

# Linear Circuit Experiment

## MAE 171a - Winter 2014

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### Overview and Aim of the Experiment

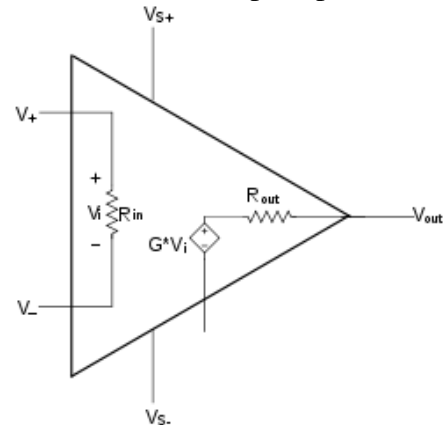
In this experiment you will learn how to build and analyze *active* linear circuits to perform basic signal conditioning: **amplification of signals, adding signals and filtering of signals**. The term *active* refers to electronic components which require external power to operate. In our case here we will be using operational amplifiers (opamps) as the active components. We will also use *passive* components (resistors, capacitors), a *sensor* (microphone) and a *transducer* (speaker). By designing/building a feedback network around an opamp, and cascading different opamp circuits, we will eventually make a **circuit that is able to amplify and filter signals coming from a microphone** and output to a speaker for audible confirmation of your circuit. **Error analysis** involves estimation of the gain and filter cut-off frequency from experimental data measured on your circuitry.

### Background on Op-Amps

An operational amplifier (op-amp) is a direct current (DC) coupled high-gain electronic voltage amplifier with differential inputs and typically a single output. In most situations the output of the opamp is controlled by negative feedback, which determines the output voltage gain. Less commonly, positive feedback can be used to produce regenerative gain and oscillation circuits. High impedance at the input terminals and low output impedance are important typical characteristics of an op-amp.

The figure on the right is the schematic of an ideal opamp. The main part in an amplifier is the dependent voltage source  $G \cdot V_i$  that amplifies with a factor  $G$  in relation to the voltage drop  $V_i = V_+ - V_-$  across the input impedance  $R_{in}$ . Many uses have been found for operational amplifiers and an ideal opamp seeks to characterize the physical phenomena that make opamps useful.

Supply voltages  $V_{s+}$  and  $V_{s-}$  are used internally to implement the dependent voltage sources. The positive source  $V_{s+}$  acts as an upper bound on the output, and the negative source  $V_{s-}$  acts as a lower bound on the output. The *internal*  $V_{s+}$  and  $V_{s-}$  connections are not shown here and will vary by implementation of the operational amplifier.



An *ideal* op-amp has the following properties:

- Infinite (open-loop) gain  $G$ . When doing theoretical analyses this means that a limit should be taken as open loop gain  $G$  goes to infinity.
- Zero output impedance or  $R_{out} = 0$ . This means that any output load connected to the op-amp will not influence the output voltage.
- No dynamics or infinite bandwidth and slewrate. This means that the transfer function or frequency response is simply  $G$ , without any delay or phase shift, implying infinitely fast settling time.

- Infinite input impedance, making  $R_{in} = \infty$ . This means that no current (zero input current) would flow from  $V_+$  terminal to the  $V_-$ .
- No leakage or bias currents. This means that together with the infinite input impedance absolutely no input currents are present.
- Zero offset voltage. This means when  $V_i = V_+ - V_- = 0$  (by connecting the input terminals), the output  $V_{out} = 0$ . With  $R_{out} = 0$  this implies that the output acts as a virtual ground.

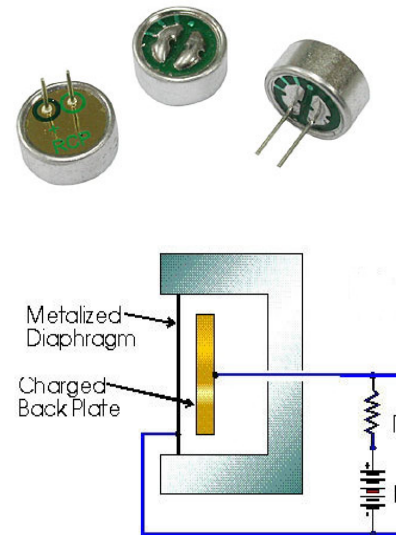
A real op-amp can only approximate these ideal conditions. However, modern opamps have sufficient performance that for a quick initial analysis of a circuit you can assume the ideal properties. In this Linear Circuit Experiment we will be using an MCP6271 opamp from Microchip Corporation. In this exercise we will determine some of the opamp performance limitations and circuit properties.

### **Background on Audio Applications**

In audio applications, signal conditioning by means of amplification, mixing (adding) and filtering mainly focuses on low-power signals composed primarily of frequencies between 20 Hertz to 20k Hertz (the human range of hearing). After all the signal conditioning has taken place these low-power signals need to be further amplified to a level suitable for driving loudspeakers. This is commonly the final stage in a typical audio playback chain. The power of an input signal to an audio signal conditioning circuit may be only a few hundred microwatts, while the output may be hundreds of watts.

In our experiment we will generate low-power signals for your signal conditioning circuitry using a DC-biased condenser (capacitor) microphone. A typical small condenser microphone and its internal structure is shown on the right. Condenser microphones use a thin diaphragm to respond to changes in sound pressure. The membrane acts electrically as one plate of a capacitor to translate fluctuations in pressure to a fluctuating electrical signal. The microphone (capacitor) is DC-biased with a fixed charge  $Q$ . Since the Voltage  $V$  in Volts across a capacitor is given by  $V = Q / C$ , where  $Q$  = charge in Coulombs and  $C$  = capacitance in Farads, fluctuations in capacitance  $C$  due to sound (air) pressure fluctuations will produce corresponding fluctuations in the voltage  $V$  across the capacitor.

