Mini Project: Computer Music Generation and Playing

Introduction

Result and analysis

4. Exercise 4 , envelope attenuation

In real life, the vibration will experience attenuation and will not continue to vibrate at a fixed amplitude when playing instruments. For this reason, envelope attenuation functions are used to simulate music generation in real life.

This part is divided into two parts. The first part is to compare performances of different parameters in exponential attenuation and the second part is to compare performance of three different envelope attenuation functions, which are exponential attenuation, linear attenuation and square attenuation.

The input signal is shown below:

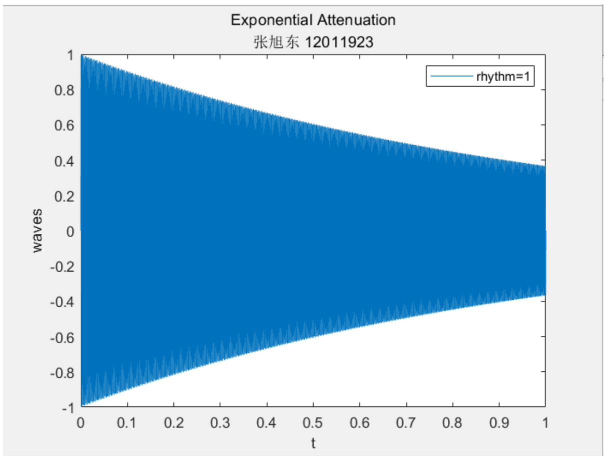
```
fs=8192;
f=440;
T=1/f;
rhythm=1;
t=linspace(0,rhythm,fs*rhythm);
y=sin(2*pi*f*t);
```

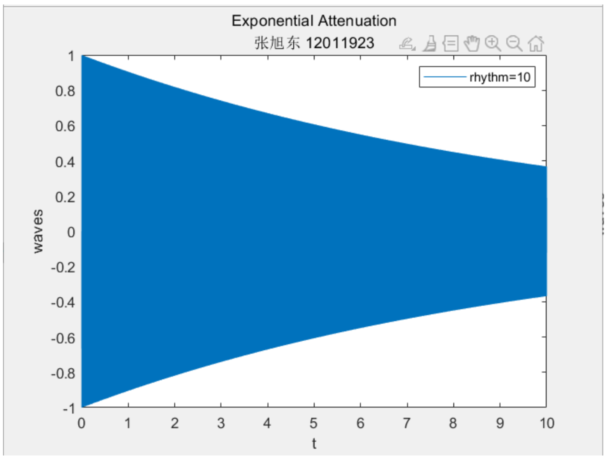
Exponential attenuation

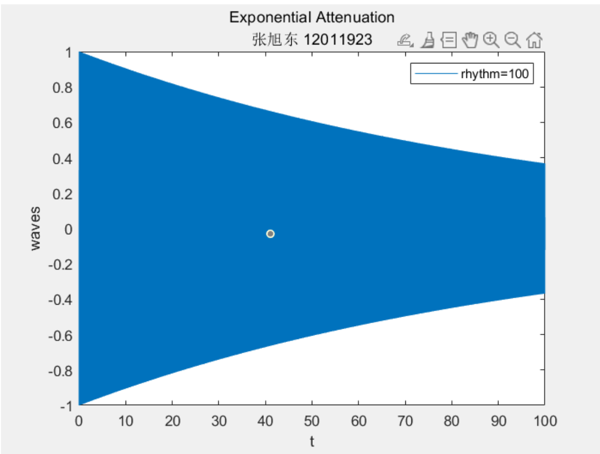
The envelope attenuation function is an exponential function which is :

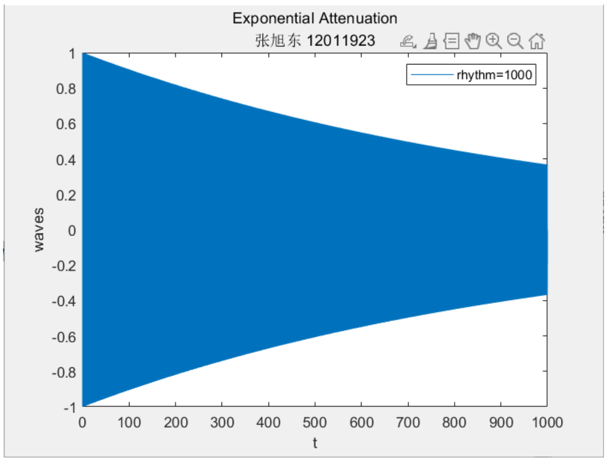
$$waves = y \times e^{-\frac{t}{rhythm}} \tag{1}$$

This function has been introduced in lab manual. Besides, what is found is that the shapes of final waves are very similar no matter rhythm increases. One possible reason is that attenuation is related to rhythm.

