LAB5 MATLAB Programming

涂峻绫(12213010) and 欧阳安男(12211831)

Contents

Introduction

MATLAB is a powerful tool to process signals.

In this lab, we are going to learn and practice how to use $flter$, $butter$, $fir2$ and so on, in order to process signals in different filters.

Meanwhile, we will be more familar with handling signals with filters by processing the audio signal and the speed-shaped noise.

Results and Analysis

1

1. Generate a speech-shaped noise (SSN) and plot the spectra of the speech signal and SSN (e.g., use Matlab function 'pwelch' or other power spectral density estimation functions).

Load audio.

```
clear;clc;
[x, fs] = audioread('C_01_02.wav');
x = x';
```

Generate white noise.

```
whitenoise = 1-2*rand(1,length(x));
```

Generate long-term spectrum of speech signal to estimate the power spectral density of the speech signal.

```
sig = repmat(x,1,10);
[Pxx,f] =pwelch(sig, [], [], 512, fs);
```

Generate filter coefficients.

```
b = fir2(3000,f/(fs/2),sqrt(Pxx/max(Pxx)));
```

Perform filtering for white noise signal to get the SSN.

```
ssn = filter(b,1,whitenoise);
```

Get the spectra of the signal and SSN.

```
X = fftshift(fft(x)) / fs;
SSN = fftshift(fft(ssn)) / fs;
omega = linspace(-pi,pi,length(X)) * fs;
```

Plot the spectra of the signal and SSN.

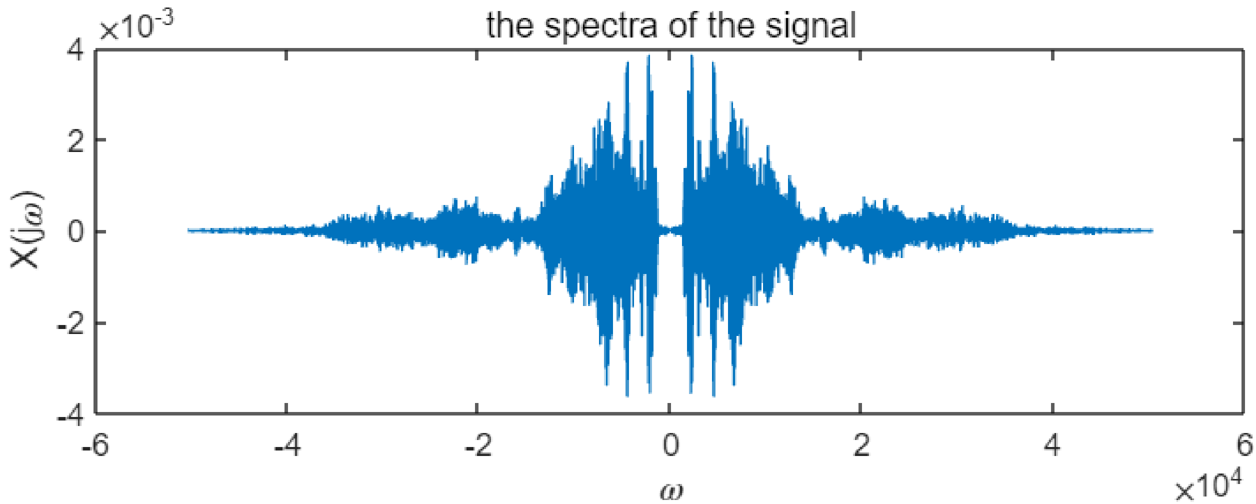
```
figure;
subplot(2,1,1);
plot(omega,X);
```

