Creating Music with Software:
Digital Audio Workstations

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Abstract—**Computer software used to be absent in the music industry but has now become a necessity. Practically all music being produced today has been influenced in some way by computer software that has acted upon original recorded material. With the ease and affordability of using a digital audio workstation (DAW), most anybody can create music that is of substantial quality. This aspect is what gives DAW software a very necessary and dominating role in the music industry today, especially for emerging artists.**

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I. INTRODUCTION

A DAW is the most powerful tool for creating music today. Its ease of use and versatility have revolutionized the manner in which music is created and have even allowed for new genres of music to present themselves. To begin to understand the importance of DAWs and why the functionality of its software is so revolutionary, certain subjects must be addressed. The first subject is a survey of the background information concerning DAWs. The next subject is an analysis of how DAWs are to be used and how they function. The last subject is an explanation of how virtual studio technologies (VSTs) are the future of DAWs and how they will grant DAW software further functionality. After gaining the vital information from each subject, the reasons for which

DAW software is now the staple of music creation will become clear.

II. SURVEY

Before DAW technology emerged, creating music was not a task that just anybody could do. Analog tape machines were the industry standard and they were producing good sounding music. The drawbacks of these machines, however, were that editing was a tedious process and the cost of the equipment was not affordable for the average person. At that point in time (1980's), computers were gaining increased processing power. And because this was the emergence of the digital age and computers were at our disposal, there had to be a better way to create music that utilized the power of the computer. Such was the idea that was responsible for the creation of DAWs.

A. Prior to DAWs

The main way that music was created before DAWs were present was by means of reel to reel tape recorders and mixing tables. Figure 1 [2] shows a reel to reel tape recorder and Figure 2 [4] shows a mixing table.

Fig. 1. Reel to Reel Tape Recorder: A reel to reel tape recorder was, years ago, state-of-the-art technology used for creating music.

The tape recorders utilized magnetic tape as the medium for carrying the sound. For those unfamiliar with what magnetic tape is, it's the film-like material found in audio cassettes. When recording, the "sound" would be put onto the magnetic tape by the tape head. The tape head is a very small electromagnet that applies a magnetic flux to the tape making the oxide on the tape magnetized and oriented in a manner proportional to the original sound signal. Figure 3 [3] depicts a tape head.

For playing music back, the process is somewhat of a reversed order. The tape runs across the tape head,

Fig. 2. Mixing Table: A mixing table was necessary for adding effects to recorded material.
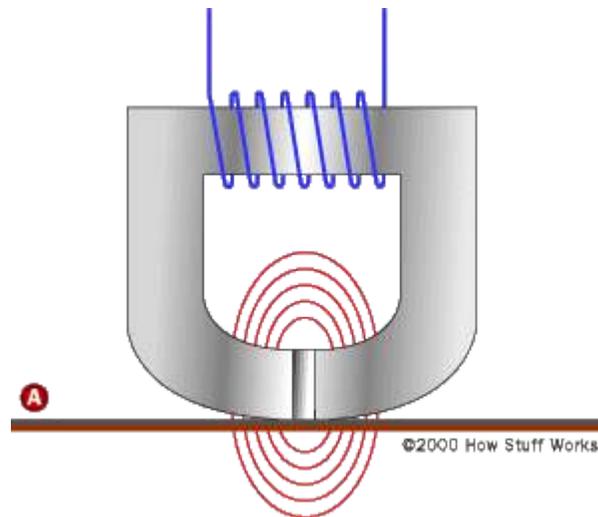
Fig. 3. Tape Head: The tape head receives the sound signal from a wire coil connected to the electromagnet. The electricity running through the wire coil creates a magnetic field in the iron core of the electromagnet and the gap in the electromagnet is where the flux (red lines) builds up. That flux is what magnetizes the tape running across the tape head [1].

creating a varying magnetic field in the core. The varying magnetic field in the core produces an electrical signal to the wire coil. This signal can then be amplified and played through speakers. When editing music with a tape recorder, pieces can literally be cut and attached together to combine certain parts of a song. The problem with editing magnetic tape, however, is that you need copies of the tape. This is because if you tried to edit the original tape recording and made a mistake, you would have no way to recover from your mistake. You would have to record the instruments and/or vocalist(s) again which would cost time and money. The problem with making copies of tape recordings, though, is that when a copy is made, it does not have the same sound quality as the original recording. This is because the signals being received from the tape that the music was recorded onto

are not as pure as the signals being sent directly from the instruments to the tape. How mixing tables fit in to the editing of the tape recordings is that they slightly alter the magnetic orientations on the tape to produce different sounds. This is how different effects such as echoes and instrument/voice sound distortions were achieved. The massive array of dials, knobs, and buttons on the mixing tables are what controlled the effects. Typically, when a recording was having effects put on it, the effects would be selected on the mixing table and then the portion of tape to be edited would run while the mixing table would be sending its altering signals to the tape head. If a mistake was made, the original music could be copied back onto the affected area given that a copy of the original recording was made. Doing so made it seem like a mistake was never made to the tape which allowed editors to readjust their effect selections and try again to make the effect sound just right. This was how music was typically made before DAWs came along and at the time (1980's), this was seen as the most efficient way to create music.

B. The Need for New Technology

Although the analog technology was getting the job done at the time, there had to be a better way to create music. The previous section clearly shows how editing using analog technology was a tedious task. Aside from that, the analog equipment required to create music was extremely expensive. A commercial tape machine could cost about $15,000 whereas a commercial mixing table could cost tens or even hundreds of thousands of dollars depending on size [12]. Also, depending on how much tape is used, creating music was very costly given that a reel of tape that yields 30 minutes of recording time was around $200 [13]. The average person could not afford such equipment. The best an average person could get was home studio equipment which was often very limited in functionality as compared to the commercial grade equipment; tape recorders were not as reliable and didn't capture the sound as well whereas mixers were laptop sized instead of table sized and didn't provide nearly as many effects or as much effect versatility. The commercial companies required technology that would enable them to create music more quickly and have more effects. The average people required affordable technology that could provide them with the means of creating studio quality music.

