# LABORATORY REPORT - CHAPTER 3

v7.5

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| Total Grade | /100 |

**Remarks:** Record all your measurements and write all your answers in the boxes provided.

## Preliminary Work

### 1. Microphone Amplifier

1. Consider the TRC-11 microphone amplifier circuit shown in Fig. 1, making use of one of the two OPAMPs in LM358 integrated circuit. Since the OPAMP operates with a single supply voltage, the input DC voltages of the OPAMP should be shifted to a voltage somewhere between VCC=+12V and GND. For this purpose, we use the regulated voltage, +6V. Therefore, +6V acts like a ground, +12V acts like the positive supply, and GND acts like the negative supply of the OPAMP.

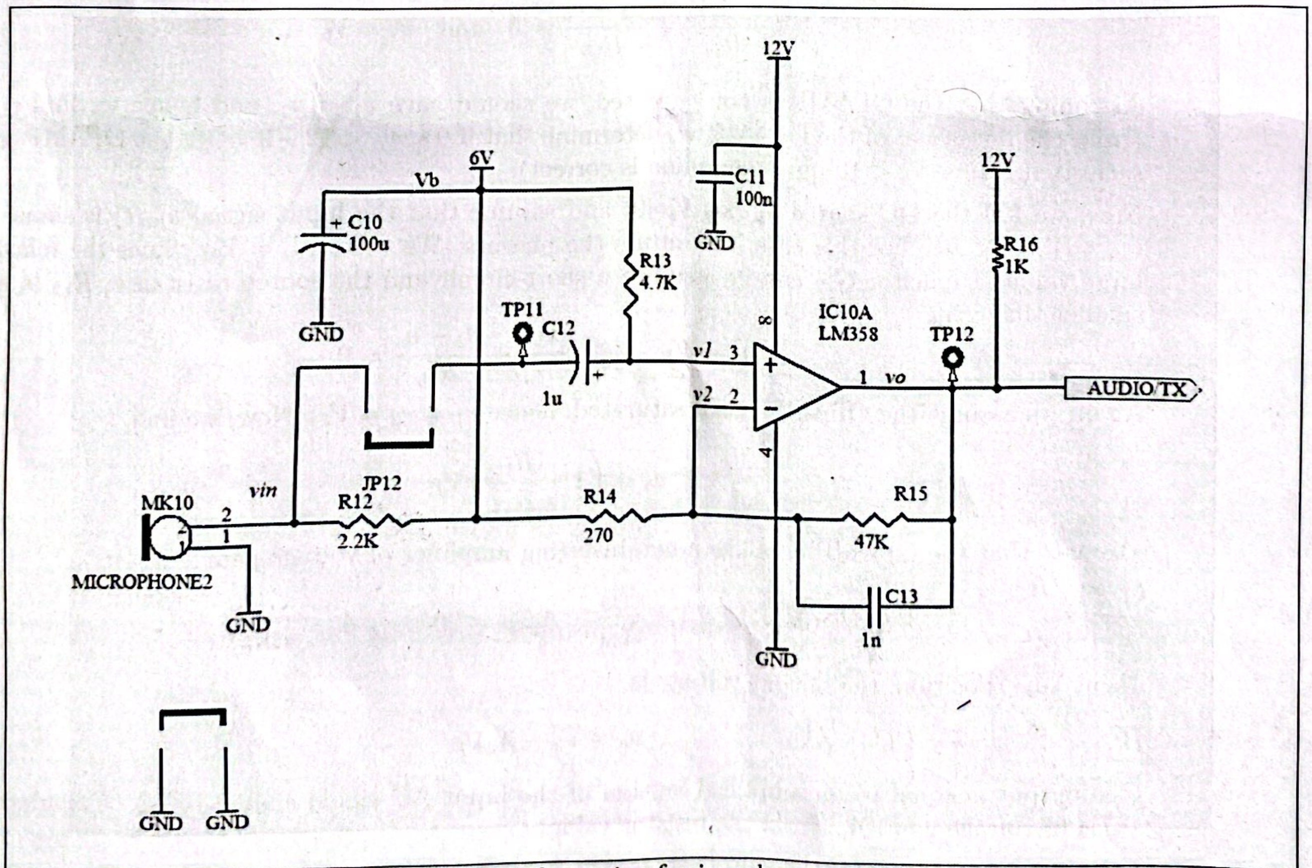


**Figure 1:** Schematic of microphone amplifier



Scanned with CamScanner

| Designator | Comment | Description |
|---|---|---|
| C10 | 100 $\mu$F | Electrolytic capacitor, polarized 16V |
| C11 | 100 nF | Capacitor, ceramic disc, 50V |
| C12 | 1 $\mu$F | Electrolytic capacitor, polarized 16V |
| C13 | 1 nF | Capacitor, ceramic disc, 50V |
| IC10 | LM358 | Dual OPAMP |
| MK10 | Microphone | Microphone capsule |
| R12 | 2.2 K | Resistor, carbon film, axial leaded, 1/4W |
| R13 | 4.7 K | Resistor, carbon film, axial leaded, 1/4W |
| R14 | 270 | Resistor, carbon film, axial leaded, 1/4W |
| R15 | 47 K | Resistor, carbon film, axial leaded, 1/4W |
| R16 | 1 K | Resistor, carbon film, axial leaded, 1/4W |

