FEDERAL UNIVERSITY OF TECHNOLOGY - PARANÁ – UTFPR
ELECTRONIC ENGINEERING
ELECTRONIC'S ACADEMIC DEPARTMENT

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GUITAR DIGITIZER

GRADUATION FINAL PROJECT

CURITIBA, BRAZIL

2017

CRISTÓVÃO DINIZ TREVISAN
VICTOR VOLOCHTCHUK DE ARAUJO

GUITAR DIGITIZER

Project presented to the Electronics Academic Department as graduation material for the course of Electronic Engineering at UTFPR

Advisor: Gustavo Benvenutti Borba

Curitiba, Brazil

2017

Cristóvão Diniz Trevisan, Victor Volochtchuk de Araujo
Guitar Digitizer/ Cristóvão Diniz Trevisan, Victor Volochtchuk de Araujo. – Curitiba, Brazil, 2017-
56 p. : il.

Supervisor: Gustavo Benvenutti Borba

Graduation Final Project – Federal University of Technology - Paraná – UTFPR
Electronic Engineering
Electronic's Academic Department, 2017.

1. Hexaphonic Guitar 2. Digitizer 3. MIDI 4. Pitch Detection I. Guitar Digitizer II. Gustavo Benvenutti BorbaIII. Federal University of Technology - Paraná IV. Electronic Engineering

CDU 02:141:005.7

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2017

This work is dedicated to those who supported us through engineering course: families, friends and teachers - even (if not more) when we were at it's hardest moments.

Acknowledgements

We give our special thanks to Eng. Mikhail Anatholy Koslowski, who gave us both intellectual (since he once used an equipament with the same purposed) and physical support (with compenents importation and equipaments). Also to our advisor who was present even in the moment that the initial idea came to life, in a situation nobody else would have stayed.

Abstract

This work presents a complete system for guitar signal acquirement and processing, in order to translate the raw audio signal to the corresponding musical notes. It is important to mention that, while this is relatively easy to be accomplished in instruments that have separated keys for each note (e.g. piano), this is not the case for instruments like the guitar – the one addressed in this work – that have a single output for multiple notes (each string has about 15 notes). The guitar is a harmonic instrument, as it can play multiple notes simultaneously, making the translation a challenging task. The proposed approach employs a specially designed pickup, developed and constructed by the authors, with dedicated coils for each string (in contrast to the single coil regular models). Audio amplifiers based on INA326 OPAMPs condition the coils' signals, which are further converted to digital by an acquisition module based on an ARM Cortex M3 microcontroller, model STM32F103C8T6. The digital samples are transferred to a PC through the USB bus and processed using pitch detection algorithms. The software is based on contemporary tools such as Node, NW.js, React+Redux and Functional Programming. Special emphasis has been dedicated to the applied nature of the project, in the sense that the goal was to obtain a *ready-to-use* prototype, despite the limitations on the accuracy of the final translation. In other words, the present work does not focus on state-of-the-art signal processing approaches in order to obtain a near perfect translation. Instead, it prioritizes an effective hardware and software architecture aiming a completely functional system for real-time guitar signal translation into the corresponding notes. Two pitch detection algorithms were evaluated: YIN and McLeod. Results show that YIN algorithm performs better for this application, achieving decent accuracy values in the practical experiments.

Key-words: Guitar digitizer, MIDI, pitch detection, hexaphonic.

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List of abbreviations and acronyms

ADC	Analog to Digital Converter
API	Application Programming Interface
AWG	American Wire Gauge
DIY	Do It Yourself
DMA	Direct Memory Access
DOM	Document Object Model
fft	Fast Fourier Transform
GUI	Graphical User Interface
IC	Integrated Circuit
IDE	Integrated Development Environment
ifft	Inverse Fast Fourier Transform
MIDI	Musical Instrument Digital Interface
npm	Node Package Manager
PCB	Printed Circuit Board
SMD	Surface Mounting Device
UI	User Interface
USB	Universal Serial Bus
n.d.	No Date

List of symbols

Ω	Ohm resistance unit
μC	Microcontroller
V	Volts
mV	milivolts

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1 Introduction

With the advance of technology, music - and musical instruments - have also evolved to use it's advantages. They are a lot of use cases, the most noticeable ones being music annotation and creation (through electronic instruments, also known as synthesizers). All of the modern musical software and hardware use the same format to communicate, called MIDI.

To translate music playing it is needed to know which note is being played at a given time. This make it very easy to translate instruments that have separated keys for each note (the most noticeable one being piano) to MIDI, but very hard do the same for instruments like the guitar, that have a single output for multiple notes (each string has about 15 notes). The guitar is also a harmonic instrument, as it can play multiple notes simultaneously, which makes it's digitization even harder.

There are already a few commercial solutions for this, but not a very performant and cheap one. Recently, a new pure software solution was released at a reasonable price which works very well for live MIDI playing, but not enough for music annotation. There are also a few hardware solutions available at the market, which perform well, but are very expensive. Table 1 shows the most relevant solutions in the current market.

Table 1 – Market Solutions

Name	Price (US$)	Usage Complexity	Live Performance	Annotation Performance
Roland GK3 + GhostHexpander + GI20	700	Hard	High	High
Godin Freeway + GhostHexpander + GI20	800	Hard	High	High
Jam Origin - Audio to MIDI	100	Easy	High	Low/Medium
Migic	40	Easy	Medium	Low

Source: authors

The price for such technologies as listed in Table 1 may not be viable for every one, thus the necessity of building a new product with a good performance and low price. Our current prototype price is US$ 77.66 (Table 9), with estimated production cost of US$ 40.21 (Table 10) for each of 1000 units. This can lower (about 30 dollars) if using a cheaper amplifier circuit. The numbers listed show how this project can be viable as a product.

A great amount of technology needs to be built around this project as software, which can also foment a market grow for this kind of product and therefore create space for new ideas in the area. The area around digital guitar processing is also barely touched

in Brazil.

1.1 Social Motivation

Guitar music annotation is very expensive as of today. The reason for this is that it takes a lot of effort to write a music sheet. In the way to make music annotation cheaper and more accessible this project also aim to do this kind of work in a cheap and easy way (that does not even require the knowledge of music annotation).

This would create great social meaning as it makes possible for everyone that play the guitar to quickly write down their own compositions and arrangements and thus share their culture to the world.

In specific, this project was thought for the people that make their own arrangements but don't share it because of the great time cost involved, most of them being YouTube guitar players all over the world. The few that do share their annotations today usually charge for it, thus blocking culture sharing.

1.2 Foremost Project Decisions

There are many ways to accomplish this objective, a few considered were through mechanical pressure detection, pure software (like the commercial solutions) and finally electromagnetic fields.

Mechanical pressure (or buttons) was discarded because it still needs to detect string vibration, which requires electromagnetic detection (to be cheap). But the latter can be used as the only input, so it does not make sense to use both.

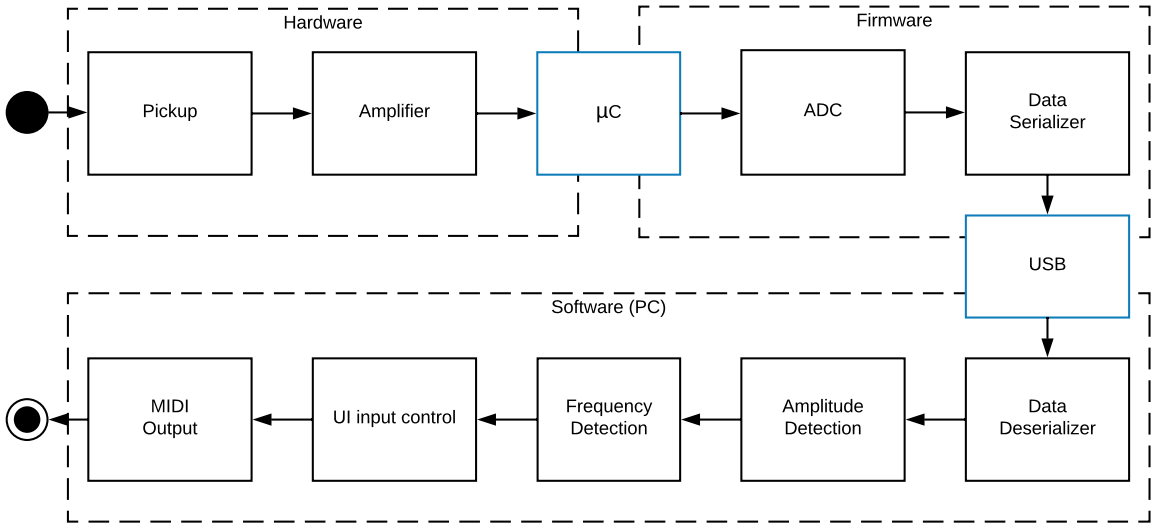
Software only detection is too hard to implement and does not guarantee good results. This approach is still being developed now days by using the most recent advanced techniques of machine learning, which is out of the scope of this project as is so state of the art that requires a really high amount of human resources to accomplish, which is not compatible to our budget (as we are mere undergrad students).

Electromagnetic detection is nothing new (it is the base principle of electric guitars). The only unusual approach we need is to get a separated signal for each string, thus removing most of the software complexity quoted above, because we can them deal with monophonic signals (as each string can only play one note at a time). It is still complex to build a project with this decision, but it is feasible, and so the chosen option.

1.3 System Overview

The Figure 1 shows a flowchart that describe how each part (at a high level abstraction) of our system connects to next one. It also divides each block by a system architecture view side.

Figure 1 – System Overview



Source: Authors

2 Hardware

2.1 Pickup

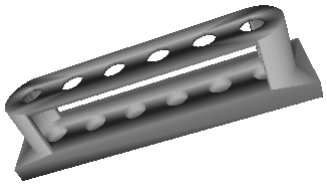
Electric guitars pickups are usually built by wrapping copper wire around magnets. The working principle is based on the variation of magnetic field, created by the string vibration. The vibration frequency of the string induces an electric signal on the output of the pickup (WALLACE, 2004; NAVE, n.d.).

