Circuits Project Manual:
Frequency Detection Security System

Hyunjun Park, Andy Ma, Paul Nieves

Problem Introduction

Our world is becoming more digitized than ever. With the advent of the information age, there is greater risk of cyber attacks including data breaches. Nearly all sensitive personal information is stored somewhere online or on our computers while companies like Meta are creating virtual realms. This begs the question of how can the identities of people be verified in an increasingly digitized world? Biometric identification can be particularly robust in specific instances[1]. because of its inability to be shared, and its difficulty to be replicated. Among them, the human voice is a unique identifier. It is well known that all sounds can be decomposed into sine waves using the fourier transform. It can be useful to demonstrate how a simplified version of the voice activation security system works. Specifically, simplified from voice detection to a sequence of single frequency detection. Our team aims to demonstrate the principles used in a voice activated security system.

Our project initially was to design a melody detector with multiple notes but we realized it was much harder to do without software components. So, In order to demonstrate the principles used in a voice activated security system and to address the larger goal of securing the metaverse, our goal is to create a single frequency detector.

Description

A general overview of our project is a frequency detector. The lock only opens when the correct frequency is heard after a certain delay.

To accomplish this, the workflow in figure 1 (shown below) was used. The input of the circuit is a sound wave going into an electret microphone which is then amplified and then filtered to the right frequency. This signal is then sent into a comparator. If this final signal is a significant value, the servo is prompted to turn, indicating that the lock has been unlocked.

Overview

The input sound signal is received by the electret microphone and transformed into a voltage. This signal is first amplified and then gets filtered with a bandpass. We are trying to detect a certain frequency, so any other frequency will not trigger the circuit. After the input signal is filtered, we need to determine if this signal is strong enough to classify as a correct input. Because sound is a sinusoidal input, we need to translate this into a DC signal for the rest of the decisions about its amplitude. A half-bridge rectifier is used to turn the AC signal to a DC voltage. Then this DC voltage will be processed with a comparator op amp which will turn on an LED to signal the unlocked state.

Milestone 1
For the first milestone, the amplifier and volume detection components of the circuit were designed and tested. To test this milestone, the input signal from an electret microphone is transmitted to an operational amplifier. This input is amplified to a certain voltage. This voltage is received by the comparator operational amplifier which will output 5V when it's above the threshold voltage and output 0V when the input voltage is below the threshold.

Milestone 2 - The second milestone is centered around designing the Timer. To test this milestone, we use a monostable vibrator with a 555 timer. Upon detecting a correct tone, the output of the circuit should trigger an led which will remain lit for 1.1 second, as specified by the monostable vibrator below. In addition, this milestone uses a half bridge rectifier to make the mic's AC voltage into a DC value for the operational amplifier to compare.

Milestone 3 - The third milestone focuses on frequency filters. The input signal needs to be filtered in order to determine whether or not the signal has the right frequency. To integrate this, the amplified signal is inputted into an active filter which will increase the voltage amplitude of the signal if the correct frequency is played and decrease the voltage when the wrong frequency is inputed. If this desired frequency remains, this signal will propagate through the rest of the circuit. In this milestone, we also use some passive filters to get a cleaner signal from the microphone and reduce noise.

Block Diagram

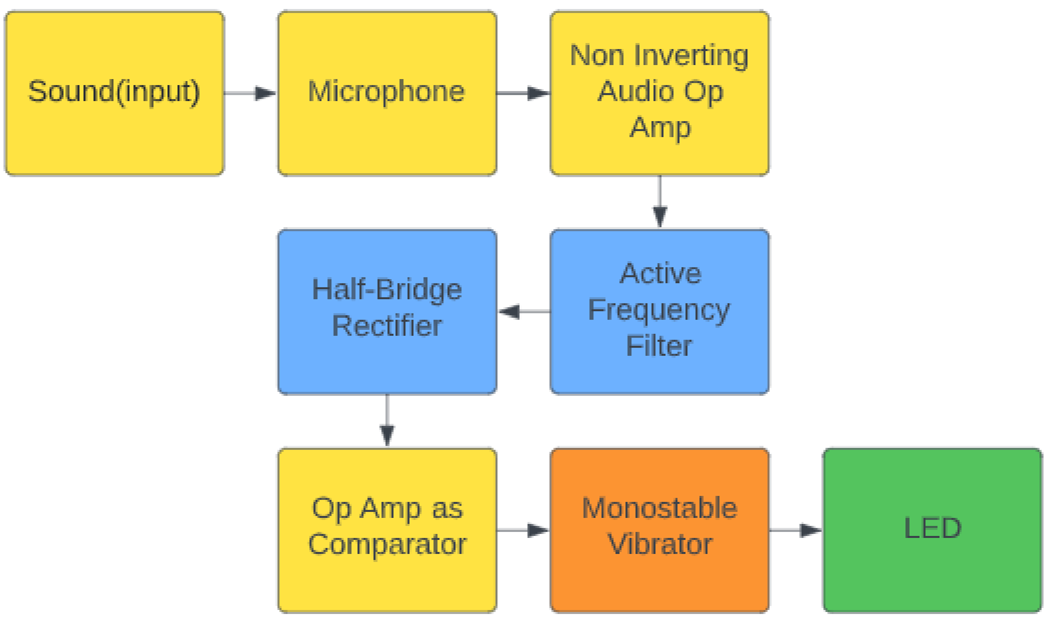


Figure 1. *Block Diagram*

Operation and Design

MS1: Input Microphone

Schematic

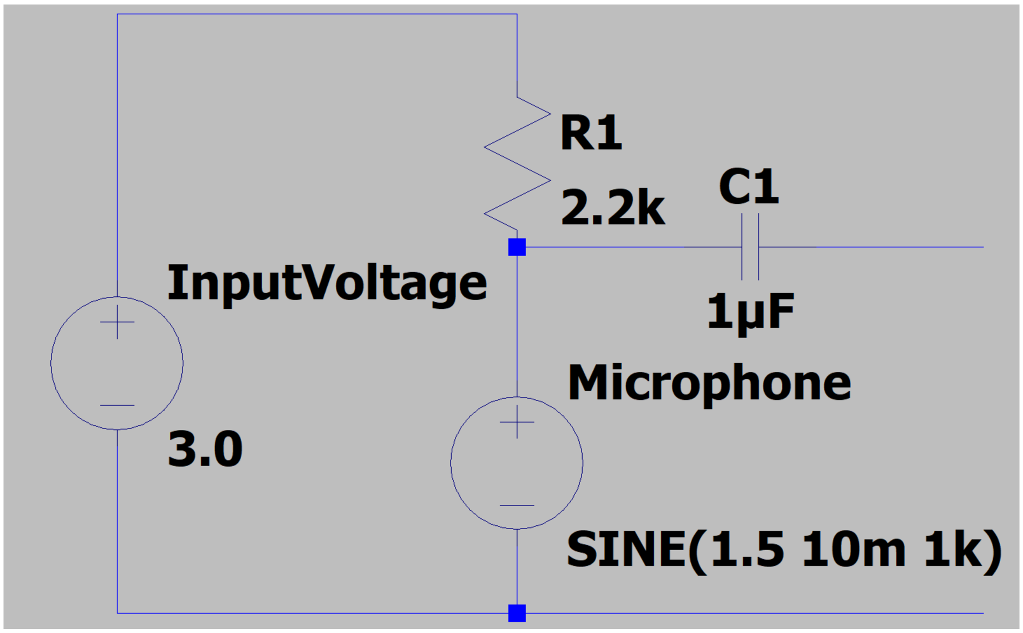


Figure 2. *Microphone Schematic*

Design Equation

There are no concrete equations for the microphone input and output.

The amplitude of AC voltage is affected by the amplitude of the sound and the frequency response graph, where voltage is proportional to the loudness for the same frequency. On differing frequencies the microphone is more sensitive when the decibel on frequency response graph is higher.

Frequency of AC voltage is roughly equal to the frequency of the sound. The microphone produces AC voltage according to the previous description.

Discussion of Component Choices

From research it is clear that there are many ways to set up the microphone with differing voltage, resistance, and capacitor so it was decided that it would be safest to use the parameter used in the data sheet for consumption of current since that would be around the parameter the manufacturer intended it to be. This includes a 2.2k ohm resistor, 1uF capacitor, and 3V voltage source, as seen in *Figure 3*. However, in the physical circuit the resistor was changed to 10k and the capacitor to 10uF because the voltage input was increased to 5V. This increase in voltage was done because the op amps in the circuit needed 5V and -5V to operate.