**Figure 2:** Bill of materials for the microphone amplifier

2. We can find the output voltage, $v_o$, of this OPAMP circuit using the superposition principle for two sources: A DC source of $V_b$ and an AC source of $v_{in}$ (with a source resistance of $R_{12}$). First, let us kill the AC source $v_{in}$ and find the output voltage, $v_o$. Since the capacitor is open-circuit at DC, we write the node equations at $v_2$ and $v_1$ as

$$\frac{v_2 - V_b}{R_{14}} + \frac{v_2 - v_o}{R_{15}} = 0 \quad \text{and} \quad v_1 = V_b$$

Assuming that the OPAMP is not saturated, we should have $v_1 = v_2$, and hence we find $v_o$=$V_b$ (from the datasheet of the OPAMP, we determine that if $0 < v_o < 12 - 2 = 10$, the OPAMP is not saturated. Since $V_b < 10$ our assumption is correct).

Now, we kill the DC source $V_b$ (set $V_b$=0) and assume that the input signal $v_{in}(t)$ is sinusoidal: $v_{in} = V_P \cos(\omega t)$. For this case, we can use the phasors: We write $v_{in} = V_P$. Since the relatively large valued capacitor $C_{12}$ can be assumed a short-circuit and the source resistance, $R_{12}$ is much smaller than $R_{13}$

$$v_1 = V_P \quad \text{and} \quad \frac{v_2}{R_{14}} + \frac{v_2 - v_o}{R_{15}} = 0$$

Again we assume the OPAMP is not saturated, hence $v_1 = v_2 = V_P$. Now, we find

$$v_o = \left(1 + \frac{R_{15}}{R_{14}}\right) V_P$$

174,07

~~We note~~ that the OPAMP acts like a non-inverting amplifier of voltage gain

$A_v = 1 + \frac{47k\Omega}{270\Omega} = 175,07$ $\quad A_v = 1 + \frac{R_{15}}{R_{14}} \quad A_{vdB} = 20 \log_{10} A_v$

Using superposition, the output voltage is

$A_{vdB} = 20 \log_{10} A_v = 44,86$

$$v_o = V_b + A_v V_P$$

The output is equal to an amplified version of the input AC signal shifted by $V_b$. Calculate the value of voltage gain, $A_v$, from the resistor values.

Because of C12 and R13, the gain decreases at frequencies lower than the corner frequency of

$\frac{1}{2\pi.(4.7k)(1\mu F)} = 33,86$ $\quad f_1 = \frac{1}{2\pi R_{13} C_{12}} = 3$

Because of C13 and R15, the gain decreases at frequencies higher than the corner frequency of

$$f_2 = \frac{1}{2\pi R_{15} C_{13}} \quad \frac{1}{2\pi \cdot (47k)(1nF)} =$$

| |
|---|
| $A_{vdB}=$ 44,86 dB $f_1=$ 33,86 Hz $f_2=$ 3386,27 Hz |

1.2. GRADE:

The transfer function can be found using the nodal analysis with the inclusion of all capacitors. We find it as

$$H(\omega) = A_v \frac{j\omega R_{13} C_{12}}{1 + j\omega R_{13} C_{12}} \frac{1 + j\omega (R_{14} \parallel R_{15}) C_{13}}{1 + j\omega R_{15} C_{13}}$$

