

ECE 4446

Fall 2015

Th 3-6pm

Lab 06

Analog Voice Scrambler

Due November 19th

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Purpose:

The purpose of this lab is to design and implement an analog voice scrambler that takes as input a sound signal (speech or music) and outputs a modified (scrambled) version of the signal through speakers. Replaying the output signal as an input should unscramble the signal and play the original signal.

Experimental Results:

The experiment was an overall success. Several test were made and were positive. For instance when playing sequentially increasing frequencies, the output was sequentially decreasing frequencies. Also, when inputting scrambled signal into the system, the result was close to the original unscrambled signal.

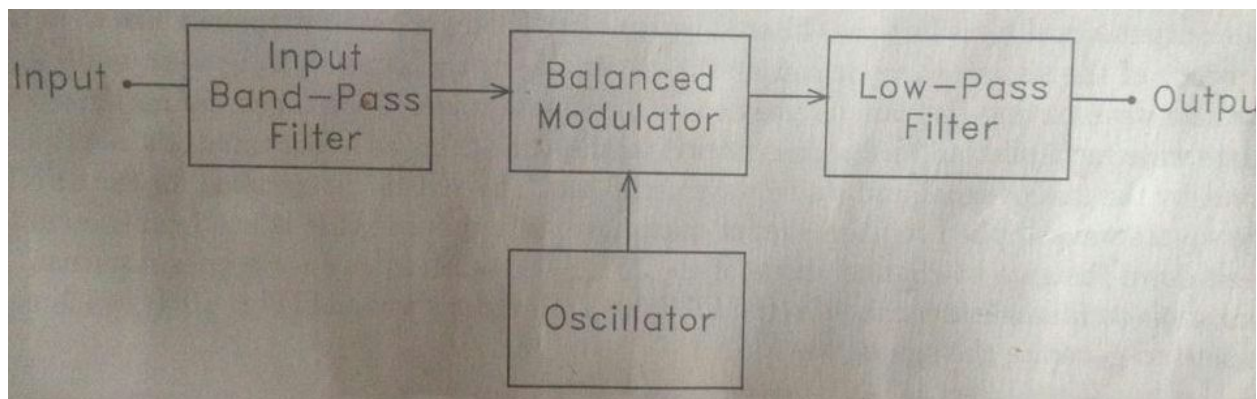


Fig 1. Full Schematic of the Speech Scrambler

Balanced Modulator:

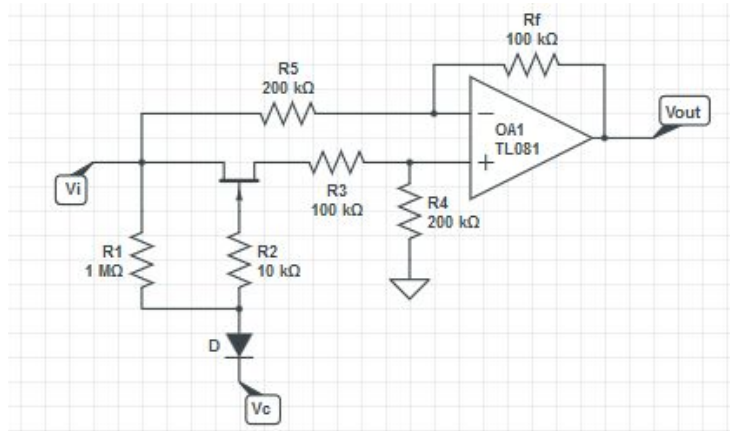


Fig. 2 Schematic of the Balanced Modulator

The input at V_C is a control voltage that is a square wave of fundamental frequency $f_0 = 3.3$ kHz. This control voltage causes the JFET to alternately switch between short and open circuit. When it is open, the gain is $-R_f/R_5 = -1/2$ (as it is a simple inverting amplifier). When the JFET is short, the gain is:

$$\frac{V_o}{V_i} = \frac{R_4 - R_3 R_F / R_5}{R_3 + R_4} = \frac{200k - 50k}{100k + 200k} = \frac{1}{2}$$

The square wave generator schematic is shown in **Figure 3**, it was then inputted into an active adder to shift and scale the square wave to switch between state 0V and state -4V.

