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An Enhanced Audio Tuner

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An Enhanced Audio Tuner

A Major Qualifying Project Report

Submitted to the Faculty

of the

WORCESTER POLYTECHNIC INSTITUTE

in partial fulfillment of the requirements for the

Degree of Bachelor of Science

in Mechanical Engineering

by

Jimmy Chang and Eric Forst

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Approved:

Professor John M. Sullivan, Major Advisor

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2. Frequency
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5. LabView
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1 Abstract

This report contains the detailed elements of the Enhanced Audio Tuner program. The Enhanced Audio Tuner program includes the ability to record, play back sound, and determine quantitatively the frequency of the sound. It compares this frequency to the chromatic scale, displays note and octave information along with the out-of-tune level. It displays a sequence of notes based on quarter note timing. The resultant program is restricted to single note inputs. A sequential change in notes requires X amount of time for stabilization.
2 Introduction

Music has many aspects associated with it besides sound. The broadest definition of music is sounds organized in time. Music involves more than just the sound that it creates; it incorporates human response to the music. Music must be melodic or pleasant to the ear to be considered music. The popularity of music is determined by each individual culture and/or subculture. As such, the pleasures of music often change over time. To understand how music works, sound must first be defined.

Sound is created by neighboring atoms rapidly vibrating and bouncing into each other. The atoms do not actually move very much, however they transfer their energy to their neighboring atoms. This can be more easily imagined via the Domino Effect. As one domino falls, it knocks over its neighboring dominoes, which in turn knock over its respective neighbors. In this way, energy is transferred from atom to atom and the atoms transfer this energy as vibrations to the eardrums, which then results in sound being heard.
3 Background Research

There are many different waves that travel through the air and the sound wave is the one that is involved with created music. The acoustic wave physics describes the governing equations required for analysis of sound waves. The frequencies of the sound waves form a pattern that the ear interprets as music. A musical tuner is a device that aids in acquiring the proper frequency of an instrument. When determining the frequency of a note, the Nyquist frequency must be taken into account by the tuner to fully reconstruct the signal. The equipment used had certain limitations in its computing capabilities that had to be accounted for.

3.1 Types of Waves

There are two types of waves: compression and transverse waves. Sound is a compression wave, comparatively, water is a transverse wave. (Kurtus, 2005) An example of what a transverse wave would look like is shown in Figure 1, while Figure 2 shows what a compression wave would look like.

![Transverse Wave](image)

Figure 1: Transverse Wave (Kurtus, 2005)
The waves shown in Figure 1 and Figure 2 are the same waves, but in transverse and compression form, respectively. For a transverse wave, the upper crests are the positive peaks, while the lower crests are the negative peaks. In a compression wave, the upper crests are represented by the heavy line densities, while the lower crests are represented by the lack of line densities.

### 3.2 Acoustic Wave Phenomena

Acoustic wave phenomena can be understood by examining its underlying physics. There are four major aspects of acoustic wave phenomena: The Principle of Superposition, reflection of waves, refraction of waves and the Doppler Effect.

The principle of superposition says that the two waves overlap, the net displacement is equal to the sum of the individual displacements. The equation for the principle of superposition can be written as:

\[ y(x,t) = y_1(x,t) + y_2(x,t) \]

Equation 1: Superposition Principle (Young at al, 569)
There are four main concepts relevant to the principle of superposition: constructive interference, destructive interference, standing waves and beats.

Constructive interference occurs when waves moving in the same medium in the same direction have the same amplitude, frequency and wavelength, causing the waves to be in-phase, resulting in the wave amplitudes being doubled.

Destructive interference occurs when waves moving in the same medium in the same direction have the same amplitude, frequency and wavelength, causing the waves to become out-of-phase, which results in the waves canceling each other out.

Standing waves occur when two waves, of same amplitude, frequency and wavelength, travel in opposite directions on the same medium. Superposition of the waves cause the resultant wave to appear as though it does not move left or right, hence a standing wave. The occurrence of standing waves can be shown by applying Equation 1 to the case of $y_1 = -A\cos(kx + wt)$ and $y_2 = A\cos(kx - wt)$.

Beats are created when two waves of equal amplitude and wave speed but different frequencies and wavelengths move in the same directions on a medium. Due to the superposition of waves, this causes interference, but in such a way that sounds like there are beats. The equation for calculating beat waves is shown in Equation 2.

\[
\sin(A) + \sin(B) = 2\cos[(A-B)/2]\sin[(A+B)/2]
\]

**Equation 2: Beat Wave Formula (Kurtus, 2006)**

In Equation 2, A is the frequency of wave A and B is the frequency of wave B.

