Sensing and Measurement Project
Stethoscope

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Spring 2023

Abstract

This project presents the design and simulation of a stethoscope circuit using Proteus 8.15 software. The stethoscope circuit is designed to amplify and filter audio signals within the frequency range of 20 Hz to 200 Hz, which is essential for effectively capturing and analyzing heart and lung sounds. The circuit consists of several key components, including a band-pass filter, and a differential amplifier. The piezoelectric sensor captures the acoustic signals and converts them into electrical signals. The non-inverting amplifier is used to amplify the weak audio signals from the sensor, ensuring optimal signal strength for further processing. The band-pass filter is implemented to attenuate unwanted noise and interference outside the desired frequency range, allowing clear and focused monitoring of the targeted sounds. Furthermore, another circuit with higher order band-pass filter is simulated and described for simulating conditions with noises and artifacts.

1. Introduction

A stethoscope is a medical instrument used by healthcare professionals, such as doctors, nurses, and medical practitioners, to listen to sounds within the body. It consists of a long, flexible tube attached to two earpieces and a chest piece. The chest piece has a diaphragm and a bell that are used to transmit and amplify sounds.

Fig. 1.(Britannica)

The primary purpose of a stethoscope is to auscultate, which means to listen to internal sounds of the body, particularly those produced by the heart, lungs, and other organs. By placing the chest piece on different areas of the body, healthcare professionals can hear various sounds, such as heartbeats, lung sounds, and bowel movements.

The stethoscope is an essential tool in diagnosing and monitoring patients. It allows healthcare professionals to detect abnormalities or irregularities in the sounds produced by the body. For example, they can identify abnormal heart murmurs, lung infections, or bowel obstructions by listening to the specific sounds generated by these conditions. Stethoscopes also help in evaluating blood flow, assessing blood pressure, and monitoring fetal heartbeats during pregnancy.

A conventional stethoscope has been generally used to hear the sound of a human heartbeat. Although the traditional systems have had the advantage of low cost and easy operation, conventional auscultation of the heart has been limited by several factors. The heart signal is a combination of high and low frequencies with low amplitude; thus, the interpretation factor of heart signal on conventional stethoscope is very subjective and much dependent on the physician’s experience, skills, hearing. Therefore, it is necessary for sensors in the electronic stethoscope to have high selective sensitivity without any noise, which is fatal to the accuracy of the diagnosis. Therefore, it is vital to advance heart auscultation in order to determine the initial diagnosis of a heart condition.

Modern and new stethoscopes typically work on the same basic principle as traditional stethoscopes, but they often incorporate advanced technology to enhance their functionality. Here is an overview of how modern stethoscopes work and what they consist of:

1. **Chest Piece:** The chest piece of a modern stethoscope contains a diaphragm and a bell. The diaphragm is a flat, circular surface that is sensitive to high-frequency sounds such as heart and lung sounds. The bell is a smaller, cup-shaped attachment used to detect low-frequency sounds. Some modern stethoscopes have an adjustable diaphragm that can switch between high and low frequencies by applying different levels of pressure.
2. **Tubing:** The tubing connects the chest piece to the earpieces and allows the transmission of sound. In newer stethoscope models, the tubing is often made from high-quality materials that minimize external noise interference and provide better sound transmission.
3. **Noise Reduction Technology:** Many modern stethoscopes incorporate noise reduction technology to filter out ambient noise and focus on the specific sounds of interest. This helps improve the clarity and accuracy of auscultation.
4. **Amplification and Electronic Features:** Some advanced stethoscope models have electronic components that amplify the sounds and provide additional features. These electronic stethoscopes may have adjustable volume levels, filters to enhance specific frequencies, and even the ability to record and store auscultation sounds for later analysis or sharing with colleagues.
5. **Wireless and Digital Connectivity:** Recent innovations in stethoscope technology include wireless connectivity and digital features. Wireless stethoscopes can transmit the auscultated sounds to a computer or smartphone in real-time, allowing for remote monitoring, telemedicine consultations, and data analysis. Digital stethoscopes may have built-in visual displays or interfaces that show sound waveforms, heart rate readings, or other relevant data.
6. **Additional Attachments:** Some modern stethoscopes come with interchangeable attachments or specialized features for specific medical purposes. For example, pediatric stethoscopes may have smaller chest pieces designed for infants and children, while cardiology stethoscopes often have enhanced acoustics for detecting subtle heart murmurs.

The below image shows different parts of the heart:

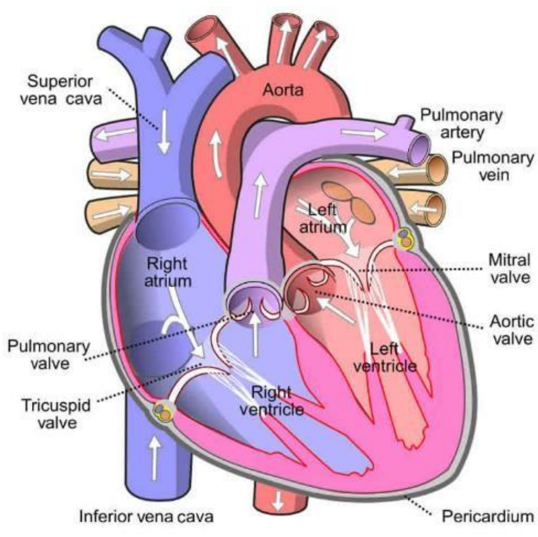


Fig. 2. ([Health Jade])

Normal heart sounds have a frequency range between 20 Hz to 200 Hz. The heart sounds consist of S1 and S2 related to valves. S1 and S2 are the “Lub” and “Dub” sound, respectively and corresponds to the two main components of the cardiac cycle, which is the rhythmic contraction and relaxation of the heart:

- ***"Lub" (S1):*** The first sound, commonly referred to as "lub" is caused by the closure of the mitral and tricuspid valves. These valves separate the upper chambers (atria) from the lower chambers (ventricles) of the heart. When the ventricles contract and blood is forced out, the mitral and tricuspid valves close, preventing blood from flowing back into the atria. The closure of these valves produces the first heart sound, which is a relatively low-frequency sound.
- ***"Dub" (S2):*** The second sound, often referred to as "dub" is caused by the closure of the aortic and pulmonary valves. These valves separate the ventricles from the major blood vessels leaving the heart the aorta and the pulmonary artery, respectively. After the ventricles have contracted and pumped blood out, the aortic and pulmonary valves close to prevent blood from flowing back into the ventricles. The closure of these valves produces the second heart sound, which is a relatively higher-frequency sound.

So, the "lub-dub" sound you hear when listening to the heart with a stethoscope represents the closing of the heart valves during the cardiac cycle. The first sound is due to the closure of the mitral and tricuspid valves (S1), and the second sound is due to the closure of the aortic and pulmonary valves (S2). These sounds provide valuable information about the functioning and condition of the heart when evaluated by healthcare professionals.

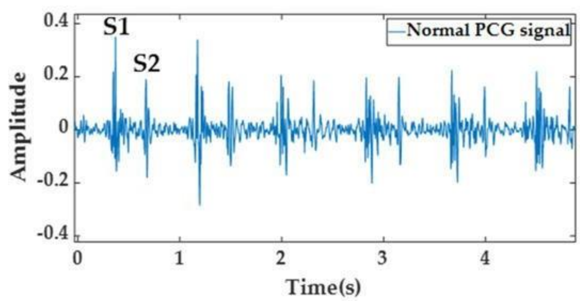


Fig. 3.(MDPI)

Our stethoscope utilizes a piezoelectric sensor to capture the heart rate signal and then transform it into an electrical signal, and later, the heartbeat signal will be visualized on a graph. Electronic stethoscopes converting acoustic sound waves into electrical signals must be amplified for optimal listening while processing. In visualizing the heart rate signal using a series of electrical signal conditioners, the signal is sensed by the sensor and then pre-amplified by the amplifier circuit and filtered to the frequency according to the desired sound characteristics using a filter circuit. These analog circuits include pre-amplifiers, Low Pass Filters (LPF) with a 200 Hz cutoff frequency, and High Pass Filters (HPF) with a frequency cut of 20 Hz. LPF will pass signals with frequencies below 200 Hz and cut off signals with frequencies above 200 Hz, while HPF will pass the signal with a frequency above 20 Hz and cut off the signal with a frequency below 20 Hz, since the main frequency components of the heart sound signal are in the range of 20-200 Hz. If the frequency was < 30 Hz, then it belongs to the category of the abnormal heart sound, as third and fourth heart sounds and the diastolic murmur of mitral stenosis. The murmur with the highest frequency noise is aortic regurgitation, the dominant frequencies around 400 Hz. Other sounds and murmurs have the major frequencies between 100 and 400 Hz. In our project, we are going to design a circuit which receives a noisy signal from a piezoelectric sensor and produces the sound of the heart in our desired range.

In the following, we will the explain which parts a stethoscope circuit should have, the specifications of its signals, which software did we use, our design and the results.

2. Design

Electronic stethoscopes offer several advantages over conventional acoustic stethoscopes. Some of these advantages include:

1. **Amplification of Sounds:** Electronic stethoscopes have the ability to amplify body sounds, making it easier for healthcare professionals to hear faint or subtle sounds. This can be particularly beneficial in noisy environments or when examining patients with certain conditions that produce low-intensity sounds.
2. **Noise Reduction:** Electronic stethoscopes often come equipped with noise reduction technology, which helps filter out background noise and improve the clarity of auscultated sounds. This can be particularly useful in busy clinical settings or emergency situations where ambient noise can interfere with accurate diagnosis.
3. **Frequency Control:** Electronic stethoscopes typically have frequency control features that allow healthcare professionals to adjust the frequency response according to the specific body sounds they want to focus on. This versatility enables better customization based on the patient's condition and the area of auscultation.
4. **Recording and Playback:** Many electronic stethoscopes have the capability to record and store auscultated sounds. This feature allows healthcare professionals to review the recorded sounds later, share them with colleagues for consultation, or use them for educational purposes. It can also serve as a helpful reference for tracking changes in a patient's condition over time.
5. **Telemedicine and Remote Monitoring:** Electronic stethoscopes with wireless connectivity enable remote auscultation, telemedicine consultations, and remote patient monitoring. Healthcare professionals can transmit the auscultated sounds to a remote location in real-time, allowing for expert consultation or monitoring of patients who may be located far away.
6. **Visual Displays and Data Integration:** Some advanced electronic stethoscopes come with visual displays that show sound waveforms, heart rate readings, or other relevant data. This visual feedback can aid in the interpretation of auscultation findings and enhance the diagnostic process.
7. **Educational Tools:** Electronic stethoscopes can serve as valuable educational tools, particularly for training healthcare professionals. The ability to record, playback, and analyze heart and lung sounds helps in teaching and improving auscultation skills.

It's important to note that while electronic stethoscopes offer these advantages, acoustic stethoscopes still have their own benefits, such as simplicity, reliability, and cost-effectiveness. The choice between the two types of stethoscopes often depends on the specific needs of the healthcare professional and the clinical setting.

The below circuit shows a simple electronic stethoscope circuit:

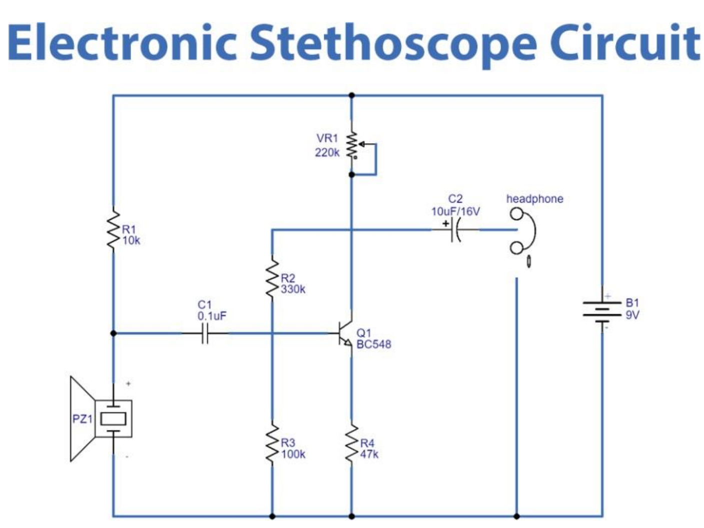


Fig. 4.(CIRCUITS DIY)

Hardware required for this circuit:

Sr No:	Components	Qty
1	TDA2005	1
2	Resistor (10K,100K,47K,330K)	1,1,1,1
3	Capacitor (0.1uf)	1
4	Speaker & Head phone	1,1
5	bc548 Transistor	1
6	Variable Resistor 220K	1

Fig. 5.(CIRCUITS DIY)

In this simple electronic stethoscope circuit, a microphone is made out of piezo speakers. This transducer has a low-frequency response power capability of up to 100mV. This signal is incredibly weak. Therefore, to boost the signal, we must employ a high-impedance input impedance preamplifier. The piezo speaker (PZ1) would then detect the heartbeats and drive a headphone output. The signal is then delivered to a signal amplifier made up of Q1, R1–R4, VR1, and C1. Others referred to it as a common-emitter amplifier. The emitter is connected to both the input and output signals.

