Optimal impulse response recording methods to emulate an acoustic guitar

Axel Strömberg
axelstr@kth.se

ABSTRACT

In this study impulse responses of an acoustic guitar are measured to create FIR-filters that give an electric guitar the timbre of an acoustic guitar. Sixteen impulse responses are measured to examine how exactly the impulse response should be measured. All impulse responses show the fundamental Helmholtz and lowest top plate resonances. The conclusion of the study is that the guitar should be struck with a wooden impulse hammer on the center, saddle of the bridge with muted strings. This gives the filter a warm, rich, acoustical timbre.

1. INTRODUCTION

The purpose of this project is to create an FIR-filter to emulate an acoustic guitar when playing on an electric guitar. Furthermore, the goal is to make a parameter study to discover how the impulse response is to be recorded to create a satisfying filter.

The motivation for constructing such a filter is to give artists an ability to emulate different guitars, all without switching instrument. Furthermore, since an acoustic guitar is sensitive to feedback an electric guitar with an FIR-filter can be more appropriate on a loud concert stage. A clear limitation to the implementation is the low pass filter characteristic from the magnetic pickups on the electric guitar.

The project was divided into four parts. First, sixteen different impulse responses were recorded when striking a guitar and ukulele with an impulse hammer. Then, two different raw samples from an electric guitar were recorded. Once the recordings were made two programs were written. The first program read the impulse responses and created FIR filters from them of desired lengths. The second read each filter together with each sample and then generated a filtered sample from the combination. The filters were then compared subjectively and analytically to find the best FIR-filter.

2. BACKGROUND

2.1 The two instruments

Even though an electric and an acoustic guitar are very similar in terms of strings, tuning and play style they produce a fundamentally different sound. This is since the strings themselves do not radiate sound efficiently. Instead, the vibrational energy of the strings must be converted to sound somehow. This difference in conversion methods are what fundamentally differentiates and acoustic and electric guitar.

In the electric guitar this is made with magnetic pickups. A magnetic pickup consists of a permanent magnet wrapped in a conductive wire. The magnet magnetizes the metal strings so that when they oscillate an electric current is induced in the conductive wire [1]. This signal is sent to a loudspeaker which amplifies the signal and produces the radiated sound. The loudspeaker is a part of the instruments and gives the electric guitar its timbre and thus the sound from an electric guitar is conventionally picked up by a microphone directed to the loudspeaker.

On the other hand, the acoustic guitar converts the energy from the strings to sound energy via the bridge and through the hollow body. The energy of the oscillating strings is converted to oscillations of the top plate through the saddle and bridge of the guitar. Here the hollow body acts like a Helmholtz resonator with a resonance frequency between $80 \leftrightarrow 100$ Hz. The resonance frequency f is described by the expression

$$f_0 = \frac{c}{2\pi}\sqrt{\frac{A}{V L_{\text{eff}}}} \tag{1}$$

were v is the speed of sound in air, $A = \pi a^2$ is the cross-sectional area of the hole with radius a, V_0 is the volume of the hollow body and L_{eff} is the effective length of the neck that is somewhat larger than the thickness of the top plate due to its curvature [2]. The importance of the formula is that the Helmholtz resonance frequency can be altered by perturbing the parameters A and L_{eff}. Furthermore, the top plate itself has resonance frequencies with fundamental modes between 150 Hz and 250 Hz and many radiating modes above that as well [3, p. 146]. Since the acoustic guitar transmits the energy to the top plate without external amplification the string vibrations of an acoustic guitar decay faster than that of an electric guitar.

