DSP i Audioteknologi
Guitar Cabinet Impulse Response

AUTE - Project

Group members:

Name	StudentID
Frodi Vestergaard Dam	201410870
Michael Engel	201309486
Ali Saleh	201509807
Minik Olsen	201600341

Contents

Chapter 1

Problem statement

Capture impulse response (IR) of guitar power amplifier and cabinet using both a linear measuring microphone (M1) and a typically applied microphone (M2).
Capture the IR's using different microphone placements relative to the center of the speaker cone in two dimensions, thus creating a 2D matrix of which an infinite number of IR's with unique placements can be created.
Apply the IR's of M1 and M2 to an electrical signal from an electric guitar running through a guitar preamp.
Describe the differences and qualities in tone using M1 and M2 and also the difference using different microphone placements.
Select the preferred IR and compare it in a listening test to the guitar amplifier in its traditional configuration and using its DI output.

Capture and analyze the IR of the linear measuring microphone (M1) and of at least two typically applied microphones (M2), in order to create perfect models of the typically applied microphones (M2).
Convolute these IR's of typically applied microphones (M2) to guitar cabinet IR's recorded with the linear measuring microphone (M1). Switch between the typically microphones digitally, without having to record the IR's with each microphone.

Chapter 2

Impulse Response

An IR is the behaviour of a system (in this case a guitar speaker miked up in a room) when subjected to a brief input signal, captured in a digital format. It is, if you like, a digital "fingerprint" of the tone of a speaker. It captures every resonance and reflection in a way that a basic EQ cannot; an impulse response is if the characteristics of a speaker, a recording studio's room or a piece of hardware gear.
An impulse response (IR) is the output of a linear time-invariant (LTI) system when the input signal is an unit impulse signal, which is a signal with short duration.
Impulse responses are highly accurate digital representations of speakers. IR's allows the user to simulate a speaker's tone through a speaker by digital means. [1]

Figure shows 2.1 a system as a blackbox where the input is a unit impulse delta(t) and output is the impulse response h(t) of the system. The impulse is modeled as a Dirac delta function assuming it's a continuous-time system.

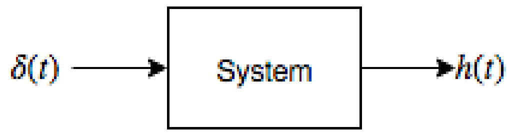

Figure 2.1 – System as a blackbox.

The impulse response is characteristic for the system and it describes the response of the system for all frequencies. Thus meaning the system can be defined with the generated impulse response and be used to predict the system's output in the time domain.

Figure 2.2 shows an example of an impulse signal of an arbitrary system and the system's output as the generated impulse response modeled as a Dirac's

[1] https://www.celestionplus.com/ir-overview/

delta function.

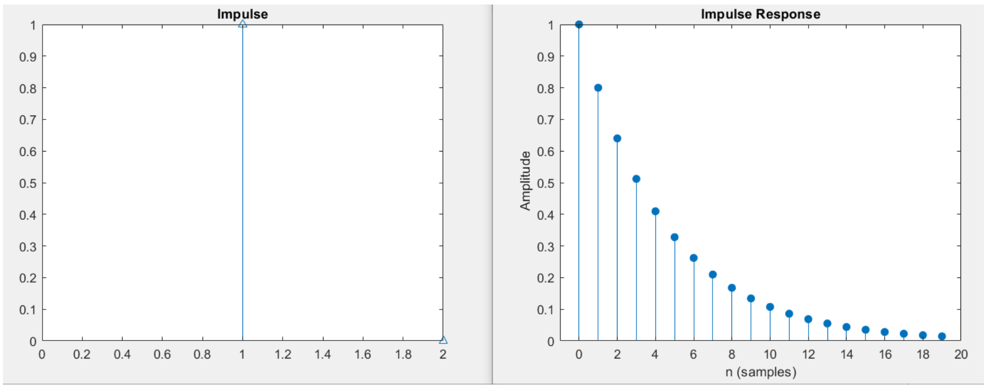

Figure 2.2 – Example of an impulse signal of an arbitrary system.

This impulse response can then be used to measure the system's frequency and phase response, shown in Figure 2.3:

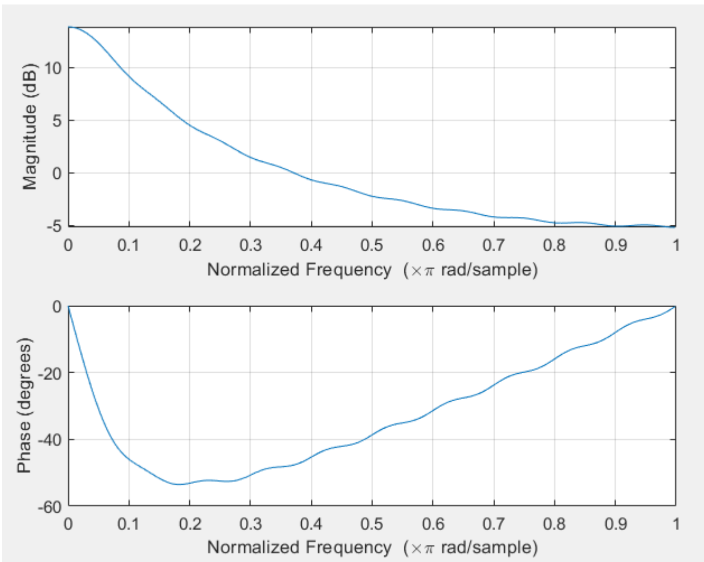

Figure 2.3 – System frequency and phase response.

Thus a very accurate description of the system is now available. It is now possible to reproduce the systen using the IR and the system itself is not needed.
The project group recorded a set of IR's for 1 speaker in different microphone setup configurations, placed in 9 different positions for each microphone type: Measurement microphone (M1), typically applied microphone Shure SM57 (M2).

In practice, the IR files are typically used to create a distinct guitar sound via a AxeFX II standalone amp modeling hardware. As tones are stored in digital format, the user can mix IR files together to create tones, blending different microphones, speakers and cabinet types to match the user's preferences.

2.1 Theory

Each IR captures the essence of a guitar speakers' tone. However in order to make effective use of an IR, it must undergo a process called convolution. This is a signal processing operation that has the effect of combining the IR with an input signal, thereby digitally applying the speaker's tone to that signal.

The convolution process is already built in to the operation of standalone amp modelling hardware, so the user do not even need to be aware of it. [2]

[2] https://www.celestionplus.com/ir-convolution/

Chapter 3

IR Capture

3.1 Recording an IR

When recording or capturing an IR of a guitar cabinet, a distinct sound signal has to be played through the cabinet's speaker. The distinct sound signal has to have a high enough amplitude which affects all the frequencies of the cabinet and therefore creating echoes over the entire frequency range. A microphone measures the cabinet's response, resulting in an IR of the cabinet.
A loud bang or a loud sine sweep typically generates the distinct sound signal used for the impulse responses.

Big Bang

Using a loud bang is the easiest way of creating an impulse response as it is a naturally occurring sound that covers all frequencies. The bang can be generated by popping a balloon or firing a gun with blank rounds.
The loud bang has its disadvantages. It is almost impossible to reproduce for multiple tests. The amplitude may vary, and the sound can easily overdrive the microphone preamp and the microphone itself.
When recording an IR of a guitar cabinet, only the guitar cabinets response to the signal is wanted. The response of the room, in which the guitar cabinet is placed, is not wanted. A loud bang can therefore recorded under controlled circumstances and played through the guitar cabinet and thereby eliminate some of the problematics. However, it must be sure that the microphone recording the loud bang is not clipping as this can affect the impulse response negatively.

