Smarter Travel Voiceover Project

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Course: Multimedia Applications Development

Module: Audio Production

Class Group: W2

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Introduction

In this project I will create a voiceover script and record it in a professional studio. Afterwards I will create a mixdown of my work. The goal of this project is making a professional voiceover that will be uploaded on the internet. To gain experience working in the audio production field.

Also, I have the intentions to enter the Smarter Travel Students Awards using my project. The category I am aiming for is “Media”, which my project would be eligible for. As I process this project, I will showcase the editing skills I have learned so far using the software Adobe Audition.

This software is used by professionals in the media industry. Therefore, giving me real-work experience throughout the process. I will go through step-by-step, what audio effects I be using to enhance my audio recording. Sharing screenshots, explaining audio terminology and comparing the original audio file to my final file.

1: Initial Script

The Positive Aspects Of Walking

These are the reasons, why you should be walking.
Honestly, you don't need to be fit for it.
It's up to you how far you want to walk.
Nobody is going to stop you, as it can be done anywhere and anytime.
For such a simple activity, it can help with your mental and physical health.

Mentally, it justs puts you in a better mood and reduce stress.
Physically, its just easier on the joints, so don't expect injuries.
Two hundred calories can be lost in a thirty minute brisk walk, better than sitting down.
Zero costs for transport.

When you are done, you feel motivated to work or study.
So, stop listening to my voice and start walking.

For this competition, I can choose a large list of topics for my Smarter Travel entry. The topic I have chosen is "Positive Aspects of Walking" because I go out walking everyday myself. I feel I could add my own experiences for my entry. I need to make sure that the content is good and not to add filler.

My target duration for the audio file is forty seconds. For the intro, I am talking to the listener and stating my theme of the story. Which are the reasons why you should walk. I want to tell the listener how easy the exercise is and that there are no limitations.

For the middle portion, I continue from my last sentence in act one/intro. In that walking can leave a great impact on the human body. It can benefit you and people close to you, emphasis is on mental health. I have dealt with heavy stress when it comes down with college work. I want students to be aware that walking can help with issue.

Act three (End) is where I am closing it off and asking the students to reflect over my voiceover, to go outside and start walking. As previously said, "it can anywhere and anytime".

2: VO SCRIPT FORMATTING

ID	**The Positive Aspects of Walking**	DURATION: 40 secs VO: Male Tone: Informational Pace: medium
TPAOW00	These are the reasons, why you should start walking.	
TPAOW01	Honestly, you don't need to be fit for it.	
TPAOW02	It's up to you how far you want to walk.	
TPAOW03	Nobody is going to stop you, as it can be done anywhere and anytime.	
TPAOW04	For such a simple activity, it can help with your mental and physical health.	
TPAOW05	Mentally, it just puts you in a better mood and reduces stress.	
TPAOW06	Physically, it's just easier on the joints, so don't expect injuries.	
TPAOW07	200 calories can be lost in a 30-minute brisk walk, better than sitting down.	"two hundred", "thirty"
TPAOW08	Zero cost for transport.	
TPAOW09	When you are done, you feel motivated to work or study.	
TPAOW10	So, stop listening to my voice and start walking.	

Once I am happy with my script, I then covert it into the voiceover script formatting. The reason I am doing this is because it makes the text easier to read for me during the recording.

Let's begin at the top, there are three columns separated. On the left is my ID, where my title is used. For example, TWPAOW is short for The Positive Aspects of Walking. The ID number is added at the end too. The font size is ten in that column.

The second column consists of the main content that I be saying out in the recording. Starting with the title of the script, which its font size is sixteen and in bold. Within TWPAOW00 contains the opening line of the script. To make it easier for me, I made sure each line are seven words at maximum.

This is because I want to have natural pauses as I am going over my script. Meaning I won't be out of breath as I am reading it out, while reduces the chances of any bad takes to occur.

I can say during my voice recording, that using these formats helped me a lot. Especially if there a specific line I am looking inside the current script. It feels less overwhelming before doing a new take in the studio. Professional voice actors and actresses uses similar voiceover formats like this too.

3: Recording

Within my college campus at WIT, is a professional recording studio. Once I am inside the studio, it cancels any noise coming out from the room due to sound reflectors. Meaning I can shout at the top of my lungs without anyone hearing me. Before I start my recording, I do some warmups to help me relax and make my recoding sound full of confidents. This involves rolling back and forth my shoulders, berating in/out and shouting as loud I can.

For setup, I have a mixing broad which takes different audio sources through its multiple input channels, adjust levels and other attributes of the sound. To mix them in a smaller number of outputs. For example, if I take audio from a singer performance session. I can add effects to it and tidy it up, then mixing them down to a stereo or mono output.

To understand how to use it, I will begin with “Channel”. Channel is a unit of the mixer or a set of controls on the broad. It can be very complex as Channel controls can be repeated on the broad. Imagine it like a waterpipe for example, water enters in as the input and must travel down to the output. It travels to different parts of the pipe (Channels) to its destination (Mixdown’s fader before sent to output).

I start with channel 1 where the INPUT is my microphone. Moving down, I adjust my GAIN around 0 to -15. I do this because the gain controls the input level of each individual channel. Now I can push my channel fader all the way up to 0. Faders control the level of each channel sent to outputs, like master.

Moving on to the MASTER FADERS, these controls the level of signal being sent to the main outputs. I then push up the faders for USB (audio file being stored), PC, STEREO, MONO. The studio closed headphones which is a output of the mixer, is what I'll be wearing throughout the recording. Therefore, I adjust my PRES.HEADPHONES level between 6-7.

The next step is to power on the USB Media player (DENON) and insert my USB into the DRIVE slot. This where my recordings will be stored after each take. I then press the record button once to start arm, another press to start recording. When I am done, I press the record button again to end take and start another take. Each take creates a wav.file for my recording.

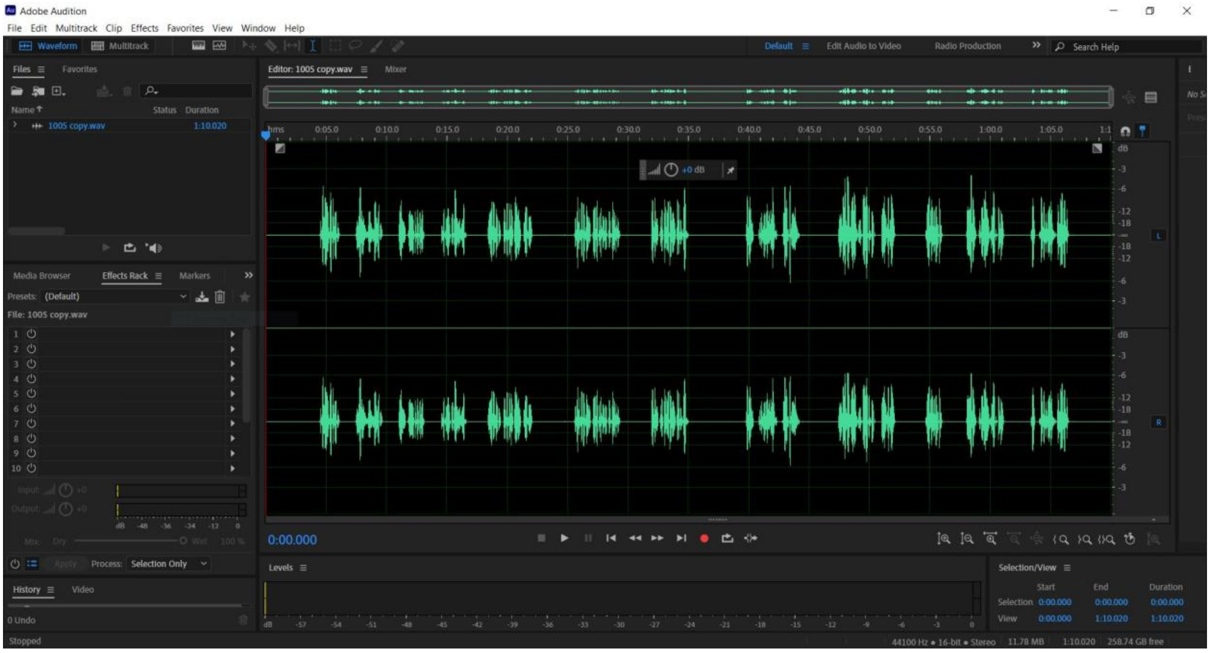
As I am recording, I make sure I am standing perpendicular to microphone and not being too close either. This is because I want the loudness meters balanced. This keeps my output volume in check because I don't want my recording to sound too low or too high.

