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Construction, Performance Testing and Result Analysis of a Noise Eliminating Signal Controlled Hearing Aid

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Abstract

This paper describes the construction of a hearing aid using analog signal controlled system and analog processing signal chain. Microphone and headphones are used for converting acoustic audio signal into the analog electrical signal and vice versa. The diode combinational circuits have been used for detecting desired signals. Analog amplifier amplifies the audio signal. Active filter is used for passing desired frequency band.

Keywords

Unidirectional Microphone, Transducer, Signal Detector, Amplitude controller, Noise Eliminator.

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Introduction

Hearing loss is one of the most common human impairments. It is estimated that by year 2015 more than 700 Million people will suffer mild deafness. Most can be helped by hearing Aid devices depending on the severity of their hearing loss. It also can be helped to detect desired signal from various signals. It makes signal which magnitude is fixed. This paper describes the implementation and characterization detail of an analog signal Controlled hearing aid that use unidirectional microphone as an audio transducer, the Diodes combinational circuit acts as a signal detector; they make decision that which signal can pass and which signal is absorbed by diodes. LDR and LED combinational circuit can contribute the control of signal. The first generation of hearing aids usually consisted of analog variable gain amplifiers, Electret microphones and speakers that compensated hearing loss. They dissipated a Considerable amount of power and had flat frequency characteristics that make the devices Uncomfortable for more patients since hearing loss usually varies across different Frequencies. The next generation of devices adopted analog filter banks in which band-pass Filters were used in parallel to amplify acoustic signal to a specific level in each different frequency band. This design, however, resulted in bulky devices that still required high power consumption.

Figure 1 shows block diagram of the first generation of hearing aid. Microphone converts the acoustic audio signal into the analog electrical signal. It produces very week electrical signal such as 0.4 V (maximum). Hence a variable gain amplifier is used to amplify that signal. Band-pass filter passes all frequencies which exist in this bandwidth. And finally speaker or headphones converts analog electrical signal into acoustic audio signal.

Figure 2 shows the expected hearing aid consist unidirectional microphone, Amplifier, Signal detector, Amplitude controller, Band-pass filter, Speaker. The unidirectional microphone receives audio signals and converts into analog audio signal from one direction. Signal detector passes desired signal from all types of signals. Amplitude controller controls the amplitude of the signal. At the backend, a speaker delivers the acoustic sound to excite the patient's ear-drum.

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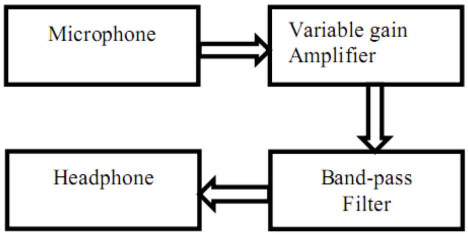

Figure 1: *Block diagram of the first generation of hearing aid*

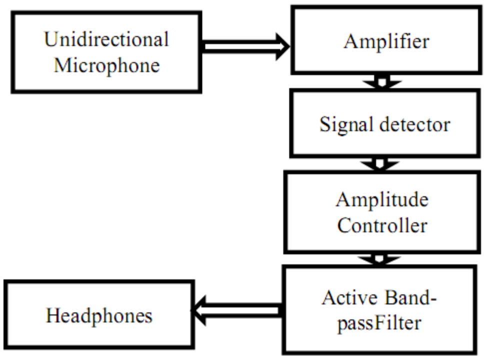

Figure 2: *Block diagram of desired hearing aid*

Characteristics of Desired Signal

At first we need to define the desired signal and noise. The signal which is to listen or detect is human audio signal as is the desired signal. The other extra signals coming from signal sources, system devices or environment are designated as noise. In a crowded area, where many people's are gathering and talking with each other, In that case the audio signal from a person is to be desired or detected. Assuming that this single person is only desired signal source. The various parameters are needed to detect this signal can be as follows:

Audio Signal with:
`Case A:`

1. Higher dB(Greater than 0.25 V_{RMS})
2. Minimum distance from microphone(Less than 3 inch)
3. Minimum angle with respect to the microphone (Zero degree)
4. Specific audio band (100 Hz - 2 kHz)

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`Case B:`
1. Lower dB (less than 0.15 V_{RMS})
2. Minimum distance from microphone (Less than 3 inch)
3. Minimum angle with respect to the microphone (Zero degree)
4. Frequency of the desired signal (100 Hz - 2 kHz)

`Case C:`
1. Lower dB (less than 0.15 V_{RMS})
2. Maximum distance from microphone (greater than 3 meter)
3. Minimum angle with respect to the microphone (Zero degree)
4. Frequency of the desired signal(100 Hz - 2 kHz)

`Case D:`
1. Lower dB (less than 0.15 V_{RMS})
2. Minimum distance from microphone (Less than 3 inch)
3. Maximum angle with respect to the microphone (Zero degree)
4. Frequency of the desired signal (100 Hz - 2 kHz)

`Case E:`
1. Lower dB (less than 0.15 V_{RMS})
2. Maximum distance from microphone (greater than 3 meter)
3. Maximum angle with respect to the microphone (greater than 30 meter)
4. Frequency of the desired signal(100 Hz - 2 kHz)

`Case F:`
1. Variable dB
2. Maximum distance from microphone (greater than 3 meter)
3. Minimum angle with respect to the microphone (Zero degree)
4. Frequency of the desired signal(100 Hz - 2 kHz)

`Case G:`
1. Variable dB (0.10 V_{RMS} to 3.7 V_{RMS})
2. Minimum distance from microphone (Less than 3 inch)
3. Maximum angle with respect to the microphone (180 degree)
4. Frequency of the desired signal (100 Hz - 2 kHz)

`Case H:`
1. Variable dB (0.10 V_{RMS} to 3.7 V_{RMS})
2. Variable distance from microphone (3 inches to 3 meter)
3. Variable angle with respect to the microphone (0 to 180 degree)
4. Frequency of the desired signal(100 Hz - 2 kHz)

Circuit Analysis

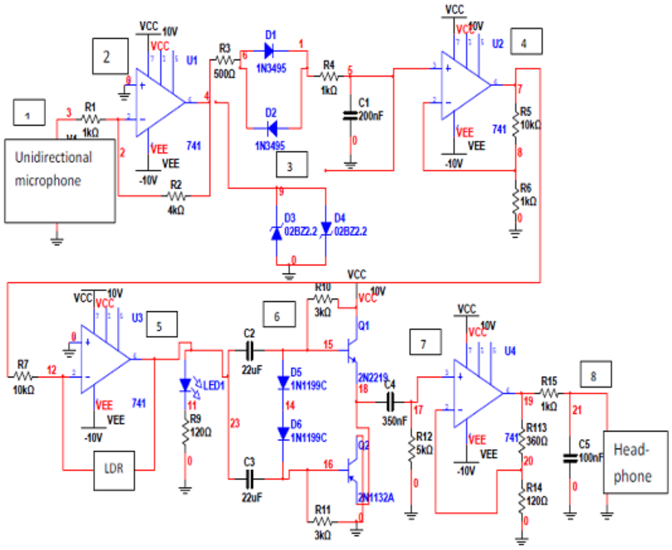

Figure 3: *Circuit diagram of desired Hearing Aid*

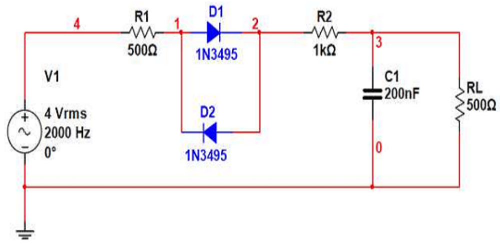

Figure 4: *Circuit diagram used for passing high amplitude desired signal*

Circuit Analysis of Case A

Figure 4 is applicable for this case. It can pass the higher amplitude signal to the load. As we know that when the PN junction is formed, any time there is a positive charge and a negative charge near each other, there is a force acting on the charges as described by Coulomb's law. In depletion region, there are positive charges and many negatives on the opposite sites of the PN junction. The force between the opposite charges forms a field of force's called an electric field. This electric field is a barrier to the free electrons in the n region and energy must be expanded to move an electron through the electric field. The