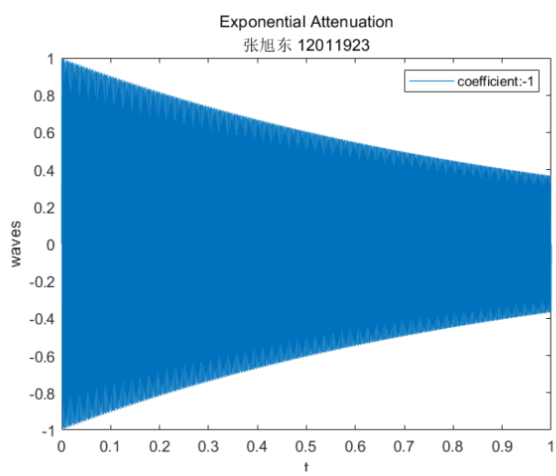


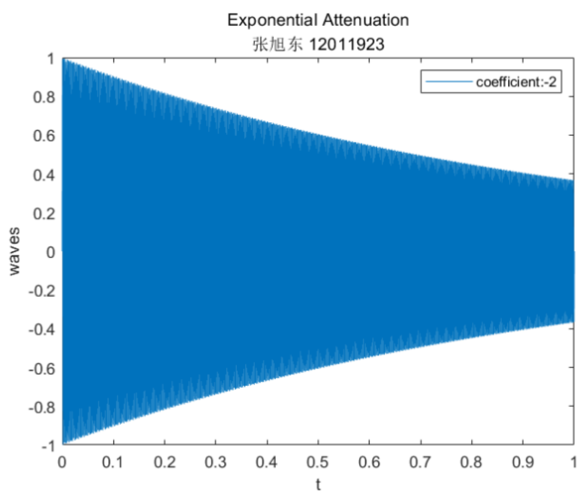


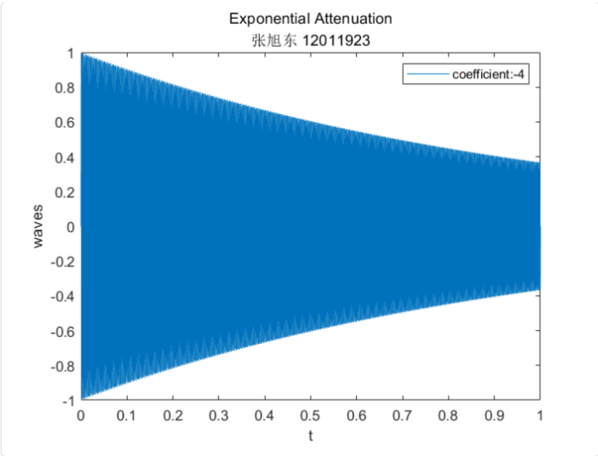


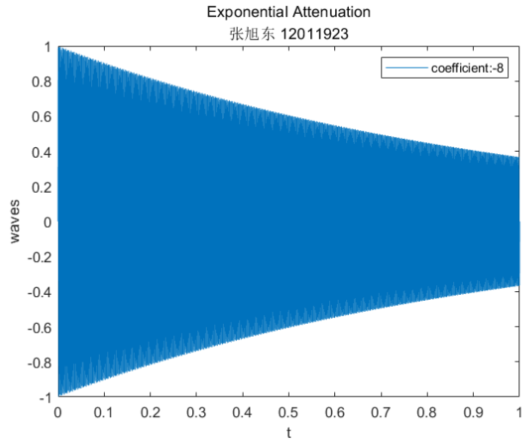


So, the value of rhythm can be set to the value you want. (In the following, the value of rhythm is 1.) And the coefficient of t is changed to $-1, -2, -4, -8, -16$ to compare the result. The waveforms are shown below:

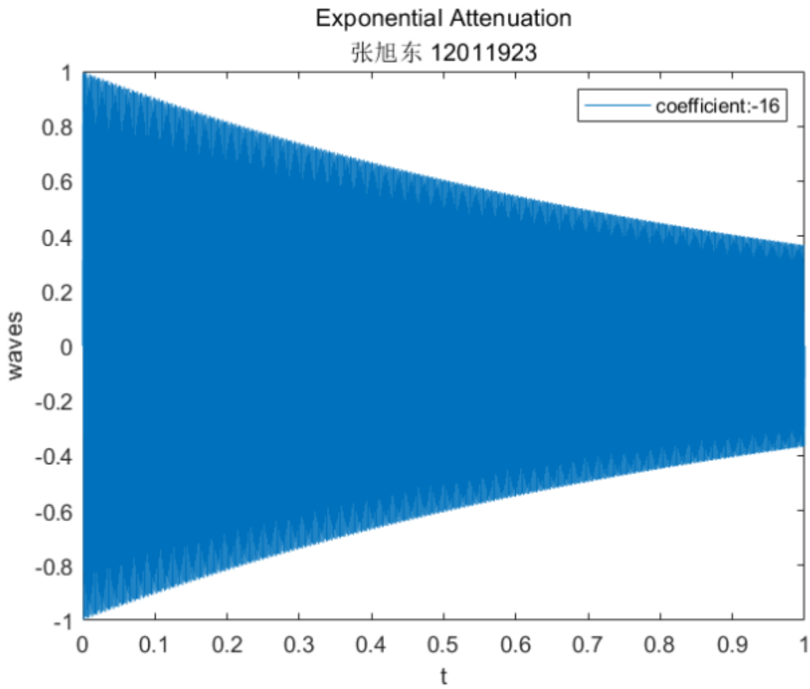








Exponential Attenuation



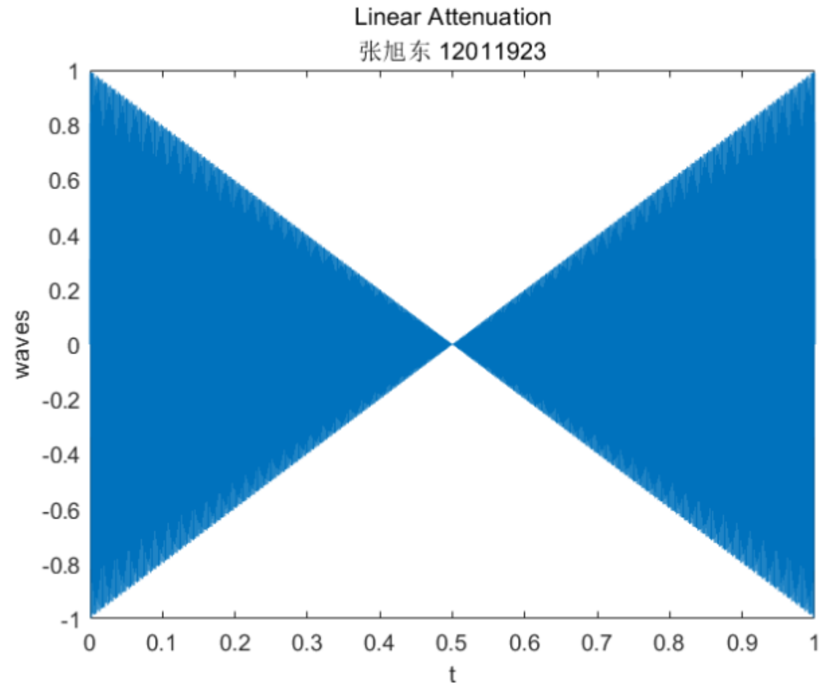
According to the sound, it is obvious that the speed of attenuation gets more and more faster with the coefficient of t increasing. What's more, the sound is like a "de" without almost the trails when the coefficient of t is 16. In my opinion, the second is the most similar to the note played in the real life of the above five. What has to be acknowledged is that the most appropriate coefficient of t may be not 2. There are more space to explore.

Linear attenuation

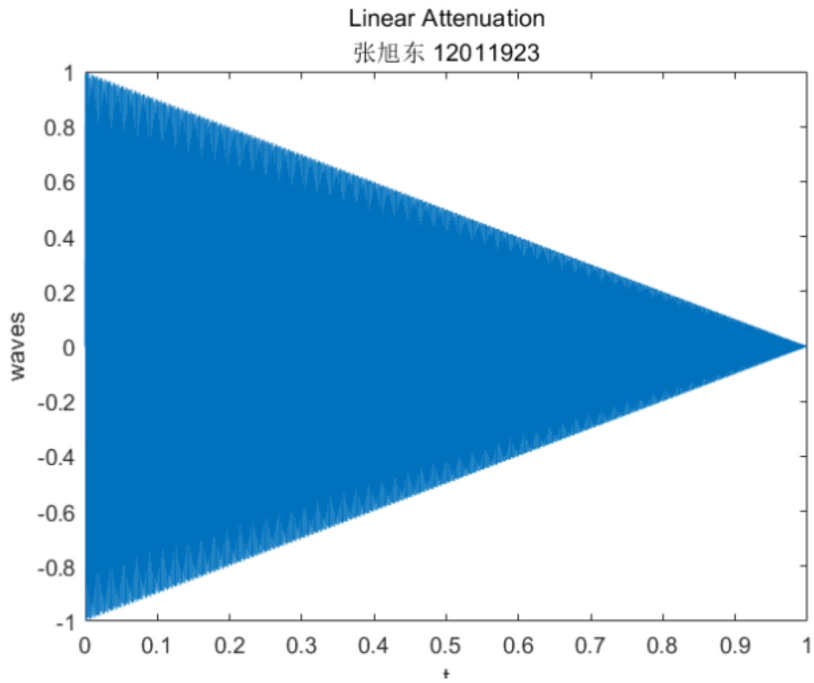
The envelope attenuation function is a linear function which is :

$$waves = y\times(1 - \frac{t}{rhythm}) \tag{2}$$

There is a detail needed to be considered. The coefficient of t can't be greater than1.Otherwise, the waveform is totally different from real life. The result is shown below when the coefficient of t is 2:



The waveform is shown below when the coefficient of t is 1 :



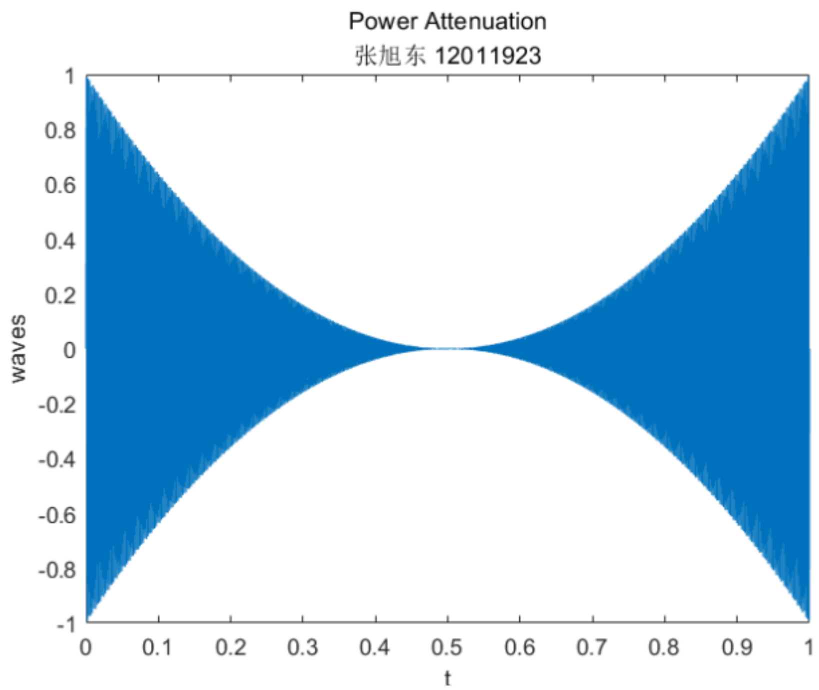
What can be seen is that the amplitude is decaying at a fixed rate, which is contrary to our common sense. Also, the sound is inanimate and lifeless.

Square attenuation

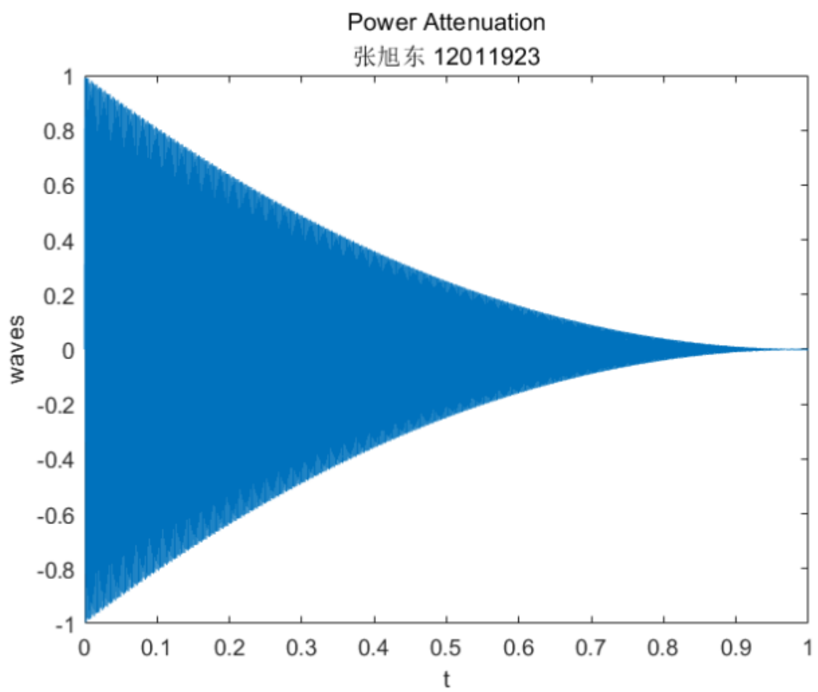
The envelope attenuation function is a linear function which is :

$$waves = y\times(1-\frac{t}{rhythm})^2 \tag{3}$$

Same as linear attenuation, The coefficient of t can't be greater than1.Otherwise, the waveform is totally different from real life. The result is shown below when the coefficient of t is 2:



The waveform is shown below when the coefficient of t is 1 :



It is pleasing that it is a little similar to the exponential attenuation. So, guess: the similarity of the two attenuation would increase with appropriate coefficient of t. One major different of the two attenuation is that square attenuation rate is less than the exponential attenuation for square grows slower than the exponential.

Then exponential attenuation function is used to simulate the generation of $Castle\ in\ the\ sky$. Compared to the sound which doesn't contain envelope attenuation, what is obvious is that the sound is smoother, exactly at the higher frequency. That means it sounds more supple, which indicates exponential attenuation function has a good effect in simulating music generation in the real life.

