Practices for Mapping a Massive Multi-Channel Composition Across a Massive Speaker Array

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ABSTRACT

Composers now have at their disposal the tools to create massive multi-channel audio mixes that can better represent a large array of voices by giving each voice its own single or multiple channels in a large array of loudspeakers to better spatialize their sound sources.

There are some issues that need to be resolved, like balancing many voices across a large array of loudspeakers spread out much farther apart than a large ensemble or orchestra and the different frequency responses of different sizes and types of loudspeakers.

I will demonstrate the work-flow I have discovered to map a massive multi-channel composition over an array of 40 channels in a digitally controlled theater and how to solve the above mentioned problems when dealing with a massive array of audio channels and loudspeakers.

1. INTRODUCTION

According to Malham [3], The ability to control the spatial aspects of audio has been pursued by composers since the emergence of remote sound transmission of the late 19th century. Prior to then, most musical events were fixed in location according to where they were performed. When this barrier was broken with the ability to transmit music, audio engineers were tasked with the problem of restoring the spatial characteristics of sound after they were transmitted. With these challenges observed and overcame, composers were met with new opportunities to explore and utilize this âĂIJnew worldâĂİ of musical possibilities. Since then, the constant exploration by composers, largely electronic composers, has constantly pushed the boundaries of spatial sound requiring engineers to constantly produce new, higher quality, and revolutionary technology to aid these composers in their journey in spatializing sound and music.

Even with all this progress, Malham argues that we composers still have a long way to go before we will achieve a synthesized sound-scape that accurately emulates a natural sound-scape.

With very few examples out in the world of techniques for diffusing a massive multi-channel composition across a massive speaker array, this paper with discuss the methods and techniques that I have discovered through my research into massive multi-channel audio to achieve such a sound-scape, the issues that I discovered, and how I resolved those issues.

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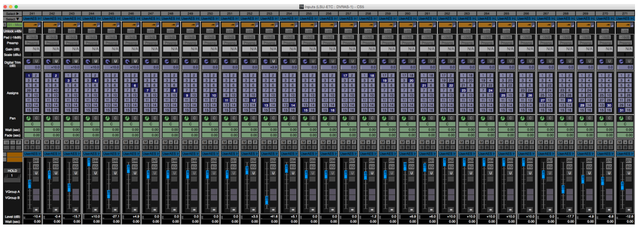

Figure 1: CueStation software.

2. METHODS AND MATERIALS

For my research into mapping massive multi-channel audio I used Apple's Logic Pro X DAW software to compose and output the audio to a secondary software, CueStation, which controls the LSU Digital Media Center Theater's 92 speaker array. CueStation acts as a live mixing environment that can be used for input and output assignment from external sources as well as save and recall user-made cues. Any DAW software capable of multi-channel output can be used in this process. Figures 1 and 2 show CueStation and Logic respectively. Figure 3 shows the matrix in CueStation. The left side of the matrix corresponds to the bus that each track is routed to; in this case tracks 1-40 from Logic were routed to buses 1-40 in CueStation. The numbers at the top of the matrix correspond to each speaker in the array in the theater.

Each of the 40 tracks of the composition in Logic was routed to its' own individual audio output channel corresponding to its track number; 1 through 40. With the Dante Virtual Soundcard chosen, the computer is able to send 40 channels of audio to CueStation.

CueStation then receives each individual channel and is represented as a track similar to a DAW software.

Each track can then be routed to the matrix, which can be used to map each track to a specific speaker and can also control the gain for that matrix channel. Using this routing method, I assigned each track to its' own individual speaker and adjusted the gain as needed to balance volumes across the theater's speaker array. Some speakers models are inherently louder than others and at different angles in reference to the listener's ear even sitting in the middle of the theater.