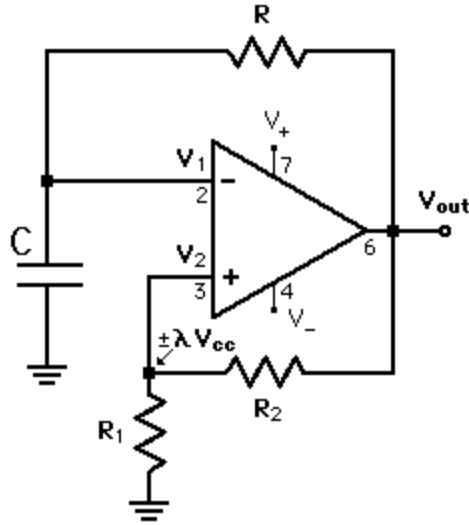


Fig. 3 Schematic of the square wave generator

Figure 4 shows the plot of the square wave generator (V_C) and **Figure 5** shows the output of the balanced modulator when V_i is a single sinusoid of frequency 400Hz.

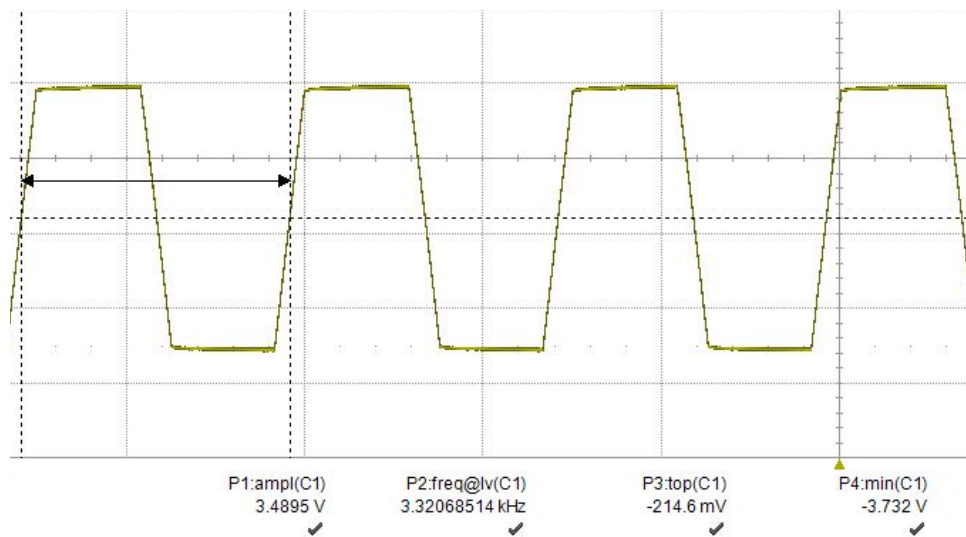


Fig. 4 Square wave generator output (V_C)

