Noise Cancelling Headphones

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1 Introduction

In today's busy and noisy world we like to go into a different zone while listening to music. However, more often than not the surrounding noise interferes us in this basic task as well. This motivates the need to have noise cancelling headphones. Noise cancelling headphones have become common these days, but are still not understood to the fullest. In this project we develop a control system to actively cancel out the noise for us to peacefully listen to music!

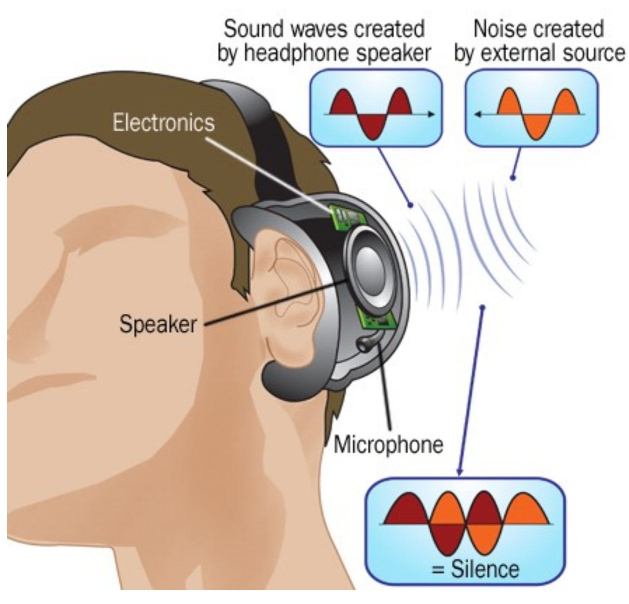

Figure 1: Inside noise cancelling headphones, source here

2 Theory

We design a feedback closed loop control system for cancelling noise. The transfer function of such a closed loop system is given as -

$$T(s) = \frac{G(s)}{1 + G(s)H(s)}$$

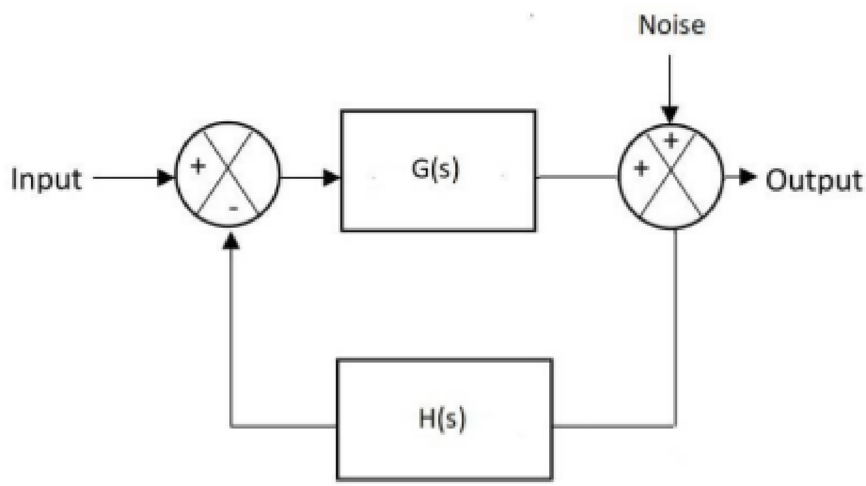

Figure 2: Closed loop Feedback system

In such a system, the noise to output transfer function is given by-

$$T_N(s) = \frac{1}{1 + G(s)H(s)}$$

This transfer $T_N(s)$ should evaluate to give an attenuation of 20 dB at a frequency of 100 Hz, without affecting other frequencies significantly. For achieving this, we intend to increase the gain of the block H(s) so that the open loop gain G(s)H(s) = 20dB at a frequency of 100 Hz. So, we first design a gain stage to achieve the required open loop gain.
Next, we see that including such a huge gain in the system pushes it to instability (if the gain added is greater than gain margin of the system). This happens because the magnitude Bode plot shifts higher by the added

gain, leading to the phase crossover frequency now having a gain of more than 0 dB.
So, we add a **lag compensator** to the system so that the system becomes stable again. The compensator helps reduce the gain near the phase crossover frequency, hence making the system stable. It effectively increases the gain margin (by changing the phase crossover frequency) for the system and hence helps in stabilizing the system.
We design the compensator, based on the measurements of the open loop systems frequency response characterization. In our simulations, we found out that a single order compensator is not enough to stabilize the system.