3. This transfer function in decibels is plotted in Fig. 3.

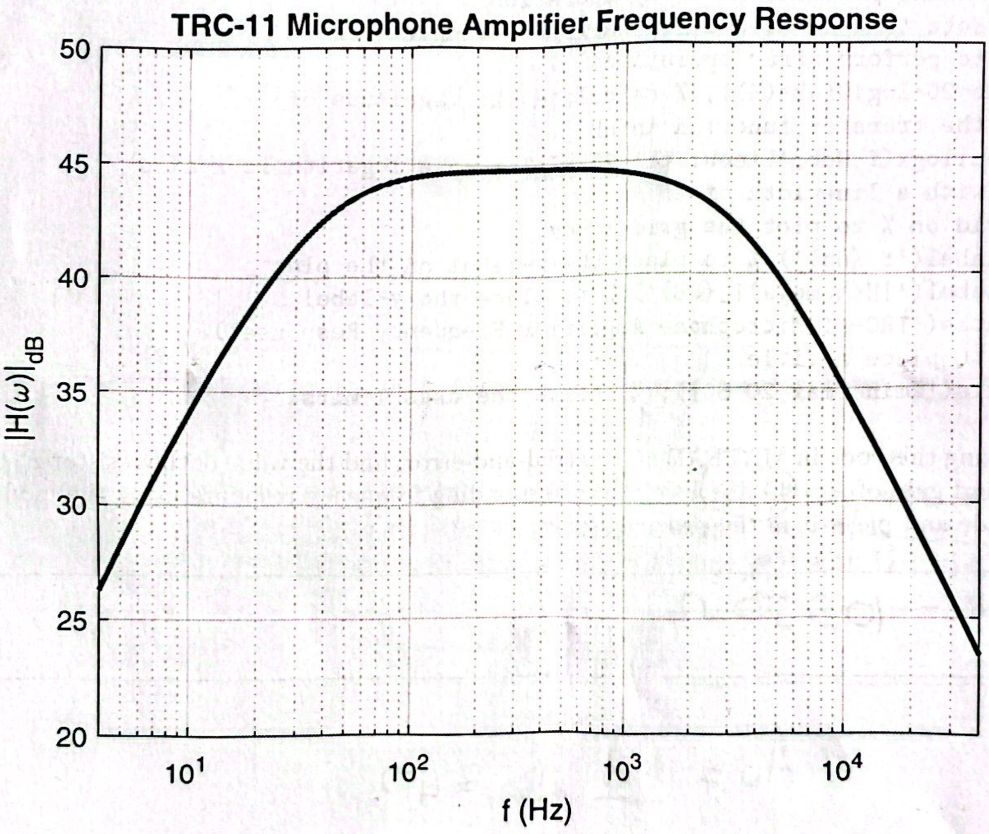


**Figure 3:** Calculated Frequency Response of the Microphone Amplifier

The MATLAB code to plot this function is

```
% MATLAB code to plot the transfer function
% of the Microphone amplifier
clear all % clear all variables in MATLAB
close all % clear all plot windows
fmin=4; %minimum frequency in Hz
fmax=40e3; % maximum frequency in Hz
C12=1e-6; % C12 capacitor value in F
C13=1e-9; % C13 capacitor value in F
R13=4.7e3; % R13 resistance in Ohms
R15=47e3; % R15 resistance in Ohms
R14=270; % R14 resistance in Ohms
Av=1+R15/R14;
f=fmin:fmin/5:fmax; % Frequency vector
w=2*pi*f; % angular frequency vector
% below, 1i represents the unit imaginary number j
H=Av*(1i*w*R13*C12)./(1+1i*w*R13*C12).*(1+1i*w*(R14*R15)...
/(R14+R15)*C13)./(1+1i*w*R15*C13);
% MATLAB performs an array operation
% Note that we need a "." in front of operators
% to perform array operations
Hdb=20*log10(abs(H)); % calculate the magnitude of
% the transfer function in dB
semilogx(f,Hdb,'LineWidth',2) % plot on a logarithmic x-axis
% with a linewidth of 2
grid on % to plot the grid lines
xlabel('f (Hz)') % to place the x-label on the plot
ylabel('|H(\omega)|_{dB}') % to place the y-label
title('TRC-11 Microphone Amplifier Frequency Response')
% to place a title
axis([fmin fmax 20 50]); % define the axes limits
```

Using this code in MATLAB and by trial-and-error, find the value of the resistor $R_{15}$ to have a mid-band gain of $A_v$=40 dB. Plot the corresponding frequency response using the modified MATLAB code and paste it in the provided space.