C. Improving Computer Technology

Improvements in computer processor power are what enabled the development of DAWs. Sound editing software relies heavily on the CPU to give it instant feedback, especially for recording because it is in real time [24]. Another technology that led to the development of DAWs was pulse code modulation (PCM) recording. An analog to digital converter would utilize PCM technology to turn analog sound signals into digital sound signals which was how sounds could be recorded digitally. Prototypes of DAWs started in the mid 1980's but processors weren't powerful enough for full versions until the late 1980's [11]. Computer processing power is ultimately what enabled the emergence of DAWs. People using analog technology had wanted faster ways to produce music but the technology was not yet present.

D. Introduction of the DAW

The first modern DAW was Steinberg Cubase (originally called Cubit) introduced by Steinberg in April of 1989. The reason it is considered the first modern DAW is because it was among the first digital sound editors to utilize musical instrument digital interface (MIDI) recording, but most importantly, it was the first to graphically implement a vertical list of tracks and a horizontal timeline. This graphical setup was much more intuitive and thus allowed for much easier editing. The graphical setup of Cubase has since been imitated by practically every DAW made since [24]. A picture of the original graphical set up is shown in Figure 4 [7].

Fig. 4. Steinberg Cubit: You can see the vertical list of instruments on the left section of the screen and from the middle to right sections of the screen you can see the timeline. Places where there are rectangles in the timeline are where the associated instruments are actually present in the recording.

Cubase was originally released for Atari ST computers but the following year Cubase was released for Apple Macintosh computers. Cubase was not available for Windows computers until 1992. Another DAW from around that time that is significant is Pro Tools which came out for Windows in 1991 [24]. A picture of the original version of Pro Tools is shown in Figure 5 [5].

When Pro Tools was released, many major recording studios "went digital". Pro Tools initially cost $6,000. Two years later Pro Tools only cost $400 [25]. This

Fig. 5. Pro Tools: Despite the waveforms, you can see how the Pro Tools setup is close to Cubase's setup. The instruments are still in a vertical list whereas the horizontal timeline still shows rectangles which are where the associated instrument is present in the recording.

satisfied the need of average people to be able to afford commercial recording technology. Although not as graphically pleasing or intricate as DAWs today, the first versions of Cubase and Pro Tools were the original major DAWs to be produced.

E. New Status Quo

With DAWs being utilized by major recording studios instead of analog tape machines, the industry of music making was now in the digital age. The digital age is the most advanced age humanity has been in technology-wise. A by-product of this is that creating and editing music can be done faster than ever before. While many at the time were still meddling with the tediousness of analog equipment, many more were making the change from analog to digital technology. And since the costs of making the change were very affordable, this made DAW software even more enticing. DAW software was then, and still is now, the newest and most advanced way to create music.

III. FUNCTIONALITY OF A DAW

A DAW is the most useful tool for creating music today. Unfortunately, the processes that must be made in order to create music with one can be ambiguous. An analysis of the software itself, however, will provide the knowledge to turn ambiguities of the subject into understanding. How a DAW actually works and the algorithms it utilizes can be understood by looking at its three main functions: recording, editing, and adding effects. Such

aspects are fundamental to the actual creation of music. DAWs were created to simplify the process of creating music which is why the three aspects themselves have been established as the basis for how DAWs function. Methods to carry out the process of utilizing DAW software efficiently will be posed along with how the three main functions are to be used.

A. Recording

The first step in creating music with a DAW is to first obtain a recording of vocal content, instrumental content, or both. This requires hardware that will be able to capture sound signals emitted from a voice or instrument and turn them into electrical signals. Such hardware is often in the form of a microphone or a digital instrument such as a keyboard that can directly send its electrical signals through an output cable. In order to record with a DAW, a track must be established and a signal input type must be selected so that the DAW software will be able to recognize where it can acquire the electrical signals from to be put into the track. A track is a structure that incoming sound signals are recorded onto. Figure 6 [26] shows an image of the REAPER DAW [26] created by Cockos that is showing a list of sound signal input devices. (Note that the REAPER DAW is the DAW that will be referenced throughout this section unless otherwise specified. Also note that other DAWs will function in a very similar manner.)
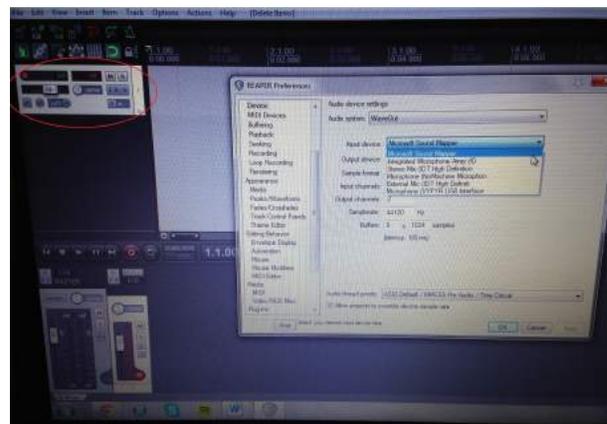
Fig. 6. REAPER Input Device Window: The track object is the gray box circled in red which has a few controls in it and is located to the left of the upper left corner of the input device window. Selected in blue is the Microsoft Sound Mapper input device. This is an internal microphone that is found in many laptops.

For a track to be ready for recording, the track must be "armed". This means that the arm/disarm switch must be toggled "on" to make a track available to be recorded on. This is useful because it allows for one track to be recorded onto at a time. If a track was armed and the

record button was pressed, any sound that the laptop's microphone picks up will be recorded onto that track. Figure 7 [26] shows an image of my voice recorded onto a track.