From what has been discussed above, in my opinion, the best attenuation function of the above three is exponential attenuation. What's more, as discussed in exponential attenuation, the coefficient of t has an effect on authenticity of simulation. There is a guess that the most appropriate coefficient is between in the interval "[2 4]" according to the performance of different coefficient of t.

5. Exercise 5 , harmonic wave

In real life, when playing instruments, in addition to the fundamental frequency, there are also varying numbers of standing waves due to the sound principle of music instruments. According to the principle of standing wave, the length of the string vibration must be an integer multiple of half wavelength, that means the frequency of the sound consists of the fundamental frequency and the integer multiple harmonic frequency of the fundamental frequency. The main energy is concentrated in the fundamental frequency. Different instruments have different proportions of harmonic energy, which generates waves with entirely different timbre. According to survey, what is found is that most of the existing research is about the digital imitation of piano sounds. What's more, the amplitude of harmonic waves of different notes is different. The `figure1` is the amplitude of harmonic waves at a simplified version while the `figure2` is the amplitude of harmonic waves at a detailed version. The simplified version and *do* notes of detailed version is chosen to do the following simulation.(which to choose is up to you.) Besides, assume the amplitude of the 7th harmonic wave is 0.

a^1	基波	二次谐波	三次谐波	四次谐波	五次谐波	六次谐波	…
频率/Hz	440.1	880.8	1320.6	1760.9	2201.2	2640.7	…
幅值	0.1262	0.0152	0.0062	0.0044	0.0059	0.0043	…

<u>figure1</u>

	基频		二次		三次		四次		五次		六次		七次	
	频率	幅值	频率	幅值	频率	幅值	频率	幅值	频率	幅值	频率	幅值	频率	幅值
do	261.9	0.044	523.6	0.081	786.1	0.017	1048	0.009	1313	0.006	1570	0.007	1832	0.013
re	293.8	0.077	587.2	0.043	882.3	0.011	1177	0.022	1467	0.019	1761	0.004	2055	0.009
mi	329.8	0.173	659.9	0.051	989.9	0.016	1318	0.005	1647	0.008	1977	0.007	2307	0.003
fa	349.3	0.134	699.3	0.035	1050	0.007	1396	0.007	1745	0.009	2095	0.005	2444	0.006
so	392.4	0.083	784.4	0.035	1179	0.006	1569	0.011	1646	0.009	1975	0.002	2746	0.004
la	440.2	0.127	880.9	0.013	1320	0.007	1760	0.005	2201	0.006	2640	0.005	3080	0.002
si	494.0	0.113	988.6	0.014	1482	0.006	1976	0.005	2469	0.006	2964	0.004	3458	0.002

figure2

This part is divided into two parts: the first part is the comparison of different amplitudes of harmonic waves about single tone and the second part is the comparison of different amplitude of harmonic waves about $Castle\ in\ the\ sky$. What's more, exponential attenuation function is added to both of them in order to simulate music more really, whose expression is:

$$waves = y \times e^{-\frac{2t}{rhythm}} \tag{4}$$

Single Tone

Two harmonic energy ratios is shown below:

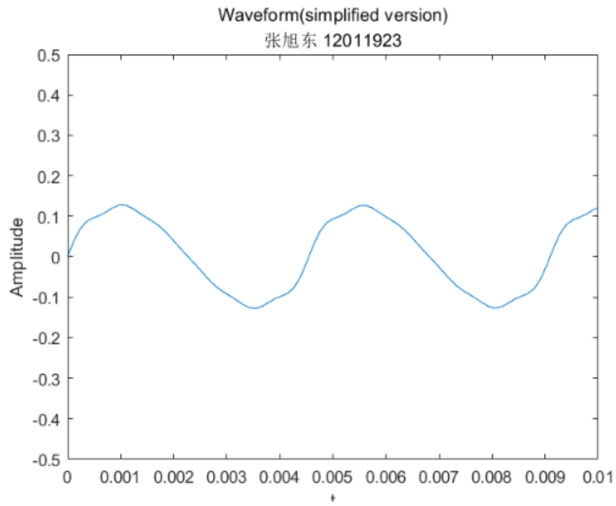
```
k1=[0.1262 0.0152 0.0062 0.0044 0.0059 0.0043 0];
k2=[0.044 0.081 0.017 0.009 0.006 0.007 0.013];
```

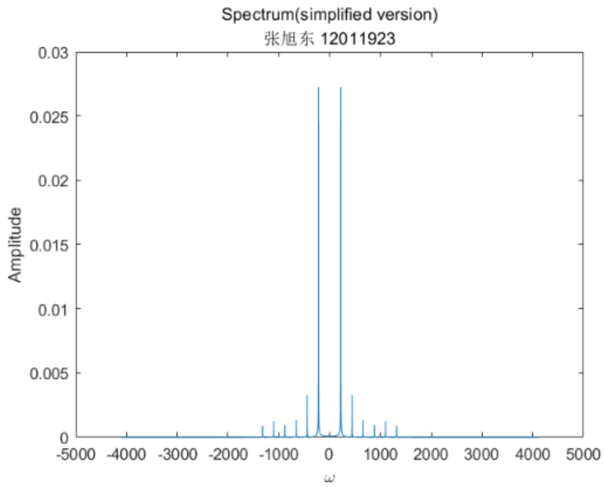
The MATLAB code is shown below:

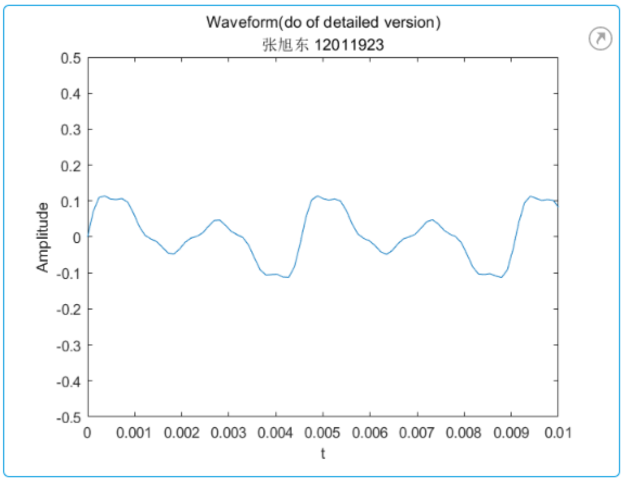
```
fs=8192;%采样率
fc=220;%基频
rhythm=1;%时宽
n=fs*rhythm;%采样点个数
t=linspace(0,rhythm,fs*rhythm);
f=linspace(-fs/2,fs/2-1,n);
k1=[0.1262 0.0152 0.0062 0.0044 0.0059 0.0043 0];
k2=[0.044 0.081 0.017 0.009 0.006 0.007 0.013];
```

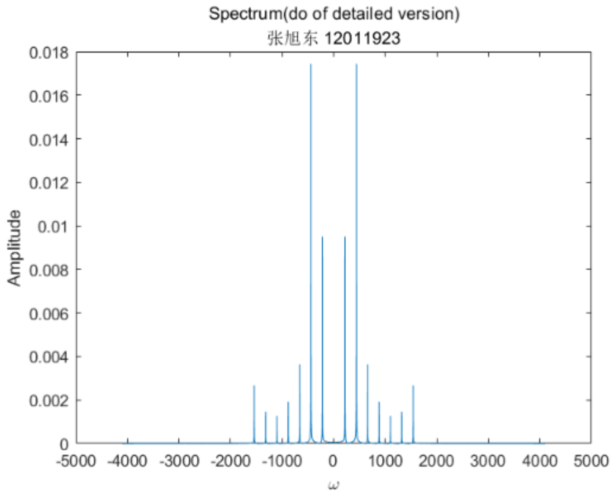
```
y1=0;
y2=0;
for i =1:length(k1)
    y1=y1+k1(i)*sin(2*pi*i*fc*t).*exp(-2*t/rhythm);
    y2=y2+k2(i)*sin(2*pi*i*fc*t).*exp(-2*t/rhythm);
end
plot(t,y1);
xlabel('t');
ylabel('Amplitude')
title(['Waveform(simplified version)' newline '张旭东 12011923'])
axis([0 0.01 -0.5 0.5]);
plot(f,abs(fftshift(fft(y1./n))));
xlabel('\omega');
ylabel('Amplitude')
title(['Spectrum(simplified version)' newline '张旭东 12011923'])
plot(t,y2);
xlabel('t');
ylabel('Amplitude')
title(['Waveform(do of detailed version)' newline '张旭东
12011923'])
axis([0 0.01 -1 1]);
plot(f,abs(fftshift(fft(y2./n))));
xlabel('\omega');
ylabel('Amplitude')
title(['Spectrum(do of detailed version)' newline '张旭东
12011923'])

sound(y1,fs);
pause(1.5);
sound(y2,fs);
```

There are some details needed to be considered. Firstly, when do `DFT`, the target function needs to be divided by the total number of samples. Secondly, according to the `Nyquist sampling theorem`, sample frequency f_s need to be greater than two times of signal frequency f_c.





