NOTGUITAR: A REAL-TIME TIMBRAL CONVERSION SYSTEM

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1. ABSTRACT

NotGuitar is a DSP system designed to perform real-time timbral conversion from electric guitar to saxophone In order to preserve the dynamic and expressive elements of the guitarist's playing as much as possible, notGuitar does not use MIDI or synthesis, but simply processes the input guitar signal. NotGuitar was implemented using a Texas Instruments DSK6713 DSP board in May 2018 at the University of Southern California.

2. INTRODUCTION

The specific sound of a particular instrument is known as the "timbre" of the instrument, and is made up primarily of two parts: frequency characteristics (harmonic structure) and an amplitude characteristics (attack, sustain, release) [1]. NotGuitar uses note onset and envelope detection combined with amplitude envelope modulation to alter the amplitude characteristics. To affect the frequency characteristics, notGuitar uses a Sliding Discrete Fourier Transform (SDFT) to perform rough pitch detection, then uses the rough pitch detection to inform an adaptive circular convolution filter. The full system architecture can be seen in Fig. 1.

3. FREQUENCY CHARACTERISTICS

The frequency component of an instrument's timbre is typically referred to as the instrument's "harmonic structure." Essentially, any note played on any instrument produces sound not just at the fundamental frequency of that note, but also at every multiple of that frequency, known as "overtones." The frequency characteristic of individual instruments exists because the relative amplitude of each harmonic is different for each instrument.

Our plan for manipulating the frequency content of the signal was to split our frequency spectrum linearly, with enough resolution to isolate individual overtones of any guitar note. We would then use guitar and saxophone sample recordings to train sets of multipliers that could act on the frequency bands to transform the harmonic structure of the guitar to that of the saxophone. Additionally, by detecting a rough frequency band that contained the fundamental, we could tune our multipliers to more accurately represent the harmonic structure relative to the fundamental frequency of the input signal.

[*] Thanks to Richard Leahy for guidance and support

3.1. Different Approaches

Our approach towards the frequency component of the guitar to saxophone timbral shift went through many iterations which will be outlined below:

3.1.1. Sub-banding Filter Bank

We initially decided to use a linear sub-banding filter bank to split the signal into linearly spaced frequency bands, detect which of the sub-bands contained the fundamental frequency, apply the corresponding set of multipliers, and then sum the signal back together for the output. We abandoned this approach due to the difficulty of summing the sub-bands back together. Since each sub-band would have it's own group delay and phase delay, summing the bands correctly would have been non-trivial.

3.1.2. Short Time Fourier Transform

Our next plan was to use a Short Time Fourier Transform (STFT) to split up the frequency spectrum. Since an N-window Fourier Transform essentially splits the positive frequencies into N linearly spaced frequency bins, we could use the STFT to find the fundamental frequency, then apply the multipliers, and use an inverse STFT to bring the signal back to the time domain. While this method was well suited for our purposes, we ended up abandoning the STFT after finding other methods with lower time complexity.

3.1.3. Sliding Discrete Fourier Transform

In the system described above, the STFT would be operating on nearly identical windows in consecutive time steps. The only differences would be one new sample entering the window, one old sample leaving the window, and all the other samples being shifted one sample over. Using this information, we next considered using a Sliding Discrete Fourier Transform [2]. While the STFT has a time complexity of $\mathcal{O}(n \log n)$ for both the forward and inverse transforms, the SDFT is $\mathcal{O}(n)$ for the forward transform. In the final system, the SDFT is used to calculate the Fourier Transform of the first four frequency bins, which then informs the system which set of frequency multipliers to use for the circular convolution.

