# Experiment 4

# Noise cancellation in Headphones

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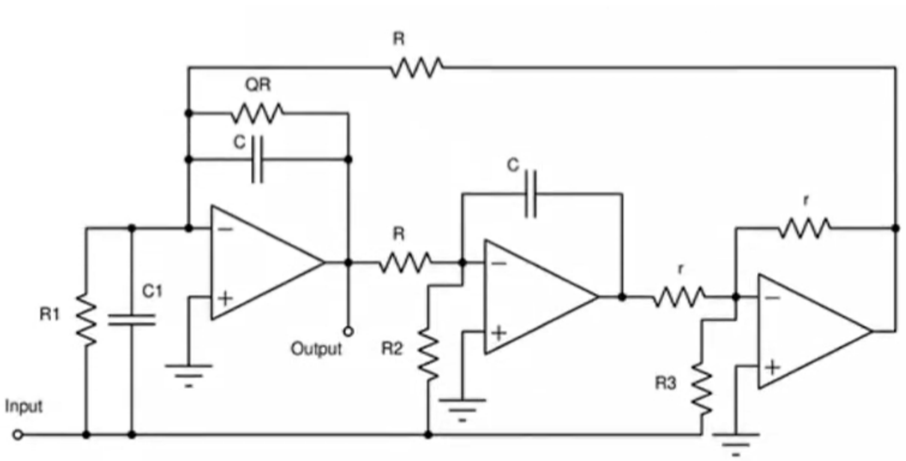
## Contents

# 1 Objective

- To design and implement an analog circuit for noise cancellation in headphones
- To achieve an attenuation of 20 dB, when a noise of 100 Hz frequency is applied
- To design an analog compensator to stabilize the system, i.e. loop shaping of the loop transfer function

# 2 Compensator Design




We first obtained the bode plots of the system without compensator and observed its characteristics to determine the transfer function of the compensator to use .
Transfer function for the above looks like this:

$$T(s)=\frac{V_o(s)}{V_i(s)}=\frac{\left(\frac{C1}{C}\right)^2 s^2+\frac{1}{C}\left(\frac{1}{R1}-\frac{r}{R.R3}\right)s+\frac{1}{C^2.R.R2}}{s^2+\frac{1}{Q.C.R}s+\frac{1}{C^2.R^2}}$$

Figure 1: Second-order Transfer function

The transfer function we used is as follows :

$$\frac{0.00197(s/53313+1)(s/13427+1)}{(s/1164+1)(s/1074.19+1)}$$

The zeros were placed 40-50 times away from origin as compared to poles which were placed near 100 (100 and 90). The cascaded design should have provided

20.59 dB gain theoretically at 100 Hz frequency.
From this transfer function we determined the corresponding values of resistors and capacitors and implemented the above analog circuit.

- $C_1 = 41pF$
- $C = 1.07nF$
- $R_1 = 11K\Omega$
- $R_2 = 48.6K\Omega$
- $R = 911.82K\Omega$
- $R_3 = 10.21M\Omega$
- $r = 998.72\Omega$

Then we moved to closed loop system and followed the same. For closed loop system we designed a subtractor circuit and implemented it on hardware.

