Noise Cancellation in headphones

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1 Objective

- The objective of this experiment was to achieve an attenuation of 20 dB at 100 Hz frequency for the given plant(Headphones connected with a biasing circuit) using analog circuitry alone.
- To design an analog compensator to stabilize the response of the headphones.

2 Mathematical Modelling

We used MATLAB to model the system, the compensator circuit and the compensated system to find the best compensator for our system and the target response. We manually wrote the data from the uncompensated system in MATLAB and stored it in Data a 32×3 matrix with the three entries being [frequency, gain, phase]. Then we plotted the data in log sale

```
frequency = data(:,1);
magnitude=data(:,2); phase = -
data(:,3); phase_rad = phase;
magnitude_dB = 20*log10(magnitude); semilogx(frequency,magnitude_dB);
semilogx(frequency,phase_rad);
```

We then defined a 2nd degree compensator transfer function using poles and zeros. The values below are post tuning.

```
K=0.01; z1=2000*2*pi;
z2=1000*2*pi;
p1=100*2*pi;
p2=50*2*pi;
S=tf([K -K*(z1+z2) K*z1*z2],[1 (p1+p2) p1*p2]);
```

We then extract the magnitude and phase at the corresponding frequency points which we observed and apply the compensation on it. We then observe the final compensated system to find if it has achieved the said requirements.

```
[magnitude_tf,phase_tf,wout]=bode(S,frequency); bode(S);
semilogx(frequency,squeeze(20*log10(magnitude_tf)));
semilogx(frequency,squeeze((phase_tf)));
magnitude_final=magnitude_dB+squeeze(20*log10(magnitude_tf));
phase_final=squeeze(phase_tf)+phase_rad;
semilogx(frequency,magnitude_final); semilogx(frequency,phase_final);
```

3 Compensator Design

Observing the bode plot we constructed, the phase margin was a whooping -106^0 which was diffcult to be compensated solely using a single pole-single zero compensator. As a result we decided to go for a Lead-Lag compensator which would compensate our system correctly. Below is the circuit diagram for the same:

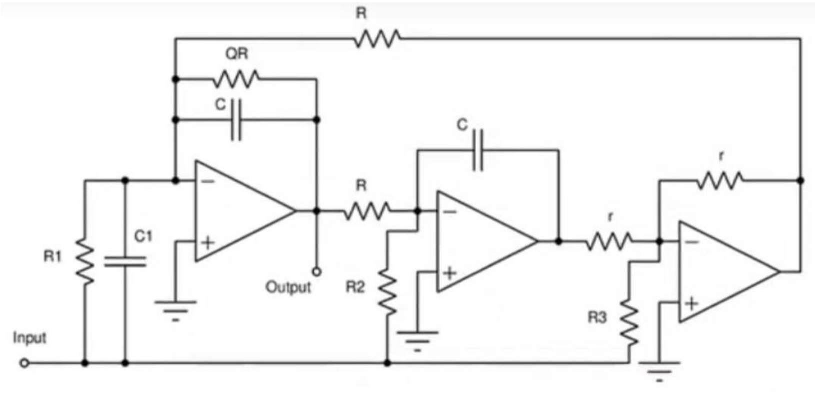


Figure 1: Lead-Lag compensator circuit

Below are the values for the circuit parameters used:

Component	Value
C_1	470pF
C	4700pF
R	220KΩ + 5.6KΩ
R_1	3.3kΩ + 1.5KΩ
R_2	120KΩ
R_3	470Ω
r	47KΩ