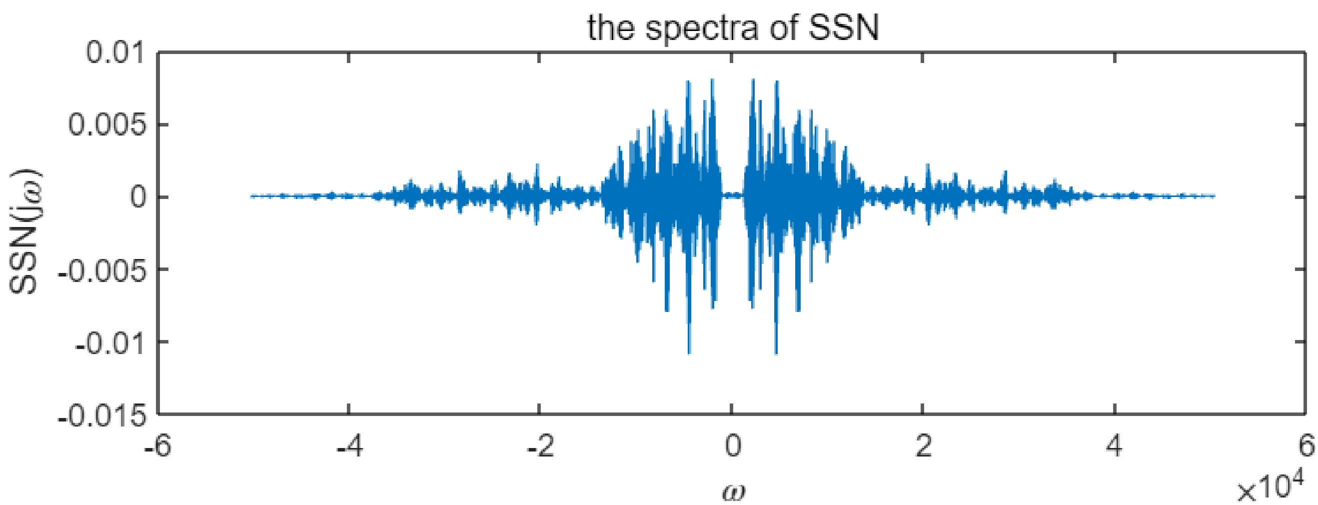
警告: 复数 X 和/或 Y 参数的虚部已忽略。

```
title('the spectra of the signal');
xlabel('\omega');
ylabel('X(j\omega)');
subplot(2,1,2);
plot(omega,SSN);
```

警告: 复数 X 和/或 Y 参数的虚部已忽略。

```
title('the spectra of SSN');
xlabel('\omega');
ylabel('SSN(j\omega)');
```





2

2. Adjust the SNR (x(t) to the above SSN) to -5 dB, let $y = x + SSN$, and normalize the energy of y in relative to x(t), i.e., modify the energy of y so that it equals to the energy of x.

Adjust SNR.

```
X2 = X / sqrt(10);
x2 = ifft(ifftshift(X2)) * fs;
```

Generate y.

```
y = x2 + ssn;
```

Modify y.

```
y = (norm(x2)/norm(y)) * y;
```

Since floating point number comparison exist precision problem, when the difference between two floating point number is smaller than a very small value, we think they're the same.

```
abs(norm(y)-norm(x2)) < 1e-12
```

```
ans = logical
   1
```

Get the spectra of y.

```
Y = fftshift(fft(y)) / fs;
```

Plot the spectra of the adjusted SNR and y.

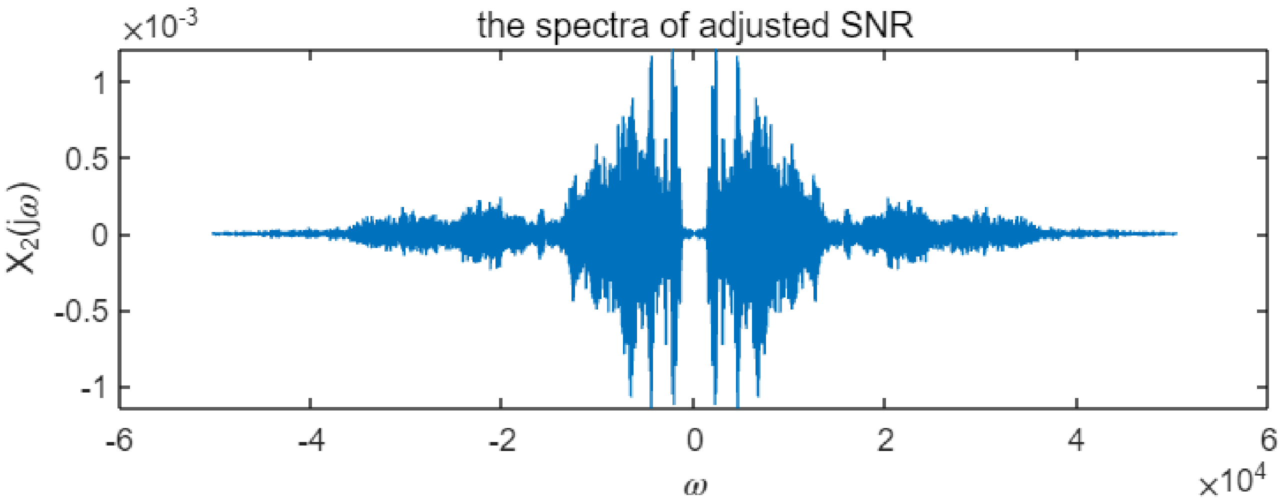
```
figure;
subplot(2,1,1);
plot(omega,X2);
```

