Claire Caplan, crc235
Amaya Murguia, amm449
Collaborated with Yiqi Jiang, yj89

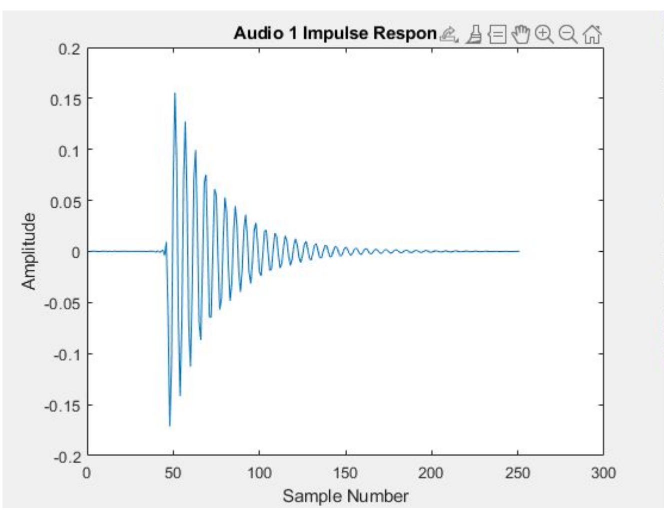
ECE 4670 Lab 1 Report

Due Friday 3/12/2021

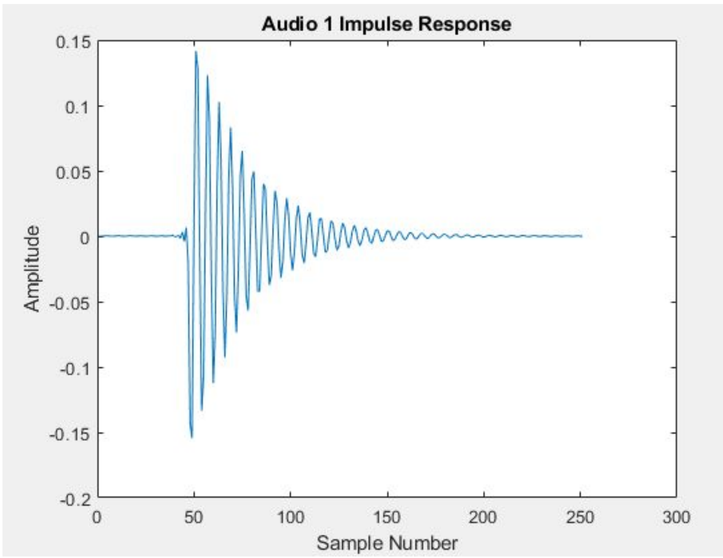
3.1 Measure and Plot the Impulse Response

The three audio 1 zoomed in impulse response plots look similar to one another with a similar amplitude of 0.15. The duration of the impulses responses are also similar. The channel tends to oscillate before returning to zero. The three audio 0 channel plots also look similar, but their amplitudes vary a bit more from one another. The channel oscillates much less before returning to zero. Between each channel, the major variation is that the audio 1 channel has more oscillation and ringing before returning to zero. We found the max peak and then zoomed in 250 indices around it and added a period of 44100 samples to find new sections of the impulse response. The x axes are all labeled starting with sample 0.