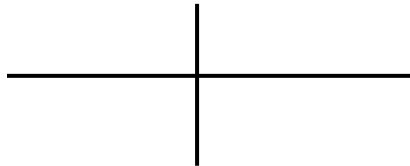
To create (amplified) sound (air pressure variations) output of a level audible by the human ear, the signals from the DC-biased condenser (capacitor) microphone need to be amplified and send to a loudspeaker. A loudspeaker (shown to the right) is an electro-acoustical transducer that converts an electrical signal into air pressure variations (sound). Typically, in a loudspeaker an electrical signal is applied to a low impedance voice coil to create a magnetic field. The magnetic field thus produced by the voice coil interacts with the speaker's magnetic field (permanent magnets in the speaker) to generate a net magnetic force which moves a lightweight cone back and forth to produce longitudinal pressure waves our ears interpret as sound.



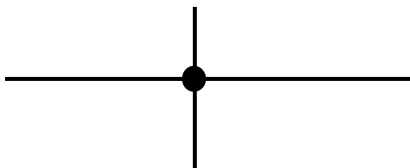
### Important tips on reading, building and measuring circuits in this lab

**Wire connections:** In the schematic diagrams provided in this lab there is an important electrical distinction between lines that cross with a *tie dot* and lines that cross with no tie dot. Be sure to maintain this distinction as you build your circuits or they will not work properly.

No tie dot = **no connection**



Tie dot = **connection**



**Wire colors:** As you build your circuits you should use a consistent color scheme for wire colors so you know which wires perform which functions. This will help later with debugging your own circuit. If you need to ask for help with your circuit, the debugging is going to be much easier if certain wire colors always mean the same thing.

+5 volt power:	<b>Red</b>
Ground:	<b>Black</b>
Output signals:	White
Input signals:	Green

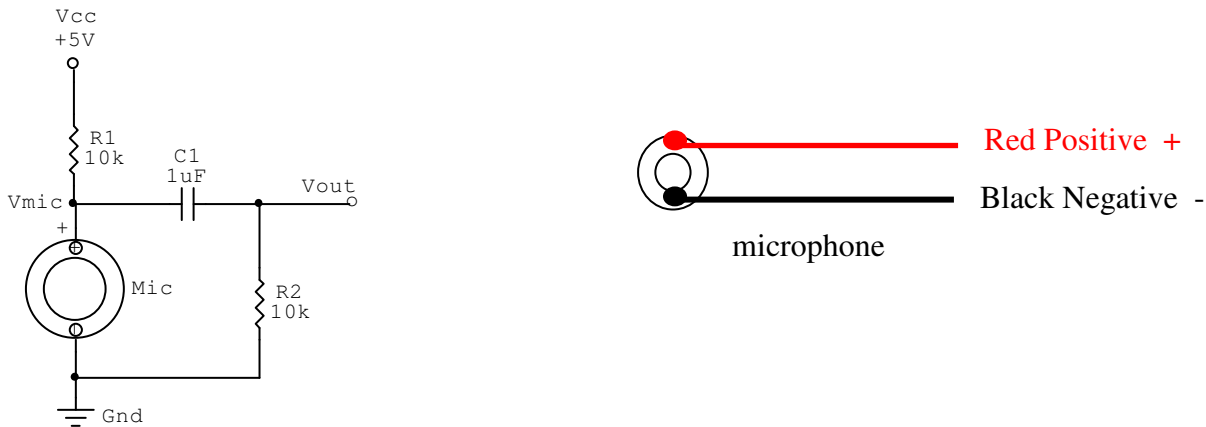
**Voltage measurements:** all voltage measurements (for example at output or input signals of a circuit) should always be taken with respect to ground.

## WEEK 1

The task of week 1 is to build a simple circuit to amplify a low-power microphone signal. During the lab you will experimentally verify the amplification and make sure intermediate and amplified signals do not get 'clipped' against the *rails* (limits) of the circuit power supply.

### Task 1-a: measure and display a signal from a microphone (RC circuit)

Using the supplied *electret* (condenser) microphone and the concept of DC-biasing, first build a DC-bias circuit for the microphone as shown above to measure (sound) pressure variations. You must **preserve the polarity of the microphone** when building the circuit: **red wire = positive**, **black wire = negative**.



**Explanation of circuitry:** The microphone circuit is powered with  $V_{cc} = +5V$  and **ground (Gnd)**, found on your proto-board. Select resistors **R1 (10 k $\Omega$ )** and **R2 (10 k $\Omega$ )**. R1 is used to create the DC bias voltage/charge for the microphone and R2 is the *load* resistor. The load resistor together with capacitor **C1 (1  $\mu F$ )** creates a high pass filter that eliminates the bias voltage  $V_{mic}$  from your output signal  $V_{out}$ . This filter means that  $V_{out}$  will just consist of voltage *fluctuations* produced by the microphone response to sound.

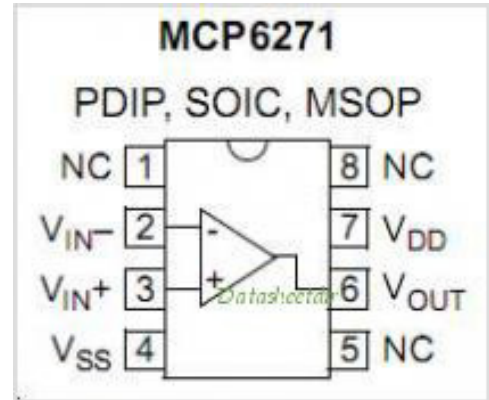
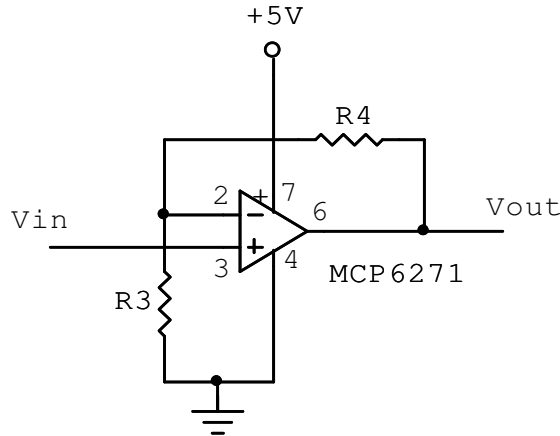
**Keep in mind: all voltage measurements (for example at points  $V_{mic}$  or  $V_{out}$ ) are always taken with respect to ground.**