Fig. 7. REAPER Voice Track: Notice the wave that is present. This wave is a graphical representation of my voice saying "Hello". The units of the graph are amplitude over time. Amplitude is in decibels, which is what the human ear interprets as loudness. Time is in seconds.

One may wonder how the DAW actually records the sound and creates such a graph. The DAW does so by means of an analog to digital converter (ADC) utilizing PCM, which was, in this case, my laptop's microphone. A voice creates a change in air pressure and an ADC senses that change in air pressure and outputs electrical signals accordingly [14]. The electrical signals produced are then received by the DAW and the DAW interprets the incoming signals and puts the data onto the graph. A recording is merely a collection of sound samples. What if two things were being recorded at the same time? What would the graph look like? Figure 8 [26] shows an image of a recording where I said "Hello" while a guitar chord was being played.
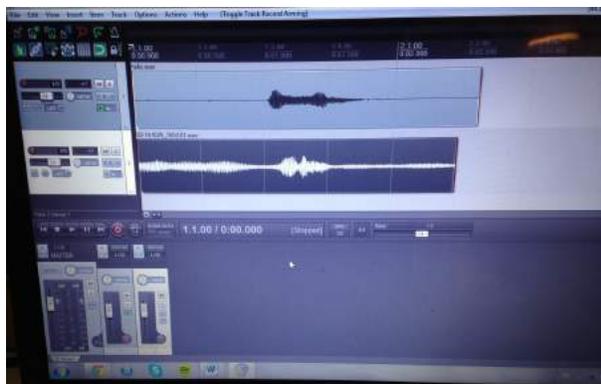

Fig. 8. REAPER Voice and Voice Over Guitar Tracks: The graph on the bottom is the one where I said "Hello" while the guitar chord was played. Notice that the moment where I say "Hello" is more distorted when the guitar chord is being played as opposed to the "Hello" graph on top. This results in a loss of sound clarity.

Because the microphone cannot get every sample from my voice and the guitar at the exact same time, it must sample the sound combination of my voice and the guitar. This is why it is desirable to have multiple tracks that each have one voice or instrument being recorded on them; there is better sound clarity. If I created a track where only the guitar chord was playing and played that track along with the original track that I said "Hello" in, my voice should sound much clearer. Figure 9 [26] shows a picture of my voice track saying "Hello", and another track where only the guitar chord is being played.
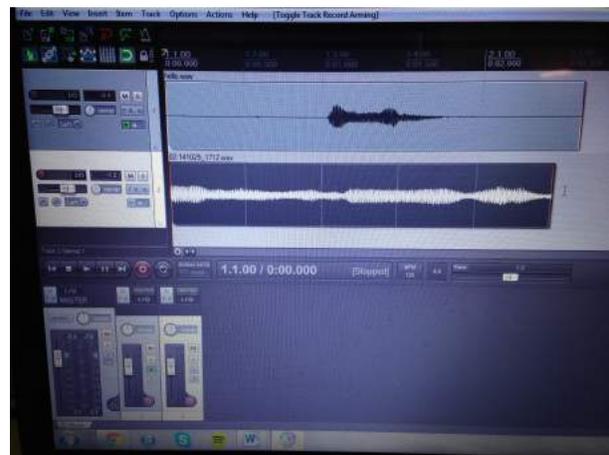

Fig. 9. REAPER Voice and Guitar Tracks: It is easy to tell that my voice is much clearer when both of these tracks are played at the same time. This method of using individual tracks for each voice or instrument is the ideal way to record music on DAWs because it provides the best clarity.

For professional music making, there would obviously be much better equipment used for recording than a laptop microphone but the recording process itself is still the same. Clarity is key when recording in a professional setting. Typically, recording takes place in a soundproof recording studio. But since not everybody has access to a soundproof room to record in, unwanted noise can sometimes occur. But instead of having to re-record something in order to produce a recording without unwanted noise, a very useful feature of a DAW comes into play: editing.

B. Editing

Editing is the process of "fixing" a recording. Editing is done to a recording before adding effects so that the recording itself can be as perfect as possible when the effects are to be applied. One of the most common edits made in music, especially for vocalists, is a pitch correcting edit. If a person who is not a particularly skilled vocalist were to sing a song, there would be some difference between the pitch of their voice and

the correct pitch that the song is to be sung at. To adjust pitch, a pitch editor must be created for a track. A pitch editor shows a graphed line which is a representation of pitch over time. In Figure 10 [26] below, there is a pitch editor, ReaTune(Cockos) [26], that is showing the pitch (red line) of my voice in the recording where I said "Hello".
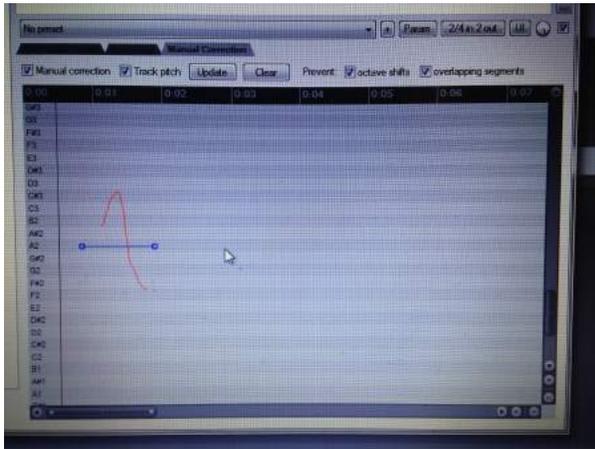

Fig. 10. REAPER Pitch Correct Window: Notice the fluctuation in pitch of the red line. If I wanted to reduce the fluctuation in order to have the word "Hello" centered around a particular pitch, I would have to add a pitch line (blue) and then run the recording again in order for the pitch line to make its effect.