The target range for the left and right meter is between the middle (4) and high range (7). I placed my voiceover script in front me, not holding as it could create unnecessary noise and my neck would turn downwards from microphone.

In conclusion, I found the first takes uncomfortable as no one like hearing your own voice. Overtime you get used to this and feel more confident with your voice. It helps that after reading each sentence, to take a breather and then move on instead of trying all in one go. I can edit it down later in Adobe Audition.

4: Editing

1) Inserting My Recording & Understanding Waveform



The software I be using for editing is Adobe Audition. On the top left corner is the Files window, where I can create or add new files. I inserted my WIT-SMT-RECORDING_MASTER-.WAV file. This will display my recording in the Editor window. The rough green lines appearing is my recording's waveform.

A waveform is a visual showing of soundwaves in my recording. Soundwaves are created by the vibration of my voice speaking out. These waves are form when air molecules are moved and fuse together near vibration, making air pressure rise. Therefore, the air molecules under pressure pushes the molecules surrounding them. Then they push the next group of molecules and repeats.

As a result, the wave is creating high pressure to move through the air. This leaves low pressure behind too. Back at the image above, what happens when the playhead reaches the waves? It makes my receptors in my ears to vibrate, making me hear my recording. Also, the horizontal green line passing through is air pressure being silent.

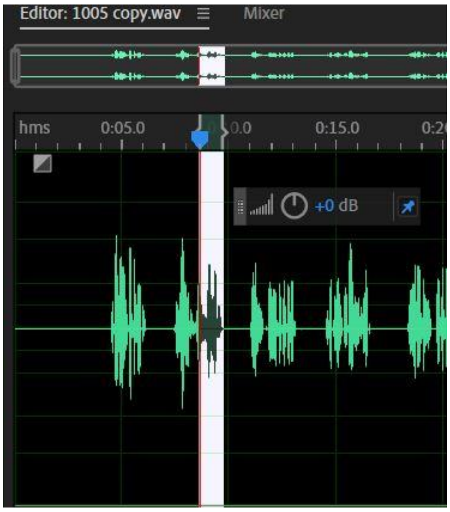
Peak is the highest pressure and trough is the lowest pressure. The difference between is called dynamic range. To better understand this, they are essential how loud or quiet this recording is. Another word for this is loudness or amplitude.

The term "cycle" describes amount of time it takes a waveform to go from one amplitude. Through the different amplitude changes until it reaches the same value again.

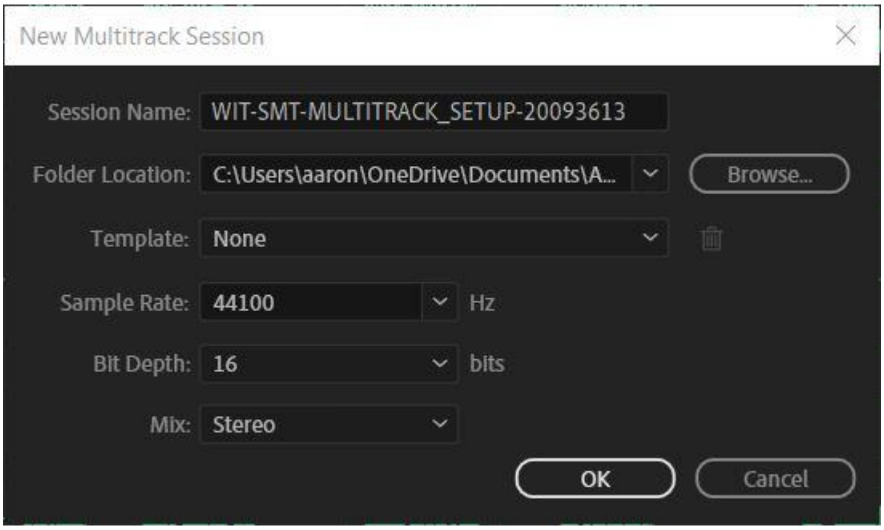
A common term Frequency (Hz) is used to describes the number of cycles per second. I will bring this point again later. Frequency is associated with pitch. This is when the faster a wave repeats itself, the higher the pitch is for the sound. Making me sound like my voice full of helium from a balloon.

If you look across the x-axis, it is the duration of the waveform. The y-axis is important because it is the air of molecules recorded using dB.

DB is a unit of measurement for amplitude of my recording. For example, the higher the peaks the louder my recording. The lower the peaks the quieter it is. Audition lets me adjust the dB in an area that may be hard to hear.



2) Using Multitrack



I am placing my audio recording inside a Multitrack session on Audition. I switch to the Multitrack tab on the top left and soon a popup appears. I name my session and change my sample rate to 44100Hz, Bit Depth to 16 bits and Mix to Stereo.

The sample rate 44100Hz is measured in hertz. The frequency is the number of cycles per second. Meaning the waveform goes through 44100 cycles every second. Sample rate tries to digitalise my original analog waveform. The higher the sample rate is, the closer it is to the original analog waveform.

Sample rate will try to limit my frequency range. It needs to be least double of my original frequency to work. Humans can hear 20,000 Hz. Therefore, 44100 is the option I have chosen.

Sixteen-bit depth creates a high dynamic range and reduces floor noise. Bit depth is the resolution of the amplitude. A bit is a number in computers that either have a value of 0 or 1.

The two values represent on/off or Truth/False. They have four different states: 0/0, 1/0, 0/1 or 1/1. Each added bit doubles the number of states.

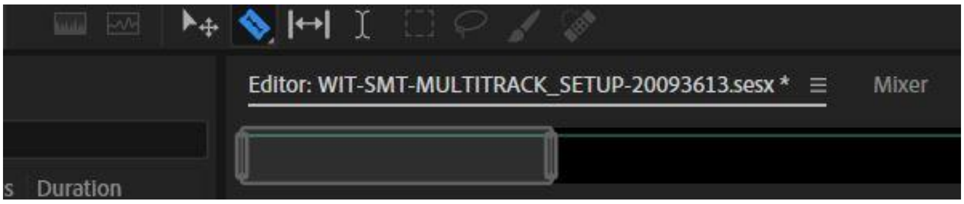
When waveform is sampled, each sample is then assigning the amplitude value closest to the original waveform. Two bits makes four possible amplitude positions in each sample. 16-bit is the CD-quality sound meaning each sample is a possibility of 65,536 values.

