For this lab, bring a music playing device if you happen to have one. Anything with an 1/8th inch out should work.

Active Filters

In this lab, you build a circuit that takes a song playing from an audio source (laptop, mp3 player, etc.), isolates its bass drum, and blinks an LED in response to each drumbeat.

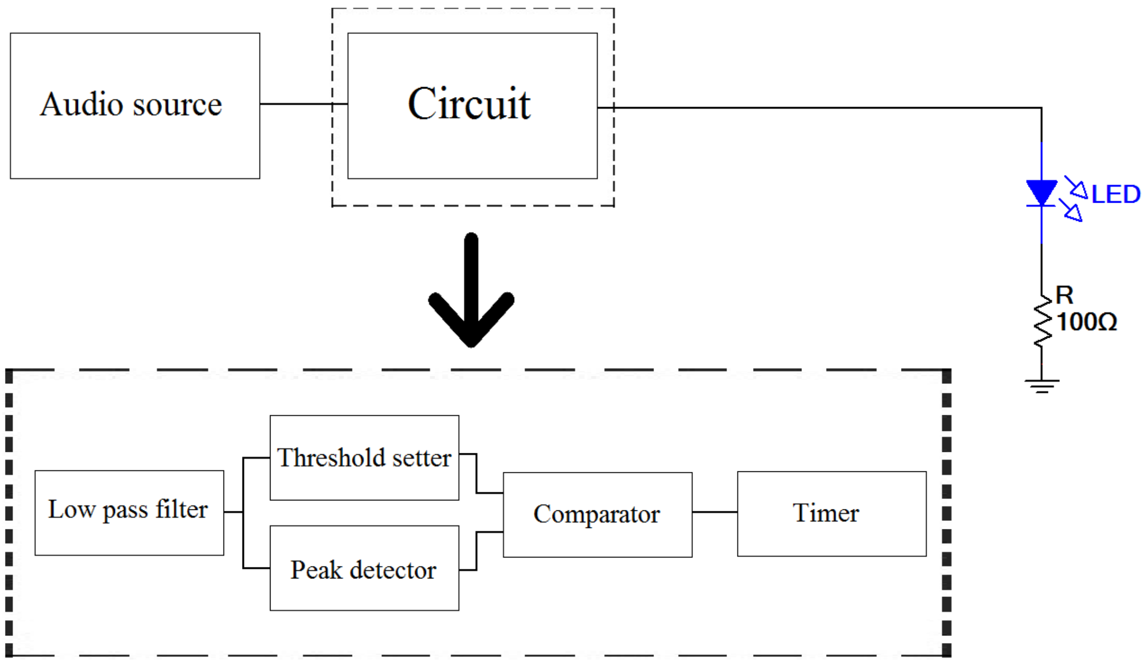


Figure 1 Block diagram of the circuit.

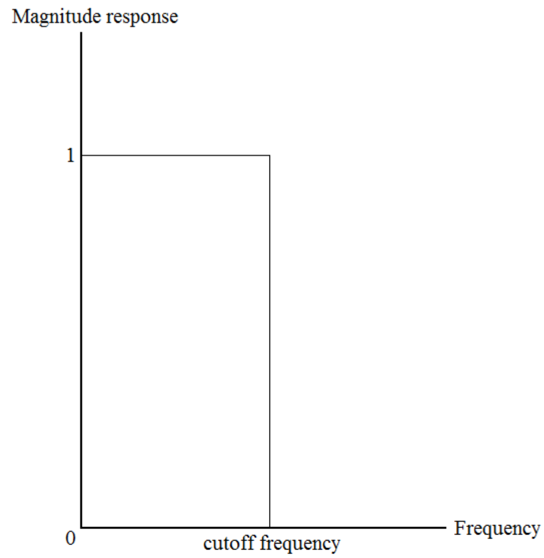
Play a song from your audio source and probe the signal with your oscilloscope. Adjust the horizontal and vertical scales so you can see several seconds of the signal at once.

The energy of a bass drum lies mainly in frequencies under 200Hz. To isolate the bass drum, we need to eliminate the song's high frequencies with a low pass filter. The ideal low pass filter has the magnitude response shown in Figure 2(a), where all frequencies below the cutoff are passed through untouched, and all frequencies above the cutoff are blocked completely. This magnitude response is impossible in reality, but you will implement one of the filters shown in Figure 2(b) as an approximation.