This means that some of my sound material, which were exact duplicates or at the same loudness, came out of different speakers at unbalanced volumes that needed to be adjusted accordingly. What I discovered was that simply routing a final mix out of Logic did not guarantee a balanced mix in CueStation through the matrix and out of the

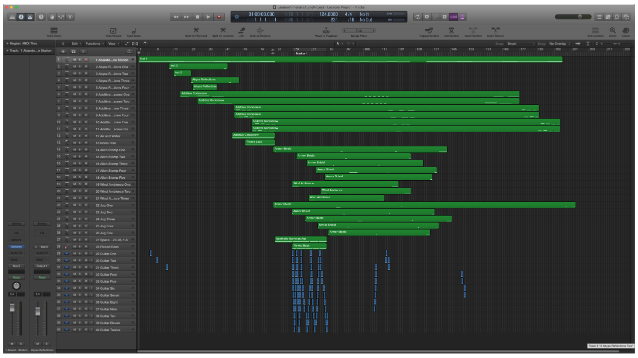

Figure 2: Logic Pro X DAW.

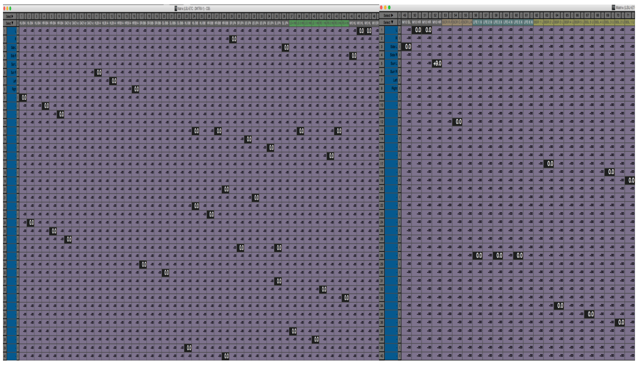

Figure 3: Matrix in CueStation.

speakers.

3. RESULTS

Figure 4 demonstrates the above mentioned issue of the final mix out of Logic not being balanced in CueStation; it shows a section of the composition that has the same guitar sample duplicated 11 times and mapped across 11 speakers; each sample being offset in time to create a motivic, rhythmic structure. In Logic, this section was perfectly balanced with each track at the same loudness, at unity gain, but after the journey to CueStation and mapping across the wide array of 11 speakers, drastic loudness changes were necessary on the CueStation faders to regain that balance lost from the final mix in Logic. In the first half of figure 4, on the left side, all of the faders are at unity gain.

To balance the mix, channel 29's loudness was changed to -4.9, channel 30's to -6.8, channel 31's to -12.6, channel 32's to -22.4, channel 33's to -16.1, channel 34's to -23.4, channel 35's to -27.7, channel 36's to -24.4, channel 37's to -19.1, channel 38's to -20.9,channel 39's to -4.0, and channel 40's to -4.6.

3.1 Discussion

As shown by the results of the final loudness of this guitar section, focused care was taken to balance these volumes. The difference between every track's loudness, excluding channels 29 and 40, is too large to be due to human error. This is why in future research, I want to measure the angle of each of these speaker's horn in relation to a listener's ear in the center of the theater (center of speaker array) to calculate whether or not a mathematical correlation can be discovered that accounts for and demonstrates the drastic change in loudness across all 11 speakers.

These possible measurements, like the ones discussed in

Figure 4: Unbalanced and balanced loudness changes.

[4], could then theoretically be used to calculate the specific loudness change needed to balance an unbalanced section out of Logic of future compositions to accurately capture the original balanced mix.

4. CONCLUSIONS

Using a DAW software to output a massive multi-channel composition to a massive speaker array was the most efficient technique that I could find for my research into mapping a massive multi-channel, of which used a total of 50 speakers. All issues that I encountered with this research were resolved and I believe more research into the phenomena of unbalanced output mixes from the DAW could provide techniques to counteract said phenomena in a mathematically accurate way if I were to discover a mathematical correlation between the change in angles in speakers in relation to a listeners ear to the change in loudness experienced after the output from the DAW.

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[5], [1], [2], [4], [3].