Computer Science Honours Project Proposal

An Investigation of Digital Mixing and Panning Algorithms

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Statement of the Problem

In the sound engineering community there is debate about the sound quality of an audio mix that has been digitally summed instead of summed using an analogue mixing console. For the purpose of this proposal the terms "summing" and "mixing" will be used interchangeably and an "analogue" sample will refer to audio that has been summed in an analogue console, even if it played from a digital system. Some professionals say that a digital mix is lacking the "undeniable depth, width, punch and realism" (Farmelo, 2011, p. 1) of an analogue mix. Others claim that digital mixing is superior to analogue mixing, but they usually acknowledge that the limitations of analogue equipment might account for the difference in the perceived character of the mix (Cooper, 2004). For sound engineers who have been recording and mixing for the past 20 years, this unexplained variation of sound is a source of frustration, and summing boxes are even being sold for the sole purpose of creating an analogue mix. Since not much research has been done in this field, there is also no standard interface available to audibly test and compare analogue and digital audio samples.

Objective of Research

The first goal of this project is to definitively prove whether a visual or audible difference in sound quality exists. If it does exist, then a second goal is to determine whether a digital mixing algorithm can be created to satisfactorily emulate the sound of a mix summed in an analogue console. The third objective is to design and implement an interface that facilitates user comparison of two audio samples. The user should also be able to control the mixing algorithm and other variables affecting the sound of a digitally-summed mix via the interface.

History and Background

In order to compare digital and analogue mixing methods, some background information about audio recording is needed. Audio was first recorded using analogue equipment, usually onto magnetized tape. A recording device, typically a microphone, would record and convert fluctuations in air pressure to a measurable electronic signal. These voltages would then be recorded onto tape (Watkinson, 2001). As new technology has been developed, digital equipment has been created to record, edit and mix audio signals. However, microphones still capture analogue signals, so some kind of conversion needs to take place for audio to be processed using a Digital Audio Workstation (DAW). This has triggered many discussions about the quality variations between analogue and digital equipment and even among the "perceivable difference in sound quality between different DAWs" themselves, as said by Leonard *et al.* (2012, pp. 1).

To convert the recorded voltages to a digital audio signal, measurements, called samples, are taken at fixed time intervals. This pulse code modulation method is generally accepted as the standard system used in the conversion process. To make sure none of the original signal is lost, samples need to be taken at a very high rate of 44.1kHz (Watkinson, 2001, pp. 207–209). This specific rate was chosen for numerous reasons - one of which is that 44.1kHz is more than twice the rate of the highest frequency humans can hear. These voltage readings are stored numerically and the wave can then be easily recreated (Watkinson, 2001).

Since a song normally consists of multiple tracks, a DAW needs to be able to sum these separate tracks to create a master stereo track. This process is not as simple as adding the two values together, as this new summed value could "over- or underflow the range available" as Vogler (2012, pp. 1) puts it. Therefore various mixing algorithms have been developed which aim to add the tracks together in such a way that the level of each individual track is not perceived to be louder or softer than it was originally. Since so many different methods have been created, there is a diverse selection of DAWs available and industry recording engineers argue about which DAW is technically the best for summing audio. There is also a big debate over whether digital software can recreate the summing technique of an analogue mixing console.

This has led to various listening and visual tests amongst DAWs and digital and analogue systems. The most recent testing was done by Leonard *et al.* (2012) where the internal summing of five different DAWs was investigated. After some initial testing they discovered an error in the panning of one of the tracks. This resulted in an examination of the effect that panning had on the summing of the tracks. They found that the different panning laws of the DAWs caused significant variations in output levels and sound quality.

Another experiment done by Aarts *et al.* (2007) compared two DAWs with different sampling rates to an analogue system in order to find the most similar sounding DAW. Their tests showed that the DAW with the higher sampling rate was said to be the system that sounded most like the analogue system. They do not, however, say if there were any audible differences between the analogue and digital systems.

Approach

First, a recording session needs to be done so that analogue and digitally mixed audio samples can be obtained in order to determine any visual or audible differences. This will be completed with the help of the Sound Technology consultant, Corinne Cooper. The best equipment available will be used to achieve the best quality possible. There will also be control samples, consisting of sinusoidal waves to be used in the visual testing. Once these samples have been recorded there will be a series of listening and visual tests to determine whether any significant differences can be found. The listening tests will be performed by an expert panel of audio professionals. Research will need to be done in order to execute the visual testing.

