A Comparative Study of Analogue and Digital Mixing Techniques

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ABSTRACT

The difference between the summing quality of analogue and digital audio is widely debated. It was hypothesized that a mechanism to provide the characteristics of an analogue mixer could be found and implemented in a digital mix. An Audio Mix Analyzer was designed and implemented to facilitate the creation of audio tracks using various mixing algorithms, and to enable the comparison of two audio samples. Listening tests were performed and a difference between the analogue sum and the digitally-summed audio. This difference was dependent on the genre of the song and the number of tracks being summed.

Categories and Subject Descriptors

H.5.5 [**Sound and Music Computing**]: Methodologies and techniques; H.5.5 [**Sound and Music Computing**]: Signal analysis, synthesis, and processing; I.1.2 [**Algorithms**]: Analysis of Algorithms

General Terms

Audio, Mixing Algorithms

Keywords

Analogue, Digital, Mixing

1. INTRODUCTION

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 paper, 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" [7, p. 1] of an analogue mix. However, as Leonard *et al.* [9, p. 1] states, there is not much "quantifiable evidence to support these claims" yet. And although others, like Cochrane [5], point out that summing with a digital audio workstation (DAW) or analogue console will not necessarily improve the quality of a song if it has not been correctly mixed (artistically speaking), it is generally accepted that the limitations of analogue equipment might account for the difference in the perceived character of the mix [6]. 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, like the Crane Song Egret or Rupert Neve Design 5059 [16], are being sold for the sole purpose of creating an analogue mix.

The hypothesis for this paper is that a mechanism to provide the characteristics of an analogue mixer can be found and implemented in a digital mix. The first goal of this project was to definitively prove whether a visual or audible difference in sound quality exists through visual and aural testing. The second objective was 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 using the interface. Lastly, it should be determined whether a digital mixing algorithm can be created to satisfactorily emulate the sound of a mix summed in an analogue console.

2. MIXING AND THE ANALOGUE/DIGITAL DEBATE

2.1 Fletcher-Munson curves and RMS Levels

Not only are human ears limited to the frequencies they can hear, but they perceive some frequencies to be louder or softer than others. A set of experiments performed by Harvey Fletcher and W.A. Munson in 1933 [8, p. 12] resulted in the Feltcher-Munson curves (as shown in Fig. 1) or equal-loudness curves. These curves show that in order to hear a bass frequency under 100 Hz as loud as a mid-range frequency of, say a 1kHz wave, the level of the audio needs to be higher. In other words, bass frequencies are not heard clearly in softer audio but our ears are very receptive to frequencies between about 2 kHz and 5 kHz [18, p. 45]. The

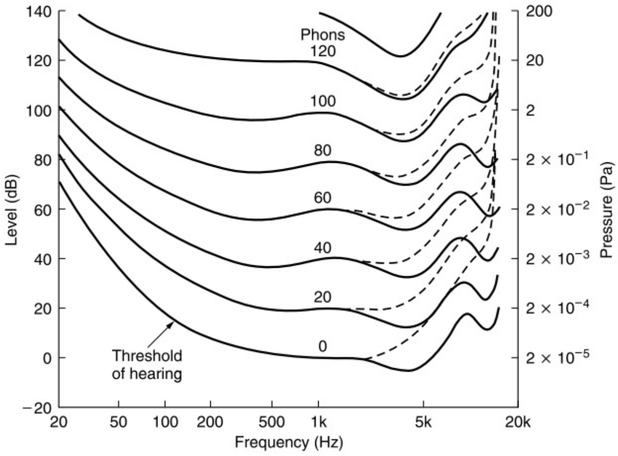

Figure 1: Fletcher-Munson curves [18, p. 46]

fact that the "ear's frequency response changes with signal level" [13, p. 30] means that the average loudness of two audio tracks need to be the same to compare them [8, p. 96]. The Root Mean Square (RMS) level can be used to measure this average loudness of an audio signal. It can be worked out mathematically using Eq. (1) [12], where x is the level of a sample and n is the number of samples.

$$rms = \sqrt{\left(\frac{\sum x^2}{n}\right)} \tag{1}$$

3. CURRENT TECHNIQUES FOR DIGITAL MIXING

3.1 Mixing Algorithms

A song typically consists of multiple tracks, therefore a DAW needs to be able to sum these separate tracks to create a master stereo track. This process is not always as simple as adding the two values together, as this new summed value could clip or "over- or underflow the range available" as [17] puts it. The source code of three open source DAWs were investigated to determine the specific summing algorithms, gain representation and panning laws used.

