An Impermeable Membrane Design for Hearing Aid Applications

By

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Abstract

A common problem when using a hearing aid is the accumulation of earwax, dust and water in the conduit that transmits the sound towards the ear. Hearing aids are damaged when unqualified people try to remove these foreign materials. An impermeable membrane can be placed at the opening of the hearing aid to avoid the entrance of dust or water and accumulation of earwax, without altering the frequency response of the hearing aid. The objective of this research project was to develop a tool to specify membrane mechanical properties and dimensions to achieve a specific frequency response that does not adversely affect the hearing aid's performance. The frequency response for various types of impermeable membranes were measured experimentally and compared to computer simulations. We found that the membranes with similar output to that of the system without membrane were made of polypropylene and latex, agreeing with the data collected in the simulations made on the computer.

Resumen

Un problema común cuando se usan audífonos es la acumulación de cerumen, polvo y agua en el conducto que transmite el sonido hacia el oído. Los audífonos se dañan cuando la gente no adiestrada intenta retirar estos materiales extraños. Una membrana impermeable se puede colocar en el orificio de salida del audífono para evitar la entrada de polvo o de agua y de la acumulación de cerumen, sin alterar la respuesta de frecuencia del audífono. El objetivo de este proyecto de investigación fue desarrollar una herramienta para especificar características y dimensiones mecánicas de la membrana para alcanzar una respuesta de frecuencia específica que no afecte adversamente el funcionamiento del audífono. La respuesta de frecuencia de varios tipos de membranas impermeables fue medida experimentalmente y comparada con las simulaciones por computadora. Se encontró que las membranas con respuesta más parecida al sistema sin membrana resultaron ser la de polipropileno y la de Látex, concordando con los datos obtenidos en las simulaciones por computadora.

Copyright

To my parents Carmen, Miguel (R.I.P.), my brother Gabriel, my sister Eloina, my daughter Laura Sofia and my wife Orisnella, I love you.

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h	thickness
ρ	density
c	sound speed
Z	acoustic impedance
k	wave number
P_i	incident pressure amplitude
P_r	reflectioned pressure amplitude
P_t	transmitted pressure amplitude
ω	frequency (radians)
R	reflection coefficient
T	transmission coefficient
$H(\omega)$	impulse response of the system
$x_f\,[n]$	input sequence
$y_f\,[n]$	output sequence
H_{nw}	transfer functions of the system without membrane
H_{wm}	transfer functions of the system with membrane
H_m	transfer function of the membrane

Chapter 1

Introduction

1.1 Clinical problem.

Hearing loss is a problem for many people, being more likely among the elderly. Most people over 65 years old suffer some kind of hearing loss problem and almost everyone over 80 years old is affected. Also, ear diseases are a major cause of hearing loss. The working environment is another important factor that influences the incidence of hearing loss among the elderly. Those who had worked in factories double the risk of suffering from hearing loss, as compared to people who had worked in offices or at administrative positions [1]. It is estimated that approximately 5 percent of the population of the United States has some sort of hearing problem [1].

Hearing aids are the most common solution to the hearing loss problem. These electronic devices amplify sound to compensate for the hearing loss of an individual. Sound pressure waves are converted into electricity by a microphone. The electric signals are then amplified through controlled electronic circuitry. The amplified electric signals are then converted to pressure waves by a speaker at a much more intense level.

According to the American Academy of Audiology, about 1.7 million hearing aid devices are sold annually. Common problems with the use of hearing aids are the accumulation of earwax, dust and the entrance of water to the device through the conduit that transmit the sound towards the ear. In addition, there are other problems such as interference with other electronic devices (cell and wireless phones), a short battery life, background noise, feedback, etc, that affect these devices. Studies [2, 3, 4] have been conducted to diminish or eliminate the adverse effects of electrical interference with hearing aids. This investigation centers its attention on the problem that results from the accumulation of earwax, dust and the entrance of water. The hearing aid can be damaged when unqualified people try to give maintenance to it and thereby, has to be taken to an audiologist for cleanup, which can be troublesome and time-consuming [5].

1.2. Methods to prevent damage of hearing aids due to water and earwax accumulation.

To prevent damage of a hearing aid from accumulation of earwax or from the entrance of water inside the device, a non-porous guard or membrane is utilized to completely cover the outside of the hearing aid. This is commonly used in two types of hearing aids: the "behind-the-ear" hearing aid and "in-the-ear" hearing aid (see figure 1). The protection is inserted in the inside of the orifice (see figure 2). This type of protection is replaceable, of a tubular form and has a collar design on the external side of the hearing aid on the inside of the ear where the membrane is located. It is placed with pressure on the part of the collar where the sound exits. It is also possible to fixate the membrane in other ways, such as, fixing the membrane with an external plastic ring or to fix it with glue to the outside of the hearing aid (see figure 3).

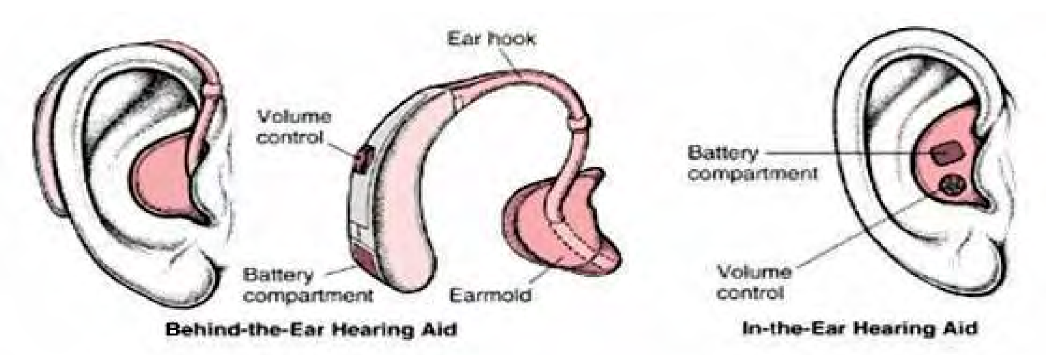


Figure 1. Hearing aids types. (Taken from [1])

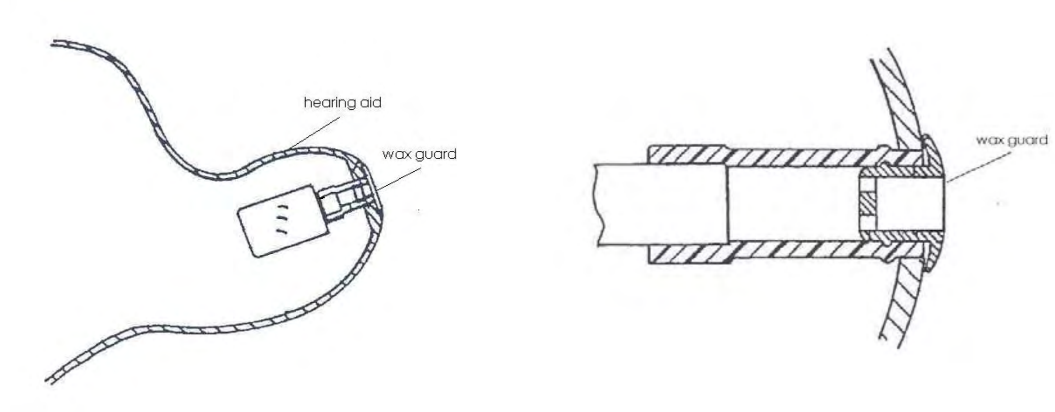


Figure 2. Wax guard for hearing aid. (Taken from [6])

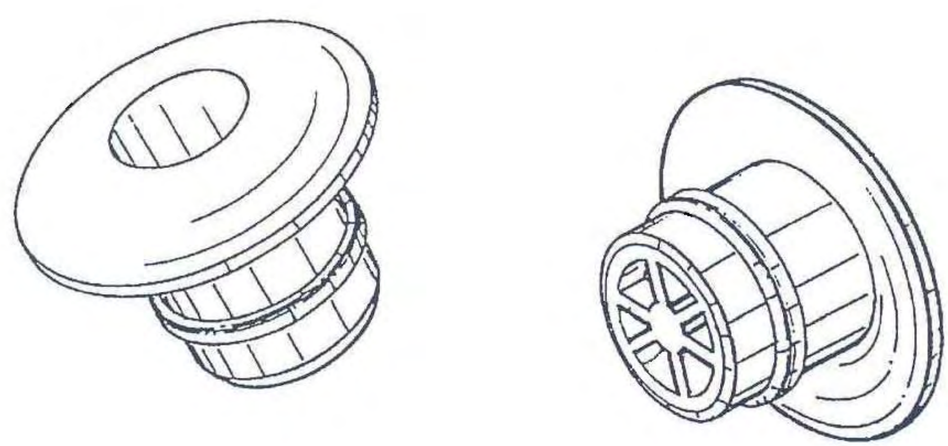


Figure 3. Ear wax membrane for hearing aid (In the Ear hearing aid). (Taken from [6])

Previous studies have been made by others on how to improve hearing aids. Berger [7] tried a non-porous guard in the form of a membrane or diaphragm completely covering the outlet of the hearing aid. They found that although wax may build up externally on the membrane, it may be easily removed with a wipe, whereas, without the wax guard, the sound outlet itself had to be cleaned with an instrument or other device, risking damage to the hearing aid. The membranes were made of plastic, metalized plastic and stainless steel. They had a diameter between 0.20 and 3/8 inches, and a thickness between 0.0005 and 0.001 inches. For the chosen thickness range and for membranes made of plastic or metalized plastic the results obtained, when a sound wave was applied, was similar in amplitude and frequency response as the first sound wave (see figures 4,5,6).

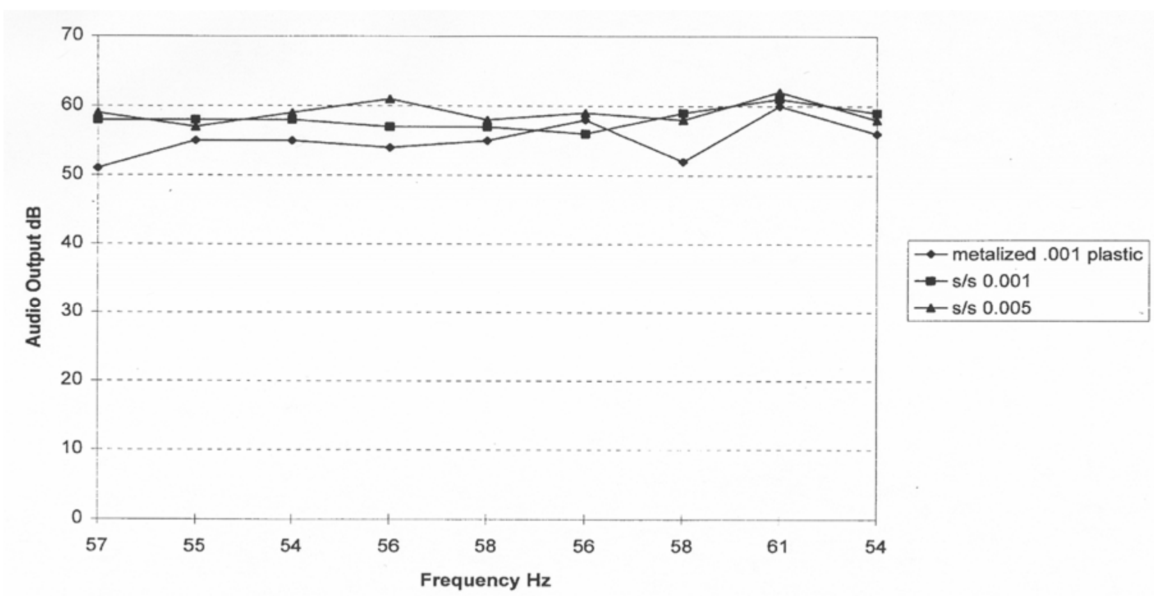


Figure 4. Frequency response for 0.20 inch diameter diaphragms.(Taken from [7])

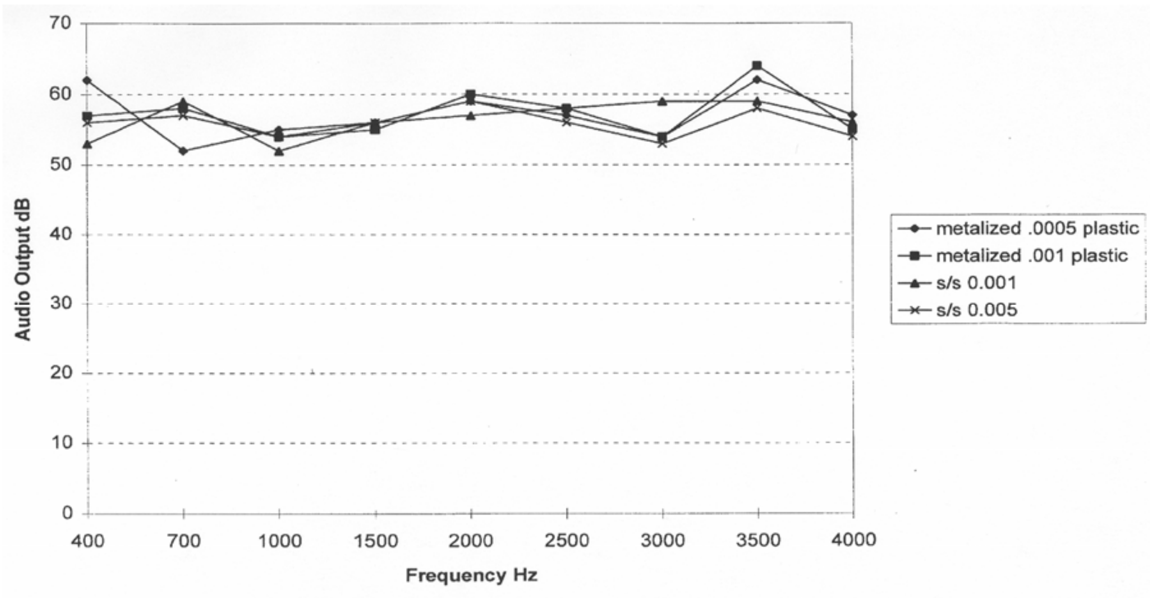


Figure 5. Frequency response for 0.40 inch diameter diaphragms. (Taken from [7])

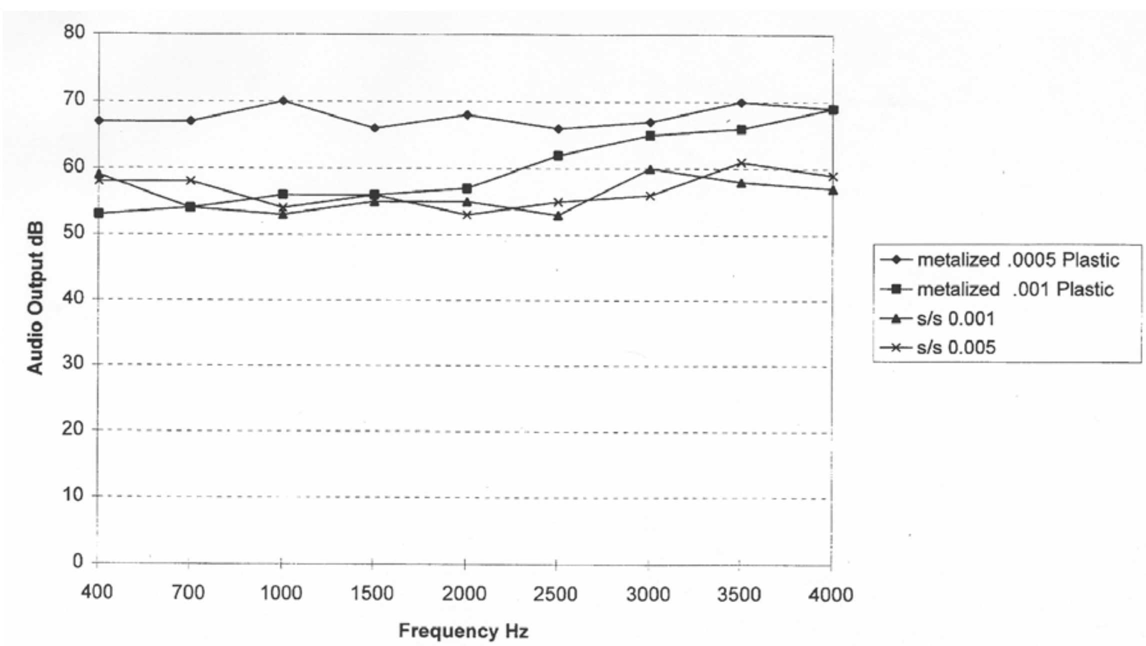


Figure 6. Frequency response for 3/8 inch diameter diaphragms. (Taken from [7])

Gunnersen et al. [6] placed very thin membranes at the distal end of the hearing aid without evaluating the behavior of this membrane with respect to the hearing necessities of the individual. This investigation was focused on creating an ear guard that could be easily removed to avoid the entrance of earwax on the device. Its canal was approximately the same diameter of the hearing aid canal (see figure 7) [6]. This study found that after almost 30 hours of use there was a buildup of approximately 20 mg of earwax on the guard, which accounted to 50% of its surface.

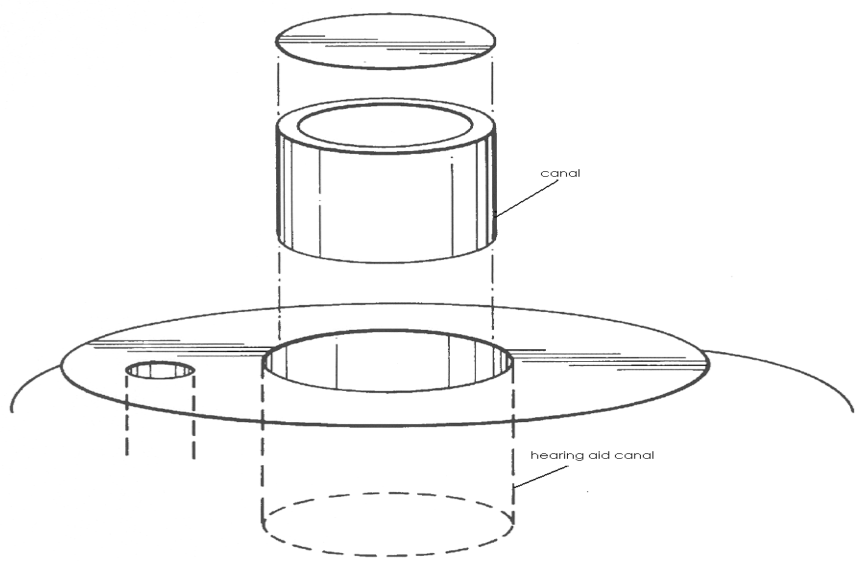


Figure 7 . Ear wax guard for an In-the-ear hearing aid. (Taken from [6])

Weiss et al. [8] created a barrier that prevented ear wax from contacting and damaging the internal components of the hearing aid but that attenuated its acoustic response (see figure 8). It had a series of partial plates in the canal of the guard that accumulated the ear wax as it started to build up on the hearing aid but at the same time, attenuated the sound due to their placement. (see figure 9)

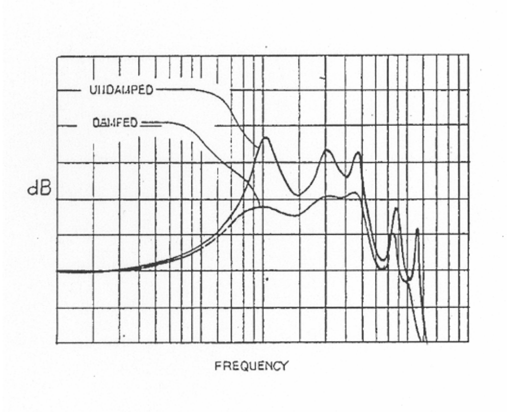


Figure 8. Damped and undamped frequency response of the hearing aid receiver. (Taken from [8])

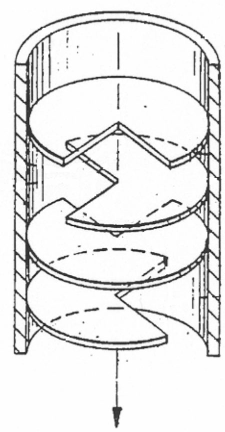


Figure 9. Partial cross sectional view of the barrier. (Taken from [8])

1.3. The usage of the reflection and transmission coefficients for a membrane design.

As sound travels through different media, it encounters different acoustical impedances that originate from changes in media or from changes in the dimension of the waveguide. A change in acoustical impedances, results in the phenomenon of reflection and transmission of sound waves [9]. This phenomenon can be quantified using the transmission and reflection coefficients, which measure the fraction of the sound wave

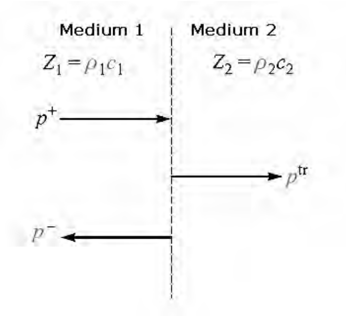


Figure 10. Reflection and transmission of sound at normal incidence. (Taken from [9])

that is transmitted from one medium to another and the fraction that is reflected back to the first medium (see figure 10), respectively.

Using these coefficients, an algorithm can be elaborated that simulates the effects of placing a membrane between two equal media, and that takes into consideration the basic characteristics and parameters of the membrane, such as thickness, density and elasticity. These effects are represented using the transmission coefficient as an acoustic filter. By using frequency-domain analysis techniques, it will be possible to determine how various types of impermeable membranes will distort sound waves as they propagate through them.

1.4. Objectives and Contributions.

The main objective of this project was to explore a mathematical model that could be used by medical device designers to appropriately select impermeable membranes to isolate acoustic transducers from water, dust, earwax or other foreign material without affecting the frequency response of the device. The sound transmission properties of various types of impermeable membranes were evaluated and compared to bench top measurements. Computer simulations were also be performed to estimate the effects of three key membrane parameters: thickness (*h*), density (ρ) and sound speed (*c*), on the membrane's overall acoustic response. The selected membrane was tested in the laboratory, and its experimental response compared to that obtained from the simulations. This comparison will help us make a tool to design any type of membrane depending on the necessities of the hearing aids and other medical device users and the specifications of the material at hand. Trials could not be conducted with people with hearing problems due to the high cost of hearing aids and lack of availability of an audiologist and its patients.

The contribution of this project to the field of biomedical engineering was to improve the design of hearing aids and other medical devices by designing an impermeable membrane that avoids the entrance of dust, earwax or water to the device, but without causing sound distortion.

1.5. Thesis Outline

This thesis is divided into six chapters. Chapter 2 presents the literature review related to this research. It describes the placement of very thin membranes at the distal end of the hearing aid and discusses the theoretical background regarding sound transmission through thin membranes. Chapter 3 explains the system requirements with details on data acquisition and the experimental setup. It also outlines the methods and materials that were used to achieve the desired impermeable membrane. The fourth chapter describes the discussion of the results obtained in the research, the experimental and model simulations results and their comparison. Chapter 5 covers conclusions and recommendations based on this research. Chapter 6 details the references used.