Other interesting studies of the acoustic wave phenomena are the reflection of waves, refraction of waves and the Doppler Effect.

There are two main cases of reflection of waves, reflection from a hard boundary and reflection from a soft boundary. Reflection from a hard boundary causes the wave to
undergo a phase change of 180 degrees, while reflection from a soft boundary does not cause the wave to undergo a phase change.

An example of a hard boundary would be shaking a rope that is tied to a wall; the oscillation of the rope travels down the rope causing it to move one way, but when it hits the wall and comes back, the oscillation is reversed. An example of a soft boundary would be a rope tied to a pole; but because the rope can move up and down the pole, the oscillation stays the same instead of being reversed. Wave reflection is an important concept that is essential to sonar and echolocation, since they both rely on waves bouncing back to the emitter from an object.

Wave refraction is when a wave is bent as it moves between mediums. This can be explained by the example of a straw and a glass of water. When one puts the straw in the glass of water, one can observe that the straw appears bent at an angle in the water, but straight when out of the water. This is due to the refraction of light waves as it travels between air and water. This phenomenon causes the straw to appear bent, when in fact that it is known that the straw is straight.

The Doppler Effect can be best described using an example. As a train speeds towards an observer, the sound of its horn is higher pitched, but as the train speeds past the observer, the sound of its horn is lower pitched, this is termed the Doppler Effect. As the train moves forward, the sound wave it emits is more concentrated in front of the train and less concentrated behind the train. This makes sense if one considers that to create a high pitched sound; an object has to vibrate faster than to create a lower pitched sound. This shows that sounds of higher pitch have more energy. Because more sound waves are concentrated in front of the train, there is more energy from sound waves in front of the
train than in back of the train, thus causing a higher pitched sound in front of the train and a lower pitched sound in back of the train.

### 3.3 Sound and Music

The different notes of music are determined by frequencies of sound waves created by an instrument. The frequencies of a piano keyboard range from 27 Hertz to 4186 Hertz. Many instruments cannot play at both extremes of the piano range, as illustrated in the Appendix. The Enhanced Audio Tuner program is valid from a C₂ to a C₇. This range encompasses the most instruments. The frequency of each note is calculated using the speed of sound and the wavelength of each note.

The frequency of each note is calculated by using the Wave Equation as shown in Equation 3.

\[ v = \lambda \times f \]

*Equation 3: Wave Equation (Kurtus)*

Where \( \lambda \) is the wavelength of the note, \( f \) is the frequency of the note and \( v \) is the velocity of the sound traveling through the air, which is 345 m/s. (Kurtus) Although there are exact frequencies of each note, there is a small range of frequencies that an instrument can play and still considered to be playing a certain note in tune.

For a sound to be in tune, it must be within a certain deviation of the note frequency. The frequency has to be within five cents of the notes frequency. Five percent of the difference between a note and the note above or below that notes frequency
is the acceptable tolerance for a specific note to be in tune. Example 1 illustrates how to calculate the acceptable in tune range/

Example 1: Calculated in tune range

\[
\begin{align*}
\text{A}_4 &= 440 \text{ Hz, } A_4^\# / B_4^b = 466.16 \text{ Hz, } \text{G}_4^\#/A_4^b = 415.30 \text{ Hz} \\
466.16 - 440 &= 26.16 \text{ Hz} \\
26.16 \times .05 &= 1.308 \text{ Hz (upper tolerance)} \\
440 - 415.30 &= 24.7 \text{ Hz} \\
24.7 \times .05 &= 1.235 \text{ Hz (lower tolerance)}
\end{align*}
\]

\[
\text{A}_4 = 438.765 \text{ to 441.308 Hz}
\]

The complete list of ranges for each note is given in the Appendix. The lower tolerance of a note is always less than the upper tolerance since the frequency of notes grows exponentially. A musical tuner must be able to use this information to display the correct note. (Lerch) A note is considered out of tune when the frequencies do not line up properly as illustrated in Figure 3.

![Figure 3: Out of tune note.](image)

The red wave is what the frequency should be, and the blue wave represents a frequency that is out of tune. (Keur)

### 3.4 Musical Tuners

A musical tuner is a device that aids in acquiring the proper frequency of an instrument. A tuner uses an input sound from an instrument and determines the
frequency of the sound waves. Every note has a different frequency so the device is able to determine the note of the instrument. The output display lets an observer know the correct note the instrument is playing.

