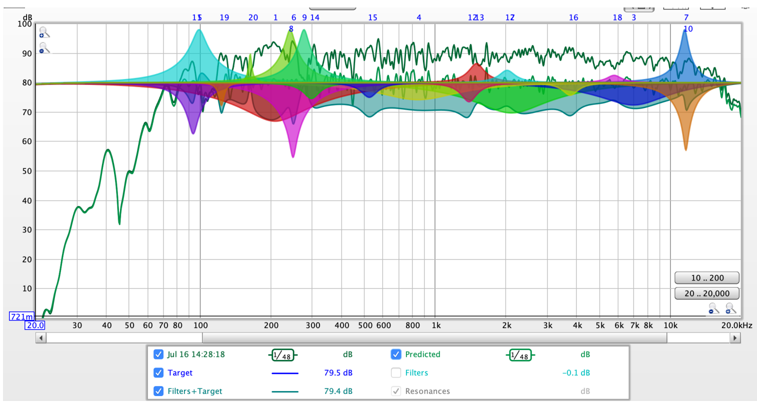

Figure 3.4 : Predicted frequency response after filter application.

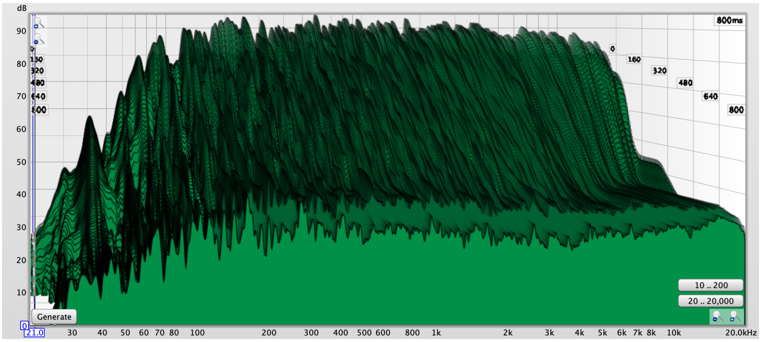

Figure 3.5 : Waterfall diagram for reverberation times over frequencies and SPL.

The frequency response of the performance hall is quite important as well. It can be considered as an additional external equalizer applied to the master track. As can be seen in Fig.3.2 the frequency response is not flat through the whole spectrum. It can be seen as the measurement result graph is the external equalizer that is applied to the source material. This is one of the reasons behind why the same source material sounds different in different rooms.

Using the room dimensions, the volume of the room is calculated as 282 m^3. As the result of the room dimensions, calculated axial room modes are shown in Table3.1.

Table 3.1 : Axial room modes of ITU, MIAM Usmanbaş Concert Hall.

Mode Number	Frequency (Hz)	Closest Pitch	n_x, n_y, n_z
1	14.78 Hz	A-1#	1-0-0
2	20.18 Hz	E0	0-1-0
4	29.57 Hz	A0#	2-0-0
6	40.35 Hz	E1	0-2-0
8	44.35 Hz	F1	3-0-0
11	59.14 Hz	A1#	4-0-0
13	59.97 Hz	A1#	0-0-1
14	60.53 Hz	B1	0-3-0
26	73.92 Hz	D2	5-0-0
32	80.71 Hz	E2	0-4-0
42	88.71 Hz	F2	6-0-0
54	100.88 Hz	G2#	0-5-0
58	103.49 Hz	G2#	7-0-0
75	118.28 Hz	A2#	8-0-0
80	119.93 Hz	A2#	0-0-2

3.1 Survey Output

The survey results of the blind test show the changes that participants would make regarding the impulse response of the performance space. By using a convolution plugin as the last plugin on the master chain, given changes applied to the original material and during the export phase the convolution plugin is bypassed. Then all the participants gathered in the concert room at the same place where the microphone was positioned. A blind test conducted between original version of the source material and the mix version made by using convolution reverb. The test is conducted whether they would choose the track A or B where the tracks are played randomly. Participants made a choice of which one translates better in the performance space. After the participants statement of preference, they made a commentary regarding their preference. The changes made to the original material, the preference of the blind test and the commentary are as follows;