Chapter 2

Literature Review

This chapter addresses previous research efforts to avoid the entrance of earwax or dust to the hearing aid, and explains the theory that can be applied to model the reflection and transmission of sound waves through circular membranes. The first part of this chapter presents the use of wax guard membranes in the hearing aid and the effect caused in the device. The second part of this chapter deals with the general principles of the usage of reflection and transmission coefficients and their application on circular membranes. The final part of this chapter explains the mathematical methods employed to analyze the behavior in frequency of the sound wave in the setups utilized (with and without membrane).

2.1. Wax guard for hearing aids.

The use of the hearing aid has become more common throughout the world. It is the most used tool to help people with auditory loss return to a normal social life [1]. Hearing aids can be damaged by their users when they try to remove the earwax from them. In order to correct this potential problem, several investigations [6, 7, 8] have been made in which a wax guard (membrane, barrier, protection) is placed outside or into the hearing aid to avoid the entrance of earwax. Usually the wax guard membrane is made of plastic (polypropylene, latex or thick latex) or metalized plastic [7], has a diameter between 0.20 and 0.375 inches and a thickness between 0.0005 and 0.001 inches. The methods used to attach the wax guard membrane may vary from completely inserted, semi inserted or external to the hearing aid canal (see figure 11).

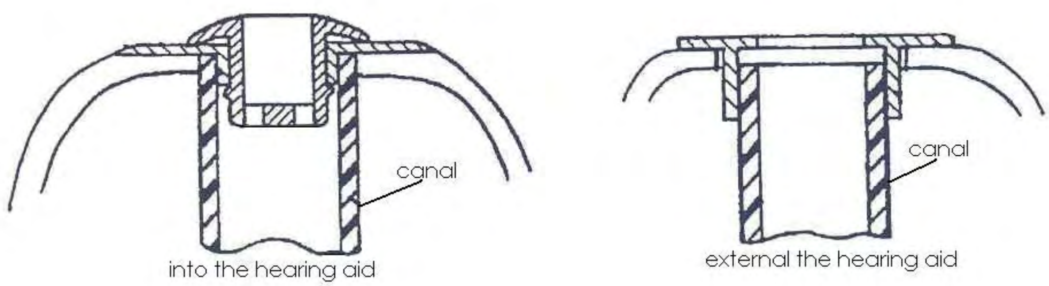


Figure 11. Cross sectional view of two different positions of the wax guard in a hearing aid housing. Taken from [6].

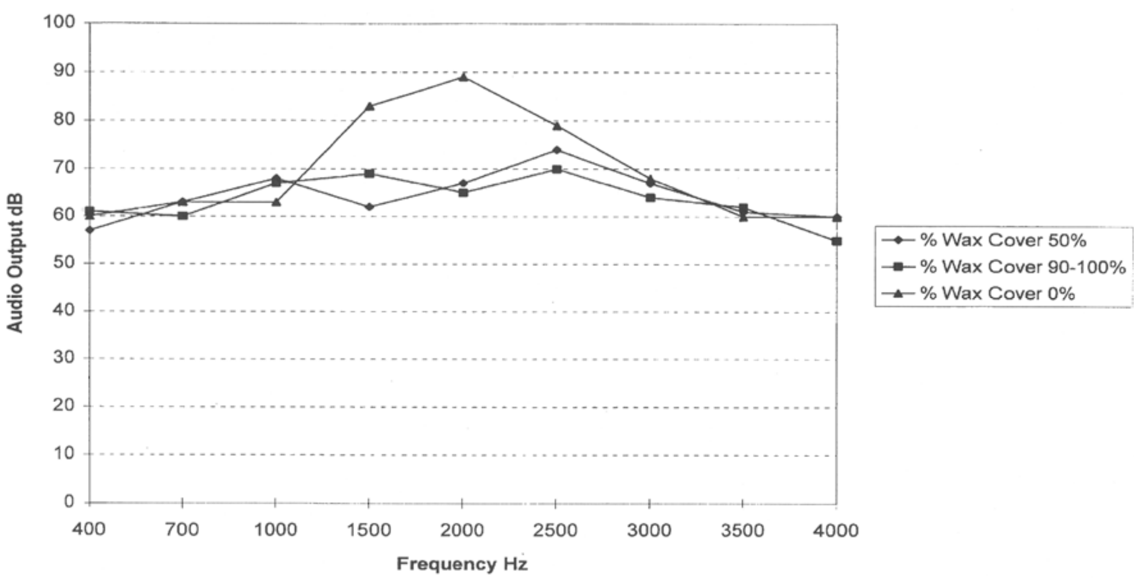


Figure 12. Effect of earwax on frequency response for differents values of wax cover. Taken from [7]

When earwax guard membranes are used at the distal end of the hearing aid, there is a probability of sound distortion because the membrane will attenuate and possibly reflect part of the sound generated by the hearing aid. Generally, this distortion can be minimized if the thickness of the wax guard membrane is thin enough so that the sound wave can completely go through the membrane. Earwax tends to accumulate externally on these membranes, and this build up may be easily removed with a wipe or tissue but it may cause attenuation in the sound wave as that accumulation increases. As shown in Figure 12 this sound attenuation is more emphasized at frequencies between 1500 and 2500 Hz [7].

There are other hearing aids (see figures 13 and 14) that are a hundred percent water-tight, exclude sweat, sand and dust, and are hermetically sealed so they can remain submerged at one meter of water for as long as 10 minutes without any problem or impairment. The fact that they can tolerate such conditions makes them convenient for people who must work under extreme weather conditions or for those who are keen of water sports. As a result of its strength, the instrument is less susceptible to breakdown, and therefore its features make it ideal for children [10].

Figure 13. Water proof hearing aid model HB-54. Taken from [10]

Figure 14. Internal Parts of Hearing aid model HB-54. Taken from [10]

The sealing of the two-section casing, the battery compartment and trimmer cover are provided by rim labyrinth seals. Receiver and microphone tubes are provided with flange-shaped protuberances which hermetically seal the instrument against moisture on fitting the ear hook. A waterproof diaphragm has been fixed in front of the microphone, while the switching elements are protected by accurate sweat-proof rubber seals [10].

2.2. Acoustic principles and previous work.

This section presents the acoustic principles and previous works in hearing aids applications. The first part shows the equations used to determine the pressure transmission coefficient filter. The second part illustrates the results of several investigations related to wax guards for hearing aids.

2.2.1. Pressure reflections and transmission coefficients.

The reflection and transmission coefficients can indicate the fraction of the sound wave that crosses from one medium to another and the fraction reflected at the first medium (see figure 15). The transmission phenomenon can be quantified using the transmission coefficient, which can be used to describe sound distortion through a membrane.

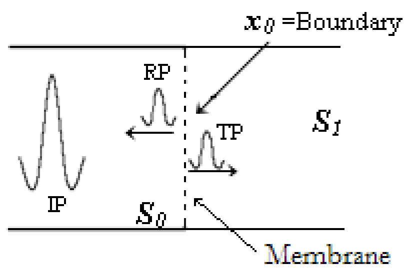


Figure 15. Transmitted Pulse in a tube with membrane. Taken from [11].

The acoustic principles applied to the reflection and transmission of a wave through different media will be used to analyze the behavior of sound waves, at different frequencies, in the presence of a membrane. A sound wave experiences a reflection whenever it encounters a change in acoustic impedance [9]. At any interface *(x_o)* between two different acoustic impedances, the pressure reflection and transmission coefficients at the boundary are given by

$$R_{x_0} = \frac{Z_2 - Z_1}{Z_2 + Z_1} \tag{1}$$

$$T_{x_0} = \frac{2\sqrt{Z_2 Z_1}}{Z_2 + Z_1} \tag{2}$$

where Z_1 and Z_2 are the acoustic impedances before and after the boundary between both media, respectively, and which depend on the densities and the sonic speed in each medium. In the case when there is a section of different acoustic impedance used to separate the acoustic media (see figure 16), the pressure reflection and transmission coefficients at the first boundary (*x=0*) are

$$R_x(\omega) = \frac{(1 - Z_1/Z_3)\cos k_2 h + j(Z_2/Z_3 - Z_1/Z_2)\sin k_2 h}{(1 + Z_1/Z_3)\cos k_2 h + j(Z_2/Z_3 + Z_1/Z_2)\sin k_2 h} \tag{3}$$

$$T_x(\omega) = \frac{2}{(1 + Z_1/Z_3)\cos k_2 h + j(Z_2/Z_3 + Z_1/Z_2)\sin k_2 h} \tag{4}$$

where Z_1, Z_2 and Z_3 are the acoustic impedances of medium 1, 2 and 3 respectively, and k_2 is the wave number of medium 2 [9, 11] (see figure 16). If the thickness h of medium 2 is very small ($k_2 h << 1$), (3) and (4) reduce to

$$R_{x_0}(\omega) = \frac{Z_3 - Z_1}{Z_3 + Z_1} \tag{5}$$

$$T_x(\omega) = \frac{2Z_3}{Z_3 + Z_1} \tag{6}$$

which are the reflection and transmission coefficients, respectively, for the ordinary two-medium problem. Therefore, thin membranes of solid materials can be placed between two media without affecting the low frequency acoustic reflection and transmission properties of the interface. At higher frequencies, the effects of the membrane on sound distortion will increase. Membrane parameters (size, dimensions, tension and type of material) will be varied in order to minimize the effects of the membrane on sound transmission.

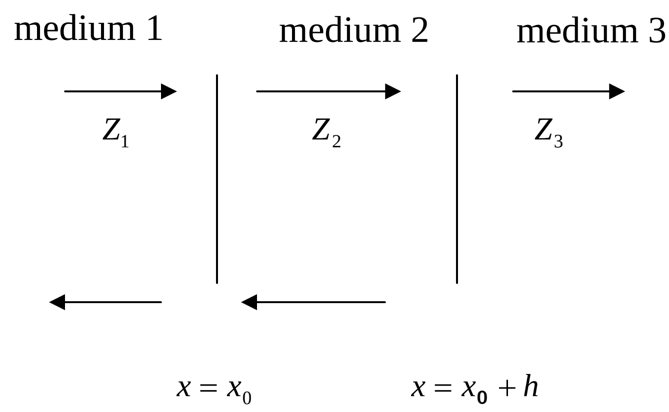


Figure 16. Incidence, Reflection and Transmission of waves of sound in three means.

According to (3) and (4), the reflection and transmission properties of a thin membrane placed between two media will depend on the thickness *h* of the membrane, density ρ and the sound speed *c* within the membrane. We can use these parameters to design membranes with a particular frequency response. Several types of membranes will be acquired and their performance will be tested experimentally and compared with results obtained via simulations.

2.2.2. Previous works in hearing aids applications.

There have been several investigations in which very thin membranes were placed at the distal end of the device, but without evaluating the behavior of this membrane with respect to the hearing necessities of the individual [6, 8]. These studies only center their attention on avoiding the entrance of some type of materials into the apparatus and the deterioration of the capacity of transmission of the sound into the ear. For the protection of an in-the-ear hearing aid against contamination by ear wax through the acoustic outlet port or a vent, a replaceable ear wax guard is inserted in the aid and comprises an essentially tubular element with a through-going cavity and an abutment collar in one end for sealing against the hearing aid housing. For an easy and safe insertion and removal of the ear wax guard, an applicator is used, which in one end has a smooth pin for introduction in the through-going cavity of the ear wax guard and in the other end a harpoon-shaped catch member. To mount the ear wax guard, a hose or tube member serving as acoustic outlet canal is connected to an abutment collar in abutment with the outside of the hearing aid. The abutment collar is designed with oversized standard dimensions and adapted to individually user-adapted hearing aid housing by preparation of its periphery edge [6].

Simultaneously, similar devices to avoid the entrance of dust or wax to the hearing aid have been patented [6, 7]. In these studies, researchers have made acoustic analysis (frequency response) of the behavior of different types of membranes without mentioning any type of element (filter) that compensates the on-speed operation of the hearing aid. A non-porous wax guard for an in-the-canal hearing aid is in the form of a membrane or diaphragm which completely covers the mouth of the round or other shape outlet of the hearing aid. The membrane is made of plastic or metalized plastic, or stainless steel, having diameters between 0.20 inches and 0.375 inch, and a thickness between 0.0005 inches and 0.001 inches. The membrane is affixed to the mouth of the sound outlet by a number of methods. It may be attached to a thin ring of plastic material, and attached with a spring clip to a recess in the sound outlet. It may simply be bonded, by adhesive or heat bonding, to the recess. It may be affixed to a cylindrical mount, and press-fit into the port.

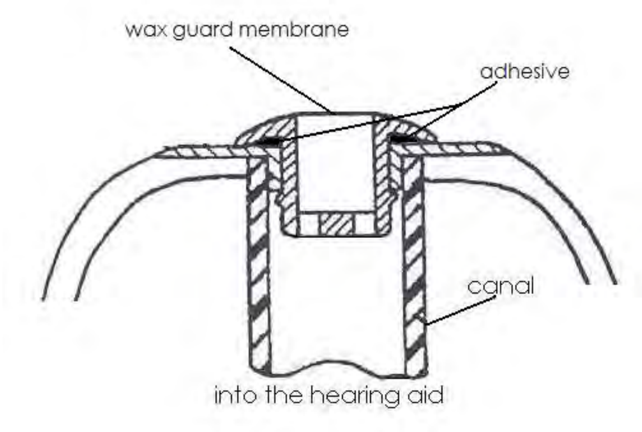


Figure 17. Outlet of the Hearing aid and the wax guard. Taken from [7]

As an alternative, the cylindrical mount may be threaded, and mated with an internal thread cut into the sound outlet. Although wax may build up externally on the membrane, it may be easily removed with a wipe or a tissue, whereas, without the wax guard, the sound outlet itself must be cleaned with an instrument or other device, risking damage to the hearing aid or requiring a hearing aid professional to do the cleaning [7].

Studies have been made to analyze the acoustic behavior of several types of impermeable membranes, using digital signal processing techniques to compensate the deterioration caused by the membrane in the sound generated by the hearing aid [8]. These studies analyzed the behavior of several types of membranes (latex 70µm of thickness, latex heavy of 140µm of thickness and polyethylene with 200µm of thickness) under an acoustic pulse.

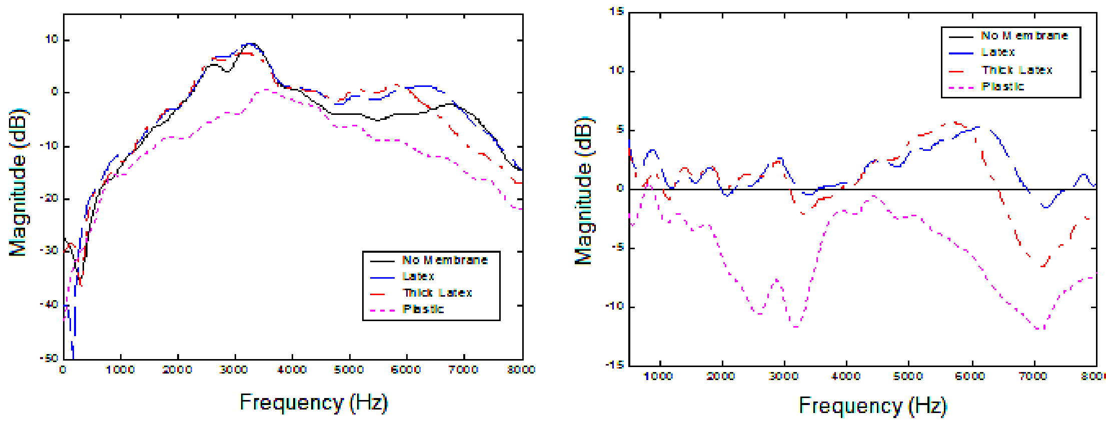


Figure 18. Acoustical Characterization of Impermeable Membranes: Hearing Aid Applications. Taken from [12]

The system consisted of acoustic transducers, amplifiers and several membranes. The study found that for frequency values below 4 kHz, the frequency responses of the overall system when thin latex and thick latex membranes were used similar to those obtained when no membrane was present (see figure 18). At higher frequencies, the responses obtained with both latex membranes deviate more from the response obtained without a membrane. Therefore, if a thin membrane (as compared to the wavelength of the sound signal) is placed between two acoustic media, one can assume that the membrane will have no effect on the sound transmission and reflection [12].

2.3. Analysis of signal frequency and system behavior with and without the impermeable membrane.

For the frequency analysis of a sound or electric signal we used one type of transform, which indicate the range of frequencies in which a certain system can be tested using an acoustic pulse. In this case, we specifically used the Fast Fourier Transform, which shows the frequency range in which the system (speaker, tube, and microphone) can respond to a sound pulse that has a particular frequency content. The acoustic behavior for a system can vary depending on the different types of membranes used to protect the hearing aid from the entrance of water or earwax to the device, as well as the behavior of the system without any type of protection.

2.3.1. Transmission from one medium to another through a layer: normal incidence.

When an acoustic wave traveling in one medium encounters the boundary of the second medium, reflected and transmitted are generated. Discussion of this phenomenon is greatly simplified if it is assumed that both the incident wave and the boundary between the media are planar [11].

The ratios of the pressure amplitudes and intensities of the reflected and transmitted waves to those of the incident wave depend both on the characteristic acoustic impedances and speeds of sound in the two media and on the angle the incident wave makes with the normal to the interface. Let the incident and reflected waves travel in a medium of characteristic acoustic impedance $Z_1 = \rho_1 c_1$ where ρ_1 is the equilibrium density and c_1 the sound speed in the medium one. Let the transmitted wave travel to a second medium of characteristic acoustic impedance $Z_2 = \rho_2 c_2$. If the complex pressure amplitude of the incident wave is P_i, that of the reflected wave P_r, and the transmitted wave P_t, then we can define the *pressure transmission and reflections coefficients*

$$T = P_t / P_i \tag{7}$$

$$R = P_r / P_i \tag{8}$$

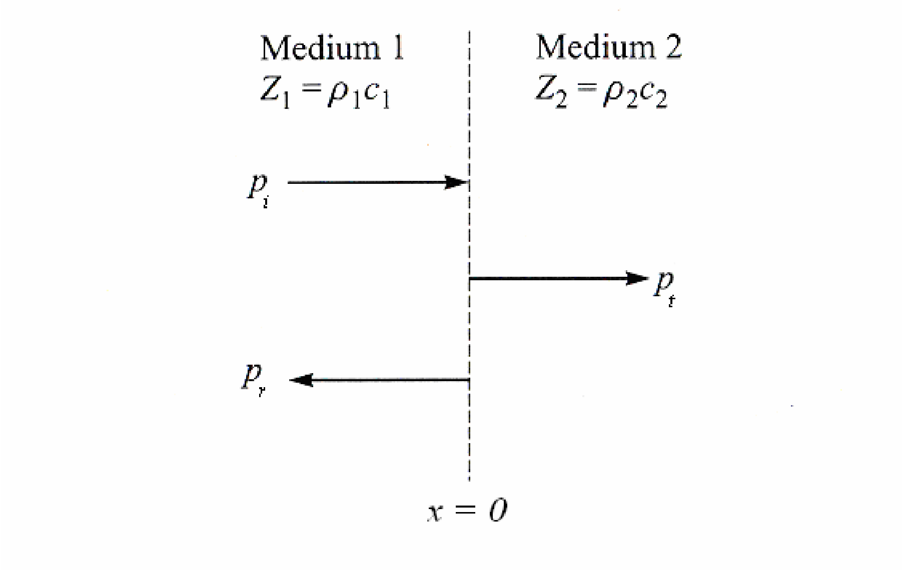


Figure 19. Reflection and transmission of plane wave normally incident on a boundary. Taken from [11]

In Figure 19, let the plane $x = 0$ be the boundary between medium 1 of the characteristic acoustic impedance Z_1 and medium 2 of the characteristic impedance Z_2. Let there the be an incident wave traveling in the positive x direction,

$$p_i = P_i e^{j(\omega t - k_1 x)} \tag{9}$$

which, when striking the boundary, generates a reflected wave

$$p_r = P_r e^{j(\omega t + k_1 x)} \tag{10}$$

and transmitted wave

$$p_t = P_t e^{j(\omega t - k_2 x)} \tag{11}$$

The transmitted wave has the same frequency as the incidents wave but, because of the different phase speeds c_1 and c_2, the wave number $k_1 = \omega/c_1$ in medium 1 and $k_2 = \omega/c_2$ in medium 2 are different [11].

There are two boundary conditions that must be satisfied for all times at all points on the boundary: first, the acoustic pressures on both sides of the boundary must be equal and second, the particle velocities normal to the boundary are equal. The first condition, *continuity of pressure*, means that there can be no net force on the plane separating the media. The second condition, *continuity of normal velocity*, requires that the fluids remain in contact [11]. The pressure and normal particle velocity in medium 1 are $p_i + p_r$ and $(u_i + u_r)$, so that the two boundary conditions are

$$p_i + p_r = p_t \qquad at\ x = 0 \tag{12}$$

$$u_i + u_r = u_t \qquad at\ x = 0 \tag{13}$$

Division of (12) by (13) yields

$$\frac{p_i + p_r}{u_i + u_r} = \frac{p_t}{u_t} \qquad at\ x = 0 \tag{14}$$

which is a statement of the continuity of normal specific acoustic impedance across the boundary. Since a plane wave has $p/u = r$, depending on the direction of propagation, (14) becomes

$$Z_1 \frac{p_i + p_r}{p_i - p_r} = Z_2$$

which leads directly to the reflection coefficient

$$R = \frac{Z_2 - Z_1}{Z_2 + Z_1} = \frac{1 - Z_1/Z_2}{1 + Z_1/Z_2} \tag{15}$$

Then, since (12) is equivalent to $1 + R = T$, we have

$$T = \frac{2Z_2}{Z_2 + Z_1} = \frac{2}{1 + Z_1/Z_2} \tag{16}$$

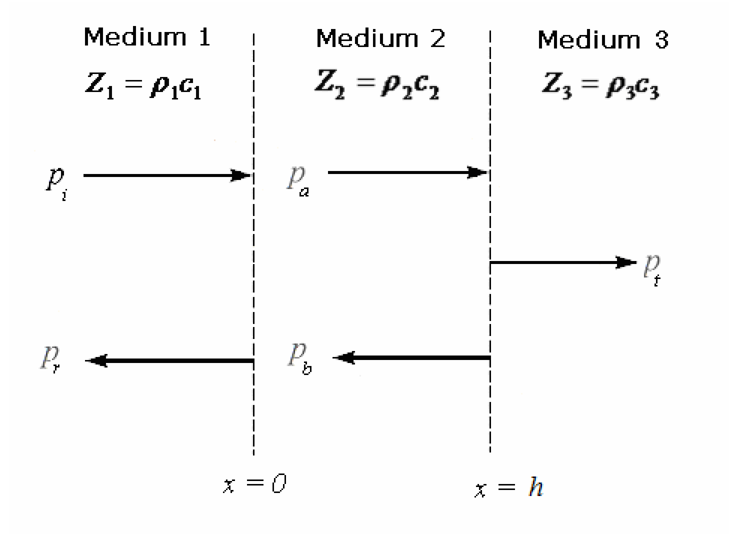


Figure 20. Reflection and transmission of plane waves normally incident on a layer. Three media. Taken from [11]

Assume that a layer of uniform thickness *h* lies between two dissimilar media and that plane wave is normally incident on its boundary (figure 20). Let the characteristic impedances of the media be Z_1, Z_2 and Z_3, respectively. When an incident signal in medium 1 arrives at the boundary between media 1 and 2, some energy is reflected and some is transmitted into the second medium. The portion of the wave transmitted will proceed through medium 2 to interact with the boundary between media 2 and 3, where again some of the energy is reflected and some transmitted. The reflected wave proceeds back to the boundary between media 1 and 2, and the whole process is repeated [9].