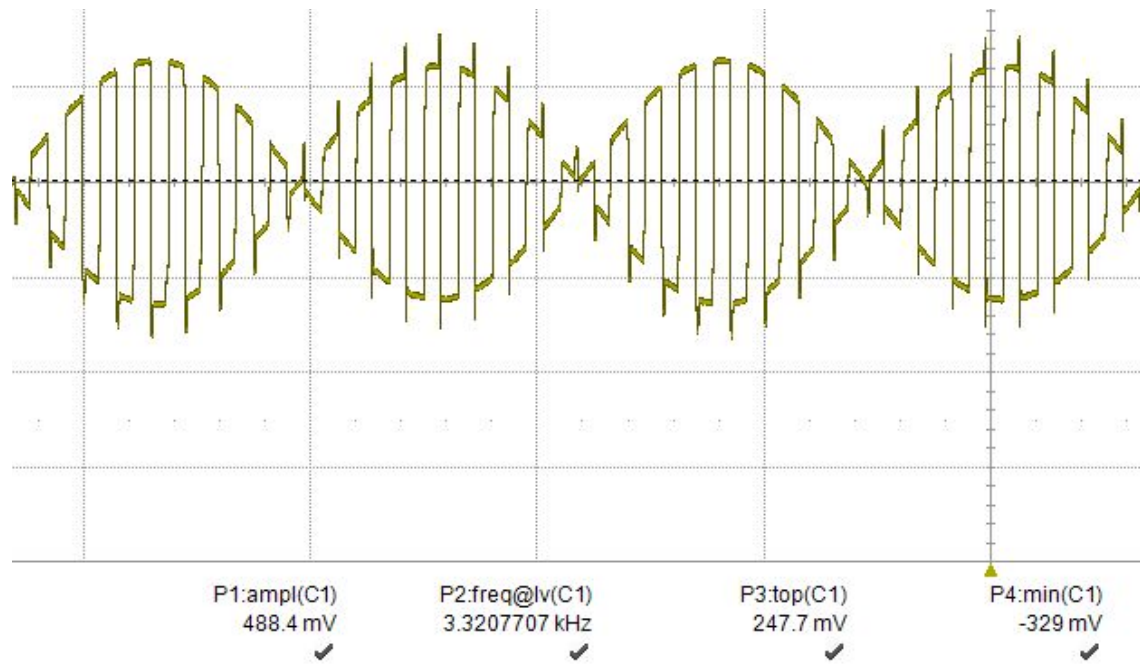


Fig. 5 Output of balanced modulator for 1Vpp 400 Hz input signal

The waveform is as expected, because the gain is oscillating between $\frac{1}{2}$ and $-\frac{1}{2}$, the output is a version of the input that is oscillating between an inverted and non-inverted version of the input.

The frequency content of the modulator output is shown in **Figure 6**, **Figure 7**, and **Figure 8**. All figures show the same output, but display different points on the frequency content.



Fig. 6 Frequency content of modulator output of 400Hz with the central 3.3kHz peak

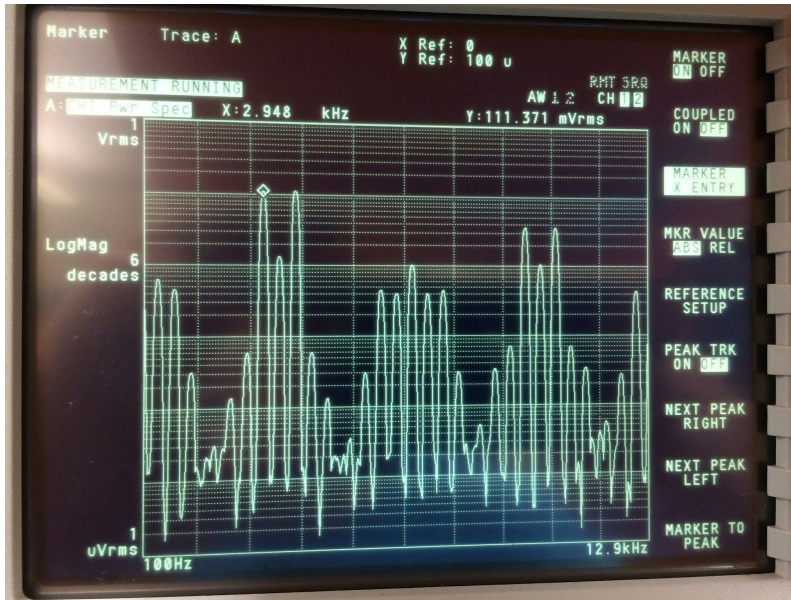


Fig. 7 Frequency content with left peak of 2.94kHz (= 3300 - 400)



Fig. 8 Frequency content with right peak of 3.7kHz (=3300 + 400)

The central peak at 3.3 kHz is the point at which the frequencies are flipped. The actual magnitude of this value does not matter that much, as the low pass filter is supposed to have a very low gain at that particular frequency, effectively canceling out the signal. Frequencies higher than 3.3 kHz are mirrored from the lower frequencies, and vice versa. The nearby peaks at 3 and 3.7 kHz are approximately the same distance away from the central frequency. The distance between the two peaks around 3.3 kHz is relative to the input signal of 400 Hz. The smaller the input signal, the smaller the distance between the center frequency and the neighboring peaks.

This is due to the convolution of the carrier signal at 3.3 kHz and the input frequency of 400 Hz. The distance between the left and right peaks are supposed to approach the input frequency of 400 Hz, which is the case in this scenario.