2.2 FIR-filter

Let $x(t)$ be the input of a system and $y(t)$ the output. Often the relationship between x and y can be described by a linear differential equation

$$\cdots + b_2\ddot{y} + b_1\dot{y} + b_0 y = \cdots + a_2\ddot{x} + a_1\dot{x} + a_0 x + c, . \tag{2}$$

A *transfer function* [4] is a way to convert the differential equation to a simple one parameter equation using the *Laplace transform*

$$\mathcal{L}\left[x(t)\right]_s = \int_0^\infty \mathrm{e}^{-st} x(t)\,\mathrm{d}t = X(s)\,. \tag{3}$$

By taking the Laplace transform on both sides of (2) and rearranging the transfer function $H(s)$ can be expressed as

$$H(s) = \frac{Y(s)}{X(s)} \,. \tag{4}$$

The Laplace transform is useful due to its linearity and that differentials can be expressed in terms of the Laplace transform of the original function

$$\mathcal{L}\big[\dot{x}(t)\big]_s = sX(s) - x(0) \,.$$

This implies that (4) simply becomes a division of two polynomials, $H(s) = \frac{n(s)}{d(s)}$.

The *impulse function* $\delta(t)$ is defined

$$\begin{gathered} \delta(t) = 0, \quad t \neq 0 \\ \int_{-\infty}^{\infty} \delta(t)\, \mathrm{d}t = 1 \end{gathered} \tag{5}$$

and the *impulse response* of the system is the output $y(t)$ when $x(t) = \delta(t)$. By inserting

$$X(s) = \int_0^{\infty} \mathrm{e}^{-st}\delta(t)\, \mathrm{d}t = 1$$

in (4) it becomes apparent that $Y(s) = H(S)$ and the impulse response $y(t)$ is the inverse Laplacian of $H(s)$. That is, $h(t)$ is the impulse response.

When the impulse response of a system is known the output y can be calculated from the input x with *convolution*

$$y(t) = x(t) * h(t) = \int_{-\infty}^{\infty} x(\tau)h(t-\tau)\, \mathrm{d}t \,. \tag{6}$$

The impulse response can be easily be measured for a system by measuring the output y when exciting the system with $x \approx \delta$. For example, $h(t)$ describing a room reverberation can be determined by measuring the audio response in a room when popping a balloon.

An *FIR-filter* calculates the convolution (6) between an impulse response $h(t)$ and an input signal $x(t)$. In practice $h(t)$ is cut to an appropriate length to reduce unwanted reverberation and (6) is replaced with a *discrete convolution*

$$y_k = \sum_{i=0}^{M} h_i x_{k-i} \tag{7}$$

where the inputs are discrete arrays; $x = \{x_0, \cdots, x_n\}$, $h = \{h_0, \cdots, h_m\}$, $y = \{y_0, \cdots, y_{n+m-1}\}$.

When analyzing the impulse response the *waveform* in the time domain is only one of the interesting aspects, often the *Fourier transform* is used to show the waveform in its frequency domain.

$$\mathcal{F}\big[x(t)\big]_\omega = \int_{-\infty}^{\infty} \mathrm{e}^{-i\omega t}x(t)\, \mathrm{d}t = X(\omega) \tag{8}$$

Due to its complex nature the Fourier transform can be difficult to understand, however its intuitive that an input signal $x(t)$ having with large oscillations with the angular frequency ω' should resonate with $e^{-i\omega' t}$ in (8) and give a large value of $X(\omega')$. Thus $X(\omega)$ is measurement of the different partial angular frequencies in $x(t)$ and their relative magnitude. In practice the numerical *discrete Fourier transformation, DFT* is used. Here x is given as a discrete array together with an integer framerate and the convolution is determined by

$$X_k = \sum_{n=0}^{N-1} x_n e^{-i\frac{2\pi}{N}kn} \,. \tag{9}$$

When analyzing a longer sample with many chords or notes in series a *spectrogram* that shows $X(\omega)$ over time as a heat map can be used.

3. METHOD

To make a good parameter study sixteen different impulse responses were measured on a guitar and ukulele. These were then read and trimmed to the durations $150, 500$ and 1000 ms. Two samples of an electric guitar were recorded and filtered with the different filters giving out $3 \times 16 \times 2 = 96$ filtered files. These were initially compared subjectively and then some were plotted in a spectrogram to compare with the corresponding transfer function.