SPECIFICATIONS					
parameter	conditions/description	min	typ	max	units
directivity	omnidirectional				
sensitivity (S)	f = 1 kHz, 1 Pa, 0 dB = 1 V/1 Pa	-46	-44	-42	dB
operating voltage			3	10	Vdc
output impedance (Zout)	f = 1 kHz, 1 Pa		2.2		KΩ
sensitivity reduction (ΔS-Vs)	f = 1 kHz, 1 Pa, Vs = 3.0 to 2.0 Vdc		-3		dB
frequency (f)		20		20,000	Hz
current consumption (IDSS)	Vs = 3.0 Vdc, RL = 2.2 KΩ			0.5	mA

Figure 3. *Part of Microphone Data Sheet(Specification)*

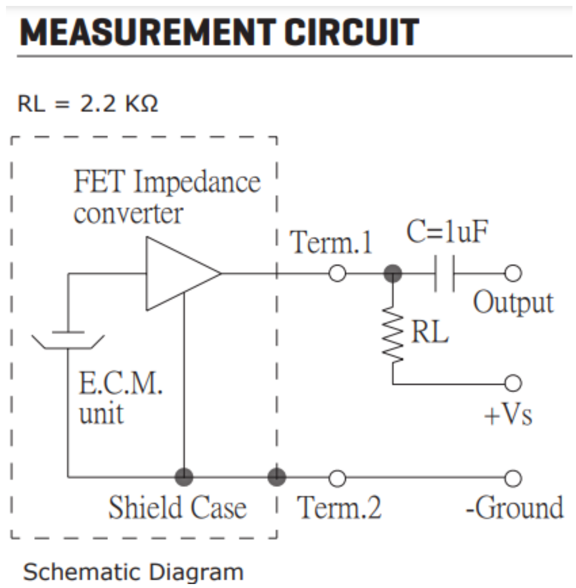


Figure 4. *Part of Microphone Data Sheet(Measurement Circuit)*

Input/Output Plot

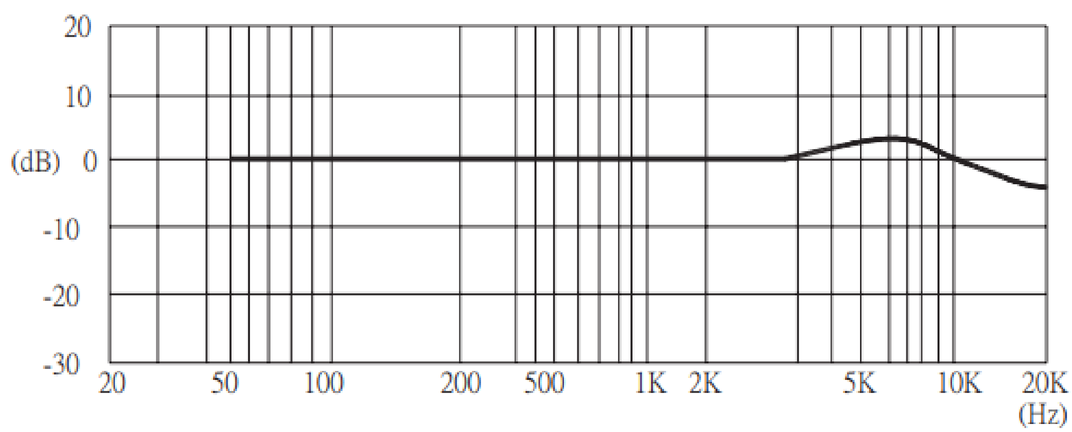


Figure 5. Microphone Frequency Response

This is the frequency response of the electret microphone and it shows what frequencies are being increased/decreased in volume (dB) as it's being picked up due to the design of the microphone. Furthermore, the microphone is more sensitive from 5k to 10k Hz and less sensitive from 10k to 20k.

MS1 Building Block 1: Non-Inverting Amplifier

Schematic

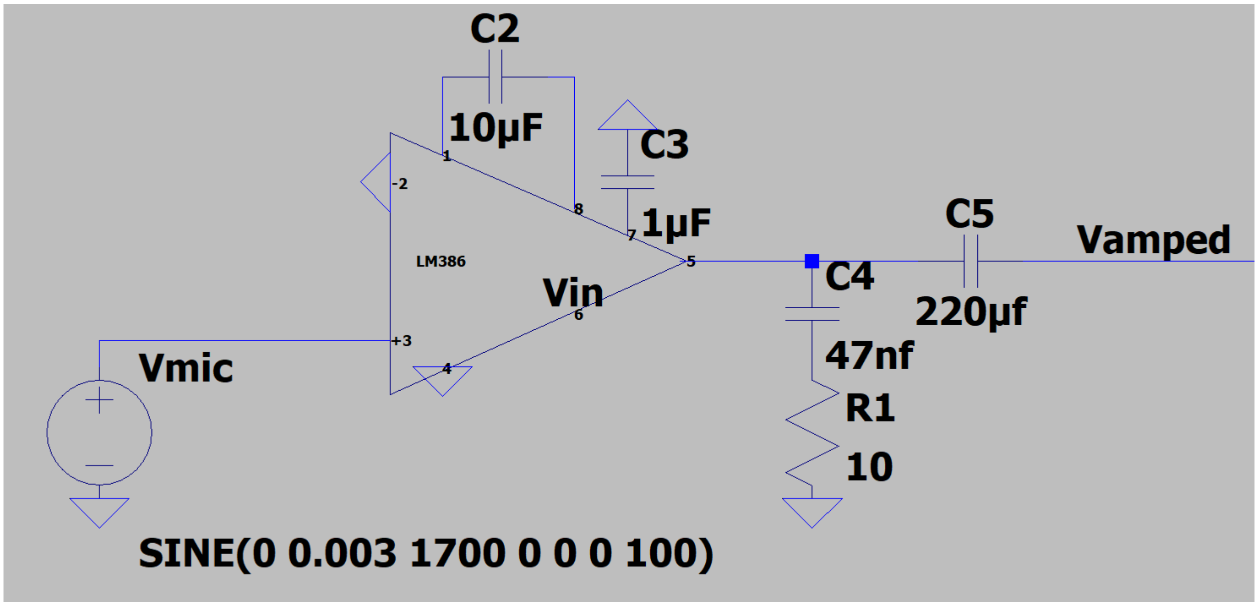


Figure 6. *Sound Amplifier (LM386)*

Design Equation

$Vamped = 200 * Vmic$

Discussion of Component Choices

In previous versions of this circuit, the OP27 was set up as an inverting amplifier to amplify the microphone input. That design had a noisy output and made it difficult to read its frequency; there was no way to adjust volume to assure a clean sinusoidal voltage. For this reason, the circuit is now using the LM386 op amp which is designed for audio amplification. In figure 6, there is one capacitor that is between pin 1 and 8 which sets the op amp with a gain of 200. This was chosen because the operational amplifier is set up in a way where the gain is controlled by what is connected from pin 1 to pin 8. If a capacitor is put from pin 1 to 8, bypassing the 1.35-kΩ resistor, which is inside of the op amp, the gain will go up to 200 (46 dB). Without this capacitor the gain is set to 20 dB. Also, this omp amp was chosen because if a potentiometer is connected to pin3, the input voltage can be adjusted.

The reason the circuit needs to amplify the voltage from the microphone is because the voltage is not significant unless the signal is amplified. The microphone's voltage before amplification is around 0.003V which would be hard to process later in the circuit. Therefore, the value chosen for C2 is the most appropriate to have the microphone at an operating voltage around 3V. In the simulation (Figure 2) below it shows that the mic's voltage is amplified to 420mV which is close to the desired operating voltage because when it goes through the half bridge rectifier its DC value is around 3V.

