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ECE 45
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Synthesizer Final Project

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OVERVIEW

The goal of the final project for ECE 45 is to employ the foundational ideas of the class, and apply it in a meaningful way through real-world application. In this case, we employed the use of continuous signals via sound waves, and altered the sound waves using multiple methods we have learned from the class. Some of these methods include the Fourier Transform paired with different types of filtering to achieve the desired effect of implementing class ideas in a useful manner: changing a sample audio to make it sound different.

CONTRIBUTORS

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REAL-WORLD APPLICATION AND USAGE

- **Chorus Filter** - The chorus filter is a filter that adds filler sound to the original wave file by amplifying the sound such that the original wav file sounds more powerful and deep. It works by amplifying the lower frequencies of the wave file, and then taking chunks of each frequency over 100 times with delay, and adding it back together and dividing by 100. This makes the wave file seem more rich in sound. Some applications of this would be if a singer is singing a song alone, but wants to make their voice sound as if more people are singing the song with them. I.E. if a single person was singing a Christmas Carol, it would sound like many people were singing at once.

- **Muffler Filter** - The muffler filter is a filter that works by running a low-pass filter to only allow lower frequencies, and then adding a reverb effect to the new sound. The resulting sound is lower, and sounds like the original sound if the original sound was some distance away from the listener, being muffled out. It is like if you walk outside of a party and hear the party noises from outside of the building, but the sound is dampened. Some applications of this in the real world would be in music production, such as a rap song. If a certain rap producer wanted the effect of an interlude conversation being dampened out while the song was playing in the background, then this filter would apply. Or another example is in movie sound production, and a scene of a battle is taking place, and the sounds of battle are being muffled while the movie scene plays out.

- **Amp Frequency Range** - This filter works by amplifying or reducing a certain range of frequencies within a wave file. This filter works by adding a multiplier within a certain range of frequencies, such that the amplitudes of the frequencies are increased or decreased. Real-world application of this filter is very useful, because by using an amplifier of zero, you could take out a section of a song. I.E. you could take out all of the bass of a song, so that only the melody remains. Another real-world example would be if a DJ wanted to bass-boost a certain song, and increased the bass of the song using the frequency-range amplifying filter demonstrated here.

HOW TO USE THE SYNTHESIZER

Installation

1. Download the Project folder from the GitHub link https://github.com/pmaddire/Audiofiltering
2. Make sure all the python files, as well as the input file you would like to edit, are within the same directory. Your input file must be in the .wav format.
3. Some key notes are that this was written using py37, and will not work without a py37 environment.

Usage

1. To start using the program, open up the ECE 45 proj.py file and run it. The program will then instruct you in how to use it and run through a few steps before creating an output file in the same folder.
2. It will ask for the name of the input file (must be in the same directory) first. Make sure to add .wav to the file name
3. Then you will be able to choose which filter you would like to use (Linear Envelope is not currently working).
4. The program will now ask for the file name that the new edited sound should be stored under. Make sure you add .wav to the filename.

5. Once a filter is chosen, the program will prompt you for specific values for filter parameters. Give inputs as integers unless specified otherwise.
6. After you have provided the necessary parameters the program will spend some time applying the filters and will create a file under the name you previously provided within the same directory.
7. Throughout the process, the program will supply you with visuals as well as updates as the program applies filters.

APPLICATION OF CLASS MATERIAL

- **Chorus Filter** - The chorus filter utilizes fourier transforms to change the wav file to frequency domain, and fourier transform shifts to return the frequency domain graph back to a center of zero. Additionally, inverse fourier transforms are utilized to transfer the graph of the waveform back into the time domain so that it can later demonstrate the difference between the graph in the time domain pre and post filtering.

- **Muffler Filter** - The muffler filter also utilizes the same fourier transforms, fourier transform shifts, and inverse fourier transforms for the same reasons listed as above, however the muffler filter also includes a low-pass filter such that we will be working only with lower frequencies for the filter. This, in turn, allows for reverb to be applied so that the lower frequencies are "muffled" and the entire sound is quieter.

- **Amp Frequency Range** - The amp frequency range, in contrast, does not need any low or high pass filtering. Instead, there are fourier transforms and fourier transform shifts (and inverse fourier transforms for the same reasons listed as for the chorus filter), but the central difference is that the the range inputted will be amplified by multiplying whatever the inputted multiplier is to the amplitudes of the frequencies within the inputted frequency range.