It is capable of producing responses at low frequencies. As a result, the output has a low-impedance connection to the headset. The potentiometer VR1 acts as an output adjuster to loud or low so that the heartbeat can be heard. The amplifier's input impedance can be raised using a Darlington transistor.

Stethoscopes can be susceptible to various artifacts and external noises that can affect the quality and accuracy of the sounds being heard. Some common artifacts and external noises that can impact stethoscope performance include:

- **Ambient Room Noise:** Background noise present in the environment, such as conversations, equipment noise, or other activities, can interfere with the clarity of the internal body sounds being detected by the stethoscope.
- **Contact Noise:** The stethoscope may pick up noise from physical contact with external objects. For example, the rubbing of clothing, movement of the stethoscope tubing, or accidental contact with nearby surfaces can produce noise that interferes with the desired sounds.
- **Electrical Interference:** Stethoscopes that incorporate electronic components or amplification circuits may be susceptible to electrical interference. This interference can come from nearby electrical equipment, power sources, or electromagnetic fields, leading to unwanted noise or distortion in the signal.
- **Tubing Noise:** Stethoscope tubing can sometimes generate noise when it rubs against clothing, skin, or other surfaces. This noise can be transmitted to the earpieces and interfere with the clarity of the internal body sounds.
- **Patient Movement:** When a patient moves during auscultation, it can create noise and artifacts. For example, shuffling or rustling sounds caused by the patient's clothing or body movements can mask or distort the desired sounds.
- **Respiration Noise:** During lung auscultation, respiratory noises, such as breathing sounds or the movement of air through the airways, can interfere with the detection of subtle abnormalities or faint sounds.
- **Environmental Factors:** Environmental factors, such as high humidity or temperature, can affect the stethoscope's performance and may introduce additional noise or distortion.

To minimize the impact of these artifacts and external noises, healthcare professionals can take several measures, including: (1) Choosing a quiet environment for auscultation whenever possible, (2) Ensuring proper positioning of the stethoscope on the patient's body to minimize contact noise and movement artifacts, (3) Using high-quality stethoscopes with good noise isolation properties, (4) Employing noise reduction or filtering techniques in the stethoscope circuitry, if available, (5) Educating patients on minimizing movement and extraneous noise during the examination.

By being mindful of these potential artifacts and external noises and taking appropriate measures, healthcare professionals can optimize the listening experience and improve the accuracy of their assessments using a stethoscope.

Generally, the voltage range of the input signal in a stethoscope can be quite small, on the order of microvolts (µV) to millivolts (mV). The mechanical vibrations of the internal body sounds, such as the heartbeat or lung sounds, are converted into corresponding electrical signals through the transducer. Therefore, after receiving the input signal we used a preamplifier. As we mentioned in previous sections, the main frequency components of the heart sound signal are in the range of 20-200 Hz. Hence, after preamplifier, we need a band-pass filter in order to extract the frequency components of the input signal within this range.

One way of creating a band-pass filter is to cascade a HPF (High Pass Filter) and LPF (Low Pass Filter). The order of cascading does not matter, that is, either LPF can placed first and HPF later or vice versa. The below circuit shows circuit diagram of 2nd order active Band Pass Filter by cascading a 2nd order active HPF with a 2nd order active LPF.

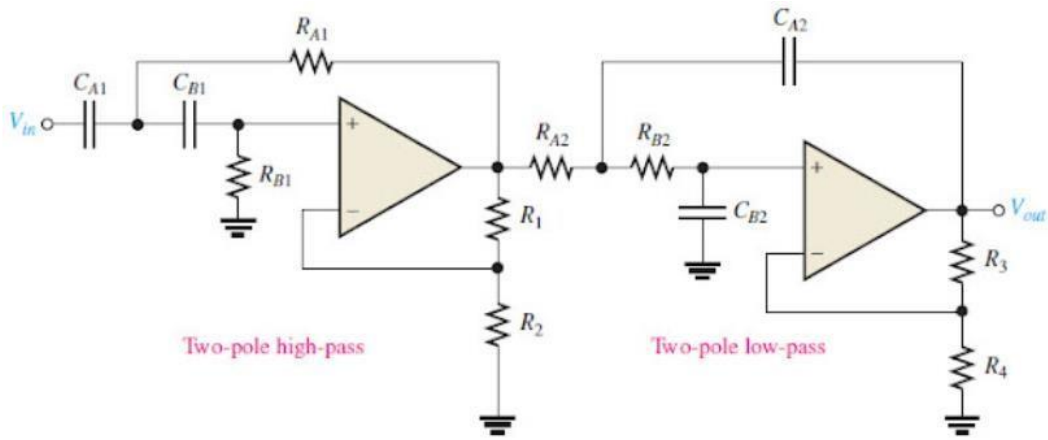


Fig. 6.(ee_diary)

In the above circuit diagram, both the low pass filter and high pass filter are using Butterworth Sallen-Key topology. The Sallen-Key filter is a second-order filter, meaning it has a roll-off slope of -40 dB per decade or -12 dB per octave. It is often used in audio applications for tasks such as low-pass, high-pass, band-pass, or band-reject filtering. The filter's transfer function can be customized to achieve the desired frequency response by selecting appropriate component values. The advantages of the Sallen-Key filter include simplicity of design, low component count, and ease of customization. However, it also has some limitations, such as limited frequency response range and sensitivity to component tolerances.

The frequency response graph of the band-pass filter is shown below.

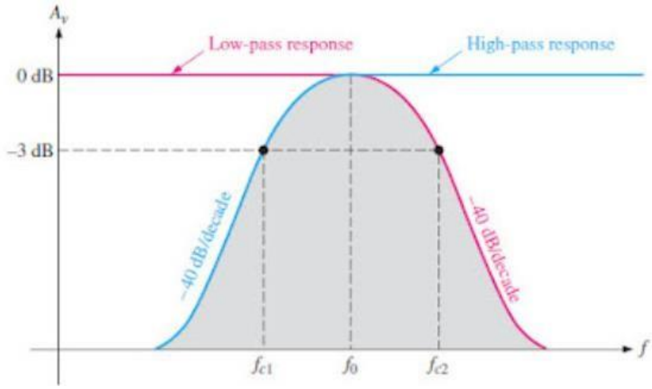


Fig. 7.(ee_diary)

The lower cutoff frequency fc1 is due to high pass filter and is given by the following equation,

$$f_{c1} = \frac{1}{2\pi\sqrt{R_{A1}R_{B1}C_{A1}C_{B1}}}$$

Similarly, the higher cutoff frequency fc2 is due to low pass filter and is given by the following equation,

$$f_{c2} = \frac{1}{2\pi\sqrt{R_{A2}R_{B2}C_{A2}C_{B2}}}$$

After filtering out the unwanted frequencies, we used an amplifier in order to amplify the amplitude of the signal to our desired range.

We will do the calculations for the 2nd order low-pass butterworth filter (high-pass is similar to this):

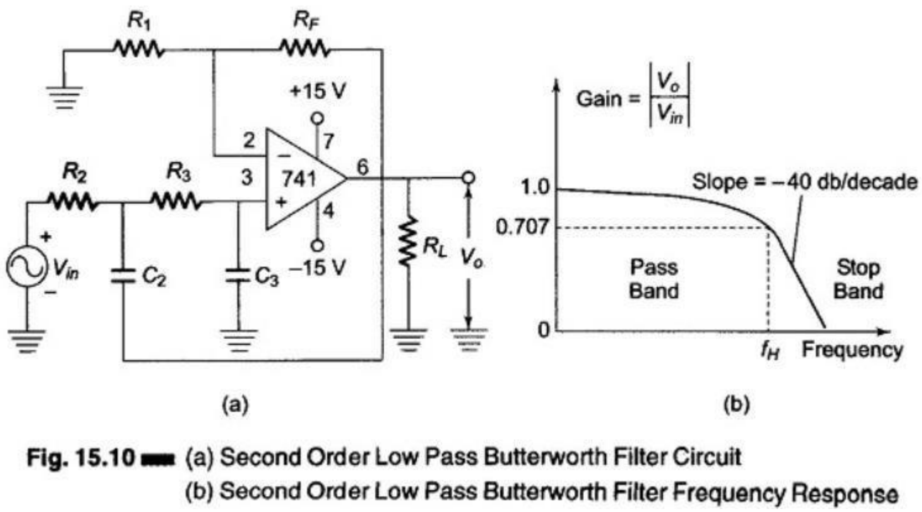


Fig. 8.(EEGUIDE.COM)

The gain of the second order filter is set by R1 and RF, while the high cutoff frequency fH is determined by R2, C2, R3 and C3 as given below.

$$f_H = \frac{1}{2\pi\sqrt{R_2R_3C_2C_3}}$$

The voltage gain magnitude equation for a second order low pass Butterworth response is given by:

$$\left|\frac{V_o}{V_{in}}\right| = \frac{A_F}{\sqrt{1+(\frac{f}{f_H})^4}}$$

where

- AF = 1+ RF/R1 = pass band gain of the filter
- f = frequency of the input signal, in Hz
- f_H = High cutoff frequencies, in Hz

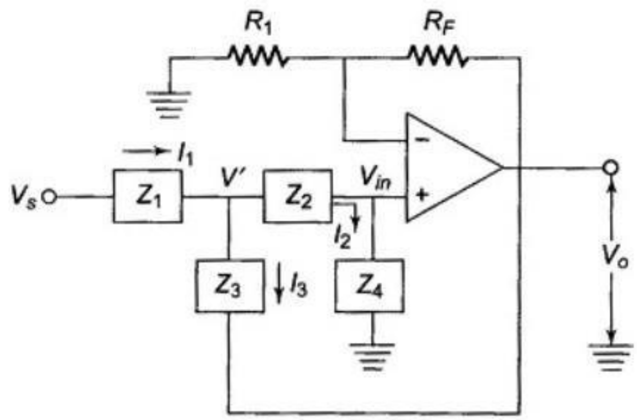


Fig. 9.(EEGUIDE.COM)

Referring to above circuit:

$$I_1 = I_2 + I_3$$

$$I_1 = \frac{V_s - V'}{Z_1}$$

$$I_2 = \frac{V' - V_{in}}{Z_2} = \frac{V'}{Z_2 + Z_4}$$

$$I_3 = \frac{V' - V_o}{Z_3}$$

Substituting the second, third and fourth equation in the first one, we have:

$$\frac{V_s - V'}{Z_1} = \frac{V'}{Z_2 + Z_4} + \frac{V' - V_o}{Z_3}$$

Also:

$$V_{in} = \left(\frac{Z_4}{Z_2 + Z_4}\right) \times V', \text{ therefore } V' = \frac{Z_2 + Z_4}{Z_4} \times V_{in}$$

But the open loop voltage gain Avo = Vo/Vin, therefore:

$$V_{in} = \frac{V_o}{A_{V_o}}$$

Therefore,

$$V' = \left(\frac{Z_2 + Z_4}{Z_4}\right) \times \frac{V_o}{A_{V_o}}$$

Substituting for Vin we have:

$$\frac{V_s - \frac{Z_2 + Z_4}{Z_4} \times \frac{V_o}{A_{Vo}}}{Z_1} = \frac{\frac{Z_2 + Z_4}{Z_4} \times \frac{V_o}{A_{Vo}}}{Z_2 + Z_4} + \frac{\frac{Z_2 + Z_4}{Z_4} \times \frac{V_o}{A_{Vo}} - V_o}{Z_3}$$

$$\frac{V_s}{Z_1} - \frac{Z_2 + Z_4}{Z_1 Z_4} \times \frac{V_o}{A_{Vo}} = \frac{Z_2 + Z_4}{Z_4(Z_2 + Z_4)} \times \frac{V_o}{A_{Vo}} + \frac{Z_2 + Z_4}{Z_3 Z_4} \times \frac{V_o}{A_{Vo}} - \frac{V_o}{Z_3}$$

$$\frac{V_s}{V_1} - \frac{Z_2 + Z_4}{Z_1 Z_4} \times \frac{V_o}{A_{Vo}} = \frac{1}{Z_4} \times \frac{V_o}{A_{Vo}} + \frac{Z_2 + Z_4}{Z_3 Z_4} \times \frac{V_o}{A_{Vo}} - \frac{V_o}{Z_3}$$

$$\frac{V_s}{Z_1} = \frac{V_o}{A_{Vo}} \left[\frac{1}{Z_4} + \frac{Z_2 + Z_4}{Z_1 Z_4} + \frac{Z_2 + Z_4}{Z_3 Z_4} - \frac{A_{Vo}}{Z_3}\right]$$

$$\frac{V_s}{Z_1} = \frac{V_o}{A_{Vo}} \left[\frac{Z_1 Z_3 + Z_3(Z_2 + Z_4) + Z_1(Z_2 + Z_4) - A_{Vo}(Z_1 Z_4)}{Z_1 Z_3 Z_4}\right]$$

$$\frac{V_s}{Z_1} = \frac{V_o}{A_{Vo}} \left[\frac{Z_1 Z_3 + Z_2 Z_3 + Z_3 Z_4 + Z_1 Z_2 + Z_1 Z_4 (1 - A_{Vo})}{Z_1 Z_3 Z_4}\right]$$

$$\frac{V_s}{Z_1} = \frac{V_o}{A_{Vo}} \left[\frac{Z_3 (Z_1 + Z_2 + Z_4) + Z_1 Z_2 + Z_1 Z_4(1 - A_{Vo})}{Z_1 Z_3 Z_4}\right]$$

$$\therefore \quad V_s = \frac{V_o}{A_{Vo}} \left[\frac{Z_3(Z_1 + Z_2 + Z_4) + Z_1 Z_2 + Z_1 Z_4(1 - A_{Vo})}{Z_3 Z_4}\right]$$

