Signals and System Project: Digital Synthesizer

Niyousha Dadkhah 401101687
Mehrshad Goljarian 401102401
Hossein Sanami 401102048

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1 MIDI

In order to read MIDI format, we need to decode it and extract neccesery informations out.
Below you can see a block of code that is supposed to decode this format

```
from mido import MidiFile

def read_midi_file(file_path):
    mid= MidiFile(r'C:\Users\golja\Desktop\file\midi.mid')
    for i,track in enumerate(mid.tracks):
        print(f'track {i}: {track.name}')
        for msg in track:
            print(msg)
```

```
import mido
import pygame

# Load a MIDI file
mid = mido.MidiFile(r'C:\Users\golja\Desktop\file\midi.mid')

# Initialize pygame mixer
pygame.mixer.init()

# Set the tempo and resolution
tempo = 500000  # Assuming the default tempo is 500000 microseconds per beat
resolution = 480  # MIDI standard resolution

# Iterate through the MIDI file and play the notes
for msg in mid.play():
    if msg.type == 'note_on':
        # Calculate the duration in seconds based on tempo, resolution, and time of the message
        time_in_seconds = mido.tick2second(msg.time, mid.ticks_per_beat, tempo)

        pygame.mixer.music.set_volume(0.5)  # Adjust volume if needed

        # Play the note with a fixed duration
        pygame.mixer.music.load(r'C:\Users\golja\Desktop\file\midi.mid')  # Replace with your sound file
        pygame.mixer.music.play()

        # Wait for the duration of the note before proceeding to the next message
        pygame.time.wait(int(time_in_seconds * 1000))
```

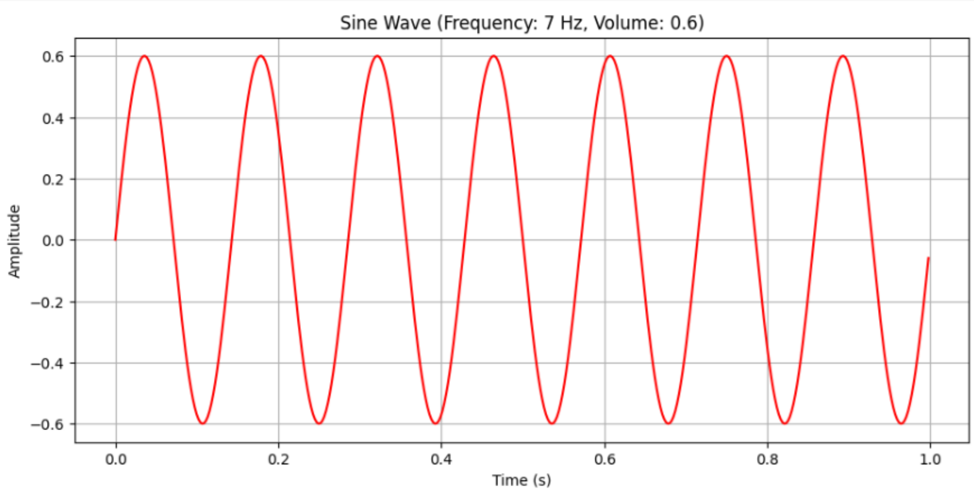
2 Synthesis

Different wave shapes are presented here:

2.1 Sine Wave

You can see the code and the wave form below :

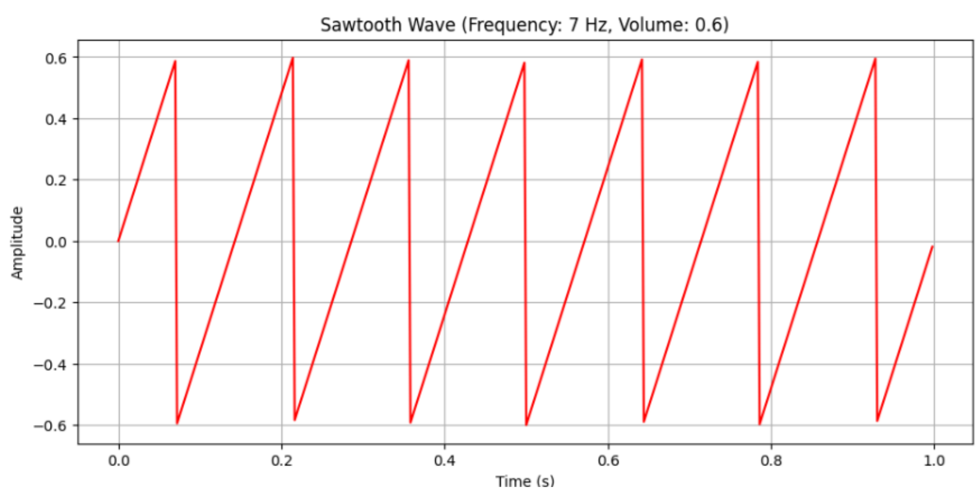
```
import numpy as np
import matplotlib.pyplot as plt

class Oscillator:
    def __init__(self, frequency, sampling_rate, volume):
        self.frequency = frequency
        self.sampling_rate = sampling_rate
        self.volume = volume
        self.phase = 0

    def generate_sine_wave(self, duration):
        num_samples = int(duration * self.sampling_rate)
        time = np.linspace(0, duration, num_samples, endpoint=False)
        sine_wave = self.volume * np.sin(2 * np.pi * self.frequency * time)
        return time, sine_wave

frequency = 7
sampling_rate = 441
duration = 1
volume = 0.6
oscillator = Oscillator(frequency, sampling_rate, volume)
time, sine_wave = oscillator.generate_sine_wave(duration)

plt.figure(figsize=(11, 5))
plt.plot(time, sine_wave,color='red')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.title('Sine Wave (Frequency: {} Hz, Volume: {})'.format(frequency, volume))
plt.grid(True)
plt.show()
```



2.2 Sawtooth Wave

You can see the code and the wave form below :

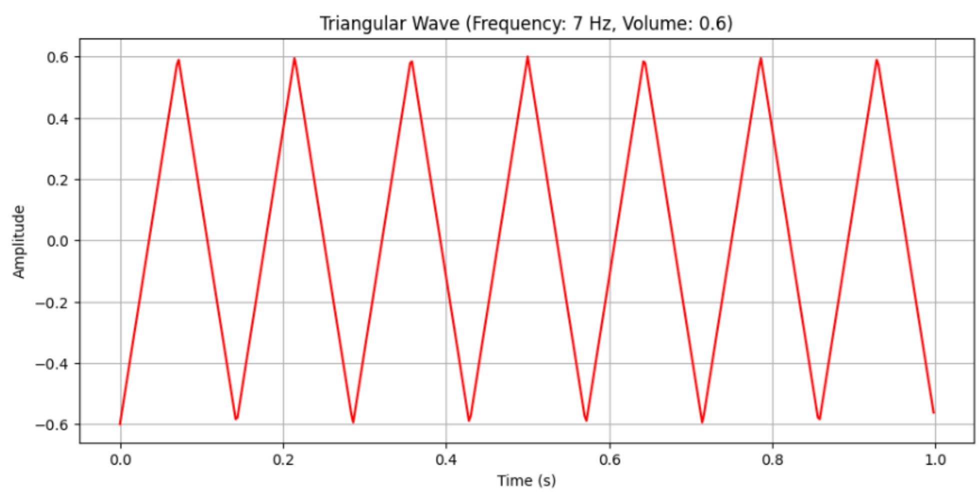
```
import numpy as np
import matplotlib.pyplot as plt

class Oscillator:
    def __init__(self, frequency, sampling_rate, volume):
        self.frequency = frequency
        self.sampling_rate = sampling_rate
        self.volume = volume
        self.phase = 0

    def generate_sawtooth_wave(self, duration):
        num_samples = int(duration * self.sampling_rate)
        time = np.linspace(0, duration, num_samples, endpoint=False)
        sawtooth_wave = self.volume * (2 * (self.frequency * time - np.floor(self.frequency * time + 0.5)))
        return time, sawtooth_wave

frequency = 7
sampling_rate = 444
duration = 1
volume = 0.6
oscillator = Oscillator(frequency, sampling_rate, volume)
time, sawtooth_wave = oscillator.generate_sawtooth_wave(duration)

plt.figure(figsize=(11, 5))
plt.plot(time, sawtooth_wave,color='red')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.title('Sawtooth Wave (Frequency: {} Hz, Volume: {})'.format(frequency, volume))
plt.grid(True)
plt.show()
```



2.3 Triangular Wave

You can see the code and the wave form below :

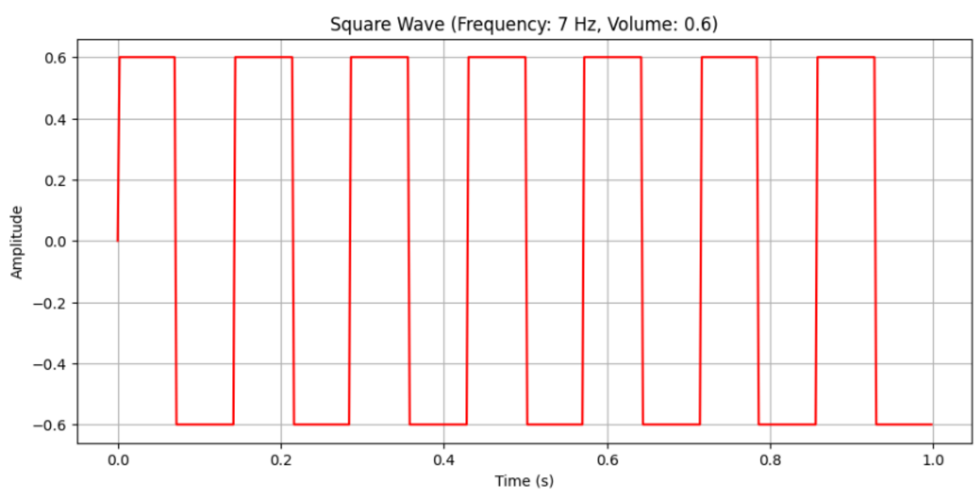
```
import numpy as np
import matplotlib.pyplot as plt

class Oscillator:
    def __init__(self, frequency, sampling_rate, volume):
        self.frequency = frequency
        self.sampling_rate = sampling_rate
        self.volume = volume
        self.phase = 0

    def generate_triangular_wave(self, duration):
        num_samples = int(duration * self.sampling_rate)
        time = np.linspace(0, duration, num_samples, endpoint=False)
        triangular_wave = self.volume * (2 * np.abs(2 * (time * self.frequency - np.floor(time * self.frequency + 0.5))) - 1)
        return time, triangular_wave

frequency = 7
sampling_rate = 444
duration = 1
volume = 0.6
oscillator = Oscillator(frequency, sampling_rate, volume)
time, triangular_wave = oscillator.generate_triangular_wave(duration)

plt.figure(figsize=(11, 5))
plt.plot(time, triangular_wave,color='red')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.title('Triangular Wave (Frequency: {} Hz, Volume: {})'.format(frequency, volume))
plt.grid(True)
plt.show()
```



