Time Frequency Representations practice

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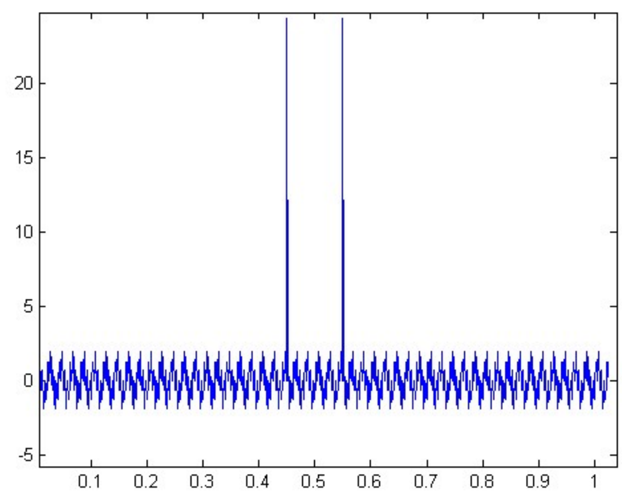
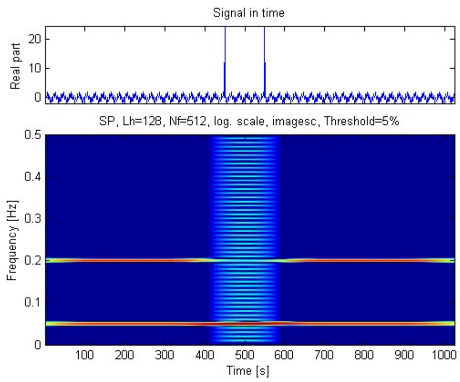
EE 7345

Q1

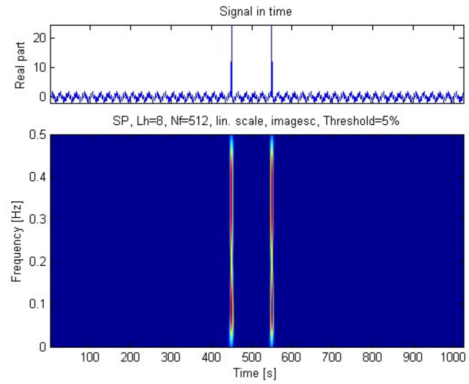
According to the problem, we need generate a time series signal with 1.024 seconds long and sampling frequency 1000Hz, That means we need divide the 1.024s to 1000 same parts. It is consists of two sinusoids with 100Hz and 400Hz frequency, and has two pulses at 0.45s and 0.55s respectively. The signal was generated below.



Then we use functions such as tfrsp() and tfrscalo() to analyze the signal. According to the uncertainty principle on the slides, time and frequency resolution depends on the length of analysis window and time and frequency could not get a good resolution at the same time, and these two parameters obey condition $\Delta t \Delta f \geq \frac{1}{4\pi}$ .That is why when we use spectrogram() to analyze signal the resolution between time and frequency is decided, which means it cannot be changed when then length of window is constant. Therefore, we can only get a good resolution in time or frequency, In another word, we can obtain good resolution in both performance.