Table 1: Circuit Design Parameters

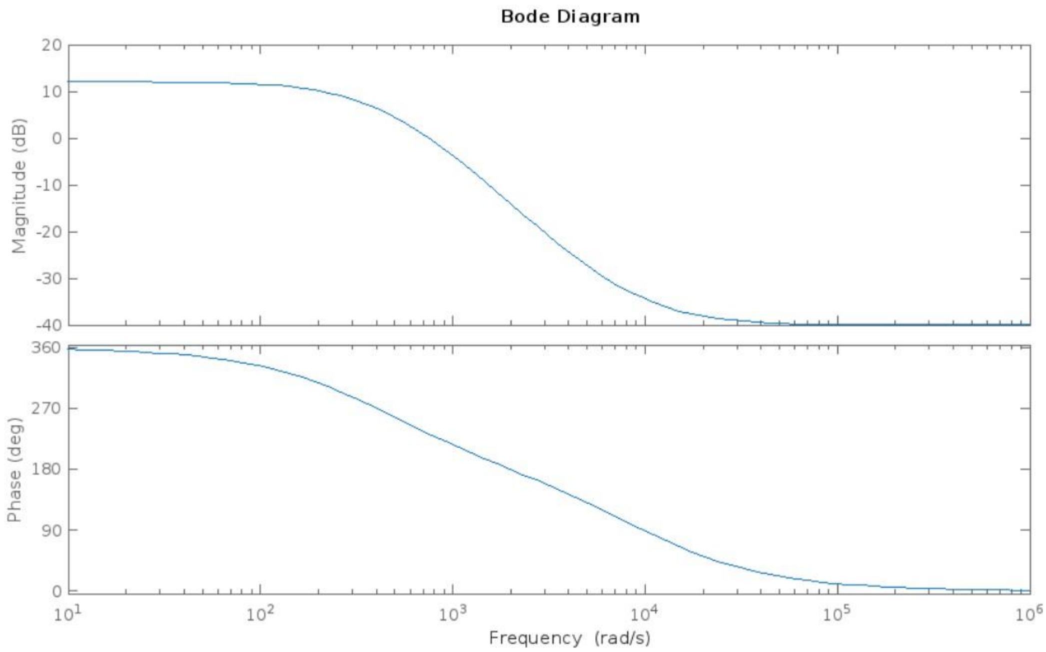


Figure 2: Ideal Bode Plots of the intended Lead-Lag compensator

4 Challenges Faced

4.1 Noise in Measurements

The entire system was disturbed heavily by any speech in the the lab. The idea was to have an airtight casing in which the headphones would be kept such that all of the vibrations only from the headphones would be transfered to the measurement sensor through which we record the output waveforms.

In our case, a tight smack on the plastic box did the trick. However, the readings were yet noisy at 100Hz value and were quite steady in the 1-3KHz range.

4.2 Instability of the given plant

Looking at the plots we constructed using MATLAB, we found that the phase margin was negative with a value of -106^0 which was peculiar to many of the lab TAs too. However, using a second order Lead Lag compensator we were able to compensate the phase quite well.

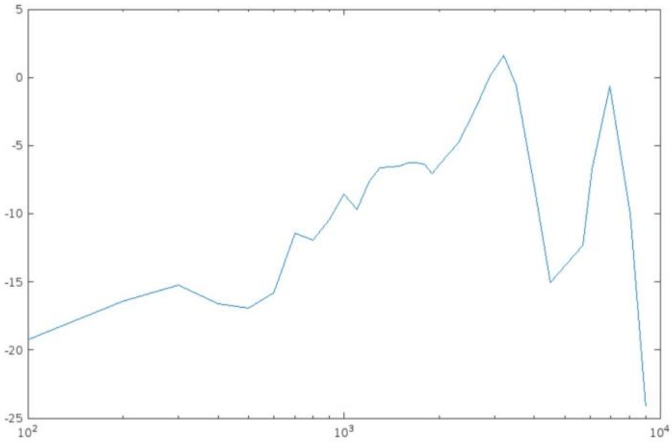
4.3 Choosing an appropriate Compensator

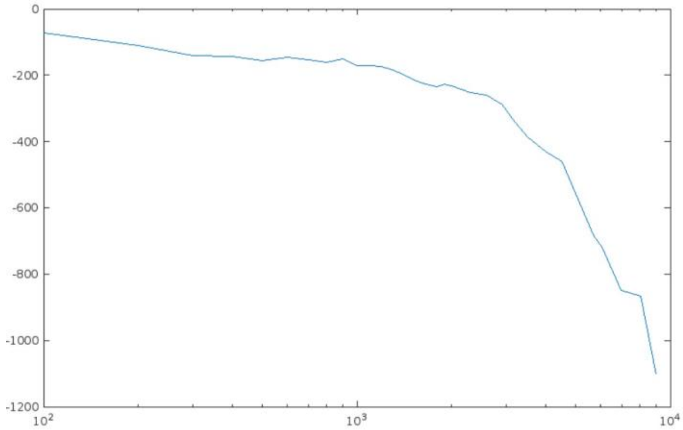
The single pole-single zero compensator is able to provide a 90^0 phase lead at best which is not able to provide positive phase margin.Even spacing 2 left half zeros for a 106^0 phase lead seemed a difficult task, theoretically tuning and let alone practically having such elegant results. As a result we decided to stick with a standard 2^{nd} order lead lag compensator.

