EE 344: Final Design
Wideband Audio Acquisition Using an Electret Microphone

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1 Objective

Conventional audio acquisition circuits suffer from one key limitation: they are unable to record low frequency audio input due to the DC blocking capacitor placed at the microphone output in these circuits. Our goal is to make a wideband audio capture circuit which is able to reliably record audio signals even in the low frequency region. We want to then transmit the recorded audio samples to an external device for further processing and storage.

2 Design Summary

We opt for a control based solution which eliminates the need for a DC blocking capacitor. Instead, a feedback circuit injects the necessary current to eliminate the DC bias of the microphone output. This DC compensated signal is passed through an op-amp gain stage. Automatic Gain Control is implemented by making the gain of this stage variable using a Digital Potentiometer. The amplified output is then recorded by a microcontroller which applies both DC Bias Compensation and Gain Control signals and also transmits the recorded audio samples using an external bluetooth module.

3 Component Selection

3.1 Electret Microphone

The CMA-4544PF-W electret microphone is a suitable choice for this experiment for several reasons.

- It has a small form factor of 9.7 x 4.5 mm^2, which is important for compact circuit designs.
- It has a high signal-to-noise ratio, which is crucial for capturing clean audio signals.
- It has a wide frequency response range that covers frequencies from 50Hz to 20KHz, making it suitable for capturing a broad range of sounds.
- It has low power consumption, which is advantageous for battery-powered applications.
- Finally, it is affordable, making it a cost-effective option for experimental setups.

The CMA-4544PF-W offers a good balance of performance and practicality, making it an excellent choice for this experiment.

3.2 Microcontroller

The TIVA C TM4C123GH6PM microcontroller has the following properties which make it a suitable choice for this experiment.

- It has a 32-bit ARM Cortex-M4F core, which provides high-performance computing and DSP capabilities, making it well-suited for audio signal processing.
- It has a high clock speed of up to 80 MHz, which allows for fast data processing and high sampling rates.
- It has a built-in ADC, which are essential for capturing analog signals.
- It has 4 SPI communication interfaces, which allows for easy data transmission to other devices using Bluetooth.
- It has a low-power mode, which is important for energy-efficient applications, especially in battery-powered systems.
- It has a robust software development kit (SDK), which makes it easy to develop and test applications using a range of programming languages and development tools.

- Finally, it has a rich set of peripherals and I/O ports, which makes it highly configurable and customizable for various experimental setups.

Overall, the TIVA C TM4C123GH6PM offers a good balance of performance, power efficiency, and flexibility, making it an excellent choice for this experiment.

3.3 Bluetooth Microcontroller Interface

The following reasons make LAUNCHXL CC2640R2F a suitable choice for this experiment.

- It is a low-power, high-performance microcontroller, which is essential for energy-efficient applications, particularly in battery-powered systems.
- Second, it has a built-in Bluetooth 5.1 Low Energy (BLE) module, which allows for wireless data transmission to other devices, making it easy to integrate into a wider range of systems.
- It has a variety of interfaces, including 1 UART, 2 SPI, 1 I2C, and 1 I2S which make it easy to communicate with other devices or sensors.
- It has 28 KB of RAM, which is sufficient for data processing and storage for the proposed experiment along with a 12 bit ADC.
- It has a compact form factor, which is important for space-constrained applications.
- Finally, it has a low cost, making it an affordable option for experimental setups.

In summary, the LAUNCHXL CC2640R2F is a cost-effective and versatile option that strikes a good balance between performance, power efficiency, and flexibility, making it a highly suitable choice for our experiment.

3.4 Digital Potentiometer

The following reasons make MCP41100-E/SN a suitable choice for this experiment.

- It provides a high-resolution 256-tap adjustment for fine-grained gain control in the audio acquisition circuit.
- It has an SPI interface, which enables easy digital communication with the microcontroller, simplifying the integration of the digital potentiometer into the circuit.
- It has a low-power consumption, making it an energy-efficient option for battery-powered applications.
- The evaluation module provides a convenient way to test and evaluate the performance of the MCP41100 digital potentiometer, which helps streamline the design process.

Overall, the MCP41100 digital potentiometer with SPI interface evaluation module offers a high level of functionality, accuracy, and convenience.

3.5 Digital to Analog Converter (DAC)

The MCP4921 DAC with SPI interface is a suitable choice for this experiment for several reasons.

- It has a high resolution of 12 bits, which ensures accurate digital-to-analog conversion.
- It has a fast settling time, which is important for applications where rapid changes in the output voltage are required.
- It has a compact form factor, which is important for space-constrained applications
- It has a wide input voltage range, which makes it suitable for a variety of input signal levels.
- It has an SPI interface, which enables easy digital communication with the microcontroller, simplifying its integration into the circuit.

The MCP4921 DAC with SPI interface offers reliable and efficient operation, making it an excellent choice.

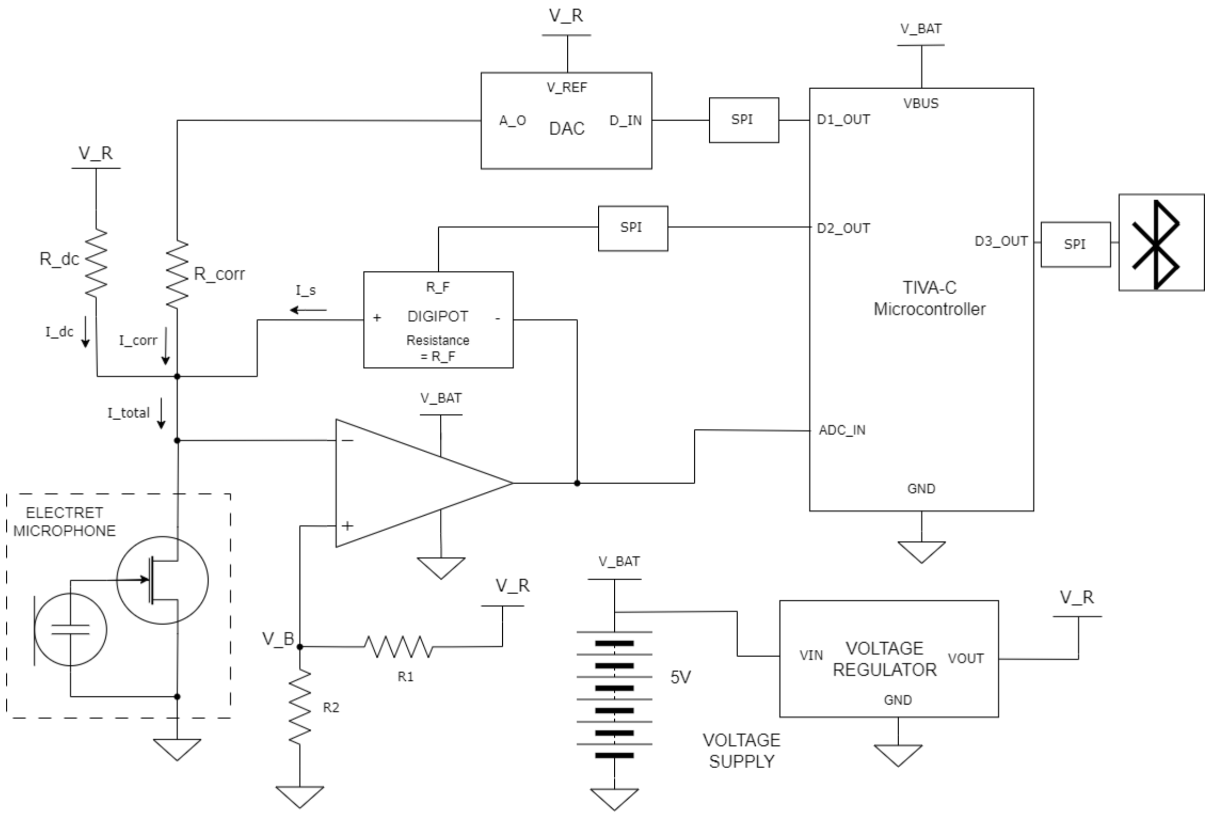
3.6 Operational Amplifier

The selection of TI TLV2472 operational amplifiers for this project is based on several factors.