Some tuners have many different configurable options, one such tuner is MuseBook. This particular software includes a meter that indicates exactly how in tune the instrument is as well as whether it is closer to sharp or flat. There are many other features that enable this software to do more than just play back a note. MuseBook has a few main features such as the ability to display the wave of sound over time; it has the option of focusing on playing one note, or allowing one to play many notes and displaying them as one reach the correct frequency; and is able to play an example of a specific note such as a middle A. (MuseBook)

**3.5 Nyquist Frequency**

The Nyquist frequency “is the highest frequency that can be coded at a given sampling rate in order to be able to fully reconstruct the signal” (Weisstein, 2006). Frequencies above the Nyquist frequency will not be represented correctly and will cause aliasing. Aliasing is distortion due to signal reconstruction. The techniques used to defeat aliasing are called anti-aliasing. An example of considerations of Nyquist frequency is in an audio Cd. Audio Cd's are sampled at a frequency of 44100 Hz, therefore the Nyquist frequency of the audio Cd is 22050Hz. This frequency is chosen because the human ear cannot hear frequencies over 20kHz.

**3.6 DAQ Board Specifications**
The main hardware used for the Enhanced Audio Tuner program is the DAQ Board. Each computer in Higgins Labs 031 is equipped with one of these boards. The data acquisition (DAQ) device shown in Figure 4 is manufactured by National Instruments and has certain specifications that aid in determining the proper number of samples and the proper sampling rate of the input device.

![Multifunction Data Acquisition model-NI-PCI-6040E](image)

**Figure 4: Multifunction Data Acquisition model-NI-PCI-6040E**

The DAQ has a maximum on-board memory limit of 512 samples, so the ideal number of samples should be below that in order to maximize computing power. The maximum sampling rate is 500 kSamples/s, which is a much higher sampling rate than necessary. It is important to note, although the DAQ Board can sample 500 kSamples/s, it does not mean one can achieve that many samples. This is due to the innate hardware limitations of the computers used. Due to these factors, the number of samples and the sampling rate were kept at a manageable level, below 5,000 samples and 50,000 Samples/s, respectively. This developer defined limit also lowered the amount of computing power required to run the Enhanced Audio Tuner program.

Trial and error was used to determine the ideal ratio of number of samples to the sampling rate as shown in Table 1.
The ideal size was to have the number of results above 25, however, this should not be at too great of a sacrifice of accuracy.

### 4 Methodology

#### 4.1 Plan of Work

Stage 1:

Use Labview hardware and software to build an elementary tuner. Determine what components need to be displayed for the user. Determine what programming needs to be done by the engineers. The elementary tuner program should be able to: Dial in a note, have an audio output of that note, have an audio input of the note, save the analysis and display the real time data and results in a meaningful method. This might be similar to the ‘manual’ mode of the MuseBook Tuner.

<table>
<thead>
<tr>
<th>Time</th>
<th>Rate</th>
<th># of Samples</th>
<th>Frequency</th>
<th>Results (Theoretical)</th>
<th>Results (Actual)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.5</td>
<td>50,000</td>
<td>1000</td>
<td>101.72</td>
<td>25</td>
<td>25</td>
</tr>
<tr>
<td>0.5</td>
<td>50,000</td>
<td>500</td>
<td>102</td>
<td>50</td>
<td>50</td>
</tr>
<tr>
<td>0.5</td>
<td>50,000</td>
<td>450</td>
<td>155.98</td>
<td>56</td>
<td>56</td>
</tr>
<tr>
<td>0.5</td>
<td>50,000</td>
<td>400</td>
<td>147.32</td>
<td>62.5</td>
<td>62</td>
</tr>
<tr>
<td>0.5</td>
<td>50,000</td>
<td>200</td>
<td>337</td>
<td>125</td>
<td>122</td>
</tr>
<tr>
<td>0.5</td>
<td>25,000</td>
<td>2500</td>
<td>101.77</td>
<td>5</td>
<td>6</td>
</tr>
<tr>
<td>0.5</td>
<td>25,000</td>
<td>1000</td>
<td>101.74</td>
<td>12.5</td>
<td>14</td>
</tr>
<tr>
<td>0.5</td>
<td>25,000</td>
<td>500</td>
<td>101.72</td>
<td>25</td>
<td>26</td>
</tr>
<tr>
<td>0.5</td>
<td>25,000</td>
<td>425</td>
<td>101.94</td>
<td>29.4</td>
<td>30</td>
</tr>
<tr>
<td>0.5</td>
<td>25,000</td>
<td>400</td>
<td>102</td>
<td>31.25</td>
<td>32</td>
</tr>
<tr>
<td>0.5</td>
<td>25,000</td>
<td>250</td>
<td>106.6</td>
<td>50</td>
<td>51</td>
</tr>
<tr>
<td>0.5</td>
<td>10,000</td>
<td>1000</td>
<td>101.79</td>
<td>5</td>
<td>6</td>
</tr>
<tr>
<td>0.5</td>
<td>10,000</td>
<td>500</td>
<td>101.79</td>
<td>10</td>
<td>11</td>
</tr>
<tr>
<td>0.5</td>
<td>10,000</td>
<td>250</td>
<td>101.83</td>
<td>20</td>
<td>21</td>
</tr>
<tr>
<td>0.5</td>
<td>10,000</td>
<td>100</td>
<td>102.09</td>
<td>50</td>
<td>50</td>
</tr>
</tbody>
</table>