4.4 Fixing small frequency gain

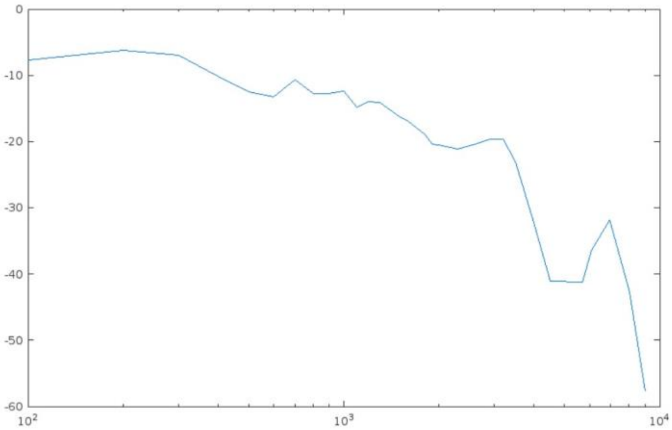
The original headphones happened to have a large attentuation for the 100Hz frequency value. We were able to lift this up to 0dB or unity gain using the base compensator design. To further amplify this nominally we used an non-inverting amplifier at the output of the compensator. This preserved the phase to a good extend and did a good job on the gain as well. We were able to achieve a 12-13dB gain with no hesitation.

5 Result

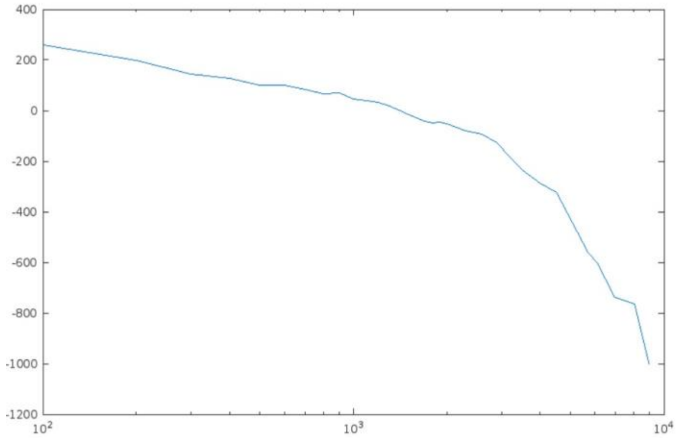




(a) Magnitude response of the uncompensated plant (b) Magnitude response of the uncompensated plant