Sweep

A sine wave sweep can also be used to record an IR. A wave generator generates a sine wave and sweeps it from 20Hz to 20kHz within a given time frame. The sweep is played through the guitar cabinet, and a measurement microphone

records the response. The result of the recording is an audio file with a duration corresponding to the sine wave sweep time frame. This is typically from 4 to 10 sec.
The recorded sine wave sweep audio file can not be used as an IR directly. The recorded audio file needs to be de-convoluted, which is the inverse of convolution, where the response of the guitar cabinet is sorted from the sine wave sweep.
In the year 2000, professor Angelo Farina at the University of Parma in Italy discovered that using an exponential sine wave sweep (ESS) instead of a linear sine wave sweep results in a better signal to noise ratio (SNR) and time-separates the harmonic distortion of the system from the linear response[1]. The IR is then generated by convolving the recorded ESS audio with the clean inverse ESS.
Angelo Farinas paper on *"Advancements in impulse response measurements by sine sweeps"* was presented as a convention paper in 2007 at the 122nd Audio Engineering Society convention in Vienna Austria[2].

Conclusion

Because of the good documentation and advantages, Angelo Farinas ESS method for creating an IR will be used in this project to create the guitar cabinet IR.

3.2 Impulse Response Generator

The algorithmn to generate the impulse responses has been developed in Matlab with the same procedure Angelo Farina discovered. The following figure shows a diagram behind the sequence of the Matlab script.

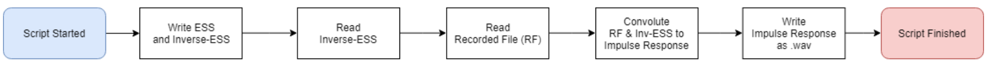

Figure 3.1 – Sequence of Matlab Algorithm

To write an ESS, the mathematical form is defined as:

$$x(t) = sin(K \cdot (e^{t/L} - 1)) \quad (3.1)$$

Where K and L can be obtained as:

$$K = \frac{T \cdot \omega_1}{ln(\frac{\omega_2}{\omega_1})} \quad (3.2)$$

[1] http://citeseerx.ist.psu.edu/viewdoc/download?doi=10.1.1.33.1614&rep=rep1&type=pdf
[2] http://pcfarina.eng.unipr.it/Public/Papers/226-AES122.pdf

$$L = \frac{T}{ln(\frac{\omega_2}{\omega_1})} \tag{3.3}$$

Where T is the length of the sweep in seconds, ω_1 is the start frequency and ω_2 is the end frequency.
Thus the full definition for the ESS,$x(t)$, can be expressed in full form as:

$$x(t) = sin(\frac{T \cdot \omega_1}{ln(\frac{\omega_2}{\omega_1})} \cdot (e^{\frac{t}{T} \cdot ln(\frac{\omega_2}{\omega_1})} - 1)) \tag{3.4}$$

The Inverse of ESS is the time-reversal of the ESS $x(t)$. This can be implemented in matlab by taking the amount of samples and decrementing by 1 until all of them have been passed, this variable is called N_{inv}. Thus the final equation in matlab for the inverse ESS is defined as:

$$f(t) = x(N_{inv}) \cdot e^{t/L} \tag{3.5}$$

The final step is to convolute the inverse ESS $f(t)$ with the recorded file $y(t)$ to find the impulse response $h(t)$:

$$h(t) = f(t) \circledast y(t) \tag{3.6}$$

The generated impulse response $h(t)$ can then be used to identify the linear system.

Results and Plots

This section will explain some of the essential results and plots obtained by the matlab algorithm.

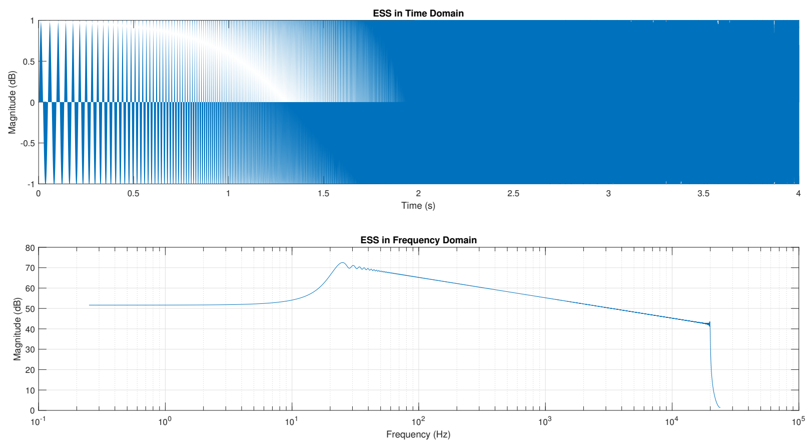

Figure 3.2 – ESS in Time- and Frequency Domain

Figure 3.2 shows the ESS in the time domain and frequency domain. The time domain shows how the exponential sine sweep is exponentially growing through the four seconds length. In the frequency domain it can be observed the sweep is lowering its amplitude in the higher frequencies and the sweep is between 20 Hz to 20 kHz as expected.

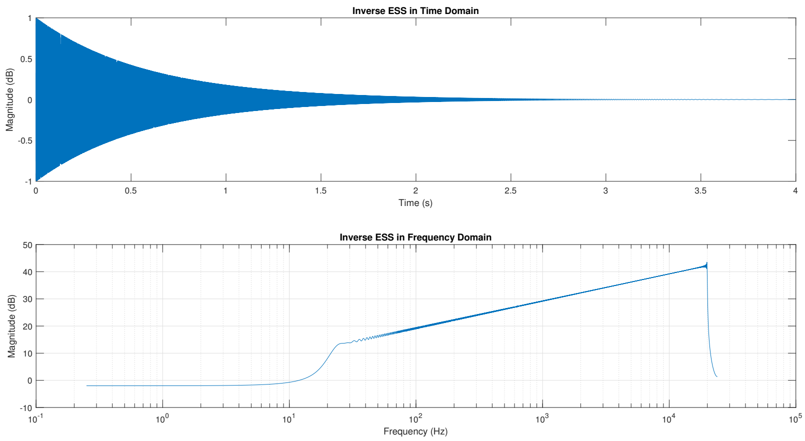

Figure 3.3 – Inverse ESS in Time- and Frequency Domain

The time and frequency domains for the inverse ESS can be seen on figure 3.3. The time domain shows how the inverse ESS is doing the exact opposite of the ESS. The inverse ESS is exponentially decreasing towards zero within the four seconds time length. Furthermore, the frequency domain is growing in amplitude proportionally with the frequency in the interval of 20 Hz to 20 kHz.

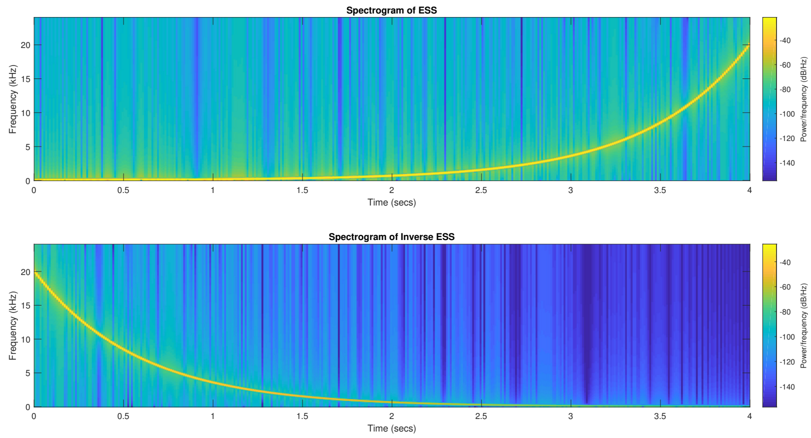

Figure 3.4 – Spectrogram of ESS and its inverse

The spectrogram on figure 3.4 shows the comparison of the ESS and the inverse ESS. The essential observation here is that the ESS is growing in the frequencies proportional with time and the inverse is doing the exact opposite. Thus the spectrogram represents that the ESS and the inverse ESS have been generated properly compared with the theory of exponential sine sweeps.