## 3 Bode Plots

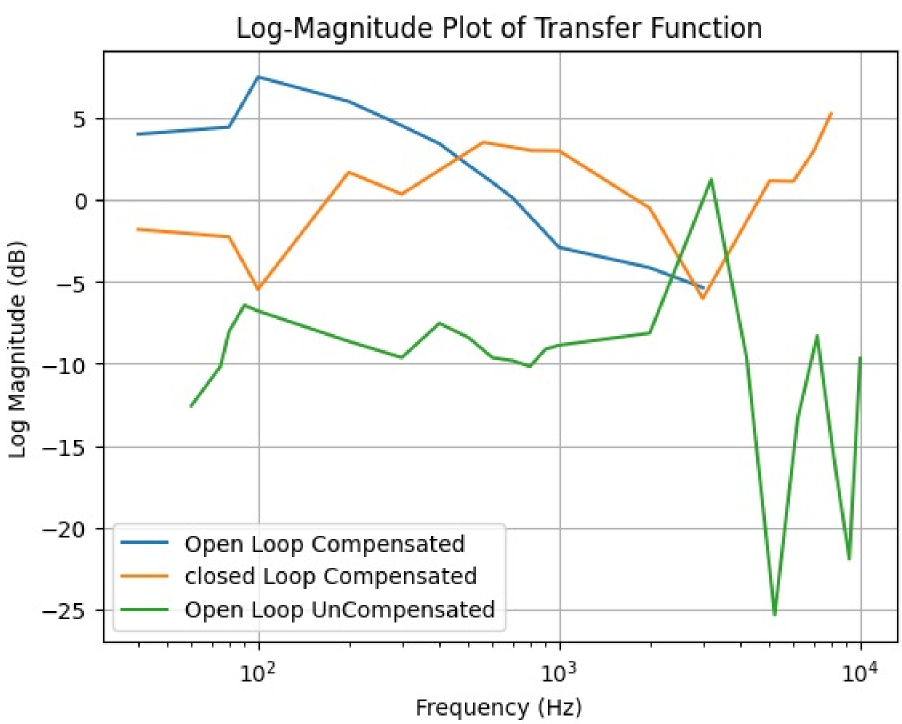



Figure 2: Combined Bode plot

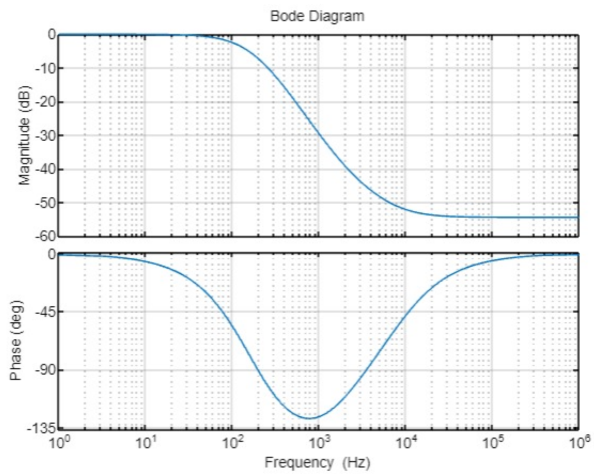



Figure 3: Bode plot of the transfer function

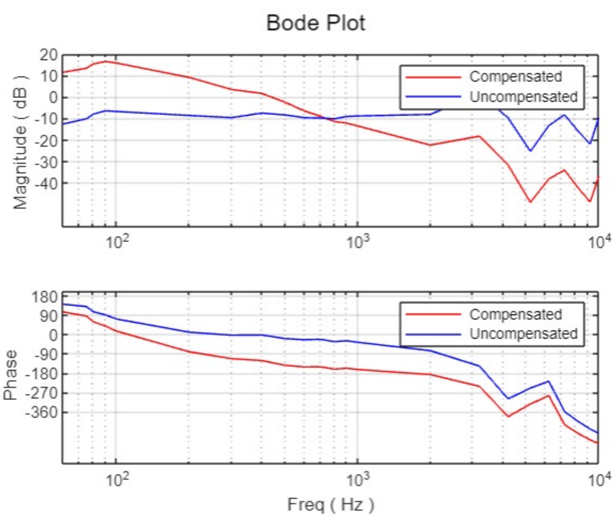



Figure 4: Bode plot compensated on matlab

## 4 Matlab Code

```
freq = [60  , 75, 80, 90, 100, 200, 300, 400, 500, 600, 700, 800, 900, 1000, 2000,
3200, 4200, 5200, 6200, 7200, 8200, 9200, 10000];
vin = [1000,1000,1000, 1000, 1000, 1000, 1000, 1000, 1000, 1000, 1000, 1000, 1000,
1000, 1000, 1000, 1000, 1000, 1000, 1000, 1000, 1000, 1000];
Vout = [235 , 310, 396, 477, 458, 370, 330, 420, 380, 330, 323, 310, 350, 360,
392, 1156, 328, 54, 215, 385, 160, 80, 328];
ph = [140 , 129, 105, 90, 71, 10, -4, -3, -20, -25, -23, -34, -30, -37, -76, -148,
60-360, 110-360, 142-360, 1-360, -45-360, -80-360, -100-360];
gain = 20* log10 ( abs ( Vout ./ vin ) );


figure (1) ;
subplot (2 ,1 ,1) ;
semilogx ( freq , gain ) ; ylabel ( ' Magnitude ( dB ) ') ; grid on ;
subplot (2 ,1 ,2) ;
semilogx ( freq , ph ) ; ylabel ( ' Phase (\circ ) ') ; xlabel ( ' Freq ( Hz ) ')
sgtitle ( 'Bode Diagram') ;

 z = -[8000 , 2000]*6.28;
 p = -[175, 172]*6.28;
 k = 0.00188 ;
 h = zpk (z ,p , k )
 figure (2) ;
 b = bodeoptions ;
 b . FreqUnits = 'Hz';
 bodeplot (h , b ) , grid ;

 s = 1i * freq * 2 *pi ;
 num = s /(8000*6.28); num1 = s /(2000*6.28);
 den = s /(175*6.28); den1 = s /(172*6.28);
 g = 1*( num +1) .( num1 +1) ./(( den +1) .( den1 +1) ) ;
mag = 20* log10 ( abs ( g ) ) + gain +25 ;
 a = angle ( g ) ;
 phase = rad2deg ( a ) + ph ;

figure (3) ;
 subplot (2 ,1 ,1) ;
semilogx ( freq , mag , 'r' , freq , gain , 'b') ; ylabel ( 'Magnitude ( dB ) ')
; grid on ; yticks ( -40:10:40 ) ;
```

```
 legend ( ' Compensated ' , ' Uncompensated ' , ' Location ' , ' southwest') ;
 subplot (2 ,1 ,2) ;
 semilogx ( freq , phase , 'r' , freq , ph , 'b')
 ; ylabel ( ' Phase') ; yticks(-360:90:360) ; grid on ;
legend ( ' Compensated ' , ' Uncompensated ' , ' Location ' , ' southwest ')
; xlabel ( ' Freq ( Hz ) ') ; sgtitle ( 'Bode Plot ') ;


C = 1e-9;   %1pF(Commented)
R = 1/(sqrt(-p(1)*-p(2))*C)
Q = 1/(R*C*(-p(1)-p(2)))
R2 = 1/(k*17.78*-z(1)*-z(2)*C*C*R)
C1 = C*sqrt(k)

% x = 1/R1 - r/(R*R3)
x = C*k*(-z(1)-z(2))
R1 = 1e4;       % 10kOhm(Commented)

% y = r/R3
y = 1/R1 - x
R3 = 1e7;       % 10MOhm(Commented)
r = y*R3
```