Input/Output Plot

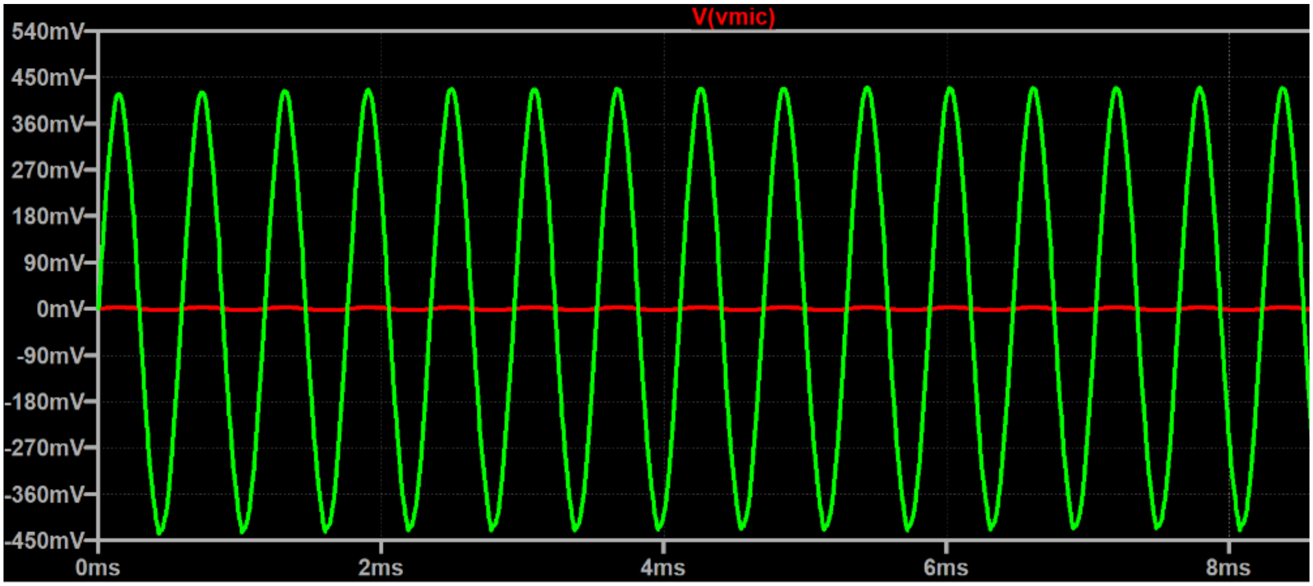


Figure 7. *Sound Amplifier Simulation (LM386), Vamped (Green)*

MS1 Building Block 2: Op Amp as Comparator

Schematic

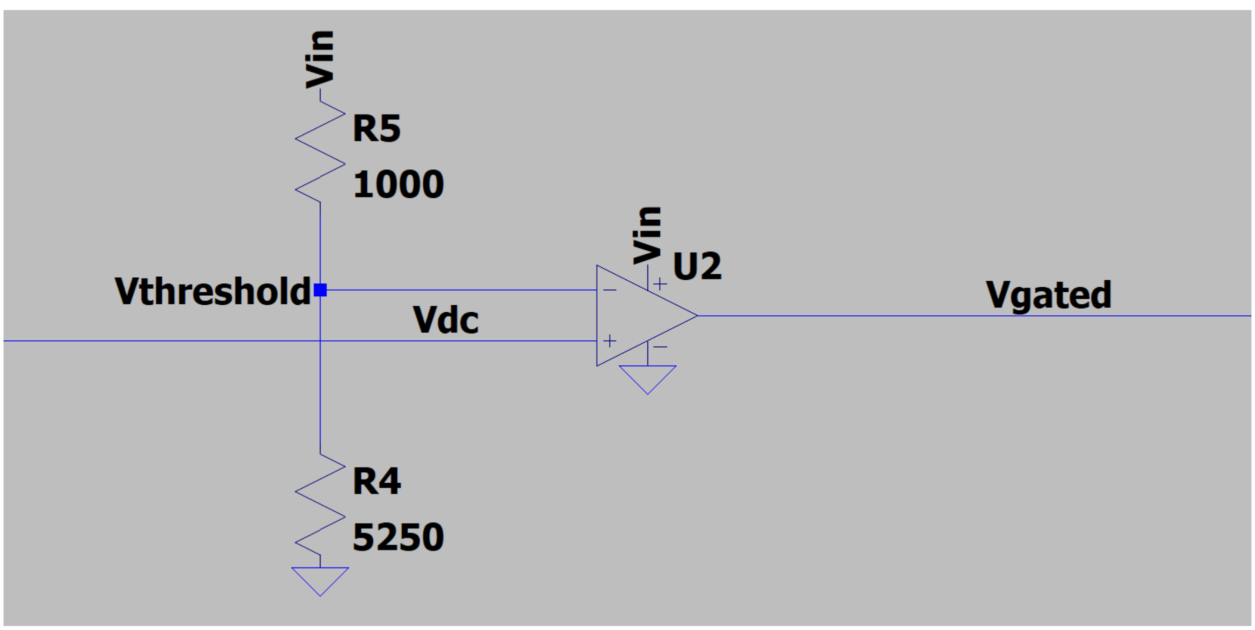


Figure 8. *Comparator Schematic*

Design Equation

$$Vthreshold = Vin \bullet \frac{R4}{R5+R4}$$

$$Vthreshold < Vdc, Vgated = Vin$$

$$Vthreshold > Vdc, Vgated = Ground$$

Discussion of Component Choices

Sounds that are received and exceed a certain threshold voltage need to be recognized by the circuit because that means the filter block before the comparator recognizes it as the right frequency input. The OP27 operational amplifier as a comparator is used to implement this. If an incorrect frequency is received, the comparator will not send voltage to the timer which will keep the LED off. The resistors values were calculated with a voltage divider formula to get a Vthreshold of 4.2V. Lastly, the OP27 is a good choice as a comparator because it is a precision amplifier which means it can compare values with smaller differences.

Input/Output Plot

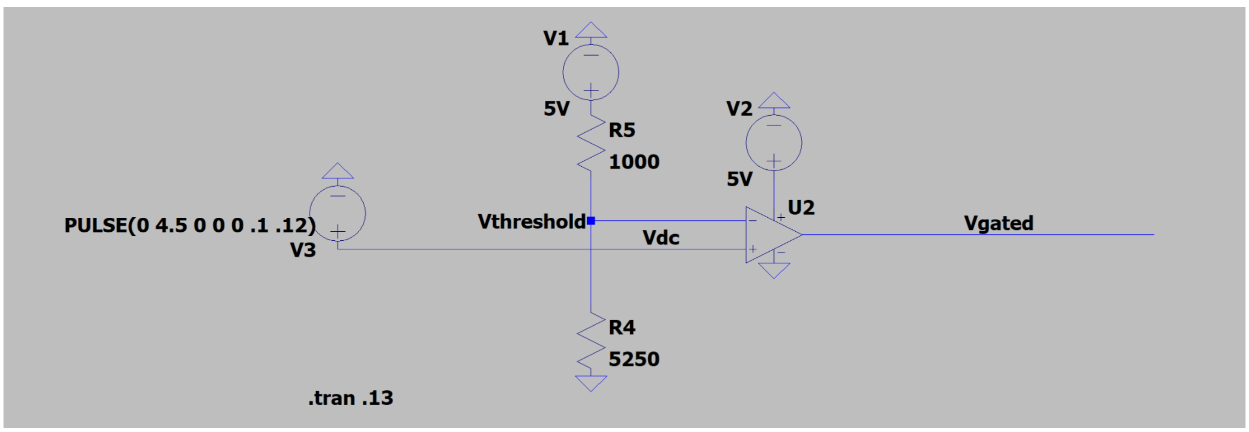


Figure 9. *Setting Positive terminal of the op amp to a pulse input (0V to 4.5V)*