3.1.4. Circular Convolution

Since the multiplication of two signals in the frequency domain corresponds to the convolution of those signals in the time domain, we ultimately decided to use circular convolution to implement the filters in our system. The SDFT needs N complex-by-complex multiplications for the forward transform, and $\frac{N}{2}$ real-by-complex

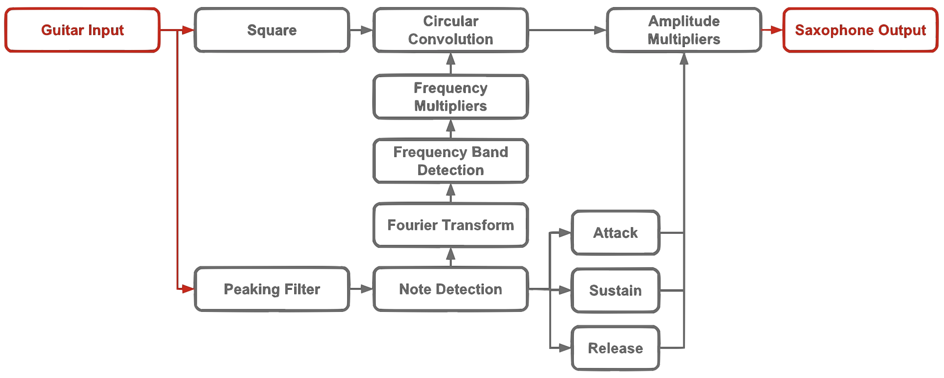


Figure 1: *notGuitar System Flowchart*

multiplications for the inverse transform. With an additional N real-by-complex multiplications for the frequency multipliers, the total time complexity of the system coumes out to $\mathcal{O}(7n)$. By contrast, circular convolution needs only N real-by-real multiplications, for a total time complexity $\mathcal{O}(n)$.

3.2. Frequency Multipliers

After making the decision to use circular convolution for frequency filtering, we needed to generate the overtone weightings to properly convert a guitar's harmonic structure into one that matches a saxophone. To do this, we recorded 25 notes on both guitar and saxophone, and used `Matlab` to detect the relative amplitudes of each overtone for each notes. These relative amplitudes allowed us to calculate a multiplier value to be applied to each overtone, imposing the frequency characteristics of a saxophone on the guitar signal. However, our implementation of these frequency multipliers needed to address limitations in frequency resolution, high frequency content, and circular convolution adaptations.

3.2.1. Frequency Resolution

Due to the use of circular convolution to split the frequency spectrum linearly, there were inherent limits on the width of each resulting frequency bin. In order to maintain an acceptable resolution for the frequency multipliers, we designed the convolution (as well as the SDFT used for pitch detection) to use frequency bins that are 128 Hz wide. This width corresponds to the frequency spacing of the overtones for the lowest note in our training system. As the input signal increases in frequency from that note, the overtone spacing will increase, and the effective resolution of the system will improve from its baseline accuracy. This fixed bin width also required a change from our initial plan to isolate each overtone. Instead, we generated one set of frequency multipliers for each frequency bin, averaged from the training notes with fundamental frequencies in that bin. These averaged multipliers contained between 5 and 12 notes per frequency bin. A backup set of multipliers was generated from the average of all 25 notes, to be used in case the SDFT was unable to detect the fundamental frequency.

3.2.2. High Frequency Content

During our frequency multiplier testing, we noted that the guitar spectrum contained fewer high frequencies than the saxophone spectrum we were trying to replicate. Since we wanted to avoid any synthesis or sampling, we alterred our system so that instead of filtering the direct input signal we filtered the input signal summed with a scaled square of itself. The scaling allowed us to control the intensity of the high frequencies from the square relative to the lower frequencies that were already present. By training with this modified signal and performing the same operation on the input signal in our real-time system, we were able to generate enough high frequency content and weight it appropriately to match the harmonic structure of a saxophone.

3.2.3. Final Adaptations

Since we had originally planned to perform a DFT and an inverse DFT, we had generated frequency weighting coefficients in the frequency domain. After switching to a circular convolution filter however, we needed the weighting coefficients to resemble a time domain signal that could be convolved in the time domain with the input signal. To make this adjustment, we took the inverse DFT of the weighting coefficients. In testing, we found that the SDFT was unable to accurately detect the fundamental frequency during the attack and release parts of the envelope (which makes sense, since these are relatively broad-band events), so we generated a set of coefficients averaged over all 25 training notes to be used during these parts of the envelope.