2.4 Square Wave

You can see the code and the wave form below :

```
import numpy as np
import matplotlib.pyplot as plt

class Oscillator:
    def __init__(self, frequency, sampling_rate, volume):
        self.frequency = frequency
        self.sampling_rate = sampling_rate
        self.volume = volume
        self.phase = 0

    def generate_square_wave(self, duration):
        num_samples = int(duration * self.sampling_rate)
        time = np.linspace(0, duration, num_samples, endpoint=False)
        square_wave = self.volume * np.sign(np.sin(2 * np.pi * self.frequency * time))
        return time, square_wave

frequency = 7
sampling_rate = 444
duration = 1
volume =0.6

oscillator = Oscillator(frequency, sampling_rate, volume)
time, square_wave = oscillator.generate_square_wave(duration)
plt.figure(figsize=(11, 5))
plt.plot(time, square_wave,color='red')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.title('Square Wave (Frequency: {} Hz, Volume: {})'.format(frequency, volume))
plt.grid(True)
plt.show()
```



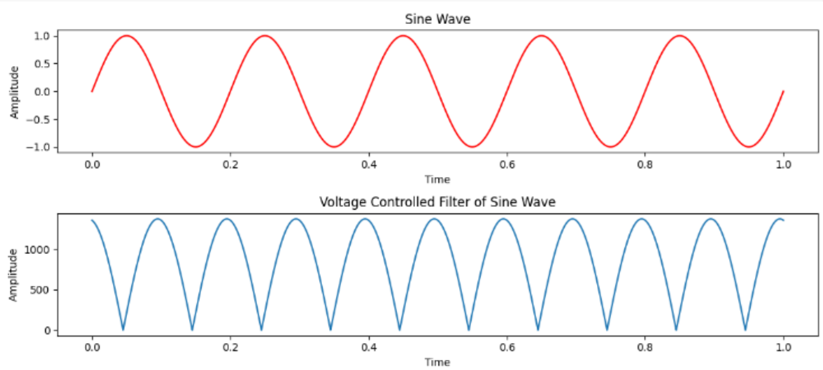
3 VCF: Voltage Controlled Filter

Below you can see the effect of VCF on different types of waves:

3.1 Sine Wave

You can see the code and the wave form below :

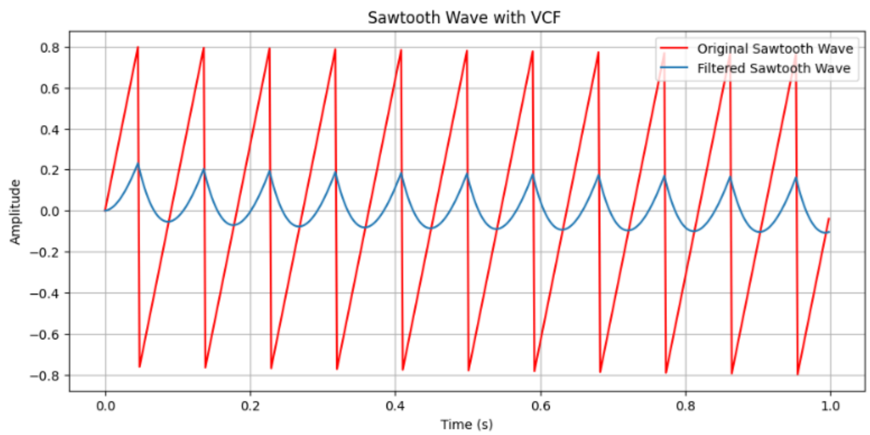
```
import numpy as np
import matplotlib.pyplot as plt

frequency = 5
sampling_rate = 44100
duration = 1
t = np.linspace(0, duration, int(sampling_rate * duration), endpoint=False)
sine_wave = np.sin(2 * np.pi * frequency * t)
cutoff_frequency = 5
vcf = np.abs(np.fft.ifft(np.fft.fft(sine_wave) * np.fft.fft(np.exp(-t * cutoff_frequency))))

plt.figure(figsize=(11, 5))
plt.subplot(2, 1, 1)
plt.plot(t, sine_wave,color='red')
plt.title('Sine Wave')
plt.xlabel('Time')
plt.ylabel('Amplitude')

plt.subplot(2, 1, 2)
plt.plot(t, vcf)
plt.title('Voltage Controlled Filter of Sine Wave')
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.tight_layout()
plt.show()
```



3.2 Sawtooth Wave

You can see the code and the wave form below :

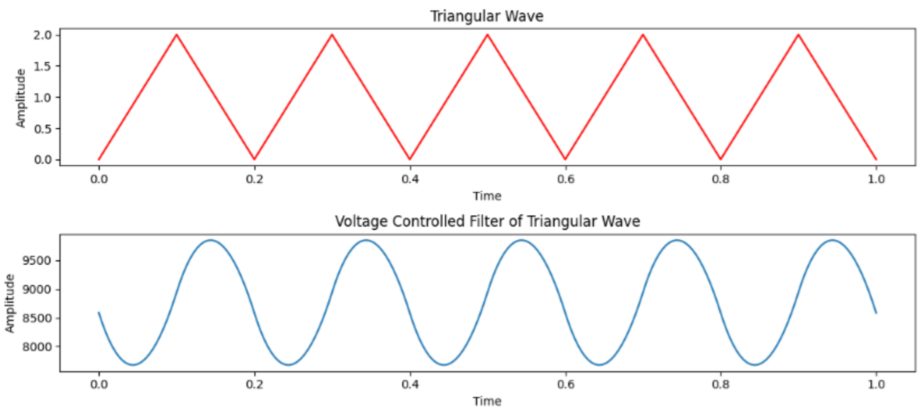
```
import matplotlib.pyplot as plt

class VCF:
    def __init__(self, cutoff_freq, resonance):
        self.cutoff_freq = cutoff_freq
        self.resonance = resonance

    def apply_filter(self, signal, sampling_rate):
        dt = 1.0 / sampling_rate
        RC = 1.0 / (2 * np.pi * self.cutoff_freq)
        alpha = RC / (RC + dt)

        output = np.zeros_like(signal)
        prev_output = 0

        for i in range(len(signal)):
            output[i] = alpha * (signal[i] - prev_output) + prev_output
            prev_output = output[i]

        return output

class Oscillator:
    def __init__(self, frequency, sampling_rate, volume):
        self.frequency = frequency
        self.sampling_rate = sampling_rate
        self.volume = volume
        self.phase = 0

    def generate_sawtooth_wave(self, duration):
        num_samples = int(duration * self.sampling_rate)
        time = np.linspace(0, duration, num_samples, endpoint=False)
        sawtooth_wave = self.volume * (2 * (self.frequency * time - np.floor(self.frequency * time + 0.5)))
        return time, sawtooth_wave

frequency = 11
sampling_rate = 441
duration = 1
volume = 0.8
cutoff_freq = 2000
resonance = 0.5

oscillator = Oscillator(frequency, sampling_rate, volume)
time, sawtooth_wave = oscillator.generate_sawtooth_wave(duration)

vcf = VCF(cutoff_freq, resonance)
filtered_wave = vcf.apply_filter(sawtooth_wave, sampling_rate)

plt.figure(figsize=(11, 5))
plt.plot(time, sawtooth_wave, label='Original Sawtooth Wave',color='red')
plt.plot(time, filtered_wave, label='Filtered Sawtooth Wave')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.title('Sawtooth Wave with VCF '.format(cutoff_freq, resonance))
plt.legend()
plt.grid(True)
plt.show()
```



3.3 Triangular Wave

You can see the code and the wave form below :

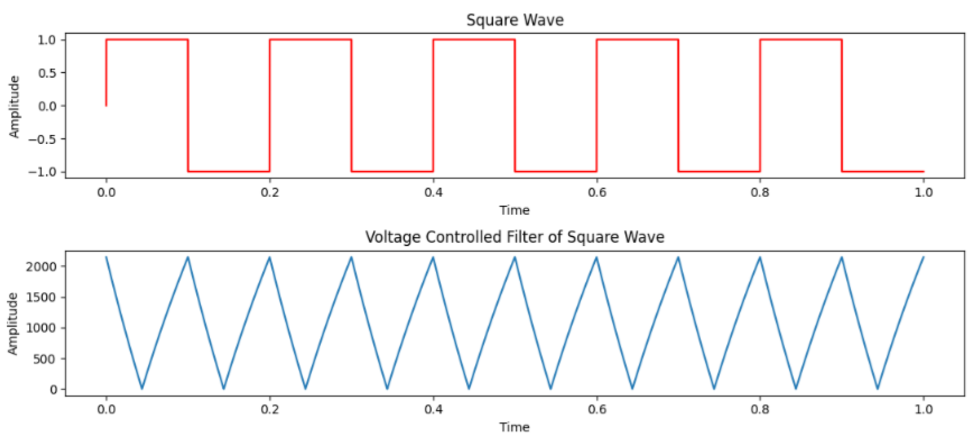
```
import numpy as np
import matplotlib.pyplot as plt

frequency = 5
sampling_rate = 44100
duration = 1
t = np.linspace(0, duration, int(sampling_rate * duration), endpoint=False)

triangular_wave = 2 * np.abs(2 * (t * frequency - np.floor(t * frequency + 0.5)))
vcf = np.abs(np.fft.ifft(np.fft.fft(triangular_wave) * np.fft.fft(np.exp(-t * cutoff_frequency))))

plt.figure(figsize=(11, 5))
plt.subplot(2, 1, 1)
plt.plot(t, triangular_wave,color='red')
plt.title('Triangular Wave')
plt.xlabel('Time')
plt.ylabel('Amplitude')

plt.subplot(2, 1, 2)
plt.plot(t, vcf)
plt.title('Voltage Controlled Filter of Triangular Wave')
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.tight_layout()
plt.show()
```