The ratio between the desired spectral components over the largest undesirable spectral component is calculated as such:

$$SNR = \left(\frac{A_{Signal}}{A_{Noise}} \right)^2 = \left(\frac{111.371 \text{ mV}_{rms}}{5 \text{ mV}_{rms}} \right)^2 = 496.14$$

The frequency content of the modulator output at 2kHz is shown in **Figure 9**, **Figure 10**, and **Figure 11**. All figures show the same output, but display different points on the frequency content.

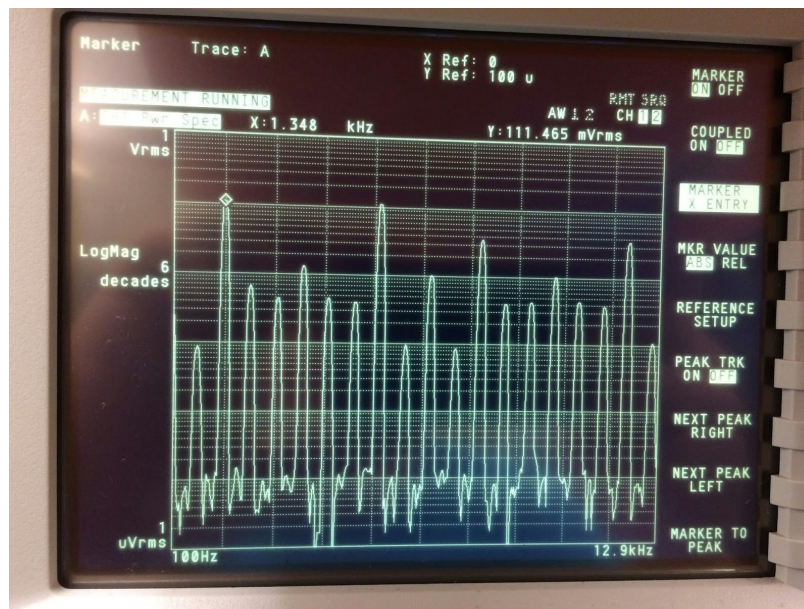


Fig. 9 Frequency content of modulator output of 2kHz with left peak of 1.35kHz



Fig. 10 Frequency content with center peak of 3.3kHz

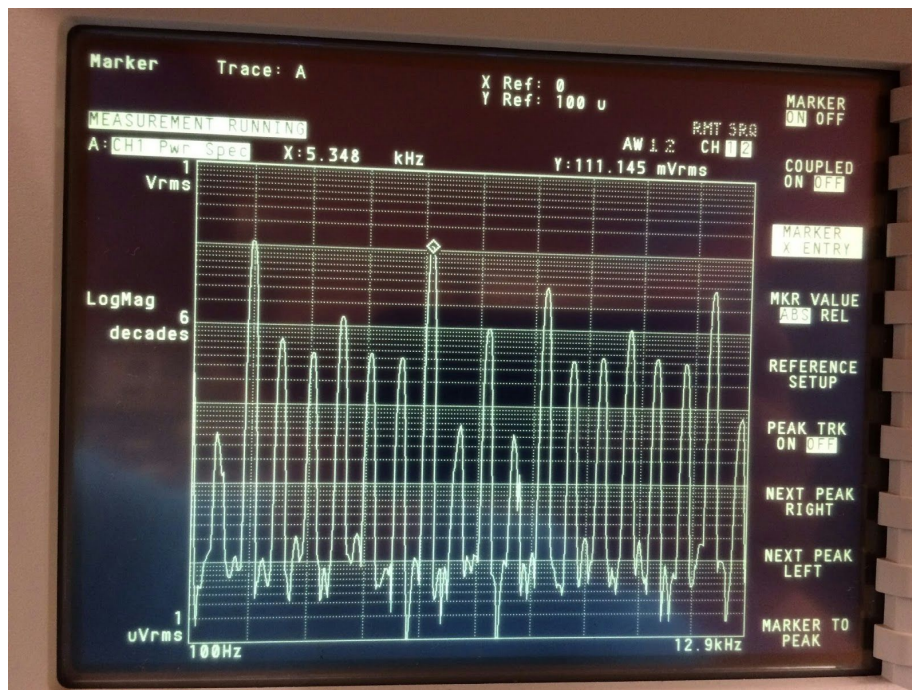


Fig. 11 Frequency content with right peak of 5.35kHz

The circuit behaves as expected. A higher frequency input of 2kHz results in a larger gap between the right and left peaks from the center frequency of 3.3kHz. In this case, the gap was around 2kHz which is the input frequency which is to be expected.