On the other hand, if the incident wave train is long compared to 2l, and monofrequency, it can be assumed to be

$$p_i = P_i e^{j(\omega t - k_1 x)} \qquad (17)$$

The various transmitted and reflected waves now combine so that in steady-state condition the wave reflected into medium 1 may be represented by

$$p_r = P_r e^{j(\omega t + k_1 x)} \tag{18}$$

the transmitted and reflected waves in medium 2 by

$$p_a = A\, e^{j(\omega t - k_2 x)} \tag{19}$$

$$p_b = B e^{j(\omega t + k_2 x)} \tag{20}$$

Respectively, and the wave transmitted into the medium 3 by

$$p_t = P_t e^{j(\omega t - k_3 x)} \tag{21}$$

Continuity the normal specific acoustic impedance at $x = 0$ gives

$$\frac{P_i + P_r}{P_i - P_r} = \frac{Z_2}{Z_1}\frac{A + B}{A - B} \tag{22}$$

Similarly, at $x = l$

$$\frac{Ae^{-jk_2 l} + Be^{jk_2 l}}{Ae^{-jk_2 l} - Be^{jk_2 l}} = \frac{Z_3}{Z_2} \tag{23}$$

Algebraic manipulation then reveal the form of the pressure reflection and transmission coefficient

$$R_{x_0} = \frac{(1 - Z_1/Z_3)\cos k_2 h + j(Z_2/Z_3 - Z_1/Z_2)\sin k_2 h}{(1 + Z_1/Z_3)\cos k_2 h + j(Z_2/Z_3 + Z_1/Z_2)\sin k_2 h} \tag{24}$$

$$T_{x_0} = \frac{2}{(1 + Z_1/Z_3)\cos k_2 h + j(Z_2/Z_3 + Z_1/Z_2)\sin k_2 h} \tag{25}$$

2.3.2. Technique to obtain the behavioral results of the system with membrane.

In this section, we demonstrate how the mathematical model presented on this chapter can be used to validate experimental results. Those equations correspond to the transmission coefficient that represent the ratio between the incoming and the outgoing pressure from the membrane. First of all, the transmission coefficient equation generates a filter which takes in consideration all the characteristics of a membrane. This equation (25) can be mathematically transformed to result in a filter that acts as a mirror-imaged among frequencies of 0-20 kHz and 20-40 kHz. The result filter multiplied by the sound wave affecting the membrane yields the exit wave.

$$T_{x_0} = \frac{2}{(1+Z_1/Z_3)\cos k_2 h + j(Z_2/Z_3 + Z_1/Z_2)\sin k_2 h} \tag{25}$$

These filter obtained on the previous equation can simulate the behavior of a membrane when the sound wave crosses it (see Figure 21).

This chapter exposed several works on the area of guards that help prevent the entrance of any type of foreign material for different types of hearing aids. Basically, four types of investigations were mentioned. They focused their attention on different approaches to avoid the entrance of earwax, dust or water to the hearing aid. Also the method used to simulate the membranes behavior as an acoustic filter was described.

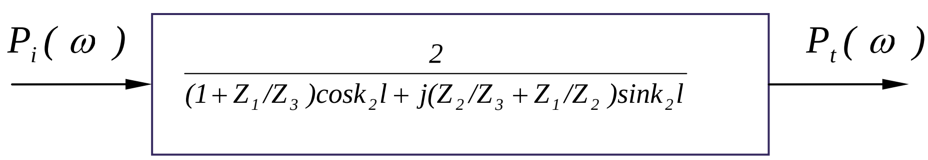


Figure 21. Transmission coefficient filter as acoustic filter, that represent the membrane behavior.

The main purpose of this project was to develop a tool for selecting impermeable membranes that can be placed at the opening of hearing aids and others medical devices, to avoid the entrance of dust and accumulation of earwax but without affecting the frequency response of the device. This will help us make a tool to design any type of membrane depending on necessities of the hearing aid user and medical device designers.

Chapter 3

Materials and Methodology

The purpose of this chapter is to describe the experimental procedures and simulations that were performed in this research. The first part of the chapter explains the various experimental setups that were employed to perform acoustic measurements in plastic tubes with and without membranes. The second part describes the computer analysis of sound pulses acquired with different types of membrane's material and in different types of setups. The last part describes the computer simulations parameters used to predict the experimental results.

The flow chart presented in Figure 22 outlines the steps followed during our investigation to achieve the goals of this research.

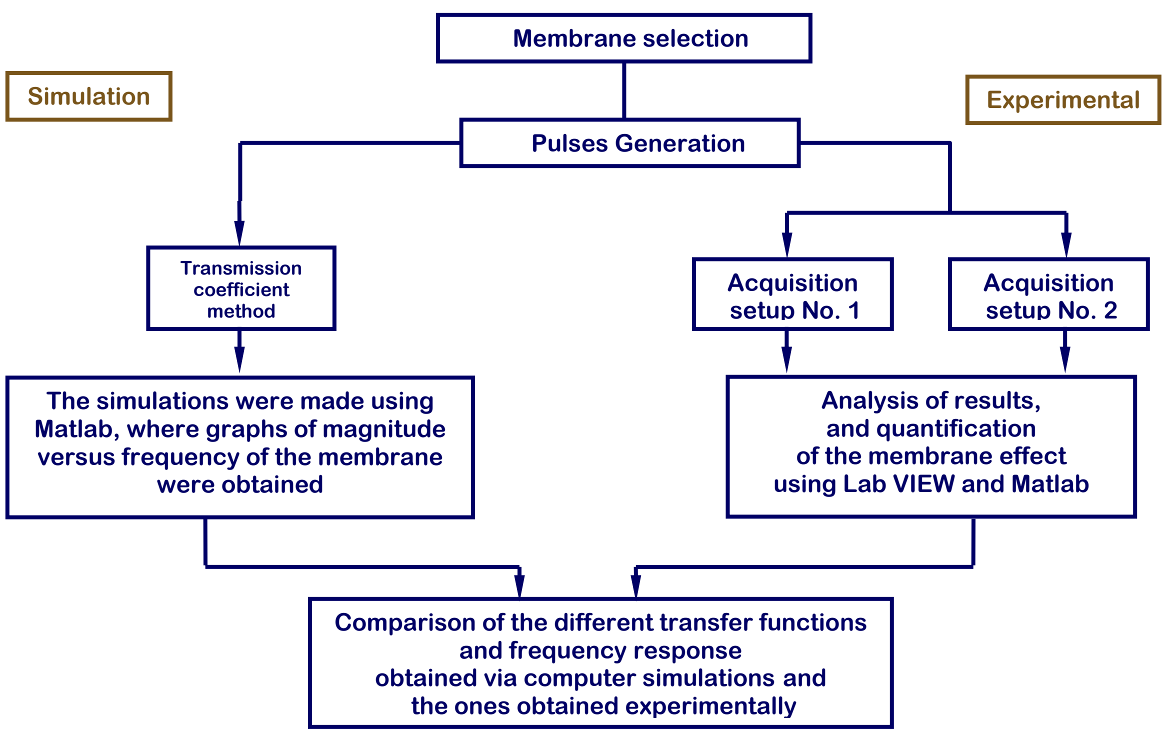


Figure 22. Flow chart of the general procedure to follow the investigation.

Initially the specific parameters (density, sound velocity and thickness) of the membrane are selected, next acoustic pulses are generated to experimentally test the membranes and simulate these membranes with the transmission coefficients filter and then the results obtained via computer simulations and experimentally are compared to validate the model.

3.1 Experimental setup and procedures employed.

To achieve the goals of this research project, we will describe the materials used at the two laboratory setup and the methodology followed to obtain the necessary data. Figures 23 and 24 are schematic representations of the setups used. They also show the equipment employed to acquire and process the acoustic signal. Both setups were made

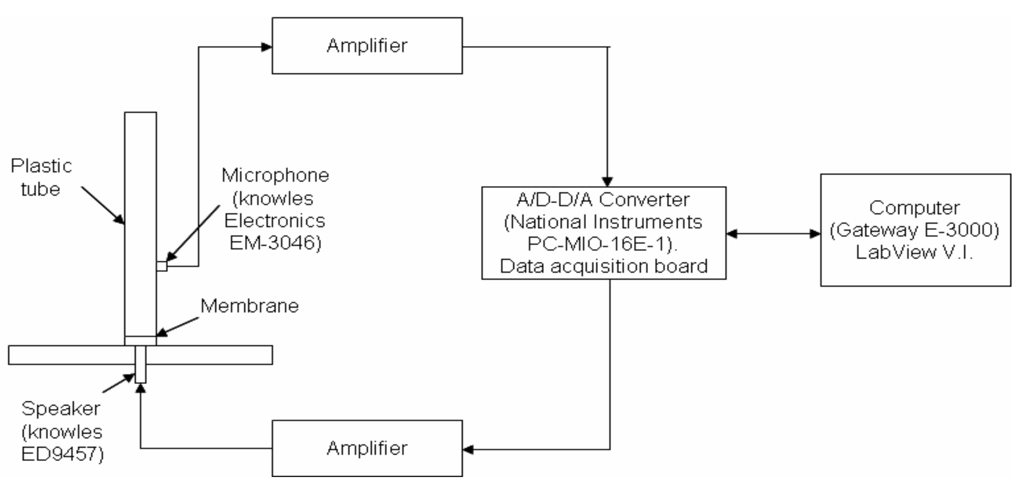


Figure 23. Schematic of experimental setup No. 1.

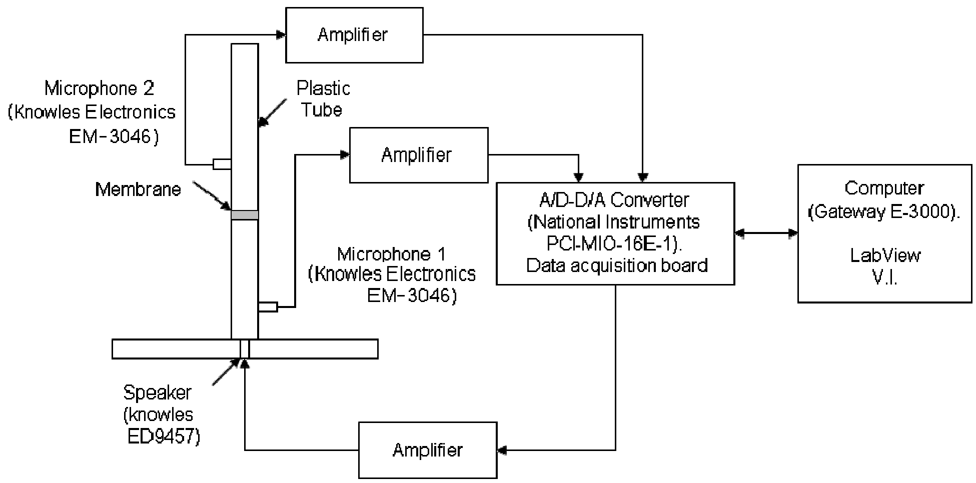


Figure 24. Schematics of experimental setup No.2.

of plastic tubes, but with different dimensions, also the way that the signal is captured varies. On the first setup, one speaker and one microphone are used whereas on the second, one speaker and two microphones are used while the rest of the components of the system remain the same. The flow chart shown in Figure 25 summarizes the process of operation of both setups.

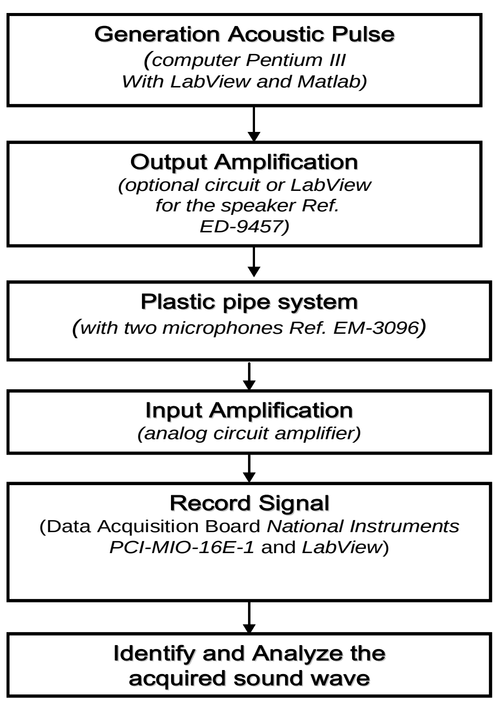


Figure 25. General procedures followed for both experimental setups.

- **Generation of the acoustic pulse**. An acoustic pulse is generated by a Pentium III computer (using Lab VIEW 7.0 and Matlab 7.0) which was used to probe the system without a membrane. This step will be explained in section 3.1.1.
- **Output amplification.** Using the Lab View software, the electric pulse is generated and then passed through the acquisition board (PCI-MIO-16E-1) to the setups (speaker ED-9457). This pulse has a duration of approximately 0.5 ms and an amplitude of 6 mV.
- **Plastic pipe system.** This system consist of a flexible, 0.5 in diameter, plastic tube with a total length of 156 cm. Depending on the setup, there were two possible membrane locations. Figure 26 depicts the dimensions and placement of the components of the two setups and demonstrates how the first setup is part of the second setup.

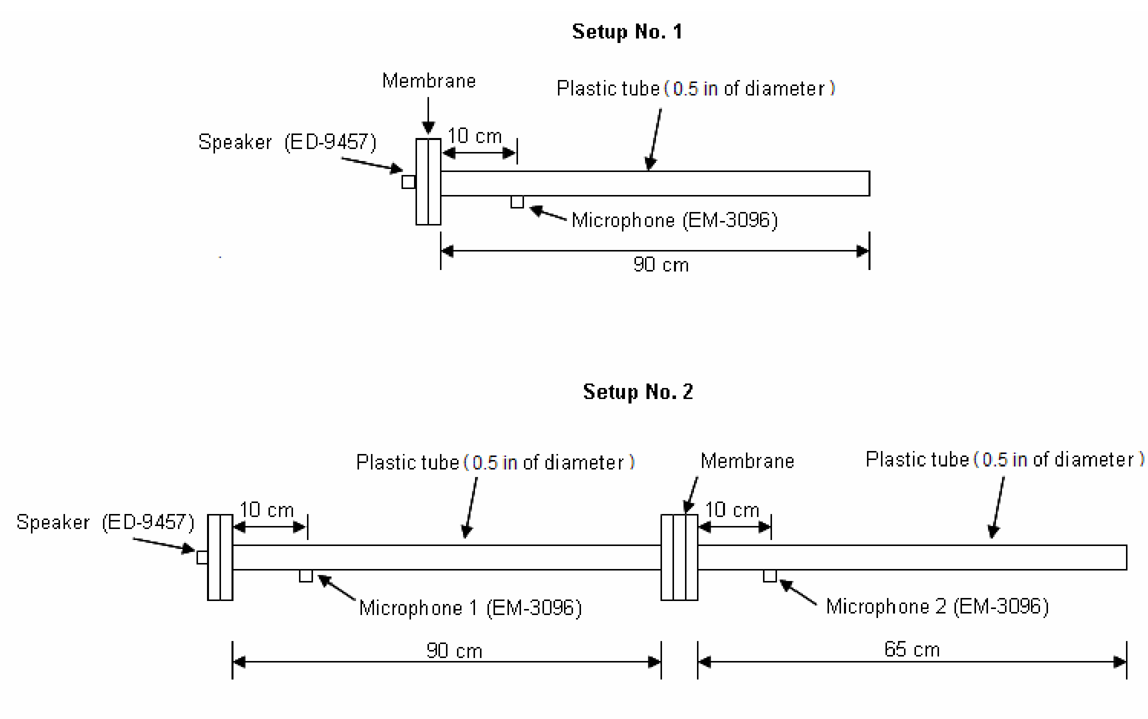


Figure 26. Dimensions of the two different setups.

- **Input amplification.** Since the signal sent by the speaker and sensed by the microphone was very small, amplification was needed. This was achieved by an electric circuit that has the function of providing the necessary voltage (with a voltage regulator lm7805) so that the microphone could sense the sound. At the same time, this signal was recorded, amplified (Gain = 100) and digitalized using a data acquisition board and analyzed in the computer with the Lab View and Mat lab software. Figure 27 shows a schematic diagram of the microphone amplifier circuit

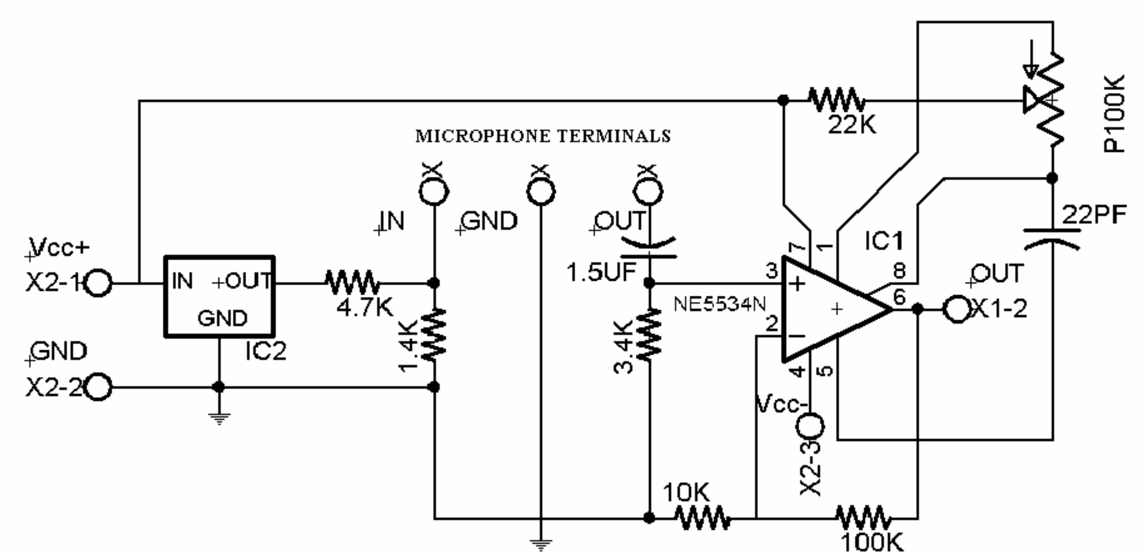


Figure 27. Amplifier Circuit.

- **Recorded Signal.** The amplified received signal was recorded using a data acquisition board (National Instruments CCA, PCI-6052E) and LabVIEW. To capture this signal we used two channels on the data acquisition board at a sampling rate of

40 kS/s. The data was recorded as a text file (.txt) in the software LabVIEW using a "virtual instrument", or VI, and analyzed in Matlab program.

- **Identify and analyze the acquired sound wave.** Each captured signal (both incident and reflection pulses) were separated using the Matlab program to read the text file and then to analyze their frequency content.

3.1.1 Acoustic pulse generation

When an acoustic pulse travels down a tube, reflections will arise but attenuation, dispersion and overlap will also occur. The only way to isolate and compare the desired incident pulse and reflections is if the incident sound wave has a known form easy to recognize. Additionally, it is required that the acoustic pulse has a specific form that will allow it to be compared with the acoustic system response and eventually with the acoustic response of the simulated model.

3.1.1.1 Inverse filtering technique.

Due to the band-pass frequency characteristics of the system (i.e., the amplifiers, transducers and tube), specifically those of the speaker, when an electrical pulse is applied to the system it does not respond as an acoustic pulse. With this technique, the impulse response of the system $H(\omega)$ is estimated in order to determine the electrical input sequence $x_f\ [n]$ required to produce a desired acoustic pulse $y_f\ [n]$. The block diagram in Figure 28 explains the inverse filtering procedure used to create the acoustic pulse.

An electric pulse $x[n]$ was generated (using a *Hanning* window in *Matlab*), using a sampling rate of 40 KS/s. The pulse was sent to the speaker using the data acquisition board and *LabVIEW*, sensed by microphones 1 and 2 and recorded in a text file with a Virtual Instrument (*LabView*) as shown in figure 28a. Using the input signal $x[n]$, and the output signal $y[n]$, the system's frequency response $H(f)$ was determined using:

$$H(f) = \frac{Y(f)X(f)^*}{X(f)X(f)^*} \tag{26}$$

where $X(f)$ is the Fourier transform of a very short input pulse and $Y(f)$ is the Fourier transform of the system's output

The electrical input sequence $x_f\,[n]$ required to obtain the acoustic pulse $y_f\,[n]$ at the output of $H(\omega)$, as shown in figure 28b, is obtained using:

$$x_f[n] = FFT^{-1}\left[\frac{Y_f(\omega)}{H(\omega)+k}W(\omega)\right] \tag{27}$$

where k is a small constraining factor to prevent division by zero or by small numbers which would result in high magnitude frequency components. $W(\omega)$ is a digital band-pass filter (with cut-off frequency set by trial and error according to the obtained response) used to reject frequency components that are not present in $H(\omega)$.

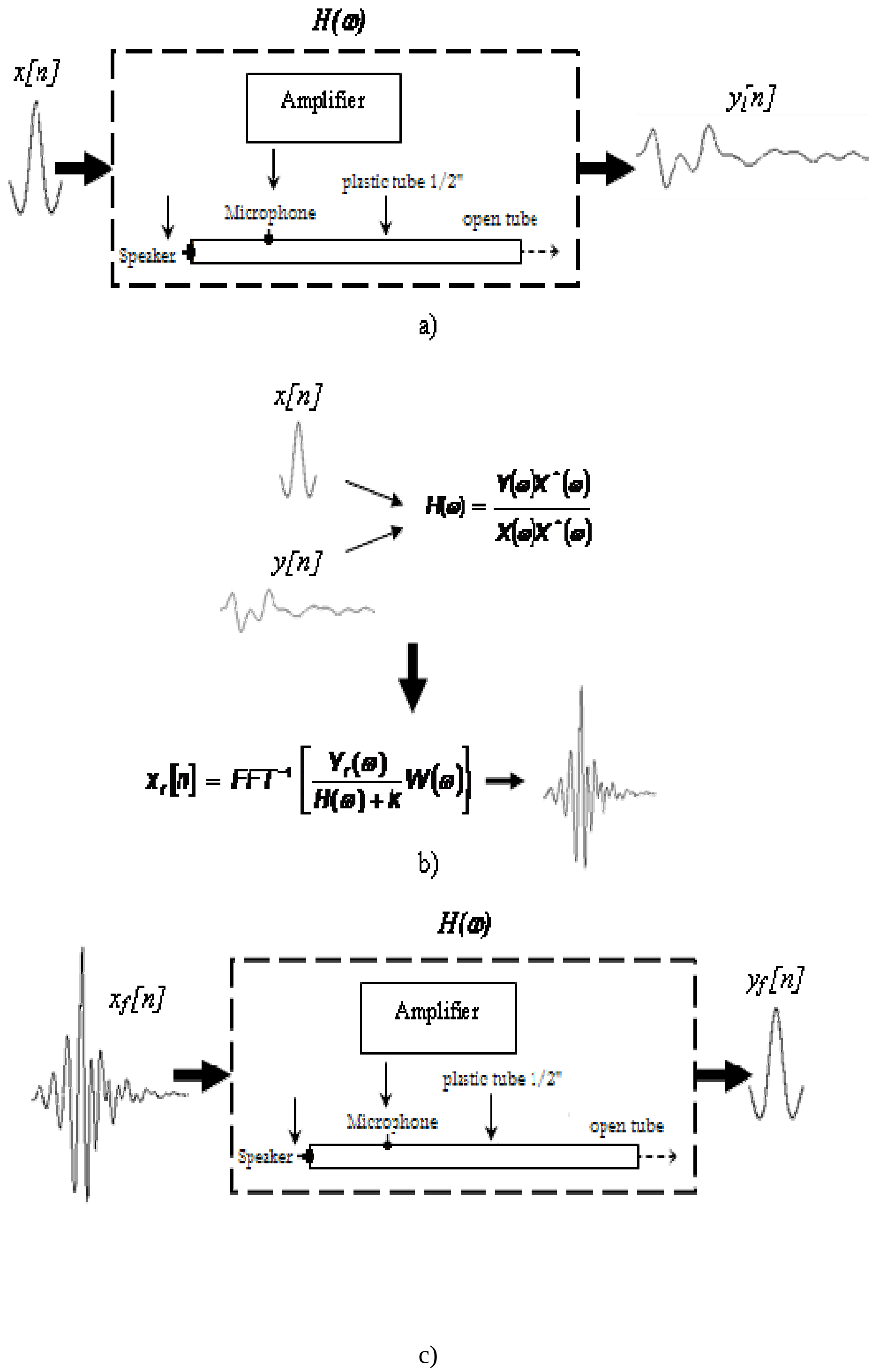


Figure 28. Inverse filtering procedure, a) Record the system response to the electric pulse, b)Determine the system frequency response and the input sequence x_f *[n]* required to create the acoustic pulse, c) Acoustic pulse resulting as a system response to the input sequence found by inverse filtering.