The designed pickup used the following main components:

- Base to assembly the set up magnet+coil
- 6 magnets
- Copper wire to wrap the magnets
- Cover to attach the set up on the guitar

After some studies it was decided to build the pickup base using a 3D printer, due the availability and reduced cost of this project. The model is showed on Figure 2 and was projected using the AutoCAD software.

Figure 2 – 3D pickup base project

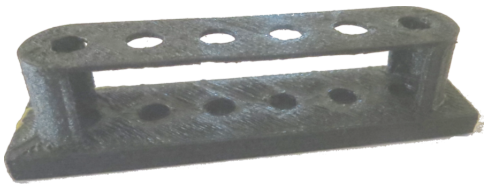


Source: Authors

There were two materials available to print the model, *PLA* (Polylactic Acid) and *ABS* (Acrylonitrile Butadiene Styrene) (3D..., 2013). It was decided to print the model on PLA because it attends the requisites of robustness of the project, is faster to print and have a lower cost when compared to the other materials. The built part is showed on Figure 3

With the pickup base ready, the coils were dimensioned. The area of a turn was considered as a square with side dimensions equal to the wire diameter, and using that it was estimated the wire diameter as in Equation 2.2.

Figure 3 – 3D pickup base



Source: Authors

$$Turns = \frac{Area\ between\ magnets}{Area\ of\ each\ turn} = \frac{w * l/2}{d^2} = \frac{5mm * 12mm/2}{d^2} = 1000 \tag{2.1}$$

$$d = \sqrt{\frac{w\ *\ l}{2\ *\ Turns}} = \sqrt{\frac{5mm * 12mm}{2 * 1000}} \cong 0.17mm \cong 34AWG \tag{2.2}$$

Equation 2.1 was used with 1000 turns as a reference, based on a similar project (HEXAPHONIC. . . , n.d.). Using the area between 2 magnets (5 mm wide and 12 mm length) and the estimated area of a wire turn it is possible to calculate the wire diameter, as in Equation 2.2 - resulting in a diameter of 0.17 mm, that can be converted to 34 AWG, approximately.

This diameter (number of turns) is only feasible with an automated process. The present project was built by manually wrapping copper wire around each magnet (Figure 4), so only 500 turns were feasible.

After mounting the coils, as showed on Figure 5 it was performed a test to verify the real voltage value induced by the variation of the magnetic field. It was used the LM 741 (LM741. . . , 2015) with a gain of 10, because it wasn't possible to verify the value on the coil output. With this simple test circuit it was possible to verify that the output value on the IC was 8 mV, this value indicate that the output value on the coil is around 1 mV.

Figure 4 – Magnets used on the project

Source: magnets (n.d.)

After testing with one coil, the complete assemble was made, as showed on Figure 6 and Figure 7.

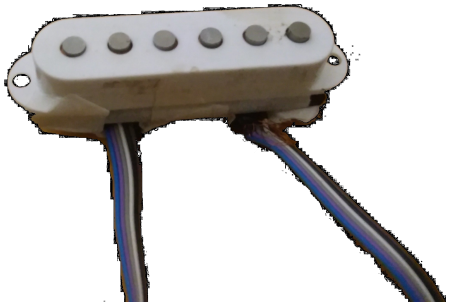
Figure 5 – Single Coil assembled

Source: Authors

Figure 6 – Coils assembled

Source: Authors

Figure 7 – Assembled Pickup



Source: Authors

2.2 Amplifier Circuits

Amplifiers circuits (AMPLIFIER..., n.d.) are circuits which increase an electrical input signal according to a specific transformation function (gain) for each different topology. The circuit for small input signals is normally composed by an operational amplifier, resistors, trimmer potentiometers to adjust the gain and capacitors for frequency filtering.

It was required by the project an amplifier with a gain of at least 1000, beacuse, as it was mentioned on section 2.1, the tested output coil presented the value around 1 mV and the circuit needs a signal of around 1V to work as expected on the AD converter, that can be powered by a sigle 5 V supply, provided by the USB port. The circuit should be cheap (and compact) when compared to the existent systems, which usually need multiple

complex modules to perform the same functionalities as proposed in this project.

It was researched some types of amplifiers circuits in Milmann e Halkias (1981) and in Carter (2000). After verifying some amplifier topologies it was selected one circuit which attend to this project's requirements. This chosen circuit uses the Texas Instruments INA326 Instrumental Amplifier.This topology can be supplied with a single 5V source and can reach the desirable gain without distortion on the desired frequency range (human audible frequencies) with a single amplifier per channel.

2.2.1 Amplification Project Based on INA326

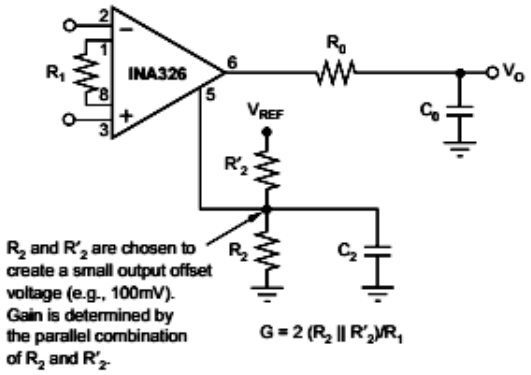
The project using the INA326 IC started by researching the *component datasheet* (INA326. . . , 2004) and the *supplier catalogue* (CARTER, 2000). In these documents it was verified the circuit topology in Figure 8 which provides the desired gain for the pickup signal respecting this project's initial requests.

This recommended circuit's gain is obtained by Equation 2.3 which is provided by the *component datasheet* (INA326. . . , 2004):

$$G = 2 * \frac{(R_2||R'_2)}{R_1} \tag{2.3}$$

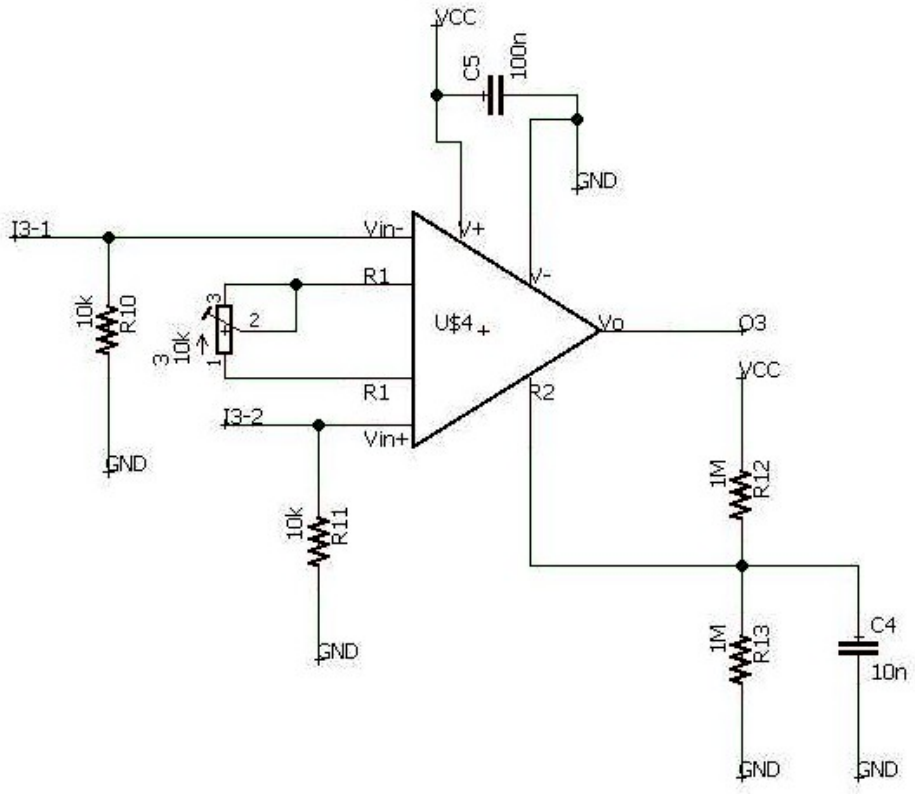
Both the R_0 resistor and the C_0 capacitor were excluded because they were not relevant for project requirements. The values for the remaining components were calculated and the circuit was build to test it's results for the desired application. After verifying the circuit does work properly, giving the desirable value on the output and without distortions on single supply method, the schematic in Figure 9 was developed using the software CadSoft Eagle Professional 7.6.0.

Figure 8 – INA326 topology



Source: INA326. . . (2004)

Figure 9 – INA326 Schematic Circuit



Source: Authors

It was decided to use trimmer potentiometers on the amplification circuit to regulate the gain for each channel. The desired gain range was achieved by selecting the component values showed at Figure 9. The complete schematic is just a replication of Figure 9 for each channel, and can be seen at Figure 30.

A PCB was then built, using the same software described on the schematic modeling, resulting in the board seen at Figure 10. It has two layers, with track width of 15 mils. It was assembled on FR4 dual layer copper board, using both through-hole and surface-mount technologies. This choice was made due the facility of the assembly of the through-hole components and the availability of the IC only in SMD (SOP-8) encapsulation. The generated gerber files were sent to an internal manufacturer at UTFPR, which gave the board seen at Figure 11. The component list for this board is as in Table 2.

After soldering all the components the assembled PCB was as showed at Figure 12, it was performed some bench tests to verify the functionality of the amplifier circuit. The result showed that the circuit attends the required functionality, amplifying the signal from a peak-to-peak of about 1mV to 1.5V for the entire range of audible frequencies, proving that the circuit is working perfectly and attending the demands of the project.