Figure 12 and Figure 13 show the output from the elliptical filter for a 400Hz and 2kHz signal respectively.

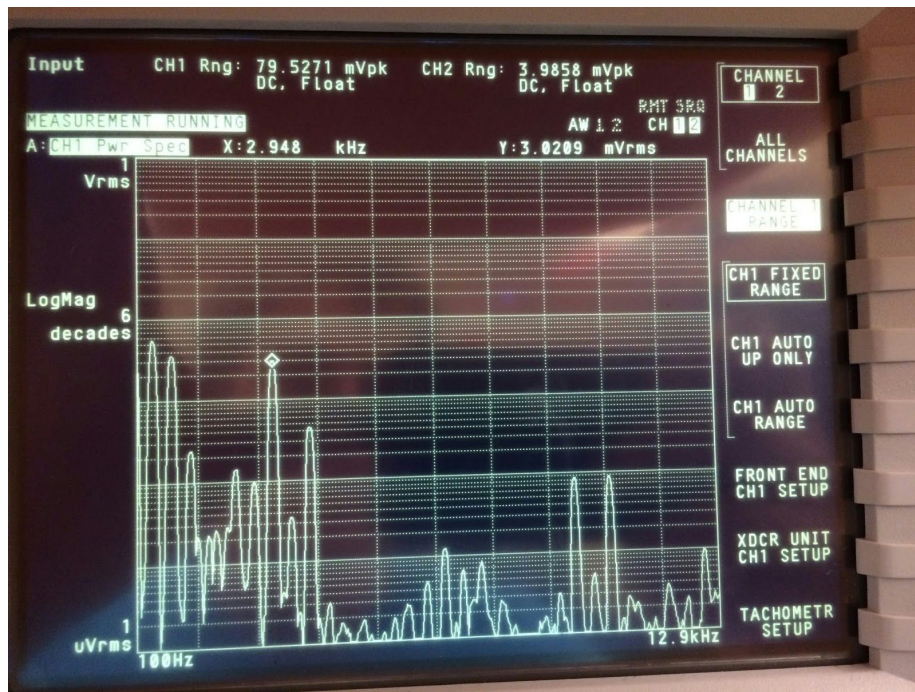


Fig. 12 Frequency content of Elliptical filter output with a 400Hz input signal

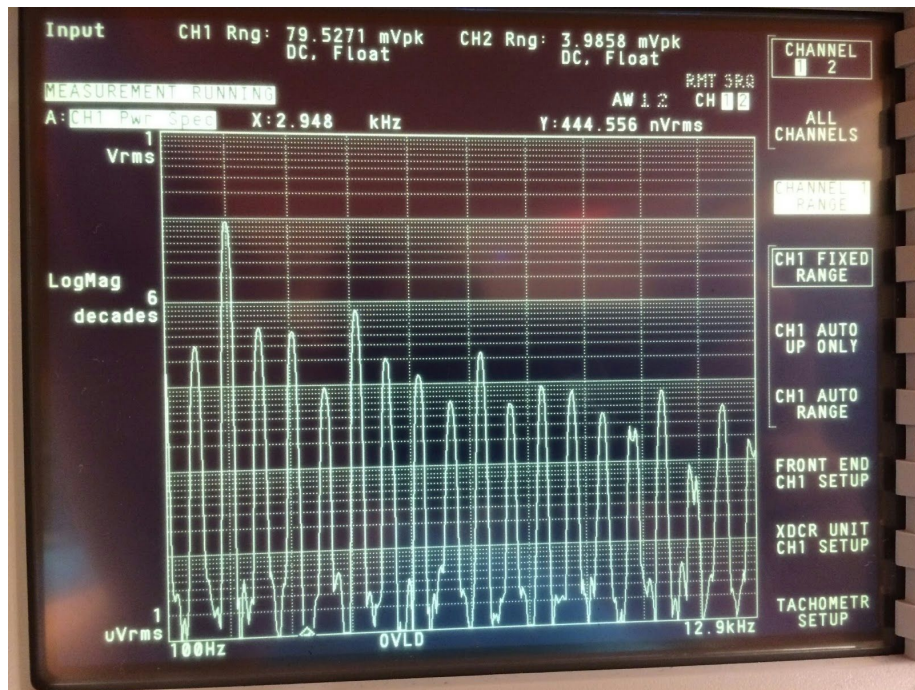


Fig. 13 Frequency content of Elliptical filter output with a 2kHz input signal

Figure 12 shows an output signal of 3kHz, with noise elsewhere. The signal is not strong relative to the noise towards 100Hz, which is a problem most likely caused by resistance and capacitance tolerances, slight changes that cause the center frequency seen in the modulator output to be 3.364kHz instead of 3.30kHz. The 3kHz signal is expected to be the largest signal, as everything else from the modulator output should be reduced drastically from the low pass filter at 400Hz.