3.1 Recordings

The sixteen different recorded **impulse responses** are listed in Table 1. The measurements were made with all combinations of guitar/ukulele, wood/metal impulse hammer, strike at the center/edge of the bridge and with muted or freely vibrating strings.

Once the guitar and ukulele were tuned they were placed in order on rubber foam and one string was removed. A small clip microphone was hung 3 cm into the body in the center of the hole. The room was connected to a corridor and wasn't well isolated. If a door was opened to the corridor during a recording this measurement was discarded.

The instruments were struck on the bridge with either a wooden cylinder or metal screwdriver. The wooden cylinder was of medium hard wood with $\varnothing \approx 0.5$ cm and the height $\approx$ 20 cm. The screwdriver was flat headed and medium-sized ≈ 15 cm long.

When the center of the bridge was struck the contact point was on the saddle of the bridge where the string had been removed and ≈ 2 cm from the edge of the impact hammer. That is, on the smooth cylindrical side. When the edge of the bridge was struck this was not possible and instead the contact point was ≈ 2 cm from the edge of the bridge and on the top edge of the cylinder or metal screwdriver.

If the strings were accidentally struck by the impulse hammer the recording was discarded. If the strings weren't muted it was important that they resonate from the bridge and not directly from the impulse hammer. If the strings were to be muted a piece rubber foam was gently pressed down on the base of the fretboard to absorb any vibrations from the strings.

The two **samples** were recorded directly from the jack output of a Squier by Fender electric guitar with GarageBand on an iPad. The first recording, `chords.wav`, is

	Instrument	Impulse hammer	Impulse position	Strings
1	Guitar	Wood	Bridge Center	Free
2				Silent
3			Bridge Edge	F
4				S
5		Metal	BC	F
6				S
7			BE	F
8				S
9	Ukulele	Wood	BC	F
10				S
11			BE	F
12				S
13		Metal	BC	F
14				S
15			BE	F
16				S

Table 1. Table over the impulse response measurements. Measurements were made with all combinations of Guitar/Ukulele, Wood/Metal impulse hammer, strike at Bridge Center/Bridge Edge and strings Free/Silent. For example, the file `6-G-M-BC-S.wav` is the sixth recording, on a guitar, with a metal impulse hammer struck at bridge center and strings are muted to be silent.

an eight-seconds long recording of some simple, looping chords. The second recording, `notes.wav`, is an eight second long recording of melody played in single notes.

3.2 Filtering programs

Firstly a program was written that created FIR-filters of desired length from all impulse responses. It reads all `.wav` files in an `impulse_responses` directory sequentially. The program then calculates the impact index as the first index where the amplitude is half of the maximum amplitude in the recording. To assert that no part of the filter is trimmed it keeps 20 ms of the recording before the impact. The user is prompted to leave a filter duration in ms and that duration will be kept after the impact. The filter duration was set to 150, 500 and 1000 ms. The rest of the recording is discarded and the result is written to a new `.wav` file in a new `FIR_filters` folder together with a `.png` time- and frequency plot of the filter.

A program was written that read all FIR filters of desired length and all the samples. It then filters the samples with each FIR filter according to (9). The two programs are available at [5].

4. RESULTS

4.1 FIR-filters

In Figure 1 two 1000 ms FIR-filters are plotted with the strings muted or freely vibrating. Both in the plot and in the audio file one can notice how the strings give a longer impulse response.