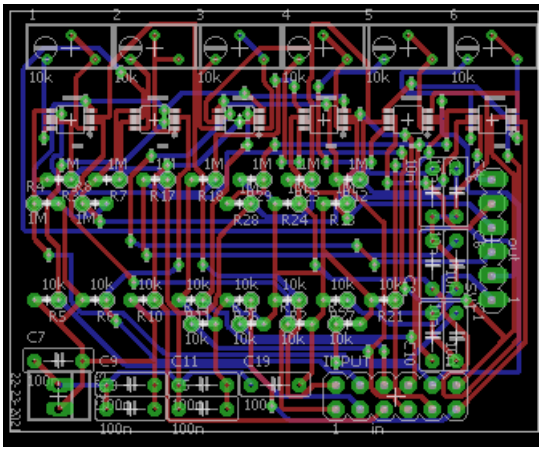
After the bench test, the system was connected to the pickup to verify if the guitar signal would be amplified as needed for the conversion process. The result was satisfactory

Table 2 – INA Board BOM

Name	Quantity	Value
INA326	6	
Trimmer Potentiometer	6	10kΩ
Ceramic Capacitor	6	10nF
Electrolytic Capacitor	6	100nF x 50V
Resistor	12	10kΩ
Resistor	12	1MΩ
Pin bar	1	6 positions
Pin bar	1	12 positions dual track
Pin bar	1	2 positions

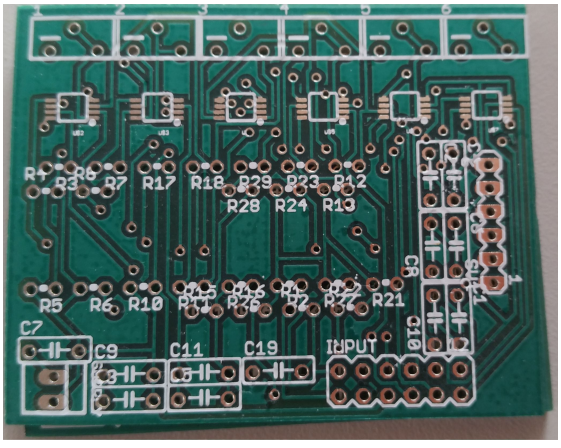
Source: Authors

Figure 10 – Projected INA PCB



Source: Authors

Figure 11 – PCB Project

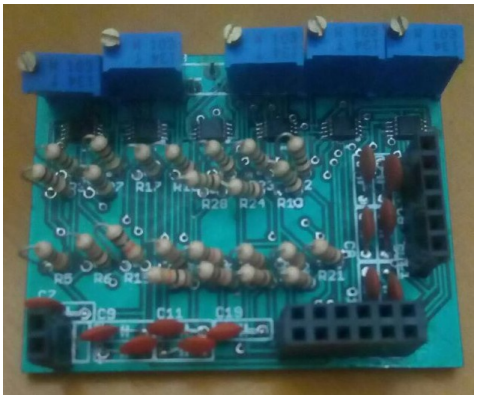


Source: Authors

and attended well the purpose of the project, as showed on Figure 13 which shows the resultant signal when one string is played.

After perform all the functionality tests it was attached the amplifier circuit to the

Figure 12 – PCB Board



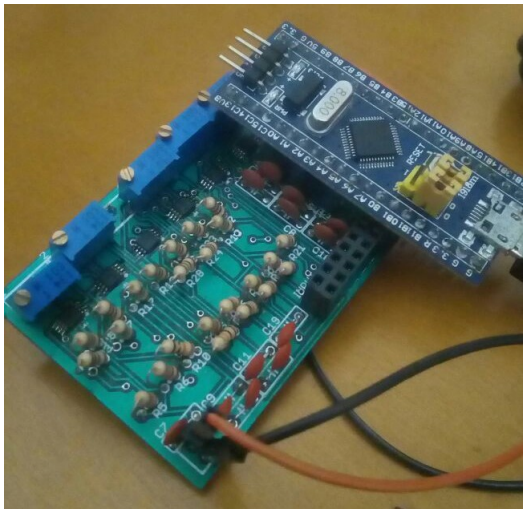
Source: Authors

Figure 13 – Amplified pickup signal with INA circuit

Source: Authors

microprocessor circuit, as showed at Figure 14.

Figure 14 – Captation system



Source: Authors

3 Firmware

3.1 Specifications

The firmware is basically an analog sampler, all it has to do is sample six analog channels, add a header (to identify the beginning and check continuity) and send it through USB. The following sections present the requirements of the system, details about the selection of the microcontroller and how the signal acquisition parameters (set by registers) were obtained.

3.1.1 Requirements

a) Super cheap

b) 6 analog channels (more precision is better)

c) High sample rate (at least 10 kHz for each channel, but ideally 44 kHz or more)

d) Fast USB support, to send the data with headers

Based on this requirements the minimum transfer speed can be calculated. Considering that a header will be set for every 252 samples (42 for each channel) and it has 4 bytes (3 of identification - to assure it is the header and not some data - and a counter). Previewing the worst case, each sample is 2 bytes long. The transfer rate given by Equation 3.1.

$$transfer\ rate = \left(\frac{channels * \frac{bytes}{sample} * \frac{samples}{package} + header\ size}{\frac{samples}{package}} \right) * f_s[B/s] \tag{3.1}$$

Considering that f_s has to be somewhere between 10 kHz and 50 kHz, the transfer rate numerical result is given by Equation 3.2.

$$transfer\ rate = \left(\frac{6 * 2 * 252 + 4}{252} \right) * f_s = 12.0159 * f_s = 120.159\ to\ 600.794[kB/s] \tag{3.2}$$

USB transfer speed is usually referred in Mbps, which gives a range between 961.27 and 4806.34 Mbps. This is too high for serial communication (typical max of 1 Mbps) so we need to add a requirement for raw USB support, which allows *bulk transfers* that can a have transfer rate up to 12Mbps (USB full-speed standard).

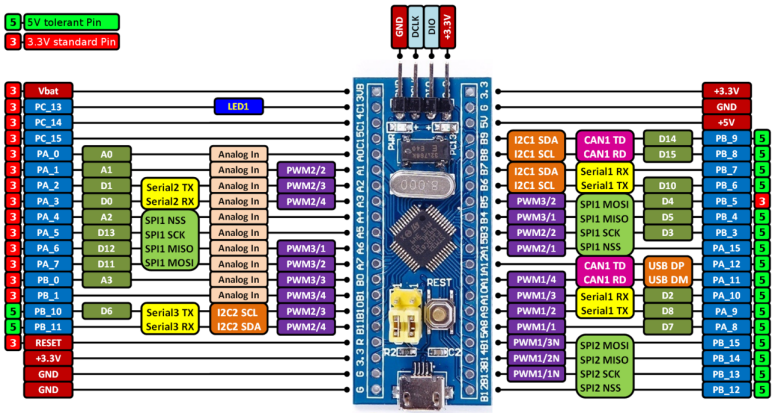
It's still needed to choose or exact sample rate (f_s), but it is first necessary select which hardware will be used.

3.1.2 Microcontroller Selection

There are too many microcontrollers that fit our requirements, but the most popular and cheap one is clearly the ARM from ST called STM32F103C8T6 (STM32F103XX..., 2015), and that is why it was selected.

It has eight 12 bits ADC inputs (with 2 parallel channels), DMA for the ADCs and USB full-speed support. It is also relatively fast (72 MHz clock, 32 bits architecture). All this for under US$ 2 in a developing board from China (the actual μC is under 0.2 U$). The board's pinout can be seen in Figure 15.

Figure 15 – STM32F103C8T6 Board



Source: Hudak (n.d.)

3.1.3 Sample Frequency

Usage of the ADCs for the selected μC can be optimized by using continuous sampling mode in conjunction with DMA (STM32F103XX..., 2015, ch. 11). In this mode, the sample frequency is controlled by a register that sets the *convolution time*. The convolution time is the amount of time the ADC takes to sample. The longer it is (more ADC clock cycles) the better the resulting precision.

The first variable chosen for this setup is the ADC clock, which is set by dividing the μC clock by 2, 4, 6 or 8. The ADC also has a maximum clock of 14 MHz. Taking in account the μC clock of 72 MHz, the highest possible value for the ADC clock is 12 MHz, which is set by a divider value of 6.

The last value to be chosen is the mentioned convolution time (T_c is calculated as the selected value plus 12.5 ADC clock cycles), which gives a sample frequency calculate by Equation 3.3.

$$f_s = \frac{ADC\ clock}{T_c + 12.5} * \frac{Parallel ADCs}{channels} = \frac{12}{T_c + 12.5} * \frac{2}{6}\ [MHz] \tag{3.3}$$

Table 3 shows the calculated results for each possible register value (using Equation 3.3). Based on it the chosen sampling frequency is 47.619 kHz. By using Equation 3.2 we can also calculate the actual data transfer rate (for all channels, including header), resulting in a total of 572,185 kHz. This last value will be used to test the USB communication.

Table 3 – ADC Sampling Frequencies

Register Value	**Convolution Time [cycles]**	**Sampling Frequency [kHz]**
000	1.5	285.71
001	7.5	200
010	13.5	153.85
011	28.5	97.56
100	41.5	74.07
101	55.5	58.82
110	**71.5**	**47.62**
111	239.5	15.87

Source: authors

3.2 Implementation

3.2.1 First Attempt

We first tried to build the firmware from scratch, using the tools given by the manufacturer, essentially a set of driver abstractions (HAL drivers). The problem found is that these abstractions are too slow, and don't work when the firmware uses the hardware close to it's limits (as we do for both transfer and sampling rates).

3.2.2 Second Attempt

In the research it was found a high quality open source project called MiniScope (MINISCOPE, n.d.), in which a few options of low budget DIY digital scope (using different μCs) are presented. One of the μCs used by MiniScope is the one selected, so for the implementation it was taken it's firmware as a base project. In this project the author claims to sample and transfer two channels at 461 kHz (but 8 bits only), which is very close to our needs (it is needed a little more transfer but much less sampling speed).

3.2.3 IDE

As it was taken MiniScope as a base project it will be used the same IDE as it, named CooCox (COOCOX, n.d.). It has a full set of tools, and it's completely free (no limitations).