3.3. Training Process

The process of generating frequency weighting coefficients can be seen in Figures 2 to 4.

3.4. Results

The task of manipulating overtones with filtering was difficult mainly due to the limitations of our real-time system, as well as the frequency content (or lack thereof) of the guitar. Given those limitations, the system performs well, and the output signal comes fairly close to that of a saxophone. The main limitation of our real-time system was that in order to complete all the neccesary processing

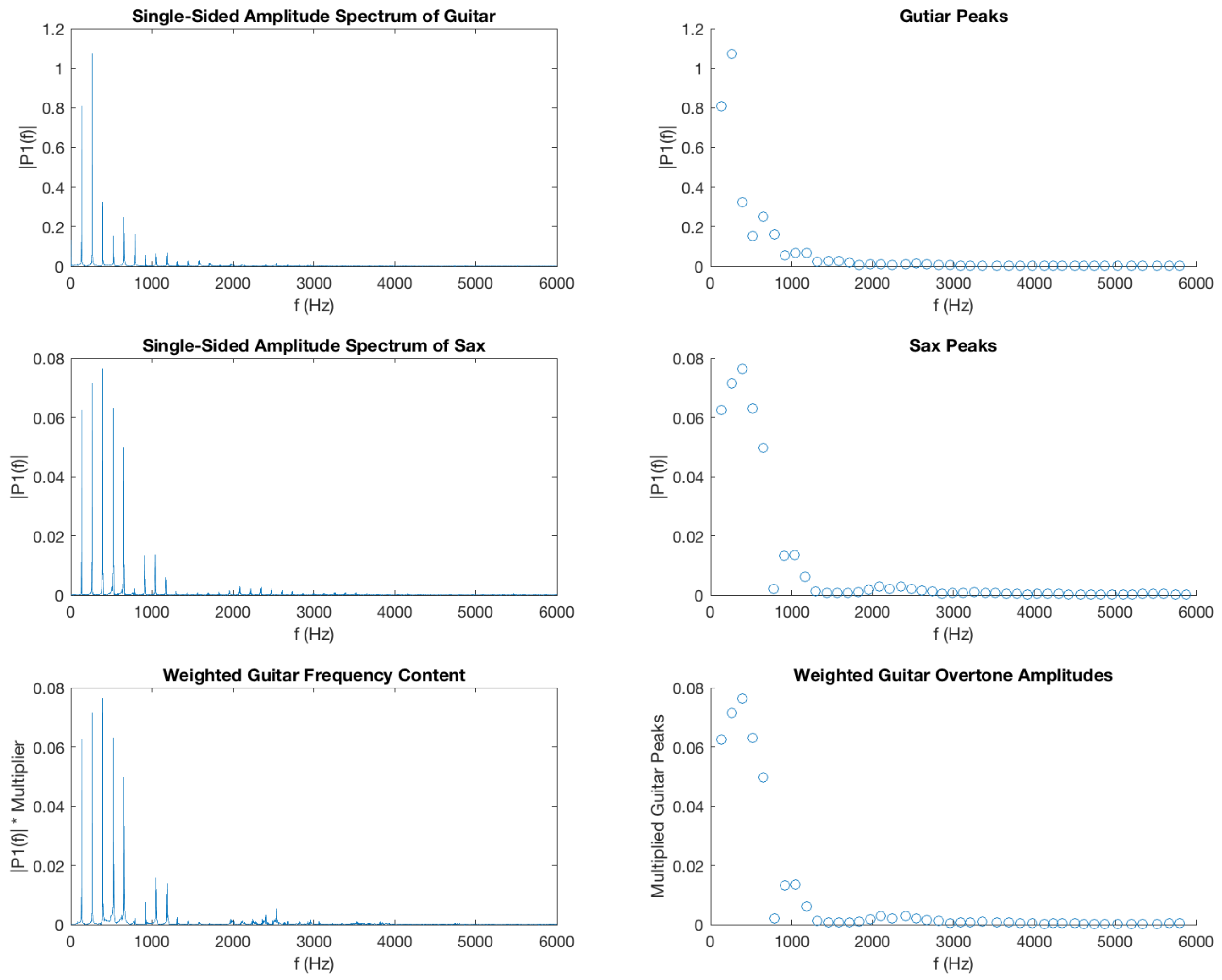
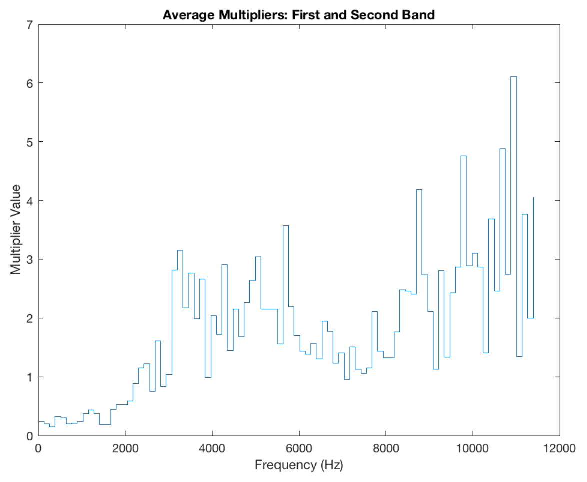


