S2014, BME 101L: Applied Circuits Lab 6
Low-power audio amplifier

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1 Design Goal

In this lab, you will design a low-power audio amplifier for an electret amplifier, using an op-amp chip and a dual-rail power supply supply. That is, we will have 3 power wires: ground (0v) and $\pm V_{pow}$.

We do not want to amplify any DC bias on the microphone, just the audio frequencies (in fact, just those greater than the resonant frequency of your loudspeaker).

For this lab, you will want to amplify sounds from a quiet room or normal conversation (30–60dBA) up to as loud as you can make the amplifier go without clipping or otherwise distorting the signal.

We would like the output to be centered at 0v when there is no sound coming into the microphone, so that the loudspeaker is not pushed away from its central position.

2 Background

2.1 DC bias

Based on the measurements you did in Lab 2 (electret microphone), you should be able to make a simple circuit that produces voltage fluctuations from an electret microphone by adding a pull-up resistor. The voltage fluctuations can be fairly high, but the changes in current through the microphone remains quite small. In other words, this circuit has a high *source impedance*, and so is not capable of delivering much power. In particular, it is not capable of driving a loudspeaker.

To get enough power to drive a loudspeaker, you'll need an amplifier that can provide substantial current and voltage. For this lab, you'll use an op amp chip (the MCP6004 quad op amp chip) to make the amplifier. Of course, the op amp chip is not designed for high power output, and so is still not really capable of driving a loudspeaker very loudly (we'll do a more powerful amplifier in a later lab).

Amplifying the DC bias on the microphone can result in the op amp output saturating (getting stuck at either the highest or lowest voltage). To get rid of any DC bias, you can use a simple high-pass filter between the microphone circuit and the input to the amplifier. This can be as simple as a capacitor and resistor, acting as a voltage divider. The impedances have the same magnitude when $\left|\frac{1}{j\omega C}\right| = R$, that is, when $\omega = \frac{1}{RC}$, where $\omega = 2\pi f$. If you want frequencies greater than f to be passed with little change in amplitude, then you want $RC > \frac{1}{2\pi f}$.

When there is no sound, the input of the amplifier should be in the middle of the voltage range (0v). Think about ways you can accomplish this. Remember that a capacitor is an open circuit for DC.

It is not good to drive loudspeakers with a DC signal (that would push the cone away from the center-rest position), so you'll want the output to be centered in the middle of the output range, that is, at 0v when the input is at 0v.

2.2 Loudness

The loudness of sound is usually expressed in *decibels*, which are logarithms of ratios:

$$D = 20\log_{10}\left(\frac{A}{A_{ref}}\right) \ ,$$

where D is the loudness in decibels, A is the amplitude of the signal, A_{ref} is the amplitude of the reference being compared to.

It is common to report sound pressure level (SPL) with a reference sound pressure of 20μPa RMS, which is about the threshold of hearing for young adults at 1kHz.

Human hearing is not uniform across all frequencies, and so standard weighting curves have been created to measure sound pressure level so that 0dB roughly corresponds to the threshold of hearing across the full range of human hearing. The most common weighting scheme is A-weighting which peaks around 2–3kHz, is 0 at 1kHz and drops 10dB by 200Hz. That means that to get the same dBA value a signal must have $10^(10/20)$ more amplitude at 200 Hz than at 1kHz. See `http://en.wikipedia.org/wiki/A-weighting` for plots of the A-weighting scheme. See `http://en.wikipedia.org/wiki/Sound_pressure` for a table of common sound levels.

Note that 40dBA means $10^{40/20} 20\mu\text{Pa} = 2\text{mPa}$ RMS sound pressure around 1kHz, but around

$$10^{(40+10)/20} 20\mu\text{Pa} = 6.3\text{mPa}$$

at 200 Hz, where human hearing is 10dB less sensitive (according to the A-weighting scale).

The frequency range of interest is from just above the resonant frequency of your loudspeaker (determined in Lab 5) up to the limits of human hearing (around 15kHz).

You will need to compute the range of RMS pressure you expect as input. What is the largest RMS pressure you expect to need to handle?

2.3 Gain

An op amp provides a very large open-loop gain (25,000 to 400,000 for the chip you are using), and so is almost always used in a feedback loop to set a reasonable gain. You will need to compute the gain you need and select the feedback circuitry appropriately.

To compute the voltage gain, you need to determine the voltage range expected at the input to the amplifier and the voltage desired at the output of the amplifier.

Determining the input voltage swing, we need to know the loudness of the input sound, the sensitivity of the microphone, and the voltage/current gain (impedance) of the current-to-voltage converter. We talked about the loudness of the input in Section 2.2, and you can find the sensitivity of the microphone on the data sheet.