Figure 13 shows a strong signal at 1.35kHz, which is about 2kHz below 3.3kHz. This is the correct frequency from the modulator output that is passed by the low pass filter. The amplitude is also at an acceptable level, as it is a decade higher than the noise below it. The elliptical filter has worked very well on the modulator output at 2kHz.

Input Band-Pass Filter (Chebyshev):

The Band-Pass Filter is used to only keep a desired range of frequencies in the signal, to reduce the noise in the signal. The design was chosen so that the two -3dB frequencies would be: 300Hz and 3kHz. **Figure 14** displays the transfer function experimentally obtained from the built Chebyshev circuit while **Figure 15** compares the experimental results with the theoretical model.

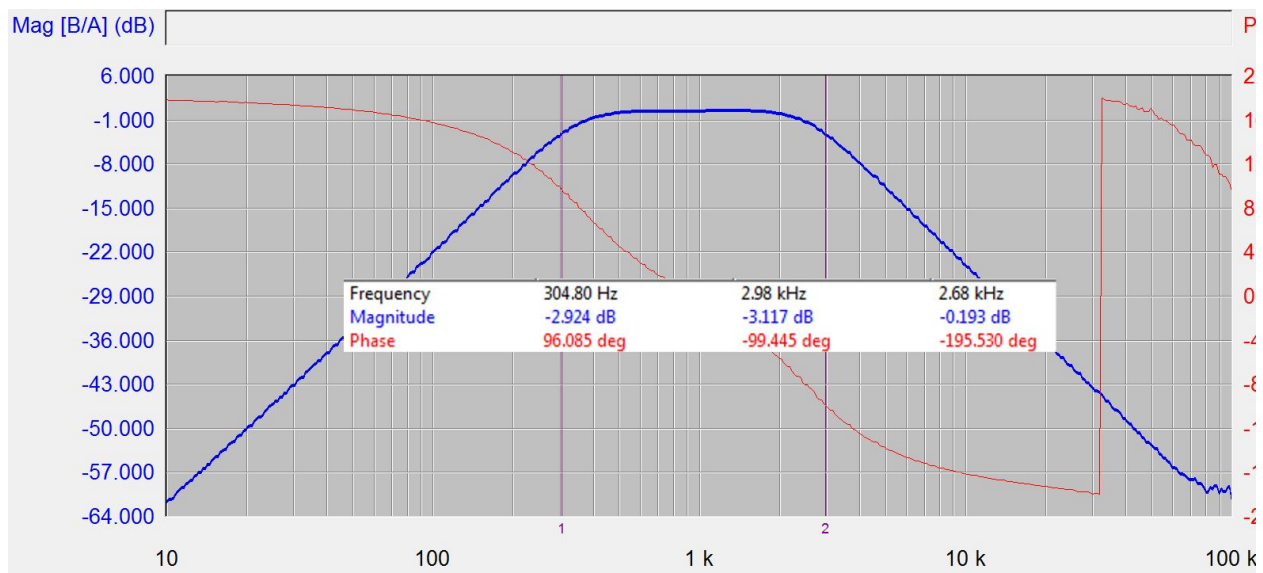


Fig. 14 Experimental measurement of Transfer function of Band-Pass filter.