Figure 10 shows the pitch line but it has not yet taken effect. Figure 11 [26] shows the pitch graph where the pitch line has taken its effect. Notice the differences.

This function of correcting pitch is not only limited to vocals. Any instrument's recording can be pitch corrected. The pitch correct edit does the exact same process for any recording. A pitch line must be set for a certain pitch and the DAW software will run an algorithm that will change the original pitch values of a recording to have values that are centered close to the pitch line.

Another widely used edit is the copy and paste edit. Songs tend to have repeated sections in them such as a chorus. Instead of having to record a whole song where all the choruses are played, one chorus track can be recorded and then be copied and pasted where needed. This allows for the choruses or other sections to all sound exactly the same which is ideal depending on the song trying to be created. Figure 12 [26] shows both tracks from Figure 9 having been copied and then pasted.

It can be assumed that the DAW software algorithm for this edit merely had to copy all of the graph values and then start adding them to the end of where the track initially ended in order to create a duplicate track or section of a track. Knowing this, one can assume the DAW uses a for loop to accomplish this task structured in this manner:
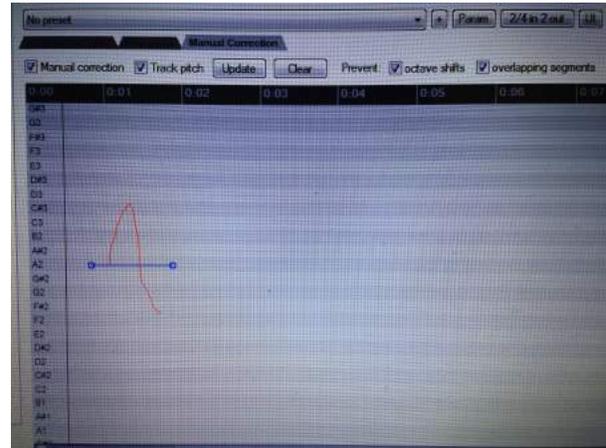
Fig. 11. REAPER Pitch Correct Window 2: The pitch line was set at an A2 pitch and after the pitch line made changes to the red line, the red line now starts at an A2 pitch. This means that my "Hello" recording sounds different than it originally did. Because of the blue pitch line being set at the pitch of A2, my "Hello" recording's pitch is now centered at A2. This means that if an instrument were to play an A note or A chord such as the guitar chord from the track shown in Figure 9, the word "Hello" would sound good when the guitar chord is played because they are at relatively the same pitch.
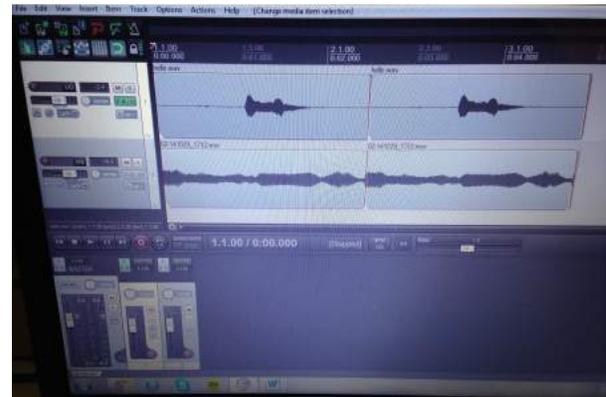
Fig. 12. REAPER Copied Tracks: When both tracks are played together, "Hello" is said twice while the guitar chord gets played twice.

```
for    (minIndex; minIndex < maxIndex; minIndex++)
{
        getSampleValue[minIndex];
        setSampleValue[minIndex + offset];
}
```

"minIndex" would be the starting index and the sound samples at that index noted by "getSampleValue[minIndex]" would then be set at an index that accounts for where the sound is to be "pasted" using "setSampleValue[minIndex + offset]". The "offset" accounts for the amount of time between

the section that was copied and the section that was "pasted". The copy and paste edit saves a lot of time when creating songs that have parts that are repeated multiple times. Time spent creating a song can essentially be cut in half by utilizing the copy and paste edit because roughly only half of the material in the song needed to be recorded.

Another invaluable edit is clipping and it allows for a track or section of a track to be "clipped" so that the particular track or section of a track can have the proper time length which means that it can start and end at the right time or on the right beat. If a track recording has unnecessary excess material on it that makes it too long or not end on the correct beat, all that must be done is the edge of the track recording section must be selected and dragged to the correct spot. There is some unnecessary excess material in the track where "Hello" is said twice. Because the track is a vocal track and nothing is being said at certain points in time, it does not make sense to have the "white noise" be a part of the track. Figure 13 [26] shows the parts of the vocal track having been clipped.

Fig. 13. REAPER Clipped Vocal Tracks: This provides for better sound clarity because although there was not really any background noise on the voice track, there is still sound being recorded.

Although it sometimes cannot even be heard, it is always best to get rid of any foreign sound so that the recording can be as clear and close to perfect as possible. The algorithm used when clipping is simply deleting sound samples. Assume a for loop to be used that has this structure:

```
for    (minIndex; minIndex < maxIndex; minIndex++)
{
    deleteSampleValue[minIndex];
}
```

The algorithm is deleting sound samples using "deleteSampleValue[minIndex]" for sample values that are in between "minIndex" and "maxIndex".

Keeping the beat exact is another way to make music close to perfect. The sliding edit can help achieve keeping everything right on the beat. If an instrument was playing but was consistently ahead or behind the beat, all that would need to be done is the track that it was recorded onto would have to be slid over to where it would be on the beat. If I wanted my "Hello" track to start exactly when the guitar chord is played, all that needs to be done is the "Hello" track must be slid over. Such is shown in Figure 14 [26].