As a result, the higher bit depth is the more range of values for dB can reach to.

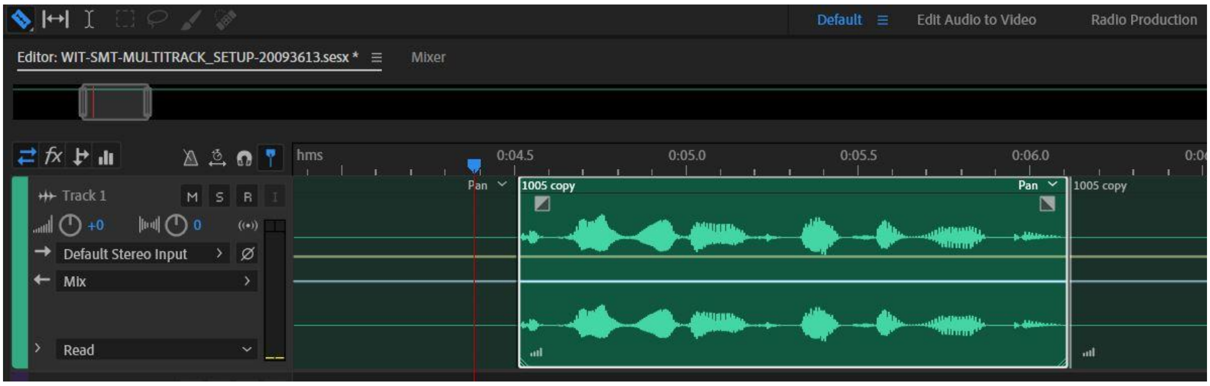
Stereo is used to represent two channels for the audio output (i.e.: speakers). Unlike Mono which only sends 1 channel signal. Stereo sounds wider and more realistic. Our ears would be able to hear more details from it.

Once session is set up, the Editor window changes to Multitrack session. Next, I will drag/open my audio recording into the empty Track 1.

Track 1 is now occupied by my recording file. Afterwards I will rename the file as WIT-SMT-RECORDING_MASTER-.WAV.



At the top of the Editor is the timeline of the selected track 1. In Audition, I can control the grey slider called the zoom navigator to zoom in/out of my waveform. I simply drag the end of the zoom navigator. The smaller it is the more its zoom in.

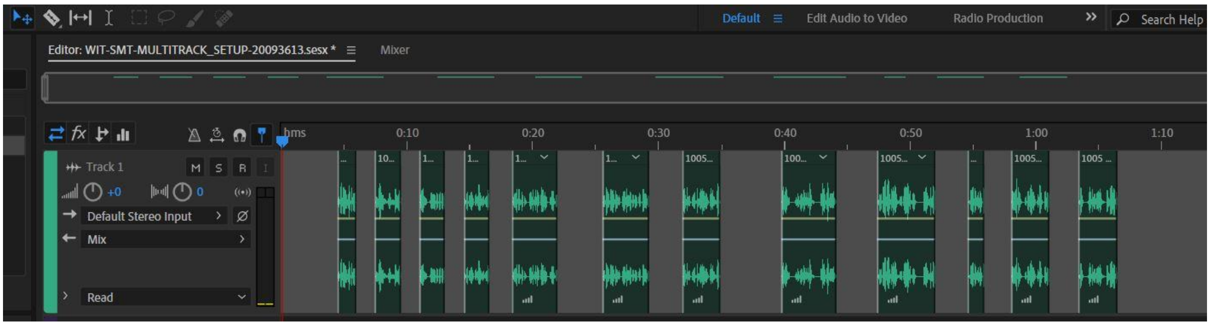


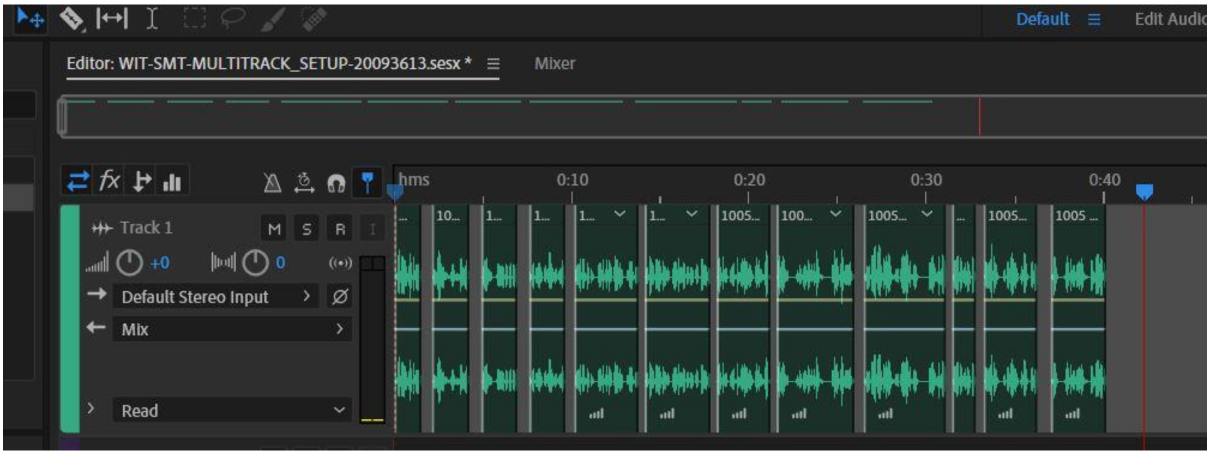
I want to zoom in my waveform so I can start my splitting at the exact point of each sentence.

Rather than leaving a long silent delay at the start of the recording. The tool I used is "Cut" which looks like a razorblade, the location of the toolbar is at the top left in the image above.

How it works is that I want to break apart my recording into clips containing a line from my script. I place my cut tool at the start of a cycle in my waveform, making a starting point. While scrubbing through the waveform with my playhead.

I use the same cut tool to make my end point at the end of my current line. Scrubbing is an interaction which a user drags a playhead across a segment of a waveform to hear it.





I repeat this process for all my lines in my recording. Now I can zoom out and see all the clips I spliced. I can select clips that have no sound waves and just delete them using the backspace key. This should just remove a part of the waveform and be replaced by the grey empty space.

Now I can rearrange all my segments of the recording to keep it under forty seconds. To make it easier for me, I turn off the "Snapping" tool so that my clips don 't clash with one another.

To move clips, I just select it and drag it across the timeline of track 1. The Snapping tool is found beside the x-axis (seconds) at the top of track 1. Its symbol is a magnet.