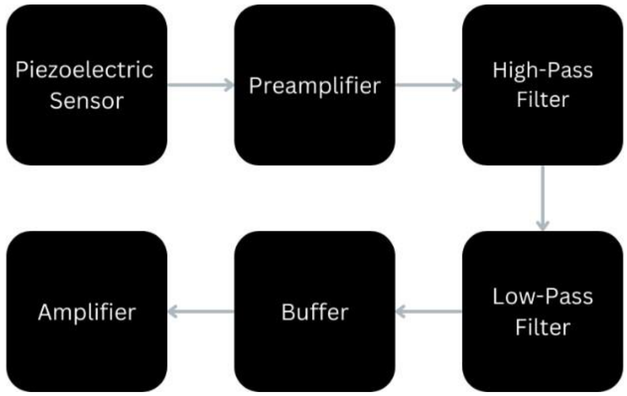
$$\therefore \quad \frac{V_o}{V_s} = \frac{Z_3 Z_4 A_{Vo}}{(Z_1 + Z_2 + Z_4) Z_3 + (1 - A_{Vo}) Z_1 Z_4 + Z_1 Z_2}$$

Which is the transfer function of our filter. For a low-pass filter Z1 and Z2 are resistors and Z3 and Z4 are capacitors, where s is the transfer function. Therefore:

$$A\ v(s) = \frac{A_{V_o} \times 1/sC_1 \times 1/sC_2}{(R_1 + R_2 + 1/sC_2)\, 1/sC_1 + (1 - A_{V_o})\, R_1/sC_2 + R_1 R_2}$$

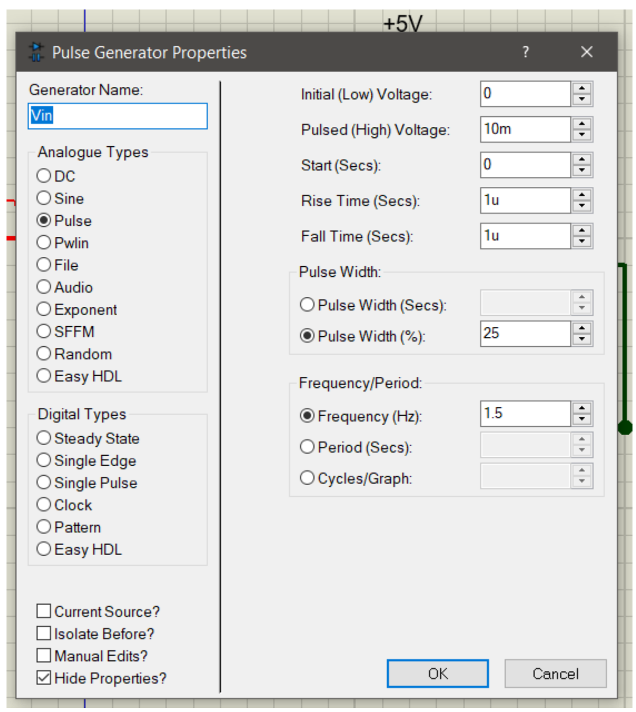
3. Project

As mentioned earlier, we have designed a stethoscope circuit using Proteus 8.15. Our circuit consists of six major parts: (1) Piezo Sensor, (2) Preamplifier, (3) Low-Pass Filter, (4) High-Pass Filter, (5) Buffer, (6) Amplifier. The block diagram of our circuit:



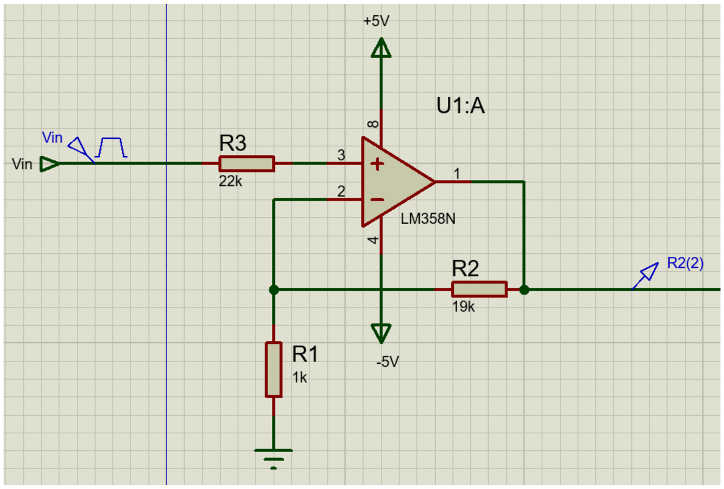
Proteus 8.15 Professional is a specific version of the Proteus software, which is an electronic design automation (EDA) tool for circuit design and simulation. It is one of the most popular and widely used versions of Proteus. Proteus 8.15 Professional is an advanced software package that provides a comprehensive set of features for designing, simulating, and testing electronic circuits. It is primarily aimed at professional engineers, electronics enthusiasts, and researchers who require a powerful tool for their circuit design projects.

As we mentioned before, the input signal is received from a piezoelectric sensor. This sensor converts mechanical stress into voltage. The characteristics of the input signal in our circuit:



Pulse voltage generator with amplitude of 10mV, pulse width of 25% and 1.5Hz frequency.

Then the signal is pre-amplified using a non-inverting amplifier:

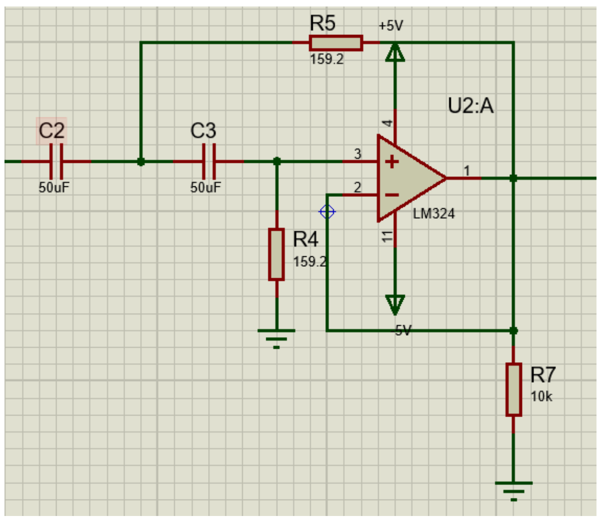


We used a LM358N operational amplifier for this part. The gain for this amplifier is calculated from below equation:

$$gain = 1 + \frac{R_2}{R_1} = 1 + \frac{19k}{1k} = 20$$

Therefore, this pre-amplifier amplifies the input signal from piezoelectric sensor by 20. We put R3 in positive port of the op-amp in order to remove the offset from output signal.

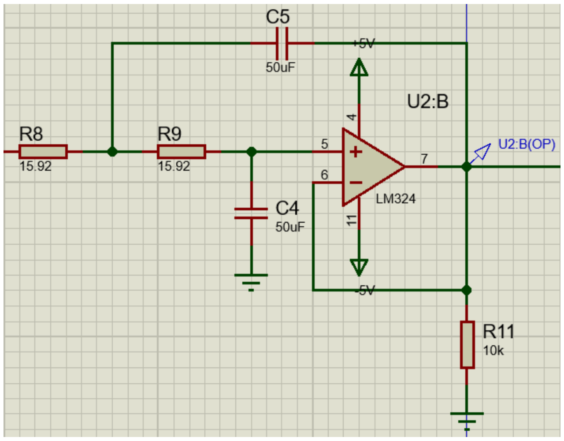
The next stage is a high-pass filter. As we mentioned before, we used a Sallen-Key filter. We defined the values of capacitors and resistors for 20Hz cutoff frequency.



Cutoff frequency:

$$f_{c1} = \frac{1}{2\pi\sqrt{(159.2)(159.2)(50\mu)(50\mu)}} \approx 20Hz$$

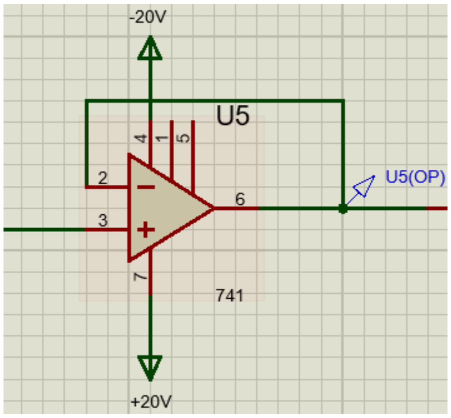
The next stage after high-pass filter is the low-pass filter which also is a Sallen-Key filter.



Cutoff frequency:

$$f_{c2} = \frac{1}{2\pi\sqrt{(15.92)(15.92)(50\mu)(50\mu)}} \approx 200Hz$$

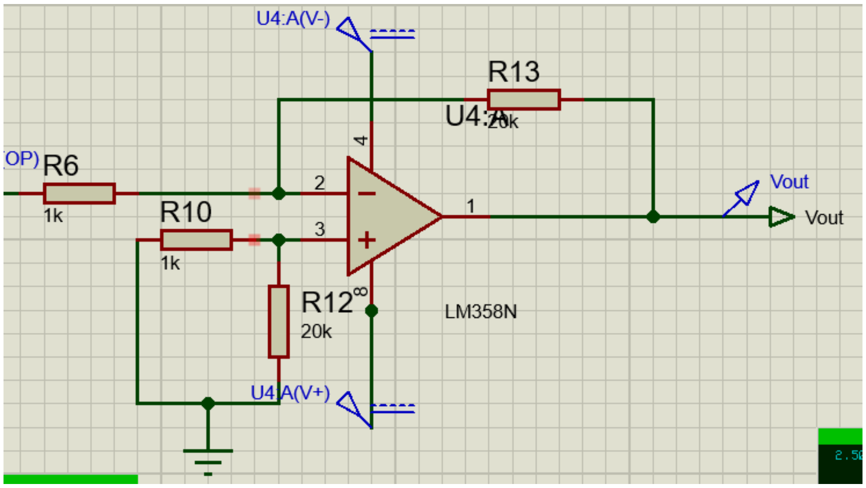
This HPF and LPF create a band-pass filter together. The bandwidth of this band-pass filter is 180Hz (200-20). The next stage is a buffer(voltage-follower):



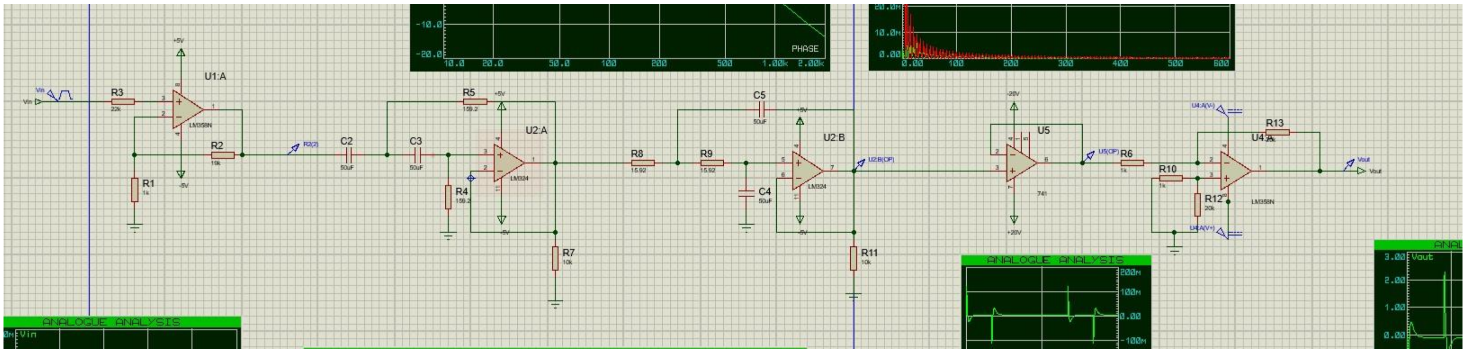
There are some benefits in using buffer between two stages:

1. **Impedance matching:** A buffer is often used to match the impedance between two circuits or components. When the output impedance of a source is significantly higher than the input impedance of the load, a buffer can be inserted in between to isolate the two and ensure proper signal transfer without loss or distortion. The buffer presents a high input impedance and a low output impedance, effectively bridging the impedance gap.
2. **Signal isolation:** Buffers can provide signal isolation, protecting sensitive circuitry from being affected by the source circuit. By using a buffer, the output circuit is decoupled from the input circuit electrically, preventing any potential interference or loading effects. This is particularly useful when driving multiple loads or when connecting circuits with different voltage levels.
3. **Voltage level shifting:** Buffers can be utilized to shift voltage levels between circuits. For example, if a signal needs to be translated from a lower voltage range to a higher voltage range, a buffer with appropriate gain can amplify the signal to the desired level without altering its waveform. Conversely, a buffer can also be used to attenuate or decrease the voltage level if necessary.
4. **Impedance isolation:** Buffers can isolate the input and output impedances of a circuit. The buffer's high input impedance prevents loading effects on the source, ensuring that the original circuit remains unaffected. At the same time, the buffer's low output impedance allows it to drive a load with minimal signal degradation, overcoming any impedance mismatch between the buffer and the load.