Fig. 14. REAPER Slid Over Vocal Tracks: The two separate parts where "Hello" is said as seen in Figure 13 have merely been shifted over so that they start at the same time that the guitar chord starts.

An algorithm for such could be comprised as a for loop like this:

```
for    (minIndex; minIndex < maxIndex; minIndex++)
{
    getSampleValue[minIndex];
    deleteSampleValue[minIndex];
    setSampleValue[minIndex + offset];
}
```

This algorithm shifts the content by getting sound samples in between "minIndex" and "maxIndex", deleting the original sound samples with "deleteSampleValue[minIndex]", and then finally putting the sound samples in the correct spot with "setSampleValue[minIndex + offset]". "offset" is the variable that accounts for sound samples being set to the correct index which corresponds to being set to the correct time.

A problem that would require a more tedious process to solve would be if an instrument started off-beat, but then got back on-beat. In order to fix this problem, clipping and sliding would still be used, but splitting

must also be used. Splitting separates a track or section of a track so that the pieces being split can be treated as separate sections, i.e. they can be slid around and clipped independently from one another. This means that if a singer were to initially start singing before the beat but then started singing with the beat, a section or sections would need to be split, then clipped, and then slid together. It's almost like putting a puzzle together to make everything fit just right. In certain cases however, such as a foreign noise being present in the recording, the same process of splitting, clipping, and sliding is used, but in a different manner. Copying must also be included in such a process. Say, for example, a guitar chord was being played and you happened to sneeze and the recording captured your sneeze. In order to fix this, the section of the track would have to be split at the beginning of the sneeze noise and at the end of it. The sneeze section must then be deleted. In order to fill this now empty space, a section of the uninterrupted guitar chord being played must be copied. Then the copied section must be pasted into the gap area where the sneeze sound used to be and must then be slid over if needed so that there is no longer any gap in the recorded material. The recording would now be very close to being completely imperfection free.

The edits described are among the most common and useful edits that occur when creating music. Such edits allow the recordings to be as imperfection free as possible. Complete perfection is essentially impossible but the idea is to use editing to get as close to perfect as possible. When a recording is finished being edited, that would be the best time to add effects to the recording to really enhance the sound thus making a more professional sounding recording.

C. Effects

Effects are to music as toppings and condiments are to food. A hamburger with just a beef patty between buns can taste good, but if cheese, lettuce, pickles, ketchup, etc. were added, the taste of the burger would be greatly enhanced. The same is for effects. Although they're not necessary, they can greatly enhance the sound of an instrument, voice of a vocalist, or any sound general. Some of the most common effects used in music are delay and reverberation, or "reverb" for short [15].

Delay is like an echo in that whatever noise is under the influence of a delay, that noise will be repeated. The time rate at which the delayed sound can repeat often ranges from a few milliseconds to several seconds. In many songs, you may notice that a vocalist's voice seems to echo a little. This is not because they are in a big room that would promote echoing, it is because there is a delay effect that has been added to their recorded voice. If delay were added to my voice track recording, the "Hello"

would be repeated with decreasing volume each repeat which is how an echo-like sound would be achieved. Figure 15 [26] shows the ReaDelay(Cockos) [26] sound effect settings window of the "Hello" voice track.
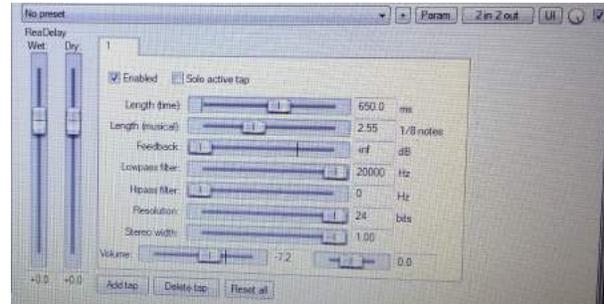

Fig. 15. REAPER Delay Window: If you look, you will see that there are time and musical length values. These values have been specifically set so that "Hello" will echo to the beat. Also notice that the volume has been decreased (it's a negative value). This means that the delay effect on "Hello" will simulate an echo because there are repeats with decreasing volume.

Surely, the DAW achieves this effect by collecting the sound samples in the range of "Hello", and then replaying the sound samples at the specified times that fit the beat all while decreasing the amplitude of the sound samples to accommodate for the decrease in volume [23]. Assume that this is achieved by means of a for loop containing an if statement as such:

```
for  (minIndex; minIndex < maxIndex; minIndex++) {
    if (getAmplitude(minIndex) > minSoundAmplitude)
        break;
    setSampleValue(minIndex + offset,
    getAmplitude[minIndex] * df);
}
```

"minSoundAmplitude" is a variable that decides when the loop is to stop given that "getAmplitude(minIndex" becomes less than "minSoundAmplitude" before "maxIndex" is reached. The "setSampleValue(minIndex + offset, getAmplitude[minIndex] * df)" determines the index where sound is to be set by means of the "offset" in its first parameter. For its second parameter, the amplitude is set but it is dampened by a dampening factor "df" value that is less than one. This is what allows for each repeat to decrease in volume. Reverb is widely used in combination with delay. This is because when the two are used in unison, a vocalist's voice or an instrument's sound will have a "full" sounding tone that also has "depth". The produced tone is audibly pleasing and thus desirable, which is why reverb and delay put together are widely used. Delay accounts more for the "depth" in tone whereas reverb accounts more for the "fullness" of tone. Reverb is similar to delay in that it