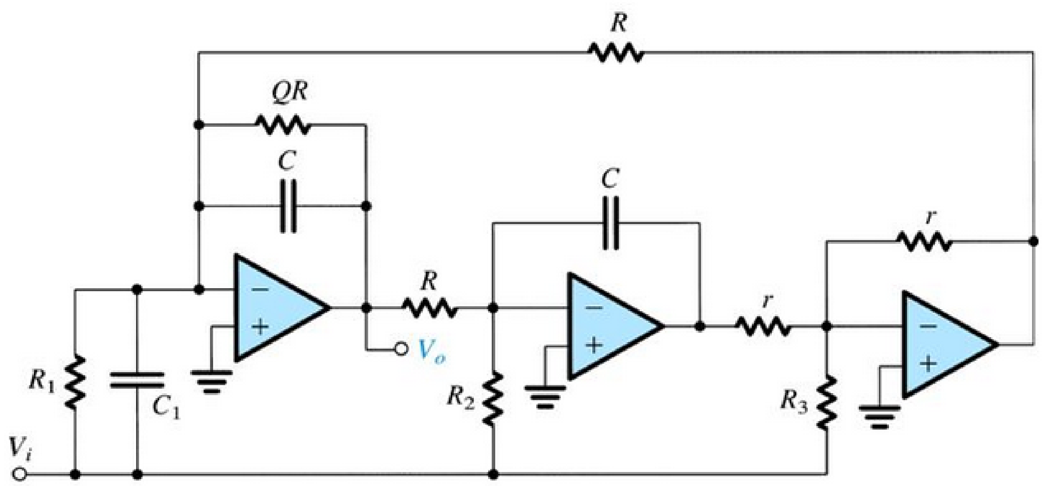

Figure 3: Compensator design: Tow-Thomas biquad filter
source: Microelectronic Circuits by Sedra Smith

We design a second order **Tow-Thomas biquad filter** which acts as the compensator. We decide our required cut offs and roll offs based on the frequency response obtained before. We then simulate the filter along with the system and fine tune our values chosen so that we have a stable system with a **positive gain margin**.

Such a compensator has a transfer function-

$$T(s) = \frac{(\frac{C1}{C})^2 s^2 + \frac{1}{C}(\frac{1}{R_1} - \frac{r}{RR_3}) + \frac{1}{C^2 RR_2}}{s^2 + \frac{1}{QCR}s + \frac{1}{C^2R^2}}$$

We use MATLAB simulations to get the values of $C1, C, R1, R, R2$ to get a stable closed loop system using the compensator.

2.1 Methods

- We first need the transfer function of the Headphone output cascaded with the microphone followed by the Biasing circuit. So, we characterize the systems frequency response (phase and magnitude)

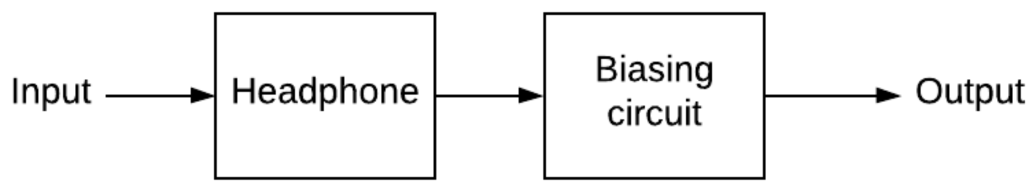

Figure 4: Block diagram for Headphone characteristics

- Next, using the frequency response, we design a gain block so that the gain at 100 Hz frequency is 20 dB.

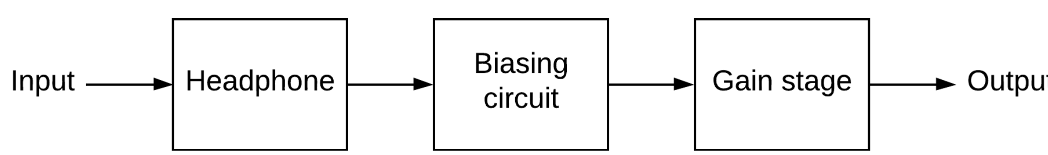

Figure 5: Block diagram for Headphone cascaded with gain

- Then, we characterize this open loop system, and get its magnitude and phase response for a wide range of frequencies.

