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EN2111 - Project Report - High Fidelity Audio Amplifier

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1 Introduction and Scope of the Project

High-fidelity audio amplification is essential for delivering clear, distortion-free sound in home theater and Hi-Fi systems. Among various amplifier classes, the Class AB amplifier is a popular choice due to its optimal balance between the low distortion of Class A and the high efficiency of Class B. This project focuses on designing a high-performance Class AB power amplifier tailored for home audio applications. The system includes a low-noise pre-amplifier stage to process weak audio signals, a tone control circuit for user-adjustable sound quality, and a robust power amplification stage using either discrete transistors. Negative feedback is employed to reduce distortion and improve linearity, ensuring high audio fidelity across the full frequency range.

The scope of the project includes the complete design, simulation, prototyping, and testing of the amplifier system. Key features comprise adjustable tone control (bass, mid, treble), compatibility with standard 8-ohm speakers, and a regulated power supply (±15V) for stable operation. The implementation process involves circuit design using LTspice , careful component selection, breadboard prototyping, and the development of a cost-effective PCB. Final testing evaluates parameters such as gain, total harmonic distortion (THD ¡ 1%), frequency response (20 Hz – 20 kHz), and output power (10W), validating the amplifier's performance both with and without negative feedback.

2 Block Diagram

The overall structure of the high-fidelity audio amplifier system is designed to ensure clean and accurate amplification of audio signals with minimal distortion. The audio signal path begins with the **Pre-Amplifier** stage, which boosts weak input signals (such as those from mobile phones, audio players, or microphones) to a suitable level for further processing.

This is followed by the **Tone Control** stage, which in this design is implemented as a three-band equalizer providing independent control over bass, midrange, and treble frequencies. This allows the user to tailor the sound output according to personal preferences or the acoustic environment.

The final stage is the **Power Amplifier**, a Class AB amplifier that drives the output load (typically an 8-ohm speaker) by delivering sufficient power while maintaining low distortion and high fidelity.

Together, these stages form a complete signal processing chain from input to output, optimized for home audio applications.

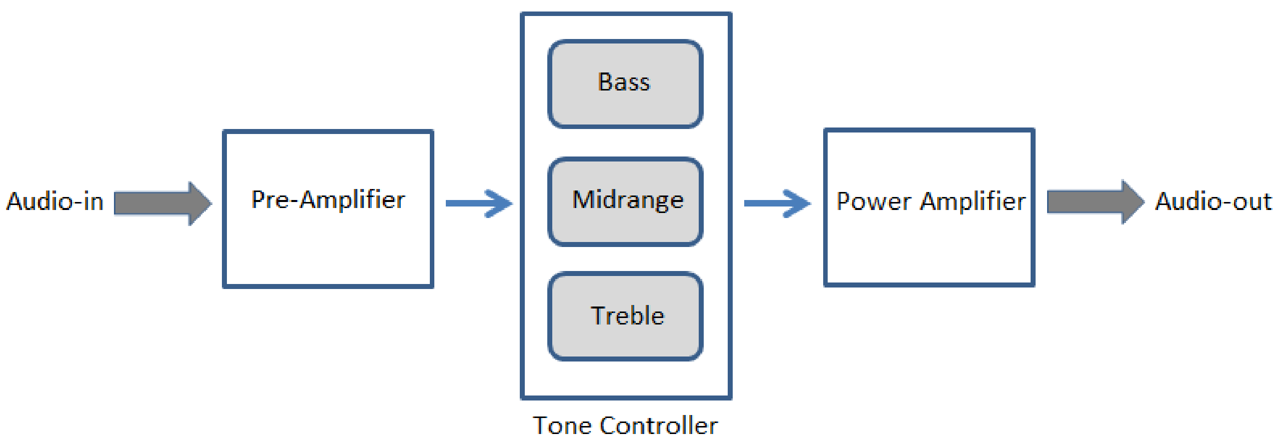


Figure 1: Block Diagram

3 Pre-Amplifier

3.1 Introduction

The pre-amplifier stage plays a crucial role in the overall design of the high-fidelity audio amplifier by boosting weak input signals to a level suitable for further processing. In this project, a non-inverting operational amplifier (op-amp) configuration is used for the pre-amplifier due to its high input impedance and stable gain characteristics, which are ideal for audio applications.

The chosen gain range of 30 to 40 ensures that even low-level signals from sources such as smartphones or audio players are amplified sufficiently before reaching the tone control and power amplifier stages. This gain is adjustable using a variable resistor, enabling fine-tuning according to input signal levels and user preference. The non-inverting configuration also preserves the phase of the signal, which is important for maintaining audio fidelity in multi-stage amplification systems.

3.2 Schematic Diagram

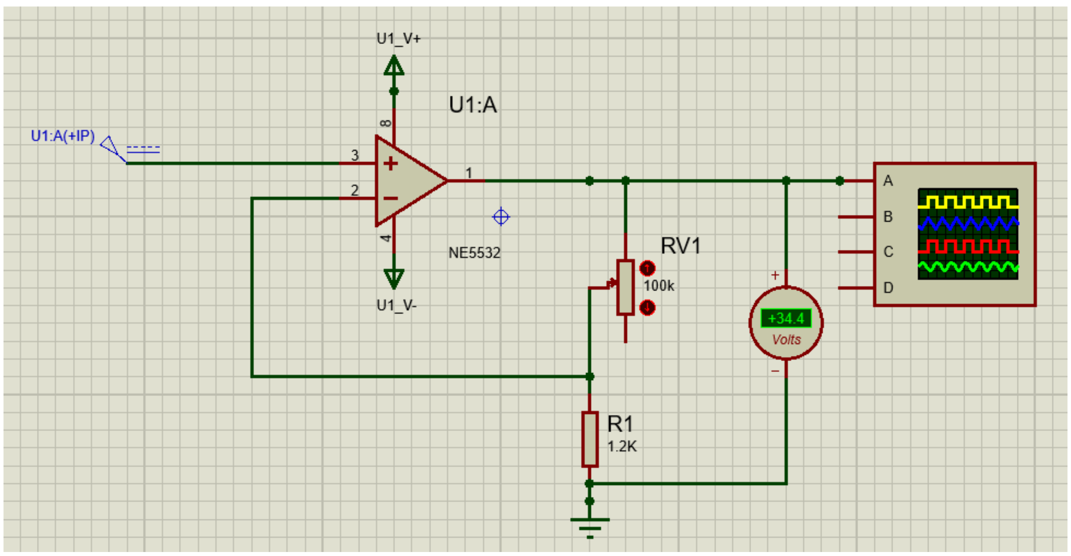


Figure 2: Schematic diagram of the pre-amplifier circuit

The schematic diagram of the pre-amplifier is based on a single op-amp in a non-inverting configuration. The gain of the amplifier is determined by the following equation:

$$\text{Gain} = 1 + \left(\frac{RV1}{R1}\right)$$

Where:

- $RV1$ is the 100kΩ variable resistor (potentiometer),
- $R1$ is the 1.2kΩ fixed resistor.

This formula allows adjustable gain within the desired range of 30 to 40.

