Detmold
HfM
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ETI

MASTER THESIS

Optimization of room acoustic parameters in artificial reverberation for enhanced perceived naturalness of musical sounds

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Abstract

Reverberation is a fundamental acoustic phenomenon that shapes how humans perceive spatial cues on a daily basis. In contemporary music production, however, reverberation is often added separately from the dry sound with reverberation processors. While this workflow is practical, it is often achieved in a perceptual manner, which motivates the main objective of establishing an objective set of guidelines that can assist sound engineers in replicating the reverberation characteristics of acoustic environments by determining which physical and psychoacoustic parameters are most critical for achieving a natural-sounding artificial reverberation that convincingly resembles a real physical room environment.

This thesis investigates the relationship between common room acoustic parameters and perceptually-oriented cues of naturalness, with the goal of formulating objective, practical guidelines for sound engineers and music producers. By examining how room-acoustic parameters align with perceptual attributes from the listener's perspective, the work aims to clarify how specific control choices on algorithmic processors can reproduce the typical characteristics of real acoustic environments.

Acknowledgements

First and foremost, I want to thank my parents for their unwavering support throughout my whole life. I am also grateful to my family, friends, to Tais, and to everyone who has ever lent me a helping hand.

My sincere thanks go to the Hochschule für Musik Detmold and to the ETI for welcoming me, to all the faculty and my fellow students for being part of this experience, and especial thanks to my supervisor, Prof. Malte Kob, for his guidance and for generously sharing his knowledge over these years.

I hope this Master's Thesis proves satisfactory and serves as a foundation for further research on this topic.

Glossary

Artificial Sample Samples generated using an algorithmic reverberation processor.

DAW Software used for recording, editing, and producing audio..

DI Box Converts unbalanced, high-impedance signals into balanced, low-impedance signals.

ITA Toolbox MATLAB toolbox for acoustics measurement, analysis, and simulation.

MATLAB Programming platform for numerical computing, data analysis, and visualization.

Processor Controls Adjustable settings within a reverberation processor.

Processor Controls A saved set of values on the processor's controls.

Room Acoustic Parameter Measurement of the acoustic behavior of an environment.

Soundscape The acoustic perception of an environment.

Acronyms

AC-GTR Acoustic Guitar.

BS Brahms Room.

C80 Clarity.

CH Church Room.

D50 Definition.

E-GTR Electric Guitar.

EDT Early Decay Time.

ETI Erich Thienhaus Institut.

FR Frequency Response.

HfM Hochschule für Musik Detmold,.

HOF Hall of Fame.

HR H-Reverb.

JND Just Noticeable Difference.

LEGATO Legato Singin.

M7 Bricasti M7.

rt60 Reverberation Time (60 dB).

SNR Signal-to-Noise Ratio.

SPEECH Speaking Voice.

ST Studio Room.

STACCATO Staccato Singing.

T30 Reverberation Time (30 dB).

TS Center Time.

TV TrueVerb.

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1. Introduction

Reverberation is one of the most important natural acoustic phenomena that occurs in everyday life; it has consequently become intrinsic to the human expression of arts and music, playing a key role in many traditions and cultural events. We cannot imagine, for example, a play in a theater, a speech in a hall, without the corresponding cues of depth and space that reverberation brings along, let alone the intermixing of voices that occurs when a choir is singing in a church or an orchestra performing on a stage. However, while these statements have been true across history, and still continue to be true in day to day life, when recorded music comes into the picture, for most contemporary pop music, the reverberation and spatial cues heard are most probably not naturally captured.

Whenever a sound engineer or music producer is recording a single instrument, the most common and practical method of dealing with reverberation is to isolate the track from the room reflections by making the recording sound as "dry" as possible, meaning using absorption screens on the back of microphones as well as recording in rooms with relative low reverberation times, and to add reverberation artificially in a later stage during production, via external methods, like analog and digital reverberation processors, or plugins within the digital audio workstation (DAW).

Figure 1.1.: Absorption screen for microphone.

Source: Image taken from: https://www.walmart.com/ip/Sound-Absorbing-Foam-Screen-Panel-Vocal-Recording-Platethe.

The main reason this is done is a practical one: to be able to have the most control possible over how the soundscape is shaped to make it fit into the mix. This is quite an interesting approach, because via these "artificial" methods, we can emulate a plethora of soundscapes and sound options that we could not normally access so easily via practical methods. And while artificial reverberation is often based on physical models, when regarding algorithmic reverberation, or on recordings of a specific physical space, for convolution reverberation.

However, this brings into consideration the following question: when trying to recreate a real physical room soundscape with artificial reverberation, how can we know that we are achieving an accurate representation of what the sound source would sound like?.

With what certainty can the engineer know that the controls knobs and faders he is moving around in the reverberation processor actually gives a faithful representation of what a church sounds like, or a small classroom, or a big conference hall?. The normal, empirical way of doing this is by "using the ears" and adjusting the artificial reverberation controls until the audio resembles what the operator considers to be similar to the real environment.

This, however, is a purely subjective method of recreating reverberation fidelity. It is by no means a poor way; countless records have been made over the years with excellent audio quality and great reverberation definition. But is there a more scientific way of achieving this?

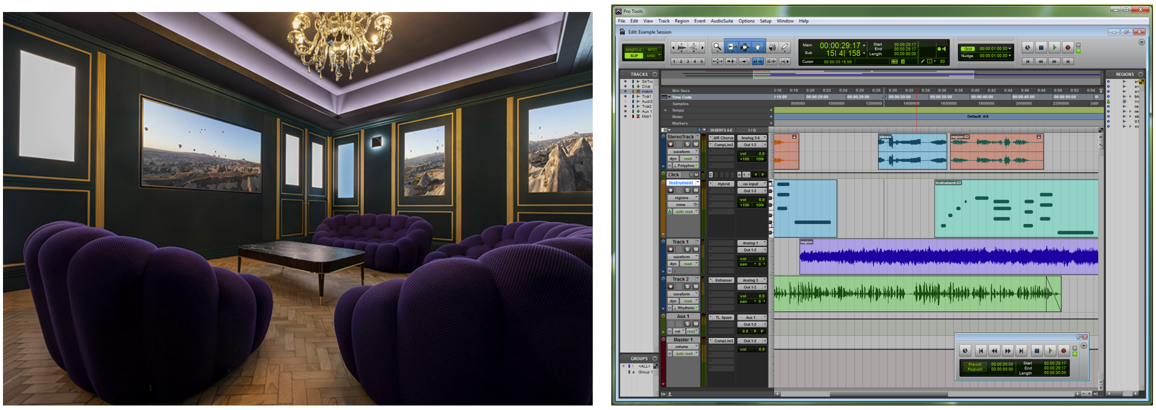

Figure 1.2.: How to recreate a soundscape in the studio?.

What are the parameters, both physical and psychoacoustic, that matter the most when analyzing the naturalness of an artificially generated reverberation that resembles a physical room?

The purpose of this thesis is to analyze the most common parameters present in room acoustics and try to link them to psychoacoustic parameters from the listener's perspective, to find an objective set of guidelines that can help new sound engineers easily replicate the reverberation characteristics of common room acoustics examples.

The focus will be on algorithmic reverberation because, due to the inherent nature of convolution reverberation, it already includes the spatial cues of the room acoustics recording, and they are generally considered to be more natural sounding. So the objective is to develop the set of guidelines specifically for algorithm-based processors. Several processors will be analyzed in order to obtain more general results on not focusing on a single one. Several rooms will also be analyzed to cover different geometries and sizes, as well as different instruments. To obtain a varying amount of interactions between room acoustics and the source's tonal characteristics.

The motivation for this topic comes from my own experiences as a sound engineering student, struggling to understand the controls of the many reverberation processors available and how they affected the signal. Hopefully, by the end of this study, a set of recommendations can be elaborated that would help to guide new and old engineers and music producers to make their artistic work easier to do.

2. Background

In this chapter, the fundamental theory necessary to understand this study will be discussed, starting with the fundamentals of Room Acoustic, the parameters to be analyzed in this project, and the measurement technique utilized to obtain them. Following that, the two more common techniques of creating artificial reverberation, algorithmic and convolution, will be discussed along with the reason to focus this study only on algorithmic reverberation. In the final section, the evaluation of reverberation, the perception of naturalness, the listening test methodologies, and the statistical analysis tools used across the research will be explained.

2.1. Fundamentals of Room Acoustics

In order to determine whether an artificial reverberation processor is actually naturally emulating a physical room soundscape, first, it's important to understand the common acoustic characteristics of typical rooms. For this, one of the most important quantifiable parameters is the room acoustic parameters, of which the one relevant for this study will be defined next.

2.1.1. Room Acoustic Parameters (T30, EDT, D50, C80, Ts)

When talking about reverberation, the three main factors determining its characteristics are the direct sound, the early reflections, and the late reverberations. The latter one is measured with the Reverberation time *RT60* number, which indicates the time required for the sound pressure level to decay by 60 dB after the source has stopped. However, for cases where the measured time is too short or the signal to weak to determine a decay of 60 dB, a very common alternative to use is the extrapolation of the -5 dB to -35 dB, called *T30*. [1].

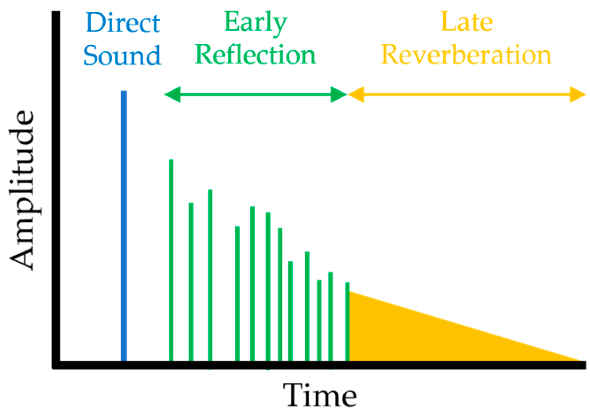

Figure 2.1.: Components of sound over time.

Source: Image taken from Eunjae Kim' article on Sound Propagation Algorithms.

Early Decay Time (*EDT*), which gives a value for early reflections, is obtained from the initial 0 dB to -10 dB portion of the decay curve and usually tends to correlate more strongly with the perceived level of reverberation at the listener position than the actual reverberation time [2, 1].

Other important room acoustic parameters that help to describe the perceived clarity of sound in a room are Definition (*D50*), which is used for spoken voice and defines the ratio of early-to-late energy arriving within the first 50 ms, and the Clarity (*C80*, which serves the same purpose but for music and is used for energy arriving in the first 80 ms. Finally, the Center Time T_s represents the temporal distribution of energy in the signal and decreases as early energy increases. All these parameters are defined in the ISO 3382-1 standard [1, 3]

$$D_{50} = \frac{\int_0^{50\text{ ms}} p^2(t)\,dt}{\int_0^{\infty} p^2(t)\,dt}, \qquad C_{80} = 10\log_{10}\frac{\int_0^{80\text{ ms}} p^2(t)\,dt}{\int_{80\text{ ms}}^{\infty} p^2(t)\,dt}, \qquad T_s = \frac{\int_0^{\infty} t\,p^2(t)\,dt}{\int_0^{\infty} p^2(t)\,dt}$$

where p(t) is the sound pressure as a function of time (pressure impulse response).

For ease of interpretation, the ISO 3382-1 provides a recommended frequency averaging table, together with the "just noticeable differences" (JNDs) for listeners, for several room acoustic parameters, with the relevant ones for this project being:

JND for acoustic room acoustic parameters

Acoustic quantity	Unit	JND	Typical range
Reverberation Time (RT60)	seconds	Rel. 5 %	
Early Decay Time (EDT)	seconds	Rel. 5 %	1.0 s to 3.0 s
Definition (D50)	%	5%	30 % to 70 %
Clarity (C80)	decibels	1 dB	-5 dB to +5 dB
Center Time (Ts)	miliseconds	10 ms	60 ms to 260 ms

Table 2.1.: Just noticeable differences values according to the ISO 3382-1.

2.1.2. Measurement Technique Standards

Impulse response-based measurement, through a swept sine excitation and de-convolution, has become the preferred method in modern practice to calculate room acoustic parameters, because it offers a robust signal-to-noise ratio *SNR* and reduced harmonic distortion *THD* [4, 1]. ISO 3382-1 recommends sufficient dynamic range to measure impulse responses designed to calculate reverberation time: " for the determination of T30, the level of the background noise must be at least 45 dB below the maximum." (ISO 3382-1:2009, 5.3.3) [1]. The sound source is recommended to be as omnidirectional as possible, and it has to be able to produce a sufficient level to achieve the minimum required range.

The microphones and recording devices used should also meet the requirements of professional-grade audio equipment. The number of microphones varies depending on the size of the rooms to be measured. For smaller rooms, as few as three measurement microphones can be used, and for very large auditoriums, there is no limit to the number of microphones required, as long as they cover the largest area of the room as possible [1, 3]. The positions of the microphone should imitate a person sitting in the room, and the distance between them has to be approximately 2 [m].

At least three measurement positions of the source have to be used and averaged to obtain an accurate representation of the room's acoustics, and if a single representative measurement for the room is wished to be obtained, all the measurements from the different microphone positions should also be averaged.

2.2. Artificial Reverberation

Most modern music produced for pop or rock audiences uses "artificial" reverberation processors to introduce depth and atmosphere to single audio recordings. These additions are usually done in a controlled post-production environment where the parameters of the reverberation can be tweaked and adjusted to fit the exact needs of the musicians and producers. Only on very few occasions, the actual acoustics of the room are captured and kept on the recordings, with the exceptions normally being multi-channel instruments and acoustic recordings. When applying this artificial reverberation, the two most common options sound engineers have to choose from are Algorithmic and Convolution reverberation processors.

2.2.1. Algorithmic vs. Convolution Reverberation

Classical "artificial" algorithmic reverberation designs use parallel comb filters and all-pass sections to produce dense, exponentially decaying responses with low coloration. According to Schroeder, such configurations can yield a “colorless artificial reverberation” when distortion and coloration are kept small [5]. More modern types of algorithmic reverberation processors include feedback loops and hybrid networks that increase echo density and control modal buildup while exposing musically meaningful parameters [6].

On the other hand, artificial convolution reverberation obtains the linear time-invariant response of a space by convolving the input signal with a measured IR. This approach reproduces a specific representation of the time and energy structure of an environment, at the cost of less control of the parameters and potential latency [3, 4].

2.2.2. Common Applications and Limitations of Algorithmic Reverberation

Algorithmic reverberation processors remain prevalent in music production and interactive audio because they are efficient and easily automated. However, their simplified structure and limited spatial detailing can make the recreation of certain perceptual cues hard to achieve without additional modeling [7, 6, 8]. Recent listening tests report broad variance in perceived naturalness across algorithmic reverberation types and settings, with the differences often hinging on early to late energy balance and temporal cues rather than the algorithm type.[6, 8]. Estimating room-like parameter target values from measurements or recordings allows algorithmic reverberation processors to approximate the desired physical environment to a certain degree [7].

2.3. Perceptual Evaluation of Reverberation

Perceptual terminologies such as *naturalness* or *realism* are used in acoustics to describe the subjective characteristics that a listener expects to hear when experiencing sound in a physical acoustic environment. Like, for example, rooms with large reverberation times tend to sound more *"spacious"*, or very absorptive environments are usually described as sounding *"dull"*. There are no actual, defined parameters to quantifiably measure this characteristic, which can vary greatly in description between listener experiences and background; however, there exist methods to evaluate the perceptual impact those qualitative characteristics have on the listener.

2.3.1. Naturalness in Sound Perception

From the interpolation of early energy timing, spectral coloration, late decay statistics, and binaural/directional cues [9, 8], experimental evidence links judgments of the defined ISO 3382 room acoustic parameters [1] to more perceptual characteristics of sound. The German Institute for Standardization defines in its Specifications and Instructions for the room acoustic design (DIN 18041) [10], a contextualized room acoustic parameter target; the tolerance range of percent variation of reverberation time *T* in relation to a target reverberation time T_{ref} over frequency, as seen in 2.2.

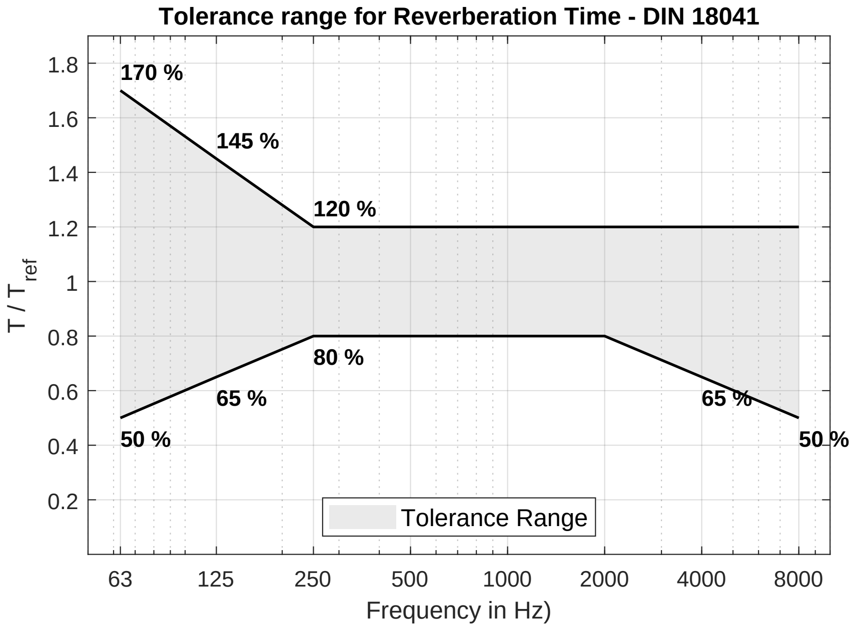

Figure 2.2.: Tolerance range of reverberation time T as a function of frequency.

The reference reverberation time, also called T_{soll} in German, is defined by the purpose and usage of the room in scale to its volume in cubic meters, where A1 is rooms used for music, A2 rooms used for speech, A3 and A4 for teaching rooms of different sizes, and A5 is for sport rooms.

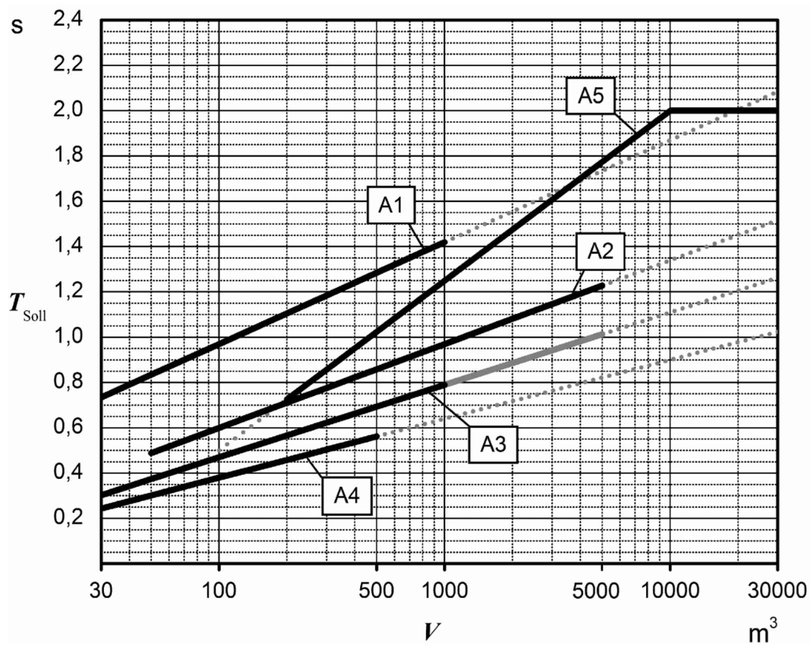

Figure 2.3.: Target values of T_{soll} for different room types.
Source: Graph taken from the DIN 18041 standard.

However, since the reverberation time of a room depends not only of its size, but of many factors such as its geometry, absorption material, amount of objects present inside, and more, and these factors can vary from room to room even on the most similar conditions, the ISO 23591 [11] was examined to provide a range of target reverberation times for music and speech rooms, where the green areas 3 and 6, seen in Figure 2.4 represent the accepted T_{mid} for amplified signals, the blue areas 2 and 5 represent T_{mid} for loud acoustic music, and the red areas 1 and 4 represent T_{mid} for quiet acoustic music. These graphs will be used to analyze the obtained results of the room acoustic measurement in this study, as well as the variations in the reverberation and early decay time.

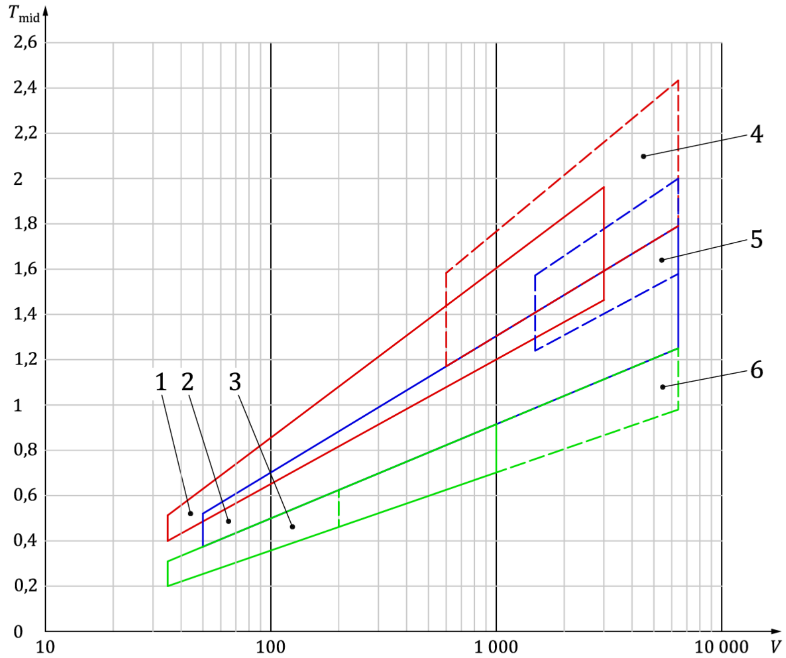

Figure 2.4.: Reverberation time (T_{mid}) relative to room Volume for different types of music.

Source: Image taken from the ISO 23591 standard.

2.3.2. Listening Test Methodologies

In order to evaluate the impact on perception that the variation of room acoustic parameters has, a certain number of tests have to be conducted with qualified listeners who have had ear training or professional music experience, to judge with a keen ear the amount of qualitative difference each type of variation produces on human perception. There are numerous types of listening test designs and methodologies, but for this study, the focus will be on two kinds.

Multiple-stimulus listening test allows for sensitive discrimination among condition variants. The ITU-R BS.1534 (MUSHRA) [12] describes a multiple-stimulus architecture with a hidden reference and one or more anchors of lower quality, rated on a 0–100 quality scale. It is widely used to compare variants of the sample signal under controlled settings. Perceptual studies have used MUSHRA directly or in modified form in the past with successful results [9]. The other type of test is "Force-Chose" one. In this type of listening test architecture, listeners must choose among a defined number of alternative signals on each trial. This method allows a more bias-resistant measurement for the preference of the test signal, although its design is more simple, it can be very effective.

3. Methodology

The goal of the study is to investigate the effect that changes in room acoustic parameters have on the perceived naturalness of reverberation when using algorithmic reverberation processors that simulate real room acoustic environments. This chapter explains the strategy used to achieve this goal by detailing each stage of the research process, as well as the considerations made in the theoretical, technical, and experimental aspects, to allow a direct comparison between recordings made in real room acoustic environments and recordings made by using artificial reverberation approximations of those same environments. The methodology was designed to ensure that differences in perception could be attributed solely to the controlled variations in acoustic parameters, rather than uncontrolled factors such as performance inconsistency or playback system coloration.

The chapter begins by describing the equipment, materials, and acoustic environments used in the study, followed by the process of sample generation, parameter modifications, and listening test design. The final sections detail the two-phase perceptual evaluation, which first identified the most natural-sounding artificial samples and then systematically tested listener sensitivity to controlled parameter changes.

3.1. Study Design

To achieve a systematic and quantifiable rating on the perceptual impact that variations in room acoustic parameters have on the naturalness of artificial reverberation, particularly Reverberation Time (T30) and Early Decay Time (EDT), its necessary to first obtain samples that use artificial reverberation and are considered to sound natural, meaning that they resemble and are indistinguishable from samples recorded in an actual physical environment.

In order to take the subjectivity of the artistic performance out of the equation, various samples were recorded in an anechoic chamber and later used as the "dry" samples for both the physical environments and artificial processors, to ensure that the performance is the same in both cases. The dry recordings were made in an anechoic chamber so that no room acoustic parameters from these original recordings bleed into the next phases.

3 Rooms × **5 Instruments** × **4 Processors** = **60 Samples**

Figure 3.1.: Amount of samples.

After obtaining the "artificial" samples (Fig. 3.1) by combining each source, enviromennt and processor, and using as a reference the "real" samples obtained (all derived from the original "dry" samples), a qualification process can begin to determine whether these artificial samples actually sound natural to the listener. After that, the samples that are judged to be natural-sounding can have their room acoustic parameters modified in stages, and a listening test comparing all of these modifications can be produced. The global structure of the study (seen in Fig. 3.2) goes as follows:

- **Controlled Dry Recordings**: Various music samples as well as speaking voice samples were first recorded in an anechoic chamber to obtain completely “dry” signals, which were later to be re-recorded in the different room environments as well as processed with the artificial reverberation processor. These samples included acoustic guitar, electric guitar, legato singing, staccato singing, and speaking voice. Using identical performances for all conditions ensured that no performance variability could influence the perceptual outcome.

- **Real Room Re-recordings and Acoustic Measurements**: The dry samples were re-recorded in three distinct acoustic environments: a music studio, a concert hall, and a church. Using a Neumann KH 120A loudspeaker and a binaural recording head. Simultaneously, room impulse responses were captured using a Globe Source and measurement microphones to extract the room acoustic parameters of the room: Reverberation Time (T30), Early Decay Time (EDT), Clarity (D50), and Center Time (C80). These measurements served as reference values for later matching the artificial reverberations to the real rooms.
- **Artificial Reverberation Processing**: The dry signals, convolved with the KH 120A loudspeaker's frequency response to ensure playback consistency, were processed using four algorithmic reverberation processors: the TC Electronic Hall of Fame 2 (HoF 2) guitar pedal, the Bricasti M7 hardware reverb unit, and two software plugins, the Waves TrueVerb and the Waves H-Reverb. For each room, processor parameters were adjusted to approximate the measured room acoustic parameter values, with a focus on the 2 main parameters of this study, Reverberation Time (T30) and Early Decay Time (EDT), while at the same time being monitored against the real recordings.
- **Qualification Test**: Participants are asked to judge whether each artificial sample sounded natural or unnatural. This step identifies the most convincing processor–room–instrument combinations, independent of processor type.
- **Perception Listening Test**: Using only the best rated samples per room-instrument combination that were previously qualified as natural sounding by the listeners, Reverberation Time (T30) and Early Decay Time (EDT) were systematically modified one at a time while keeping the other parameter as constant as possible by tweaking the processor's settings This allowed for a controlled investigation of the perceptual sensitivity to each room acoustic parameter (Reverberation Time (T30) and Early Decay Time (EDT).

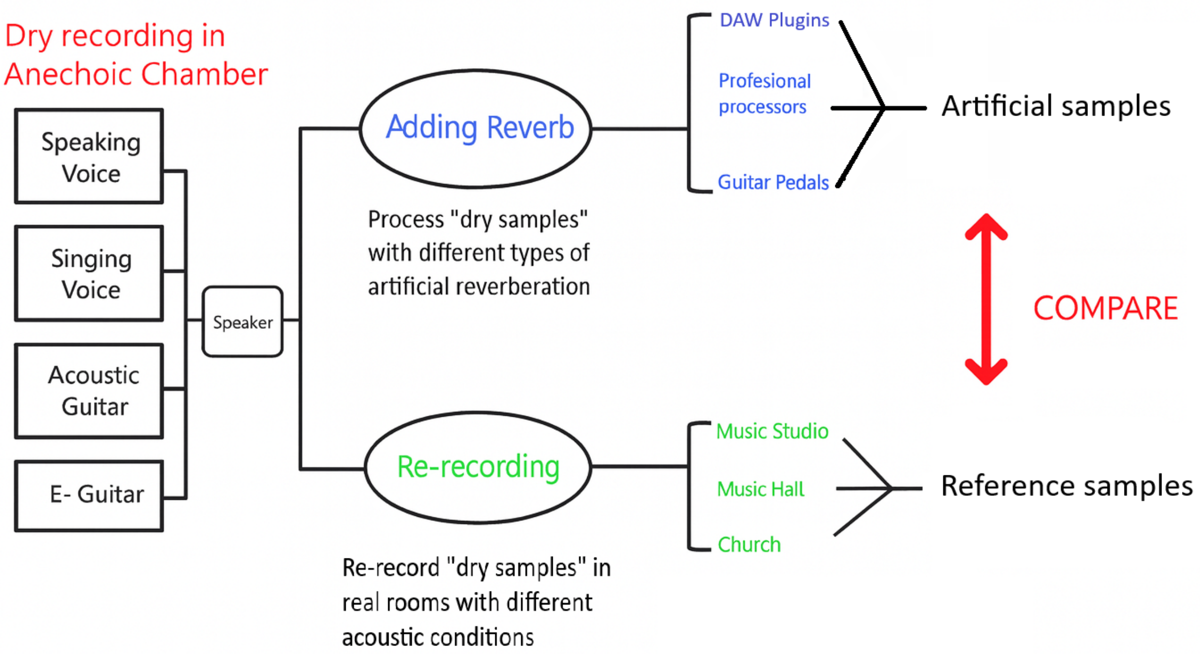

Figure 3.2.: Study Design.

3.2. Equipment, Materials and Environments

This section comprises a detailed description of all the elements used in the process of of acquisition, reproduction and processing of the audio samples needed for this investigation, including the specifications of the different recording and playback equipment used for capturing the musical samples and obtaining the room acoustic measurements, the different musical sources used and the criteria for choosing them, and the various physical environments selected, along with the artificial reverberation processors used to simulate them and generate the test stumulis.

3.2.1. Recording and Playback Equipment

The starting anechoic recordings were made using microphones that fitted the nature of the sound sources, applying conventional techniques used in standard music studio productions, like the large diaphragm condenser microphone Neumann TLM 127 [13] for the human voices and acoustic guitar, and the Radial J48 DI box [14] for the electric guitar. For the playback in the different physical environments, the KH 120A directional speaker [15] seen in Fig. 3.4 was used to reproduce the sources in a controlled and repeatable manner. This speaker was selected for its relatively flat on-axis frequency response and suitability for accurate vocal and musical reproduction. The sources were captured with the acoustic binaural dummy head Neumann KU 100 [16], which has two calibrated microphones located in the ear canals, which imitate the spatial cues and timbral characteristics perceived by a human listener. All the different recording equipment can be seen in Fig. 3.3. The recording chain was completed using a BabyFace Pro interface [17], a transparent, low-noise audio interface and pre-amplification stage, and ProTools as the Digital Audio Workstation.

Figure 3.3.: Recording equipment: Neumann TLM 127, Radial J48 DI box, NTI M2010, Neumann KU 100.
Source: Images taken from the respective manufacturer's manual.

For the room acoustic parameters measurements, calibrated NTI M2010 omnidirectional measurement microphones [18] were used, positioned across the rooms, in combination with the Outline Globe Source Radiator, an omnidirectional speaker seen in 3.4, located in several positions and emitting a sweep signal to provide a uniform sound radiation for impulse response acquisition. The recording chain was completed using the BabyFace Pro interface, along with an OctaMic Mic Preamp & DA converter [19], which can be seen in 3.5, to enable the use of several measurement microphones at the same time, and MATLAB together with ITA ToolBox as the platform for generating and recording the measurement signals, data analysis, and visualization.

Figure 3.4.: Playback equipment: Outline GRS and Neumann KH 120A.

Source: Images taken from the respective manufacturer's manual.

The artificial reverberation processors were selected from an ample number of options to represent different audio quality categories and pricing, but all with the same principle of algorithmic processing and limited to real physical room acoustic environment modeling. These were the TC Electronic Hall of Fame 2 [20] guitar pedal, the Bricasti M7 [21] digital processor, the Waves H-Reverb [22] plugin, and the Waves TrueVerb [23] plugin. The complete list of parameters specifications for each processor can be read in the Appendix B. For the process of reverb approximation with the artificial reverberation processors, the dry signals were re-recorded in the anechoic chamber with the KH 120A speaker and the NTI M2010 measurement microphone to account for the speaker's frequency response as in the real room's playback.

Figure 3.5.: Interface Babyface Pro and Preamp OctaMic II.

Source: Images taken from the respective manufacturer's manual.

3.2.2. Instruments Sources

The source material consisted of five different signal types selected to cover a broad range of spectral and temporal characteristics: acoustic guitar, electric guitar, legato singing, staccato singing, and speaking voice, as seen in Fig. 3.6. These sources were chosen to represent both sustained and transient-rich types of signals, as well as tonal and speech variations of the human voice, to ensure covering a variety of signal excitations in the room environments and reverberation modelling. All fragments were designed to be short (6 to 10 seconds) to avoid listener fatigue during the perceptual tests.

Figure 3.6.: Sound sources recorded in anechoic chamber to use as dry samples.
Source: Photos taken with permission of the msusicions.

In the same vein, the musical fragments were designed to cover a wide range of the instruments' frequency register and dynamic possibilities and were executed by experienced players to ensure a professional and consistent delivery of dynamics and articulation. For the spoken voice fragment, the paragraph was chosen for phonetic diversity, providing a balanced distribution of vowel and consonant sounds, capturing the natural rhythm of speech, and using a neutral tone and steady delivery. These approaches were important to maintain a stable spectral content and temporal envelope across all test samples.

3.2.3. Recording Enviroments

Four different recording environments were used in this study: an anechoic chamber for the initial dry sample recording, and three acoustic environments that were later to be modelled by the reverberation processors: a music studio, a concert hall, and a church. Each environment was selected to represent a distinct range of acoustic conditions, from a relatively dry, low reverberation room, a medium-sized hall with medium-long reverberation, to a highly reverberant church. When convolved with the variety of sound sources, this allowed for a large range of reverberation scenarios.

- **Anechoic Chamber**: The anechoic chamber (Fig. 3.7) provided an environment completely free of reflections, which allowed for recording the sources consistently without any influence from room acoustics. It was used for the initial "dry" source recordings as well as for the re-recordings made to capture the frequency response of the KH 120A loudspeaker. Its use ensured that all the test samples were generated from identical, uncoloured source performances and that the loudspeaker coloration was consistently present in both the real and artificial conditions.

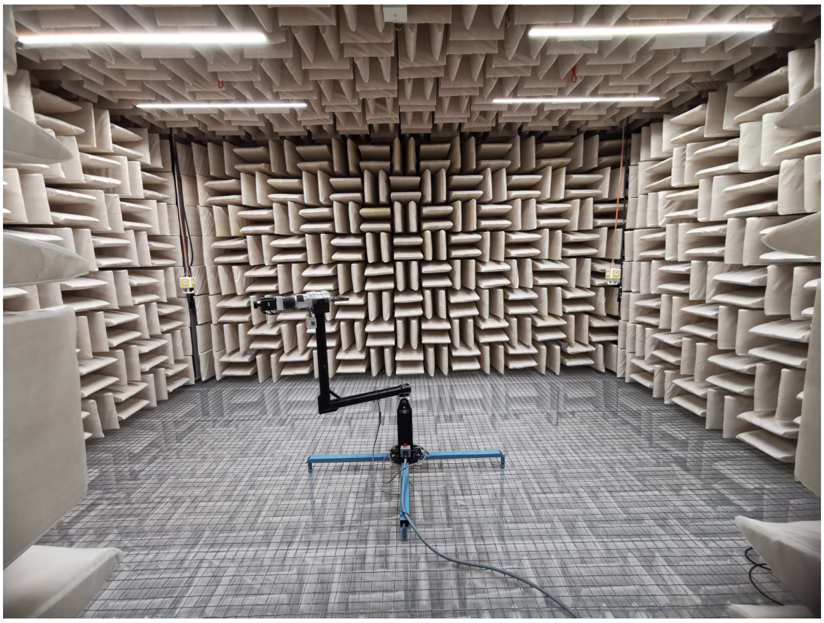

Figure 3.7.: Anechoic Chamber.

- **Studio Room**: The small room chosen for this study was the recording studio number 2 of the ETI, called Aufnahmeraum 2 and seen in Fig. 3.8, which represented a small-sized, controlled reverberation recording space, like those used in professional music production. It offers a pretty natural-sounding combination of short decay times, even shorter early reflections, and great acoustic clarity, which made it an ideal case for approximating real, smaller physical environments.

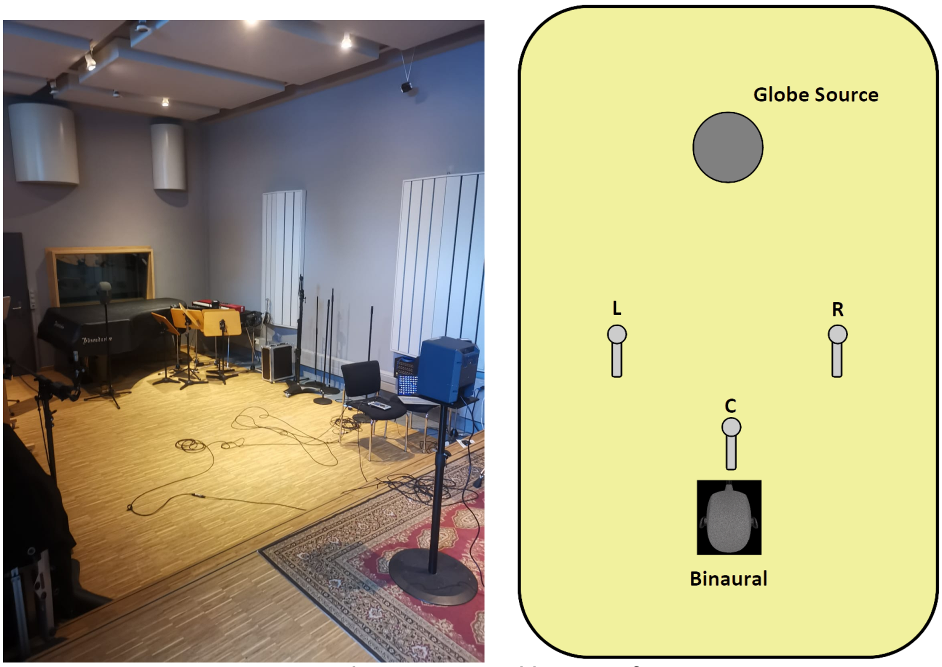

Figure 3.8.: Studio picture and layout of equipment.

- **Concert Hall**: The concert hall chosen was the Brahmssaal (Fig. 3.9) of the Musikhochschule of Detmold, a performance hall designed for classical music and normally used for chamber music and vocal recitals. Its medium-sized volume and longer decay times offer a noticeable reverberation time without being overwhelming. It provided an environment with almost an equal amount of Early Decay Time (EDT) and Reverberation Time (T30).

Figure 3.9.: Brahms picture and layout of equipment.

- **Church**: The church chosen for this project was the Erlöserkirche, seen in Fig. 3.10. It is located downtown in the marketplace of Detmold, and it served to represent the most extreme acoustic condition in the study, with a very high volume and ceiling, long reverberation tails, and strongly pronounced early reflections that even seemed to surpass the late reverberation times.

Figure 3.10.: Church photo and layout of equipment.

3.2.4. Reverberation Processors

Four different algorithmic reverberation processors were selected to model the acoustic parameters of the physical environments; these had a wide range of audio quality ranking, price points, and designs, while at the same time, all shared the same principle of using digital algorithmic processing to process the input the dry signals into reverberated outputs. This selection included an accessible multi-use guitar pedal, a high-end professional hardware processor, and two mid-range software plugins part of to the Waves Audio library in ProTools. Together, these four processors provided a diverse set of tools for generating artificial reverberation, enabling this study to achieve a wide range of room simulations. *All diagrams of the measurement setups in this section were created with the use of an AI image generator.*

- **Guitar pedal - Hall of Fame**: The first reverberation processor selected for this study was the TC Electronic Hall of Fame 2 [20], a digital multi-reverb guitar pedal designed primarily for electric guitar. It offers a range of built-in reverberation algorithms (like Room, Hall, and Church presets) and a high amount of customization and fine editing capabilities through its companion computer software called TonePrint [24], seen in Fig. 3.11, which allows access to a large amount of parameters not seen in the physical unit. To record the signals through the pedal, an IR signal generated in MATLAB was played back from the computer via an RME Babyface Pro interface, which was routed through a DI box to convert the line signal to instrument level and then entered the Hall of Fame stereo input. Then the pedal's output was sent via a second DI box and returned to the audio interface for recording. All this while the Hall of Fame was also connected via USB to the computer to allow real-time TonePrint control.

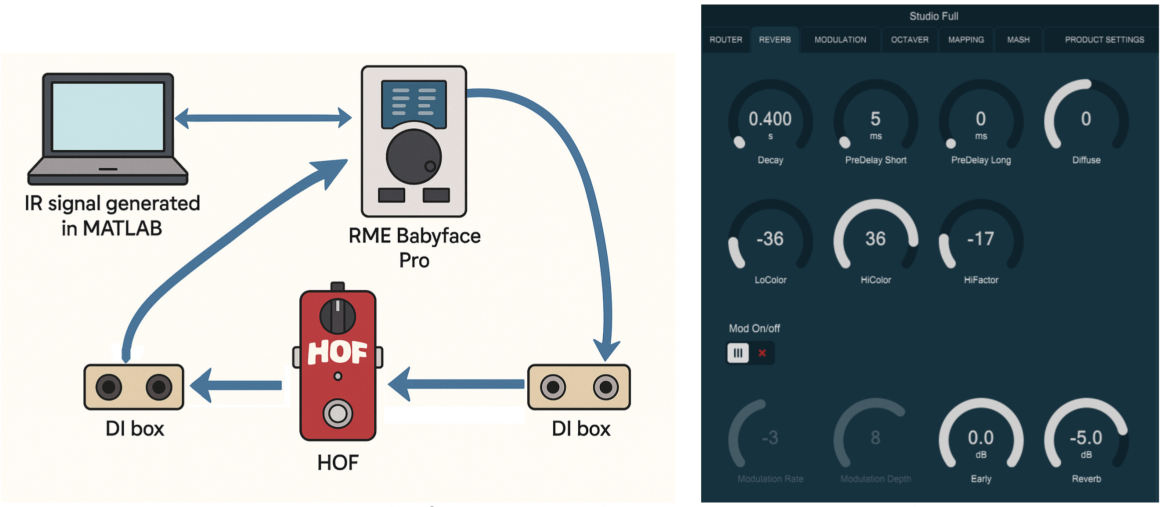

Figure 3.11.: Hall of Fame 2 Reverb measurement setup and UI.