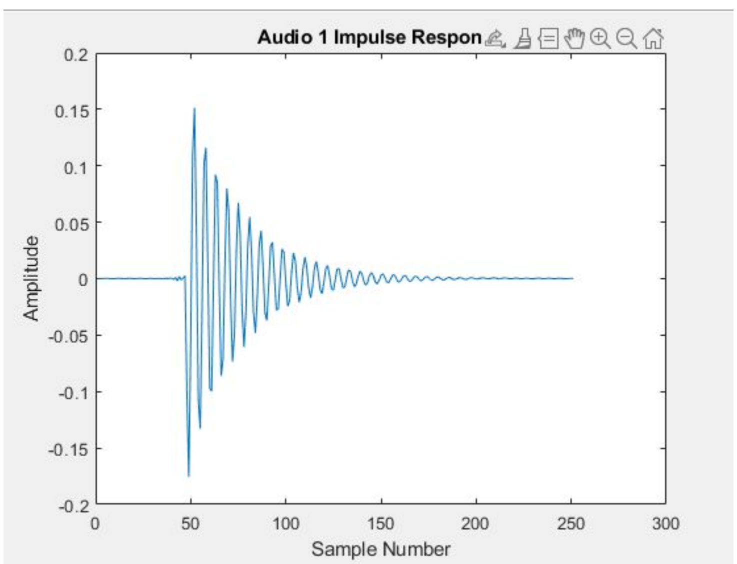
Audio 1: zoomed in impulse response 1



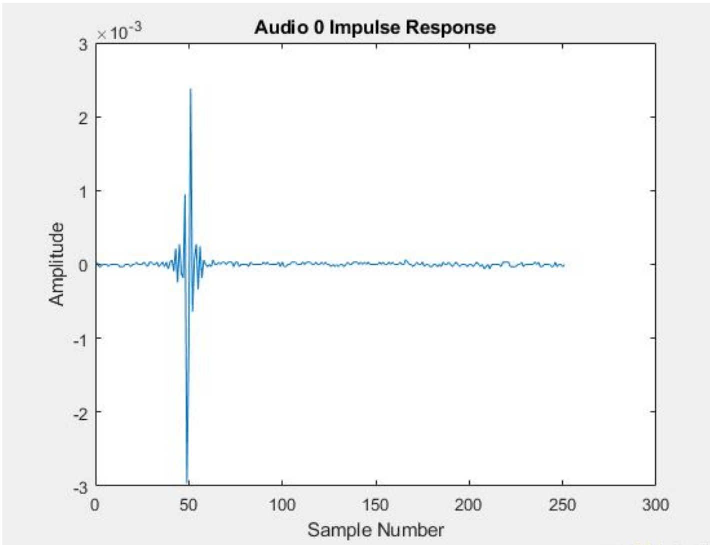
Audio 1: zoomed in impulse response 2



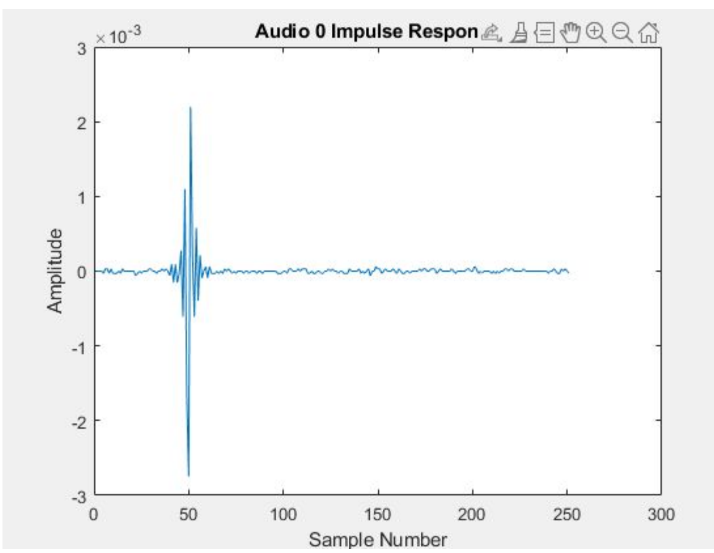
Audio 1: zoomed in impulse response 3



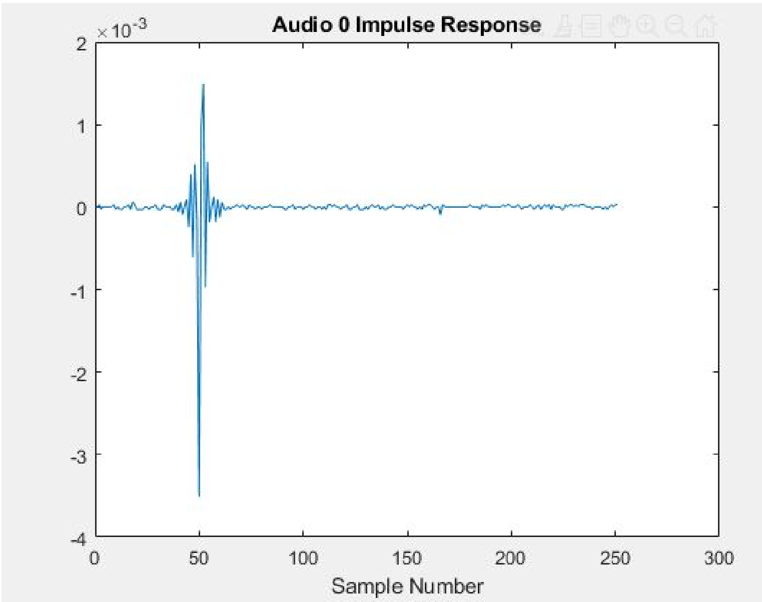
Audio 0: zoomed in impulse response 1



Audio 0: zoomed in impulse response 2

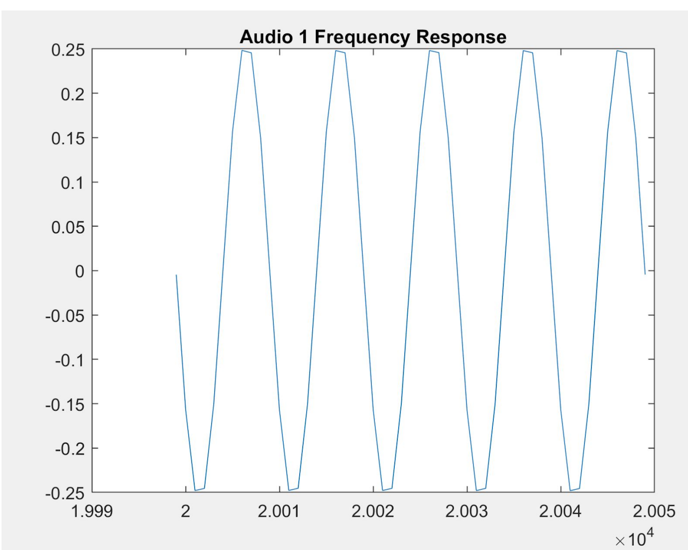


Audio 0: zoomed in impulse response 3