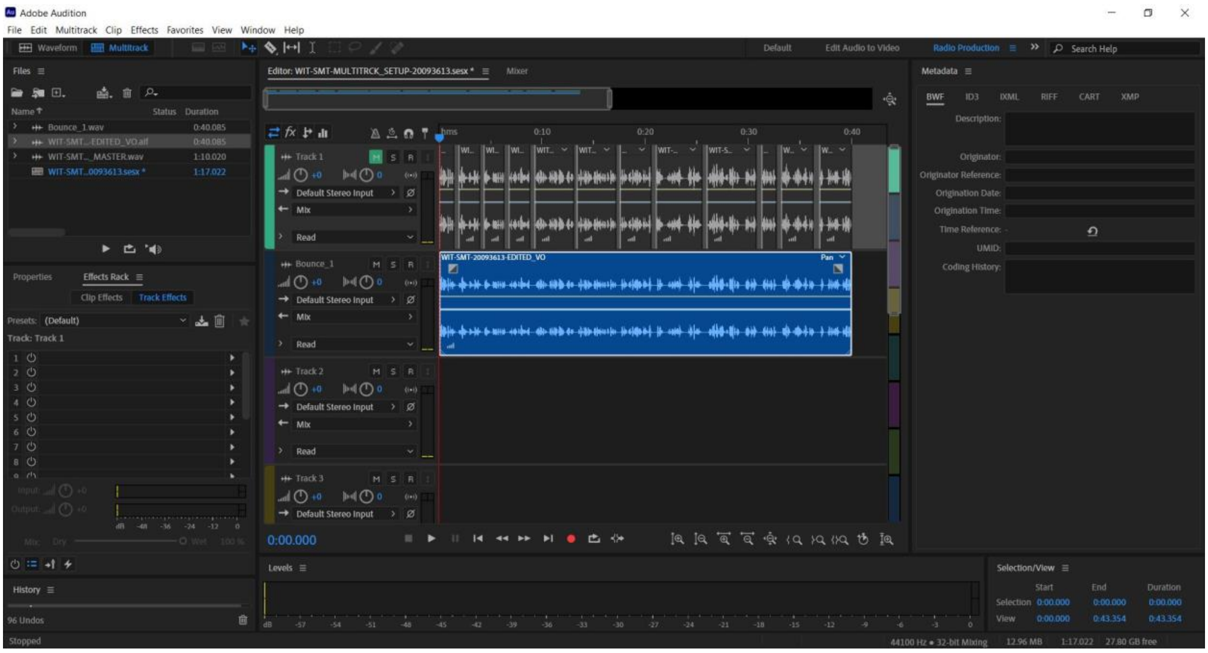
3) Creating Bounce File

After finishing splicing and tidying my recording, I now want all my clips to be mix into one file. It's a file of my recording but now without the awkward silence and to have it flow in a consisted form.

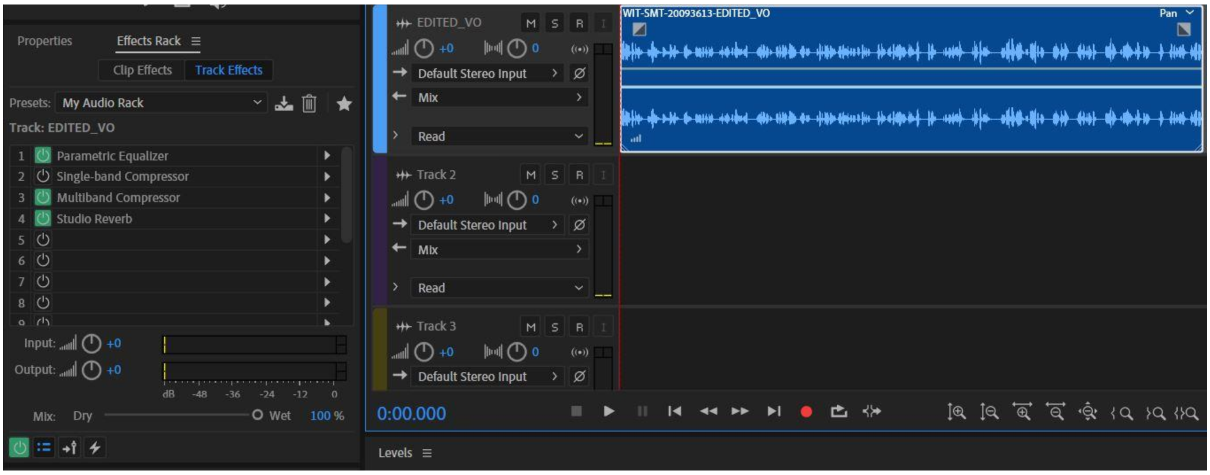
I double click at the empty space at the end of my last clip. This selects all my clips.

Next, I right-click and select Bounce To New Track > Selected Clips Only. Once it done, it automatically creates the file and adds its waveform to the next track tab.

Automatically it will be named Bounce 1, but in the screenshot below I renamed my bounce file to WIT-SMT-20093613-EDITED_VO. Let's say I only want to listen to my bounce file, I can simply mute Track 1 and then press play. Another option would be to have the Solo button turn on in my EDITED_VO tab.



4.1: FX Settings

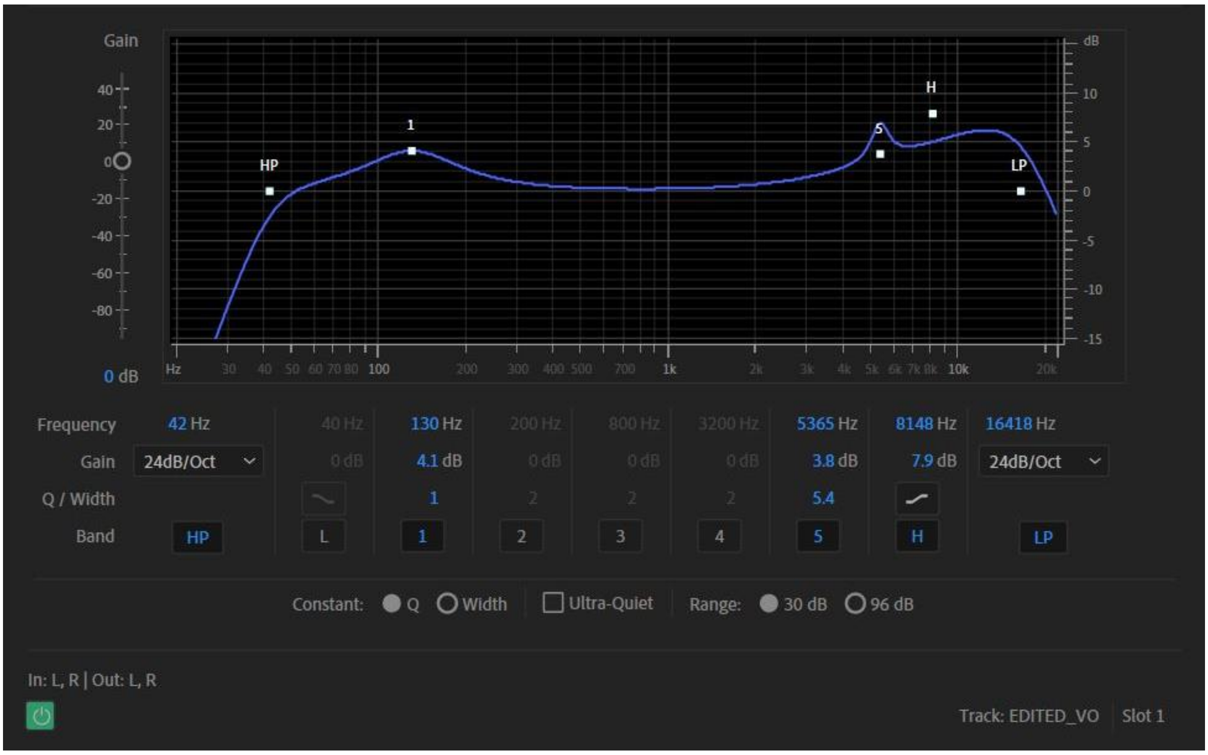


With my EDITED_VO selected, over to the right is my Effects Rack. This where I can use variety of effects that can enhance my recording in any way I want. To create a new effect, I click the right arrow of slot 1 in the rack window. This will provide a list of options such as Amplitude & Compression, Filter & EQ, Reverb and many more. I can apply more than one effect to my recording if I wish.