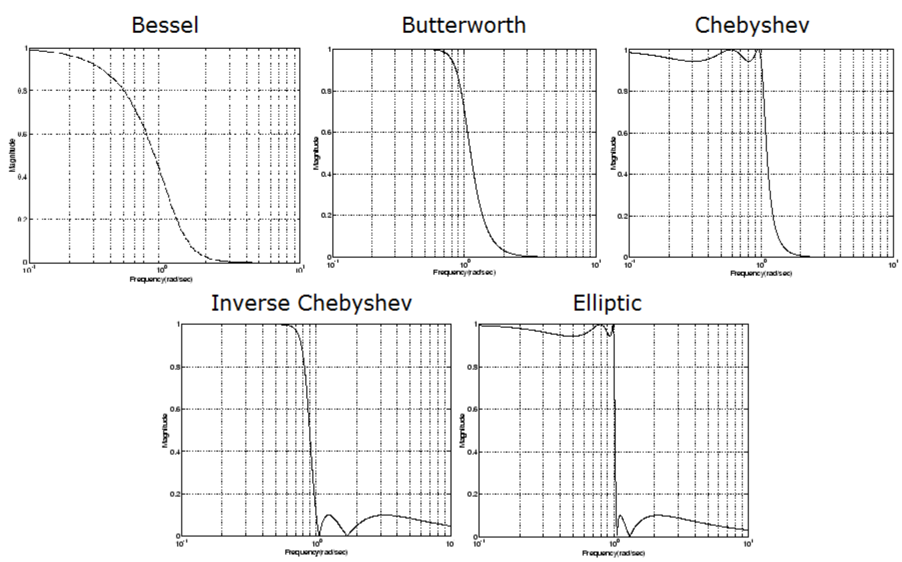
When you design your low pass filter, you will be faced with many important choices. As an engineer, you must guide your decisions by considering the cost and effectiveness of each implementation:

1. Effectiveness. The filter must attenuate the high frequencies so the LED blinks only in response to the bass drum, and not any other instruments.

2. Cost. The filter must be as simple as possible (but still be effective). We *could* use a monstrous 10^{th} order filter, but that is far more complicated than necessary. In the real world, higher complexity translates to higher cost.



(a)



(b)

Figure 2 Magnitude responses of different filters. (a) Ideal filter. (b) Real filters.

You will now design a low pass filter that satisfies both of these design objectives. To help you, we have coded a LabVIEW virtual instrument (VI) that simulates your circuit and allows you to experiment with different filter topologies.

Open the VI and upload your WAV file (or a WAV file that we have provided) under "path" in the front panel. Run the VI. In the waveform graph on the front panel, you can see the original song's signal at the top, and at the bottom, a slew of signals superimposed over each other. These are the signals you will see at various stages in your circuit.