Figure 2: *Overtone Amplitude Detection and Simulation of Applied Multipliers*



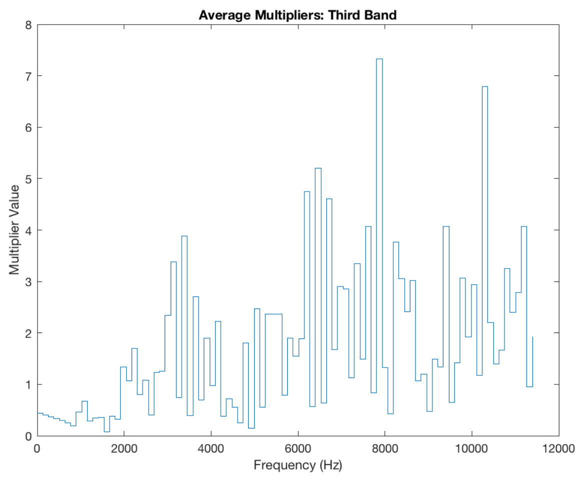
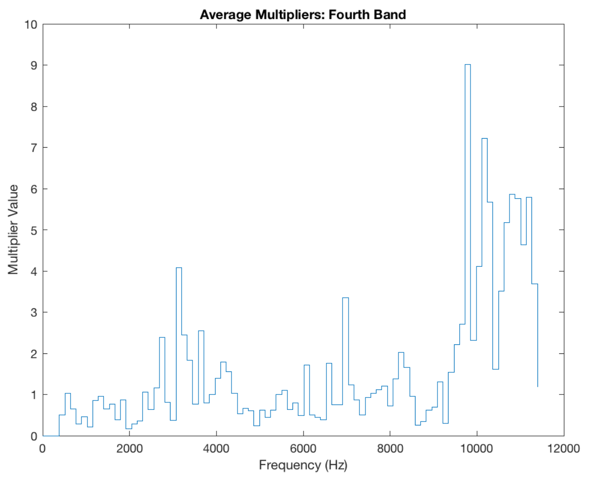


Figure 3: *Frequency Multipliers for the first and second frequency bin (left) and for the third frequency bin (right)*