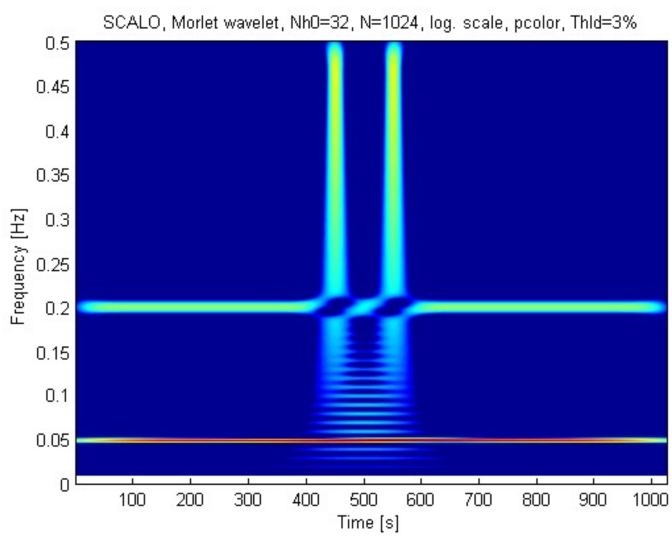


Window length 256



window length 16

However, When we use for CWT the Δt and Δf are not fixed with analysis window length, So the performance of resolution is constantly changing, which means a small Δf leads to a larger Δt while a small Δt makes a larger Δf . For this property, we can get relatively better resolution for both time and frequency of the signal in tfrscalo(). We can see clearly two frequency features located at 0.05 and 0.2 normalized frequencies along the entire time axis which represent two sinusoids of this signal, and also two frequency that is very clear enough for us to recognize that the two impulse are located at 0.45 sec and 0.55 sec.



Matlab Code

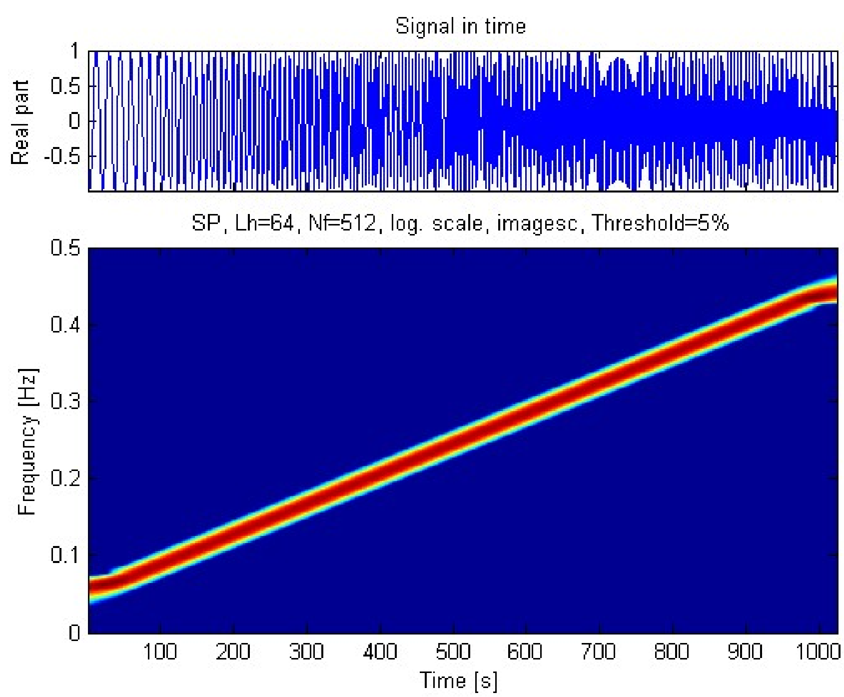
```
clear all
clc
Fs = 1000;
N=1024;
t = 1/Fs:1/Fs:1024/Fs
```

```
x = sin(0.1*pi*[0:N-1])+sin(0.4*pi*[0:N-1])
x(450) = x(450)+25
x(550) = x(550)+25
% original signal
figure(1)
plot(t,x)

%Spectrogram distribution 256
M = transpose(hilbert(x));
figure(2)
tfrsp(M, [1:N],N,hamming(odd(256)));
%Spectrogram distribution 16
figure(3)
tfrsp(M, [1:N],N,hamming(odd(16)));

%Scalogram, for Morlet or Mexican hat wavelet
figure(4)
tfrscalo(M)
```