creates an echo like sound, but it does not produce a true echo sound like delay does because reverb does not repeat the sound. Reverb provides a more "dying down" effect to sound. Its effect is comparable to letting a guitar chord "ring out". If a guitar chord were to be played, initially the chord would be loud. But after letting the chord continue to "ring out", the volume of the chord would decrease as time went on until it eventually could no longer be heard. Such is the effect that reverb creates. Of course, reverb can be adjusted in ways that make the time taken for the sound "dying down" shorter or longer. Times for such are often a few milliseconds to several seconds. Figure 16 [26] shows the ReaVerb(Cockos) [26] sound effects settings window.
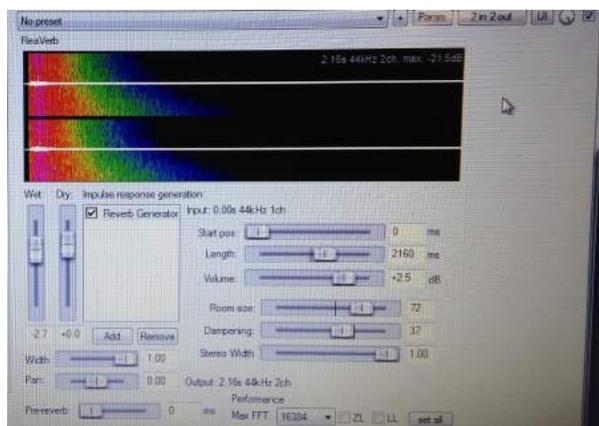

Fig. 16. REAPER Reverb Window: Notice the rainbow colored graph. This is a representation of the reverb. The left to right "spread" of the graph is to do with the length of time it takes for the reverb to become silent. The colors of the graph represent the intensity of the reverb effect with warm colors being more intense and cool colors being less intense. As can be seen, the intensity is initially high and then lessens as time goes on, thus adhering to the properties of reverb. The slope of the graph has to do with damping which decreases the intensity in a parabolic manner rather than linearly.

It can be surmised that the DAW achieves this effect by taking the sound samples towards the end of "Hello" and repeating them over and over while decreasing volume to account for the "dying down" effect. An algorithm for reverb can be assumed to be a for loop containing an if statement as such:

```
for (minIndex; minIndex < maxIndex; minIndex++) {
    if (getAmplitude(minIndex) > minSoundAmplitude)
        break;
    setSampleValue(minIndex,
    getAmplitude(minIndex) * df);
}
```

This for loop for reverb is very similar to the for loop for delay except for an alteration in the first

parameter of the "setSampleValue" method. There is no "offset" variable because reverb applies only to the ending sound samples and there is no need for an "offset" variable. "minIndex" would be an index at the end of the section having reverb applied and "maxIndex" would account for the "spread" or time that it takes for the sound to "die down" under the reverb effect. Just like for delay, in reverb there is an if statement that controls when the for loop is to stop depending on whether or not "getAmplitude[minIndex]" becomes less than "minSoundAmplitude". The dampening factor "df" accounts for the reduction of sample amplitudes thus making the sound "die down" as time goes on. The reason that the sound samples towards the end of the "Hello" would be taken and repeated is because that is what makes the word "Hello" seem like it is "dying down". If all the sound samples of "Hello" were repeated, a delay effect would be achieved instead of reverb. Thus the difference between delay and reverb can now be seen; it's a difference between the sound samples that are to be repeated. The combined effect of the delay and reverb on the "Hello" track sounds better than it did without the effects. The spoken word "Hello" now contains more "fullness" and "depth" and is reasonably more interesting to listen to as opposed to just a plainly spoken "Hello" without any effects.

There are thousands of different effects that can be added to music but describing several of them with little detail would have less of an impact compared to describing a couple of the most used effects in more detail in an effort to keep this paper an acceptable length. The most important aspect to understand about how effects work is that the sound samples of a recording are taken and manipulated by algorithms in a DAW in such a way that creates the type of sound a certain effect is supposed to make. Such has been shown above with delay and reverb; describing the bigger picture concerning effects in general seemed a necessary clarification to "connect the dots".

IV. FUTURE TRENDS

DAWs are currently the most modern tools used for creating music and they are implemented by the most advanced form of technology currently available: computers. As technology advances, so does the process of creating music; this has always been the trend. However, since music production is currently utilizing the highest-order technology available, where does that leave room for advancements in such production? Given that no more advanced form of technology than computers presents itself, it is proposed that the actual platform that is a DAW will not make any significant advancements in the next several years to come. It is proposed that advancements relative to a DAW will be made in the

software that is placed upon the DAW platform. Such software comes in the form of virtual studio technologies (VSTs). It is also proposed that because the DAW platform itself will not see significant advancements, the state-of-the-field concerning DAWs will remain relatively the same during the next several years to come, with exception to VSTs.

A. Advances in Technology, Advances in Music Creation

Years ago, tape machines used to record sound on magnetic tape were the pinnacle of music creating technology. At that time, there was no more advanced form of technology that could have been practically used to create music. Computers were present at the time, but they were more in their state of infancy and lacked the necessary computing power to be used for creating music. But as technology improved, music had made the transition from being analog, to being digital. The happening of the transition was due to the introduction and usage of compact discs (CDs) in the early 1980's [16]. CDs were the first medium that allowed music to be stored digitally. Computers were being used more in the industry at the time, but computing power was still not strong enough to fully implement a DAW. As time went on however, advancements in computing technology allowed computers to have the necessary processing power. The first fully functional and modern form of a DAW was released by Steinberg Media Technologies in 1989 [17]. Steinberg's Cubase (originally called Cubit) DAW revolutionized the process and ease of how music could be produced. A trend can be seen from these events: as technology advances, methods for creating music advance as well. This is because the new methods for creating music utilize the new advancements in technology. Because it is often desired that tasks get accomplished as quickly and as efficiently as possible, it is ensured that when any new technology comes out that could aid in creating music more efficiently, it will be utilized. For DAWs today, this means that unless a more advanced form of technology than computers presents itself, DAWs will remain the most efficient tools being used.