- The Op-amp provides reliable and accurate signal acquisition, which is essential for high-quality audio recording as the amplifier supports rail-to-rail operation, which enables precise acquisition of signals across the full input range.
- This feature is particularly important in low-voltage applications, where the voltage range can be limited.
- Furthermore, the Op-amp has low input noise and low offset voltage, which ensures that the acquired signal is free from unwanted noise and distortion.
- The amplifiers also has a high gain bandwidth product, which allows for fast signal processing and ensures that the acquired signal is faithfully represented in the output.

Overall, the TI TLV2472 operational amplifiers provide a robust and reliable solution for signal acquisition in this project, which is crucial for obtaining high-quality audio recordings.

4 Principle of Operation



4.1 Bias Current Compensation

The principle of operation of the experiment involves the conversion of incoming sound signals into voltage signals using an electret microphone. This voltage signal is then fed into the gate of the FET, and a negative feedback opamp configuration sets the voltage at the drain to V_B. A

current $I = I_{bias} + i_{in}$ flows into the drain of the FET, where i_{in} is the small signal component carrying the sound signal information, and I_{bias} needs to be compensated.
To compensate for the bias current, compensating currents from the sources I_{DC} and $I_{correction}$ are brought in. The current I_C cancels most of the bias current and is fixed, while $I_{correction}$ is a variable output of the microcontroller, which handles small variations in the bias current. The TIVA C muC and the MCP4921 DAC with an SPI interface are used to efficiently convert the digital input to analog output for DC bias compensation.
The microcontroller uses a digitized version of the opamp output as input to control the gain R_F and decide $V_{control}$ in a feedback loop. The digitized signal is also transmitted to an external device through a Bluetooth module, where further operations may be performed on the signal. The TPL0102 256-tap dual-channel digital potentiometer with an I2C interface evaluation module is used to set the voltage $V_{control}$, which controls the gain of the opamp. The LAUNCHXL CC2640R2F Bluetooth microcontroller interface is used to transmit the digitized signal to external devices.
Finally, the TI TLV2472 operational amplifiers are used in the circuit to support rail-to-rail operation, ensuring precise acquisition of signals. Overall, the principle of operation of the experiment involves integrating various components and techniques to ensure the accurate acquisition and transmission of sound signals while compensating for DC bias.

4.1.1 Compensation Algorithm

In our design we utilise a successive approximation algorithm for eliminating bias. The signal average over a window is roughly constant and equals the DC offset. We thus, compute the average over a sliding window and after certain number of samples are collected we set one bit of the DAC so as to reduce the offset. We set bits from the MSB towards the LSB to progressively eliminate the error. The compensation algorithm runs for roughly 2 seconds on device startup after which it enters the desired operating mode with gain compensation.

4.1.2 Limitations and Alternatives

As discussed with Professor Pandey, the successive approximation approach is not robust to noise. If the input signal causes the output to drift significantly, the windowed average may be computed incorrectly and thus the algorithm would end up setting the DAC compensation bits wrongly.

A more robust alternative would be to do a continuous search from one extreme to another and observe the output error. In case of noise fluctuations leading to incorrect compensation, one can simply run the search algorithm in the reverse direction over a smaller window until the error is fully eliminated.

4.2 Automatic Gain Control

Once DC bias compensation is achieved, the device enters the gain compensation stage. We have a constant resistance in series with a digital potentiometer to achieve the desired gain range. Similar to the windowed averaging employed for bias current compensation, here we record a window of samples and compute the maximum and minimum values in the window. We compute a difference of these values to estimate the peak-to-peak variation of the signal. The signal gain is then adjusted incrementally by the digipot until a suitable peak-to-peak variation is achieved.

If the signal variation lies below a chosen threshold we treat it as noise and do not amplify it. This is done to prevent sudden amplification of background noise when the speaker is silent. Moreover, when the signal peak to peak variation is above a chosen threshold we decrease it in steps until the threshold is reached.

4.2.1 Limitations and Alternatives

The gain compensation algorithm we designed adjusts the gain in steps. However, this is not ideal since the target gain takes time to be reached. A more efficient alternative would be to calculate the exact gain needed to transform the current peak-to-peak variation to the desired target value and compute the digipot resistance accordingly.

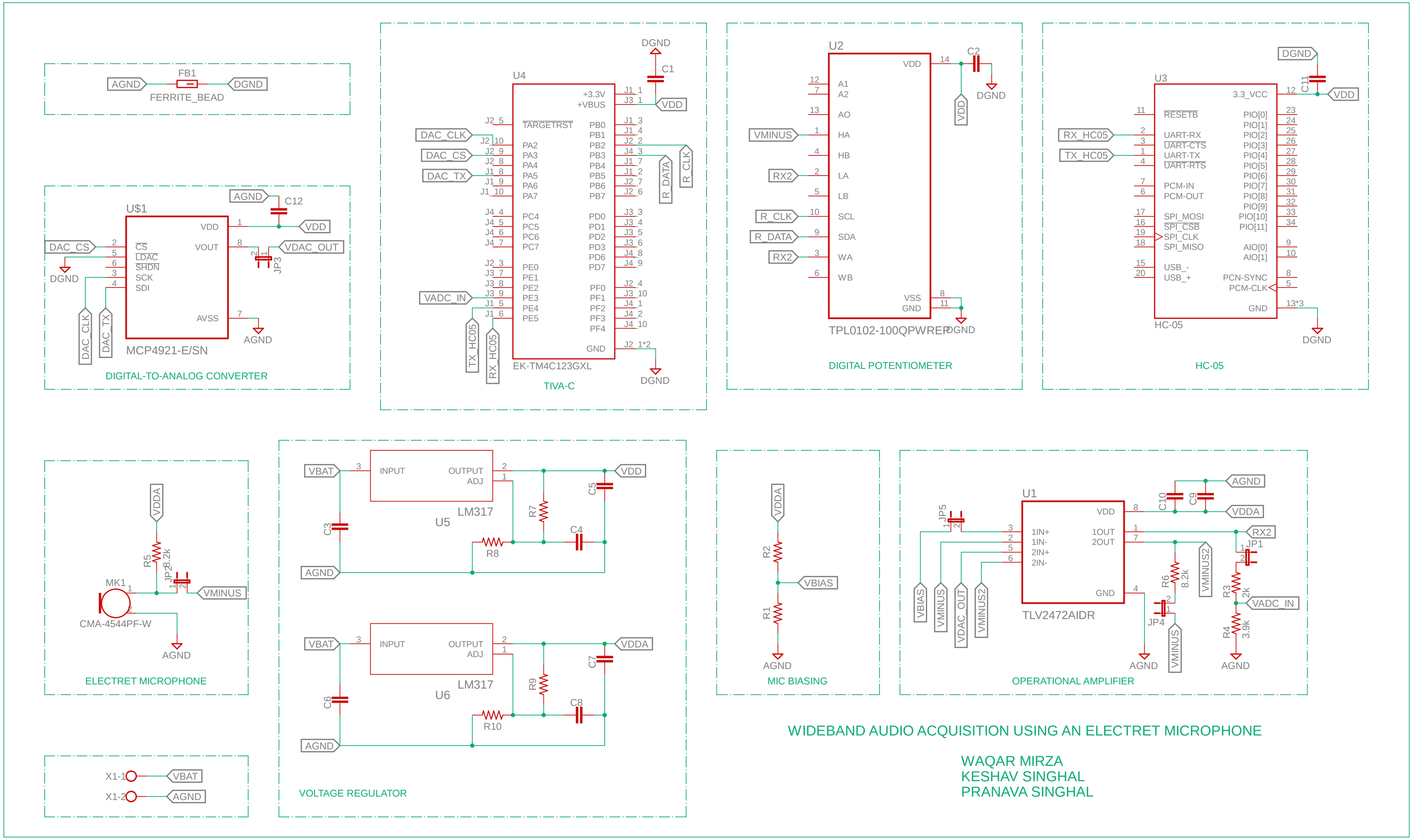
Another limitation of our circuit for gain compensation is its single stage nature. We can only vary the feedback resistance in a range of $20k\Omega$ to $40k\Omega$ giving us only upto 2x gain variation. A better circuit design would have two amplifier stages as follows:

- The first stage would have a constant gain and also a low pass filter to eliminate high frequency noise.
- The second stage would have only the variable resistance and thus give us a much larger gain variation ability.

In addition to eliminating noise and larger gain variation, the noise elimination also improves the reliability of computing the peak-to-peak signal variation and thus improves the robustness of the exact gain transformation approach for gain control.

5 CAD Files

5.1 Schematic



WIDEBAND AUDIO ACQUISITION USING AN ELECTRET MICROPHONE

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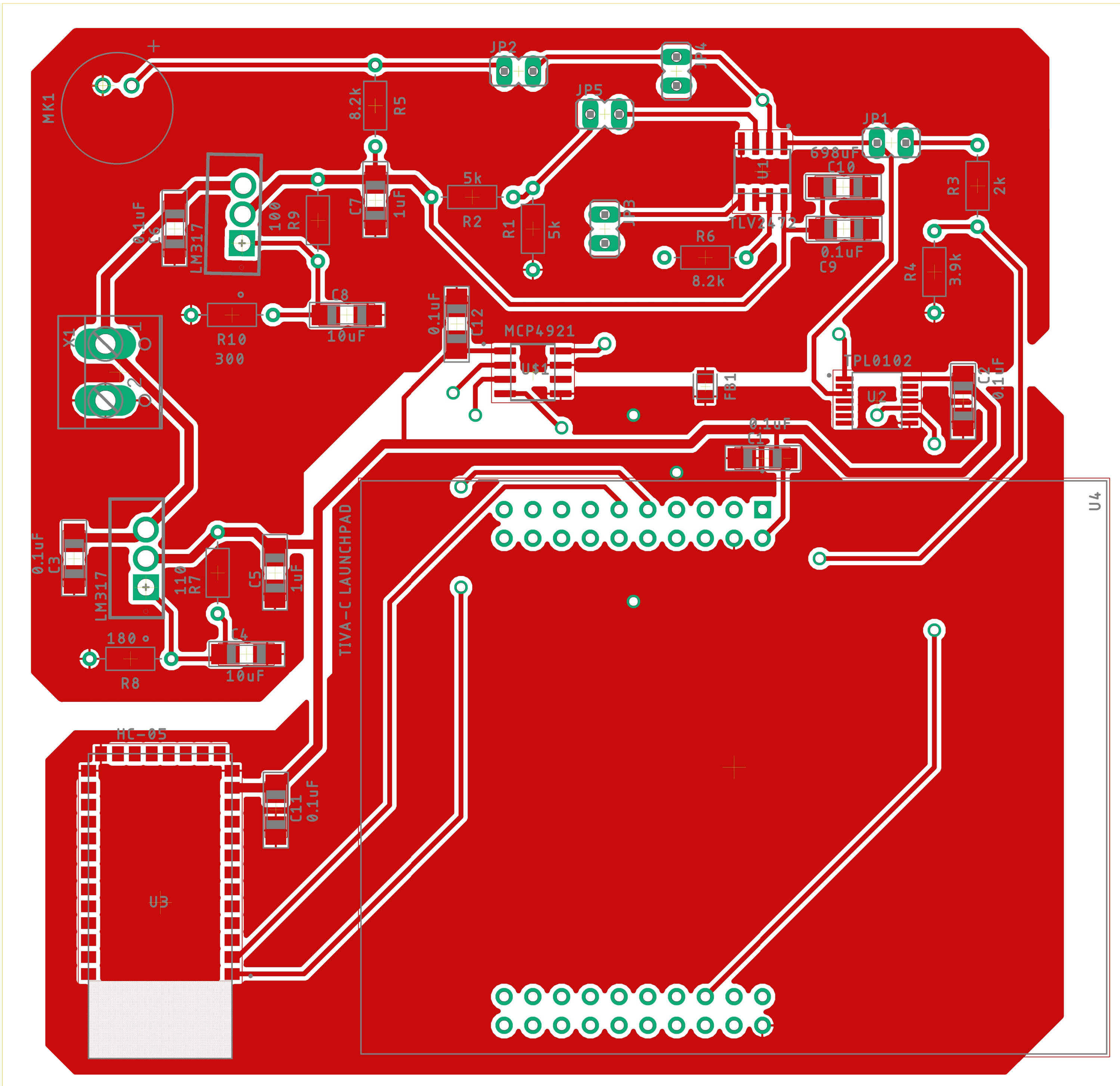
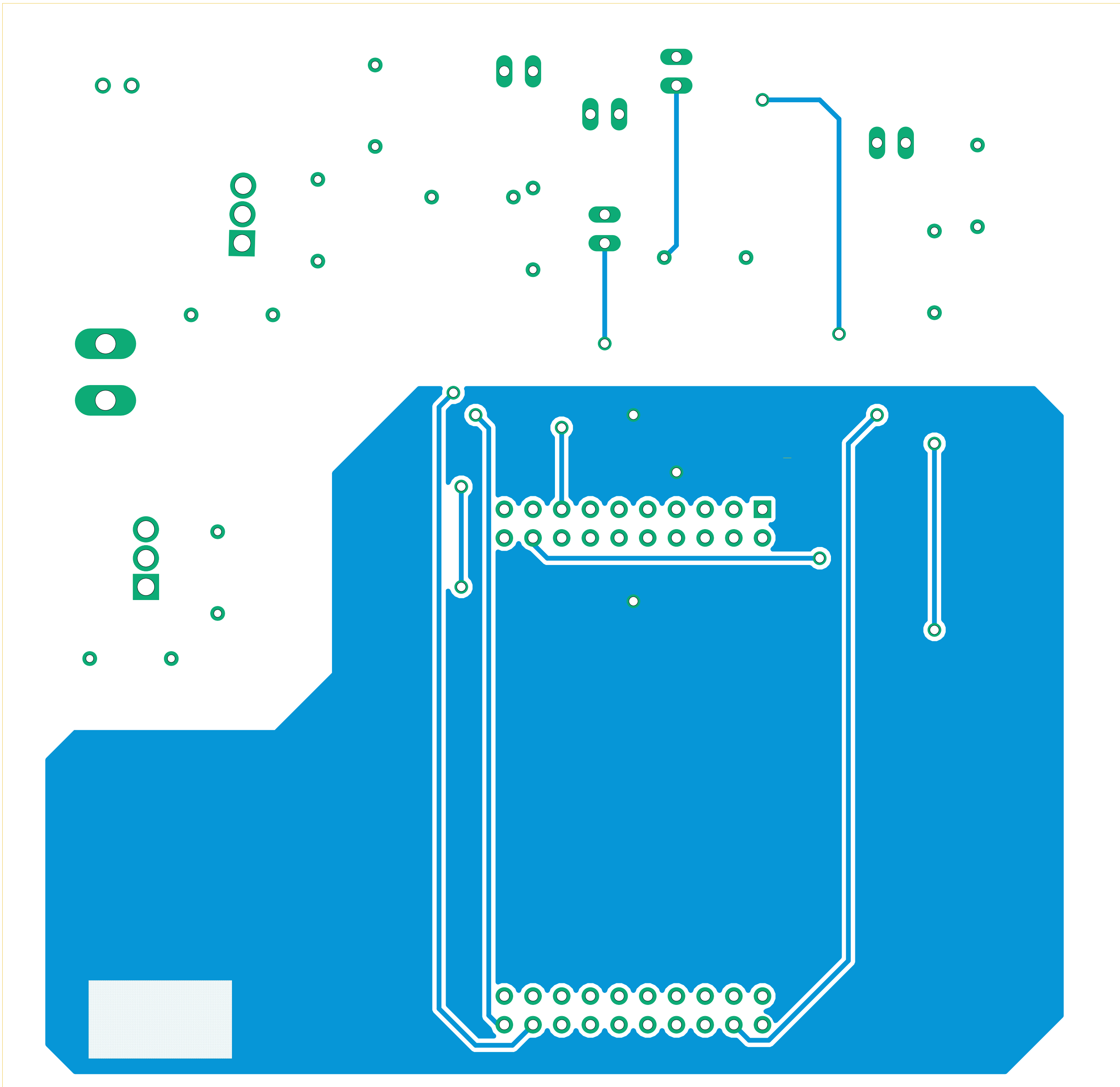
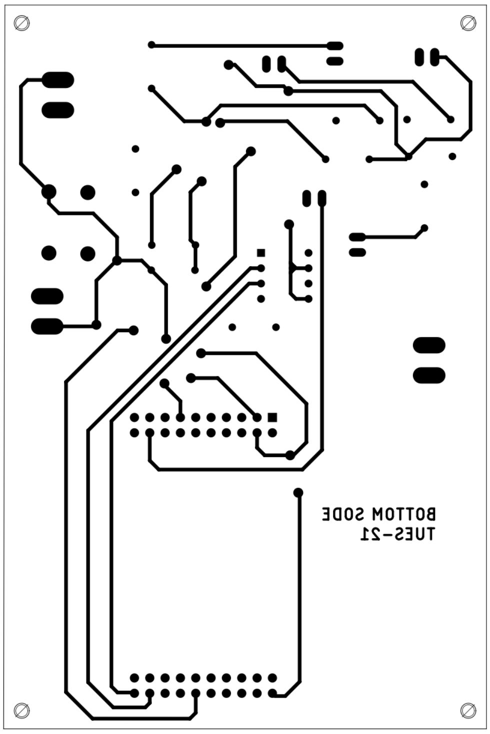
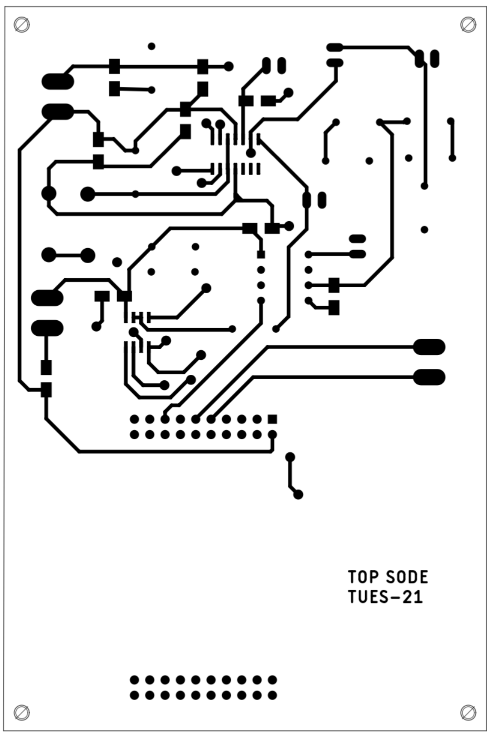


MK1
JP2
JP4
JP5
JP1
8.2k
R5
698uF
C10
R3
2k
5k
R2
R1
5k
C7
1uF
LM317
100
R9
0.1uF
C6
JP3
TLV2472
U1
R6
8.2k
0.1uF
C9
R4
3.9k
C8
10uF
R10
300
X1
0.1uF
C12
MCP4921
U$1
TPL0102
U2
C2
0.1uF
FB1
0.1uF
C1
U4
TIVA-C LAUNCHPAD
0.1uF
C3
LM317
110
R7
C5
1uF
180
R8
C4
10uF
HC-05
C11
0.1uF
U3