3.4 Square Wave

You can see the code and the wave form below :

```
import numpy as np
import matplotlib.pyplot as plt
frequency = 5
sampling_rate = 44100

duration = 1
t = np.linspace(0, duration, int(sampling_rate * duration), endpoint=False)
square_wave = np.sign(np.sin(2 * np.pi * frequency * t))
cutoff_frequency = 5
vcf = np.abs(np.fft.ifft(np.fft.fft(square_wave) * np.fft.fft(np.exp(-t * cutoff_frequency))))

plt.figure(figsize=(11, 5))
plt.subplot(2, 1, 1)
plt.plot(t, square_wave,color='red')
plt.title('Square Wave')
plt.xlabel('Time')
plt.ylabel('Amplitude')

plt.subplot(2, 1, 2)
plt.plot(t, vcf)
plt.title('Voltage Controlled Filter of Square Wave')
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.tight_layout()
plt.show()
```



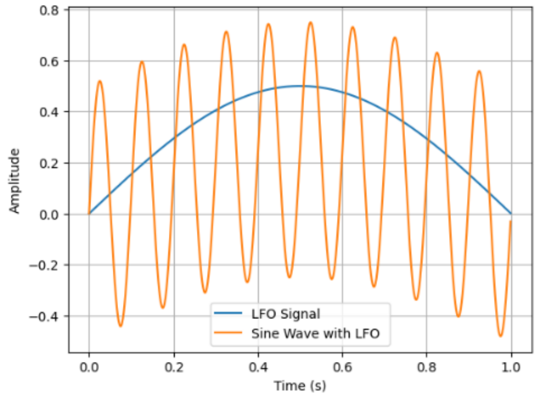
4 LFO

Below you can see the effect of LFO on different types of waves:

4.1 Sine Wave

You can see the code and the wave form below :

```
import numpy as np
import matplotlib.pyplot as plt

def generate_sine_wave(freq, amplitude, duration, sampling_rate):
    t = np.linspace(0, duration, int(sampling_rate*duration), endpoint=False)
    wave = amplitude * np.sin(2 * np.pi * freq * t)
    return wave

def apply_volume_control(signal, volume):
    return signal * volume

def apply_vcf(signal, cutoff_freq):
    filtered_signal = np.convolve(signal, [1, -1], mode='same')
    return filtered_signal

def generate_lfo(freq, amplitude, duration, sampling_rate):
    t = np.linspace(0, duration, int(sampling_rate*duration), endpoint=False)
    lfo = amplitude * np.sin(2 * np.pi * freq * t)
    return lfo

freq = 10
amplitude = 1
duration = 1
sampling_rate = 1000
volume = 0.5
cutoff_freq = 100
lfo_freq = 0.5
lfo_amplitude = 0.5

sine_wave = generate_sine_wave(freq, amplitude, duration, sampling_rate)
lfo_signal = generate_lfo(lfo_freq, lfo_amplitude, duration, sampling_rate)
modulated_wave = sine_wave + lfo_signal
modulated_wave_volume = apply_volume_control(modulated_wave, volume)
modulated_wave_vcf = apply_vcf(modulated_wave_volume, cutoff_freq)

plt.figure()
plt.plot(np.arange(len(lfo_signal))/sampling_rate, lfo_signal, label='LFO Signal')
plt.plot(np.arange(len(modulated_wave))/sampling_rate, modulated_wave_volume, label='Sine Wave with LFO ')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.legend()
plt.grid()
plt.show()
```

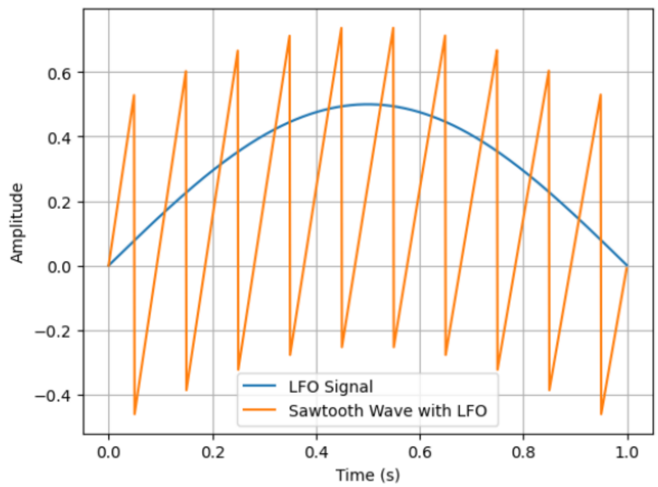


0.8
0.6
0.4
0.2
0.0
−0.2
−0.4
Amplitude
LFO Signal
Sine Wave with LFO
0.0
0.2
0.4
0.6
0.8
1.0
Time (s)

4.2 Sawtooth Wave

You can see the code and the wave form below :

```
import numpy as np
import matplotlib.pyplot as plt

def generate_sawtooth_wave(freq, amplitude, duration, sampling_rate):
    t = np.linspace(0, duration, int(sampling_rate*duration), endpoint=False)
    wave = amplitude * (2 * (t * freq - np.floor(t * freq + 0.5)))
    return wave

def apply_volume_control(signal, volume):
    return signal * volume

def apply_vcf(signal, cutoff_freq):
    filtered_signal = np.convolve(signal, [1, -1], mode='same')
    return filtered_signal

def generate_lfo(freq, amplitude, duration, sampling_rate):
    t = np.linspace(0, duration, int(sampling_rate*duration), endpoint=False)
    lfo = amplitude * np.sin(2 * np.pi * freq * t)
    return lfo

freq = 10
amplitude = 1
duration = 1
sampling_rate = 1000
volume = 0.5
cutoff_freq = 100
lfo_freq = 0.5
lfo_amplitude = 0.5

sawtooth_wave = generate_sawtooth_wave(freq, amplitude, duration, sampling_rate)
lfo_signal = generate_lfo(lfo_freq, lfo_amplitude, duration, sampling_rate)
modulated_wave = sawtooth_wave + lfo_signal
modulated_wave_volume = apply_volume_control(modulated_wave, volume)
modulated_wave_vcf = apply_vcf(modulated_wave_volume, cutoff_freq)

plt.figure()
plt.plot(np.arange(len(lfo_signal))/sampling_rate, lfo_signal, label='LFO Signal')
plt.plot(np.arange(len(modulated_wave))/sampling_rate, modulated_wave_volume, label='Sawtooth Wave with LFO ')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.legend()
plt.grid()
plt.show()
```

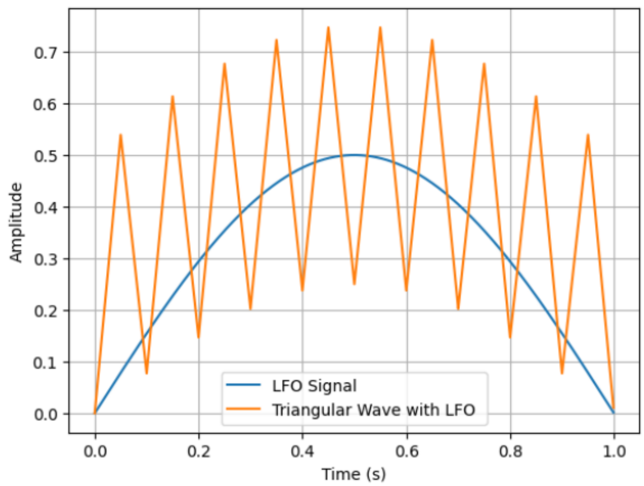


0.6
0.4
0.2
0.0
−0.2
−0.4
Amplitude
0.0
0.2
0.4
0.6
0.8
1.0
Time (s)
LFO Signal
Sawtooth Wave with LFO

4.3 Triangular Wave

You can see the code and the wave form below :

```
import numpy as np
import matplotlib.pyplot as plt

def generate_triangular_wave(freq, amplitude, duration, sampling_rate):
    t = np.linspace(0, duration, int(sampling_rate*duration), endpoint=False)
    wave = amplitude * np.abs(2 * (t * freq - np.floor(t * freq + 0.5)))
    return wave

def apply_volume_control(signal, volume):
    return signal * volume

def apply_vcf(signal, cutoff_freq):
    filtered_signal = np.convolve(signal, [1, -1], mode='same')
    return filtered_signal

def generate_lfo(freq, amplitude, duration, sampling_rate):
    t = np.linspace(0, duration, int(sampling_rate*duration), endpoint=False)
    lfo = amplitude * np.sin(2 * np.pi * freq * t)
    return lfo

freq = 10
amplitude = 1
duration = 1
sampling_rate = 1000
volume = 0.5
cutoff_freq = 100
lfo_freq = 0.5
lfo_amplitude = 0.5

triangular_wave = generate_triangular_wave(freq, amplitude, duration, sampling_rate)
lfo_signal = generate_lfo(lfo_freq, lfo_amplitude, duration, sampling_rate)
modulated_wave = triangular_wave + lfo_signal
modulated_wave_volume = apply_volume_control(modulated_wave, volume)
modulated_wave_vcf = apply_vcf(modulated_wave_volume, cutoff_freq)

plt.figure()
plt.plot(np.arange(len(lfo_signal))/sampling_rate, lfo_signal, label='LFO Signal')
plt.plot(np.arange(len(modulated_wave))/sampling_rate, modulated_wave_volume, label='Triangular Wave with LFO ')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.legend()
plt.grid()
plt.show()
```

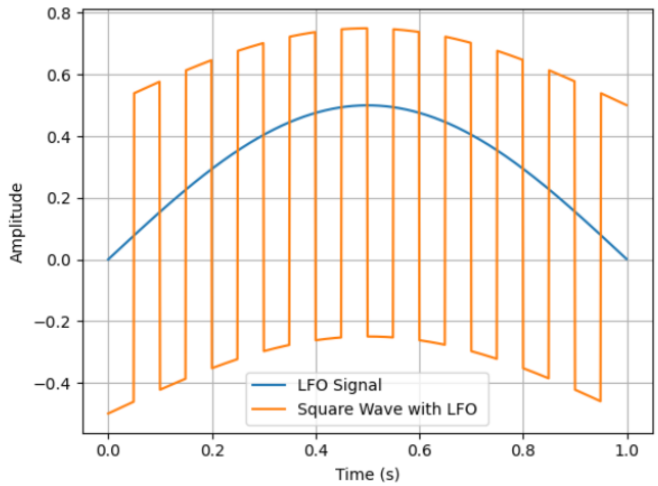


0.7
0.6
0.5
0.4
0.3
0.2
0.1
0.0
Amplitude
0.0
0.2
0.4
0.6
0.8
1.0
Time (s)
LFO Signal
Triangular Wave with LFO