- **Plugin - H-Reverb**: The Waves H-Reverb plugin [22] combines algorithmic and FIR-based reverb design, allowing control over decay envelopes, early reflections, and reverb equalization. It also includes several experimental shaping tools, modulation options, and time-varying filters, enabling both natural and unnatural sounding spaces. For this study, all the parameters that do not reflect real room acoustic parameters were deactivated. Its flexibility and broad parameter range make it quite interesting for many types of situations, whether these be more normal or more creative cases. The connection scheme for the impulse response measurements consisted of using a Babyface interface to output a sweep signal generated with MATLAB, which was fed into ProTools, passed through the plugin, and then returned to MATLAB again for recording, as seen in Fig. 3.12.

Figure 3.12.: H-Reverb measurement setup and UI.

- **Hardware unit - Bricasti M7**: The Bricasti M7 [21] is a high-end digital reverb hardware design for studio productions. The M7 organizes its reverb algorithms into six main categories: Halls, Rooms, Plates, Ambience, Spaces, and Chambers. Each group reflects a commonly used type of reverberation in the music and audio post-production industries. While some, like Halls and Rooms, aim to emulate real architectural spaces, others—such as Spaces and Ambience—offer more experimental or non-traditional spatial effects. Within each category, the M7 includes a wide selection of preset variations for different applications. In addition to editing reverb-specific parameters, the Bricasti M7 allows users to access a set of global system parameters that affect signal routing, level, and gain staging. These settings are not tied to individual presets but instead control how the processor interacts with connected equipment. The connection seen in 3.13 consisted of simply connecting the input and out of the processor to those of the babyface.

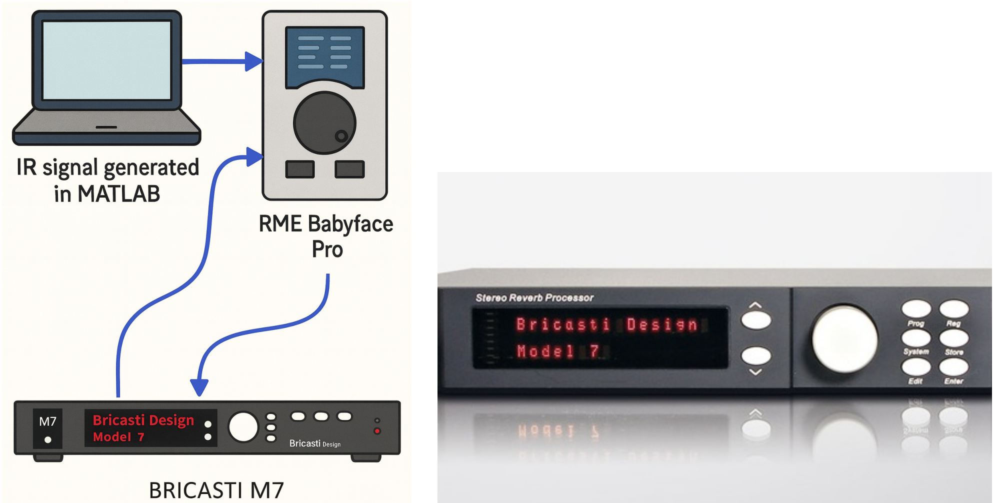

Figure 3.13.: M7 measurement setup and UI.

- **Plugin - TrueVerb**: The Waves TrueVerb plugin [23] focuses on simulating real-world spaces through a combination of direct sound and reverb and early-reflection modeling. It integrates room size, decay, and early reflection controls, allowing users to precisely shape a spatial impression that emulates real room acoustic environments. Its large flexibility makes it perfect for this study because its layout is simple and straightforward for applications requiring realistic room sound without excessive complexity. Being also a ProTools plugin, it was measured the same way as the H-Reverb. The whole list of available parameters for this and the other reverberation processors can be seen in the Appendix B.

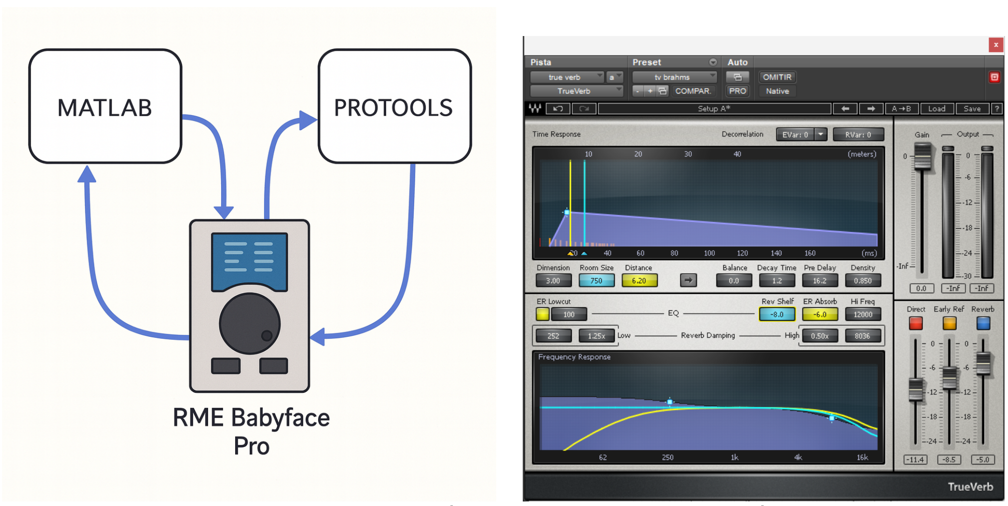

Figure 3.14.: TrueVerb measurement setup and UI.

3.2.5. Summary

The equipment, materials, and environments described in this section formed the foundation of the experimental workflow of this study. In the following sections, the process of recording the sources, measuring the room acoustic parameters, and preparing the samples for the listening tests will be discussed. Since this study covers a wide variety of combinations of Room–Instrument–Processor, with three physical environments, five sources, and four different reverberation processors, as seen in Fig. 3.1, for ease of reference, Table 3.1 was created to summarize the abbreviations or “working names” used throughout the document to identify the different rooms, instruments, and reverberation processors. The Brahmssaal presents are referred to as both "Hall" and "Brahms".

Types of Rooms

Real Environment	Working Name
Aufnahme 2	Studio
Brahmsaal	Hall / Brahms
Erlöserkirche	Church

Types of Instruments

Instrument	Working Name
Acoustic Guitar	Ac-gtr
Electric Guitar	E-gtr
Legato Singing	Legato
Speaking Voice	Speech
Staccato Singing	Staccato

Types of Reverbs

Processor Name	Working Name
TC Hall of Fame 2	HoF
Waves H-Reverb	HR
Bricasti M7	M7
Waves True Reverb	TV

Table 3.1.: Working names used for rooms, processors, and instruments

3.3. Recording Procedure

The process of obtaining the test samples consisted of the sources being recorded inside the anechoic chamber and later reproduced through the KH 120A loudspeaker [15] positioned at the center of the stage in the different room environments and re-recorded with the binaural Neumann KU 100 head [16] placed in the center of the room to obtain the "reference" samples. Then, before processing the "artificial" samples, the "dry" samples were also re-recorded through the KH 120A, but in the anechoic chamber instead, to convolve the samples with the frequency response of the loudspeaker, like in the real room environments.

To obtain the room acoustic parameters of the environments, impulse responses were measured in each of the three real acoustic spaces using the Outline Globe Source Radiator and NTI M2010 omnidirectional microphones, with microphone positions distributed throughout the spaces. These measurements provided the reference values for the reverberation time (T30), early decay time (EDT), and additional parameters such as clarity (C80), definition (D50), and center time (Ts), which were later used in the artificial reverberation matching process. For averaging the results, a single figure was obtained by averaging over the six one-third-octave bands from 400 Hz to 1.250 Hz [1].

The number of microphones per room for the acoustic measurements, and the positioning of each, varied depending on the size of the room. According to the ISO 3382-1 [1]: "Microphone positions should be at positions representative of positions where listeners would normally be located. For reverberation time measurements, it is important that the measurement positions sample the entire space." (ISO 3382-1:2009, 4.3).

All the microphones, including the binaural head, were positioned at 1.2 [m] from the floor, mimicking the height of the ears of a sitting person. At the same time, a distance of approximately 2 [m] was kept between microphones, and 1 [m] between the microphones and the walls. The sound source was positioned at 1.5 [m] from the floor and with a minimum distance of 1.5 [m] from the microphones while moving it around to obtain several takes to average. "Source positions should be located where the natural sound sources in the room would typically be located. A minimum of two source positions shall be used" (ISO 3382-1:2009, 4.3). The measurements were done without a subwoofer, only with the globe source, meaning that frequencies below 80 Hz are not accurately represented in the results. All the measurements were recorded at 24-bit / 44.1 kHz in MATLAB, and post-processing and analysis were done with the help of ITA Toolbox to extract the acoustic parameters.

3.3.1. Anechoic Chamber Recording

The anechoic chamber recordings capture under identical conditions using the appropriate microphone or DI configuration (as described in Section 3.2.1) for each one of the 5 sources. For the musical sources, performers were positioned in the chamber, facing the Neumann TLM 127 with a distance of approximately 20 [cm] as per the standard rule of thumb when making vocal recordings. For the acoustic guitar, the TLM was used again, but this time at a distance of approximately 50 [cm], pointing at the 12th fret. The electric guitar was recorded directly by line using the J48 DI box. Gain staging was performed to ensure consistent recording gain across all takes. All signals were recorded at 24-bit / 44.1 kHz into Pro Tools via the RME Babyface Pro interface.

Several takes were performed of each instrument, consisting of different variations of musical patterns and speech phrases, and at different volumes and levels of intensity. These recordings were later analyzed, and a small fragment of between 5 to 10 seconds was selected for each sound source to be the core "dry" sample for the rest of the study. These dry samples had their peak level normalized to ensure a standardized volume range when playing back the samples.

In addition to capturing the original dry performances, the anechoic chamber was used in a second set of recordings in which the selected dry signals were re-recorded, using a measurement microphone on axis at 1 [m] of the Neumann KH 120A loudspeaker. This step imprinted the loudspeaker's frequency response (Fig. 3.15) onto the signals, ensuring that playback coloration would be consistent between the "reference" and "artificial" samples.

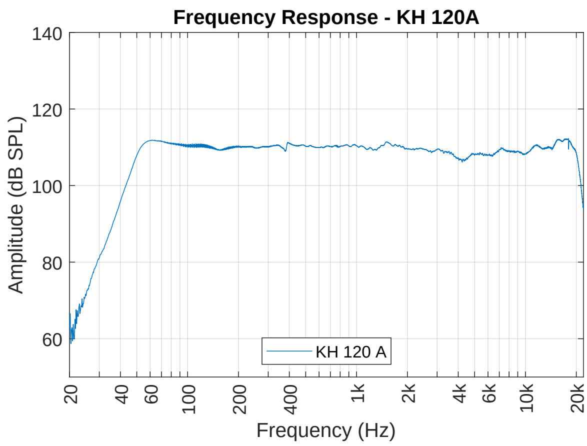

Figure 3.15.: Neumann KH 120A speaker Frequency Response.

3.3.2. Studio - Aufnahmeraum 2, ETI

The "Studio" recordings were realized in the Aufnahmeraum 2 at the Erich Thienhaus Institut (ETI). For the musical fragments re-recording, the Neumann KH 120A directional loudspeaker [15] was placed in the center of a platform located in one extreme of the room, while the Neumann KU 100 binaural head [16] was positioned at the other side of the room. For the impulse response measurements, three NTI M2010 measurement microphones [18] were distributed around the room: left mic, center mic, and right mic, as seen in Fig. 3.8, while the Globe Source was placed on the same platform as the KH 120A directional speaker. The source was moved to three positions across the platform: left stage, center stage (same position as the KH 120A speaker), and right stage. The impulse responses of the three measurement micpositions, as well as the three Globe Source positions, were later averaged to obtain a single set of room acoustic measurements

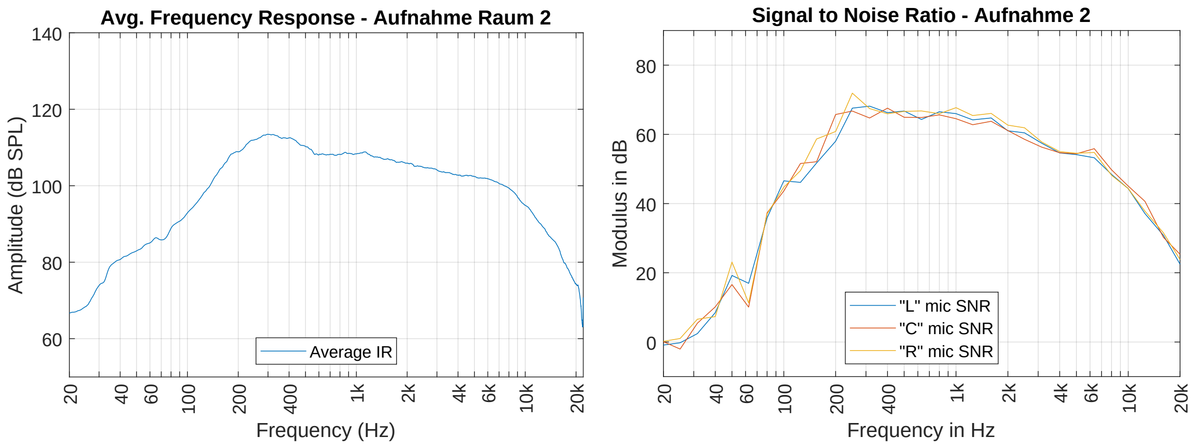

Figure 3.16.: Frequency response and Signal to Noise Ratio of Studio.

Figure 3.16 shows the averaged frequency response (FR) of the measured impulse responses in the studio. The curve exhibits a relatively flat midrange, with a low-frequency bump around 300 Hz, probably due to a mode present in the room, and a gradual decay of frequencies above 8 kHz. The signal-to-noise ratio (SNR) for the three microphone positions shows values above 60 dB across most of the spectrum, indicating a relatively low noise floor and reliable measurement conditions, in accordance with the (ISO 3382-1:2009, 5.3.2), which states: "For the determination of T30, the level of the background noise must be at least 45 dB below the maximum."

Table 3.2 summarizes the averaged room acoustic parameter values obtained between 400 Hz and 1250 Hz, with averaged values being: T30 = 0.48 [s], EDT = 0.44 [s], D50 = 80.63 [%], C80 = 11.16 [dB], Ts = 0.03 [s]. These values are the benchmark to which all the artificial processors will try to approximate. The results for the Studio show that this particular room has low reverberation, with early reflections slightly shorter, a high clarity for both musical and speech purposes, and a close environment feel with relatively good intelligibility.

Room Acoustic Paramemters - Aufnahme 2

Frequency Band	T30 (s)	EDT (s)	D50 (%)	C80 (dB)	Ts (s)
400 Hz	0.44	0.49	80.41	11.88	0.03
500 Hz	0.47	0.46	77.06	10.53	0.03
630 Hz	0.46	0.44	80.82	10.64	0.03
800 Hz	0.49	0.38	83.00	11.37	0.03
1000 Hz	0.52	0.44	81.22	11.06	0.03
1250 Hz	0.53	0.43	81.26	11.47	0.03
Average (400–1250 Hz)	**0.48**	**0.44**	**80.63**	**11.16**	**0.03**

Table 3.2.: Acoustic parameters across 1/3 octave bands for Studio.

The Reverberation Time (T30) and Early Decay Time (EDT) curves shown in Fig. 3.17 present the frequency bands from 125 Hz to 8000 Hz, leaving out of extremely low frequency values which are not present in this study due to only realizing the impulse response measurement with a globe source and not sub woofer, and the high frequency which tend to dissipate and get absorbed in the room environments. The T30 shows a steady value above 0.4 [s] until the 1 kHz mark and then jumps up above 0.6 [s] for the higher end of the curve. The EDT, on the other hand, remains around 0.44 [s] for the whole graph. Clarity (C80), Definition (D50), and Center Time (Ts) also show steady values across their frequency spectrum, with slight disturbances on the low-mid frequencies around 100 Hz to 300 Hz. D50 and C80 show that the room has a high ratio of early to late energy. The center time (Ts) is consistently short at 0.03 [s], indicating quick energy arrival and low unintelligibility.

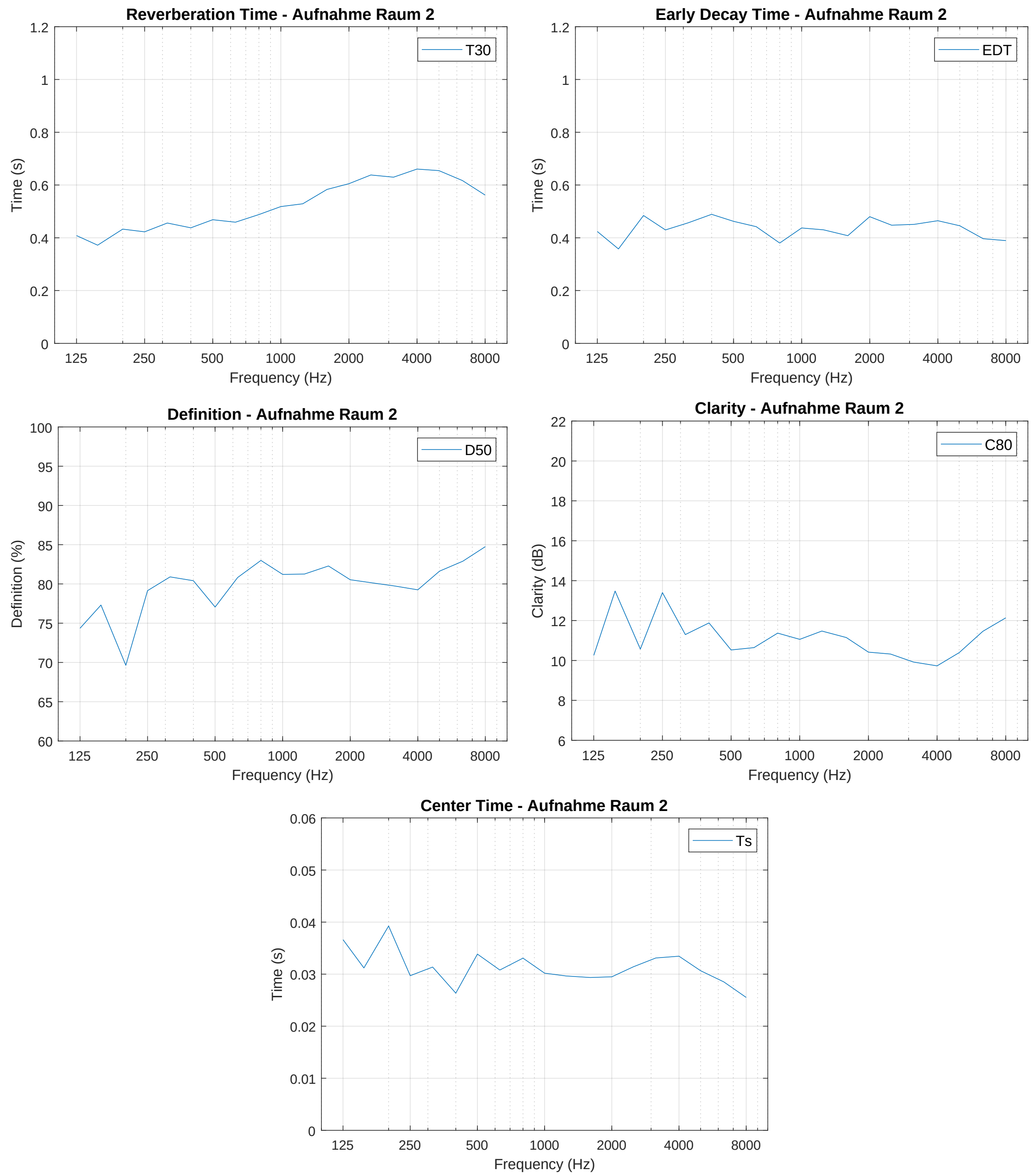

Figure 3.17.: Room Acoustic Parameters Graphs for Studio: T30, EDT, D50, C80, Ts.

3.3.3. Hall - Brahmssaal, HfM

The "Hall" recordings were realized in the Brahmssal of Hochschule für Musik Detmold, a medium-sized performance hall primarily used for chamber music and vocal recitals. The process was the same as before, first for the musical fragments re-recording, the Neumann KH 120A directional loudspeaker [15] was used, placed in the center of the stage, while the Neumann KU 100 binaural head [16] was located in the middle room at a representative listening location.

For the impulse response measurements, five NTI M2010 measurement microphones [18] were placed around the room: left mic (L), center mic (C), right mic (R), left surround mic (Ls), and right surround mic (Rs), as can be seen in Fig. 3.9. The Globe Source was placed on the stage and moved around three positions across the platform: left stage, center stage, and right stage, the center one being the same as the directional speaker. The impulse responses of these five positions, as well as the three Globe Source positions, were later averaged to obtain a single set of representative room acoustics parameters for the hall.

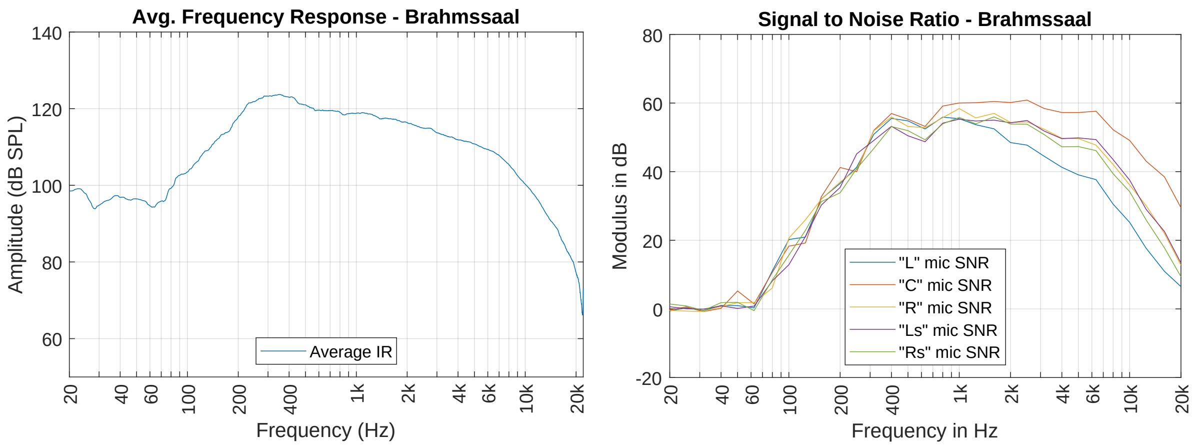

Figure 3.18.: Frequency response and Signal to Noise Ratio of Hall.

The averaged frequency response (FR) and signal-to-noise ratio (SNR) for the Brahmssaals are shown in Figure 3.18. The frequency response displays larger amplitudes for low frequencies around 300 Hz, due to probably due to a strong background noise present in the room caused by the ventilation system, which was not possible to turn off. followed by a relatively flat mid-range and a gradual decrease after 8 kHz. The SNR remains above 50 dB for most of the spectrum, which is slightly above the recommended minimum value for room acoustic measurements [1]. We can see, however, that for higher frequencies the SNR starts to vary a lot between microphones, once again probably due to the constant ventilation system background noise and the distance of its air outlets to the microphones.

Room Acoustic Paramemters - Brahmssaal

Frequency Band	T30 (s)	EDT (s)	D50 (%)	C80 (dB)	Ts (s)
400 Hz	1.38	1.39	32.21	0.68	0.11
500 Hz	1.28	1.43	35.68	0.09	0.11
630 Hz	1.25	1.21	40.14	1.00	0.10
800 Hz	1.22	1.17	38.25	1.02	0.09
1000 Hz	1.14	1.19	43.32	1.70	0.09
1250 Hz	1.11	1.07	45.29	2.46	0.08
Average (400–1250 Hz)	**1.23**	**1.24**	**39.15**	**1.16**	**0.09**

Table 3.3.: Acoustic parameters across 1/3 octave bands for Hall.

The averaged room acoustic parameter values are: T30 = 1.23 [s], EDT = 1.24 [s], D50 = 39.15 [%], C80 = 1.16 [dB], Ts = 0.09 [s]. Compared to the studio, these results show a significantly longer reverberation time and reduced definition, which is expected for larger performance spaces intended for live music rather than music recording. The Reverberation Time (T30) and Early Decay Time (EDT) curves shown in Fig. 3.19 cover frequency range from 125 Hz to 8 kHz and shows that both parameters are almost identically matched, both having higher values at low frequencies (approximately 1.5 [s] around 200 [Hz]) and decreases constantly toward the higher end of the spectrum.

The Clarity (C80) and Definition (D50) graphs show a constant increase in values above 250 [Hz] onward, basically opposite to the T30 and EDT, which means that with decreased reverb times, the clarity gets stronger, indicating improved articulation in the upper range. The Center Time (Ts) values also reflect a more spacious and reverberant listening environment. The averaged values from Table 3.3 serve as the target optimal values for the artificial reverberation approximation in the “hall” category of artificial reverberation processors.

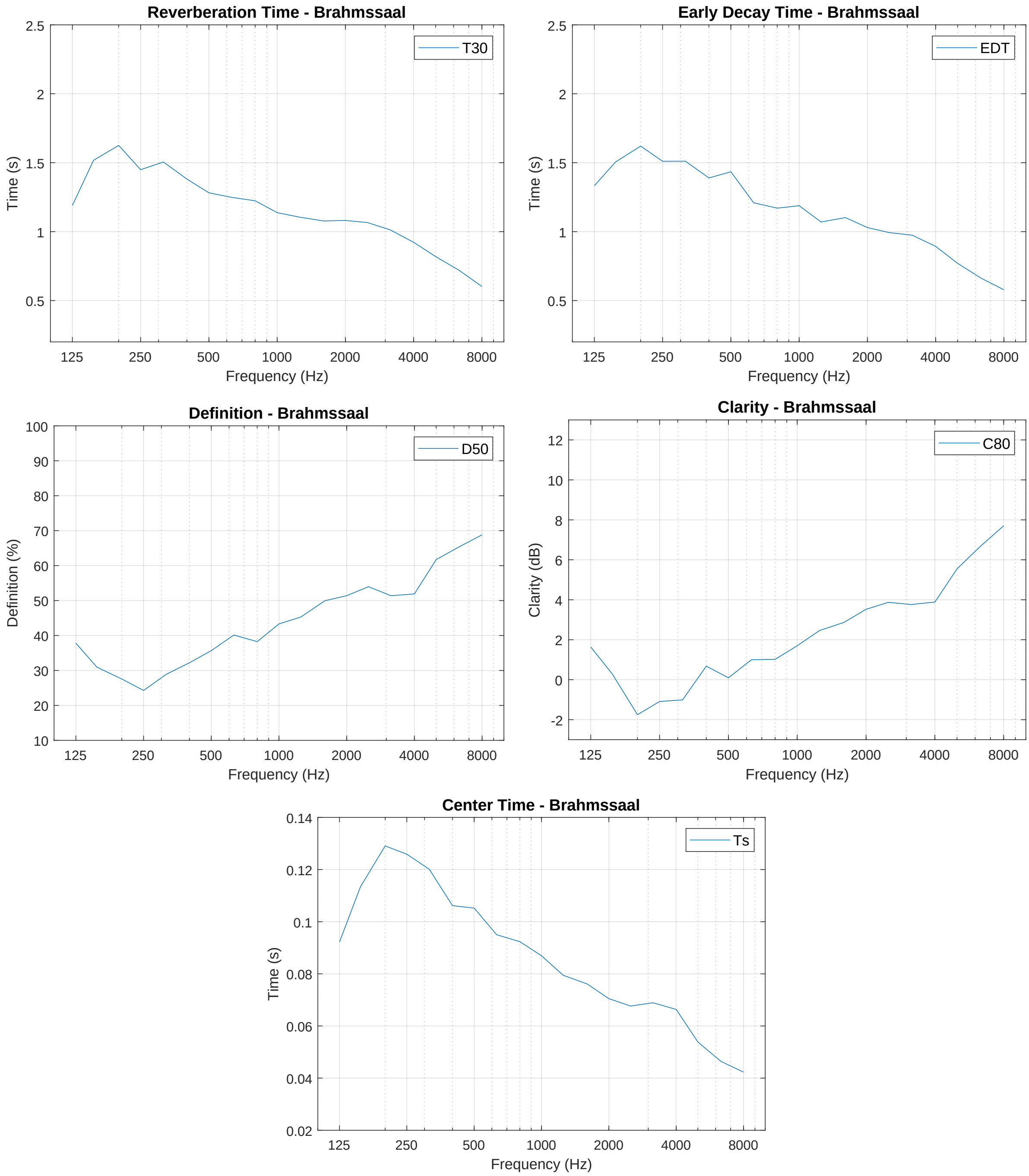

Figure 3.19.: Room Acoustic Parameters Graphs for Hall: T30, EDT, D50, C80, Ts.

3.3.4. Church - Erlöserkirche, Detmold

The "Church" recordings were done in the Erlöserkirche in the city center of Detmold. This large space has a high ceiling and highly reflective walls, and represents the most extreme acoustic condition in this investigation. Once again, the process of acquiring the room musical fragments consisted of placing the Neumann KH 120A directional loudspeaker [15] on the front of the church, close to the pulpit, while the Neumann KU 100 binaural head [16] was located in the center of the church's audience room. For the impulse response measurements, this time six NTI M2010 measurement microphones [18] were placed around the room: left mic (L), center mic (C), right mic (R), left surround mic (Ls), center surround mic (Cs), and right surround mic (Rs), as seen in Fig. 3.10. The Globe Source was once again placed on the stage and moved around three positions across the platform, the center one being the same as the directional speaker. The impulse responses of these six positions, as well as the three Globe Source positions, were later averaged to obtain a single set of room acoustic measurements.

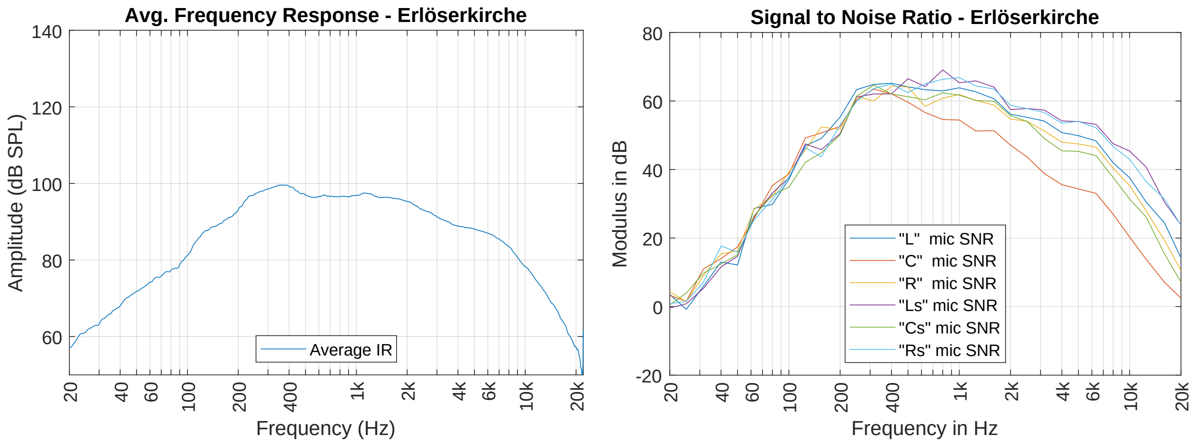

Figure 3.20.: Frequency response and Signal to Noise Ratio of Church.

The averaged frequency response (FR) shows a fairly steady amplitude in the mid-frequency range, with reduced energy in the low and high extremes of the spectrum. The signal-to-noise ratio (SNR) values are around 60 dB across most microphone positions, with lower values above 6 kHz, as shown in Fig. 3.20. It has to be noted that the average amplitude of the frequency response is around 20 [dB] lower than that of the Brahmssaal 3.18 but still achieves similar SNR values, meaning that the background noise of this environment was lower.

Room Acoustic Paramemters - Erlöserkirche

Frequency Band	**T30 (s)**	**EDT (s)**	**D50 (%)**	**C80 (dB)**	**Ts (s)**
400 Hz	1.96	2.10	27.66	-1.76	0.15
500 Hz	1.97	2.09	31.00	-1.72	0.15
630 Hz	1.98	2.12	30.33	-1.43	0.15
800 Hz	2.07	2.29	27.99	-2.20	0.16
1000 Hz	2.23	2.30	31.07	-1.79	0.16
1250 Hz	2.29	2.32	28.48	-2.08	0.16
Average (400–1250 Hz)	**2.08**	**2.20**	**29.42**	**-1.83**	**0.16**

Table 3.4.: Church Room Acoustic parameters

Table 3.4 details the averaged room acoustic parameters for this room: T30 = 4.89 [s], EDT = 4.15 [s], D50 = 17.42 [%], C80 = –5.84 [dB], and Ts = 0.19 [s]. These results clearly mark the church as the most reverberant environment in the study, with strong late reverberation and early reflections, and consequently, really low definition and clarity values. The T30 and EDT curves shown in Fig. 3.21 show consistently high reverb times across the frequency range, peaking at over 2.2 seconds in the mid frequencies around 1k [Hz] and around 1.8 [dB] in the low frequencies range. However, for frequencies above 4k [Hz] the reverb times suddenly decrease. In general, the EDT values are slightly above the T30, meaning that the early reflections in the room are a little bit stronger than the reverberation tail.

The Definition (D50) and Clarity (C80) graphs confirm the strong late reverberation energy of the room, with a poor early to late energy ratio. The D50 values are all below 35 [%] except above 4k [Hz] (which is the same point at which the reverb decays for T30 and EDT). In the same vein, C80 is consistently negative across most of the spectrum, only surpassing the 0 [dB] threshold after 4k [Hz]. Center Time (Ts) values average around 0.15 s, more than double that of the hall and nearly six times longer than in the studio, reflecting the much greater perceived distance to the sound source. All these values served as the optimal values that the artificial reverberation processors tried to match for the “church” category in the next section.

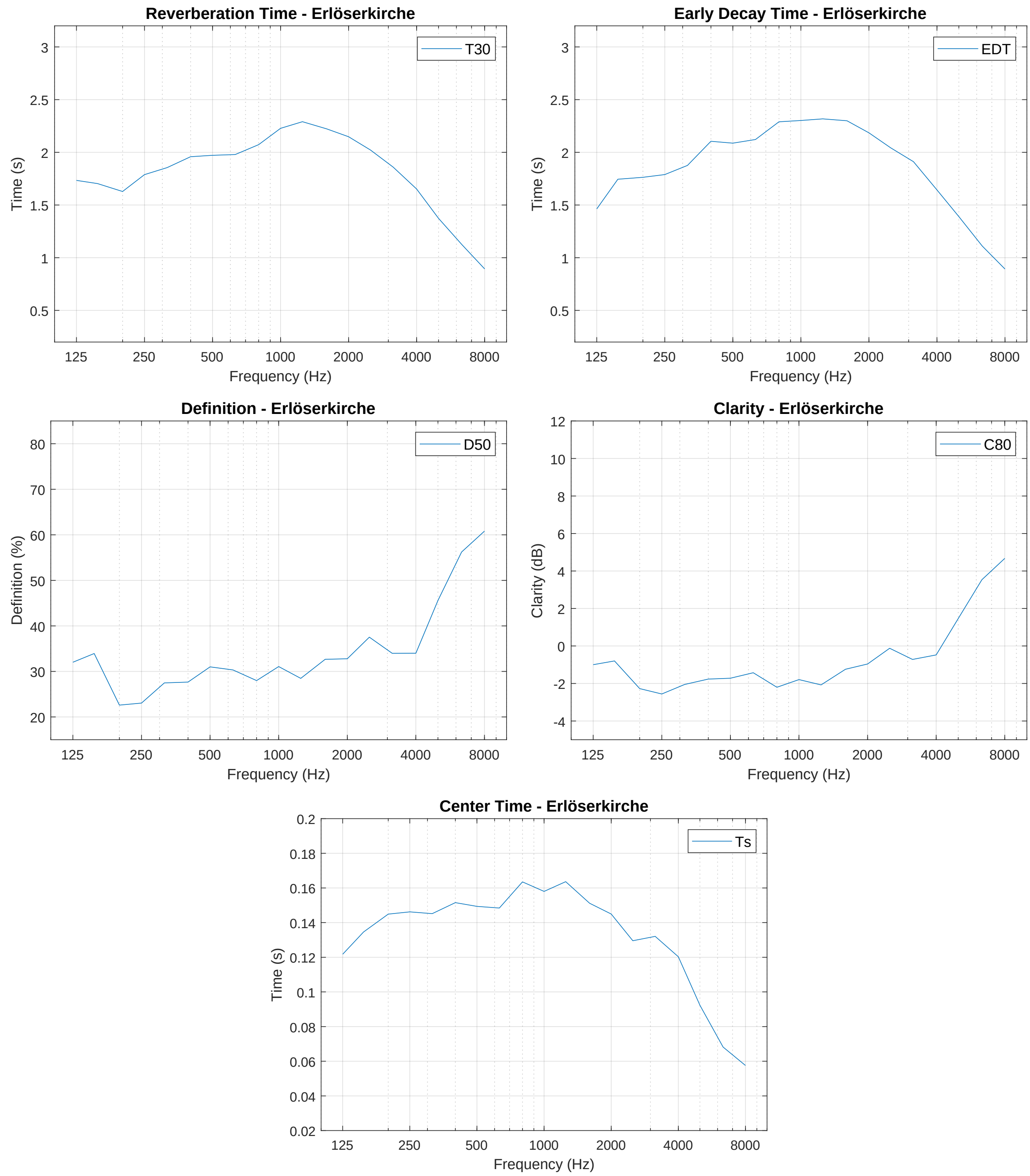

Figure 3.21.: Room Acoustic Parameters Graphs for Church: T30, EDT, D50, C80, Ts.

3.3.5. Real signals processing

Following the recording procedures described in this section, the acoustic measurements underwent a post-processing work to obtain the averaged impulse responses for each environment. For every room, all individual impulse responses were first averaged across the repeated takes of all Globe Source positions, giving the average result across takes for each microphone position. These averages were then combined across all microphone positions to obtain an overall single average for the room that covers the listener's area in multiple positions [1]. A compilation table of all measured parameters for every room can be seen in Table 3.5.

Average Room Acoustic Parameters - All Rooms

Room Name	T30 (s)	EDT (s)	D50 (%)	C80 (dB)	Ts (s)
Aufnahme 2	0.48	0.44	80.63	11.1	0.03
Brahmssaal	1.23	1.24	39.15	1.16	0.09
Erlöserkirche	2.08	2.20	29.42	-1.83	0.16

Table 3.5.: Average Room Acoustic Parameters for every type of room.

As well, with the reverberation time average of each room plus its volume, each room can be positioned within the range of standard reverberation times relative to room size, in accordance with the ISO 23591 [11], which defines the standard expected reverberation times depending on the room's size and uses.

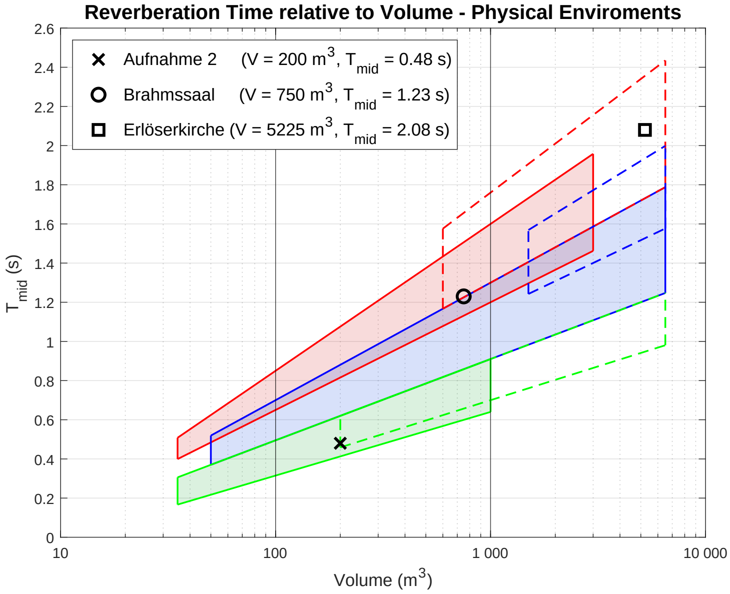

Figure 3.22.: T30 relative to room size for every physical environment.

For the musical fragment recordings reproduced through the directional loudspeaker, several takes were performed with varying orientations of the acoustic head and the speaker. After a comparative perceptual evaluation, the central head orientation (meaning the speaker in the center of the stage and the head in the center of the room) was selected for all real musical samples, as it provided the most balanced and "average" representation of the acoustic environments.

Figure 3.23.: Noise removal plugin, Waves X-Noise.

A problem that arose while analyzing the samples was that the signals captured in the Brahmssaal (Hall) contained excessive background noise, due to a constant noise from the ventilation system, which could not be switched off during the recordings. To mitigate the impact of this on the perception of musical fragments, a high-pass filter was applied at 120 [Hz] to attenuate low-frequency rumble, and a noise reduction plugin was used to clean the signals. This was only done to the musical samples and not to the room acoustic parameter measurements. The plugin used for the noise reduction process was the Waves X-Noise, a broadband noise reduction processor based on frequency domain analysis [25]. The plugin operates by analyzing a short segment of the audio that contains only the unwanted noise, creating a spectral noise profile that serves as the reference for subtracting noise content from the full signal. As noted in the users' manual: "X-Noise is ideal for removing background noise caused by tape hiss and air conditioner/ventilation systems" (Waves Audio, X-Noise User Guide, p. 2), which meant it was perfectly tailored to solve this problem.

3.4. Artificial Reverberation Processors

This section describes the process of utilizing the artificial reverberation processors described in section 3.2.1 to emulate the acoustic environments of the different selected rooms, having as reference the measurements and musical recordings done in the last section 3.3.Each processor was configured to match the measured parameters of the real rooms as closely as possible, with special care to match the Reverberation Time *(T30)* and Early Decay Time *(EDT)*, and checking the clarity and center times measurements as secondary references. The goal was to create reference presets of each room that could later be systematically modified for perceptual evaluation.

It is important to note that in most algorithmic reverb processors, modifying one control, such as "Decay Time", that would usually be associated with the *RT60* parameter, probably will affect other parameters like Clarity or EDT. This all depends on the internal architecture of the algorithm. This interaction is common but not universal, and it presented a significant challenge during the matching process. Ensuring that changes in one parameter did not unintentionally alter others required constant iterative measurement and fine-tuning. The following subsections detail the specific processors used: the TC Electronic Hall of Fame 2, the Waves H-Reverb, the Bricasti M7, and the Waves TrueVerb, along with their configurations, strategies, and limitations encountered.