Participant A made changes of 6 dB low shelf cut in 23 Hz (q = 1). 1.3 dB bell boost in 30 Hz (q = 0.21). 1.6 dB bell boost in 78 Hz (q = 1). 2.2 dB bell cut in 595 Hz (q = 0.48). 2.3 dB bell cut in 2053 Hz (q = 4.421). 1.4 dB bell cut in 4728 Hz (q = 3.4). 0.7 dB high shelf cut in 9143 Hz (q = 1). 4.1 dB bell cut in 21677 Hz (q= 1.2). Extra compression applied with 3.7:1 ratio 10 ms attack time and 400 ms release time. No change in time based effects. The mix made through convolution is preferred. On the commentary Participant A stated that every decision made during the mix is made according the convolution and by doing that the bass absence is well tolerated by the changes and overall mix sounded better with the listening space. In addition to that, the participant stated that more boost could be done with the high frequencies due to lack of air in the mix but overall the impulse response is really beneficial. Spectrum analysis of both versions shown in Fig.3.6 with integrated loudness levels (LUFS) and dynamic range (DR). In Fig.3.7 a representation for the equalizer changes made by the artist is shown.

Participant B made changes of 18 dB low shelf boost in 118 Hz (q = 1). 3.5 dB bell cut in 295 Hz (q = 1). 8.7 dB bell boost in 1685 Hz (q = 1.2). 6.9 dB high shelf boost in 6221 Hz (q = 1). Extra compression applied with 6.2:1 ratio 33 ms attack time and 18 ms release time. No change in time based effects. The original mix is preferred. On the commentary Participant B stated that both versions sounded bad in

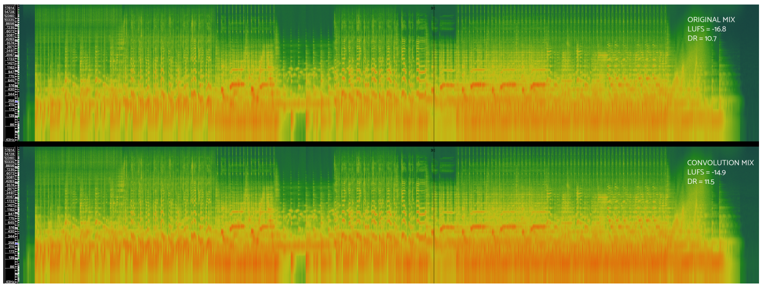

Figure 3.6 : Spectrum analysis of original mix and the mix made through convolution by Participant A.

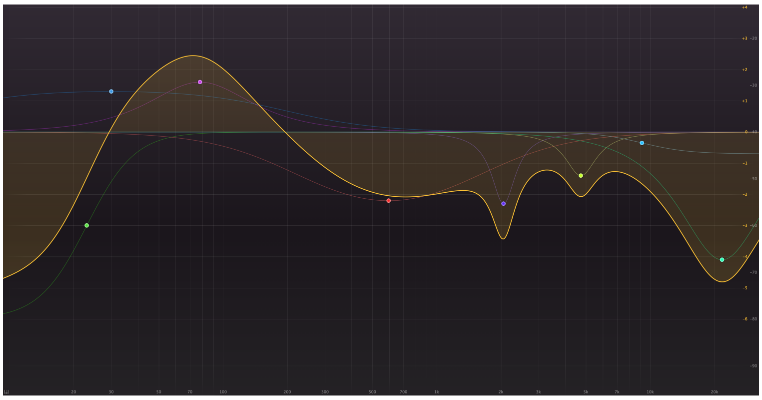

Figure 3.7 : Equalizer representation for the changes made by Participant A.

the room and has actually no clear preference. For the both versions the intended bass was not achieved. The mix made through convolution has too much bass where as the original has insufficient of bass frequencies. Although, the participant stated that this way of monitoring the room beforehand is beneficial by the reason that while doing the changes the feel of the impulse response convolution was really similar to the original one. The changes might work if made in a more subtle manner. Spectrum analysis of both versions shown in Fig.3.8 with integrated loudness levels (LUFS) and dynamic range (DR). In Fig.3.9 a representation for the equalizer changes made by the artist is shown.