Table 1: Trial and Error of Ideal Rate and Number of Samples for the DAQ
Stage 2:

Use the Labview hardware and software to enhance the tuner to perform chromatic operations. As notes change the VI should recognize the note and display how to tune it automatically. A full suite of statistics is performed on the input. An enhancement to this would be: the user supplies an input file (such as a scale with timings) and the program displays the note to be played. The VI receives the input tone from the user, assesses its ‘in tune’ value and proceeds to the next note (with timing) in the input file. The software would know the timings, trim off the input from the lead and tail of the data to get the ‘steady-state’ tone from the musical instrument. Once the entire file has been run, an overall analysis of the tuning is provided to the user. One option might be a) for a single tuning adjustment (like a flute, clarinet, whistle, etc.), a single level of flatness or sharpness is displayed b) for a 6-string guitar, perhaps the statistics are displayed per string.

Stage 3:

The Labview software is made to run with the computer's internal sound card. The Labview software is compiled into a standalone executable program. The application package should be able to run on a computer that does not contain Labview. Many of the menu features (of the MuseBook Tuner, for example) should be available with the MQP product. Some features should be unique to the product. Labview is the easiest mode to perform frequency analysis for the input data. However, in Stage 3 and/or 4, Labview should not be a requirement.

Stage 4:
Stage 4 is an enhancement of Stage 3. One possible option would be to be able to take input data from another music format, such as midi or mp3. Second possible options would be to program the VI to be able to play the input file along with the user. The program would record and analyze the user’s input and assesses the level of ‘in tuneness’. This may be accomplished with tempo or some other measure associated with music. Third possible option would be to play a full scale and assess the overall quality of ‘in tuneness’ for an instrument based on the number of tuning parameters available, i.e. a component of stage 3. A last possible option would be for the program to play a tune from an input file, such as a wav or mp3 file, and have a microphone input followed by an analysis of the player's ability to keep in tune with the music file or keep in tempo. Final note: It is important that all program options from elementary through sophisticated be intuitive for the user to operate.

### 4.2 Equipment Basics

The equipments used for the LabView program were the internal Data Acquisition (DAQ) board, the BNC-2120, Labtec LCS-1012 speaker, and Sennheiser e835 microphone. The DAQ board is a part of the computer that allows any device to input a signal. The BNC board is the device that connects to the DAQ board and has many functions it can perform. The microphone is connected to the BNC board through one of the many device connectors. The speaker is used to play a note from the computer so that the microphone can acquire the sound.
4.3 LabView Essentials

The most important parts of the Enhanced Audio Tuner program are the major functions and VI's used in the program. The main software basis for the entire program is the DAQ Assistant.

The DAQ Assistant has various possible inputs such as the number of samples and the sampling rate, which can be altered to acquire the desired sample size for the VI's. The sampling rate is the number of data points collected per second. The number of samples groups that many data points into one sampling packet. An example of the DAQ Assistant is shown in Figure 5.

![Figure 5: DAQ Assistant](image)

Frequency from the Tone and Measurements function is collected with an elapsed time block to display the frequency of the signal at a specific time.

4.4 The Math Behind the LabView Program

Various formulas were used in order to create a meter to display whether how close a note is to the perfect frequency of that note. To determine the upper and lower
range of tolerance for an in tune note, $\Delta F_{lo}$, a relationship between the low tolerance, $F_{lt}$, and the perfect frequency of a note, $F_p$, has to be established as illustrated in Equation 4.

$$\Delta F = \Delta F_{lo} = F_p - F_{lt}$$  

**Equation 4: Delta Frequency Equation**

Only the lower tolerance was used to determine the range of the tolerance because the meter has to be linear. If the actual range for the high tolerance was used, then the needle on the meter would not be the same distance from the perfect frequency as compared to the same displacement on the lower tolerance range. Once $\Delta F$ is obtained, more mathematical calculations must be made to determine the lower and upper half of the out of tune distance of a note, $F_{lo}$ and $F_{hi}$ respectively, as shown in Equation 5 and Equation 6, respectively.