4.4 Square Wave

You can see the code and the wave form below :

```
import numpy as np
import matplotlib.pyplot as plt

def generate_square_wave(freq, amplitude, duration, sampling_rate):
    t = np.linspace(0, duration, int(sampling_rate*duration), endpoint=False)
    wave = amplitude * (2 * (np.floor(2 * freq * t) % 2) - 1)
    return wave

def apply_volume_control(signal, volume):
    return signal * volume

def apply_vcf(signal, cutoff_freq):
    filtered_signal = np.convolve(signal, [1, -1], mode='same')
    return filtered_signal

def generate_lfo(freq, amplitude, duration, sampling_rate):
    t = np.linspace(0, duration, int(sampling_rate*duration), endpoint=False)
    lfo = amplitude * np.sin(2 * np.pi * freq * t)
    return lfo

freq = 10
amplitude = 1
duration = 1
sampling_rate = 1000
volume = 0.5
cutoff_freq = 100
lfo_freq = 0.5
lfo_amplitude = 0.5

square_wave = generate_square_wave(freq, amplitude, duration, sampling_rate)
lfo_signal = generate_lfo(lfo_freq, lfo_amplitude, duration, sampling_rate)
modulated_wave = square_wave + lfo_signal
modulated_wave_volume = apply_volume_control(modulated_wave, volume)
modulated_wave_vcf = apply_vcf(modulated_wave_volume, cutoff_freq)

plt.figure()
plt.plot(np.arange(len(lfo_signal))/sampling_rate, lfo_signal, label='LFO Signal')
plt.plot(np.arange(len(modulated_wave))/sampling_rate, modulated_wave_volume, label='Square Wave with LFO ')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.legend()
plt.grid()
plt.show()
```

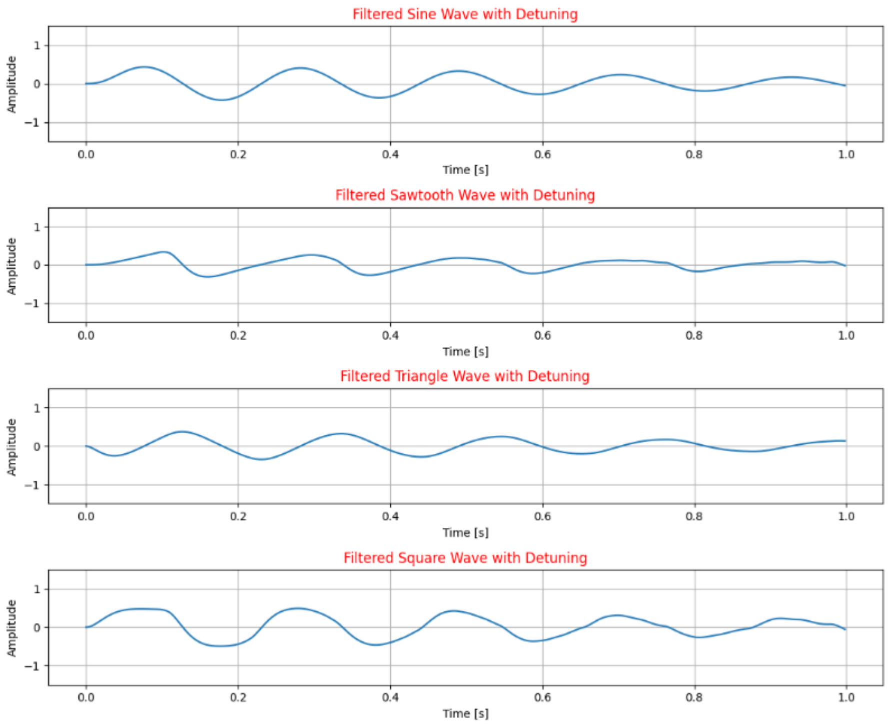


0.8
0.6
0.4
0.2
0.0
−0.2
−0.4
Amplitude
LFO Signal
Square Wave with LFO
0.0
0.2
0.4
0.6
0.8
1.0
Time (s)

5 Detuning

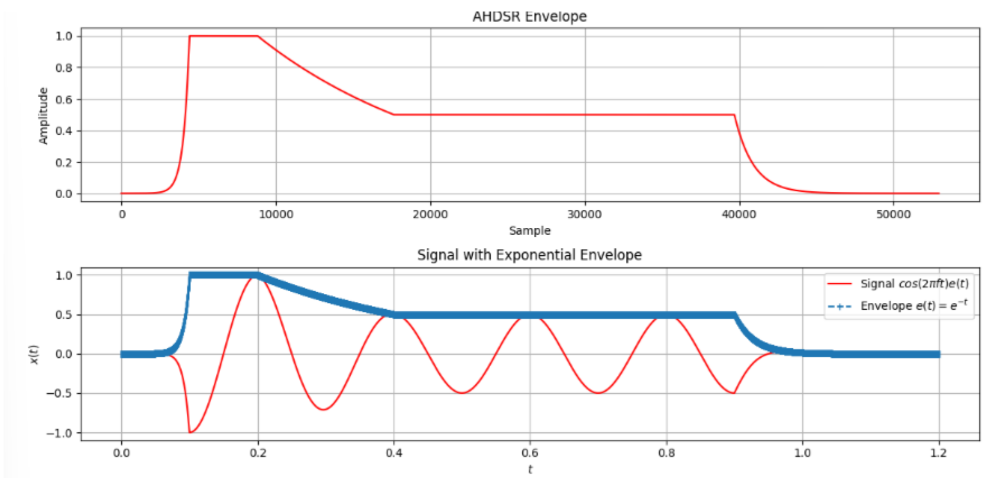
```python
import numpy as np
import matplotlib.pyplot as plt
from scipy.signal import butter, lfilter


class Oscillator:
    def __init__(self, shape='sine', phase=0, frequency=5, volume=1, detuning_factor=0, num_detuned=5):
        self.shape = shape
        self.phase = phase
        self.frequency = frequency
        self.volume = volume
        self.detuning_factor = detuning_factor
        self.num_detuned = num_detuned

    def generate_wave(self, duration=1, fs=500):
        t = np.linspace(0, duration, int(fs*duration), endpoint=False)
        base_freq = self.frequency
        wave = np.zeros_like(t)
        for i in range(self.num_detuned):
            detuned_freq = base_freq + self.detuning_factor * (i - self.num_detuned // 2)
            if self.shape == 'sine':
                partial_wave = np.sin(2 * np.pi * detuned_freq * t + self.phase)
            elif self.shape == 'sawtooth':
                partial_wave = 2 * (t * detuned_freq - np.floor(0.5 + t * detuned_freq))
            elif self.shape == 'triangle':
                partial_wave = 2 * np.abs(2 * (t * detuned_freq - np.floor(0.5 + t * detuned_freq))) - 1
            elif self.shape == 'square':
                partial_wave = np.sign(np.sin(2 * np.pi * detuned_freq * t + self.phase))
            else:
                raise ValueError("Shape not recognized. Use 'sine', 'sawtooth', 'triangle', or 'square'.")

            amplitude = self.volume / (i + 1)
            wave += amplitude * partial_wave
        return wave / self.num_detuned

def butter_lowpass(cutoff, fs, order=5):
    nyq = 0.5 * fs
    normal_cutoff = cutoff / nyq
    b, a = butter(order, normal_cutoff, btype='low', analog=False)
    return b, a

def butter_lowpass_filter(data, cutoff, fs, order=5):
    b, a = butter_lowpass(cutoff, fs, order=order)
    y = lfilter(b, a, data)
    return y

osc_sine = Oscillator(shape='sine', phase=0, frequency=5, volume=1, detuning_factor=0.2, num_detuned=5)
osc_saw = Oscillator(shape='sawtooth', phase=0, frequency=5, volume=1, detuning_factor=0.2, num_detuned=5)
osc_triangle = Oscillator(shape='triangle', phase=0, frequency=5, volume=1, detuning_factor=0.2, num_detuned=5)
osc_square = Oscillator(shape='square', phase=0, frequency=5, volume=1, detuning_factor=0.2, num_detuned=5)
duration = 1
fs = 500

wave_sine = osc_sine.generate_wave(duration, fs)
wave_saw = osc_saw.generate_wave(duration, fs)
wave_triangle = osc_triangle.generate_wave(duration, fs)
wave_square = osc_square.generate_wave(duration, fs)
cutoff = 10
order = 2

filtered_sine = butter_lowpass_filter(wave_sine, cutoff, fs, order)
filtered_saw = butter_lowpass_filter(wave_saw, cutoff, fs, order)
filtered_triangle = butter_lowpass_filter(wave_triangle, cutoff, fs, order)
filtered_square = butter_lowpass_filter(wave_square, cutoff, fs, order)

t = np.linspace(0, duration, int(fs*duration), endpoint=False)

fig, axs = plt.subplots(4, 1, figsize=(11, 9))

axs[0].plot(t, filtered_sine)
axs[0].set_title('Filtered Sine Wave with Detuning',color='red')
axs[0].set_ylim(-1.5, 1.5)
axs[0].grid(True)

axs[1].plot(t, filtered_saw)
axs[1].set_title('Filtered Sawtooth Wave with Detuning',color='red')
axs[1].set_ylim(-1.5, 1.5)
axs[1].grid(True)

axs[2].plot(t, filtered_triangle)
axs[2].set_title('Filtered Triangle Wave with Detuning',color='red')
axs[2].set_ylim(-1.5, 1.5)
axs[2].grid(True)

axs[3].plot(t, filtered_square)
axs[3].set_title('Filtered Square Wave with Detuning',color='red')
axs[3].set_ylim(-1.5, 1.5)
axs[3].grid(True)
```

```
for ax in axs:
    ax.set_xlabel('Time [s]')
    ax.set_ylabel('Amplitude')

plt.tight_layout()
plt.show()
```