3.1.1.2 Hanning pulse generation.

Using the Hanning window function in the Matlab software, we generated the Hanning pulse using the "hann(x)" command, where x is the total number of samples that comprise the pulse. Depending on the necessary energy and the frequency range which the system operates, we selected the adequate durations for the Hanning pulse. For our system, we needed enough energy for the pulse travel the whole length of the system (155 cm) before being completely attenuated and of adequate frequency (0-12KHz) to test the system. In figures 29 and 30 the Hanning pulse ($x\,[n]$) provided to the system, and the resulting electric pulse ($y\,[n]$).

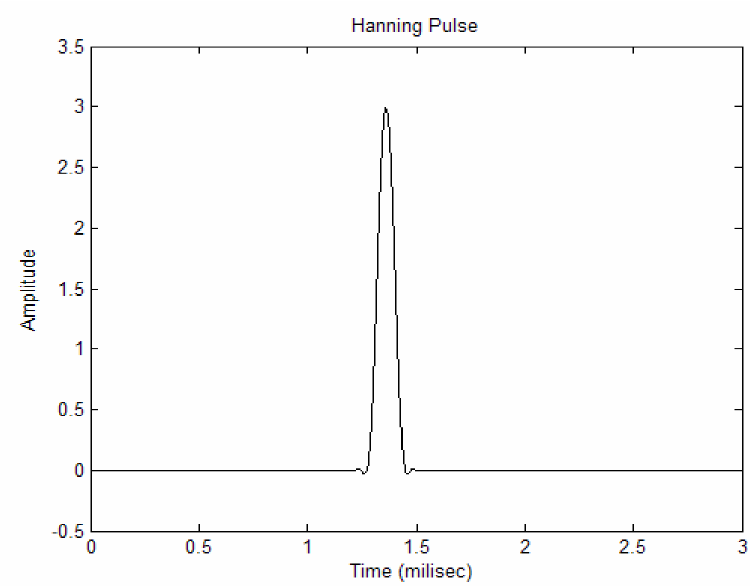


Figure 29. Hanning pulse.

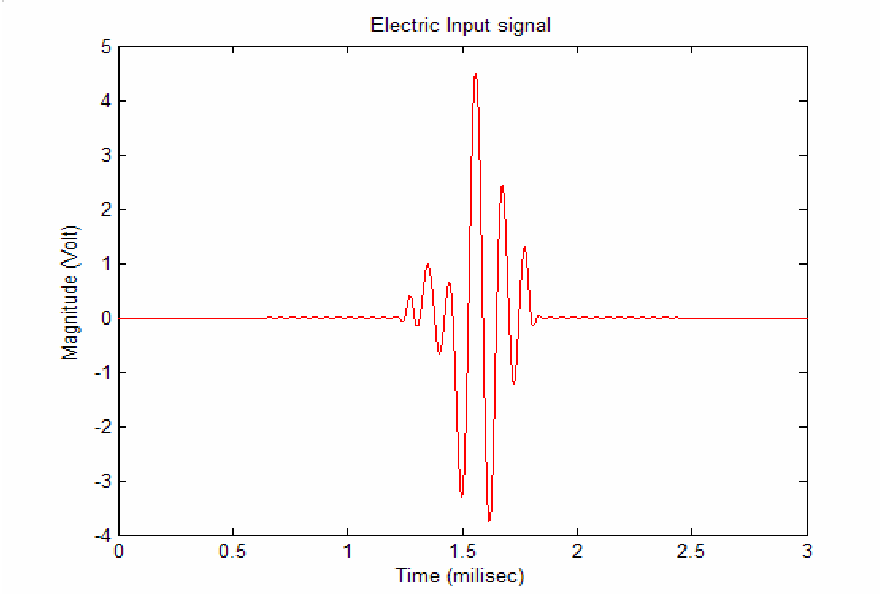


Figure 30. Electric input signal.

3.1.1.3 System performance and analysis to obtain an electric pulse.

In order to find the electric pulse, a Hanning pulse was generated and sent to the system so the transfer function (*H(f)*) of the system could be obtained. After identifying *H(f)*, and with the desired output signal, the input electric signal for our system was obtained by deconvolution. This process was made to obtain the desired acoustic pulse so we could easily identify both incident and reflection pulses (see figure 28c). In the previous section 3.1.1.1 the different mathematical equations (26 and 27) were observed to find the input signal that derived in the desired sound pulse.

Proper system performance is based on the adequate selection of the pulse parameter that will be used. In other words, the energy should be enough so that the pulse could be detected by microphone 2 (setup 2) and the frequency range of the pulse should be equal or greater than the one emitted, transmitted or detected by the system (speaker, tube, microphones). In our case, the desired ranges of amplitude and frequency were mentioned on section 3.1.1.2.

3.1.2 Electric pulse transmission and sound reflection results for two types of setups.

Whenever an acoustic pulse travels along a tube, reflections will occur but attenuation, dispersion and overlapping will also take place. The simplest approach to isolate and evaluate the desired incident pulse and reflections separately is if the incident sound wave has a known form easy to recognize in the received signal. Also, it is required that the length of the system tube and the distance that separates the speaker and the microphones be enough so that there is no overlap between the incident and reflected acoustic pulses. In our case, the most critical situation is on the second setup where we have two microphones, separated 91 cm one from the other, and with a membrane between both which increases the possibility of overlapping on the first microphone. The next section explains the signal analysis methods used for each of experimental setups.

3.1.2.1. One-microphone setup and system analysis.

When the acoustic response behavior of setup No. 1 was analyzed, the entire acoustic signal (incident and reflection pulses, see figure 31) was recorded. Next, the incident pulse was separated from the reflection pulses using a computer program developed in Matlab 7.0. This process was made for two systems, with and without a membrane outside the speaker. Afterwards, the frequency response was analyzed for the incident pulses in both cases and then compared one with respect to the other.

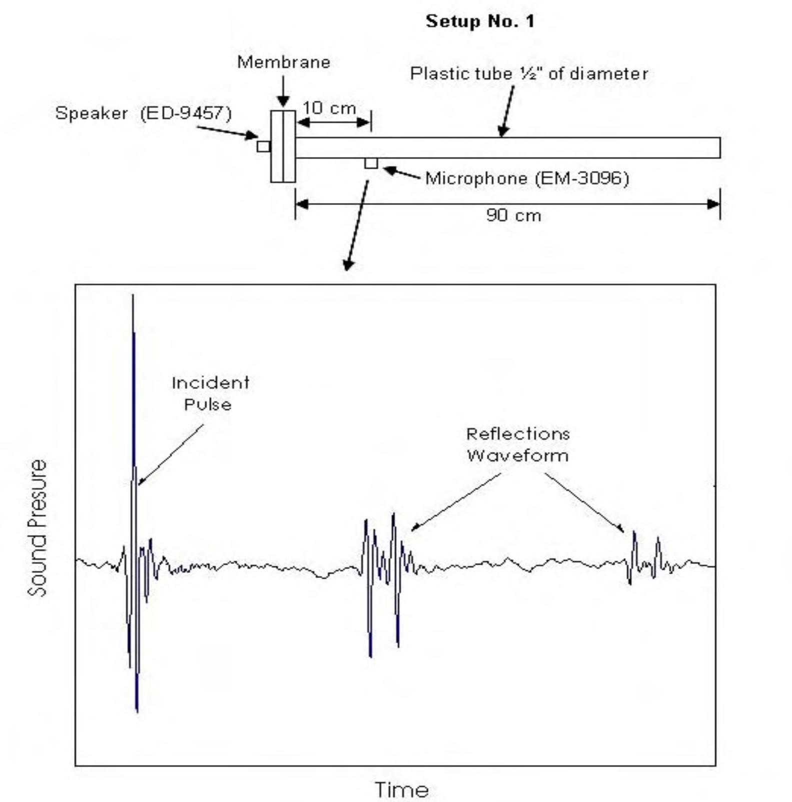


Figure 31. Signal received by the microphone in setup No.1 .

Initially the system's frequency response was determined using

$$H(f) = \frac{Y(f)X(f)^*}{X(f)X(f)^*} \tag{26}$$

$Y(f)$ and $X(f)$ are the Fourier transforms of the recorded acoustic signal and the input electrical signal, respectively. The transfer functions of the system without membrane $H_{nw}(f)$ and with membrane $H_{wm}(f)$ were estimated. The effective transfer function of the membrane $H_m(f)$ could be found using and compared for each type of membrane with the transfer function without membrane.

$$H_m(f) = \frac{H_{nm}(f)}{H_{wm}(f)} \tag{28}$$

3.1.2.2. Two-microphone setup, system analysis and calibration.

As we did in setup No. 1, the complete sound wave (incident and reflection pulses, see Figure 32) was recorded by microphones 1 and 2, and saved in the computer using the LabVIEW software. Then, the incident pulses for both microphones (1 and 2) were separated from the reflections pulses. This previous step was made using the Matlab software with which the Fourier transform for the different types of membranes were calculated. In this case, the membrane location was as shown in Figure 32; it was situated behind microphone one and in front of microphone two. The system analysis was made taking the signal from the first microphone as the input to the system and the one of the second microphone as the output and then the transfer function was calculated. The transfer functions of the system without membrane $H_{nw}(f)$ and with membrane $H_{wm}(f)$ were estimated and the effective transfer function of the membrane $H_m(f)$ was found using equation (28).

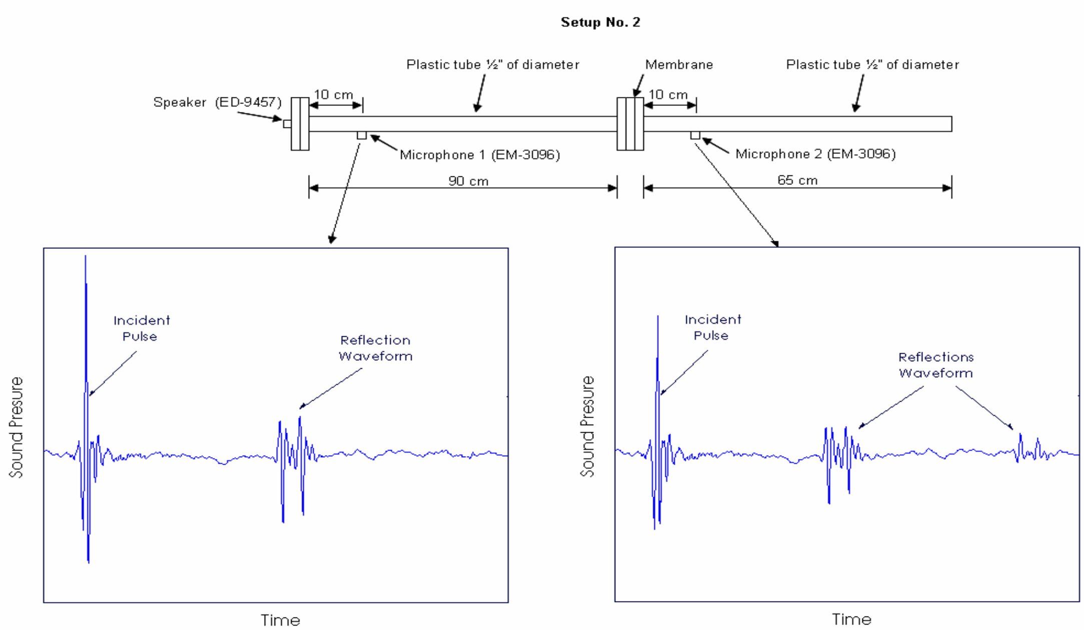


Figure 32. Signal received by microphones 1 and 2, in setup No.2.

Before making the analysis, the system was calibrated taking (microphone No.1) as reference. Initially, both microphones were situated equidistantly to the speaker as shown in the Figure 33. The calibration was made using the incident pulses of both microphones and obtaining a factor $X_2(f)/X_1(f)$ (the rate in frequency between the incident pulses), which multiplies the signal captured with the two microphones in setup No. 2.

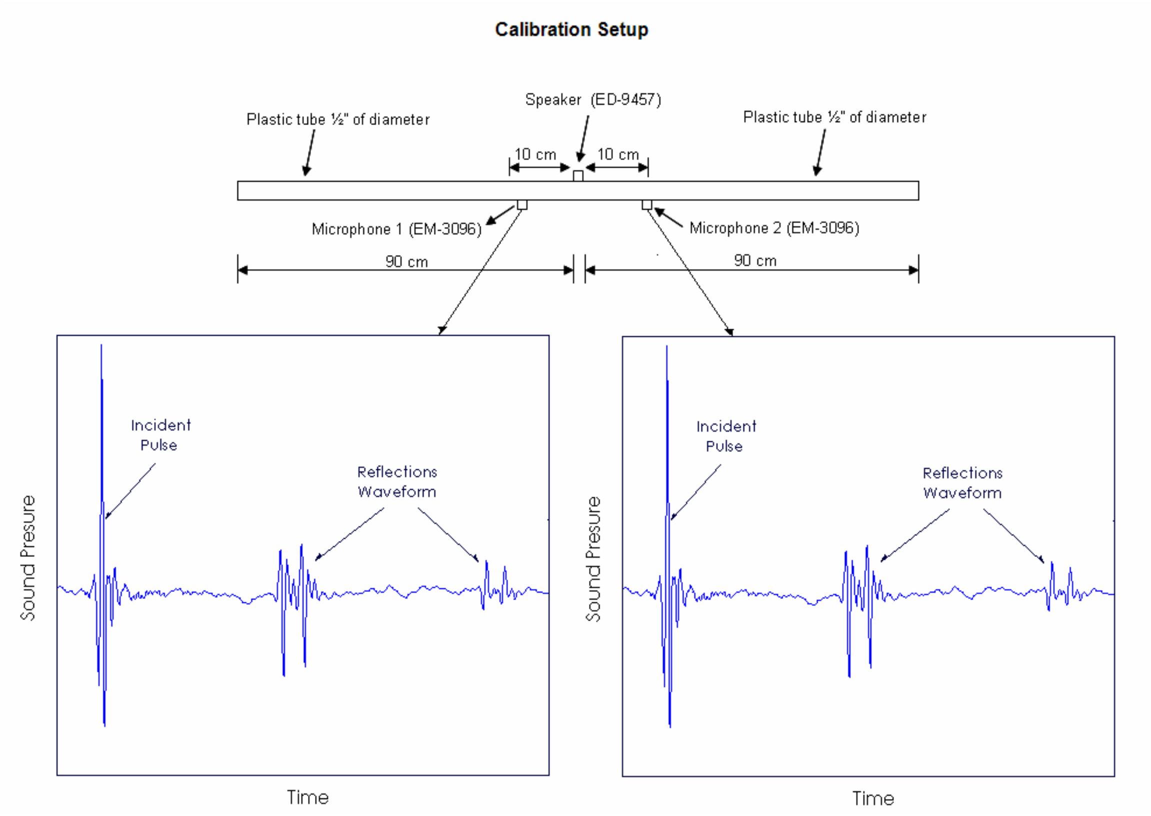


Figure 33. Setup to Calibration the microphones 1 and 2.

3.2. Computer simulations and parameter selection for sound propagation in a plastic tube with ½" diameter.

To characterize the sound propagation in a plastic pipe system (crossing a membrane) used in this research, we consider the effects of transmission and reflection, which occur when the pressure wave travels inside the tube system in presence of a membrane and these effects were modeled with the transmission coefficient filter.

3.2.1. Computer analysis for one and two-microphone setups using the coefficient transmission.

While analyzing the behavior of a system through which a sound wave is traveling, such as a uniform plastic tube, we have to take in consideration whether or not the sound wave senses a change in medium (such as a change in density or in dimensions). In other words, such a change results in a variation of the acoustic impedance of the system. As consequence, these changes cause that signals are transmitted and others that are reflected on the interior of the tube. To measure this signals, we use what is called transmission and reflection coefficients which indicate the approximate percentage of a signal that gets through (transmission coefficient) and the amount that is reflected (reflection coefficient). In this particular thesis research, we used the transmission coefficient to simulate the attenuation of an acoustic pulse when it encounters an impermeable membrane. In order to get the correct value, we took in consideration the characteristics and dimensions of the membrane.

To make the simulations, we parted from equations (24) and (25) (described previously) by which we obtain the expression for the transmission coefficient. It is a vector (of frequency) that we processed to obtain a filter which multiplies the acoustic pulse that is sent to the system.

$$R(x) = \frac{(1 - Z_1 / Z_3)\cos k_2 l + j(Z_2 / Z_3 - Z_1 / Z_2)\sin k_2 l}{(1 + Z_1 / Z_3)\cos k_2 l + j(Z_2 / Z_3 + Z_1 / Z_2)\sin k_2 l} \tag{24}$$

$$T_{x_0} = \frac{2}{(1 + Z_1 / Z_3)\cos k_2 h + j(Z_2 / Z_3 + Z_1 / Z_2)\sin k_2 h} \tag{25}$$

To find the resulting vector of the transmission coefficient equation, first we cut the vector in half, we find its conjugate and flip it. We then generate another vector, constituted on its first part by the initial vector cut in half plus the conjugated vector inverted on the end; its central part filled with zeros. Afterwards, we multiply each component by the vector of the acoustic pulse to obtain the resultant vector or filtered vector. It should be noted that this process is done at the frequency domain and includes the parameters of the three specific mediums (air-plastic-air).

During our simulations, we did not take in consideration the effects of attenuation and energy loss when a pulse travels through a uniform system on a plastic tube. The reason for that was that we were comparing the effects of the membrane at the same point of the setups and the pulse used for the simulations was the same one captured at that same point but on a system without a membrane. The filter resulting from the transmission coefficient equation simulated the behavior of the membrane.

In summary, in this chapter we described the experimental setups, the procedures and the simulations that were made in this investigation. Basically, we used two types of experimental setups and one type of theoretical simulations to demonstrate the behavior of the sound pulse traveling through an uniform system and its behavior when it suffers a change in impedances; specifically, when an impermeable membrane prevents its normal propagation. Based on the resulting equations, we developed a methodology for designing impermeable membranes so that they can be placed at the opening of the hearing aid to avoid the entrance of dust and accumulation of earwax but without affecting its frequency response.

Chapter 4

Results and Discussion

The purpose of this chapter is to show, discuss and compare the results from the computer simulations with the results from the experiments performed during this research, in order to establish how well the model would predict a response to a specific acoustic pulse and determine the conditions where the model would be valid for use in hearing aids or other applications. The chapter is divided in four sections. The first section shows and analyzes the experimental results. The second section presents and discusses the results obtained from computer simulations (based on the transmission coefficient) used to model and predict the sound propagation in plastic tubes (0.5 in diameter). The third section establishes how well the experimental results from the first part fulfill the expected results from the second part. The last section of the chapter shows the analysis of predicted results for the use of transmission coefficient in different membranes and different setups, establishing specific parameters where this method could be used.

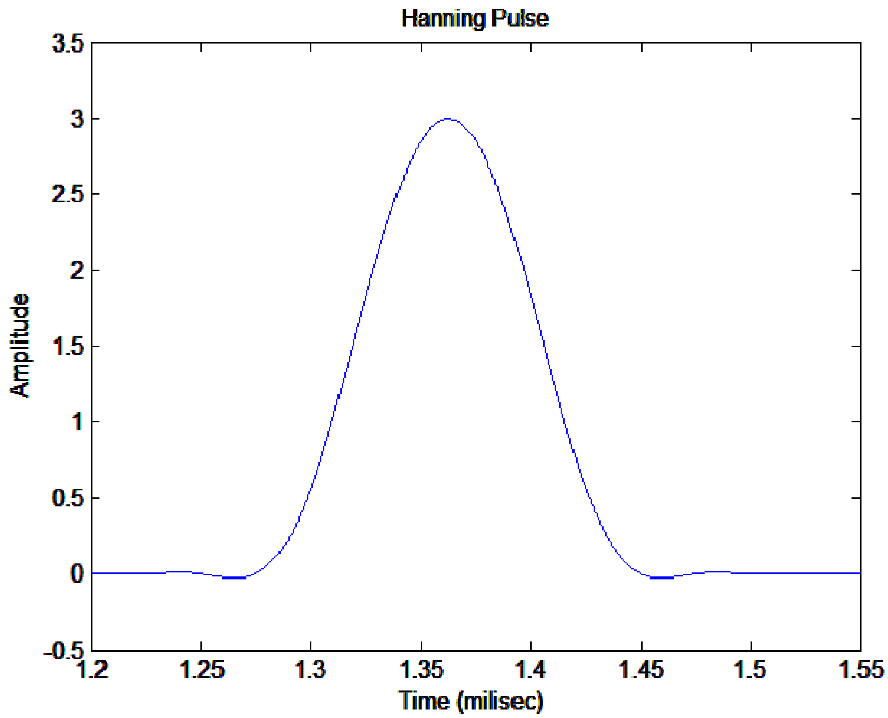
4.1 Experimental Results

This section presents the results of the experiments performed to understand how sound propagates through the different membranes and into the 0.5 inches diameter plastic tube. The first part shows the results of using the inverse filtering technique explained in sections 3.1.1.1 and 3.1.1.2. The second part illustrates the system behavior in the frequency domain. The third section explains the system behavior in presence of the three different membranes using the first setup (one microphone). The last section explains the system behavior under the existence of the different membranes using the second setup (two microphones).

