

# **Project Report for EC516 Project**

## **Assignment 01 (Fall 2018)**

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### **1. Assignment**

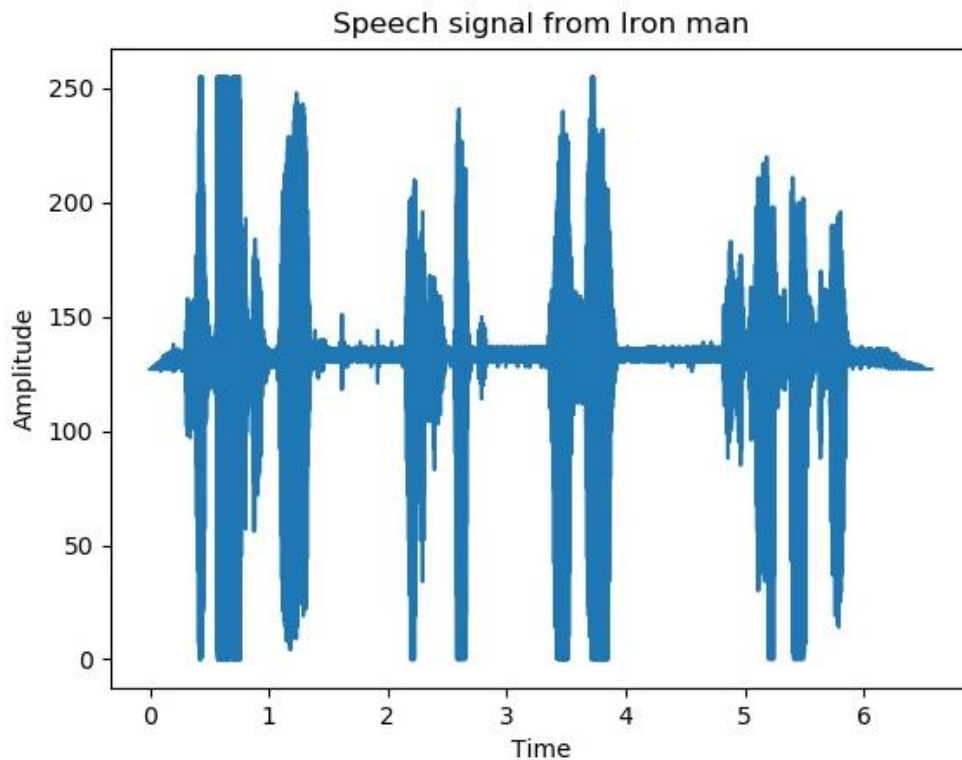
- 1.1 (3 points) Record and display a speech signal.
- 1.2 (3 points) Compute and display the discrete TDFT of the recorded speech signal using a rectangular (“box”) window whose duration is approximately 20 ms.
- 1.3 (4 points) Use GFBS Method to compute the speech signal back from its discrete TDFT.

### **2. The process and the result of the project**

#### **2.1 Record and display a speech signal.**

I got the signal from the movie *Iron Man*.

First, I used the wavfile function from scipy.io in python to read the sample rate and the amplitude of the signal. The sample rate for this project is 11025Hz. Second, I calculated the duration of the signal using the sample rate and the length of the recorded signal. Then, I displayed the signal using matplotlib in python with time as x and amplitude as y.



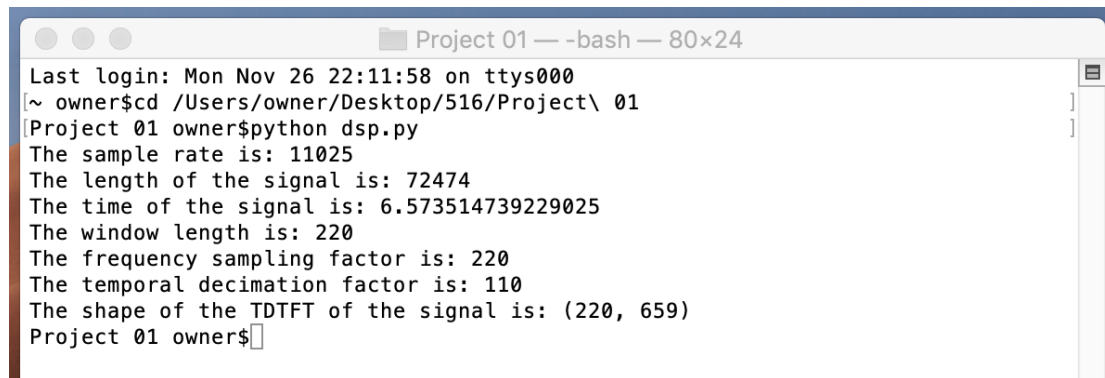
2.2 Compute and display the discrete TDFT of the recorded speech signal using a rectangular (“box”) window whose duration is approximately 20 ms.

Now, I have a discrete time signal. From the definition of TDFT, I am going to observe the signal using a shifted window function. From the assignment, I am using a window which has a length of  $0.02 \times \text{sample rate} = 220$ .

Since we want to use GFBS method, the  $L$  which is the Temporal Decimation Factor has to be smaller than the length of the window. So I set  $L$  equals to the half of the length of the window which is 110.

For each column of the DTDFT, we are actually doing a DTFT of the

observed signal. In this way, I decided to use the FFT function provided by numpy in python. So the M which is the frequency-sampling factor is the same as the length of the window which is 220. By combining all those FFTs together, we got the DTDTFT of the speech signal.

