LABORATORY REPORT - CHAPTER 3

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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 V_{CC} and GND. For this purpose, we use the regulated voltage, +6V.

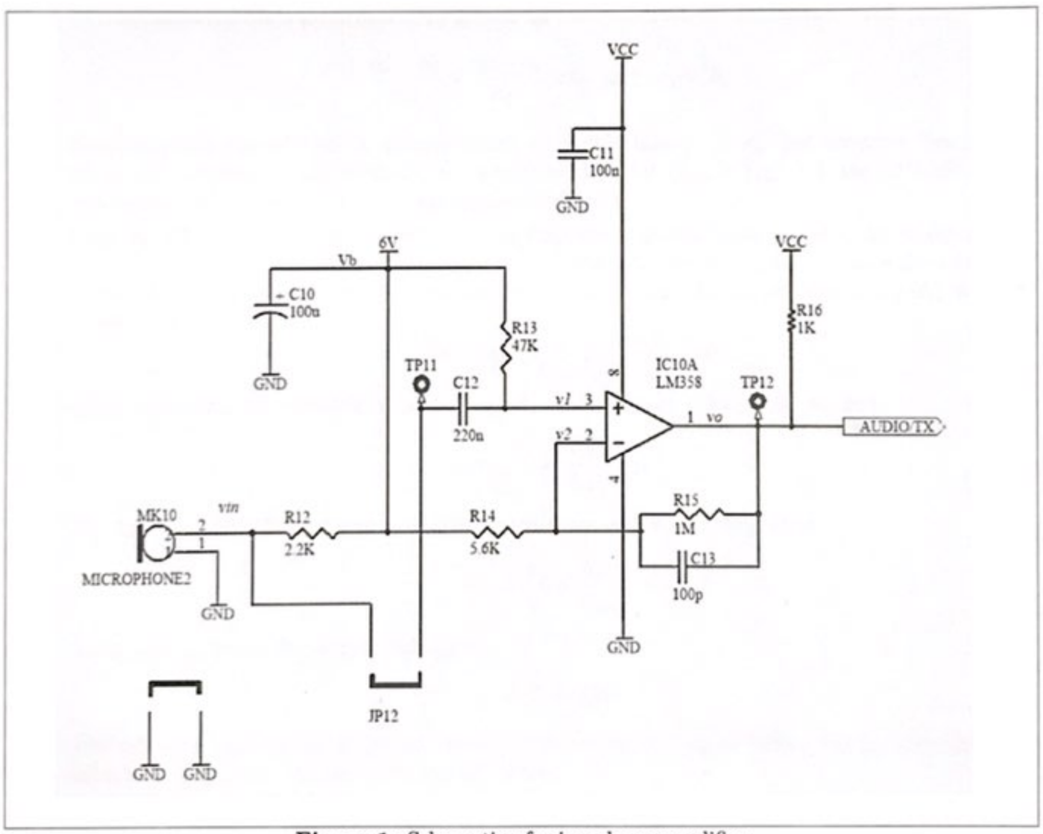


Figure 1: Schematic of microphone amplifier

Designator	Comment	Description
C10	100u	Electrolytic Capacitor, 16V
C11	100n	Capacitor, ceramic disk, 50V
C12	220n	Capacitor, ceramic disk, 50V
C13	100p	Capacitor, ceramic disk, 50V
IC10	LM358	Dual OPAMP
MK10	MICROPHONE2	Microphone Capsule
R12	2.2K	Resistor, carbon film, axial leaded, 1/4W
R13	47K	Resistor, carbon film, axial leaded, 1/4W
R14	5.6K	Resistor, carbon film, axial leaded, 1/4W
R15	1M	Resistor, carbon film, axial leaded, 1/4W
R16	1K	Resistor

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 $0 < v_o < V_{CC} - 2$, the OPAMP is not saturated. Since $V_b < V_{CC} - 2$ 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

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

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

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

Using superposition, the output voltage is

$$v_o = V_b + A_v V_P$$

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

$$f_1 = \frac{1}{2\pi R_{13} C_{12}} = \frac{1}{2\pi \cdot 47 \cdot 10^3 \cdot 22 \cdot 10^{-8}} = 15.4\ \text{Hz}$$

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

$$f_2 = \frac{1}{2\pi R_{15} C_{13}} = \frac{1}{2\pi \cdot 10^6 \cdot 10^{-10}} = \frac{10^4}{2\pi} = 1591.5 \text{ Hz}$$

Calculate these frequencies.

