DSP i Audioteknologi
Guitar Cabinet Impulse Response

AUTE - Project

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Chapter 1

Project Description

Measure impulse response (IR) of guitar power amplifier and cabinet using both a linear measuring microphone (M1) and a typically applied microphone (M2). Measure 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 made.
Calculate the difference between the IR's of both the linear measuring microphone (M1) and the typically applied microphone (M2), in order to create a model of the typically applied microphone.
Use this model as a filter to the M1 IR's and create M3 IR's that therotically should be similar to M2 IR's.
Apply the IR's of M1, M2 and M3 to an electrical signal from an electric guitar running through a guitar preamp*. Describe the differences and qualities in tone using M1, M2 and M3 and also the difference in the microphone placement.
Finally attempt to tweak one or more of the IR's to create an IR that sounds as good as possible.

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. More than that, with a Celestion IR you get the bottled essence of the speaker, a state of the art recording studio, equipment, time and expertise all distilled into an Impulse Response file. A file that gives you great tone, quickly and easily, time and time again. [1] 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 ahighly accurate digital representations of speakers. IR’s allows the user to simulate a speaker’s tone through a speaker by digital means. [2]

The following figure shows 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.

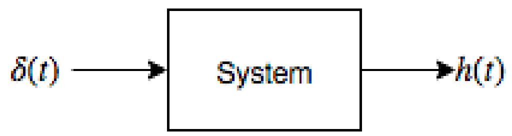

Figure 2.1 – System as a blackbox.

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.

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

The following figure shows an example of an impulse signal of an unknown system and the system's output as the generated impulse response.

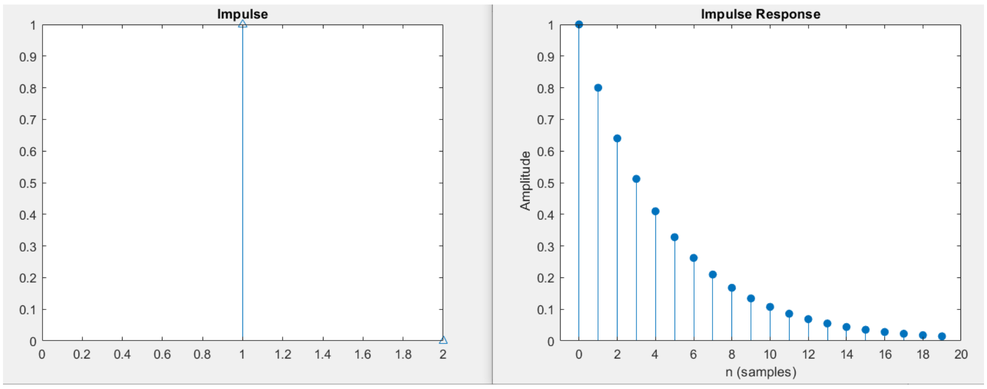

Figure 2.2 – Example of an impulse signal of an unknown system.

This impulse response can then be used to measure the system's frequency and phase response, which is shown in the following figure.

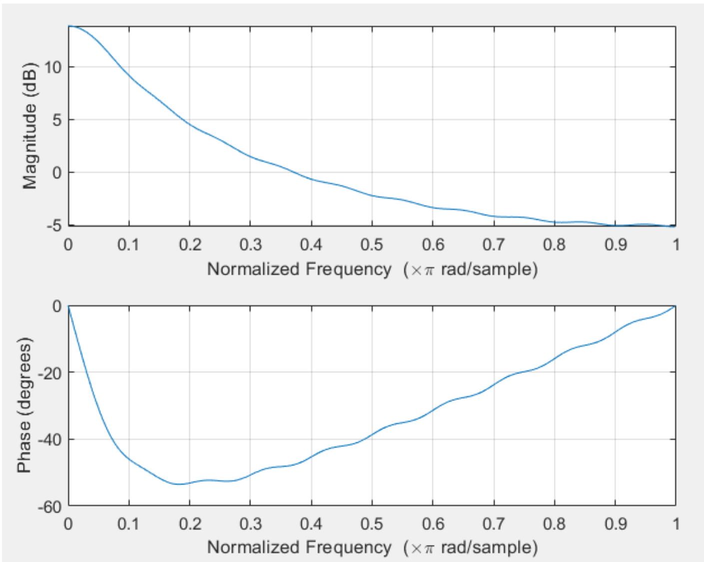

Figure 2.3 – System frequency and phase response.

