Digital Signal Processing

Flanger

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I. Background of the Study

We use sounds on a daily basis, primarily in communication, in creating music or patterns, in studying the behavior of our surrounding to grasp a better understanding of it. Over the years sound engineers has been studying and using different techniques and technologies in sound modulation to further understand signals or create distinct sounds. One of the common modulation techniques is flanging, it is achieved by simultaneously playing two identical sound signals while time delay or shift is applied on another copy[1] . This creates a comb filter which produce a sweeping sound effect.

Flanging is a phase modulation which is similar to that of Chorus effects however at a shorter time delay between 1ms to 5ms, creating a slight difference on a specific interval then sync with the original sound signals somewhat creating a satisfying sound to its listener. This also create output feedback signal back to the input to perform signal processing[2].

Recent studies seek to improve the Flanging model [3], creates neural model on Flanging [4], to simply creates a better approach on creating this effect. This project seeks to understand the mathematical and digital signal processing concept behind the Flanging effect.

II. Statement of the Problem

Flanging is achieved using three common approaches, the traditional tape where audio engineering or producers taps on the actual tape record, the pedal flanging which most musician used or the software flanging used as plugins or extension in audio editing software[5]. Both the traditional and the pedal flanging requires the instrument or physical copy of the audio, and as well as the skill to operate the device. Furthermore, the cost of utilizing or owning the device is another factor that hinder this study. For software, there may be free or open-source programs to be used to recreate these effects however there are only limited program that let you observe graphical representation of this signal on each interval. Hence, this study aims to understand the operating principle and mathematical implementation of flanging and digitally recreate the effects.

III. Methodology

a. Design

The design of this project is experimental, where in the proponent test approach of digital signal processing discussed throughout the semester. The flanging involve phase shifting of signals to where delay is applied to the system hence creating another copy of the audio system, to arrive at this result, the concept of Fourier Transform is applied on the system. Lastly the experimental process of the project revolves on the delay and sweep frequency applied on the audio and analysis will be done through listening and observance of the result of the sum.

b. Mathematical Solution

The basic mathematical implementation of this project, is the summation of the system minus its delayed parameter.

$$X[k] = \sum_{n=0}^{N-1} x[n] - x[n-d]$$

Where k is an integer, and d is the delay are the shift applied on the duplicate copy of the audio.

This equation already imitates the audio effect as the phase is shifted however its sill needs to smoothen the effect and provide a much more pleasing audio.

To further improve the effect, the concept of Fourier transform is followed.

$$X[k] = \sum_{n=0}^{N-1} x[n]e^{-2\pi i \frac{kn}{N}}$$

Whereas the actual equation implemented in the program is

$$X[k] = \sum_{n=0}^{N-1} x[n] + x\left[n + d + (r\, sin\frac{2\pi ns}{K})\right]$$

Where:

r is the is the amplitude of change, s is sweep frequency which is the total delay change of the system; and K is the sample rate of the audio system.

This smoothens out the audio signal thus providing a more appealing result.

c. Program

```
function [ y ] = Flange( x, fs)
x = x(:,1);
delay = 15;
range = 12; % amplitude of change
sweep_freq = 0.3125; % provide a distinguishable sweep

   for i = 1:length(x) - delay - range
       y(i) = x(i) + x(i+delay+round(range*sin(2*pi*i*sweep_freq/fs)));
   end
end

clc
[x,fs] = audioread('raw.wav'); % read audio file

y = Flange(x,fs);

% Plotting the results
figure;
subplot(2,1,1)
plot(x)
title('Raw Audio')
xlabel('Time')
ylabel('Audio Signal')

subplot(2,1,2)
plot(y)
```

```
title('Flanged Audio')
xlabel('Time')
ylabel('Audio Signal')

%% Plotting Raw Audio
% Plot Time Domain for input
xt = linspace(0,length(x)/fs,length(x)); %vector of raw audio
figure;
subplot(2,1,1)
plot(xt,abs(x))
title('time domain')
xlabel('Time')
ylabel('amplitude')

% Plot Freq Domain
xnfft = 1024;
f = linspace(0,fs,xnfft);
X = abs(fft(x,xnfft));

subplot(2,1,2)
plot(f(1:xnfft/2),X(1:xnfft/2));
title('freq domain')
xlabel('freq')
ylabel('abs')

%% Plotting Flanged Audio
yt = linspace(0,length(y)/fs,length(y)); %vector of flanged audio
figure;
subplot(2,1,1)
plot(yt,abs(y))
title('time domain')
xlabel('Time')
ylabel('amplitude')

% Plot Freq Domain
ynfft = 1024;
f = linspace(0,fs,ynfft);
Y = abs(fft(y,ynfft));

subplot(2,1,2)
plot(f(1:ynfft/2),Y(1:ynfft/2));
title('freq domain')
xlabel('freq')
ylabel('abs')

%% Audio Output
% play the raw audio
sound(x,fs)

% play and save the output audio
% sound(y,fs)
% audiowrite('output.wav',y,fs);
```