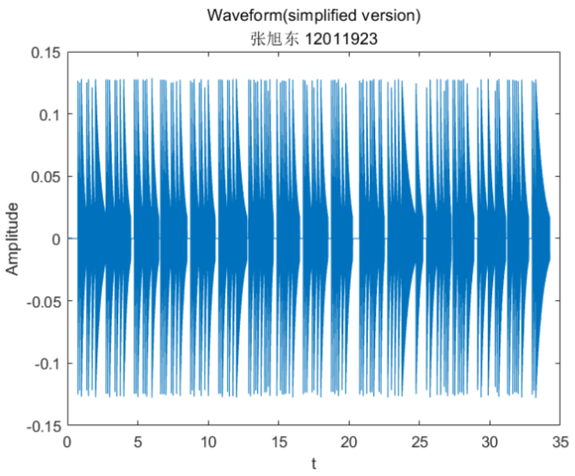


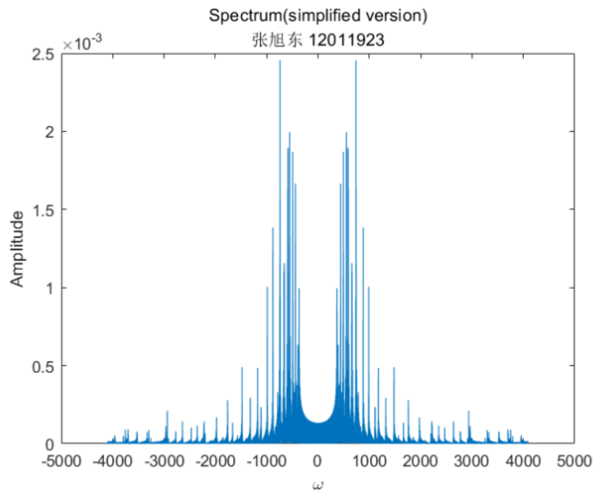


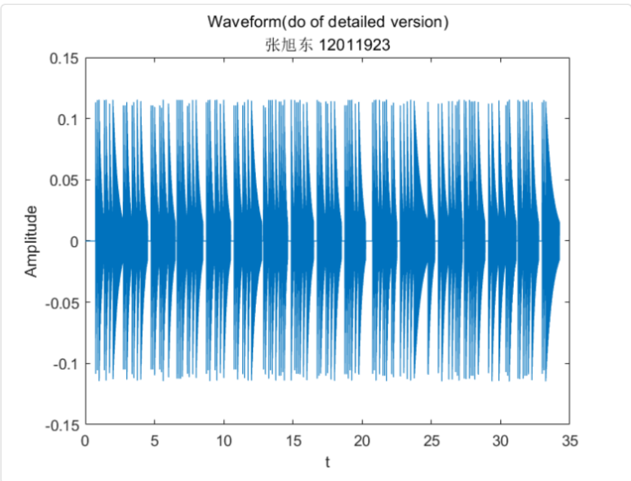
The first thing to do is to compare their waveform and it is obvious that the waveform of simplified version is smoother than the waveform of detailed version. The second thing to do is to compare their spectrum in frequency domain. It is concluded that the larger the amplitude of the i_{th} harmonic wave in time domain, the larger energy of the i_{th} harmonic wave in frequency domain, which is the same as the theory. At last, timbre difference between is analyzed. For simplified version, the started frequency is low and gives a soft feeling. For *do* of detailed version, the started frequency is high and gives a vigilant feeling. In my opinion, I prefer the first one.

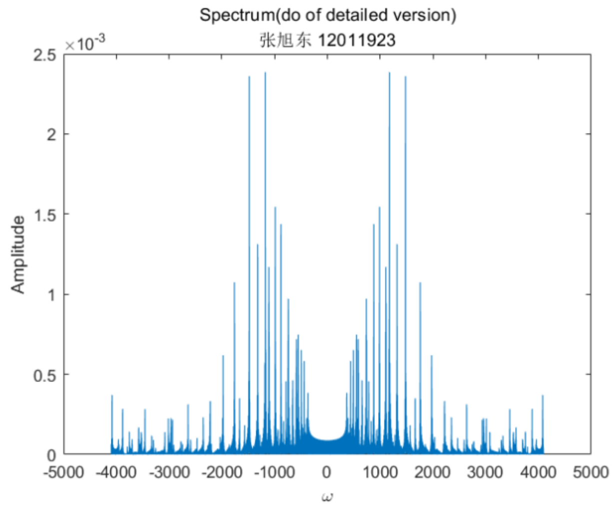
Castle in the sky

The two harmonic energy ratios is the same as that in single tone. The result is shown below.