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| $R_{15}$= 10530 Ω |

$$A_v = \frac{R_{15}}{R_{14}} + 1 = 40$$

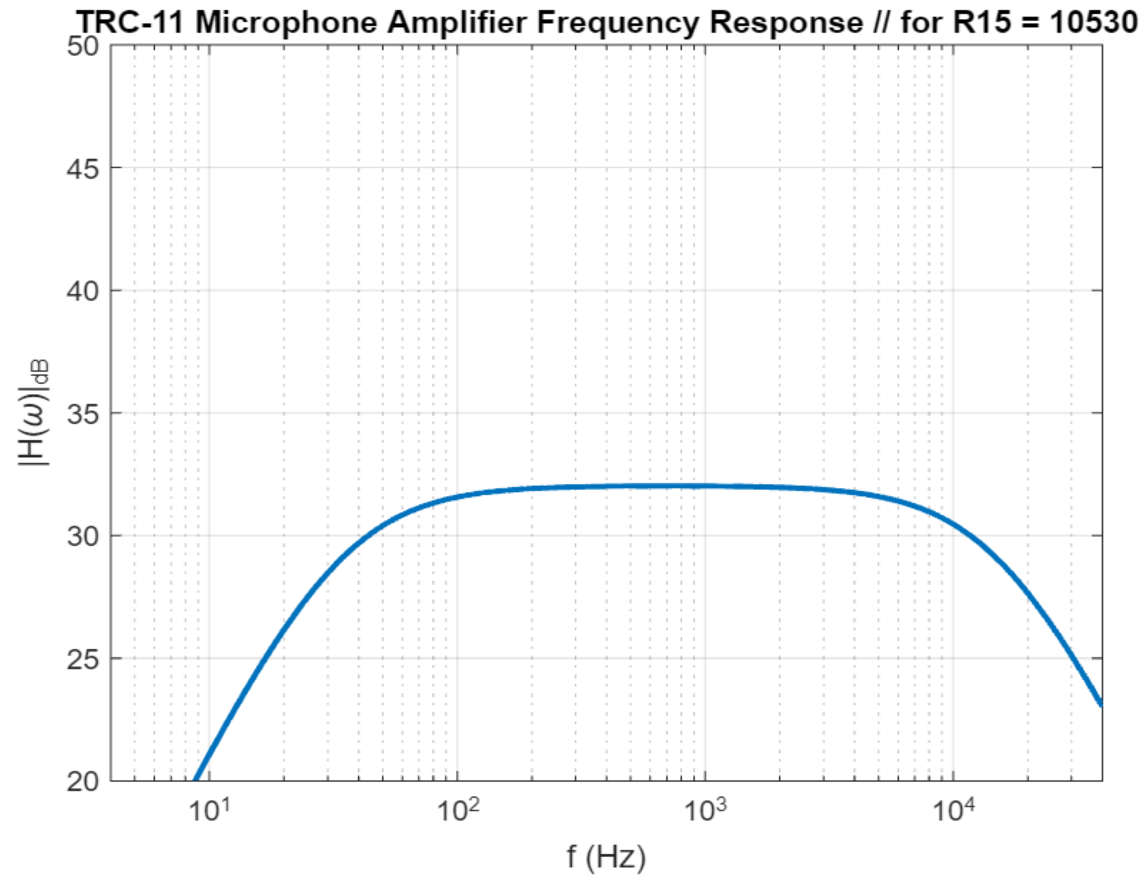
Should be 39

$x = [1, 2, 3]$

$y = [3, 4, 7]$

$$\frac{R_{15}}{270\,\Omega} = 39 \Rightarrow R_{15} = 10530$$

$x.\hat{}y = [3, 8, 21$




| 1.3. GRADE: |
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4. From the datasheet of the OPAMP LM358 on page ~~269~~ 331, the typical gain factor $A$ of the OPAMP is found as 110 dB. What is this amplifier's approximate supply current, $I_S$, from +12 V supply? What is the open-loop voltage gain, $A_0$, at 10 kHz in dB (page ~~269~~ 331) with $V_{CC}$=12 V?

$I_S$= 0,6mA   $A_0$= ~35dB

| 1.4. GRADE: |
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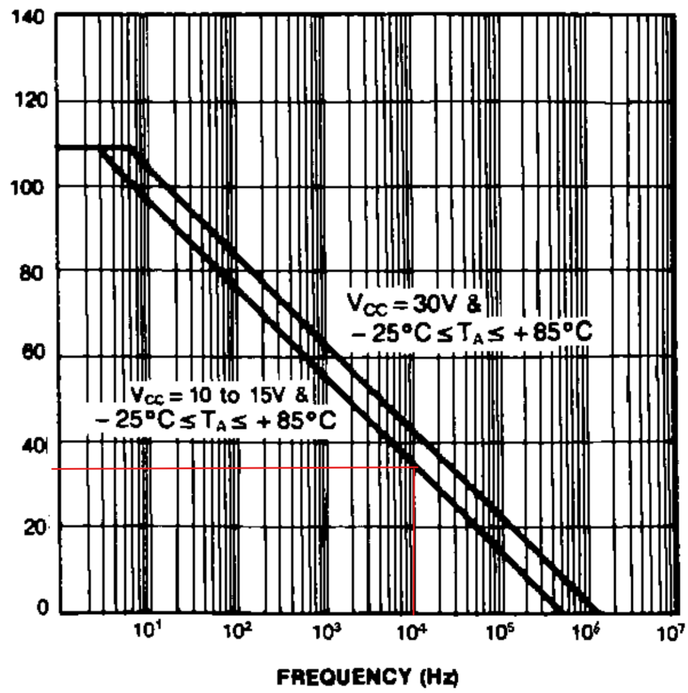