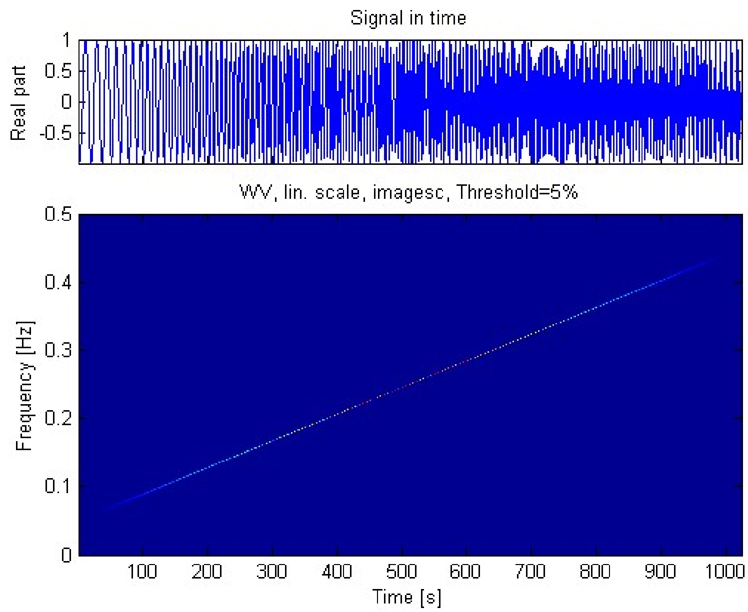
Q2

In this question, we need generate a linear chirp signal with length N=1024, the frequency increasing from normalized frequency 0.05 at N=1 to normalized frequency 0.45 at N=1000.

According to the problem, we display the time and frequency feature by using STFT and Wigner distribution respectively.



Spectrogram



Wigner-ville distribution

In these two time-frequency planes of spectrogram and wigner distribution, we can observe that frequency increased by time linearly, while the waveform wigner distribution has a better resolution to show this result. Also, during the time at starting and ending it becomes non-linear.

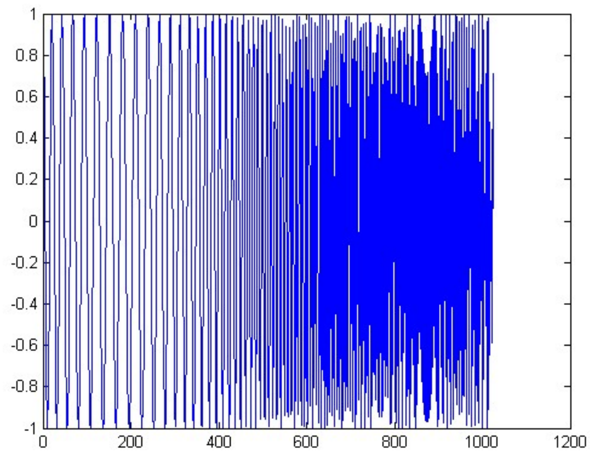
Matlab code

```
clear all;
clc;
N=1023;
% Signal with linear frequency modulation
y = fmlin(N,0.05,0.45)
figure(1)
%spectrogram method
tfrsp(y);

%Wigner-ville distribution
figure(2)
tfrwv(y,[0:N],N)
```

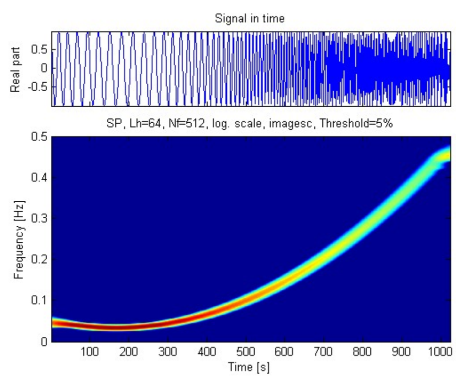
Q3

In this problem, we need to generate a parabolic instantaneous frequency signal having a length of N = 1024 that passes through the time-frequency points (n = 1, f = 0.05),(n = 500, f = 0.1),(n = 1000, f = 0.45).