3.4.1. Hall of Fame 2

The Hall of Fame 2 offers a wide range of preset reverb types (e.g., Hall, Church, Plate), which can be modified via three physical knobs: Decay, Tone, and Level [20], as well as a Pre-Delay switch. However, it is through the use of the TonePrint Editor that users can access deeper control over a multitude of hidden parameters, including high and low frequency equalization (through the "HiColor" and "LoColor" controls), access to pre-delay fine tuning ("PreDelay" control), early reflection modifications ("Early" control), and even more obscure parameters like diffusion, modulation, octavator, damping, and more [24]. These parameters were largely left untouched. A whole list of the available parameters for the TC Electronics Hall of Fame 2 Reverb pedal can be seen in the Appendix B and the physical setup specifications and the global controls, meaning the controls that remain constant for the processor across all Presets, are defined in table 3.6.

Global Settings - HoF

Parameter	Value
In Level	0 dB
Octaver	-100 dB
Kill Dry	On
Modulation	Off
Diffuse	0 dB

Physical Setup - HoF

Parameter	Routing	Value
DAW Output	Mono	-12 dB
1st DI	Mono/Stereo	-20 dB PAD
2nd DI	Stereo/Stereo	Max Gain
DAW Input	Stereo	+25 dB

Table 3.6.: Hall of Fame 2 Reverb general settings

For this processor, three custom presets were created to simulate the acoustic characteristics of the real environments: the Erlöserkirche church, the Brahmssaal concert hall, and the Aufnahme 2 recording studio. These presets were constructed based on the room acoustic parameters of the real environments obtained in the previous section 3.3, with specific focus on Reverberation Time (RT60) and Early Decay Time (EDT). While some controls of the processor are analogous to their real-world room acoustic parameter counterparts, like the "Decay" control in the pedal being basically the value of the RT60, other room acoustic parameters required deeper adjustments to mimic the spectral envelope observed in the real room IRs. To enable objective approximation between the real and artificial reverberation responses, a setup was designed where impulse responses were passed through the Hall of Fame 2 pedal and measured for analysis and comparison as seen in Fig. 3.11.

Before recording the processed IRs, a level check and calibration process was done with the pedal in bypass mode, so the gain of both the pedal and the computer was optimal. Unity gain was achieved through a gain stage where the output level of the interface was -12 dB, then a -20 dB pad on the first DI box, maximum input level on the pedal, full output level on the second DI box, and 25 dB of gain on the interface input. As well, a calibration step was performed, which minimized latency from 2114 samples (approximately 48 ms at 44.1 kHz) to 1 sample.

The creation of the Presets began with getting familiarized with the Manual for the pedal and identifying similarities between real acoustic parameters and the software's internal labels. One notable challenge was the pedal's internal signal routing: it splits the input into a dry and a processed path, but the dry signal has no independent level control. This behavior complicated the ability to maintain a dry/wet balance. The solution was to engage the pedal's Kill Dry, which removes the direct path entirely, leaving only the processed signal, adding a copy of the dry signal through a duplicated channel in ProTools, and mixing them. The final values of the controls for each preset can be seen in the next table 3.7.

Studio Preset - HoF

Parameter	Value
Fx Level	-16 dB
Decay	0.40 s
Pre Delay	5 ms
LoColor	-36
HiColor	+36
HiFactor	-17
Early	0 dB
Reverb	-5.0 dB
Direct Signal	-6.2 dB

Hall Preset - HoF

Parameter	Value
Fx Level	-19 dB
Decay	1.20 s
Pre Delay	8 ms
LoColor	+4
HiColor	-26
HiFactor	0
Early	0 dB
Reverb	-4.5 dB
Direct Signal	-10.2 dB

Church Preset - HoF

Parameter	Value
Fx Level	-21 dB
Decay	2.09 s
Pre Delay	13 ms
LoColor	-28
HiColor	+5
HiFactor	0
Early	0 dB
Reverb	-8.0 dB
Direct Signal	-14.0 dB

Table 3.7.: Hall of Fame 2 Reverb Presets

For every type of room preset, the T30 was fairly well approximated to the real room parameter. For this, besides the "Decay" control, the equalization controls "LoColor" and "HiColor" were key to shaping the processors' time over frequency curve into the desired one. For EDT, however, things were a little more difficult; The pedal had problems managing to emulate really short early reflection times, which, as can be seen in the "EDT - Studio Present - HoF" graph, made it impossible to reduce the time of the response below 0.7 [s], and for the Brahms Preset, the low end of the spectrum was closely matched, but the higher end shows larger values than the real Brahmssaal.

Finally, a table 3.8 was compiled with all the room acoustic parameter values of the obtained Preset. These values are now treated as the "optimal" values for each combination of Room and Processor. The Reverberation Time (T30) and Early Decay Time (EDT) curves of each preset can be seen in Figure 3.24

Average Values - HoF Presets

Preset Name	T30 (s)	EDT (s)	D50 (%)	C80 (dB)	Ts (s)
HoF "Studio"	0.55	0.83	65.12	5.23	0.05
HoF "Hall"	1.25	1.44	49.72	1.19	0.09
HoF "Church"	1.99	2.04	39.62	-0.92	0.14

Table 3.8.: Room Acoustic Parameters average values for the Hall of Fame Reverb Presets

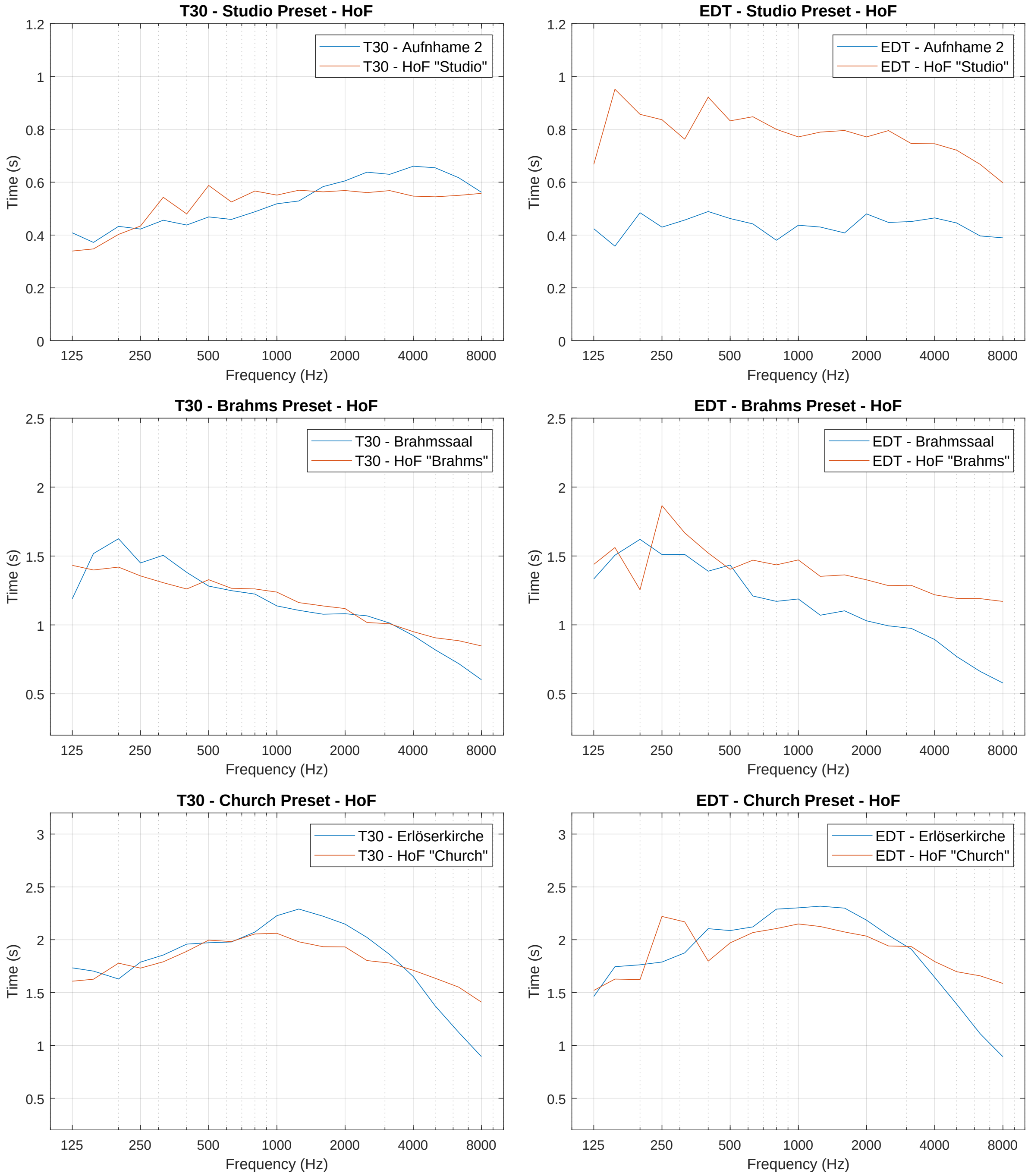

Figure 3.24.: T30 and EDT for every Preset created with the Hall of Fame 2 pedal.

3.4.2. Waves H-Reverb

The Waves H-Reverb is a Finite Impulse Response (FIR) algorithmic reverb designed to merge the flexibility of synthetic reverberation with the sonic smoothness of convolution-based approaches [22]. Its architecture offers more precise control over the decay envelope, enabling deeper shaping of the time frequency response. Its interface is divided into two main views: a collapsed mode with only the essential controls, and an expanded mode exposing a broad range of advanced parameters such as multi-band EQ, dynamics processing, modulation, and resonant time filtering, most of which are not used in this project. To develop the three custom Presets to match the acoustic characteristics of the three real environments measured in Section 3.3, a particular focus on Reverberation Time (T30) and Early Decay Time (EDT), using the processor "Reverb Time" control as a direct link to T30, and adjusting complementary control (such as "Buildup Time", "ER/Tail Balance", and the" multi-band EQ sections"), to approximate the early reflections emulation.

Global Settings - HR

Parameter	Value
Reverb Size	1.00
Reverse	Off
Density	100%
Input Echoes	Off
Output Echoes	Off
Dynamics	Off
Damping	Off
Resonant Filter	Off
Modulation	Off
Drive	0%
Analog	Off
Digital	Off

Studio Preset - HR

Parameter	Value
Pre-Delay	3 ms
Buildup Time	17 ms
Reverb Time	0.49 s
ER Select	9
ER/Tail Balance	85.0%
Dry/Wet	81.1%
Output	-6.0 dB
X-Time	75.1%
X-Gain	-51.5 dB
ER Filter	10.1 KHz
Gain	-14.2 dB
LoShelf EQ	80 Hz
Gain	-6.8 dB
LoBell EQ	Off
HiBell EQ	4.1 kHz
Gain	+2.6 dB
Q	0.46
HiShelf EQ	10.2 kHz
Gain	-14.0 dB

Hall Preset - HR

Parameter	Value
Pre-Delay	9 ms
Buildup Time	31 ms
Reverb Time	1.15 s
ER Select	3
ER/Tail Balance	55.2%
Dry/Wet	90.0%
Output	-8.7 dB
X-Time	92.3%
X-Gain	-64.0 dB
ER Filter	8.6 kHz
Gain	-14.2 dB
LoShelf EQ	90 Hz
Gain	-17.7 dB
LoBell EQ	265 Hz
Gain	+6.0 dB
Q	1.33
HiBell EQ	6.7 kHz
Gain	-5.6 dB
Q	0.46
HiShelf EQ	7.2 kHz
Gain	-19.2 dB

Church Preset - HR

Parameter	Value
Pre-Delay	0 ms
Buildup Time	32 ms
Reverb Time	2.05 s
ER Select	7
ER/Tail Balance	58.0%
Dry/Wet	65.0%
Output	-9.0 dB
X-Time	80.0%
X-Gain	-46.0 dB
ER Filter	10.5 KHz
Gain	-80.0 dB
LoShelf EQ	130 Hz
Gain	-25.0 dB
LoBell EQ	Off
HiBell EQ	6.7 kHz
Gain	-5.6 dB
Q	0.46
HiShelf EQ	7.7 kHz
Gain	-40.2 dB

Table 3.9.: H-Reverb Presets

The global settings and fixed controls for the H-Reverb used in all presets are shown in Table 3.9 together with control values for each individual Preset. The design process involved iterative listening and measurement cycles, using swept-sine excitation through the plugin to verify that the frequency decay curves followed the target curves of the corresponding real rooms. One advantage of the H-Reverb over the Hall of Fame 2 pedal was the availability of independent Dry/Wet control and fully parametric EQ sections for both early reflections and the reverb tail.

For the Studio preset, T30 was closely matched across the spectrum, while EDT remained consistently higher than the real room, particularly in the low–mid range. The Brahms preset achieved a good approximation in the mid-frequency range for both T30 and EDT, though the plugin tended to produce slightly shorter early decay in the low frequencies. The Church preset demonstrated the largest deviations, with T30 and EDT values generally exceeding those of the real Erlöserkirche, particularly above 2 kHz, indicating limitations in the plugin's difficulties to reduce the high-frequency decay without also reducing overall reverberation time.

Average Values - HR Presets

Preset Name	T30 (s)	EDT (s)	D50 (%)	C80 (dB)	Ts (s)
HR "Studio"	0.46	0.67	56.73	6.38	0.05
HR "Hall"	1.13	1.01	43.14	1.89	0.08
HR "Church"	2.40	1.74	52.92	2.40	0.09

Table 3.10.: Room Acoustic Parameters average values for the H-Reverb Presets

The resulting T30 and EDT measurements for the three presets are presented in Figure 3.25 and the average values for all room acoustic parameters derived from the processed IRs are summarized in Table 3.10.

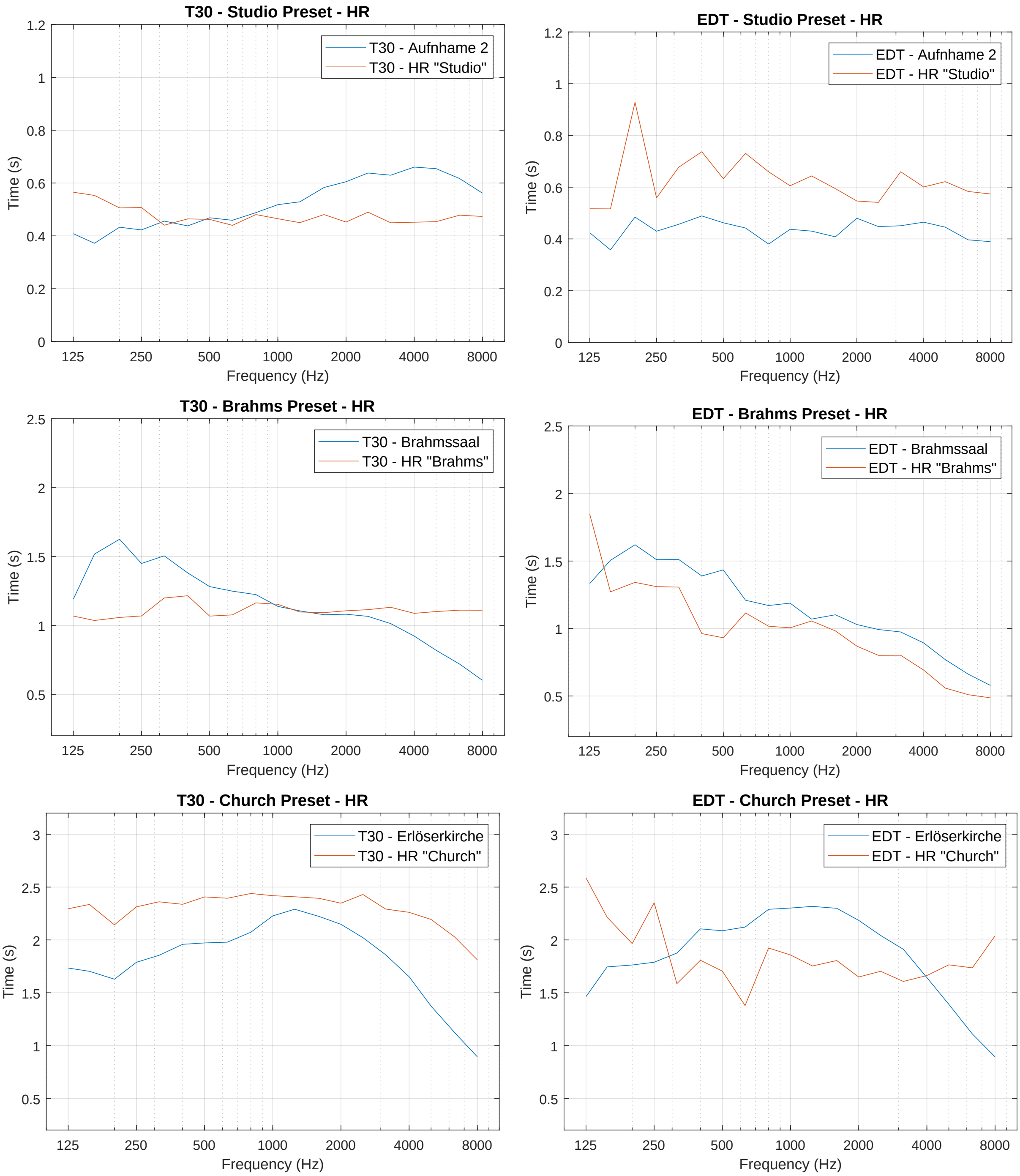

Figure 3.25.: T30 and EDT for every Presets created with the H-Reverb plugin.

3.4.3. Bricasti M7

The Bricasti M7 is a high-end reverberation hardware unit [21] used in professional music studios. Its main controls include "Reverb Time" and "Early/Reverb Mix", several early reflection modification options, along with frequency shaping controls for both high and low frequencies. An important key feature of the M7 is that its internal algorithm is constantly processing the passing signals, which results in slightly different results every time, resulting in varying outcomes for every take, which is similar to natural physical reverberation.

Global Settings - M7

Parameter	Value
Dry Gain	Off
Wet Gain	0 dB
Audio Routing	Mono/Stereo
Modulation	Off
Delay	Off

Physical Setup - M7

Parameter	Routing	Value
DAW Output	Mono	-12 dB
Processor Input	Mono	+5 dB
Processor Output	Stereo	+8 dB
DAW Input	Stereo	+5 dB

Table 3.11.: M7 Reverb general settings

The presets on the Bricasty M7 were designed by selecting three presets from the M7 (Rooms, Halls, and Spaces) and adjusting its available controls to approximate the measured Reverberation Time (T30) and Early Decay Time (EDT) of each real room. The controls on the M7 are accessed through a small display screen on the front of the device and manipulated via a series of buttons. The whole list of available parameters can be found in Appendix B. In the table 3.11, the global settings of the processor as well as its physical setup values are defined. For the presets, besides the main two controls of "Reverb Time and "Early/Reverb Mix", additional controls like frequency "Multiply" and crossovers were used to shape the frequency decay. The detailed control values for each individual preset are shown in Table 3.12.

Studio Preset - M7

Parameter	Value
Reverb Time	0.45 s
Size	12
Pre-delay	4 ms
Diffusion	8
Density	2
Rolloff	13.2 kHz
HF RT Multiply	1.00 x
HF RT Crossover	12 kHz
LF RT Multiply	0.35 x
LF RT Crossover	160 Hz
VLF Cut	-10 dB
Early/Reverb Mix	20 // 19
Early Rolloff	12 kHz
Early Select	16
Master Gain	+5.0 dB

Hall Preset - M7

Parameter	Value
Reverb Time	1.15 s
Size	18
Pre-delay	6 ms
Diffusion	5
Density	5
Rolloff	11.6 kHz
HF RT Multiply	0.45 x
HF RT Crossover	8 kHz
LF RT Multiply	1.25 x
LF RT Crossover	200 Hz
VLF Cut	0 dB
Early/Reverb Mix	20 // 15
Early Rolloff	11.6 kHz
Early Select	20
Master Gain	+2.8 dB

Church Preset - M7

Parameter	Value
Reverb Time	2.05 s
Size	Small
Pre-delay	12 ms
Diffusion	6
Density	8
Rolloff	12k Hz
HF RT Multiply	0.50 x
HF RT Crossover	8k Hz
LF RT Multiply	0.45 x
LF RT Crossover	240 Hz
VLF Cut	-20 dB
Early/Reverb Mix	20 // 12
Early Rolloff	12 kHz
Early Select	10
Master Gain	+0.0 dB

Table 3.12.: M7 Reverb Presets

As with the previous processors, the calibration process included sending swept-sine excitations through the unit and recording the resulting impulse responses. Latency was corrected from 2155 to 1 sample, which is a reduction of almost 50ms. and the dry path was disabled so that only the processed reverberation can be measured. For the Studio preset, the processor was able to closely match both T30 and EDT across the spectrum, though with slightly lower values in the high-frequency range. The

Brahms preset also followed the target curves with good accuracy, especially in the mid frequencies, although it tended to have shorter times at low frequencies. The Church preset was the most accurate of the three, with both T30 and EDT curves matching the measured Erlöserkirche values across the whole spectrum. The average values of the room acoustic parameters are summarized in Table 3.13 and the T30 and EDT curves for the three M7 presets are presented in Figure 3.26.

Average Values - M7 Presets

Preset Name	T30 (s)	T30 (s)	D50 (%)	C80 (dB)	Ts (s)
M7 "Studio"	0.48	0.58	65.94	7.46	0.05
M7 "Hall"	1.20	1.06	62.00	4.09	0.07
M7 "Church"	2.01	1.96	52.50	1.06	0.12

Table 3.13.: Room Acoustic Parameters average values for the M7 Reverb Presets

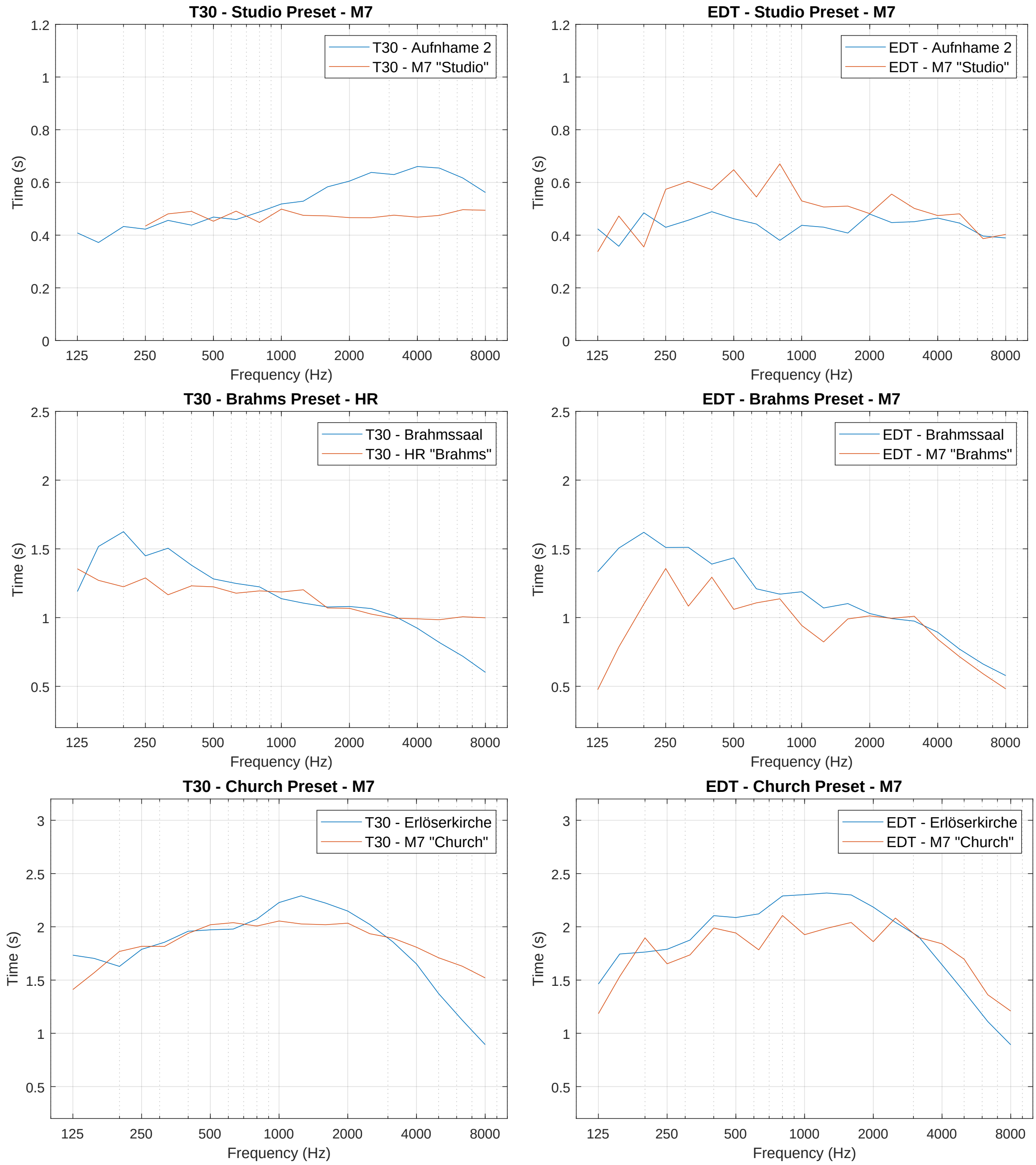

Figure 3.26.: T30 and EDT for every Preset created with the Bricasti M7 hardware.

3.4.4. Waves TrueVerb

Waves TrueVerb is an algorithmic reverberation plugin from the Waves Audio [23] that combines direct sound, early reflections, and reverberation tail to create its outcome. It has practical controls that align well with physical room acoustic parameters: "Size", "Distance", "Decay", "Pre-Delay", and more parameters defined in the Appendix B that serve to modify the equalization of the reverberation decay. It also counts with independent output levels for: Direct Sound", "Early Reflection", and "Reverberation". The three custom Presets built to approximate the measured acoustic behavior of the Aufnahme 2, Brahmssaal, and Erlöserkirche were created by using the "Decay" control as the main link to reverberation time, while the other parameters were used to shape the early reflections perception.

Global Settings - TV

Parameter	Value
Dimension	3.00
Link	Off
Density	0.850
Input Gain	0 dB
EVar	0
RVar	0

Studio Preset - TV

Parameter	Value
Room Size	200 m^3
Distance	4.60 m
Balance	0.0
Decay Time	0.5 s
Pre-Delay	10.5 ms
ER Lowcut	Off
ER Absorb Gain	12 kHz -6.0 dB
Rev Shelf Gain	12 kHz -10.6 dB
Low Damping	Off
High Damping Ratio	4.8k Hz 1.12x
Direct	-4.3 dB
EarlyRef	-8.4 dB
Reverb	-6.9 dB
Master Gain	+3.0 dB

Hall Preset - TV

Parameter	Value
Room Size	750 m^3
Distance	6.20 m
Balance	0.0
Decay Time	1.2 s
Pre-Delay	16.2 ms
ER Lowcut	100 Hz
ER Absorb Gain	12 kHz -6.0 dB
Rev Shelf Gain	12 kHz -8.0 dB
Low Damping Ratio	252 Hz 1.25x
High Damping Ratio	8.0k Hz 0.50x
Direct	-11.4 dB
EarlyRef	-8.5 dB
Reverb	-5.0 dB
Master Gain	+1.7 dB

Church Preset - TV

Parameter	Value
Room Size	5200 m^3
Distance	6.15 m
Balance	0.0
Decay Time	1.9 s
Pre-Delay	18.0 ms
ER Lowcut	100 Hz
ER Absorb Gain	10k Hz -6.0 dB
Rev Shelf Gain	10k Hz -8.3 dB
Low Damping Ratio	245 Hz 0.62x
High Damping Ratio	7.4k Hz 0.12x
Direct	-11.9 dB
EarlyRef	-13.6 dB
Reverb	-10.7 dB
Master Gain	+3.2 dB

Table 3.14.: TrueVerb Presets

Global settings were kept constant across all Presets: Dimension = 3.00 was recommended in the manual [23] as the best value to emulate 3D spaces, Link = Off was done to be able to manually control the pre delay time, and Density = 0.850 was also a recommendation of the manual as the most common value for the plugin. All these values, along with the Preset values, are shown in Table 3.14, showing a clear increase in the Room Size control, longer Decay Time, and higher Pre-Delay times from the studio preset to the Brahms and church ones.

The design process followed the same loop used with the other processors: sweep excitation through the plugin was measured to process the IR, followed by an iterative adjustment until the frequency decay curves approximated the real room acoustic parameter values. The list of average acoustic parameter values of the plugin is available in Table 3.15. and the T30 and EDT curves can be shown in Figure 3.27,

Average Values - TV Presets

Preset Name	T30 (s)	T30 (s)	D50 (%)	C80 (dB)	Ts (s)
HR "Studio"	0.51	0.39	83.51	12.20	0.03
HR "Hall"	1.26	1.15	48.72	2.78	0.08
HR "Church"	1.79	1.62	44.01	2.02	0.10

Table 3.15.: Room Acoustic Parameters average values for the TrueVerb Presets

In the Studio Preset, the T30 and EDT closely matched the Aufnahme 2 across the spectrum, except for a small difference in the high end for T30. The Brahms Preset T30 alignment is almost identical, but the EDT presents a big dip in the frequencies around 200 [Hz]. Finally, the Church Present was the most difficult to approximate; both T30 and EDT remain below the measured Erlöserkirche curves across much of the spectrum,

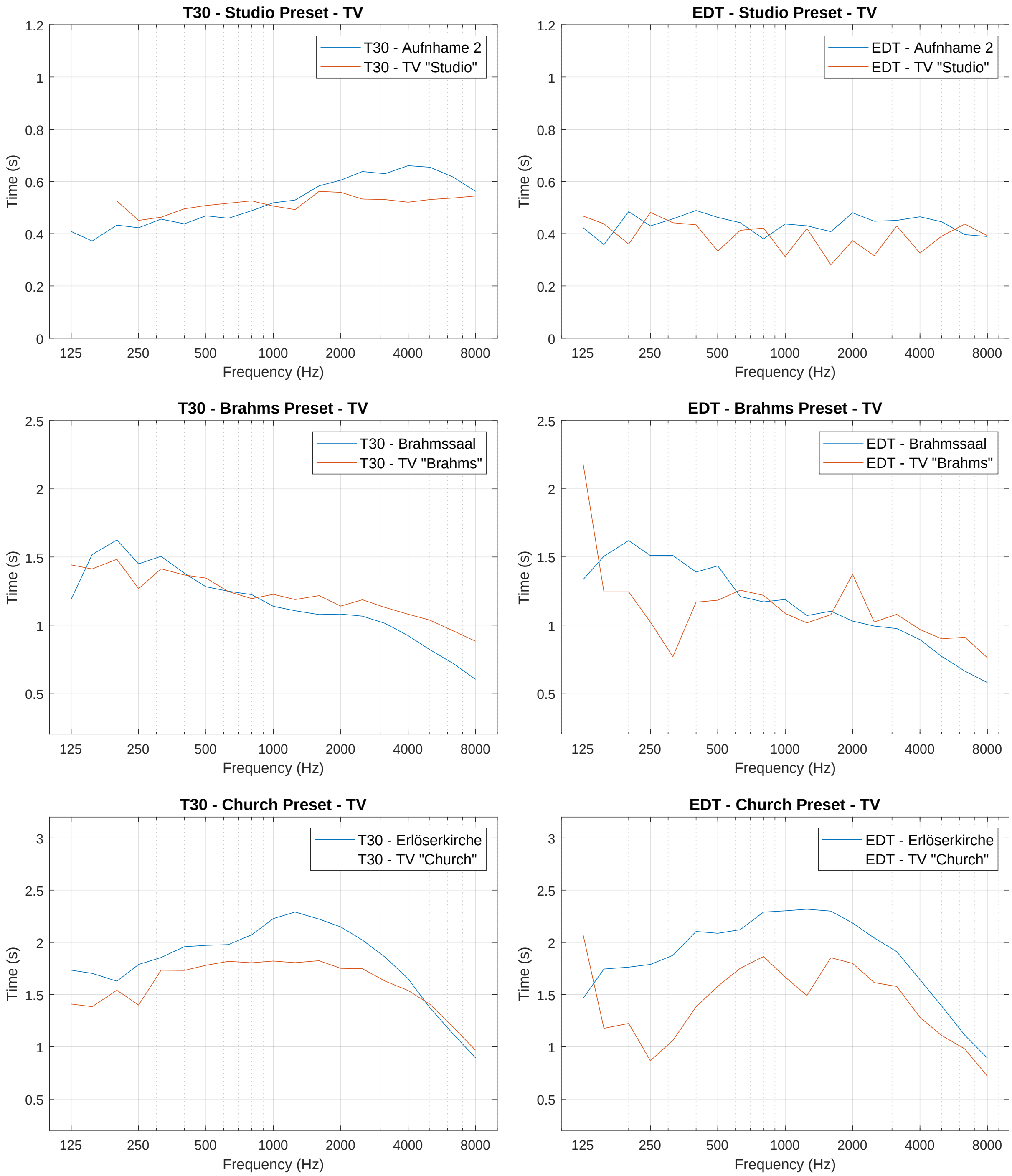

Figure 3.27.: T30 and EDT for every Preset created with the TrueVerb plugin.

3.4.5. Artificial Signals Processing

All the Presets for the artificial reverberation processors (Hall of Fame 2, H-Reverb, Bricasti M7, and TrueVerb) were created following the same steps: First the processor's global controls were set, meaning deactivating or bypassing any control that did not contribute to the recreation of a natural sounding soundscape; any control designed to introduce "experimental" characteristics. Then a calibration process followed for the processors that were physical hardware units (Hall of Fame and Bricasti M7), including unity gain and latency correction. After that, a base preset was selected to serve as the initial starting point (For example, starting with the native "Halls" preset of the M7 for the designing of the Brahmssaal preset). Then, the room acoustic parameters measured in section 3.3 were the guideline, inputting the T30 values in controls such as "Decay" or "Reverb Time". Finally, a sweep signal was run through the processor and measured to obtain its room acoustic parameters, and compared them to the actual physical rooms

This last step was iterated as many times as needed, adjusting the controls of the processors while comparing the room acoustic parameter values, and monitoring via professional mixing headphones the sounding result of the processor convolution with the instrument's "dry" samples against the "real" samples of the instruments in the physical rooms. The reverberation times relative to room size for each processor/plugin can be seen in Appendix C, and a compilation table can be seen in 3.16, where the percentage difference in reverberation time and early decay time is shown against the real measurements of the rooms.

T30 and EDT average times for all Presets.

Room / Preset	T30 (s)	*Δ* T30 (%)	EDT (s)	*Δ* EDT (%)
Aufnahme 2 (Studio)	0,48	-	0,44	-
Studio HoF	0,55	+14,6%	0,83	+88,6%
Studio HR	0,46	-4,2%	0,67	+52,3%
Studio M7	0,48	0,00%	0,58	+31,8%
Studio TV	0,51	+6,3%	0,39	-11,4%
Brahmssaal (Hall)	1,23	-	1,24	-
Hall HoF	1,25	+1,6%	1,44	16,1%
Hall HR	1,13	-8,1%	1,01	-18,6%
Hall M7	1,20	-2,4%	1,06	-14,5%
Hall TV	1,26	+2,4%	1,15	-7,3%
Erlöserkirche (Church)	2,08	-	2,20	-
Church HoF	1,99	-4,3%	2,04	-7,3%
Church HR	2,40	+15,4%	1,74	-20,9%
Church M7	2,01	-3,4%	1,96	-10,9%
Church TV	1,79	-13,9%	1,62	-26,4%

Table 3.16.: T30 and EDT average times difference between every Processor's Presets and Real Rooms.

Each processor had its own characteristics that affected the whole preset approximation process: The Hall of Fame's internal signal split required the activation of the "Kill Dry" switch, the H-Reverb's large amount of experimental parameters that were all deactivated, the Bricasti M7's unfriendly user interface of a small screen and sub menus for modification in controls, and the TrueVerb's simpler design that did not allow for much tweaking.

Since every processor has its own internal architectures and algorithm logic to create the reverberation, most of which we don't have access to, it is not realistic to expect an exact matching of every parameter value of the different environments emulated. The goal of this study is to create a general approximation, to learn the general behavior of algorithmic processors, and hopefully learn how to use them to achieve naturalness when recreating real physical environments. That is also why several processors were chosen; to randomize processes and obtain several working conditions, taking advantage of each processor's capabilities.

The Table 3.16 and the Figure 3.28 show that, for the most part, the process of approximating the processor's controls to emulate the acoustics of the different rooms was successful. All Reverberation Times (T30) fall within the accepted threshold, according to the ISO 23591[11]. The Early Decay Times, however, present bigger variations for some processors' presets. Since this study not only focuses on the objective measurement and approximation of the room acoustic parameters, but also on the perceptive sensation of naturalness when listening to reverberation, in the next sections, all the musical samples are processed through the processor's approximations, and a process of naturalness qualification and a listening test rating will be realized.

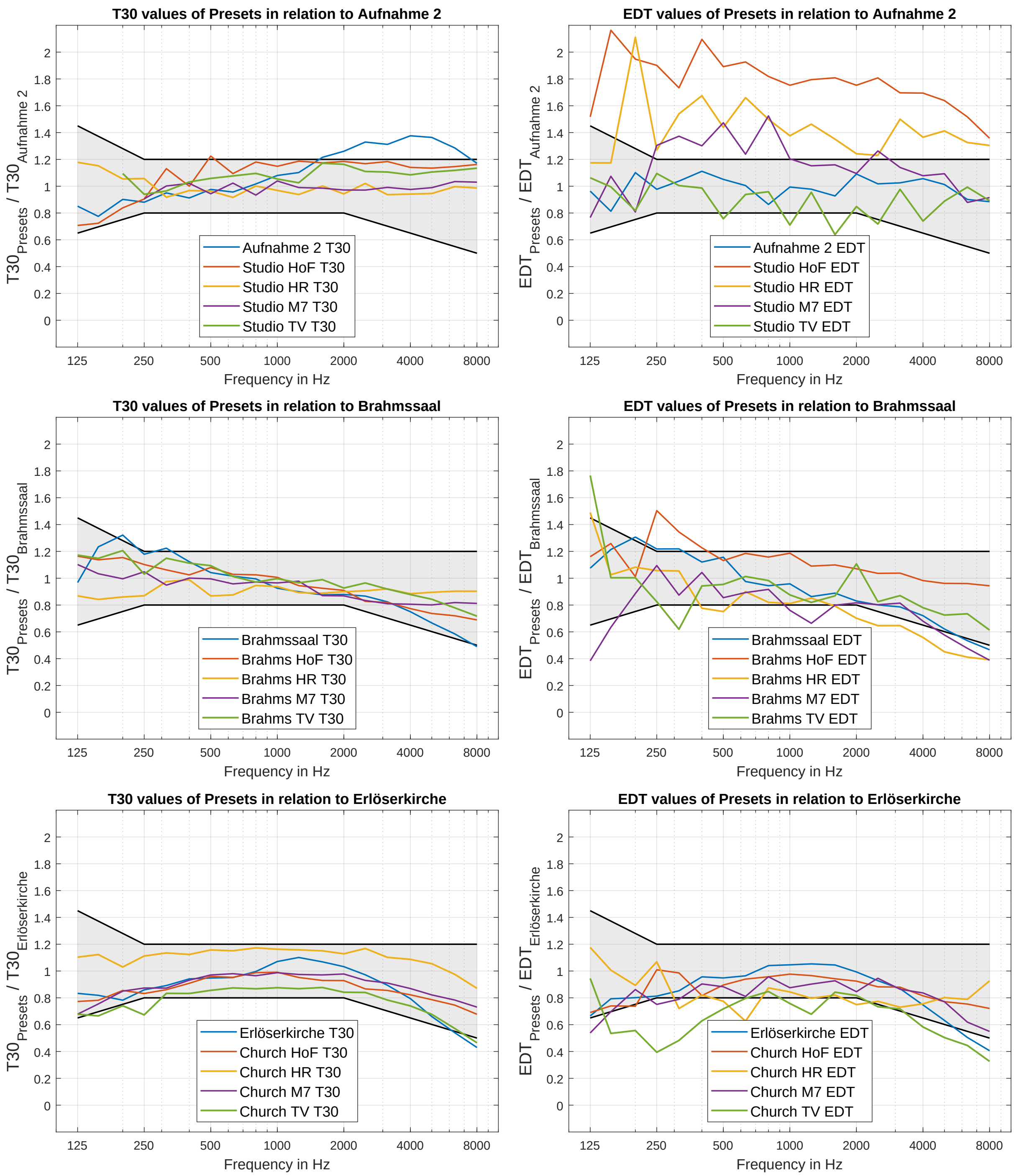

Figure 3.28.: T30 and EDT comparison for every Preset with its respective Room.

3.5. Qualification of Samples

The purpose of this section is to integrate the perceptual naturalness judgment before moving to the rating of the parameters in the final listening test. In the previous sections, the focus was on the physical measurements of the rooms and the processors. Now the process of obtaining the musical samples with "artificial" reverberation and the strategy of ensuring these are accurate perceptual representations of the rooms selected for this study in section 3.3 will be described.

3.5.1. Purpose and Design of Qualification Test

To avoid falling into an over exploration of a single condition, whether this be a particular room's acoustics, the characteristics of a single instrument, or the inner structure of a processor, a varying amount of conditions were chosen: three rooms, five instruments and four processors, to randomize all the possible combinations of conditions and avoid bias when analyzing the perceived naturalness of the reverberation. However, since there were so many samples, a process of qualification needed to be done in order to determine which samples sounded most natural. For that, a qualification test was developed, where participants were asked to judge whether each sample sounded natural or non-natural.

The criteria for choosing the participants consisted of people with ear training and familiarity with music, acoustics, or sound engineering, like tonmeisters, professional musicians, or acoustic engineers. The samples were ranked based on the overall rating of all participants, choosing the best-qualified ones to enter the next step of acoustic parameter modification.

It is important to note that all "artificial" musical samples, which were obtained by processing each artificial reverberation processor's Preset with the "dry" musical samples, were monitored to have their overall loudness compensated so that they were equal in LUFS to the "real" musical samples recorded in the different environments. This loudness compensation process was done in ProTools with the Waves Loudness Meter (WLM) [26]. First, each "real" musical sample's loudness was measured, and that value became the "target" to which each corresponding "artificial" sample of the same room-instrument combination had to be adjusted, whether that meant an increase or reduction in volume. The whole Table of loudness compensation can be found in the Appendix D.