- Using these responses, we design a **second order** Tow-Thomas biquad filter, by simulating the compensator's transfer function on MATLAB along with the frequency response we have. The values of components we used: (nearest to calculated values)
  $C = C_1 = 10nF$
  $R = 8.2k\Omega$, $R_1 = 8.2k\Omega$
  $R_2 = 8.2k\Omega$, $R_3 = 120k\Omega$
  $Q = 1/14$, $r = 1k\Omega$

3 Block Diagram

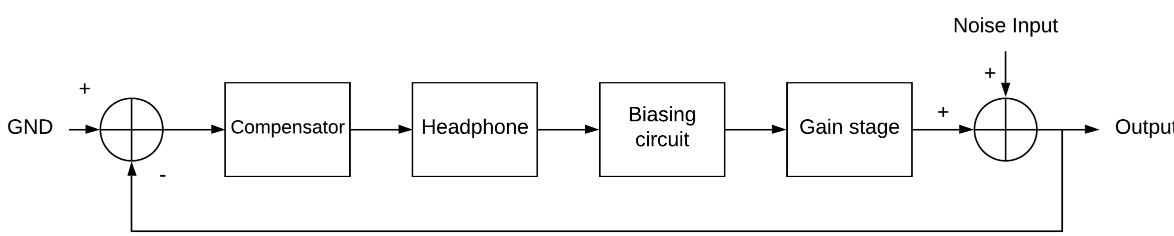

Figure 6: Block diagram for the closed loop noise-cancelling headphones

4 Results

Following are the results obtained:

- The Bode Magnitude Plot and Phase Plot of Headphone cascaded by biasing circuit and the gain block