Although different gain and pan algorithms are used, each DAW sums the samples in very similar ways. This means there is an opportunity to modify and experiment with this process when implementing new mixing algorithms to emulate the analogue mixing process.

Since the effect of audio summing methods has been so widely debated, alternative methods for adding the samples together have been developed. For example, Toth devised Equation (2) to calculate the summed value of two samples a and b that themselves have values between 0 and 1. This formula can be extended to account for samples that could be added together without clipping to get Equation (3). However, as Voegler [17] points out, this method is not symmetrical (see Figure 2) and favours extreme values in the subsequent mix.

$$sum = a + b - ab \tag{2}$$

$$sum = \begin{cases} 2ab & a < 0.5 \cap b < 0.5 \\ 2(a+b) - 2ab - 1 & a \geq 0.5 \cup b \geq 0.5 \end{cases} \tag{3}$$

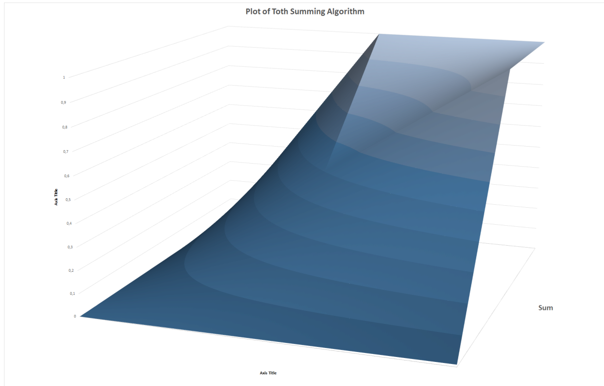

Figure 2: Viktor T. Toth Method

Voegler [17] himself developed a method for adding two samples together which he calls "Loudness Normalization by Logarithmic Dynamic Range Compression". First, he developed the Linear Dynamic Range Compression method presented in Equation (4) where $x = a+b$ and is a floating point value between 0 and 2 and t is the chosen threshold between 0 and 1. As can be seen in Fig. 3, where a threshold of 0.6 is used, the compression takes effect pretty dramatically, which results in an audible reduction in level. To lessen the impact of this, Voegler [17] used mathematics beyond the scope of this paper to produce a Logarithmic Dynamic Range Compression method given in Equation (5). A list of $f_\alpha(t)$ values corresponding to various thresholds was given in Voegler's publication [17]. Since a threshold of 0.6 was used, a value of 7.48 was used for $f_\alpha(t)$. This results in a smoother transition as compression is applied as shown in Fig. 4. A goal of this project was to determine what effect implementing these alternative summing methods would have on the sound quality of a digital mix.

$$sum = \begin{cases} x & -t \leq x \leq t \\ \frac{x}{|x|} \cdot (t + \frac{1-t}{2-t} \cdot (|x| - t) & |x| > t \end{cases} \tag{4}$$

$$sum = \begin{cases} x & -t \leq x \leq t \\ \frac{x}{|x|} \cdot (t + (1-t) \cdot \frac{ln(1+f_\alpha(t) \cdot \frac{|x|-t}{2-t})}{ln(1+f_\alpha(t))} \cdot & |x| > t \end{cases} \tag{5}$$

4. A SYSTEM FOR TESTING MIXING TECHNIQUES

4.1 Design

After investigation it became clear that there was no established method for testing two audio samples. There was also no interface available that enabled a user to instantly and seamlessly switch between an analogue sample and a digital one to hear if differences between them existed. Therefore, it was decided that a program with this functionality should be created to assist with listening tests. This would accomplish the first goal of the project. The second goal was

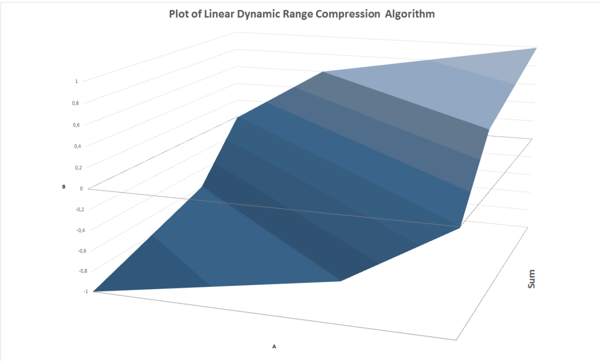

Figure 3: Linear Dynamic Range Compression
(t = 0.6)