$$F_{lo} = F_p - \Delta F$$  

**Equation 5: Lower Frequency Tolerance Equation**

$$F_{hi} = F_p + \Delta F$$  

**Equation 6: Higher Frequency Tolerance Equation**

The meter indicates how close a frequency is to the perfect frequency by way of a percent. This frequency percent, $F_{per}$, can be obtained by using the measured frequency, $F_m$, the perfect frequency, $F_p$, and the range of tolerance, $\Delta F$ as illustrated in Equation 7.

$$F_{per} = (F_m - F_p) / \Delta F$$  

**Equation 7: Frequency Percentage Equation**

Example 2 demonstrates the process involved in creating this meter.
Example 2: Calculations behind creating the meter.

\[
\begin{align*}
\text{Note} &= B_o = \text{Natural B} \\
F_p &= 30.8706 \text{ Hz} \\
F_{lt} &= 30.0015 \text{ Hz} \\
F_m &= 30.2536 \text{ Hz} \\
\Delta F &= \Delta F_{lo} = F_p - F_{lt} = 30.8706 - 30.0015 = .8691 \text{ Hz} \\
F_{lo} &= F_p - \Delta F = 30.8706 - .8691 = 30.0015 \text{ Hz} \\
F_{hi} &= F_p + \Delta F = 30.8706 + .8691 = 31.7397 \text{ Hz} \\
F_{per} &= (F_m - F_p) / \Delta F = (30.2536 - 30.8706) / .8691 \\
&= -.70993 = 70.993\% \text{ closer to the next lowest note.}
\end{align*}
\]

4.5 LabView Programs

Several iterations of the Enhanced Audio Tuner program were created and tested before the final LabView VI could be devised. All of the data of the Enhanced Audio Tuner program was collected through a National Instruments BNC-2120 board via the DAQ Assistant of LabView. The BNC-2120 board is an internal hardware component common to all computers in Higgins Labs 031, the Structural Dynamics, Vibrations, and Experimentation Lab of WPI.

4.5.1 Basic Audio Tuner Functions: 11.30.jc.2006.vi

Version 11.30.jc.2006 introduces the basic functions of the Enhanced Audio Tuner program. This version contains all the components necessary to determine the frequency of an input signal. Notable formula nodes in version 11.30.jc.2006 are the Octave formula node, the Notes formula node and the Meter formula node. Notable VI’s and functions used are the Amplitude and Level Measurements function, the Extract Single Tone Information.vi, and the Tone Measurement function.

The Octaves formula node reads in the frequency provided by the Extract Single Tone Information.vi and assigns an octave variable to the frequency. This octave variable is then fed to the Notes formula node. There is an artificial floor implemented in the
Octaves formula node, so that any frequency lower than 25.9515 Hz is set to 25.9515 Hz. The Octaves formula node is shown in Figure 6.

![Octave Formula Node](image)

The Notes formula node takes the frequency and octave variable from the Extract Single Tones Information.vi and the Octaves formula node, respectively. This case structure ranges from 0 to 8 since there are eight octaves on a standard piano keyboard. The Notes formula node uses the octave variable to select which case to use, from 0 to 8. For each octave, there is an IF statement that governs the ranges of each note within that octave. For each note range, the frequency is assigned a Note variable, a Perfect Frequency, a Lowest Frequency for the Note, a Highest Frequency for the Note, a Lowest Tolerance for being In Tune and a Highest Tolerance for being In Tune, these variables are known in the program as Note, Fp, Flo, Fhi, FLT and FHT, respectively which are
calculated in the appendix. If the note is out of tune, it is assigned the variable Tune as equal to 0. The Note formula node is shown in Figure 7.

Fp is calculated by taking (FHT-FLT)/2 + FLT. Flo and Fhi are calculated by taking the difference between two adjacent notes, divided the result by two, and then added to the lower note. The value of each note was taken from MuseBook. FHT and FLT are half the distance between the value of the note as provided by MuseBook and the adjacent note.

The Meter formula node takes the difference between F and Fp and divides that by the difference between Fp and Flo. This sets the frequency to be a number ranging
from negative one to one. The frequency output of the Meter formula node is inputted into two meter displays. One named Tune and the other named Fine Tuning. The Tune meter displays the frequency from negative one to one. The scale for the Fine Tuning meter is +/- 20% of the Tune meter. The Meter formula node is shown in Figure 8.