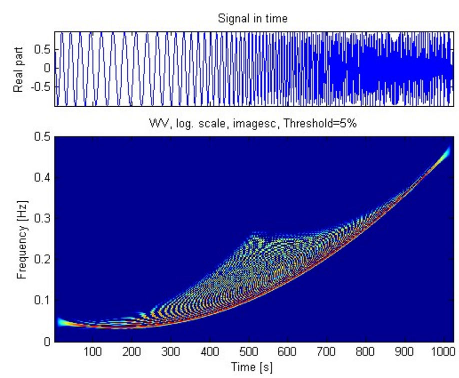
First of all, we have a general look for original parabolic instantaneous signal.



Secondly, we use Spectrogram time-frequency distribution and Wigner distribution to show the relation between time and frequency.



Spectrogram



wigner distribution

From the two time-frequency plane, we can observe that the frequency increased with time in a parabolic way, which can display the time-frequency features of original signal. Particularly, the waveform of Wigner distribution can display interference term between frequency and time. Therefore, Wigner distribution can preserve the property of time shift and frequency shift.

$$W_x(t,f) = \int_{\tau} x(t+\tau/2)x^*(t-\tau/2)e^{-j2\pi f\tau}d\tau$$
$$= \int_{\nu} X(f+\nu/2)X^*(f-\nu/2)e^{j2\pi t\nu}d\nu$$

According to the expression above, then will generate interaction term, when $W_x(t,f) \neq 0$, but w(t) = 0. In the Wigner distribution, we can find the range of interference terms clearly in the Wigner distribution.

Matlab code

```
clear all
clc;
N = 1024
%Parabolic frequency modulated signal
z=fmpar(1024,[1 0.05],[500 0.1],[1000 0.45]);
%original signal
figure(1)
plot((z))

figure(2)
%Spectrogram time-frequency distribution
tfrsp(z,[1:N],N,hamming(odd(128)));
%Wigner distribution
figure(3)
tfrwv(z,[1:N],N);
```

Q4

A phonocardiogram or PCG is a plot of high-fidelity recording of the soundsand murmurs made by the heart with the help of the machine called the phonocardiograph; thus, phonocardiography is the recording of all the sounds made by the heart during a cardiac cycle. The sounds are thought to result from vibrations created by closure of the heart valves. There are at least two: the first when the atrio-ventricular valves close at the beginning of systole and the second when the aortic valve and pulmonary valve close at the end of systole. It allows the detection of subaudible sounds and murmurs, and makes a permanent record of these events. In contrast, the ordinary stethoscope cannot always detect all such sounds or murmurs, and it provides no record of their occurrence. The ability to quantitate the sounds made by the heart provides information not readily available from more sophisticated tests, and it provides vital information about the effects of certain drugs on the heart. It is also an effective method for tracking the progress of the patient's disease.

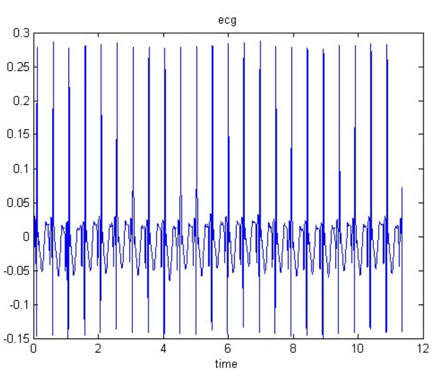
The PCG signal also includes some data with regard to the function of the heart and they are the result of mechanical vibrations. The sounds of the heart are made by the heart valves and also the murmurs. Therefore, useful information about the situation of the valves, the types of the diseases or the possible openings in the walls of the ventricle and vestibule can be obtained thanks to the process of PCG signal. The discovery of the signals of the heart and their analysis has always been interesting for the cardiologists. The main issue in processing of these signals is the discovery of vital components, the recognition of abnormal rhythms and the improvement of efficacy of signal recognition. The importance of the statement of the problem and the type of the problems in processing of such signals has intrigued not only the physicians, engineers but also the technicians of signal processing. Stethoscope is a recognized device in the medicine practice and physicians have always trusted it for the recognition and diagnosis of heart diseases. Nowadays, the main function of stethoscope is a primary test for the assurance of the health of heart performance. The people with abnormal heart beats need to be sent to cardiological clinics. Today, the medical technology is moving towards the economy and the reduction of diagnosis expenses and the preservation of human health.