3.3 Component Selection

Operational Amplifier (NE5532):
The NE5532 op-amp is selected for this application due to its low noise, high slew rate, and excellent audio performance. It is a dual op-amp IC, making it suitable for stereo applications or future expansion. Its characteristics make it particularly well-suited for high-fidelity audio signal processing, minimizing unwanted noise and distortion in the pre-amplifier stage.

100kΩ Potentiometer (RV1):
Used to adjust the gain dynamically. By varying the resistance, the user can set the amplification factor between approximately 30 to 40, as required for different input signal strengths.

1.2kΩ Resistor (R1):
This resistor sets the minimum gain and works with the potentiometer to define the upper gain limit. Its relatively small value ensures a wide gain range while maintaining circuit stability.

3.4 Simulation

The pre-amplifier circuit was simulated using Proteus simulation software. The simulation verified that the gain of the amplifier could be adjusted within the intended range of 30 to 40, confirming that the design meets the project requirements.

The output signal was amplified cleanly without distortion or clipping, demonstrating the effectiveness of the NE5532 op-amp and the overall circuit design. This validated the circuit's functionality before moving to physical implementation and prototyping.

4 Tone Controller

4.1 Introduction

To shape and enhance the quality of the audio signal before amplification, a tone control circuit was implemented. This section provides users with the ability to adjust bass, midrange, and treble frequencies manually. The system incorporates low pass, band pass, and high pass filters for effective frequency segmentation and control. Each filter stage is carefully designed to cover a specific portion of the audio spectrum to ensure accurate tonal adjustments.

4.2 Schematic Diagram

The tone control circuit comprises three major filter stages connected to an operational amplifier. These include:

- A bass control implemented using a low-pass filter.
- A midrange control using a band-pass filter.
- A treble control using a high-pass filter.

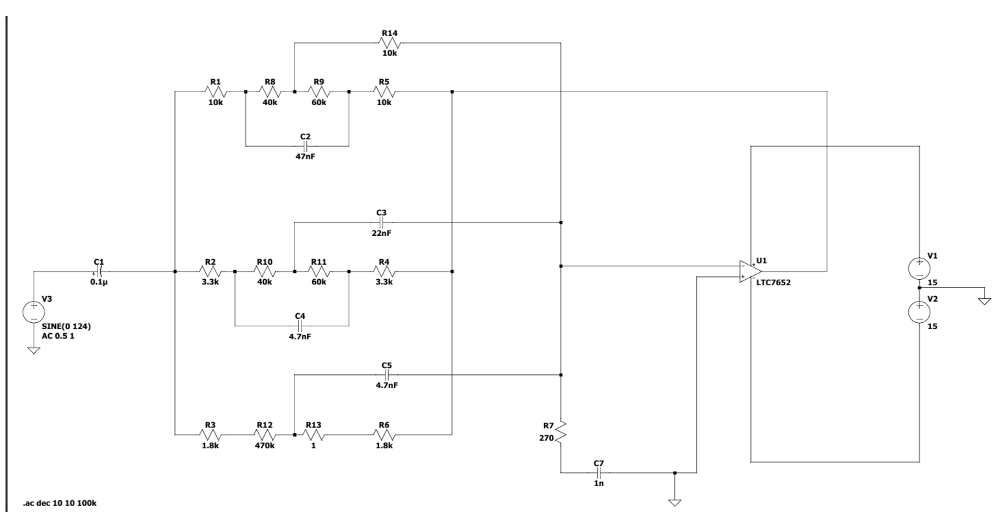


Figure 3: Tone Controller Schematic Diagram

4.3 Component Selection

Passive components such as resistors and capacitors were chosen based on standard availability and desired cutoff frequencies. Variable resistors (potentiometers) were used to allow user control over each band. The operational amplifier used is the NE5532, which is suitable for audio applications due to its low noise and wide bandwidth.

To ensure optimal performance in audio signal processing, several operational amplifiers were evaluated based on critical parameters such as noise, bandwidth, and supply voltage. The NE5532 was chosen for its superior performance in low-noise and high-fidelity audio applications.

Table 1: Comparison of Commonly Used Audio Op-Amps

Parameter	**NE5532**	**LM741**	**TL072**	**LM358**	**OPA2134**
Input Offset Voltage (mV)	0.5	1–5	3	2	0.5
Input Noise (nV/Hz @1kHz)	5	20	18	40	8
Slew Rate (V/µs)	9	0.5	13	0.3	20
Gain Bandwidth Product (MHz)	10	1	3	1	8
Supply Voltage (V)	±3 to ±20	±10 to ±22	±3 to ±18	3–32 (single)	±2.5 to ±18
THD (% @ 1kHz)	0.0005	0.01	0.003	0.01	0.00008
Typical Applications	High-fidelity audio	General-purpose	Audio, filters	Low-power	High-end audio

Justification for NE5532 Selection

- **Low Noise:** With an input noise density of 5 nV/Hz, the NE5532 is well-suited for audio applications where minimal background noise is critical.
- **Low THD:** Its extremely low Total Harmonic Distortion ensures accurate sound reproduction.
- **High Slew Rate:** At 9 V/µs, it handles fast signal changes effectively, preserving signal integrity.
- **Cost-effective:** The NE5532 offers high-performance characteristics at a relatively low cost compared to premium op-amps like the OPA2134.

4.4 Simulation and Frequency Analysis

Simulation of the tone controller was conducted to analyze the frequency response of each filter stage. The tone controller circuit was simulated using LTspice to observe the frequency response across the three adjustable tone ranges: bass, midrange, and treble. By individually varying the potentiometers associated with each filter section, we were able to analyze how the gain responded across different frequencies. The low-pass filter (bass) showed significant attenuation of higher frequencies while amplifying lower ones. The band-pass filter (midrange) effectively isolated a specific frequency band, demonstrating a peak around its center frequency and attenuation outside the band. The high-pass filter (treble) suppressed low-frequency signals while enhancing the higher end of the spectrum. This analysis confirmed the expected tone shaping behavior, validating both the theoretical calculations and the practical implementation of the design.The theoretical cutoff frequencies were calculated using the formula:

$$f_c = \frac{1}{2\pi RC}$$

Bassband (Low Pass Filter):

$$R = 10k + 100k + 10k = 120k\Omega$$

$$f_c = \frac{1}{2\pi \times 120 \times 10^3 \times 47 \times 10^{-9}} = 28.12 \text{ Hz}$$

Midband (Band Pass Filter):

$$\text{Upper Cutoff: } f = \frac{1}{2\pi \times 3.3 \times 10^3 \times 4.7 \times 10^{-9}} = 10.26 \text{ kHz}$$

$$\text{Lower Cutoff: } f = \frac{1}{2\pi \times (100 + 3.3) \times 22 \times 10^{-6}} = 70 \text{ Hz}$$

Treble Tone Control (High Pass Filter):

$$f_1 = \frac{1}{2\pi \times (470 + 1.8) \times 10^3 \times 4.7 \times 10^{-9}} = 71.77 \text{ Hz}$$

$$f_2 = \frac{1}{2\pi \times 1.8 \times 10^3 \times 4.7 \times 10^{-9}} = 18.8 \text{ kHz}$$