I have done previous editing on a different project before on Audition. Audition can let you save your previous presents. This saves time from starting all over again. I can still edit these effects if I want and save it again.

"My Audio Rack" is the name of my presents that I will using. It has the following: Parametric Equaliser, Single-Band Compressor, Multiband Compressor, Studio Reverb.

1) Parametric Equalizer



To find this effect use Filter And Eq > Parametric Equalizer. Parametric Equalisers gives me total control over frequency, Q and gain settings. Frequency sets the center frequency for bands 1-5 above in the image. Gain controls overall amplitude. Q controls width of selected frequency band.

The graph above visualises my frequency in the recording. The y-axis is the amplitude. The graph ranges from lowest to highest in a logarithmic fashion and spaced evenly by octaves.

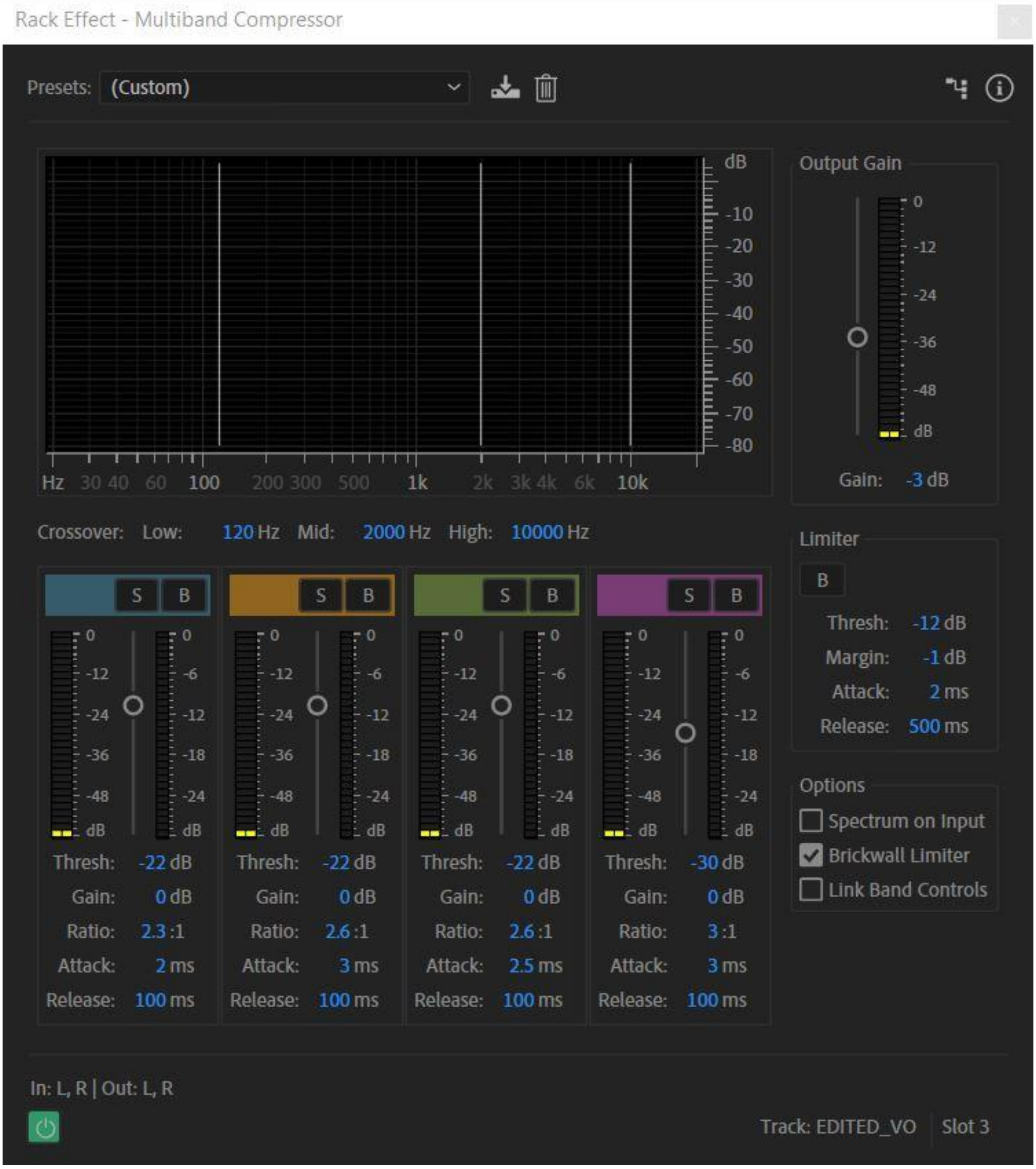
Parametric equalisers allow me to pick spots of the audio frequencies. I can choose to either boost or cut. At default, the blue graph line will appear horizontal across the x-axis (0hz to 20hz). The y-axis is the amplitude using the unit molecules dB (-15dB to 15dB)

For band 1, I am boosting the frequency around 130Hz because I want to make a boom effect with my recording. For band 5 and its next point (H) or High Shelving.

This range is important because its where 60% of human voice can be heard (1k - 8k). Using the High Shelving filter reduces hiss or amplifier noise.

I don’t need to manual change it as I can use other pre-sets in Audition. There are two ways I can adjust the frequencies. I can create adjustments by moving the dots on the graph where I want them or typing my desired frequency. If my volume is too loud, I can use the Master Gain slider on the left to control volume.

2) Multiband Compressor



Located by Amplitude And Compressor > Multiband Compressor.

Compressors is squashing the sound, usually by having a main threshold level. Anything above this level will be reduce. Essentially it makes the sound volume lower by reducing dynamic range (difference between the loudest and quieter sound).

Multiband compressor lets me compress four different frequency bands separately.

Each band has a unique dynamic range, so this effect is good for audio mastering. This effect lets me choose crossover frequencies and apply band-specific compression settings.

Crossover frequency is determining the width of each band, either low, midrange or high. Threshold slider sets the input level when compression begins. For example, if I wanted to highly reduce the dynamic range. I would set the input level around 15 dB giving high compression.

Gain can be used to control amplitude of a specific band after compression. Ratio sets compression ratio between 1 to 1 and 30 to 1. For example, 3:0 outputs 1 dB for every 3dB increase in compression threshold. Attack is how quick compression is applied when it the audio exceeds threshold.

Release is how quick compression stops after audio is below threshold. Output Gain controls the overall amplitude after compression. Limiter is used after Output Gain, at the end of the signal path, optimizing overall levels.

It specifies Threshold, Attack, and Release settings that are less aggressive than similar band-specific settings. Then specifies a Margin setting to determine the absolute ceiling relative to 0 dBFS.

3) Studio Reverb

Reverb is where if I was in a room doing a recording, sound would bounce off to other areas of the room towards my ears. Audition let me use reverb effects to simulate many room environments. I mainly looking for a professional studio environment.