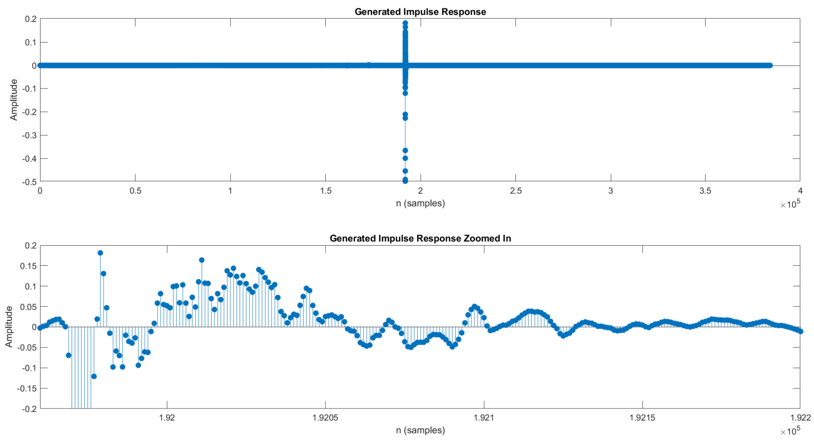

Figure 3.5 – Generated Impulse Response original and zoomed view

The impulse response $h(t)$ generated by the inverse ESS and the recorded file can be seen on figure 3.5. The upper figure shows the impulse response and the lower one shows the same impulse response but zoomed on the focus point where the samples $N = T \cdot fs = 4s \cdot 48kHz = 192000samples$. The impulse response has been generated with a very low scaling of amplitude in order to reduce the harmonic responses. The harmonic responses can be seen more clearly in the following figure.

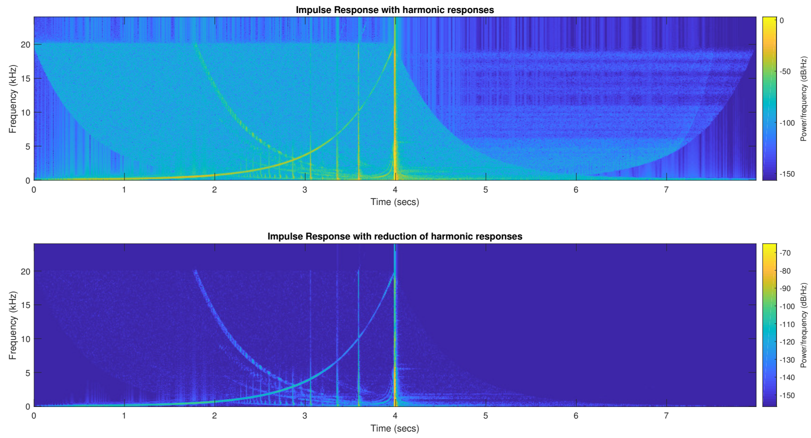

Figure 3.6 – Spectrogram of Impulse Response with- and without harmonic responses

Figure 3.6 shows the impulse response with harmonic responses and reduction of the harmonic responses by lowering the amplitude. The upper figure shows the harmonic responses have a clear effect on the output. However, when the impulse response has its amplitude lowered, the harmonic responses do not affect the end results as drastically.

3.3 Choosing the Right Microphone

When capturing an Impulse Response from a speaker cabinet, the acquired sample is the combination of the signal's DAC (Digital/Analog Converter), speaker cabinet, the microphone, microphone preamplifier, the ADC (Analog/Digital Converter), the cables in use and the room itself. Essentially, everything that comes in contact with the impulse as it makes a round-trip from the chosen DAW (Digital Audio Workstation) and back again. Using high quality equipment is therefore essential.

There are several configuration techniques to obtain an IR: Anything from mono to multichannel formats as long as the convolution software supports the configuration. As far as polar patterns and microphone types, no hard and fast rules exist – It is entirely up to the user to decide according to taste.

Location for the measurements are conducted in ASE's AudioLAB. An anechoic chamber is a shielded room designed to attenuate sound or electromagnetic energy. Anechoic chambers were originally used in the context of absorbing acoustic (sound) echoes caused by internal reflections of a room.[3] This is highly useful for loudspeaker measurements, noise emission studies, studies of attenuation and diffraction of sound and also for creating an artificial listening environment where all the acoustic properties are determined by the experimental setup. [4] Within the scope of the project, 2 different microphones will be investigated : A Linear Measuring Microphone (M1) and a typically applied microphone (M2).

The guitar cabinet's impulse responses will be captured from 9 fixed microphone placements in the anechoic chamber at ASE's AudioLAB which will be used to compare the qualities of each microphone. After capture of the IR data, one or two IR's will be tweaked to create an IR with the highest quality possible based on the acquired data.

Measurement Microphone

Generally, measurement microphones are scalar sensors of pressure. They exhibit omnidirectional response only limited by the scattering profile of their physical dimensions.

For scientific measurements, its precise sensitivity (in volts per pascal) must be known. Measurement microphones are almost always omnidirectional, which limits their practical mic'ing application somewhat. Measurement microphones tend to have very small-diameter capsules (approx. 12mm or less), making them susceptible to noise than the small and large capsules commonly used for music recordings; low noise is usually not a requirement in measurement applications as it is far more common to want to know how loud something is than how quiet.

The Audix T1 PLUS is designed to capture acoustic measurement for room analysis software programs, real time analyzers etc. For the project, the group chose the Audix TM1 PLUS measurement microphone due to its linearity, accurate response and consistency.

[3] https://ualr.edu/systemsengineering/anechoic-chamber/
[4] https://www.elektro.dtu.dk/english/research/research-facilities/anechoic-chamber

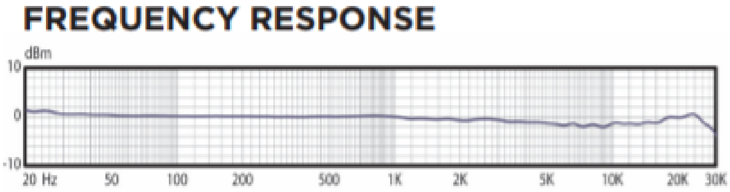

Figure 3.7 – Audix TM1 Measurement microphone frequency response. [5]

Giving its near-flat frequency response, the Audix TM1 PLUS' measurement data can be compared with the typically applied microphone M2's measurement data to determine the usefulness for using the M2 for guitar cabinet impulse response modeling.

Shure SM57

The Shure SM57 frequency response is not linear but is typically used for recording electric guitars or amplifying a guitar cabinet on stage to the audience through the P.A as its frequency response it tailored for guitars, drums and vocals. The SM57 has become an industry standard for mic'ing guitar amps as it makes the guitar cut through the mix with little or no equalization. The SM57 is chosen as the typically applied microphone M2 for measuring impulse responses.