True peak was monitored during the loudness compensation of the samples to ensure that no sample exceeded safe playback levels or introduced digital clipping. All samples were checked to ensure that the true peak level never exceeded -1.0 dBTP. This threshold was used to avoid playback distortion, especially after gain adjustments during the loudness compensation. Similarly, the duration of each "artificial" sample was checked to be of the same length as its corresponding "real samples" to ensure a consistent comparison. To ensure that the full reverberation decay was perceptually represented, each sample included additional time after the end of the direct sound, corresponding to the estimated full decay of reverberation time for each environment 3.3. Specifically, 1.21 [s] were added for Brahms samples, 2.09 [s] for Church samples, and 0.49 [s] for Studio samples. This ensured that the reverberation tail was audible and complete in both real and artificial samples.

3.5.2. Samples and User Interface

In order to realize the qualification test, a Table of all samples was created 3.17, organizing every sample by the room that is being emulated, the instrument used as the excitation signal, and the processor used to create the reverberation. A Sample Name was formulated by utilizing acronyms of every condition that was part of creating the sample. The codes for the rooms were: ST = Studio, BS = Brahmssaal, CH = Church. Each instrument was assigned a quick visual differentiation, a color coding, as well as the abbreviation: A-gtr = Acoustic guitar (yellow), E-gtr = Electric guitar (green), Legato = Legato singing (red), Speech = Speaking voice (blue), Staccato = Staccato singing (orange). The processor coding was: HoF = Hall of Fame, HR = H-Reverb, M7 = Bricasti M7, TV = TrueVerb. With a combination of three rooms, five instruments, and four processors, the whole list amounts to 60 artificial samples.

Master Table (1/2)

#	Room	Instrument	Processor	Sample Name	Group
1	Studio	Ac-gtr	HoF	ST a-gtr HoF	A
2			HR	ST a-gtr HR	
3			M7	ST a-gtr M7	
4			TV	ST a-gtr TV	
5		E-gtr	HoF	ST e-gtr HoF	B
6			HR	ST e-gtr HR	
7			M7	ST e-gtr M7	
8			TV	ST e-gtr TV	
9		Legato	HoF	ST legato HoF	C
10			HR	ST legato HR	
11			M7	ST legato M7	
12			TV	ST legato TV	
13		Speech	HoF	ST speech HoF	D
14			HR	ST speech HR	
15			M7	ST speech M7	
16			TV	ST speech TV	
17		Staccato	HoF	ST staccato HoF	E
18			HR	ST staccato HR	
19			M7	ST staccato M7	
20			TV	ST staccato TV	
21	Hall	Ac-gtr	HoF	BS a-gtr HoF	F
22			HR	BS a-gtr HR	
23			M7	BS a-gtr M7	
24			TV	BS a-gtr TV	
25		E-gtr	HoF	BS e-gtr HoF	G
26			HR	BS e-gtr HR	
27			M7	BS e-gtr M7	
28			TV	BS e-gtr TV	
29		Legato	HoF	BS legato HoF	H
30			HR	BS legato HR	

Master Table (2/2)

#	Room	Instrument	Processor	Sample Name	Group
31	Hall	Legato	M7	BS legato M7	H
32			TV	BS legato TV	
33		Speech	HoF	BS speech HoF	I
34			HR	BS speech HR	
35			M7	BS speech M7	
36			TV	BS speech TV	
37		Staccato	HoF	BS staccato HoF	J
38			HR	BS staccato HR	
39			M7	BS staccato M7	
40			TV	BS staccato TV	
41	Church	Ac-gtr	HoF	CH a-gtr HoF	K
42			HR	CH a-gtr HR	
43			M7	CH a-gtr M7	
44			TV	CH a-gtr TV	
45		E-gtr	HoF	CH e-gtr HoF	L
46			HR	CH e-gtr HR	
47			M7	CH e-gtr M7	
48			TV	CH e-gtr TV	
49		Legato	HoF	CH legato HoF	M
50			HR	CH legato HR	
51			M7	CH legato M7	
52			TV	CH legato TV	
53		Speech	HoF	CH speech HoF	N
54			HR	CH speech HR	
55			M7	CH speech M7	
56			TV	CH speech TV	
57		Staccato	HoF	CH staccato HoF	O
58			HR	CH staccato HR	
59			M7	CH staccato M7	
60			TV	CH staccato TV	

Table 3.17.: Structure and naming logic for all samples

The grouping of each sample consists of bundling up each room–instrument combination, with the goal of picking the best-rated one from each group from the qualification test to carry forward into the next test. Each group contains the four processors (HoF, HR, M7, TV) utilizing the same signal with the same type of room preset. Grouping was also done to simplify tracking and post-processing.

With the samples ready, the qualification test was developed in MATLAB. The participants were asked to first listen to a familiarization phase, where they were able to adjust the volume and listen to the "real" samples of the instruments recorded in one of the rooms: the Aufnahme 2, the Brahmssaal, or the Erloserkirche. After this, the test phase for that room began, where the participant would listen one by one to the 20 "artificial" samples made to emulate the acoustics of that room and qualify them as either "Natural" or "Non-Natural" sounding.

The UI of the test was pretty simple; it consisted of one sample per test screen, a photograph of the room, a "Play Sample" button, a "Stop" button, and the "Natural" and "Non-Natural" buttons, along with the option to go "Back" or "Next" 3.29. The Next button remained disabled until a response was given, preventing skipped trials. Participants were allowed to replay the samples as many times as they desired, and the volume could not be modified during the test phase, only during the familiarization phase. The "real samples" played during the familiarization phase were recorded with the binaural head so listeners could hear the spatial impression of each space. The test lasts approximately 20 minutes.

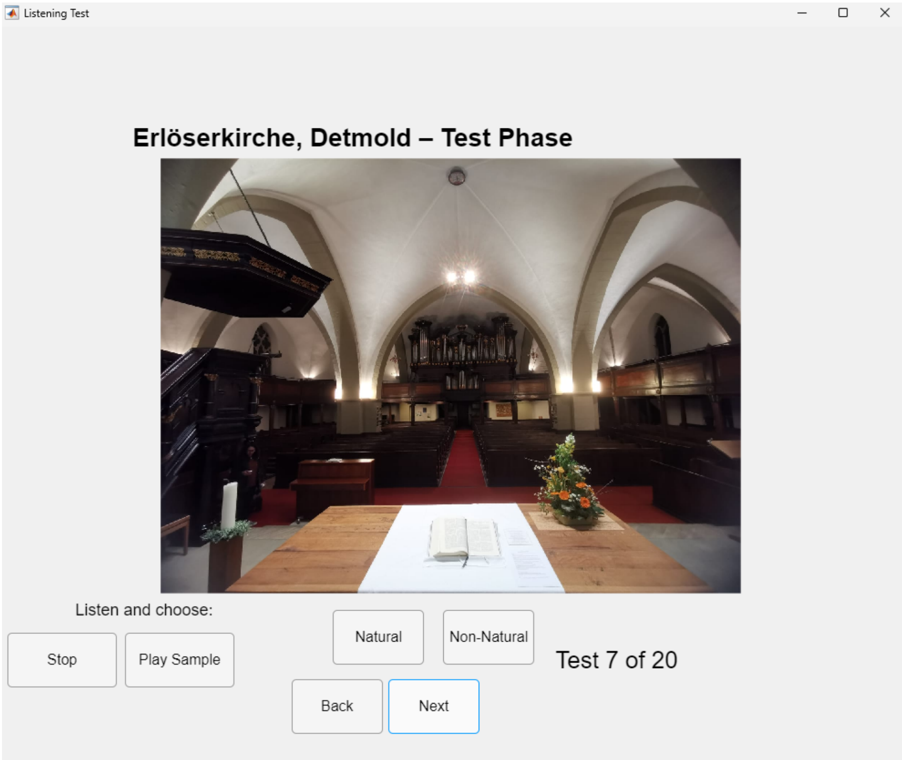

Figure 3.29.: Qualification Test's User Interface.

3.5.3. Analysis Criteria for Qualification Test

Participants answered the simple question of whether each sample sounded natural or non-natural. The results from this qualification test allowed the selection of samples that are agreed as sounding "natural" to use them as reference samples in the next section, where the modification and comparison of room acoustic parameters will take place. The perceived naturalness ratings were treated as the dependent variable, and the room and instrument combinations, including their interaction, were considered a fixed effect. The processor used in the creation of the sample was considered independent of the rating, meaning that the best-ranked samples were to be chosen no matter the type of processor they used. To account for listener differences, the participant's ID was considered a random intercept. This allowed testing whether naturalness ratings varied across instruments and rooms, and whether the effect of one factor depended on the other.

Examining the distribution of the "Natural" and "Non-Natural" ratings collected in the qualification test, besides showing how all samples were rated overall, will allow us to infer: Which Instrument tended to be rated more natural, which Room tended to be rated more natural, and how Processor choices behaved across each room–instrument combination. Finally, for the selection of the samples, the best-ranked one per group will be picked within each group to allow for a representation of every single room and instrument combination in the final test.

3.6. Sample Preparation and Listening Test

3.6.1. Strategy for Modification of Parameters

After the qualification process, one sample per combination of room and instrument is to be chosen to serve as the reference samples for this section. Starting from the reference, or "optimal" sample, the processor controls are to be adjusted to modify one room acoustic parameter between T30 and EDT at a time, in stages. When modifying T30, only the controls directly linked to reverberation time, like "Decay" or "Reverb Time," are to be changed, keeping all other controls fixed to the reference. In the same way, when modifying EDT, early reflection related controls like "Balance" or "Early Level" are to be adjusted while leaving decay controls fixed if possible.

The equalization controls of the processors, such as damping, shelves, and coloration, are not to be modified from the "optimal" samples' positions. Each modification of the reference's presets is to be measured through an IR sweep, and the values are analyzed and stored to verify that the intended parameter changes were achieved. As an anchor, the threshold of tolerance for reverberation times relative to frequency, seen in Chapter 2 and defined in section 4.2.3 of the DIN 18041 [10], will be kept in the background as a reference for the parameter modifications. Though the ranges of modifications of the parameters will surpass the defined threshold in this study, it is important to note that in the DIN 18041, the definition is made for real physical rooms, and this study focused on artificial reverberation processors. Lastly, the same process of loudness compensation process explained in section 3.5.1 is to be done with all samples obtained after the room acoustic parameters' modifications to maintain coherence with the "optimal samples.

3.6.2. Purpose and Design of Listening Test

The purpose of this test is to evaluate how systematic changes in T30 and EDT affect perceived naturalness, using the qualified reference samples as baselines. For each sample, listeners are tasked with evaluating a series of conditions based on how similar they sound in comparison with a reference. The reference is always the "optimal" samples obtained from the previous sections, and the conditions are their corresponding variations of T30 and EDT. A modified MUSHRA-based [12] test design, seen in Figure 3.30, is to be used to judge how the parameter modification changes the perceived naturalness of the already "natural-sounding" qualified reference samples.

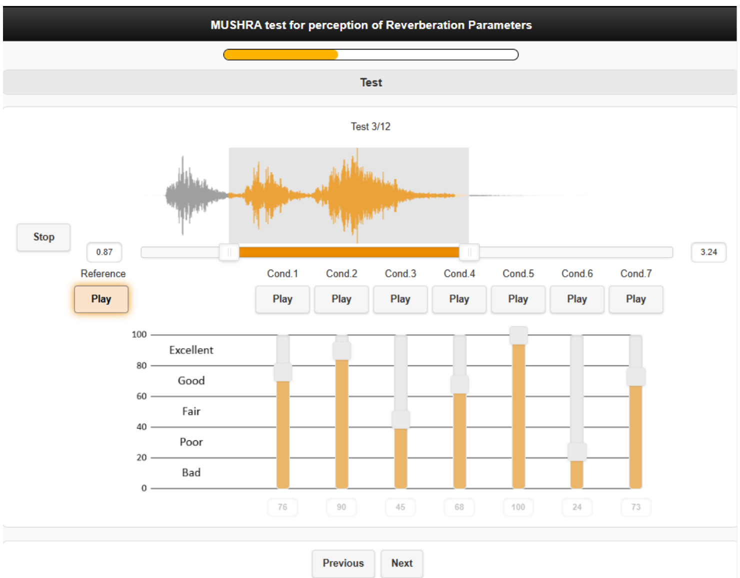

Figure 3.30.: Listening Test's User Interface.

Core elements of MUSHRA style are used, like the multi-stimulus with hidden reference structure, the continuous 0–100 quality scale, the randomized trials, the training session, and the short audio samples. To keep judgments consistent and comparable, the test design was modified to keep the session manageable and avoid listener fatigue. A single reference is to serve simultaneously for two sets of parameter changes in the same trial, T30 and EDT modifications. This helps reduce the number of trials without changing the basic task, to compare the "modified" samples against their corresponding "optimal" sample and rate their quality on the MUSHRA scale. This choice aligns with the ITU-R BS.1534-3 recommendation on limiting the number of stimuli per trial and keeping audio samples short to prevent listeners from being tired.

Because of the decisions to have two parameter modifications between the same test's conditions, the threshold for exclusion of participants was reduced from having to rate the hidden reference with a minimum 90% score in more than 15% of the trials, to 80% minimum score. The trials were randomized for each test screen, and a short training phase at the beginning of the test ensures that the participants will comprehend the objective of the test and make their best judgment. The samples last between 5 to 12 seconds, with seven stimulus conditions per screen and fixed volume during the test phase. The whole test was calculated to last around 30 to 40 minutes. Overall, the listening test stays within the MUSHRA framework while making modifications necessary to achieve the desired functionality for this study.

The number of conditions being seven was chosen after pilot testing revealed that having too many stimuli close to the reference created perceptual ambiguity, making it difficult for listeners to identify the reference or perceive meaningful differences, which resulted in a bigger percentage separation gap between the conditions' parameter modifications.

3.6.3. Analysis Criteria for Listening Test

This listening test delivers the final step in the experimental process of this study. The goal is to obtain a systematically rated ranking of the qualified reference "optimal" samples versus their reverberation time modified versions to determine the impact of the modifications on the perceived naturalness of the reverberation. The dependent variable is the MUSHRA score for the condition. The scores are to be examined to define the median trend, and their distribution, to visualize how each condition behaved, and therefore how the acoustic parameter modification affected the sample's perceived naturalness

Each room and instrument combination is to be analyzed both as a coupled structure, to compare the T30 versus EDT variation impact relative to the reference, as well as variables dependent on each other, to try to find tendencies in ratings across different instruments in the same room, and ratings of different rooms for the same instrument. The goal here is the determine whether scores change only because of the acoustic parameter's modifications, or because the naturalness is affected depending on the room, the instrument, or both.

4. Results

This chapter shows the results of the experimental work of this thesis. It first details the results for the qualification test used to select the natural sounding reference samples. It then presents the main listening test results, where versions of those references that had their room acoustic parameters modified were rated. Results are given per room and instrument, with the corresponding figures and tables for T30 and EDT comparisons. Statistical outputs are shown as distributions (quartiles, medians, means) and mixed-effects estimates for Room, Instrument, and parameter condition. Interpretation of the results, along with the analysis of the trends, will be done in the following chapter; here only the graphs and results numbers will be shown. For clarity, the figures follow the same conventions established in Chapter 3: abbreviated Sample Names and consistent color coding by instrument. Percent values are reported as percentages of votes and rounded to two decimals.

4.1. Results of Qualification Test

39 participants took part in the qualification test. Each sample received a naturalness score defined as the percentage of Natural responses against the Non-Natural responses. The whole dataset contains 60 artificial samples: 3 rooms, times 5 instruments, times 4 processors. Scores are presented in three complementary views: per room, per instrument, and per processor. For each view, the figures include:

- A graph of all corresponding samples with the 50% acceptance line.
- An ordered list with the samples ranked by score.
- A vote-distribution plot that shows the percentage of Natural votes per sample relative to the 50% line.

The Sample Name coding used throughout the figures follows the convention defined in Chapter 3.5: room code (ST, BS, CH), instrument code (a-gtr, e-gtr, legato, speech, staccato), and processor code (HoF, HR, M7, TV). The summary of the statistics for the 60 scores is the Mean at the 53,25% point, the Median at 57,69%, the Mode is 61,54%, and the Maximum and Minimum values are 84,62% and 2,56%, respectively. The Standard deviation of the mean is 0,18 (or 18,33 percentage points). Each room graph shows 20 items (5 instruments multiplied by 4 processors), each instrument graph shows 12 items (3 rooms multiplied by 4 processors), and each processor graph shows 15 samples (3 rooms multiplied by 5 instruments).

4.1.1. Ranking per Room

In this section, for each room, a graph containing the ranking of the 20 samples per room is shown. These samples are clustered by group of instruments, marked by their coloring, and with the processor displayed in sequence. The graphs also include a 50% reference line that represents the acceptance threshold. Then, a table with the samples ordered by ranking of naturalness scores, together with a vote distribution plot, is presented.

Scores for the studio presets span from 2.56% to 84.62%, meaning that it contains both the highest and

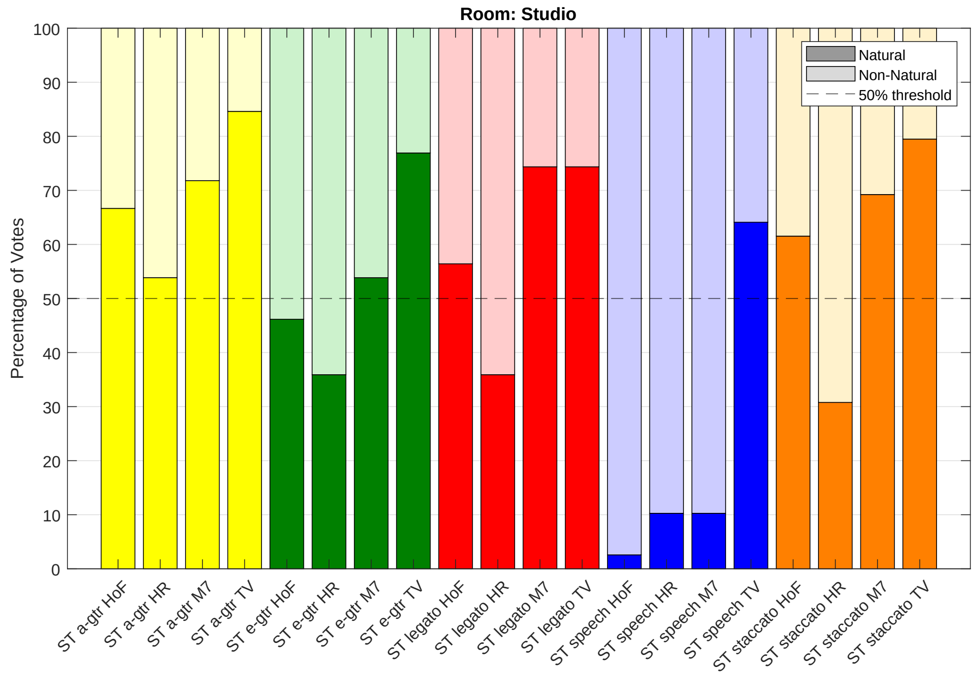

Figure 4.1.: Distribution of perceived naturalness across samples for Studio room.

lowest ranked samples of the whole test. The top three samples were all from the TrueVerb processor, marking it as the best processor for this room. Out of the 20 samples, 13 were ranked above the 50% acceptance threshold, and 10 above the all-sample median of 57.69%. The lowest rated instrument for this room was the Speech, with only one sample ranked above the threshold, while the other three were among the lowest ranked samples in the whole test. The general distribution for this room shows that while most samples are highly natural sounding, the rating from the participants is polarized.

Naturalness Scores - Studio					
#	**Room**	**Instrument**	**Processor**	**Sample Name**	**% Score**
1	Studio	Ac-gtr	TV	ST a-gtr TV	84,62%
2		Staccato	TV	ST staccato TV	79,49%
3		E-gtr	TV	ST e-gtr TV	76,92%
4		Legato	M7	ST legato M7	74,36%
5		Legato	TV	ST legato TV	74,36%
6		Ac-gtr	M7	ST a-gtr M7	71,79%
7		Staccato	M7	ST staccato M7	69,23%
8		Ac-gtr	HoF	ST a-gtr HoF	66,67%
9		Speech	TV	ST speech TV	64,10%
10		Staccato	HoF	ST staccato HoF	61,54%
11		Legato	HoF	ST legato HoF	56,41%
12		Ac-gtr	HR	ST a-gtr HR	53,85%
13		E-gtr	M7	ST e-gtr M7	53,85%
14		E-gtr	HoF	ST e-gtr HoF	46,15%
15		E-gtr	HR	ST e-gtr HR	35,90%
16		Legato	HR	ST legato HR	35,90%
17		Staccato	HR	ST staccato HR	30,77%
18		Speech	HR	ST speech HR	10,26%
19		Speech	M7	ST speech M7	10,26%
20		Speech	HoF	ST speech HoF	2,56%

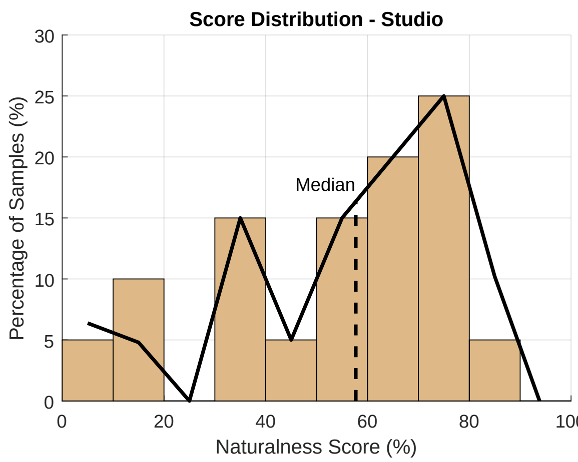

Figure 4.2.: Naturalness Scores and Vote Distribution for Studio room

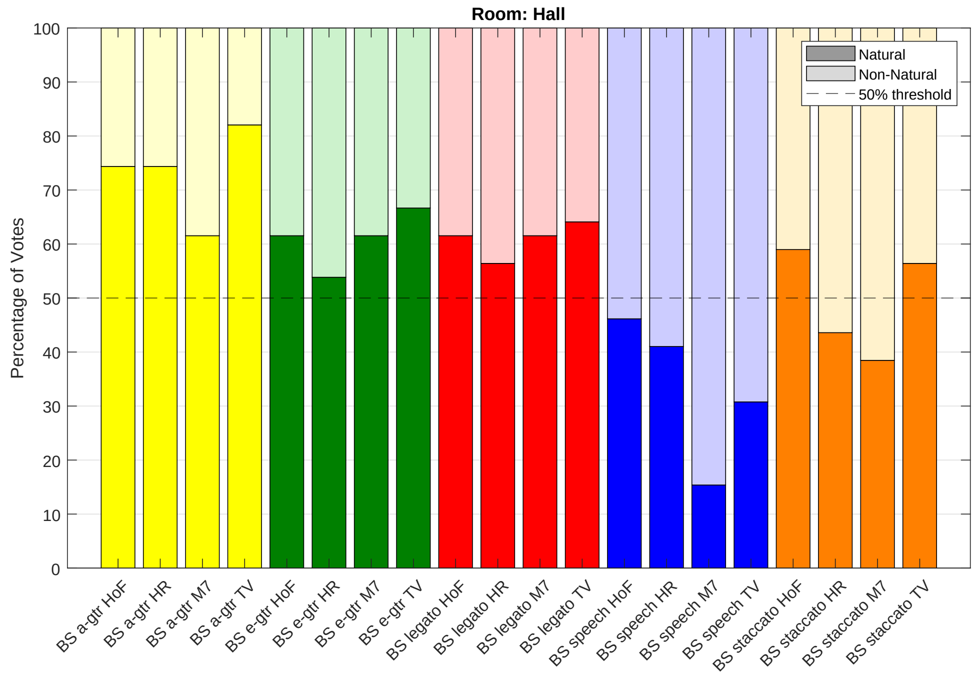

Figure 4.3.: Perceived naturalness across samples for Hall room.

Scores for the hall presets span from 15.38% to 82.05%. Out of the 20 samples, 14 were at or above the 50% acceptance threshold, and 11 were above the all-sample median (57.69%), meaning that it is the room with the most samples above the median. The lowest rated instrument for this room was Speech, with none of its samples above the threshold. The highest rated sample was "BS a-gtr TV" and the worst rated sample was "BS speech M7". The distribution shows that this room has a fairly stable rating from the participants.

Naturalness Scores - Hall					
#	Room	Instrument	Processor	Sample Name	% Score
1	Hall	Ac-gtr	TV	BS a-gtr TV	82,05%
2		Ac-gtr	HoF	BS a-gtr HoF	74,36%
3		Ac-gtr	HR	BS a-gtr HR	74,36%
4		E-gtr	TV	BS e-gtr TV	66,67%
5		Legato	TV	BS legato TV	64,10%
6		Ac-gtr	M7	BS a-gtr M7	61,54%
7		E-gtr	HoF	BS e-gtr HoF	61,54%
8		E-gtr	M7	BS e-gtr M7	61,54%
9		Legato	HoF	BS legato HoF	61,54%
10		Legato	M7	BS legato M7	61,54%
11		Staccato	HoF	BS staccato HoF	58,97%
12		Legato	HR	BS legato HR	56,41%
13		Staccato	TV	BS staccato TV	56,41%
14		E-gtr	HR	BS e-gtr HR	53,85%
15		Speech	HoF	BS speech HoF	46,15%
16		Staccato	HR	BS staccato HR	43,59%
17		Speech	HR	BS speech HR	41,03%
18		Staccato	M7	BS staccato M7	38,46%
19		Speech	TV	BS speech TV	30,77%
20		Speech	M7	BS speech M7	15,38%

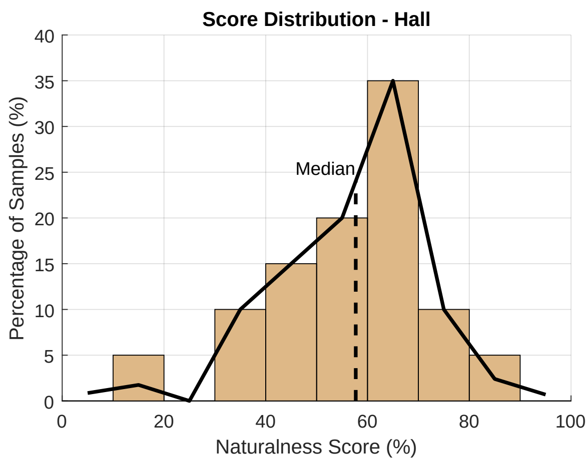

Figure 4.4.: Naturalness Scores and Vote Distribution for Hall room

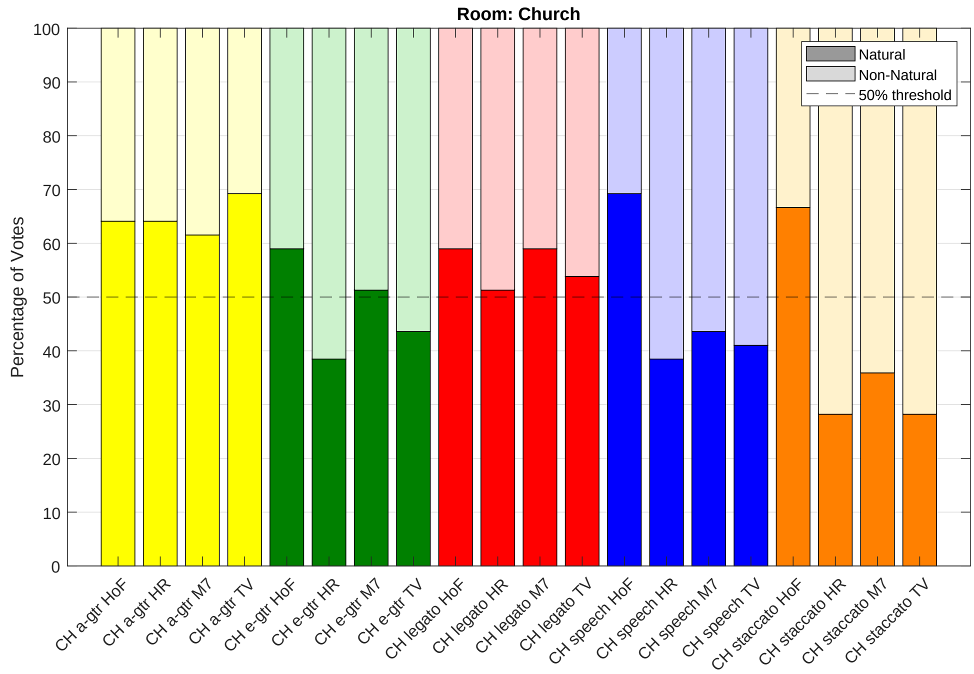

Figure 4.5.: Perceived naturalness across samples for Church room.

Scores for the church presets span from 28.21% to 69.23%. The top position was shared by "CH a-gtr TV" and "CH speech HoF" (both 69.23%). Out of the 20 samples, 12 were at or above the 50% threshold, and 9 were above the all-sample median (57.69%). The lowest rated instrument for this room was Staccato, with 2 samples ("CH staccato HR" and "CH staccato TV") sharing the minimum (28.21%). The distribution is comparatively narrow and centered near 50–60%, with fewer extreme scores than in the studio or hall categories.

Naturalness Scores - Church					
#	**Room**	**Instrument**	**Processor**	**Sample Name**	**% Score**
1	Church	Ac-gtr	TV	CH a-gtr TV	69,23%
2		Speech	HoF	CH speech HoF	69,23%
3		Staccato	HoF	CH staccato HoF	66,67%
4		Ac-gtr	HoF	CH a-gtr HoF	64,10%
5		Ac-gtr	HR	CH a-gtr HR	64,10%
6		Ac-gtr	M7	CH a-gtr M7	61,54%
7		E-gtr	HoF	CH e-gtr HoF	58,97%
8		Legato	HoF	CH legato HoF	58,97%
9		Legato	M7	CH legato M7	58,97%
10		Legato	TV	CH legato TV	53,85%
11		E-gtr	M7	CH e-gtr M7	51,28%
12		Legato	HR	CH legato HR	51,28%
13		E-gtr	TV	CH e-gtr TV	43,59%
14		Speech	M7	CH speech M7	43,59%
15		Speech	TV	CH speech TV	41,03%
16		E-gtr	HR	CH e-gtr HR	38,46%
17		Speech	HR	CH speech HR	38,46%
18		Staccato	M7	CH staccato M7	35,90%
19		Staccato	HR	CH staccato HR	28,21%
20		Staccato	TV	CH staccato TV	28,21%

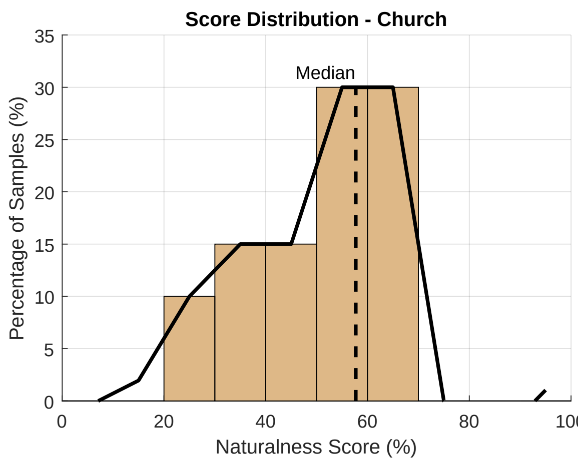

Figure 4.6.: Naturalness Scores and Vote Distribution for Church room.

4.1.2. Ranking per Instrument

For each instrument, each graph ranks its 12 samples (3 rooms multiplied by 4 processors), all with the corresponding instrument coloring and shown in clusters of room grouping, with processors in sequence, with a 50% acceptance threshold line as reference. A table then lists the samples ordered by naturalness score, followed by a vote-distribution plot.

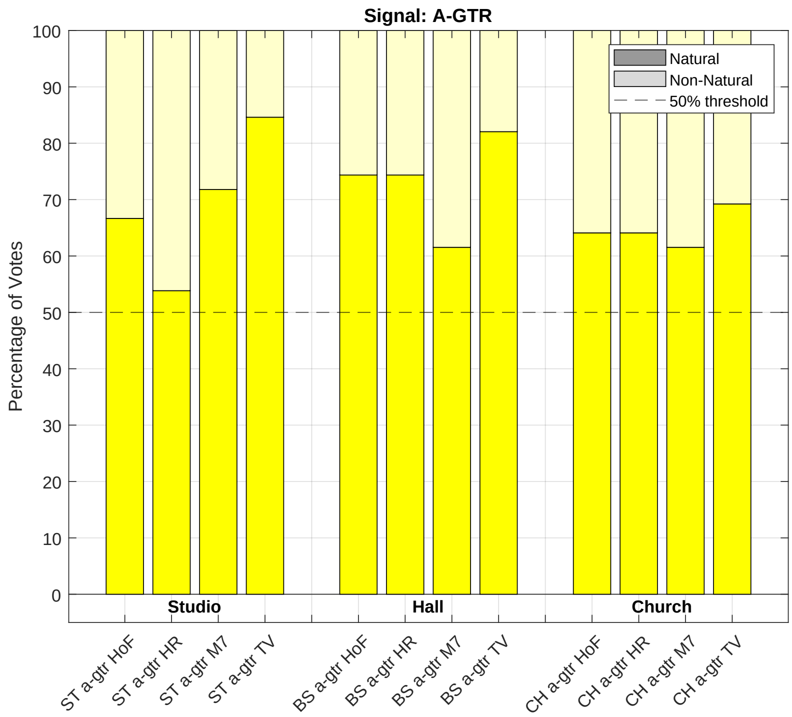

Figure 4.7.: Distribution of perceived naturalness across samples for Acoustic Guitar.

Scores for Ac-gtr range from 53.85% to 84.62%. All 12 samples are above the 50% acceptance threshold, and 11 are above the all-sample median (57.69%), meaning that this is a highly natural sounding instrument. The highest-rated sample is “ST a-gtr TV” with an 84.62% score. The lowest rated sample is “ST a-gtr HR” with 53.85%, still above the 50% line and above the overall mean. The score distribution is skewed toward higher values (instrument median is 67.95%), and the vote-distribution plot shows Natural votes dominating for all samples, with the largest margins for the TrueVerb processor, especially in Studio and Hall.

Naturalness Scores - Acoustic guitar					
#	**Instrument**	**Room**	**Processor**	**Real Name**	**% Score**
1	Ac-gtr	Studio	TV	ST a-gtr TV	84,62%
2		Hall	TV	BS a-gtr TV	82,05%
3		Hall	HoF	BS a-gtr HoF	74,36%
4		Hall	HR	BS a-gtr HR	74,36%
5		Studio	M7	ST a-gtr M7	71,79%
6		Church	TV	CH a-gtr TV	69,23%
7		Studio	HoF	ST a-gtr HoF	66,67%
8		Church	HoF	CH a-gtr HoF	64,10%
9		Church	HR	CH a-gtr HR	64,10%
10		Church	M7	CH a-gtr M7	61,54%
11		Hall	M7	BS a-gtr M7	61,54%
12		Studio	HR	ST a-gtr HR	53,85%

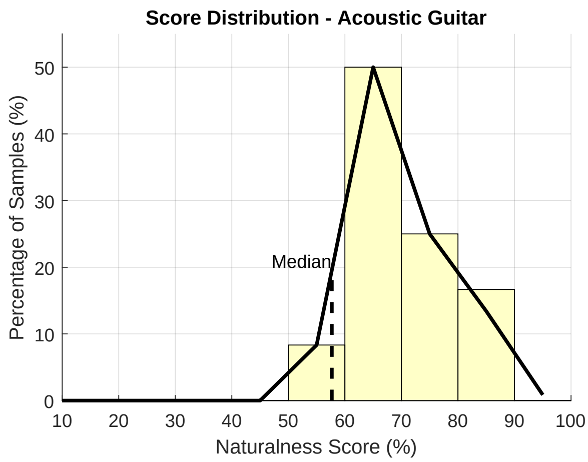

Figure 4.8.: Naturalness Scores and Vote Distribution for Acoustic guitar.

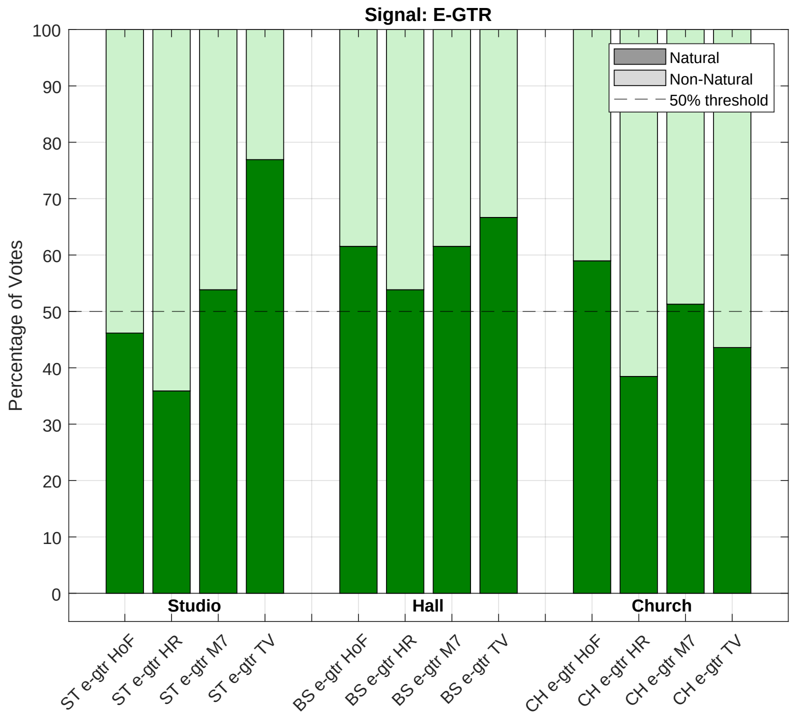

Figure 4.9.: Distribution of perceived naturalness across samples for Electric Guitar.

Scores distribution for E-gtr ranges from 35.90% to 76.92%. 8 of the 12 samples are above the 50% acceptance threshold, and five are above the all-sample median (57.69%). The highest rated sample is "ST e-gtr TV" with 76.92%, followed by "BS e-gtr TV" at 66.67%; the lowest rated sample is "ST e-gtr HR" with 35.90%. The score distribution is centered only slightly above the threshold (instrument median is 53.85%), indicating a mixed response. The naturalness score table shows that the best-ranked processor is the TrueVerb in the Studio and Hall rooms, while several Church room samples fall below the threshold. The Studio room shows the widest spread and contains both the best and worst samples. Hall is consistently highly ranked, with all four samples at or above 53.85%, and the Church room seems to trend lower. From a processor point of view, the TrueVerb processor leads overall, then the HoF, the M7, and finally, the worst rated processor is the HR, including the overall minimum.

Naturalness Scores - Electric guitar					
#	**Instrument**	**Room**	**Processor**	**Real Name**	**% Score**
1	E-gtr	Studio	TV	ST e-gtr TV	76,92%
2		Hall	TV	BS e-gtr TV	66,67%
3		Hall	HoF	BS e-gtr HoF	61,54%
4		Hall	M7	BS e-gtr M7	61,54%
5		Church	HoF	CH e-gtr HoF	58,97%
6		Hall	HR	BS e-gtr HR	53,85%
7		Studio	M7	ST e-gtr M7	53,85%
8		Church	M7	CH e-gtr M7	51,28%
9		Studio	HoF	ST e-gtr HoF	46,15%
10		Church	TV	CH e-gtr TV	43,59%
11		Church	HR	CH e-gtr HR	38,46%
12		Studio	HR	ST e-gtr HR	35,90%

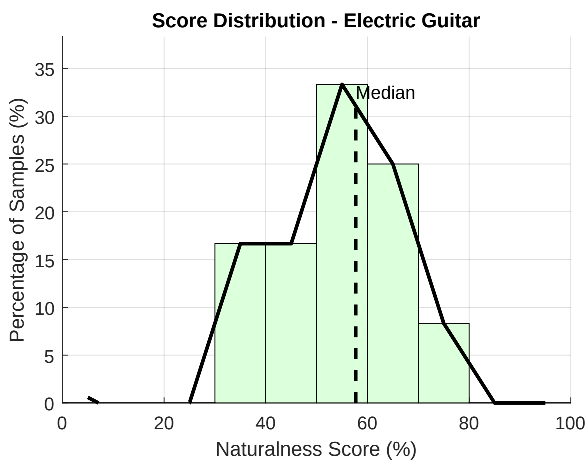

Figure 4.10.: Naturalness Scores and Vote Distribution for Electric Guitar.

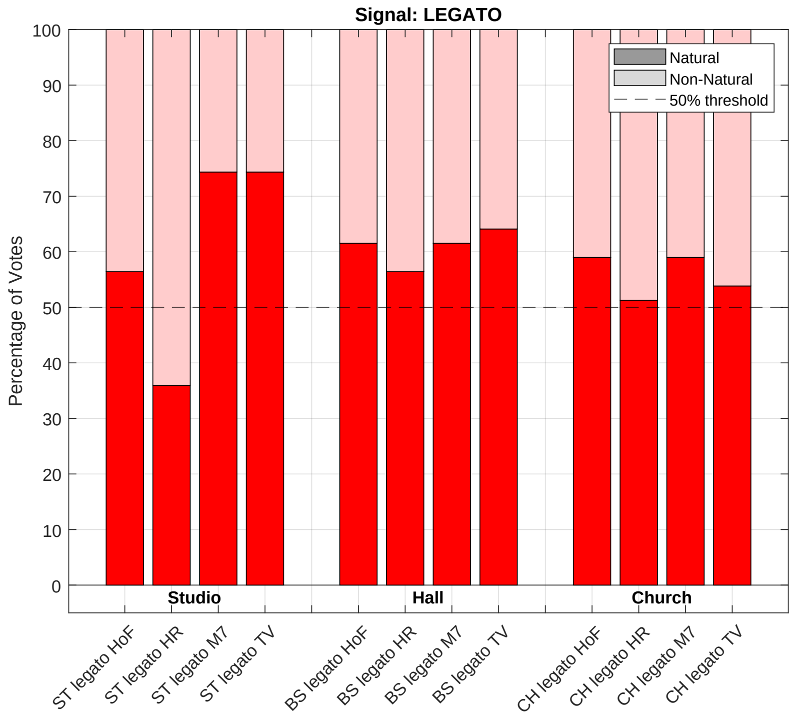

Figure 4.11.: Distribution of perceived naturalness across samples for Legato singing.

Scores distribution for Legato ranges from 35.90% to 74.36%. 11 of the 12 samples are above the 50% acceptance threshold, and 7 are above the all-sample median (57.69%). The highest rated samples are "ST legato M7" and "ST legato TV", both at 74.36%. The lowest rated sample is "ST legato HR" with 35.90%. The score distribution is slightly skewed toward higher values (instrument median is 58.97%). By room, the Studio room shows the widest spread and contains both the maximum and the minimum scored sample. The Hall room is stable, with all four samples at or above 53.85%; the Church room clusters narrowly around 51–59%, with all samples above the threshold. From a processor view, the TrueVerb processor and M7 are the strongest, leading in the Studio room and ranking high in the Hall room, while the HR processor is the weakest, though it remains above threshold in the Hall and Church rooms.

