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DESIGN OF A HEARING AUXILIARY FOR BILATERAL HYPOACUSIA

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ABSTRACT

In this work a box type hearing aid was designed as a convenient and low cost alternative for people with bilateral hearing loss. The hearing aid will be equipped with the following stages: a microphone to receive the acoustic signals, a preamplifier with automatic gain control to protect the user from excessive sounds, a 3-band equalization (bass, mid and treble) for each ear canal, a volume control for each and finally a 3.5mm jack output for the user to equip this device, with the headphones of his choice.

KEYWORDS: Bilateral Hypoacusia, auditory aid, audio amplifier, three band equalizer, hearing aid.

I. INTRODUCTION

Hearing is one of the most important senses we have as human beings, since it not only allows us to interact socially but also with our environment, allowing our evolution to be faster and our survival easier. However, this sense is continually compromised, both for congenital issues and for causes due to the environment, leading us to suffer different levels of hearing loss (hearing impairment). According to the World Health Organization (WHO) more than 5% of the world population (360 million people) suffer from disabling hearing loss, this is a loss greater than 30 dB in children and 40 dB in adults (328 million adults and 32 million children) [1]. These percentages of incidence are increasing as we are continuously exposed to high levels of noise due to industries, traffic, among other environmental factors, where the majority of people with disabling hearing loss live in countries with low and middle income.

Due to the importance of hearing in our society and the incidence of hearing loss in this, has sought since ancient times to solve this condition, which leads us today to have various types of hearing aids, from the ancient trumpets of the 19th century [2] to implantable devices such as in [3] where it is implanted directly into the brainstem. In this project, a hearing aid for bilateral hearing loss was designed with a 3.5mm dual-channel female Jack output with audio adjustment, as well as independent three-band EQ (bass, mid, treble) for each channel, so that the patient can dispose of a hearing aid, which adapts to your particular hearing impairment, with excellent quality and at a low cost.

It was in 1980 several companies were already incorporating digital processing to the hearing aids, these models debuted until 1996 with programmable models that could be adjusted depending on the needs of the patient, these models were already available for 2000. By the year 2005, the hearing aids Digitals had captured more than 80 percent of the market. Thanks to the technological advances that we have in 2016 in terms of miniaturization in electronic technology, as well as in biomedicine, so that we can already have very small hearing aids, as well as the newest of implantable hearing devices , these can be implanted directly on the surface of the brainstem [3], with the advantage that these can solve serious and profound hearing loss problems, the drawback is that surgical intervention is required and they are extremely costly.

The problem today is in terms of background noise. Excel in the amplification and control of acoustic feedback since digital hearing aids also tend to incorporate strange sounds that can make conversation difficult. So the main development is focused on filtering the noise.

II. DEVELOPMENT

The hearing aid will be equipped with the following stages: a microphone to receive the acoustic signals, a preamplifier with automatic gain control to protect the user from excessive sounds, a 3-band equalization (bass, mid and treble) for each ear canal, volume control for each channel and finally a 3.5mm jack output for the user to equip this device, with the headphones of his choice, figure 1.

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Figure 1. Block diagram of the hearing aid

Entry stage

For the input stage we use a CMC-2242PBL-A microphone, it is electret condenser type. It was chosen based on its low current consumption, operating frequency within the range of human hearing (20 HZ -20 KHz), small size, without losing sight of the price and quality, so its main characteristics are shown in the table 1. *Table 1: Main properties of the CMC-2242PBL-A microphone*

| Directional properties | Omnidirectional |
|--------------------------------|--|
| Output impedance (Z_{out}) | 2.2 ΚΩ |
| Sensitivity (S) | -42 ± 3 dB |
| Operating voltage | $2 V_{dc}$ (estándar) $10 V_{dc}$ (max.) |
| Frequency of operation (f) | 100 ~ 20,000 Hz |
| Current consumption (IDSS) | 0.5 mA max. |
| Signal to noise ratio (S / N) | 58 dBA |
| Operating temperature | -20 ~+70 °C |
| Dimensions | ø6.0 x 2.2 mm |

For the design of the circuit we will build on the configuration of figure 2 and in equation 1. Where we have: the load resistance $R_L = 22 k\Omega$, the supply voltage $+V_S = 3.3 V$ and the capacitor $C = 0.1 \mu F$.

$$f_0 = \frac{1}{2\pi (RL)(C)} = 72 \, Hz \tag{1}$$



Figure 2. Schematic and connection diagram of the microphone CMC-2242PBL-A]

Pre amplification and automatic gain control (AGC)

For the stage of pre-amplification and automatic gain control we use the integrated circuit MAX9814, since this IC, in addition to being designed to amplify the signal coming from the microphone, has an integrated automatic gain control (AGC) ideal to work under the conditions desired, without the risk of harm to the user. In addition, the THD is well below 1%, which is the value requested that does not exceed the audio quality

In addition, the THD is well below 1%, which is the value requested that does not exceed the audio quality standard, among other properties as shown in table 2.