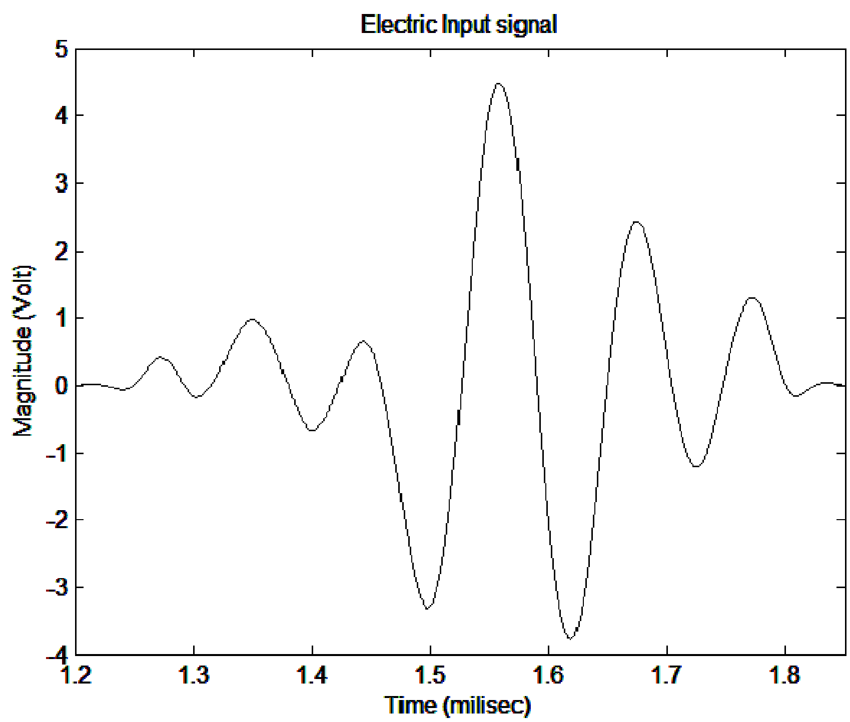
4.1.1. Acoustic pulse created using the inverse filtering technique.

Using a Hanning pulse, we could find the electric pulse to create the acoustic pulse, used to differentiate the incident from the reflection pulses. As shown in Figure 34, the Hanning pulse was generated with a length of 0.25 milliseconds and a frequency content between 0 to 11.5 kHz in order to work with the limitations of the speaker-microphone system. Once we have found the frequency response of the acoustic and the electric pulse we can determine the behavior of the system.

Notice, as shown in Figure 35, the spectrum of frequency of the electric pulse which was influenced by the limitations of the speaker response at low and high frequencies. In the next section we will see the speaker frequency response using the electric pulse as input and the acoustic pulse as output. As it could be seen, the obtained electrical pulse depends basically on the response of the system but even more on the speaker. The frequency response of the speaker was between 1.5–10 KHz approximately.



a)



b)

Figure 34. Input pulses used to determine the acoustic pulse. a) Hanning pulse, b) Electric pulse.

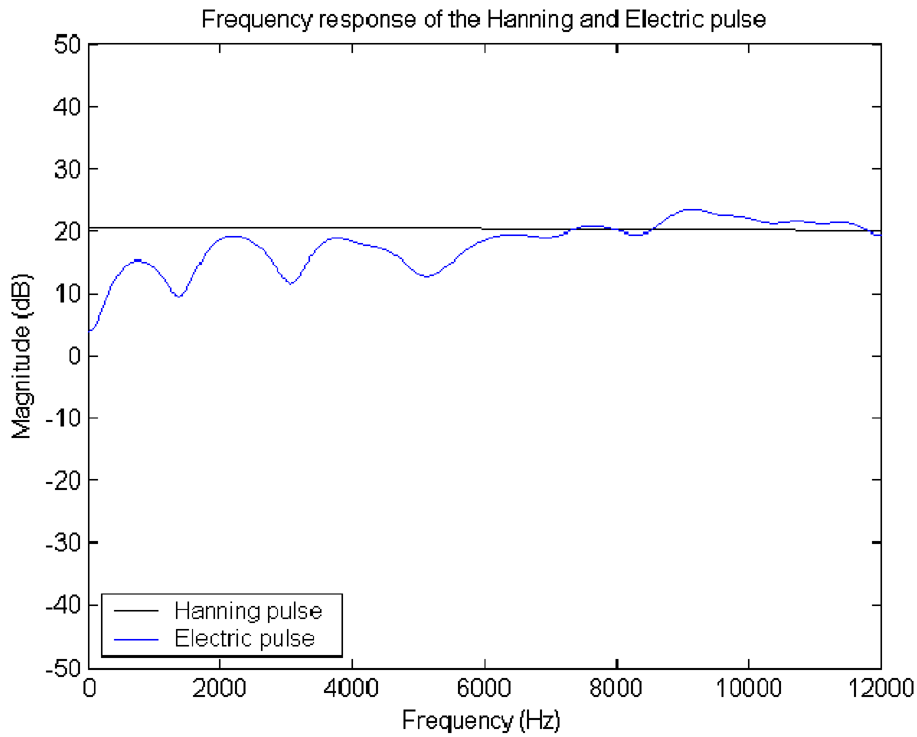
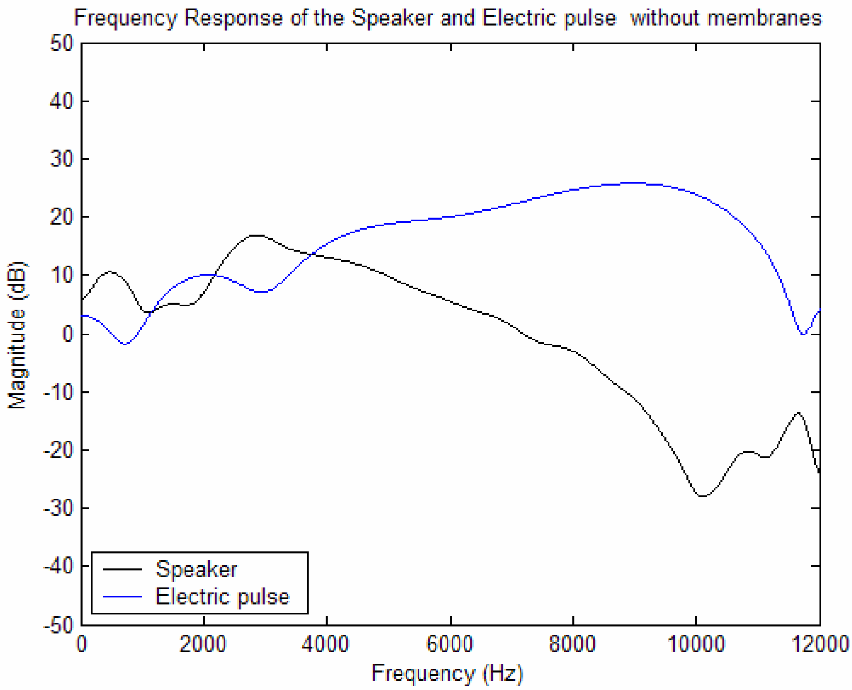


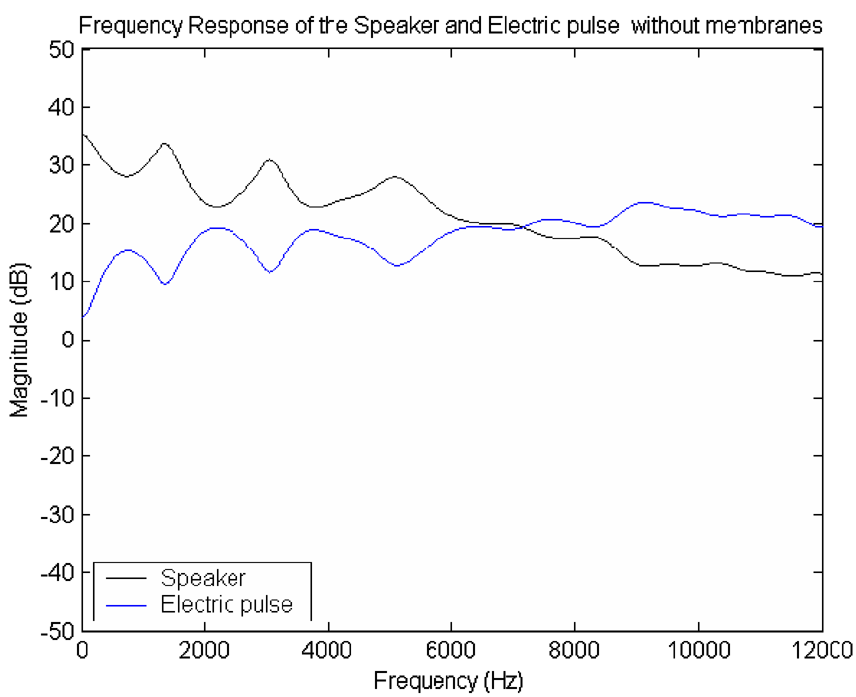
Figure 35 Content in frequency of the Hanning and the electric pulse.

4.1.2 The behavior of the system in the frequency domain.

In this section, we will explain the frequency response of the system constituted by the speaker-plastic tube-membrane-microphone(s). Initially, the speaker behavior was tested applying the electric pulse obtained according to the procedure described in sections 3.1.1.1, 3.1.1.2 and 3.1.1.3. When the electric pulse was sent, an acoustic pulse was sensed by the microphone to obtain the frequency behavior of the speakers (see figure 37) and then compared with its performance specifications as show in figure 36. We can see that the speaker attenuates, causes energy loss and limits the frequency range of the electric pulse. Due to this, the behavior in frequency of the entire system depends on the performance specifications of the speaker which means that the speaker works as a band pass filter.



a)



b)

Figure 36. Frequency Content in the Electrical Pulse and the Speaker. a) Setup 1, b) Setup 2.

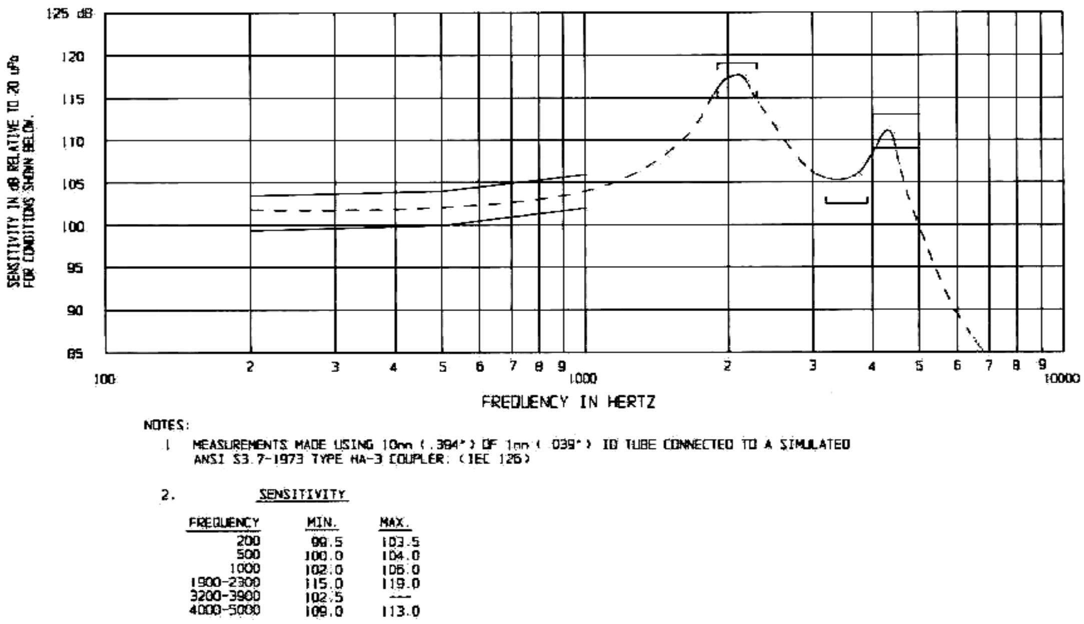


Figure 37. Speaker Frequency response, Ref XL – 9457. Setup 1.

On the other hand, the frequency response of the microphones can be seen in figure 38 and it practically does not have any type of restriction for the range of frequencies to which this speaker was working, and the system was controlled by the frequency response of the speaker. In this case, the range used is enough to develop the investigation.

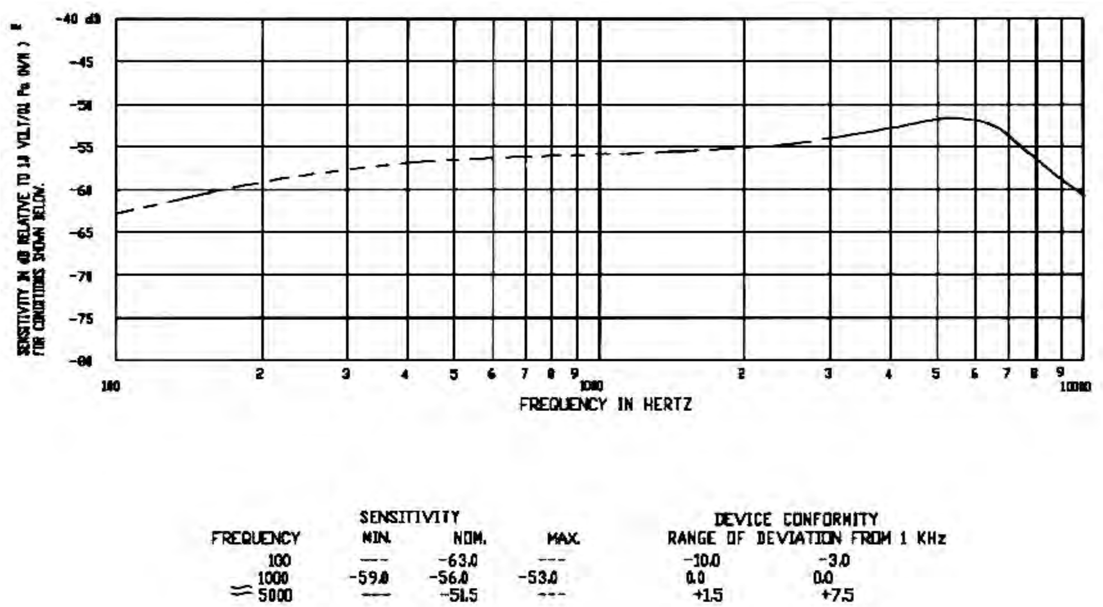


Figure 38. Microphone Frequency response, Ref EM – 3046.

4.1.3. The system behavior in the frequency domain using different membranes.

Knowing the behavior of the speaker, we will now compare its frequency response in the presence of three types of membranes. First, we will show the different acoustic pulses found when the system is in the presence or absence of membranes. Figure 39 shows the incident and reflected pulses (sensed by the microphone), Figure 40a shows the incident pulse previously separated when no membrane is used, and Figure 40b shows the same incident pulse but in the presence of different types of membranes (latex, thick latex and polypropylene). The different thickness and densities of the different materials of the membranes are shown in table 1.

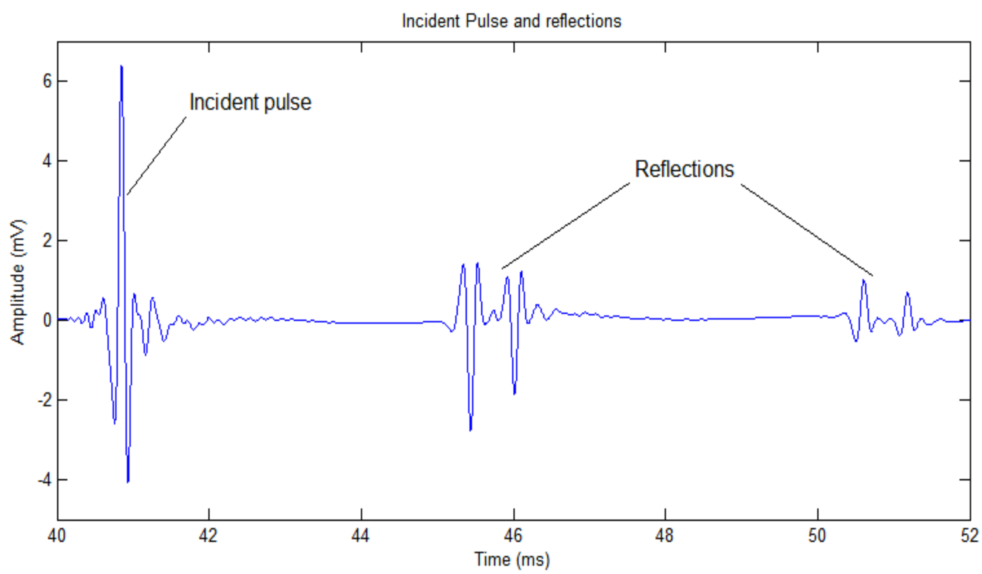
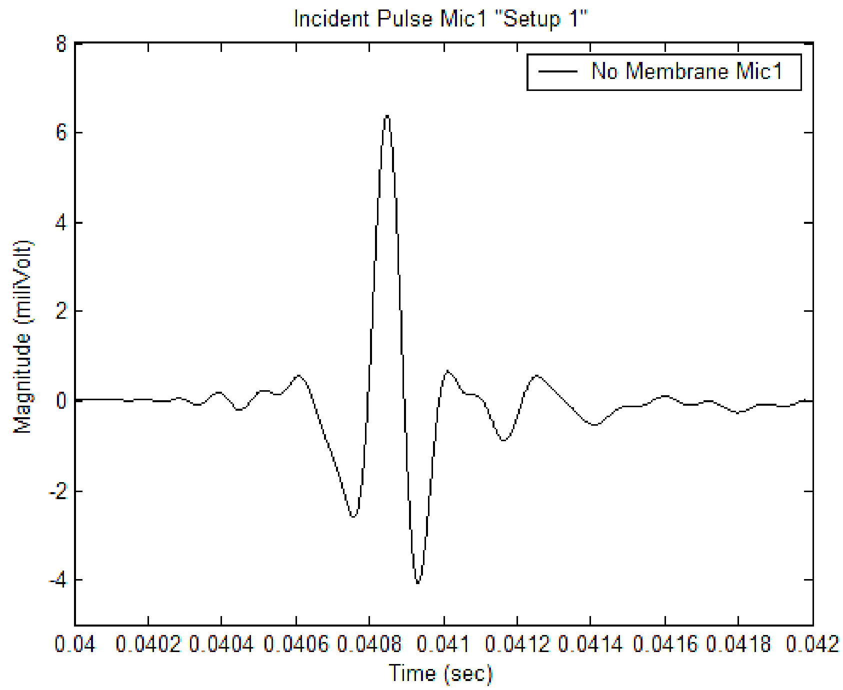
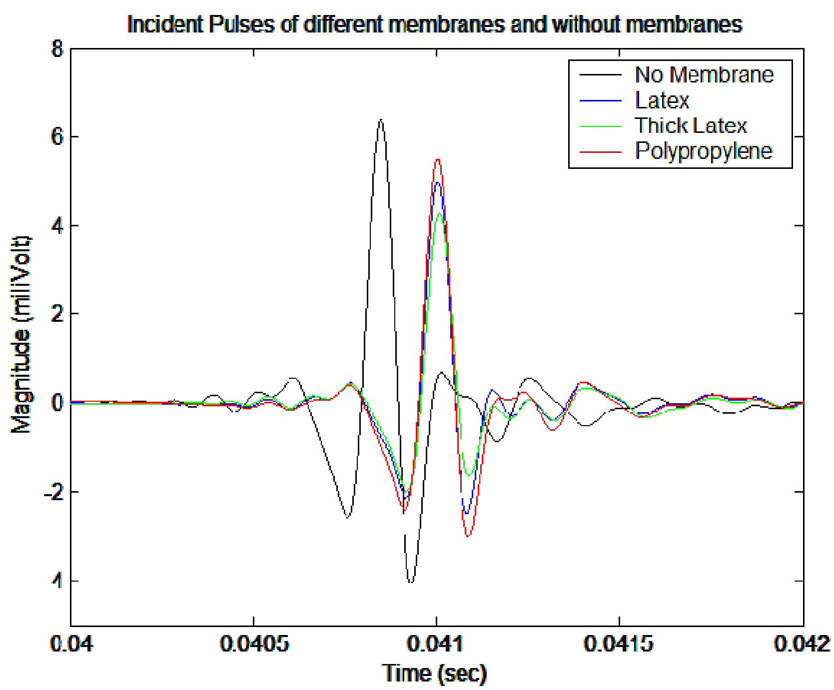


Figure 39. Incident and reflections pulses obtained in the first setup.



a)



b)

Figure 40. Incident pulses to differents membranes and without membrane for the setup No. 1, a) Incident pulse without membrane, b) Incident pulses with and without membranes.

	Material	Thickness (mm)	Density (Kg/m^3)	Sound velocity (m/s)
1	Air	0	1.204	343
2	Latex	0.08	950	1050
3	Thick Latex	0.12	950	1050
4	Polypropylene	0.01	930	2698

Table 1. Differents thickness, densities and sound velocities of the different materials and the air.

Additionally, in Figure 40b we can observe the phase shift between the incident pulse without membrane and that with membrane due to the phase effects of the system (membrane) and also the different densities, thickness and velocities of the different materials. The frequency response of the speaker for each incident pulse is similar enough, unless the membrane is too thick (double thick latex) and the attenuation is too great. Also, we can observe that the range of frequencies that are handle by the speaker are kept very similar but once they fall outside, it sets limitations to the system.

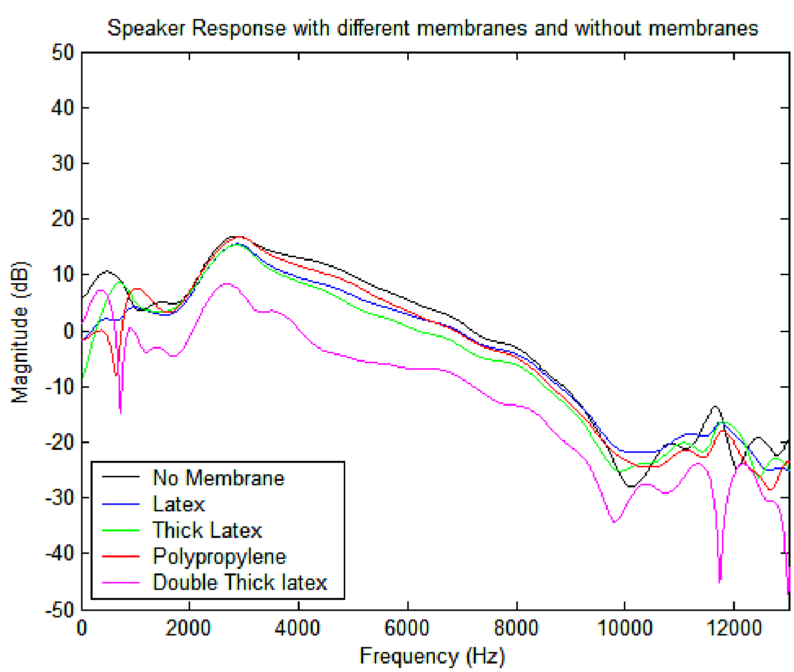
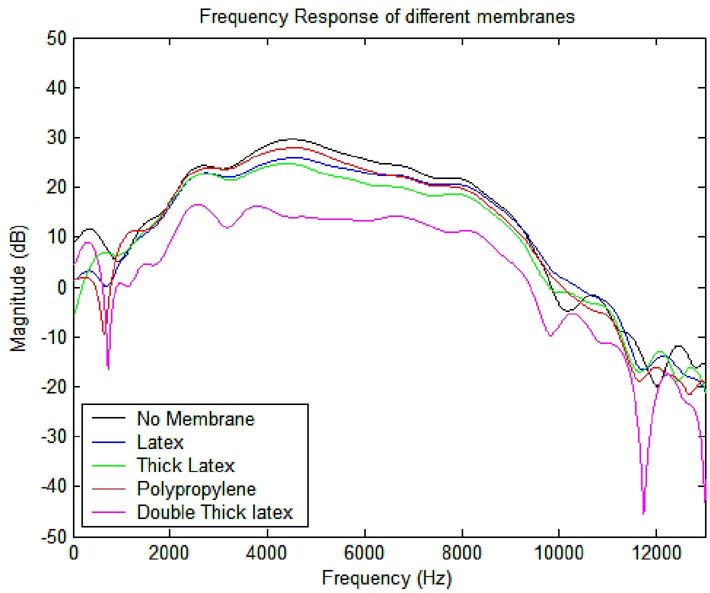


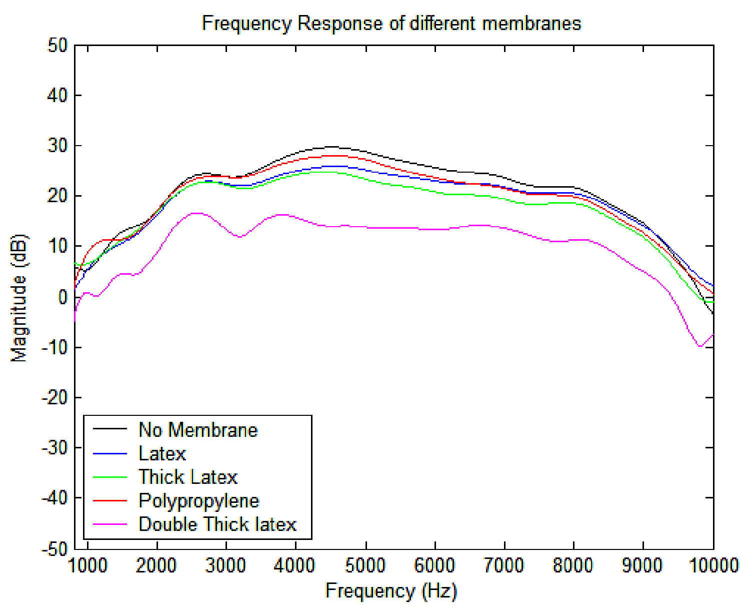
Figure 41 Speaker response with differents membranes and without membranes. Interval 0 -12500 Hz.