- Build the circuit shown above. Keep in mind the polarity of the microphone.
- Measure the DC-bias (offset) voltage at  $V_{mic}$ . Based on your measurements, compute the apparent resistance  $R$  value of the microphone MIC.
- Measure the DC-bias (offset) voltage at  $V_{out}$ . Explain (in report) why it is (close to) 0.
- Display and save time plots of the signal  $V_{out}$  on the oscilloscope and measure the pk-pk  $V_{out}$  response for the following cases:
  - Short the microphone ( $V_{mic} = 0$ ) by connecting a *jumper* wire directly from the  $V_{mic}$  node to ground. Theoretically  $V_{out} = 0$ , but due to noise this will not be the case. Hence, this measurement gives an estimate of the basic *noise floor* of your circuit.
  - Remove the short. The mic is now responding to the ambient sound in the room. This will give you the general level of response due to background noise and sounds in the room picked up by the microphone.
  - Now speak into the microphone and observe/measure the signal on the scope. This will give you estimate of the signal that needs to be conditioned and amplified by the circuitry you will be building further in this exercise.

**Task 1-b:** analyze and build op-amp amplifier.

This task can be done independently from Task 1-a and **should be done simultaneously with Task 1-a** by the other members of your group in this laboratory.

Using the MCP6271 opamp we will set up a basic non-inverting amplifier to amplify the microphone signal as shown in the schematic below.

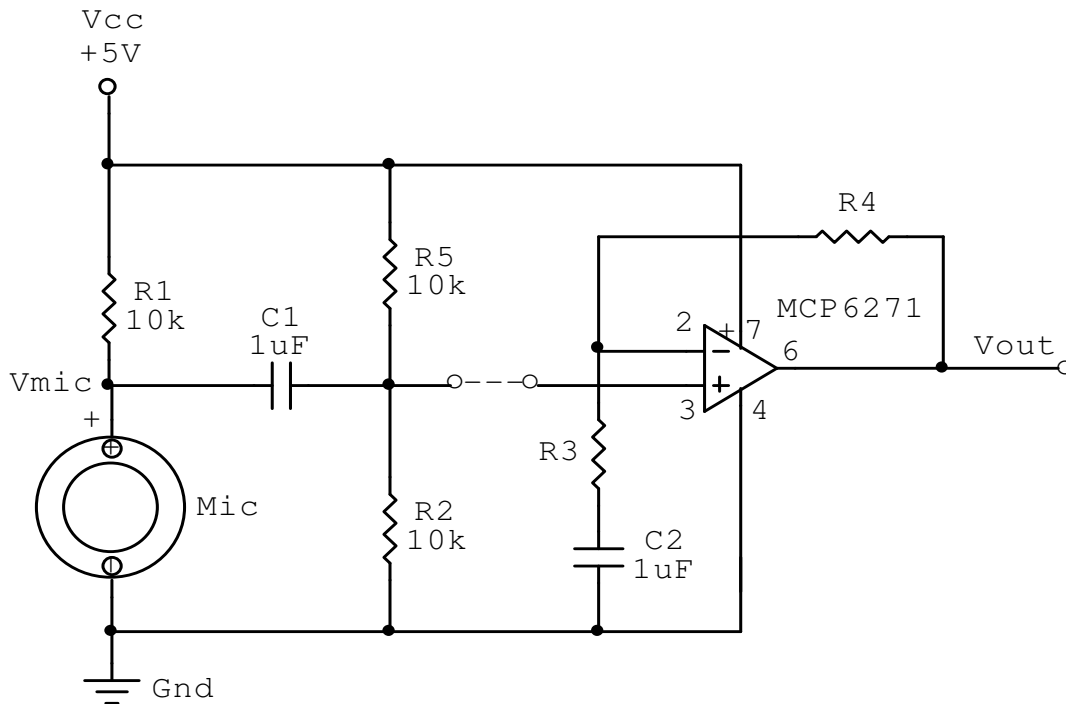
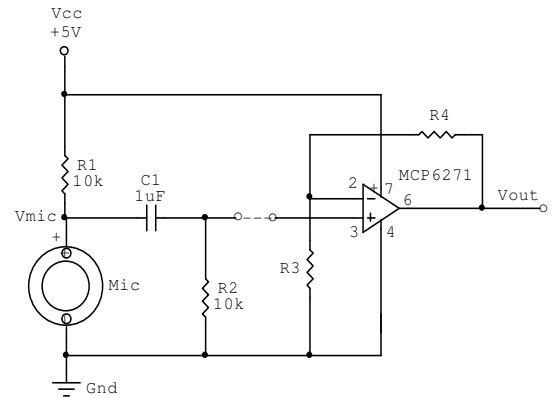


*“Pinout” of the Microchip MCP6271 Rail-to-Rail Operational Amplifier*

- Build the circuitry above, using **R3 = R4 = 10k**. With these resistor values, what is the theoretical **gain** (amplification) of the amplifier?
- Short Vin (using a jumper wire from Vin to ground) and measure Vout. Explain (in report) the difference between theoretical value of Vout = 0 and your actual measurement.
- Experimentally verify the gain of your op-amp circuitry as follows:
  - Use the function generator to create a **100 mV peak-peak 1 kHz** sinewave input signal with a **0 Volt offset** and connect to Vin.
  - Display and save time plots of the signal Vout on the scope. Explain (in report) why the output signal Vout gets **clipped** (squared off) for negative values of the input signal.
  - Use the function generator to create a **100 mV peak-peak** sinewave input signal and choose an **offset to avoid clipping** of the amplified output signal.
- Now replace resistor **R3 = 1k** resistor and **R4 = 100k** resistor. With these resistor values, what is the gain/amplification of the (non-inverting) amplifier?
- Experimentally verify the gain of your op-amp circuitry as follows:
  - Use the function generator to create now a **20 mV peak-peak 1 kHz** sinewave input signal Vin. The 20 mV represents the typical signal amplitude from a microphone.
  - Display and save time plots of the signal Vin and Vout on the scope and **make sure Vout does not get `clipped` for both positive and negative values** of the input signal by and choosing an appropriate offset for Vin.
  - Change the frequency of the sinusoid input Vin and observe gain and phase shift between the input Vin and output Vout signals. Are you able to see changes in gain and phase shift with different frequency? Save time plots for 100 Hz, 1 kHz, 10 kHz and 20 kHz and try to explain (in report) the observed results.
  - **Error analysis:** estimate gain of your circuitry (several times for statistical analysis).