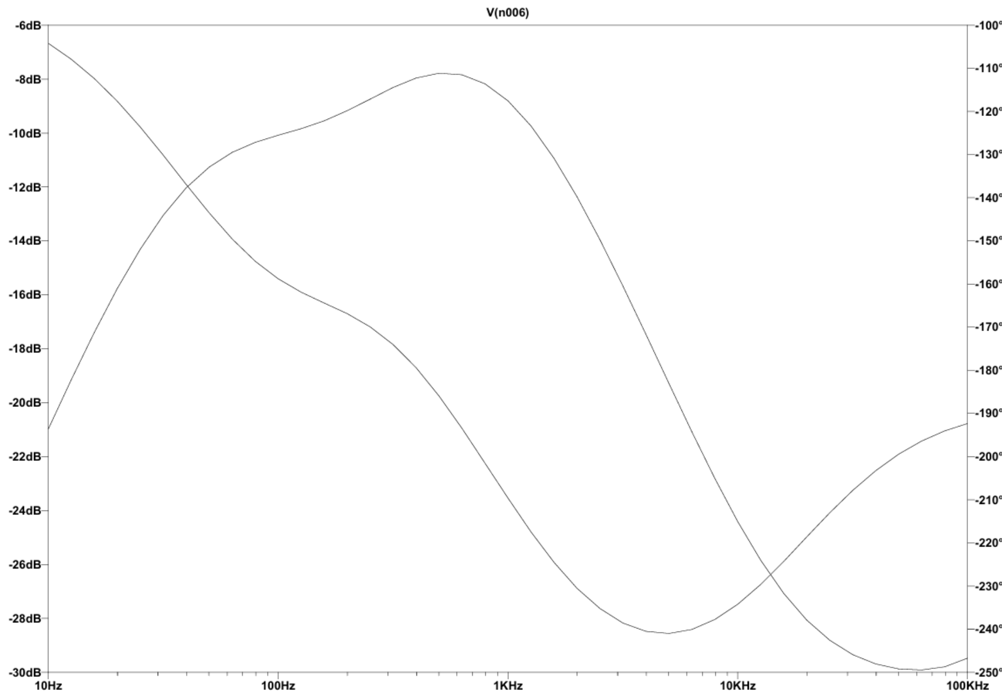


Figure 4: Frequency Response of the Tone Controller when the High Frequencies are Attenuated

4.5 Conclusion

The tone controller successfully enables precise adjustment of audio frequency bands. With a well-designed combination of passive components and operational amplification, the circuit allows users to tailor bass, midrange, and treble levels. The calculated cutoff frequencies fall within ideal audible ranges for home audio enhancement, ensuring clarity and customization of sound output.

5 Power Amplifier

5.1 Introduction

The Class AB power amplifier stage serves as the final and most critical stage in the Hi-Fi audio amplifier, responsible for delivering sufficient power to drive an 8-ohm, 10-watt speaker with high fidelity. Combining the efficiency of Class B operation with the linearity of Class A, the Class AB topology minimizes crossover distortion while maintaining reasonable power efficiency—making it ideal for high-quality audio applications. This configuration uses a pair of complementary transistors operating slightly above cutoff to ensure smooth signal transition between the positive and negative halves of the waveform. The result is a well-balanced trade-off between thermal performance, linearity, and output power, which is essential for accurate reproduction of audio signals in Hi-Fi systems. Also, It should be mentioned that this has a high power efficiency. (a maximum of around 78.5%)

5.2 Schematic Diagram

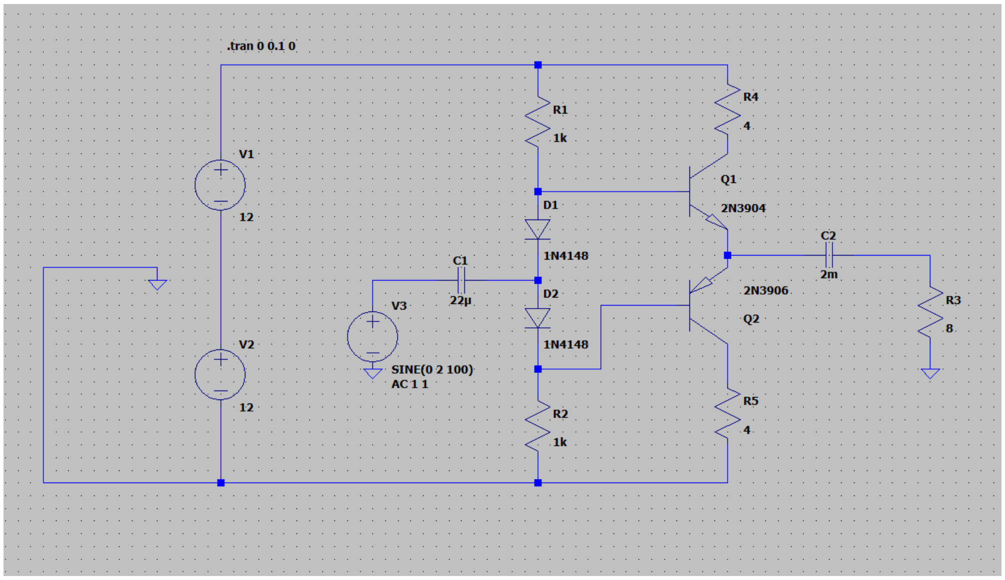


Figure 5: Initial Schematic Diagram

This is an initial schematic diagram used to design the Class AB Power Amplifier and to simulate and verify whether the functionality of the Power Amplifier is achieved.

5.3 Simulation

Here, We used LtSpice to simulate and verify the functionality of the Power Amplifier.

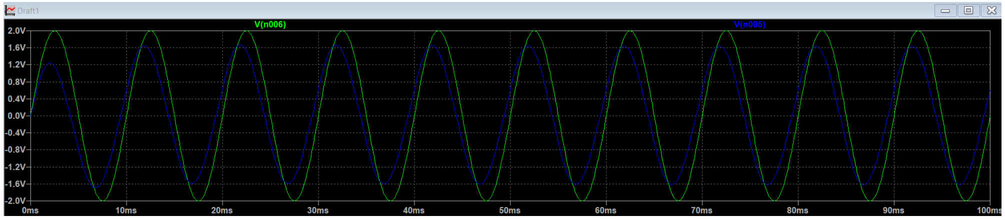


Figure 6: Input Voltage (Green) Vs. Output Voltage (Blue)

Generally in Power Amplifiers the voltage does not vary much. So our design satisfy that, as there isn't a considerable output voltage drop.

When Considering the current gain, a Class AB Power Amplifier should provide a huge current gain to drive the load. As it is visible below we have satisfied that as well.