3.2 Measure the Frequency Response by Method 2 Above

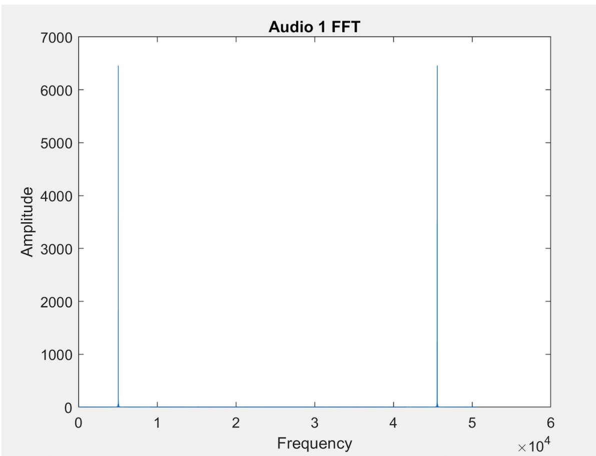
The frequency response plotted over a few periods is shown below. The period is 10 samples. The amplitude of the received waveform is about 0.25, which is less than the input waveform's unit amplitude.

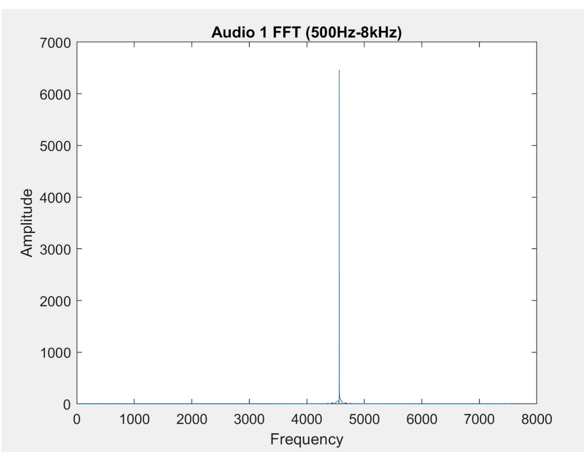


The phase shift is difficult to measure with the existing setup since the transmitter and the receiver are not time synchronized.