Naturalness Scores - Legato					
#	Instrument	Room	Processor	Real Name	% Score
1	Legato	Studio	M7	ST legato M7	74,36%
2		Studio	TV	ST legato TV	74,36%
3		Hall	TV	BS legato TV	64,10%
4		Hall	HoF	BS legato HoF	61,54%
5		Hall	M7	BS legato M7	61,54%
6		Church	HoF	CH legato HoF	58,97%
7		Church	M7	CH legato M7	58,97%
8		Studio	HoF	ST legato HoF	56,41%
9		Hall	HR	BS legato HR	53,85%
10		Church	TV	CH legato TV	53,85%
11		Church	HR	CH legato HR	51,28%
12		Studio	HR	ST legato HR	35,90%

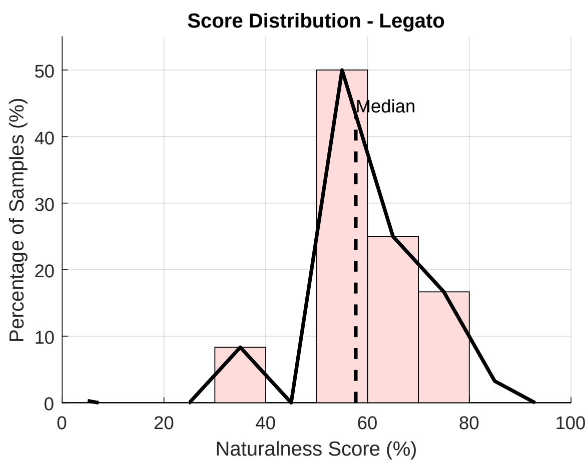

Figure 4.12.: Naturalness Scores and Vote Distribution for Legato.

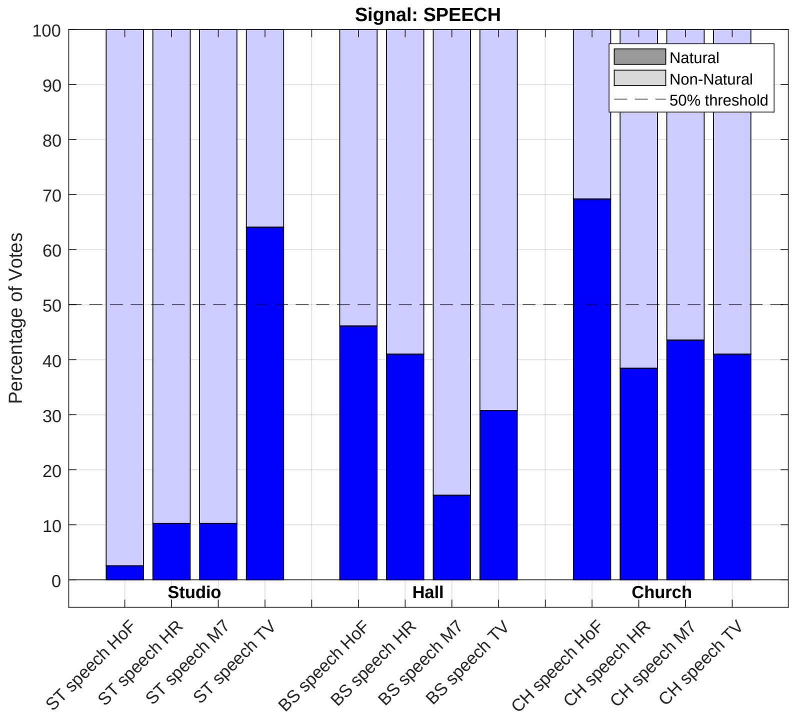

Figure 4.13.: Distribution of perceived naturalness across samples for Speech.

Scores distribution for Speech ranges from 2.56% to 69.23%. Only 2 of the 12 samples are above the 50% acceptance threshold, and the same 2 samples are above the all-sample median (57.69%). The highest rated sample is “CH speech HoF” with 69.23%, followed by “ST speech TV” at 64.10%; the lowest rated sample is “ST speech HoF” with 2.56%, being also the worst rated sample in the whole test. The score distribution is skewed toward lower values (instrument median is 39.74%), and the naturalness score table shows Non-Natural votes dominating for most samples. Only the Church room with the HoF processor and the Studio room with the TrueVerb processor are ranked natural. In general, this instrument was highly ranked as not natural.

Naturalness Scores - Speech					
#	Instrument	Room	Processor	Real Name	% Score
1	Speech	Church	HoF	CH speech HoF	69,23%
2		Studio	TV	ST speech TV	64,10%
3		Hall	HoF	BS speech HoF	46,15%
4		Church	M7	BS speech M7	43,59%
5		Hall	HR	BS speech HR	41,03%
6		Church	TV	CH speech TV	41,03%
7		Church	HR	CH speech HR	38,46%
8		Hall	TV	BS speech TV	30,77%
9		Hall	M7	BS speech M7	15,38%
10		Studio	HR	ST speech HR	10,26%
11		Studio	M7	ST speech M7	10,26%
12		Studio	HoF	ST speech HoF	2,56%

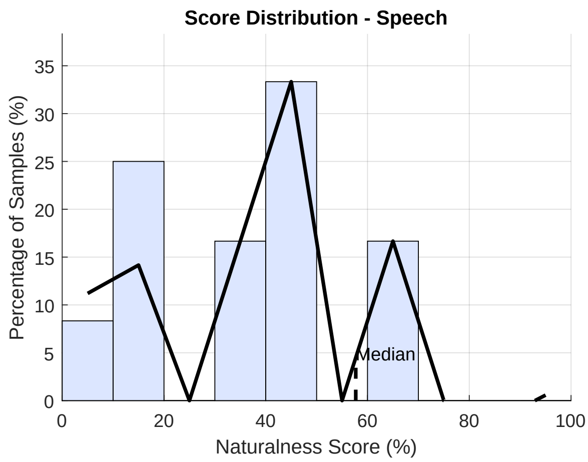

Figure 4.14.: Naturalness Scores and Vote Distribution for Speech.

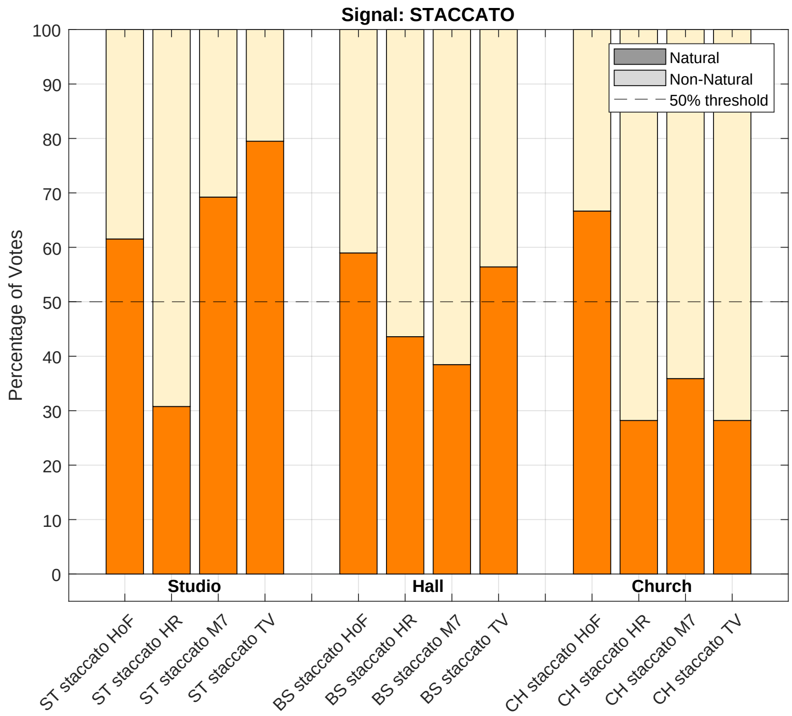

Figure 4.15.: Distribution of perceived naturalness across samples for Staccato singing.

Scores distribution for Staccato ranges from 28.21% to 79.49%. 6 of the 12 samples are above the 50% acceptance threshold, and 5 are above the all-sample median (57.69%). The highest rated sample is "ST staccato TV" with 79.49%; the lowest rated samples are "CH staccato HR" and "CH staccato TV," both at 28.21%. The naturalness score table centers at the threshold (instrument median is 50.00%), and the vote distribution plot shows a peak of Natural votes and Non-Natural majorities in most Church room samples, meaning a polarizing instrument. By room, the Studio room is the strongest (three samples above threshold, including the maximum). The Hall room is in the middle range, with two samples above threshold and two below. From a processor point of view, the TrueVerb processor is best ranked in the Studio room but and worst in the Church room, the HoF processor is best in the Church room and good in the Studio room, the M7 processor is highly ranked in the Studio room but low in the Church room and the HR processor is the worst overall.

Naturalness Scores - Staccato

#	Instrument	Room	Processor	Real Name	% Score
1	Staccato	Studio	TV	ST staccato TV	79,49%
2		Studio	M7	ST staccato M7	69,23%
3		Church	HoF	CH staccato HoF	66,67%
4		Studio	HoF	ST staccato HoF	61,54%
5		Hall	M7	BS staccato M7	58,97%
6		Hall	TV	BS staccato TV	56,41%
7		Hall	HR	BS staccato HR	43,59%
8		Hall	M7	BS staccato M7	38,46%
9		Church	M7	CH staccato M7	35,90%
10		Studio	HR	ST staccato HR	30,77%
11		Church	HR	CH staccato HR	28,21%
12		Church	TV	CH staccato TV	28,21%

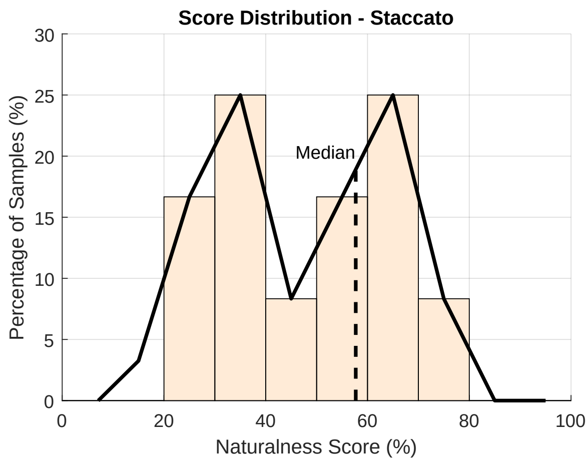

Figure 4.16.: Naturalness Scores and Vote Distribution for Staccato.

4.1.3. Ranking per Processor

For each processor, a graph showing its 15 samples (3 rooms times 5 instruments), clustered by room and colored by instrument, is shown, with a 50% acceptance line as reference. A table then lists the samples ordered by naturalness score, followed by a vote-distribution plot.

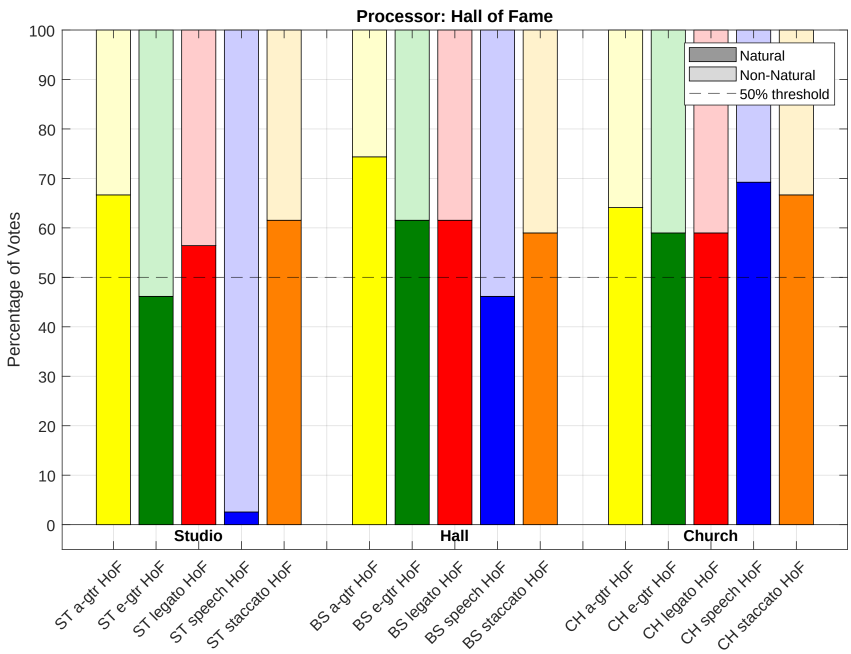

Figure 4.17.: Distribution of perceived naturalness across samples for HoF processor.

Scores distribution for the HoF processor ranges from 2.56% to 74.36%. 12 of the 15 samples are above the 50% acceptance threshold, and 11 are above the all-sample median (57.69%). The highest rated sample is “BS a-gtr HoF” and the lowest rated sample is “ST speech HoF” with 2.56%. The score distribution is skewed toward higher values (processor median is 61.54%), with most samples clustering around 59–66%. By room, the Church room is strongest (all five samples above threshold), the Hall room has four above threshold, and the Studio room is polarized with three above threshold and two well below.

Naturalness Scores - Hall Of Fame					
#	Processor	Room	Instrument	Sample Name	% Score
1	HoF	Hall	Ac-gtr	BS a-gtr HoF	74,36%
2		Church	Speech	CH speech HoF	69,23%
3		Studio	Ac-gtr	ST a-gtr HoF	66,67%
4		Church	Staccato	CH staccato HoF	66,67%
5		Church	Ac-gtr	CH a-gtr HoF	64,10%
6		Studio	Staccato	ST staccato HoF	61,54%
7		Hall	E-gtr	BS e-gtr HoF	61,54%
8		Hall	Legato	BS legato HoF	61,54%
9		Hall	Staccato	BS staccato HoF	58,97%
10		Church	E-gtr	CH e-gtr HoF	58,97%
11		Church	Legato	CH legato HoF	58,97%
12		Studio	Legato	ST legato HoF	56,41%
13		Studio	E-gtr	ST e-gtr HoF	46,15%
14		Hall	Speech	BS speech HoF	46,15%
15		Studio	Speech	ST speech HoF	2,56%

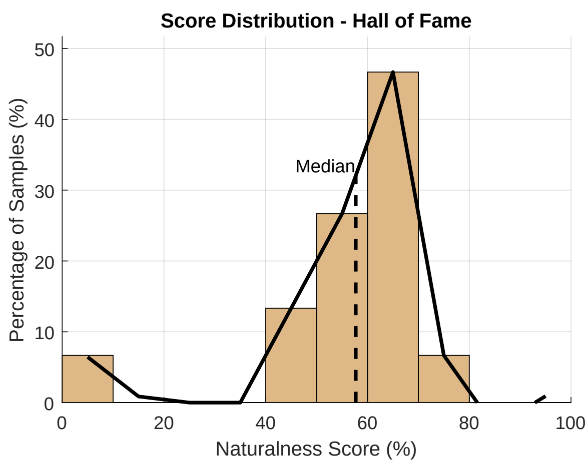

Figure 4.18.: Naturalness Scores and Vote Distribution for Hall of Fame processor.

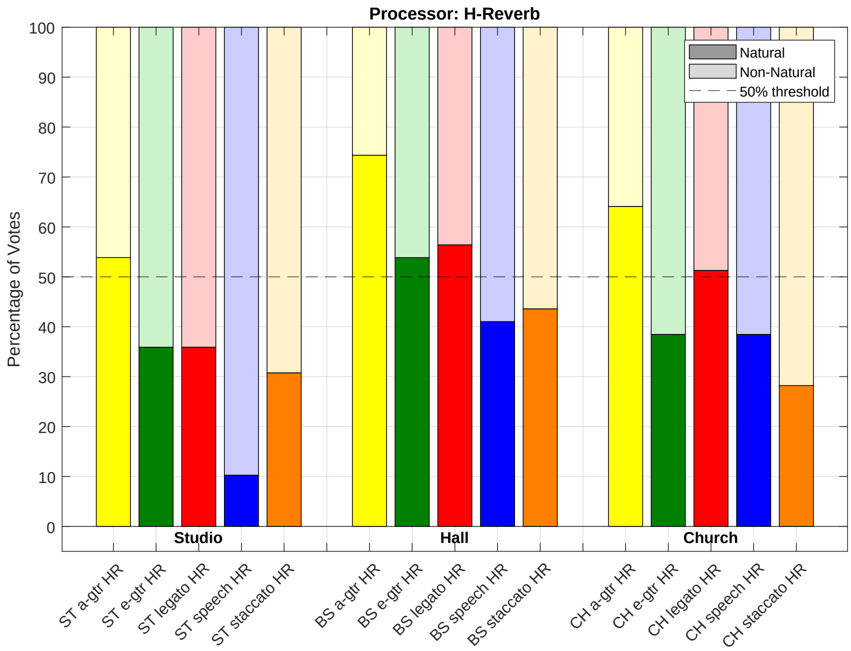

Figure 4.19.: Distribution of perceived naturalness across samples for H-Reverb processor.

Scores distribution for the HR processor ranges from 10.26% to 74.36%. 6 of the 15 samples are above the 50% acceptance threshold, and only 2 are above the all-sample median (57.69%). The highest rated sample is “BS a-gtr HR” with 74.36%, and the lowest rated sample is “ST speech HR” with 10.26%. The score distribution is skewed toward lower values (processor median is 41.03%), with most samples between 35–56%. By room, the Hall room is strongest, with 3 of 5 samples above threshold, including the maximum. The Church room is mixed, with 2 above threshold and 3 below, and the Studio room is the weakest. The vote distribution plot shows Natural votes concentrated in the few above-threshold cases, and clear Non-Natural majorities across most samples.

Naturalness Scores - H-Reverb					
#	**Processor**	**Room**	**Instrument**	**Sample Name**	**% Score**
1	HR	Hall	Ac-gtr	BS a-gtr HR	74,36%
2		Church	Ac-gtr	CH a-gtr HR	64,10%
3		Hall	Legato	BS legato HR	56,41%
4		Studio	Ac-gtr	ST a-gtr HR	53,85%
5		Hall	E-gtr	BS e-gtr HR	53,85%
6		Church	Legato	CH legato HR	51,28%
7		Hall	Staccato	BS staccato HR	43,59%
8		Hall	Speech	BS speech HR	41,03%
9		Church	E-gtr	CH e-gtr HR	38,46%
10		Church	Speech	CH speech HR	38,46%
11		Studio	E-gtr	ST e-gtr HR	35,90%
12		Studio	Legato	ST legato HR	35,90%
13		Studio	Staccato	ST staccato HR	30,77%
14		Church	Staccato	CH staccato HR	28,21%
15		Studio	Speech	ST speech HR	10,26%

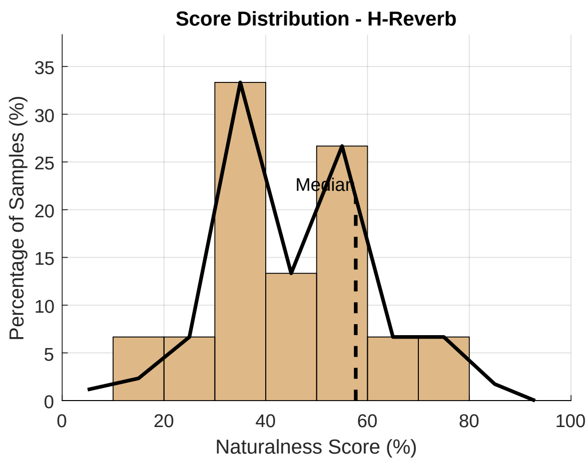

Figure 4.20.: Naturalness Scores and Vote Distribution for H-Reverb processor.

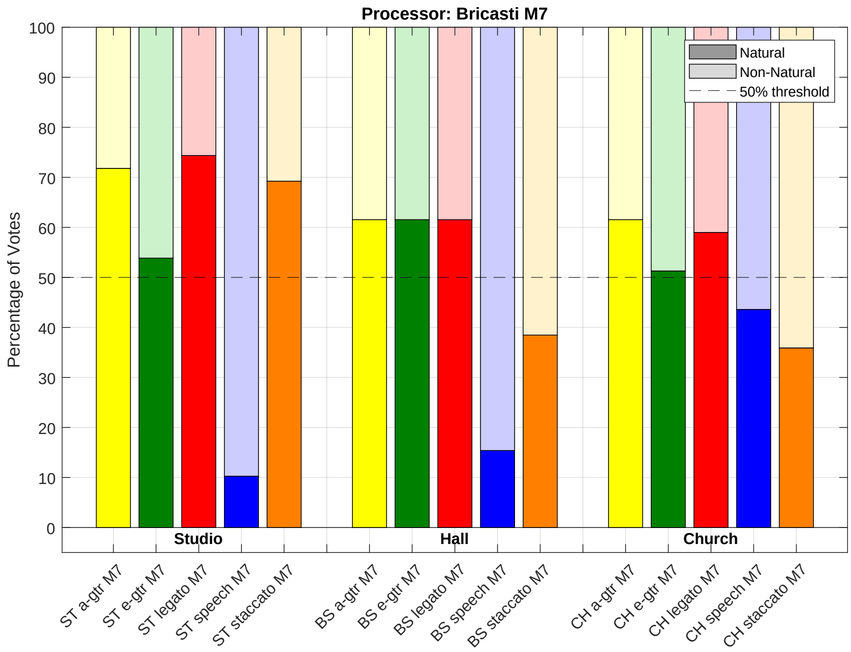

Figure 4.21.: Distribution of perceived naturalness across samples for M7 processor.

Scores distribution for the M7 processor ranges from 10.26% to 74.36%. 10 of the 15 samples are above the 50% acceptance threshold, and 8 are above the all-sample median (57.69%). The highest rated sample is “ST legato M7” with 74.36%, and the lowest rated sample is “ST speech M7” with 10.26%. The score distribution is tilted toward higher values (processor median is 58.97%), with a clear cluster around 58–62%. By room, the Studio room is strongest, with 4 of 5 samples above threshold and the top three overall, the Hall room has 3 above threshold and 2 below, and the Church room also has 3 above threshold and 2 below. The vote distribution plot shows Natural votes dominating for the above-threshold samples, while Non-Natural majorities for speech can be seen in the naturalness scores table.

Naturalness Scores - Bricasti M7

#	Processor	Room	Instrument	Sample Name	% Score
1		Studio	Legato	ST legato M7	74,36%
2		Studio	Ac-gtr	ST a-gtr M7	71,79%
3		Studio	Staccato	ST staccato M7	69,23%
4		Hall	Ac-gtr	BS a-gtr M7	61,54%
5		Hall	E-gtr	BS e-gtr M7	61,54%
6		Hall	Legato	BS legato M7	61,54%
7		Church	Ac-gtr	CH a-gtr M7	61,54%
8	M7	Church	Legato	CH legato M7	58,97%
9		Studio	E-gtr	ST e-gtr M7	53,85%
10		Church	E-gtr	CH e-gtr M7	51,28%
11		Church	Speech	CH speech M7	43,59%
12		Hall	Staccato	BS staccato M7	38,46%
13		Church	Staccato	CH staccato M7	35,90%
14		Hall	Speech	BS speech M7	15,38%
15		Studio	Speech	ST speech M7	10,26%

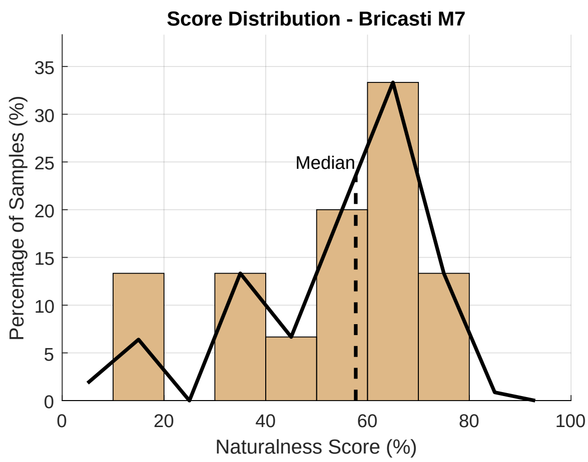

Figure 4.22.: Naturalness Scores and Vote Distribution for M7 processor.

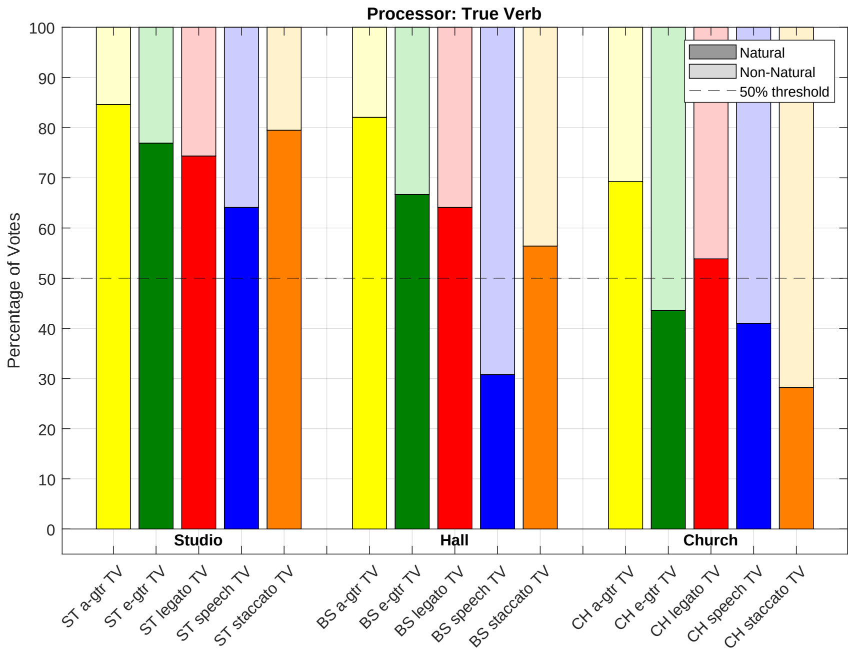

Figure 4.23.: Distribution of perceived naturalness across samples for Staccato singing.

The scores distribution for the TrueVerb processor ranges from 28.21% to 84.62%. 11 of the 15 samples are above the 50% acceptance threshold, and 9 are above the all-sample median (57.69%). The highest rated sample is "ST a-gtr TV" with 84.62%, and the lowest rated sample is "CH staccato TV" with 28.21%. The score distribution is skewed toward higher values (processor median is 64.10%), with a large cluster between 64–84%. By room, the Studio room is the strongest, all five samples are above the threshold and above the all-sample median. From the Naturalness Scores table and the vote-distribution plot, the "Natural" votes dominate in the Studio room and most of the Hall room, while the Church room shows mostly "Non-Natural" rankings.

Naturalness Scores - True Reverb					
#	**Processor**	**Room**	**Instrument**	**Sample Name**	**% Score**
1	TV	Studio	Ac-gtr	ST a-gtr TV	84,62%
2		Hall	Ac-gtr	BS a-gtr TV	82,05%
3		Studio	Staccato	ST staccato TV	79,49%
4		Studio	E-gtr	ST e-gtr TV	76,92%
5		Studio	Legato	ST legato TV	74,36%
6		Church	Ac-gtr	CH a-gtr TV	69,23%
7		Hall	E-gtr	BS e-gtr TV	66,67%
8		Studio	Speech	ST speech TV	64,10%
9		Hall	Legato	BS legato TV	64,10%
10		Hall	Staccato	BS staccato TV	56,41%
11		Church	Legato	CH legato TV	53,85%
12		Church	E-gtr	CH e-gtr TV	43,59%
13		Church	Speech	CH speech TV	41,03%
14		Hall	Speech	BS speech TV	30,77%
15		Church	Staccato	CH staccato TV	28,21%

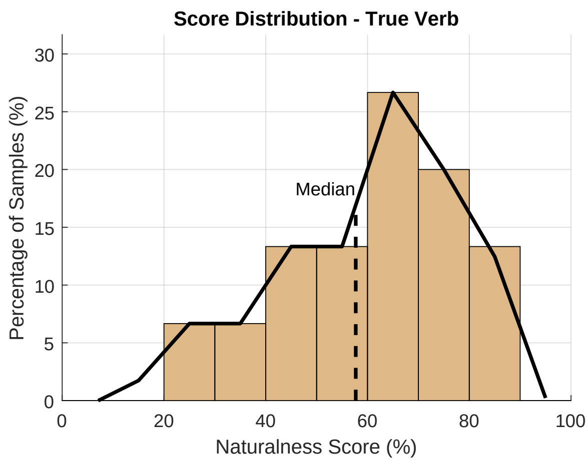

Figure 4.24.: Naturalness Scores and Vote Distribution for True Verb processor.

4.1.4. Analysis and Selection

This section lists the global results of all the samples above the 50% acceptance threshold shown in Table 4.1 and analyzes them to explain the selection process from the next stage of the study. Then the score distribution for all samples, and summary statistics by room, instrument, and processor. The approach chosen to select the samples that will move to the next stage consists of picking the best ranked one within the same room and instrument combination, meaning one for each label group, with the second exclusion factor being that at least the participants ranked that sample as a natural sounding at least 50% of the time. Throughout the different graphs and tables, the 50% acceptance threshold line is shown together with the median (57.69%) as a reference point.

Best Rated Samples (1/2)

#	Sample Name	% Score	Group	Selection
1	ST a-gtr TV	84,62%	A	Best
2	BS a-gtr TV	82,05%	F	Best
3	ST staccato TV	79,49%	E	Best
4	ST e-gtr TV	76,92%	B	Best
5	ST legato M7	74,36%	C	Best
6	ST legato TV	74,36%	C	
7	BS a-gtr HoF	74,36%	F	
8	BS a-gtr HR	74,36%	F	
9	ST a-gtr M7	71,79%	A	
10	ST staccato M7	69,23%	E	
11	CH a-gtr TV	69,23%	K	Best
12	CH speech HoF	69,23%	N	Best
13	ST a-gtr HoF	66,67%	A	
14	BS e-gtr TV	66,67%	G	Best
15	CH staccato HoF	66,67%	O	Best
16	ST speech TV	64,10%	D	Best
17	BS legato TV	64,10%	H	Best
18	CH a-gtr HoF	64,10%	K	
19	CH a-gtr HR	64,10%	K	
20	ST staccato HoF	61,54%	E	

Best Rated Samples (2/2)

#	RealName	% Score	Group	Best Rated	
21	BS a-gtr M7	61,54%	F		
22	BS e-gtr HoF	61,54%	G		
23	BS e-gtr M7	61,54%	G		
24	BS legato M7	61,54%	H		
25	BS legato M7	61,54%	H		
26	CH a-gtr M7	61,54%	K		
27	BS staccato HoF	58,97%	J	Best	
28	CH e-gtr HoF	58,97%	L	Best	
29	CH legato HoF	58,97%	M		
30	CH legato M7	58,97%	M	Best	Median
31	ST legato HoF	56,41%	C		
32	BS legato HR	56,41%	H		
33	BS staccato TV	56,41%	J		
34	ST a-gtr HR	53,85%	A		
35	ST e-gtr HR	53,85%	B		
36	BS e-gtr HR	53,85%	G		
37	CH legato TV	53,85%	M		Mean
38	CH e-gtr M7	51,28%	L		
39	CH legato HR	51,28%	M		
41	CH speech HoF	46.15%	I	Best	< 50%

Table 4.1.: Ranking of samples with perceived naturalness above 50% of the votes.

The overall score distribution shown in Figure 4.2 spans from 2.56% to 84.62%, the average value being 53.25%. The distribution is skewed towards the higher range of the votes, with a peak in the range of 60 to 70% (the mode being 61.54%), meaning that most values, 37 out of 40, are ranked above the mean of the whole test. The standard deviation of the test is 0.1833 (18.33% points), indicating a moderately wide spread in ranking for the samples. In total, 39 of the 60 samples, meaning 65% of the samples, are above the 50% threshold. The top best ranked samples 20 are dominated by the Studio room, the TrueVerb processor, and the acoustic guitar instrument. The single highest-rated sample is a combination of those three conditions. “ST a-gtr TV”, with 84.62% naturalness rank.

Table 4.3 shows that Acoustic has all its samples above the threshold of acceptance and also the narrowest vote spread, with a standard deviation of 0.09. Legato also performs well, with 92% of the samples in this category above threshold and a 0.10 standard deviation, followed by Electric with 67% samples above the acceptance threshold. The worst ranked instrument is Speech, with only 17% of its samples (2 out of 12) above the threshold and a large spread distribution of 0.21 standard deviation. Staccato sits right in the middle of the sample pool, with half of its samples ranked as natural sound and the other half as non-natural.

The Hall room was the highest ranked room category overall, with 70% of its sample above the threshold, but with a small margin in comparison to the other rooms; the Studio has 65% acceptance, and the Church has 60%, indicating that the room category had more homogeneous judgments across the whole test. The Studio room shows the largest standard deviation at 0.25 points. However, that is probably due to the outlier samples "ST speech HR", "ST speech M7", and "ST speech HoF", which all sit at or below the 10% acceptance

Global Statistics

Mean	53,25 %
Median	57,69%
Mode	61,54%
Max Value	84.62%
Min Value	2.56%
Std. Dev	0.1833

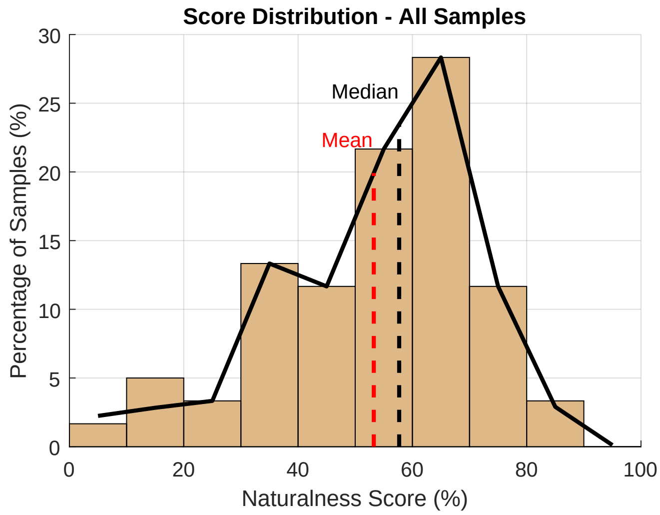

Table 4.2.: General statistics and Score Distribution for all samples.

Of all the processors, the Hall of Fame had the largest number of samples ranked as natural sounding, with 80% of its samples above the acceptance threshold. The TrueVerb processor followed closely at 73% above threshold, but with the consideration that it contains most of the best ranked samples of the ranking list in table 4.1. The Bricasti M7 processor is also well ranked with 67% of samples above threshold, and the H-Reverb (HR) processor is the weakest one with only 40% above threshold. In general, the spread of results across the processors is pretty similar, with their standard deviations ranging from 0.16 to 0.19.

Performance of each category

	$>$ **50%**	$>$ **Median**	**Std. Dev**
Studio	65%	50%	0.25
Hall	70%	55%	0.16
Church	60%	45%	0.13
Ac-gtr	100%	92%	0.09
E-gtr	67%	42%	0.12
Legato	92%	58%	0.10
Speech	17%	17%	0.21
Staccato	50%	42%	0.18
Hall of Fame	80%	73%	0.17
H-Reverb	40%	13%	0.16
Bricasti M7	67%	53%	0.19
True Verb	73%	60%	0.18
TOTAL	65%	MEAN 62%	0.18

Table 4.3.: Performance of each category in reference to the global statistics.

With these results, a list of qualified samples can be generated (Fig. 4.4 containing the best ranked samples per group, to focus on the best-ranked while also ensuring diversity in the types of samples that will move to the next stage. Out of the 15 possible room and instrument combinations, 14 were above the threshold of acceptance; only the samples in the Brahmssaal for the Speech were all considered "Non-Natural" sounding.

Qualified Samples

#	Sample name	Group
1	ST a-gtr TV	A
2	BS a-gtr TV	F
3	ST staccato TV	E
4	ST e-gtr TV	B
5	ST legato M7	C
6	CH a-gtr TV	K
7	CH speech HoF	N
8	BS e-gtr TV	G
9	CH staccato HoF	O
10	ST speech TV	D
11	BS legato TV	H
12	BS staccato HoF	J
13	CH e-gtr HoF	L
14	CH legato M7	M

Not Qualified

#	Sample name	Group
15	BS speech HoF	I

Table 4.4.: Qualified and Not Qualified Samples based on 50% acceptance threshold

This results in the "Qualified Matrix" shown in Table 4.5, which illustrates the distribution of room and instrument combination on the left side with the corresponding selected processor for every type of combination, and on the right side shows a list of the Preset names for every selected processor. Meaning that only the H-Reverb processor was not qualified for the next step.

Rooms and Instruments Distribution Matrix

	Ac-gtr	E-gtr	Legato	Speech	Staccato
Studio	TV	TV	M7	TV	TV
Hall	TV	TV	TV		HoF
Church	TV	HoF	M7	HoF	HoF

Selected Preset Names

1	TV Studio
2	M7 Studio
3	TV Hall
4	HoF Hall
5	TV Church
6	HoF Church
7	M7 Church

Table 4.5.: Matrix of best rated processors across rooms and instruments and selected Samples.

Considerations on the Hall of Fame 2 processor

While the previously shown results were supposed to move this study onto the next section of the experimentation process, a problem arose when trying to process the selected samples created with the Hall of Fame processor. While the manual and specifications of the processor [20][24] claim that the processor has controls for deep editing of specific parameters, like "Reverb" or "Early" to modify reverberation time and early decay time respectively, and while the processor's editing software (Fig 3.11) actually has those controls listed, as shown in the Appendix B. The truth is that in reality those controls barely do anything in regards of the room acoustic parameters response of the processor. When running an IR through the processor, any modification happening was only due to the "Decay" control and the equalization section, which affects both T30 and EDT, meaning that we are unable to actually modify the early reflections independently of each other.

This situation created a huge problem in the study, not only because of the great amount of research and experimentation already done on this processor, but especially because the Hall of Fame processor was one of the highest ranked processors in the results of the Qualification Test. Thus, the decision was made to also exclude the Hall of Fame processor from the study from now on.

Updated selected samples

With these considerations, the previous analysis was done again to determine the new selected samples of the qualification stage. The new updated ranking Table can be seen next 4.6. This time, only 12 out of 14 sample combinations of room and instruments were able to score above a ranking above the 50% threshold. Furthermore, while on the previous list (Table 4.1) all the selected samples were above the mean (meaning above the middle point of all samples), in this new list, one sample falls below the median line and one falls below the average, barely above the 50% threshold.

Updated Best Rated Samples (1/2)
(Excluding HoF processor)

#	Sample Name	% Score	Group	Selection
1	ST a-gtr TV	84,62%	A	Best
2	BS a-gtr TV	82,05%	F	Best
3	ST staccato TV	79,49%	E	Best
4	ST e-gtr TV	76,92%	B	Best
5	ST legato M7	74,36%	C	Best
6	ST legato TV	74,36%	C	
7	BS a-gtr HoF	74,36%	F	*Excluded*
8	BS a-gtr HR	74,36%	F	
9	ST a-gtr M7	71,79%	A	
10	ST staccato M7	69,23%	E	
11	CH a-gtr TV	69,23%	K	Best
12	CH speech HoF	69,23%	N	*Excluded*
13	ST a-gtr HoF	66,67%	A	*Excluded*
14	BS e-gtr TV	66,67%	G	Best
15	CH staccato HoF	66,67%	O	*Excluded*
16	ST speech TV	64,10%	D	Best
17	BS legato TV	64,10%	H	Best
18	CH a-gtr HoF	64,10%	K	*Excluded*
19	CH a-gtr HR	64,10%	K	
20	ST staccato HoF	61,54%	E	*Excluded*
21	BS a-gtr M7	61,54%	F	

Updated Best Rated Samples (2/2)
(Excluding HoF processor)

#	RealName	% Score	Group	**Best Rated**	
22	BS e-gtr HoF	61,54%	G	*Excluded*	
23	BS e-gtr M7	61,54%	G		
24	BS legato M7	61,54%	H		
25	BS legato M7	61,54%	H		
26	CH a-gtr M7	61,54%	K		
27	BS staccato HoF	58,97%	J	*Excluded*	
28	CH e-gtr HoF	58,97%	L	*Excluded*	
29	CH legato HoF	58,97%	M	*Excluded*	
30	CH legato M7	58,97%	M	Best	Median
31	ST legato HoF	56,41%	C	*Excluded*	
32	BS legato HR	56,41%	H		
33	BS staccato TV	56,41%	J	Best	
34	ST a-gtr HR	53,85%	A		
35	ST e-gtr HR	53,85%	B		
36	BS e-gtr HR	53,85%	G		
37	CH legato TV	53,85%	M		Mean
38	CH e-gtr M7	51,28%	L	Best	
39	CH legato HR	51,28%	M		
41	BS speech HoF	46.15%	I	Best	< 50%
44	CH speech M7	43.59%	N	Best	
52	CH staccato M7	35.90%	O	Best	

Table 4.6.: Ranking of samples with perceived naturalness above 50% of the votes.

These results reduce the list of qualified samples to 12, which is good for the purposes of the listening test duration, but reduces the variety of processors in the final listening test, narrowing the scope of the results to only two types of artificial reverberation: The Bricasti M7 hardware unit and the TrueVerb plugin.

Updated Qualified Samples (Excluding HoF processor)

#	Sample name	Group
1	ST a-gtr TV	A
2	BS a-gtr TV	F
3	ST staccato TV	E
4	ST e-gtr TV	B
5	ST legato M7	C
6	CH a-gtr TV	K
7	BS e-gtr TV	G
8	ST speech TV	D
9	BS legato TV	H
10	CH legato M7	M
11	BS staccato TV	J
12	CH e-gtr M7	L

Updated Not Qualified (Excluding HoF processor)

#	Sample name	Group
13	BS speech HoF	I
14	CH speech M7	N
15	CH staccato M7	O

Table 4.7.: Qualified and Not Qualified Samples based on 50% acceptance threshold

The new list of qualified and not qualified samples can be seen in Table 4.7. In the end, for the Studio room category, 4 out of the 5 instruments selected had the same processor's preset, the TruVerb Studio; the other one (for Legato) is the M7 Studio. For the Church, out of the three qualified combinations, two use the Bricasti M7 and one the TrueVerb. And for the Brahms category, all of the combinations selected use the Trueverb. This can be seen in the distribution matrix (Table 4.8), where for the processor name of every selected sample, together with their corresponding processor and Preset name, is shown in the next table

Updated Rooms and Instruments Distribution Matrix (Excluding HoF processor)

	Ac-gtr	E-gtr	Legato	Speech	Staccato
Studio	TV	TV	M7	TV	TV
Hall	TV	TV	TV		TV
Church	TV	M7	M7		

Selected Preset Names (Excluding HoF processor)

1	TV Studio
2	M7 Studio
3	TV Hall
4	TV Church
5	M7 Church

Table 4.8.: New matrix of best rated processors across rooms and instruments and selected Presets.