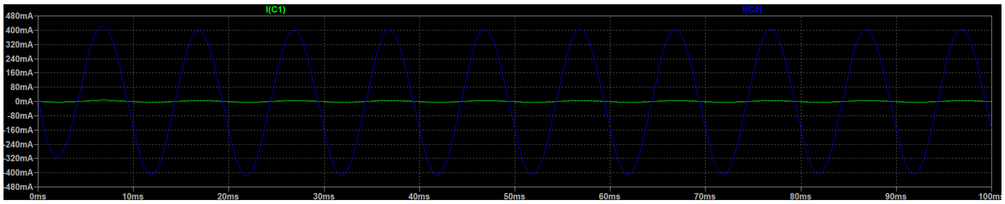


Figure 7: Input Current (Green) Vs. Output Current (Blue)

5.4 Component Selection

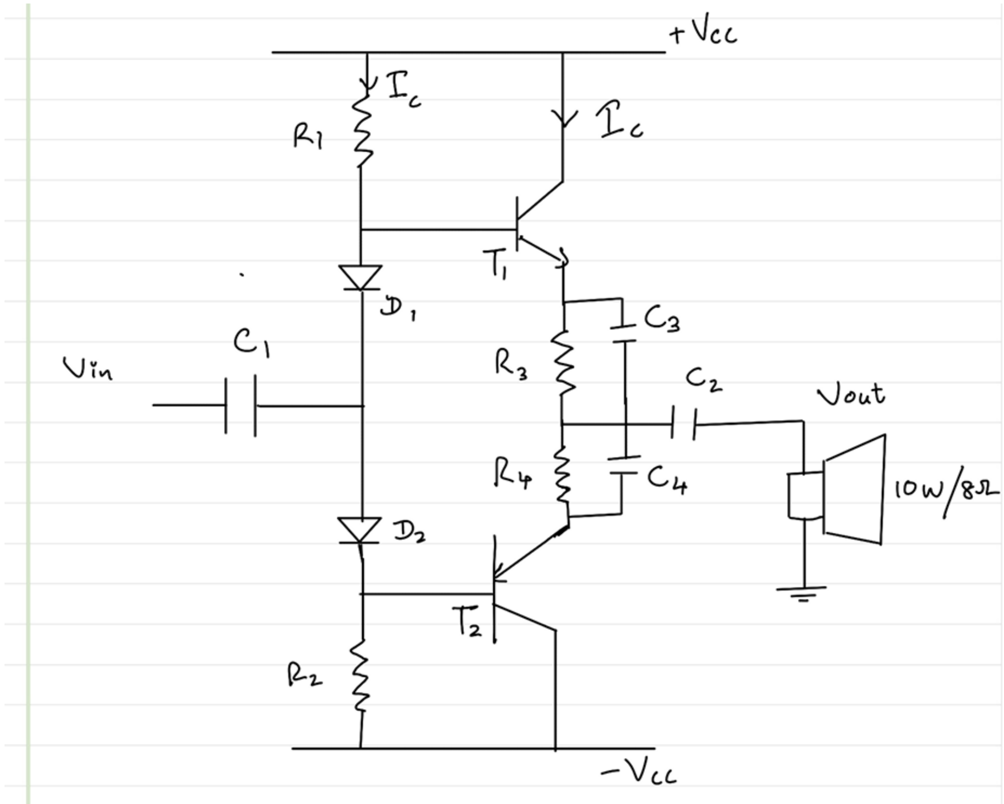


Figure 8: Class AB Power Amplifier Design

Now, let's do the calculations to come up with the required components.

$$P_L = \frac{V_{out}^2}{R_L}$$

$$V_{out,\ rms} = \sqrt{10 \times 8} = 8.95\ V$$

$$V_{out,\ max} = 12.65\ V \approx 13\ V$$

So, We choose, $V_{cc} = 15\ V$.

$$V_{CE,\ 1} = V_{cc} - V_{out}$$

$$V_{CE,\ 1} = V_{cc} - V_m \cdot \sin(\omega t)$$

$$I_c = I_{c,Q} - I_L$$

$$I_c = I_{c,Q} - I_m \cdot \sin(\omega t)$$

$$I_m = \frac{V_m}{R_L} = \frac{13}{8} = 1.625\ A$$

$$P_D = V_{CE,\ 1} \times I_c$$

Since, the calculation gets complexed to find the maximum Power Dissipation. We used the software *Maxima* to get the computations done.

```
phi:atan(2·%pi·f·L/R_L);
```

$$\operatorname{atan}\left(\frac{2\,\pi\,L\,f}{R_L}\right)$$

```
Im:V_m/R_L;
```

$$\frac{V_m}{R_L}$$

```
Vce:V_CC/2−V_m·sin(2·%pi·f·t);
```

$$\frac{V_{CC}}{2}-V_m\sin(2\,\pi\,f\,t)$$

```
Ic:I_Q+Im·sin(2·%pi·f·t);
```

$$\frac{V_m\sin(2\,\pi\,f\,t)}{R_L}+I_Q$$

```
Pd:trigreduce(Vce·Ic);
```

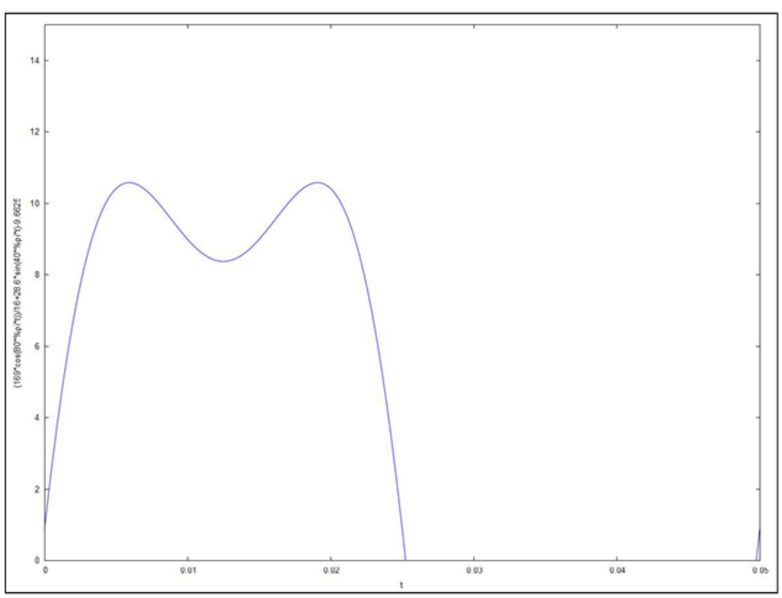
$$\frac{V_m^2\cos(4\,\pi\,f\,t)}{2\,R_L}+\frac{V_{CC}\,V_m\sin(2\,\pi\,f\,t)}{2\,R_L}-I_Q\,V_m\sin(2\,\pi\,f\,t)-\frac{V_m^2}{2\,R_L}+\frac{I_Q\,V_{CC}}{2}$$

```
E1:subst([V_CC=30,V_m=13,R_L=8,f=20,I_Q=50e−3],Pd);
```

Pd

```
E2:wxplot2d(E1,[t,0,5e−2],[y,0,15]);
```

Above is the code used in *Maxima* to compute the maximum power dissipation. Then we plotted the graph for the power dissipation.