The final stage is an amplifier which amplifies the output signal to our desired range (from millivolts to volts):



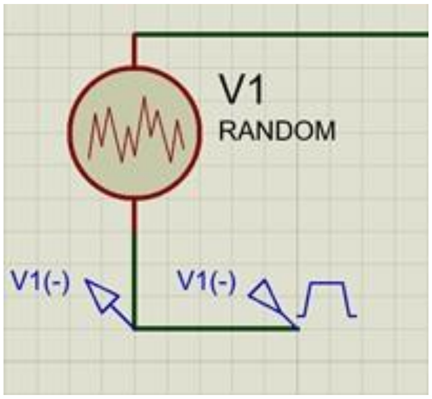
The entire circuit:



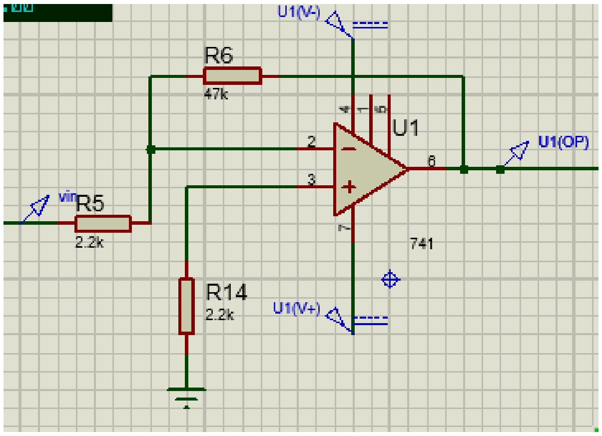
Additional circuit:

As an additional work, we considered the fact that in reality, the input signal is always noisy. We designed another circuit, which is effective for detecting noisy signals of the heartbeat. We also increased the order of our filters to 4. Although the circuit will be more complicated to analyze, it will be more effective.

First, we used a pulse voltage generator with 10mV amplitude, 25% pulse width and 1.5Hz frequency in order to simulate the heartbeats. Then, we made this series with a random voltage generator in order to add the effect of noise (the random voltage generator has 2.5mV amplitude and 300Hz frequency):



In our design for the circuit, we first used a pre-amplifier circuit. As mentioned before, the amplitude of the input signal is too small. This amplification makes the signal ready for filtering. It also compensates for possible unwanted attenuations during the filtering stages and avoids data loss. Our design for the pre-amplifier circuit is a simple inverting amplifier:

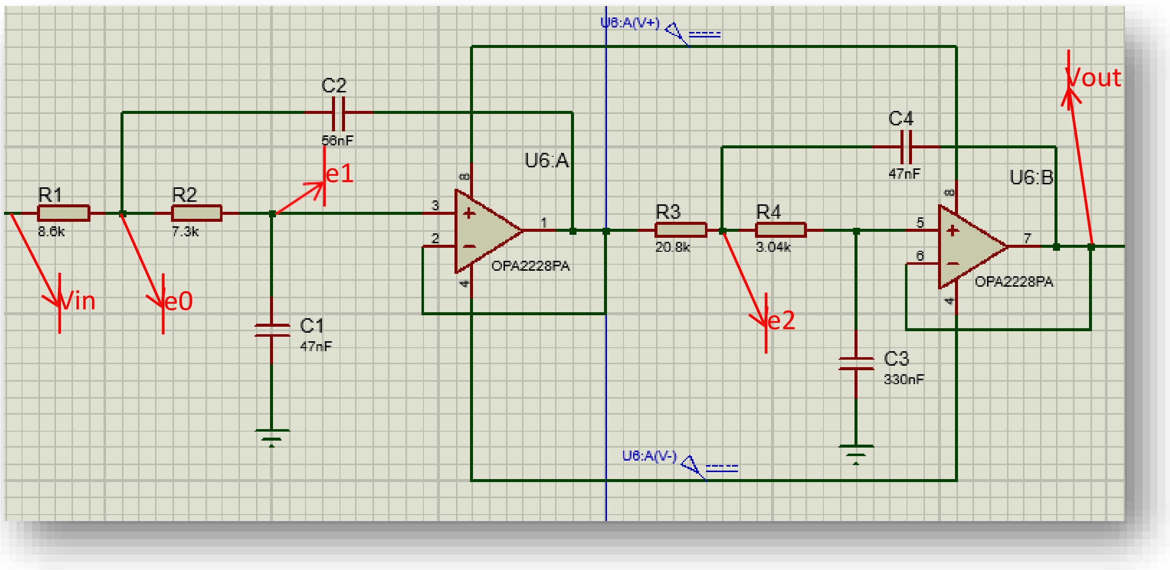


We know that the gain for this amplifier is:

$$g = -\frac{R6}{R5}$$

Hence, the signal is amplified by almost 21 using this circuit.

After that, we used a low-pass filter to remove high-frequency noises from the signal. As discussed before, the highest required frequency for our analysis is almost 200Hz. Therefore, we set f=200Hz as the cut-off frequency for our filter. To do so, we need to extract the frequency response of the circuit. Here is the designed circuit for low-pass filter:



This is a 4th-order filter. Considering the nodes e0, e1, e2, Vin, Vout in the figure above, we can write 4 KCL equations in Laplace domain like this:

$$\frac{E_0 - Vin(s)}{R1} + \frac{E_0 - E_1}{R2} + (E_0 - E_1) * (C2 * s) = 0$$

$$E_1 * (C1 * s) + \frac{E_1 - E_0}{R2} = 0$$

$$\frac{E_2 - E_1}{R3} + \frac{E_2 - Vout(s)}{R4} + (E_2 - Vout(s)) * (C4 * s) = 0$$

$$Vout(s) * (C3 * s) + \frac{Vout(s) - E_2}{R4} = 0$$

Using these 4 equations, we can form a linear 4-variable system (which can be solved by defining a 4*4 coefficients matrix) and compute E0, E1, E2, Vout(s) as functions of Vin(s). We only need Vout(s). After some calculations, the transfer function for this filter is as follows:

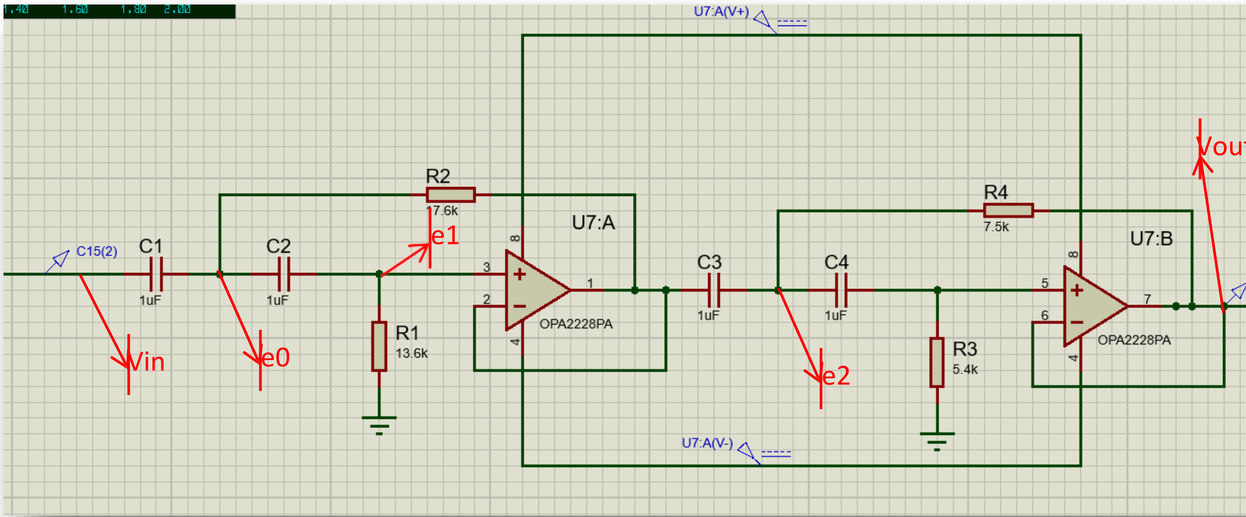
$$H(s) = \frac{Vout(s)}{Vin(s)}$$

$$= -\frac{1}{(1 + (C1 * R1 + C1 * R2)s + C1 * C2 * R1 * R2 * s^2) * (1 + (C3 * R3 + C3 * R4)s + C3 * C4 * R3 * R4 * s^2)}$$

Here we can see this is a 4th order filter. It is clear that the maximum absolute value for this transfer function is 1(With → ∞). As a result, in order to compute the cut-off frequency, we need to equate $|H(j\omega)|$ to $1/\sqrt{2}$.

Now if we set the values R1 = 8.6 kΩ, R2 = 7.3 kΩ, R3= 20.8 kΩ, R4 = 3.04 kΩ, C1 = 47nF, C2 = 56nF, C3 = 330nF, C4 = 47nF for the resistors and capacitors, the equation $|H(j\omega)| = 1/\sqrt{2}$ is held for $f = 200Hz$ (or equivalently,$\omega = 400\pi\ rad/s$).

The next stage is to pass the signal through a high-pass filter. We have already concluded that the lowest frequency to keep in our signal should be about 20Hz. Therefore, we used a 4th order high-pass filter which is very similar to the low-pass filter we discussed before. The circuit is as follows:



In a similar way, we can define the nodes e0, e1, e2, Vin, Vout in this circuit. Again, we can write 4 independent KCL equations and solve the linear system to obtain the transfer function:

$$\begin{pmatrix} (C1+C2)*s+\frac{1}{R2} & -C2*s-\frac{1}{R2} & 0 & 0 \\ -C2*s & C2*s+\frac{1}{R1} & 0 & 0 \\ 0 & -C3*s & (C3+C4)*s+\frac{1}{R4} & -C4*s-\frac{1}{R4} \\ 0 & 0 & -C4*s & C4*s+\frac{1}{R3} \end{pmatrix} \begin{pmatrix} E0 \\ E1 \\ E2 \\ Vout(s) \end{pmatrix} = \begin{pmatrix} C1*s*Vin(s) \\ 0 \\ 0 \\ 0 \end{pmatrix}$$

After calculation, the transfer function can be obtained:

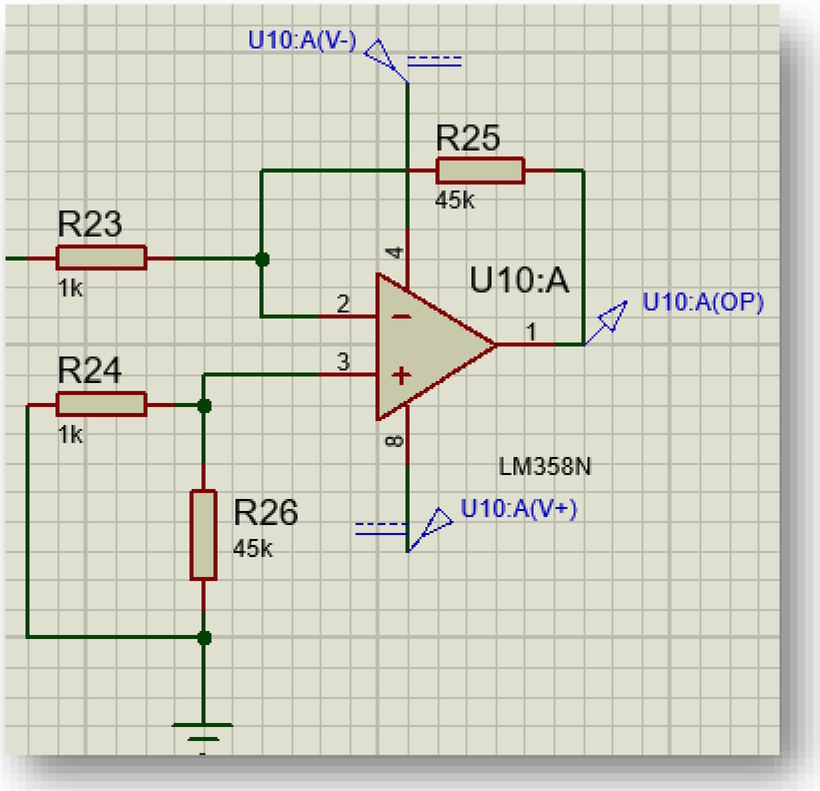
$$H(s)=\frac{Vout(s)}{Vin(s)}$$

$$=-\frac{C1*C2*C3*C4*R1*R2*R3*R4*s^4}{(1+(C2*R2+C1*R2)s+C1*C2*R1*R2*s^2)*(1+(C4*R3+C4*R4)s+C3*C4*R3*R4*s^2)}$$

Again, the maximum absolute value for the transfer function is 1(With $s\to\infty$). As a result, in order to compute the cut-off frequency, we need to equate $|H(j\omega)|$ to $1/\sqrt{2}$.