6 Push Design

```
import numpy as np
import matplotlib.pyplot as plt

def exponential_segment(start_level, end_level, duration, sample_rate):
    t = np.linspace(0, duration, int(sample_rate * duration), endpoint=False)
    segment = start_level * np.exp(np.log(end_level / start_level) * t / duration)
    return segment

def ahdsr_envelope(attack, hold, decay, sustain_level, sustain_duration, release, sample_rate=44100):
    attack_segment = exponential_segment(1e-5, 1, attack, sample_rate)
    hold_segment = np.ones(int(sample_rate * hold))
    decay_segment = exponential_segment(1, sustain_level, decay, sample_rate)
    sustain_segment = np.ones(int(sample_rate * sustain_duration)) * sustain_level
    release_segment = exponential_segment(sustain_level, 1e-5, release, sample_rate)
    envelope = np.concatenate((attack_segment, hold_segment, decay_segment, sustain_segment, release_segment))
    return envelope

attack = 0.1
hold = 0.1
decay = 0.2
sustain_level = 0.5
sustain_duration = 0.5
release = 0.3
sample_rate = 44100

envelope = ahdsr_envelope(attack, hold, decay, sustain_level, sustain_duration, release, sample_rate)
t = np.linspace(0, len(envelope) / sample_rate, len(envelope), endpoint=False)
frequency = 5
signal = np.cos(2 * np.pi * frequency * t) * envelope

plt.figure(figsize=(12, 6))
plt.subplot(2, 1, 1)
plt.plot(envelope,color='red')
plt.title('AHDSR Envelope')
plt.xlabel('Sample')
plt.ylabel('Amplitude')
plt.grid(True)

plt.subplot(2, 1, 2)
plt.plot(t, signal, 'r', label='Signal $cos(2\pi ft)e(t)$')
plt.plot(t, envelope, '|--', label='Envelope $e(t) = e^{-t}$')
plt.title('Signal with Exponential Envelope')
plt.xlabel('$t$')
plt.ylabel('$x(t)$')
plt.legend()
plt.grid(True)

plt.tight_layout()
plt.show()
```


AHDSR Envelope
Amplitude
Sample
Signal with Exponential Envelope
Signal $cos(2\pi ft)e(t)$
Envelope $e(t) = e^{-t}$
$x(t)$
t

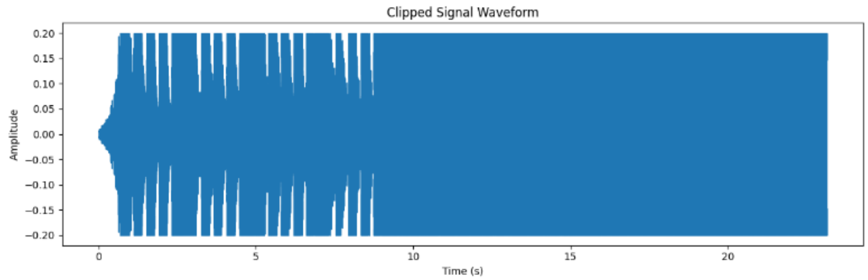
7 Effects

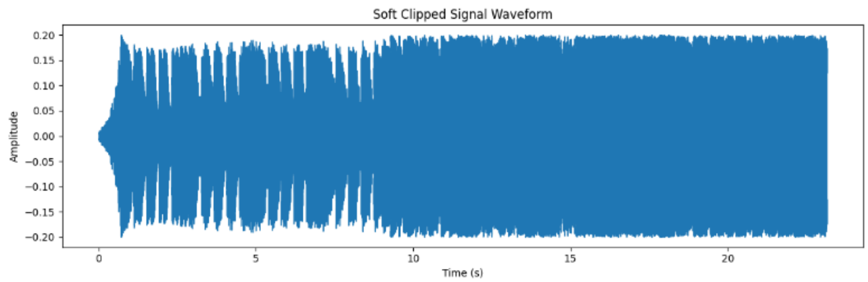
7.1 Saturation

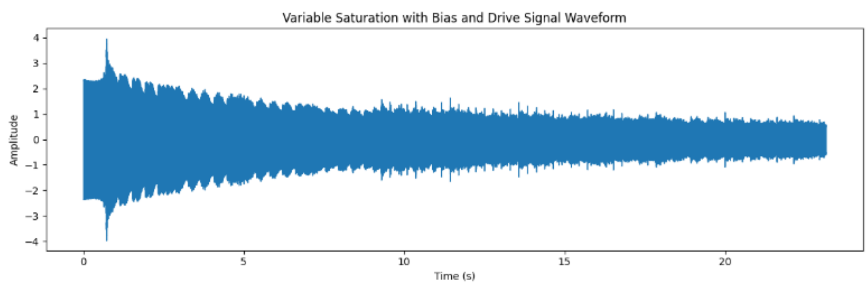
There are three different types of saturation effect. In order to add some warmth to the music. In the fallowing you can see three functions to generate three types of saturation and their effect on both music tracks and the signal plots.

```
 import os
import librosa
import numpy as np
import soundfile as sf
import matplotlib.pyplot as plt
import sounddevice as sd

def play_audio(audio_file):
    y, sr = librosa.load(audio_file, sr=None)
    sd.play(y, sr)
    status = sd.wait()

def hard_clip_audio(audio_file, threshold):
    y, sr = librosa.load(audio_file, sr=None)

    # Apply hard clipping saturation
    clipped_y = np.clip(y, -threshold, threshold)

    # Define the save path for the clipped audio file
    save_dir, save_filename = os.path.split(audio_file)
    save_path = os.path.join(save_dir, "hard-clipped_" + save_filename)

    # Save the clipped audio to a new file
    sf.write(save_path, clipped_y, sr)

    # Plot the signal waveform
    plt.figure(figsize=(12, 4))
    librosa.display.waveshow(clipped_y, sr=sr)
    plt.title('Clipped Signal Waveform')
    plt.xlabel('Time (s)')
    plt.ylabel('Amplitude')
    plt.tight_layout()
    plt.show()

    return save_path

# Example usage
audio_file = r'C:\Users\golja\Desktop\file\music.wav'
threshold = 0.2
clipped_audio_file = hard_clip_audio(audio_file, threshold)
def soft_clip_audio(audio_file, threshold):
    y, sr = librosa.load(audio_file, sr=None)

    # Apply soft clipping saturation
    clipped_y = np.tanh(y / threshold) * threshold

    # Define the path to save the clipped audio file
    save_dir, save_filename = os.path.split(audio_file)
    save_path = os.path.join(save_dir, "soft-clipped_" + save_filename)
    # Save the clipped audio to a new file
    sf.write(save_path, clipped_y, sr)

    # Plot the signal waveform
    plt.figure(figsize=(12, 4))
    librosa.display.waveshow(clipped_y, sr=sr)
    plt.title('Soft Clipped Signal Waveform')
    plt.xlabel('Time (s)')
    plt.ylabel('Amplitude')
```

```
    plt.tight_layout()
    plt.show()

    return save_path

# Example usage
audio_file = r'C:\Users\golja\Desktop\file\music.wav'
threshold = 0.2
clipped_audio_file = soft_clip_audio(audio_file, threshold)


def variable_saturation_with_bias_drive(audio_file, saturation_curve, bias, drive):
    y, sr = librosa.load(audio_file, sr=None)

    # Normalize saturation curve to be within the range [-1, 1]
    saturation_curve_normalized = 2 * ((saturation_curve - np.min(saturation_curve)) / (np.max(saturation_curve) - np.min(saturat

    # Apply variable saturation with bias and drive
    clipped_y = (y + bias) / (1 + drive * saturation_curve_normalized)

    # Define the path to save the clipped audio file
    save_dir, save_filename = os.path.split(audio_file)
    save_path = os.path.join(save_dir, "variable-clipped_" + save_filename)

    # Save the clipped audio to a new file
    sf.write(save_path, clipped_y, sr)


    # Plot the signal waveform
    plt.figure(figsize=(12, 4))
    librosa.display.waveshow(clipped_y, sr=sr)
    plt.title('Variable Saturation with Bias and Drive Signal Waveform')
    plt.xlabel('Time (s)')
    plt.ylabel('Amplitude')
    plt.tight_layout()
    plt.show()

    return save_path

# Example usage
audio_file = r'C:\Users\golja\Desktop\file\music.wav'
y, sr = librosa.load(audio_file, sr=None)
saturation_curve = np.linspace(0.5, 1.0, len(y))  # Example of a linear saturation curve
bias = 0.7
drive = 0.7

clipped_audio_file = variable_saturation_with_bias_drive(audio_file, saturation_curve, bias, drive)

print("Audio file after variable saturation with bias and drive saved as:", clipped_audio_file)

print("Audio file after soft clipping saturation saved as:", clipped_audio_file)
print("Audio file after hard clipping saturation saved as:", clipped_audio_file)
```


Clipped Signal Waveform
Amplitude
Time (s)


Soft Clipped Signal Waveform
Amplitude
Time (s)


Variable Saturation with Bias and Drive Signal Waveform
Amplitude
Time (s)

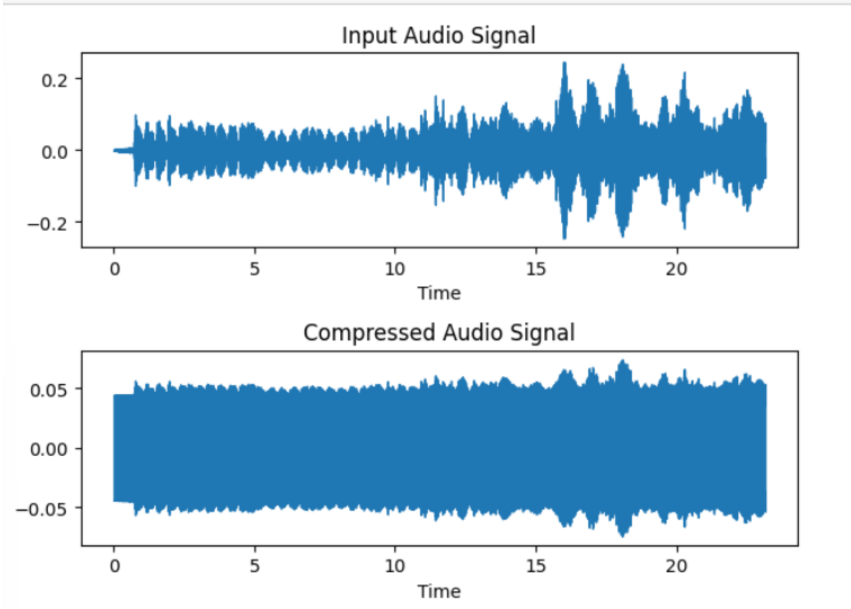
7.2 Compressor

In order to make more smooth sound tracks we can use compressor effect.