Participant C made changes of 3 dB bell cut in 180 Hz (q = 0.75) on the bass channel. 3 dB boost in 1 kHz (q = 0.5) and 2.1 dB high shelf boost in 7941 Hz (q = 0.3) in guitar channel. No change in time based effects and dynamics. The original mix is preferred. On the commentary Participant C stated that the original mix sounded less sharp and better overall on the high mid frequencies. Also stated, while using convolution the mix started to sound muddy and to prevent the muddiness some bass frequencies are cut

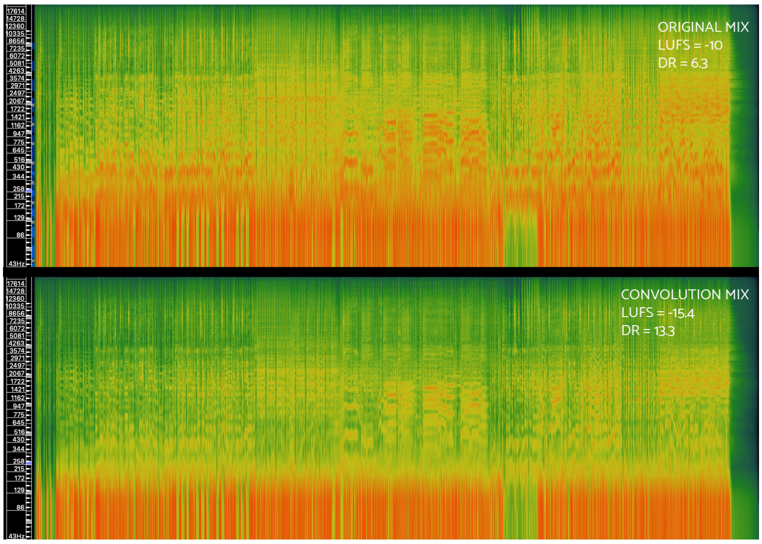

Figure 3.8 : Spectrum analysis of original mix and the mix made through convolution by Participant B.

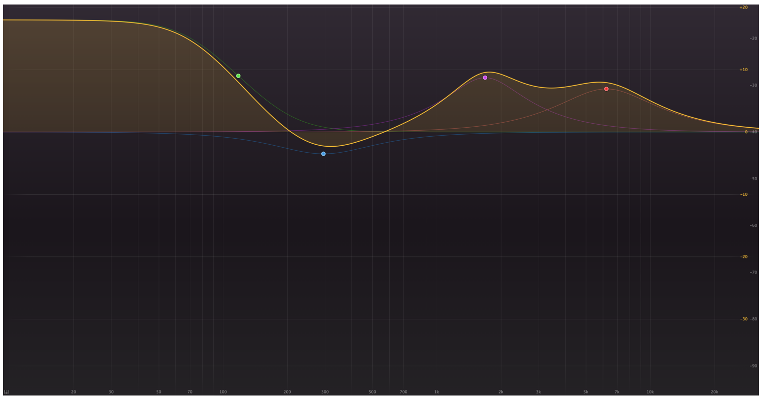

Figure 3.9 : Equalizer representation for the changes made by Participant B.

and the high frequencies of the guitar is boosted. Spectrum analysis of both versions shown in Fig.3.10 with integrated loudness levels (LUFS) and dynamic range (DR). In Fig.3.11 a representation for the equalizer changes made by the artist is shown.

Participant D made changes on the kick channel, 1.2 dB bell cut in 120 Hz (q = 13), 2.2 dB of bell boost in 3.9 kHz (q = 0.76). On the bass channel, 2.5 dB bell boost in 100 Hz (q = 0.8) and 1.3 dB bell cut in 1 kHz (q = 2.5). On the guitar channel 5.4 dB bell cut in 130 Hz (q = 12), 4.6 dB bell cut in 660 Hz (q = 13), 2 dB high shelf boost in 10 kHz (q = 0.3). 4 dB gain in snare reverb. No change in dynamics. On the commentary Participant D stated that the reverb change and bass boost regarding the convolution worked really well in the room. Also stated, the way of impulse response monitoring might be used for the live sound situations to get an extra insight beforehand. Spectrum