5.2 PCB Layout

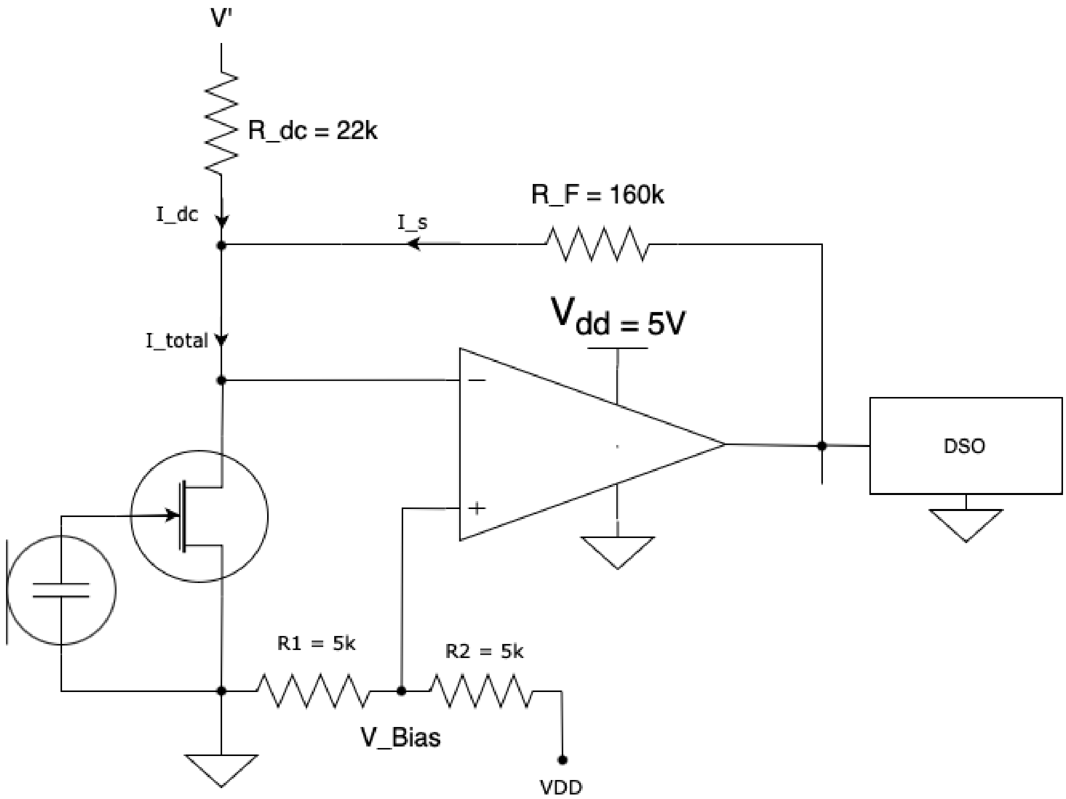




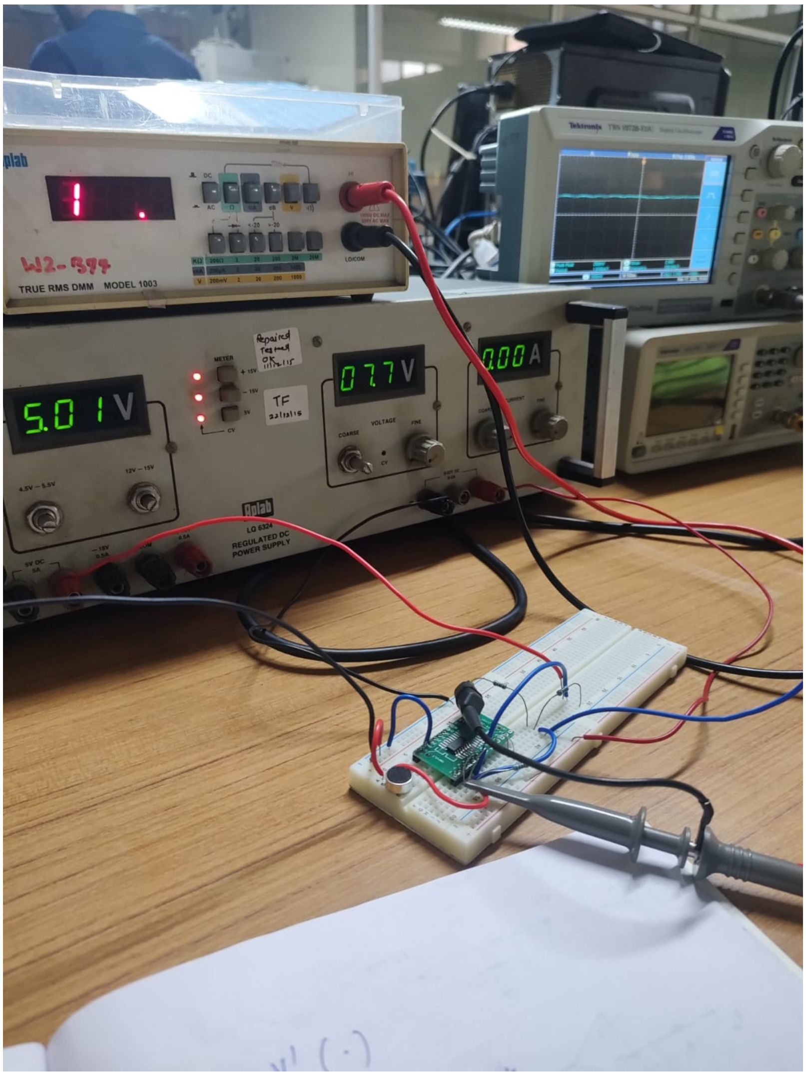
6 Experiments

6.1 Microphone Amplifier Interfacing

6.1.1 Description of Test Setup and Method



We checked the interfacing of the microphone with amplifier. The key objectives of this experiment were to identify a suitable biasing voltage for the microphone, DC compensation resistance and the appropriate feedback resistance for the op-amp to get the desired output voltage swing. In order to perform this test we used the following circuit, where the microphone biasing voltage V_{bias} and the DC-compensation voltage $V^{'}$ were set using a variable voltage supply. Audio signal used is a 1KHz sinusoidal tone played using a mobile phone speaker at fixed volume at a particular distance from the microphone. The experiments were performed with low ambient noise (like people speaking) in the background.
The first part was aimed at identifying a suitable biasing point. The value of feedback resistance R_F was fixed initially. Even the resistor on the DC compensation branch, R_{dc} is constant. Thus, we tried different values of V_{bias} using the variable voltage source and for each value of V_{bias} we adjusted $V^{'}$ so that the DC current is close to zero on the feedback resistor. The output voltage fed to the oscilloscope (DSO in figure) is given by $V_{out} = V_{minus} + I_s \cdot R_F$. Thus, if the current I_s only carries the AC component of the signal, the output is centred around $V_{minus} = V_{bias}$. We use this knowledge to vary $V^{'}$ while observing the signal output on the oscilloscope until the **mean** of V_{out} is close to V_{bias} on the oscilloscope. Knowing $V^{'}$ at the biasing point and R_{dc} we also get to know the DC compensation current for this particular microphone I_{dc}.
For each value of V_{bias} we also note down the output peak-to-peak voltage swing while keeping the input audio signal identical at a fixed distance. We obtain similar waveforms and output swing for a large range of biasing values any of which can be used for operation. We choose a convenient middle value of $V_{bias} = 2.5V$. This simplifies design in a few key ways. First, the output signal is centred around $2.5V$ which allows us to make best use of the op-amp output swing from 0 to $5V$. Moreover, in the final circuit, the control feedback from the DAC will have voltage



outputs between 0 to $5V$ and using a biasing of $2.5V$ allows us to use equal range of positive and negative DC-compensating control currents, maximising the useful range of the DAC feedback. We generate this $V_{bias} = 2.5V$ using a resistive divider with $R_1 = R_2 = 5k\Omega$.