**Task 1-c: combining microphone and amplifier circuitry**

This task **combines the results from Task 1-a & Task 1-b**. A direct combination of the previous two circuits leads to the schematic to the right. Unfortunately, because the signal from the microphone can be both positive or negative with respect to ground, the amplified microphone signal will be clipped in the op-amp circuitry for negative values and only amplified for the positive values. To overcome this problem, we modify the combined circuitry as shown below, where we add **R5 (10k)** from Vcc to R2, and we insert a capacitor **C2 (1µF)** between R3 and ground.



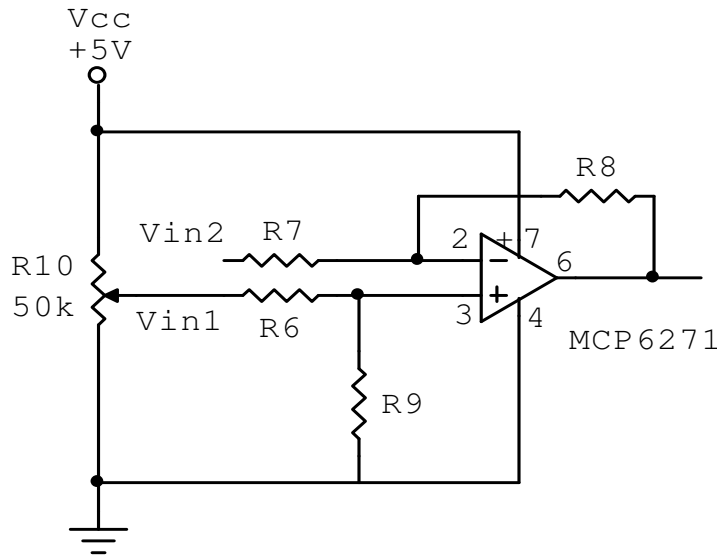
- Explain (in report) the purpose/role of capacitor C2. You can explain this via the filter properties of this circuit (see also lecture notes)
- Combine both microphone and op-amp circuitry by building the circuit above.
- Measure the DC-bias (offset) at the + input of the op-amp and DC-bias (offset) at Vout. Explain why both of them should theoretically be at 2.5V. Explain (in your report) any differences between the theoretical values and your measured values.
- Display and save time plots of the signal Vmic and Vout on the scope and comment on the performance of your microphone pre-amplifier (you can create interesting sounds yourself). Verify that you are able to amplify the microphone signal Vmic about 100 times without clipping the signal.

**WEEK 2**

The task for week 2 is to build a simple **difference amplifier**. This circuit configuration will be used as a *mixer*, which allows you to add a signal to your microphone signal. During the lab you will experimentally verify the gain of a difference amplifier and avoid clipping of signals.

**Task 2-a: difference amplifier** (See also the appendix for additional information on this task)

We are going to create a basic differential amplifier to **amplify the difference between two signals**. Using a rail-to-rail opamp such as the MCP6271, the circuit topology shown below will demonstrate the function of the difference amplifier.