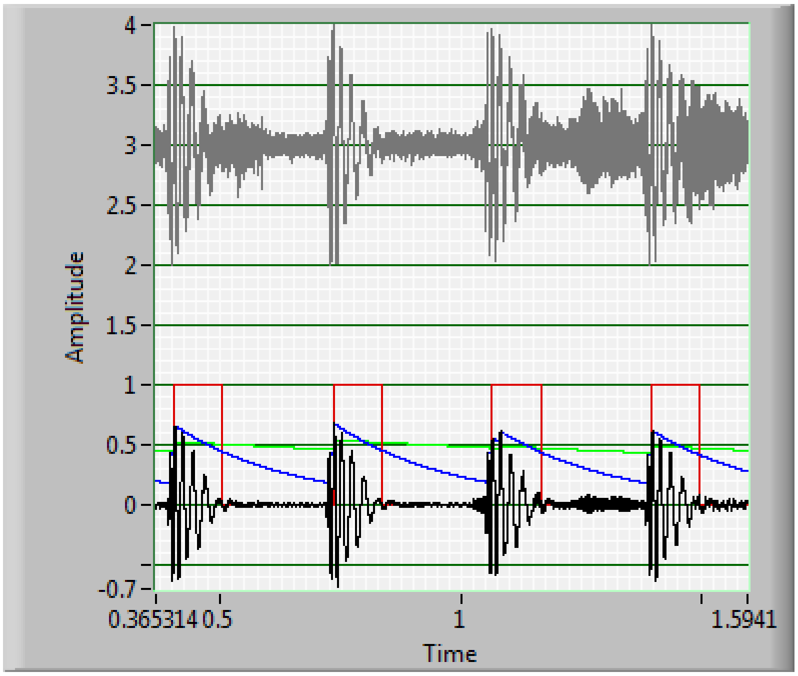


Figure 3 Waveform graph in LabVIEW. The gray signal is the original song. The black signal is the filtered song. The blue signal is the output of the peak detector. The light green signal is the threshold signal. The red signal is the final output of the entire circuit; it is the signal fed to the LED.

As the VI runs, tweak the filter parameters on the front panel – topology, order, cutoff frequency, stopband attenuation, and ripple – and the VI will simulate the new filter.

Your concern right now is the red signal. It is the final output of your circuit. When it is high, the LED is on. When it is low, the LED is off. You want to have it be high for each drumbeat and low otherwise. Below, you will find useful information about the different filter topologies.

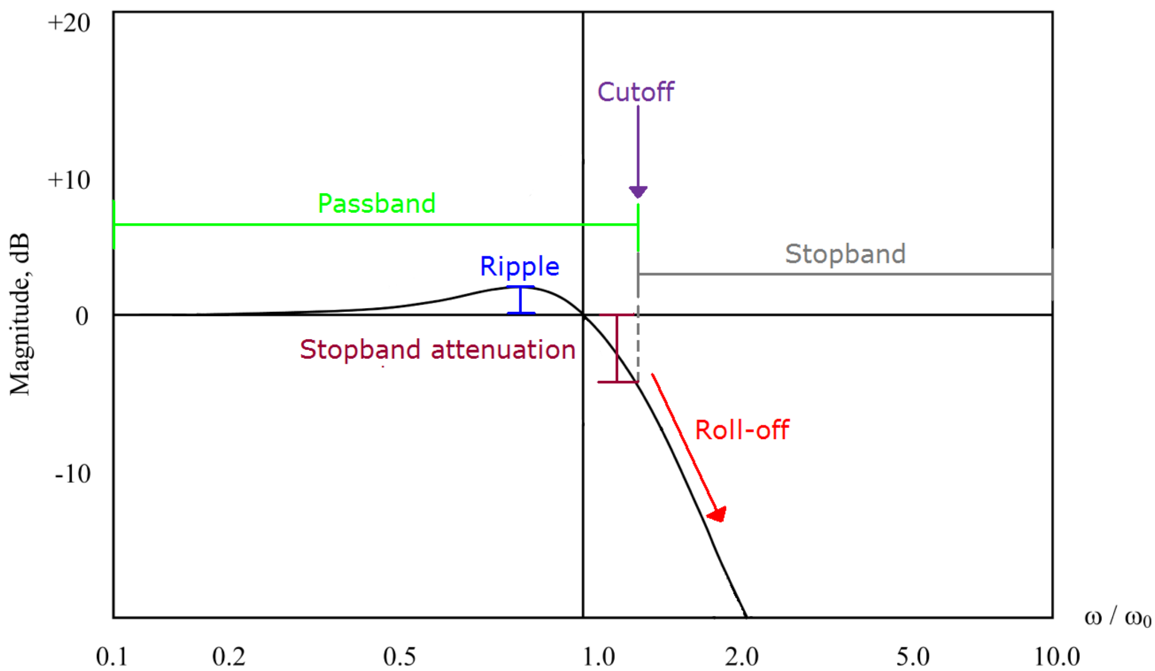


Figure 4 Example filter magnitude response and general terminology. The passband and stopband are the ranges of frequency that are nominally passed through and blocked by the filter. The cutoff frequency separates them. Stopband attenuation is the minimum attenuation of all stopband frequencies. Roll-off is how quickly stopband frequencies are attenuated as frequency increases. Ripple is how much the magnitude response ripples in passband or stopband.