As it is visible from the graph, the maximum power dissipation is around 10 - 12 W.

$$P_{D,\ max} \approx 10 - 12\ W$$

After analyzing few Transistor pairs, we found two pairs that will satisfy our requirements and those are especially used in audio amplifiers.

$$T_1 \implies 2SD1047$$

$$T_2 \implies 2SB817$$

Both these transistors has below core specs. that satsify our requirements.

$$P_{D,\ max} = 100\ W > 12\ W$$

$$I_{c,\ max} = 8\ A > 1.7\ A$$

$$V_{CE,\ max} = 140\ V$$

Let's choose **1N4148** silicone diodes for D_1, D_2,

$$V_D = 0.7\ V$$

R_1 is given by the formula,

$$R_1 = \frac{V_{cc} - V_d}{I_{c,Q}}$$

So, with a safety margin we choose,

$$R_1 = R_2 = 1.2\ k\Omega$$

We calculate C_2 as,

$$C_2 = \frac{1}{2\pi f R_L} = \frac{1}{2\pi \times 20 \times 8} = 994.7\mu F$$

So we choose, $C_2 \implies 1\ mF$
For C_2 we choose, $C_2 \implies 1\ mF$

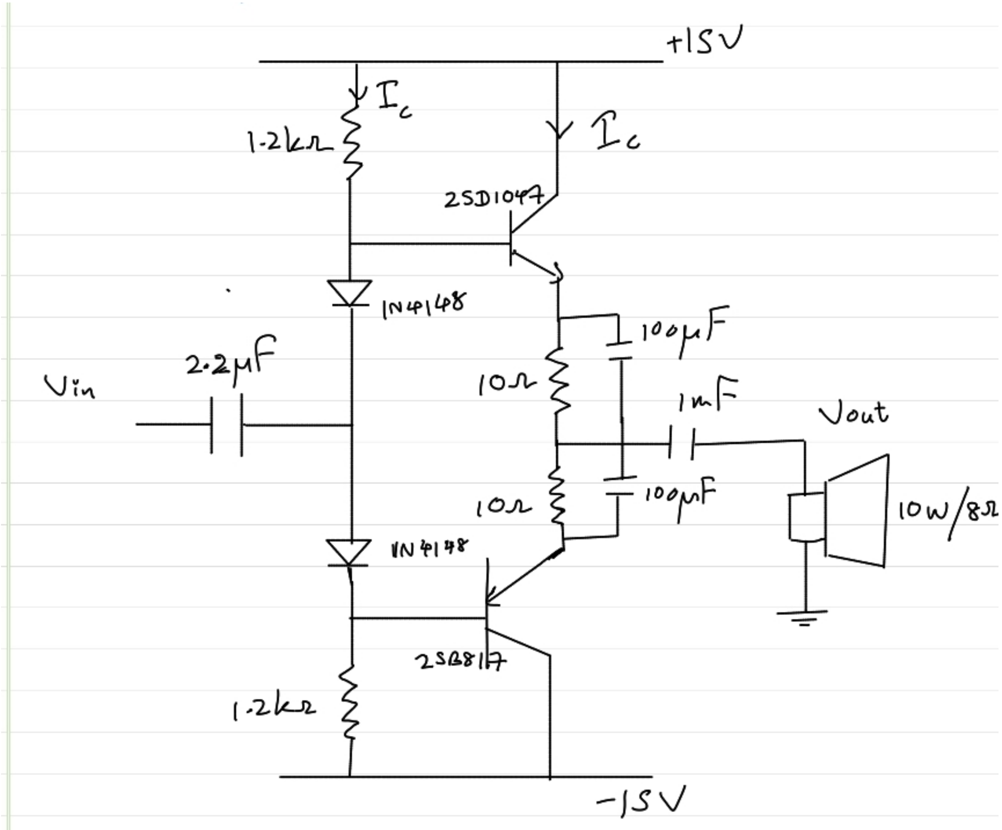
In the final circuitry we have used resistors and capacitors for **Emitter Degeneration**. Therefore, the values we used there were,

$$R_3 = R_4 = 10\ \ (0.5\ W)$$

$$C_3 = C_4 = 100\ \mu F$$

5.5 Final Circuitry

This is the final Circuitry of the Class AB Power Amplifier we implemented,


+15V
Ic
1.2kΩ
Ic
2SD1047
1N4148
100μF
2.2μF
Vin
10Ω
1mF
Vout
10Ω
100μF
10W/8Ω
1N4148
2SB817
1.2kΩ
-15V

6 Breadboard Implementation

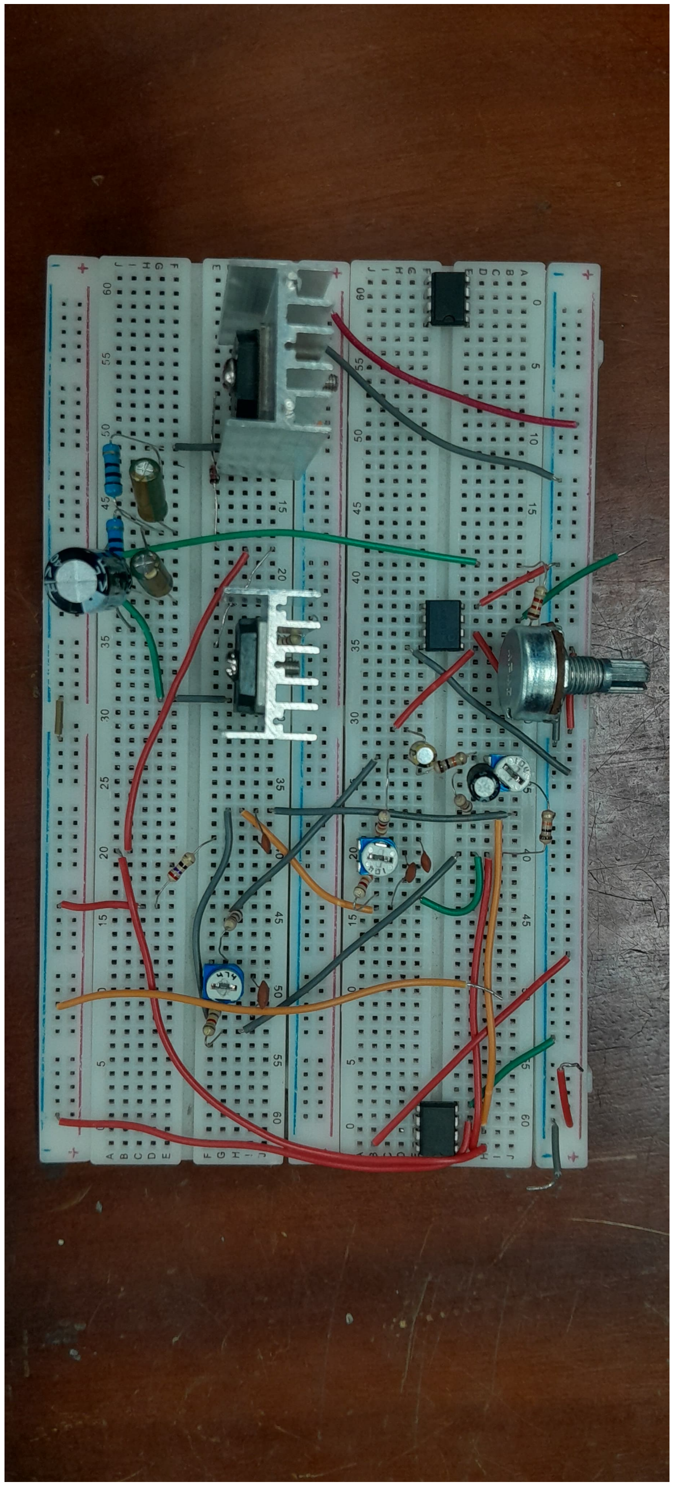


Figure 9: Breadboard Implementation

7 Distortion Analysis

In our Hi-Fidelity Audio Amplifier design, two primary types of distortion are observed at the output stage: **harmonic distortion** and **crossover distortion**.
Harmonic distortion arises whenever the amplifier's transfer characteristic deviates from perfect linearity, generating frequency components at integer multiples of the input frequency.
Crossover distortion, on the other hand, is inherent to push–pull output stages operating around the zero-crossing of the waveform; it appears as a small "dead-zone" near the point where one transistor hands off conduction to its complement.