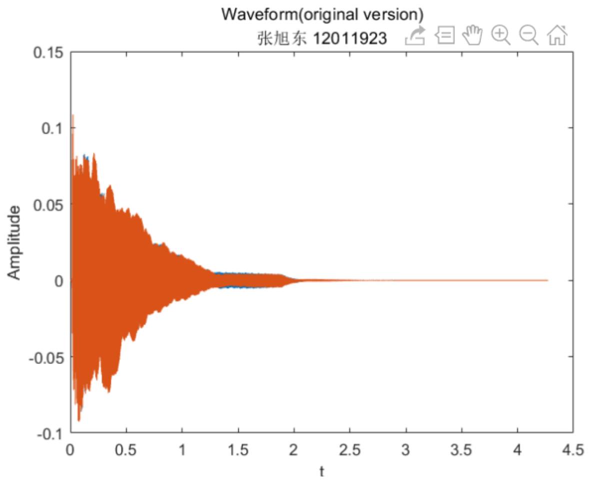


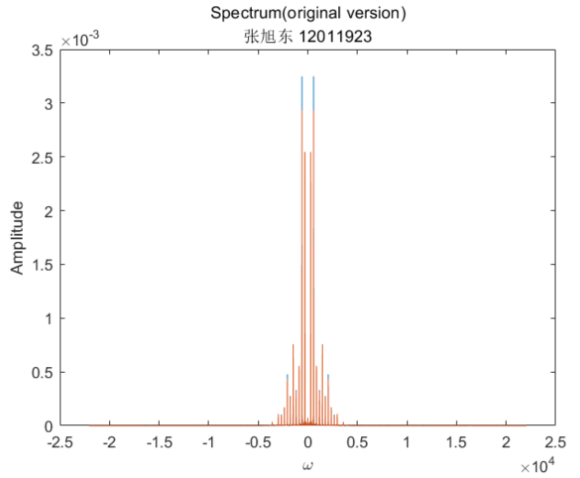


Analyzing from time domain, nothing can be gotten. However, there are lots of information in frequency domain. Firstly, most of energy is concentrated in low frequencies for simplified version and most of energy is concentrated in intermediate frequencies for *do* of detailed version. Secondly, energy distributed in high frequency for *do* of detailed version is more than that for simplified version. After that, timbre difference between is analyzed. For simplified version, the sound gives a light and ethereal feeling. For *do* of detailed version, the sound gives a stagnant and slightly hoarse feeling. Absolutely, I prefer the first one.

6.Simutation For Piano

Before simulating the piano, what is needed to do is download the standard $1 = D$ tone and analyze the waveform and spectrum of real piano. The figures are follows.

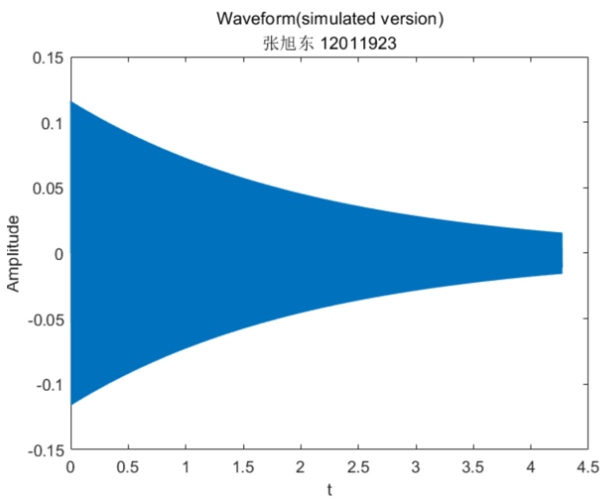


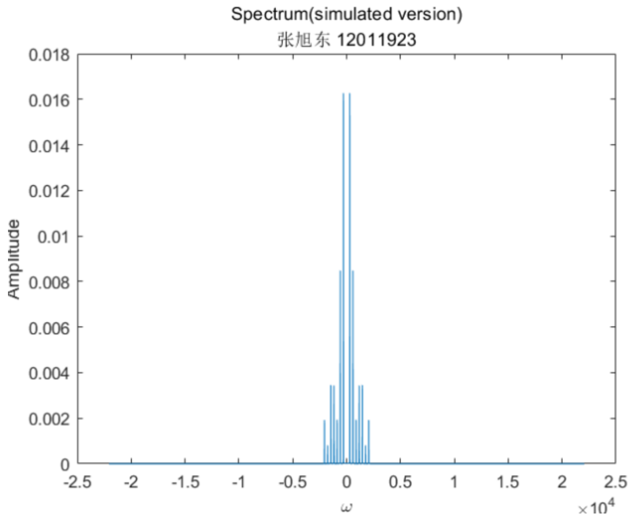


Then, the amplitude of harmonic waves for piano is found according to the reference. The amplitude of harmonic waves is:

```
k1=[0.077 0.043 0.011 0.022 0.019 0.004 0.009];
```

After that, exponential attenuation function, $waves = y \times e^{-\frac{2t}{rhythm}}$ is used to simulate the sound of piano. The waveform and spectrum of simulation are follows.





Starting analysis with waveform in time domain. It is obvious that the speed of attenuation of the waveform of real piano is faster than that of waveform of simulated sound. Also, the amplitude variation law of the former is relatively disordered relative to that of the latter. Then,analyze frequency domain. Most of the energy is concentrated in roughly the same range of frequencies. However, the bandwidth of the former is larger than that of the latter and the amplitude of the former is an order of magnitude smaller than that of the latter.

Conclusion

Reference

[1] 曹莎莎. 一种钢琴乐音仿真模型的研究[D].合肥工业大学,2017.

[2] 刘超.基于频谱包络的钢琴乐音仿真模型构建[D].咸阳师范学院，2021.