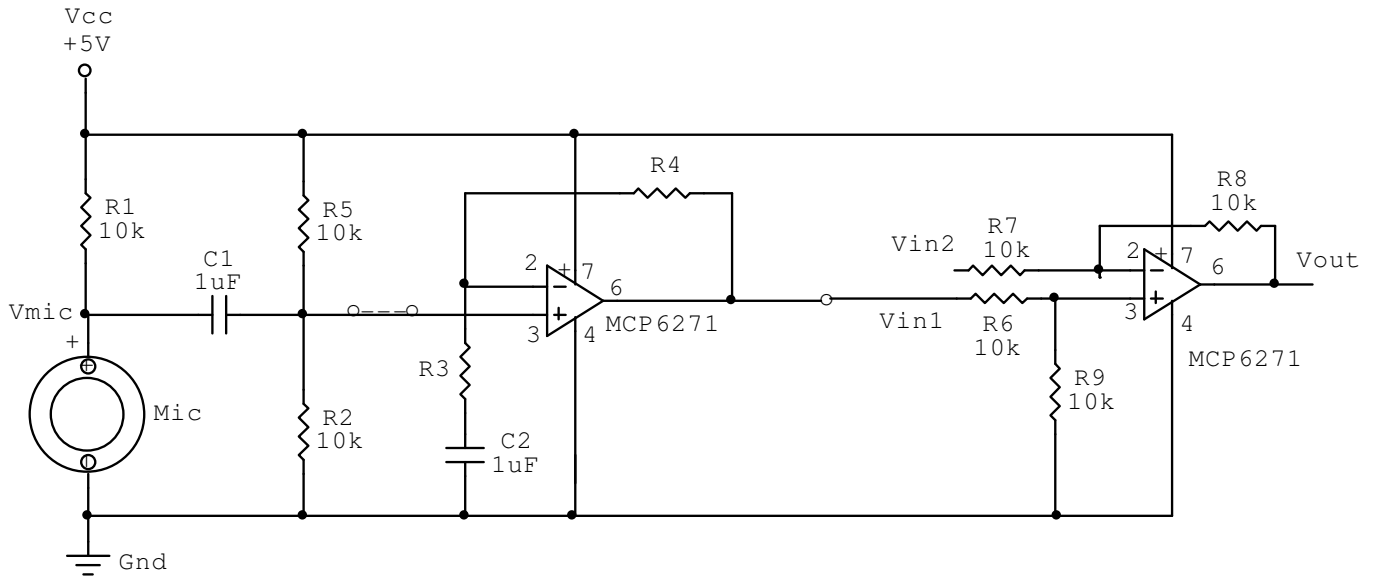


Difference amplifier configuration

- Build the circuitry above using **R6 = R7 = R8 = R9 = 10kΩ** and **R10 = 50 kΩ potentiometer** (variable resistor).
- Explain (in your report) that  $V_{out} = V_{in1} - V_{in2}$  in case **R6 = R7 = R8 = R9**.
- Connect  $V_{in2}$  to ground, measure  $V_{in1}$  and adjust the potentiometer R10 so that  $V_{in1} = 2.5V$ . This adjustment provides a DC-bias voltage which makes  $V_{out} = 2.5V$  when  $V_{in2} = 0V$ .
- Experimentally verify the functioning of your difference amplifier (see also appendix) via:
  - Use the function generator to create a **100mV pk-pk 1 kHz** sinewave input signal with a **0 Volt offset** and connect this signal to  $V_{in2}$ .
  - Display and save time plots of the signal  $V_{in2}$  and  $V_{out}$  on the scope.
  - Verify that indeed  $V_{out} = V_{in1} - V_{in2}$  and explain (in report) why you can use the potentiometer R10 to adjust the DC-offset (bias) of the output signal  $V_{out}$ .
  - Adjust R10 so that the DC-offset (bias) of  $V_{out}$  is *as small as possible*, but without ‘clipping’ the output signal  $V_{out}$ .
- Leave **R8 = R9 = 10k** but make **R6 = R7 = 1k**. Based on these resistor values, compute the theoretical differential gain  $\gamma$  in  $V_{out} = \gamma(V_{in2} - V_{in1})$ .
- Determine the gain of the differential amplifier by **creating a graph** in which you plot the amplitude of  $V_{in}$  (x-axis) against  $V_{out}$  (y-axis) for different  $V_{in}$  pk-pk amplitudes.
- For **error analysis**: fit curves to the measured data in the graph and estimate **gain & offset** of your differential amplifier circuit.
- Explain (in your report) the difference in gain/amplitude and offset of  $V_{out}$ . Adjust the potentiometer R10 as necessary to avoid clipping of the output signal  $V_{out}$ .

**Task 2-b: combining microphone amplifier circuitry of week 1 with difference amplifier**

This task combines the end results from Task 1-c (Week 1) and the difference amplifier of Task 2-a above. A direct combination of the circuits leads to the schematics below. Set  $R6 = R7 = R8 = R9 = 10k\Omega$  and the extra input signal  $Vin2$  is simply subtracted from the amplified microphone signal circuitry designed in Week 1.



- Build the combined Week 1 amplified microphone signal circuit and the differential amplifier circuitry from Task 2-a as shown in the above schematic.
- Explain (in report) why the potentiometer R10 has been eliminated from the circuit, e.g. why is there no need to introduce a bias voltage at  $Vin1$  of the differential amplifier?
- Again measure the DC-bias at  $Vin1$  and the DC-bias (offset) at  $Vout$ . Explain why both of them should (theoretically) be at 2.5V. Explain any differences between this theoretical value and your measured value.
- Experimentally verify the functioning of your circuitry as follows:
  - Use the function generator to create a **100mV pk-pk 1 kHz** sinewave input signal with a **0 Volt offset** and connect it to  **$Vin2$** .
  - Display and save time plots of the  $Vout$  signal on the scope by also creating sound signals for the microphone. Verify that you can see the 100 mV sine wave on your amplified microphone signal without clipping the output signal  $Vout$ .

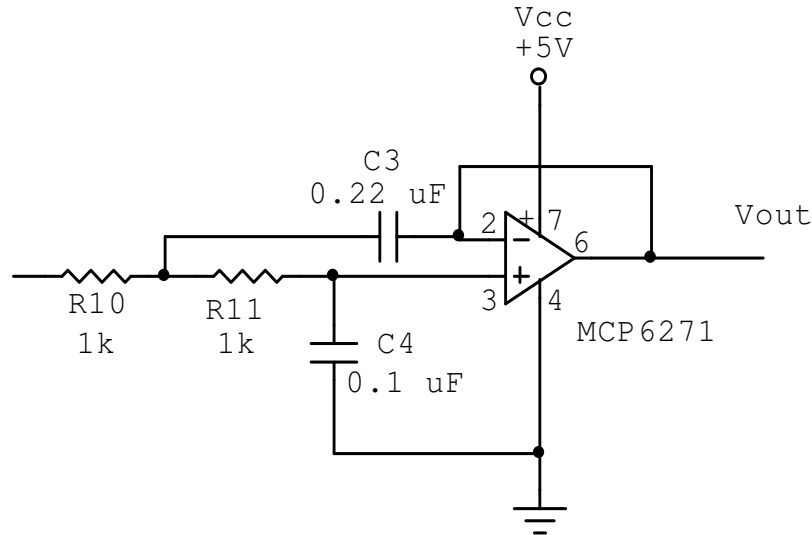


### WEEK 3

The task of week 3 is to build a simple **low pass filter** to filter both the microphone and the mixer signals. During the lab you will experimentally verify the frequency response of your filter by applying signals with different frequencies. Again, you must verify that the filtered signals are not getting clipped.

#### Task 3-a: 2<sup>nd</sup>-order low pass Butterworth filter

To reduce of the amplitude of the filtered signal at higher frequencies, you are going to build and test a 2<sup>nd</sup> order (Butterworth) filter:

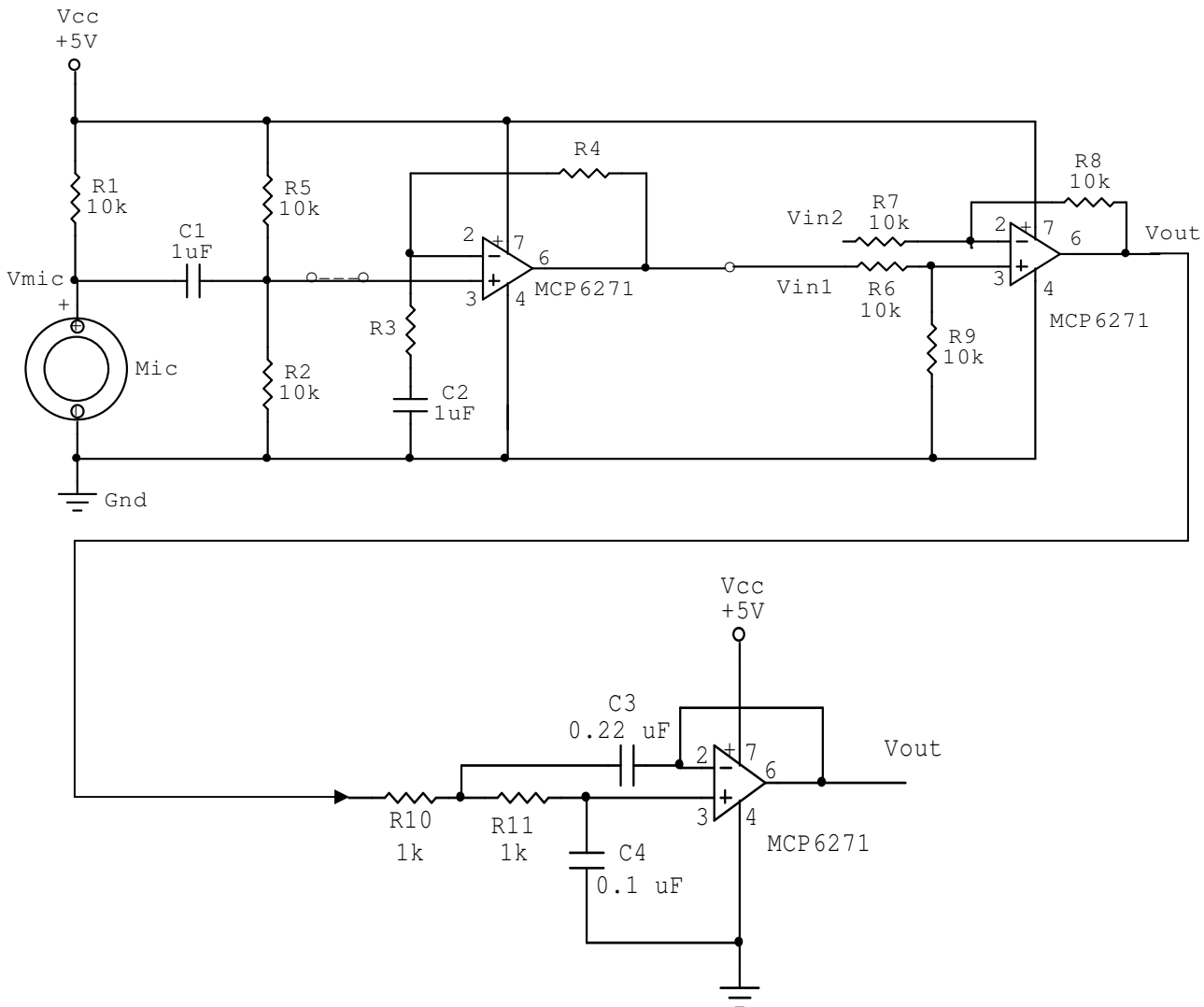


2<sup>nd</sup>-order low pass Butterworth filter

- **Build the above circuitry.** The capacitor values of 0.1 uF and 0.22 uF shown in the schematic are real-world available components, approximating but not equal to the ideal values of 0.141 uF and 0.283 uF, respectively, to obtain a so-called 2<sup>nd</sup> order low-pass Butterworth filter characterized by a resonant frequency of  $\omega_n = 5000\text{rad/s}$  (approximately 800 Hz) and a damping ratio of  $\beta = \sqrt{1/2}$  (approximately 0.707).
- For your report, **compute the Bode response with Matlab** using the ideal numerical value for  $\omega_n$  above (see also lecture notes).
- Experimentally verify the amplitude frequency response of this filter as follows:
  - Use the function generator to create a **1V pk-pk** sinewave input signal with a **2 V offset** and connect to  $V_{in}$ . The offset ensures the output does not get clipped or distorted.
  - **Vary the input signal frequency** from 10Hz – 10kHz and record the pk-pk amplitude of the output signal  $V_{out}$  for various (at least 10) frequency points. Best is to choose the frequency points with a logarithmic progression, e.g. 10Hz, 25Hz, 50Hz, 100Hz, 250Hz, 500Hz, 1kHz, 2kHz, 5kHz, 10kHz.
  - **Create a log-log graph** in which you plot the applied frequency (log x-axis) against amplitude of  $V_{out}$  (log y-axis) for the different frequencies. Compare the graph (or measured points) with the frequency response created by Matlab. Keep in mind that gain in dB =  $20 \cdot \log_{10}(\text{gain})$  and verify that this filter gives a 40 dB/decade reduction at frequencies above the cut-off frequency.
  - **For error analysis: estimate the cut-off frequency** of your filter by experimentally determining the frequency at which the pk-pk  $V_{out}$  amplitude is reduced by a factor of -3 dB =  $\sqrt{1/2} \approx 0.707$ . Perform the estimation several times for statistical analysis.

**Task 3-b:** combine microphone amplifier (Week 1), mixing circuitry (Week 2) with low pass filter

This last task is to combine the low pass filters you designed and tested this week with the completed circuitry of week 2. This is your final circuit that contains all the sub-circuits you have designed and by now looks pretty complicated. If you have made it this far – congratulations!