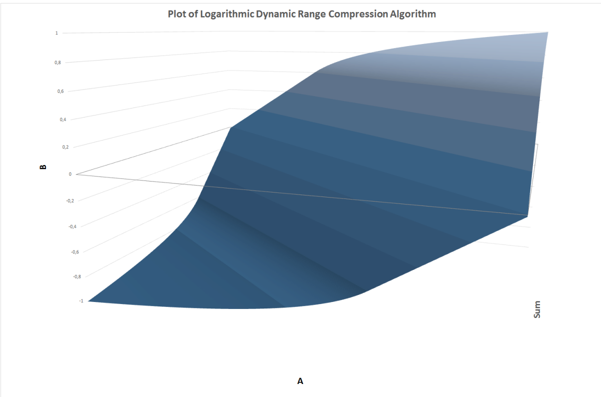

Figure 4: Logarithmic Dynamic Range Compression
(t = 0.6, $f_{\alpha}(t)$ = 7.48)

to determine if a digital mixing algorithm could emulate the summing of an analogue console. Mixing and saving multiple samples using various mixing algorithms would be easier to achieve with a user-friendly interface.

These two goals motivated the development of a program, the Audio Mix Analyzer, that could both create a digital mix of up to eight audio tracks with different mixing algorithms and allow a user to evaluate the audio samples that were created. When creating a mix the user would be required to:

- Start the application
- Load tracks to create a mix
- Create a mix
- Save a mix

When evaluating a mix the user should be able to:

- Load two tracks for comparison
- Play the tracks out of an audio device
- Switch between the tracks

Based on the design of the Audio Mix Analyzer and the requirements of audio programming, it was decided that C++ [1] and the Jules' Utility Class Extensions (JUCE) library [4] would be used for the implementation.

4.2 Audio Mix Analyzer

After the requirements and design of the program had been finalized, and the programming environment had been chosen, the Audio Mix Analyzer could be implemented. There were five main components to this process:

- the interface,
- the mixing algorithms,
- the adjustment of the RMS level,
- the playback of tracks, and
- the timing of the algorithms.

The Mix Creation tab of the Audio Mix Analyzer (see Fig. 5) allows the user to create two different mixes at once and the interface is split into two identical halves. Each half has the potential for eight 'Load Track' buttons, pan and gain sliders, a combo box used to select a mix, a slider showing the RMS level and 'Mix Sample' and 'Save Sample' buttons.

Each row of track controls corresponds to one track. From the beginning of the project, the decision was made to only allow a maximum of eight tracks per mix for each half. When a user begins the program only one row of track controls is visible for each mix. Once the user loads the first track, the second row of controls becomes visible. After the second track is loaded, controls for the third are shown, and so on.

Once the user has loaded all the tracks they want, they can select the mixing algorithm they want to use using the 'Select Mix' combo box. When the 'Mix Sample' button is pressed the samples will be summed using the specified algorithm. During the summing process, the program also ensures that the RMS level is within a certain range. When the 'Save Sample' button is pressed a file browser is opened for the user to save the newly created audio file.

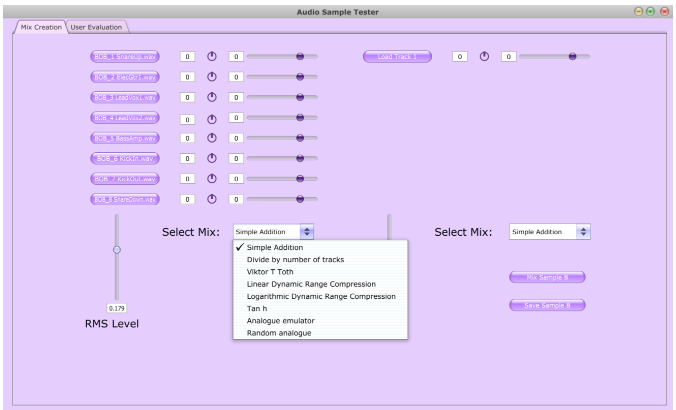

Figure 5: Mix Creation tab
of Audio Mix Analyzer

After all the audio wave files required have been created, the user can use the User Evaluation tab (see Fig. 6) to compare two tracks. This tab allows the user to load two tracks using the 'Load Track buttons'. It also allows the user to control the playback of audio files using 'Play', 'Pause' and 'Stop' buttons and to switch between audios files by pressing the 'Play Track A' and 'Play Track B' buttons. The RMS level of the audio file is displayed after a file has been loaded.