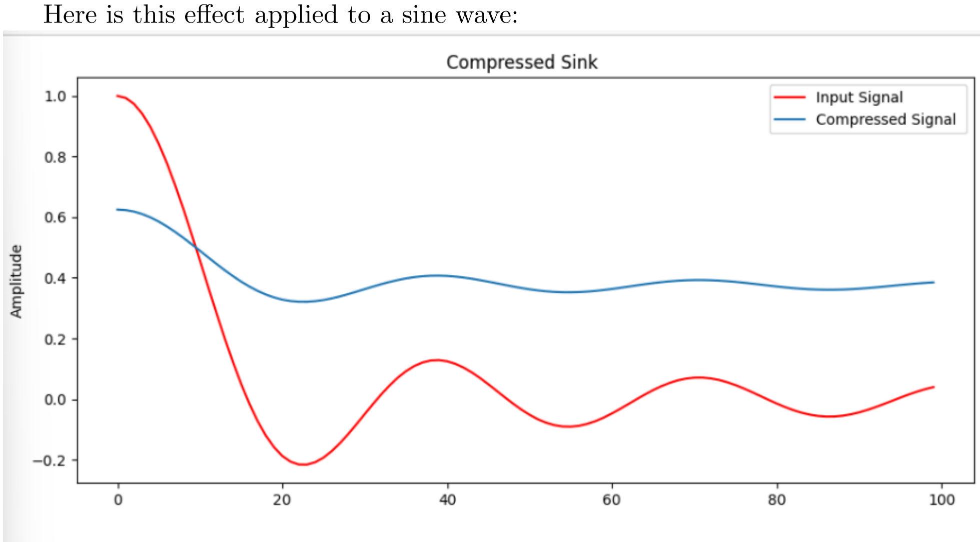
The fallowing code applies this effect to an audio track:

```
 import numpy as np
import matplotlib.pyplot as plt
import librosa
import librosa.display
import soundfile as sf

class Compressor:
    def __init__(self, threshold, ratio, attack, release, fs, N):
        self.threshold = threshold
        self.ratio = ratio
        self.attack = attack
        self.release = release
        self.fs = fs
        self.N = N
        self.gain = 0
        self.x_buffer = np.zeros(N)

    def update_gain(self, x):
        abs_x = np.abs(x)
        if abs_x > self.threshold:
            increment = -1/self.attack if x > 0 else 1/self.release
            self.gain += increment / self.fs
            if self.gain > 0:
                self.gain = 0
        else:
            self.gain = 0

    def compress(self, x):
        self.x_buffer[:-1] = self.x_buffer[1:]
        self.x_buffer[-1] = x
        mean_x = np.mean(np.abs(self.x_buffer))
        self.update_gain(mean_x)
        y = self.threshold + ((x - self.threshold) / self.ratio * np.power(10, self.gain/20))
        return y

compressor = Compressor(threshold=0.05, ratio=8, attack=20, release=40, fs=44100, N=10)
audio_file = r'C:\Users\golja\Desktop\file\music.wav'

audio_signal, sr = librosa.load(audio_file, sr=441)
output_signal = [compressor.compress(x) for x in audio_signal]
sf.write('compressed_audio.wav', np.array(output_signal), sr)

plt.figure()
plt.subplot(2, 1, 1)
librosa.display.waveshow(audio_signal, sr=sr)
plt.title('Input Audio Signal')

plt.subplot(2, 1, 2)
librosa.display.waveshow(np.array(output_signal), sr=sr)
plt.title('Compressed Audio Signal')
```

```
plt.tight_layout()
plt.show()
```

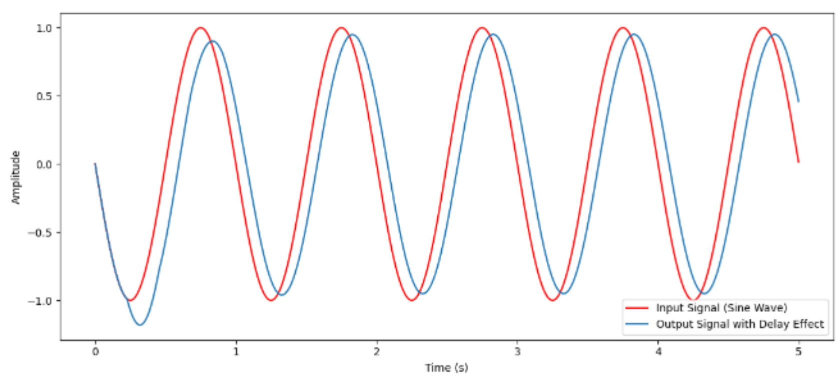


Here is this effect applied to a sine wave:

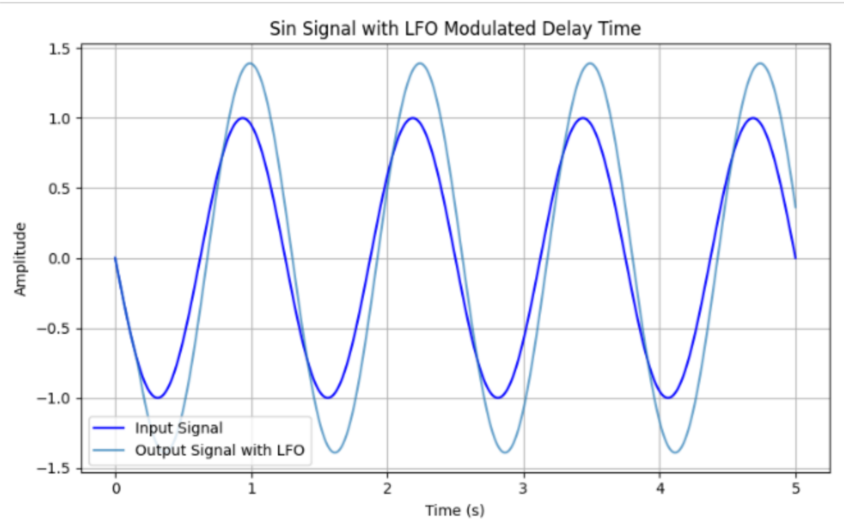


7.3 Delay

```
import numpy as np

def apply_delay_effect(signal, delay_samples, feedback_gain):
    output_signal = np.zeros_like(signal)

    for i in range(len(signal)):
        if i - delay_samples >= 0:
            output_signal[i] = signal[i] + feedback_gain * output_signal[i - delay_samples]
        else:
            output_signal[i] = signal[i]

    return output_signal

delay_samples = 100
feedback_gain = 0.5
fs = 441
duration = 5
t = np.linspace(0, duration, int(fs * duration), endpoint=False)
input_signal = np.sin(2 * np.pi * 440 * t)
output_signal = apply_delay_effect(input_signal, delay_samples, feedback_gain)
import matplotlib.pyplot as plt
plt.figure(figsize=(11, 5))
plt.plot(t, input_signal, label='Input Signal (Sine Wave)',color='red')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.legend()

plt.plot(t, output_signal, label='Output Signal with Delay Effect')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.legend()
plt.tight_layout()
plt.show()
```

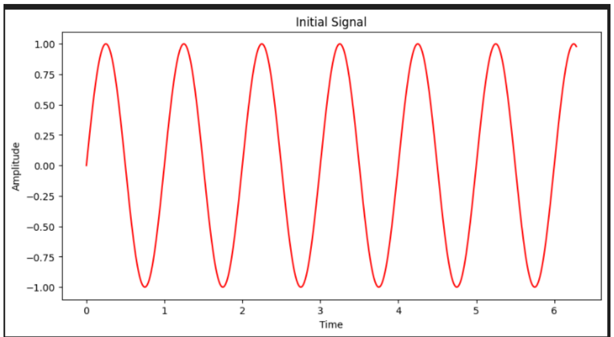


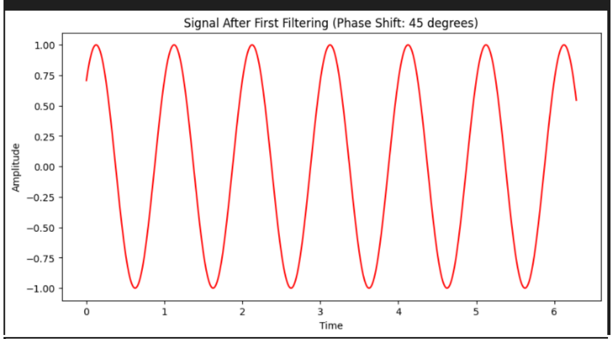
```
 import numpy as np
import matplotlib.pyplot as plt

fs = 441
n = 50
cutoff_freq = 20
delay_gain = 0.5
duration = 5
t = np.linspace(0, duration, int(fs * duration))
input_signal = np.sin(2 * np.pi * 440 * t)
lfo_freq = 0.7
lfo_amplitude = 20
lfo_signal = lfo_amplitude * np.sin(2 * np.pi * lfo_freq * t)
delay_time = n + lfo_signal

def apply_delay_vcf_echo_lfo(input_signal, delay_time, cutoff_freq, fs, delay_gain):
    output_signal = np.zeros_like(input_signal)
    delay_signal = np.concatenate((np.zeros(int(max(delay_time))), input_signal[:-int(max(delay_time))])) * delay_gain
    RC = 1/(2*np.pi*cutoff_freq)
    alpha = 1/(RC*fs)
    filtered_delay_signal = np.zeros_like(delay_signal)
    for i in range(1, len(delay_signal)):
        filtered_delay_signal[i] = alpha*delay_signal[i] + (1-alpha)*filtered_delay_signal[i-1]

    output_signal = input_signal + filtered_delay_signal
    return output_signal

output_signal_lfo = apply_delay_vcf_echo_lfo(input_signal, delay_time, cutoff_freq, fs, delay_gain)

plt.figure(figsize=(8, 5))
plt.plot(t, input_signal, label='Input Signal', color='blue')
plt.plot(t, output_signal_lfo, label='Output Signal with LFO',  alpha=0.7)
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.title('Sin Signal with LFO Modulated Delay Time')
plt.legend()
plt.grid(True)
plt.tight_layout()
plt.show()
```