4.1.3.1 Experimental results for the first setup using several types of membranes.

Starting off from the previous section and knowing the effect of the speaker in the system, this subsection will present the results for the first setup using a single microphone. Figure 42 shows the frequency response of different membranes and Figure



a)



b)

Figure 42 Frequency response of differents membranes, a) interval between 0 - 12500 Hz, b) interval between 800-10000 Hz.

43 shows how the noise floor affects the frequency response of the different membranes.

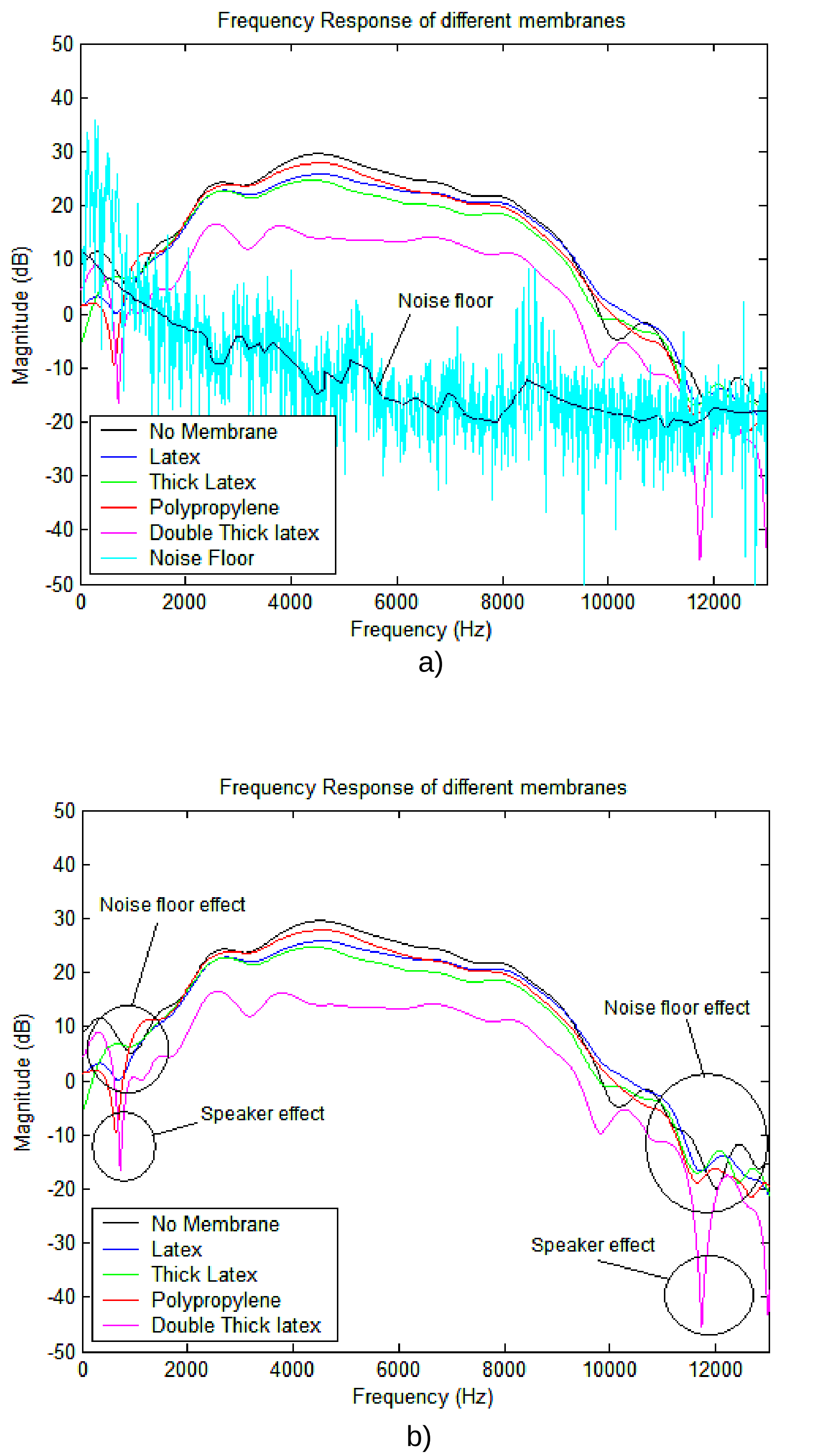
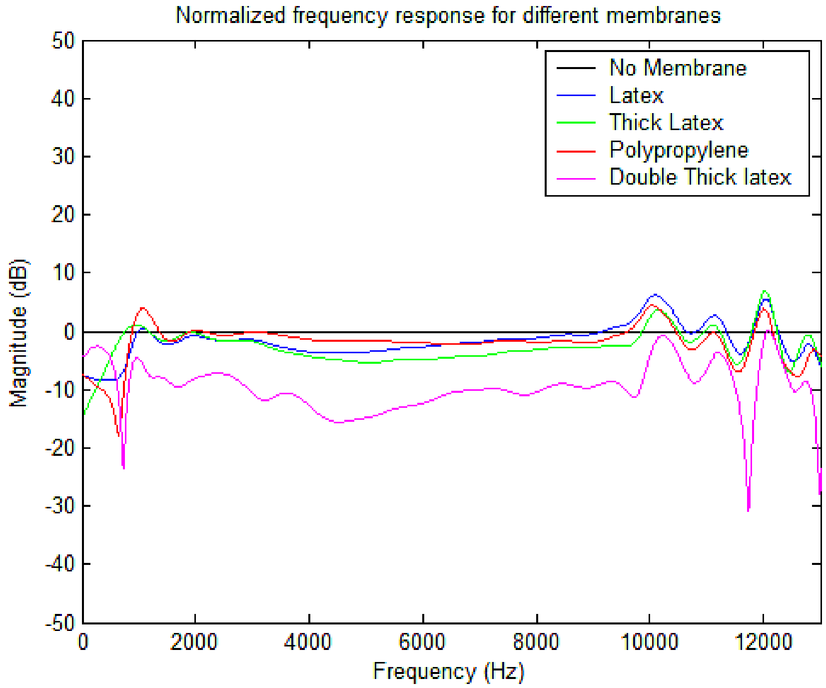


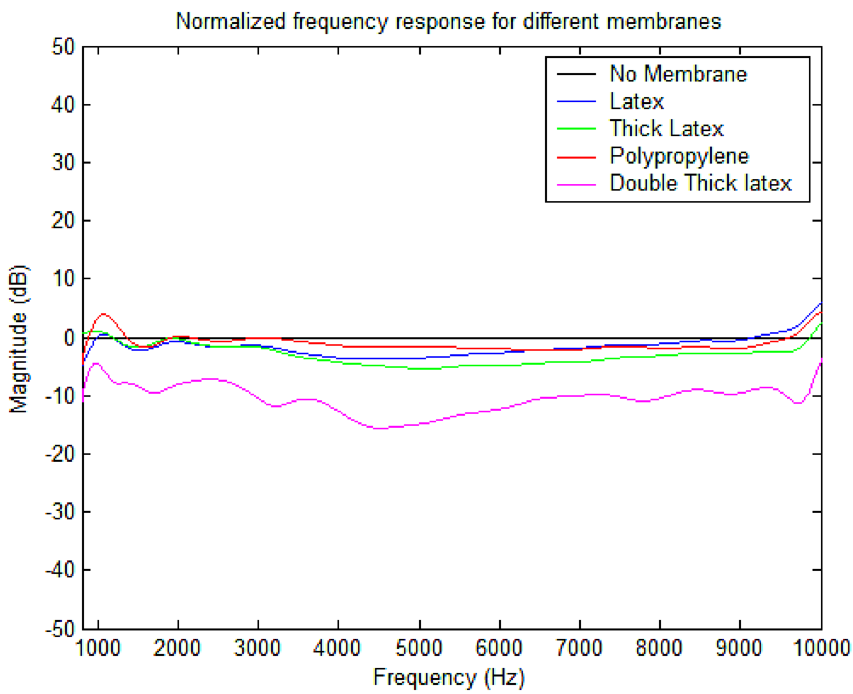
Figure 43. Effects of the noise floor and the Speaker system in setup No 1.

As it is observed in Figures 42 and 43, the frequency response of the system in the presence of different membranes is affected by the different thicknesses and densities, as well as external factors such as noise floor and the effect of the speaker in the system. In figure 41, we can observe that the effect of the different membranes is not as significant for the first three membranes (latex, polypropylene and thick latex) as it is for the double-membrane of thick latex. From these results we can conclude that the best behavior comes from membranes made of polypropylene, latex and thick latex (with thickness of 0.01 mm, 0.08 mm and 0.12 mm respectively) since the attenuation as well as the content in frequency is almost the same one as when there is no membrane. Notice on Figure 43 how the noise floor, as well as the speaker effect, affects the system response on both ends (high and low) of the frequency range of work.

Additionally, we show in Figure 44 the behavior of the different membranes with respect to the system without membranes; that is, the normalized plots respect to the incident pulse without membrane. For this graphics we can also see that the effects (frequency response and attenuation) in the system with the first three membranes are very similar to the behavior without membranes. In addition, the effective range of the response in frequency of the system was framed between 800 and 10000 Hertz using the three thinner membranes.



a)



b)

Figure 44. Normalized Frequency response for different membranes, a) Interval between 0-12500 Hz, b) Interval between 800 – 10000 Hz.

4.1.3.2. Experimental Results for the second setup (two microphones) for different membranes.

The performance and the operation of this setup were described in section 3.1.2.2 and consist of one speaker, two microphone and a membrane situated between the microphones. In this setup, we reference a microphone with respect to the other since in spite of being similar in characteristics they are not perfectly equal. This calibration was also described in the section 3.1.2.2.

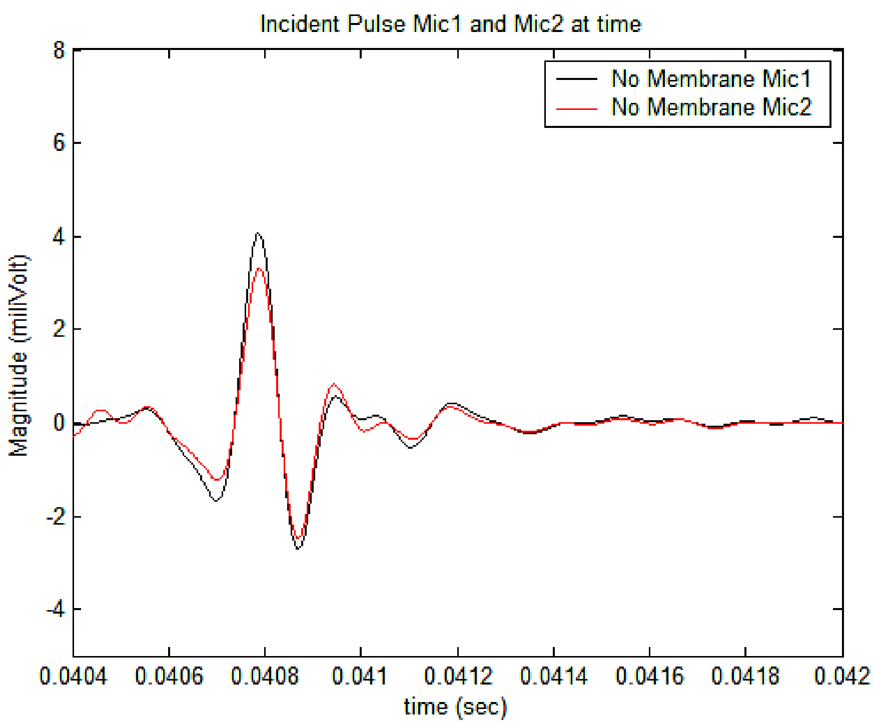


Figure 45. Incident pulses captured in the calibration setup.

In the graph shown in Figure 45 we can see the two pulses acquired simultaneously in the calibration setup with some variation in amplitude but very similar in time. On the other hand, in Figure 46 we can see the incident pulses in microphones one and two, respectively, separated 3.05 ms from each other. By this we are verifying that theoretically the sound speed in the air corresponds approximately to 343 m/s. It is observed that due to the distance between both microphones, the attenuation of the pulse in the second microphone is more accentuated and the magnitude of the pulse is reduced by approximately 2,2 mV.

In this second setup, the signals were taken from the second microphone with and without a membrane. The different thicknesses used in this setup were the same as those shown in table 1.

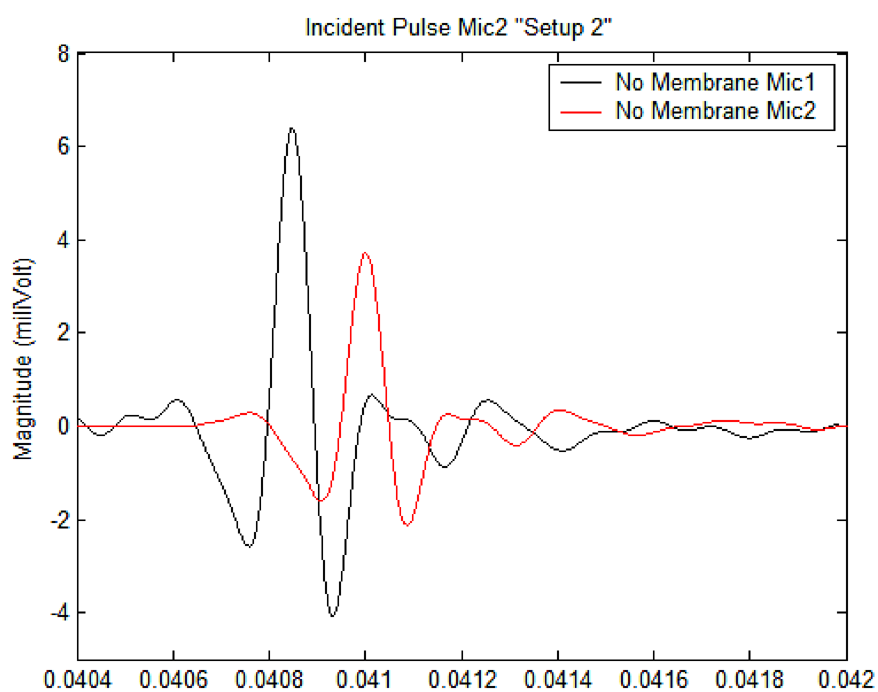


Figure 46. Incident pulses of setup two for microphones 1 and 2 respectively.

Next we show in figure 47 the different incident pulses that were obtained in the second microphone for setup two. As we can see, the pulses obtained in the presence of different membranes have a time delay compared to the pulse obtained without any membrane in spite of having the second microphone at the same position.

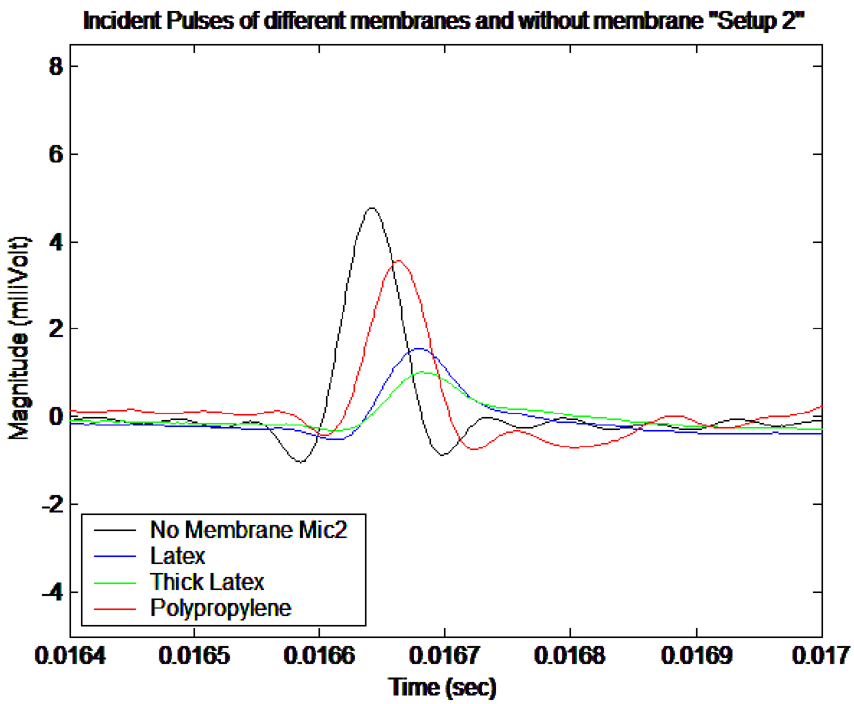


Figure 47. Incident pulses in setup two and microphone two, with no membrane and for different membranes

In figure 48 we can observe that the behavior for membranes made of plastic or latex was very similar to the behavior of the system when there was no membrane. In this setup, the membranes were put under a greater tension with respect to the one used in setup one, due to the construction of the setup on its second part. However, the different ranges of frequencies obtained using this setup were the same to those obtained in the first setup.

The effects of the noise floor and the limitations of speaker also affect the behavior of the system as well as the increase in thickness of the membranes used. The tension applied to the membrane was another factor that contributed to the behavior of the system since it affected the value of its resonance frequency. Figure 49 shows specifically the areas (low frequencies) where the noise floor affects each one of the responses of the different types of membranes. It shows how an increment in the thickness of the membrane diminishes the range of frequencies that can be obtained but always within the range allowed by the speaker.

Comparing the effectiveness of both setups, we could see that each one has its limitations but they give a good approach and agree with the expected results, even though when these results were compared with those obtained theoretically, factors such as the floor noise and the speaker response were not considered. Finally, we can affirm that as less tension is applied to the membrane and as its thickness decreases, the pulse that is obtained after it goes through the membrane is practically the same as the one that entered it and with the same content in frequency. Figure 50 shows the normalized frequency response for the different membranes at intervals between 0-12000 Hz

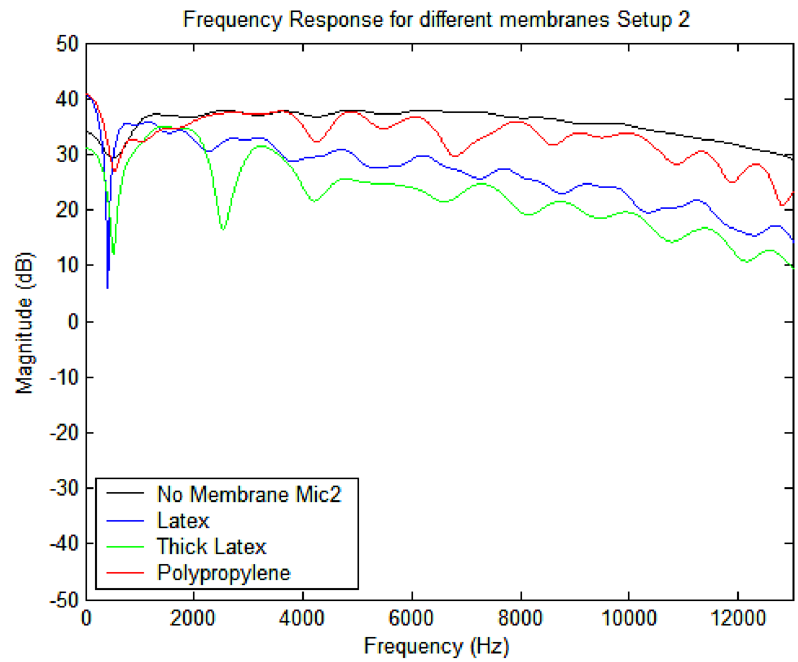


Figure 48. Frequency response for different membranes, interval 0- 13000Hz.

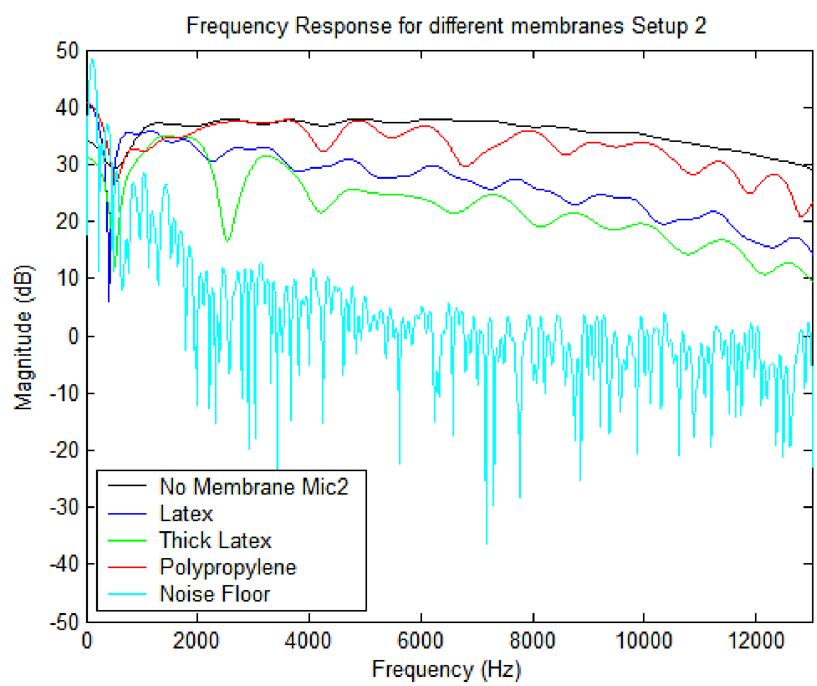


Figure 49. Effects of the noise floor and the Speaker system in setup No 2.