3.3 Measure the Frequency Response by Method 2 Above (continued)

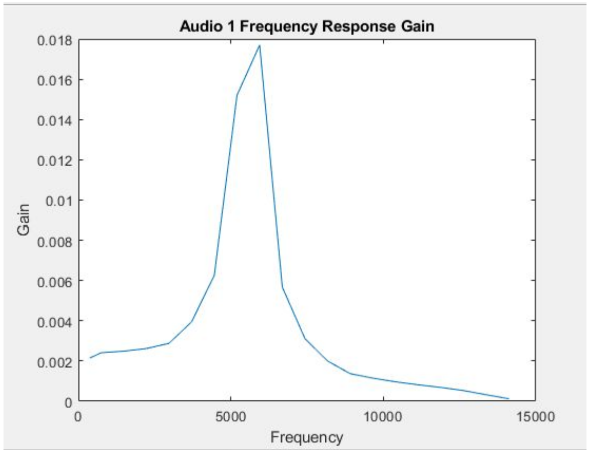
Below is the plot for the DFT of the received signal for the audio 1 channel. Our highest value of n that we plugged into the cosine is 50,000 and since this number is large the cosine was spread out in time, and so the delta function is narrow in frequency and has a large amplitude. The second plot of audio 1 highlights the sample of Z that corresponds to 4.41 kHz.



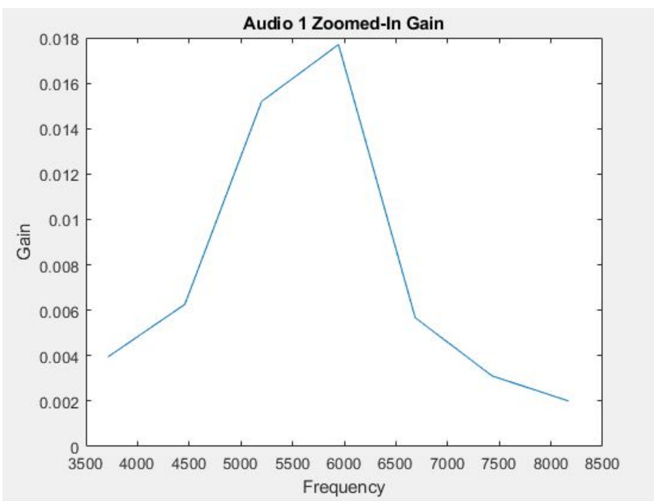


3.4 Measure the Frequency Response by Method 2 Above (further continued)

The gain of the frequency response for a multiple frequency sinusoid input over the range 500 Hz to 20 kHz is shown below. When generating our multiple frequency sinusoid, we normalized by dividing all elements of the signal by the maximum amplitude value, to avoid clipping in the audiowrite function.