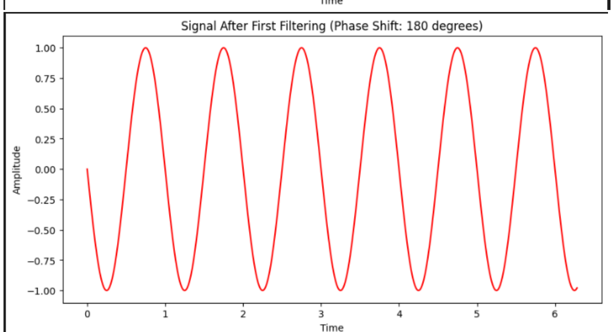


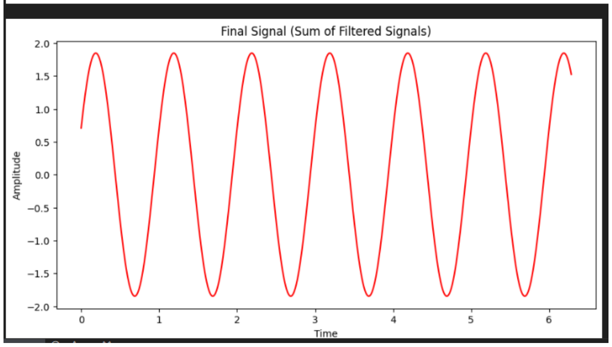
7.4 Phaser Effect

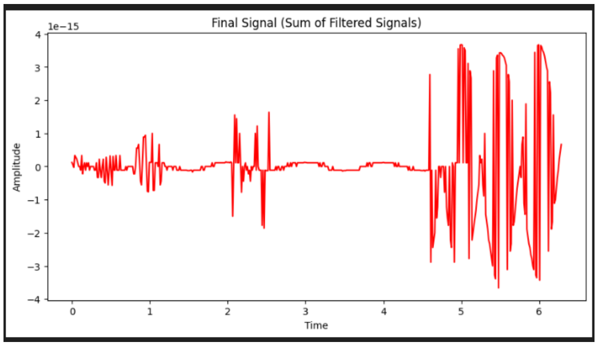
```
 import numpy as np
import matplotlib.pyplot as plt


freq = 440
phase = 0

t = np.linspace(0, 2*np.pi, 500)
A = 1
f = 1
phase = 0
sin_wave= A * np.sin(2 * np.pi * f * t + phase)
new_phase1 = np.pi/4
phase_shifted_sin_wave = np.sin(2 * np.pi * f * t + new_phase1)

new_phase2=np.pi
phase_shifted_sin_wave1= np.sin(2 * np.pi * f * t + new_phase2)
final_signal1 = sin_wave + phase_shifted_sin_wave
final_signal2 = sin_wave + phase_shifted_sin_wave1

plt.figure(figsize=(10, 5))
plt.plot(t, sin_wave,color='red')
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.title('Initial Signal')

plt.figure(figsize=(10, 5))
plt.plot(t, phase_shifted_sin_wave,color='red')
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.title('Signal After First Filtering (Phase Shift: 45 degrees)')

plt.figure(figsize=(10, 5))
plt.plot(t, phase_shifted_sin_wave1,color='red')
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.title('Signal After First Filtering (Phase Shift: 180 degrees)')

plt.figure(figsize=(10, 5))
plt.plot(t, final_signal1,color='red')
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.title('Final Signal (Sum of Filtered Signals)')

plt.figure(figsize=(10, 5))
plt.plot(t, final_signal2,color='red')
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.title('Final Signal (Sum of Filtered Signals)')

plt.show()
```


Initial Signal
Amplitude
1.00
0.75
0.50
0.25
0.00
−0.25
−0.50
−0.75
−1.00
0
1
2
3
4
5
6
Time


Signal After First Filtering (Phase Shift: 45 degrees)
Amplitude
1.00
0.75
0.50
0.25
0.00
−0.25
−0.50
−0.75
−1.00
0
1
2
3
4
5
6
Time


Signal After First Filtering (Phase Shift: 180 degrees)
Amplitude
1.00
0.75
0.50
0.25
0.00
−0.25
−0.50
−0.75
−1.00
0
1
2
3
4
5
6
Time


Final Signal (Sum of Filtered Signals)
Amplitude
2.0
1.5
1.0
0.5
0.0
−0.5
−1.0
−1.5
−2.0
0
1
2
3
4
5
6
Time


Final Signal (Sum of Filtered Signals)
1e−15
Amplitude
Time
4
3
2
1
0
−1
−2
−3
−4
0
1
2
3
4
5
6

7.5 Flanger

This effect basically generates a modulated delay of the music track and creates a jetplane like sound to the track.
This effect have been tested on both a music track and a short speech.
Here is the function codes for the electric guitar input:

```
def feedback_modulated_delay(data, modwave, dry, wet):
    out = data.copy()
    for i in range(len(data)):
        index = int(i - modwave[i])
        if index >= 0 and index < len(data):
            out[i] = out[i] * dry + out[index] * wet
    return out

def flanger(data, freq, dry=0.5, wet=0.5, depth=20.0, delay=1.0, rate=44100):
    length = float(len(data)) / rate
    mil = float(rate) / 1000
    delay *= mil
    depth *= mil
    modwave = (np.sin(2 * np.pi * freq * np.linspace(0, length, len(data))) / 2 + 0.5) * depth + delay
    return feedback_modulated_delay(data, modwave, dry, wet)


input_file_path = r'C:\Users\golja\Desktop\file\Electric.wav'
rate, data = wav.read(input_file_path)

data_with_flanger = flanger(data, freq=3.14159)

output_file_path = r'C:\Users\golja\Desktop\file\ElectricFlanger.wav'
wav.write(output_file_path, rate, np.int16(data_with_flanger))
```

7.6 Chorus

This effect basically adds delayed versions of the original track into the main one, so that the output sounds like more wide.
This effect have been tested on both a music track and a short speech.
Here is the function codes for the electric guitar input:

```
def modulated_delay(data, modwave, dry, wet):

    out = data.copy()
    for i in range(len(data)):
        index = int(i - modwave[i])
        if index >= 0 and index < len(data):
            out[i] = data[i] * dry + data[index] * wet
    return out
def chorus(data, freq, dry=0.5, wet=0.5, depth=5.0, delay=25.0, rate=44100):
    length = float(len(data)) / rate
    mil = float(rate) / 1000
    delay *= mil
    depth *= mil
    t = np.linspace(0.0, length, len(data), endpoint=False)
    modwave = (np.sin(2 * np.pi * freq * t) / 2 + 0.5) * depth + delay
    return modulated_delay(data, modwave, dry, wet)


input_file_path = r'C:\Users\golja\Desktop\file\Talk.wav'
rate, data = wav.read(input_file_path)

data_with_chorus = chorus(data, freq=3.14159/2)

output_file_path = r'C:\Users\golja\Desktop\file\talkWithChorus.WAV'
wav.write(output_file_path, rate, np.int16(data_with_chorus))
```

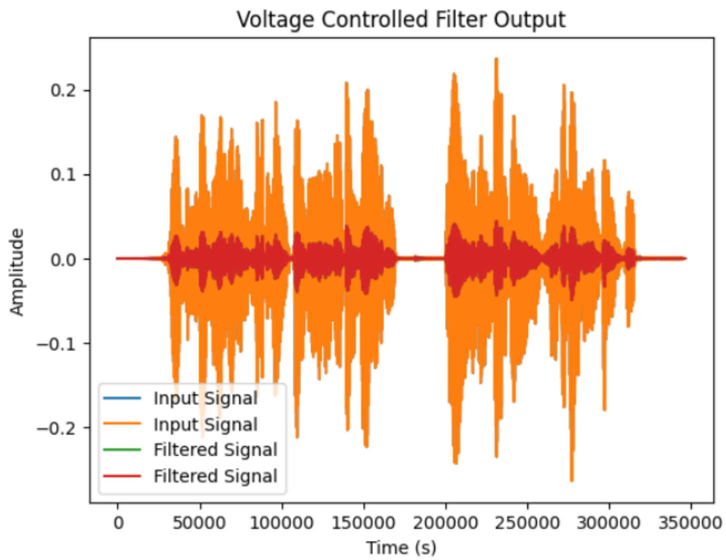
7.7 Reverberation

We implemented this effect by the second way, convolution of the sound and the desired impulse response.
Input and Output waves are available to play.

```
 rir, sample_rate = sf.read(r'C:\Users\golja\Desktop\file\Room1.wav')
rir = rir[int(sample_rate * 1.01):int(sample_rate * 1.3)]
rir = rir / np.linalg.norm(rir, 2)

speech, _ = sf.read(r'C:\Users\golja\Desktop\file\Electric.wav')

speech_with_reverb = signal.fftconvolve(speech, rir, mode='full')

sf.write(r'C:\Users\golja\Desktop\file\ElectricInRoom1Reverb.WAV', speech_with_reverb, sample_rate)
```