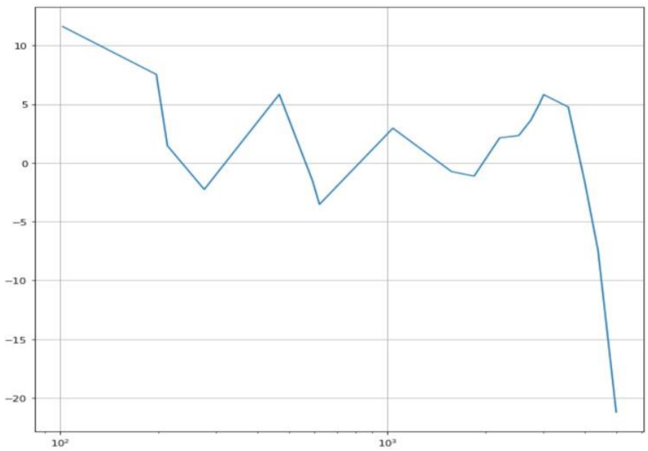


(c)IdealMagnituderesponseofthecompensated plant

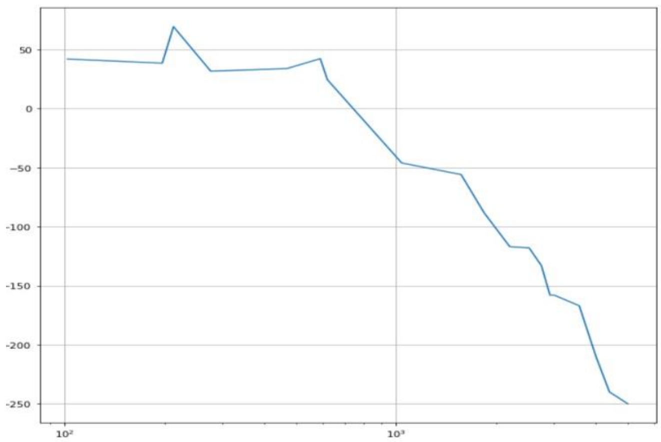


(d) Ideal Phase response of the compensated plant

Figure 3: Plant System and Ideal Response

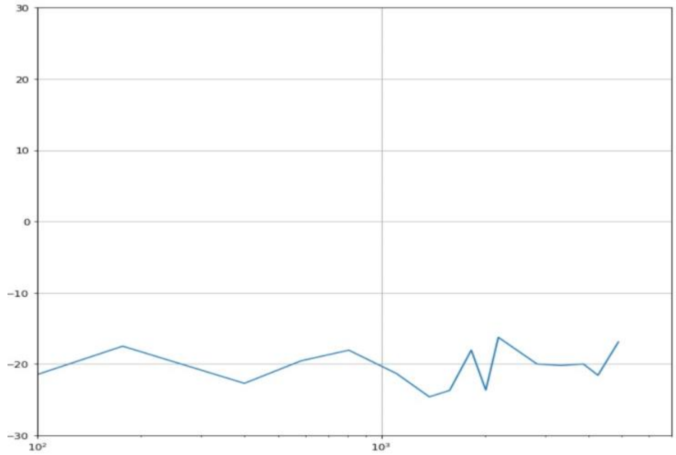


(a)ActualMagnituderesponseofthecompensated plant

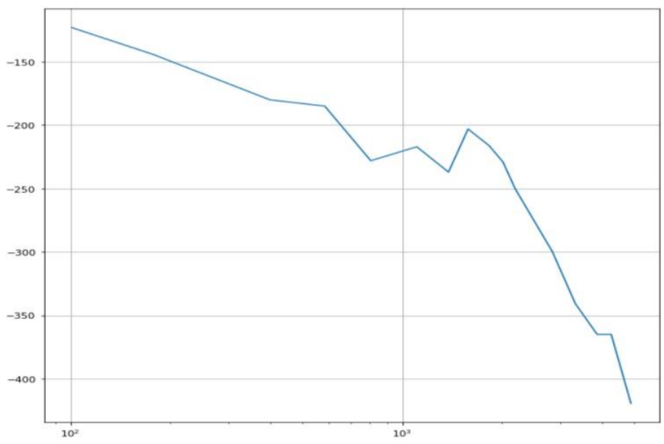


(b) Actual Phase response of the compensated plant

Figure 4: Open loop Response



(a) Actual Magnitude Response after unity feedback



(b) Actual Phase Response after unity feedback

Figure 5: Closed loop Response

6 Observations and Inferences

6.1 Issues with large passive analog circuitry

All of us know that passive components come with a large tolerance rating especially cheap ones. This is typically in the 10-20% range. To add to the misery we know that the breadboard rails due to its parallel strip structure ends up adding stray capacitance and resistance. As a result such errors cascade to large deviations in intended design post primary amplification stage.

As a result the experimental bode plot is in no way similar to the one designed in MATLAB. However it performs the job of a roll-off pretty well in the higher frequency band which will be demonstrated with an oscilloscope image.

6.2 Tuning the circuit

We were pretty confident the implementing the textbook circuit was not going to yield us the complete objective hence we decided to critically analyze the response of the textbook circuit. We found that although the 100Hz response was not 20dB, the gain at 100Hz was pretty good compared to the response of the plant which gave such a small response that was masked by noise itself.

Thus we went about designing a non-inverting amplifier(as the phase was already well compensated) and received a final response of about 12-13dB.

6.3 Understanding system limitation

We do realized that while loop shaping we face a rather complicated trade-off. In layman terms, if we improve the frequency response in one region of the spectrum, the remaining bands experience a rather worsened frequency response. This trade-off is governed by the **Bode Sensitivity Integral**.

6.4 Implementing Closed-Loop Feedback

Post designing an open loop compensator we were asked to connect the feed-forwards system back to its input in a unity gain negative feedback configuration and analyze the results. We found that the gain margin of the system was surely stability due to this. Also to mention that there was no gain crossover in this setup, the reason begin that the magnitude of the transfer function was such that at any value of G(s) the system would not end up crossing the 0dB point and would always remain negative.