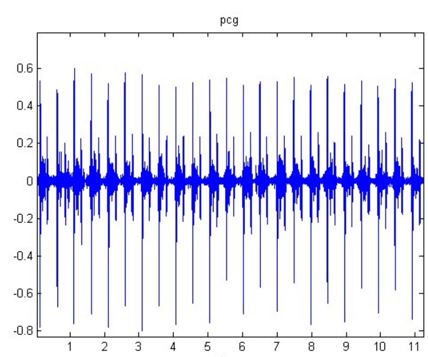
Sources of my phonocardiogram information:

[1] Wikipedia.http://en.wikipedia.org/wiki/Phonocardiogram

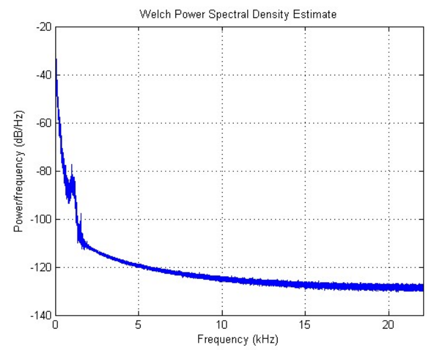
[2] Farzam B, Shirazi J. The Diagnosis of Heart Diseases Based on PCG Signals using MFCC Coefficients and SVM Classifier[J]. IJISET-International Journal of Innovative Science, Engineering & Technology, 2014, 1(10).

First of all, I download the file contains PCG and ECG signals, then I plot them in time domain.





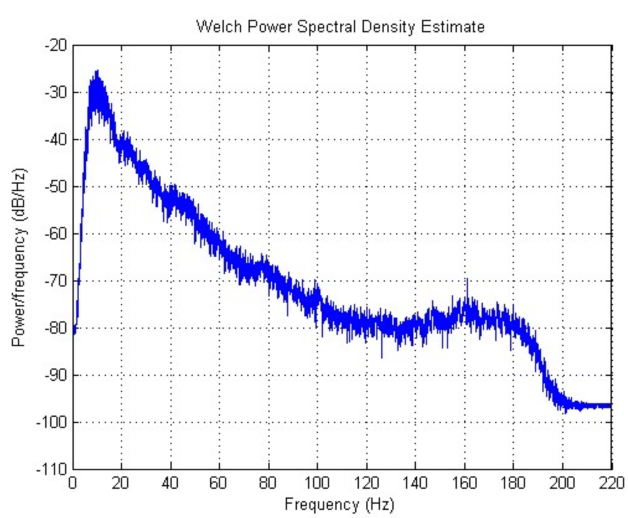
Then, I analyze the signal using the Welch averaged periodogram and the spectrogram. We need decimate () since the sample frequency 44100 Hz is too big to make it hard to analyze. If we did not decimate the signal, we will obtain that