Figure 2 shows the impulse responses of a guitar and

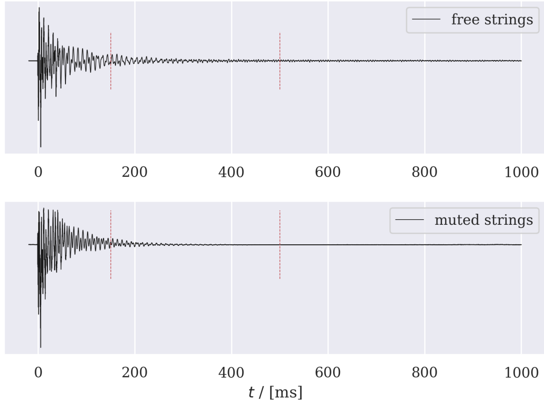

Figure 1. Comparison between freely vibrating or muted strings. The impulse responses are from measurements on a guitar, with a wooden impulse hammer, struck on the center of the bridge. The vertical dotted lines shows the cut of mark for the 150 ms and 500 ms filters.

ukulele. The Helmholtz resonance frequencies are 100 Hz and 200 Hz for the guitar and ukulele respectively. This correlates well with (1) if the ukulele is half as long. If the the Helmholtz resonance frequency of the guitar is given by

$$f_0 = \frac{c}{2\pi}\sqrt{\frac{A}{VL_{\text{eff}}}}$$

a ukulele of half the length but same body proportion will have a Helmholtz resonance frequency of

$$\frac{c}{2\pi}\sqrt{\frac{A/4}{V/8L_{\text{eff}}/2}} = 2f_0\,.$$

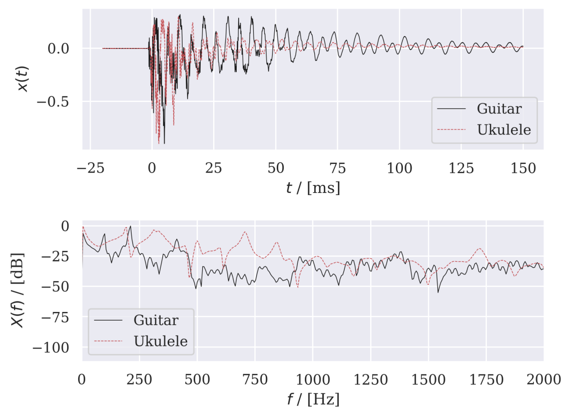

Figure 2. Impulse responses of a guitar and ukulele. The measurements are made with a wooden impulse hammer, struck on the center of the bridge and muted strings.

The impulse responses from different impulse hammers are shown in Figure 3. The impulse response from the wooden hammer initially decays faster than the metal impulse hammer. The larger more continuous oscillations from the metal hammer correlate with more intense lower resonances in the filtered sample.

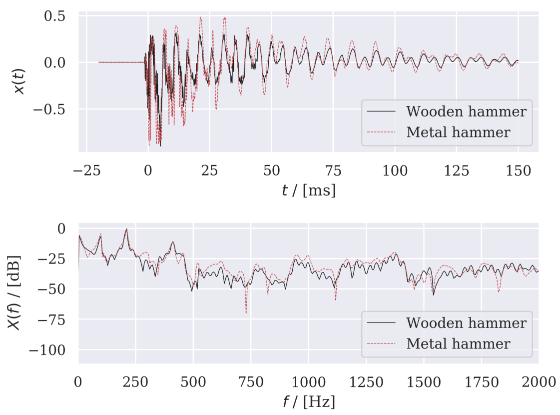

Figure 3. The impulse responses of a guitar struck with a wooden and metal impulse hammer. The guitar is struck on the center of the bridge and the strings are muted.

In Figure 4 the impulse responses of a guitar struck on the center or edge of the bridge are shown. The waveforms look similar but in the frequency domain one can see that by striking on the center of the bridge modes of higher frequencies are excited more intensely which correlates to a result with more presence of the higher frequencies.