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potential difference of the electric field across the depletion region is the amount of voltage required to move electrons through the electric field. This potential difference is called the barrier potential. The barrier voltage of silicon diode is 0.7 V. It acts as an open circuit if the amplitude of signal is less than 0.7 V. Hence the signals whose are weak cannot accepted by load, and the signals which are strongest is passed by the diode to the load. Figure 4 shows that two are back to back connected; it is due to bias the diodes. Forward bias is the condition that allows current through the PN junction. The resistor which connects in series to diodes limits the current to a value that will not damage the PN structure. Notice that only one diode keeps in on state in both positive and negative cycle. On the other hand, one diode must be stayed at off state. This is due to reverse bias of diode. Reverse is the condition that essentially prevents current through the diode.

Circuit Analysis of Case B

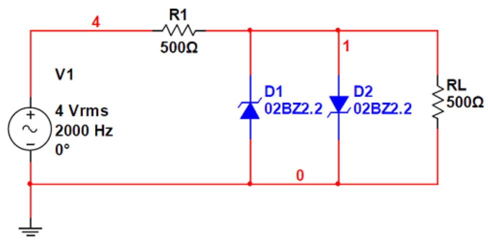

Figure 5: *Circuit diagram used for passing low amplitude desired signal*

We have applied this circuit for this case. Figure 5 shows the parallel connection of zener diodes, its limits the signal level. When the audio signal has higher amplitude and passes through the zener diodes, the diodes act as short, therefore there is no current flow through the load. But if the audio signal has lower amplitude, it passes to the load. Case 3 also solved by this circuit. We did not use normal PN junction or rectifier diode instead of zener diode, because the increasing rate of current of zener diode is higher than other. Though it is designed for operation in the reverse breakdown region, we have used it in forward region. The forward breakdown region is shorter than that of reverse breakdown region. When these diodes reach forward breakdown, its voltage remains almost constant even though the current change drastically. This is due to the heavily doped to reduce the breakdown voltage. This causes a very thin depletion region. As a result intense electric field exists within the depletion region. Hence it is acted as short by applied very small voltage.

Figure 5 also shows a small amount of resistance is connected to the source and diode, because it will decrease circulating current.

Circuit Analysis of Case D

One of the reasons for finding lower dB is that maximum angle between source and microphone. In that case we need to decrease the angle between source and microphone. If we reach the minimum angle we may find maximum dB. Higher amplitude passing circuit might be used for finding signal to the load is this case. If the amplitude maintained at lower dB though the angle between source and microphone is minimum, in that case we have to apply the circuit of lower amplitude passing signal.

And rest of the cases, at first, it is necessary to consider that the distance between source and microphone is an independent parameter, so that we cannot change the distance from microphone and source. Sometimes the magnitude of the audio signal is varied though the distance and angle are fixed. That is why it is impossible to detect the signal by using circuit 16 and circuit 17.there parameter one and two does not work successfully it the dB level increase and decrease with respect to time. Only one parameter can be applied for receiving signal is frequency. These problems ought to solve through the band pass filter. As we know that we use band pass filter for passing signal of desired frequency level, hence filter circuit may be detected the signal which has variable dB if frequency keeps fixed. Assume that the frequency of the desired signal is different than others. Than we need to change the value of bandwidth to our desired level of band-pass filter. To find sharp bandwidth we may use sallen key or other high efficient filter and whose roll off -80 dB/decade. After all it can pass the signal smoothly and perfectly which frequency exists in this bandwidth. And decrease the amplitude of other signals. Then high amplitude passing circuit can be used for receive that signal.