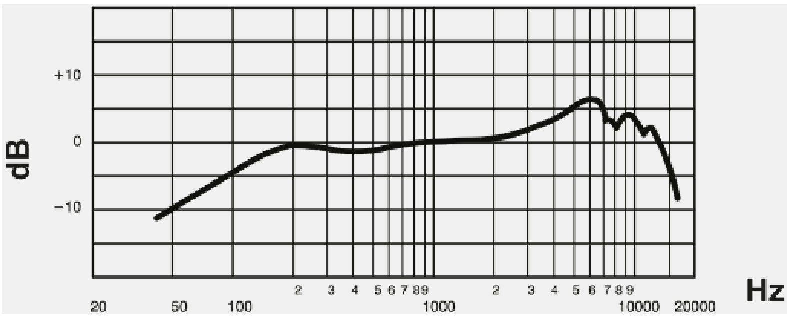

Figure 3.8 – SM57 Frequency Response [6]

The SM57 low frequencies below 180 Hz are attenuated with a slight dip in the low mid range between 250-630 Hz and varying increases in the high frequent (HF) range between 2kHz-15kHz and a roll-off attenuation from 15 kHz.

3.4 Microphone Placement Matrix

Microphone placement positions and angles affect measurements as well as recordings. The Marshall cabinet's speaker was measured from 9 positions: On cap, on cone and edge of cone, each position with 3 different distances to the speaker. Figure 3.9

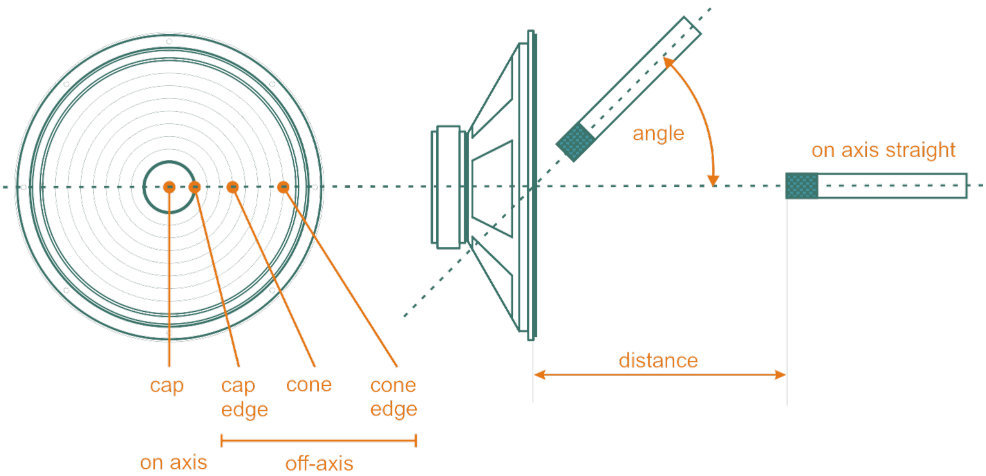

Figure 3.9 – Speaker Diagram. Frontal view of guitar cabinet speaker. 7

Figure 3.10 shows the top view positions of the microphones in a matrix .

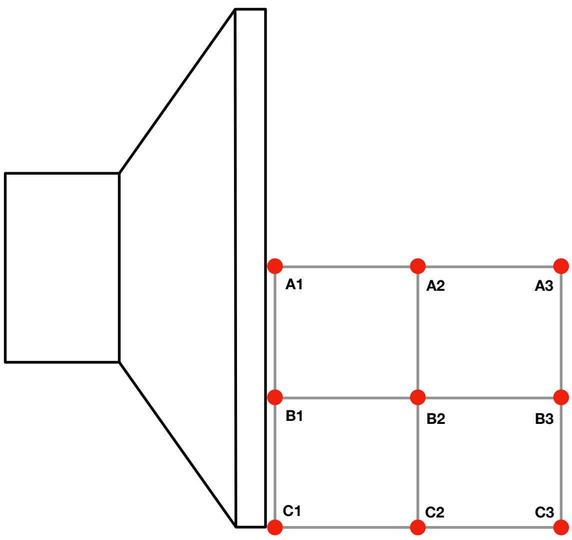

Figure 3.10 – Top view of microphone positions in matrix.

The distance is critical when positioning a microphone. Closer to the speaker right up agains the cabinet will yiels the proximity effect (lower frequencies build up and may overload the microphone), whereas farther distances adds noise sources such as the room sound.
Positioning the microphone at the center yields a trebly sound, and moving away from center yields the lower frequencies to be more dominant.
The group chose to measure the guitar cabinet with the microphones on-axis (perpendicular to the speaker) to minimize data processing. Keep in mind that off-axis positioning yields roll-offs in the high frequencies and reduction of lower frequencies.

3.5 Microphone IR

Figure 3.11 – Measurement setup with Adam Audio S3V and Shure SM57

All microphones have a different frequency response, making them especially suitable for different situations. The idea is to record the IR of different microphones so that these microphones can be simulated as well as the guitar amplifier cabinet.
A linear speaker is needed to record a microphone's IR. A non-linear speaker will affect the microphone's IR so that the final frequency response will be a mix of the speaker and the microphone.

An Adam Audio S3V active midfield studio monitor is used for the IR microphone recording. The S3V has a 9-inch low-frequency driver, a 4-inch mid-frequency driver and a ribbon tweeter. This combination of drivers will ensure an excellent reproduction of all the frequencies in the hearable range.

The speaker is seen on figure 3.11 [8] together with a Shure SM57 microphone. The S3V is factory tuned. The frequency response of the S3V in figure 3.12 shows that the speaker is linear down to about 50Hz.

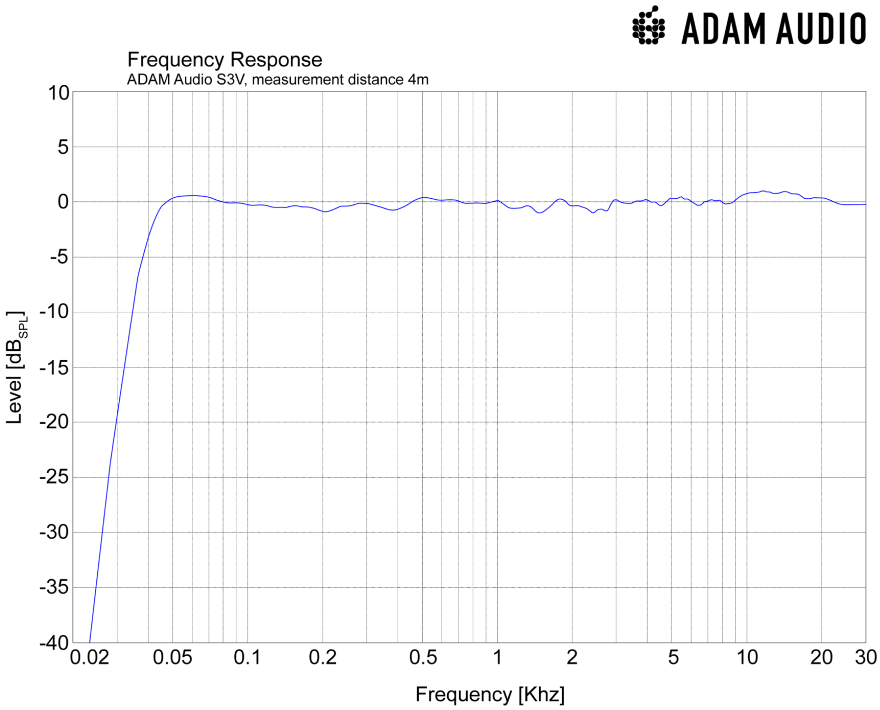

Figure 3.12 – Adam Audio S3V frequency response.

To ensure the linearity of the speaker in our measurement setup, the speaker's frequency response is measured by using the software Smaart 8 by Rational Acoustics. Figure 3.13 shows the result of the measurement.

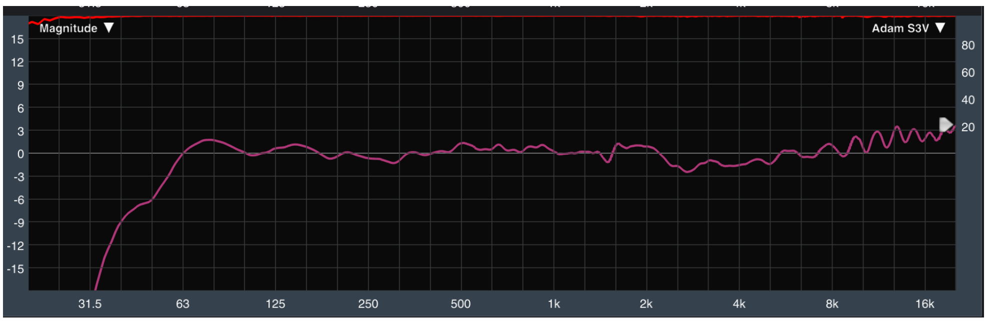

Figure 3.13 – Measurement result of Adam Audio S3V frequency response

[8] https://www.adam-audio.com/content/uploads/2018/04/adam-audio-s3v-studio-monitor-frequency-response-1920x1463.png

The measurement is quite similar to the datasheet. There is a small drop from 2 kHz and a small boost at 10 kHz. Comparing the measurement result with the Audix TM1 frequency response in figure 3.7 shows that the microphone might be causing the drop and the boost. However the drop and boost are within reasonable limits.

Capturing the microphone IR

The IR is captured from three different microphones. These three microphones are:

- **Audix TM 1** condenser microphone. See section 3.3 for description.
- **Shure SM57** dynamic microphone. See section 3.3 for description.
- **Cloud JSR-34p**[9] ribbon microphone.

An ESS is played through the Adam speaker. The microphone is placed in the speakers mid-field and the ESS is recorded. The recorded exponential sine wave is then convolved with the inverse ESS to produce the IR for the microphone. The convolution process is described in section 3.2.

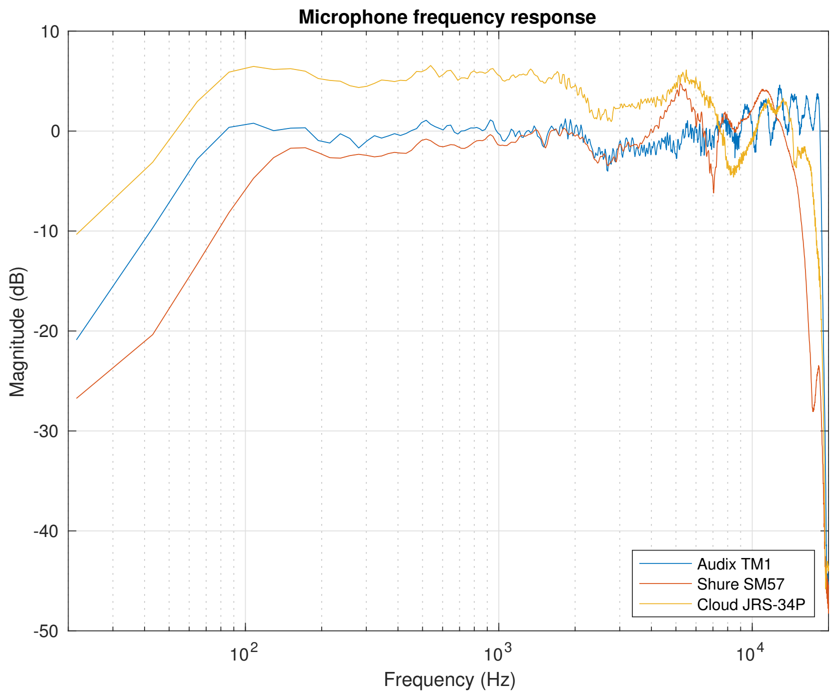

Figure 3.14 – Selected microphone frequency responses from recorded IR

[9]https://www.cloudmicrophones.com/microphones-jrs-34p

The frequency responses from the three microphones recorded IR's are shown in figure 3.14.
Comparing Shure SM57 frequency response in figure 3.14 with the frequency response from the manufacturer in 3.8, it shows that they are quite similar with peaks and valleys at the same frequencies.
Comparing the Audix TM1 frequency response in figure 3.14 with the frequency response in figure 3.13, which is the frequency response of the Audix TM1 measuring a pink noise signal from the Adam S3V studio monitor, and the frequency response from the Audix TM1 datasheet in figure 3.7, the frequency responses are quite similar.
These comparisons also show a coherence in the measurements and recorded IR's.

3.6 Recording Setup

After acquisition of the microphone impulse responses through the linear speaker Adam Audio S3V in ASEs anechoic chamber, it is now possible to record the impulse responses for the microphones through the guitar cabinet with a sine sweep in the anechoic chamber.

The following gear was used for the recordings:

1. **Sine wave**. Generated from MATLAB.
2. **Guitar cabinet**: 20W Marshall 2554 Silver Jubilee tube amplifier.
3. **Microphone**: **SM57** *or* **Audix TM1**.
4. **Audio Interface**: Focusrite Scarlett 2i2.
5. **Logic Pro X**: Recording software.
6. **MATLAB**: For data processing.

The figures below displays the block diagrams for the recording signal path:

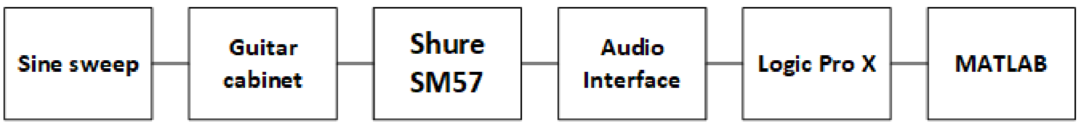

Figure 3.15 – Block diagram of signal path for recording SM57 IR.

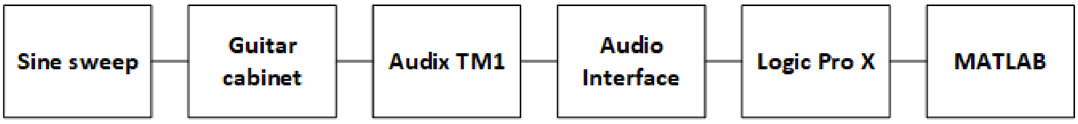

Figure 3.16 – Block diagram of signal path for recording Audix TM1 IR.

The procedure for the recordings are completely identical in terms of signal path and microphone placement. Using the microphone placement matrix from Figure 3.10, the necessary impulse responses was acquired:

Figure 3.17 – Picture of setup. The Audix TM1 is recording from the B1 position

Chapter 4

IR Implementation

4.1 Interpolation of Matrix

Interpolation is a method of constructing new data points within a set of known discrete data points. In music production this is done to upsample a file for higher sample rate. Often just to ensure compatibility. In a similer same way the 3x3 matrix to a 5x5 matrix will be interpolated, increasing the number of IR's from 9 to 25.
This will give more flexibility for choosing microphone placement.

Taking the matrix from figure 3.10 and transforming it to a 5x5 matrix. In order to make room for the in between IR's, A2 becomes A3, A3 becomes A5 and so forth.

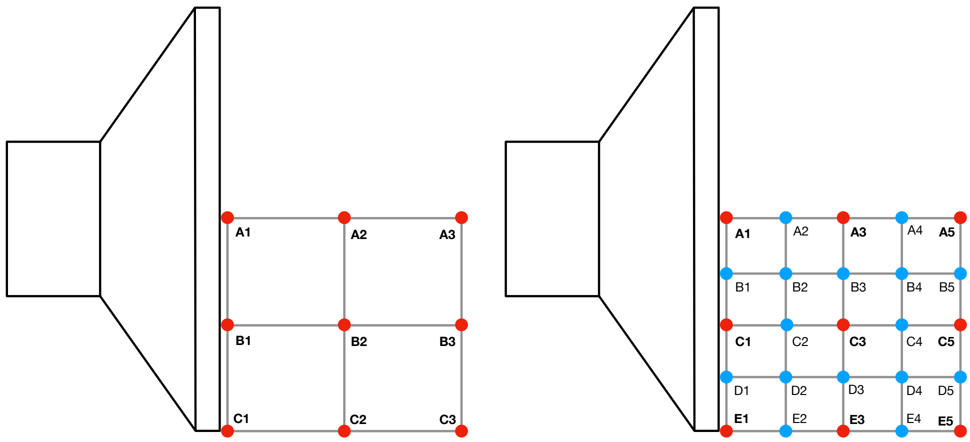

Figure 4.1 – Top view of microphone positions in new interpolated matrix.

The blue dots represent the missing IR's in the matrix.

Our research for this problem concluded it as not being a typical problem, hence little theory was found, which is why we came up with our own solution.

By looking at A2 as being the average of A3 and B2 being the average of A1, A3, B1 and B3. This assumes any change in position results in an equal linear shift in frequency respons.

This might seem naive - especially if the magnitudes of the filter coefficient are simply averaged.
By first Fourier transforming the IR's, converting to dB, meaning the IR's together, converting to magnitude and finally inverse Fourier transforming them, should give a better result.
See the difference in the frequency dB plot in figure 4.2.

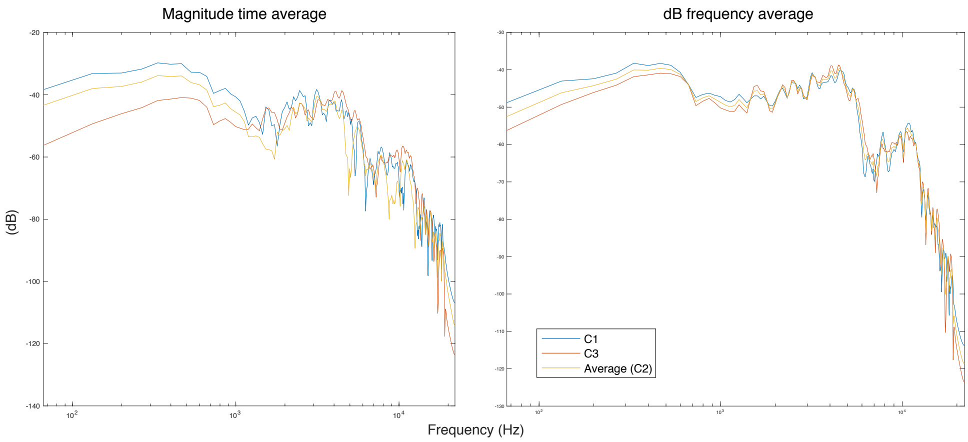

Figure 4.2 – Time mag. averaging and frequency dB averaging, plotted in frequency dB

The graphs reveal that the left figure, averaging magnitude in time, does not result in a perfect average when seen on a frequency dB chart. It seems to work okay in the low frequencies but quickly becomes unusable.
The left figure, averaging dB in frequency looks perfect, with the combination in the exact middle.

4.2 Analyzing Matrix

The following plots shows how the microphone placement affects the response. The figures shows the Audiox TM1 (M1). Using SM57 (M2) resulted in similar results but with a slightly higher proximity effect.

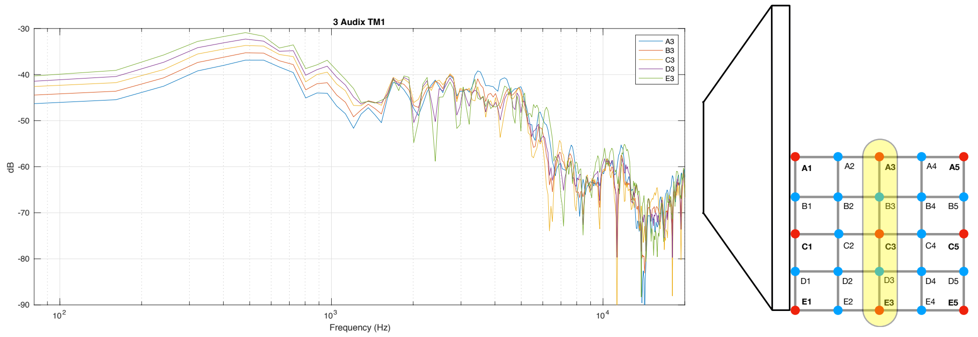

Figure 4.3 – A3 to E3 - going from center of speaker to edge of speaker

The figure concludes that the closer to the edge of the speaker the more bass and midrange.

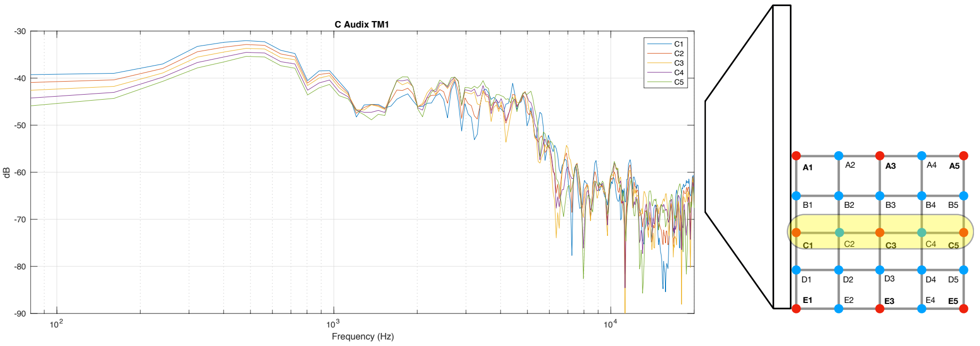

Figure 4.4 – C1 to C5 - going from close to speaker to far from speaker

The figure concludes that the further away from the speaker less bass and a little less midrange.

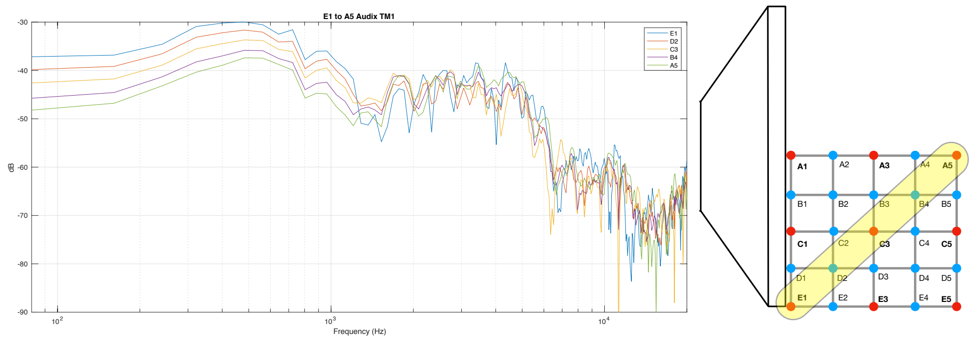

Figure 4.5 – E1 to A5 - going from close and edge of speaker to far and center of speaker

The figure concludes that going from close and edge of speaker to far and center of speaker equals to the greatest loss in bass and midrange.