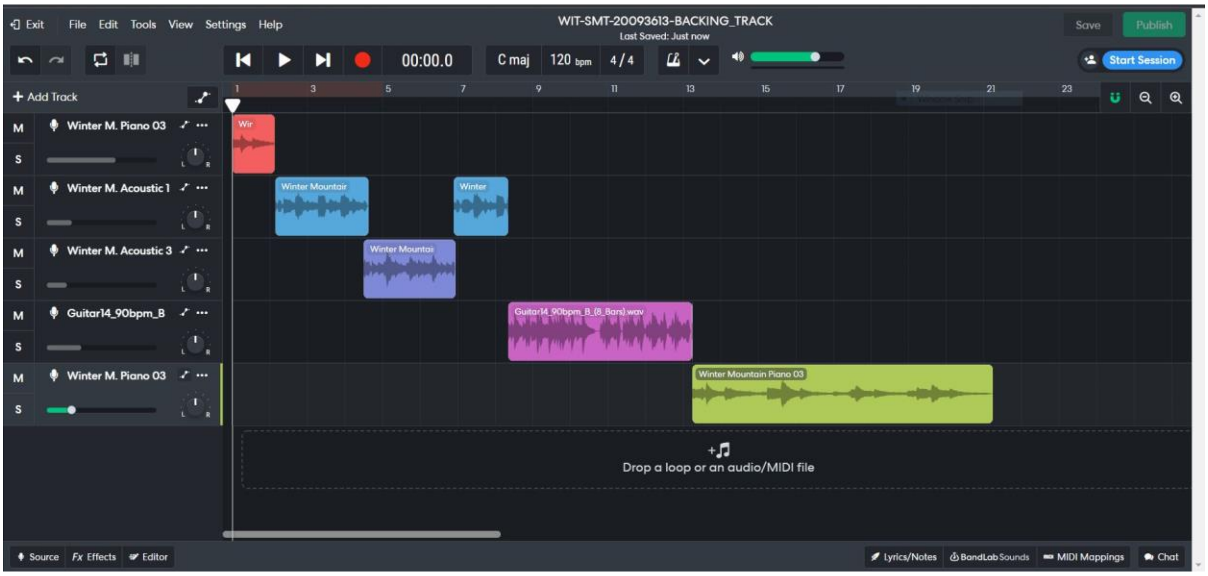
What makes studio reverb unique is that it is faster and less processor-intensive than other reverbs effects in Audition. Roomsize sets room size, Decay reduces amount of reverb in milliseconds. Early Reflections controls the percentage of echoes.

Width controls the spread across the stereo channels. High and Low Frequency Cut specifies where a reverb can happen at its highest and lowest frequency. Damping adjusts the amount of attenuation applied to the high frequencies of the reverb signal over time.

Diffusion simulates the absorption of the reverberated signal as it is reflected off surfaces. Finally, the dry effect sets how much of the original audio to be outputted with this effect. Wet is setting the amount of reverb to be outputted.

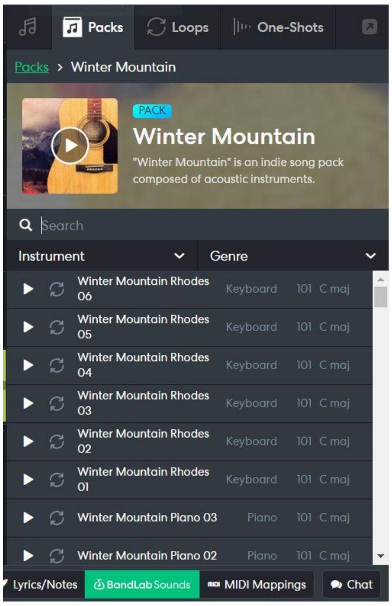
I want my recording to have reverb that is not too high (loud echo) or too low. I need it to be heard but not distracting the listener but to have a presence in the background.

5: CREATING THE BACKING TRACK



I am using Band labs to create my backing track. It is a simple process as you don't need to use instruments connecting to the PC. Band lab provides MIDI instruments, sound loops and sound effects to pick from. Then all I need to do is to insert them into separate tracks.

I want my track to have a laid-back theme, as you are doing your walk. Instruments like stumming the guitar or piano keys I feel would suit my theme. I found loops such as "Winter Mountain" which contains different guitar strumming and piano loops. As a result, I want to make sure it flows nicely and fits the theme of my project. I may adjust the volume(dB) to create dips or fade in/out in between tracks.



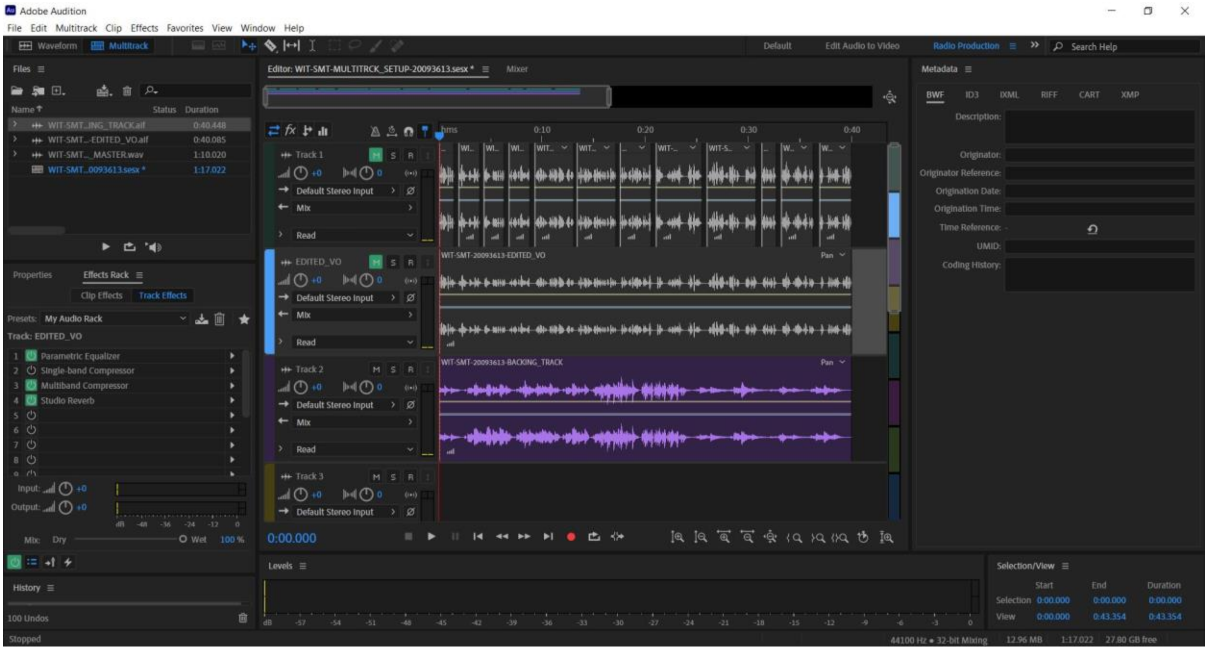
I use the zoom tool here to get a better look at my region for the first track.

I can edit my region by adding a second loop after its finish its first cycle. I drag the Loop symbol ↻ to create another loop.