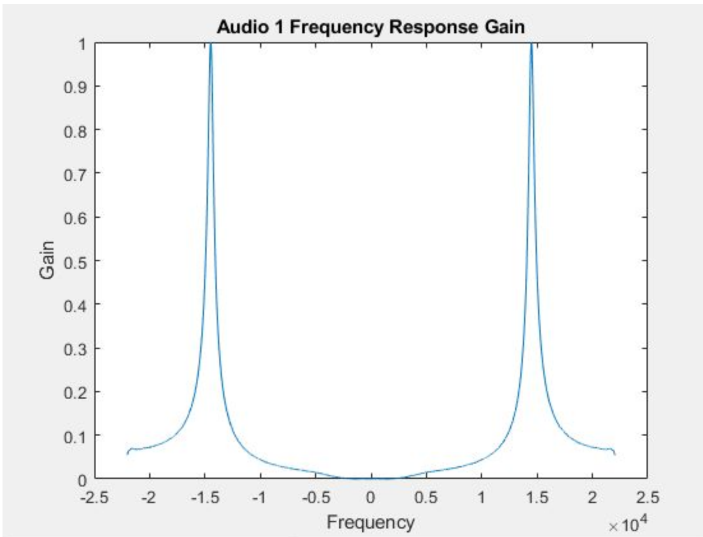


The gain does exhibit a peak, and the plot below shows the frequency response where the gain is in the vicinity of the peak. The peak is at about 5944Hz, which we determined from the zoomed-in plot.



3.6 Finish Measurement of the Frequency Response by Method 1 Above

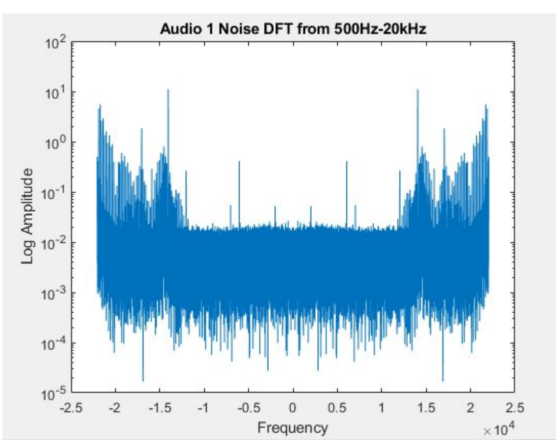
Below is the gain for audio1 over the frequency bands 500 Hz to 20 kHz. We changed the frequency axis to discrete frequencies according to section 6 of the lab handout.

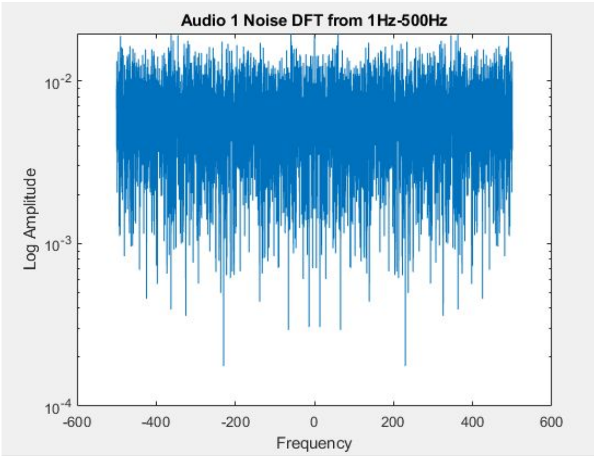


4 Determining the Noise Spectrum and Power

The received power for audio0 is: **2.3767x10^{-10} W** and the received power for audio1 is: **9.6704x10^{-9} W**. These are the estimates of the standard deviations of the two channels because they are the estimates of the noise, which is how much the received signal can vary.

Below are the DFTs of the noise outputs for both channels. The spectra look mostly flat with peaks at frequencies around 1.7x10^{4} Hz. The likely source of the spikes is noise from the RLC filter at specific resonant frequencies.





5 Difference in Sampling Rates

f_1 and f_2 are the sampling rates of the transmitter and receiver. We know f_1 and need to estimate f_2. We do this by sending two impulses and determining the number of zeros between them on the sending and receiving sides. Then, knowing the number of zeros between the impulses on the receiving end, we can compute the resulting f_2. The equation relating f and N with 20 seconds between the impulses is: f = N/20.

f_1 = 44100 Hz
T_1 = 1/(44100)
number of zeros to ensure 20 seconds between the impulses = $20/T_1$ = 882,000
N_{in} = 882,000
N_{out_audio0} = 884718
N_{out_audio1} = 787364
$N_{out} = 20/T_2 \rightarrow f_2 = (N_{out}/20)$

f_{2_audio0} = 44.23 kHz
f_{2_audio1} = 45.04 kHz

We counted the number of zeros by finding the number of values in the output less than 0.0005. f_1 and f_2 are different due to differences in the clocks in the hardware.