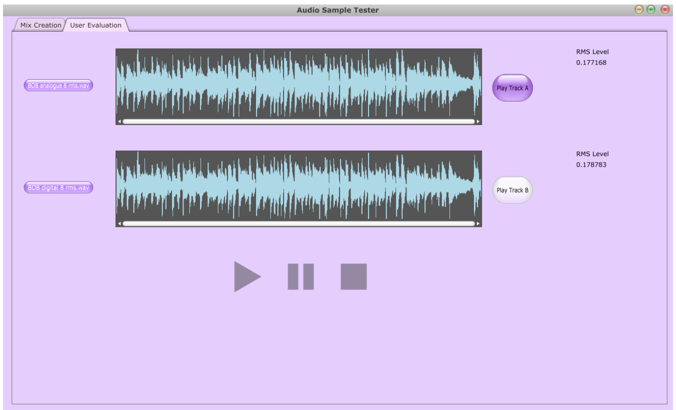

Figure 6: User Evaluation tab
of Audio Mix Analyzer

5. AURAL FEEDBACK

Before the Audio Mix Analyzer could be used to facilitate any listening tests, audio sample tracks needed to be recorded. It was decided that songs of various genres would be used for the listening tests. All audio samples were summed using both analogue and digital devices to allow for a direct comparison. A Mackie 24•8 [11] analogue mixing console was used for the analogue summing and the DAW chosen for digital summing was Cubase Artist 6.5 [2] created by Steinberg. The converter used was the RME Fireface UFX [3]. The "-3dB @ center" [10, p. 2] panning law is used by the Mackie desk and therefore the Cubase and RME panning laws were set to the same panning law manually to guarantee that this did not affect the recordings. No additional panning or gain adjustments were used in this process.

5.1 Multitrack recordings

The audio sample tracks were recorded with the listening tests in mind. It was determined that contemporary songs of different genres should be used. This would establish whether the genre of a song influenced the sound of the summing. Three of the more popular broad genre classifications were picked to represent the diverse nature of contemporary music. These classifications were 'Country', 'Rock' and 'RnB/Electronic'. Eight WAV tracks for each song were needed to be summed and recorded. The *'Mixing Secrets' Free Multitrack Download Library* contains multiple 'raw multitrack projects' [14] of uncompressed and unedited WAV tracks. From this library three songs that could be classified by the indicated genres were chosen. These songs were

- 'Black Out Betty' by Banned From The Zoo [1] (Pop/Rock)
- 'Catching Up' by Lyndsey Ollard (Singer-Songwriter)
- 'Rockshow' by ANiMAL [2] (Hip-Hop/Electronic)

All the tracks were sampled at a rate of 44.1 kHz and had a 24-bit depth. It was not necessary to have a higher sample rate or bit depth as these are the standard settings used in the sound engineering community and the tests were designed to emulate realistic scenarios. The project files for each song consisted of more than eight separate WAV files. The Audio Mix Analyzer was built to sum a maximum of eight tracks and consequently, 30 seconds of eight of the most significant tracks were chosen for each song. An effort was made to select a section of the song where all eight of the tracks produced a signal.

Another aspect that could impact the difference between analogue and digital summing was the number of tracks summed. For this reason each song was summed four times - with the number of tracks shown below:

1. Two tracks
2. Three tracks
3. Five tracks
4. Eight tracks

After all of these tracks had been recorded, they were loaded into the Audio Mix Analyzer and the gain was then adjusted to ensure that the RMS of each track was between -14.94 dB and -15.04 dB.

Once both digital and analogue sums of the three songs were recorded, the Audio Mix Analyzer was used to sum the tracks using the digital mixing algorithms implemented. The panning of the tracks was not altered in any way but the gain was modified until the RMS was in the designated range.