## 2. Loudspeaker/Earphone Amplifier

1. A schematic diagram of the loudspeaker amplifier is given in Fig. ??.

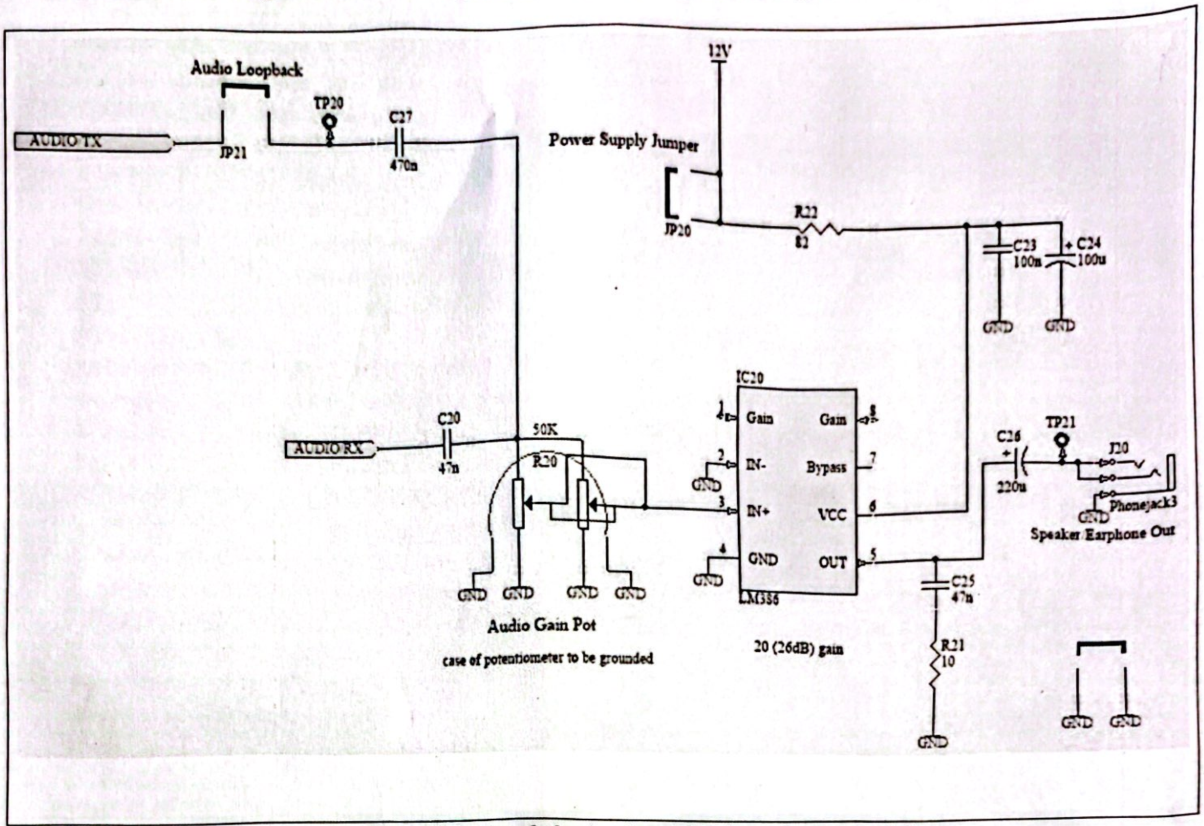


**Figure 4:** Schematic of the loudspeaker/earphone amplifier

2. IC20 is (LM386) a low-voltage audio amplifier integrated circuit. Examine the datasheet given on page 371. Which type of package is your integrated circuit? What is the supply voltage range of this IC?

| | |
|---|---|
| Package type | DIP 8 |
| Min. Supply voltage | 4 V |
| Max. Supply voltage | 12 V |

Absolute max. sup. voltage 15V

| 2.2. GRADE: |
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| Designator | Comment | Description |
|---|---|---|
| C20 | 47 nF | Capacitor, ceramic disc, 50V |
| C23 | 100 nF | Capacitor, ceramic disc, 50V |
| C24 | 100 $\mu$F | Electrolytic capacitor, polarized, 16V |
| C25 | 47 nF | Capacitor, ceramic disc, 50V |
| C26 | 220 $\mu$F | Electrolytic capacitor, polarized, 16V |
| C27 | 470 nF | Capacitor, ceramic disc, 50V |
| IC20 | LM386 | Low voltage audio amplifier |
| J20 | Phonejack | Speaker/earphone jack, PCB mount |
| R20 | 50 K | Potentiometer, stereo |
| R21 | 10 | Resistor, carbon film, axial leaded, 1/4W |
| R22 | 82 | Resistor, carbon film, axial leaded, 1/4W |