**Code Explanation**

- We plot the Bode diagram, which is a graphical representation of the gain (magnitude) and phase response of a system as a function of frequency using the arrays given above.
- The semilogx function is used to plot the data on a logarithmic scale for the frequency axis.
- We define zeros (z) and poles (p) of a system, along with a gain factor (k). These are used to create a zero-pole-gain (ZPK) model of a system, typically a transfer function in control systems.
- A Bode plot is then generated for this model using *bodeplot*, which visualizes how the frequency response (both magnitude and phase) of the system varies with frequency.
- A custom transfer function (g) is created using the frequency array, zeros, and poles previously defined.

- Then we calculate a modified gain (mag) and phase (phase) by incorporating the gain and phase shift from the earlier part, along with additional calculations.
- A Bode plot is then generated for both the compensated (modified) and uncompensated (original) systems, showing the effect of the compensation.
- We calculate values for different components of an electrical filter (like resistance R, quality factor Q, another resistance R2, and capacitance C1).
- We perform some additional calculations to determine the values of other components (x, y, r) in the circuit based on predefined and calculated values.

## 5 Challenges faced

- **Extensive Debugging for Complex Circuit:** Debugging a complex circuit featuring multiple components, including capacitors and resistors, necessitated extensive effort and numerous alterations to component values. This undertaking was time-consuming and demanded a systematic approach to pinpoint and address issues effectively
- **Environmental Variations:** Noise-canceling headphones aim to minimize or eradicate external ambient noise. Nevertheless, the degree of noise reduction achieved is contingent upon environmental conditions. Elements like the intensity of background noise, acoustic properties of the surroundings, and the frequency spectrum of ambient sounds can collectively influence the headphones' effectiveness.
- **Pole and Zero Tuning:** Achieving the targeted 20dB gain at 100Hz through the adjustment of poles and zeros proved to be a intricate and demanding undertaking. Initially, the poles were positioned nearer to the origin, while the zeros were placed deeper into the left half complex plane. Fine-tuning around these parameters presented a substantial challenge.
- **Damaged Box:** Damage to the internal biasing circuit of the headphones resulted in inaccurate output or complete system malfunction. Addressing this issue by repairing or replacing the affected components presented a substantial challenge, demanding specialized knowledge and equipment for resolution

# 6 Results

## 6.1 System

- Gain Margin : 0.17 dB
- Phase Margin : 31.62°
- Gain at 100Hz : -6.21 dB

## 6.2 Compensator Transfer Function

$$\frac{0.00197(s/53313 + 1)(s/13427 + 1)}{(s/1164 + 1)(s/1074.19 + 1)}$$

## 6.3 After Compensator

### 6.3.1 Open Loop

Gain Margin : 5.28 dB
Phase Margin : 44.23°
Gain at 100HZ : 6.09 dB

### 6.3.2 Closed Loop

Gain Margin : 2.92 dB
Phase Margin : 35.6°
Gain at 100Hz : -5.37 dB

## 7 Observations and Inference

The following observations have been noted:

- Phase margin and gain margin are terms used in control systems engineering to quantify the stability and robustness of a feedback control system. They are key indicators that help engineers assess the system's performance and make adjustments to improve stability and response.
  **Phase margin** represents the angular difference between the phase of the system and -180 degrees at the frequency where the magnitude of the open-loop transfer function is unity (i.e., the gain crossover frequency).
  Whereas,**Gain Margin** represents the amount by which the gain of the system can be increased before the magnitude of the open-loop transfer function becomes unity at the gain crossover frequency.

- Second-degree transfer functions provide a steeper roll-off in the frequency response compared to first-degree transfer functions. This means that as you move away from the system's bandwidth or cutoff frequency, the gain decreases more rapidly with a second-degree transfer function. This is advantageous here as a rapid roll-off is desirable to attenuate unwanted frequencies effectively.

- When feedback is applied and the compensator is part of a closed-loop system, the system's overall transfer function changes. The feedback modifies the response of the system, and a particular behavior often observed is the mirror image effect about the horizontal axis in the Bode plot of the closed-loop transfer function. This can be explained by the closed-loop transfer function.