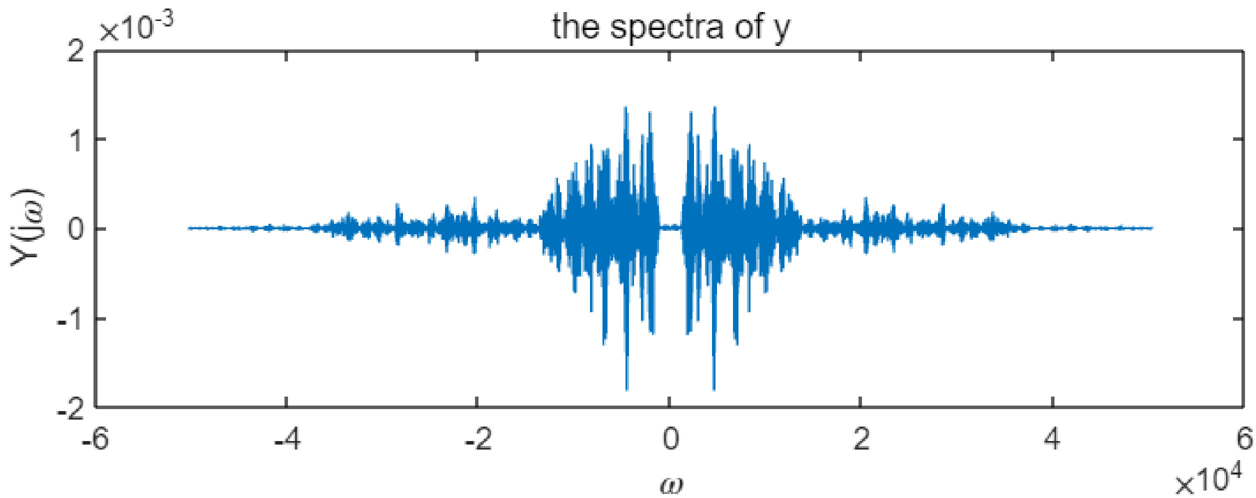
```
警告: 复数 X 和/或 Y 参数的虚部已忽略。
```

```
title('the spectra of adjusted SNR');
xlabel('\omega');
ylabel('X_2(j\omega)');
subplot(2,1,2);
plot(omega,Y);
```

```
警告: 复数 X 和/或 Y 参数的虚部已忽略。
```

```
title('the spectra of y');
xlabel('\omega');
ylabel('Y(j\omega)');
```





3

1)

3. Extract speech envelope
    1) with 2nd-order low-pass filter and cutoff frequency f_{cut}= 100, 200, and 300 Hz. Plot these three envelope waveforms in one plot and describe the difference among them.

Get the filter coefficients of the three 2^{nd}-order low-pass filters.

```
[b1,a1] = butter(2,100/(fs/2));
[b2,a2] = butter(2,200/(fs/2));
[b3,a3] = butter(2,300/(fs/2));
```

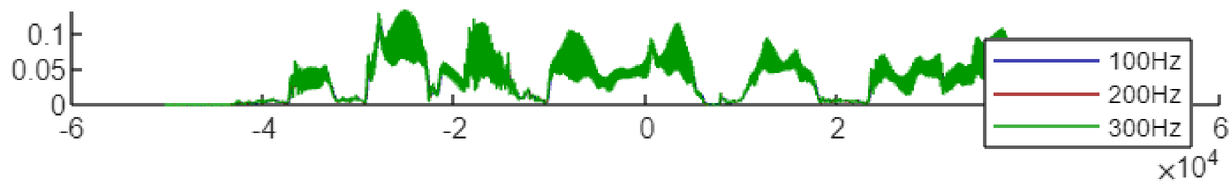
Get the three envelope waveforms of the signal.

