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Proposal for
COMP 4905 – Honours Project

RT-Tuner: A real-time audio analysis tool

It is my intention to develop a piece of software which provides an avenue for the inspection and analysis of real-time (ie. live) audio data, ultimately resulting in a user-friendly graphical tool which may be used to perform some degree of real-time musical instrument tuning. In order to achieve this, I intend to develop a tool in the form of a PC application written in the C++ programming language. The tool's two primary functions will be divided into two modules, with each of two development phases focusing primarily on one of these modules: Phase 1 will focus on the design and development of a real-time audio-data representation/visualization tool; and Phase 2 will focus on the design and development of a decision-making module, which will make use of the audio data and visualization mechanism from Phase 1 in order to determine (and display) some useful information which will aid the user in tuning their instrument.

In more detail, the first module – which I will call **AVIS**[1] here – is intended to be a piece of software whose function it will be to receive live audio data from some source (most likely a microphone or similar input device), and display this data visually in some meaningful way. Precisely which information will be displayed and in what format is yet to be determined, but some example possibilities include the display of a waveform or audio fingerprint, a frequency-spectrum display showing the frequency make-up of the audio signal, and a power/decibel indicator. The decision of what data to use will depend on the results of the requirements analysis to be done during the early stages of the design process. In order to parse out the necessary data for such a display, a variety of tasks will need to be performed which will make use of existing libraries, such as the conversion of the audio signal to some sort of digital format, and perhaps the application of a mathematical Fourier transform package to convert the data from the time domain to the frequency domain which will allow for further analysis. This means that my primary tasks for the AVIS will include parsing the digitally-converted audio input, intelligently applying any necessary mathematical transformations in order to extract the desired data, processing this data in order to obtain the desired results in a display-friendly format, and designing and implementing the graphical user interface to display this data to the user.

The second module (which I've taken to calling the **BACKSTOP,** for fairly obscure reasons[2]) will focus, as mentioned above, on some decision-making process using the AVIS module developed in Phase 1. Specifically, this module will, using AVIS as its basis, attempt to find the nearest fundamental note to that of the live audio and then display this information in some user-friendly way, in the hopes of providing a simple visual cue that will allow a user to correct or maintain the pitch/tuning of an instrument. In order to achieve this, it will be necessary to carefully analyze the frequency-spectrum data, likely through the application of various Fourier analysis techniques (eg. digital filtering, etc.), and compare these results against known values for common frequencies for the fundamental notes[3]. As the stated purpose of this tool is to allow for a user to perform real-time adjustments to correct or

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1 "AVIS" is simply an abbreviation of "audio visualization"

2 In baseball, "backstop" is a slang name for the catcher, who is responsible for helping keep the pitcher in line with respect to his control and pitch selection – which is more or less what this tool is intended to do for its users.

3 In the interest of keeping things simple, an even-tempered scale with 'A_4' at 440Hz will be assumed, but if time allows it may be worthwhile to implement a function allowing a user to select from a variety of other tunings.

maintain the tuning of an instrument, these comparisons would ideally be performed over some time window, allowing a user to see how the signal is behaving over time, but whether this is feasible to implement will need to be determined. Thus, the primary tasks for this module include performing in-depth signal analysis, performing a data comparison to known values and deciding the most likely correlate, and implementing a user-friendly visualization of this information.

With regard to the logistics of this project, it is expected that minimal external hardware or software will be required. In terms of hardware, only a **laptop PC** (to be used both for development and as a test-bed), and a **microphone** (or similar audio input device) are likely to be required. As for software, at this stage it is expected that only two external packages will be needed, outside of the standard C++ libraries: some **audio signal conversion library** will be required in order to render the analog audio signal into a usable digital format; and a **"fast Fourier transform" library** will be required to quickly perform the calculations for the time-to-frequency domain conversion which is necessary for in-depth analysis of the audio signal. With respect to deliverables, the three primary end-products are expected to be the "**AVIS"** visualization tool resulting from Phase 1; the "**BACKSTOP"** instrument-tuning extension resulting from Phase 2; and the **final report**. Additional secondary, intermediate deliverables – which are intended to be submitted only to the project supervisor – include a ***mid-term report*** and a ***first draft*** of the final report. An overview of my tentative expected schedule for these deliverables follows:

1. **Phase 1, "AVIS"** – to be completed on or about the week of **Feb. 16th**
2. ***Mid-term report*** – to be provided to supervisor on or about ***Feb. 25th***
3. **Phase 2, "BACKSTOP"** – to be completed on or about the week of **Mar. 16th**
4. ***First draft*** of final report – to be submitted on or about ***Apr. 3rd***
5. **Final report** – to be completed on or before **Apr. 17th**

As someone with a limited formal background in either music or digital signal processing, but some amount of amateur experience with both, I expect this project to be a challenging undertaking, but not an overwhelming one. Having done a good deal of research, and being very interested in both of these subjects, I am confident that I have sufficient domain knowledge, that I will remain thoroughly engaged throughout this endeavour, and that I will be able to effectively complete all objectives outlined here in the required time frame.

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