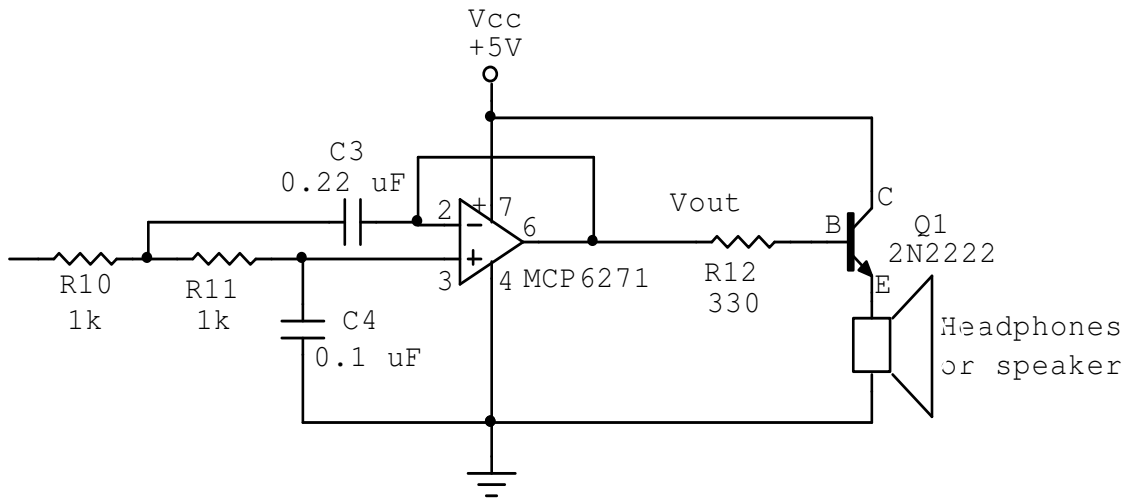
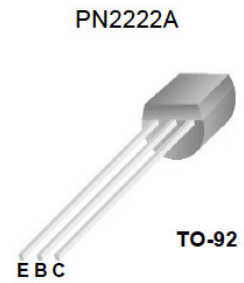
Final circuit: Microphone preamp (Week 1), differential amp (Week 2) and filter (Week 3)

To finish up the final testing of your circuit, the last final tests should be included in your report:

- Connect Vin2 (mixer/differential amplifier input) to ground. Display on the scope and save time plots of the signal Vin1 (the amplified microphone signal) and Vout (the amplified and filtered microphone signal).
- Connect Vin2 (mixer/differential amplifier input) to the function generator and generate a 100 mV 5 kHz zero offset sine wave. This signal simulates a disturbance on your microphone signal that was mixed in. Again display Vout (the amplified and filtered microphone signal) on the scope and verify that the 5 kHz disturbance has been reduced.
- As a final last fun test you can listen to the amplified and filtered microphone signal via a (high impedance) ear phone speaker. You do this as follows: **connect the output of the last op-amp (filter design stage) via a 1 μF capacitor (to remove the bias voltage) to a set of headphones.** Now you can listen to the amplified and filtered microphone signal. An optional power boost can be given to the last op-amp to also power a low impedance speaker (see next page).

**Task 3-c: (optional) simple power boost**

The output of the MCP6271 opamp is limited to about 15 milliamps (mA). This is not enough current to drive a speaker to listening sound volume. To boost the power of the op-amp in our last filter stage, an additional output stage can be added. A simple way to do this is by adding an NPN transistor Q1 (2N2222 or similar) that can carry more current (sourced from the +5V power supply), and base resistor R12 (330 ohms). The transistor acts as a voltage follower as indicated in the schematic below and its function is to provide sufficient current to drive the headphone or speakers:



Final filter section with added transistor and speaker or headphones

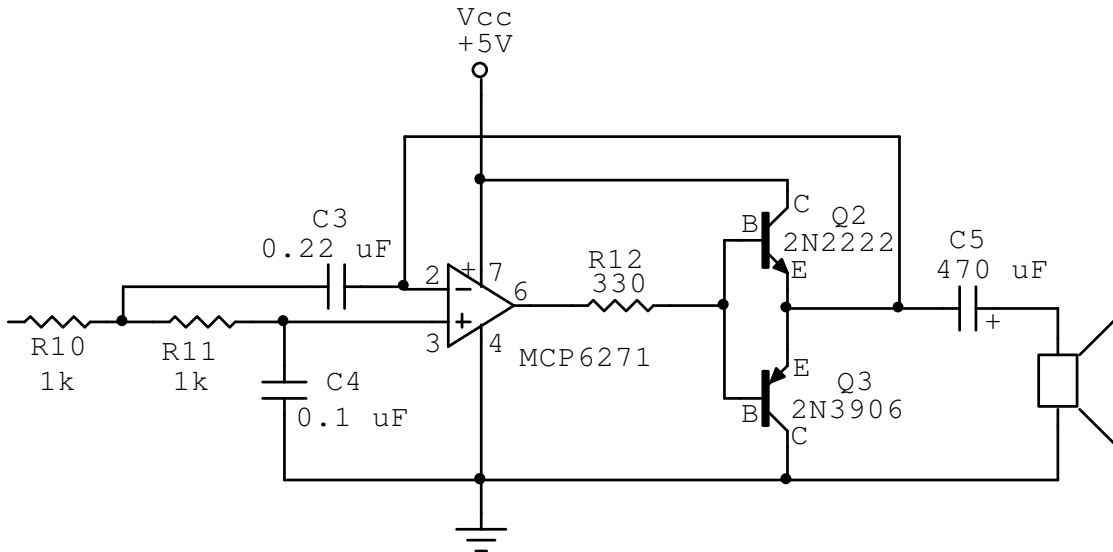
The 330 ohm resistor R12 limits the current demand out of the Butterworth filter opamp. The final output from the transistor emitter E can be connected to a low impedance load such as an 8-Ohm speaker or headphones to reproduce the sound picked up by the microphone in the first stage of the circuitry. This approach will work for small speakers or headphones, but the output power is still limited by the current limitation the small transistor. In addition the speaker is subjected to a constant bias voltage and current flow under quiescent conditions. The constant current may tend to heat the speaker voice coil.

You can use the transistor and this circuitry to power the 8-Ohm speaker provided in the lab (optional).

Once you hook up this simple power boost, listen to the sound the speaker is producing. How is the quality of the sound? What happens when you bring the speaker close to the microphone?