3.2.4 Modifications

The base project, MiniScope, samples 2 channels at a different speed, bit rate and does not add any headers to the data. It also has some code to answer a few commands. It was as simple as setting up the registers for 6 channels, changing the sample size, placing the already chosen speed (subsection 3.1.3) and removing any unused code.

The act of changing the sample size was not done by registers. MiniScope was already sampling with 12 bits, but it was ignoring the least significant ones when filling the USB buffer. What was done is to change the bits alignment and putting all the data received from the ADC to the USB buffer.

3.2.5 Testing

A first attempt using the OS (Windows at that time, later Ubuntu) default driver was made. That did not work well, as it is too generic and thus slow.

At a second try, a simple libusb (LIBUSB, n.d.) program was built to test the transfer rate (calculated in subsection 3.1.3). The received data rate matched almost perfectly the calculated one.

3.2.5.1 Repository

All code is available at GitHub (TREVISAN, 2017a).

4 Software

4.1 Tools Selection

4.1.1 Top Level Requirements

The exact implementation of each of the items will be discussed later, but a brief description of the requirements is presented bellow:

a) Desktop GUI

b) Efficient signal processing (for pitch detection)

c) Real time graph visualization of the signals (oscilloscope like)

d) Access to `libusb` (LIBUSB, n.d.) API

e) Access to MIDI API

4.1.2 Language Choice

4.1.2.1 Java

The first choice was Java, as it meets all requirements. Desktop GUI can be done using Swing. A good library for pitch detection is available (called TarsosDSP (SIX; CORNELIS; LEMAN, 2014)). There is also a binding to `libusb` called `usb4java` and native MIDI support.

Following that idea, a functional prototype was built, but a few problems came to rise. The first is that `usb4java's` high-level API had bugs and was not working correctly. The solution was to fall back to the lower level API, but that made things much more complicated as threading and synchronization problems had to be dealt with. There was no good library for real time visualization either, which made really hard to both debug and tune the frequency detection algorithm. On top of that, Swing is at least non-pleasant compared to more modern UI programming, so a second approach came to be.

4.1.2.2 JavaScript

In alignment with both current work experience and world programming tendencies JavaScript was taken as a choice. It was seen that requirements fit much better now, for the following reasons.

For the desktop GUI, JavaScript has a few nice and mature Desktop GUI frameworks, like Electron and NW.js.

JavaScript is an interpreted language, therefor that has a low efficiency when compared to C++ or Java. That is huge problem, but there is an well known overcome. As this project tries to build a desktop application, Node.js will be used ultimately, and it has support for C++ bindings. That means the JavaScript code can call a compiled C++ library to calculate the pitch, thus solving the problem.

Graph visualization should not be a problem either as there are a lot of libraries for that. The most problematic requirement in Java was *libusb* support. It is available in JavaScript using *node-usb* (NODE-USB, n.d.), and a few simple tests returned good results with a much simpler API. MIDI was also tested and worked just fine.

4.1.3 Desktop Framework

Now that JavaScript is set as the final selection we need an environment to run it. There are two already listed really mature and popular choices: Electron and NW.js.

At first Electron was used to build a test application, because it is the most popular of the two (in fact even the editor used to write this words is built with it), but the pitch detection call was running slowly. As a mater of fact it was running much faster using pure JS code rather than the C++ library. A deeper research was needed, and the way Electron worked was getting in the project's way, but first it's necessary to know what Node.js is.

4.1.3.1 Interpreter

JavaScript is a interpreted language and thus needs an interpreter. The most common one is Google V8, which happens to be the same one used in most web browsers as well as in Node. The difference between browsers and Node is simply the API that comes with them. Web needs firstly to access the UI (html) and ways to modify it, it also needs secure and limited access to hardware and internet calls. Of course that means web JavaScript code cannot use C++ libraries directly.

On the other hand, Node is a more pure version of V8, it also gives the possibility to write and call C++ code (feature needed for this project), which ultimately makes it as capable as any desktop program can be. Node also comes with a hand-full set of native resources (like file system and full communication access), but it does **not** provide any kind of GUI. Knowing that, it is possible to have a better understanding on how the two desktop environments (Electron and NW.js) work.

4.1.3.2 Electron

Electron by design has at least two processes running (HOW..., 2017), one for the "web" and other for Node access. The answer for how the web process access native

resources is also the reason why the execution of the C++ processing library was slow: it uses inter-process communication (IPC). IPC makes things a lot slower, which ultimately makes impossible to use Electron in this project.

4.1.3.3 Node Webkit

Differently from Electron, NW.js (NODE..., n.d.; HOW..., 2017) takes the Node environment and combines it with Chromium into a single process, removing the use of IPC. Initial tests reported that the pitch detection library has fast execution as expected.

4.1.4 Architectural Tools

NW.js will go only as far as to give access to both Node and DOM API's. But that is too crude, and not what it was wanted by given up on Java Swing. Again based on current work experience and world tendencies the setup chosen is React + Redux.

React (REACT, n.d.) is a library created by Facebook and world wide used for UI applications. It uses a declarative component-based system that makes it easy to build scalable and reusable code.

React only go as far to help building the UI, but it is also needed to pass the state of the application to the UI components, and this is where Redux (REDUX, n.d.) comes in. It keeps all the application state stored in a single place, described by transition functions. That makes the storage system easy to be tested and used, because all actions (that modify the state) must be well defined and it doesn't rely on the UI (React), making it easy to test.

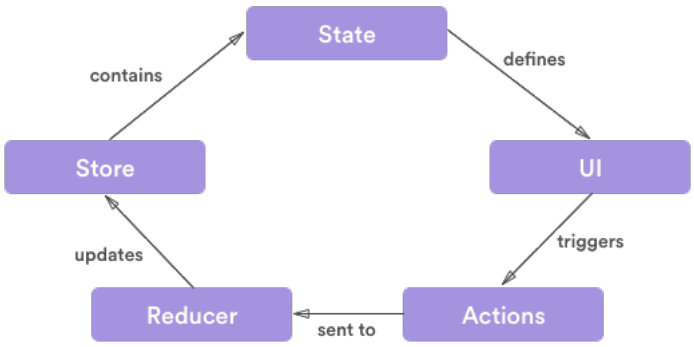
Figure 16 shows the flow of an application that uses React + Redux. It is obvious to see the simplicity it has, a single path must be followed. This simplicity is what makes it much easier to use against other frameworks like Java Swing.

There is still the choice of the visual library to use, and the chosen one is Semantic UI React (SEMANTIC..., n.d.). It has some nice and robust React components to build a well designed application.

4.1.5 Fast Signal Processing

Pitch detection is a challenging task, and good implementations are time consuming, since it is needed efficiency to run it in real-time. The library already said to be used didn't actually existed, the only one available was a pure JavaScript library (PITCHFINDER, n.d.) which is not suitable for this project. The solution was to build our own library based on both the pure JavaScript one and TarsosDSP (SIX; CORNELIS; LEMAN, 2014). Implementation details discussed further on section 4.2.

Figure 16 – React Redux Flow Diagram



Source: Getting... (2016)

4.1.6 Real Time Visualization

There are lots of charting libraries available for use with web interfaces (and by extension NW.js), unfortunately none was good fit for real time high density signals such as audio. The solution was again to build one, since all other things are looking to run smoothly in JavaScript. Implementation details on section 4.3.

4.2 Pitch Detection

Pitch detection can be defined as frequency detection with the addition of note quantization. Table 4 shows the base frequency for each of the 12 existent notes. Multiples of the same frequency are seen as the same note on a different range, known as octave. Even though the quantization makes things simpler it's still a hard task, even more for

Table 4 – Notes Frequencies

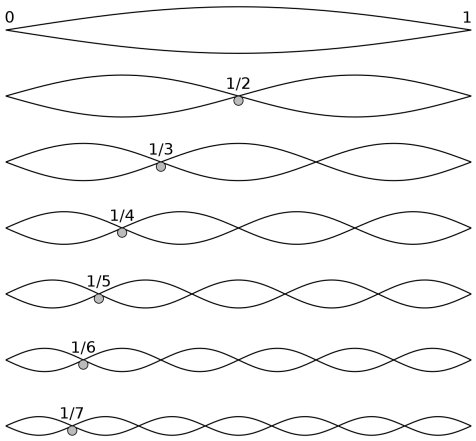
Note Name	Frequency
A	440.00
A#	466.16
B	493.88
C	523.25
C#	554.37
D	587.33
D#	622.25
E	659.25
F	698.46
F#	739.99
G	783.99
G#	830.61

Source: authors

instruments where there is the presence of harmonic series. Harmonic series notes are

multiples of the fundamental frequency (most important note) produced by integer sections of the instrument vibration. Figure 17 shows an visual representation of why they exist. The existence of them as well as the presence of both inter-signal and white noise makes necessary the use of non-trivial algorithms for pitch detection, and two of them will be discussed next.

Figure 17 – Harmonic Series



Source: Wikipedia (2017)

4.2.1 YIN algorithm

Difference function is a well know time domain technique to calculate a signal's fundamental frequency, however it's sole usage does not deliver satisfactory accuracy for this project. YIN (CHEVEIGNÉ; KAWAHARA, 2002) is a method that provide improvements to the difference function method, achieving a much higher precision. It can also be implemented with logarithmic growth as the difference function can be calculated using the FFT and IFFT algorithms. The algorithm can be divided in 5 steps, 4 which are implemented, as follows:

1. Difference Function
2. Cumulative mean normalized difference
3. Absolute threshold
4. Parabolic interpolation
5. Best local estimate (not implemented yet)

It's important to notice that the absolute threshold is a controlled attempt to regulate the error introduced by the harmonic series (as in Figure 17), thus it gives preference to lower frequencies (below the threshold).