7.1 Harmonic Distortion

Harmonic distortion occurs whenever the amplifier's active devices (transistors or op-amps) exhibit nonlinear behavior. In practice, the output voltage $v_{\text{out}}(t)$ can be viewed as a Fourier series expansion:

$$v_{\text{out}}(t) \;=\; \sum_{n=1}^{\infty} V_n \,\sin\bigl(n\omega t + \varphi_n\bigr),$$

where:

- V_1 is the amplitude of the *fundamental* component at angular frequency $\omega = 2\pi f_{\text{in}}$,
- V_n (for $n \geq 2$) are the amplitudes of the *nth* harmonic components at $n\omega$, and
- φ_n are the corresponding phase offsets.

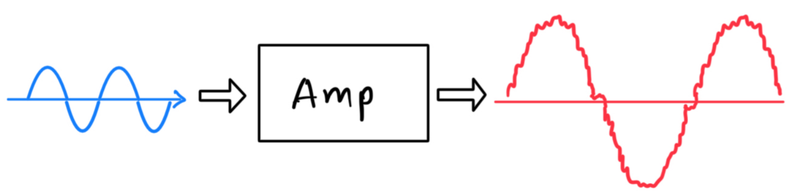


Figure 10: Distortion occuring at otput

Because the amplifier is not perfectly linear, energy "leaks" into these higher-order harmonics $\{2\omega,\, 3\omega,\, \ldots\}$. The Total Harmonic Distortion (THD) quantifies the ratio of the combined RMS amplitude of all harmonic components to that of the fundamental. By definition,

$$\text{THD} \;=\; \frac{\sqrt{\displaystyle\sum_{n=2}^{\infty} V_n^2}}{V_1}.$$

Equivalently, if we denote

$$V_{\text{H}} \;=\; \sqrt{V_2^2 + V_3^2 + V_4^2 + \cdots} \quad ,$$

then

$$\text{THD} \;=\; \frac{V_{\text{H}}}{V_1} \quad\Longrightarrow\quad \text{THD}\,(\%) \;=\; 100\% \;\times\; \frac{\sqrt{\sum_{n=2}^{\infty} V_n^2}}{V_1}\,.$$

7.1.1 Derivation of the THD Formula:

Starting from the Fourier expansion

$$v_{\text{out}}(t) \;=\; V_1 \sin(\omega t + \varphi_1) \;+\; \sum_{n=2}^{\infty} V_n \sin(n\omega t + \varphi_n),$$

we compute the RMS value of each sinusoidal component:

$$V_{n,\mathrm{RMS}} = \frac{V_n}{\sqrt{2}}, \quad n = 1, 2, 3, \ldots$$

By convention, THD is concerned only with the ratio of the RMS sum of the *harmonic* terms to the RMS of the *fundamental* alone. Thus:

$$V_{1,\mathrm{RMS}} = \frac{V_1}{\sqrt{2}},$$

$$V_{\mathrm{H,RMS}} = \sqrt{\sum_{n=2}^{\infty} \left(V_{n,\mathrm{RMS}}\right)^2} = \sqrt{\sum_{n=2}^{\infty} \frac{V_n^2}{2}} = \frac{1}{\sqrt{2}} \sqrt{\sum_{n=2}^{\infty} V_n^2}.$$

Hence,

$$\mathrm{THD} = \frac{V_{\mathrm{H,RMS}}}{V_{1,\mathrm{RMS}}} = \frac{\left(\frac{1}{\sqrt{2}} \sqrt{\sum_{n=2}^{\infty} V_n^2}\right)}{\left(\frac{V_1}{\sqrt{2}}\right)} = \frac{\sqrt{\sum_{n=2}^{\infty} V_n^2}}{V_1}.$$

This matches the familiar form used in our harmonic distortion analysis.

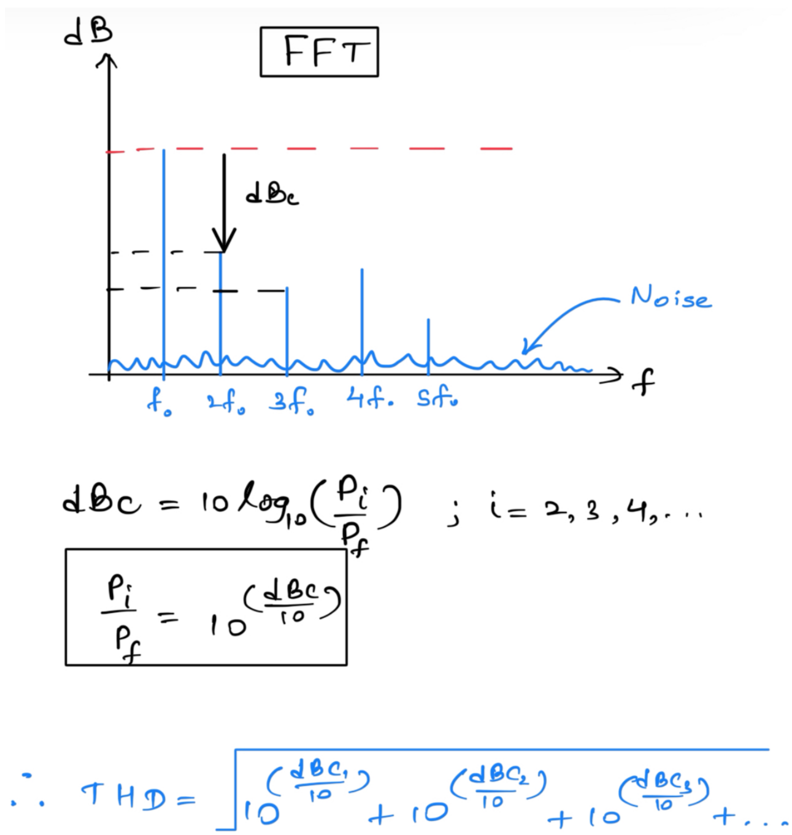


Figure 11: THD derivation for graphical method

7.1.2 Calculation Procedure:

1. *Signal Acquisition:* Apply a pure sinusoidal input of frequency f_{in} (e.g., 1 kHz) to the amplifier under test and measure the output waveform $v_{\text{out}}(t)$ using a spectrum analyzer or FFT-capable oscilloscope.

2. *Harmonic Extraction:* Compute (or read off) the amplitudes V_1, V_2, V_3, ... of the fundamental and the first several harmonics. In practice, harmonics with power less than 30 dB level with respect to the fundamental can be considered as negligible, because higher-order components are typically negligible.

