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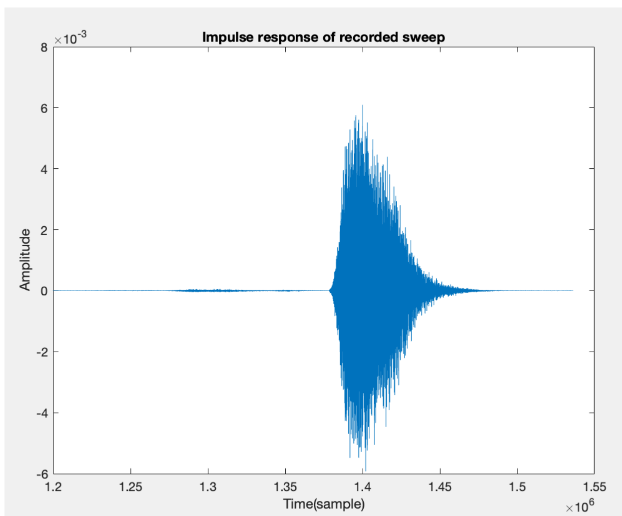
Homework 6b

Problem 1

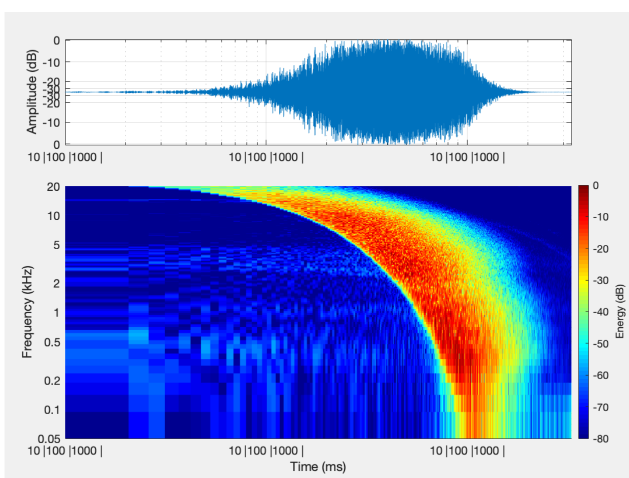
I worded with Jatin, Kim and Jingjie for measuring the Impulse response. We measured the east staircase, because the space of staircase, which crosses three stairs, is super large. The sound reverberation is obvious. Then we set the speaker at the third floor and the microphone at the first floor. The measured sounds (1a, 1b, 1c, 1d and 1e)are in the submitting folder.

Problem 2

(a) The plot of impulse response can be shown as below:



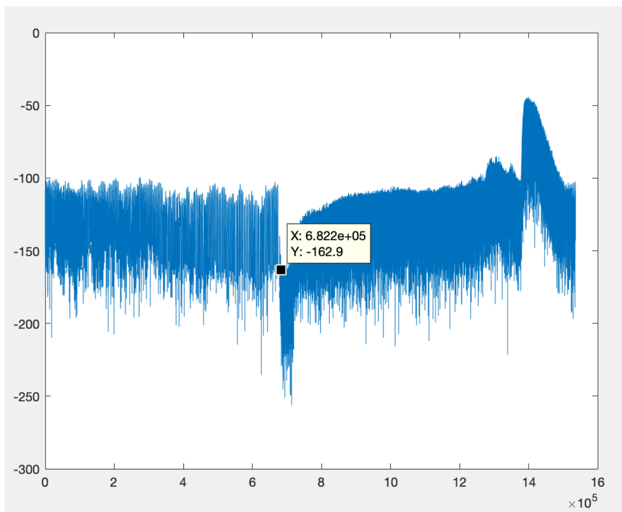
Then, the ftgram can be shown as below:



(b)

The SNR can be calculated as below:
noise = 2.5709e-07
peak value = 0.0061
Then the SNR can be calculated as mag2db(peak/noise) = 87.5dB

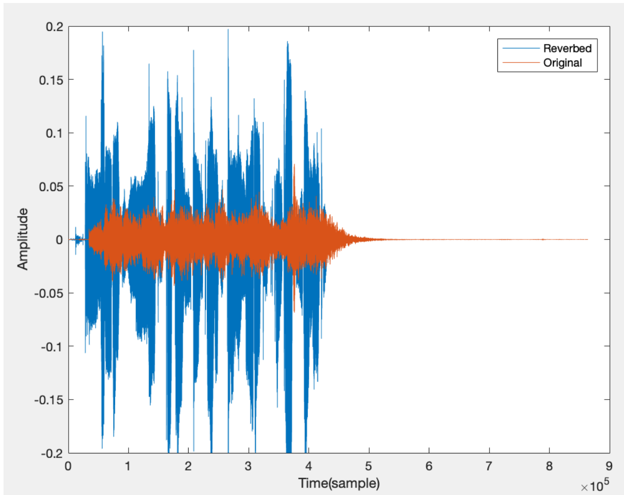
(c) T_{60} is the time of one signal decaying 60dB, therefore we can see from the dB plot of impulse response, $T_{60} = 682200/48000 = 3.26s$



(d) The code for convolving the measured impulse response with Suzanne Vega is:

```
music = audioread('TomsDiner_full.wav');
music = music(:,1);
music_reverb = conv(music,impulse_response);
audiowrite('music_reverb',music_reverb,44100);
```

The processed audio is in the submitted file. This is the plot of original sound and reverbed sound.



Problem 3

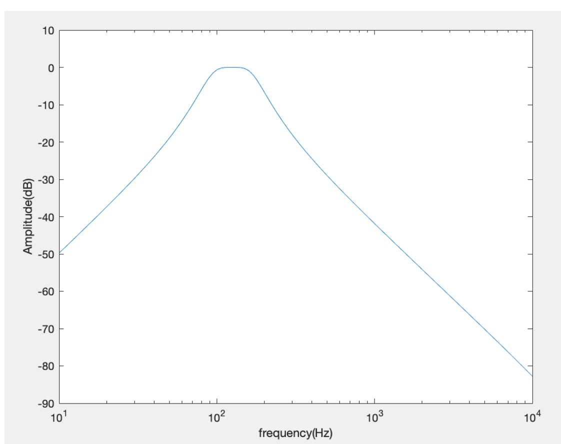
(a) The MATLAB code for creating bandpass filter is shown:

```
fc = [62.5/sqrt(2),62.5*sqrt(2)];
[b,a] = butter(2,fc/(fs/2),'bandpass');
```

Then the coefficients of b and a can be obtained. So next step is read the noise file and apply the zero-phase filter:

```
noise = audioread('1d.wav');
noise_filtered = filtfilt(b,a,noise);
```

Then the plot of frequency responses of the filters on a logarithmic frequency axis can be shown: The $T60$ can be measured from the dB plot of noise filtered signal, which is



0.4s.