**Task 3-d:** (optional): real power boost stage

To really boost the power of the op-amp and eliminate the bias voltage on the speaker, an output stage can be added that consists of a pair of power transistors which can handle larger currents. Since the currents will be fluctuating in both positive and negative directions with respect to the 2.5V “quiescent” state of the circuit (no microphone signal), the output stage consists of an NPN (Q2) and a PNP (Q3) transistor that act as voltage followers:



Final filter section with high-power output stage added

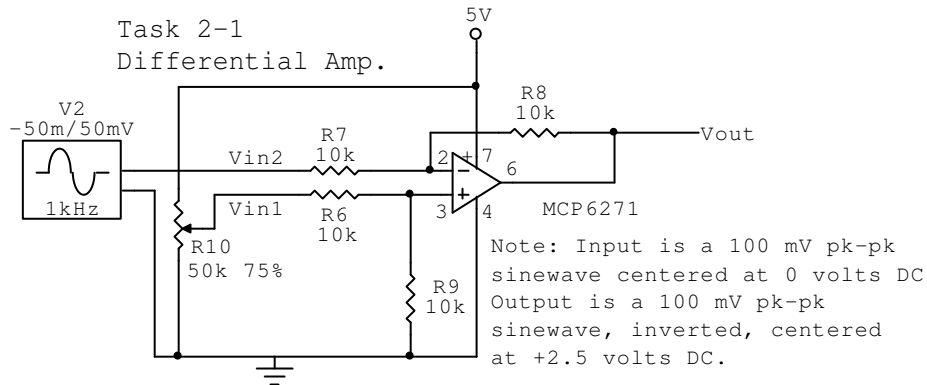
This output stage is quite simple and can deliver quite a bit of power into an 8-Ohm speaker. Notice the following:

- The feedback into the inverting (-) input of the opamp now comes from the transistor output stage instead of the output of the opamp. This is to compensate for the (non-linear) behavior of the NPN/PNP transistor pair.
- The bias voltage at the output the opamp is approximate 0.5V lower than the input bias voltage applied to the non-inverting (+) input of the op-amp. This is due to the fact that the NPN and PNP transistors with their internal diode structure always present a series voltage drop of roughly ~0.5 V.
- Since a bias voltage is present on Vin (and thus also at the output of the opamp), both NPN and PNP transistors always conduct current and no resistor/diode pairs are needed to keep the transistors from conducting current. Otherwise an additional double resistor and diode pair has to be added to each transistor base B.
- The large electrolytic capacitor C5 is used to filter the bias voltage. The capacitor C5 along with a low impedance load (such as an 8-Ohm speaker) creates a high pass filter that removes the bias voltage. A large capacitor is used to make the cut-off frequency as low as possible to handle the low impedance 8-Ohm load.

You can use this circuitry to power the 8-Ohm speakers provided in the lab (optional).

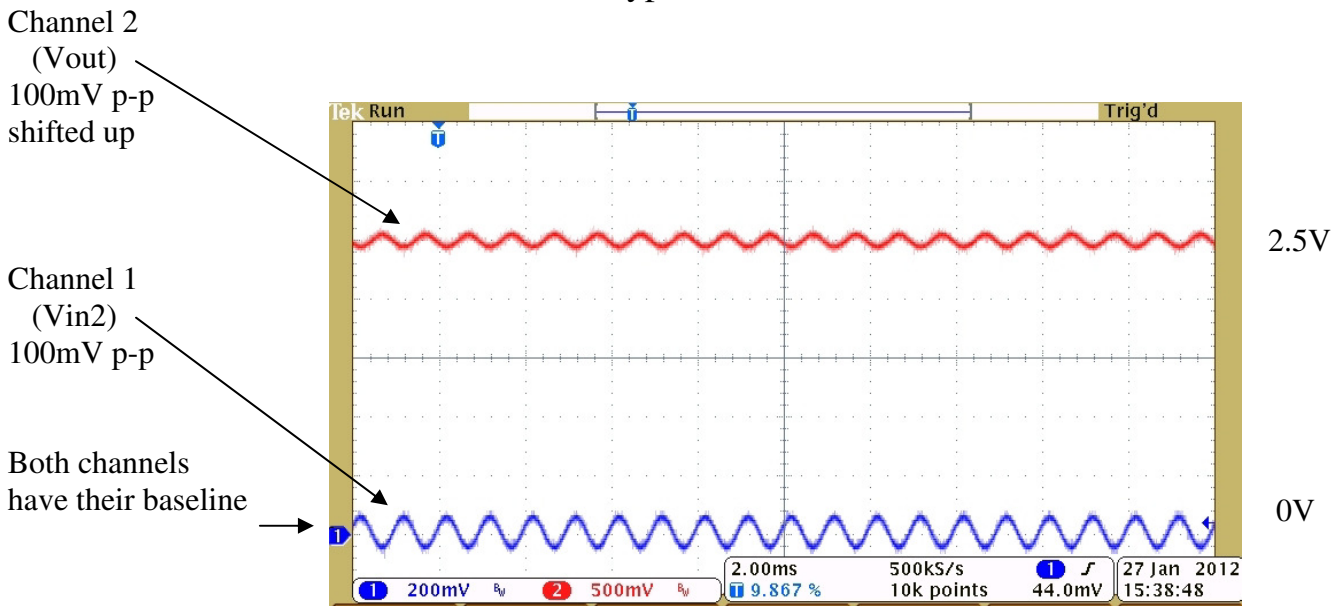
**Appendix**

Additional details on Task 2-1:



Scope setup: Horizontal timebase: 2 mS/div  
 Trigger : on channel 1  
 Trigger mode: Auto  
 Channel 1: Vin2 Scale 200 mV/div DC coupled  
 Channel 2: Vout Scale 500 mV/div DC coupled

Task 2-1 typical screenshot



**Notes:** This screen shot is what your Task 2-1 should look like. Observe that the circuit output (Vout on channel 2, 100 mV p-p) is the same as the Vin2 100 mV p-p input, except that it is riding at a level 2.5 volts above the input which is centered on the zero volt level. The 2.5 volt offset is adjusted by clipping the red multimeter lead on the Vin1 voltage, and the black multimeter lead on ground. Then adjust the potentiometer and watch the scope and multimeter until you get a 2.5V offset as shown. Notice that channels 1 and 2 have different vertical scales. This is required so you can get both channels on the scope screen at the same time.