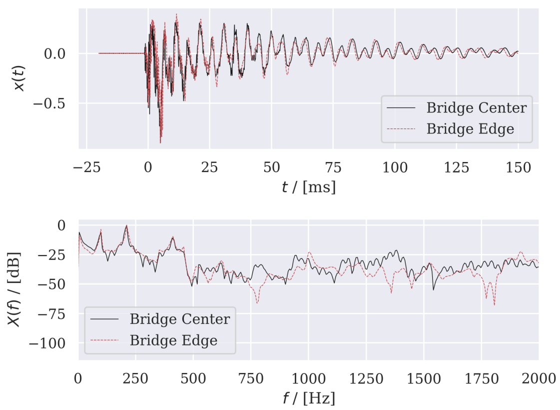

Figure 4. The impulse responses of a guitar struck on the center and edge of the bridge. The guitar is struck with a wooden impulse hammer and the strings are muted.

4.2 Samples

A logged spectrogram of the chords sample is shown in Figure 5.

4.3 Filtered samples

The filter in the study giving the best acoustical sound is the filter from the impulse response recording `02-G-W-BC-S.wav` shown in Figure 6. The spectrum of the chords sample filtered with this filter is shown in Figure 7. When comparing Figure 5, 6 & 7 one can see that the sample and filtered sample contains the same partials from the strings. However the Helmholtz frequency around 100 Hz as well as the fundamental top plate frequencies 200 Hz and 400 Hz are strongly resonated in the filtered sample. This correlates with the observed warm acoustical timbre.

5. DISCUSSION

The optimal **filter duration** was 150 ms. With a duration higher than this an unnatural small-room reverberation occurred. Long duration was extra problematic with freely vibrating strings which can be seen in Figure 1.

A comparison between the different **instruments** is shown in Figure 2. Most acoustic guitar filters gave a rich acoustical timbre while the ukulele filter sounded neither like a ukulele nor an acoustic guitar. This is since the ukulele filter had an impulse response from a ukulele body with high fundamental resonances but the string sample was from an electric guitar. The electric guitar has longer strings than a ukulele and thus lower fundamentals. Because of this, the strings of the electric guitar resonated well with the Helmholtz resonance in the acoustic guitar but the higher resonance in the ukulele gave the samples an unnatural sound.

As for the different **impulse hammers** the wooden impulse hammer gave a warm, wooden acoustic sound. The metal impulse hammer gave the sample a more artificial sound. The higher frequencies were more present in the filters from the wooden impulse hammer. This result is unexpected since a metal impulse hammer is more rigid and should give in impulse function closer to (5) and thus a more accurate impulse response h. It is, however, possible that the wooden impulse hammer better represented how strings interact with the bridge and thus these filters gave a more authentic sound compared to the artificial sounding filters from the metal impulse hammer. When analyzing the spectra in Figure 3 one can see that the wooden impulse response have more defined resonance peaks as well as a more regular pattern corresponding to the regularity of the resonances in the top plate.

The best **position** to strike the bridge was on the center of the saddle. This strike gave the filter a more natural and rich sound with more presence in the higher frequencies above 1 kHz which is illustrated in Figure 4. Furthermore, when striking on the center of the bridge the filtered samples included more of the direct sound from the electric guitar strings.

The best impulse responses was recorded with muted **strings**. This is since some notes in the sample resonate greater than others with the tuning of the strings in the impulse response. With free strings, the guitar should be tuned to the same notes as the current notes in the sample which is inconvenient.

6. CONCLUSION

The conclusion is that the optimal FIR-filter is generated from an impulse response measured when striking the center of the bridge with a wooden hammer while the strings are muted. This will give a rich acoustical timbre to the electric guitar signal.

Further studies to improve the filters should use a microphone and a guitar of higher quality. Then, microphone

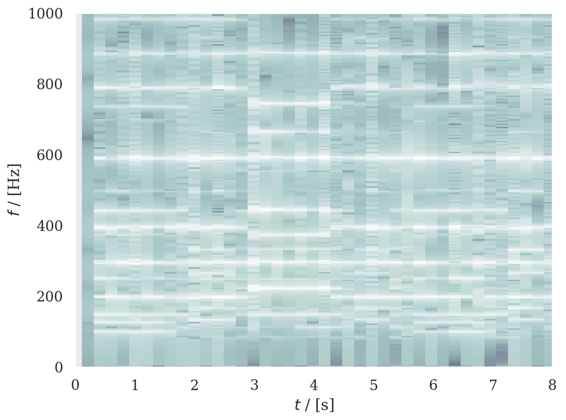

Figure 5. Logarithmic spectrogram from the recorded sample of chords played on an electric guitar.