A_v= 179.6	f_1= 15.4 Hz	f_2= 1591.5 Hz or 1.6 kHz

1.2. GRADE:

3. The gain function in decibels is plotted in Fig. 3. If $f_1 \ll f \ll f_2$, then $|v_o/v_{in}| = A_v = 20\log_{10} A_v = A_{vdB}$. If $f = f_1$ of $f = f_2$, then $|v_o/v_{in}| = A_v/\sqrt{2} = A_{vdB} - 3\text{dB}$ (3 dB less than the low frequency value).

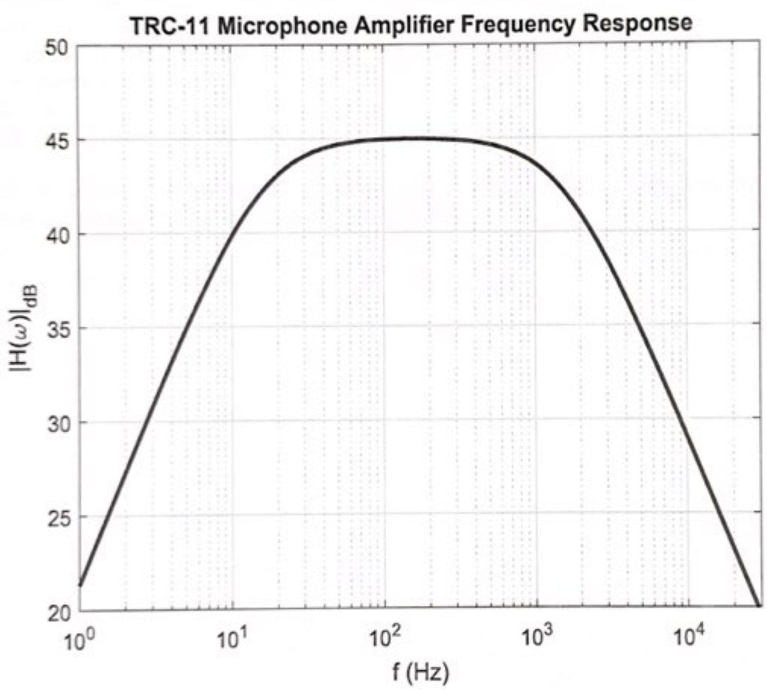


Figure 3: Calculated Frequency Response of the Microphone Amplifier

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
hold off
fmin=1; %minimum frequency in Hz
fmax=30e3; %maximum frequency in Hz
C12=220e-9; % C12 capacitor value in F
C13=100e-12; % C13 capacitor value in F
```

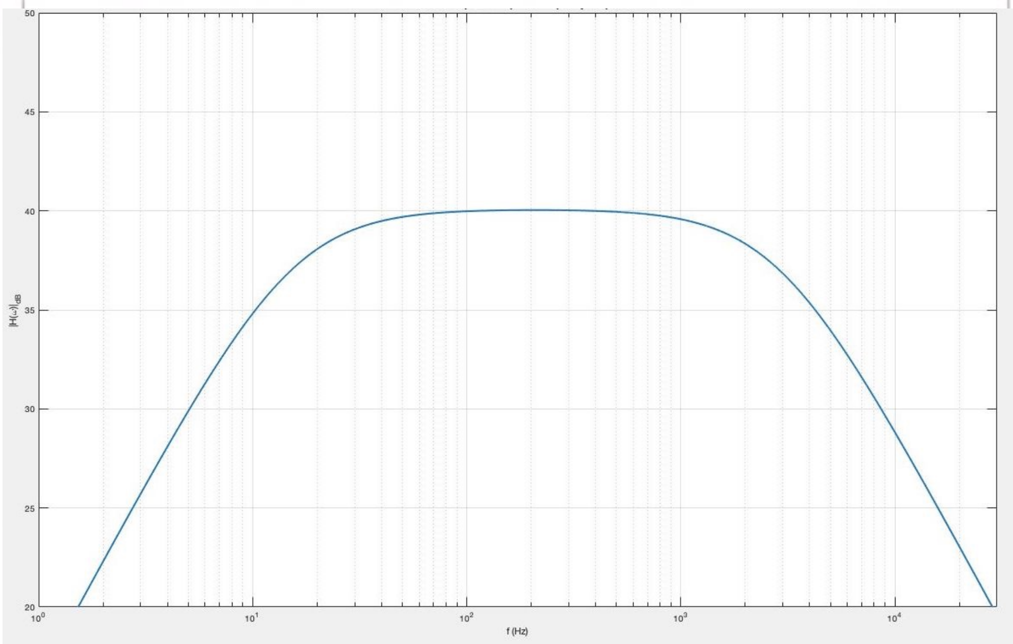
```
R13=47e3; % R13 resistance in Ohms
R15=1000e3; % R15 resistance in Ohms
R14=5.6e3; % R14 resistance in Ohms
Av=1+R15/R14;