4.2. Preparation of Selected Samples

While the original objective of this study was to analyze a wide variety of room acoustic parameters, two problems appeared in the process of the research, which made it necessary to slightly narrow the focus of the investigation. The first issue was the sheer amount of parameters to dwell on, which would make the duration of any listening test surpass the recommended amount of time by far, and second, the interconnection of room acoustic parameters means that modifying any parameters will result in changes other parameters, which introduces even more variability into an already large number of study possibilities.

Thus, it was decided to focus the study on analyzing the two most well known parameters, which most algorithmic reverberation processors already deal with: Early Reflections (EDT) and Reverberation Time (T30). Both parameters are to be modified in stages and compared to a reference sample considered to have "optimal" room acoustic parameters and rated as "natural" sounding in accordance with the previous "qualification" stage. The step-by-step modification of the processors' controls is detailed together with the resulting variations on the two key room acoustic parameters, T30 and EDT. Then, a graph of these variations will be shown. For a complete detailing of the variation of all acoustic parameters, including definition (D50), clarity (C80), and Center Time (TS), see the Appendix E

Due to inherent constraints of the way the algorithmic processors behave, for most cases, it was only feasible to vary the EDT in a decreasing manner while maintaining the T30 constant. Attempts to increase the EDT inherently result in increases in the whole reverberation time, which reflects common behavior of room acoustics where early reflection times tend to be shorter than the late reflections. Thus, the variation strategy became increasing the T30 in relation to the EDT and decreasing the EDT in relation to the T30.

4.2.1. Modification of Studio Presets

For the studio presets, due to the already low reverberation times and the limitations of the processors, it was not possible to modify the controls in a way that only 10% of T30 would change, because the values were so low, So the strategy for these types of presets was to modify the controls until achieving variations of 20% of the original optimal preset. However, for the most extreme control modification case, only a 50% reduction on the EDT could be achieved, meaning that the modification logic for the room category presets would be:

- **Studio EDT modification**: optimal EDT $>$ -20% EDT $>$ -40% EDT $>$ -50% EDT.
- **Studio T30 modification**: optimal T30 $>$ +20% T30 $>$ +40% T30 $>$ +60% T30.

True Verb - Studio Preset

Starting with the True Verb processor, the details of the modifications made on its Studio preset are specified in Table 4.9, keeping track of the exact amount of variation on the controls of the processor and how it affects the room acoustic parameters. For the EDT modification, the "Early" control value was reduced, achieving a final modification of -49% on the EDT, while the T30 remained within the JND of 5% for all stages except the last one. For the T30 modification, EDT was able to be maintained at a perfect value while increasing the T30 to +59% by adjusting the "Decay" and "Early" controls.

True Verb - "Studio" Settings

Sample	Processor Settings			Room Acous. Parameters			
Name	Decay	Early	Reverb	T30	Δ	EDT	Δ
EDT optimal	0.5	-8.4	-8.9	0.51	0%	0.39	0%
EDT -10%	0.5	-6.1	-8.9	0.50	-2%	0.35	-10%
EDT -20%	0.5	-4.6	-8.9	0.50	-2%	0.31	-21%
EDT -30%	0.5	-3.1	-8.9	0.49	-4%	0.27	-31%
EDT -40%	0.5	-1.4	-8.9	0.49	-4%	0.23	-41%
EDT -50%	0.5	0.0	-8.2	0.48	-6%	0.20	-49%
T30 +60%	0.8	0.0	-8.5	0.81	59%	0.39	0%
T30 +40%	0.7	-3.4	-8.5	0.71	39%	0.39	0%
T30 +20%	0.6	-4.4	-8.5	0.60	18%	0.39	0%
T30 optimal	0.5	-8.4	-8.9	0.51	0%	0.39	0%

Table 4.9.: Processor settings and changes in T30 and EDT values for TrueVerb, Studio Preset

Figure 4.25 illustrates how the two types of parameter modifications interact with each other. First, the EDT modification can be appreciated on the top left, showing how the EDT is varying per stage of modification, and on the top right, how the T30 manages to remain constant while the EDT is being modified. The opposite effect can be seen at the bottom of the figure, on to bottom left, the resulting EDT from the T30 modifications is shown to have some variation, especially on the lower end of the spectrum, while on the bottom, the actual T30 variation can be seen.

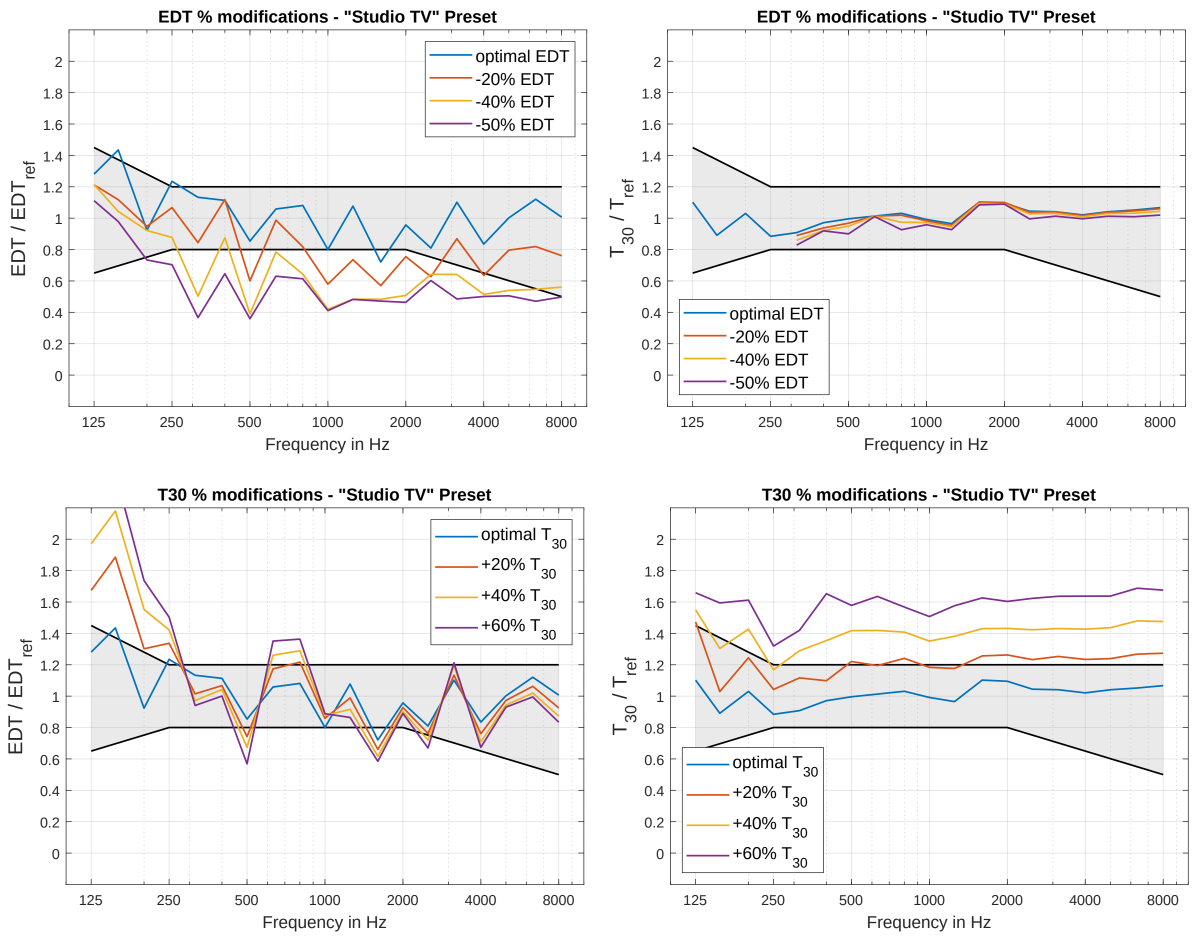

Figure 4.25.: Modification on EDT and T30 parameters for Studio preset of the TV processor.

Bricasti M7 - Studio Preset

For the Bricasti M7 processor, the details of the modifications made to its Studio room preset are specified in Table 4.10, keeping track of the exact amount of variation on the controls of the processor and how these changes affect the room acoustic parameters. For the EDT modification, "Rvb Time" is held at 0.45 [s] while the "Early / Reverb" balance is kept with the early at 20 and reverb being reduced from 19 to 11. The resulting EDT decreases from 0.58 [s] to 0.30 [s], reported as 0% to 48% modification in the table. The "T30" remains close to 0.47 [s] with only small variation below the JND. For the T30 modification, "Rvb Time" is raised from 0.45 [s] all the way up to 0.75 [s], achieving a final modification of +64%, while the EDT values remain mostly constant below the JND of 5%.

Bricasti M7 - "Studio" Settings

Sample	Processor Settings			Room Acous. Parameters			
Name	Rvb Time	Early	Reverb	T30	Δ	EDT	Δ
EDT optimal	0.45	20	19	0.47	0%	0.58	0%
EDT -10%	0.45	20	18	0.48	2%	0.53	-9%
EDT -20%	0.45	20	16	0.48	2%	0.46	-21%
EDT -30%	0.45	20	15	0.48	2%	0.41	-29%
EDT -40%	0.45	20	13	0.48	2%	0.34	-41%
EDT -50%	0.45	20	11	0.47	0%	0.30	-48%
T30 +60%	0.75	20	14	0.77	64%	0.59	2%
T30 +40%	0.65	20	15	0.66	40%	0.56	-3%
T30 +20%	0.55	20	17	0.56	19%	0.58	0%
T30 optimal	0.45	20	19	0.47	0%	0.58	0%

Table 4.10.: Processor settings and changes in T30 and EDT values for M7, Studio Preset

Figure 4.26 shows, on the top left, the EDT modification where the EDT curves fall below the optimal line across the spectrum, while the T30 curve on the top right remains almost intact. On the bottom left, the graph shows that during the T30 modification, the EDT curve had small variations but remained close to the optimal curve, and on the bottom right, the intended T30 variations are achieved.

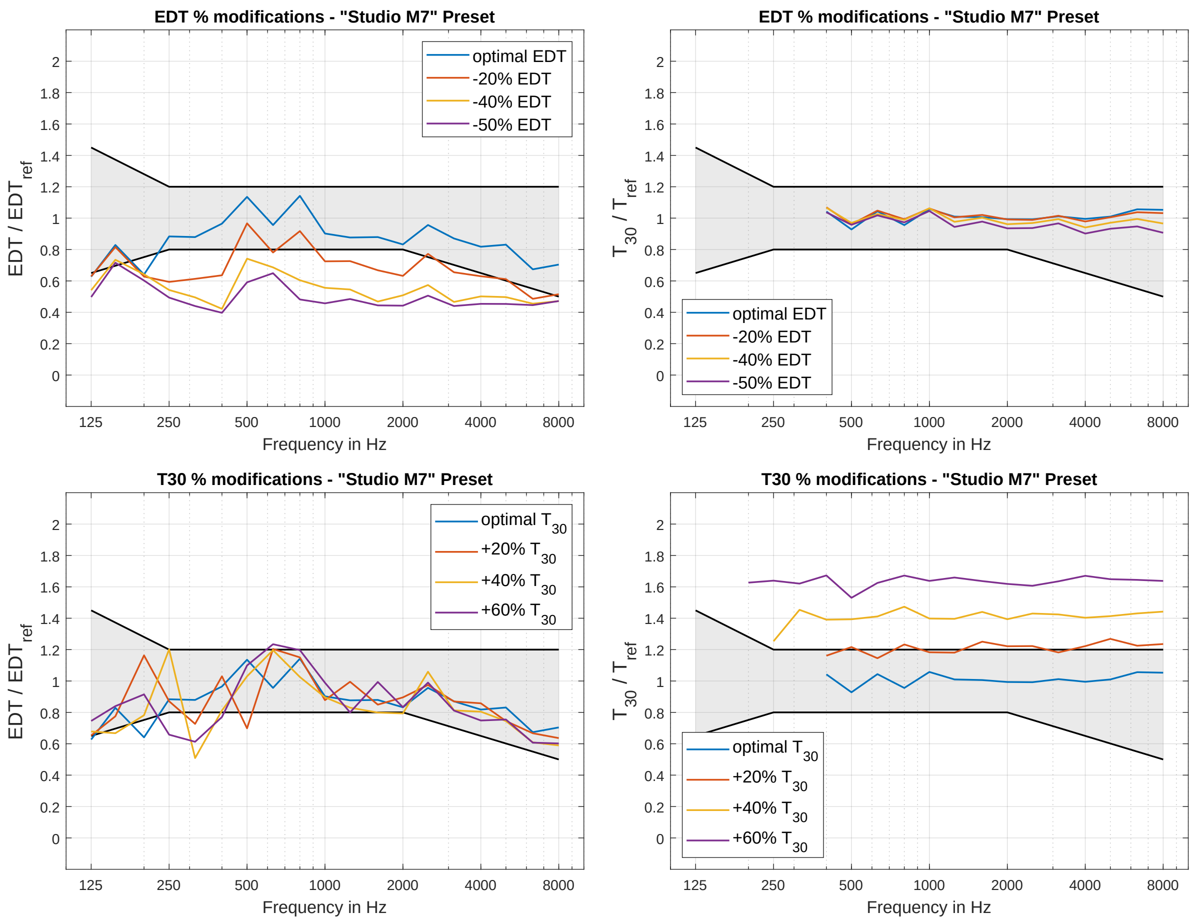

Figure 4.26.: Modification on EDT and T30 parameters for Studio preset of the M7 processor.

4.2.2. Modification of Hall Presets

For the hall presets, the longer base reverberation times allowed FOR smaller and more controlled changes on the processor without reaching its functional limits. For this room category and the next one, the variation on the processor's control were applied to achieve EDT reductions of -10%, -30%, and -50% while keeping T30 approximately constant (the non-target parameter is always kept within the 5% JND), and on the other hand, increases of +10%, +30%, and +50% to the T30 while keeping EDT constant where done.

- **Hall EDT modification**: optimal EDT > -10% EDT > -30% EDT > -50% EDT.
- **Hall T30 modification**: optimal T30 > +10% T30 > +30% T30 > +50% T30.

TrueVerb - Hall Preset

The only type of processor qualified for this room category was the TrueVerb. The details of the modifications made to its Brahms preset are specified in Table 4.11, keeping track of the exact variation on the processor controls and how these changes affect the room acoustic parameters. For the EDT modification, "Decay" was constant at 1.2 [s] while "Early" was increased from -8.5 to 0 [dB]. The resulting EDT decreases from 1.15 [s] to 0.58 [s] while "T30" stays close to 1.26 [s] with a small variation of between 0 to 2%. For the T30 modification, the "Decay" control was raised from 1.2 [s] up to 1.9 [s] to reach a +56% increase on T30, while the "Early" control was moved toward 0 [dB] to keep EDT near the 1.15 [s] mark.

True Verb - "Hall" Settings

Sample Name	Processor Settings			Room Acous. Parameters			
	Decay	Early	Reverb	T30	Δ	EDT	Δ
EDT optimal	1.2	-8.5	-5.0	1.26	0%	1.15	0%
EDT -10%	1.2	-4.9	-5.0	1.27	1%	1.04	-10%
EDT -20%	1.2	-2.8	-5.0	1.27	1%	0.92	-20%
EDT -30%	1.2	-1.2	-5.0	1.28	2%	0.81	-30%
EDT -40%	1.2	0.0	-5.2	1.28	2%	0.69	-40%
EDT -50%	1.2	0.0	-6.2	1.27	1%	0.58	-50%
T30 +50%	1.9	0.0	-6.2	1.97	56%	1.16	0%
T30 +40%	1.7	0.0	-6.2	1.76	46%	1.16	0%
T30 +30%	1.6	0.7	-6.2	1.65	31%	1.15	0%
T30 +20%	1.5	-1.7	-6.2	1.55	23%	1.15	0%
T30 +10%	1.3	-4.7	-5.0	1.36	8%	1.15	0%
T30 optimal	1.2	-8.5	-5.0	1.26	0%	1.15	0%

Table 4.11.: Processor settings and changes in T30 and EDT values for TrueVerb, Brahms Preset

Figure 4.27 shows how both types of modifications behave across the frequency spectrum. On the top left, the EDT curves for -10%, -30%, and -50% can be visualized, and on the top right, the T30 curves during the EDT modification remain near the optimal value. On the bottom left, during the T30 modification, the EDT curves stay relatively close to the optimal line, but with small deviations in low frequencies. On the bottom right, the T30 curves show the intended variation steps as near parallel curves across frequency.

4.2.3. Modification of Church Presets

For the church presets, the modification strategy is the same one applied to the hall presets; EDT reductions of -10%, -30%, and -50% while keeping T30 approximately within the 5% JND, and T30

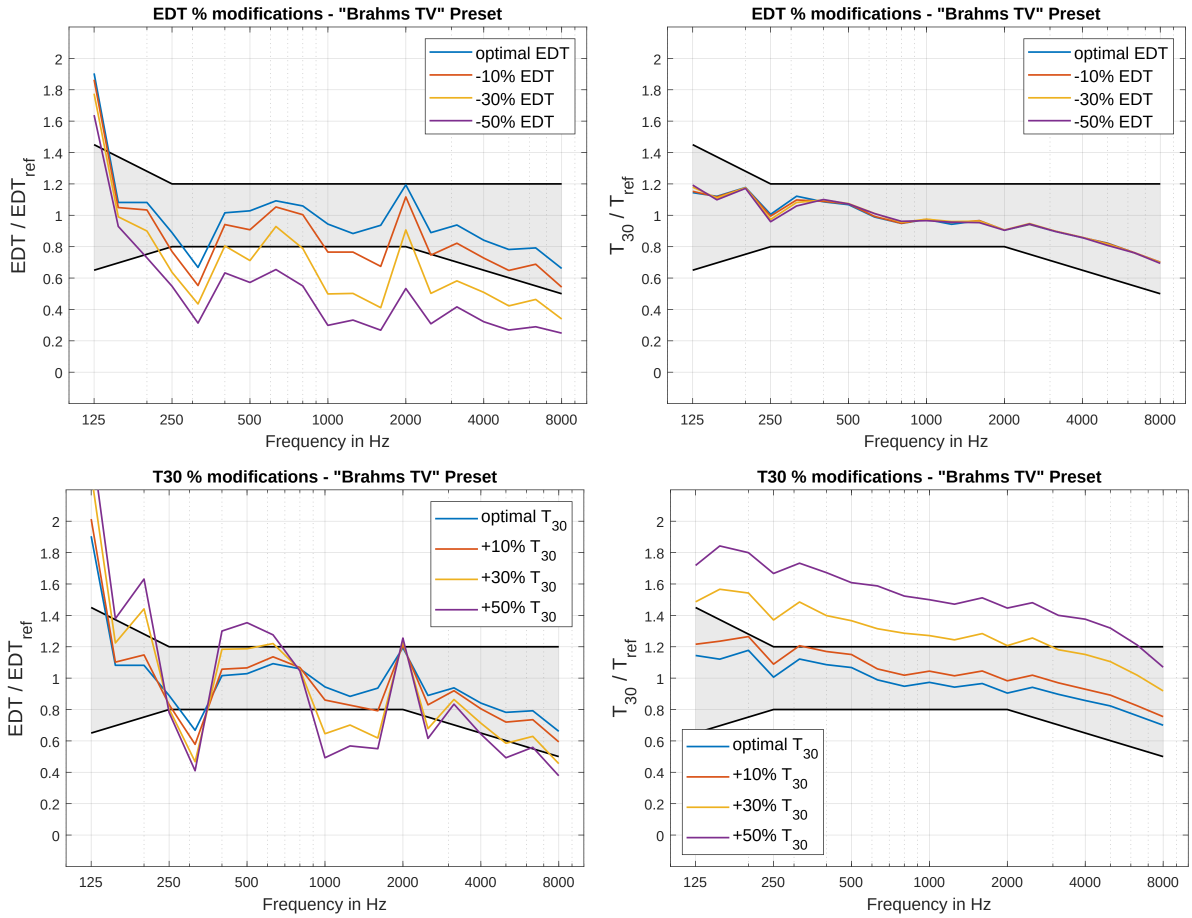

Figure 4.27.: Modification on EDT and T30 parameters for Brahms preset of the TV processor.

increases of +10%, +30%, and +50% while maintaining EDT approximately within the 5% JND. All time quantities are reported in [s].

- **Church EDT modification**: optimal EDT $>$ -10% EDT $>$ -30% EDT $>$ -50% EDT.
- **Church T30 modification**: optimal T30 $>$ +10% T30 $>$ +30% T30 $>$ +50% T30.

TrueVerb - Church Preset

Table 4.12 specifies how the variation on the processor controls and affects the room acoustic parameters for the Church room preset of the TrueVerb processor. For the EDT modification, the "Decay" control is kept at 1.9 [s], the "Reverb" control is kept at -10.7 [dB], and the "Early" control is modified by stages from -13.6 [dB] until -4.2 [dB]. The resulting EDT decreases from 1.62 [s] to 0.81 [s], which means a 0% to -50% variation, while "T30" stays close to 1.79 [s] with only -1% variation. For the T30 modifications, the "Decay" control is raised from 1.9 [s] up to 2.9 [s] to reach +54% on T30, with the "Early" control brought up to -4.5 [dB] to maintain EDT near 1.62 [s].

Figure 4.28 shows how both types of modifications behave across frequency. On the top left, the EDT curves for EDT modifications show the desired change, and on the top right, the T30 curves during the EDT modification stay close to the optimal curve. On the bottom left, during the T30 modification, the

True Verb - "Church" Settings

Sample Name	Processor Settings			Room Acous. Parameters			
	Decay	Early	Reverb	T30	Δ	EDT	Δ
EDT optimal	1.9	-13.6	-10.7	1.79	0%	1.62	0%
EDT -10%	1.9	-10.4	-10.7	1.79	0%	1.46	-10%
EDT -20%	1.9	-8.5	-10.7	1.79	0%	1.30	-20%
EDT -30%	1.9	-6.9	-10.7	1.79	0%	1.13	-30%
EDT -40%	1.9	-5.5	-10.7	1.78	-1%	0.97	-40%
EDT -50%	1.9	-4.2	-10.7	1.78	-1%	0.81	-50%
T30 +50%	2.9	-4.5	-10.7	2.75	54%	1.62	0%
T30 +40%	2.7	-5.4	-10.7	2.56	43%	1.63	1%
T30 +30%	2.5	-6.5	-10.7	2.37	32%	1.63	1%
T30 +20%	2.3	-7.9	-10.7	2.18	22%	1.62	0%
T30 +10%	2.1	-9.9	-10.7	1.99	11%	1.62	0%
T30 optimal	1.9	-13.6	-10.7	1.79	0%	1.62	0%

Table 4.12.: Processor settings and changes in T30 and EDT values for True Verb, Church Preset

EDT curves remain near the optimal value, and on the bottom right, the intended +10%, +30%, and +50% changes in T30 are achieved across frequency.

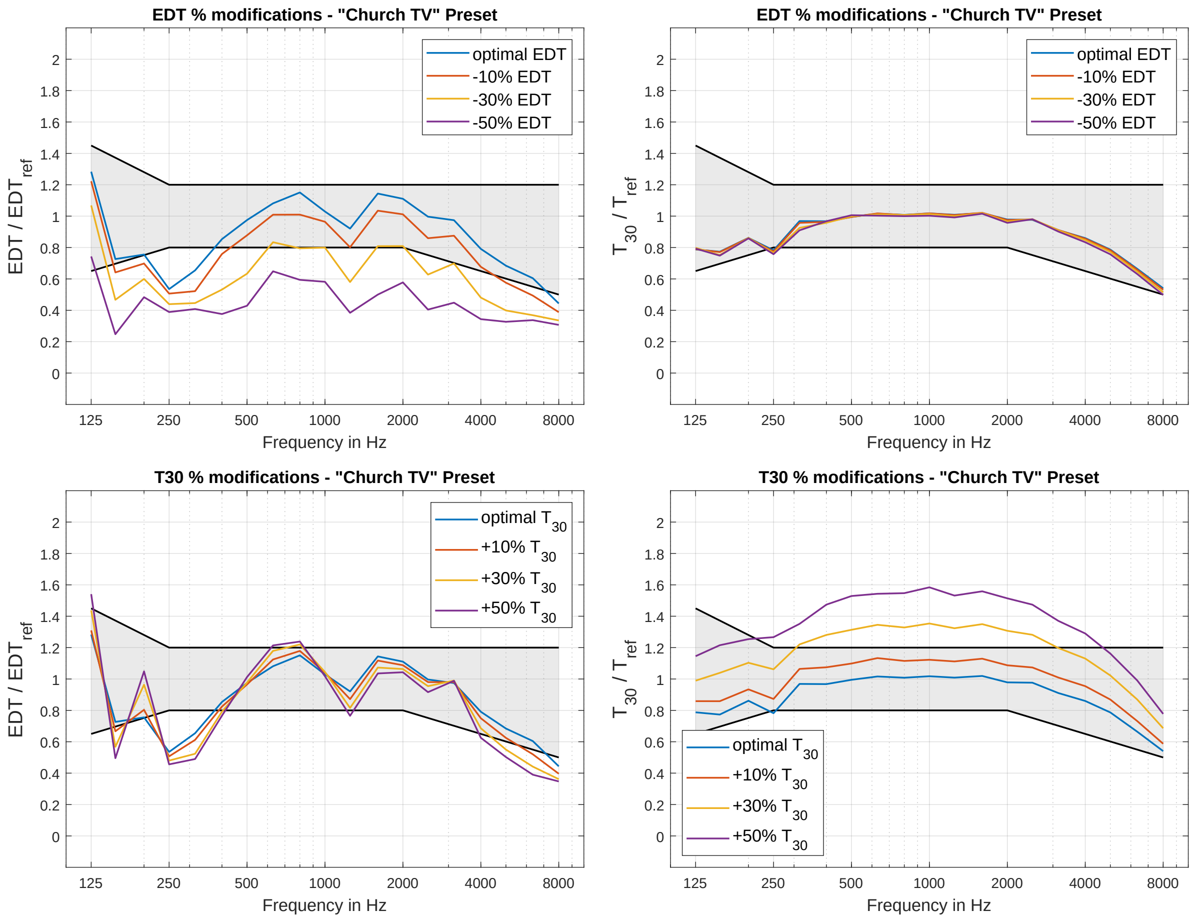

Figure 4.28.: Modification on EDT and T30 parameters for Church preset of the TV processor.

Bricasti M7 - Church Preset

For the Bricasti M7's Church Preset, Table 4.13 shows the details of the modification process. The "Rvb Time" control was maintained at 2.05 [s], and the "Early / Reverb" mix was modified from 12 to 5 on the reverb side, for the EDT modification, resulting in a reduction of -57% in EDT while T30 stayed constant. For the T30 modification, the "Rvb Time" control was raised from up to 3.10 [s] to reach a +50% increase on T30. Figure 4.29 shows that T30 curves remained constant across the EDT modification, but the EDT curves presented small variations in the extreme ends of the spectrum during the T30 modifications.

Bricasti M7 - "Church" Settings

Sample	Processor Settings			Room Acous. Parameters			
Name	Rvb Time	Early	Reverb	T30	Δ	EDT	Δ
EDT optimal	2.05	20	12	2.01	0%	1.95	0%
EDT -10%	2.05	20	10	2.02	0%	1.76	-10%
EDT -20%	2.05	20	8	2.02	0%	1.51	-23%
EDT -30%	2.05	20	7	2.02	0%	1.31	-33%
EDT -40%	2.05	20	6	2.02	0%	1.07	-45%
EDT -50%	2.05	20	5	2.01	0%	0.84	-57%
T30 +50%	3.10	20	5	3.01	50%	2.04	5%
T30 +40%	2.90	20	6	2.83	41%	2.12	9%
T30 +30%	2.65	20	7	2.61	30%	2.08	7%
T30 +20%	2.45	20	8	2.50	19%	2.05	5%
T30 +10%	2.25	20	10	2.21	10%	2.02	4%
T30 optimal	2.05	20	12	2.01	0%	1.95	0%

Table 4.13.: Processor settings and changes in T30 and EDT values for M7, Church Preset

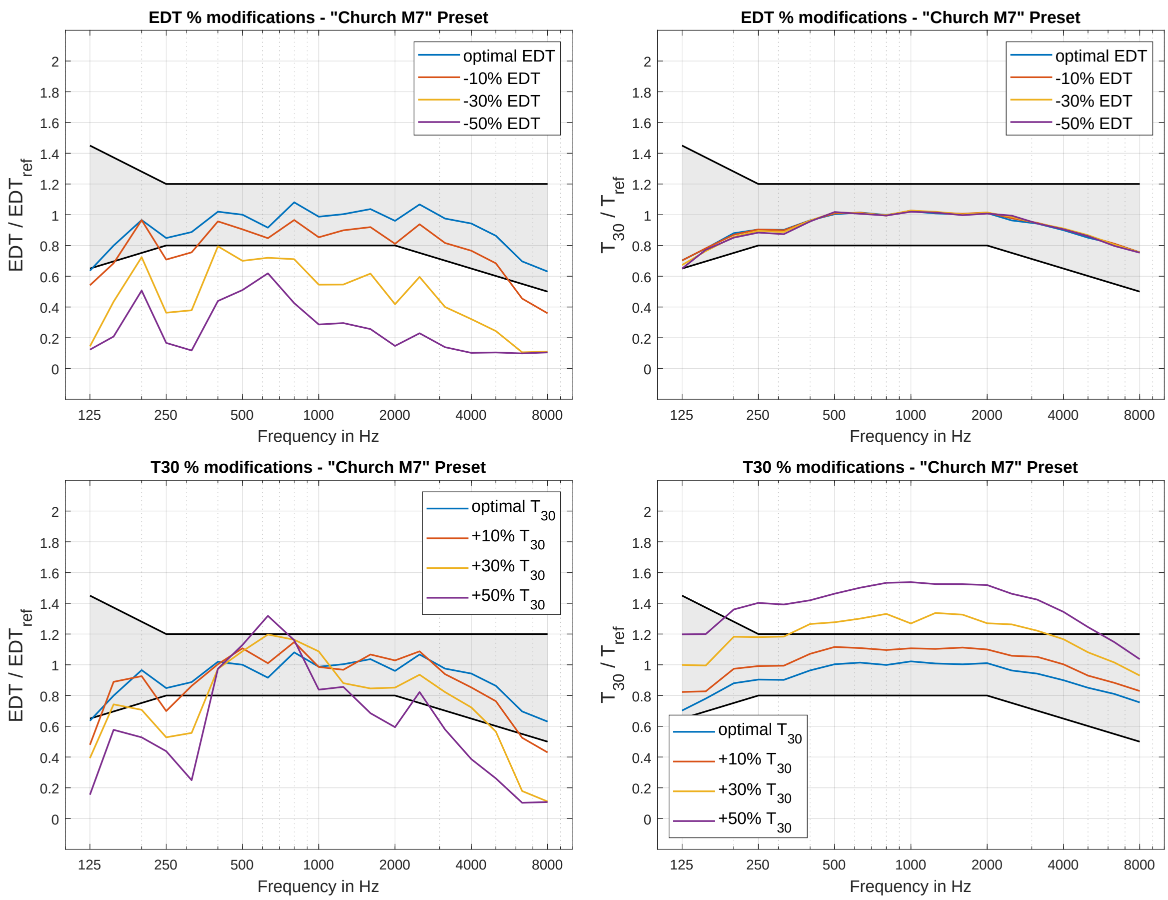

Figure 4.29.: Modification on EDT and T30 parameters for Church preset of the M7 processor.

4.3. Results of Listening Test

This section presents the results of the listening test on the perception of naturalness in reverberation. 32 participants took part in the test, but following the MUSHRA recommendations [12], in analyzing the data, 9 participants were excluded for not meeting the requirement of recognizing and rating the hidden reference with at least an 80% score, more than 15% of the time. The analysis shows first the total distribution in box plots, and then breaks down the results by rooms and instruments.

4.3.1. General Distribution of Scores

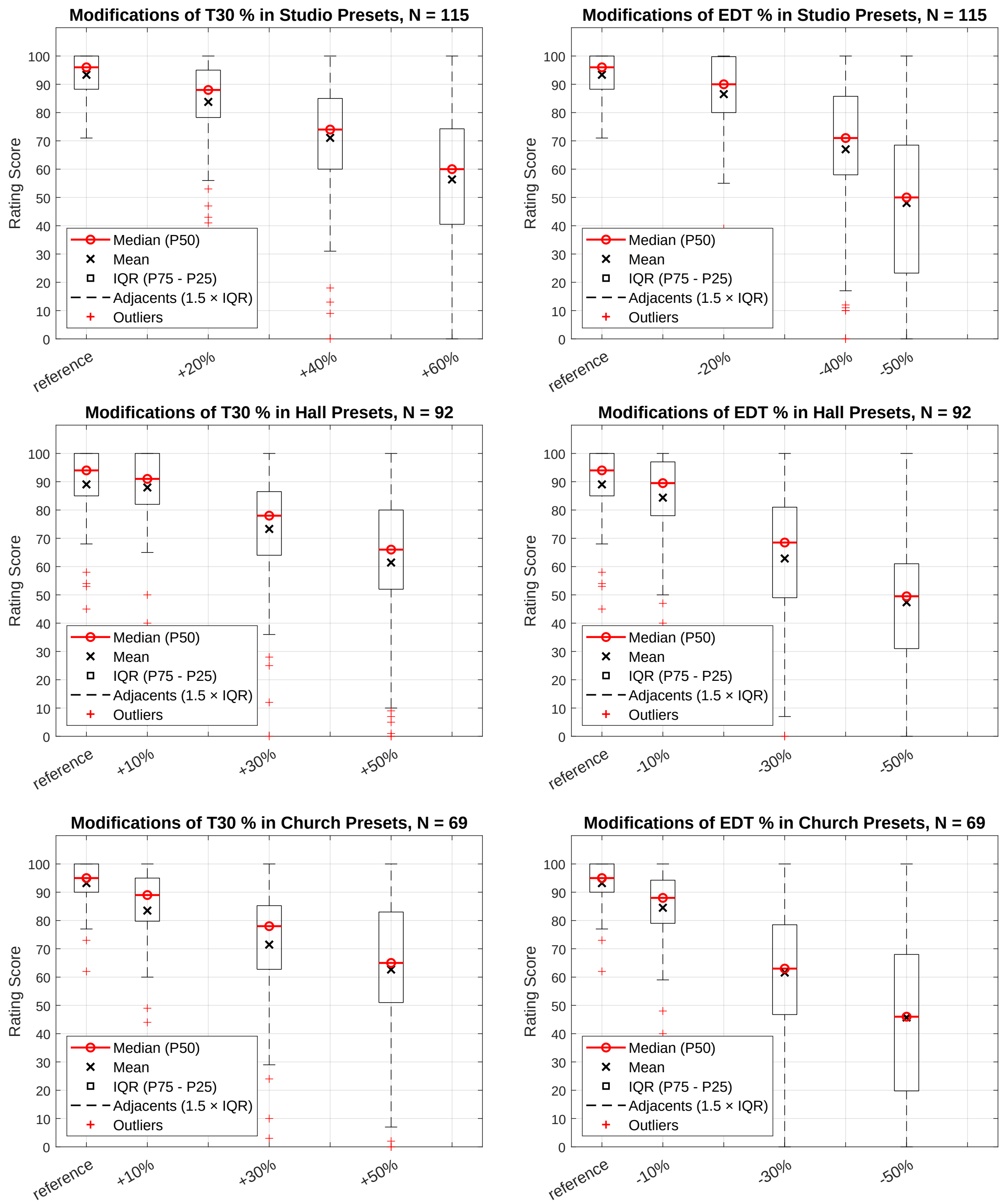

Figure 4.30.: T30 and EDT rating scores across all rooms.

Each graph from figure 4.30 depicts the resulting MUSHRA test score of all accepted participants, separated by room preset category, and split into the type of parameter modification, wheter that be T30 or EDT, both sharing the same reference. The title indicates the type of modification, the type of room present, and the N represents the total amount of scores recorded for each particular boxplot in that plot. The graphs display first the reference condition starting from the left of the x-axis, and continuing with each incremental modification done to the corresponding room acoustic parameter. The vertical axis is the rating score from 0 to 100. The legend indicates the plotted statistics: Median (P50), Mean, IQR (P75–P25), Adjacent values (1.5 × IQR), and Outliers.

The first two graphs at the top of the figure show the score distribution across all Presets emulation of the Studio room, independent of the type of instrument or processor used in the emulation. N is 115 for the studio because the 5 instruments are multiplied by 23 participants. The median score for the reference sample is approximately 95%, with the interquartile range (IQR) relatively tight above 90% score. For the 20% modification condition, the median decreases slightly for both T30 and EDT, but only by 5% of the reference; however, the IQR starts to spread, and more outliers are found in the lower range. For the 40% modification condition, a difference between T30 and EDT perception scores starts to be appreciated. The median and the whiskers for the EDT modification decrease further. Finally, for the -50% EDT modification, a huge dip in the median score can be seen, as well as a big IQR, indicating a large spread of votes. In comparison, the +60% T30 condition continued the steady low trend decline of its previous modification stages. These results show that the median score decreases across the x-axis, and the spread of votes increases, with a particular dip for the EDT modification.

The two middle graphs on Figure 4.30 show the distributions of the Hall room presets, independent of the type of instrument or processor used in the emulation. N is 92 as indicated in the figures (4 instruments times 23 participants). Both plots show a median rating above 95% for the median that decreases across the x-axis, but the EDT modification plot shows a steeper decline, with a median rating score of 70% for the 30% modification condition, and of 50% rating score for the 50% modification condition. Compared to the median scores of 80% and 65% of the T30 modification graph. As well, the spread of the IQR and the adjacent values is larger for the EDT plots. However, the T30 graph has many more outlier values that are in the lower range of the rating. This room in general, has a tighter IQR than the Studio or Church room. The church room graphs show an N value of 69, composed of 3 instruments times the 23 participants. The trend in median decline is similar to the one observed in the Hall room; the EDT median decreases more sharply than the T30. However, the spread across ratings for this room is larger, with the lower end of the IQR, the 25th % point of the votes, positioned at the 20% rating score mark, the lowest for all the test samples.

For comparison, the same vote distribution plots for all the 32 participants, including the excluded ones, can be found in the Appendix F. From there, it can be seen that while the score plots in Figure 4.30 show a skew distribution with median values generally larger than the mean, which indicates that most scores are above the average, for the graph containing all participants, the mean tends to be equal to the median. As well, in the Appendix, the distribution graph for every single test question can be found, meaning the result for each one of the 12 qualified samples, and its corresponding conditions with the modified parameter. This can be helpful if any particular combination of room, instrument, and processor is wished to be analyzed.

Finally, in Figure 4.31, a comparison between T30 and EDT median trends for each room is displayed, where it can be seen that generally, and especially in the larger room, the rating of the EDT modification tended to be lower than the rating of the T30 modifications.

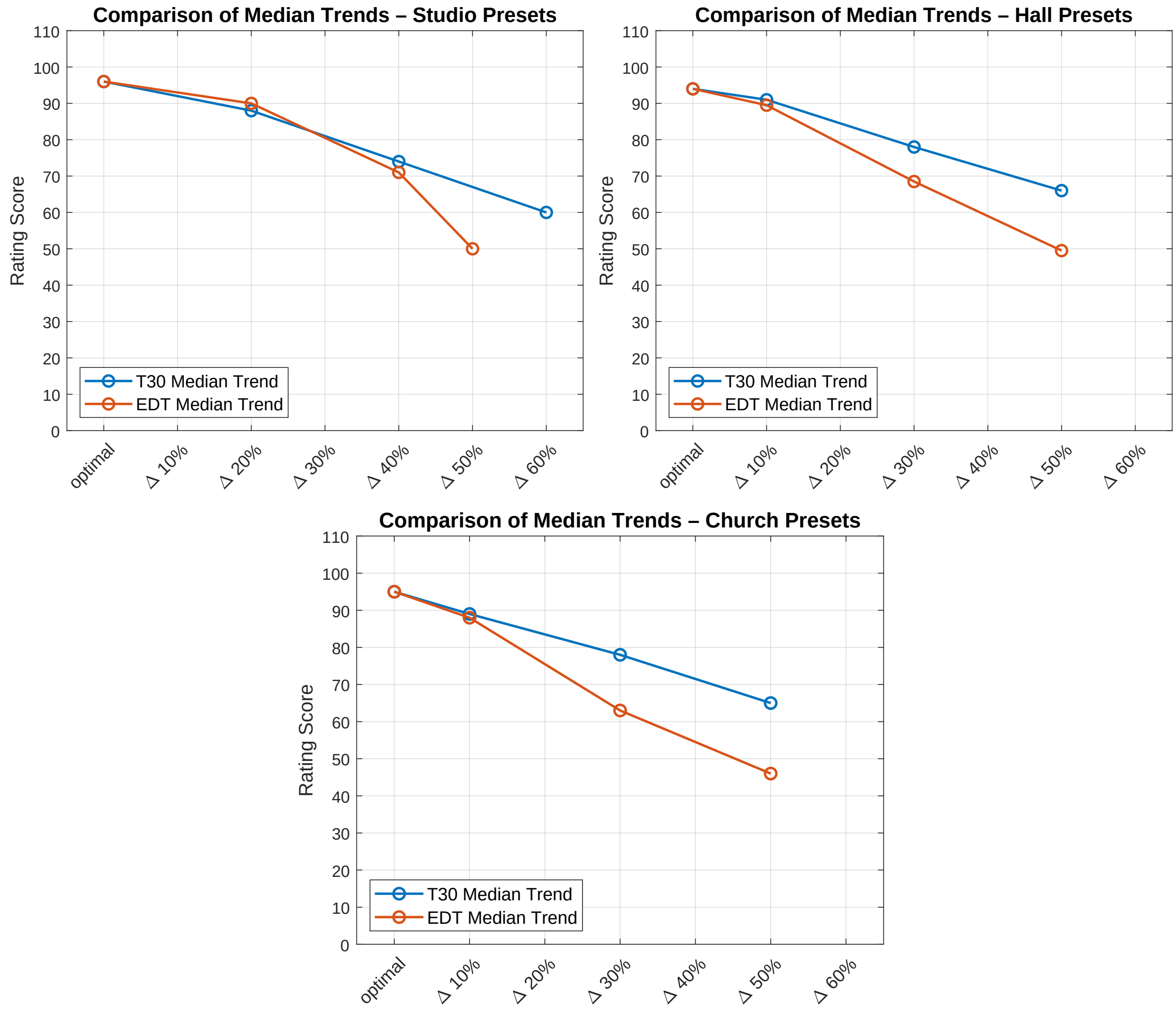

Figure 4.31.: Comparison of T30 and EDT average median rating scores across all rooms.