Thus we can make a very accurate description of the system without needing the system itself, since we can use the impulse response to reproduce the system.

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).

The IR files are 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 don't even need to be aware of it. [3]

2.2 Recording an IR

When we wish to record or capture an impulse response of a guitar cabinet, a loud sound has to be played thru the cabinet's speaker which affects all the frequencies of the cabinet and creates echoes over the entire frequency range. A microphone measures the cabinet's response, resulting in an impulse response of the cabinet.
A loud bang or a sine sweep typically generates the loud sound 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. The amplitude may vary, and the sound can easily overdrive the microphone preamp and the microphone itself. In our case, we need the impulse response of the guitar cabinet itself and not the room. Therefore we can record a loud bang under controlled circumstances and play the recorded bang thru the guitar cabinet and thereby eliminate some of the problematics. However, it must be sure that the microphone recording the loud bang is not overdriven as this can affect the impulse response negatively.

Sweep

A sinewave sweep can also be used to create an impulse response. A wave generator generates the sinewave and sweeps it from 20Hz to 20kHz within

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

a given timeframe. The sweep is played thru the guitar cabinet, and a measurement microphone records the response. The result is an audio file with a duration of 5 to 10 sec. The audio file can not be used as an impulse response file straight away. The recorded audio file needs to be deconvoluted, which is the inverse of convolution, where the response of the guitar cabinet is sorted from the sinewave sweep. (put in reference to deconvolution).
In the year 2000, professor Angelo Farina at the University of Parma in Italy discovered that using an exponential sinewave sweep (ESS) instead of a linear sinewave sweep results in a better signal to noise ratio (SNR) and time-separates the harmonic distortion of the system from the linear response. The impulse response is then generated by convolving the recorded ESS audio with the clean inverse ESS.

2.3 Convolution algorithm

Her skal der skrivest om python algo

2.4 Choosing the Right Microphone

When capturing an Impulse Response from a speaker cabinet, by necessity we're sampling the signal's DAC (Digital/Analog Converter), power amplifier, 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.[4] This is hughly useful for loudspeaker measurements, noise emission studies, studies of attenuation and diffraction of sound and also for creating an artificial listening environment where alle the acoustic properties are determined by the experimental setup. [5] Within the scope of the project, we will investigate 2 different microphones: A Linear Measuring Microphone (M1) and a typically applied microphone (M2).
We will capture the guitar cabinet's impulse response from 9 fixed microphone

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

placements in the anechoic chamber at ASE's AudioLAB which we will use to compare the qualities of each microphone. After capture of the IR data we will tweak one or two IR's to create an IR with the highest quality possible based on our 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 l imits 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.
For the project, the group chose the Audix TM1 PLUS measurement microphone due to its linearity, accurate response, consistency, ease of use and affordability.
The Audix is designed to capture acoustic measurement for room analysis software programs, real time analyzers etc.

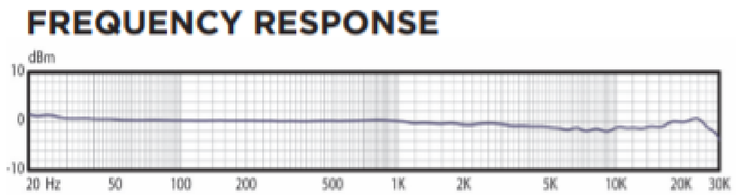

Figure 2.4 – Audix TM1 Measurement microphone frequency response.
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Giving its near-flat frequency response, we can compare the Audix TM1 PLUS' measurement data with the typically applied microphone M2's measurement data to determine the usefulness for 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. We will use the SM57 for our typically applied microphone M2 for measuring impulse responses.

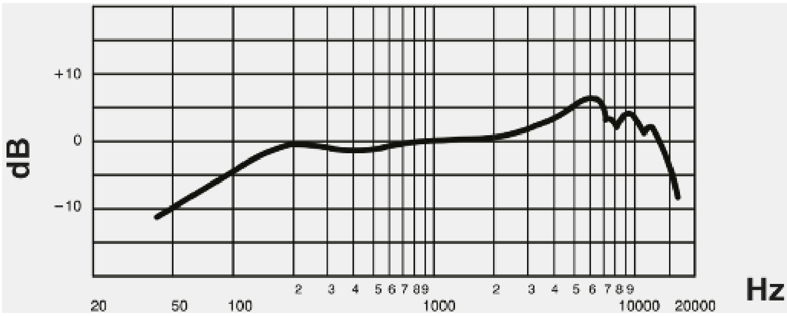

Figure 2.5 – SM57 Frequency Response
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2.5 Microphone Placement Matrix

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 2.6

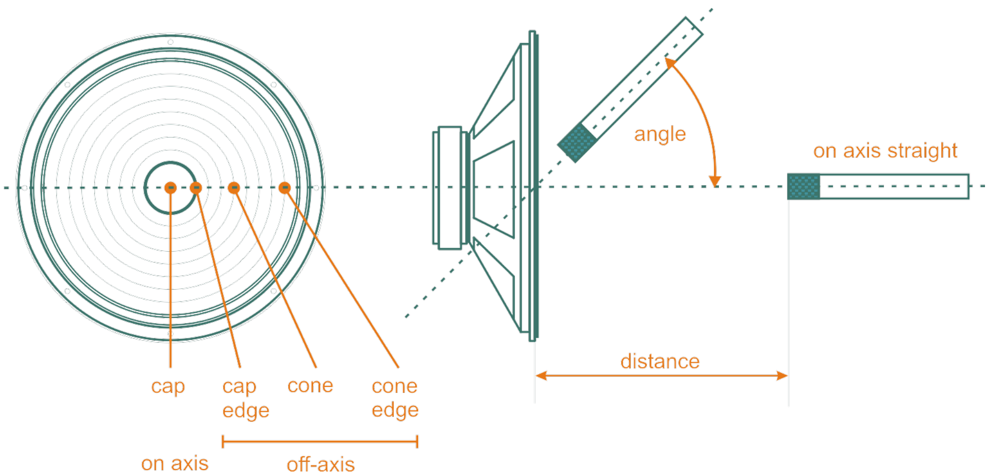

Figure 2.6 – Speaker Diagram. Frontal view of guitar cabinet speaker.
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Figure 2.7 shows the top view positions of the microphones in a matrix .

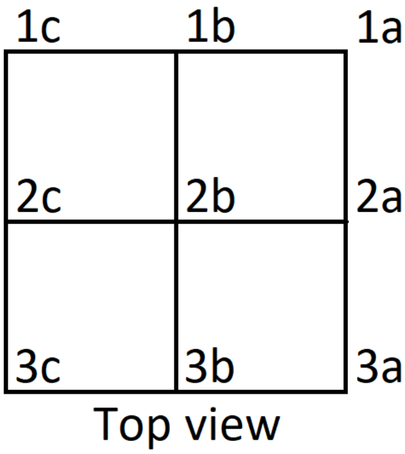

Figure 2.7 – Top view of microphone positions in a matrix.

Top view of microphone placement matrix. Column is the position from the cap, cone and the edge of cone, whereas the row represent the distance from the speaker to the microphone. The distance is critical when positiioning 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. The user positions the microphone as desired. 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.

2.6 Microphone IR

Figure 2.8 – 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. In our case, an Adam Audio S3V active midfield studio monitor is used for the IR 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 2.8 [9] together with a Shure SM57 microphone. The S3V is factory tuned. The frequency response of the S3V in figure 2.9 shows that the speaker is linear down to about 50Hz.

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

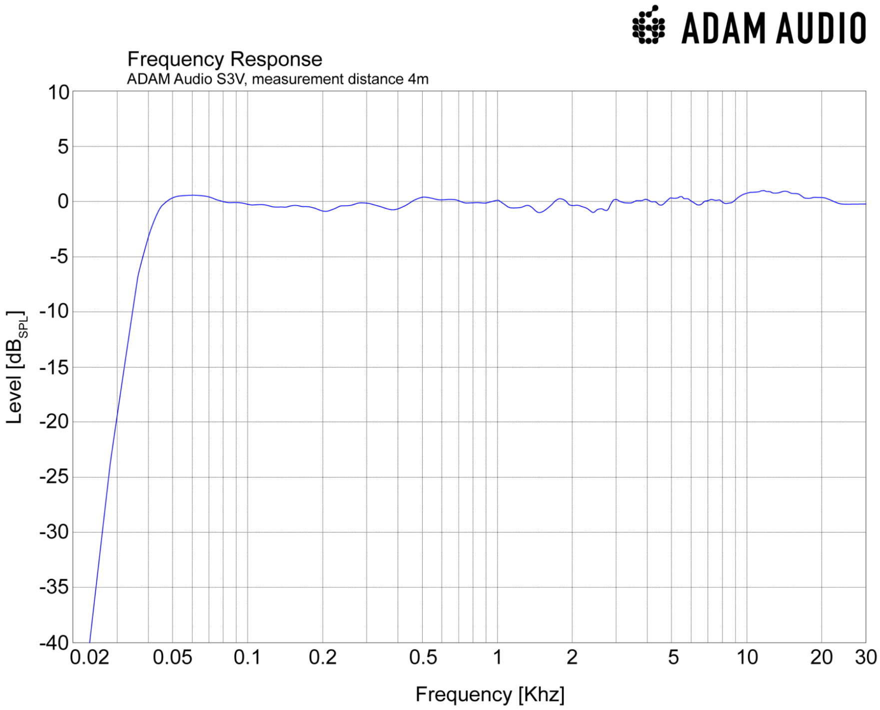

Figure 2.9 – Adam Audio S3V frequency response.

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 2.10 shows the result of the measurement.

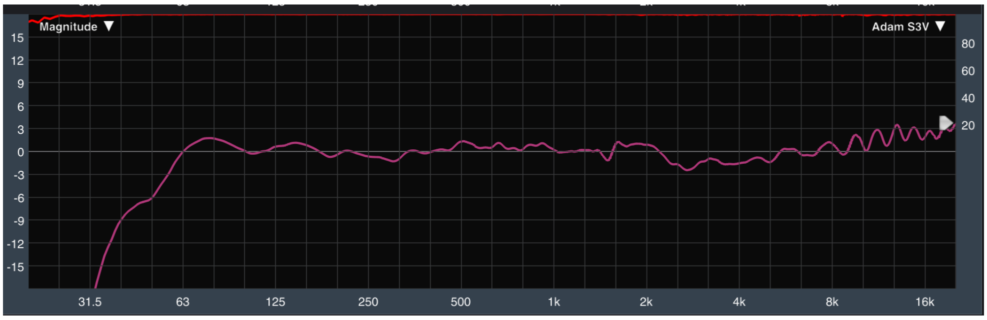

Figure 2.10 – Measurement result of Adam Audio S3V frequency response

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 **??** shows that the microphone might be causing the drop and the boost. But the drop and boost are within reasonable limits.

List of Figures