f=fmin:fmin/5:fmax; % Frequency vector
w=2*pi*f; % angular frequency vector
H=Av./((1+j*w*R15*C13).*(1+j./(w*C12*R13)));
% 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
hold on
axis([fmin fmax  20 50]); % define the axes limits
```

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.

R_{15}= 554400 Ω or 0.55 MΩ



1.3. GRADE:

4. From the datasheet of the OPAMP LM358 on page 355, 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 355)?

I_S= 0.65mA A_0= 40dB

1.4. GRADE:

2. Loudspeaker/Earphone Amplifier

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

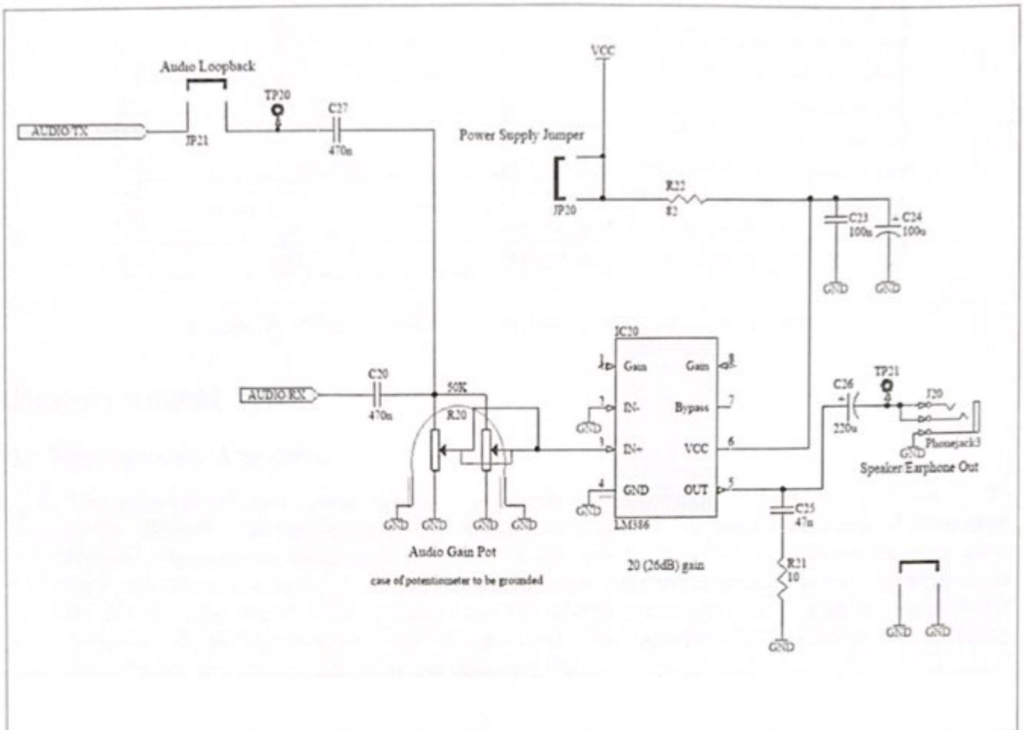


Figure 4: Schematic of the loudspeaker/earphone amplifier

2. IC20 is (LM386) a low-voltage audio amplifier integrated circuit. Examine the datasheet given on page 357. Which type of package is your integrated circuit? What is the supply voltage range of this IC?

Package type	DO8: DIP08
Min. Supply voltage	4V
Max. Supply voltage	12V

2.2. GRADE:

Designator	Comment	Description
C20, C27	470n	Capacitor, ceramic disk, 50V
C23	100n	Capacitor, ceramic disk, 50V
C24	100u	Electrolytic Capacitor, 16V
C25	47n	Capacitor, ceramic disk, 50V
C26	220u	Electrolytic Capacitor, 16V
IC20	LM386	Low Voltage Audio amplifier
J20	Phonejack3	Speaker/Earphone jack, PCB mount
R20	50K	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, V_{CC} (+12 or +9 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 10×10^4=100 nF. Mount and solder the capacitor C11.
2. C10 is an electrolytic capacitor to provide a clean bias voltage, V_b, 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 220 nF capacitor C12. The marking on the capacitor is 224.
5. Mount and solder 100 pF capacitor C13. The marking on the capacitors is 101.
6. Mount and solder MK10. Note the direction of microphone capsule when mounting.
7. Solder loops of wires to TP11 and TP12.
8. Solder a piece of 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= 6.64 V

1.10. GRADE:

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 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.

   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.