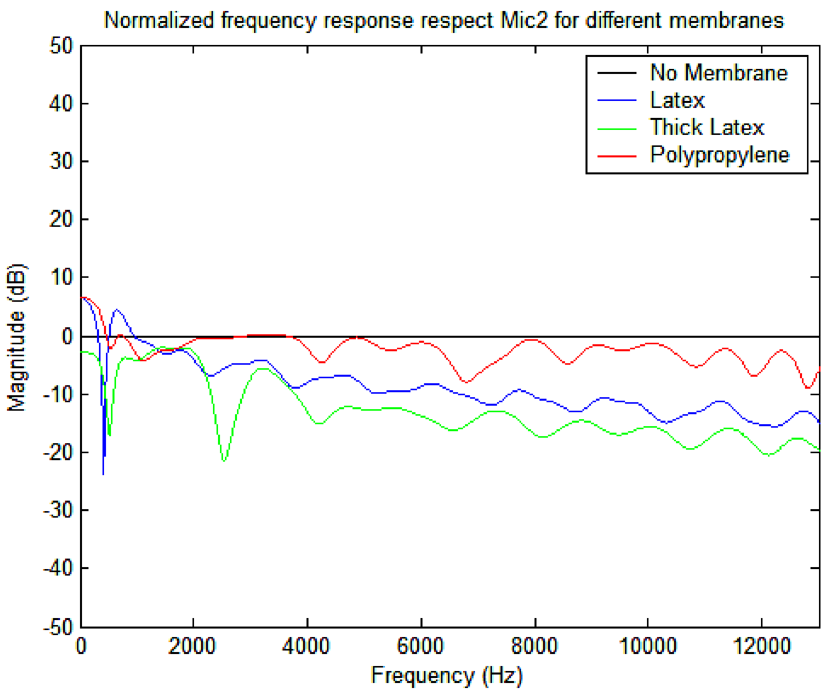


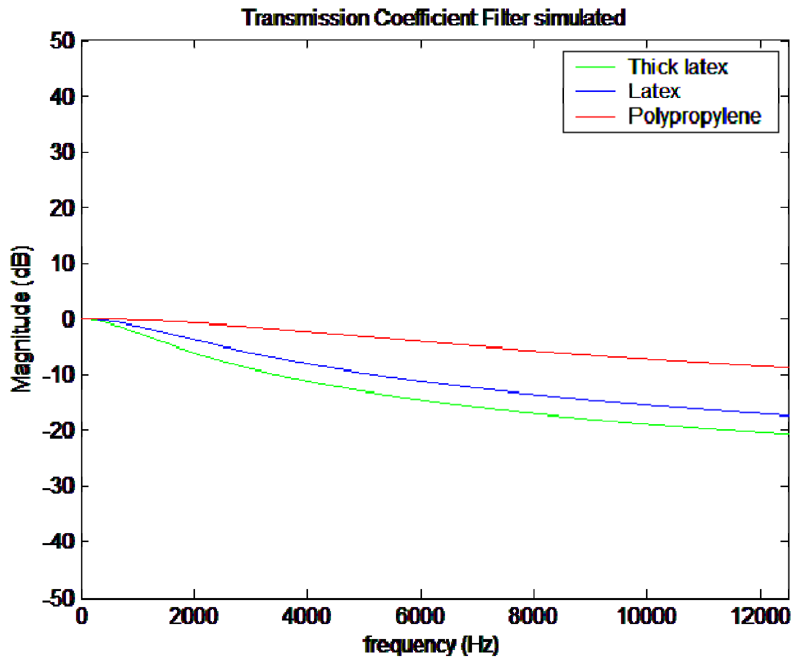
Figure 50. Normalized frequency response of the different membranes, interval 0-13000 Hz,

4.2. Computer Simulations Results.

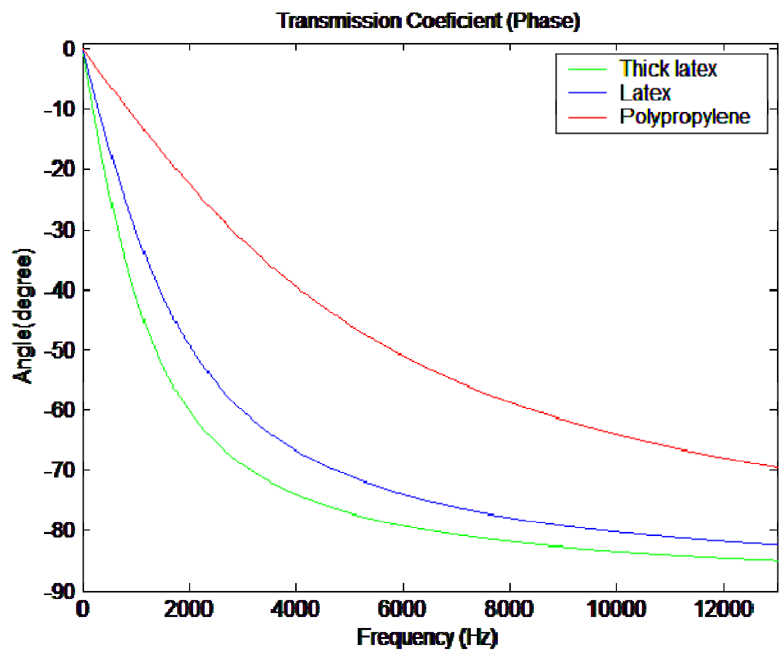
To measure this signals, we used what is called transmission and reflection coefficients which indicates the approximate percentage of a signal that gets through (transmission coefficient) and the amount that is reflected (reflection coefficient). In this particular thesis research, we used the transmission coefficient to simulate the attenuation of an acoustic pulse when it encounters an impermeable membrane. In order to get the correct value, we took into consideration the characteristics and dimensions of the membrane but this method did not quantify the effect introduced by the noise floor.

4.2.1. Transmission coefficient filter simulation results.

In order to make the simulations with the transmission coefficient, the deductions were obtained in section 3.2.1 using the equation (26) as described in Chapter 2. To be able to use these equations, the density characteristics and sound speeds for each membrane materials and the air were required as well as their thickness. Initially, the transmission coefficient filter for each membranes used in this investigation were calculated and drawn. Figure 51 shows the behavior in frequency for the different filters their magnitude (dB) and angle (degree). These filters behaved as a low pass filters and presented a shift phase as the membrane increased in thickness.



a)



b)

Figure 51. Transmission coefficient filter for different membranes, a) Magnitude dB, b) Phase

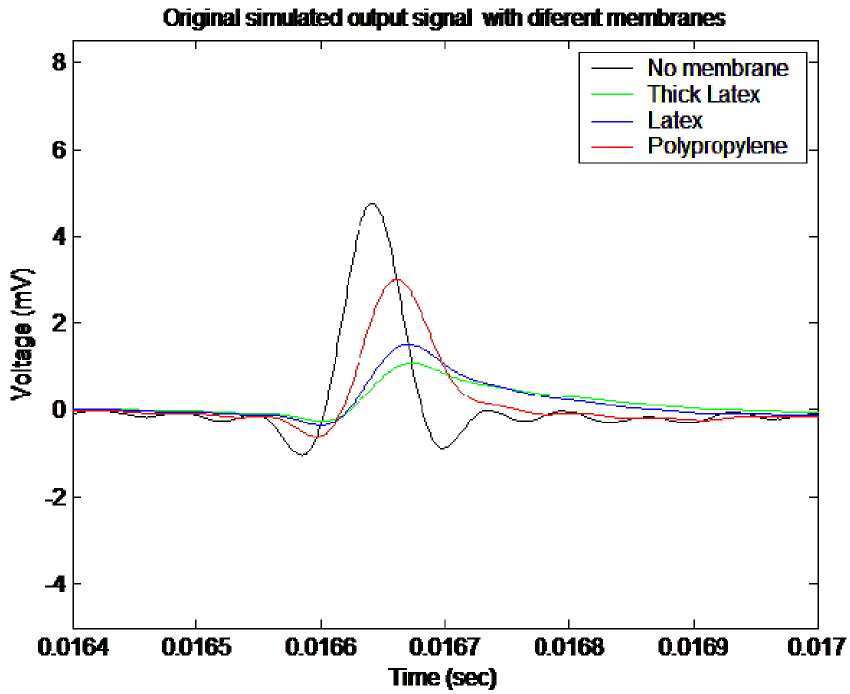


Figure 52. Incident pulse for different simulated membranes.

In Figure 52 the different pulses obtained from the simulations with and without membrane can be observed. The resulting behaviors were very similar to the ones obtained experimentally; although in these simulations the noise floor effect, another limitations of the system, was not taken in consideration. Figure 51a and 51b, show the magnitude and the phase delay of the transmission coefficient, respectively. The phase for each type of response was negative having a delay with a magnitude near to one. We can see that the phase angle of the original pulse with respect to the others experimentally also has the same delay.

The following Figure 53, is a comparison of the content in frequency among different simulated membranes. For the membranes made of latex and plastic, we obtained similar results to those obtained experimentally but with a better output, that is, very similar to the pulse obtained without a membrane.

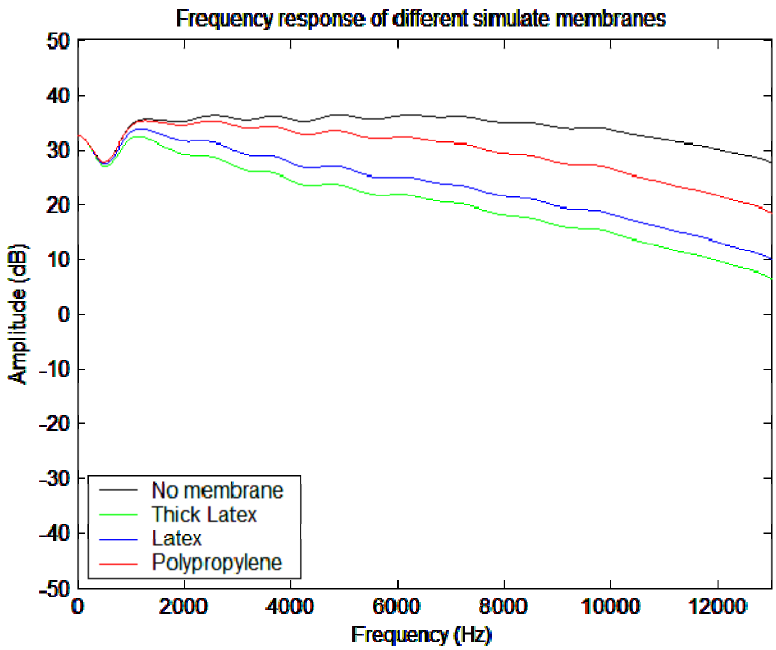


Figure 53. Frequency response for different membranes using the transmission coefficient.

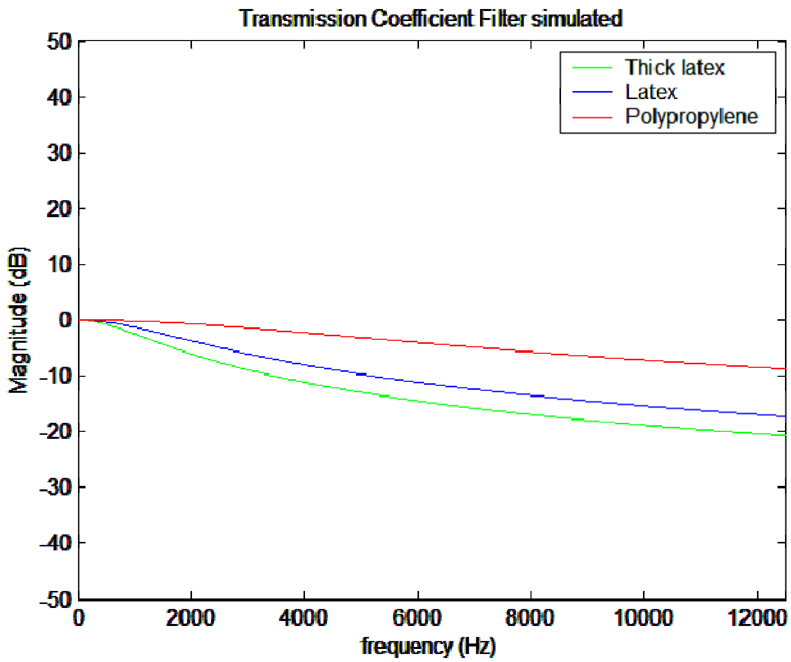
4.3. Experimental vs. Simulated results.

This section compares the results obtained experimentally with the simulated ones to determine how well the data collected experimentally correlates with the one found in the simulations. Initially, we compared the different filters obtained via simulation versus those obtained experimentally. Afterwards, we compared the frequency response of the different membranes obtained through experiments and simulations.

4.3.1. Coefficient transmission filter obtained experimentally vs. simulated.

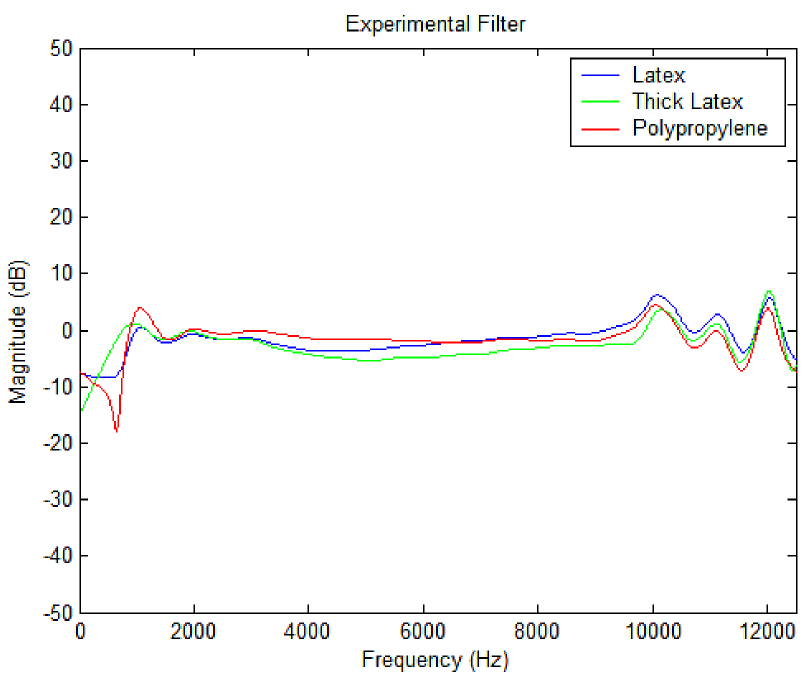
When we obtained the transmission coefficient filter for the first setup, the attenuation, the speaker performance and the noise floor of the system were not considered. In spite of it, the results obtained for the range of frequencies between 1000-10000 Hertz were reasonably acceptable. Figure 54 shows that the behavior of the different membranes was very similar between the ranges in frequencies of 1800 – 10000 Hz. As shown, the plastic membrane was the one with the best frequency response and the results were very similar to the simulated ones. The behavior of the latex and thick latex membrane were also good and their frequency response was very similar to the one found during the simulations. Over 7500 through 10000 Hertz, the experimental behavior changed slightly for the plastic membrane reducing its performance, while the response for the latex membrane improved.

On the other hand, when the factors that affect the system in the theoretical calculations were not considered, the filter error for different frequencies increased significantly for some membranes and diminished for others, not maintaining therefore a uniform trend as the frequency varied.



a)

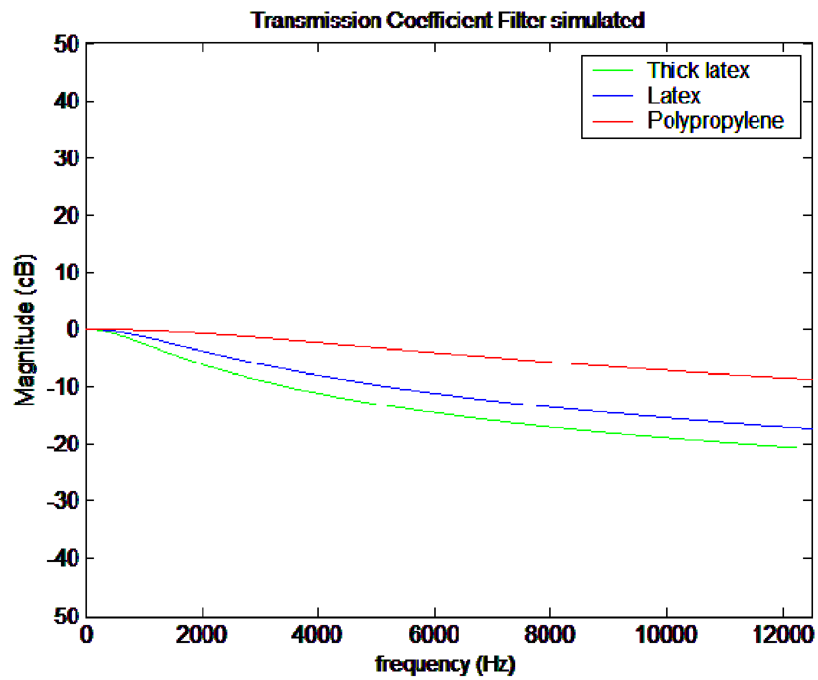
a)



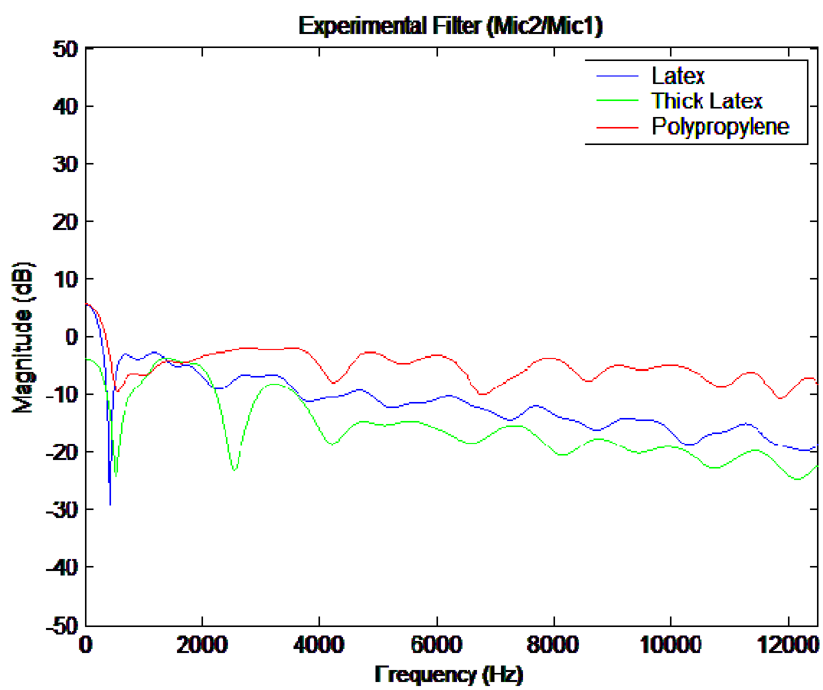
b)

Figure 54. Theoretical and experimental filter using setup No. 1 (one microphone) a) Transmission Coefficient Filter, b) Experimental Filter.

In figure 55 can be appreciated the behavior of the filter generated by the transmission coefficient and the one found experimentally in setup number two using as reference signal the one captured by the microphone one. As we can see in the figure, in this setup the behavior of different membranes goes accordingly to the one found by the filter of the transmission coefficient; that is, the membrane with the best response was the one made of polypropylene, followed by the one of Latex and finally the thick Latex. These results are better shown in Figure 56.



a)



b)

Figure 55. Theoretical and experimental filter using setup No. 2 (two microphones) a) Transmission Coefficient Filter, b) Experimental Filter.

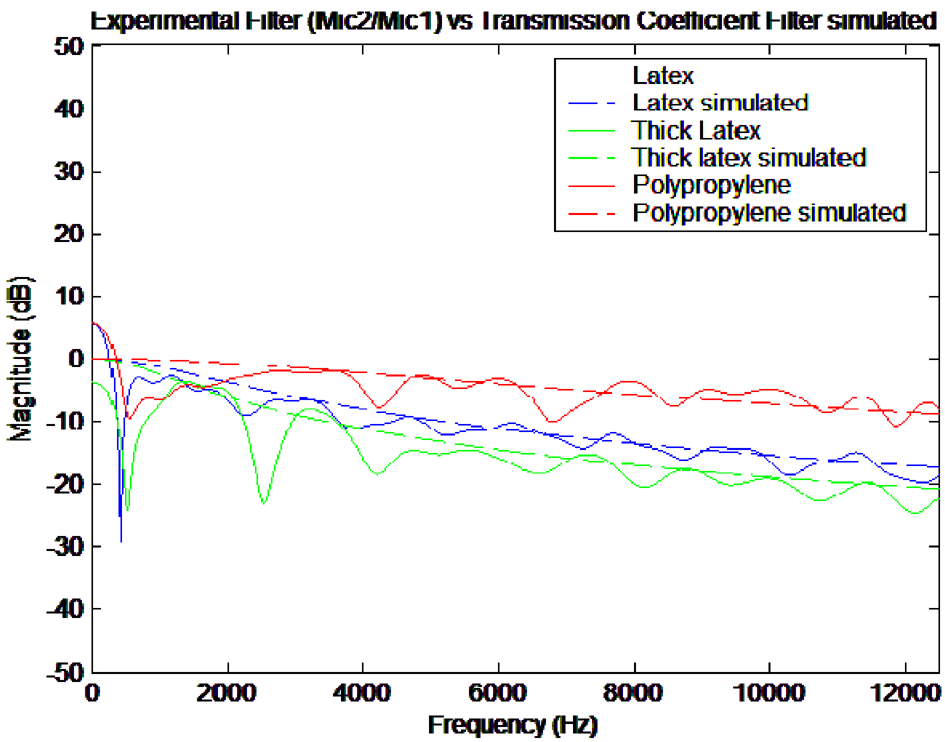
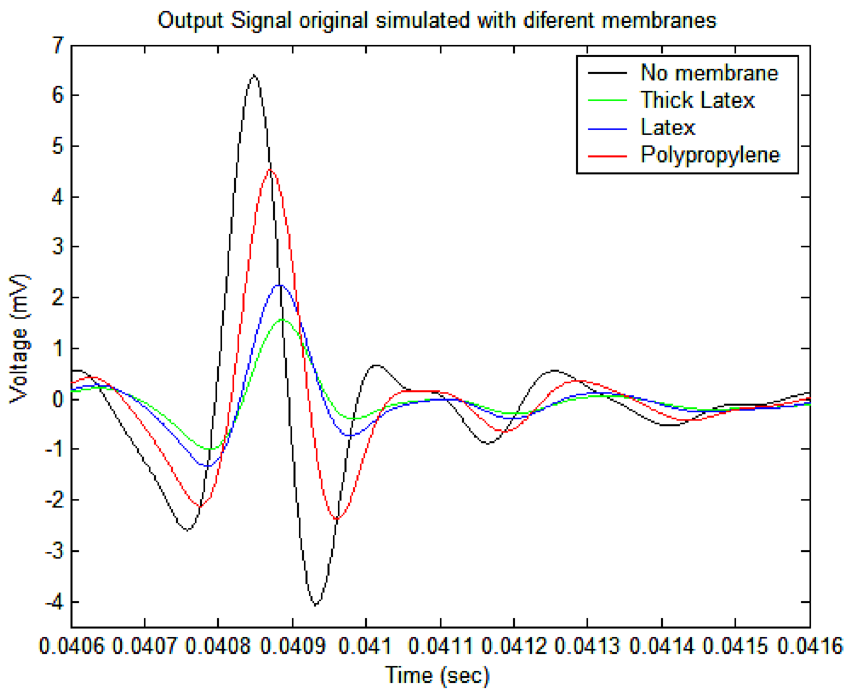


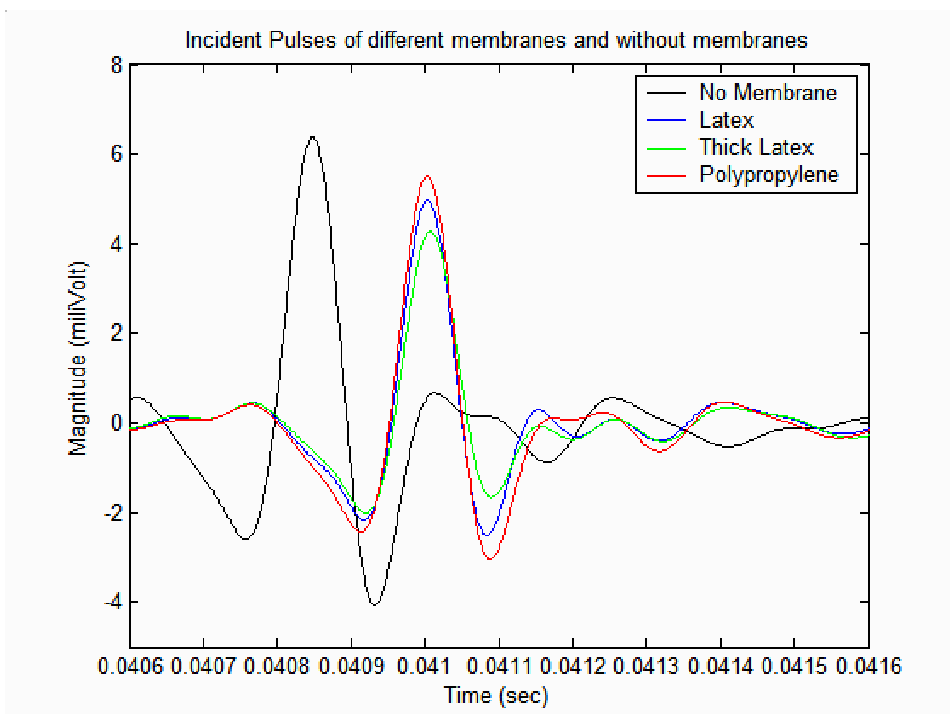
Figure 56. Experimental filter (Mic2/Mic1) vs. Transmission Coefficient filter.