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Table 2: Main properties of the MAX9814 integrated circuit

| Total harmonic distortion (THD) plus noise | 0.04 % |
|--|------------------|
| Maximum input impedance | 10 kΩ |
| Supply voltage | 2.7 V - 0.5 V |
| Functional temperature | -40 °C a + 85 °C |
| Operating supply current | 3.1 mA |

The circuit of figure 3 has been designed with an attack time of 1.1 ms, Gain 40 dB, and an attack and release ratio of 1: 500. The output of the MAX9814 is skewed at 1.23V. To eliminate DC displacement, a coupling capacitor (COUT) was used. So finally at the output of this we have a high pass filter set at 72 Hz to avoid lower frequencies and above all 60 Hz noise.



Figure 3: Diagram of the typical / Functional circuit of the MAX9814 CI.

Equalization stage

For this stage the integrated circuit MCP604-E / SL was used. Three bands with second order pass band active filters with input attenuator were used to modify the gain in low (450 Hz - 900 Hz), medium (1125 Hz) and treble (3 KHz). Later, an inverting adder was added where each one of its inputs will be controlled with a trimpot to give the necessary gain to each band depending on the needs of the patient, later the 3 bands of the signals will be added and finally with another trimpot for have volume control on the device.

Second order band pass filter with input attenuator, for bass (450 Hz - 900 Hz)

Figure 4 shows the second order band pass filter with input attenuator designed to operate between the frequencies of 450 Hz and 900 Hz, which correspond to the bass tones of the device. We start with the filter design by setting the following parameters: initial frequency $f_1 = 450$ Hz, final frequency $f_2 = 900$ Hz, capacitors $C_1 = C_2 = 0.01 \,\mu\text{F}$, gain HOBP = 1. The following data are available with these parameters: center frequency $f_c = 636$ Hz, quality factor Q = 1.4, bandwidth BW = 450 Hz.

We start with the filter design by setting the following parameters: initial frequency $f_1 = 450$ Hz, final frequency $f_2 = 900$ Hz, capacitors $C_1 = C_2 = 0.01 \,\mu\text{F}$, gain HOBP = 1. With these parameters we have the following data: central frequency $f_c = 636$ Hz, quality factor Q = 1.4, bandwidth BW = 450 Hz.

With the help of equations 2, 3 and 4 the following values of the resistances shown in table 3 are obtained.

$$R_2 = \frac{Q}{\pi f_0 C}$$
(2)
$$R_A = \frac{R_2}{2 |H_{0BP}|}$$
(3)



 $R_B = \frac{R_2}{2} \left(\frac{1}{2Q^2 - |H_{0BP}|} \right) \quad ^{(4)}$

 Table 3: Resistance values obtained theoretically and the closest ones that are commercially available

| Resistance | Theoretical resistance | Commercial resistance |
|------------|------------------------|-----------------------|
| R2 | 70 ΚΩ | 68 ΚΩ |
| RA | 34 ΚΩ | 33 ΚΩ |
| RB | 11.7 ΚΩ | 12 ΚΩ |



Figure 4: Second order band pass filter with input attenuator for Graves

Second order band pass filter with input attenuator, for media (1 KHz - 2KHz)

Figure 5 shows the second order bandpass filter with input attenuator designed to operate between the frequencies of 1 kHz and 2 kHz, which correspond to the bass tones of the device.

We start with the filter design by setting the following parameters: initial frequency $f_1 = 1$ kHz, final frequency $f_2 = 2$ kHz, capacitors $C_1 = C_2 = 0.01 \mu$ F, gain HOBP = 1. With these parameters we have the following data: central frequency $f_c = 1.4$ kHz, quality factor Q = 1.4, bandwidth BW = 1 kHz.

With the help of equations 2, 3 and 4 the following values of the resistances shown in table 4 are obtained. *Table 4: Resistance values obtained theoretically and the closest ones that are commercially available*



Figure 5: Second order bandpass filter with input attenuator for Media.



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Second order bandpass filter with input attenuator, for treble (2 KHz - 4 KHz).

Figure 6 shows the second order band pass filter with input attenuator designed to operate between the 2 kHz and 4 kHz frequencies, which correspond to the bass tones of the device.

We start with the filter design by setting the following parameters: initial frequency $f_1 = 2$ kHz, final frequency $f_2 = 4$ kHz, capacitors $C_1 = C_2 = 0.01 \,\mu\text{F}$, gain HOBP = 1. The following data are available with these parameters: center frequency $f_c = 2.8$ kHz, quality factor Q = 1.4, bandwidth BW = 1 kHz.