4.2.2 MacLeod algorithm

MacLeod (MCLEOD; WYVILL, 2005) goes for a similar approach, using the square difference function. More precisely it uses a special normalized version of it. The best result is then calculated choosing the best peak. This is done by means of using a parabolic interpolation of the highest peak and its two neighbors, this process also gives a threshold constant that limits the detection of the neighbors, and thus the possibility of some tuning. It can be divided in two steps:

1. Normalized Square Difference Function
2. Peak Picking (Using parabolic interpolation of neighbors)

The normalized square difference function can also be calculated using FFT and IFFT for logarithmic growth.

4.2.3 Implementation

Implementation for both algorithms follow the same pattern, taking the pretty Java code of TarsosDSP (SIX; CORNELIS; LEMAN, 2014) and replacing the syntax and data structures with C++ ones (using standard library for containers). There is also a JavaScript bridge for data type conversion, so we can use the library calls with simple arrays of numbers.

The implementation is not using FFT for logarithm growth yet (but quadratic growth instead), following the TarsosDSP library. The faster implementation is kept as a goal for future improvement. All code is available as an open source project at Github and npm (NODE-PITCHFINDER, n.d.).

4.3 Data Visualization

The objective of data visualization is to live debug and tune both hardware gain and algorithms control constants. The ideal case is to have both real time chart as well as a buffered/triggered one, essentially something like an oscilloscope. That has to be performant for real time audio signals, sampled at more than 47 kHz.

There are a great amount of JavaScript DOM libraries for charting, but most of them are too automatic or have too much details, making them too slow for the project

need. The solution was to build our own library for that, again available as an open source project at both GitHub and npm (REACT-PLOTTER, n.d.).

4.3.1 Requirements

a) Be a React Component

b) Automatic calculations for array input

c) Option for triggering

d) Option for buffering

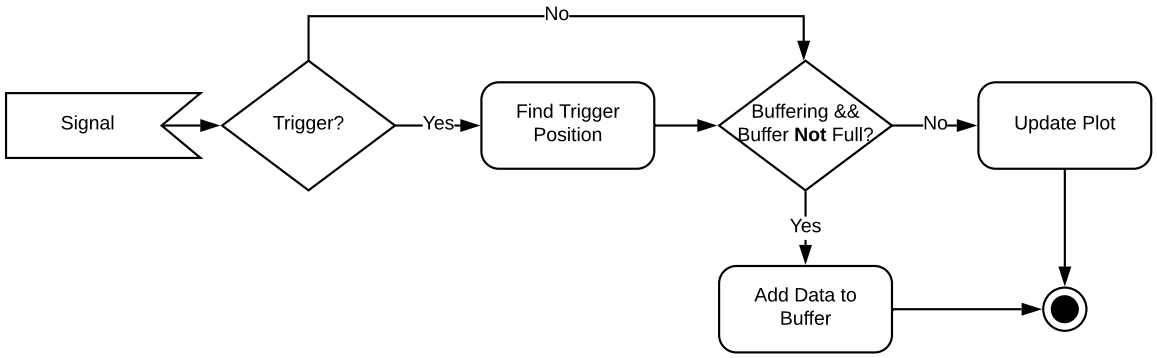
e) Minimum Redraw

f) Fixed Height/Width

4.3.2 Algorithm

Triggering and buffering are achieved by using a filter that only calls the plotting function when the options are met. This filter is simply the function called to add data, and it is represented by Figure 18.

Figure 18 – Add Function Diagram



Source: authors

For the actual drawing a triple buffer technique is used, one for holding the last state (called plotBuffer), one for drawing (called drawingBuffer) and finally the one actually rendered (called canvas). That last one is needed so the cartesian axis don't get saved on the drawing scene.

For linear plot time a translation is established, in a way that only the new points will be drawn, the past ones are only translated to the left. The steps of the algorithm are as follow:

1. Clear drawingBuffer

2. Copy plotBuffer to drawingBuffer translating (removing) extra data
3. Draw new data on drawingBuffer
4. Copy drawingBuffer to plotBuffer
5. Copy plotBuffer to canvas
6. Draw axis on canvas

4.3.3 Implementation

Using the listed requirements (subsection 4.3.1) a minimum API was built as a React component. Being such, all it gives is a set of properties, for which the chart is drawn (when needed). They are listed in Table 5. The style property is a function that is

Table 5 – React Plotter Props

Property	**Type**	**Description**
style	Function	Style function (called to print the data)
[trigger]	number	Use trigger
[onlyFull=true]	bool	When using trigger it tells if the view should wait for a complete data set before updating
[width=300]	number	
[height=150]	number	
[initialData=[ ]]	number[ ]	
[appendData=[ ]]	number[ ]	
[dataSize=100]	number	
[pixelSkip=1]	number	Pixels between points
[max=100]	number	Maximum Y Value
[min=-100]	number	Minimum Y Value
[useMean=true]	bool	Use mean calculation, otherwise median

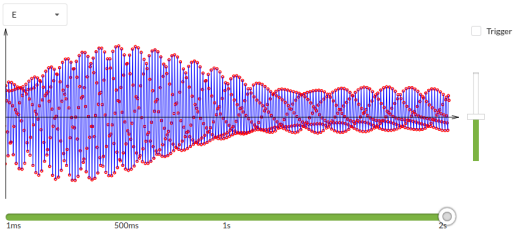
Source: react-plotter (n.d.)

called to render each point. Two styles were built, a line plot (points are connected by a straight line) and a digital plot (digital signal standard chart, not used in the final version of this project).

4.3.4 Results

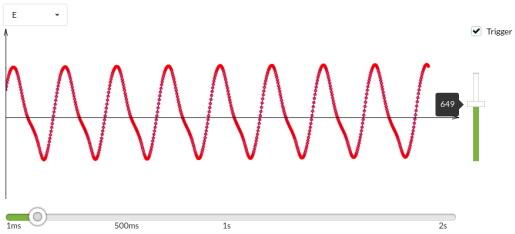
The results are more than satisfactory, tested to be able to run multiple plots of audio speed signals at the same time without much effort. Figure 19 and Figure 20 show how the visualization looks on the project, but full details and working examples are also available at GitHub (REACT-PLOTTER, n.d.).

Figure 19 – Real Time Plot



Source: authors

Figure 20 – Triggered Plot



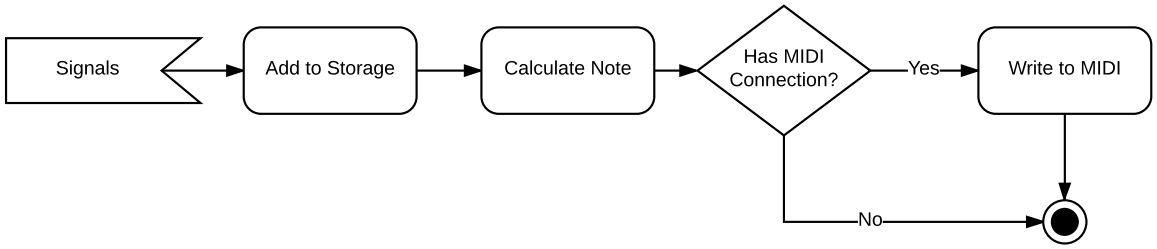
Source: authors

4.4 Main Program

The general idea is to build an UI with three main divisions: Home (with device selection), midi selection and signal-to-MIDI connections, Plot (with the signal visualization) and Options (with the algorithm tuning options, as well as virtual MIDI creation).

When a device is selected it's signals will go through a simple process, as in Figure 21.

Figure 21 – Signals Flow Diagram



Source: authors

4.4.1 GUI

4.4.1.1 Home

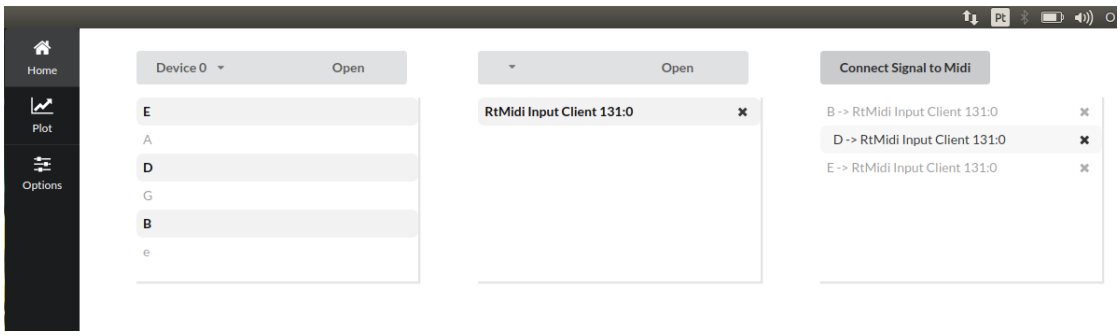
The home page has three horizontally divided sections: Device and signals, MIDI selection and list, and finally connections, as in Figure 22.

The first is used to open the device (there can be only one used at a time). When opened a list of signals will be displayed (six of them, named as each guitar note). Each signal can be selected so it can be connected to a MIDI device.

The second is to open any given number of existing MIDI devices, which will be listed bellow (can be also closed). The listed devices can also be selected, but only one at a time.

The final section is used to establish the connections, given the selected signals and MIDI, the connections are showed in the box bellow and can be deleted.