Now if we set the values R1 = 13.6 kΩ, R2 = 7.6 kΩ, R3= 5.4 kΩ, R4 = 7.5 kΩ and C1=C2=C3=C4=1uF for the resistors and capacitors, the equation $|H(j\omega)| = 1/\sqrt{2}$ is held for $f = 20Hz$ (or equivalently,$\omega = 40\pi\ rad/s$).

After the filtering stages, we used a buffer similar to previous circuit. Finally, we need to amplify the signal again. This stage is necessary because the amplitude of the signal is too small that cannot be heard by speakers. We used a simple differential amplifier for this final amplification:

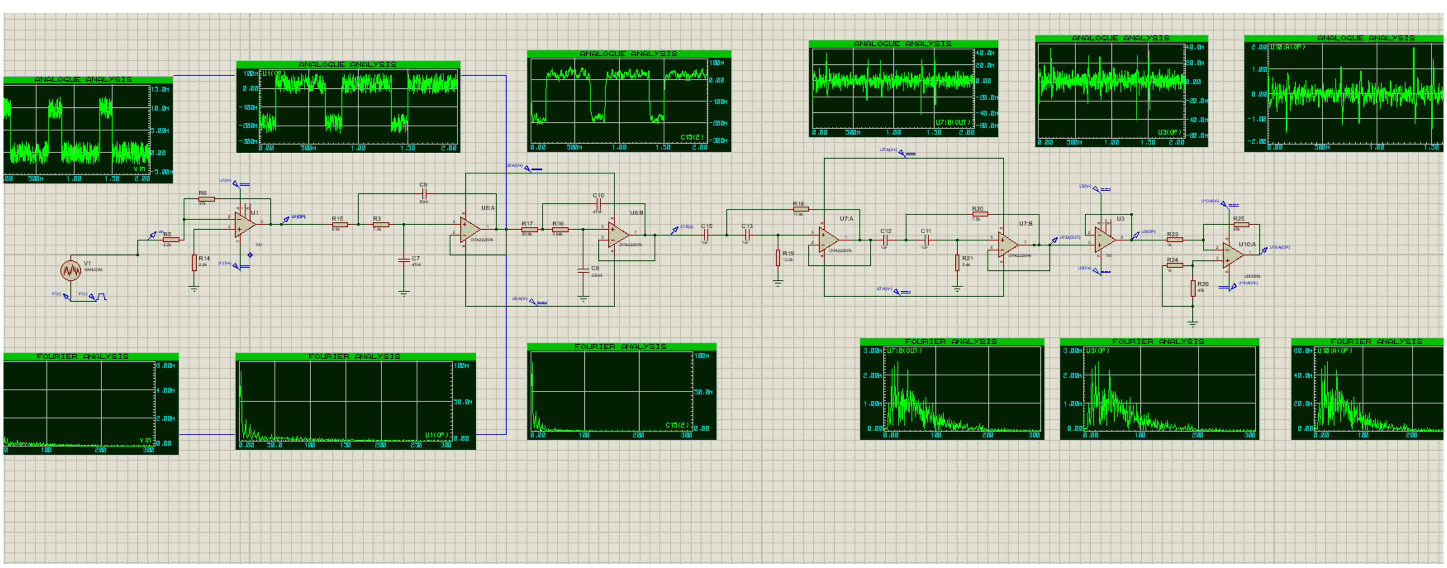


The negative input of the amplifier is connected to GND. Considering that $R26 = R25 = 45k\Omega$ and $R23 = R24 = 1k\Omega$, The differential gain of this amplifier is obtained by:

$$g = \frac{R25}{R23}$$

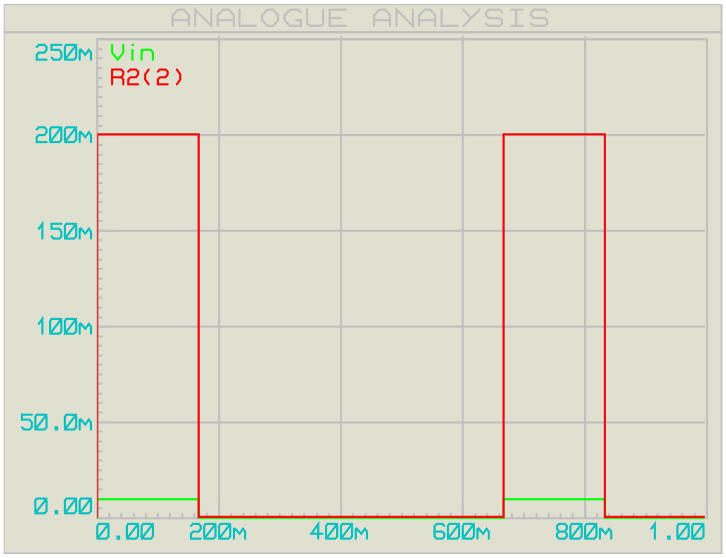
Hence, the output of this circuit is 45 times its input.

Finally, here is a schematic for the whole circuit:

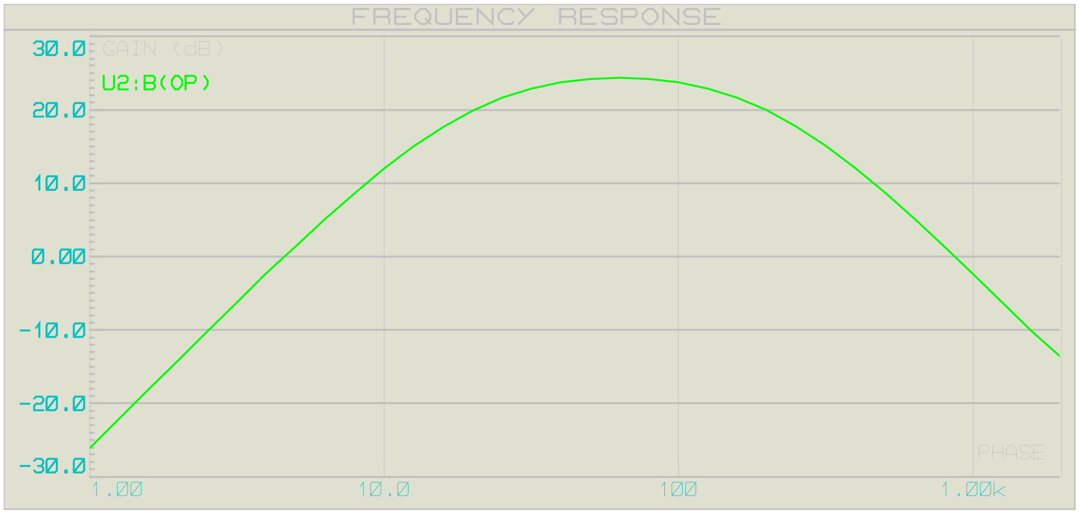


4. Results

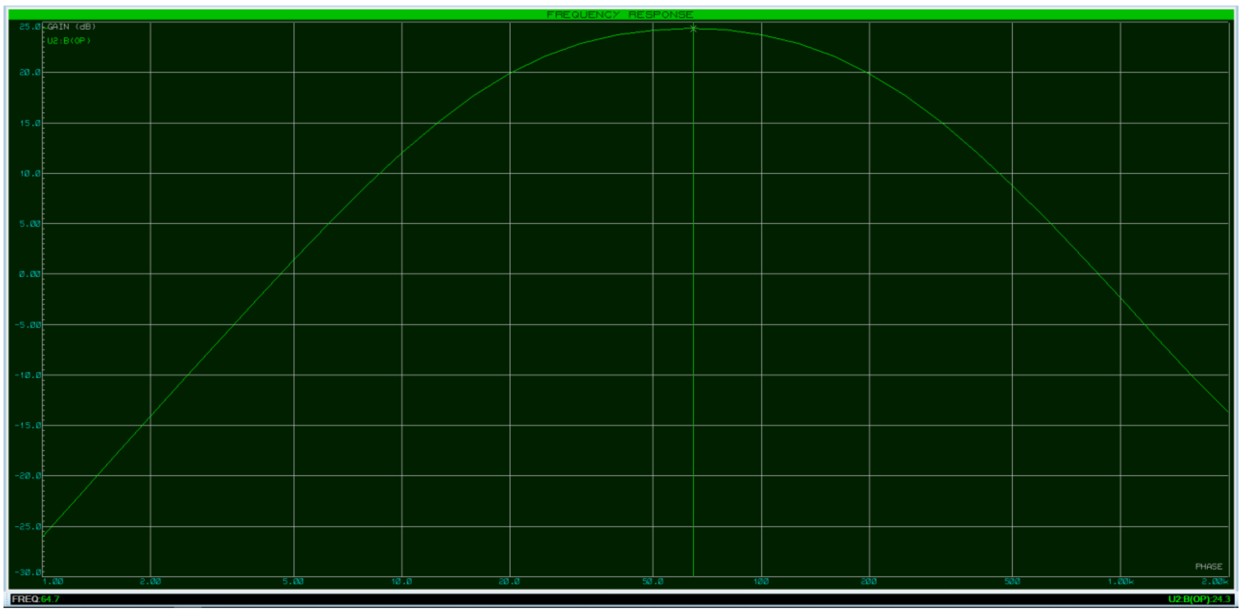
The simulation results of the main circuit:



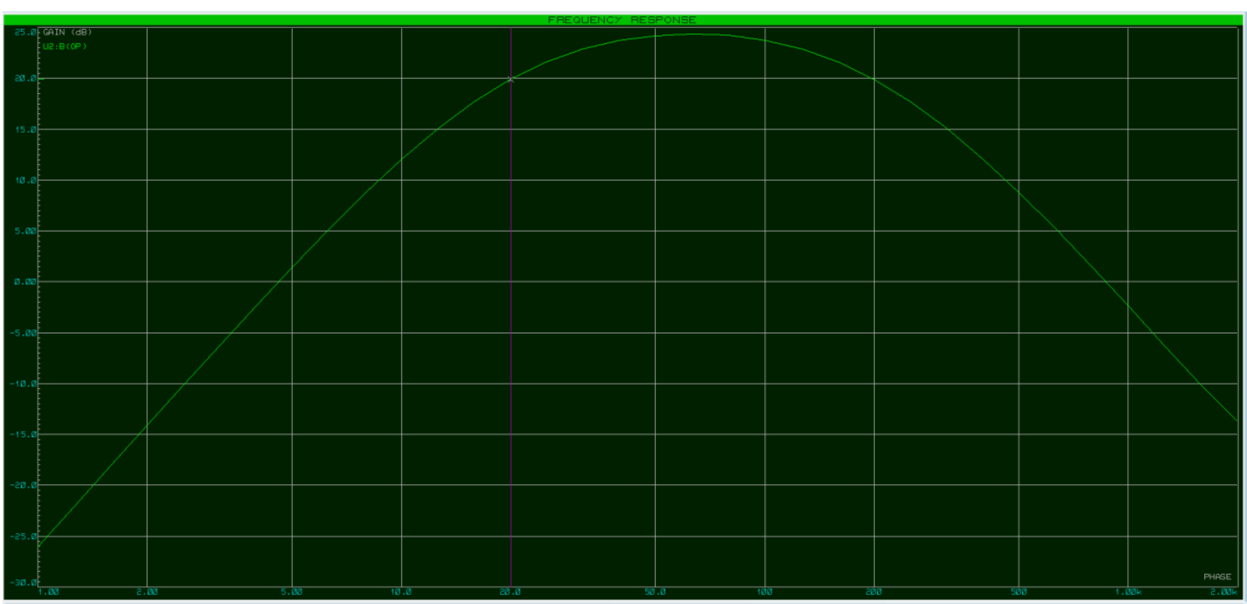
The above graph shows the input signal before and after the pre-amplifier stage.