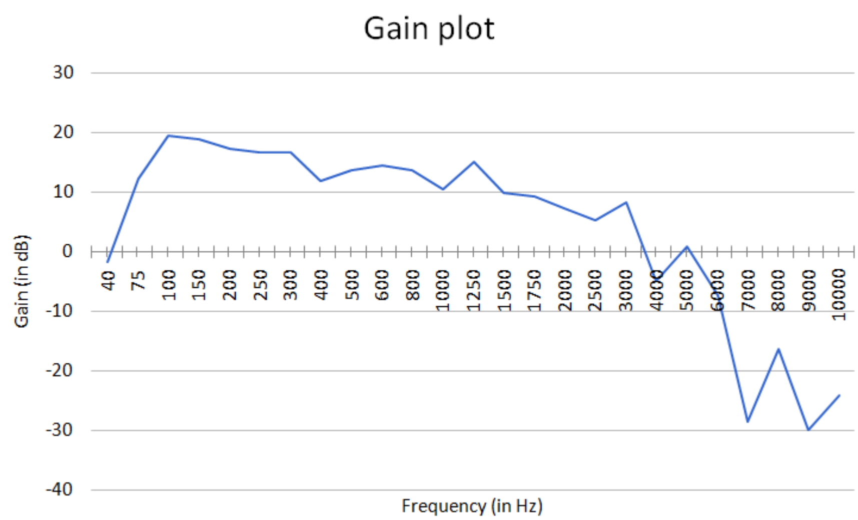

Figure 7: Gain Plot after adding gain

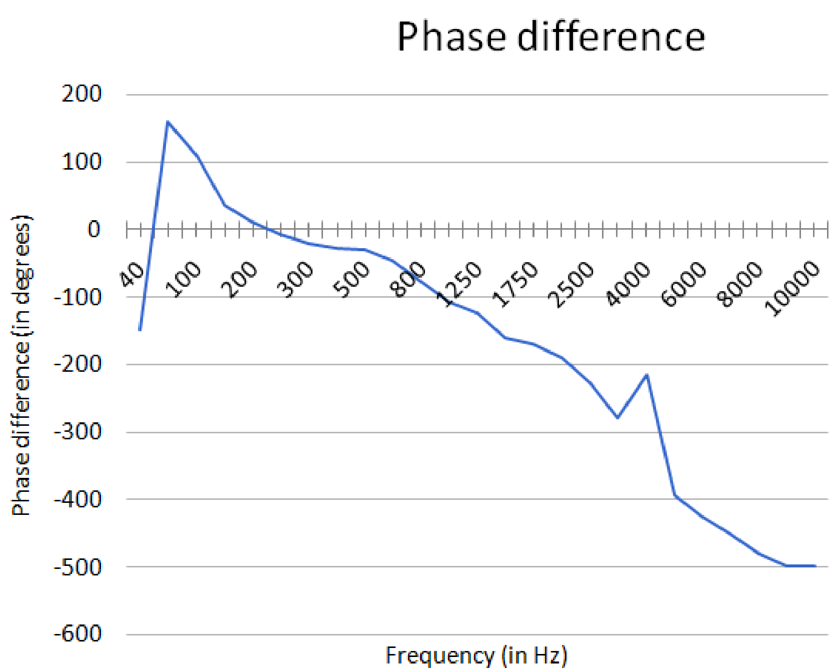

Figure 8: Phase plot after adding gain

- The Plot of Compensator: Tow-Thomas biquad filter obtained via MATLAB Simulation is as follows:

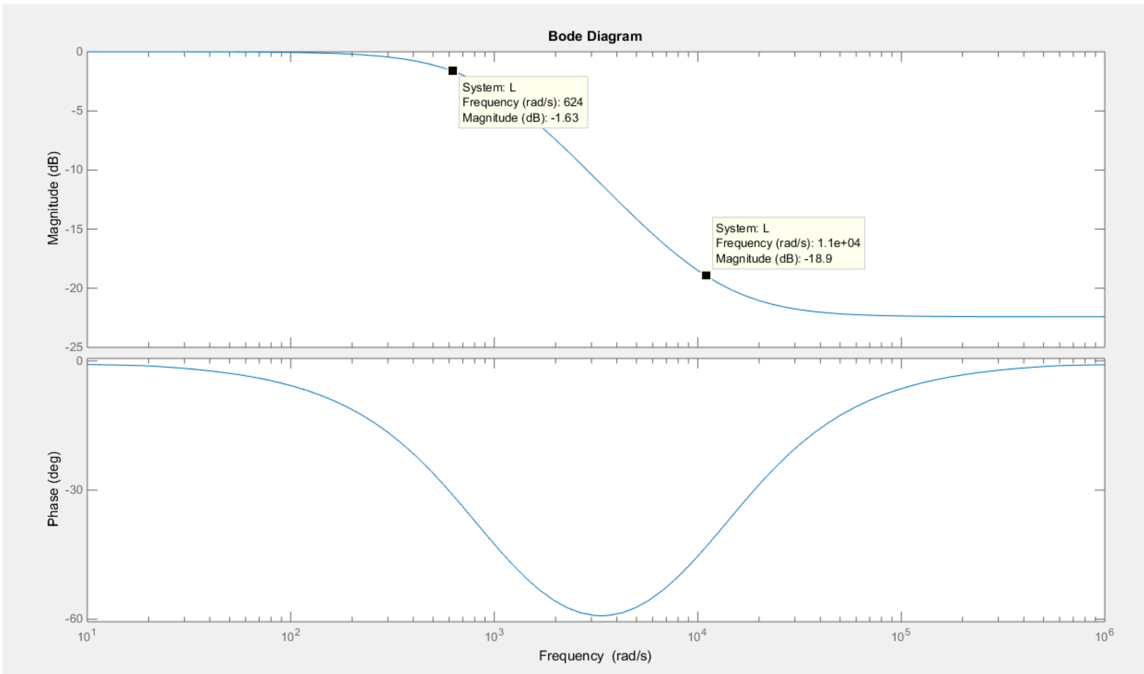

Figure 9: Compensator MATLAB Simulation

- The open loop magnitude and phase response after adding compensator is given in the following image showing clearly what happens at -180^o. We get a gain margin (GM) of 5.4dB