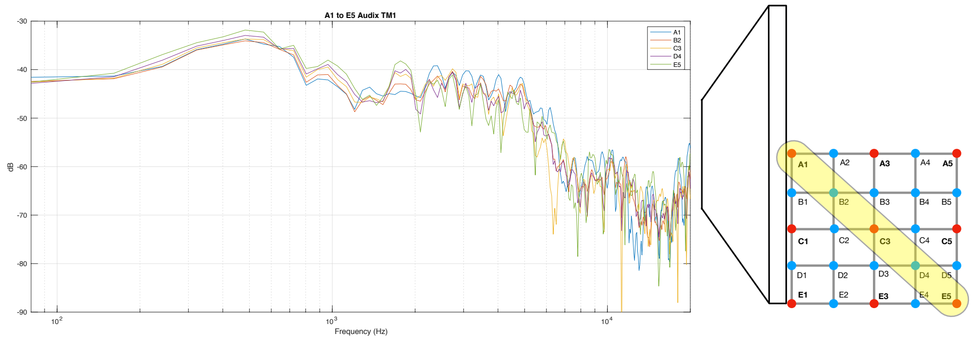

Figure 4.6 – C1 to C5 - going from close and center of speaker to far and edge of speaker

The figure concludes that going from close and center of speaker to far and edge of speaker equals almost no change in response.

4.3 Conclusion

By interpolating the matrix of 3x3 IR's we have created a 5x5 matrix, increasing the number of IR's to 25, giving more choices when selecting microphone placement.
The analysis revealed that the closer to the edge and closer to the speaker gives more bass and midrange, and that the loss in bass and midrange of going away from the speaker, can be almost completely neutralized by moving the microphone closer to the edge.
In practical scenarios, there could be several side effects of moving the microphone further away such as more ambiance and room captured when further away.

Chapter 5

Choosing an IR

By listening to the different IR's through the Adam S3V studio monitor and comparing them to the guitar cabinet, the position C5 is chosen because of the simulariti to the guitar cabinet but also the wantet fullness in the sound. This IR is uploaded to the Mini-DSP and used for the listening test in chapter 6. Figure 5.1 shows the frequency response of position C5 in Matlab and figure 5.2 shows the frequency response of position C5 loaded into the Mini-DSP.

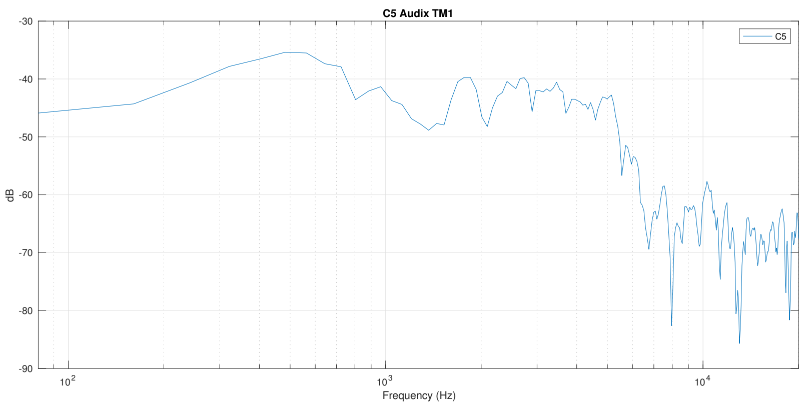

Figure 5.1 – Frequency response of position C5 in matlab

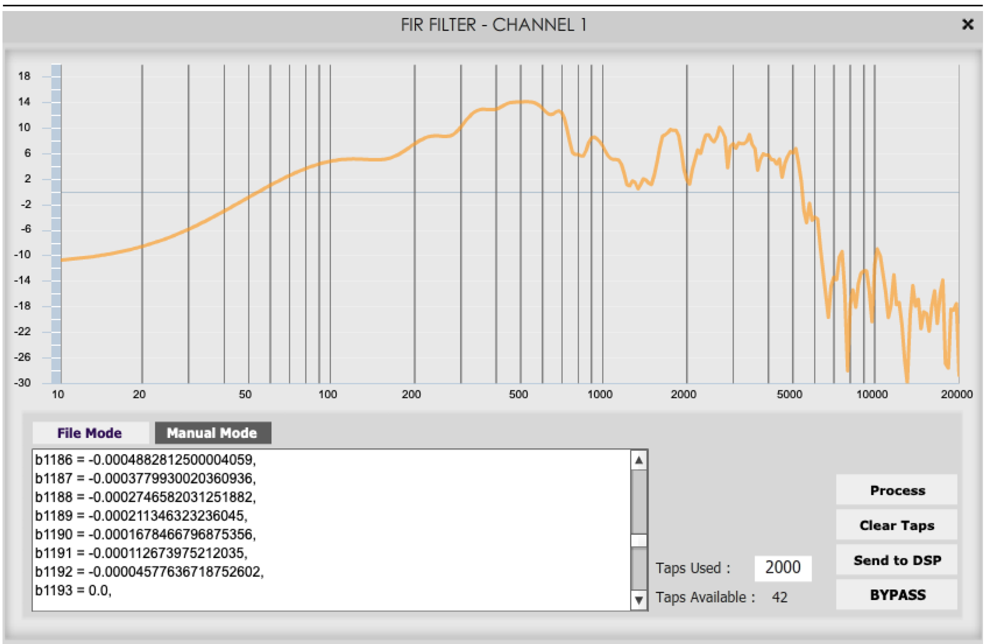

Figure 5.2 – Frequency response of position C5 in Mini-DSP

Chapter 6

Listening Test

In order to evaluate whether we succeed in capturing an impulse response, modifying it by placement and equalization, as to create a sound just as good or better than the analog amplifier, we have to do a qualitative listening test. During the test we will be comparing our IR's to the amplifier, using its power amp and built in speaker and also the amplifier using its built in DI circuit. The DI is an analog circuit designed to match the response of the speaker. It is basically just an analog low-pass filter.

The listening test will be conducted using the setup shown in figure 6.1:

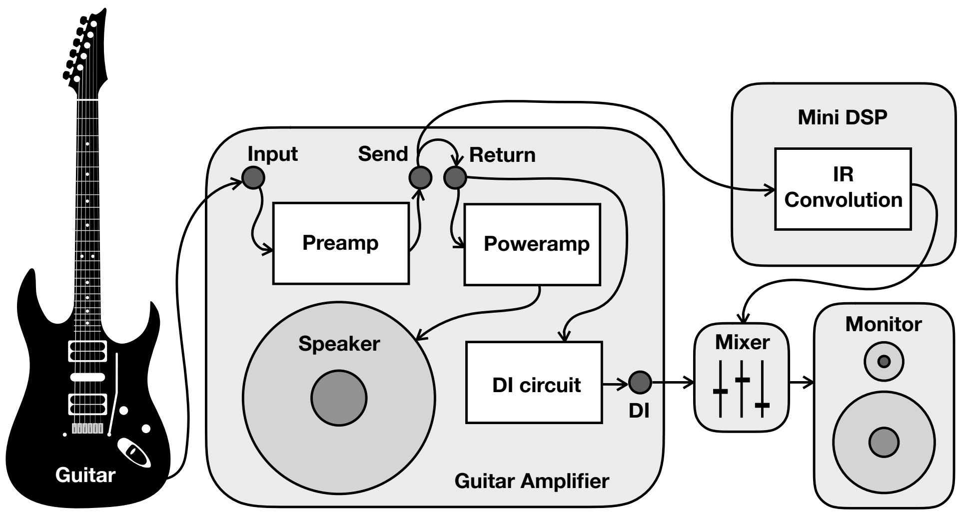

Figure 6.1 – Block diagram of listening test

The 3 configurations are shown in the following figures 6.2, 6.3 and 6.4, where the green highlighted modules shows the signalpath.