The data sheet presents the sensitivity in a rather confusing way, though, because it does not give it in μA/Pa or some other comprehensible unit. Instead it gives it in dB with a reference of 1V/Pa. This measurement is made with a 2.2kΩ load resistor (the output impedance of the specs), so 0dB corresponds to a current of 454.5μA/Pa. The -44dB spec with a 3v power supply, means that the current is $10^{-44/20} 454.5\mu A/\text{Pa} = 2.87\mu A/\text{Pa}$. Reducing the voltage from 3v to 2v reduces the sensitivity by 3dB (a factor of 0.7). Note that the sensitivity is not very precisely specified: $\pm$2dB is a factor of 0.79 to 1.26.

The output of the amplifier is going to be limited both by the supply voltage and and by the current output limits of the op amp.

Look up the power-supply voltage limits for the MCP6004 op-amp chip. You want to make sure that your $V_{pow} - (-V_{pow}) = 2V_{pow}$ power-supply voltage stays well below the $V_{DD} - V_{SS}$ limits of the chip—it is best to stay at least 10–15% below the absolute max ratings.

One you have determined the power supply voltage, you can choose the size of your pull-up resistor to get a reasonable DC bias on your microphone. You would like the output impedance to be fairly high (to get large voltage swings for small changes in current), but you want to make sure that current fluctuations you expect do not produce such large voltage swings that the microphone bias moves out of the saturation region of the curve.

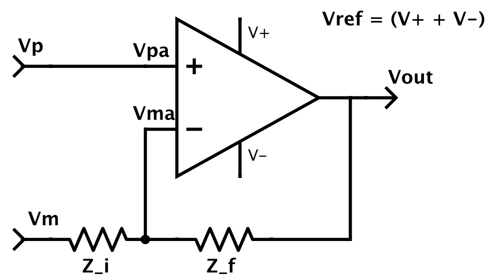

Figure 1: Generic negative feedback amplifier using an op-amp. The same circuit is used for positive gain, negative gain, low-pass active filters, high-pass active filters, and (with non-linear devices replacing the impedances) logarithmic amplifiers.

Look up the maximum output current (the output short-circuit current) of the MCP6004 op amp. If your loudspeaker is about an 8Ω load, which is going to limit the power to your loudspeaker more: the voltage limits of the op-amp power supply or the current limit of the op amp?

2.4 Negative-feedback amplifier

Figure 1 shows the standard negative-feedback circuit used for most op-amp amplifier circuits. We'll look at the general formula for the output voltage of the amplifier, then some special cases that we'll use in this lab.

The first thing to realize is that the op amp output is between the power rails V_+ and V_-, and that the mid-point between the power rails, $V_{ref} = (V_+ + V_-)/2$, is an important point of symmetry.

The output of the amplifier can be expressed in terms of the inputs to the op amp:

$$V_{out} - V_{ref} = A(V_{pa} - V_{ma}) \ ,$$

where A is a large number (the *open-loop gain* of the op amp). The op amp inputs are related to the external inputs:

$$\begin{aligned} V_{pa} &= V_p \\ V_{ma} &= \frac{Z_i V_{out} + Z_f V_m}{Z_i + Z_f} \end{aligned}$$

The formula for V_{ma} is just another way of writing the voltage-divider formula.

We can put the equations together to get the general form

$$V_{out} - V_{ref} = A\left(V_p - \frac{Z_f V_m + Z_i V_{out}}{Z_i + Z_f}\right) \ .$$

We can rearrange this to

$$V_{out}\left(1 + A\frac{Z_i}{Z_i + Z_f}\right) = V_{ref} + A\left(V_p - \frac{Z_f V_m}{Z_i + Z_f}\right) \ ,$$

or (multiplying both sides by $Z_i + Z_f$)

$$V_{out}(Z_i + Z_f + AZ_i) = (V_{ref} + AV_p)(Z_i + Z_f) - AZ_f V_m \ ,$$

and

$$V_{out} = \frac{(V_{ref} + AV_p)(Z_i + Z_f) - AZ_f V_m}{Z_i + Z_f + AZ_i} \ .$$

That looks like an awful mess, but see what happens as $A \to \infty$. The terms not multiplied by A become insignificant, and we get

$$V_{out} \to \frac{V_p(Z_i + Z_f) - Z_f V_m}{Z_i} = V_p + (V_p - V_m)\frac{Z_f}{Z_i} \ .$$

Let's look at some special cases.

- If we have symmetric power supplies, $V_- = -V_+$, then $V_{ref} = 0$, which simplifies the math. This is the usual condition for beginning op amp circuits, and we'll use it in this lab. (In all future labs, we'll be doing single-power-supply op amp circuits, in which $V_- = 0$.) Note that for very large A, V_{ref} doesn't really matter anyway.