5.2 Aural Testing

Many factors influenced the selection of the samples used in the listening tests. As each test would have to be completed individually there was a limit to the number of samples that could be used. Hence only the most important comparisons could be chosen. In an attempt to restrict the time needed for each test session to under 30 minutes, only six tests per song were selected, resulting in a total of 18 tests. As one goal of the project was to determine whether there is a difference in analogue and digital summing, it was decided that an analogue sample should be compared to a digital sample for every test. For each song a control test was used, where the analogue sum of all eight tracks was used for both samples. This was to establish whether a participant could tell when there was no difference between two samples. For clarity, the six comparisons per song are shown below:

1. Analogue vs. Analogue (8 tracks)
2. Analogue vs. Digital (8 tracks)
3. Analogue vs. Digital (5 tracks)
4. Analogue vs. Digital (2 or 3 tracks)
5. Analogue vs. Simple Addition (8 tracks)
6. Analogue vs. Digital Mixing Algorithm (8 tracks)

This resulted in a total of 18 tests. A 19th test of Simple Addition vs. Digital (8 tracks) was added after the first listening test had taken place. This was due to a suspicion that comparing the digital sum to a sum using the Simple Addition algorithm would produce an interesting result.

An expert panel of 13 participants, between the ages of 18 and 55, was selected to take part in the listening tests.

[1] http://www.bannedfromthezoo.com/
[2] https://soundcloud.com/animalmusicuk

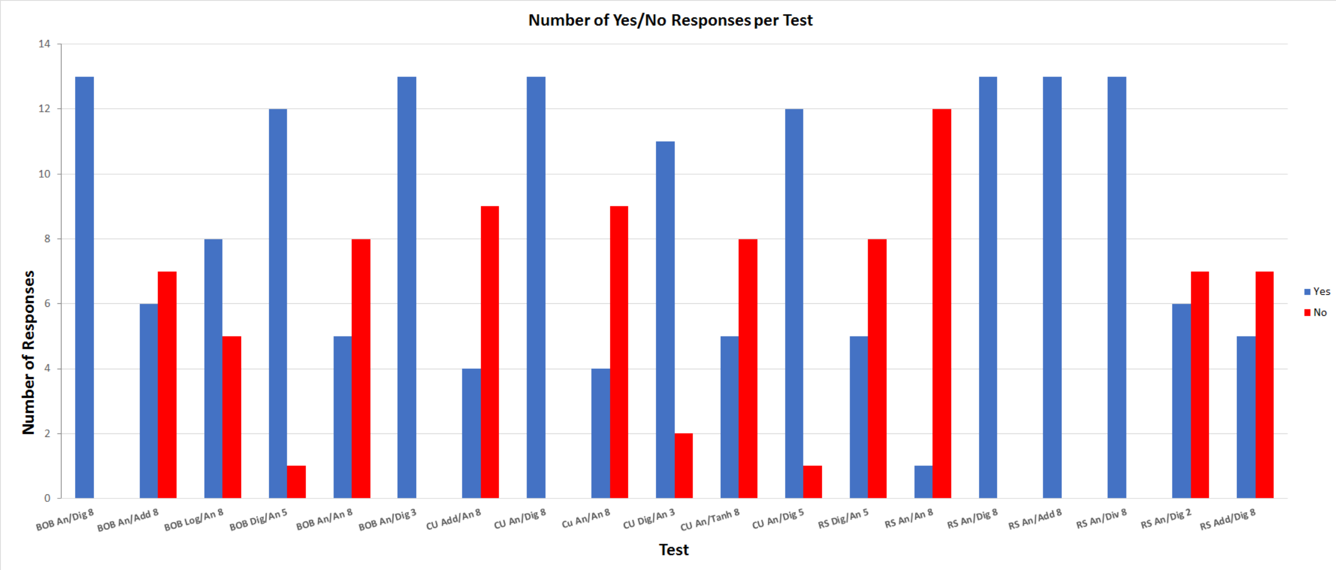

Figure 7: Graph showing Question 1 Responses[3]

Participants were selected based on their music expertise and capabilities, from self taught musicians to experienced sound engineers. Before the listening tests began, participants were told that they would be listening to two audio samples and would be attempting to hear differences between the two. They were also told that sometimes the two samples would be identical and therefore there would be no difference for those tests. However, they were not told how many tests would have identical samples.