1. **Bessel.** The Bessel filter has the slowest roll-off of all 5 filters. Its chief advantage is its maximally flat group delay in the passband. Group delay is simply the derivative of the phase response (both are plotted vs. frequency). It is a measure of the time it takes for a signal to pass through a filter. Group delay increases with increasing filter order and decreasing cutoff frequency, which is bad. We do not want the LED to blink AFTER we hear the beat; we want the blink and the beat to coincide or be so close together we cannot perceive the delay. The Bessel's flat group delay means all frequencies in the passband are delayed the same amount, which means there is no signal distortion at the output due to different frequencies being delayed different amounts. The filter parameters stopband attenuation and ripple have no effect on this filter.
2. **Butterworth.** The Butterworth filter is the easiest of the 5 topologies to implement in analog circuits. Its magnitude response is characterized by a maximally flat passband, -3dB attenuation at the cutoff frequency, and a roll-off of –20N dB/decade beyond cutoff, where N is the order. This roll-off is the slowest, with the exception of the Bessel filter. Stopband attenuation and ripple have no effect on the Butterworth.
3. **Chebyshev/Inverse Chebyshev.** These filters roll off more steeply than the Butterworth, but exhibit ripple in passband (Chebyshev) or stopband (Inverse).

The value of stopband attenuation has no effect on the Chebyshev filter; ripple has no effect on the Inverse Chebyshev.

4. **Elliptic.** The elliptic filter rolls off more steeply than all other filters, but exhibits ripple in both passband and stopband.

Write down your choice of filter, and explain your choice.

It turns out that a first order Butterworth is effective enough for our circuit. A first order Butterworth also happens to be the very first filter you studied, the RC low pass filter. Because we want you to learn something new, however, you will instead be using the Sallen-Key topology. (Later, if you wish to test the Butterworth, replace the Sallen-Key filter with the RC filter followed by a voltage buffer. It should work as well as the Sallen-Key filter.)

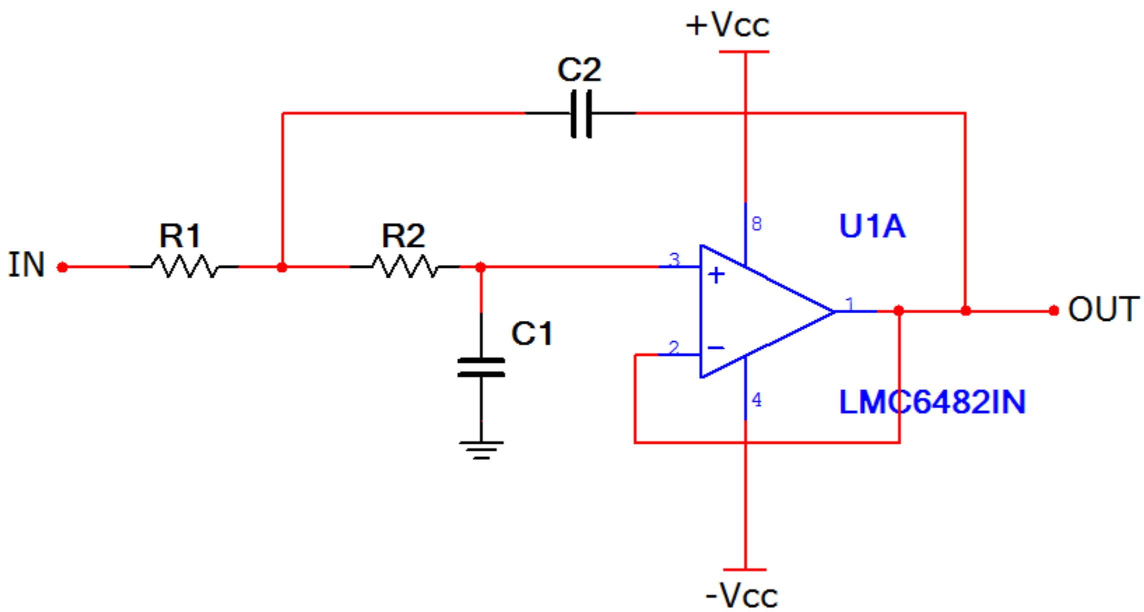


Figure 5 Sallen-Key low pass filter schematic.