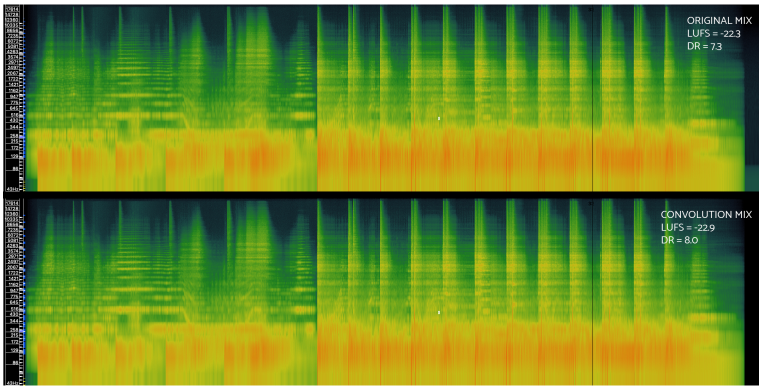

Figure 3.10 : Spectrum analysis of original mix and the mix made through convolution by Participant C.

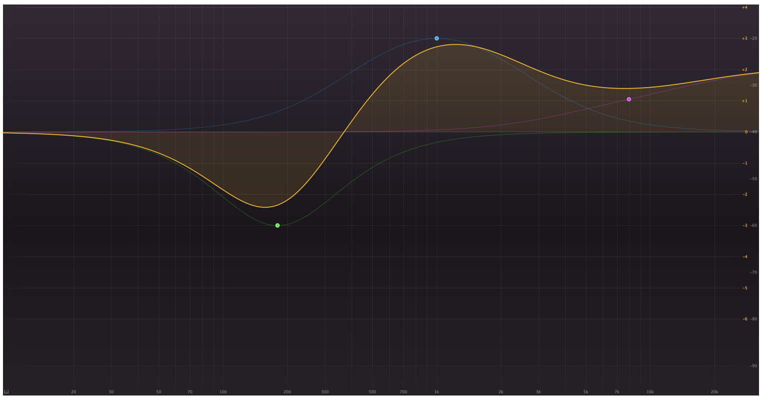

Figure 3.11 : Equalizer representation for the changes made by Participant C.

analysis of both versions shown in Fig.3.12 with integrated loudness levels (LUFS) and dynamic range (DR). In Fig.3.13 a representation for the equalizer changes made by the artist is shown.

Participant E made changes of 2 dB cut of reverb send. 3 dB cut of distortion send. In distortion send 3 dB bell cut in 550 Hz (q = 0.71). 8 dB bell cut in 918 Hz (q = 1.719), 2.6 dB high shelf cut in 3 kHz. In bass channel 2 dB bell boost in 58 Hz (q = 12.0), 6 dB bell cut in 69 Hz (q = 3.92), 5 dB bell cut in 110 Hz (q = 1.94). In kick channel 6 dB bell boost in 90 Hz (q = 10). Snare reverb is 3 dB boosted. Melody channel 5 dB cut. Bass channel 3 dB boosted. Hi-hat channel 3 dB cut. Hi-hat and crash reverb 2 dB cut. No change in dynamics. The mix made through convolution is preferred. On the commentary Participant E stated that instead of doing subtle changes on the master track, making changes in the overall mix using convolution is preferred. Also stated, The transients are more defined, bass frequencies sound more full, in general the whole mix sounds fuller. Participant stated that being more cautious about the

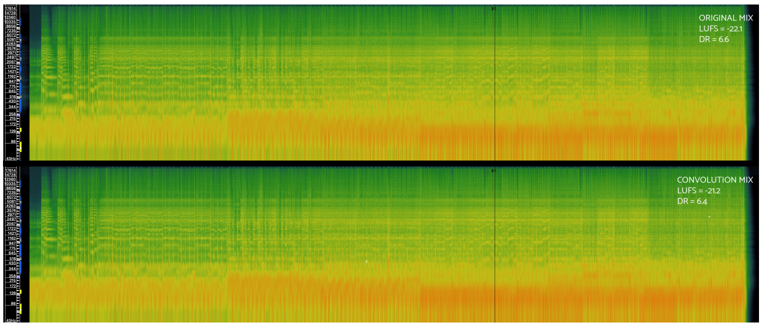

Figure 3.12 : Spectrum analysis of original mix and the mix made through convolution by Participant D.