**Figure 5:** Bill of materials for the loudspeaker/earphone amplifier

## Experimental Work

### 1. Microphone Amplifier

1. The capacitor C11 is a *bypass* capacitor, to provide a cleaner supply voltage, +12 V to the OPAMP. The bypass capacitors should always used at the supply terminals of integrated circuits. They provide energy reserve to meet the demand by the IC when there are short duration high currents needed by the IC. Bypass capacitors meet this current demand at the closest point to the IC, instead of drawing that current all the way up from the supply circuit. They are particularly important at the high-frequency part of the circuit. The capacitor C11 is marked 104, meaning $10\times10^4$=100 nF. Mount and solder the capacitor C11.

2. C10 is an electrolytic capacitor to provide a clean bias voltage, $V_b$=+6 V, to the OPAMP and the microphone. Mount and solder it with the correct polarity.

3. Mount and solder the resistors, R12, R13, R14, R15, and R16. Look at the color codes of the resistors to identify them.

4. Mount and solder 1 $\mu$F capacitor C12. This is a polarized capacitor. The negative pin of the capacitor should look toward R13.

5. Mount and solder 1 nF capacitor C13. The marking on the capacitors is 102.

6. Mount and solder MK10. Note the direction of microphone capsule when mounting. If you mount it backwards, the capsule may be damaged.

7. Solder loops of wires to TP11 and TP12. Make sure that the loops are small, so that they do not touch any other conductor when bent.

8. Solder a piece of short wire between GND pins. Do not connect the jumper JP12 yet.

9. Place LM358 (IC10) on the component side of the PCB into its holes. Check the orientation of the IC before soldering. Pin 1 of LM358 should enter the rectangular-shaped pad on the PCB, while the other pads are oval. Solder all eight pins.

10. Measure the DC voltage $v_o$ (between TP12 and GND). The output voltage, $v_o$, should be within 1 V of $V_b$=6 V. Disconnect the power adapter.

$v_o$= 6.29 V

1.10. GRADE:

5mV 10mV 0.010
0.005

11. We use a signal generator to supply a signal to the microphone amplifier. Set the output level of the signal generator DS345 to 10 mV$_{pp}$ and the frequency to 1 KHz. Connect the signal generator leads between TP11 (red) and GND (black). High-frequency signal generators assume that they have a load of 50 Ω. If the load value were much higher than 50 Ω which is the case here, the actual voltage would be about twice (20 mV$_{pp}$) the value displayed on the signal generator. Connect CH 1 probe between TP12 and GND. Connect a BNC cable between SYNC output of the signal generator and EXT TRIG input of the oscilloscope. Set the trigger input to Ext. This triggering type should always be preferred, since the triggering is independent of the signal amplitude. You should see green Trig'd on the screen when the triggering is done. Press the TRIG VIEW button to see the triggering level and the SYNC signal. The triggering level should intersect the waveform somewhere in the middle. If necessary change the coupling to AC in the Trig Menu for better triggering.

You should now see a 1 KHz sine wave. Use the MEASURE button of the oscilloscope to read the peak-to-peak voltage, $v_{opp}$, at the output of the OPAMP. If the input voltage is increased further, the output sine wave is clipped. Try it and see the clipped sine wave. Clipping is a sign of OPAMP saturation. Under the clipped condition, use the MEASURE button to read the maximum ($v_{omax}$) and minimum ($v_{omin}$) values of the output voltage. Make sure that the channel coupling is DC to be able to measure the maximum and minimum values.

$v_{opp}$= 11,2 V $v_{omax}$= 12,6 V $v_{omin}$ = 1,40 V

1.11. GRADE:

12. Measure the amplifier's transfer function from TP11 to TP12 for the frequency range of 5 Hz to 20 kHz. The transfer function in dB is given as

40mV 160mV

20log ( 0,4

$$|H(\omega)|_{dB} = 20\log_{10}\left(\frac{v_{opp}}{v_{inpp}}\right) \quad \frac{0,04}{0,01}$$

0,16 / 0,01 160 / 10

56 / 20

Make sure that the output sine wave signal is not clipped. 13 frequency points covering the frequency range should be sufficient to show the variation. Do not forget the fact that the signal generator output is twice the voltage shown on its display with high impedance load. The frequencies listed below are chosen exponentially, so that they map linearly on the logarithmic scale. Measure the gain at the corner frequencies, $f_1$ and $f_2$, as you calculated in the preliminary work.