3. *THD Computation:* Compute

$$\text{THD} \;=\; 100\% \times \frac{\sqrt{V_2^2 + V_3^2 + \cdots + V_N^2}}{V_1},$$

   where N is the highest harmonic considered. For example, if only the 2nd and 3rd harmonics are significant, then

$$\text{THD} \;=\; 100\% \times \frac{\sqrt{V_2^2 + V_3^2}}{V_1}.$$

4. *Tabulate Results:* Record the measured values $V_1, V_2, V_3, \ldots$ and compute THD as above. A low THD (e.g., below 0.1%) confirms high fidelity; higher THD indicates more energy in distortion products.

7.1.3 THD Calculation for Each Band

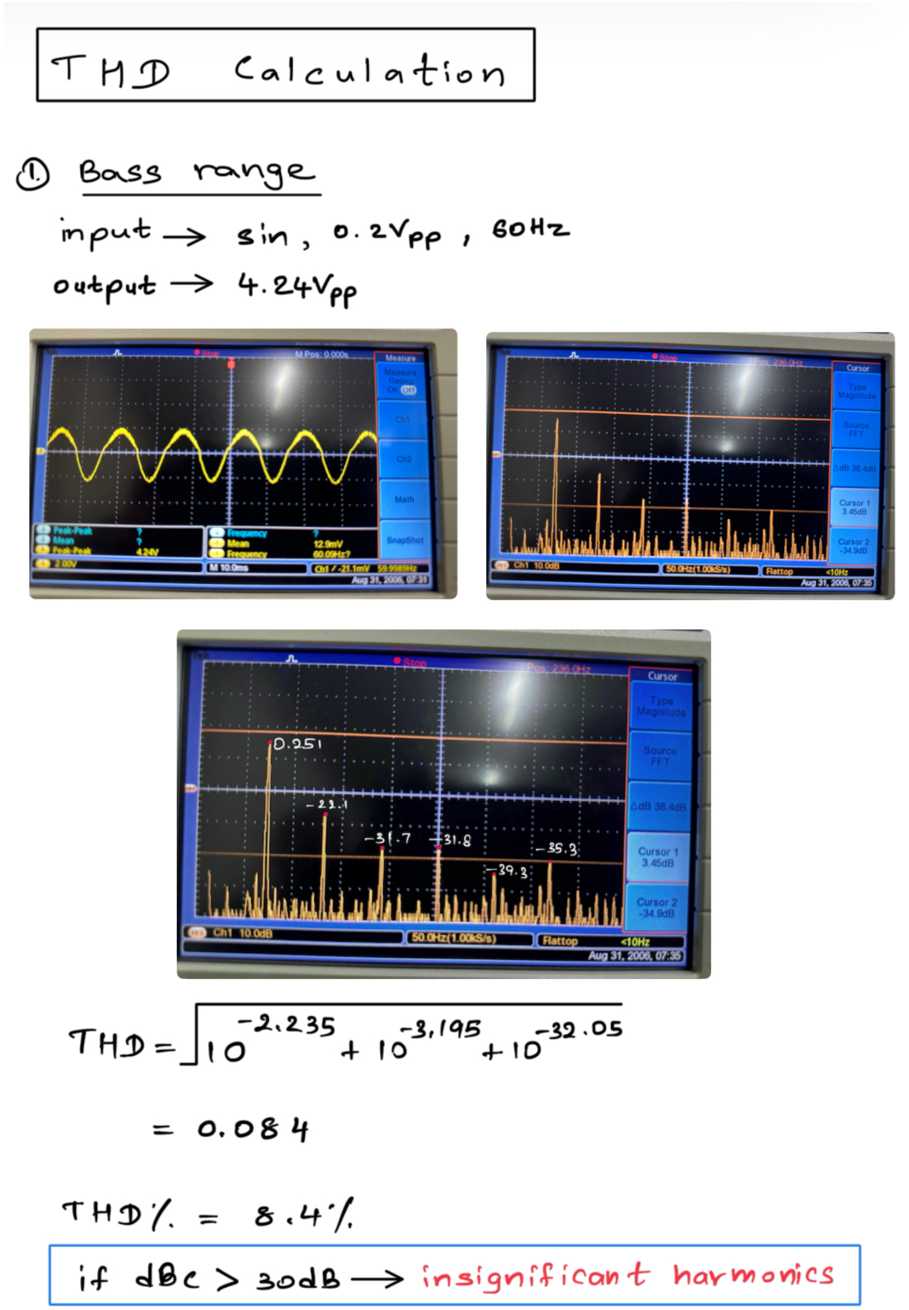
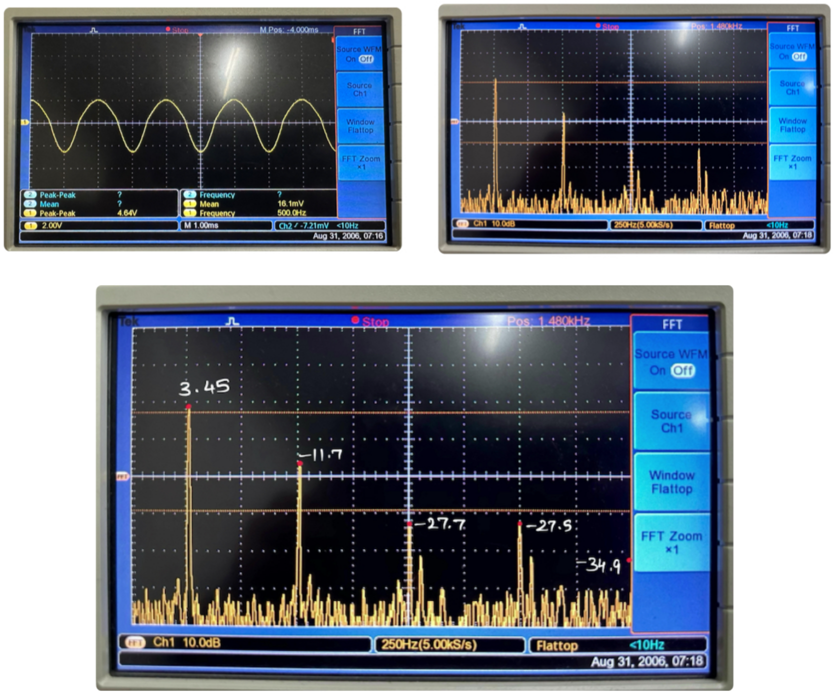


Figure 12: THD at Bass (60 Hz)

② Mid range

input → sin, 0.2V_{pp}, 500 Hz

output → 4.64V_{pp}



$$THD = \sqrt{10^{-1.515} + 10^{-3.115} + 10^{-3.095}}$$

$$= 0.179$$

$$THD\% = 17.9\%$$

Figure 13: THD at Mid (500 Hz)

input → sin, 0.2 V_{pp}, 5 kHz

output → V_{pp}

$$THD = \sqrt{10^{-1.68} + 10^{-2.471} + 10^{-3.071}}$$

$$= 0.1585$$

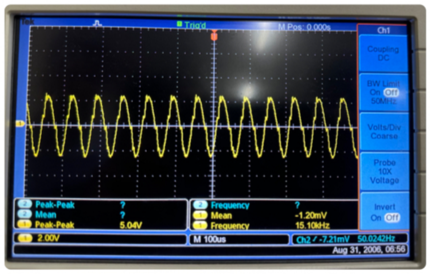
$$THD\% = 15.85\%$$

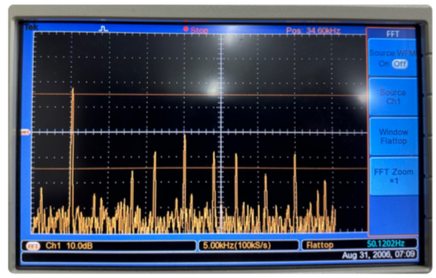
Figure 14: THD at Mid (5 kHz)