The second part of the experiment aims to identify a suitable range for the feedback resistance R_F in order to provide the necessary output voltage swing. In the final circuit R_F will be replaced by a digipot (digital potentiometer) which will allow us to vary the amplification factor based on the input loudness so that the op-amp output does not enter saturation. We fix $V_{bias} = 2.5V$ for the remaining experiments and try two different values of R_F which are $260k\Omega$ and $380k\Omega$ respectively. For a fixed value of R_F we vary the distance of our audio source from the microphone and observe the peak-to-peak swing of V_{out} on the oscilloscope. We also remove the audio source and observe the peak-to-peak swing due to ambient noise. This information will later help us design the gain-control algorithm, where we need to set a noise threshold before the gain control sets in. We take three different distances: placed on the microphone surface (nearest), roughly a centimetre from the microphone (near), and a comfortable speaking distance for a usual microphone (speaking distance). With our output centred around $2.5V$, the amplifier allows for a swing of 0 to $5V$. However, we want to keep some headroom for the signal in our design so that clipping doesn't happen. Thus, we would ideally like to have a peak to peak variation of $4V$ keeping $0.5V$ of headroom for the signal on both sides.

6.1.2 Test Results

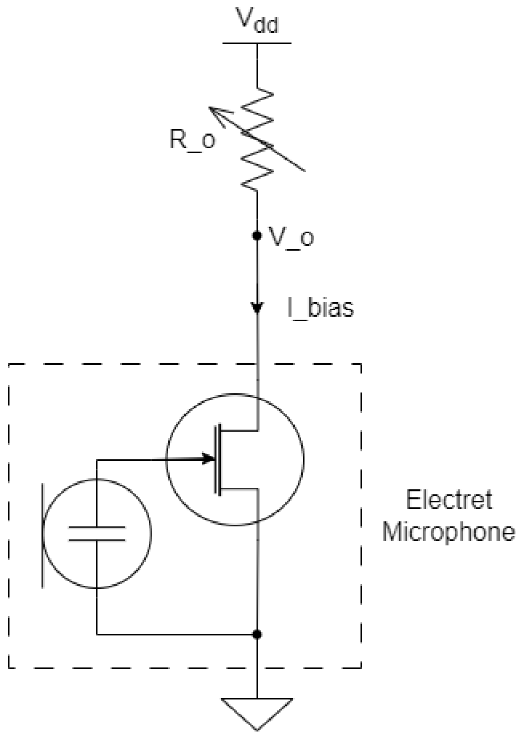
First we look at the approximate peak to peak output signal for different V_{bias} values and also calculate the bias current that needs to be compensated for that V_{bias}. Upon compensation, the signal mean should be close to V_{bias}. We can see that the amplification (peak to peak value) and bias current do not vary significantly for different V_{bias} values. So we pick V_{bias} = 2.5V for our design as mentioned earlier.

V_{pk-pk}	V_{bias}	$V'_{compensation}$	$I_{bias} = (V' - V_{bias})/R_{dc}$	Signal Mean
800mV	1.5V	6.5V	0.227mA	1.53V
800mV	2.0V	7.2V	0.236mA	1.95V
800mV	2.48V	7. 7V	0.237mA	2.50V
800mV	2.56V	7.8V	0.238mA	2.60V
800mV	3.0V	8.3V	0.241mA	3.0V

Next we have readings for the variation of the output signal peak to peak value with distance. As expected, we see drop in amplitude as distance increases. We see the amplification of signal is proportional to R_F as expected. Thus the amplifier microphone interfacing worked as expected.

Distance	V_{pk-pk}
$R_F = 260K$	
Nearest	1.8V
Near	1.5V
Speaking Distance	1.0V
Ambient Noise	0.65V
$R_F = 380K$	
Nearest	2.8V
Near	2.2V
Speaking Distance	1.3V
Ambient Noise	0.90V

6.2 Process Variation in Bias Current



6.2.1 Description of Test Setup and Method

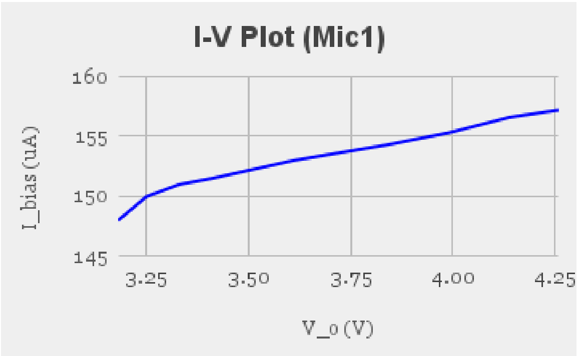
It is important that our circuit should be able to compensate the bias current for any electret microphone that the user decides to use, that is, we should be able to handle the process variation in the bias current. For this reason, we experimented with multiple electret microphones to get an idea of this process variation. The experimental setup is shown in the below circuit diagram.

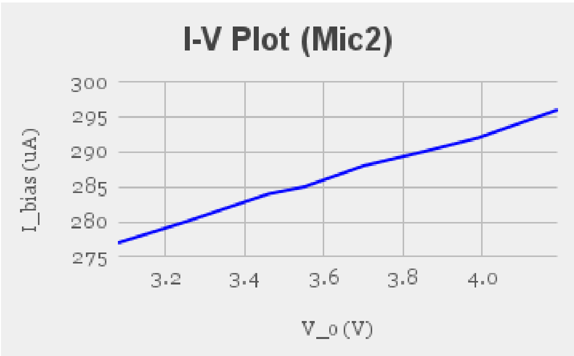
We used three different microphones for the experiment. For every microphone, the supply V_{dd} was fixed, and potentiometer value R_o was varied. The drain voltage V_o and the bias current I_{bias} were measured and recorded. Since the internal gate voltage of the electret is fixed, this gives us the I_D vs V_{DS} characteristic of the FET at fixed V_{GS}.

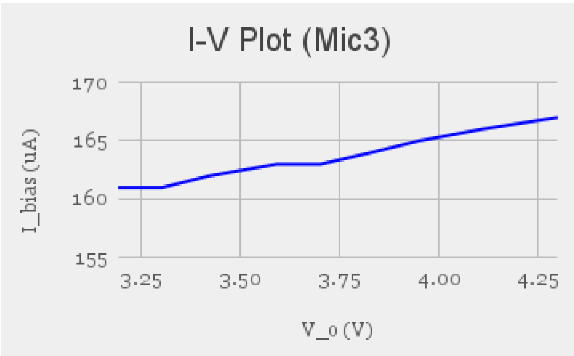
6.2.2 Test Results

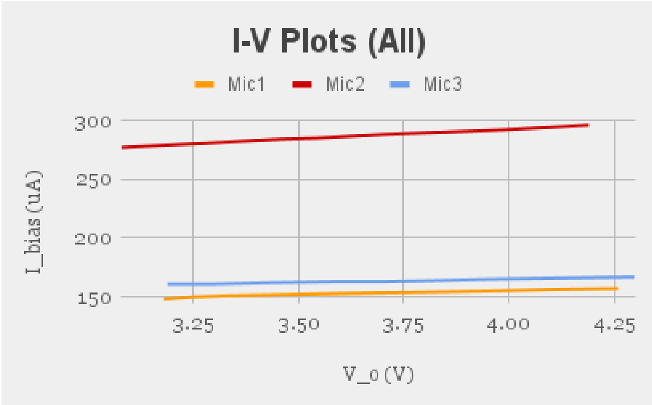
The readings and plots for the 3 microphones are as shown:

MIC 1		**MIC 2**		**MIC 3**	
V_o (V)	**I_bias (uA)**	**V_o (V)**	**I_bias (uA)**	**V_o (V)**	**I_bias (uA)**
3.18	148	4.19	296	4.3	167
3.25	150	3.99	292	4.11	166
3.33	151	3.85	290	3.95	165
3.41	151.5	3.7	288	3.83	164
3.61	153	3.55	285	3.7	163
3.84	154.3	3.46	284	3.59	163
3.99	155.3	3.25	280	3.42	162
4.14	156.6	3.08	277	3.3	161
4.26	157.2	-	-	3.19	161









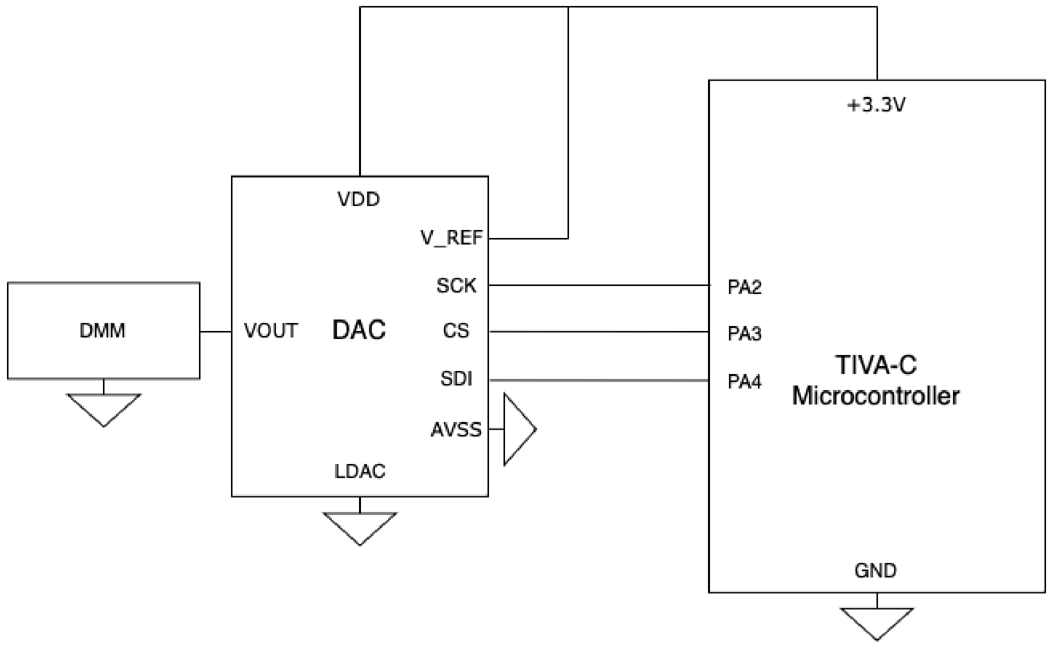
From these readings, we see a significant process variation in the bias current, and we can use this data to select a suitable range of bias currents that our circuit will be designed to compensate.

6.3 Key Calculations for DC Current Compensation

The above experiments allow us to choose suitable component values for the DC compensation part of the circuit. Referring to the circuit schematic, we can see two DC current compensation branches: one is a constant current $I_{dc} = \frac{V_{DD}-V_{minus}}{R_{dc}} = \frac{V_{DD}-V_{bias}}{R_{dc}}$, the second one is the DAC output passed through the voltage buffer which gives us an output current of $I_{corr} = \frac{V_{DAC}-V_{minus}}{R_{corr}} = \frac{V_{DAC}-V_{bias}}{R_{corr}}$. The total bias current is $I_{dc} + I_{corr}$ in which only I_{corr} is variable. By observing the biasing currents for various microphones we feel that a circuit capable of handling a process variation of 0 to $6mA$ would be sufficient. Thus, we set $I_{dc} \approx 0.3mA$, the mean value and let I_{corr} handle the swing of $\pm 0.3mA$. We set $R_{dc} = \frac{5V}{0.3mA} \approx 8.2k\Omega$ (closest standard resistor). Since $V_{DAC} \in (0,5)V$ and $V_{bias} = 2.5V$ we have a swing of $\Delta_V = 5V$ ($-2.5V$ to $2.5V$). Our desired current variation handling ability is $\Delta_i = 0.6mA$ Thus, we set $R_{corr} = \frac{\Delta_V}{\Delta_i} = \frac{5V}{0.6mA} \approx 8.2k\Omega$ (the nearest standard resistor). Since our DAC is 12-bit, we have a minimum voltage resolution of $\delta_V = \frac{5V}{2^{12}} = 1.22mV$. For the chosen R_{corr} this gives a current resolution of $\delta_i = \frac{\delta_V}{R_{corr}} = \frac{1.22mV}{8.2k\Omega} = 0.148\mu A$ which is more than sufficient accuracy for our application. We say this because a typical value of feedback resistance $R_F = 380k\Omega$ will only amplify this error to $\delta_i \cdot R_F = 56.53mV$. This is the maximum DC offset we can expect at the output if our controller is giving the ideal feedback signal (this is only a first approximation). Thus we have obtained $R_{dc} = R_{corr} = 8.2k\Omega$.

6.4 DAC Testing

6.4.1 Description of Test Setup and Method



We tested the DAC MCP4921 to check if it operates as required. We interfaced it using SPI with the μC as indicated in the figure. We sent digital input to the DAC from the μC as a 12 bit value and measured the analog voltage output of the DAC using a DMM. We set the reference voltage of the DAC to $V_{ref} = 3.3$V. We expect the output to be $V_{out} = V_{ref} \frac{Input}{2^{12}}$.

6.4.2 Test Results

We note the analog output of the DAC for different digital inputs and compare with the expected output. As we can see, the DAC is functioning almost as expected.

Digital Input	Expected Analog Value	Experimental Analog Value
50	0.040V	0.038V
100	0.080V	0.78V
150	0.120V	0.120V
250	0.201V	0.199V
400	0.321V	0.320V
700	0.562V	0.570V
1000	0.806V	0.812V
1150	0.924V	0.920V
1650	1.329V	1.322V
2048	1.645V	1.630V
2300	1.848V	1.854V
2500	2.014V	2.020V
2900	2.330	2.320
3400	2.740V	2.738V
3800	3.061V	3.058V
4000	3.214V	3.211V
4095	3.290V	3.290V

Input Frequency	Output Frequency
100 Hz	100 Hz
200 Hz	200.5 Hz
500 Hz	499.6 Hz
1 kHz	1 kHz
1.5 kHz	1499 kHz
1.9 kHz	1.9 kHz
2 kHz	$\approx$ 0 Hz
2.1 kHz	1.9 kHz
2.5 kHz and above	distorted

Table 1: ADC Testing

6.5 ADC Testing

6.5.1 Description of Test Setup and Method

We set the Analog-to-Digital Converter (ADC) to a fixed sampling rate of 4kHz. We send a fixed frequency sinusoidal voltage input to the ADC using an Arbitrary Function Generator (AFG). The frequency of the AFG can be configured as desired. The ADC is configured to record each sample and subsequently load that sample onto the DAC via SPI interface, before recording the next sample. The DAC output is recorded with an oscilloscope and the output signal frequency is measured using the DSO measurement function. We compare the input signal frequency and measured output signal frequency for various values of input frequency. Our aim is to increase the frequency until Nyquist rate and then observe aliasing at higher frequencies. This will prove that the ADC is working as expected.