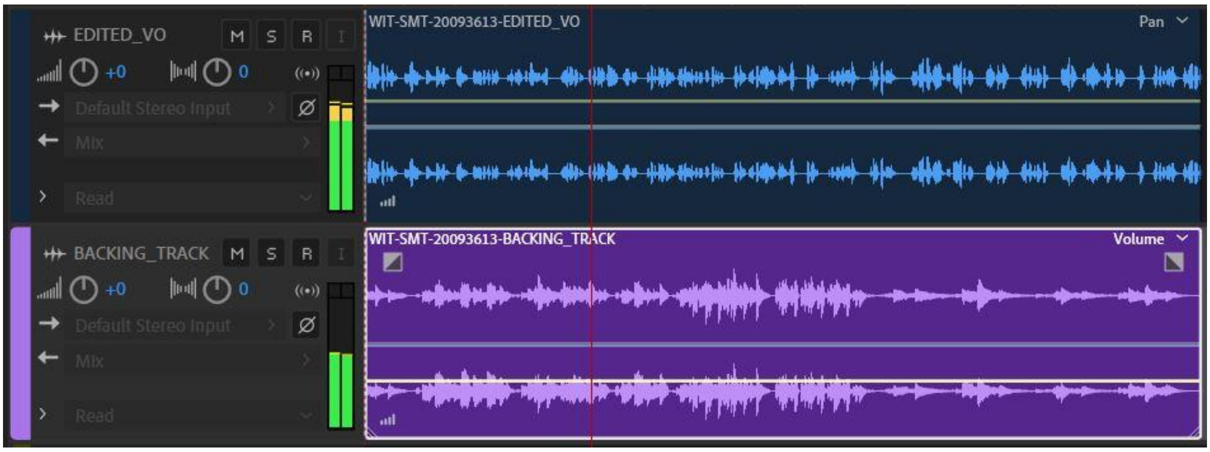
Or I can resize the original loop by dragging the symbol at the bottom right corner.

With automation enable, I can adjust the volume of each track separately instead of effecting all the tracks. I move the white horizontal lines to adjusts the volume(dB). The white points are used to create fade ins and fade outs of each music track.

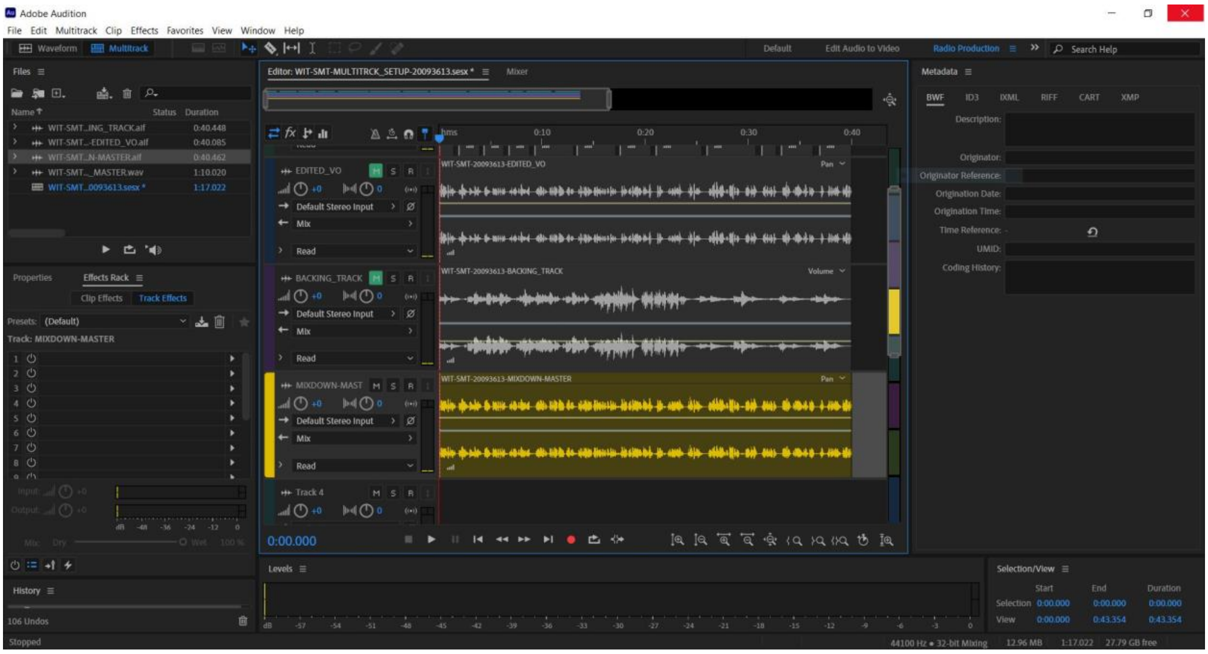
6: MIXING VO & BACKING TRACK



After making my backing track on BandLab, I can now insert it into my Mixdown session. I first saved my backing track as WIT-SMT-20093613-BACKING-TRACK.AIF file. Drag it from my folder and into the files window while in the Multitrack tab, I drag it into a new track under EDITED_VO.

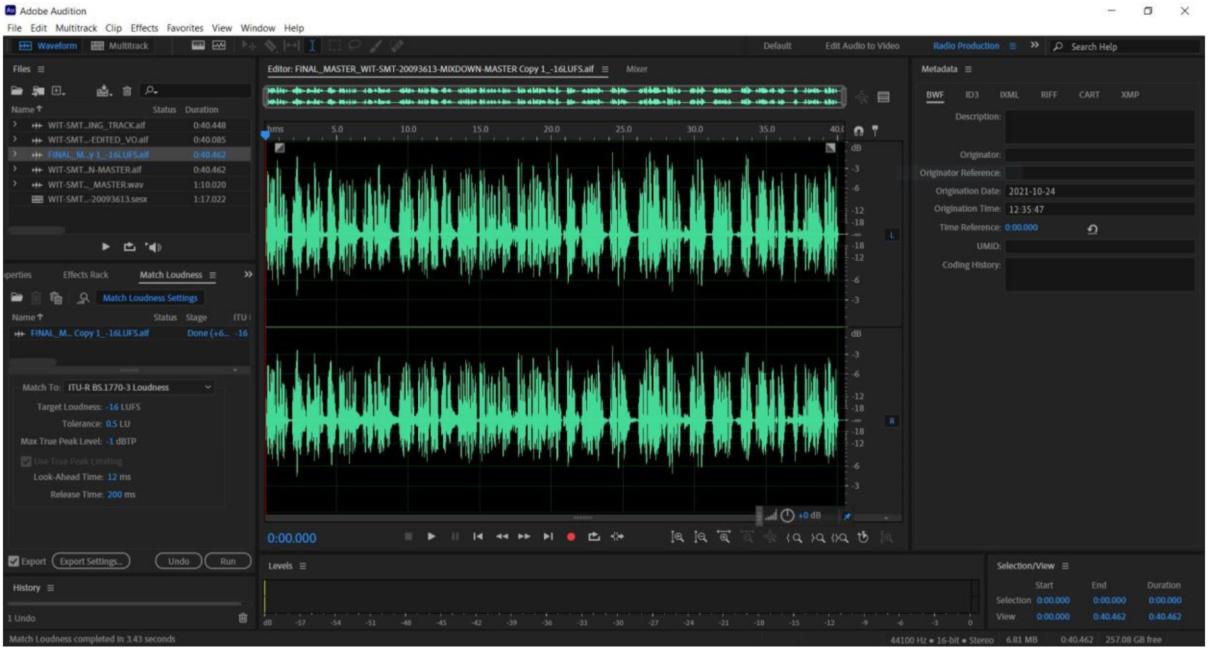


With BACKING_TRACK selected, I click on the yellow horizontal line. This line controls my amplitude of this track. It is like the “Show Automations” feature in GarageBand. For the backing track I want it to be quieter so I can hear my voice clearly. I reduce it down or cutting it to -15dB to achieve this.