1600

160 8,94

| $f$ (Hz) | $\|H(\omega)\|$ (dB) | $f$ (Hz) | $\|H(\omega)\|$ (dB) | $f$ (Hz) | $\|H(\omega)\|$ (dB) |
|---|---|---|---|---|---|
| 5 | 24,08 dB | 160 | 44,71 dB | 5000 | 38,06 dB |
| 10 | 28,94 dB | 320 | 44,71 dB | 10000 | 33,62 dB |
| 20 | 38,88 dB | 640 | 44,71 dB | 20000 | 30.10 dB |
| 40 | 42,41 dB | 1250 | 43,40 dB | | |
| 80 | 44,29 dB | 2500 | 40,66 dB | | |

567 / 20 = 26

1700 / 20

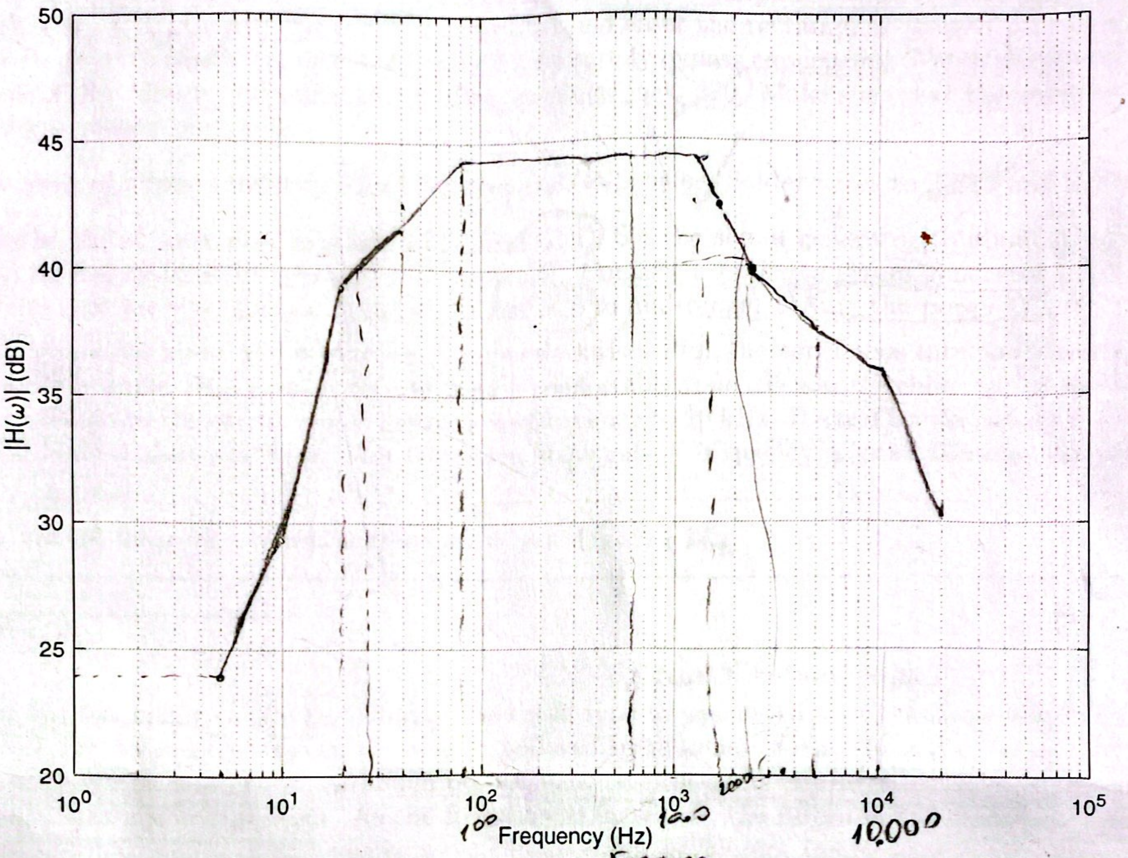
2889 / 7

| $f_1$ (Hz) | $\|H(\omega)\|$ (dB) | $f_2$ (Hz) | $\|H(\omega)\|$ (dB) |
|---|---|---|---|
| 33,86 Hz | 27,60 dB | 3386,7 Hz | 39,64 dB |

1929 / 29 = 2400 969 / 29 3360

480 / 20 = 1600 / 29 3440

13. Plot the transfer function on the log-dB grid below (transfer function on y-axis in dB and frequency on x-axis in logarithmic scale). Do not forget to mark the vertical scale properly.