A terminal window titled "Project 01 — -bash — 80x24" showing the execution of a Python script. The output displays various signal processing parameters: sample rate, signal length, time, window length, frequency sampling factor, temporal decimation factor, and the final shape of the TDTFT matrix.

```
Project 01 — -bash — 80x24
Last login: Mon Nov 26 22:11:58 on ttys000
[~ owner$cd /Users/owner/Desktop/516/Project\ 01
Project 01 owner$python dsp.py
The sample rate is: 11025
The length of the signal is: 72474
The time of the signal is: 6.573514739229025
The window length is: 220
The frequency sampling factor is: 220
The temporal decimation factor is: 110
The shape of the TDTFT of the signal is: (220, 659)
Project 01 owner$
```

### *Programming result*

The picture shown above is the result of my programming. We can see that all information we need is listed there. One thing we should pay attention to is the shape of the TDTFT of the signal. The TDTFT of the signal should be a (M, Signal length/L) matrix. Since the M=220, L=110, Signal length=72474, the shape of the matrix meets the right answer. The matrix is too big to display here, so I put it in my Github.

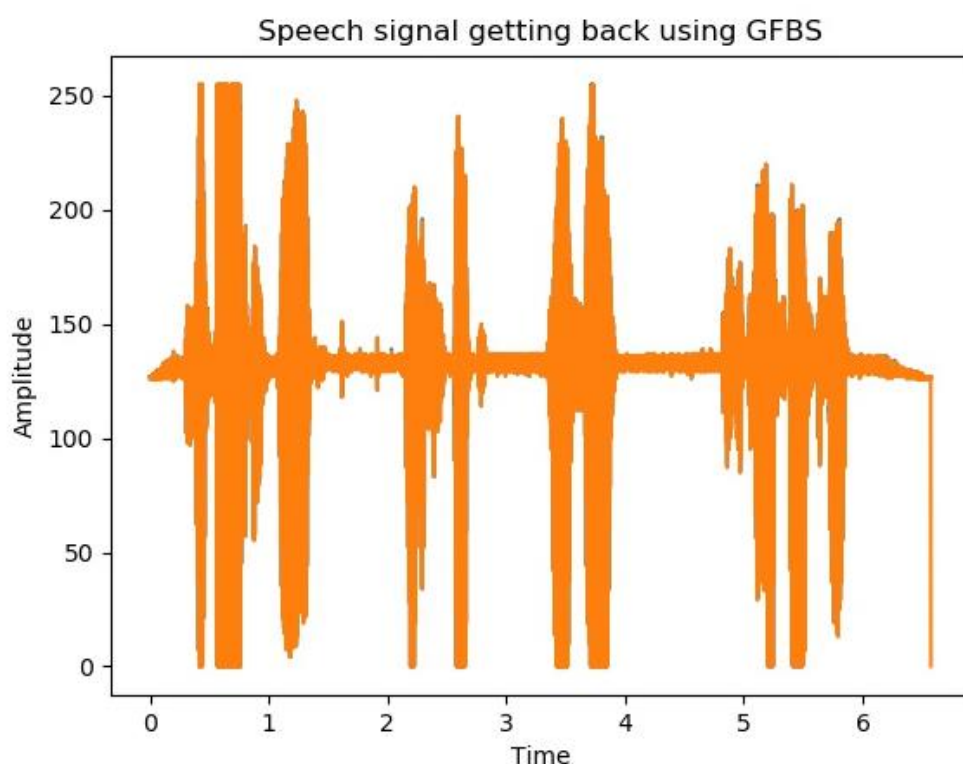
Github link: <https://github.com/cuicyz/EC516-Project1>

## 2.3 Use GFBS Method to compute the speech signal back from its discrete TDFT.

From the information given, we can calculate out the synthesis window used in the GFBS method. The synthesis window  $f[n]$  is 0.5 at  $0 \leq n \leq 109$

and 0 at otherwise.

For the realization part, I used the IFFT function. I did IFFT to each column of the DTDFT and link all those IFFTs together. In this way, I got a combination of different part of the original signal observed by the window linking together. By sampling this list, I got the original signal back. Then I used the same way in 2.1 to display the signal.



As we can see, the speech signal from iron man and the speech signal getting back using GFBS is exactly the same except the last part. I think the reason for this is I added zeros at the last of the discrete time signal to meet with the need of python before doing TDTFT. I'm not very sure about it but I will figure it out by making other attempmts.

### **3. Challenges I faced in the program**

Since I used the wavfile function in python, the read and write of the signal did not confuse me a lot. I believe the most difficult part of the project is how to do the TDTFT of the signal.

The first method I tried is writing the expression of TDTFT but it is complex to write in python, so I think there might be a better way out. The second way I tried is writing in the filtering view of TDFT but since I have no experience in doing convolution in python, my code did not work. The third way is using the FFT function provided by python. It is fast and high efficiency and I got what I want. Actually, I remembered the professor mentioned in the lecture that it is a big problem for computer to do DFT until FFT comes out. Maybe I should think of FFT earlier.

### **4. The code and github link**

I put my Github link here in case there is something I didn't clarify.

Github link: <https://github.com/cuicyz/EC516-Project1>

You can find all things you need in my Github including the wavfile, the code, the TDTFT of the signal and the displayed signals.