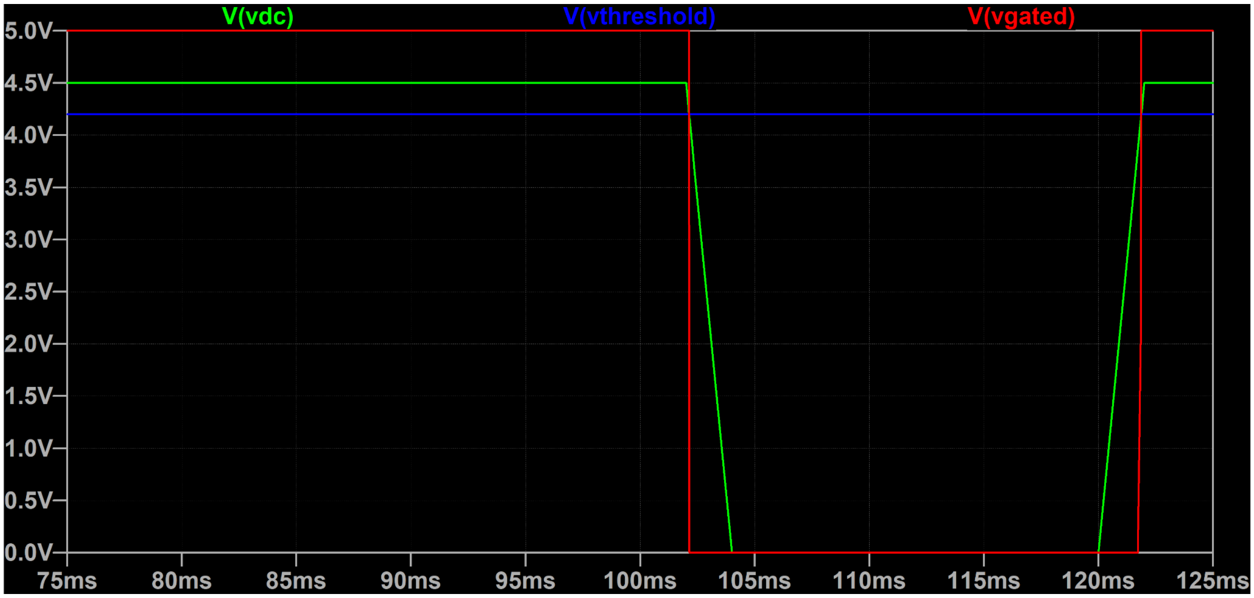


Figure 10. *Output plot of comparator with pulse input (0V to 4.5V), Vthreshold (blue)*

MS2 Building Block 1: Monostable Vibrator

Schematic

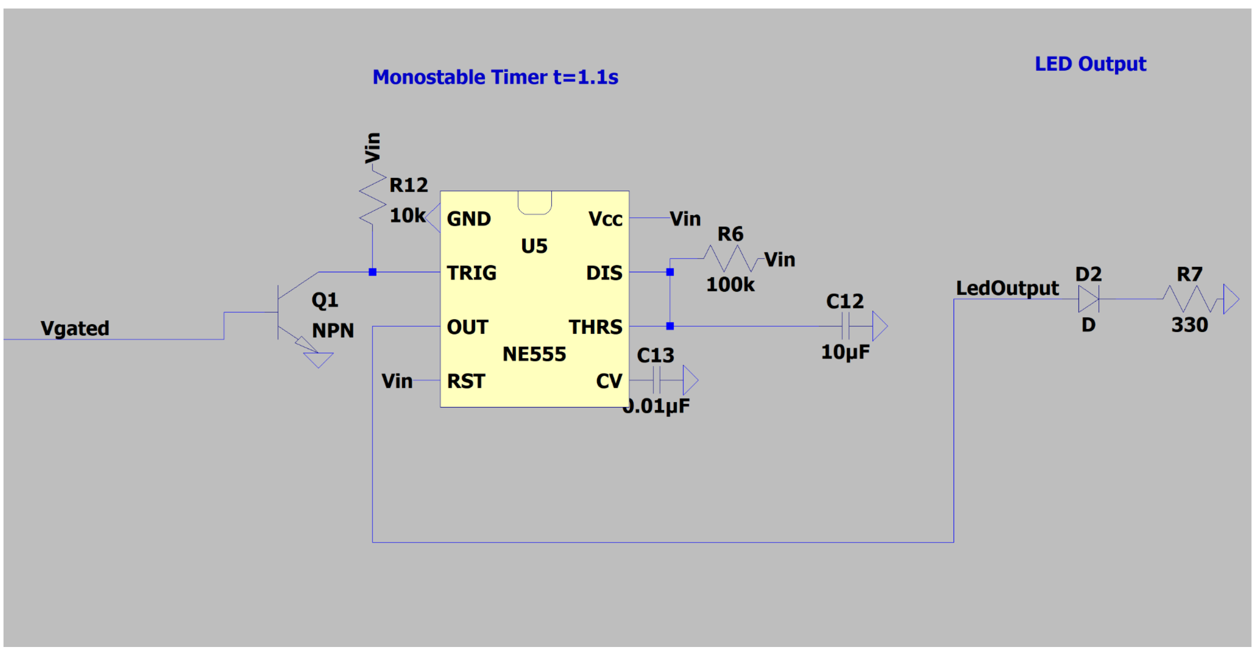


Figure 11. *555 Timer Schematic*

Design Equation

$$time \equiv t = 1.1 * R * C$$

Discussion of Component Choices

In order to light an LED for a certain amount of time, a monostable vibrator was used to trigger when the filtered output signal was above a certain threshold. The two parameters for adjusting the desired amount of time are defined by the design equation above. A capacitor value of 10uF and a resistor value of 100,000 ohms were used to time 1.1 seconds from the equation above.

Input/Output Plot

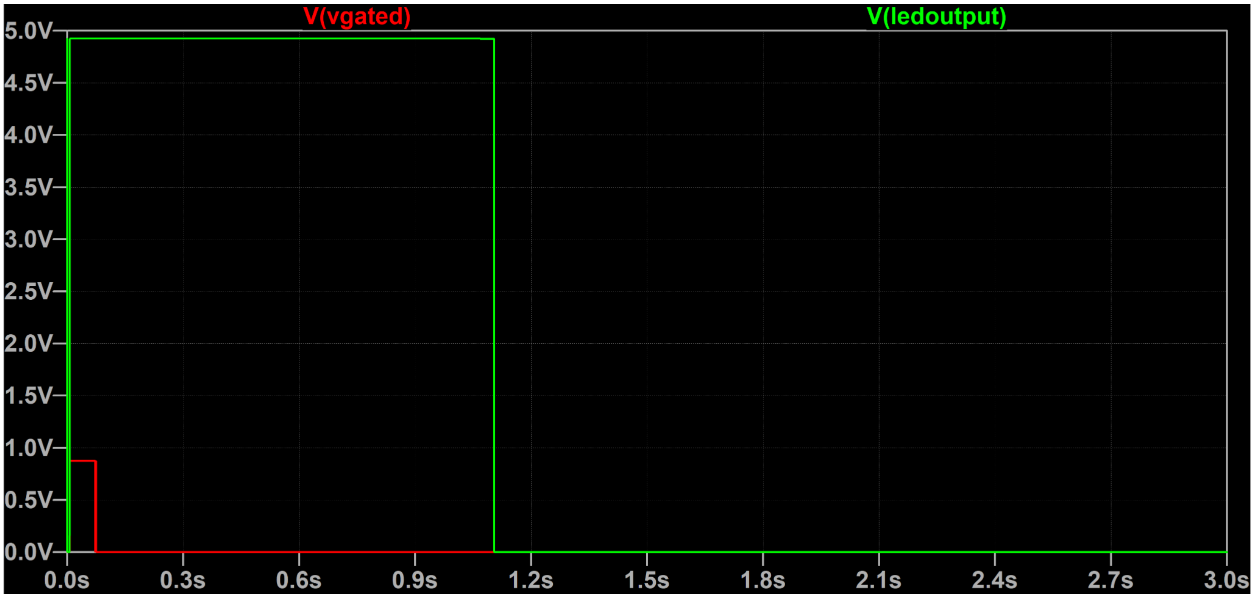


Figure 12. *555 Timer Schematic Output*

MS2 Building Block 2: Half-Bridge Rectifier

Schematic

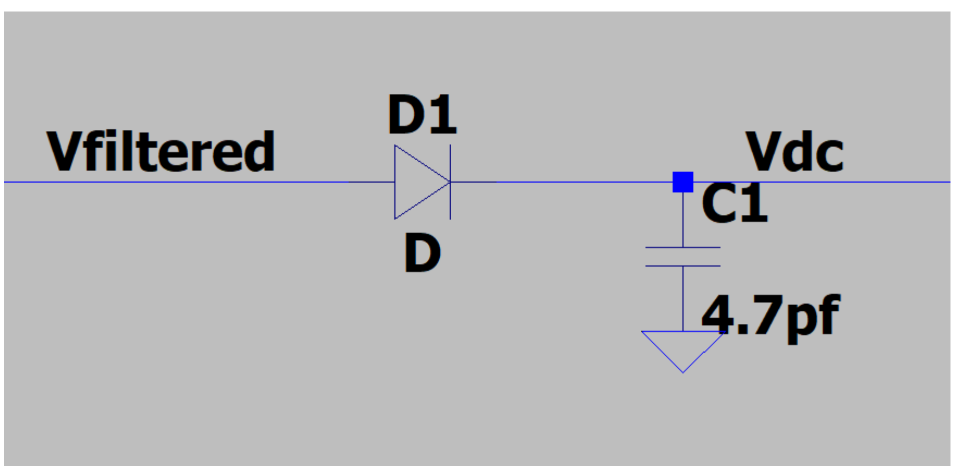


Figure 13. *AC to DC converter(half bridge rectifier)*

Design Equation

Ic = C($\frac{dVc}{dt}$) or $(1/C)\int I_C dt = V_c$

$Vc(t_0^-) = Vc(t_0^+)$

V_L = L($\frac{dI_L}{dt}$) or $(1/L)\int V_L dt = I_L$

$I_L(t_0^-) = I_L(t_0^+)$

Using KCL KVL and other circuit analysis methods first then replace I of capacitor or inductor or V of capacitor or inductor with equations above then using differential equation solve for final answer.