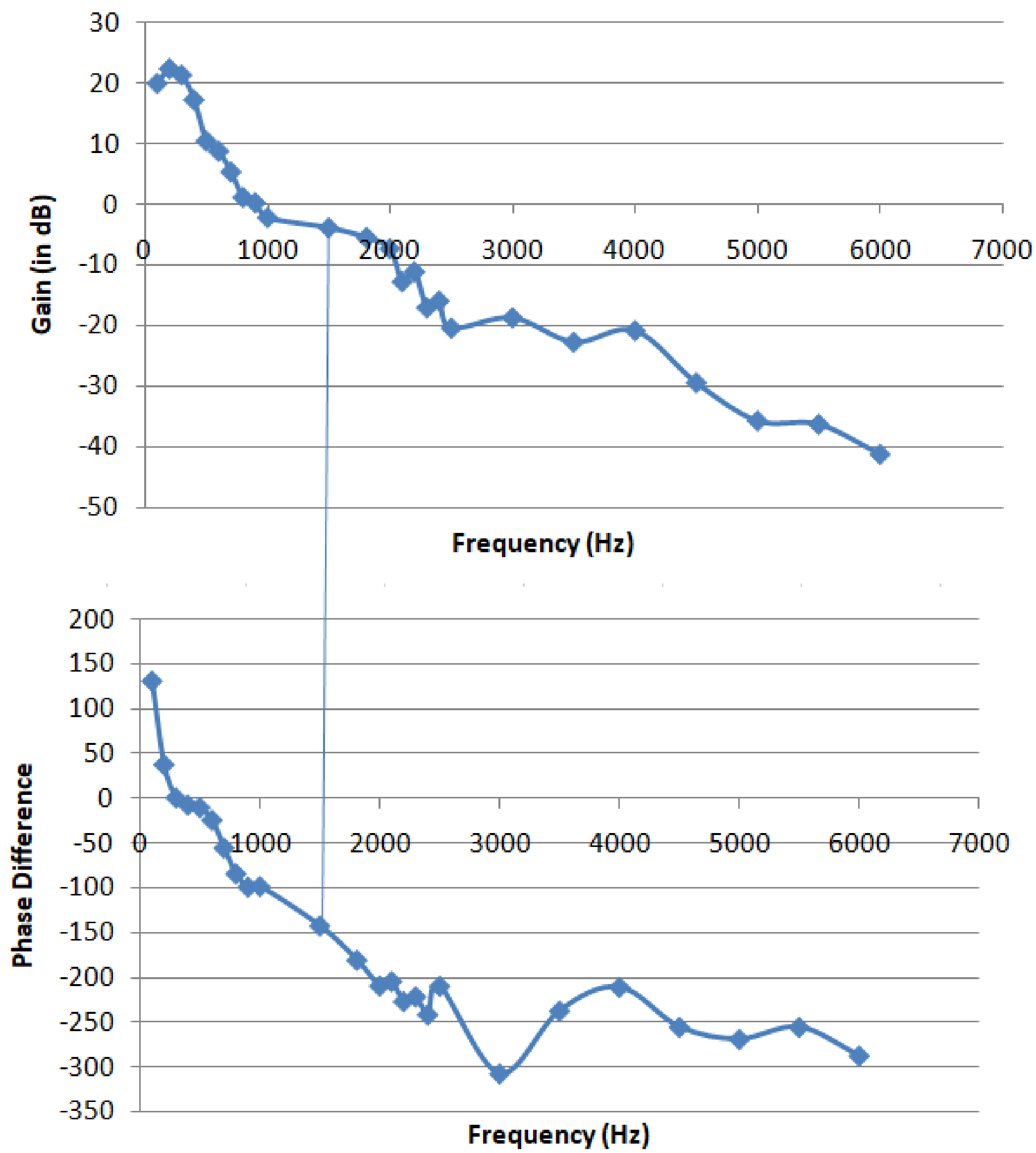

Figure 10: Open Loop characteristics shows that the gain margin is 5.4dB

- The table consisting of our final results is given below.

Frequency (Hz)	$V_{out_{pp}}$ **(mV)**	$V_{in_{pp}}$ **(mV)**	**Gain (in dB)**
100	63.4	920	-23.23
500	88	920	-20.38
1000	104	920	-18.96

Table 1: Result Data Table

5 Challenges Faced

- Design of the compensator was a very challenging task we were not able to get a stable system for 1^{st} order so we moved to 2^{nd} order compensator.
- The bode plot of the headphone would change every lab session, so we had to ensure that we finished our experiment in one go.
- Ambient noise would change the bode plot to a great extent so we had to ensure low ambient noise.
- Fine-tuning the compensator parameters was very challenging as we never used to get exact vales for the components.

6 Conclusion

With a careful design of noise-cancellation we have achieved a significant noise reduction for a noise frequency of 100 Hz. The future of this project involves making a hybrid system of analog and DSP-based noise cancelling systems. The future work is motivated by the fact that the analog systems are not able to achieve superior performance and DSP-based systems are slow in achieving a good performance. So a hybrid may be able to achieve the best of both realms.