Next, a program will be coded to statically add and play an audio track in Waveform Audio File (WAV) Format. This program will be extended to take multiple audio tracks and statically mix them using various known mixing algorithms and panning laws such as in Griesinger (2002). The algorithms to be coded will be taken from various resources, including published open source DAWs. Once these algorithms have been implemented, new algorithms will be designed in an attempt to achieve digital mixes that sound the same as analogue mixes.

At this stage, an interface for the listening tests will be designed and implemented. The user will be able to alter various variables, including the panning, volume and mixing algorithms used, to quickly change between and compare two different samples. A mock-up of a possible interface design for comparing an analogue test sample with a digital mix is shown in Fig. 1. Although only four tracks are shown in the mock-up, any number of tracks may be used in the digital mix.

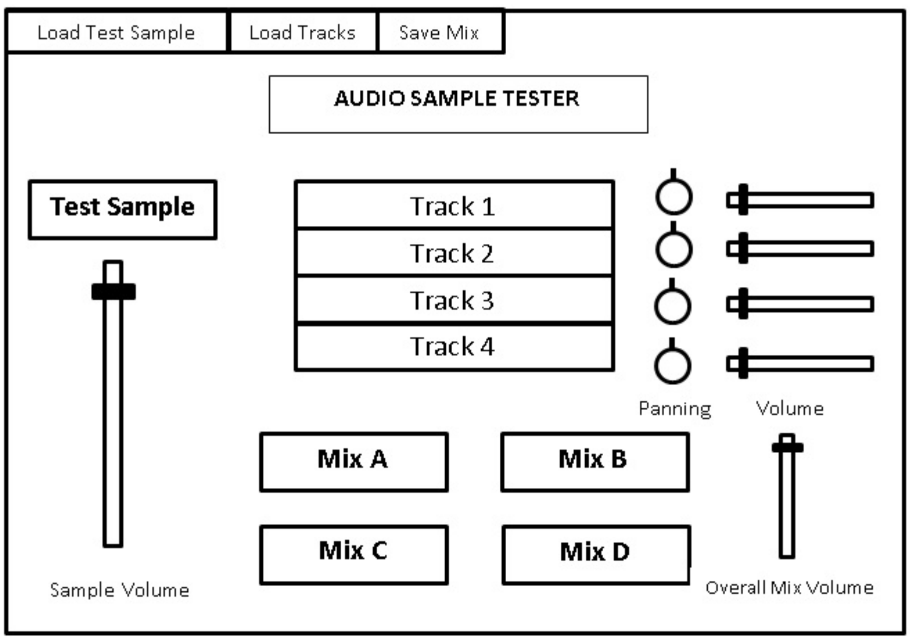

Figure 1: Interface Mock-up

This interface will then be used in a second set of listening tests to compare the analogue samples with digital samples created with the new mixing algorithms. More visual testing will also be done. After testing, the results will be analyzed to determine whether the new mixing algorithms sound similar to the sound of the analogue samples.

Timeline

First Semester	Proposed Dates
First Term	
Read Literature related to Project	1 March - 30 April
Record Analogue and Digital Samples	3 March - 9 March
Prepare Presentation	Due 11 March
Read in and Play WAV Files Statically	14 March - 3 April
Listening and Visual Testing	24 March - 30 March
Second Term	
Have Multiple Tracks Summed Statically	14 April -30 April
Implement Known Mixing Algorithms	1 May - 15 June
Design Testing Interface	28 April - 20 May
Literature Review	Due 30 May
Experiment with Mixing Algorithms	1 June - 13 July
Second Semester	
First Term	
Experiment with Panning Laws	1 July - 1 August
Implement Testing Interface	1 August - 20 August
Second set of Listening and Visual Tests	25 August - 5 September
Second Term	
First Draft of Paper	1 September
Short Paper Handed In	Due 15 September
First Draft of Thesis	15 October
Final Draft of Thesis	25 October
Final Project Write Up	Due 31 October
Website Complete	Due 7 November

Requirements and Resources

- The playing of WAV files will be coded in C++ using the JUCE library.
- An Audio Stream Input/Output (ASIO) driver with a Steinberg UR22 USB audio interface .
- Mixing and panning algorithms will be found by looking at the code of open source DAWs such as Ardour and Rosegarden, as well as references such as Vogler (2012) and Aarts *et al.* (2007).

Possible Extensions

- Once the tracks can be statically played and mixed, a possible extension would be a program that could mix and play the tracks in real-time.
- The interface will only allow for built-in mixing algorithms. A possible modification would be to allow the user to plug-in their own mixing algorithms to test.

References

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