the final equation will be in the shape of $y(t) = y_h(t) + y_p(t)$

where $y_p(t)$ will be constant while $y_h(t)$ will be in the form of $Ae^{-t/\tau}$

τ= RC for RC circuits and L/R for RL circuits.

Plug in th initial and final condition to find the constants in the formula

Discussion of Component Choices

We used 4.7uF capacitor after we roughly calculated the value in the ballpark using the formula above and changed the capacitor multiple times until we found the 4.7uF to work the best since it keeps the voltage high enough and long enough that the it ives smooth ish dc output while not lasting long enough that it lingers too long after input ac voltage disappears.

Input/Output Plot

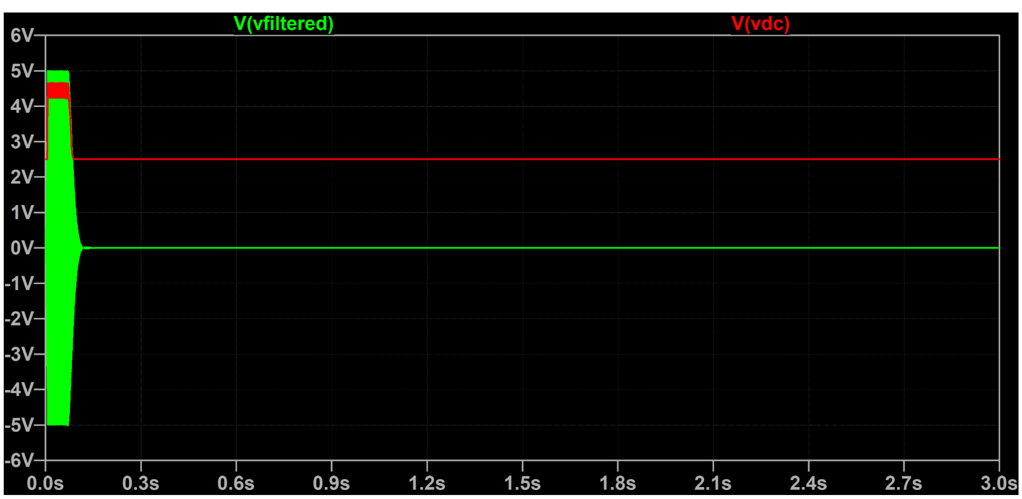


Figure 14. *AC to DC converter(half bridge rectifier) input and output*

MS3 Building Block 1: 2nd Order passive bandpass Filter

Schematic

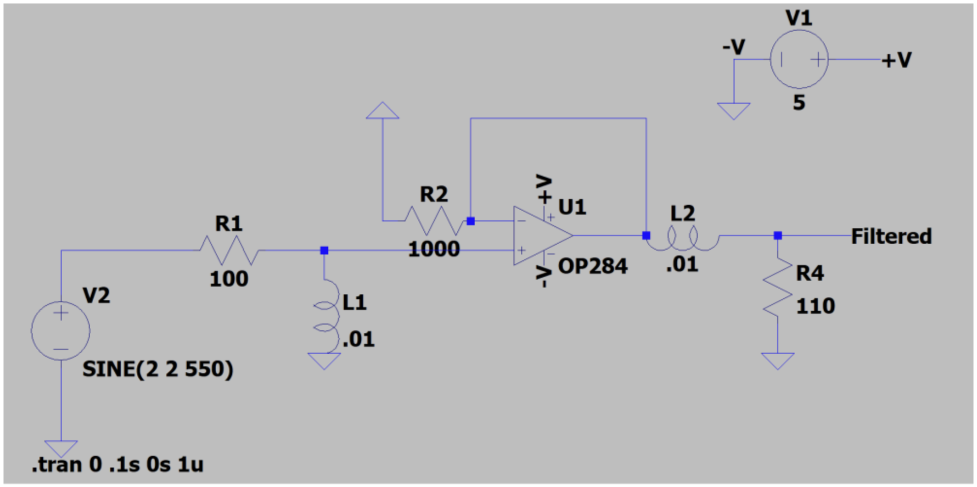


Figure 15. *2nd Order passive filter*

Design Equation

Z_L = jwL = sL

Z_C = 1/(jwC) = 1/(sC)

Z_R = R H(s) = H1 * H2 * H3

correction calculation = n*-3db

Discussion of Component Choices

We've calculated for the frequency range we wanted that the values of the components closest to those are ones shown in the circuit which include a 100 ohm resistor, 10000 ohm resistor, 110 ohm resistor, and two .01H inductors. We've used up 284 because 84 family of op amps has faster slew rate which means it can change output faster than op27 we were using before. We ended up not using this design because the stop band area doesn't have as steep of slope to allow us to differentiate reject frequencies and pass frequencies so we then try the active filter in the next building block

Input/Output Plot

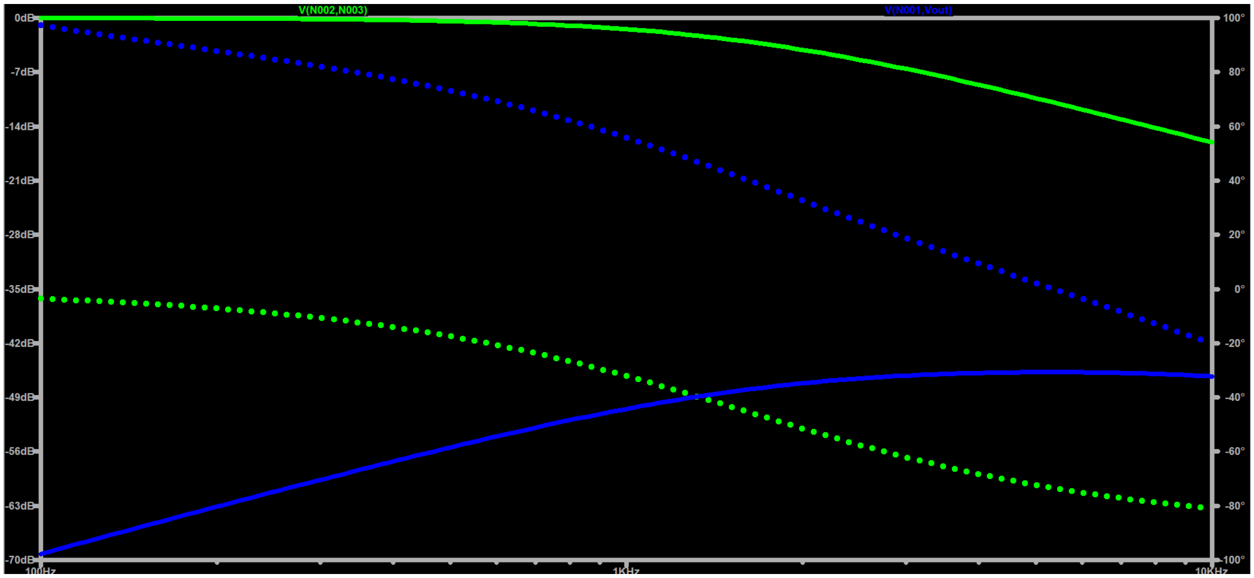


Figure 16. *2nd Order passive filter*

The top green part of the bode plot shows the filter of the beginning part and the bottom blue plot is the bode plot of the second part of the filter. When both of them are joined by an op amp it creates a second order passive band pass filter which has the pass band between both of the stop bands.