![Figure 8: Meter Formula Node.](image)

Figure 9 shows the user interface of the Meter formula node.

![Figure 9: Front panel of the frequency conversion to note in-tuneness.](image)

The Amplitude and Level Measurements.vi was used for visual aid during the testing of Version 11.30.jc.2006. The Extract Single Tone Information.vi was used to
convert the voltage output from the DAQ Assistant into frequency, which then can be used by the Octaves and Notes formula node. The Tone Measurements.vi was used to compare the frequency outputs of the different functions. Figure 10 shows the configurations of the different VI’s in Version 11.30.jc.2006.

Figure 10: Block diagram of basic functions in Audio Tuner program

The user interface for Version 11.30.jc.2006 is shown in Figure 11.
The indicator shows the value of the data coming out of the DAQ Assistant. The waveform graph shows the amplitude and time of the data. The Amplitude and Level Measurements function generates a Peak to Peak indicator and a Positive Peak indicator. The Tone Measurements function is used to generate a frequency indicator. The Extract Single Tone.vi is used to generate amplitude and a frequency array. The amplitude and frequency array of the Extract Single Tones Information.vi is the leading link to the formula nodes used in the rest of the program.

Version 11.30.jc.2006 failed because the program would always display the harmonics of the desired frequency. The desired signal is the fundamental frequency. This issue is further discussed in Section 4.6.

4.5.2 A Different Method to Obtain Frequency: 1.13.2007.vi

Version 1.13.2007 was built in an attempt to overcome the harmonic frequency issue as shown in Figure 12. A equal to within Tolerance.vi takes the waveform signal from the DAQ Assistant and compares it to the waveform signal of the Negative Peak
output from the Amplitude and Measurements.vi. The Negative Peak output is analyzed to see if it is within 15% tolerance of the waveform signal from the DAQ Assistant. If the data is within 15% tolerance, the results are sent into the Tone Measurements.vi

![Figure 12: Comparison of DAQ Assistant Voltage to Negative Peak Voltage](image)

The addition of the equal to within Tolerance.vi was not effective in solving the harmonics issue. A simple comparison of the DAQ Assistant voltage and the negative peak voltage before they are converted to frequency did not rid the program of the harmonics issue. This is due to the fact that this comparison method was not a sufficient comparison of the frequency data.
4.5.3 Statistics Collector Approach: 1.29.2007.vi

Version 1.29.2007.vi was built also in an attempt to fix the harmonics issue. This version attempted to read the frequency by using the peaks of the amplitude. A peak detector function calculates the number of peaks for given interval of time. The time interval is determined by the ratio of the rate and number of samples which was .08 seconds. This ratio was multiplied by the number of peaks found to obtain the frequency. The frequency was always twice as much as it was supposed to be, the peak function read the high and low peaks when the low peaks were the only ones that were desired, so the final frequency was halved to get the correct frequency. The peak detector function worked properly but was desired to be more accurate. A collector function was placed to collect the frequencies and bundle them into groups of 2000. The grouped frequencies were then averaged to obtain a more accurate frequency. The VI’s used is shown in Figure 13.

![Figure 13: Statistics Collector](image)

4.5.4 Error Analysis: JMS-3.12.2007a.vi

Version JMS-3.12.2007a was built by Prof. Sullivan of WPI to help remedy the perpetual problem of overcoming harmonics. The JMS formula node divides the number of samples into three section, then compares the second third of the samples to the first third of the samples. The difference between the two samples is compiled into an error
array. Each element in the error array has their value squared so that the result is always positive as well as it multiplies the error so that it is easier to see on a waveform graph. The mean and the standard deviation of this error are calculated. A ceiling value is set as the total mean of the error minus twice the total standard deviation of the error. A For loop determines the voltage with the minimum error. This point of minimum error is the correct voltage of the input signal. The rate is divided by the number of points between each minimum error peak to give the proper frequency. This frequency should be the desired frequency and should match the frequency of the incoming signal from the DAQ Assistant.