Figure 22 – Home Page

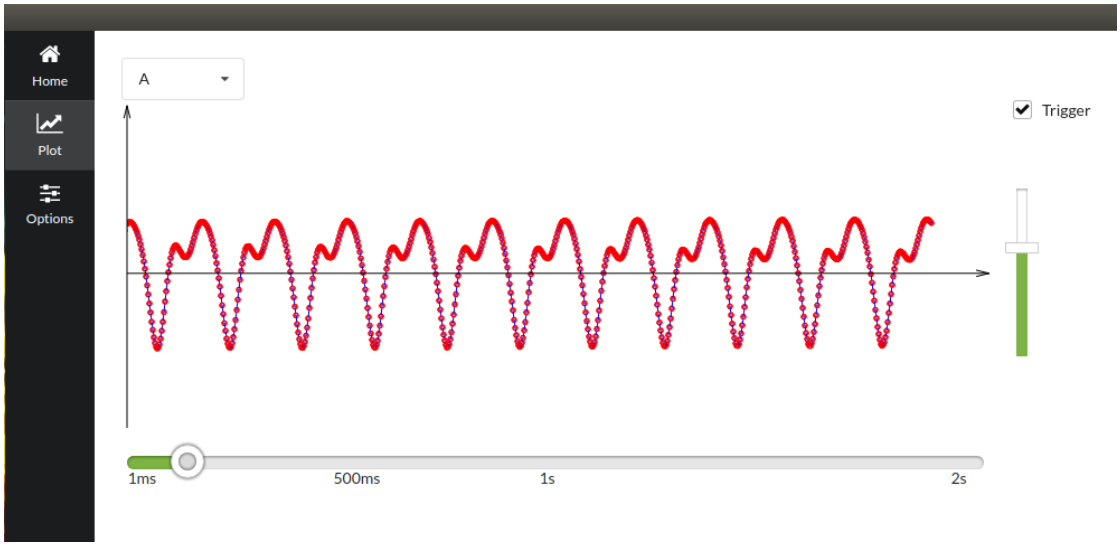


Source: authors

4.4.1.2 Plot

The plot page is a react-plotter (REACT-PLOTTER, n.d.) component with a few visual controls, being: a dropdown to select which signal is being displayed, a checkbox to enable trigger, a slider to control the time range and another slider to control the trigger value. The full page is as in Figure 23. It was chosen to show only one plot at a time so it's size is bigger, easier to see. This also makes the program a little more performant.

Figure 23 – Plot Page

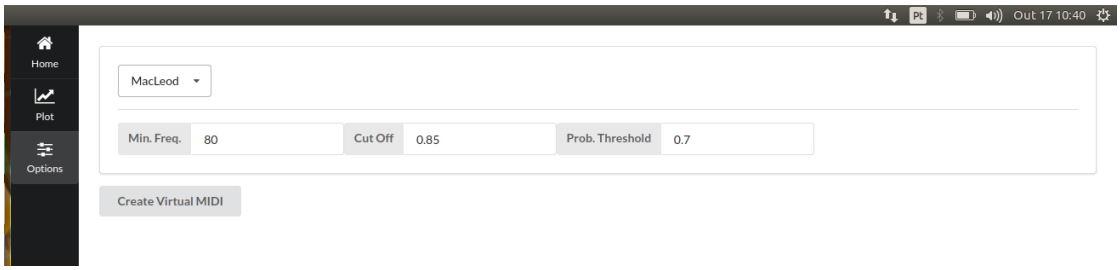


Source: authors

4.4.1.3 Options Page

The options page is used to control the current algorithm (YIN or MacLeod, section 4.2) and it's parameters. For both Linux and macOS it is also possible to create virtual MIDI devices, to which our program can write and a synthesizer can read. For Windows this is still possible, but using a hacky solution (since Windows does not provide any official API for this) - the easiest option being LoopBe1 (LOOPBE1, n.d.). The page can be seen at Figure 24.

Figure 24 – Options Page showing MacLeod configurations



Source: authors

4.4.2 Implementation

4.4.2.1 PC Operational System Resources

The PC operational system resources list is as follows:

a) USB device: only one can be connected

b) MIDI devices: multiple of them can be used at any time

Since they are all global, they can be represented as static classes. But JavaScript has good functional programming capabilities, which are very suitable for global resources. JavaScript imported modules are also scoped by default, meaning that they work like a C++ namespace, keeping our static resources separated in a nice way. Taking into account these mentions, two modules were built for resource easy access, being MIDI and USB.

4.4.2.2 Functional Programming

Functional programming is a paradigm that focus on software functionality over modeling. The state management library already chosen (Redux) is functional, which makes it very logical to choose this paradigm.

So far this document has not said a word about how to model the following software structures using OOP: resources, state or functionality. And it won't ever, because it

does not have any classes, except for the UI, which uses the class syntax to declare components. However, they don't fall into the OOP paradigm, but into a specific UI component paradigm instead.

As the *Redux store* keeps the all usable state (Table 6), it is also used as a trigger for all given functionality. This means that any functionality that needs to be implemented will, directly or indirectly, listen and/or write to the Redux store.

4.4.2.3 Entry Point

The entry point for this program is both a declarator and connector. The entry point allocates all needed structures (or calls the module that does it), the most significant one being the Redux store (REDUX, n.d.) - store being a short for storage, which is where all of the application visible state is held.

The entry point also connects all callbacks and logic in a declarative way. In this single file all of the program's internal functionalities are declared, so much that if you read it you should also understand the entire program.

4.4.2.4 Reducers

Redux stored data is not defined by a set a properties, like typical OOP applications, instead it uses the functional programming paradigm. The storage is defined by transfer functions, each of them describing actions that can modify the current state by returning a new one. Each of these functions, called reducer, receive two parameters - the current state and the action to be processed - and should return the new state for the action (or the current one if there are no changes).

Our program has nine reducers, but five of them share the same transfer function, as in Table 6.

Table 6 – Reducers

Name	**Data Type**	**Actions**	**Used for**
device	string	set, remove	current selected device
devices	string[ ]	add, remove	list of devices
signals	string[ ]	set, clear	list of signals
signalsData	object: { name: number[ ] }	set, clear	list of signals data (name and values)
object	object	set, clear, remove	Plot Page options General options MIDI devices Signal to MIDI connections

Source: authors

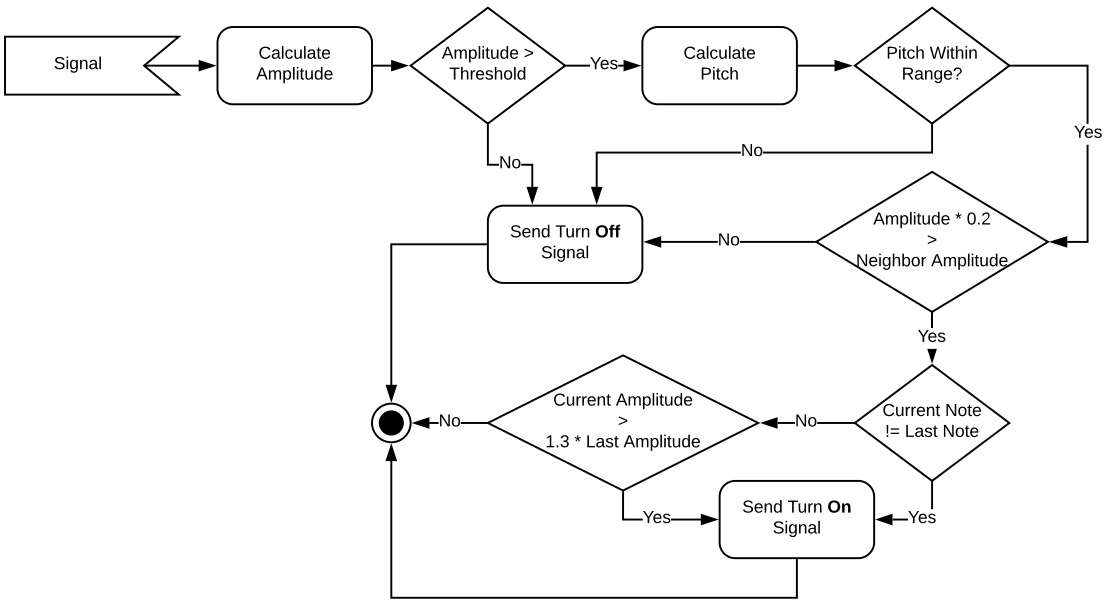
4.4.2.5 Note Calculation

The flow chart in Figure 25 presents how the note calculation process work. The amplitude calculation is an absolute average removing the mean (Equation 4.1), as in Equation 4.2. The *pitch within range* is a function that limits each string frequency to being close to it's known possible values, as in Table 7.

$$Mean = \frac{\sum_{i=0}^{N} |x_i|}{N} \tag{4.1}$$

$$Amplitude = \frac{\sum_{i=0}^{N} |x_i - Mean|}{N} \tag{4.2}$$

Figure 25 – Note Detection Diagram



Source: authors

Table 7 – Pitch Range

String	**Min. Freq.[Hz]**	**Max. Freq.[Hz]**
E	70	265
A	95	350
D	130	470
G	170	625
B	215	785
e	290	1050

Source: authors

4.4.2.6 PC Software Testing

The tests implemented at this point do not regard the acquired signals from musical instrument, but focus on simulation of the functionalities of each software module. The tests fall into one of two categories: unity or timing. Unity tests are made for the signal processing relating modules and also for every reducer, these are all automated and sum up to a total of 32 tests over 9 modules, listed at Appendix D, it order to give an idea of how they work. The timing tests were used to check if each separated functionality that may cause processing issues can run in real-time. There are timing tests for: signal average value calculation, signal window buffering, raw data conversion, pitch detection and USB polling.

4.4.2.7 Repository

Again, all code is available at GitHub (TREVISAN, 2017b).

5 Experiments and Discussions

For the quantification of how well our system works we need some measurements. Unfortunately one of our input channels is not functioning (corresponding to the D string), hence it will be ignored. To make it simple our measurements will be based on mean and standard deviation. A few measurements were proposed, as listed:

- Single Note Accuracy, Figure 26: by hitting each string at a time on only a single note, measurements were taken. The standard deviation line is based on the frequency instead of the note, showing how the detected frequency fluctuates around the note.
- A minor Chord Error, Figure 27: by hitting all the strings that compose the Am chord and comparing the readings with the expected result an error rate (and standard deviation) is calculated based on the note (considering cents).
- E major Chord Error, Figure 28: same as above, but for the E major chord.
- Chromatic Scale Error, Figure 29: For each string, a chromatic scale was plucked and with the measurements an error rate was calculated. An error is considered when two neighbor notes are not integer consecutive values.
- Pluck Counting, Table 8: by quickly (4 notes/s approximately) hitting a string 20 times, the actual number of detected notes are counted.