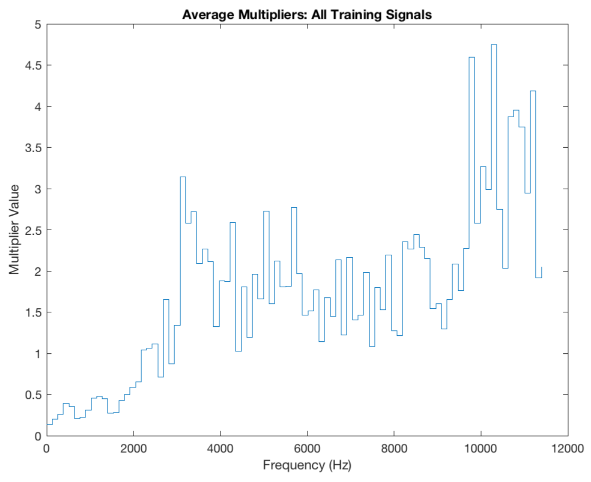


Figure 4: *Frequency Multipliers for the fourth frequency bin (left) and for all training signals (right)*

without introducing distortion from dropped samples, we needed to lower the sampling rate to 16 kHz. Additionally, the lack of high frequency content in the guitar signal made it difficult to produce a fully natural sound as taking the square of the input signal introduced some high frequency noise and distortion, that seemed to distract from the intended sound. When paired with the amplitude modulation part of the system, the output can resemble a saxophone, though with a slight dependence on playing style (e.g. picking with fingers sounds better than picking with a pick).

Some improvements that could be made to this system include more rigorous training, optimization of frequency bin widths and average multipliers, and hardware improvements. Training the multipliers with multiple versions of each note from different guitars would produce a more robust system that would sound good on many types of guitars. Similarly, optimizing the system to increase the frequency resolution would allow for more isolated overtone multipliers. This in turn would improve the effectiveness of the averaged multipliers. Finally, using a more modern DSP board would allow us to increase the sample rate, bringing us closer to the industry standard for audio technology, and allowing some of the noise and distortion from the squared signal to be separated from the input signal due to the increased frequency spectrum before the Nyquist frequency.

4. AMPLITUDE CHARACTERISTICS

Along with a unique harmonic structure, each musical instrument also has a unique amplitude envelope, helping give it a unique timbre. The amplitude envelope has three parts: the attack, sustain and release.

- Attack: The attack of a note is the event that triggers the onset of the note. For example, the attack of a saxophone would be the sound of the pick hitting the string, while the attack of a saxophone would be the sound of the tongue

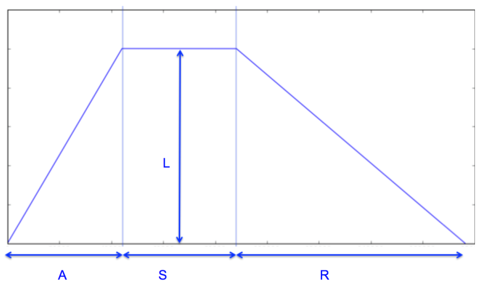


Figure 5: *The ASR Amplitude Envelope*

hitting the reed.

- Sustain: The sustain of a sound characterizes the way that the amplitude of a note changes as it is held out. For a guitar, the amplitude decays as the vibration energy of the string gradually diminishes. For the saxophone, however, the amplitude stays roughly constant as the saxophone player uses their breath support to maintain the volume of the note.
- Release: Just as the attack is the event that triggers the onset of the note, the release is the corresponding event that turns the note "off." For the saxophone the release is triggered by the player choking off their air, while the release of the guitar can be heard as the sound of the player's fingers muting the strings.

We converted the amplitude envelope of a guitar to that of a saxophone by using use note detection methods to determine when the attack, sustain, and release parts of a note began and then use multiplier tables to scale the signal to match the desired amplitude envelope.

4.1. Envelope Detection

In order to generate the amplitude multiplier tables mentioned above, it was first necessary to analyze saxophone and guitar samples offline to characterize the natural amplitude envelopes of each instrument. We implemented an envelope detection method as described in [3] which consists of rectification followed by a peaking filter followed by a low pass filter. We were then able to store the envelope values for the attack and release parts of the waveform to be used in real-time (see Fig. 7).