A Select function is used to compare the data and error out to the voltage in the formula node. The waveform data of the Select function is outputted to an Extract Single Tones Information.vi, a waveform chart, and the JMS formula node. The waveform from the Select function is changed to an array before it is sent into the formula node. The JMS formula node has a t, s, frequency, mean, standard deviation, ceiling, error plot, error, voltage and a status output. The t, s and status output are used in the formula loops to calculate frequency. Their importance lies in their assistance in debugging the JMS formula node. All other outputs are used to understand the mechanics of the formula in the formula loop. The voltage output is converted from an array to waveform data, and then it is fed into the Select function. The JMS formula node is shown in Figure 14.
The JMS formula node succeeds in overcoming the harmonics of the input signal. However, it also introduces its own error. The error with this formula node is that the
program will fail after a certain amount of time. This time to failure is based on the number of samples and the sampling rate. The table for time to failure is shown in the Appendix. This error occurs because the DAQ Assistant is attempting to access data that has already been dumped by the DAQ Board buffer. This access error is in part due to the hardware limitations of the DAQ Board and also due to the JMS formula node code.

**4.5.5 Power Spectrum: 3.20.2007.vi**

Version 3.20.2007 was also built in attempt to overcome the harmonics issue. In Version 3.20.2007, a Power Spectrum.vi was used to obtain the correct frequency since it took into consideration the effect of harmonics. The input data to the Power Spectrum.vi was converted to a waveform so the Power Spectrum function can properly interpret the data. The Power Spectrum.vi generates a visual graph of what the frequency power spectrum looks like. A comparison function was placed to compare the amplitude of the power spectrum to a tolerance of .001, in order to only get the peaks of the Power Spectrum. A statistical measurement was performed on the resultant data and it outputted the time of the first maximum. The time of the first maximum is not a time measurement but rather a frequency measurement since the X-axis of the Power Spectrum function is measured in frequency instead of time. A formula node was created in an attempt to use the data from the Spectrum Analysis function to derive the proper voltage, and from that voltage, to derive the proper frequency. Version 3.20.2007 is shown in Figure 15.
This version was not at the point at which it could properly output the proper frequency. However, it was believed that this method could also work in order to solve the issues of harmonics. The advantage of using this method is that there would be no buffer error as is common to the JMS formula node. The disadvantage was that this program would be dependant on the microphone volume. A high or low microphone volume would affect the results of this program. Because the instrument volume is likely to change rapidly throughout the tuning process, this program was abandoned at this stage, due to the fact that the JMS formula node had no problem with changing microphone volumes.
4.6 Major Setbacks

Through development of the audio tuner program, many problems arose. Those of the most importance had to do with the way the program calculated the frequency of the input signal. The problem was that the program reads in volts. The program itself had to convert volts into frequency. This, in of itself, is not a problem. The problem arises when the program interprets the harmonics of a frequency instead of the actual frequency. It was difficult to deal with this problem, because many solutions that would fix the overall program required the program to have the correct frequency. One of the biggest hurdles was to figure out how to retrieve the correct frequency. This was achieved through many different means and each solution had a different problem.

The harmonics of different notes caused the majority of the problems with the Enhanced Audio Tuner program.. The problem with harmonics is trying to force the program to ignore it. This is because since notes, like all sound are waves and every wave has peaks. These peaks represent the points in the wave where there are the highest signal strengths. The desired frequency is the fundamental frequency; however the stronger signal being picked up is a harmonic of the fundamental frequency. The fundamental frequency is the true frequency that the program should extract out of the input signal. However, because the signal strength of the harmonics of the fundamental frequency is higher than the signal strength of the fundamental frequency, the program detects the strongest signal and outputs that as the frequency of the system. In reality, the frequency that is desired is not the strongest frequency but the fundamental frequency, which happens to be the first frequency. An example of this concept is shown in Figure 16.
Many attempts were tried at solving the harmonics issue. Some failed attempts were using peak to peak analysis and the Power Spectrum function in LabView.

Peak to Peak analysis involved taking data between peaks and using its internal functions to output data. The issue with this was there were minor peaks between the large peaks. The peaks of interests are only the large peaks. There was no viable way to extract only the data from the large peaks.

The Power Spectrum function did not give the proper results. This was due to the various methods used for extracting data only exported the strongest signal, which is the harmonic of the fundamental frequency. The data desired is the first peak, the fundamental frequency. Different tolerance functions were used to try to extract the data from the Power Spectrum, but it was all to no avail. This method was completely abandoned when it was tested with a microphone to find that the Power Spectrum data would vary with change in microphone volume. This would mean a change in microphone volume would introduce errors into the results of the Power Spectrum. In

Figure 16: Power Spectrum for an A4, 440 Hz. 
other words, the program would then only work if the instrument was played within a specific volume range.