The Sallen-Key low pass filter is a second-order active filter that rolls off at –40dB/decade after the cutoff frequency (this is just a fancy way of saying that the magnitude of the frequency response decreases with ω^2, we'll talk more about this on Wednesday). As mentioned in class, we call this an active filter, because it is a filter that uses at least 1 active component. Shown in Figure 5, the Sallen-Key's active component is an operational amplifier. The operational amplifier acts as a buffer between the Sallen-Key filter and the previous and following stages, thus making the Sallen-Key's frequency response independent of the rest of the circuit. This is one advantage of an active filter over a passive filter, a filter that employs only passive elements like resistors, capacitors, and inductors.

Prelab #1: Derive the transfer function of Figure 5. Summing point constraint does work, so that will save you a lot of effort. Your first step should be to replace the capacitors with their equivalent impedance $Z_c = \frac{1}{jwC}$. Show that the transfer function can be written as:

$$H(j\omega) = \frac{1}{1 + j\omega C_1 R_1 + j\omega C_1 R_2 - C_1 C_2 R_1 R_2 \omega^2}$$

If you get stuck on the algebra, just write the node voltage equations and that will be good enough.

Prelab #2: The above expression is correct, but not necessarily the most useful expression. Show that this expression can be rewritten in terms of the cutoff frequency $\omega_c = \frac{1}{\sqrt{R_1 R_2 C_1 C_2}}$ as:

$$H(j\omega) = \frac{\omega_c^2}{\omega_c^2 + j\omega C_1 R_1 \omega_c^2 + j\omega C_1 R_2 \omega_c^2 - \omega^2}$$

$$\omega_c = 1/\sqrt{R_1 R_2 C_1 C_2}$$

Note that w_c is not a property of the input signal, but is instead a property of our system and is based on R and C values.

Prelab #3:

It is possible to show that the magnitude response of this filter is $|H(j\omega)| = \frac{\omega_c^2}{\sqrt{\omega^4 + \omega_c^4 - k\omega^2}}$, where k is a constant which doesn't matter. What would be the ratio of the amplitude of V_{out} to the amplitude of V_{in} for sinusoidal input signal with frequency much smaller than ω_c?

Prelab #4

What would be the ratio of the amplitude of V_{out} to the amplitude of V_{in} for sinusoidal input signal with frequency exactly 100 times ω_c? (Assume the $k\omega^2$ term is dominated by ω^4). Give a numerical answer.

Prelab #5: Explain in words what you'd expect to hear if you used a song as the input to your filter, and connected a pair of headphones to the output with $w_c = 1256\ rad/sec$ or equivalently $f_c = 200\ Hz$?

Prelab #6:

For simplicity, let R1=R2=R and C1=C2=C. Choose values for R and C to give you your cutoff frequency (keep in mind ω_c is the cutoff frequency in rad/sec, and the frequency target given earlier in this lab is in Hz). Write down your values of R, and C.

Build your filter, setting positive and negative supply to ±5V. Wire your audio source to its input, and put probes at the input and output of the filter. Compare the two signals. Depending on your song, your oscilloscope may show that the filtered signal is very small, with its maximum oscillation above or below DC level somewhere around 20 to

40mV. The next stage includes a diode, which will basically be an open circuit to such a small signal. You need to amplify your signal's maximum oscillation to a minimum of 300mV. Use an inverting amplifier.

Now, build a simple voltage buffer (completely separate from your circuit), and wire the audio source to this buffer too. Put a speaker at the buffer output.

Place an LED in series with a 100Ω from the output of the inverting amplifier to ground. Play your song. With the speaker and the LED, you can listen to the song and manually verify that the LED blinks on the drumbeats.

You may also connect a set of headphones to the circuit if you'd like to hear the filtered signal.

You should see the LED blink with the bass drum, but also that the blinks flicker rapidly, which is really unattractive. This is because the filtered signal wiggles rapidly between positive and negative voltages. You will fix this in the next section.

Optional

Replace one of your resistors with a potentiometer and observe how the sound signal changes as your vary the resistance.

The peak detector

The peak detector will steady the flicker. It smoothes out the input signal by "tracing" along the signal's peaks.