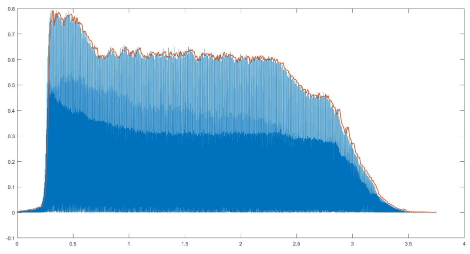


Figure 6: *Envelope Detection for Saxophone Sample*

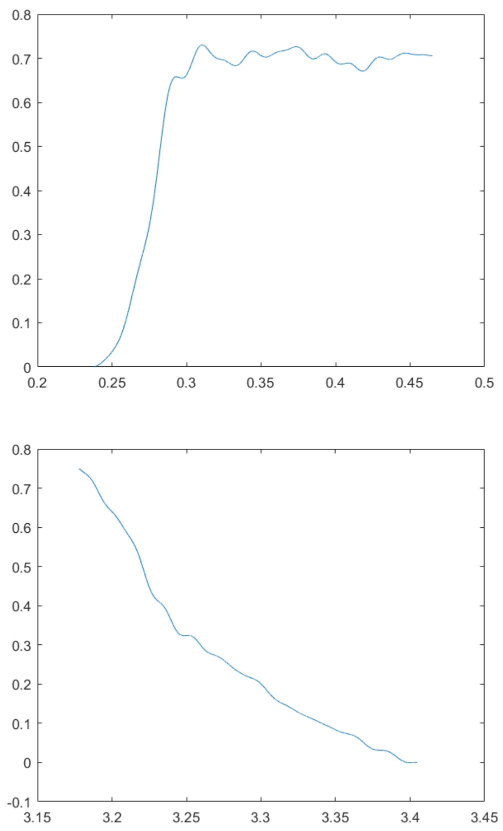


Figure 7: *Attack and Release Multipliers*

4.2. Note Detection

In order to implement the envelope modulation in real-time, it is necessary to detect the onset and offset of every note, and use those to trigger the attack and release multipliers. Following the process outlined in [4] we first implemented a short window peaking filter, and then used a threshold to determine when a new note had begun (see Fig. 8). Using the same method, but with a different threshold, we were able to detect the ends of notes as well. The issue with this method, was that if a note was already being sustained while the next note was played, the algorithm would not be able to detect the onset of the new note. To remedy this, we added another method of note detection: during the sustain part of the waveform, if the output of the peaking filter was some threshold above the average of the peaking filter window, we consider that a new note, and thus trigger the attack envelope modulation.

4.3. Sustain

Finding a transfer function for the sustain part of the waveform proved to be somewhat more difficult, because unlike the attack and release, the sustain can last for an indeterminate amount of time - as long as the player holds the note. As such, our sustain transfer function needed to be able to operate over an arbitrarily long signal without losing stability. In order to meet these demands, we model the guitar sustain as an exponential decay, and the saxophone sustain as being roughly flat. Therefore, in order

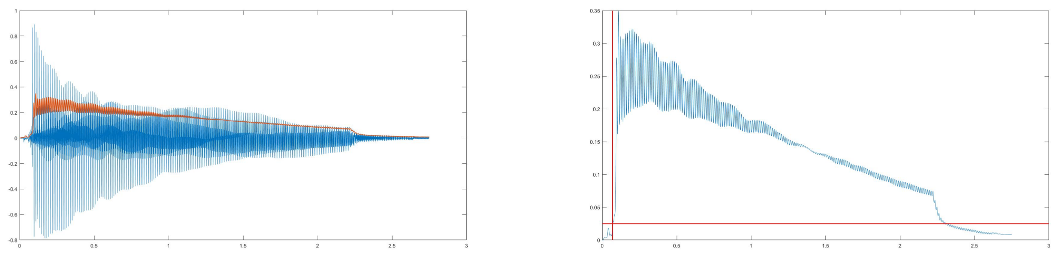


Figure 8: *Note Onset Detection for Guitar Sample*

to obtain the flat response from the exponential decay, we need to multiply the signal by an exponentially increasing function (see Fig.9). Along with giving our sound a much more saxophone-like sustain, the advantage of modeling the sustain in this way, is that the transfer function approaches a constant multiplier value, as the sustain extends to be arbitrarily long, thus ensuring stability.