- If $V_m = V_{ref} = 0$, we have $V_{out} = V_p\left(1 + \frac{Z_f}{Z_i}\right)$. This is the standard setup for a *positive-gain* amplifier.

- If $V_p = V_{ref} = 0$, we have $V_{out} = -V_m \frac{Z_f}{Z_i}$, which is the standard setup for a *negative-gain* amplifier.

Note that we've used complex impedances (rather than simple resistances) for all the calculations, so we can make the gain vary as a function of frequency, by having Z_i, Z_f, or both vary with frequency (for example, by using capacitors).

For this lab, you will mainly need a positive-gain amplifier, with gain independent of frequency, as figured out in Section 2.3.

There is a rule of thumb that you can use to help figure out how a negative-feedback amplifier will behave: as long as the open-loop gain A is high enough and the output voltage doesn't hit the power-supply limits, the two inputs to the op-amp will be held by the amplifier to be the same: $V_{pa} = V_{ma}$. The voltage-divider formula for V_{ma} then tells you what what is going on. For example, for a positive-gain amplifier, we have $V_{pa} = V_{ma} = V_{out}\frac{Z_i}{Z_i + Z_f}$ or $V_{out} = V_p\left(1 + \frac{Z_f}{Z_i}\right)$.

3 Pre-lab assignment

Read the Wikipedia articles about sound-pressure level and A-weighting mentions in Section 2.2, plus ones about op amps:
`http://en.wikipedia.org/wiki/Operational_amplifiers`
`http://en.wikipedia.org/wiki/Operational_amplifier_applications`

Reread the data sheet for the microphone.

Make a block diagram of your circuit showing the major sections of the circuit (for example, the microphone, the pull-up resistor for the mic, DC-blocking RC input filter, the op amp, the feedback for the op amp, and the loudspeaker).

Choose a power-supply voltage for your system.

Determine the largest sound pressure level you are likely to want to amplify, and the corresponding current changes you expect from the microphone.

Design your microphone current-to-voltage converter. Using the current vs. voltage measurements from Lab 2 (electret microphone), determine the resistor size needed to get a DC voltage across the microphone sufficient to get a DC bias well into the saturation region. Note that you can be connecting to V_{pow}, ground, or $-V_{pow}$. Choose a pull-up resistor that will give you a large voltage swing, but one that should not cause clipping. Use that resistance and the expected current swings to compute the expected input voltage swing from the microphone.

Design any level-shifting (high-pass) filter you need between the microphone and the op amp. You probably want to make the corner frequency of this filter just a little above the resonant frequency of your loudspeaker.

Given the expected actual voltage limits of the signal to the loudspeaker (based on the power supply or the current limitation, which ever is more limiting), and the expected input voltage swing, compute how much voltage gain you need from your op amp amplifier.

Design the feedback circuit for the op amp to get the desired gain.

Design your circuit using one or two op amps, and draw a schematic for the whole circuit. Put boxes around parts of the schematic to show the relationship to the block diagram. Put pin numbers on all the op amp connections, to speed wiring.

You need to have a block diagram and circuit diagram before coming to lab, though it need not be the one that you end up with after debugging your design.

Note that to test the design, you must have some expectations about what you should see on the oscilloscope or with the multimeter at each stage. It would help a lot if you wrote some of those expectations down before the lab time (like what DC voltage you expect to see at the microphone, and how large an AC signal you expect to see added to that). What about the DC voltage and AC signal after the input high-pass filter? At the output of the op amp?

4 Procedures

Wire up your circuit one block of the block diagram at a time, starting with the microphone and pull-up resistor. After each block is wired up, test it with the oscilloscope. If it does not behave as you expect, change the design (either of the part already wired or of the next block it connects to).

When the whole amplifier is working, measure the voltage gain of the amplifier by measuring the AC input voltage and AC output voltage of the amplifier. Do these measurements with and without the loudspeaker connected.

5 Demo and writeup

Demonstrate to the instructor a working amplifier. Show the input and output of the amplifier as two traces on a dual-trace oscilloscope.

Demonstrate the amplifier working by speaking into the microphone. (You may also be able to get some feedback squeal, but don't irritate people too much by doing it a lot.)

Measure the voltage gain of the amplifier at various input frequencies, using either the multimeters or the oscilloscope to simultaneously measure voltage at the mic and voltage at the loudspeaker. Make sure you measure frequencies from below the corner frequency of your level-shifting filter up to high frequencies (say 10kHz). You can excite the microphone by playing sine waves from the frequency generator through another loudspeaker.

6 Bonus

If you finish the amplifier early, you can try adding the following features:

- a volume control, to adjust the gain of the amplifier.

- a tone control, to boost or cut the bass or treble. See, for example, the Baxandall tone control circuit at `http://www.learnabout-electronics.org/Amplifiers/amplifiers42.php`.