MS3 Building Block 2: 2nd Order Active bandpass Filter

Schematic

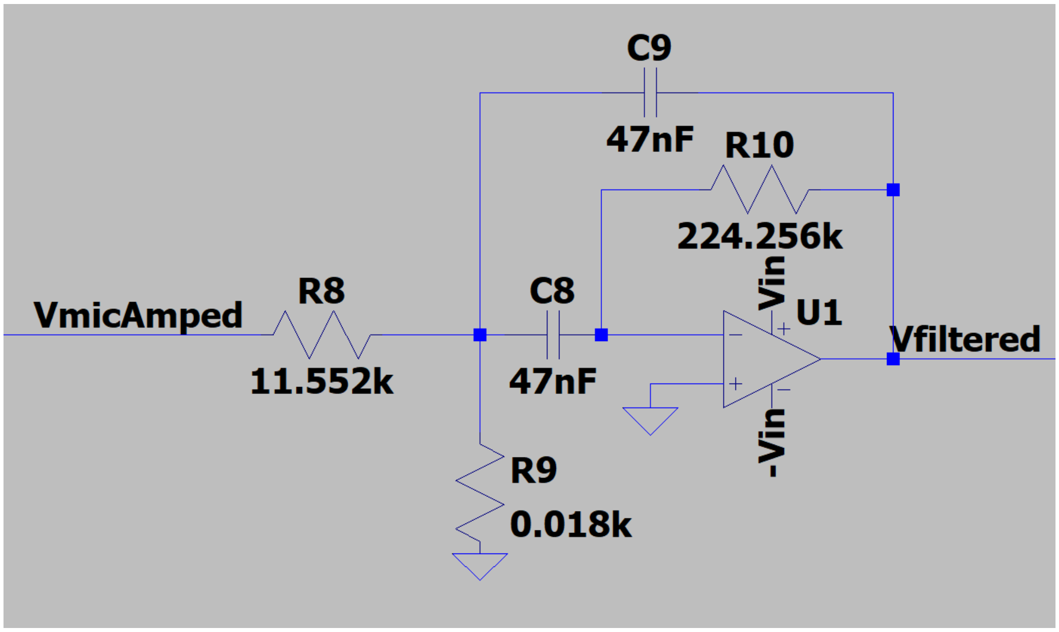


Figure 17. *2nd Order active* multiple feedback bandpass *filter*

Design Equation

Transfer Function

$$H(s) = \frac{-\frac{1}{R1C}s}{s^2 + (\frac{2}{R2C})s + \frac{1}{R2C^2}(\frac{1}{R1} + \frac{1}{R3})}$$

Q:

$$Filter\ quality = \pi f_m R_2 C$$

Resistor Values Formula Derivations:

$$R2 = \frac{Q}{\pi f_m C}$$

$$R1 = \frac{R2}{-2A_m}$$

$$R3 = \frac{-A_m R1}{2Q^2 + A_m}$$

Discussion of Component Choices

The final circuit uses an active bandpass filter because of its advantages over passive filters. Since the purpose of our circuit is to generate an output when it detects a certain frequency, the range of acceptable frequencies is determined by the slope of the rolloff or the quality factor, Q. The desired Q for the frequency detector needs to be high to avoid a wider margin of error. We achieved the desired pass band that is narrow and centered at almost 1700hz in the circuit.

Because the needs of this circuit were specifically targeting this small range of frequencies, active filters provided a much steeper roll off than passive filters. In addition, active filters can provide gain to the signal allowing more accurate control of the output signal. For instance, consider staging passive filters to replicate the same quality factor. This would result in decreasing output voltages for each additional passive filter due to attenuation. However, an active filter that can apply gain to the voltage doesn't have that problem. Therefore, an active filter was a more logical solution for the design of the frequency detector.

Furthermore, the active filter is set up as an multiple feedback bandpass (MFB) because the alternative setup, sallen key, has an increase in db in the higher frequencies which can not work for this project. The MFB has a constant slope that decreases in both sides with no gain in the higher frequencies, which makes it reliable for voltage comparator op-amp to detect if it is the right frequency. Lastly, the op amp used in the physical circuit is the 484 op amp because it has 4 op amps and if filters needed to be chained together then it could have been done with a single op amp.

Input/Output Plot

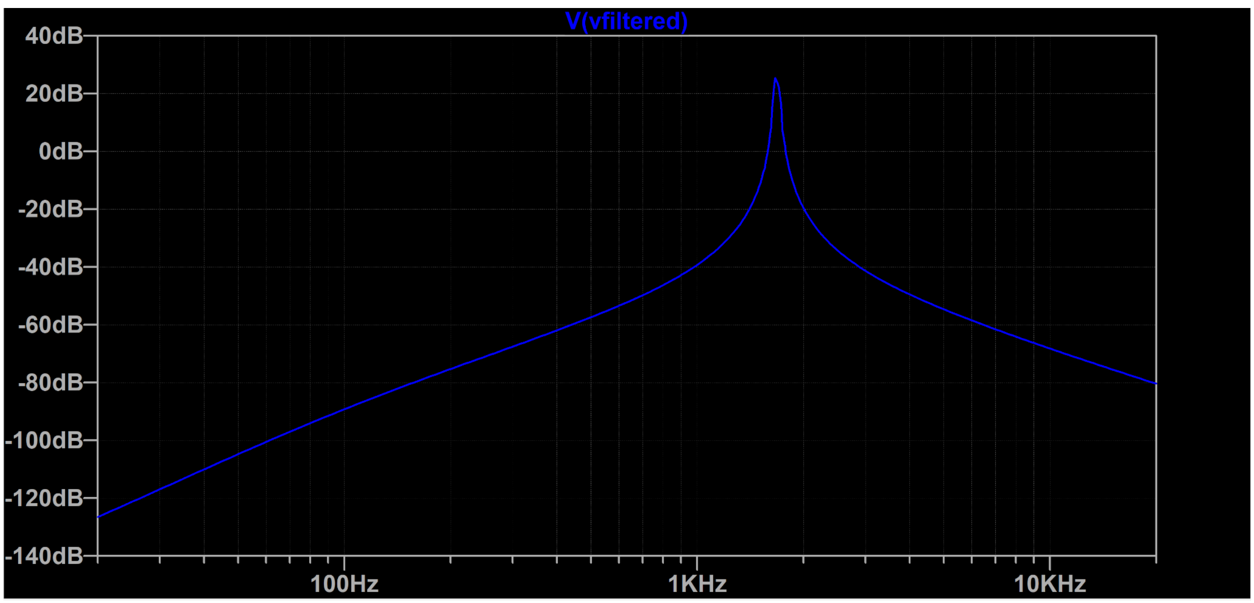


Figure 18. Bode Plot/frequency response for Active MFB filter

Figure 19. given a 1700 Hz sine wave as input (Vmicamped)