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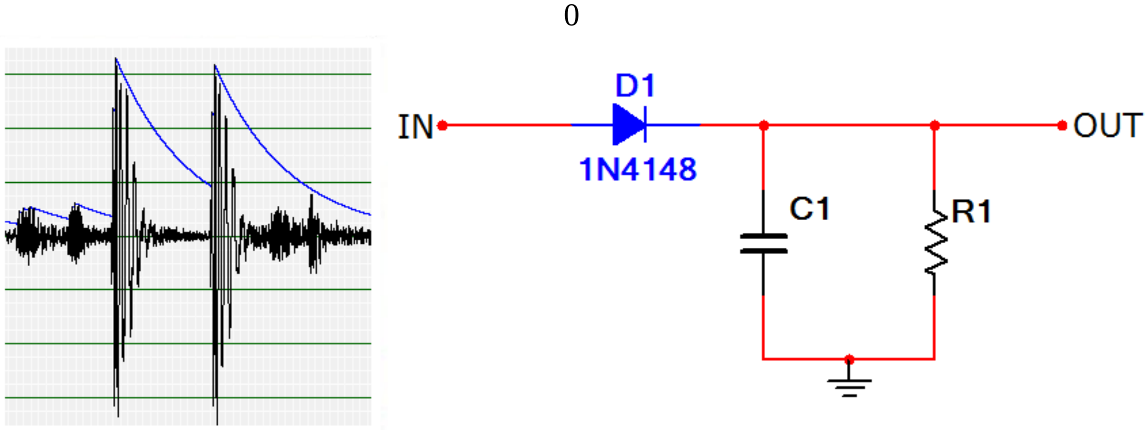


(a) (b)

Figure 6 The peak detector. (a) The black waveform is the input to the peak detector. The blue waveform is the output of the peak detector. (b) Peak detector circuit.

The input voltage (IN) is the filtered song signal, and the output voltage (OUT) is the voltage on the capacitor, C1. We want the peak detector to capture only the peak of a drumbeat, and none of its wiggle. To do so, we need a rectifier, a device that converts an AC signal to a DC signal. A diode can be used as a rectifier. Crudely speaking, a diode is a wire when IN>OUT, and an open circuit when IN<OUT.

When IN>OUT, the diode shorts and OUT is charged to IN, as you know from your study of capacitors. When IN<OUT, the diode opens and OUT decays exponentially, discharging through R with its rate of decay characterized by the time constant τ_{pd} = R1*C1. The larger the resistance and capacitance, the slower the capacitor discharges OUT.

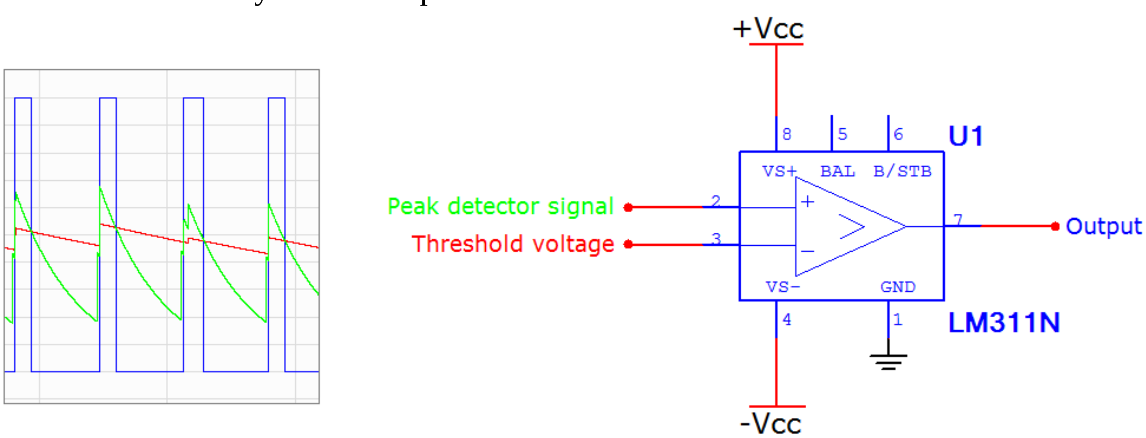
Given the filtered song signal, OUT will quickly trace the initial rise of voltage corresponding to the sharp strike of the bass drum, and then decay as the drumbeat softens and dies, until the next drumbeat occurs, giving the blue waveform in Fig 6(a).