B. Advances in the Form of VSTs

The DAW platform has remained relatively the same since it was initially created by Steinberg Media Technologies. Advances (minor) over the years of DAWs themselves have come in the form of more appealing graphical user interfaces (GUIs) and VSTs that have been included with the DAW. Figure 17 [7] [26] shows Steinberg's Cubase (Cubit version) DAW and the more modern Cockos REAPER DAW.
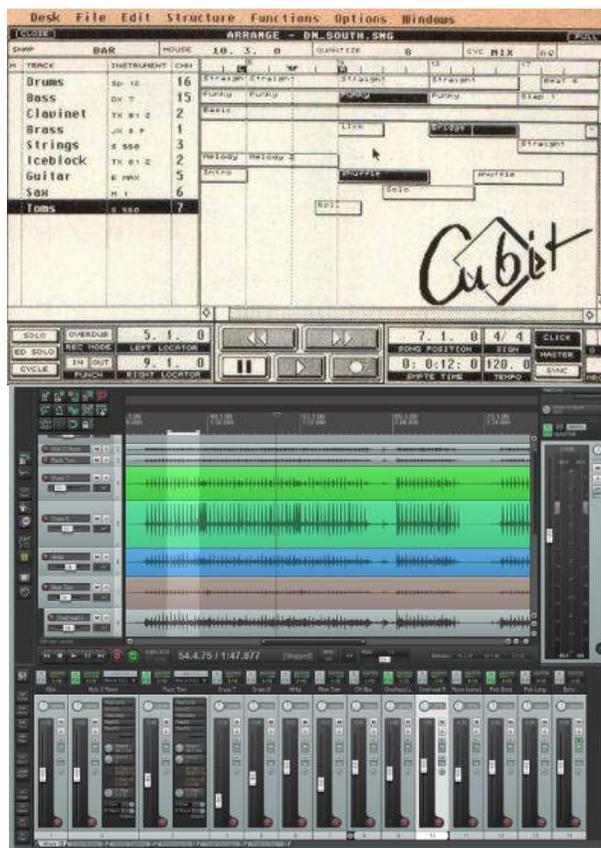

Fig. 17. Steinberg Cubit and Cockos REAPER: Although these two DAWs (Cubase top image, REAPER bottom image) are different graphically, they are structured very similarly. Both DAWs contain multiple tracks that contain recordings, as can be seen. Both DAWs also have a timeline that the recorded material on the tracks is placed on, allow for editing of the recorded material, and allow for VSTs to act upon the tracks they contain. The two DAWs were created more than 20 years apart, yet they still have a very similar structure.

As the DAW environment remains relatively the same as time goes on, the VSTs that can be added into that environment have been getting more sophisticated. Such is where the future of DAWs stands: advances in VSTs. The production of VST products has been rapidly increasing as time has passed, as has the sophistication of these products. Figure 18 [8] [6] shows a synthesizer VST created by Steinberg in 1999 [22] and a synthesizer VST created by Novaflash [21] that is still being maintained with software updates.

The recent surge of popularity in the dubstep genre of music has only become possible because of the new functions emerging in VSTs to manipulate sound. Techno was once seen as a radical genre of music because it was music that was created with "electronic sounds", i.e., not created with physical musical instruments such as a piano or violin. Dubstep is mostly "electronic" as well, but because it utilizes newer and more
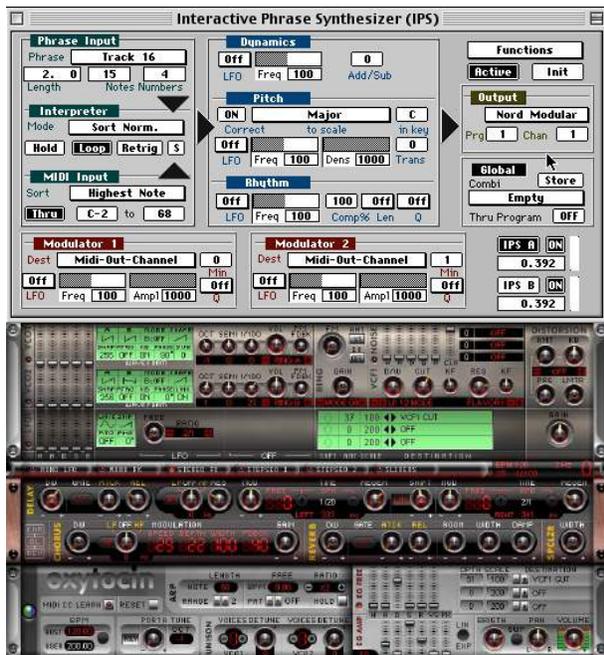
Fig. 18. Steinberg and Novaflash VSTs: Notice that Steinberg's synthesizer VST (top image) from 1999 is, by today's graphical standards, visually unappealing, unlike Novaflash's synthesizer VST (bottom image). The GUI of Novaflash's VST is more intuitive than that of Steinberg's old VST due to the knob and button controls of the GUI. These knobs and buttons make the digital interface seem more like a physically tangible interface, thus creating more ease of use. More importantly however, notice the difference in the amount of controls between the two VSTs. Because it has so many more controls, Novaflash's VST is far more sophisticated and provides more versatility in the manipulation of sound than Steinberg's old VST does.

Fig. 19. Korg Keyboard VSTi: Notice the VSTi has keys and controls just like a physical keyboard does. When the keys are set to be played in a sequence and the tone of the keyboard is set to that of a piano, the sound created from this digital keyboard will very accurately simulate the playing of a real piano.

distorted manipulations of sound found in today's newest dubstep VSTs such as DUBturbo [18] and Wobble-BOSS [19], dubstep has replaced techno as the new form of radical "electronic" music. VSTs are steering music in the "electronic sound" direction. This is because as VSTs become more versatile, they are able to manipulate natural sounds, such as those made by physical musical instruments, and turn them into unnatural sounds that resemble the sound originally created from an instrument in no way whatsoever. Sounds being manipulated by VSTs today however, most often come not from physical musical instruments, but digital instruments called virtual studio technology instruments (VSTis). A VSTi can most easily be thought of as a digital representation of a physical musical instrument. The most common VSTi is a keyboard but there are VSTis that can simulate most every physical musical instrument that exists. Figure 19 [9] shows a keyboard VSTi created by KORG [20].