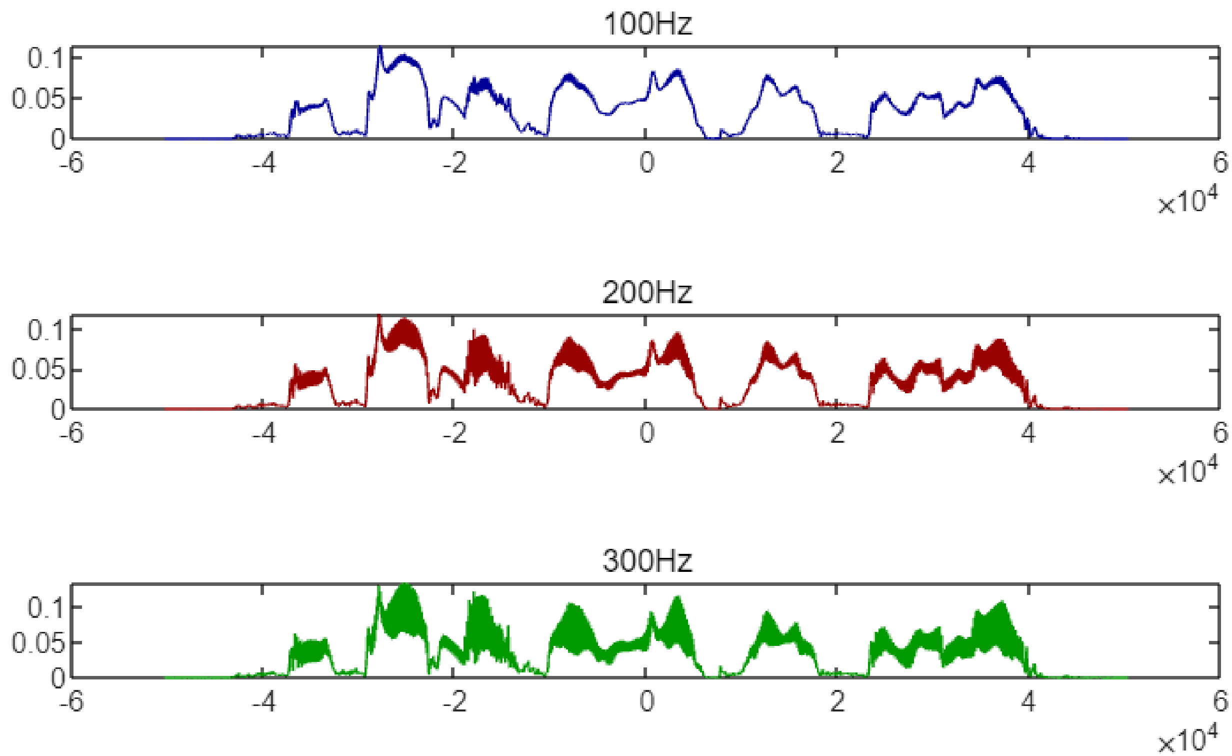
```
y1 = filter(b1,a1,abs(x));
y2 = filter(b2,a2,abs(x));
```

```
y3 = filter(b3,a3,abs(x));
```

Plot them in one plot.

```
figure;
tiledlayout(4,1);
nexttile();
hold on;
plot(omega,y1,'Color',[0,0,0.6]);
plot(omega,y2,'Color',[0.6,0,0]);
plot(omega,y3,'Color',[0,0.6,0]);
legend('100Hz','200Hz','300Hz');
nexttile();
plot(omega,y1,'Color',[0,0,0.6]);
title('100Hz');
nexttile();
plot(omega,y2,'Color',[0.6,0,0]);
title('200Hz');
nexttile();
plot(omega,y3,'Color',[0,0.6,0]);
title('300Hz');
```





Since the dots of plot of 300Hz cutoff frequency are too dense, we re-plot the three different plots respectively.

From the comparsion of the plots, we can tell that with a lower cutoff frequency, the envelope waveforms can be more smooth.

With the increase of the cutoff frequency, the envelope waveform will be with more heavy fluctuation.

2)

2) with 2nd and 6th-order low-pass filter and cutoff frequency 200 Hz. Plot these two envelope waveforms in one plot and describe the difference between them.

Get the filter coefficients of the 2^{nd}-order and 6^{th}-order low-pass filters.