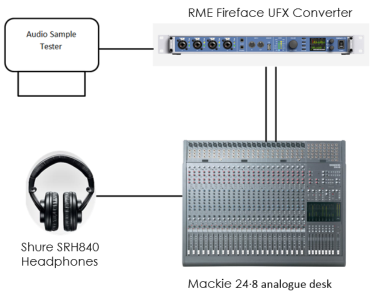

Figure 8: Listening Test Equipment Setup

An effort was made to guarantee that the setup was identical for each participant. Audio was played out from the Audio Mix Analyzer to the Mackie desk via the RME Converter. Participants listened to the tracks through Shure SRH840 headphones [15], which were connected to outputs on the Mackie desk. An illustration of the setup is shown in Fig. 8. The same audio level was used throughout all of the listening tests to prevent differences in perception of frequencies (see Section 2.1).

During the testing, each participant was asked to answer four questions on the response sheet for each of the 19 tests. The main objectives of the test were to determine whether differences between tracks could be heard and, if so, how clear that difference was. An opportunity to provide any other feedback about the difference was also needed. For this reason, the following questions were asked:

1. Is there a difference between the samples?
2. How you would describe this difference?
3. How would you rate the clarity of Sample A, from 1 (clear) to 5 (muffled).
4. How would you rate the clarity of Sample B, from 1 (clear) to 5 (muffled).

5.3 Analysis

Observation of Sample Differences

The first result worth noting is the number of differences heard between samples. Fig. 7 shows the number of 'Yes' and 'No' responses to question 1 for every test. The percentage of participants that heard a difference between samples for each of the six comparisons were calculated and can be seen in Table 1.

These figures indicate some interesting findings. First, every participant heard a difference between the analogue and digital eight-track samples for all three songs. In fact, over 70% of participants could hear a difference between all of the analogue and digital samples, for any number of tracks. Only two-thirds of participants heard a difference between the analogue sample and one of the digital mixing algorithms, which would imply that these are slightly better summing algorithms than the Cubase summing algorithm. However, although the same track was used in the Analogue vs. Analogue comparisons 25.64% of participants still heard a difference.

Difference in Clarity

Clearly, participants could hear differences between many of the samples but these results cannot reveal which of the two samples sounds clearer. A graph showing the average difference in clarity rating between the two samples for each algorithm was generated. This is shown in Fig. 9. This graph shows a very interesting phenomenon when observing the values for each song. As previously stated, the clarity values were evaluated by subtracting the digital rating from the analogue rating. A positive value, therefore, means that

[3] Tests are labeled using the format [Song Name] [Sample A/Sample B] [Number of Tracks]