VSTis are very convenient in that many of them are free, and that they do not need to be played in real time as opposed to physical musical instruments which can cost thousands of dollars and take years to learn how

to play. An issue concerning VSTis however, is that because the digital instrument does not need to be played in real time, people who have little to no skill in playing an instrument could create the same soundtracks as a talented musician that actually played the instrument in real time. For this reason, many musicians that play real musical instruments argue that technology is ruining the music industry because it allows for anybody to be able "play" music even though they have not dedicated many hours of their lives to learn how to play real instruments. Such issues arising show how sophisticated VSTis, and VSTs in general, really are because they allow anybody to create professional sounding music.

C. Current and Future State-of-the-Field

The current state of the field for DAWs is very stable. Now that music playing, editing, and recording are all digital, unless some more advanced form of technology than computers presents itself, music playing, editing, and recording are all in their most stable, modern state; not a whole lot can be done to change the way that music is currently produced. The main things that are improving DAWs are the VSTs that are being added to them. Essentially, the backbones of a DAW are remaining the same and the only thing changing is the amount of effects in the form of VSTs being piled onto those backbones. There are not many pressing issues in the state-of-the-field of DAWs. The most significant, which are still relatively minor in the scope of DAWs, are record label profits and propositions that analog equipment produces better sound. For record label profits, although labels use DAWs, DAWs have dealt a significant blow to profits that record labels used to make due to the fact that DAWs create .mp3 or related format audio files. And what can be done with computer files? They can be

shared. Because music files are massively (and illegally) shared, record labels are not making money on every file that actually exists in the world. This was not a problem years ago when tapes, CDs, and vinyl records were common. You could burn a CD and create a copy but that amount of profit loss from copying of material comes nowhere close to the amount profit loss due to the amount of file sharing that happens today. In the last 10 years, album sales have fallen an average of 8% per year. Music sales in 1999 were about $14.6 billion whereas in 2009, music sales were about $6.3 billion. The volume of unauthorized downloads continues to represent about 90% of the market, according to online download tracker BigChampagne Media Measurement [10]. This is more an issue of the effect of file sharing music but DAWs still play a small role in the issue. Now on the issue of analog units creating better sound, there is partial truth to this based on opinions, but it is mainly hype caused by people who are still attached to the old analog technology. The fact that the original tape film manufacturers have stopped producing tape film, which was usually a costly $200 per roll, supports the previous statement [12]. So why are some people still fixated on analog technology? The answer is because mixing recorded music digitally is not actually using hardware to do so, but rather mimicking it with software. That is why some people feel that DAWs are not producing the "true" sound that analog technology produces. But even though that is technically true, the software that mimics its analog cousin is so accurate that it is impossible for the human ear to distinguish a difference. That wasn't necessarily true in the early days of DAWs because computing technology back then was not as sophisticated as it currently is. But now with our advanced computer technology, VST mixers are so accurate compared to their analog cousins that the differences are indistinguishable [12]. More accurate and more functional VSTs will surface in the years to come. And because the number of different manipulations that can be done to sound is potentially infinite, more and more VSTs will be produced to accommodate for desired different sounds. The only way that the state-of-the-field for DAWs will change is if more advanced technology than computers presents itself. For the next few years to come, such technology being created so soon is an unforeseeable event which means that the state-of-the-field for DAWs will remain status quo.

V. CONCLUSION

After gaining the necessary information from the three subjects this work is comprised of, it is clearly seen why DAWs are so important and necessary in the music industry today. Commercial studio quality music can essentially be created by anybody because DAWs provide the necessary functions, ease of use, and affordability. New musical genres can even be created due to the functions that DAWs provide in association with the ever-improving VSTs that run on top of them. Such feats are the by-product of utilizing computer software wherever it can be practically applied. And in the case of modern music that millions of people listen to today, it has all been subject to DAW software. Such shows that DAW software has not only played a role in the production of music, but it has played a role in the lives of people that listen to the produced music as well.

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APPENDIX

The computer science courses I have taken have given me a lot of necessary knowledge for completing this project. If I had not taken any computer science classes, I would not understand how DAWs, or any software in general, function underneath the surface. Most of the information I know about how DAWs function has heavily relied on my understanding of data structures and object orientated programming. I have noticed that I can look at many computer programs now and have a very good sense of how the code running the programs is structured. Such was the case for this project. At

the beginning of the semester, I had never even used a DAW before. But because I was knowledgeable about how software works due to the computer science classes I have completed, I was able to learn and understand very quickly how DAWs worked and how they were to be used. Many people who take one look at DAWs are instantly deterred from trying to learn how to use them because they can look so complex. But for me, I have gained a very logical sense of thinking from taking computer science classes and I have also developed very logical problem-solving skills. These abilities enabled me to very easily accomplish the task of learning how to use a DAW and such abilities will aid me throughout the rest of my life when I am put to the task of figuring out how to solve other complex problems. This project has made me deeper understand how data structures and objects work along with how graphics can represent the data structures and objects. A DAW is really just one big GUI whose controls allow for the creation and manipulation of data structures and of the objects contained in those structures. And because I have had to study how DAWs work, by default I have had to gain knowledge about the data structures and objects in them, which has thus given me more understanding about how data structures and objects work in general; along with the graphics representing the structures of course. Without the knowledge I have accumulated over the years as a student studying computer science, I surely could not have produced a project of this caliber. I owe it to computer science for being able to accomplish tasks now that I could have never accomplished without the knowledge I have learned in the field.