Build and connect a peak detector to the filtered signal. Considering we want τ_{pd} large enough to ensure a smooth output but small enough to capture rapid successive drumbeats, choose a suitable τ_{pd} and pick values for R1 and C1 to give you your τ_{pd}. Write down your choices.

Place a voltage buffer in series with the peak detector, and wire an LED and a 100Ω resistor to the voltage buffer output. Play your song. The blinking LED should not flicker. However, the blinks are soft; they fade away rather than cut off abruptly, which is due to the decaying voltages in the peak detector output.

The comparator

To sharpen the blinks, we need to convert the peak detector signal into a digital signal that alternates between 2 voltages, positive supply on a drumbeat and ground otherwise. We can do this easily with a comparator.



(a) (b)

Figure 7 The comparator. (a) The green signal is the peak detector signal. The red signal is the threshold voltage. They are the inputs to the comparator. The blue signal is the output. (b) Schematic of the LM311N comparator with pins numbered.

The comparator outputs positive supply (+Vcc) when the peak detector signal is greater than the threshold voltage, and ground (0V) otherwise, yielding the blue waveform in Figure 7(a).

We can set the threshold voltage by stepping down the positive supply with a voltage divider. Pick 2 resistors to give a threshold voltage a little under the peaks in your peak detector signal. For example, if your peaks are 3V, and positive supply is 5V, pick 2 1kΩ resistors. Wire them in series from positive supply to ground, then wire the node between the two resistors to pin 3 of the LM311N. The threshold voltage is 2.5V, which is just under the 3V peaks. Put an LED in series with a 100Ω resistor from the comparator output to ground. Your LED should now emit sharp blinks.

This voltage divider limits our circuit's region of operation because it fixes the threshold voltage with no possibility of adjustment. If, say, we turn the volume up, the threshold will be too low, and the LED will be on all the time. Conversely, if we turn the volume down, the threshold will be too high, and the LED will never go on. Even a potentiometer in place of one of the fixed resistors is only a marginal improvement. What we want is a self-adjusting threshold voltage. We can produce this signal with another peak detector.

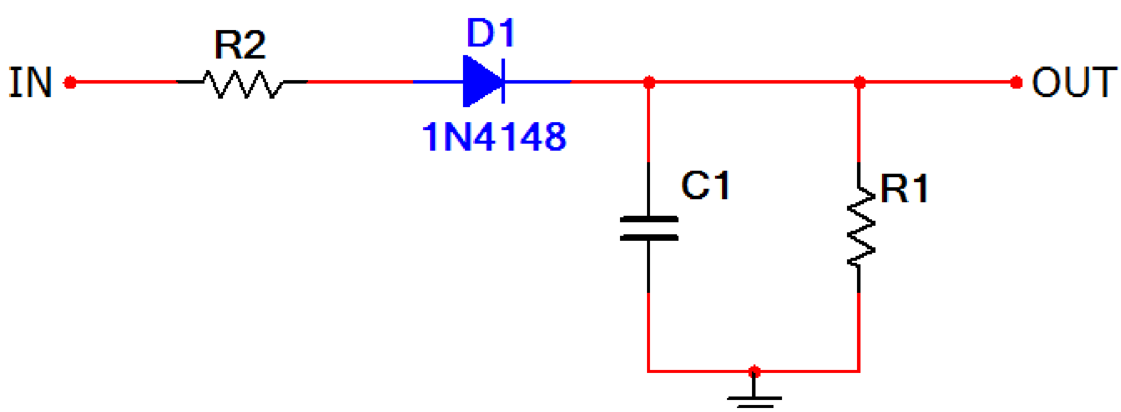


Figure 8 The threshold setter.

IN is the filtered song like in the first peak detector, so this peak detector (from now on, referred to as the threshold setter) mirrors the operation of the first. However, there are 2 important differences.

1. There is now a resistor R2 in series with the diode. Recall that the time it takes for capacitor C1 to charge OUT to IN (for IN>OUT) is characterized by $\tau = R2*C1$; R2 slows C1's charging speed, so the threshold signal's peaks will be smaller than the peak detector's. R2 will be small, around 300Ω or so.

2. τ_{th} = R1*C1 controls the rate of decay of OUT (for IN<OUT) just like in the peak detector. Since we want a relatively constant threshold voltage, pick R1 so $\tau_{th} >> \tau_{pd}$, e.g. $\tau_{th} = 10\tau_{pd}$.