Amplitude Controller

Figure 6 shows intensity response of output with respect to input. In a typical hearing aid, the value output increases linearly, as a result increases dB But it causes danger to the ear of patient or user. Therefore we need to control the amplitude of output signal to our desired level. We have made a voltage controlling circuit which can control the intensity of the signal.

Figure 7 has shown an amplitude controlling circuit. The amplitude de-

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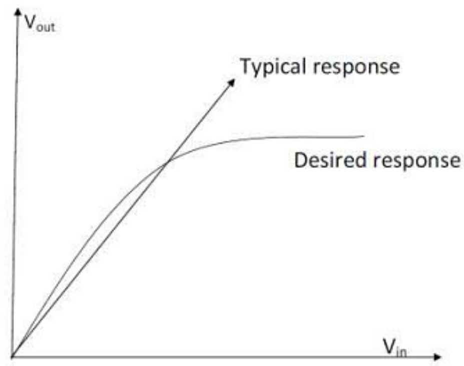

Figure 6: *Output response of hearing aid*

termines the `intensity` of any wave. Amplitude of the signal is controlled with an inverting amplifier and LED combinational circuit. From this figure we can that the gain of this operational amplifier depends upon the value of LDR (light dependent resistor), the gain of the amplifier will be increased when the resistance of the LDR is increased. The variation of resistance depends upon the intensity of LED (light emitting diode). If the input signal with higher amplitude passes to the output, the intensity of LDR keeps high which forces the resistance to of LDR to minimum. As we know that the resistance of LDR will decrease if high intensity of light falls to the LDR. Therefore the gain of amplifier would be decreased. This method has been applied for controlling the amplitude of the signal.

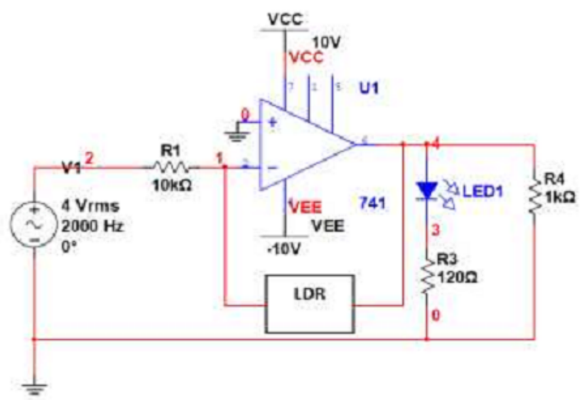

Figure 7: *Circuit diagram of amplitude controller*

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Working Procedure

The acoustic audio signal is collected by unidirectional microphone. The unidirectional microphone is a transducer which converts acoustic audio signal into analog electrical signal. These analog signals are very weak (up to 0.33 V). Hence an amplifier has to use which amplify the signal up to a desired level. There are two signal detecting circuit are connected to the amplifier. One is higher dB passing circuit other is lower dB passing circuit. The back to back connection of diodes is connected in series with other circuit. It blocks the low amplitude signals. As we know that diode goes to its on state after finding voltage which greater than 0.7 V (barrier potential). So that, the signal which is greater than 0.7 V, can overcome through this circuit. In other hand, the signal which is less than 0.7 V may not overcome through this circuit.

In the case of passing weak signals and blocking strong signal, we need to connect two zener diodes in parallel with respect to the source and load. The diode acts as short or increase current rapidly when amplitude is greater than 0.7V (forward breakdown voltage of zener diode is 0.7 V). In this case, there is no signal goes to the headphone. But the signals whose are very weak (such as less than 0.7 V) can pass to the load (headphone) perfectly.

Higher amplitude passing circuit gives distorted output signal; therefore it is necessary to use a low pass filter. A low pass filter blocks high frequency. Finally it can make smooth signal. The signal which comes from passive low pass filter is weak, that is why another amplifier is used to amplify that weak signal. Moreover, signal must be amplified to perform for next stage.

Then the desired signal goes to the amplitude controlling circuit. Output voltage of the amplitude controlling circuit depends upon the resistance of LDR. Again the resistance of LDR depends on the intensity of LED. If input voltage is higher dB, LED emits lighter and decrease resistance of LDR. We can control the amplitude by applying this circuit.