4.3.2. Rating Score Distribution across Rooms

This section describes the median score rating trends for every room across the different instruments for the two types of modifications, T30 and EDT. Each graph in the Figure 4.32 displays the median trends in score ratings across the x-axis from the reference at the left through the successive parameter modification stages.

For the Studio room, the reference median score sits near 95% for all the instruments, and while this score decreases with the accordingly with parameter modifications for both T30 and EDT, For the EDT we can observe the same dip in rating from the -40% to -50% modification observe in Figure 4.30 across all instruments, but for the T30 modification, 2 particular instruments (E-gtr and Legato) sit scored noticeable above the others in rating for the most extreme modification of +60% T30. There is no apparent difference in rating score between the conditions created with the TrueVerb processor and the one created with the Bricasti M7.

For the Brahms room, the reference, as always, is located in the upper range of the scale. For the T30 modification, the score ratings of the median trends are spread generally equally across the x-axis, with no particular instrument standing out as good or bad rated. The EDT graph, on the other hand, shows a clear lower median rating score for the E-gtr and the Staccato for the -30% modification condition.

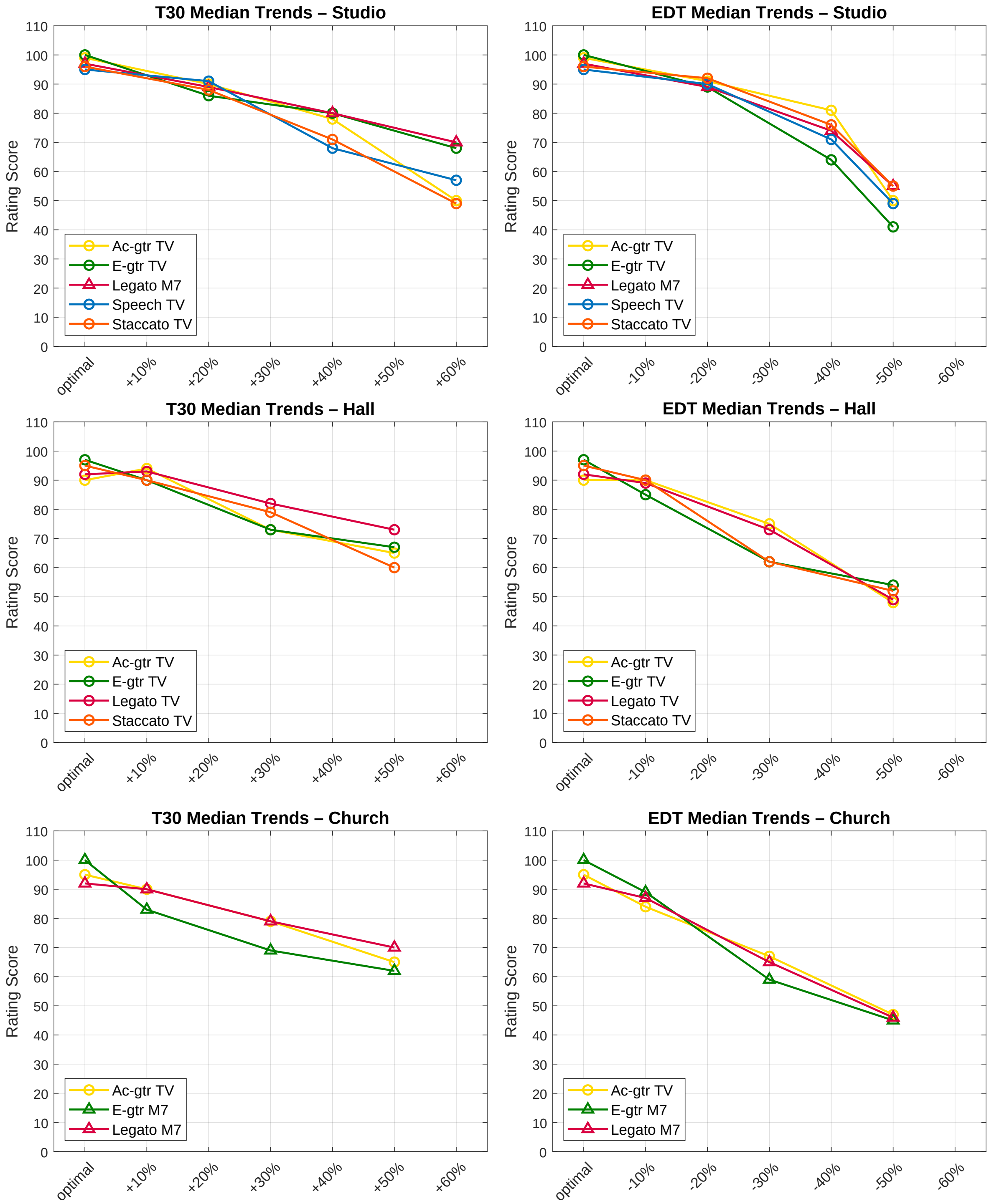

Figure 4.32.: T30 and EDT median rating scores for every instrument across all rooms.

The median trend curves of the score ratings for the Church room across all instruments begin with the reference on the upper portion of the scale, showing a steady decrease for the T30 modification graph, with a clear lower rating for the Electric guitar instrument, while the EDT modification graph presents no clear difference in between the different instruments, but it does show a sharp overall decrease in score rating for all. No clear pattern difference can be perceived between the sample created with the True Verb proccessor and those created with the Bricasti M7.

4.3.3. Rating Score Distribution per Instrument

This section describes the median score rating trends for every instrument across the different rooms for both types of modification, T30 and EDT. Each graph displays the median trends in score ratings across the x-axis from the reference at the left through the successive parameter modification stages. The top graphs in Figure 4.33 show the median trend for the Acoustic guitar for both the T30 and EDT modifications. In general, the Studio room tends to have a higher rating than the Hall and Church rooms for the EDT modifications, but not in the T30 modification conditions. The bottom right graph show a similar distribution of score rating for the Electric guitar for the EDT conditions, with a clearly higher median curve for the Studio room, but in difference to the Acoustic guitar, in the T30 conditions, the Electric guitar median trends in the studio also shows a noticeable difference in comparison to the other two rooms.

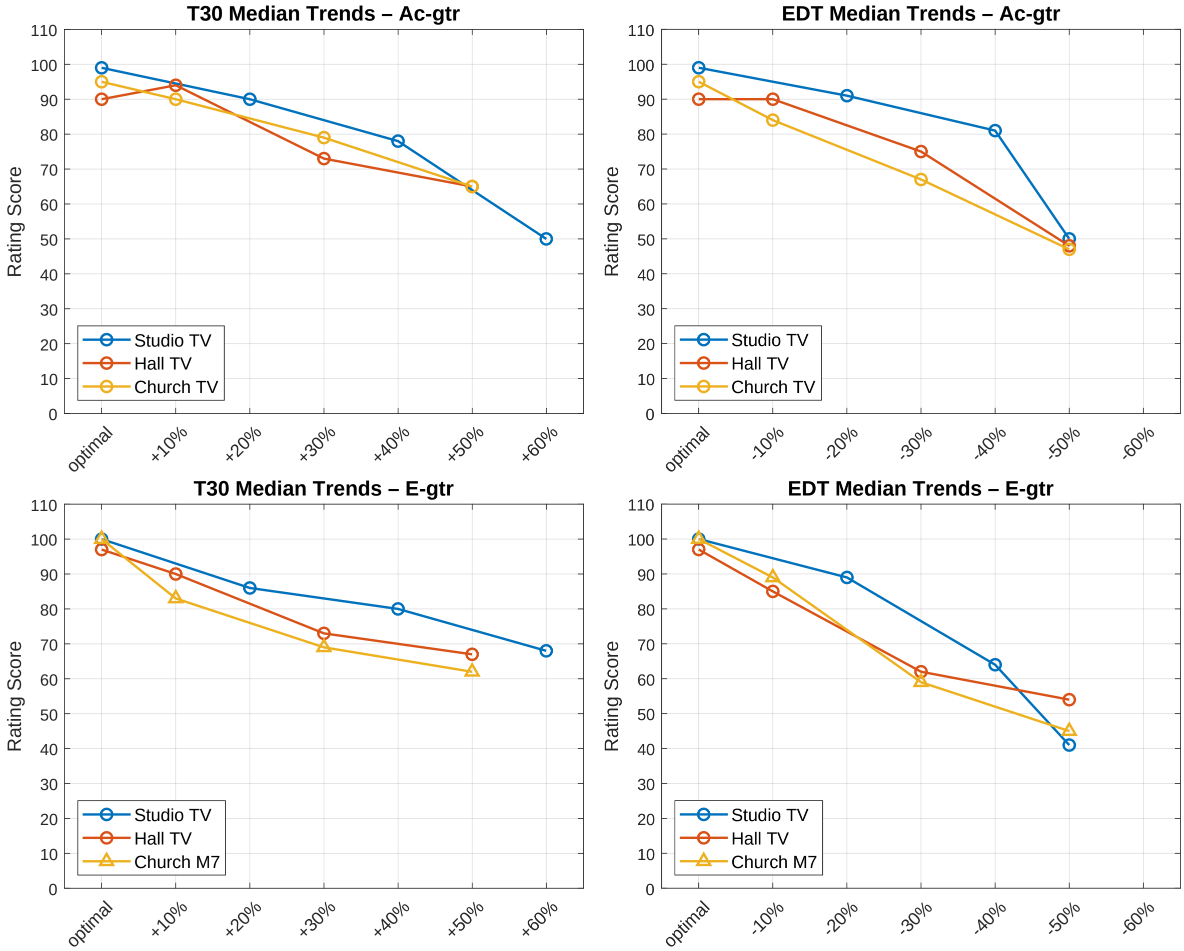

Figure 4.33.: T30 and EDT median rating scores for Ac-gtr and E-gtr across rooms.

The acoustic guitar medians tend to generally lie above the electric guitar median trend curves, especially for the Studio room and the Hall room. For the Church room, however, both instruments show equally low median trends. Across both instruments and all rooms, the EDT median plots present a sharp drop in score rating for the most extreme modification conditions, while the T30 median curves decline more steadily.

The Figure 4.35 shows median trends for the singing voices samples, meaning Legato and Staccato. The T30 modification for both instruments presents the most stable decrease in score rating out of all the instruments, with virtually no difference across rooms for both types of signals. For EDT modifications however, the Studio room score rating curve is above those of the Hall and Church rooms, with a specially large gap for Staccato at the -30% modification point (is important to consider that the Studio room does not have an actual condition at that modification point, its just an approximation via the adjacent points' median score rating).

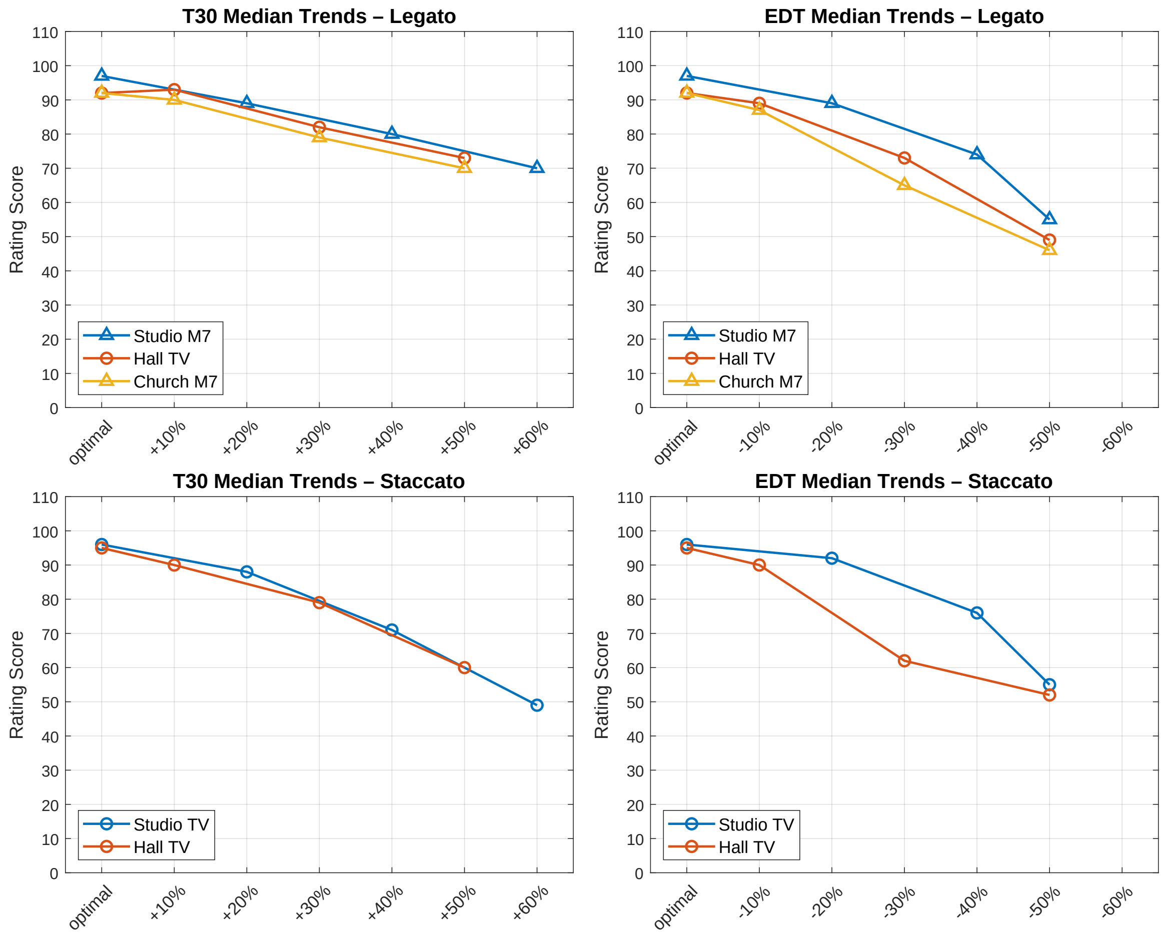

Figure 4.34.: T30 and EDT median rating scores for Legato and Stacato across rooms.

Finally, the Figure 4.35 illustrates the special case of the Speech signal, representing the only case of speaking voice in the Listening Test. While the median rating for both types of modifications, T30 and EDT, decreases with the increase of the conditions, the reduction for T30 is more progressive, while the EDT score reduction occurs more suddenly with the last two conditions, a trend that repeats across most categories of instruments or rooms.

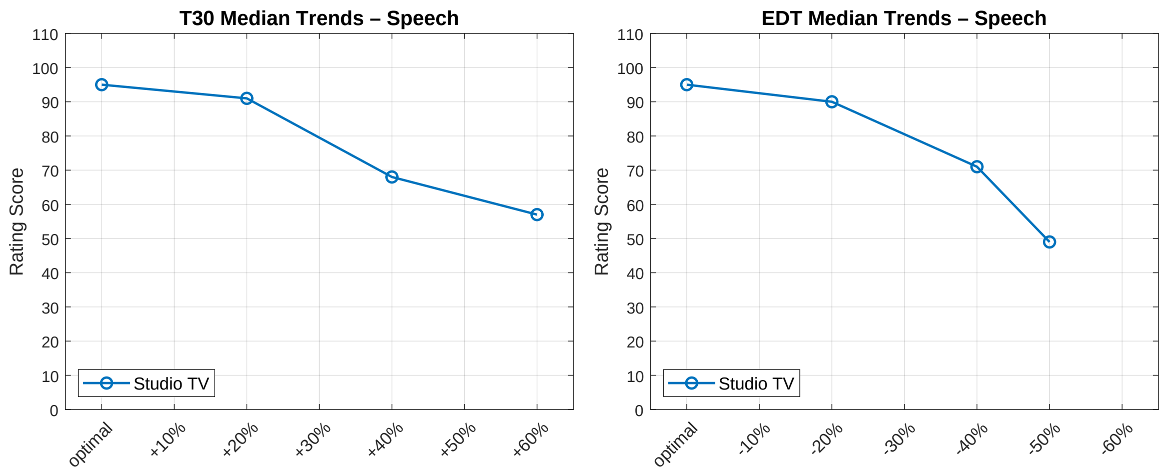

Figure 4.35.: T30 and EDT median rating scores for Speech for Studio Room.

4.3.4. Participants Survey

The last experimental process of this study consisted of a short survey post-listening test to get the feedback from the participants and their perception of the test's difficulty. The questions were completely dependent on the participants' own experience realizing the test, asking them to rate on a scale of 1 to 5, whether the test was very easy, easy, neutral, hard, or very hard to complete. The rating scale is represented on the y-axis, with the x-axis showing the name of the object in question. For the overall listening test question, the distribution of answers ranges from "Very Easy" to "Neutral", with the median answer being "Easy'. For the question regarding the difficulty evaluating the different rooms, the Studio room shows the highest answer spread, going from almost "Very Easy" to almost "Very Hard". The Hall room was rated highly consistent as being "Hard" to evaluate, while the Church room was almost universally rated as "Easy" to evaluate.

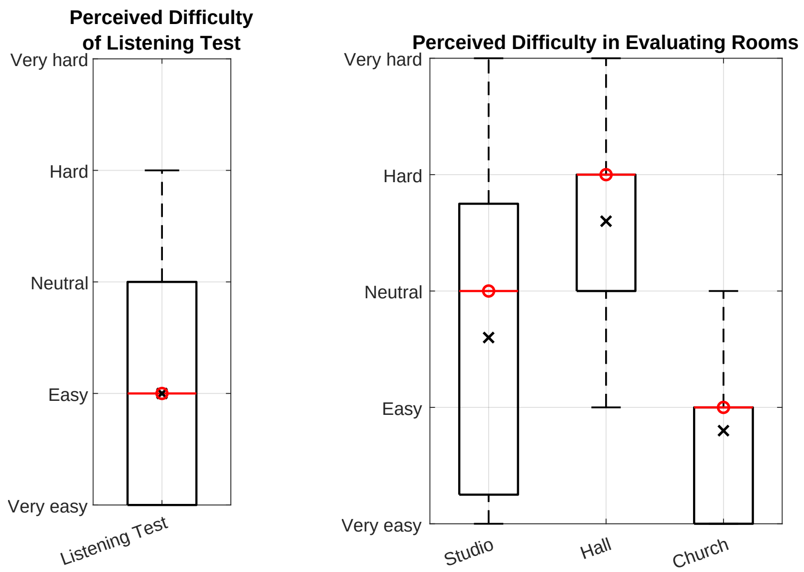

Figure 4.36.: Participants' perceived difficulty in performing the Listening Test and Evaluating Rooms

For the question regarding the difficulty of evaluating the different instruments, both of the singing signals (Legato and Staccato) were generally considered "Easy" to evaluate. The Speech and E-gtr signals were considered "Neutral" in difficulty to evaluate, but both presented a large spread of answers. The acoustic guitar was considered the most "Neutral" instrument overall.

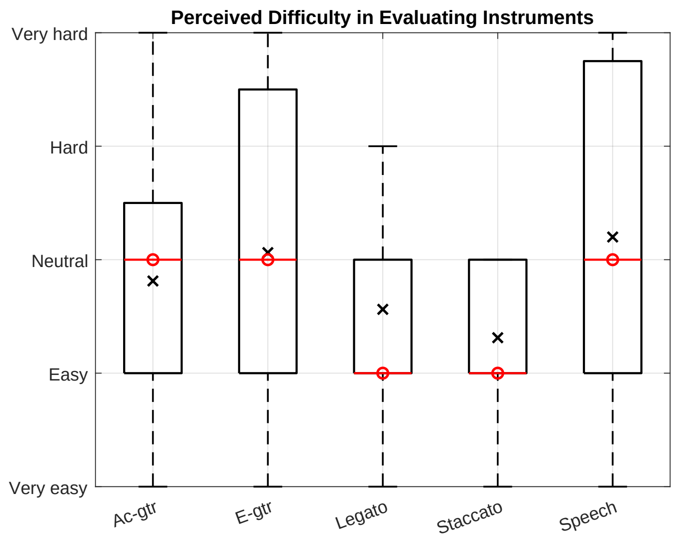

Figure 4.37.: Participants' perceived difficulty in evaluating the different Instruments.

5. Discussion and Conclusions

5.1. Interpretation of Findings

This work set out to analyze how controlled changes in room acoustic parameters, particularly *T30* and *EDT*, affected the perceived naturalness of artificial reverberation, with the goal of deriving objective recommendations and guidelines for creating reverberation presets for music production and mixing contexts (Chapter 1). The experimental strategy was separated into two steps: first, a qualification test to identify artificial samples that listeners have already judged to be natural sounding, and a second MUSHRA-based rating test in which only those qualified references were systematically modified in their *T30* or *EDT* parameters and then re-evaluated, as explained in Chapter 3.

The qualification results show that 65% of the 60 artificial samples were rated above the 50% naturalness threshold, which gives validity to this step. The best-ranked samples concentrate in the *Studio* and *Hall* rooms. By instrument, *Acoustic guitar* had all samples above the acceptance threshold and the tightest spread, *Legato* followed closely, *Electric guitar* was mixed received, *Staccato* stayed near the middle of the rank, and *Speech* was the weakest with only 17% of samples above the threshold. These varying outcomes justify the selection of several room and instrument combinations to achieve a wide possible of results.

It was clear from the results in Chapter 4 that listeners are more sensitive to *EDT* modifications than to *T30* deviations. In the listening test, median trends and distributional statistics consistently show that decreasing *EDT* leads to a more pronounced decrease in naturalness ratings than the increase of *T30* for the same reference, with the effect getting more pronounced the larger the room. The box plots results across rooms show that *EDT* modifications produce larger interquartile spreads and lower minimum values than *T30* modifications. This pattern is visible in the per-room figures 4.30 and is corroborated in the per-sample figures for the same references. For example, in “ST a-gtr TV”, where the *EDT* decreases more dramatically than the corresponding *T30*.

Because *EDT* contains the initial 0 to 10 dB portion of the energy decay, it is highly linked to the initial perception of reverberation; listeners appear to identify small changes in that early stage of reverberation more easily than for the late reflections.

5.2. Effect of Room and Instrument Type

The qualification test resulted in the processor’s naturalness ranking, and those processors that did not meet the expected naturalness ranking were left out of the following stages of the study, in that way, it could be ensured that the second listening test, about the perception of naturalness in reverberation, could go on without the conderation of the processor as a main factor, and in that way focus only on the type of the room and instrument combination, along with the room acoustic parameter’s modifications.

Regarding the qualification test process, in broad terms, it was observed that musicians were most insistent and questioned many things about the test, making many suppositions and scenarios in their heads, while tonmeisters were more easy-going. This could be because of their familiarity with how listening tests work.

Room effects. For the qualification test, many people mentioned that the Studio Room was the least natural of them all, but this is in contrast with the results of that room being the most polarizing. Also, it was frequently mentioned that the church was the hardest to judge because the test samples mostly sounded natural enough.

In the Perceptual Listening Test, the *Hall* presets maintained high reference medians and narrower IQRs across modifications, but still showed the characteristic *EDT* sensitivity. The *Studio* showed the widest spread overall, including best and worst cases at the qualification stage, and a clear difference between *T30* and *EDT* only at the extreme modifications of 40 to 50%. The *Church* room showed the sharpest declines of *EDT* rating reductions and the broadest IQRs, with the 25th percentile reaching only 20% rating at the strongest modification

Instrument effects. For the qualification test, the samples of the spoken voice were received poorly, probably due to a proximity effect present on the recording, which made low-frequency sounds stand out in a non pleasant way. During the Perceptual Listening Test, in consistency with the qualification stage, *Acoustic guitar* and *Legato* tended to maintain higher ratings and tighter spreads than *Electric guitar* and *Staccato*, with *Speech* being the least robust overall. Per-instrument median trend figures indicate that *EDT* median trends drop for all instruments without a clear distinction.

5.3. Key Findings and Recommendations

Findings tied to the research question

- **Perceived importance of *EDT* versus *T30***. Across all combinations of samples, the sensitivity to EDT modifications was greater than that of T30 modifications. Changes in *EDT* produced steeper median decreases and wider dispersions from the same reference sample
- **Room context matters.** Larger, more reverberant rooms show stronger sensitivity to *EDT* modification. In the *Studio* presets, the difference between the two was not so huge except for the more extreme conditions.
- **Source ytpe matters.** Sustained and harmonic-rich instruments such as the (*Acoustic guitar* or the *Legato*) obtained a higher naturalness rating in most cases than the transient-rich signals like the (*Speech*, *Staccato*)

Practical recommendations for emulation rooms with artificial reverberation processors

- **Determining the Reverberation Time**. As described in Chapter 3, the processor's controls that manipulate reverberation time tend to be mostly straightforward. Whether they are named "Decay Time" or "Reverb Time", they are typically an analogous approximation of T30. This means that setting that value can be as easy as inputting the desired number.
- **Early reflection matching.** On the other hand, this parameter tends to be harder to define. It is not normally not directly represented in the processor's controls. Many names, such as "Balance", "Early", or "Mix", that are used in several processors, evoke a sense of the early energy behavior, but they tend not to be an accurate representation.
- **Room considerations.** Depending on the type of physical environment that one wishes to emulate, the differences between early and late reflection have to be noted, in that for short reverberation times, it is fairly enough to set up the "Decay Reverb" controls to achieve a natural representation of the desired room, but for larger and reflection-richer rooms, key details has to

be payed to all the other more ambiguous controls of the processor.

- **Source considerations.** For sources that do not have many pauses in between, meaning a more continuous pacing, the type of reverberation processor applied is not that important, as long as the basic reverberation parameters are defined. However, for more percussive signals, the early reflection part of the processor has time to come through in the mix signal, and could expose the artificial reverberation if the controls have not been satisfactorily set up
- **Loudness bias.** Maintaining a stable loudness between signals is key to avoiding bias when judging the naturalness of a source. When a signal is louder, the human ear tends to automatically associate it with sounding better.

Some limitations of this project are that, even though the experimental manipulations of the two parameters were done in an objective manner to avoid bias at any point of the research, and whenever subjective consideration had to be made, the decisions were later put to the test by the average results of the participants in the listening tests, some subjective assumptions may have slips through the process.

As well, the impossibility of using the Hall of Fame guitar pedal as a processor type was an oversight in the initial design of the methodologies, and more testing should have been done with this device before starting the measurement process and doing the qualification test with it. As well, errors in recording the spoken voice sample may have led to an underperformance of that particular source.

5.4. Conclusion and Future Work

The thesis asked whether there is a more scientific way, beyond "using the ears", to determine if artificial reverberation correctly reflects the naturalness of a real physical room environment, and which parameters matter most for that end. Within the analyzed conditions, the answer is partially affirmative; while naturalness judgments are systematically affected by the controlled changes in reverberation time and early reflections, with changes in *EDT* impacting more on the naturalness perception. These conditions are intrinsically tied to the source being processed and the room being emulated. So, in other words, the room acoustic parameters should not be used as an all-encompassing metric to determine naturalness, but in context with the desired outcome.

The study demonstrates that *EDT* is a primary perceptual parameter on the naturalness of artificial reverberation emulating real room environments, especially in larger spaces. *T30* remains important but plays more of a starting point in the perception of reverberation, while room type and instrument type modify this interaction. These insights can translate into guidance for day-to-day music production.

Building on these results, future research could be conducted by evaluating more directly the clarity parameters like *D50*/*C80*. Also, extended investigation with a greater variety of processors, sources, and emulated rooms, although that would increase the listening test times to a duration beyond the recommendation.

Appendix A

Extra Room Acoustic Parameters Graphs

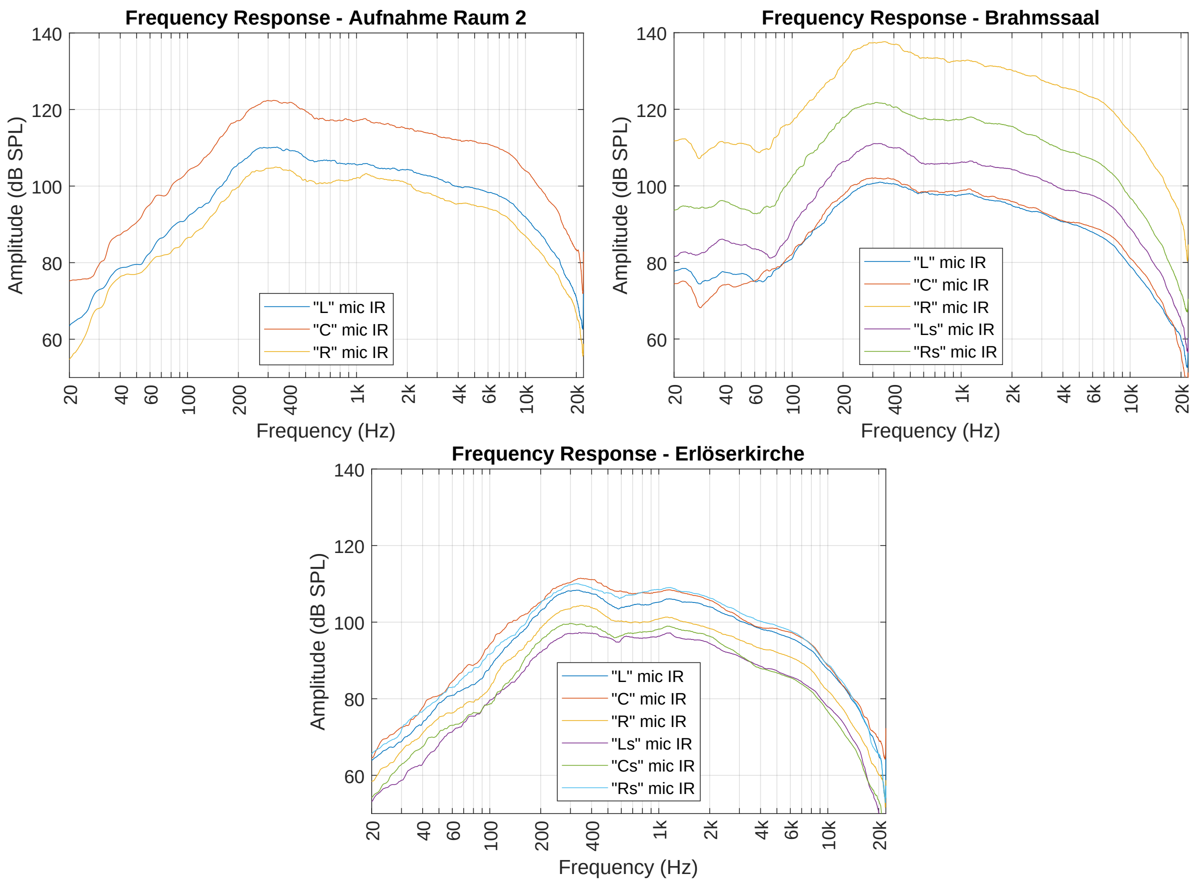

Figure A.1.: Frequency responses of all microphones for all Room Environments.

Appendix B

Processors Specifications

Parameters - Hall of Fame 2

Parameter	Control Range	Function
In Level	-100 to 0 dB	Adjusts the input gain of the reverb algorithm.
Fx Level	-100 to 0 dB	Sets the amount of effect mixed with the dry signal.
Octaver	-100 to 0 dB	Adds pitch-shifted reverb tail.
Kill Dry	On / Off	Removes dry signal.
Decay	20 ms to 20 s	Sets reverb tail length.
Pre Delay	0 to 300 ms	Sets the delay time before the reverb begins.
Diffuse	-50 to 50	Adjusts diffusion density (smoothness of tail).
LoColor	-50 to 50	Low-frequency EQ.
HiColor	-50 to 50	high-frequency EQ.
HiFactor	-25 to 25	Accentuates HiColor.
Modulation	On / Off	Enables modulation of the reverb tail.
Early	-24.5 to 0 dB	Level of early reflections.
Reverb	-24.5 to 0 dB	Level of late reverberation.

Table B.1.: Hall of Fame 2 Reverb Parameters

Parameter List – H-Reverb

Parameter	Control Range	Function
Pre-Delay	0 to 500 ms	Sets the delay time before the reverb begins.
Buildup Time	0 to 2 s	Time for reverb to reach peak level.
Reverb Size	0.5 to 2 x	Controls early reflection density.
Reverb Time	0.1 to 6 s	Duration for reverb decay.
ER Select	1 to 10 x	Selects an early reflection model.
ER/Tail Balance	0 to 100%	Adjust mix between early reflections and reverb tail.
Dry/Wet	0 to 100%	Adjust mix between direct and reverb signal.
Output	-24 to +18 dB	Controls output level.
Reverse	On/Off	Reverses reverb tail.
X-Time	0 to 100%	Crossfade between decay and release.
X-Gain	-120 to +12 dB	Gain at decay/release crossover point.
Density	0– to 100%	Controls smoothness in reverb tail.
Input Echoes	On/Off	Inserts echoes replicates of the input.
Output Echoes	On/Off	Inserts echoes replicates of the output.
Dynamics	On/Off	Applies compression to the wet signal.
ER Filter	1k to 14k Hz	Shelf EQ for early reflections.
LoShelf EQ	20 to 1000 Hz	Low shelf EQ for reverb tail.
LoBell EQ	60 to 4000 Hz	Low-mid parametric EQ for reverb tail.
HiBell EQ	1k to 15k Hz	High-mid parametric EQ for reverb tail.
HiShelf EQ	2k to 16k Hz	High shelf EQ for reverb tail.
Damping	On/Off	Adds damping for the reverb tail
Resonant Filter	On/Off	Envelope or LFO modes.
Modulation	On/Off	Add modulation to the reverb tail.
Drive	0 to 100%	Adds overdrive to the input.
Analog	On/Off	Enables analog modeling of reverb output.
Digital	Off/12/8 bit	Quantizes output for lo-fi effects.

Table B.2.: H-Reverb Parameters.

Parameter List – M7 Reverb

Parameter	Control Range	Function
Reverb Time	0.1 to 30 s	Sets the decay time.
Size	1 to 24	Controls the perceived size of the reverb.
Pre-delay	0 to 500 ms	Sets the delay time before the reverb begins.
Diffusion	1 to 9	Sets the echo density and smoothness of early reflections.
Density	1 to 9	Controls the speed of reflection buildup.
Modulation	1 to 9	Adds pitch modulation to the reverb tail.
Rolloff	80 to 28k Hz	Low-pass filter applied to the overall reverb output.
HF RT Multiply	0.2 to 1.0	Scales the reverb time of high frequencies.
HF RT Crossover	200 – 16k Hz	Crossover frequency for the HF RT multiplier.
LF RT Multiply	0.2 to 4.0	Scales the reverb time of low frequencies.
LF RT Crossover	80 – 4.8k Hz	Crossover frequency for the LF RT multiplier.
VLF Cut	0 to -18 dB	Reduces very low-frequency content in early and late reflections.
Early/Reverb Mix	0-20 / 20–0	Balances the early reflections and the reverb tail.
Early Rolloff	80 – 20k Hz	Low-pass filter for the early reflections.
Early Select	0 – 20	Selects type of early reflection.

Table B.3.: Bricasti M7 Reverb Parameters.

Parameter List – TrueVerb

Parameter	Control Range	Function
Dimension	1.00 to 3.99	Sets the pattern of early reflections.
Room Size	50 to 20000 m^3	Adjusts simulated room volume.
Distance	0.50 to 40.00 m	Adjust distance from source.
Link	On/Off	Links reverb and pre-delay to match early reflections.
Balance	-48.0 to +12.0 dB	Sets balance between early reflections and reverb tail.
Decay Time	0.2 to 100.0 s	Sets reverb decay time.
Pre-Delay	10.5 to 160.0 ms	Sets the delay time before the reverb begins.
Density	0.000 to 1.000	Controls buildup rate of reverb reflections.
ER Lowcut	16 to 512 Hz	Sets low filter for early reflections.
ER Absorb	1.6k to 21k Hz	Sets high filter for early reflections.
Rev Shelf	1.6k to 21k Hz	Sets high filter for reverb.
Low Damping	16 to 1.6k Hz	Sets low damping for reverb.
High Damping	1.6k to 21k Hz	Sets high damping for reverb.
Input Gain	-inf to 0 dB	Adjusts plugin input level.
Direct	-24.0 to 0.0 dB	Adjust direct signal output.
EarlyRef	-24.0 to 0.0 dB	Adjust early reflections output.
Reverb	-24.0 to 0.0 dB	Adjust reverb output.
EVar	0 to 6	Selects early reflection pattern variations.
RVar	0 or 1	Selects reverb pattern variations.

Table B.4.: True Verb Parameters

Appendix C

T30 relative to Room size for Processors' Presets

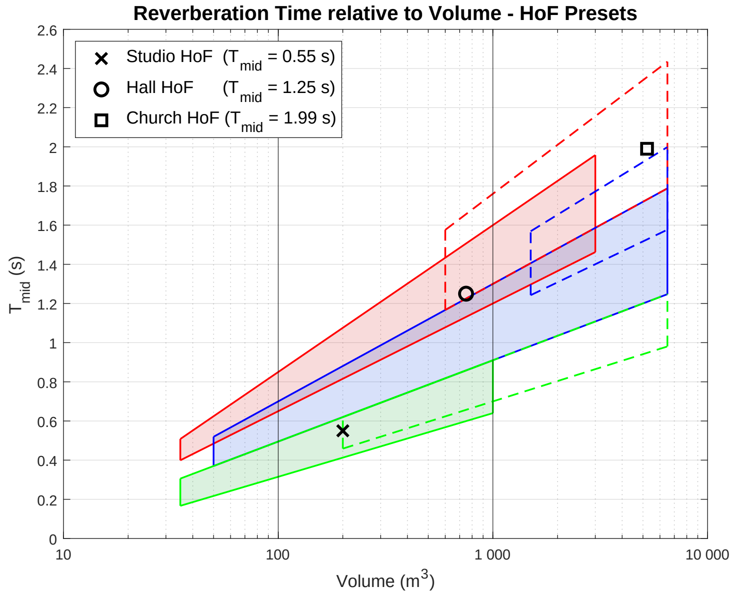

Figure C.1.: T30 relative to room size HoF presets.

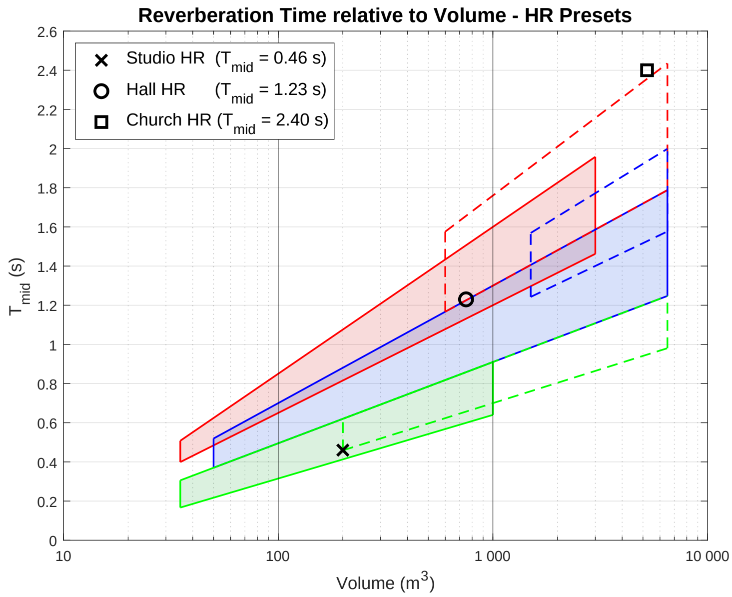

Figure C.2.: T30 relative to room size HR presets.

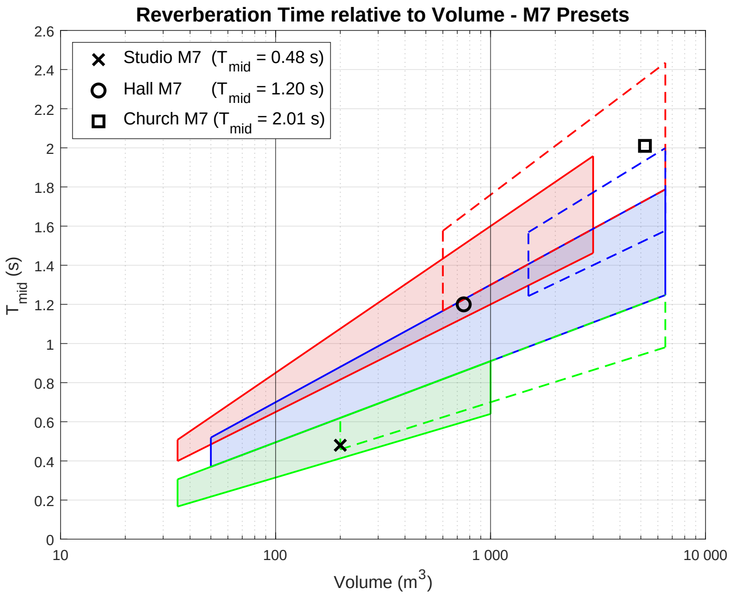

Figure C.3.: T30 relative to room size M7 presets.

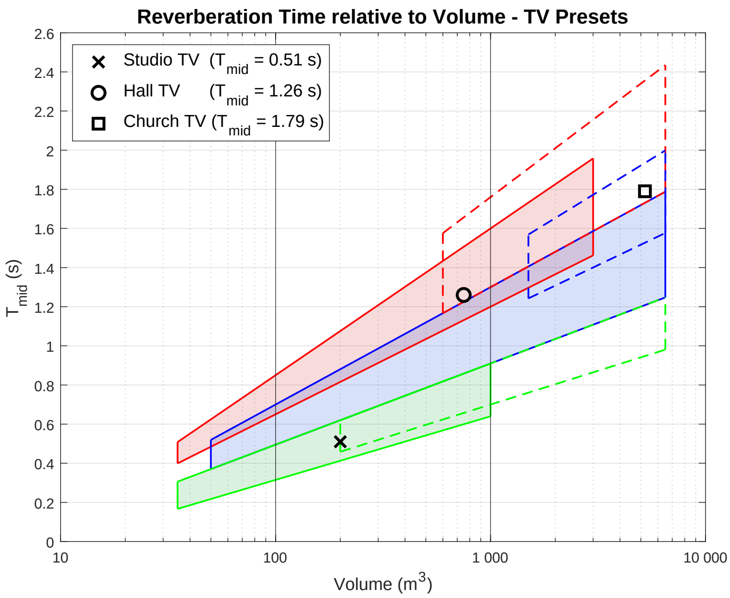

Figure C.4.: T30 relative to room size TV presets.