Table 8 – Pluck Hits Counting

Correct Value	MacLeod	YIN
20	42	28

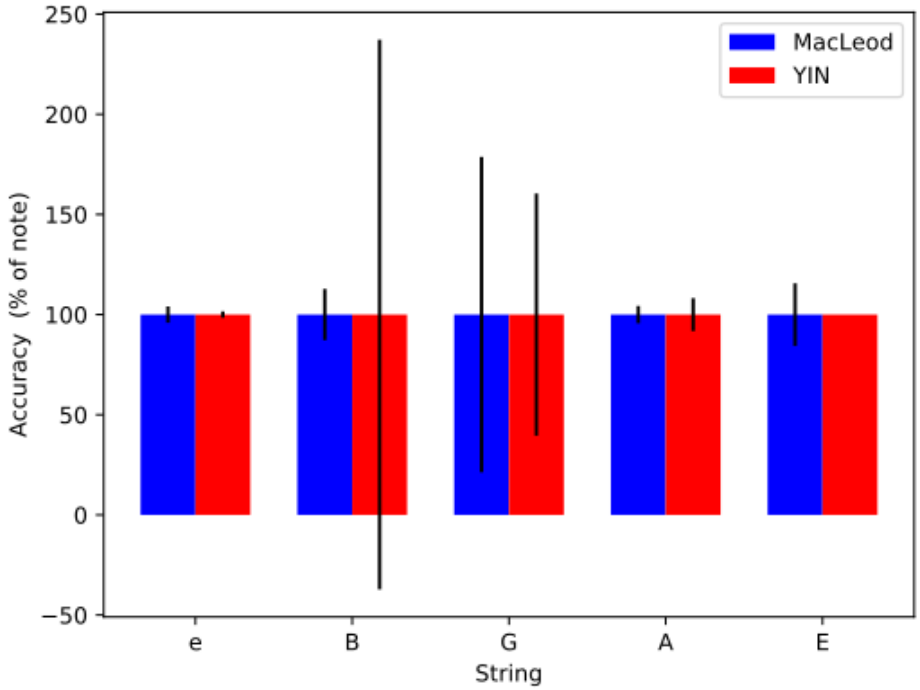
Source: authors

5.1 Discussions

Strings Single Note Accuracy test gave satisfactory results (Figure 26). B string has a high standard deviation because it's signal has DC noise, due to resistor imprecisions. This limited the channel gain which ultimately made this channel results lower than the others. The same analysis is true for all other experiments.

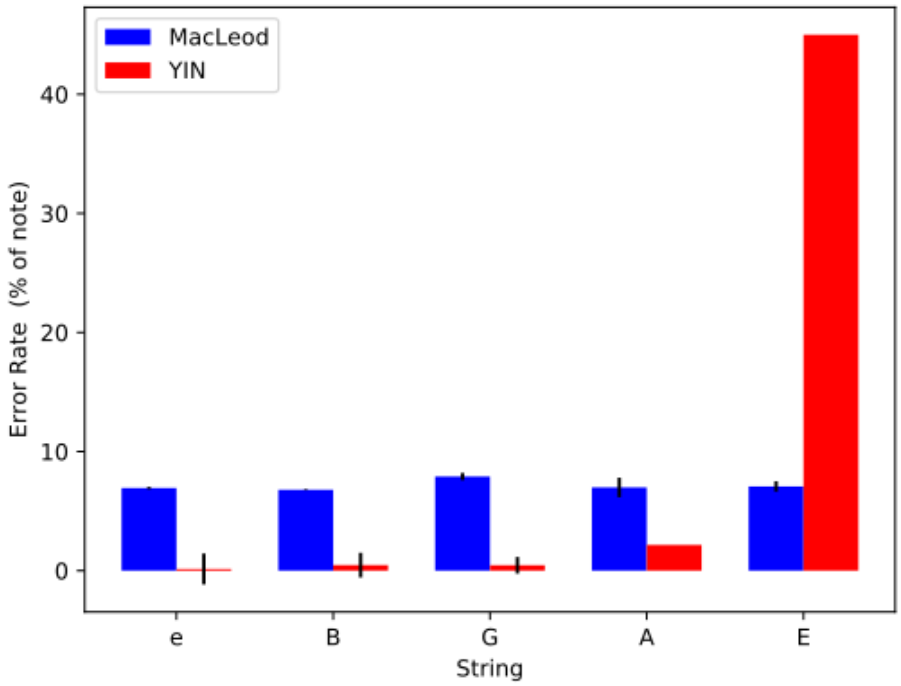
Both the chord detection experiments (Figure 27, Figure 28) gave reasonable results, but they show our system still has lots of room for improvement. The exception is for the E string error on the Am chord using the YIN algorithm. The reason for this is that

Figure 26 – Strings Single Note Accuracy



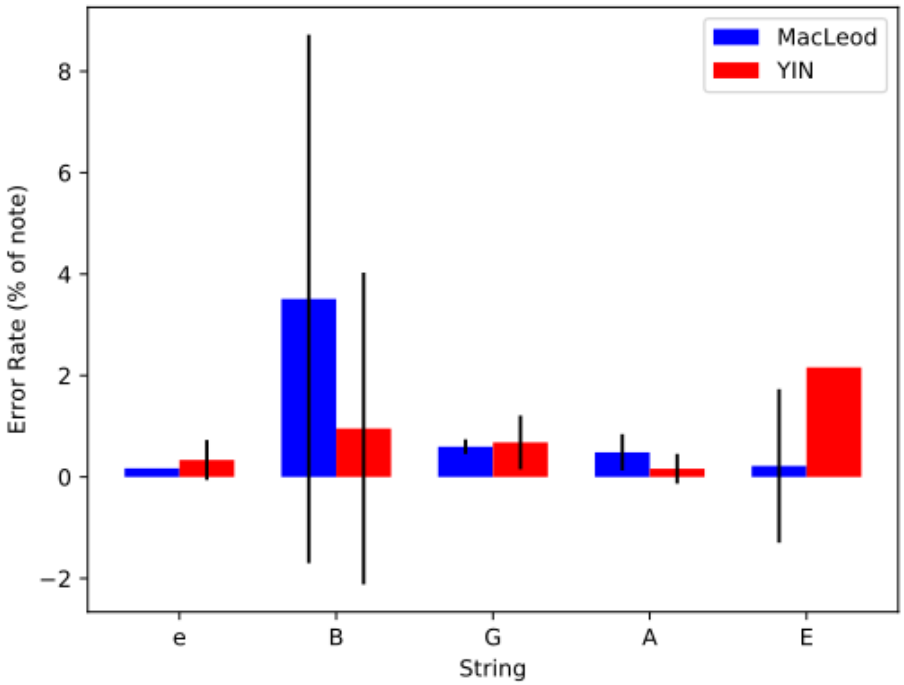
Source: Authors

Figure 27 – Am Chord Error



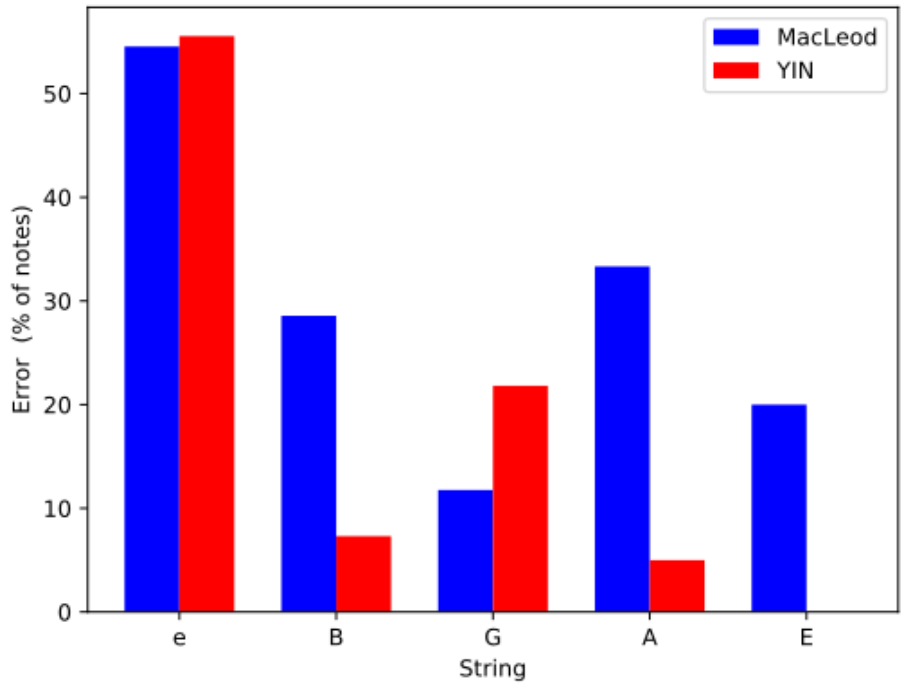
Source: Authors

Figure 28 – E Chord Error



Source: Authors

Figure 29 – Chromatic Scale Error



Source: Authors

YIN is not working well for low frequency notes, because it still needs a greater period (samples) for analysis, which is not currently possible due to processing time limitations - but a solution is proposed at the next chapter.

The chromatic scale results (Figure 29) show that our system is not working well along time. By using a small period of measurement (so it can run at real-time) the range where a note is changing to the next one is misread as an incorrect result. This is a big problem for music annotation, but has the same solution as above, which needs performance improvement.

The last test (Table 8) also shows our time related issues, that, although being natural, still need to be dealt with, for the same reason above, transitions through time that need a larger sample period to work well. The reason for the plucking issue is indeed not given by the amplitude detection, but for frequency variations.

6 Conclusions

The hardware project showed be reliable and robust. The amplification worked well for the musical instrument signal in order to be further acquired by the microcontroller ADC, digitizing the signal as expected. The project proved to be challenging – completely assembling a hexaphonic pickup and the amplifier system, but it has performed successfully. The resulting firmware may be considered final, as it meets all of our current requirements perfectly.