With the help of equations 2, 3 and 4 the following values of the resistances shown in table 5 are obtained.

| Table 5: Values of the theoretically obtained resistances and the most approximate values found | d |
|---|---|
| commercially | |

| Resistance | Theoretical resistance | Commercial resistance |
|------------|------------------------|------------------------------|
| R2 | 15.7 ΚΩ | 15 KΩ |
| RA | 5.3 ΚΩ | 6.8 KΩ |
| RB | 2.6 ΚΩ | 2.7 ΚΩ |



Figure 6: Second order band pass filter with input attenuator for Treble.

Gain and Volume Control

For the gain control stage, a set of resistors is added in series (see figure 7), one with a tripot in voltage divider configuration to attenuate or give gain to each frequency and the other set that is connected to the input of the inverter adder which is fed back with a trimpot to give the gain or attenuation (see equation 5) necessary to each band depending on the needs of the patient to add the 3 bands of signals coming and finally it is given to this with a trimpot for the control of volume in the device giving us a maximum gain of 20 dB so that in sum with the 40 dB of the preamplifier we finally have 60 dB enough to treat mild to moderate hearing loss without compromising the user to hearing damage excessive sound.

$$V_{out} = -\left(\frac{R_f}{R_1}V_1 + \frac{R_f}{R_2}V_2 + \frac{R_f}{R_3}V_3\right)_{(5)}$$



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Figure 7: Gain and volume control

Figure 8 shows the designed circuit where the different stages of pre-amplification filtering and gain control are observed.



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Figure 8. Circuit corresponding to the Hearing aid for bilateral hearing loss

III. RESULTS AND DISCUSSION

Passive high pass filters

In this analysis we can observe how effectively we have a cutoff frequency in 72.5 Hz since when we put a commercial resistance it moved slightly from 72 to 72.5, which is within the expected (see figure 9).



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Figure 9. Second order bandpass filter with input attenuator, for Graves

In figure 10 we can see the analysis of bode where we can observe the frequency curve against dB, which are very close to the calculated ones. The results are not exact given that we did not use the calculated resistances, but the closest resistances that can be found commercially were used. So finally in table 5 we have the final values of operation that correspond to this filter.



Figure 10. Bode diagram for second order bandpass filter with input attenuator for Low frequencies. Table 5: Operation values of the second order bandpass filter with input attenuator for Low frequencies.

| Initial frequency | f_1 | 447 Hz |
|-------------------|-------|--------|
| Central frequency | f_c | 650 Hz |
| Final frequency | f_2 | 943 Hz |
| Gain | HOBP | 1 V |

Second order bandpass filter with input attenuator, for Media

In figure 11 we can see the analysis of bode where we can observe the frequency curve against dB, which are very close to the calculated ones. The results are not exact given that we did not use the calculated resistances, but the closest resistances that can be found commercially were used. So finally in table 6 we have the final values of operation that correspond to this filter.



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Figure 11. Bode diagram for second order bandpass filter with input attenuator for Medium frequencies.

Table 6: Operation values of the second order bandpass filter with input attenuator for Medium frequencies.

| Initial frequency | f_1 | 913 Hz |
|-------------------|-------|----------|
| Central frequency | f_c | 1.37 KHz |
| Final frequency | f_2 | 2 KHz |
| Gain | HOBP | 1 V |

Second order bandpass filter with input attenuator, for Treble.

In figure 12 we can see the analysis of bode where we can observe the frequency curve against dB, which are very close to the calculated ones. The results are not exact given that we did not use the calculated resistances, but the closest resistances that can be found commercially were used. So finally in table 7 we have the final operation values that correspond to this filter.



Figure 12: Bode diagram for second order bandpass filter with input attenuator for High frequencies.

Tabla 7: Valores de operación del filtro paso banda de segundo orden con atenuador de entrada parafrecuencias Altas.

| Initial frequency | f_1 | 1.94 KHz |
|-------------------|-------|----------|
| Central frequency | f_c | 2.9 KHz |
| Final frequency | f_2 | 4.48 KHz |
| Gain | HOBP | 1.1 V |



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For this last stage of processing the audio signal we will analyze how the filters and the adder work together when varying the gains. So we start by observing how the curves of the filters behave, in a voltage-frequency relationship, individually (see figure 13), compared to figure 14 where we can see how the filters work (Serious, Medium and Sharp) when going through the adder.



Figure 13: Filters (High Pass, High Pass for Medium and Treble Bass)



Figure 14: Filters (High pass for medium and high tones) after passing through the adder

We can see in Figure 14 that we have an approximate gain of 2 V that correspond to the middle frequencies, this is because it is where the curves of both low and high frequencies intersect, which favors us since regularly the necessary frequencies and that commonly they require to be reinforced, they are those that are included between 1 KHZ and 3 KHz, since these are the frequencies where most of the sounds of the human voice are comprised and therefore the most necessary to interact and be able to communicate with others Humans.