Appendix D

Loudness Compensation for Processor's Presets

Hall Presets LUFS across processors

Sample	Real LUFS	TV	Δ (dB)	HR	Δ (dB)	M7	Δ (dB)	HoF	Δ (dB)
A-gtr	-24,3	-28,0	+3,7	-30,3	+6,0	-25,9	+1,6	-27,1	+2,8
E-gtr	-23,7	-24,8	+1,1	-26,9	+3,2	-23,7	0,0	-23,4	-0,3
Legato	-25,3	-21,9	-3,4	-24,5	-0,8	-22,2	-3,1	-22,8	-2,5
Staccato	-19,6	-17,9	-1,7	-20,8	+1,2	-18,1	-1,5	-18,1	-1,5
Speech	-24,9	-26,0	+1,1	-28,1	+3,2	-24,5	-0,4	-24,9	0,0

Table D.1.: LUFS values for each processor's Preset and difference to real samples.

Hall Presets LUFS across processors

Sample	Real LUFS	TV	Δ (dB)	HR	Δ (dB)	M7	Δ (dB)	HoF	Δ (dB)
A-gtr	-26,2	-29,0	+2,8	-29,8	+3,6	-28,7	+2,5	-29,5	+3,3
E-gtr	-24,9	-26,3	+1,4	-25,3	+0,4	-26,9	+2,0	-24,4	-0,5
Legato	-26,0	-23,9	-2,1	-26,0	0,0	-24,7	-1,3	-24,9	-1,1
Staccato	-21,7	-19,5	-2,2	-22,5	+0,8	-19,6	-2,1	-20,9	-0,8
Speech	-26,7	-27,3	+0,6	-27,8	+1,1	-27,6	+0,9	-27,1	+0,4

Table D.2.: LUFS values for each processor's Preset and difference to real samples.

Church Presets LUFS across processors

Sample	Real LUFS	TV	Δ (dB)	HR	Δ (dB)	M7	Δ (dB)	HoF	Δ (dB)
A-gtr	-30,6	-33,3	+2,7	-33,3	+2,7	-33,6	+3,0	-33,1	+2,5
E-gtr	-31,5	-30,4	-1,1	-29,5	-2,0	-29,6	-1,9	-29,2	-2,3
Legato	-28,3	-27,4	-0,9	-28,1	-0,2	-28,4	+0,1	-27,9	-0,4
Staccato	-24,3	-23,3	-1,0	-24,9	+0,6	-24,6	+0,3	-24,3	0,0
Speech	-32,6	-31,1	-1,5	-31,5	-1,1	-31,4	-1,2	-31,1	-1,5

Table D.3.: LUFS values for each processor's Preset and difference to real samples.

Appendix E

Extra Tables for Preset Selection

True Verb - "Studio" - Room Acoustic Parameters

Sample Name	T30 (s)		EDT (s)		D50 (%)		C80 (dB)		Ts (s)	
	T30	Δ	EDT	Δ	D50	Δ	C80	Δ	Ts	Δ
EDT +20%	0.51	0%	0.47	21%	79.6	-3.9	11.19	-1.0	0.03	0.00
EDT +10%	0.51	0%	0.43	10%	81.9	-1.6	11.78	-0.4	0.03	0.00
EDT optimal	0.51	0%	0.39	0%	83.5	0.0	12.21	0.0	0.03	0.00
EDT -10%	0.50	-2%	0.35	-10%	85.9	2.4	12.91	0.7	0.02	-0.01
EDT -20%	0.50	-2%	0.31	-21%	87.6	4.0	13.50	1.3	0.02	-0.01
EDT -30%	0.49	-4%	0.27	-31%	89.3	5.8	14.19	2.0	0.02	-0.01
EDT -40%	0.49	-4%	0.23	-41%	91.3	7.7	15.11	2.9	0.02	-0.01
EDT -50%	0.48	-6%	0.20	-49%	94.2	10.6	16.96	4.8	0.02	-0.01
T30 +100%	1.02	100%	0.39	0%	86.3	2.8	10.82	-1.4	0.03	0.00
T30 +80%	0.91	78%	0.39	0%	85.9	2.3	10.90	-1.3	0.03	0.00
T30 +60%	0.81	59%	0.39	0%	85.2	1.7	11.00	-1.2	0.03	0.00
T30 +40%	0.71	39%	0.39	0%	84.9	1.4	11.31	-0.9	0.03	0.00
T30 +20%	0.60	18%	0.39	0%	84.4	0.9	11.71	-0.5	0.03	0.00
T30 optimal	0.51	0%	0.39	0%	83.5	0.0	12.21	0.0	0.03	0.00
T30 -20%	0.40	-22%	0.39	0%	83.9	0.4	13.45	1.2	0.02	-0.01

Table E.1.: Changes in T30 and EDT and Secondary parameters for Processor-True Verb, Room-Studio

TV Studio LUFS across Preset EDT modifications

Sample	TV Optimal LUFS	-20%	Δ (dB)	-40%	Δ (dB)	-50%	Δ (dB)
ST a-gtr TV	-24,3	-27,0	+2,7	-25,4	+1,1	-24,9	+0,6
ST e-gtr TV	-23,7	-23,4	-0,3	-21,7	-2,0	-21,1	-2,6
ST speech TV	-24,9	-24,6	-0,3	-22,9	-2,0	-22,3	-2,6
ST staccato TV	-19,6	-16,7	-2,9	-14,7	-4,9	-14,1	-5,5

Table E.2.: LUFS values for TV Studio Preset and its EDT modifications.

TV Studio LUFS across Preset T30 modifications

Sample	TV Optimal LUFS	+20%	Δ (dB)	+40%	Δ (dB)	+60%	Δ (dB)
ST a-gtr TV	-24,3	-27,3	+3,0	-26,6	+2,3	-26,1	+1,8
ST e-gtr TV	-23,7	-23,7	0,0	-23,0	-0,7	-22,4	-1,3
ST speech TV	-24,9	-25,0	+0,1	-24,3	-0,6	-23,6	-1,3
ST staccato TV	-19,6	-16,8	-2,8	-16,2	-3,4	-15,6	-4,0

Table E.3.: LUFS values for TV Studio preset and its T30 modifications.

Bricasti M7 - "Studio" - Room Acoustic Parameters

Sample Name	T30 (s)		EDT (s)		D50 (%)		C80 (dB)		Ts (s)	
	T30	Δ	EDT	Δ	D50	Δ	C80	Δ	Ts	Δ
EDT +20%	0.47	0%	0.69	19%	48.7	-17.3	4.85	-2.6	0.06	0.01
EDT +10%	0.47	0%	0.64	10%	57.6	-8.5	6.10	-1.3	0.05	0.00
EDT optimal	0.47	0%	0.58	0%	66.0	0.0	7.44	0.0	0.05	0.00
EDT -10%	0.48	2%	0.53	-9%	70.1	4.1	8.29	0.8	0.04	-0.01
EDT -20%	0.48	2%	0.46	-21%	76.4	10.4	9.72	2.3	0.04	-0.01
EDT -30%	0.48	2%	0.41	-29%	79.7	13.7	10.91	3.5	0.04	-0.01
EDT -40%	0.48	2%	0.34	-41%	84.7	18.7	12.75	5.3	0.03	-0.02
EDT -50%	0.47	0%	0.30	-48%	87.8	21.8	14.48	7.0	0.03	-0.02
T30 +100%	0.97	106%	0.57	-2%	80.0	14.0	8.25	0.8	0.04	-0.01
T30 +80%	0.88	87%	0.55	-5%	78.3	12.3	8.48	1.0	0.04	0.01
T30 +60%	0.77	64%	0.59	2%	73.2	7.2	7.50	0.1	0.05	0.00
T30 +40%	0.66	40%	0.56	-3%	74.3	8.3	7.99	0.5	0.04	-0.01
T30 +20%	0.56	19%	0.58	0%	67.6	1.6	7.28	-0.2	0.05	0.00
T30 optimal	0.47	0%	0.58	0%	66.0	0.0	7.44	0.0	0.05	0.00
T30 -20%	0.36	-23%	0.58	0%	49.9	-16.1	7.33	-0.1	0.05	0.00

Table E.4.: Changes in T30 and EDT and Secondary parameters for Processor-M7, Room-Studio

M7 Studio LUFS across Preset EDT modifications

Sample	M7 Optimal LUFS	-20%	Δ (dB)	-40%	Δ (dB)	-50%	Δ (dB)
ST legato M7	-25.3	-23.2	-2.1	-23.5	-1.8	-23.7	-1.6

Table E.5.: LUFS values for M7 Studio Preset and its EDT modifications.

M7 Studio LUFS across Preset T30 modifications

Sample	M7 Optimal LUFS	+20%	Δ (dB)	+40%	Δ (dB)	+60%	Δ (dB)
ST legato M7	-25.3	-21.6	-3.7	-22.3	-3.0	-22.5	-2.8

Table E.6.: LUFS values for M7 Studio preset and its T30 modifications.

TrueVerb - "Brahms" - Room Acoustic Parameters

Sample	T30 (s)		EDT (s)		D50 (%)		C80 (dB)		Ts (s)	
Name	T30	Δ	EDT	Δ	D50	Δ	C80	Δ	Ts	Δ
EDT +10%	1.26	0%	1.22	6%	38.5	-10.3	1.45	-1.3	0.09	0.01
EDT optimal	1.26	0%	1.15	0%	48.7	0.0	2.78	0.0	0.08	0.00
EDT -10%	1.27	1%	1.04	-10%	58.2	9.5	4.14	1.4	0.07	-0.01
EDT -20%	1.27	1%	0.92	-20%	65.2	16.5	5.27	2.5	0.06	-0.02
EDT -30%	1.28	2%	0.81	-30%	70.9	22.2	6.30	3.5	0.05	-0.03
EDT -40%	1.28	2%	0.69	-40%	75.8	27.1	7.31	4.5	0.05	-0.03
EDT -50%	1.27	1%	0.58	-50%	79.2	30.4	8.10	5.3	0.04	-0.04
T30 +50%	1.97	56%	1.16	1%	74.5	25.8	6.24	3.5	0.06	-0.02
T30 +40%	1.76	40%	1.16	1%	71.1	22.4	5.70	2.9	0.07	-0.01
T30 +30%	1.65	31%	1.15	0%	69.1	20.4	5.42	2.6	0.07	-0.01
T30 +20%	1.55	23%	1.15	0%	66.1	17.4	4.98	2.2	0.07	-0.01
T30 +10%	1.36	8%	1.15	0%	57.5	8.7	3.85	1.1	0.07	-0.01
T30 optimal	1.26	0%	1.15	0%	48.7	0.0	2.78	0.0	0.08	0.00
T30 -10%	1.17	-7%	1.11	-3%	42.2	-6.6	2.16	-0.6	0.08	0.00

Table E.7.: Changes in T30 and EDT and Secondary parameters for Processor-TrueVerb, Room-Brahms

TV Hall LUFS across Preset EDT modifications

Sample	TV Optimal LUFS	-20%	Δ (dB)	-40%	Δ (dB)	-50%	Δ (dB)
ST a-gtr TV	-26.2	-28.0	+1.9	-26.3	+0.2	-26.1	-0.1
ST e-gtr TV	-24.9	-25.1	+0.4	-23.4	-1.3	-23.1	-1.6
BS legato TV	-26.0	-22.9	-3.1	-21.2	-4.8	-20.9	-5.1
ST staccato TV	-21.7	-18.6	-3.0	-17.1	-4.6	-16.8	-4.9

Table E.8.: LUFS values for TV Hall Preset and its EDT modifications.

TV Hall LUFS across Preset T30 modifications

Sample	TV Optimal LUFS	+20%	Δ (dB)	+40%	Δ (dB)	+60%	Δ (dB)
ST a-gtr TV	-26.2	-28.0	+1.8	-26,1	-0.1	-26.0	+0.1
ST e-gtr TV	-24.9	-25.0	+0.3	-23,1	-1.6	-23.0	-1.7
BS legato TV	-26.0	-22.8	-3.2	-20.9	-5.1	-20.9	-5.1
ST staccato TV	-21.7	-18.6	-3.1	-16.7	-5.0	-16.7	-5.0

Table E.9.: LUFS values for TV Hall preset and its T30 modifications.

TrueVerb - "Church" - Room Acoustic Parameters

Sample Name	T30 (s)		EDT (s)		D50 (%)		C80 (dB)		Ts (s)	
	T30	Δ	EDT	Δ	D50	Δ	C80	Δ	Ts	Δ
EDT +10%	1.79	0%	1.78	10%	40.4	-3.6	0.90	-1.1	0.11	0.01
EDT optimal	1.79	0%	1.62	0%	44.0	0.0	2.03	0.0	0.10	0.00
EDT -10%	1.79	0%	1.46	-10%	47.1	3.0	2.99	1.0	0.09	-0.01
EDT -20%	1.79	0%	1.30	-20%	49.6	5.5	3.79	1.8	0.08	-0.02
EDT -30%	1.79	0%	1.13	-30%	52.0	7.9	4.60	2.6	0.07	-0.03
EDT -40%	1.78	-1%	0.97	-40%	54.2	10.1	5.40	3.4	0.07	-0.03
EDT -50%	1.78	-1%	0.81	-50%	56.2	12.2	6.20	4.2	0.06	-0.04
T30 +50%	2.75	54%	1.62	0%	50.1	6.0	4.49	2.5	0.09	-0.01
T30 +40%	2.56	43%	1.63	1%	49.3	5.3	4.16	2.1	0.09	-0.01
T30 +30%	2.37	32%	1.63	1%	48.4	4.4	3.77	1.7	0.09	-0.01
T30 +20%	2.18	22%	1.62	0%	47.3	3.2	3.32	1.3	0.09	-0.01
T30 +10%	1.99	11%	1.62	0%	45.9	1.8	2.75	0.7	0.10	00.00
T30 optimal	1.79	0%	1.62	0%	44.0	0.0	2.03	0.0	0.10	0.00
T30 -10%	1.60	-11%	1.58	-2%	41.5	-2.5	1.28	-0.7	0.10	0.00

Table E.10.: Changes in T30 and EDT and Secondary parameters for Processor-TrueVerb, Room-Church

TV Church LUFS across Preset EDT modifications

Sample	TV Optimal LUFS	-10%	Δ (dB)	-30%	Δ (dB)	-50%	Δ (dB)
CH a-gtr TV	-30.6	-34.1	+3.5	-32.2	+1.6	-30.6	-1.8

Table E.11.: LUFS values for TV Church Preset and its EDT modifications.

TV Church LUFS across Preset T30 modifications

Sample	TV Optimal LUFS	+10%	Δ (dB)	+30%	Δ (dB)	+60%	Δ (dB)
CH a-gtr TV	-30.6	-33.8	+3,2	-32.1	+1.5	-30.3	-1.7

Table E.12.: LUFS values for TV Church Preset and its T30 modifications.

Bricasti M7 - "Church" - Room Acoustic Parameters

Sample Name	T30 (s)		EDT (s)		D50 (%)		C80 (dB)		Ts (s)	
	T30	Δ	EDT	Δ	D50	Δ	C80	Δ	Ts	Δ
EDT +10%	2.01	0%	2.12	9%	28.5	-24.3	-3.07	-4.2	0.16	0.05
EDT optimal	2.01	0%	1.95	0%	52.8	0.0	1.11	0.0	0.11	0.00
EDT -10%	2.02	0%	1.76	-10%	63.3	10.5	2.97	1.9	0.09	-0.02
EDT -20%	2.02	0%	1.51	-23%	72.3	19.6	4.82	3.7	0.08	-0.03
EDT -30%	2.02	0%	1.31	-33%	76.6	23.9	5.82	4.7	0.07	-0.04
EDT -40%	2.02	0%	1.07	-45%	80.4	27.6	6.82	5.7	0.06	-0.05
EDT -50%	2.01	0%	0.84	-57%	83.4	30.6	7.73	6.6	0.06	-0.05
T30 +50%	3.01	50%	2.04	5%	79.1	26.3	6.14	5.0	0.08	-0.03
T30 +40%	2.83	41%	2.12	9%	76.2	23.4	5.39	4.3	0.09	-0.02
T30 +30%	2.61	30%	2.08	7%	73.1	20.4	4.75	3.6	0.09	-0.02
T30 +20%	2.40	19%	2.05	5%	69.4	16.6	3.96	2.9	0.09	-0.02
T30 +10%	2.21	10%	2.02	4%	61.5	8.8	2.61	1.5	0.10	-0.01
T30 optimal	2.01	0%	1.95	0%	52.8	0.0	1.11	0.0	0.11	0.00
T30 -10%	1.81	-10%	1.95	0%	35.0	-17.8	-1.75	-2.9	0.14	0.03

Table E.13.: Changes in T30 and EDT and Secondary parameters for Processor-M7, Room: Church

M7 Church LUFS across Preset EDT modifications

Sample	M7 Optimal LUFS	-10%	Δ (dB)	-30%	Δ (dB)	-50%	Δ (dB)
CH e-gtr M7	-31.5	-26.1	-5.4	-30.2	-1.3	-31.5	+0.5
CH legato M7	-28.3	-25.5	-2.8	-29.0	+0.7	-29.8	+1.9

Table E.14.: LUFS values for M7 Church Preset and its EDT modifications.

M7 Church LUFS across Preset T30 modifications

Sample	M7 Optimal LUFS	+10%	Δ (dB)	+30%	Δ (dB)	+60%	Δ (dB)
CH e-gtr M7	-31.5	-27.3	+4.2	-30.3	-1.2	-32.1	+0.3
CH legato M7	-28.3	-26.4	-1.9	-28.7	+0.4	-29.6	+1.7

Table E.15.: LUFS values for M7 Church Preset and its T30 modifications.

Appendix F

Extra Figures for Listening Test Results

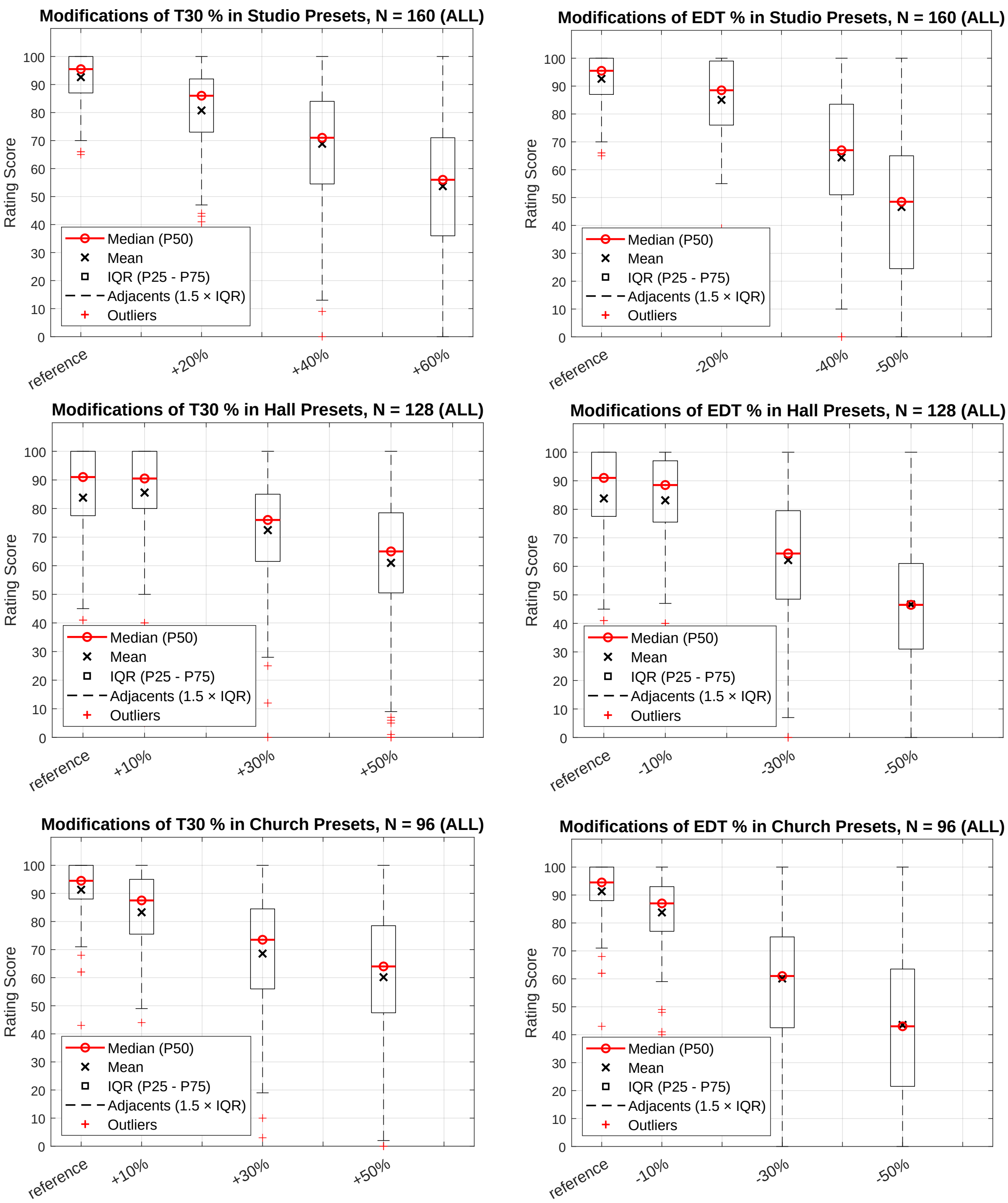

Figure F.1.: T30 and EDT rating scores across rooms for all participants (including excluded ones).

Score Rating for every sample in the listening test

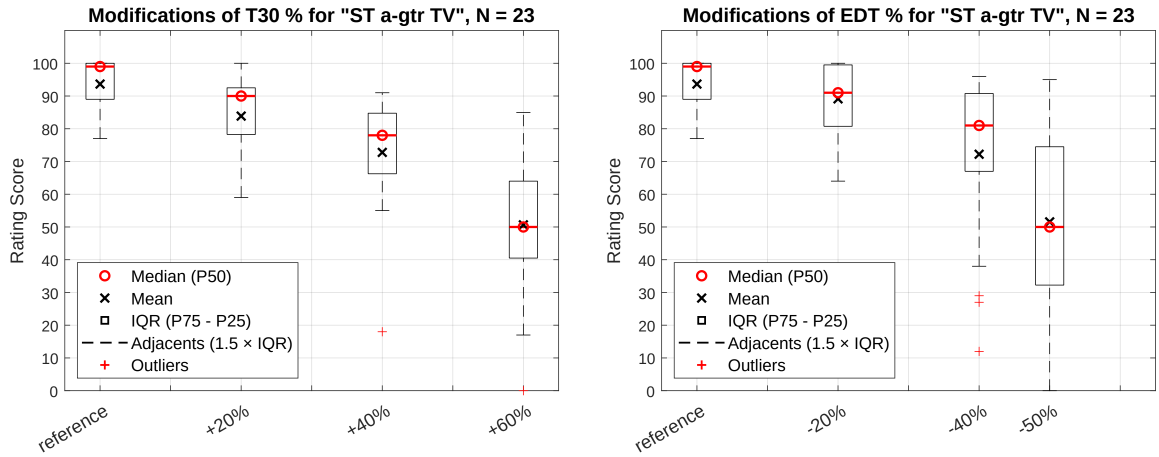

Figure F.2.: T30 and EDT rating for sample "ST Ac-gtr TV".

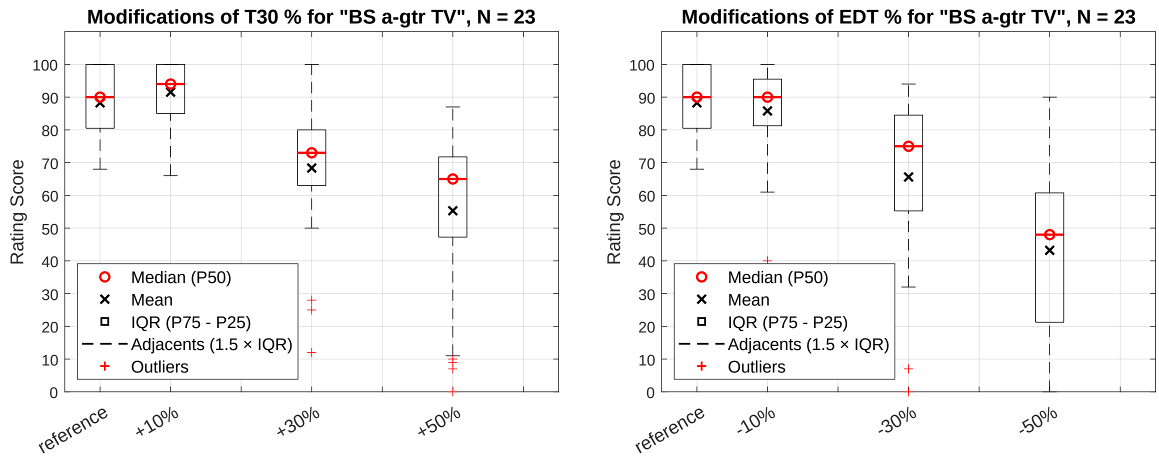

Figure F.3.: T30 and EDT rating for sample "BS Ac-gtr TV".

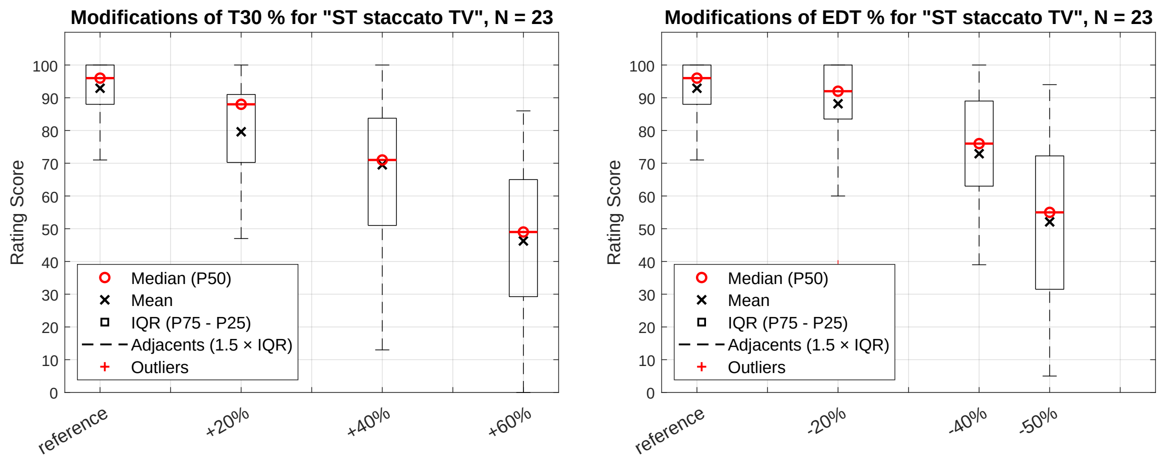

Figure F.4.: T30 and EDT rating for sample "ST Staccato TV".

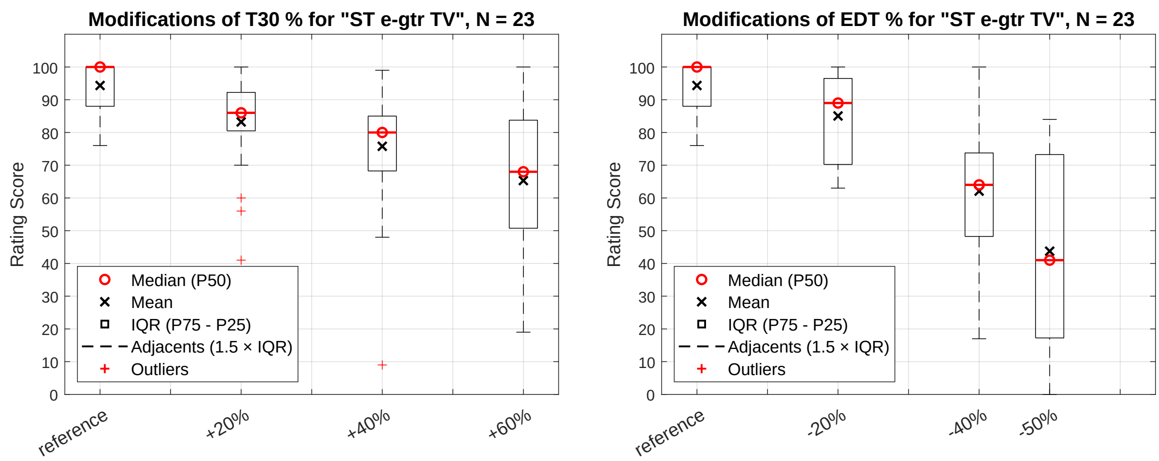

Figure F.5.: T30 and EDT rating for sample "ST E-gtr TV".

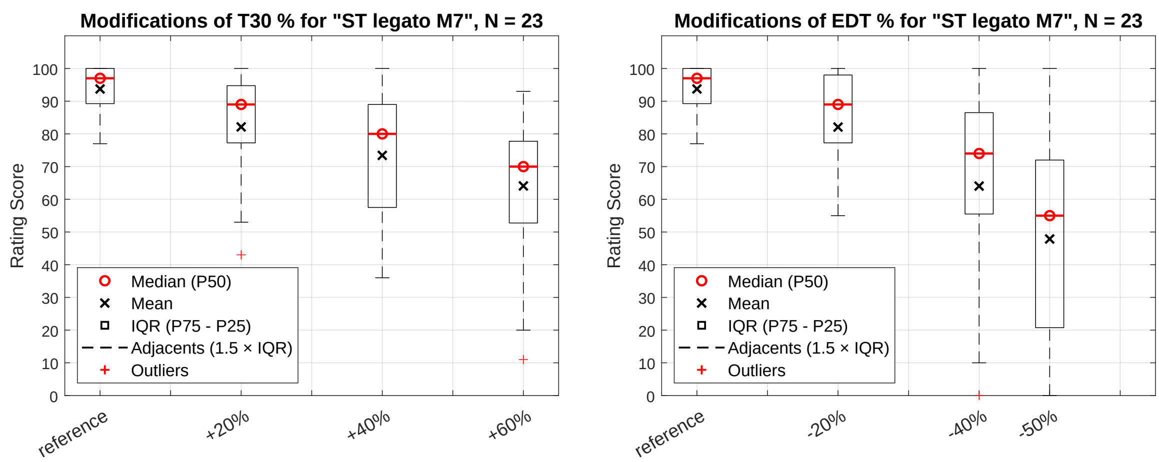

Figure F.6.: T30 and EDT rating for sample "ST legato M7".

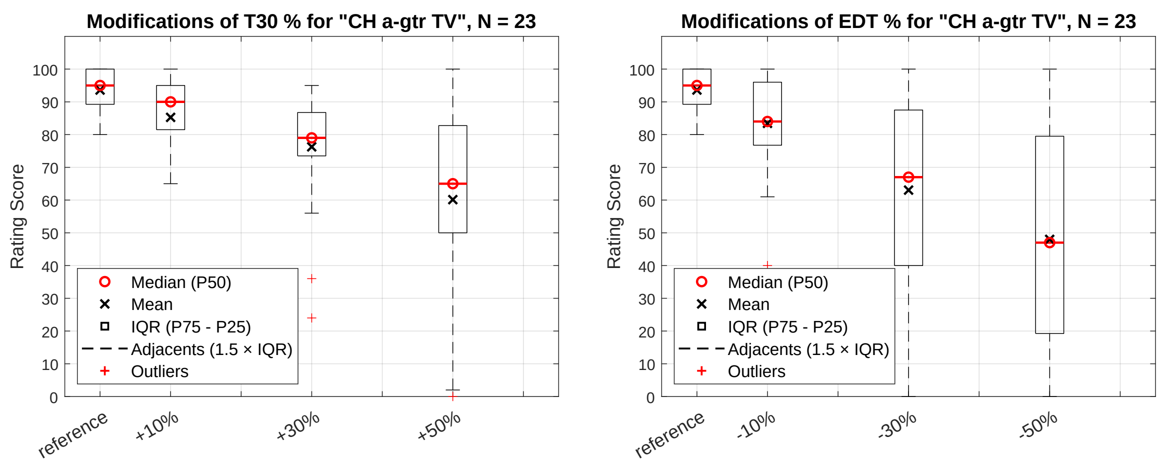

Figure F.7.: T30 and EDT rating for sample "CH Ac-gtr TV".

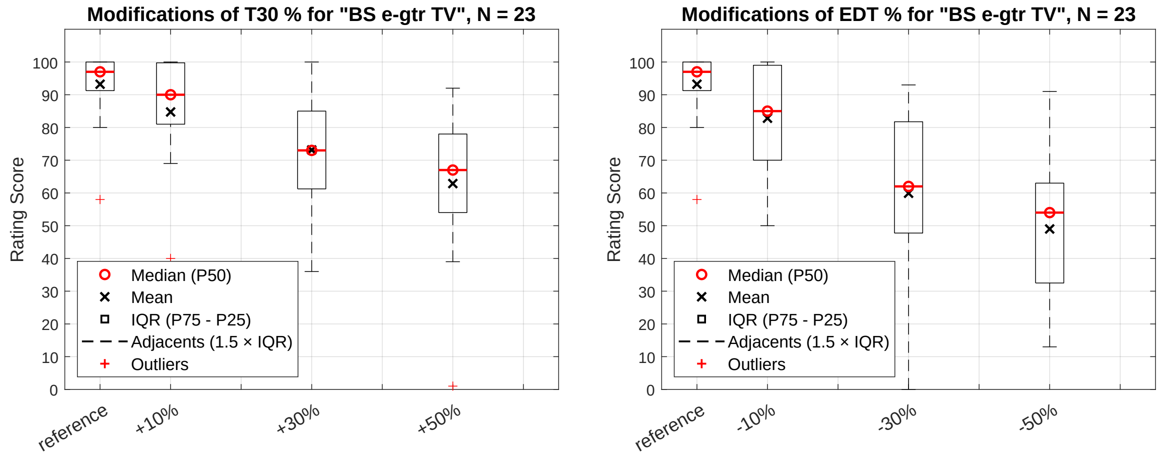

Figure F.8.: T30 and EDT rating for sample "BS E-gtr TV".

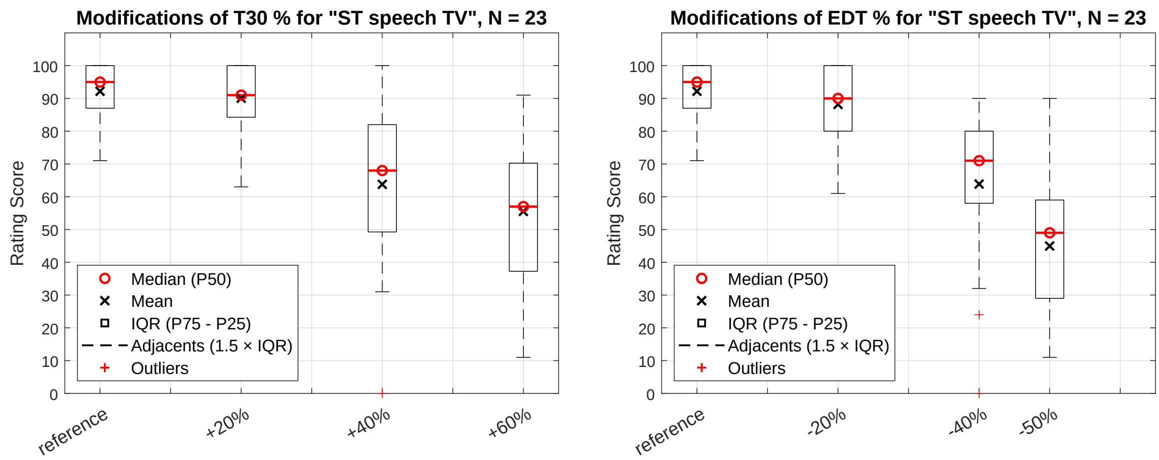

Figure F.9.: T30 and EDT rating for sample "ST speech TV".

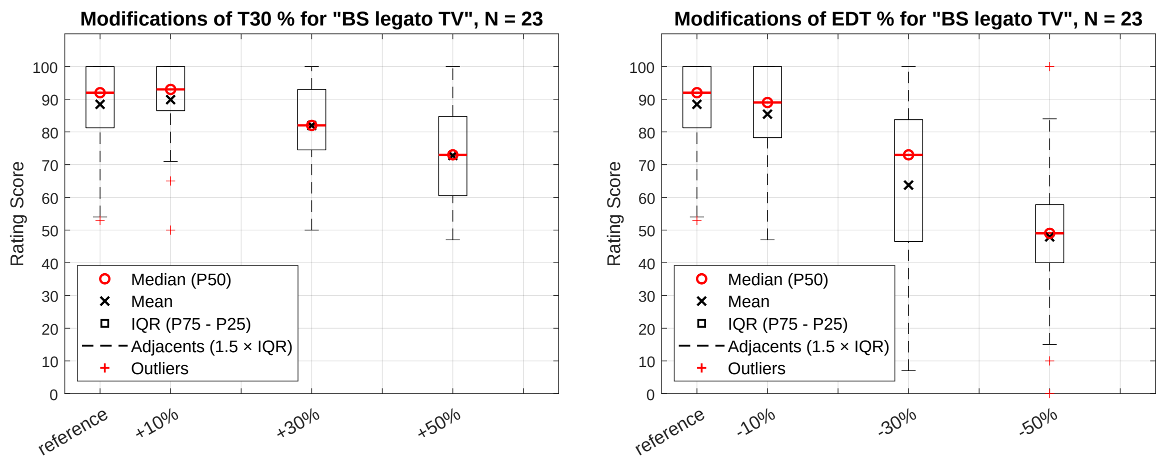

Figure F.10.: T30 and EDT rating for sample "BS legato TV".

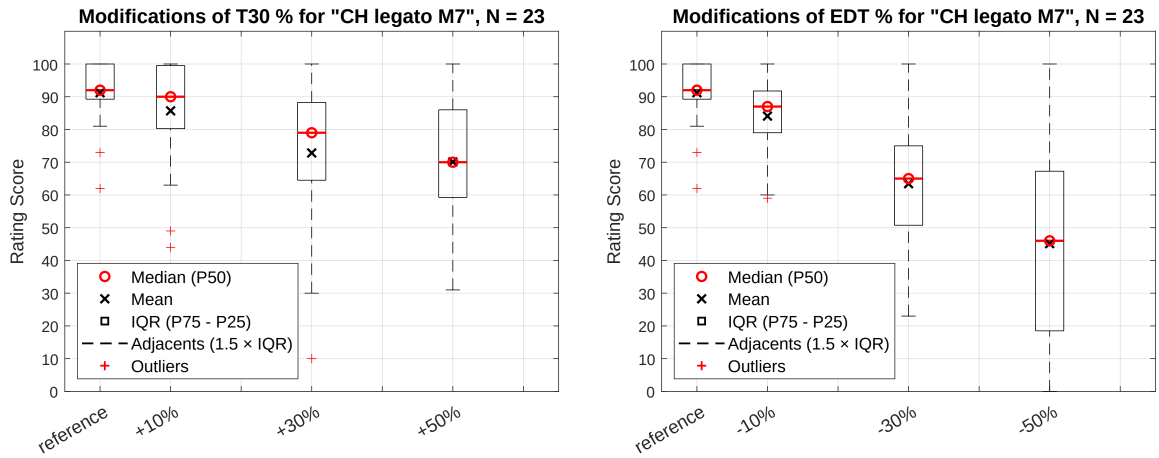

Figure F.11.: T30 and EDT rating for sample "CH legato M7".

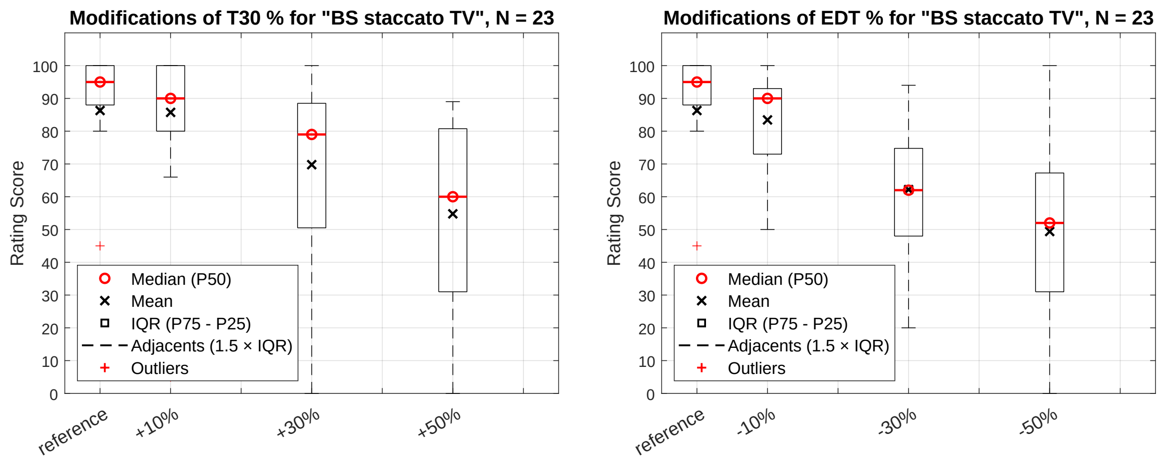

Figure F.12.: T30 and EDT rating for sample "BS staccato TV".

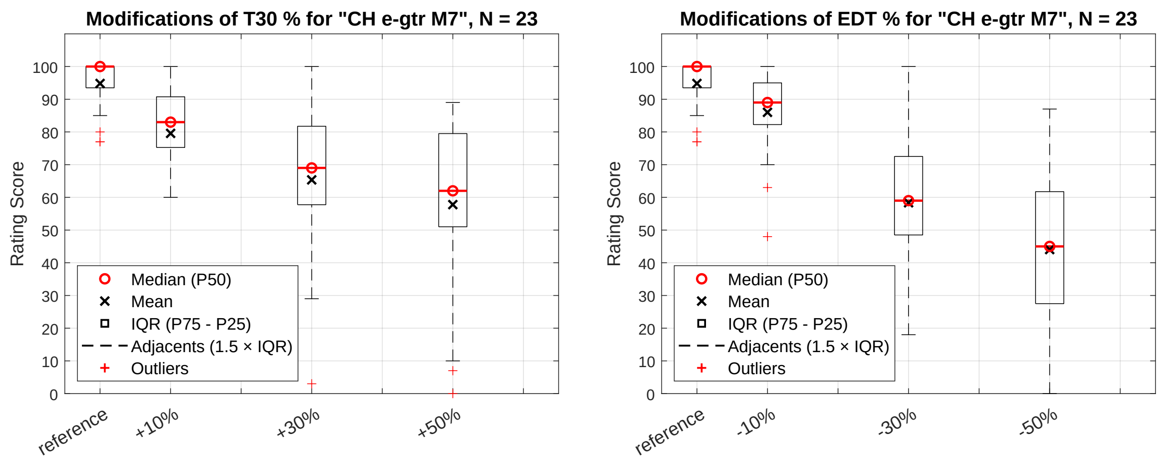

Figure F.13.: T30 and EDT rating for sample "CH E-gtr M7".

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