Real time note detection proved not to be so accurate. Legato (connected) notes may cause a middle note detection and there are miss-detections (mostly at frequency transitions). It works well enough for live play of MIDI instruments, but not for music notation. It has a lot of room for improvement, and the means to achieve it are discussed next.

Even if not having robustness to be a commercial product this project is still a big source of learning and a proof of concept. To build it, knowledge about almost every area of electronic engineering was needed at some level. Lots of work still need to be put on it, but it does have potential to evolve into a useful solution for guitar digitization.

As for engineering effort, it required knowledge from passive hardware, to active hardware (amplifiers), low level software (firmware), digital communication (USB) and up to high level software (that can be further divided to even more areas). Summarizing, it was a great way of learning and proofing our proeficiency in electronic engineering.

The area of musical signal processing proved to be as complex as it popularly said to be. Still with a fair amount of effort made it is possible to use it in a real world application, even for non-specialists in signal processing, like ourselves.

Finally our project was fun to build and even more fun to use and play with, and achieved the desired result of showing that it is possible to make guitar digitization cheap and efficient, even if not yet to that point.

6.1 Future Work

6.1.1 Hardware

For the future, thinking about production ready devices, it is needed to test INA326 ICs from other suppliers. Using chinese versions of the same IC, it might be possible to drastically reduce costs without changing the main amplifier circuit. Also to change the through-hole components to the surface mount (SMD), for a cheaper and higher quality

product. This would also lower the board dimensions, making the product more compact. Another option is to test other amplifier circuits, that are cheaper and still meet our requirements.

A crucial change would also be to change from trimmers potentiometers to programmable resistors, making the gain adjustment automatic. It is also ideal to perform the magnet wrapping automatically. This last improvement will increase component quality, passive signal output and also make it possible to add more turns by using a thinner wire, making a final product of much higher quality.

6.1.2 Software

- **Note Detection**: For real world music notation a different solution is needed: record the signals and post-process them. This way it is possible to use a more detailed analysis of each signal and thus get very good results, as there is much more computing power available when not being limited by real-time processing.

- **Performance**: Though real time analysis work, a few limitations were detected. Buffering can only process each data point one time (can't use a sliding buffer that recycles data). This means that we are close to the processing limits, due to two causes: single core processing and slow algorithms.

  One step of the solution is to use multi-core processing for the pitch detection, which can be done using Node.js support for it.

  As already stated (subsection 4.2.3) the current implementation uses quadratic growth algorithms, while it can be improved to logarithmic growth. Fixing this will improve the performance to a point where multi-core processing won't be even necessary.

- **Algorithm Improvements**: A few algorithm modifications may also improve the project's results. Most noticeable would be the above mentioned buffering to process each input multiple times, with a sliding window - which needs performance improvements first.

  Another change necessary would be to remove co-channel interference in a more fashioned way other than amplitude check, making it easier for the pitch detection algorithm to work.

- **New Features**: A main feature not yet implemented is also necessary: vibrato detection. Vibrato is the note's small variation of frequency without changing to another note. This is widely used by musicians and needs to be implemented, but has not been considered yet.

As already mentioned recording will be a breaking change to the system. The ability to have more processing power due to not run in real time can greatly improve the system accuracy. It also makes possible to look ahead of time to check with more precision a single moment (as notes are frequencies time is an important matter).

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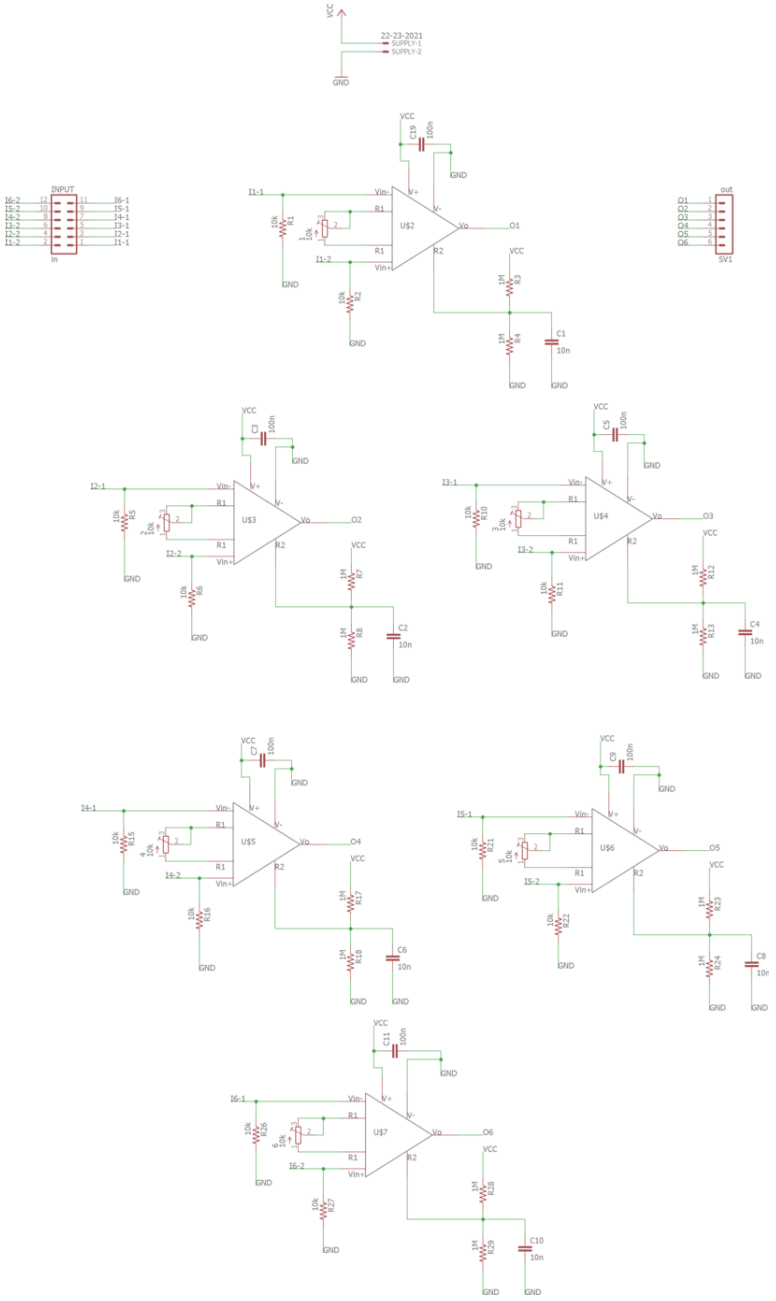
Annex

ANNEX A – INA326 complete Schematic.

Figure 30 – INA326 Complete Schematic



Source: authors

ANNEX B – Prototype Cost Table.

Costs are given on United States Dolar.

Table 9 – Prototype Board Cost Table

Component	**Quantity**	**Cost**
INA326	6	37.38
10kΩ Trimmer Potentiometer	6	7.50
10nF Ceramic Capacitor	6	2.34
100nF Electrolytic Capacitor	6	0.47
10kΩ Resistor 5% tolerance	12	0.62
1MΩ Resistor 5% tolerance	12	0.62
Pin bar 1x40	1	0.93
PCB	2	15.57
Pickup Base and Cover	1	4.68
Magnets	6	9.37
Copper wire	0.2969oz	0.18

Source: authors

Total production cost for the prototype - 79.66

ANNEX C – Production Cost Table.

Costs are given on United States Dollar for production of 1000 devices.

Table 10 – Production Board Prediction Cost Table

Component	**Quantity**	**Cost per Component**	**Total Cost**
INA326	6	2.53071	15.18
10kΩ SMD Trimmer Potentiometer	6	2.16050	12.96
10nF SMD Ceramic Capacitor	6	0.00233	0.01
100nF Electrolytic Capacitor	6	0.04431	0.27
10kΩ SMD Resistor 1% tolerance	12	0.00649	0.08
1MΩ SMD Resistor 1% tolerance	12	0.00671	0.08
Pin bar 1x40	1	0.62240	0.62
PCB	1	0.22	0.22
Pickup Base and Cover	1	4.68	4.68
Magnets	6	0.82	4.92
Copper wire	0.5938oz	2.01/oz	1.19

Source: Digikey (), PCB-shopper (), pickup... () and copper... ()

Total production cost considering 1000 boards - 40.21 per board

ANNEX D – Main Program Test List.

- Guitar Signal Processor
  - interpreter
    - should interpret correct data (152ms)
    - should throw error for wrong header (1ms)
    - should throw error for missing counter (1ms)
    - should throw error for missing data (1ms)
    - should start working again after a fail (5ms)
  - windowBuffer
    - should buffer until full and keep windowSize - windowDelta (29ms)
    - should throw error if full (1ms)
  - guitarWindowBuffer
    - it's just 6 window buffers (32ms)
  - processor
    - should detect correct note (22ms)
    - should not detect small amplitude note (8ms)
    - should detect amplitude changes (21ms)
- calculator
  - calculateAverageAmplitude
    - should calculate correct value (3ms)
    - should remove DC component (2ms)
- frequencyDetector
  - YIN
    - should resolve correct frequency (5ms)
  - MacLeod
    - should resolve correct frequency (6ms)
- helpers
  - repeat
    - should return correct data (2ms)

-
- object reducer
  - default value (3ms)
  - should set correctly (3ms)
  - should remove correctly
  - should clear correctly (1ms)
- signals reducer
  - default value (3ms)
  - should add correctly (3ms)
  - should remove correctly (1ms)
- devices reducer
  - default value (2ms)
  - should add correctly (2ms)
  - should remove correctly (2ms)
- device reducer
  - default value (4ms)
  - should set correctly (3ms)
  - should remove correctly
- signalsData reducer
  - default value (1ms)
  - should add correctly (2ms)
  - should remove correctly (1ms)