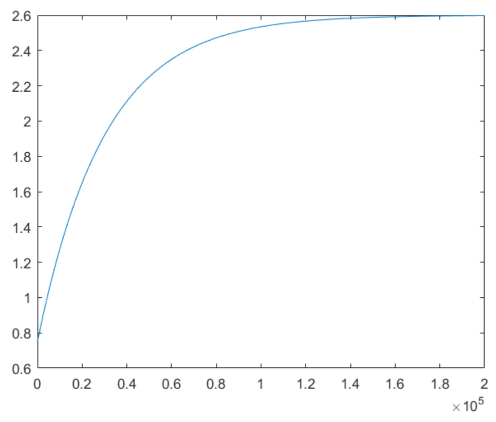


Figure 9: *Sustain Envelope*

4.4. Results

Ultimately, the amplitude envelopes worked as expected for all three parts of the waveform, creating the smooth attack that listeners expect from the saxophone, and maintaining a roughly flat sustain while allowing the performer some expressiveness. The note detection algorithm worked reasonably well for single notes, but even with the enhanced note detection scheme described above, had difficulty detecting every new note in a multi-note melody, particularly if notes were played very quickly in succession, resulting in a slightly “choppy” sound. An additional improvement would be to make the threshold values more robust, perhaps by comparing to a noise floor, rather than simply using hard-coded values. Fig. 10 shows input and output waveforms of a single guitar note passed through the real-time system.

5. FUTURE IMPROVEMENTS

Our system is currently trained to be used for one note at a time, which limits the number of things that a guitar player can do with

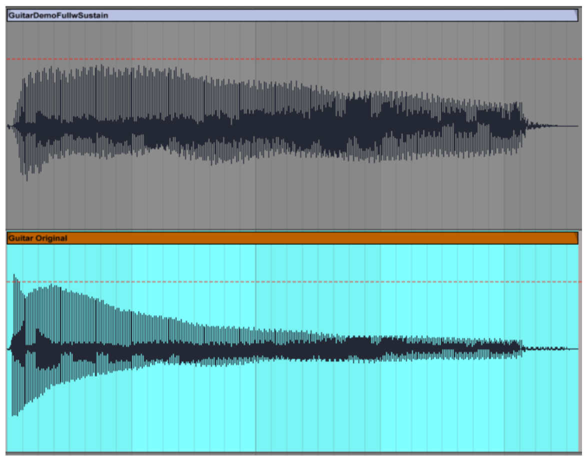


Figure 10: *Amplitude Envelope Results: Guitar Input (below), notGuitar Output (above)*

it. Even though saxophones technically cannot play chords, extending the system so the user can play a chord and have it sound like a saxophones ensemble would be interesting and useful. However, this feature would require a multi-note frequency detection algorithm, requiring more computation power.

In theory the system could be extended to work not only for guitar to saxophone, but with any melodic instrument as the input or output. Examples would include trumpet, violin, flute, etc.. This would require more off-line training to identify the frequency and amplitude characteristics of each instrument. Another aspect that would enhance the user experience would be to make the system be adaptable for users’ desired control. For example, the system could allow for the user to vary the timbre intensity and/or have a foot pedal for sustain length.

6. DOCUMENTATION OF REAL-TIME SYSTEM

A video demonstration of the notGuitar system working in real-time can be found here: `https://www.youtube.com/watch?v=0Gk7cZHvSLI&feature=youtu.be`.

7. ACKNOWLEDGMENTS

Many thanks to Richard Leahy and Antonio Ortega for guidance and support.

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