Next, a class B amplifier is added to the circuit. It amplify the power, thus increase dB. Now the desired signals which have found from previous circuits contain different kinds of external and internal noises and harmonics. So that we need to neglect noises, harmonics or other unwanted signal, otherwise we cannot listen clearly. We have used an active band- pass filter to solve those problems. The band-pass filter attacks or blocks the unwanted signal frequency and pass the signals which have desired frequency level. At the end, a smooth,

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perfect and desired signal has been found in headphones. Finally headphones converts analog electrical signal into acoustic audio signal.

Noise Elimination Process

What people call microphone noise may not be the fault of the microphone. There are three main sources of noise when using a microphone:

1. Background noise in the environment, such as sounds made by air conditioners, road noise, or the coughing of an audience.

2. Noise inherent in the preamp that the microphone connects to. All electronics add noise; cheap microphone preamps noticeably so.

3. The self-noise of the microphone.

Microphone noise can be an unwelcome nuisance to anyone attempting to record audio. Thankfully, there are simple actions that can be taken to reduce or eliminate it completely. At first, noise comes from microphone. Sound levels, including noise, are measured in decibels of sound pressure level (dB SPL), Microphones and preamps each have their own noise floors. When selecting a mic preamp we want to know to what degree the preamp's noise degrades the noise of your microphone. Different microphone technologies use different terminology to describe noise. Unidirectional microphones generally don't include a noise spec because their self noise only depends on their impedance and their temperature.

Firstly, we would like to choose MEMS based unidirectional condenser microphone. It can eliminate large amount of noises. The (Micro Electrical-Mechanical System) microphone is also called a microphone chip or silicon microphone. The pressure sensitive diaphragm is etched directly into a silicon chip by MEMS techniques, and is usually accompanied with integrated pre amplifier. It is highly sensitive. It can detect the desired signal.

In the case of passing high amplitude signal, Signal is distorted after diode operation. That is why a passive low pass filter has used to eliminate distortion.

Figure 8 shows a passive low pass single pole filter whose critical frequency is 800 Hz. The critical frequency is determined from the value of resistor and capacitor. It provides roll off of -20 dB/decade above the critical frequency. Hence if the frequency is increased above critical frequency, the output voltage

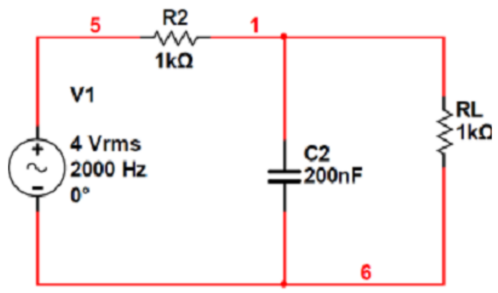

Figure 8: *Passive low pass filter*

will decrease. Distortion produces harmonics and harmonics is blocked by this low pass filter.

Moreover, it is possible to include or produce noise, harmonics, DC signal from amplifier circuit, so that we need to eliminate them by using band-pass filter. The bandwidth of audio signal is about 2.2 kHz; lower frequency is 150 Hz and upper frequencies 3 kHz. But normally it seems to be 2 kHz.

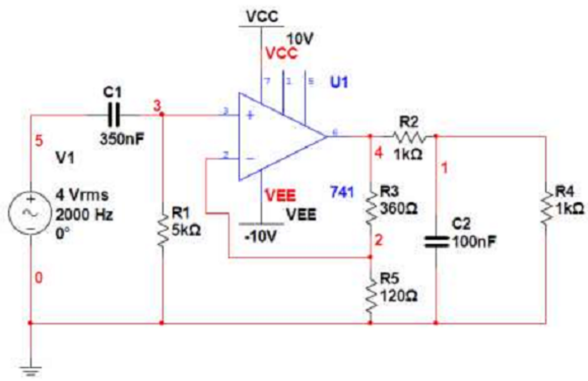

Figure 9: *Active band-pass filter*

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