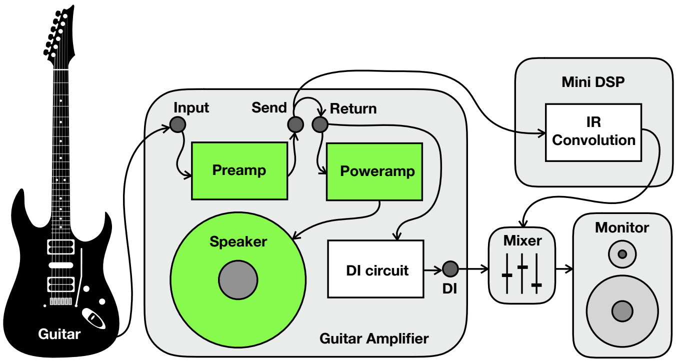

Figure 6.2 – Test 1: Guitar -> Preamp -> Poweramp -> Speaker

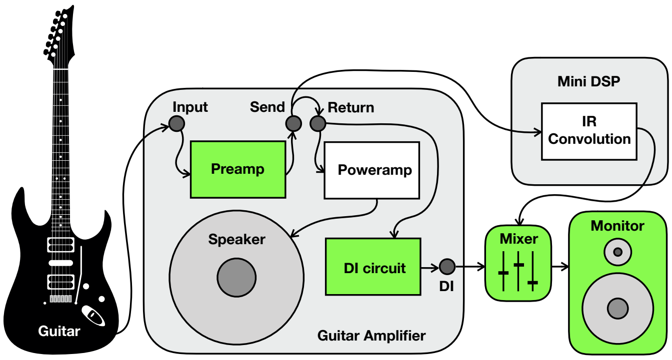

Figure 6.3 – Test 2: Guitar -> Preamp -> DI circuit -> Mixer -> Monitor

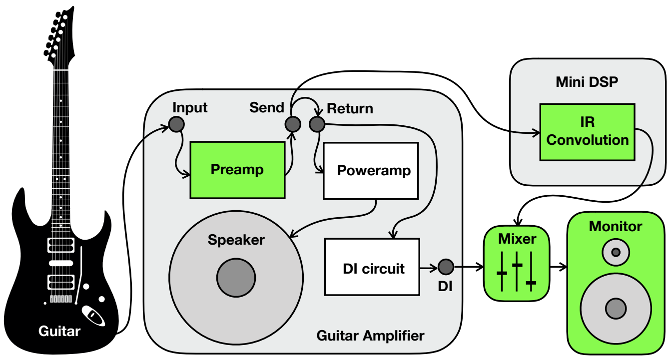

Figure 6.4 – Test 3: Guitar -> Preamp -> IR Convolution -> Mixer -> Monitor

In order to get reliable results for the listening test, there must be defined some practical boundaries.[1]

Participants

- Age between 20 to 40.
- Gender is irrelevant.
- Quantitative test, as many participants as possible.
- No information about the test must be given in advance.

The boundaries for participants will ensure the test will be attended by adults and gender will not be a determining factor. The attendees will not be given any information about the test setup to avoid any biased results. Furthermore by creating a quantitative test, the desired result statistics will be easier to determine.

Practical Setup

- 2-to-3 participants at a time to evaluate the test.
- The amplitude of the sound coming from the speakers will be at a constant level.
- Rhytmn of the played sound must be identical in all tests.

The intention behind these practical setup boundaries is to ensure all test participants experience the same tests.

Subjective Parameters

The participants will be evaluating the following three parameters

- Clarity
- Fullness
- Sharpness vs. softness

[1] Subjektiv Evaluering af Audiosystemer af Lars G Johansen, 2014

Presentation

The presentation will be a testsequence of ABR, with two processed signals (A,B) and one original reference (R). Process (A) is the built in DI circuit, Process (B) is the generated Impulse Response.

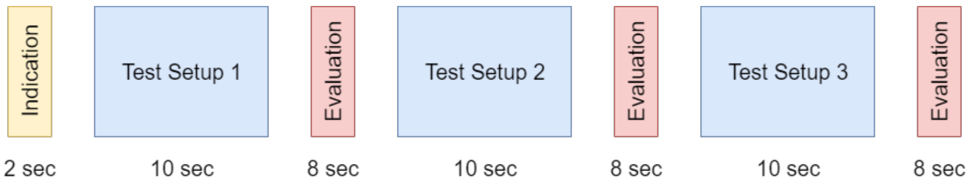

Figure 6.5 – Presentation of the ABR testsequence

To avoid biased results in the listening test, the presentation will be executed as ABR- and RAB testsequences. The evaluation part includes a pause of 2 seconds, each person will have to evaluate each setup within the given 8 seconds.

Evaluation

The evalution will take form of a basic template where the participants can cross an integer number between 1-10 to evaluate the parameters.

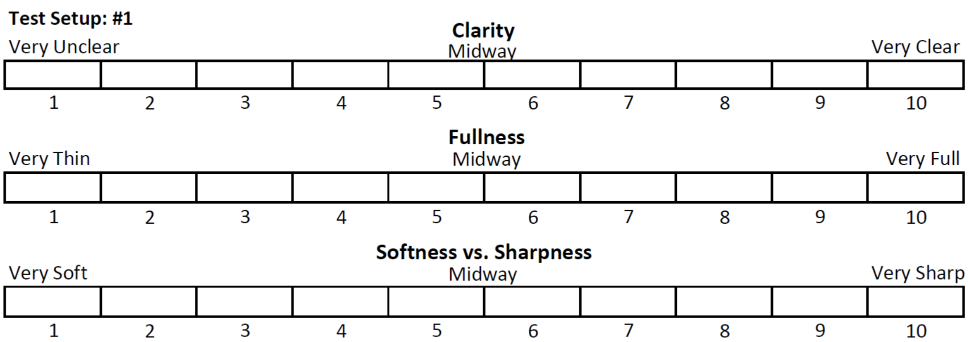

Figure 6.6 – Listening Tests Evaluation Templates

Chapter 7

Future work

More IR's of different guitar cabinets and microphones can be recorded and implemented in the project, opening the opportunity to developed a product for the guitarist marked. This product can be a pedal board sized unit with pre-loaded IR's that a guitarist can use in his effect loop. A second option could be a guitar rig unit with a preamp and a DI output, with the possibility of selecting different cabinet IR's and microphone IR's.

The Matlab code behind the project is rather static. This can be made more dynamic by introducing variables and functions that can improve the usability of the Matlab scripts.

The interpolation of the recorded signals could be made more robust by introducing an anti-aliasing filter to the process.

Chapter 8

Conclusion

We have succeeded in creating an IR of a guitar cabinet and digitally simulated the guitar cabinet sound through a studio monitor. From 9 measurement positions, 25 microphone positions was created which can be digitally selected and thereby shape the sound as desired.

The team have proved that it is possible to record an IR of a microphone, with the resulting frequency response almost identical to the frequency response given by the microphone manufacturer. It is therefore possible to convolute the microphone IR with the guitar cabinet IR and simulating the sound of a specific microphone.

Using methods from the courses, a listening test was conducted with the test sequence of ABR with two processed signals (A,B) and one original reference R. The test participants evaluated the listening test by filling a survey.

List of Figures