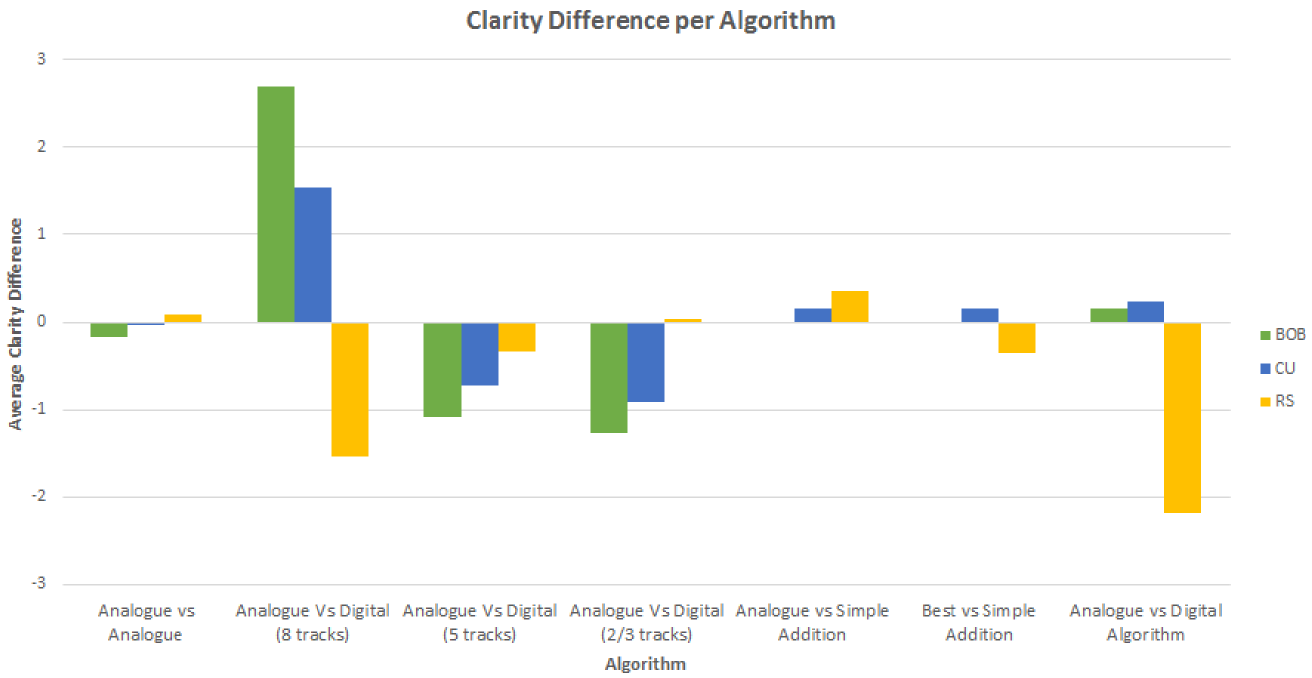

Figure 9: Graph showing Clarity Difference per Algorithm

Table 1: Percentage of Participants Hearing a Difference for each Algorithm

Tracks	Test	Yes
8	Analogue vs Analogue	25.64%
8	Analogue vs Digital	100.00%
5	Analogue vs Digital	74.36%
2 or 3	Analogue vs Digital	76.92%
8	Analogue vs Other	66.67%
8	Best vs Simple Addition	39.47%

the participants thought that the analogue sample sounded clearer, while a negative value implies the digital sample was clearer.

$$Clarity = (DigitalRating - AnalogueRating)/Responses \tag{6}$$

Figure 9 consequently seems to suggest that the analogue summed tracks sounded much clearer for 'Black Out Betty' and 'Catching Up', the Pop/Rock and Singer-Songwriter songs, while the digital sum was clearer for 'Rockshow', the Hip-Hop/Electronic song. This would imply that the best summing method is highly dependent on the genre of the song.

The graph also illustrates that, although about $\frac{1}{4}$ of participants heard a difference between the identical analogue samples, they could not hear a big difference in clarity. This helps establish the validity of this set of listening tests.

Another interesting observation is the impact that the number of tracks has on the clarity. The digital sum was perceived to be more clear when only five tracks were summed, irrespective of the song. This was also the case for when three tracks were summed. The comparison of two summed tracks from 'Rockshow' seems to be an exception to this pattern, as participants noticed the analogue sum to be slightly clearer than the digital sum. Overall, the number of tracks definitely appears to affect the clarity of the digitally summed tracks.

5.4 Testing Limitations

In hindsight the listening test process definitely had some limitations. Although every effort was made to ensure that all tests were identical, the nature of individual testing makes it impossible to guarantee that external factors did not affect the listening tests.

Many participants seemed to get frustrated at having to repeatedly listen to the same song and noted that they were not sure if the difference they were hearing actually existed or if it was psychological. Changing the configuration of the tests every time could have eliminated the influence of the order in which comparisons were asked on the results. It is also important to note that listening tests are inherently subjective and the difference between audio samples can be difficult to quantify. Therefore the interpretation of the term 'clarity' was left up to each participant.

Due to time constraints a bigger sample size of participants was not possible. Human error also plays a significant role in the outcome of the tests, both with the participant's ability to distinguish sounds and the recording of the results.

6. CONCLUSION

During the listening tests all participants noticed a difference between the analogue and digital summing of eight audio tracks, with the clearer summing method being genre dependent. About $\frac{3}{4}$ of participants noticed a difference between analogue and digital summing of two, three and five tracks. Depending on the genre, the method that sounded clearer for eight tracks was often reversed when fewer tracks were summed. This proves that the number of tracks being summed influenced the clarity.

Various mixing algorithms were implemented in the Audio Mix Analyzer and, after testing, it was found that about 60% of the participants could not hear a difference between the Simple Addition algorithm and analogue summing. These results, which corroborated the study of the open source DAWs and the experiments with an analogue mixing console, all indicate that simply adding the samples together is the summing method most similar to the analogue summing. However, from the results of the listening tests, it appears that Cubase, the closed-source DAW that was used to digitally sum audio tracks, is not using this approach.

The Audio Mix Analyzer interface enabled both the efficient creation of digitally-summed audio tracks and allowed users to easily compare two audio files during the listening tests.

6.1 Future Work

It is clear that much more work in the field of audio summing could be performed. The investigation into the effect of panning was beyond the scope of this project. This topic could be explored further to fully explain the variation in panning that can be heard when different summing algorithms are used.

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