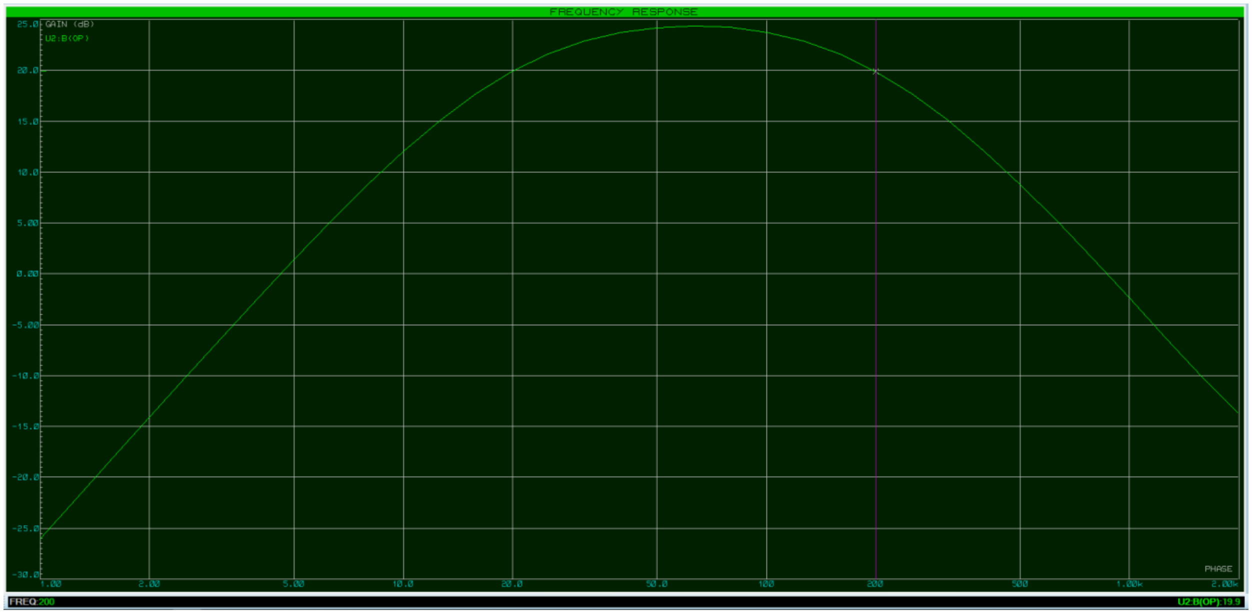


The above graph shows the frequency response of the band-pass filter.

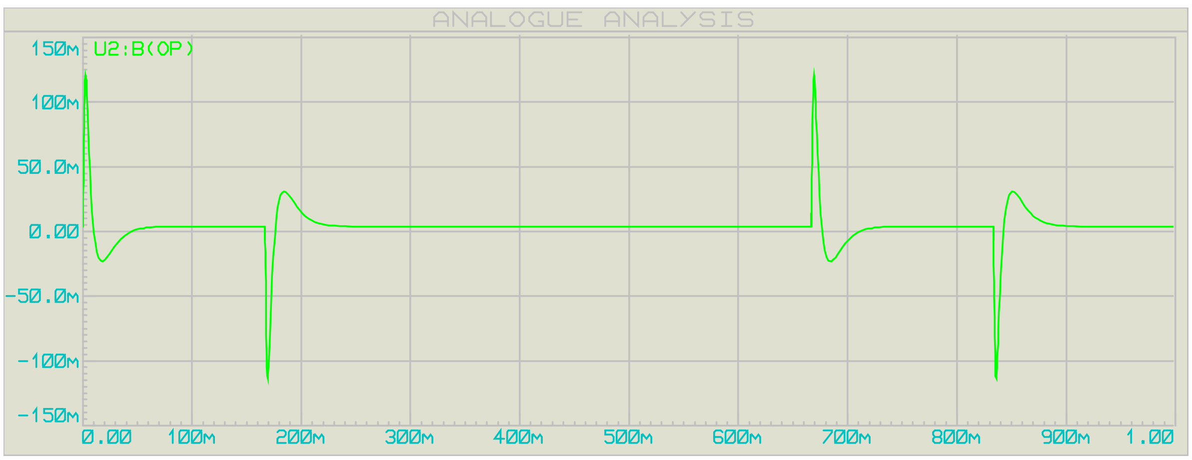


The gain at the peak is about 24.3dB.

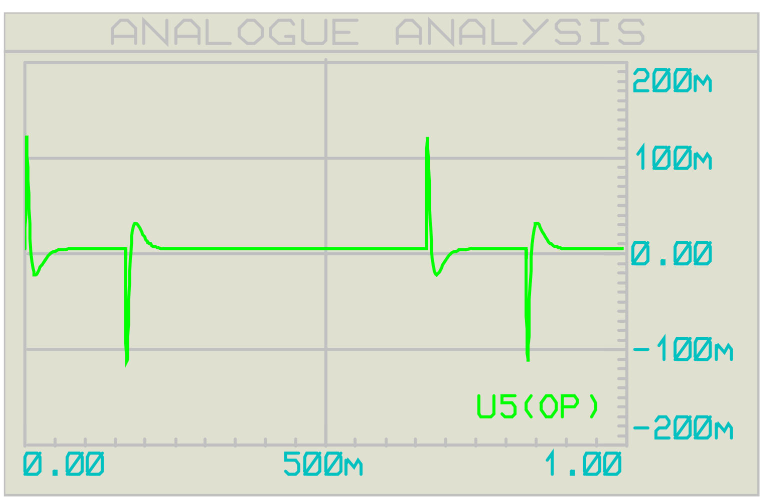




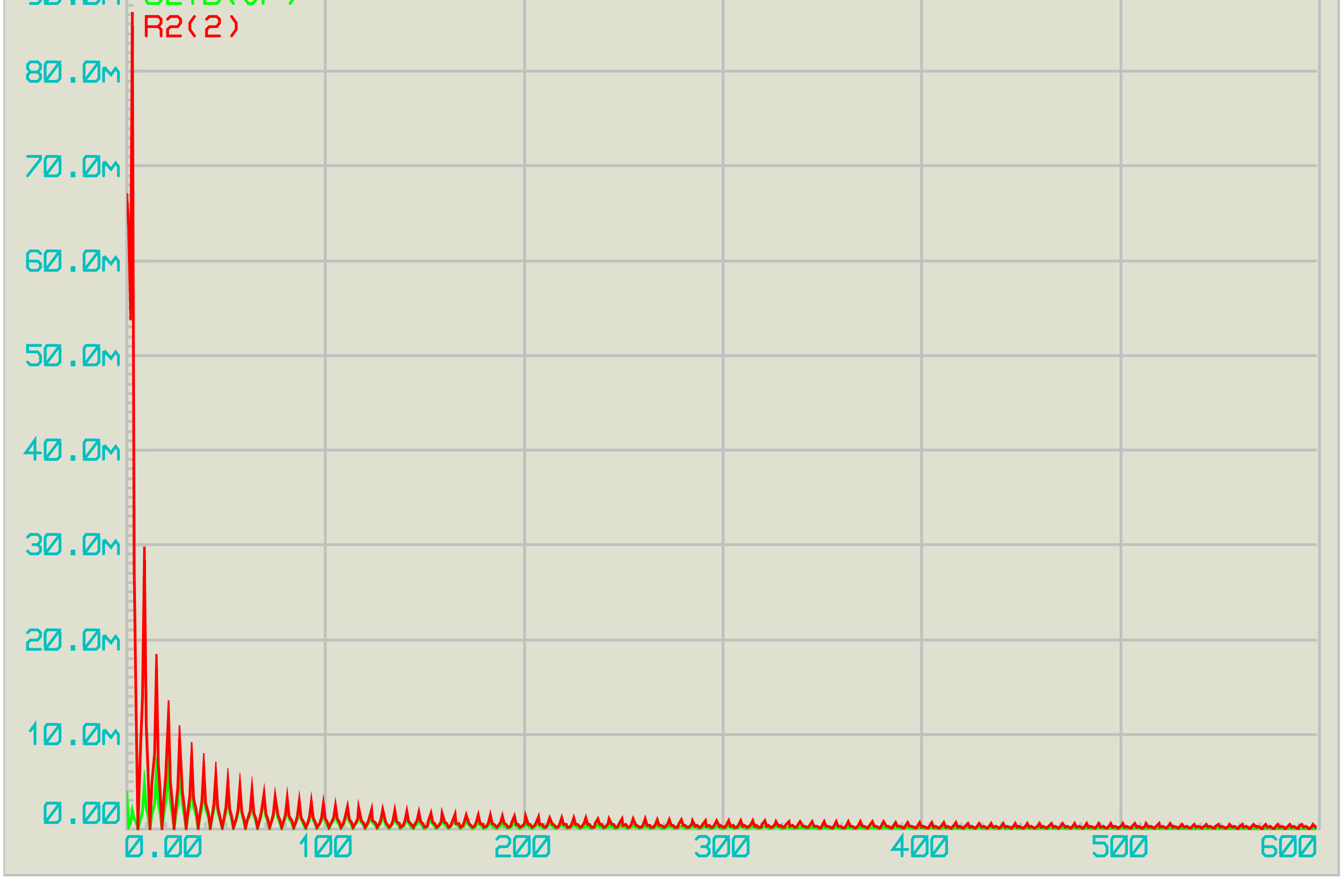
The gain at 20 and 200 Hz are 19.9Db which is about 3dB lower than the peak value. Therefore, the cutoff frequencies are 20 and 200Hz.



The above graph shows the analog signal after the band-pass filter.

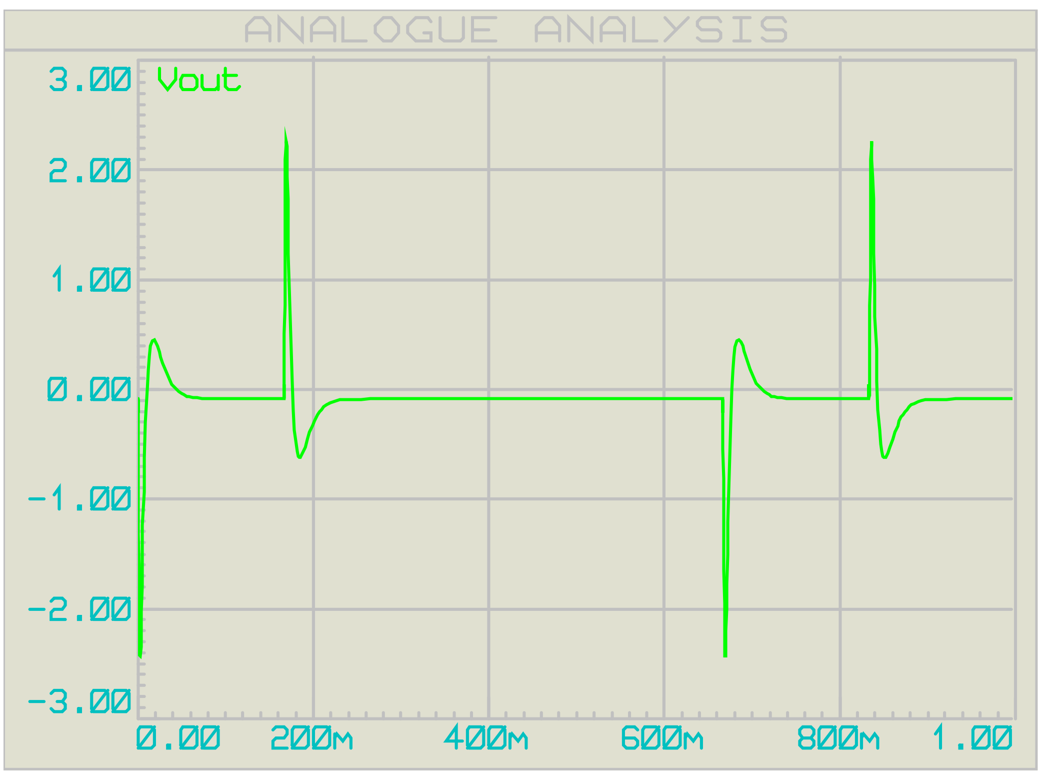


The above graph shows the analog signal after the buffer.



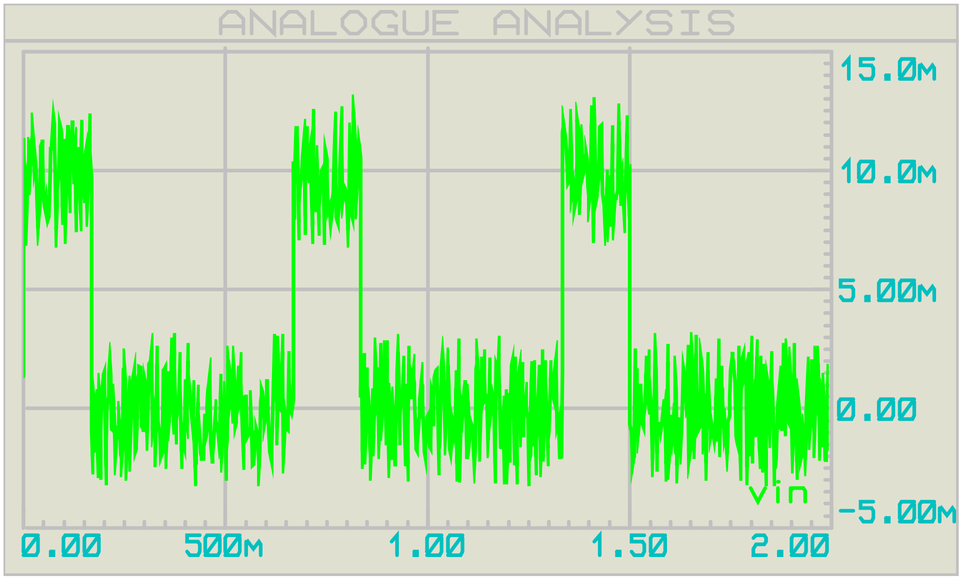
The above graph shows the fourier analysis of the signal before and after the band-pass stage.

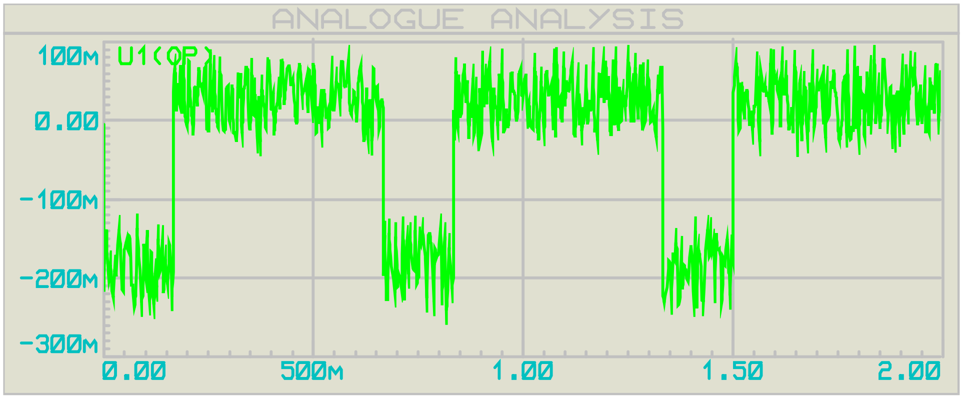
Finally, the below graph shows the output signal (lub-dub) in an appropriate range:

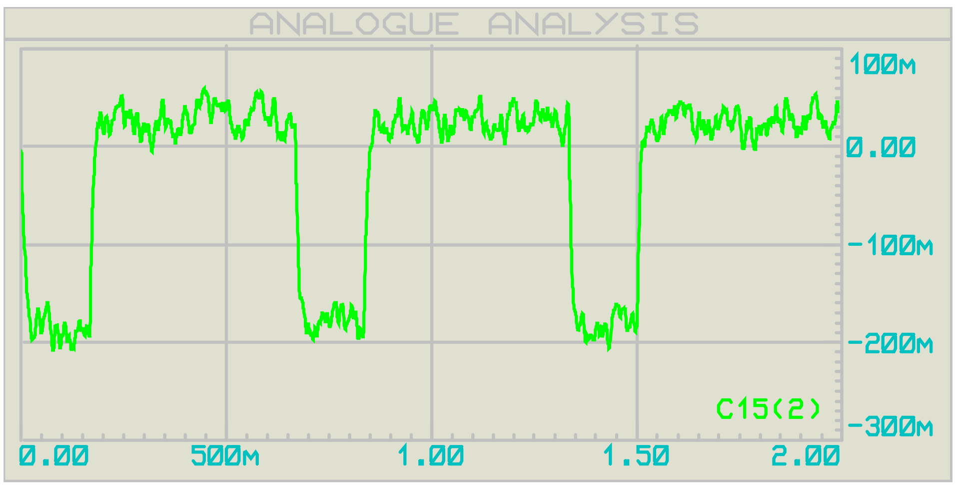


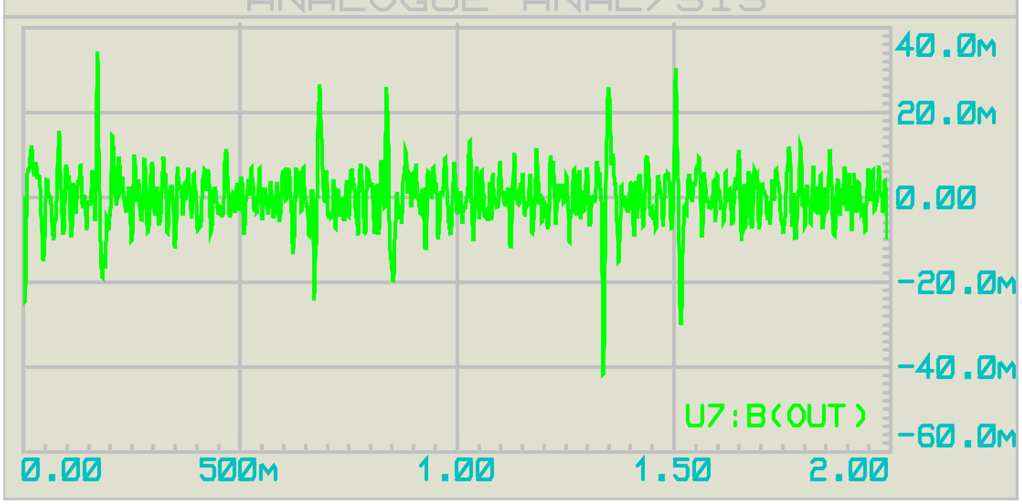
The results of the additional circuit:

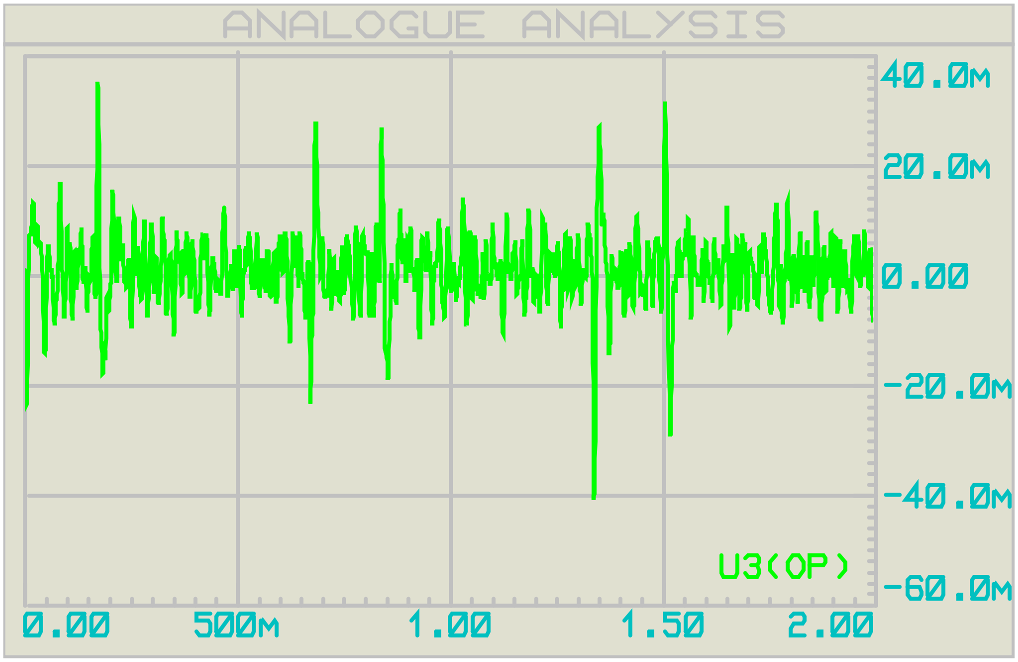
We plotted analog signals after every stage to have a step-by-step visualization for the signal. The following figures demonstrate the time-domain signal after: 1) the noisy input, 2) pre-amplifier, 3) low-pass filter, 4) high-pass filter, 5) buffer, and 6) amplifier, respectively:

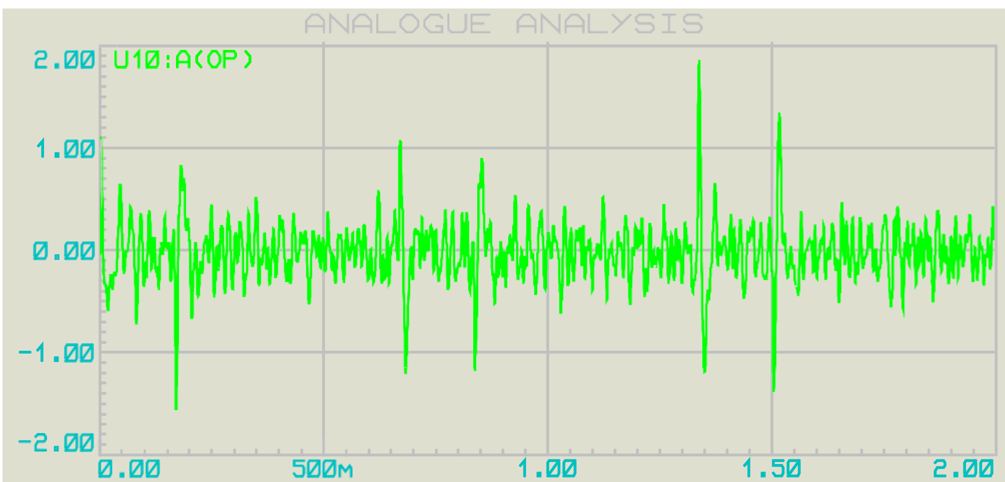






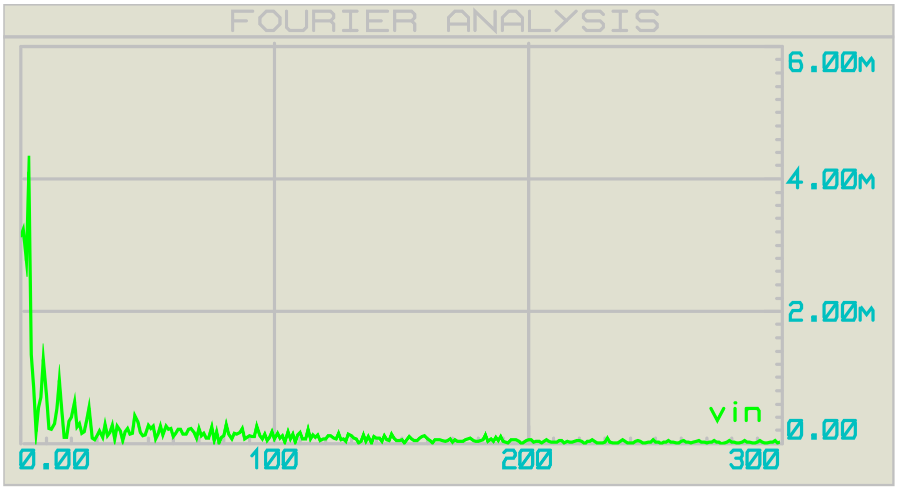


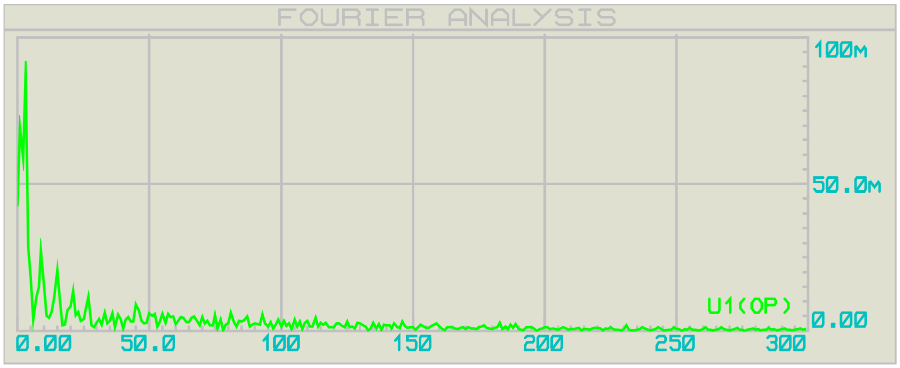


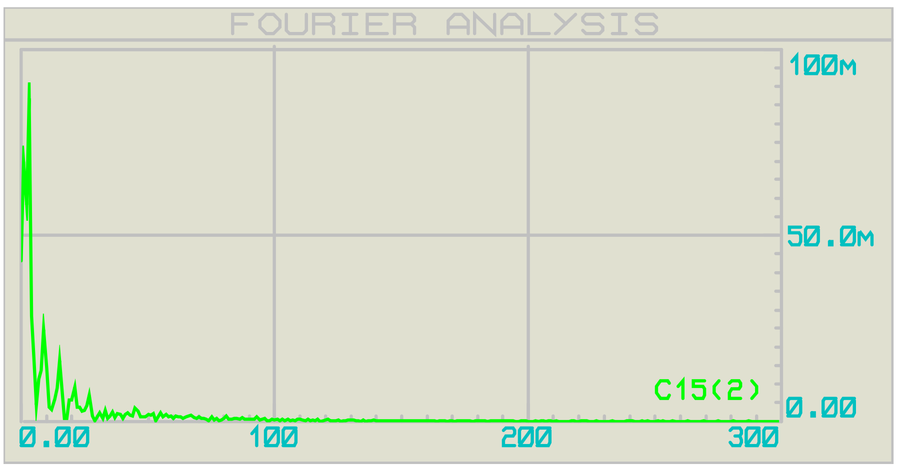


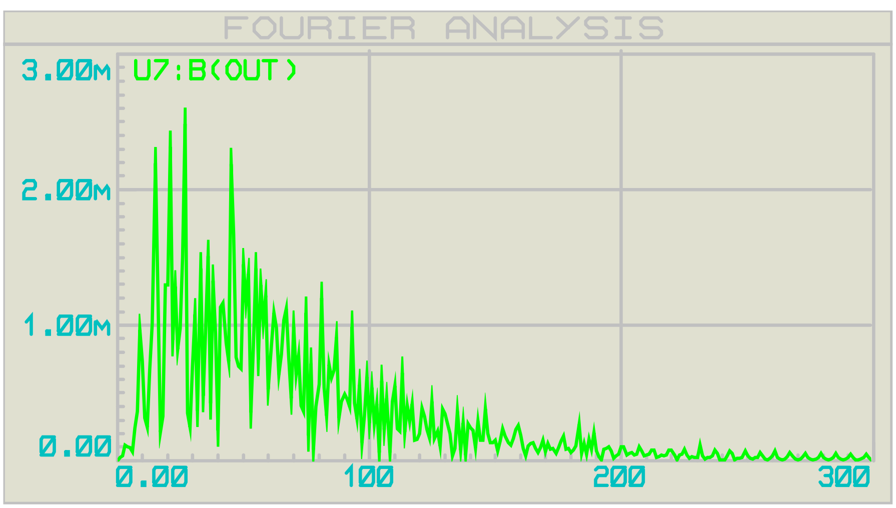
In our design, a single pulse is a representation of a 'lub-dub'. It means that, a single pulse contains both expansion and contraction of the heart. We can see that in the output signal, two prominent peaks take place for a single pulse. These peaks can simulate the sound of expansion and contraction in the output speaker.

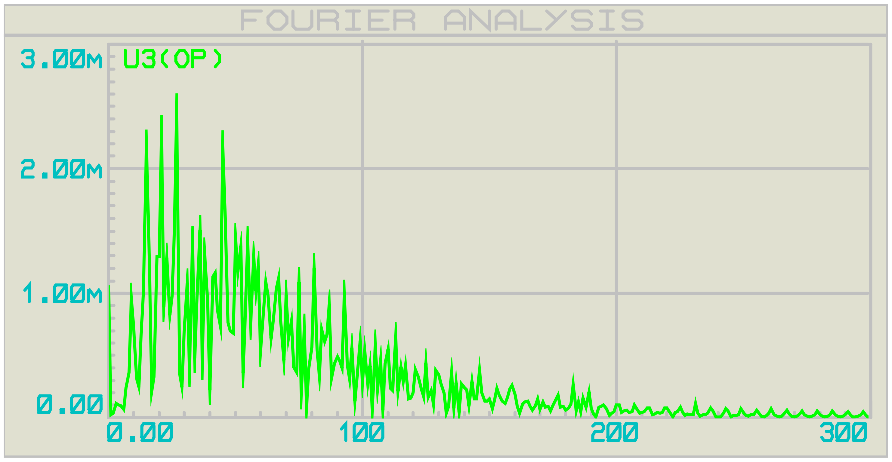
We also analyzed the signals in Fourier domain. The following figures are FFT of the discussed signals, in the same order as before:

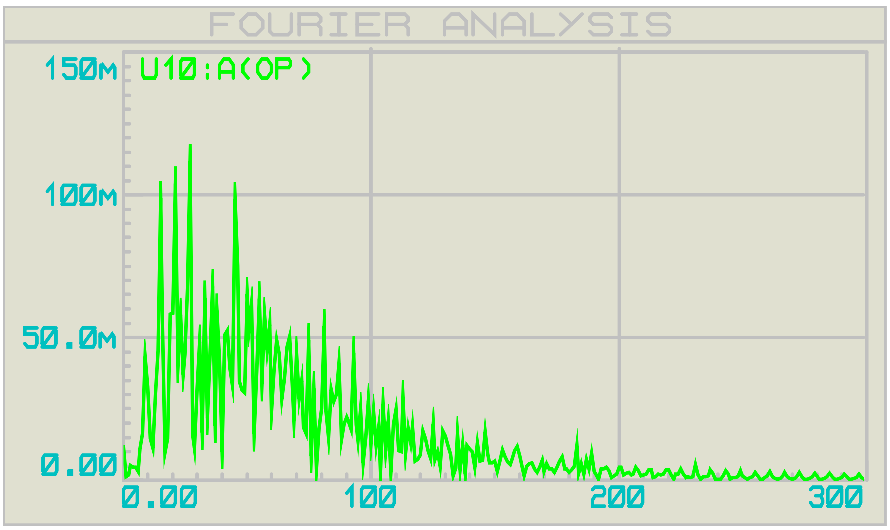

FOURIER ANALYSIS
6.00M
4.00M
2.00M
0.00
vin
0.00
100
200
300


FOURIER ANALYSIS
100M
50.0M
0.00
U1(OP)
0.00
50.0
100
150
200
250
300


FOURIER ANALYSIS
100M
50.0M
0.00
C15(2)
0.00
100
200
300


FOURIER ANALYSIS
3.00M
2.00M
1.00M
0.00
U7:B(OUT)
0.00
100
200
300


FOURIER ANALYSIS
3.00m
2.00m
1.00m
0.00
U3(OP)
0.00
100
200
300


FOURIER ANALYSIS
150m
100m
50.0m
0.00
U10:A(OP)
0.00
100
200
300

Some details about amplifiers and filters

We used two amplifier circuits in our design. Here are some required details about these amplifiers:

- Pre-amplifier details
  1. $Gain = 1 + \frac{R2}{R1} = 20$
  2. $R_{in} = R3 + \infty = \infty$
  3. CMRR: According to the fact that our negative input is GND, we can conclude that CMRR in an ideal condition equates to ∞ because of this equation for non-inverting amplifier: $Vout = k * vi + \frac{k*v}{2CMRR}$

- Final amplifier details
  1. $Differential\,Gain = \frac{R13}{R6} = \frac{20k}{1k} = 20$
  2. $R_{in} = R6 + \infty = \infty$
  3. CMRR: We know that for a differential amplifier, the following equation can be obtained: $Vout = \varepsilon * \frac{R13}{R6} V_{CM} + \frac{R13}{R6} V_{dif}$. Here, $\varepsilon = 0$ because the resistors R13 and R12, and R6 and R10 are completely matched. Hence, $Vout = \frac{R13}{R6} V_{dif}$ and $CMRR = \infty$

We also used two filtering circuits. These two together can be considered as a single band-pass filter. According to the details we discussed before, here are some main properties for the filter:

- Band-pass filter:
  1. Cut-off frequencies: $fc1 = 20Hz$ & $fc2 = 200Hz$
  2. Bandwidth: $BW = fc2 - fc1 = 180Hz$
  3. Gain: Our design for the filters is such that no amplification occurs during this stage. It means that, $g = 1$ for frequencies inside the bandwidth. This can be easily checked using Transform functions that we obtained for the filters (using special conditions: $s \to \infty$ for high-pass, and $s \to 0$ for low-pass)

5. Conclusion

In this project, we have simulated a circuit, which has been specifically designed for sensing heartbeats using a stethoscope. It is necessary to account for the main challenges and problems for this purpose. We considered the problem that the amplitude of the input vibrations is too small. Therefore, one of the main steps is to amplify the signal. Furthermore, we noted that there is a wide frequency range with unnecessary information through the input signal. Hence, we used filters to remove noises outside the useful frequency range (20-200 Hz).

Looking at the output figures we discussed before, we can conclude that the designed circuit is nicely able to identify pulses that stem from heartbeats. In our second design (for noisy input), the circuit is also able to extract important information (heartbeats) through a noisy signal. These are some positive points about the designed circuit.

However, sometimes there are some noises, which do not stem from random processes. For example, motions of the subject’s organs can affect our signal as an additional noise. In many cases, these noises have a very high amplitude and are hard to remove. Furthermore, in many cases, the spectrum of the unwanted noise is inside our desired bandwidth. Therefore, the noise cannot be attenuated using simple filters and other mechanisms should be used for removing the noises. These are some challenges that can be solved using more-complicated circuits.

6. References

- Stethoscope Instrument (2023). https://www.britannica.com/technology/stethoscope
- A Real-Time Heart Rate Signal Detection using an Electronic Stethoscope with Labview
- Study and Analysis of Electronic Stethoscope Signal Using MATLAB & LabVIEW as Effective Tools
- Proteus Design Suite
- ResearchGate
- ee-diary
- EEGUIDE.COM
- Allaboutcircuits

7. Datasheets

The choice of specific op-amps for different stages of an amplifier circuit depends on various factors such as performance requirements, cost, availability, and design constraints. Here are some reasons why we used the LM385, LM324, and 741 op-amps for our circuit:

1. LM385 (Pre-amplifier and Final Amplifier stages):
    - **Low input offset voltage:** The LM385 has a low input offset voltage, which helps minimize error and distortion in amplification.
    - **High gain bandwidth product:** With a high gain bandwidth product, the LM385 is capable of amplifying signals accurately across a wide range of frequencies.
    - **Low noise:** The LM385 has low noise characteristics, making it suitable for sensitive audio or instrumentation applications.
    - **Low power consumption:** The LM385 typically operates at low power, making it suitable for battery-powered or energy-efficient designs.
    - **Availability:** The LM385 is a commonly used op-amp and is widely available in the market, making it easy to source for various projects.
2. LM324 (Band-pass Filter):
    - **Quad op-amp package:** The LM324 is a quad op-amp, meaning it contains four separate op-amps in a single package. This allows for compact and space-efficient designs when multiple filters are required.
    - **Wide supply voltage range:** The LM324 can operate over a wide supply voltage range, accommodating different power supply configurations.
    - **Cost-effective:** The LM324 is known for being a cost-effective op-amp, making it suitable for applications where budget constraints are a consideration.
    - **General-purpose versatility:** The LM324 is a general-purpose op-amp that can be used in a wide range of applications, including filters, comparators, oscillators, and amplifiers.
3. 741 (Buffer):
    - **High open-loop gain:** The 741 has a high open-loop gain, allowing it to accurately buffer and isolate the input signal from the output, without significant signal degradation.
    - **Low output impedance:** The 741 has a low output impedance, enabling it to drive low impedance loads and maintain signal integrity.
    - **Familiarity and availability:** The 741 is one of the oldest and most widely used op-amps, making it familiar to many engineers and readily available in the market.
    - **Robustness:** The 741 is known for its robustness and ability to handle a wide range of operating conditions, making it suitable for various applications where reliability is crucial.

Here are their data sheets links:

- LM385N
- LM324
- LM741