6.5.2 Test Results

We set the AFG frequency at the following values and noted corresponding output frequencies. We can observe that at $f = \frac{f_s}{2} = 2kHz$ we observe aliasing at slightly higher frequencies and then distortion of the signal. This shows that the ADC works as expected up to Nyquist rate.

7 Demo Video

https://drive.google.com/file/d/1v451EZN24-vraX$_jAyer-HpxaoUAOmEQ/view?usp=sharing$

8 Observations

Our final experimental setup to test all the features in closed loop was as follows:

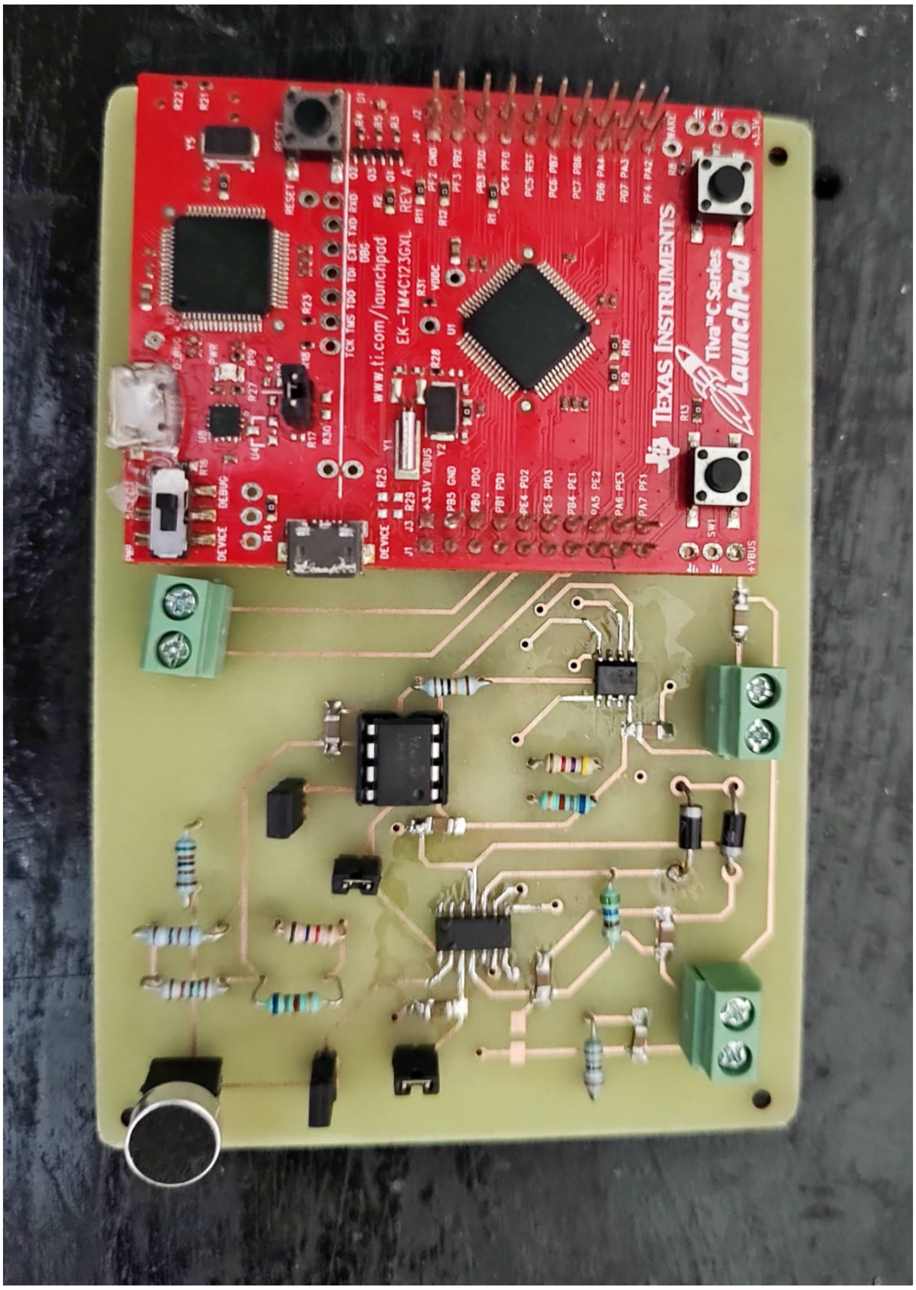
1. First, we tap the op-amp output from the jumper placed in the PCB and observe it at the oscilloscope
2. When the device is first powered on it enters a DC compensation stage. Here we can clearly observe a binary search (Successive approximation algorithm) running to find the optimal compensation value so that the output is centred around the midpoint (1.65V) of the desired op-amp range (0 to 3.3V)
3. After the compensation algorithm is run, the device enters a gain compensation stage. We connect an Arbitrary Function Generator (AGC) to the input of a speaker circuit to create sound with known characteristics. We first apply a sinusoidal input of 200 Hz and there is ambient background noise in addition to it.
4. The audio signal is also passed through an alternative constant gain branch and the output is observed on the oscilloscope for comparison with the gain controlled output.

5. We vary the distance of the sound source from the microphone gradually and observe that gain compensation becomes active once the input loudness exceeds a threshold. Initially the algorithm tries to amplify the feeble input to reach a desired threshold. However, as we bring the sound source even closer the gain compensation then starts to act in the opposite direction and reduce the output amplitude so that it approaches the target threshold. Throughout this process, the uncompensated signal in the other channel of the oscilloscope gives us an indication of the strength and the direction of the gain compensation.

6. We also remove the controlled audio source and instead speak directly into the microphone. For a comfortable speaking distance we can observe gain compensation acting exactly as desired.

9 Results

We have demonstrated a circuit that can successfully record audio input and apply DC-Offset Compensation and Gain Compensation to produce a desirable signal. We are also able to see this desired signal at the ADC input (op-amp output) in our experiment. We have realised the entire design on a compact PCB with mechanisms to make it robust to noise as well as suitable test points for debugging. We have placed components so that the power supply and microcontroller input can be accessed easily from outside the device packaging.



10 Conclusion and Future Work

We have described above several limitation of our design including the bias-compensation algorithm and the gain control algorithm and design and also suggested better alternatives for each. Moreover, our final recorded samples need to transmitted out of the device using a cyclic buffer or two buffer implementation since our device output is via UART over bluetooth and it is unable to handle sample by sample transmission. We were unable to implement this final piece of the project. Once we are able to transmit samples to another, computationally more powerful device, like a computer, we can process the samples further to store or modify our audio recording. On the software front, we could create tools to transform the recorded sample over a window into an audio file with appropriate encoding (such as a WAV file). This will enable easy playback, analysis and post-processing of the recorded audio and help us verify whether the device actually outperforms conventional microphone circuits in the low frequency regime and achieves its goal of 'wideband' acquisition

11 References

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