↓ Frequen

1.13. GRADE:

Volt 29mV

f → Volt 0,02 20mV

14. Find the values of the corner frequencies experimentally by finding the frequencies, $f_1'$ and $f_2'$, where the gain is 3-dB less than (or 0.707 ×) its mid-frequency gain value. Compare your results with the calculated values above and the curve shown in Fig. 3. = 44,71 -3 = 41,71

Measured $f_1'$= ~28 Hz and $f_2'$= ~ 2700 Hz

Comparison: $f_1 = 33,86$ $f_2 = 3386$ There is a slight difference between experimental and theoretical values.

1.14. GRADE:

15. Remove the signal generator cable. Solder a piece of wire to JP12. This jumper will connect the microphone to the OPAMP input. Connect the oscilloscope probe between TP12 and GND. Whistle into the microphone, you should be able to see a sine wave on the oscilloscope screen.

The microphone amplifier circuit is now finished.

**CHECK POINT:**

## 2. Loudspeaker/earphone amplifier

1. ~~The input signal is connected to IC20 through a volume control potentiometer, R20.~~ A three-pin potentiometer would have been sufficient, but a stereo potentiometer with six pins is used here to provide mechanical robustness. Mount and solder R20. Mount a long stripped wire through the two side holes such that the metal case of the potentiometer is grounded. Tighten the wire before soldering the wires. Grounding the case shields the environmental noise.

2. Mount and solder all the remaining components in loudspeaker amplifier circuit: R21, R22, C20, C23, C24, C25, C26, C27; IC20. Pin 1 of LM386 should enter the rectangular-shaped pad on the PCB, while the other pads are oval. C23 and C24 are supply bypass capacitors. Watch the polarity of C24 and C26. Mount and solder the speaker/earphone jack, J20. Make sure that the entry hole of the jack is placed outwards.

0,1

3. Solder a piece of clipped resistance lead between the GND holes. Solder wires to ~~TP20~~ and ~~TP21.~~

4. Connect the signal generator between TP20 and GND. Set the signal generator to about 300 Hz sine wave of approximately 100mV peak amplitude. Connect a probe to signal generator output and see the sine wave on the oscilloscope. Adjust R20 to mid-range. Switch the power ON.

Reduce the volume knob R20 all the way counterclockwise. Plug the earphones into the earphone jack. Adjust volume (R20) until you can hear a comfortable tone. While watching the signal on the screen, listen to the sound as you increase the frequency to 20 kHz. Record the frequency above which you cannot hear anything. This frequency is the cut-off frequency of your auditory system.

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| Upper cut-off frequency of auditory system= ~ 15.5 kHz |

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| 2.4. GRADE: |

square → harmonic

Decrease the frequency to 300 Hz. Change the signal type to square wave of the same amplitude. Try to feel the difference between the sound produced by a square wave and a sine wave of the same frequency and amplitude. Although both signals have the same period, the square wave has additional harmonic components. As the frequency is increased, the filters in your audio circuits attenuate the harmonics of the square wave. In addition, harmonics frequencies eventually fall beyond your hearing cut-off frequency.

5. Increase the frequency while switching between sine and square waves until the difference in the sound you hear from them becomes insignificant. Record that frequency. Can you comment on the reason? (Hint: Consider the low-pass-filters along the way and your auditory system transfer response)

| |
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| $f=$ 7kHz |

Comment: While frequency increasing, our auditory system stops to differenciate between sinusoidal and square waves because the ~~systems~~ is mainly focused on mid range frequencies. Low pass aka high cut, kind of cuts the high frequencies because high frequencies also have high energy basically and it can damage our ~~auditory~~ system. Aud system is based on low pass filter to protect our ~~ears~~ basically.

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| 2.5. GRADE: |

Connect the oscilloscope probe to TP21. Set the signal generator to sweep mode, between 300 Hz and 3 kHz. Set the sweep mode on by pressing [SWEEP ON/OFF] button. A LED lights when it is on. Choose LIN SWP or LOG SWP option using up/down arrow keys. Enter the sweep start frequency by pressing [START FREQ]. Enter the sweep end frequency by pressing [SHIFT][STOP

F]. In sweep mode, the signal generator output frequency is continuously varied linearly (or logarithmically) between lower and upper limits and in a specified period. Set the sweep period to 1 Hz by pressing [RATE] button. Listen to the sound produced through the earphones while watching the waveform.

6. Remove the signal generator. Solder a jumper at JP21. The microphone amplifier output is connected to the loudspeaker amplifier input with this jumper. Whisper into the microphone. You should hear your own sound plus any surrounding sound from the earphone. If you cannot hear your own sound there is something wrong.

| |
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| Whisper clearly audible from the earphones? Yes/No |

| |
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| 2.6. GRADE: |

7. Turn off the power. Disconnect the jumper wire at JP21, the audio loopback jumper. This way, the microphone amplifier is disconnected from the earphone amplifier.

   You are now ready to proceed with RF circuits :)

   **CHECK POINT:**