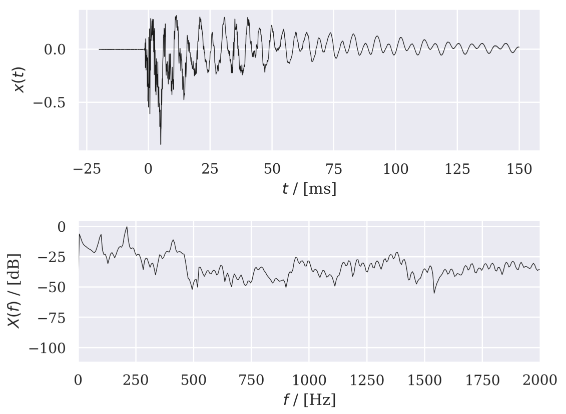

Figure 6. The 150 ms FIR-filter from the recording `02-G-W-BC-S.wav`, concluded to be the best filter in the study. The impulse response is from a guitar struck with a wooden impulse hammer on the center of the bridge with muted strings.

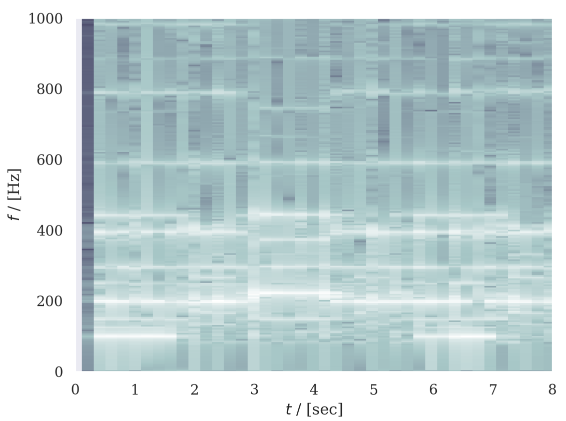

Figure 7. Logarithmic spectrogram from the chords sample filtered with the 150 ms FIR-filter in Figure 6.

and guitar placement should be examined in a sound isolated room. Furthermore one could generate the impulse response with deconvolution similar to the second method in [6]. This, however, has its own problems since the method itself lowers the important Helmholtz frequency.

7. REFERENCES

[1] Wikipedia, "Pickup (music technology) — Wikipedia, the free encyclopedia," http://en.wikipedia.org/w/index.php?title=Pickup%20(music%20technology)&oldid=883756915, 2019, [Online; accessed 22-February-2019].

[2] ——, "Helmholtz resonance — Wikipedia, the free encyclopedia," http://en.wikipedia.org/w/index.php?title=Helmholtz%20resonance&oldid=873097581, 2019, [Online; accessed 22-February-2019].

[3] N. H. Fletcher and T. D. Rossing, *The physics of musical instruments / Neville H. Fletcher, Thomas D. Rossing*, 2nd ed. Springer New York ; London, 1998.

[4] Wikibooks, "Control systems/transfer functions — wikibooks, the free textbook project," https://en.wikibooks.org/w/index.php?title=Control_Systems/Transfer_Functions&oldid=3513676, 2019, [Online; accessed 22-February-2019].

[5] A. Stromberg, "Fir_filter," https://github.com/axelstr/FIR_filter, 2019.

[6] P. Henri, V. Välimäki, and M. Karjalainen, "A digital filtering approach to obtain a more acoustic timbre for an electric guitar," vol. 2015, 09 2000.