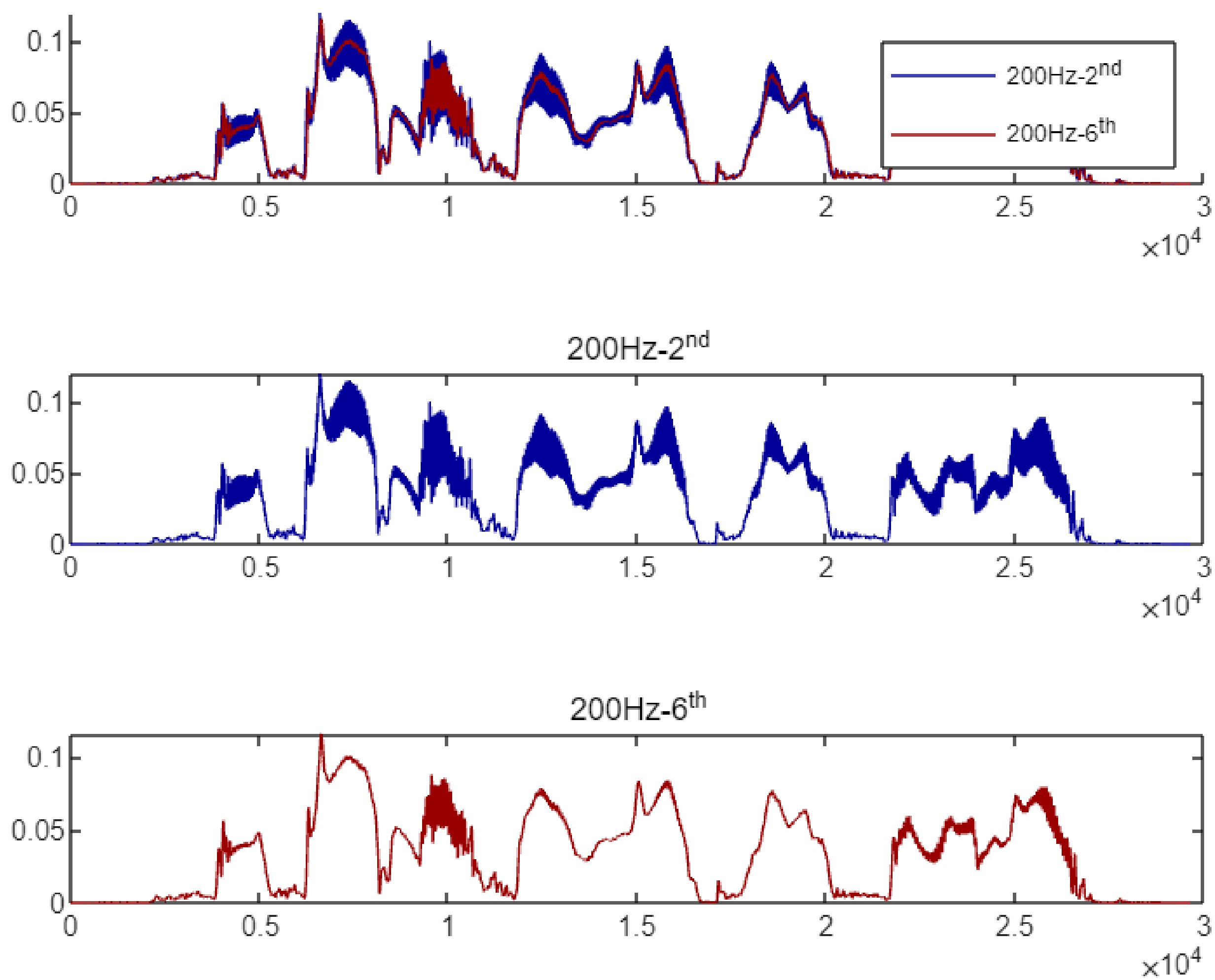
```
[b4,a4] = butter(2,200/(fs/2));
[b5,a5] = butter(6,200/(fs/2));
```

Get the two envelope waveforms of the signal.

```
y4 = filter(b4,a4,abs(x));
y5 = filter(b5,a5,abs(x));
```

Plot them in one plot.

```
figure;
tiledlayout(6,4);
nexttile([2 4]);
hold on;
plot(abs(y4),'Color',[0,0,0.6]);
plot(abs(y5),'Color',[0.6,0,0]);
legend('200Hz-2^{nd}','200Hz-6^{th}');
nexttile([2 4]);
plot(abs(y4),'Color',[0,0,0.6]);
title('200Hz-2^{nd}');
nexttile([2 4]);
plot(abs(y5),'Color',[0.6,0,0]);
title('200Hz-6^{th}');
```



Since the dots of plot of 2^{nd}-order are too dense, we re-plot the two different plots respectively.

From the plots, we can tell that, with a higher order, we can get a more smooth envelope waveform.

Expeience

```
Through Matlab work, I have a preliminary understanding of Matlab simulation tools in
the signal and system of the application of the discipline. I'm more famililar with the
usage of filter and these filter design tools. And I have a better understanding of the
audio signal processing and the speed-shaped noise. I learn how to use MATLAB to generate
different filter and what's the meaning of these filter's coefficients and how to use them
to process the signal as what I think.
```

Score

涂峻绫（100） 欧阳安男（100）