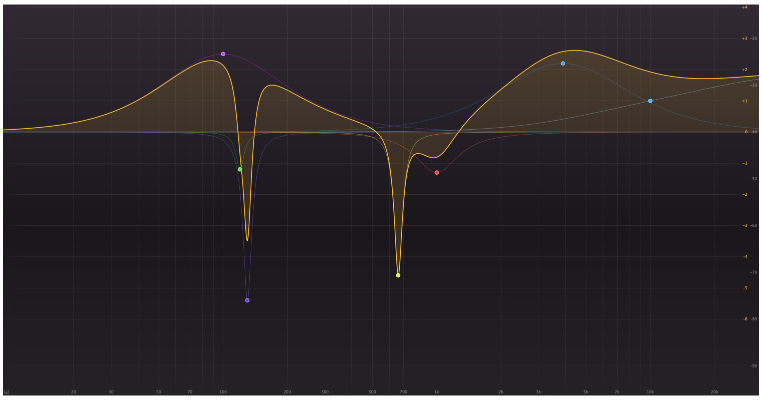

Figure 3.13 : Equalizer representation for the changes made by Participant D.

bass frequencies and room tones and did some changes accordingly for resonant bass frequencies and the changes worked really well. Spectrum analysis of both versions shown in Fig.3.14 with integrated loudness levels (LUFS) and dynamic range (DR). In Fig.3.15 a representation for the equalizer changes made by the artist is shown.

Participant F made changes of 4 dB high shelf boost in 170 Hz (q = 1.0). 2 dB bell boost in 193 Hz (q = 3.5). 3.5 dB high shelf boost in 2400 Hz (q = 1.0). Master channel compressor setting changed by longer attack time and shorter release time. (8ms to 72ms and 100 ms to 8 ms). No change in time based effects. The mix made through convolution is preferred. On the commentary Participant F stated that the bass frequencies sounds better, the transients defined better. Also stated, more changes could be done to the high frequencies that might make the whole mix sound better. Participant also stated that this whole experience was surprisingly good. Spectrum analysis of both versions shown in Fig.3.16 with integrated loudness levels (LUFS) and dynamic range (DR). In Fig.3.17 a representation for the equalizer changes made by the artist is shown.

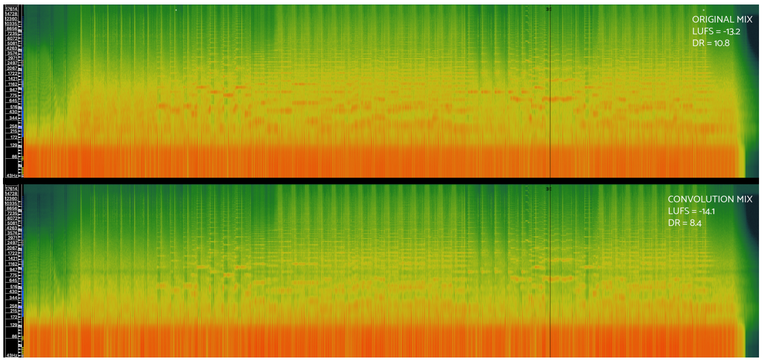

Figure 3.14 : Spectrum analysis of original mix and the mix made through convolution by Participant E.

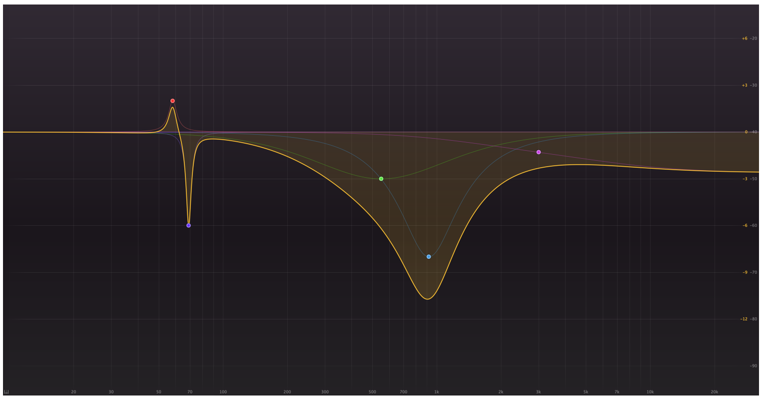

Figure 3.15 : Equalizer representation for the changes made by Participant E.