IV. Analysis

After the experiment has been performed and conducted, the following figures were obtained:

Figure 1, present the plot of the raw and the flanged audio, at first glance the difference is not noticeable except from the color of the raw audio and the flanged audio however much details is presented in Figure 2 where the magnitude at the last unit is different. Furthermore, the magnitude of the flanged audio is likely doubled compared to that of the raw audio. This is because the result is the summation of its original valued increased by its shifted value.

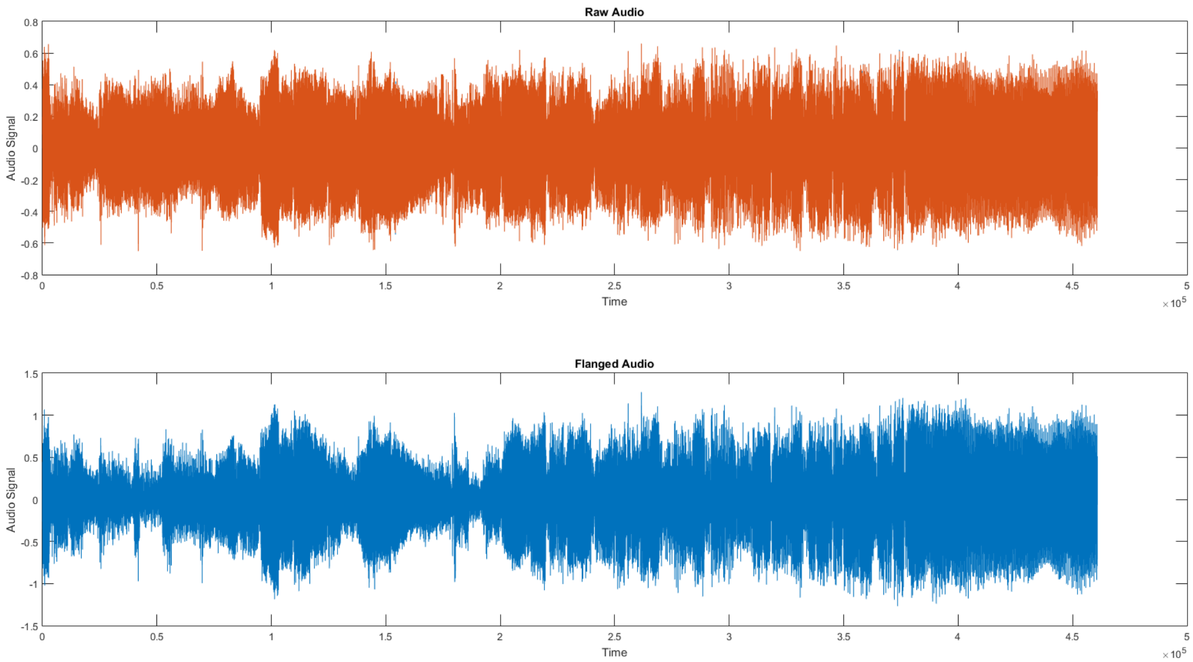

Figure 1. Amplitude Plot

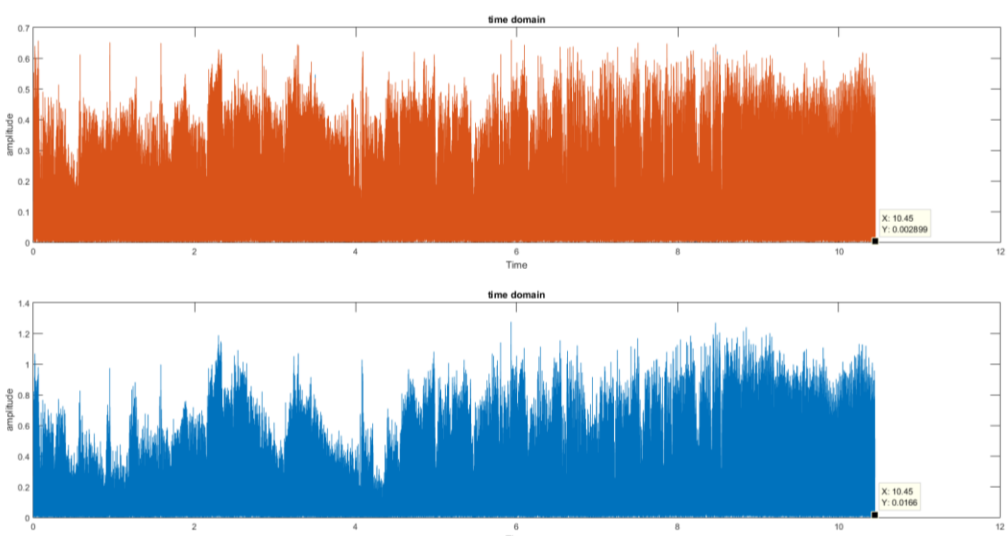

Figure 2. Time Domain Plot

By playing the two vectors through the MATLAB sound function, the shift in the audio is noticeable with a delayed effect creating a chorus effect when listening to it.

Furthermore, it can be observed in Figure 3 where the frequency domain is presented the value also doubled, the same observation can be seen in the previous figures.

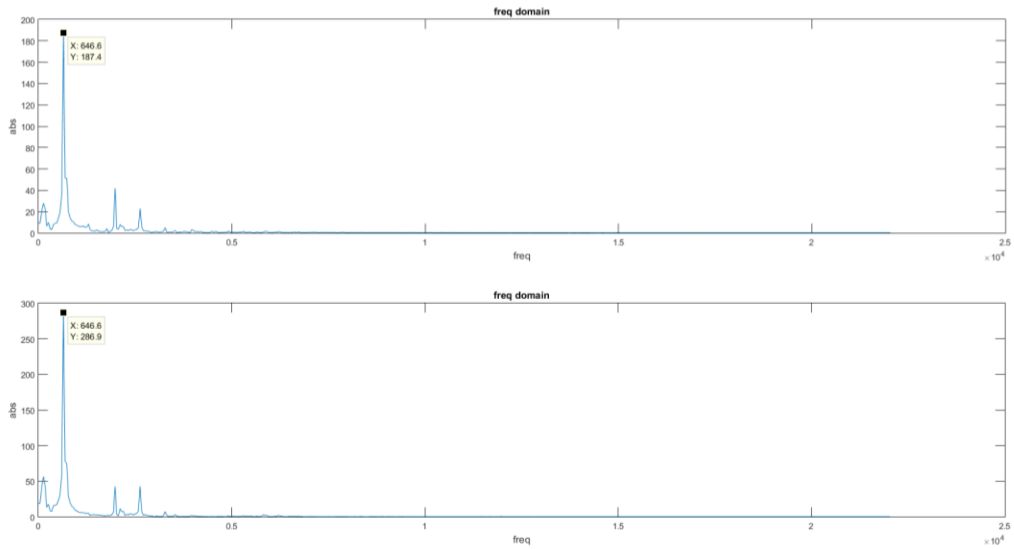

Figure 3. Frequency Domain Plot

V. Conclusion

The flanging effect was successfully replicated in this project using MATLAB through different approach and concepts of Digital signal processing thought throughout the course of the semester. Studying and observing the behavior of the signals provide in depth understanding on the mathematical concept behind signal processing with the help of different built-in