Many different frequency functions were tested to try to obtain the fundamental frequency. None of the functions could obtain the fundamental frequency flawlessly. Each function had a working range. If the signal is outside of that range, it would only output the harmonics of the intended frequency. None of the different attempts would provide a consistent and accurate result across the entire range of the program. One idea was to combine two partially working solutions and have them activate only when the fundamental frequency were within their working range. The error analysis method work for low frequencies and the Tone function obtained the correct frequency for higher frequencies. This was achieved by building a For loop that would separate the signal. The For loop in question did not solve the error as intended since the program would still not cover the entire range. After several attempts at different For loop formula, the frequency at which the consistency between the two solutions changed was isolated. A solution was implemented at this point. The solution was simply to input both signals into the Octave loop and determine the control for the solution inside the Octave loop. This was done by setting the solution, which functioned for the lower range, to be the general solution. When the frequency was above 1kHz, the higher range solution was set to the general solution, which was the same style as the original VI. This made sure that the rest of the program would not have to be changed to reflect new and changing symbols.
5 Results

The final VI contains many different capabilities. It displays the frequency of an input note and octave. It displays the note and octave of an input frequency. It display whether or not the input frequency is in tune with a note. It has two meters to show how far a note is in or out of tune. The Tune meter shows the note and its proximity to the adjacent notes. The Fine Tuning meter shows if a note is perfectly in tune, in tune or out of tune. The data from the sound can be recorded to an excel file through a record sound button, refer to Figure 9 for this function.

The program can record sound as a wav file as illustrated in Figure 17 and Figure 18.
The program also contains a library of notes for the entire range of the program as illustrated in Figure 19 and Figure 20.
The final VI mostly works. Due to the way that the JMS formula node works in combination with the sampling rate and the number of samples of the DAQ assistant, the program shuts down around one minute and thirty seconds. The break down is due to attempting to access data that has already been dumped by the hardware. The JMS formula node is illustrated in Figure 21.
Figure 21: Block Diagram of JMS formula node.

The Quarter Timing Display did not fulfill its original intent. Instead of displaying the note in quarter note timing, the frequency is displayed in quarter note timing as illustrated in Figure 22 and Figure 23. This makes it harder to read than just displaying the note.
Figure 22: Block Diagram to playback recorded data
Figure 23: Front panel to playback recorded data

The final VI is capable of taking a note inputted from the user and displaying the frequency of that note as illustrated in Figure 24 and Figure 25.

Figure 24: Block diagram for displaying the frequency from a selected note.
The VI is still reliant on the DAQ board and LabView to run properly. Currently a microphone has to be connected to the DAQ board and the program has to be run in LabView for it to work. The original intent for this part was to make the program be able to accept a microphone input from the computer's microphone jack. The program was to be made an executable so it can be used without LabView.
6 Conclusion

The final VI works great for one minute and thirty seconds, after which it breaks down. However, one minute and thirty seconds is a reasonable amount of time for one to tune an instrument. Within the time constraint, all the functions of the program worked as intended.

The combination of the DAQ Board and the JMS formula node code caused the break down error. The DAQ Board attributed to the break down error due to the computer hardware’s limitations in computing capability. The JMS formula node code attributed error due to it trying to collect samples that have already been removed from memory. To solve this error, one would need to change how the strings of data are compared. [was this a function of the CPU processor? What recommendations did Labview suggest and what were their outcomes?]

The issue with displaying notes with the Quarter Timing Display is trying to convert number data into string data with LabView. One cannot assign a string to a variable in the formula nodes of LabView. Instead, one would need to use a switch which would hold strings inside the switch loop. The Quarter Timing Display was left as a frequency display instead of a note display due to time constraints. The team wanted to be able to present this function at Project Presentation Day. Due to this, the Quarter Timing Display was left as a frequency display since it would not introduce any errors in this way.

The single greatest disappointment of the project was the inability to convert this program into a standalone program. Changing the data acquisition input in LabView opened up a new horizon of problems. This is due to different data acquisition functions having different inputs and outputs. To use a different input function, one would need to
redesign many aspects of the LabView program around this new input function. The team estimates that would take an extra week or two's worth of work to complete the transformation. And so, this aspect of the project was abandoned in favor of making other aspects of the program fully functional by presentation day.

For future experimentation with this project, the team had the following suggestions. Firstly, is to change the way the JMS formula node works. There are always multiple ways to write the same code, the change needed in this code is to change the way the samples are compared for error. An idea for expansion to the project would be to account for power chords in the data recording. Another idea would be, instead of outputting the Quarter Timing Display on screen. This program has an option to print out a sheet of music. This combined with LabView code may make printing out a full sheet of music feasible.
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Figure 26: Frequency Range of Musical Instruments. (Alten)
Table 2: Calculation involved with acquiring the correct note

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Table 3: Time to Failure
Figure 28: MuseBook Tuner version 2.2