Participant G made changes of 1.5 bell boost in 10 Hz (q = 0.4). 2.5 dB bell boost in 40 Hz (q = 0.4). 1 dB bell boost in 160 Hz (q = 0.4). 0.7 dB bell cut in 251 Hz (q = 6.3). 1 dB bell cut in 506 Hz (q = 9.7). 1.2 bell cut in 1020 Hz (q = 10.7). 0.8 dB bell cut in 2400 Hz (q = 4.3). 1.5 dB bell boost in 2500 Hz (q = 0.4). 3.5 dB boost in 20 kHz (q = 0.6). 2dB of extra limiting applied. No change in time based effects. The mix made through convolution is preferred. On the commentary Participant G stated that the bass frequencies sounds more clean and well defined, the transients sound more defined and the convoluted version sounds less lush which is is preferred by the participant for their music. Spectrum analysis of both versions shown in Fig.3.18 with integrated loudness levels (LUFS) and dynamic range (DR). In Fig.3.19 a representation for the equalizer changes made by the artist is shown.

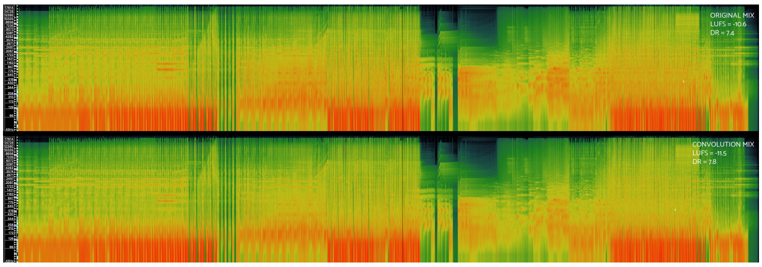

Figure 3.16 : Spectrum analysis of original mix and the mix made through convolution by Participant F.

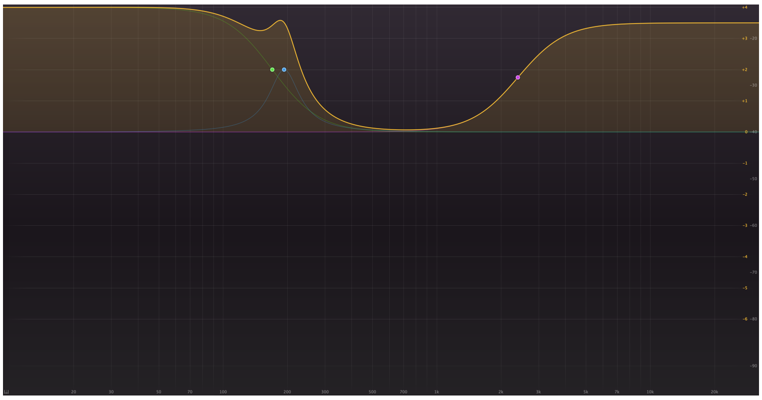

Figure 3.17 : Equalizer representation for the changes made by Participant F.

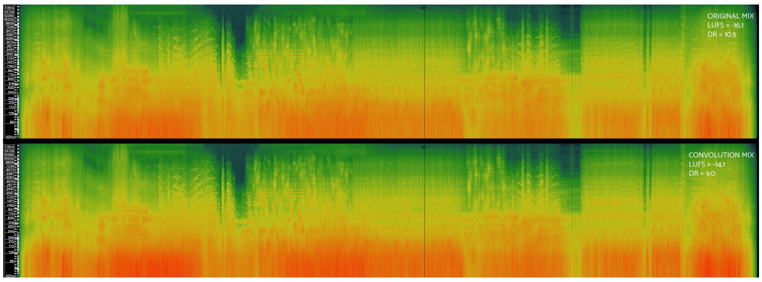

Figure 3.18 : Spectrum analysis of original mix and the mix made through convolution by Participant G.

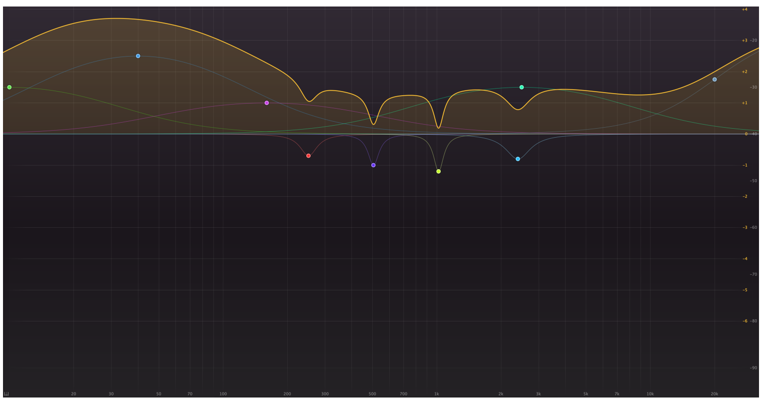

Figure 3.19 : Equalizer representation for the changes made by Participant G.