Bode analysis and corroboration in the oscilloscope

Now the Bode analysis will be done and the results obtained with the oscilloscope tests will be corroborated for different possible configurations that can be done with the variable resistances, and in this way it will be possible to understand how the resistances can be modified and thus be able to reinforce certain frequencies in the case of being necessary in an optimal way, as well as attenuating or giving profit to the rest of the frequencies depending on the final needs of the user.

Figure 15 shows Bode's analysis of the filters operating in conjunction with the adder by setting the gains of the resistances of each band (bass, mid and treble) and volume to 100%. With which we can see that the three bands reach almost 20 dB that in sum with the 40 dB of the preamplifier we will finally have a maximum gain very close to 60 dB, so we finally achieve the expected result.



Figure 15. Bode analysis of the second order bandpass filters with input attenuator (Serious, Medium and Treble) when passing through the adder with the maximum gain.

We see that the tests made in the oscilloscope (see figure 16) in comparison with those made in the analysis of Bode (see figure 15), coincide since the gain of 19.6 dB is equal to 9.5 V and the scale of the output signal is 10 V unlike the input that is 1 V.



Figure 16. Oscilloscope test of the filters together with the adder. CH1 input signal (1 Vp scale), CH2 output signal (10 Vp scale) A) 618 Hz. B) 1.37 KHz. C) 3.11 KHz

In the case of figure 17 the resistances were adjusted in 20% for bass, 0% in media, 90% in treble and finally 40% in volume to finally have an approximate gain of 37.1 dB in bass, 34.29 dB in media and 32 dB in treble, because you have to remember that you have to add the 40 dB of the preamplifier.



Figure 17. Bode analysis of the second order bands step filters with input attenuator with the resistances in 20% for Graves, 0% for Media, 90% for Trebles and 40% in the volume of the adder.

We see that the tests made in the oscilloscope (see figure 18) in comparison with those made in the analysis of Bode (see figure 17), coincide and since the scale of the output signal is the same as that of the input signal .



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Figure 18. Oscilloscope test of the filters together with the adder. CH1 input signal (1 Vp scale), CH2 output signal (1 Vp scale) A) 595 Hz. B) 905 Hz C) 2.98 KHz

In the analysis of figure 19 we can see how you can give a higher gain to the bass (16.9 dB with 100% resistance), a much lower gain to the media (4.16 dB with resistance to 1%) and a gain approximate to 10 dB in the highs with the resistance to 40% and finally the volume to 1%.



Figure 19. Bode analysis of second-order bandpass filters with input attenuator with resistances in 100% for Graves, 1% for Media, 40% for Trebles and 1% in the volume of the adder.

We can appreciate that the tests made in the oscilloscope (see figure 20) in comparison with those made in the analysis of Bode (see figure 19), coincide with finding that the input signal is on a scale of 1V and the output signal in 100 mV.



Figure 20. Test on the oscilloscope of the filters together with the adder. CH1 input signal (1 Vp scale), CH2 output signal (100 mVp scale) A) 638 Hz B) 1.8 Hz C) 3.1 KHz

Finally, in Figure 21, the analysis was carried out in the extreme case of having all the resistances variable at 0% which gives us as a result the maximum attenuation of the signal, which corresponds approximately to 0 in the output signal. All this with the sole purpose of demonstrating that the circuit has the versatility to be modified according to the needs of the user.





Figure 21. Bode analysis of second-order bandpass filters with input attenuator with all resistors at 0%.

IV. CONCLUSION

Based on the numerical analysis, simulations and tests, we can conclude that the purpose with the design of the hearing aid has been reached because:

- This can satisfy the hearing needs of people with bilateral hearing loss with a maximum hearing loss of 60 dB.

- The device has the versatility to be modified, a number of ways to modify the variable resistances, in order to adjust the gains or attenuations of the bass, mid, treble and final volume (from -220 dB to 60 dB of gain) depending on the needs of the user.

- The hearing aid has an appropriate sweep of the average hearing frequencies ranging from 72 Hz to 20,000 Hz.

- They can be perfectly adjusted in a bandwidth ranging from 600 Hz to 3100 Hz. This was determined as this is where the human voice is comprised on average.

- In addition, we managed to further minimize any risk to the user by incorporating a preamplifier with automatic gain control into the device that reacts in a matter of 1.1 milliseconds.

- The design based on surface electronic circuits offers the possibility of having a final device of dimensions less than 10×15 cm, making this an easily portable device.

Finally, we have a design of the device of excellent quality that uses current technology and that can be used by people with bilateral hearing loss of high or low resources, thanks to the low price of integrated circuits.

V. ACKNOWLEDGEMENTS

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