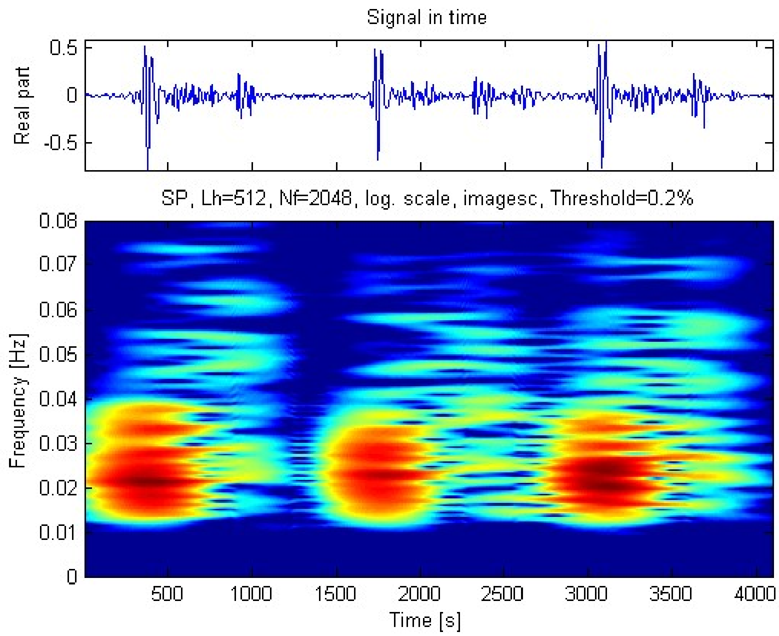
PSD without decimating

After try different decimate level, I choose 16 as decimate rate that means our sample frequency becomes smaller than original frequency.Then we apply pwelch() to show PSD and tfrsp() function to generate the time-frequency feature, When I deal with spectrogram by using STFT for the decimate signal, I change the frequency threshold and frequency bound to make we can see more important things.

From PSD calculation we can find that data mainly focus in the frequency between 10 to 100Hz, Which means that phonocardiogram signals are consists of S1 and S2 have low frequency below 100 Hz.

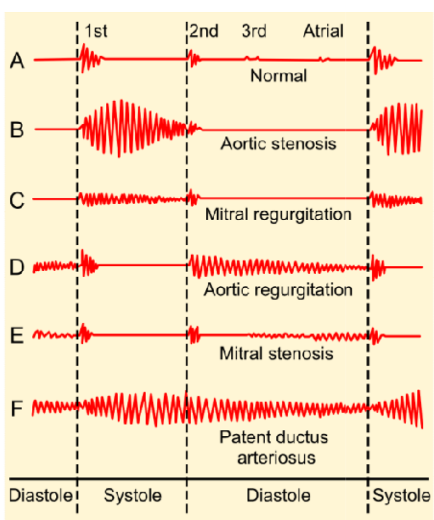


PSD calculation



The spectrogram of decimate phonocardiogram signal

From the spectrogram of decimated signal, comparing with real part of signal, I find that there are some abnormal noise between S1 and S2 which is corresponding to the spectrogram feature. The spectrogram shows that there are some noise stay in S1 and S2 and some of them are in higher frequency. Otherwise In real part the amplitude of signal between S1 and S2 is lower. Also, the signal between S1 and S2 is decayed.



Phonocardiograms from normal and abnormal heart sounds

Let us see the performance of the noise signal. It is in higher frequency and the amplitude is continuous decreasing between S1 and S2.

According to the picture above, I find that this situation is corresponding to the D type- Aortic regurgitation. Therefore, in my opinion, the patient get the disease is **Aortic regurgitation**.

Matlab code

```
clear all
close all
clc
z = audioread('phono1');
L = length(z);
Fs = 44100;
t=[1:L]/Fs;
y1 = z(:,1)
y2 = z(:,2)
figure(1)

plot(t,y1)
title('ecg')
xlabel('time')
figure(2)

plot(t,y2)
title('pcg')
xlabel('time')
figure(5)
pwelch(y2)
x = decimate(y2,16)
figure(3)
plot(x)

figure(4)
pwelch(x,[],[],[],440)
pcg2=hilbert(x)
figure(6)
tfrsp(pcg2(1:2^12))
figure(7)
tfrsp(x(1:2^12))
```