4. CONCLUSION

The study and survey show that having an impulse response beforehand would make an impact on mixing stage. The result of the survey and the blind test show that 71% of the participants preferred the mix made through convolution of the impulse response. As a personal comment I believe that, this monitoring concept with impulse response is similar to an instrument which you can learn to use in time and get better using it. In home mixing situations where the artist is used to how things sound in their room with their playback equipments (for both headphones or reference monitors) having convolution plugin with the right impulse response file on the master channel will be a valid monitoring because the overall sound will be coloured by both the equipment and room that the artist is used to and the performance space impulse response. I believe this would make an impact on composing stage as well, it can open some possibilities that an artist might not think from the first sight. By using the impulse response of an art installation space, the artist may use the acoustic information beforehand to use the space as resonator similar to Mark Bain's work "Live Room" (1998). Being able to monitor the sound of a musical piece beforehand like it's played in the venue is a simple yet powerful concept.

There are downsides of this approach as well. These downsides can be considered as future works to enhance this concept. The impulse response used for this project is taken from only one position. Therefore, all the calculations and graphs are valid for that particular position. Changing the position would generate different results. To be able to overcome this issue, the impulse response should be taken as many spots as possible where listeners can be positioned in a normal concert situation. All the graphs should be averaged to be able to create a better distributed solution which can suit to many cases. Also, another downside of the project is that the impulse response recorded while there are 2 people in the concert space. The capacity of the concert spaces and venues can be high and the resulting impulse response will be more reverberant than a filled up space with listeners. When there are more people in the performance space the sound will be absorbed more by the bodies of the listeners. This will create a different result as well. The amount of people could be averaged by looking at the ticket sales and the impulse response could be recorded or modeled like there are average amount of listeners inside. This would create a better result overall. In addition to that, the experiment group consist of 7 people. To be able to generalize the overall effect for this kind of project the sample size should be much more. Total of 7 people with 71% means 4,97 people. The 50% barrier is passed by only two people's decision. This can be critical. More samples lead better generalization.

Another concept that can be applied with the averaged results mentioned in above paragraph is room equalization and room modeling. In Fig.3.4 a possible correction for the performance space is shown. These filter settings are calculated by windowing the direct sound recorded by the microphone. An impulse response for the filters that are shown in Fig.3.4 can be generated. By convoluting the output of the system with the filter impulse response, flatter frequency response can be achieved for the room then overall sound can be tweaked to taste. The room equalization can work only if phasing taken into consideration. To overcome possible phasing problems every speaker should be calculated seperately and phase aligned accordingly. Also, this concept can be used for monitoring different types of speakers or headphones. In anechoic chamber some headphones (with dummy head recording microphones) or some speakers can be modeled then in a mixing situation as an extra perspective, the engineer can monitor how the music would sound on different types of speakers etc.

As a conclusion, I will suggest a utopian solution that may improve some of the problems. The music technology evolved into a digital era. We listen music through computers, smart phones by mp3, web or streaming services. The music stored as a pack of digital data then converted into analog signal and pushed out through speakers. Instead of trying to find an optimum spectrum that gives good result by music mastering, a relative room equalization IR might be applied as a convolution on the final stage to the digital audio. This can be achieved by measuring the performance room once. The house engineer would have less work if this would be used in concert halls. This also can be implemented to the headphones or earbuds by using dummy head microphones. Knowing the filter coefficients or having the IR file will be enough for the application. The convolution is computationally expensive yet the computers and smart phones are much powerful today. A static equalizer can be applied after the calibration as a taste. I believe this will ensure a better listening experience. One can measure their own listening environment and apply the corrections to their personal devices for a tailor cut listening experience. I believe this approach can lead better results in overall performance and concert situations.

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