If you're having trouble picking a value of R2 that works, put probes at the outputs of the peak detector and threshold setter, and tweak R2 (a potentiometer could be useful here) until you get the threshold signal you want. Write down your values of τ_{th}, R1, R2, and C1.

Now you have a self-adjusting threshold voltage: if the amplitude of the filtered signal increases due to increased volume, your threshold voltage will increase too because OUT tracks IN. Wire your threshold voltage to the comparator, and play your song at different volumes (too quiet though, and your circuit will not work) to verify that your circuit is working correctly.

The timer

The final step is to set the length of each blink. Without the timer, the length of a blink is how long the peak from the peak detector signal stays above the threshold voltage. This time differs from beat to beat. We want to make this time the same for all beats. We can do this with the monostable timer circuit from Lab 4.

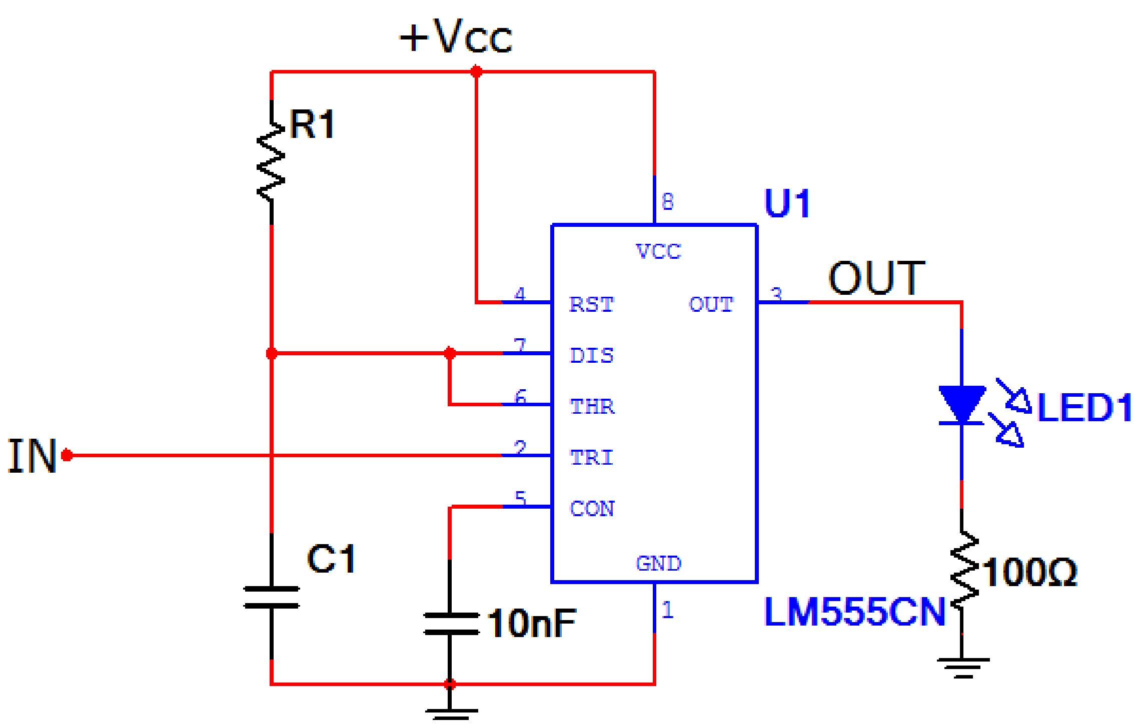


Figure 9 The monostable timer circuit.

Recall that OUT is +Vcc when IN<Vcc/3, i.e. OUT goes high when IN goes low. OUT remains high until C1, initially with zero voltage across its terminals, charges to (2/3)*Vcc. The time it takes for C1 to charge to (2/3)*Vcc is t_{pulse} = ln(3)*R1*C1. Pick values for t_{pulse}, R1, and C1, and write them down.

The comparator signal goes high on the beat, but we need a trigger signal that goes low on the beat. How can we produce this trigger signal from the comparator output? Simply switch the two comparator inputs.

Wire the LED and the 100Ω resistor to the output of the timer and play your song.