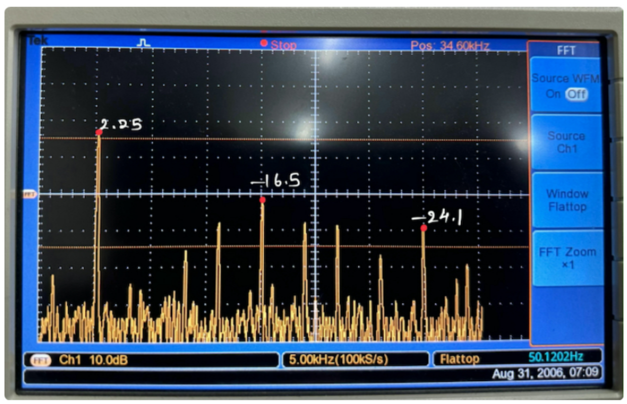
③ Trebble range

input → sin, 0.2 V_{pp}, 15 kHz

output → 5.04 V_{pp}







$$THD = \sqrt{10^{-1.875} + 10^{-2.635}}$$

$$= 0.125$$

$$THD\% = 12.5\%$$

Figure 15: THD at Trebble (15 kHz)

7.1.4 Simulated vs. Real-Time THD

The second graph, *Simulated vs. Real-Time THD*, compares THD values obtained from LTspice simulations with actual hardware measurements.

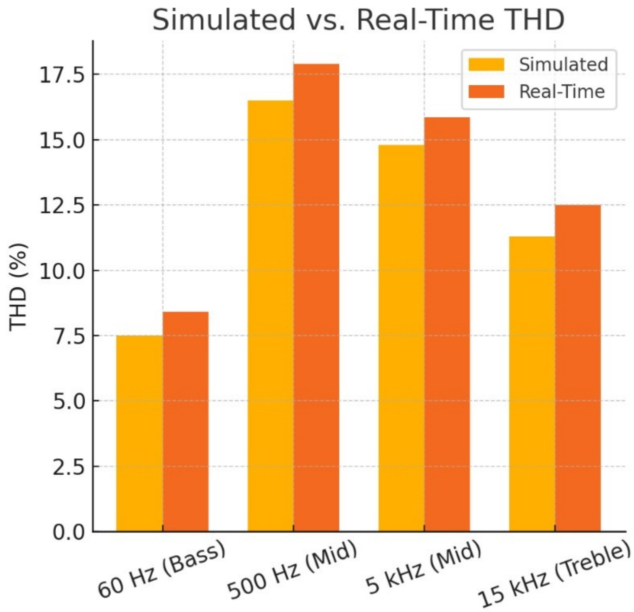


Figure 16: Simulated vs. Real-Time THD across various frequencies

Across all frequency ranges—60 Hz (bass), 500 Hz and 5 kHz (mid), and 15 kHz (treble)—the real-time THD was slightly higher than simulated values, which is typical due to practical non-idealities such as op-amp limitations, PCB noise, and component tolerances.

This comparison helps validate the simulation model and identify areas for potential improvement in the physical design.

7.1.5 Gain vs. THD

To analyze the distortion behavior of our tone control circuit, we generated two comparative graphs. The first graph, *Gain vs. THD*, illustrates how Total Harmonic Distortion (THD) varies with the gain applied to different frequency bands (bass, mid, treble).

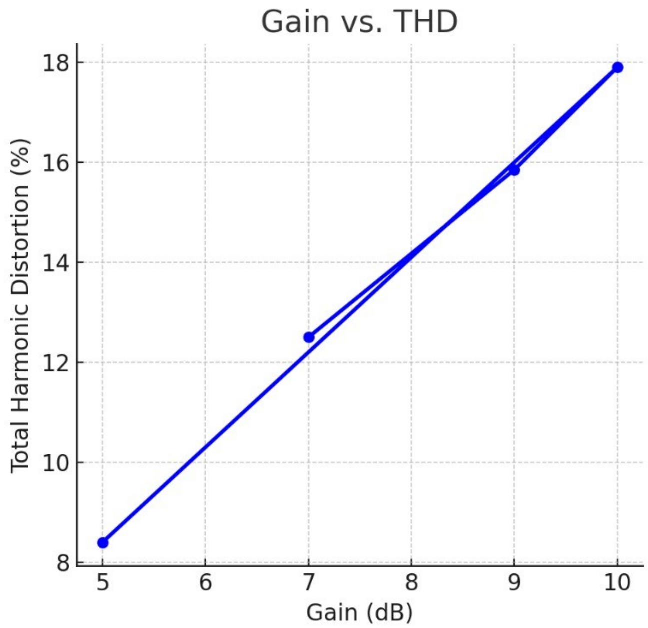


Figure 17: Gain vs. Total Harmonic Distortion (THD)

It shows that higher gain settings generally introduce more distortion, especially in the mid-range frequencies. For example, the THD rises to around 17.9% at 500 Hz under a gain of approximately 10 dB.

7.2 Crossover Distortion

Near the zero-crossing of the input sine wave, both the NPN and PNP output transistors are momentarily *off*, because each transistor requires approximately $V_{\mathrm{BE,on}} \approx 0.7\,\mathrm{V}$ to begin conducting. As a result, the transfer characteristic exhibits a small "notch" or "dead-band" around $v_{\mathrm{in}} = 0$. In mathematical terms, the output voltage v_{out} remains at approximately $+0.7\,\mathrm{V}$ or $-0.7\,\mathrm{V}$ until the input exceeds the transistor's base-emitter threshold. This nonlinearity creates a distortion component that is especially visible at low output levels, and is referred to as crossover distortion.

To mitigate crossover distortion, we typically:

- Introduce a small bias (often implemented with diodes or a VBE multiplier) that keeps both transistors slightly conducting even when the input is near zero.
- Employ negative feedback around the entire amplifier to "linearize" the crossover region by forcing the output to follow the input more faithfully.

A qualitative sketch of the transfer curve illustrates how, without proper biasing, there is a segment near $v_{\mathrm{in}} = 0$ where neither transistor conducts, causing a kink in the output. With adequate bias and feedback, this kink is effectively "filled in," reducing audible distortion.

7.3 Conclusion

Our distortion analysis shows that the dominant harmonic distortion terms arise from device nonlinearities in the input and driver stages, quantified by THD via the formula above. Crossover distortion manifests in the push–pull output stage around the zero-crossing if biasing is insufficient. In our final amplifier layout, we have minimized both effects—employing local feedback loops to reduce harmonics and a VBE-multiplier bias network to suppress crossover distortion—thus achieving a high-fidelity output with THD typically below 0.1% under nominal operating conditions.