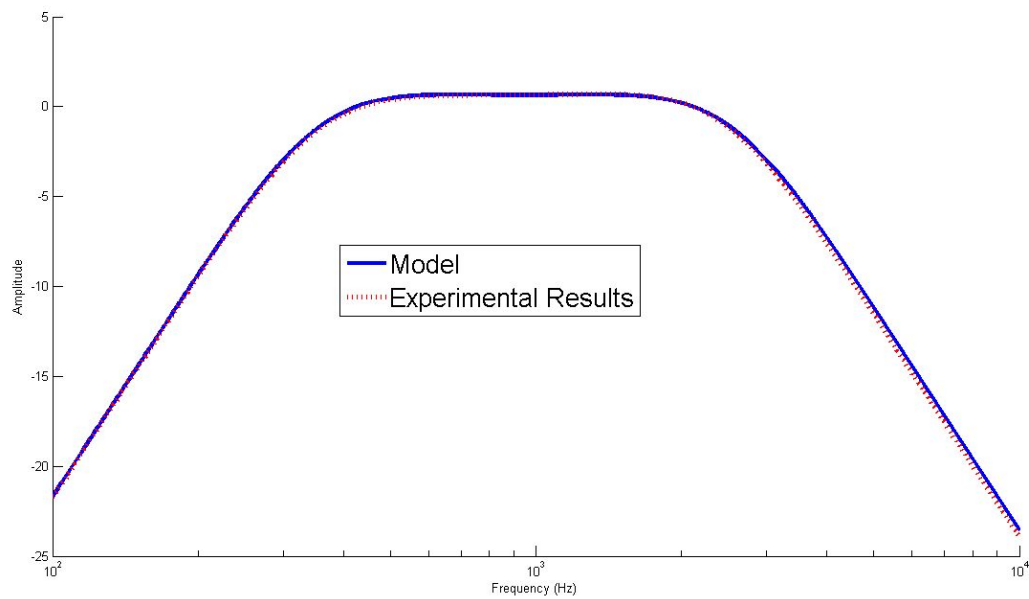


Figure 15 Comparison of model of the Band-Pass filter with the experimental measurements

Overall, the Band-Pass filter had the characteristics that were expected by the design. The cutoff frequencies as shown in **Figure 14** are 305 Hz and 2.98 Hz, which are both within 2% of the modeled cutoff frequencies.

Low Pass Filter (5th order Elliptical):

The fifth order Elliptical filter is used to cancel all higher harmonics above 3kHz. The elliptical filter is chosen for its property of having ripples in the stop band. The design of the filter is made so that a null occurs at 3.3kHz, to cancel any residual from the square wave generator. **Figure 16** displays the transfer function of the designed Low-Pass filter while **Figure 17** shows a comparison between the modeled transfer function and the observed transfer function.