functions in the MATLAB software. Furthermore, the flanging effect works best with instruments, as it creates a distinct prolonged audio which provide additional melody in the background. However, when it is applied to voice audio it creates a distortion as the voice would.

VI. Outlook or future Study

The knowledge acquired throughout the process of creating this project could be use to further improve the project or may be used as reference for similar projects. The parameter in the function could also be added to increase customizability of the function. Graphic user interface could also be incorporated for non-technical user for user friend operation. Lastly, the addition of hardware may also provide a better functionality for the application for easier audio processing.

VII. Reference

[1] Fuentes Ros en Others, “Analysis of flanging and phasing algorithms in music technology”, 2019.

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[3] Ślęzak, D., Pal, S. K., Kang, B.-H., Gu, J., Kuroda, H., & Kim, T. (Eds.). (2009). *Signal Processing, Image Processing and Pattern Recognition. Communications in Computer and Information Science.* doi:10.1007/978-3-642-10546-3

[4] Wright en V. Välimäki, “Neural Modeling of Phaser and Flanging Effects”, *Journal of the Audio Engineering Society*, vol 69, no 7/8, bll 517–529, 2021.

[5] J. Merlot, "What Is A Flanger? Explained!", *Reboot Recording*, 2022. [Online]. Available: https://rebootrecording.com/flanger/. [Accessed: 24- Feb- 2022].