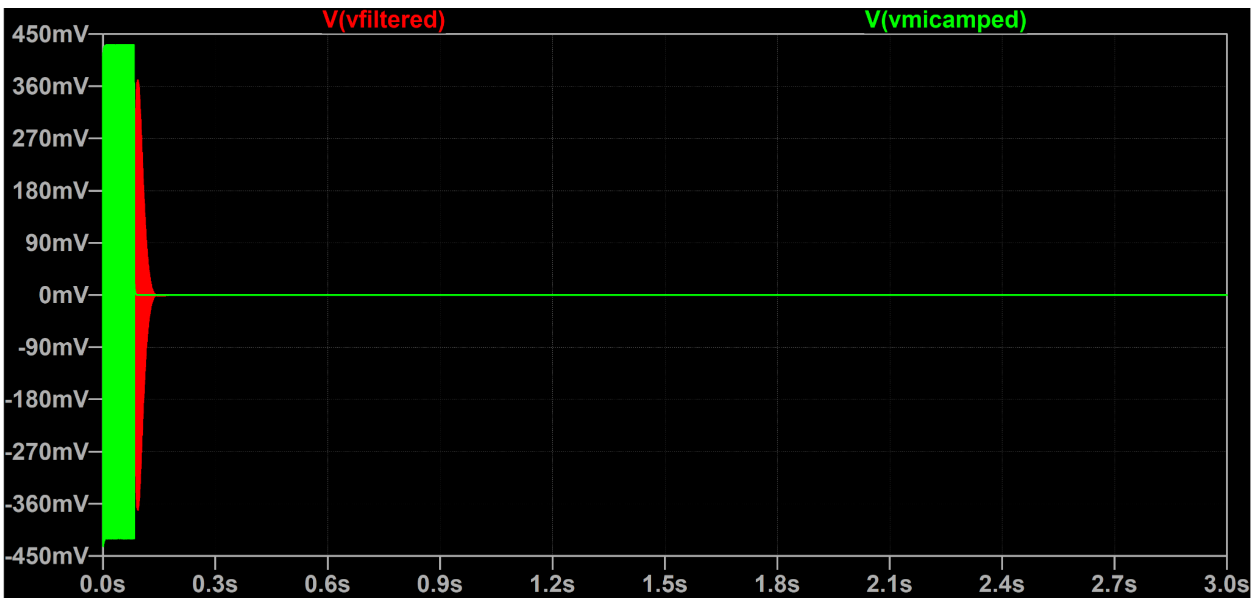
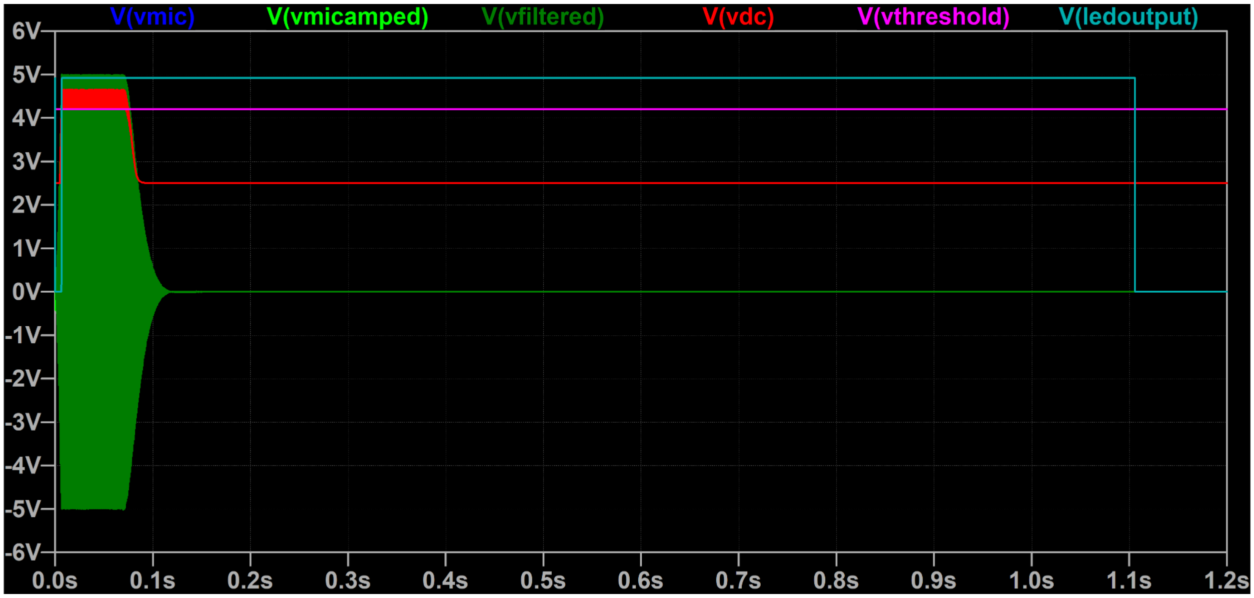


Figure 20. given a 1200 Hz sine wave as input (Vmicamped)

Integration and Optimization

The signal goes through the circuit as follows: Input sounds are picked up by the electret microphone between -46 and -42 dB. The microphone converts this input to a 20 Hz to 20 kHz AC voltage. The signal is amplified by the audio amplifier of a factor of 200. This signal is then filtered through the active filter, allowing only frequencies around 1700 Hz to continue. This is approximately the target range for the design specifications. In order to compare this signal to a voltage threshold, the output from the active filter is converted to a DC signal using a half-bridge rectifier. The output from the half-bridge rectifier is fed into a comparator which will output a signal to an LED if the input to the comparator is above a threshold.

Figure 21. given a 1700 Hz sine wave as input (Vmicamped)



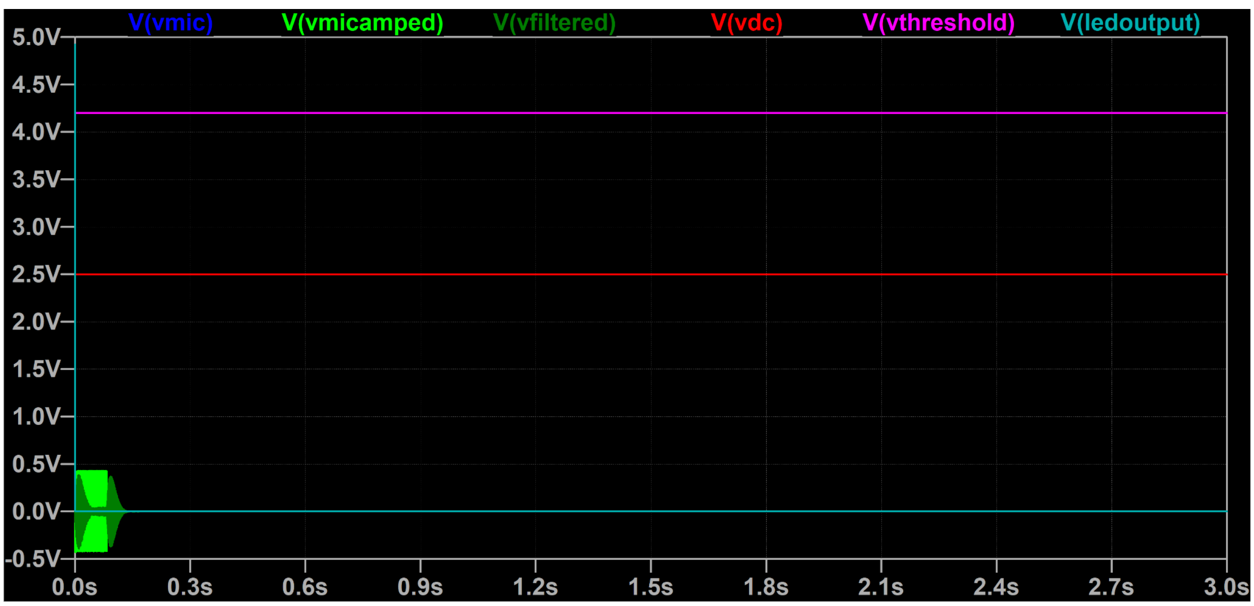


Figure 22. given a 1200 Hz sine wave as input (Vmicamped)

The main difficulty during the first milestone was integrating the microphone into the circuit. The original operational amplifier did not do a good job amplifying the signal. We opted for the LM386 audio amplifier and that improved the signal quality. The old system's volume relied on how charged the capacitor was when in use and we had no reliable way to charge the capacitor to a specific value to get a consistent result. With the new audio amplifier, the circuit has the specifications to sink and source a higher threshold of voltage without saturating the sinusoidal input from the microphone.

To optimize the volume threshold for the comparator, depending on the environment the circuit is operated in, the comparator can be tuned to a higher or lower threshold by a potentiometer. In the instance where there is a lot of noise, a higher threshold can be used to guarantee a signal.

In order to create the most ideal bandpass filter, component values were chosen to optimize the threshold for which “acceptable” frequencies would be let through. Initially two active bandpass filters were staged in an attempt to get a small range of allowed frequencies. One of the problems with this approach was the two stage active filter had the unintended consequence of not letting through any frequencies. We think it's either that pass band became so sharp that it was impossible to catch the right frequencies or that small tolerances used in the parts added up so both of the filter had slightly different pass band frequencies so in the end we ended up using only one active filter since we found out solving the transfer function that we can make the slope of the filter higher or lower without using multiple active filter.

The entire design was also optimized for the purposes of this class. The original melody detection system was replaced with a single tone frequency detector since it adequately accomplishes demonstrating the principles for voice detections in a more efficient way.

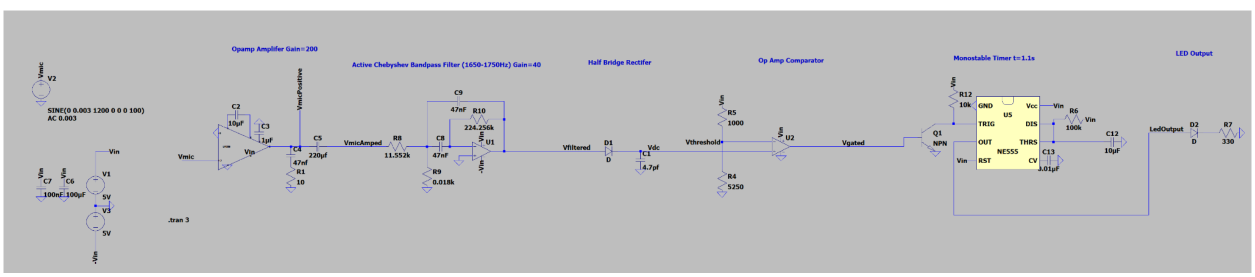


Figure 23. *Circuit Simulation All Part Integrated Overview*

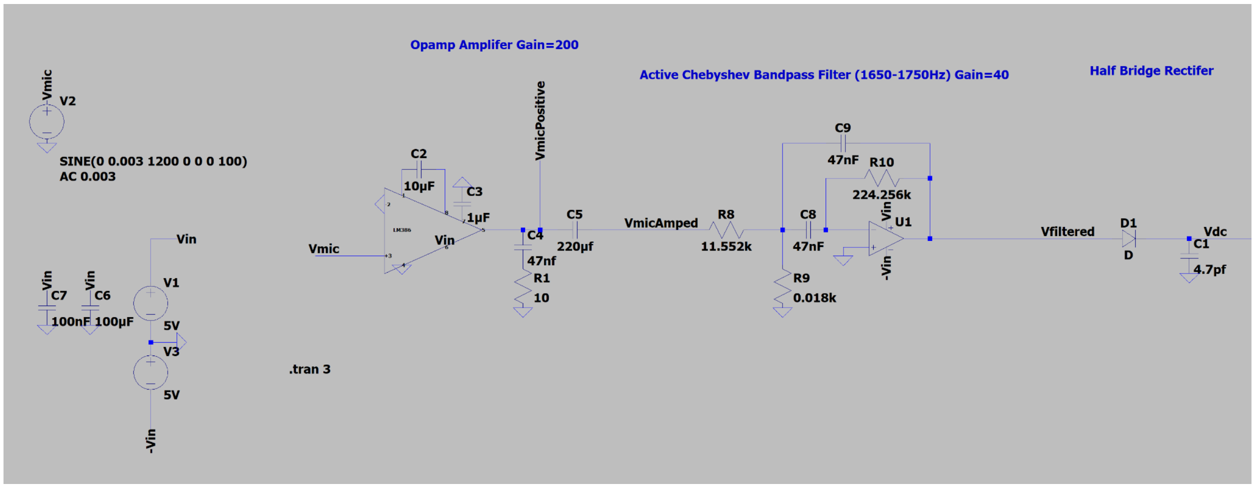


Figure 24. *Circuit Simulation All Part Integrated Overview Zoom Left*

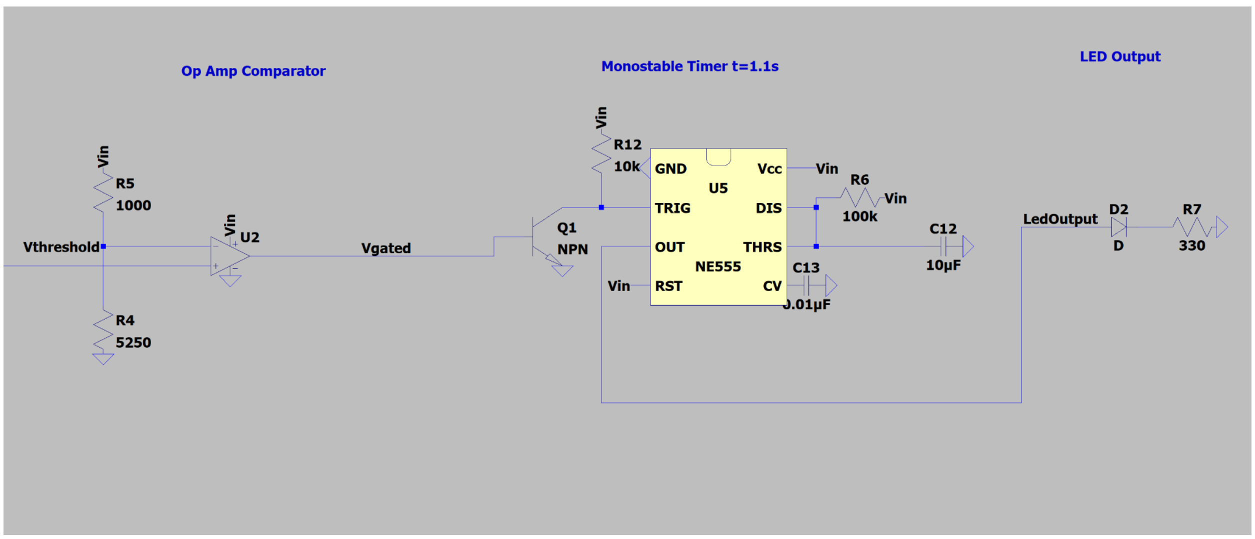


Figure 25. *Circuit Simulation All Part Integrated Overview Zoom Right*

Operating Conditions

There are some operating limitations within the design. The overall use of the system requires the tuning of two potentiometers which depend on the environment in which it is operated. One which controls the threshold voltage of the acceptable signal and one that controls the

A major limitation in this design is that even if many frequencies are heard, and one happens to fall in the accepted frequency range, the circuit will still register the tones as correct. This is unacceptable for a security system since an input will be considered correct despite containing incorrect tones. One possible solution is an additional notch filter in parallel with the active filter that checks for the existence of these incorrect tones. If there are no incorrect tones, the signal from that side of the circuit can trigger a transistor switch which acts as an additional input to the monostable vibrator.

The circuit would not be able to be used in some circumstances such as when there is a fire alarm or any loud noise in the background since if there is a loud enough noise the circuit will let through the signal even if it is the wrong frequency. The circuit will not be able to filter out significant noise. This restricts the circuit's usefulness in an outdoor environment where even a passing car, or even indoor foot traffic could nullify the sound. A potential solution is to isolate the circuit to guarantee the same, or very similar operating conditions. For instance, a personal bank vault that is isolated from sonic disturbances would be an ideal condition for the detector to operate in. This encapsulates the tradeoff from increasing the sensitivity of the comparator; this makes it easier for the user to input frequencies, but it also will pick up extraneous noise.

The active filter's bandpass has a tradeoff between the margin of acceptable values and the accuracy of detecting a correct frequency. The quality factor can be lowered or raised which increases or decreases the slope of the rolloff. Thus the margin of acceptable frequencies can be widened or narrowed. However, this carries an implicit trade off where if the band is widened, the user can more easily enter the correct frequency, but it will also have a higher margin of error for intruders to "guess" the correct tone. Ideally, this tone detector can be used synchronously for multiple tones to increase the security of the intended usage.

References

"Science Direct." *Biometric Authentication*, https://www.sciencedirect.com/topics/computer-science/biometric-authentication. Accessed 1 March 2022.

CUI Devices, "Electret Condenser Microphones," CMA-4544PF-W Datasheet, January 2020 .

Analog Devices. "Low Noise, Precision Operational Amplifier OP27",

https://www.analog.com/media/en/technical-documentation/data-sheets/op27.pdf Accessed March 1 2022.

Analog Devices, "Precision Rail-to-Rail Input and Output Operational Amplifiers",

D00293-0-4/11(J), 1996–2011.

Texas Instruments, "LM386 Low Voltage Audio Power Amplifier", SNAS545C, May 2017

Texas Instruments, , "LM555 Timer", SNAS548D, January 2015

Texas Instruments, "Chapter 16 - Active Filter Design Techniques." *changpuak.ch*,

https://www.changpuak.ch/electronics/downloads/sloa088.pdf. Accessed 1 May 2022