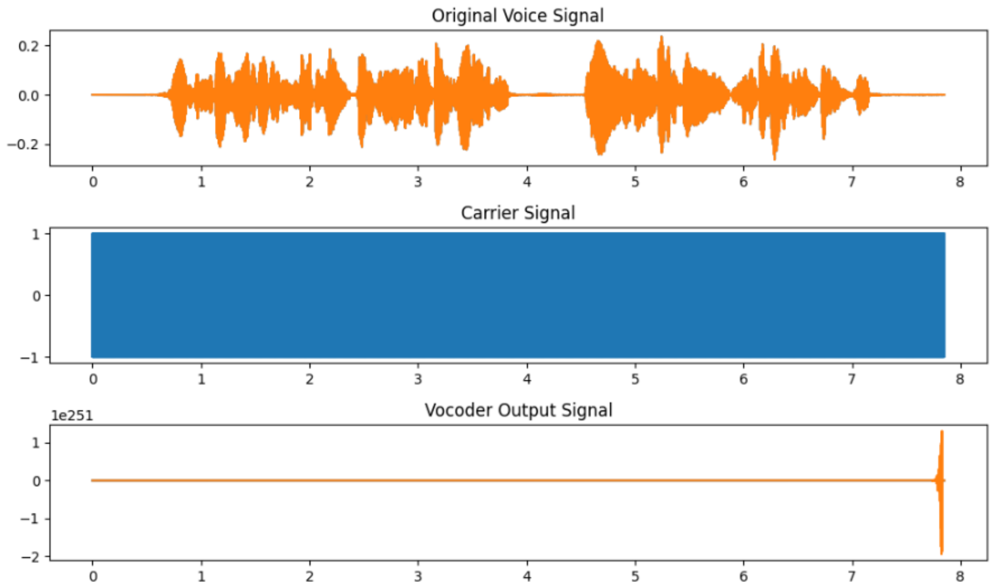
7.8 Voltage Controlled Filter

```
 import numpy as np
from scipy.signal import lfilter
import soundfile as sf

class VoltageControlledFilter:
    def __init__(self, fs):
        self.fs = fs
        self.g = 0.5
        self.y_prev = 0

    def update(self, x, cutoff, resonance):
        f0 = 2 * cutoff / self.fs
        damping = 2 * resonance
        g = np.tan(np.pi * f0)
        h = 1 / (1 + g * (g + damping))

        y = h * (x - self.y_prev + g * (1 + g) * self.y_prev)
        self.y_prev = y
        return self.g * y

fs = 441
cutoff_frequency = 500
resonance = 0.8

vcf = VoltageControlledFilter(fs)

# Read input audio file
input_signal, fs = sf.read(r'C:\Users\golja\Desktop\file\music1.wav')

# Apply the filter to the audio data
filtered_signal = np.array([vcf.update(xi, cutoff_frequency, resonance) for xi in input_signal])

# Write the filtered signal to a new audio file
sf.write("Voltage_Controlled_Filter.wav", filtered_signal, fs)

# Plot the filtered signal if needed
plt.plot(input_signal, label='Input Signal')
plt.plot(filtered_signal, label='Filtered Signal')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.title('Voltage Controlled Filter Output')
plt.legend()
plt.show()
```


Voltage Controlled Filter Output
Amplitude
Time (s)
0.2
0.1
0.0
−0.1
−0.2
0
50000
100000
150000
200000
250000
300000
350000
Input Signal
Input Signal
Filtered Signal
Filtered Signal

7.9 voice Encoder

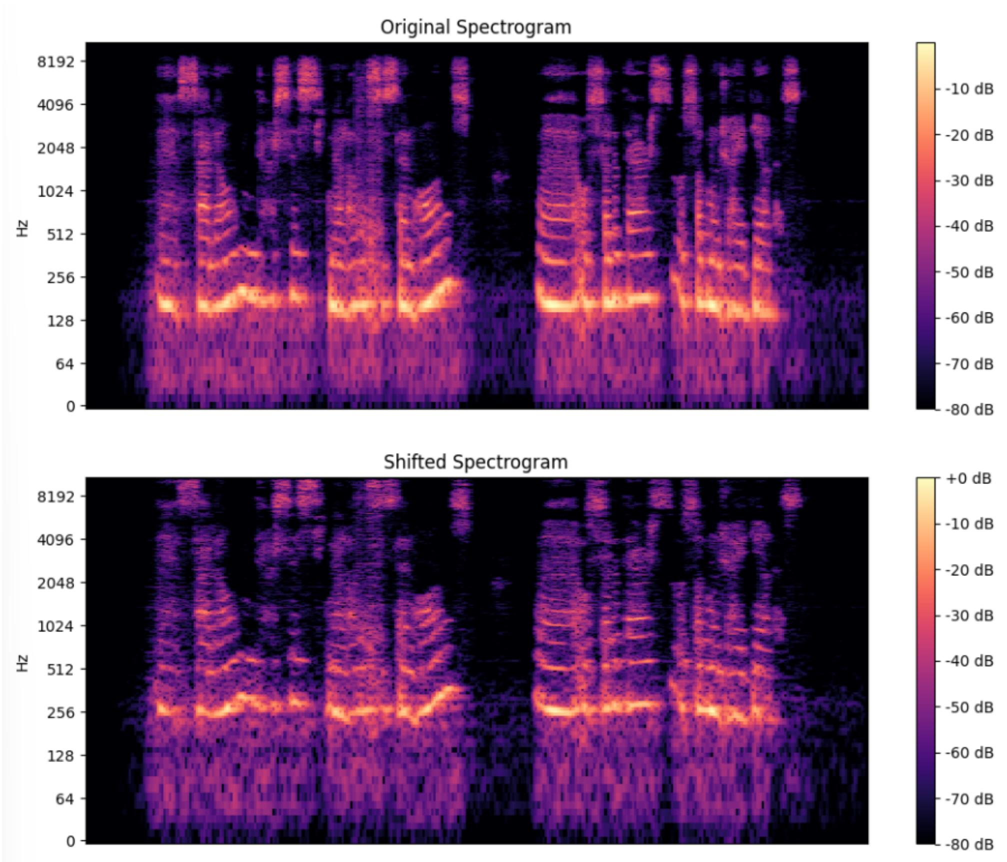
```
import numpy as np
import scipy.signal as signal
import matplotlib.pyplot as plt
import soundfile as sf

# Get input parameters from the user
fs = 26100  # Assuming standard audio sampling rate
n_bands = 20
f_low = 100
f_high = 8000  # Typical voice frequency range

# Load voice signal from a WAV file
voice_signal, fs_voice = sf.read(r'C:\Users\golja\Desktop\file\music1.wav')

# Generate a carrier signal (sine wave) based on the voice signal length
t = np.linspace(0, len(voice_signal) / fs_voice, len(voice_signal), endpoint=False)
carrier_signal = np.sin(2 * np.pi * 1000 * t)  # Use your desired carrier frequency

# Define frequency band limits
band_limits = np.logspace(np.log10(f_low), np.log10(f_high), n_bands + 1)

# Filter both voice and carrier signals in each band
filtered_voice = []
filtered_carrier = []

for i in range(n_bands):
    low_cutoff = band_limits[i]
    high_cutoff = band_limits[i + 1]
    b, a = signal.butter(4, [low_cutoff, high_cutoff], fs=fs, btype='band')
    filtered_voice.append(signal.lfilter(b, a, voice_signal))
    filtered_carrier.append(signal.lfilter(b, a, carrier_signal))

# Perform modulation in each band
# Perform modulation in each band
modulated_bands = []
for i in range(n_bands):
    # Modulate the carrier signal with each channel of the filtered voice
    modulated_band_channel1 = filtered_carrier[i][:len(filtered_voice[i])] * filtered_voice[i][:, 0]
    modulated_band_channel2 = filtered_carrier[i][:len(filtered_voice[i])] * filtered_voice[i][:, 1]
    modulated_bands.append(np.stack((modulated_band_channel1, modulated_band_channel2), axis=1))

# Recombine modulated bands
output_signal = np.sum(modulated_bands, axis=0)

# Ensure the output signal length matches the input signal length
if len(output_signal) < len(voice_signal):
    output_signal = np.pad(output_signal, (0, len(voice_signal) - len(output_signal)), 'constant')
elif len(output_signal) > len(voice_signal):
    output_signal = output_signal[:len(voice_signal)]

# Plot the original voice, carrier, and output signals
plt.figure(figsize=(10, 6))
plt.subplot(3, 1, 1)
plt.plot(t, voice_signal)
plt.title('Original Voice Signal')
plt.subplot(3, 1, 2)
plt.plot(t, carrier_signal[:len(voice_signal)])
plt.title('Carrier Signal')
plt.subplot(3, 1, 3)
plt.plot(t, output_signal)
plt.title('Vocoder Output Signal')
plt.tight_layout()
plt.show()

# Save the output signal as a WAV file
sf.write('vocoder_output.wav', output_signal, fs)
```



Original Voice Signal
0.2
0.0
−0.2
0
1
2
3
4
5
6
7
8
Carrier Signal
1
0
−1
0
1
2
3
4
5
6
7
8
1e251
Vocoder Output Signal
1
0
−1
−2
0
1
2
3
4
5
6
7
8

7.10 Pitch Shifter

```
 import numpy as np
import matplotlib.pyplot as plt
import librosa
import librosa.display
import soundfile as sf

def pitch_shift_and_visualize(audio_file, semitones):
    audio_signal, sampling_rate = librosa.load(audio_file)

    plt.figure(figsize=(10, 4))
    D = librosa.amplitude_to_db(np.abs(librosa.stft(audio_signal)), ref=np.max)
    librosa.display.specshow(D, y_axis='log', sr=sampling_rate, hop_length=512)
    plt.colorbar(format='%+2.0f dB')
    plt.title('Original Spectrogram')
    plt.tight_layout()
    plt.show()

    shifted_audio = librosa.effects.pitch_shift(audio_signal, sr=sampling_rate, n_steps=semitones)

    plt.figure(figsize=(10, 4))
    D_shifted = librosa.amplitude_to_db(np.abs(librosa.stft(shifted_audio)), ref=np.max)
    librosa.display.specshow(D_shifted, y_axis='log', sr=sampling_rate, hop_length=512)
    plt.colorbar(format='%+2.0f dB')
    plt.title('Shifted Spectrogram')
    plt.tight_layout()
    plt.show()
    output_file = audio_file.replace('.wav', '_shifted.wav')
    sf.write(output_file, shifted_audio, sampling_rate)
    return output_file

audio_file = r'C:\Users\golja\Desktop\file\music1.wav'
semitones = 8
output_file = pitch_shift_and_visualize(audio_file, semitones)
print("Pitch-shifted audio file saved as:", output_file)
```


Original Spectrogram
8192
4096
2048
1024
512
256
128
64
0
Hz
-10 dB
-20 dB
-30 dB
-40 dB
-50 dB
-60 dB
-70 dB
-80 dB
Shifted Spectrogram
8192
4096
2048
1024
512
256
128
64
0
Hz
+0 dB
-10 dB
-20 dB
-30 dB
-40 dB
-50 dB
-60 dB
-70 dB
-80 dB