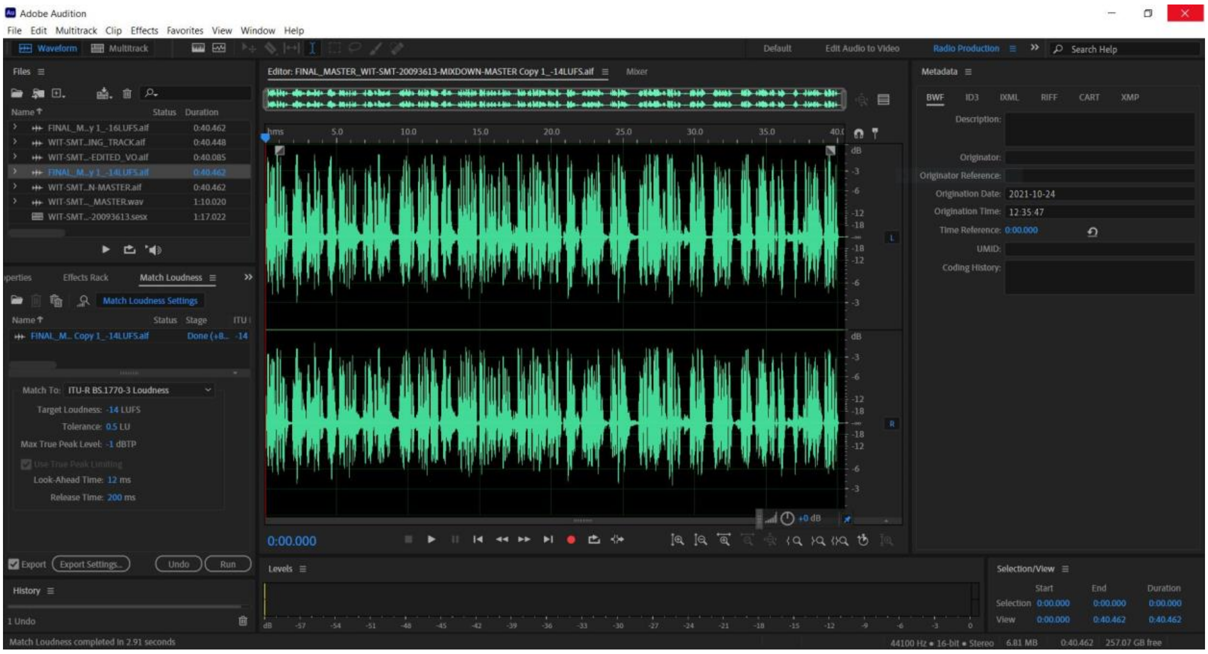
After inserting my backing track, I need to combine my EDITED_VO file and Backing Track. As a result, this file would be called my WIT-SMT-20093613-MIXDOWN-MIX.AIF file. I right click on my EDITED_VO file and select Mixdown Session To New File > Entire Session. Now I have a new file with a processed voice on top of backing track bounced to a single file.

7: MASTERING



Now its time for finale part of the project. I am going to export my MIXDOWN-MIX file to make it suitable to online services. The first export will be using -16 LUFS. I select Window > Match Loudness, then drag my mixdown onto the new panel opened.

In the Match Loudness settings, I choose ITU-R BS.1770-3 Loudness and change the LUFS to -16. This value is used by services like Spotify and YouTube. I then click Run, which will automatically create a new version of my file with the -16 LUFS applied into it. I repeat the same process below for -14 LUFS



8: EXPORTING

The following files are exports of my FINAL_MASTER -14 LUFS file. I am comparing the type of compression used, the sound quality and its file size. To start with, my WIT-SMT-20093613-FINAL_MASTER-14 LUFS.AIF file is 6.99MB. AIF (Audio Interchange Format) was developed by Apple and is the native uncompressed audio format for MAC OSX.

a) MP3 (MPEG -1 LAYER -3)

This the standard for lossy compression. A popular choice format for online music services. The size of the file will reduce significantly, making it easy for me to share with other users. However, it is lossy meaning it cannot be rebuilt because the information is lost from my original file.

The sound quality may be good for consumers, but I think for professional audio editors it is not the best sound quality when comparing to the original.

File size = 958 KB

b) M4A (MPEG 4 AUDIO)

Mainly used by Apple machines. The only container format that contains either ACC (lossy) or ALAC (lossless). MP3 needs to be encoded at a high bitrate i.e.: 192kbps to get a similar sound quality to my original audio. M4A bitrate can use less bits than MP3, meaning my new file can be a smaller size but with great sound quality.

File size = 373 KB

c) FLAC (Free Lossless Audio Codec)

This is a lossless compression, the file size is much bigger than the previous formats I used. Meaning the high-quality sound is identical to my original audio file. With no data lost, the audio quality is equal to CD-Quality.

File size = 3.28 MB

d) APE (Monkey's Audio)

It is a lossless compression algorithm. There is little difference when comparing to other lossless formats such as FLAC. APE must take more processor power to decode. Sound quality is great because of the lossless format, giving me a high resolution unlike MP3.

File Size = 3.15 MB

d) OGG (Ogg Vorbis)

It is an open-source file format for multimedia. Can contain video, text, metadata and music/audio. This is a lossy audio format, meaning some data will be taken away from my original file. To compare with the other lossy format MP3, OGG bitrate varies while MP3 bitrate is constant.

Ogg Vorbis file format is used at 320kbps, which is higher than MP3. Giving my new file a better sound quality. Although Ogg Vorbis is better, people choose MP3 because its more compatible than Ogg Vorbis.

File size = 2.14 MB

9: CONCLUSION

The best lesson I learned from this entire process is trial and error. I can apply this to my workflow during the project. At the beginning, I found writing scripts not as difficult, but the actual recording was an interesting process. It's an environment where I am acting but there is no physical audience. Making me visualise what the other person is listening to during my recording.

Adobe Audition felt overwhelming at first because of the amount options and windows to understand. After going through the official Adobe Audition tutorials, I felt more confident to use the software. Splitting the recording into clips was satisfying because it wasn't a recording with awkward pauses anymore.

Learning how the effects (i.e.: Parametric Equaliser, Multiband Compressor, Studio Reverb) work was interesting. The way I can edit my voice can create such a different tone. How they work was very important to me because I wanted my project to be high quality and to sound like a professional voice over.

Making the backing track was fun because I didn't need an instrument. Band Lab and GarageBand are great software as they provide large categories to choose from. Especially to improve my voice over and to put more emphasis on my theme(chill/laidback).

I had no trouble making the mixdown as I had all my requirements needed. The EDITED_VO and BACKING_TRACK. After it's created, I had to reduce the volume for my backing track to make my vocals easier to hear. Then use my MIXDOWN-MASTER to export both a -16 LUFS and -14 LUFS FINAL MASTER.