v_{opp}= 10.4V v_{omax}= 5.52V v_{omin} = -4.88V

1.11. GRADE:

12. Measure the amplifier's transfer function from TP11 to TP12 for the frequency range of 1 Hz to 20 kHz. Make sure that the output sine wave signal is not clipped. Calculate the transfer function in dB, and plot the transfer function on a graph paper with logarithmic scales (transfer function on the y-axis and frequency on the x-axis). 15 frequency points covering the frequency range should be sufficient to show the variation. The frequencies listed below are chosen exponentially, so that they map linearly on the logarithmic scale.

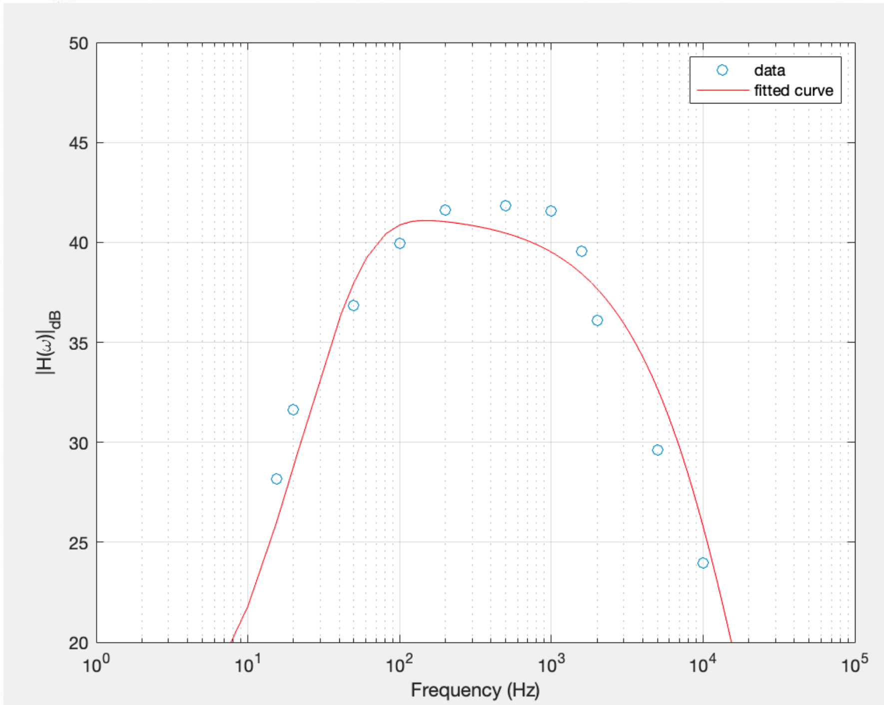
   Measure the gain at the corner frequencies, f_1 and f_2, as you calculated earlier.

f (Hz)	$\|H(\omega)\|$ (dB)	f (Hz)	$\|H(\omega)\|$ (dB)	f (Hz)	$\|H(\omega)\|$ (dB)
1	13.56	50	39.94	2000	36.10
2	14.42	100	41.62	5000	29.60
5	13.76	200	41.84	10000	23.97
10	17.28	500	41.56	20000	18.91
20	31.62	1000	39.55		

f_1 (Hz)	$\|H(\omega)\|$ (dB)	f_2 (Hz)	$\|H(\omega)\|$ (dB)
15.39	28.15	1591.5	36.83

1.12. GRADE:

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.



1.13. GRADE:

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.

Measured f_1'= 15.39 Hz and f_2'= 1591.5 Hz

Comparison: From calculated values and measured values they are almost same which is an exellent result.

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.

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.

Upper cut-off frequency of auditory system= 18 kHz

2.4. GRADE:

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)

$f=$ 8.8 kHz

Comment: when switching between sine and square waves there were a slight difference but this difference dissipated as frequency increased.

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. 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 between lower and upper limits and in a specified period. Set the sweep period to 1 Hz by pressing [RATE] button. You may change the mode of sweep by up/down arrow keys. 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.

Whisper clearly audible from the earphones? Yes/No

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: