BE1.HEEL - Heart Sound Amplifier Lab Report

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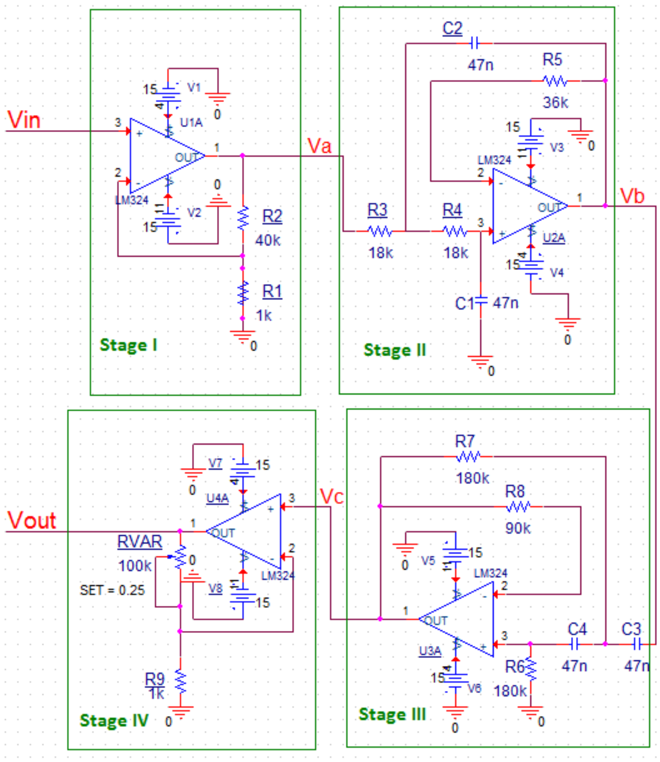
I. Introduction

The aim of this project is to design a heart rate amplifier and connect it to a microphone and headphones to create an electronic stethoscope. In the first part of the report, the heart sound amplifier is designed and modelled using PSpice (OrCAD Capture 16.6 2012). In the second part, a stethoscope is implemented in the lab using an electret microphone, the designed amplifier and headphones.

II. Design and Modelling



Schematic 1:
Design based on conditions stated in PSpice labs manual

Conditions specified[1]:
- Use LM324 opamps.
- Microphone signal (V_{in}) has amplitude 1-5mV.
- A signal of 2-3V is needed to drive the headphones. (Vout)
- Filter noise outside heart sound frequencies.
- Operate opamps within max gain capabilities. ($Gain_{max}$=200)
- Make sure DC offset does not saturate opamps.
- Make sure V_{out} has minimal DC offset.

In order to achieve the above, we first identify that the most significant heart sounds range in frequency from 20Hz to 200Hz[2]. We thus conclude that we need to filter out frequencies higher than 200Hz (e.g. ambient sounds) and frequencies lower than 20Hz (e.g. movements of the subject). Finally we acknowledge that there will be some 50Hz noise from the mains, however we assume that it will be in the order of tens of μV and hence negligible.

Our amplifier consists of 4 stages:

Stage I - A non-inverting amplifier of gain (G=1+R2/R1=41)

a) This will give us $41\text{mV} \leq V_a \leq 205\text{mV}$
b) It will only saturate the opamp if
$V_{in}\text{offset} > V_{saturation}/\text{Gain} = 13.3\,/41 = 324\text{mV}$
This means that we can be certain of getting a reliable V_a for V_{in}offset < 300mV.
c) Finally as the LM324 datasheet suggests a maximum input offset voltage of 3mV, the DC offset of V_a may be up to $41 \cdot 3 = 123\text{mV}$[3].

Stage II - A 2nd order active low pass filter with $f_{cut\text{-}off}$=188.1Hz.

a) In our particular design we set R=R3=R4=18kΩ and C=C1=C2=47nF and R5=2·R=36kΩ, which implies fc=1/(2·π·R·C)=188Hz
b) This serves to remove high frequency noise such as ambient sounds.

We simulate the frequency response of the designed low-pass filter.

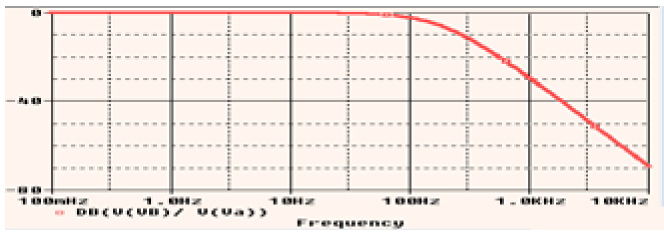


Figure 1: Frequency response of Stage II

We expect the graph in Figure 1 to follow the equation of a unity gain Sallen-Key low pass filter:

$$Gain = \left|\frac{1}{s^2 \cdot C1 \cdot C2 \cdot R3 \cdot R4 + C1(R3 + R4)s + 1}\right|$$

Where s=j·2·π·f and C1 is the capacitor to ground[4].
At f_c=188.1Hz we find Gain=-6dB, for $f \gg f_c$ the roll off is 40dB per decade and for $f \ll f_c$ Gain=0dB. All of which are consistent with our equation. For f close to f_c the signal is still attenuated to some degree and the gain continues to follow the equation described above.

Stage III - A 2nd order active high pass filter with $f_{cut\text{-}off}$=18.8Hz.

a) In our particular design we set R=R6=R7=180kΩ and C=C3=C4=47nF and R8=R/2=90kΩ, which implies fc=1/(2·π·R·C)=18.8Hz.
b) This stage serves the dual purpose of removing low frequency noise such as slight movements of the subject and also to remove DC offset (0 Hz) before the final amplification stage.

We simulate the frequency response of the designed high-pass filter.

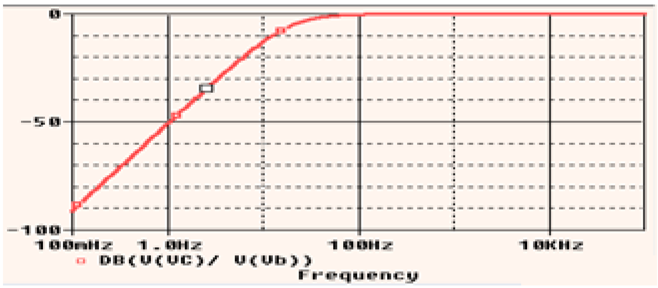


Figure 2: Frequency response of Stage III

We expect the graph in Figure 2 to follow the equation of a unity gain Sallen-Key high pass filter:

$$Gain = \left|\frac{s^2(R6 \cdot R7 \cdot C3 \cdot C4)}{s^2(R6 \cdot R7 \cdot C3 \cdot C4) + s(R7 \cdot C3 + R7 \cdot C4) + 1}\right|$$

Where s=j·2·π·f and R6 is the capacitor to ground[4].

At f_c=18.8Hz we find Gain=-6dB, for f<<f_c the roll off is 40dB per decade and for f>>f_c, Gain=0dB. All of which are consistent with our equation. For f close to f_c the signal is still attenuated to some degree and the gain continues to follow the equation described above.

By plotting the combined frequency response of the two filters (graph not included) we observe that the frequency range between 30Hz and 120Hz is attenuated by a factor of 0.7 to 0.8. Hence for our calculations we assume that the combined attenuation of the passband due to our filters is 0.75.

Stage IV - We design a non-inverting amplifier of variable gain by shorting the middle and one of the outer pins of a 100k potentiometer. We thus create a variable resistor whose resistance varies linearly with potentiometer rotation. The gain of our last stage (G_{var}=1+R_{VAR}/R9) varies between (1+0k/1k=1 and 1+100k/1k=101). This allows us to adjust the total gain of our system to accommodate variations in V_{in}.

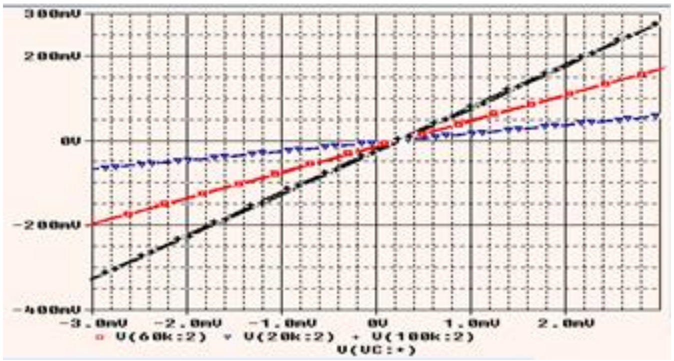


Figure 3: Output signal plotted vs Input signal for Stage IV

In Figure 3 each line represents a different value of R_{VAR}. [Blue-20kΩ, Red-60kΩ, Black-100kΩ].

The slope of each line corresponds to the gain of Stage IV for the respective value of R_{VAR}. [Blue-Gain=21, Red-Gain=61, Black-Gain=101], which follows the non-inverting amplifier equation. If we assume that the high pass filter removes any pre-existing dc offset, the Stage IV offset will be the total offset of our system. The y-intercept of each graph represents the DC offset of Stage IV and are all under 40mV.

However, as the LM324 datasheet suggests a maximum input offset voltage of 3mV, in practice the DC offset of V_{out} may be up to 101·3=303mV in the case where we need Stage IV to provide maximum gain[3].

III. Implementation

In order to construct the circuit we first had to verify that the specifications assumed in the design part of our report were valid. By connecting the microphone we were provided with directly to the oscilloscope we found that it produced an input signal of a 12V DC offset (when setting the oscilloscope to DC coupling) and an amplitude in the range of 50mV to 500mV, instead of 1-5mV, (setting the oscilloscope to AC coupling) when measuring various subjects' heart sounds.

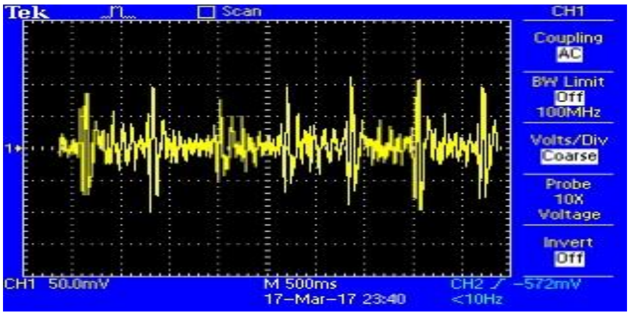


Figure 4: Heart sounds as captured by microphone (AC coupling)

In Figure 4 we observe a maximum amplitude of 100mV.

In order to identify the voltage necessary to get a reasonably loud sound from our headphones we connected them directly to a function generator and inputted a 100Hz signal. By measuring the voltage across the headphones we concluded that sufficient sound volume was provided when $V_{Headphones}$ was in the range of 0.2V to 0.3V.

However, two more factors need to be taken into consideration: the maximum power rating of our headphones (60mW) and the maximum output current of the LM324 (40mA)[3]. In order to satisfy both conditions we will need to introduce a resistor (R) in series with our headphones at the output of our amplifier. Following the instructions provided we set the series resistor value at R=1kΩ[5].

By connecting the headphones in series with a 16Ω resistor and inputting sine waves of varying frequency we concluded that for frequencies under 1kHz the headphone impedance Z_H is a constant 11.6Ω. Hence we calculated that:

$$V_H = Vout\frac{11.6}{R + 11.6}$$

Thus for $V_H \approx 0.25V$ *and* $R = 1k\Omega$ *we need* $Vout \approx 22V$

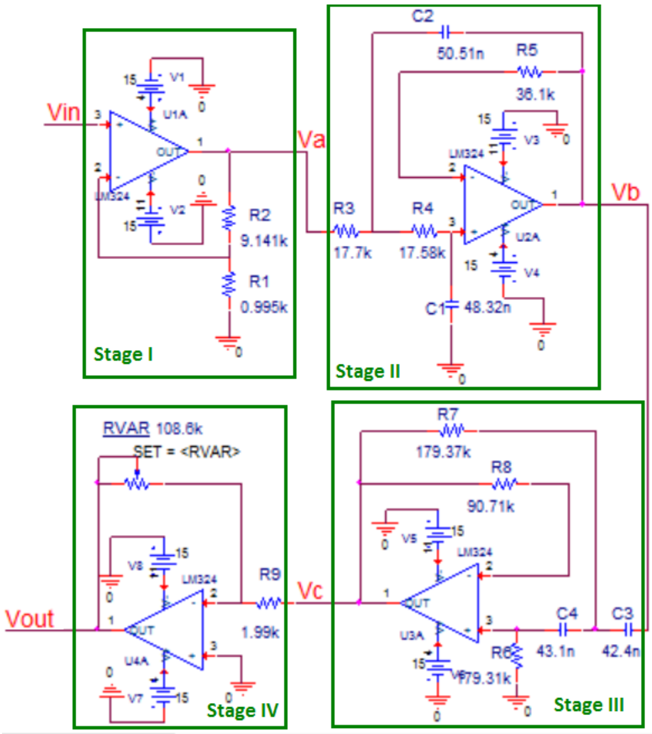
Which is problematic as the $V_{saturation}$=13.3V[1].

We therefore conclude that we need to approach the saturation limit as much as possible while also taking care not to clip our signal. Within those constraints we settled for a desired voltage of V_{out} = 10V.

Given the change in V_{in} and V_{out} we adjust our schematic appropriately. Namely, change Stage I to a non-inverting amplifier of Gain 10 so as to not saturate the opamp when a large signal is provided and changed stage IV to an inverting amplifier of variable gain from 0 to -50. The latter stage was adjusted so as to increase the possible range of inputs that can give V_{out}=10V, to:

V_{inmin} =20mV [When gain is set at -50]

V_{inmax} =1.2V [the maximum voltage at which we can safely say Stage I will not saturate].



Schematic 2: Amplifier implemented in the lab

The blue line represents the data measured, while the orange line represents the expected values according to the equation described in the design section. The -6 dB point lays at about 180Hz and the curve closely approximates the relationship expected. However as the voltage was measured using the peak to peak function of the oscilloscope, the output signal appears to have a systematic error, i.e. measured voltage is higher than predicted. This becomes significant for small enough output signals.

Stage III: We implement the High pass filter designed in Pspice.

R6=179.31kΩ	R7=179.37kΩ	R8=90.71kΩ	C3=42.4nF	C4=43.1nF

Table 2: Actual components measured

Hence, the $f_C = \frac{1}{2 \cdot \pi \cdot \sqrt{R6 \cdot R7 \cdot C3 \cdot C4}} = 20.76$Hz.

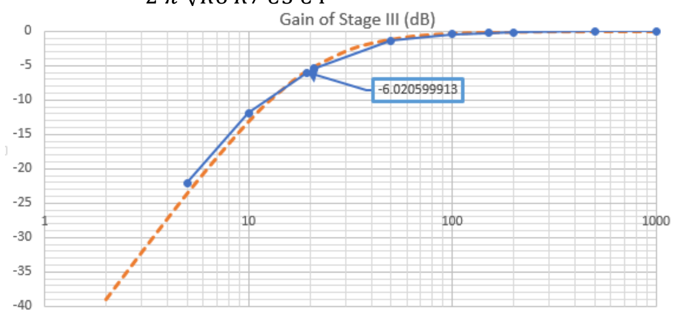


Figure 7: Frequency response of the implemented high pass filter

We test the filter on its own by inputting a 2.29V sine wave and changing the frequency to observe the frequency response.

The blue line represents the data measured, while the orange line represents the expected values according to the equation described in the design section. The -6 dB point lays at about 20Hz and the curve closely approximates the relationship expected. We observe the same systematic error described in the previous stage.

Stage I: Measured R1=0.995kΩ, R2=9.141kΩ, thus Gain= 1+R2/R1=10.19. We use the function generator to input a 100Hz sine wave of different amplitudes (V_{in}) and record the output of Stage I (V_a).

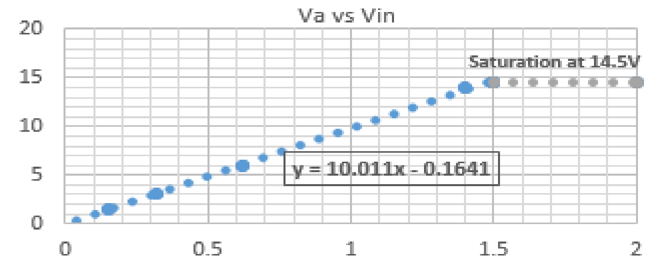


Figure 5: Gain is equal to the slope (10.01) which is consistent with the expected Gain (10.19)

The graph in Figure 5 levels off at the saturation voltage which is found to be 14.5V and the y-intercept is -0.164 which represents the DC offset.

Stage II: We implement the Low pas filter designed in PSpice.

R3=17.7kΩ	R4=17.58kΩ	R5=36.1kΩ	C1=48.32nF	C2=50.51nF

Table 1: Actual components measured

Hence, the $f_C = \frac{1}{2 \cdot \pi \cdot \sqrt{R3 \cdot R4 \cdot C1 \cdot C2}} = 182.63$Hz.

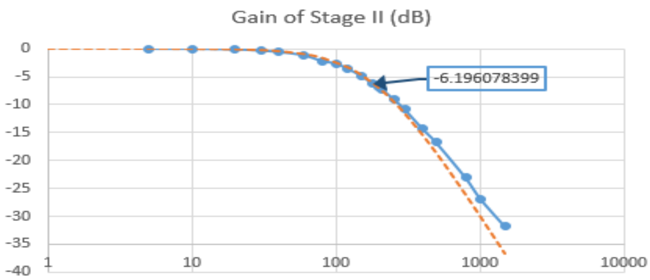


Figure 6: Frequency response of the implemented low pass filter

We test the filter on its own by inputting a 1.6V sine wave and changing the frequency to observe the frequency response.

Stage IV: Measured R9=1.99kΩ, R_{VARmax}=108.6kΩ, R_{VARmin}=0.7Ω
The gain of an inverting amplifier follows the formula:
Gain= -R_{VAR}/R9. Thus the |Gain| can become less than 1 and can reach 108.6/1.99=54.57. We test the amplifier on its own by inputting 100Hz sine waves of varying amplitudes and plottting the relationship between Vc and V_{OUT}. The slope of the graph will be equal to |Gain| and the y-intercept to DC offset. This is repeated for various rotations of the potentiometer, i.e. different values of R_{VAR}.

Stage IV V_{OUT} vs V_C

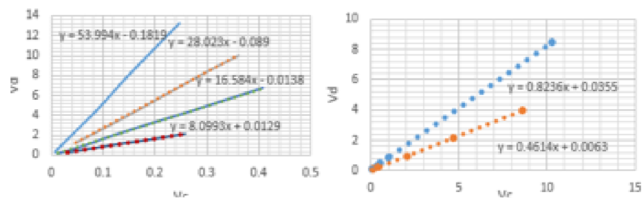


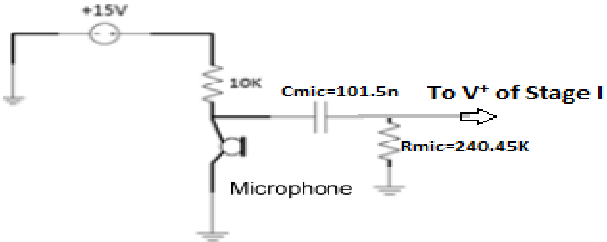
Figure 8: Output signal plotted vs Input signal for Stage IV

RVAR (kΩ)	104.5	54.1	33.4	15.6	1.63	0.975
$\lvert GAIN \rvert_{expected}$	52.50	27.20	16.78	7.84	0.82	0.49
$\lvert GAIN \rvert_{measured}$	53.99	28.03	16.58	8.10	0.82	0.46

Table 3: Actual components measured

Note: The fact that we invert our signal will not affect output sound, as audio is phase independent.

Having established that the amplifier works as expected we attempted to connect the microphone as proposed in page 10 of "R. Dickinson (2016) *Bioengineering EE Practical on Op-Amps* v4.6". However, as our first stage is a non-inverting amplifier the bias current has no route to ground and charges the capacitor. Thus when enough time passes, the signal saturates. To mitigate this, we connect a resistor from the V^+ input of the op amp to ground.

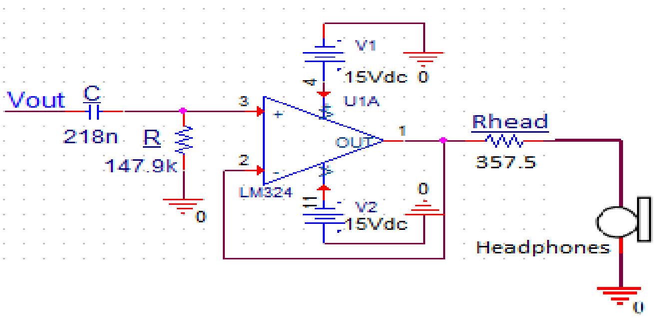


Schematic 3: Microphone stage implemented in the lab

However, C_{mic} and R_{mic} form a 1st order passive high pass filter. In order for our passband to not be attenuated, we therefore need f_C << 20Hz so we choose R_{mic}=240.45kΩ, C_{mic}=101.5nF which results in f_C=6.52Hz.

We observe that, as expected, the 12V DC offset that is present in the microphone input is completely attenuated by the time it reaches V^+ of the Stage I amplifier.

Finally, we need to connect the headphones to the output of our amplifier. It is proposed that we connect the output of our last stage first to a 220nF capacitor (to remove any DC offset) and then in series with a 1kΩ resistor and our headphones. However by doing so we create a filtering effect that is equivalent to that of a 1st order passive high pass filter with $f_c = \frac{1}{2 \cdot \pi \cdot 220nF \cdot (1k\Omega + Z_H)} = 715Hz$ which will greatly attenuate our passband.



Schematic 4: Headphone stage implemented in the lab

We therefore decided to remove the output DC offset by introducing a low cut off passive high pass filter and a buffer (Schematic 4). This will result in a minimal output signal DC offset (<10mV) and $V_H = V_{OUT} \cdot Z_h/(R+Z_H)$ which is independent of frequency since we found Z_H to be constant for f < 1kHz. We choose R=150kΩ in order to have a cut-off at 4.82Hz. Finally, we need to calculate the minimum value of R_{Head} for which the output current of the LM324 and the power across the headphones are both within limits. Since we need:

$$I_{Max} = 40\text{mA} > \frac{V_{out_{max}}}{R_{min} + Z_h} \; and \; P_{H_{max}} = 60\text{mW} > \frac{{V_{out_{max}}}^2 \cdot Z_h}{(R_{min} + Z_h)^2}$$

We find that for R_{Head}=360Ω both the headphones and opamp are not in risk of damage.

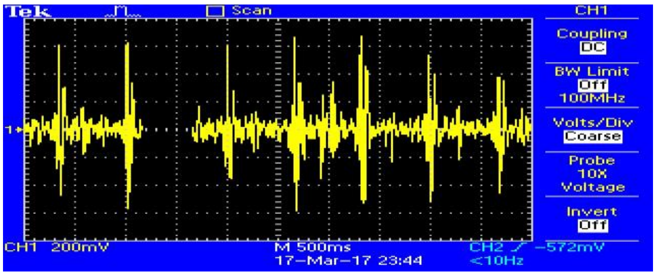


Figure 9A-Waveform after Stages I, II, III (V_C)

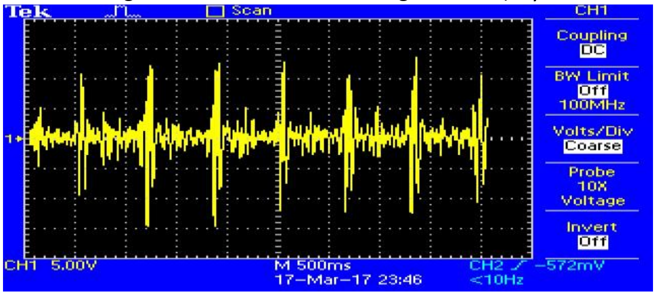


Figure 9B-Waveform after Stage IV amplification (V_{OUT})

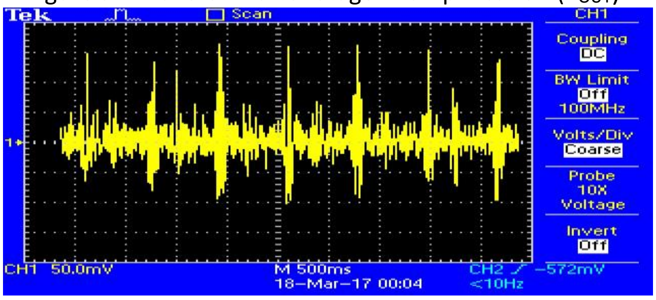


Figure 9C-Waveform across the headphones (V_H)

Figures 9A to 9C show the heart sound waveform of a subject as measured at different points of our system. The potentiometer was set to 25.6kΩ so as to accommodate our particular microphone. Each waveform exhibits the characteristics expected and the output was clearly audible. It was possible to distinguish between the "lub" and "dub" sounds by ear as is visible in Figure 9C.

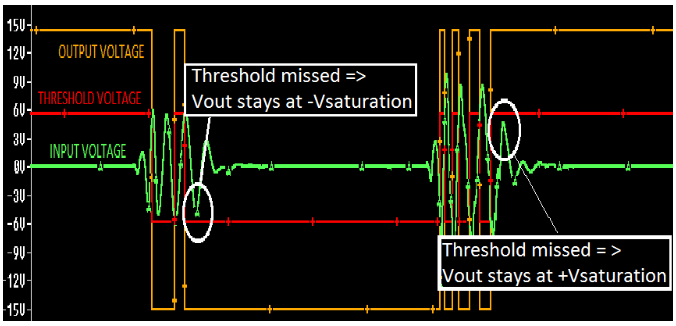
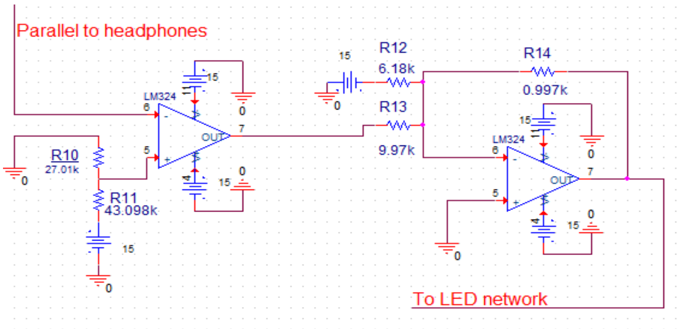


Figure 10: Schmitt trigger Pspice simulation

To digitize our output signal (V_{OUT}) we used an inverting comparator and an inverting summing amplifier. We initially considered using a Schmitt trigger, however when simulating the circuit we found that because our input waveform is not completely symmetric, the negative and positive peaks may differ thus producing errors such as in the example above (PSpice simulation). We may be able to tune the Schmitt trigger to produce accurate results but will be much harder.



Schematic 5: Inverting Comparator and inverting summing amplifier

R10=27.01kΩ	R11=43.98kΩ	R12=6.18kΩ	R13=9.99KΩ	R14=0.99kΩ

Table 4: Actual components measured

Comparator reference Voltage= 15V
Comparator threshold=15·R10/(R10+R11)=5.8V
Comparator output= {-15V if V_{OUT} > 5.8V or 15V if V_{OUT} < 5.8V}
Inverting Summing amplifier reference Voltage = -15V
Inverting summing amplifier output = $V_{DIGITAL} \Rightarrow$
$V_{DIGITAL} = -R14 \cdot \left(\frac{Comparator\ Output}{R13} + \frac{-15}{R12}\right) \Rightarrow$
$V_{DIGITAL}$ = {1V for V_{out} < 5.8V and 4V from V_{out} > 5.8V}
which is connected to a series resistor and then an LED that flashes on each heartbeat.

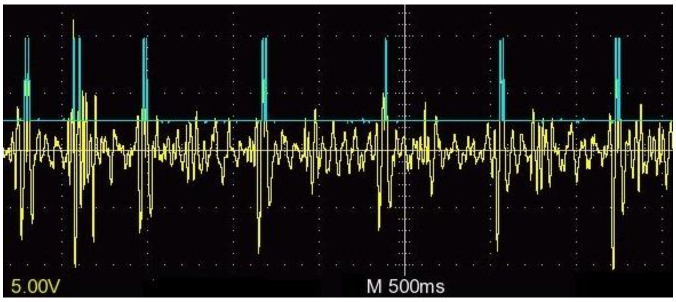


Figure 11: Heart Sound waveform after comparator and inverting summing amplifier [Blue - digital output, Yellow – V_{OUT}]

IV. Improvements

Since we have introduced a variable gain stage to our system in order to accommodate a range of input signals, we need an indicator of the value of R_{VAR} at which the signal starts clipping. For this purpose we have created a second comparator (in parallel to the headphone network), with a threshold of 12V and connected it to an LED network. Then whenever the LED flashes we know we are close to clipping and we need to stop increasing R_{VAR}. We could also have placed a potential divider before the headphone network to serve as volume control. A further improvement would be to input our digital output to an Arduino which would allow us to measure heart rate and print its value on an LCD screen.

Lastly, we realised that the number of opamps used in our design is larger than necessary. First of all, having such a high input from our microphone would have allowed us to completely remove the first amplification stage. Moreover, our design contains two passive high pass filters (one at the microphone stage and one at the headphone stage). By setting their cut-off frequency to 20Hz we can remove the active high pass filter. By using a large enough capacitor at the headphone stage we can remove the buffer and connect the headphones as proposed in the lab manual while still maintaining a cut-off frequency of 20Hz and having no DC offset in our output signal. ($C_{necessary}$=1/[2·π·(R_{head}+Z_H)·20]=22.1μF).

Thus the heart sound amplifier can be implemented by using only two opamps - one for low pass filtering and one as a variable gain stage.

V.References

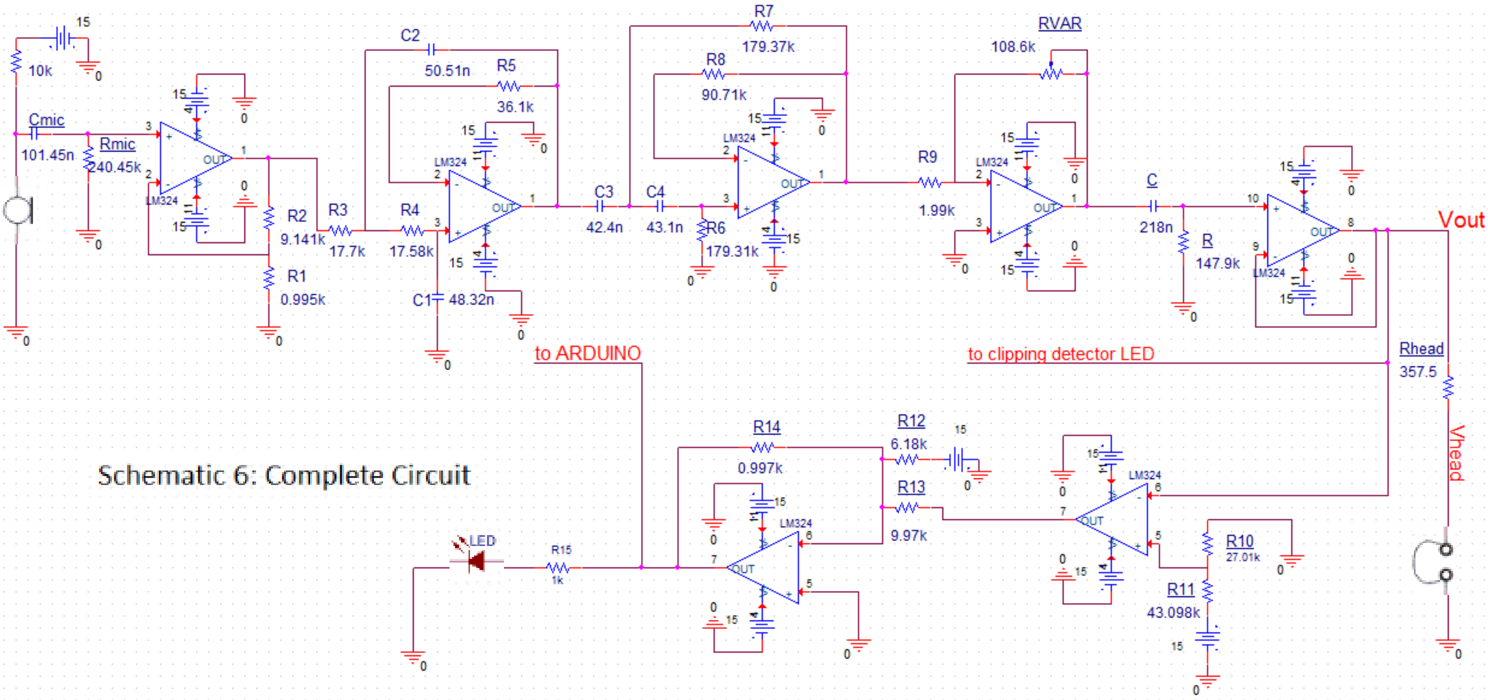
[1] - Imperial College London(2017) *1st year SPICE lab 2017 Edition* version 7.0.6

[2] - P.J. Arnott, G.W. Pfeiffer,M.E. Tavei(April 1984) *Spectral Analysis of Heart Sounds*, J. Biomed Eng. Vol. 6.

[3] - Texas Instruments Incorporated (2015) *LMx24-N, LM2902-N Low-Power, Quad-Operational Amplifiers*. Available from: http://www.ti.com/lit/ds/symlink/lm2902-n.pdf

[4] - Texas Instruments Incorporated, Karki J. (2002) *Analysis of the Sallen-Key Architecture.* Available from: http://www.ti.com/lit/an/sloa024b/sloa024b.pdf

[5] - R.Dickinson (2016) *Bioengineering EE Practical on Op-Amps* v4.6.



Schematic 6: Complete Circuit