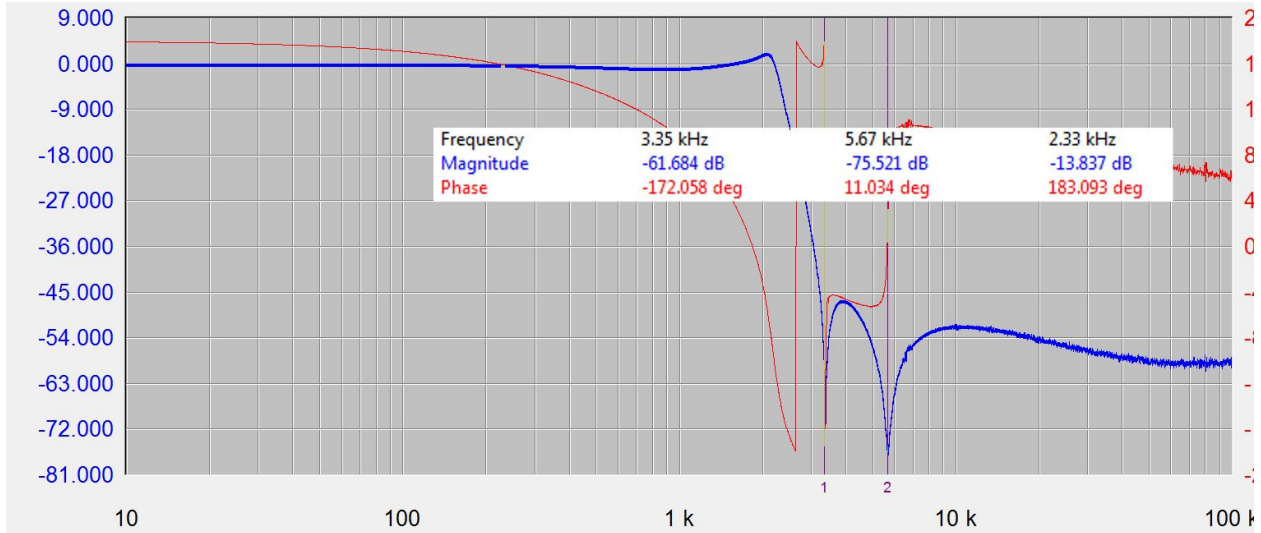


Fig. 16 Experimental measurement of Transfer function of Low-Pass filter.

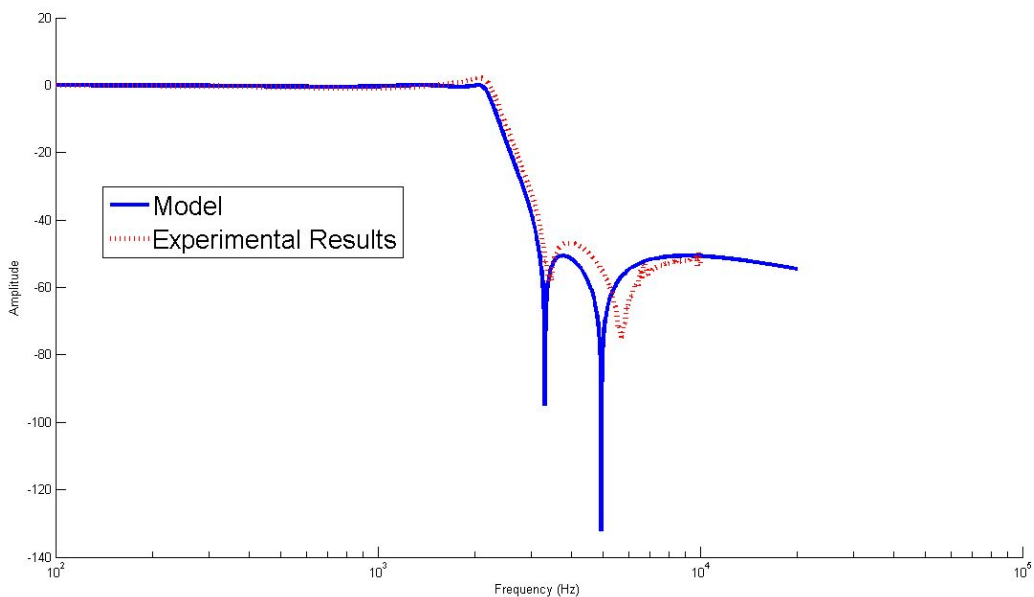


Fig. 17 Comparison of model with experimental measurements of Low-Pass filter

Using an audio input and computer speakers attached to the output of the system, phrases were used to listen to the transformation of audio through the system. It was determined difficult to replicate the output of the system. The circuit somewhat unscrambled words and phrases that were replicated from the system's output. It was noted that higher frequencies tended to work better, as the frequency spectrum has some low frequency noise with a lower frequency input signal and high frequency output signal.

The output phrase was recorded and played back into the system, which resulted in perfect unscrambling. The output still undergoes the bandpass filter, and the result sounded as though it was spoken through a telephone, which has a bandpass.

Conclusion:

The overall experiment was a success. Each subsystem built had the appropriate transfer function and the system functioned as expected. When single frequency signals were inputted, the output was single frequencies at the expected place: $f_{\text{scramble}} = 3300 - f_{\text{signal}}$. When inputting scrambled speech into the system, the output was unscrambled and was similar to the original signal, although some noise was added by the system.