4.3.2. Frequency behavior using the coefficient transmission filter vs. behavior obtained experimentally.

In this section we show the frequency response for the different pulses filtered experimentally and theorically and we compared each one. Initially we show the acoustic pulses obtained via simulations vs. those obtained in the experimental setup (Figure 57a, b). where we can observe the time delay present between the original pulse vs. the filtered signal.

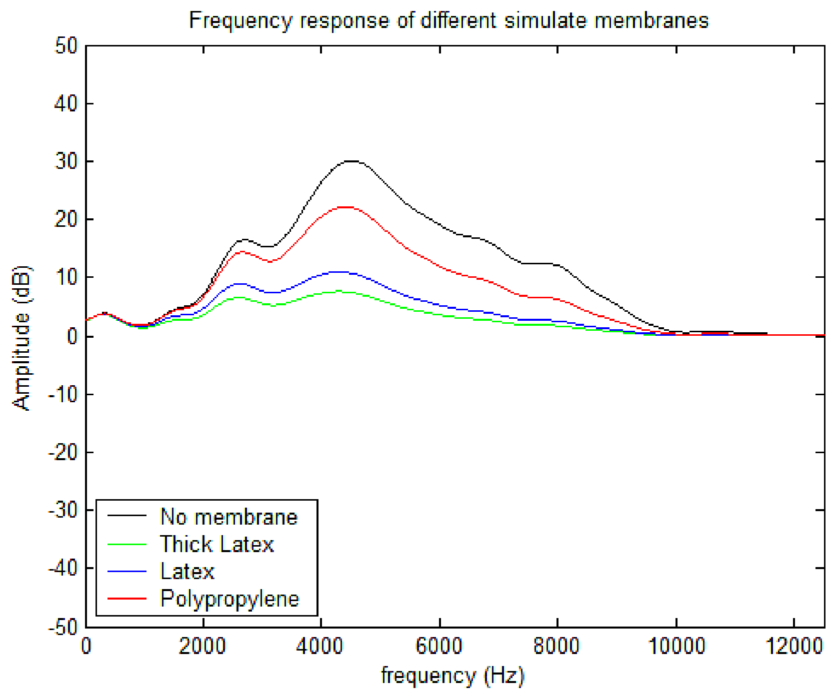


a)

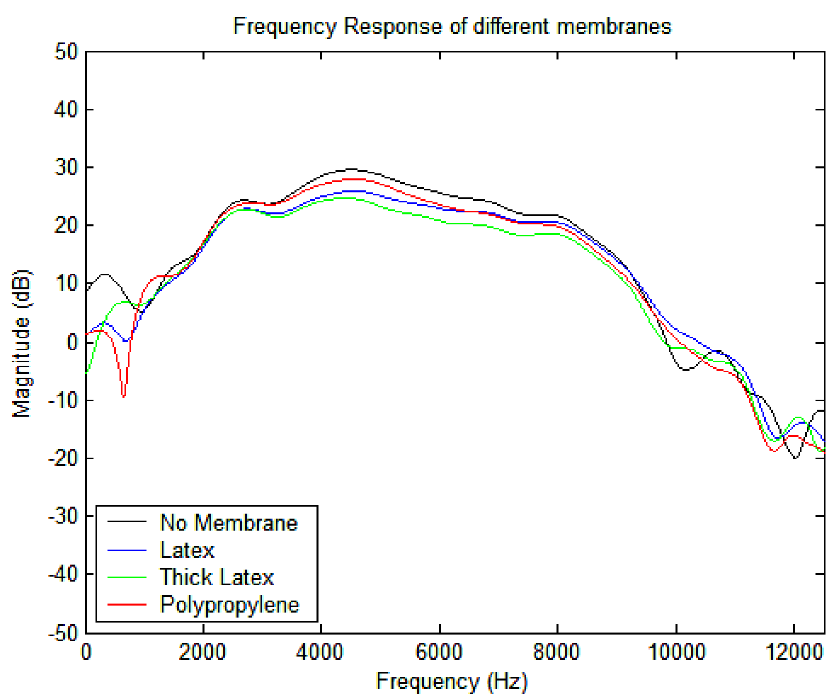


b)

Figure 57. Acoustic pulses obtained via simulations vs. obtained in the experimental setup No.1. a) Simulations, b) Experimentally



a)

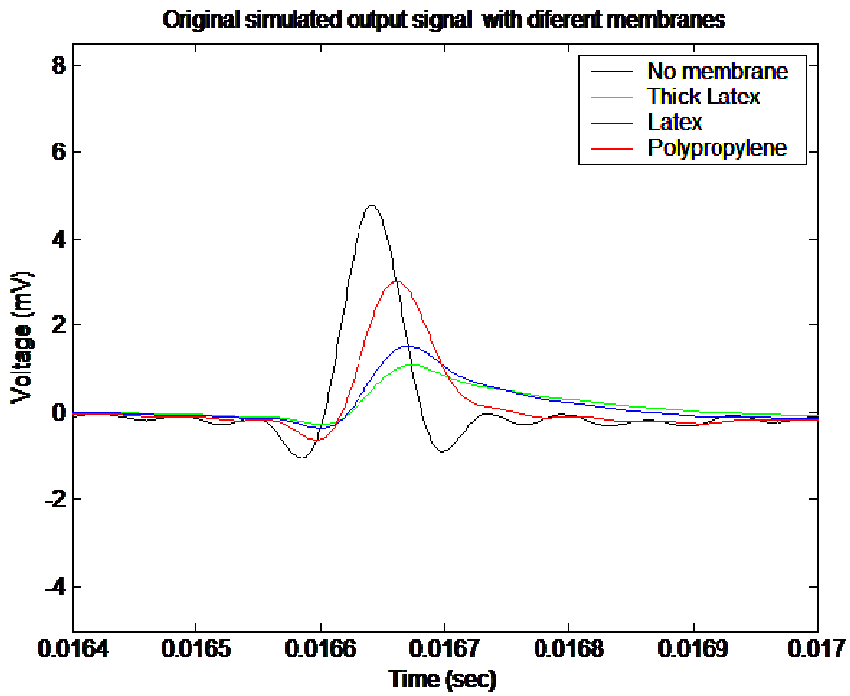


b)

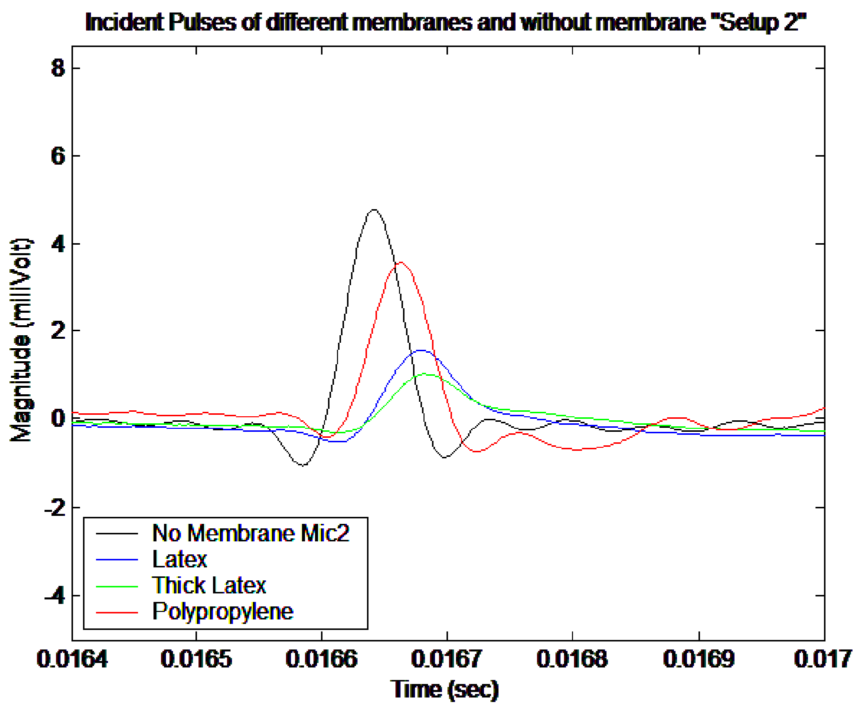
Figure 58. Frequency response of different membranes. a) Simulated results, b) Experimental results

The Figure 58 shows the frequency response for the different membranes used in the setup No. 1. This data was acquired using one microphone, where the membrane was situated between the output of the speaker and the microphone. The results showed similar frequency response and attenuation among each type of membrane, both simulated and experimental. Notice that the range in frequency where the behavior was better, was restricted for the factors such as the performance specification of the speaker, the noise floor in the laboratory and the attenuation in the system, although this last factor was not so influential.

Figure 59a, 59b show the acoustic pulses obtained via simulations versus those obtained during the experiment with setup No.2. We can see the difference between them, but the trend found in the experimental data demonstrate a variation in several factors such as the phase angle, the pulse attenuation and the frequency response. In Figure 60, we observe the differences among the frequency responses of the different types of membrane materials during the simulations and the experiment in the second setup. First of all, the range in frequency obtained experimentally was wider than the one found during simulations, but since the experimental results were limited by the system conditions, the resulting range was the same for both ranging from 1000 to 12000 Hz. Also, the order in behavior most closely to the desired one for the different types of membranes was the same (first polypropylene, second Latex and third thick Latex), but for the simulated results, the attenuation of the different pulses varied.

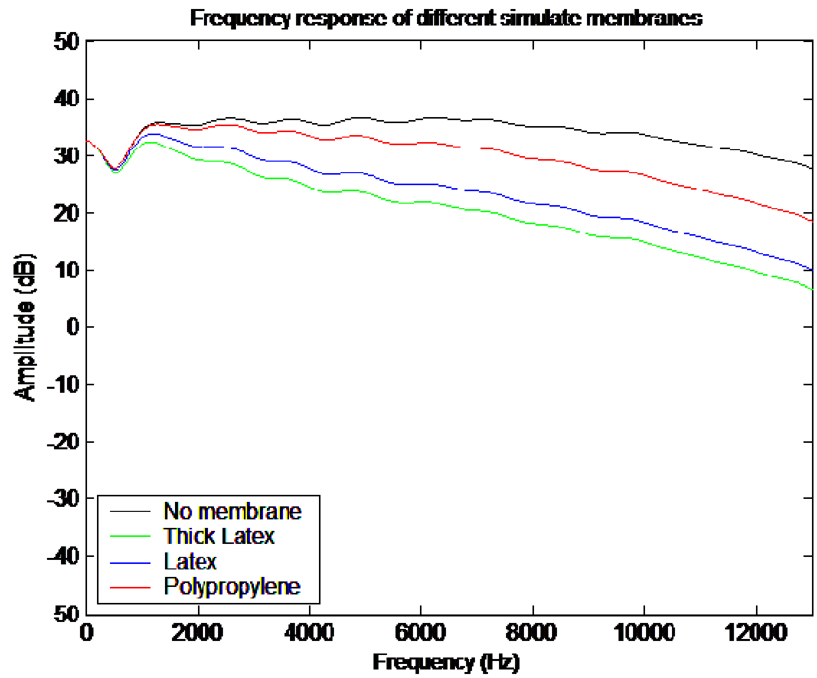


a)

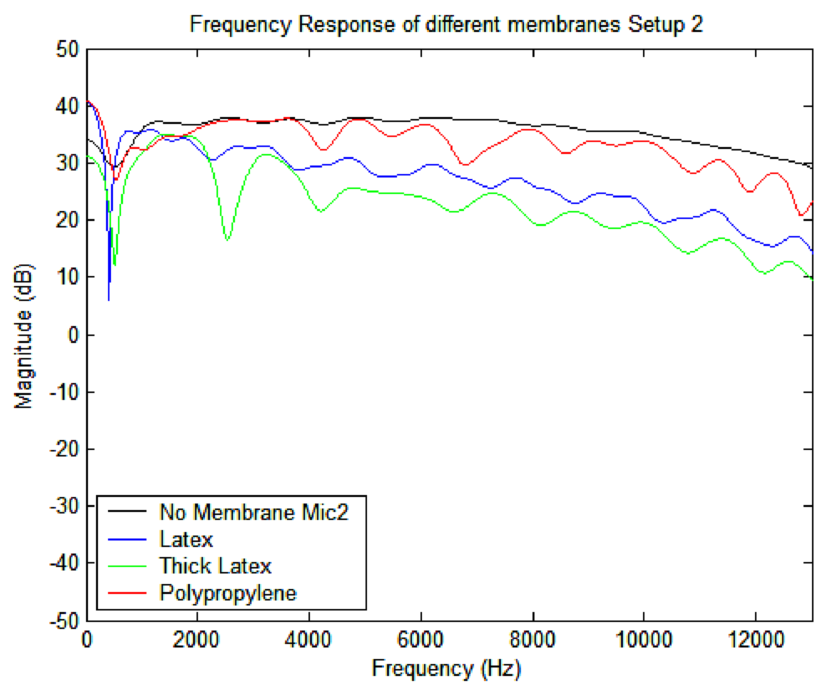


b)

Figure 59. Acoustic pulses obtained via simulations vs. obtained in the experimental setup No.2. a) Simulations, b) Experimentally



a)



b)

Figure 60. Frequency response of different membranes for setup No. 2. a)Simulated results, and b) Experimental results.

4.3.3. Sensitive analysis using the transmission coefficient filter varying the three membrane parameters.

A sensitivity analysis was performed to gain insight into the effects that each of the three membrane parameters (thickness, density and sound speed) had on sound transmission across the membrane. The sensitivity analysis was carried out using equation (25) applied to the polypropylene membrane. Each parameter was varied from its nominal value by factors of 0.01, 0.1, 10 and 100.

Figures 61, 62 and 63 show frequency-dependant transmission coefficient curves for different values of density, membrane thickness and sound speed respectively. It is important to note that when the magnitude of the transmission coefficient is unity (0dB), the membrane becomes acoustically transparent.

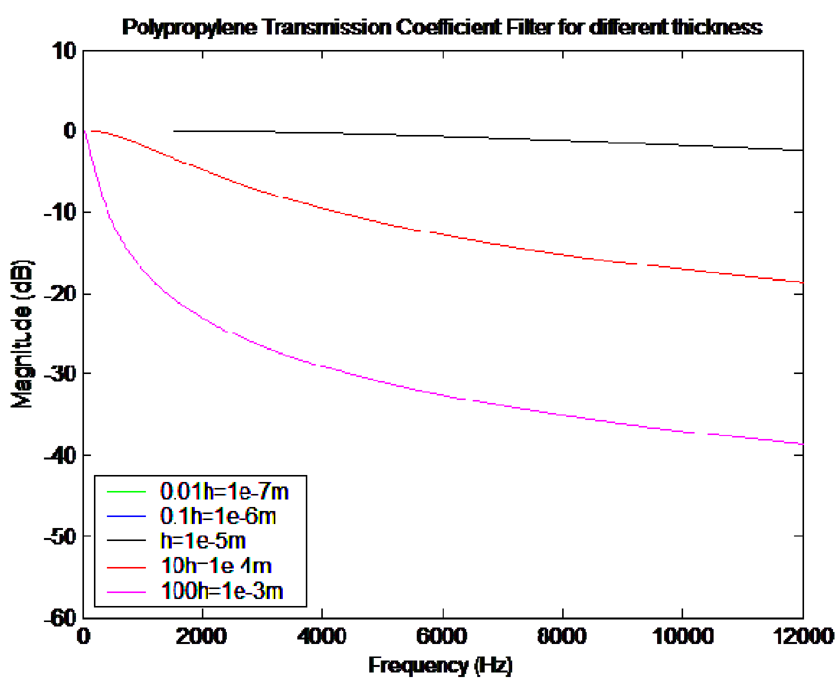


Figure 61. This plot shows the theoretical transmission coefficient of a polypropylene membrane as a function of frequency, for multiple values of membrane thickness *h*.

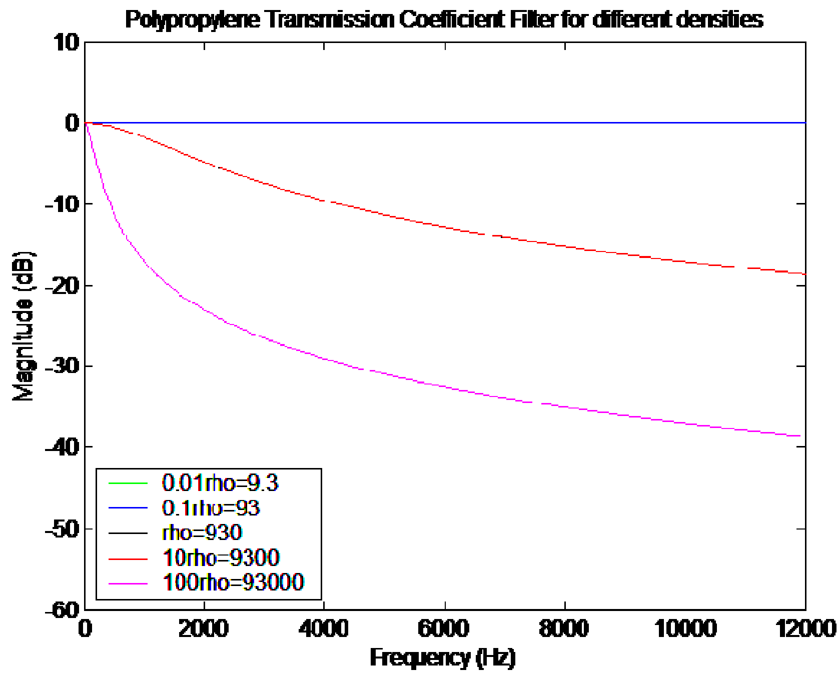


Figure 62. This plot shows the theoretical transmission coefficient of a polypropylene membrane as a function of frequency, for multiple values of membrane density *ρ*.

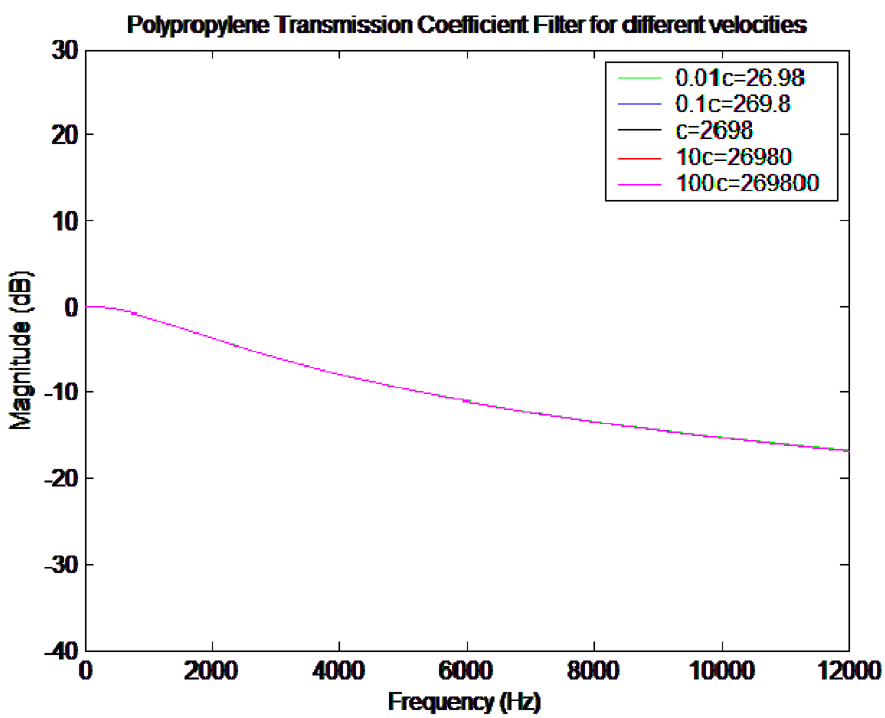


Figure 63. This plot shows the theoretical transmission coefficient of a polypropylene membrane as a function of frequency, for multiple values of membrane sound speed *c*.

4.4. Analysis of predicted results for the use of transmission coefficient in different membranes and different setups.

Although there were differences among the results, the trend observed between the simulations and the experimental results were very much alike. This differences were due to the fact that there were factors which were not considered during the simulations. One of them was the distance from the microphones to the position of the membrane for each type of setup. Furthermore, the setup construction and its dimensions influenced the exactitude in the data acquisition for the system, not to mention the parameters previously described as the noise floor, the attenuation and the performance specification of the speaker. But in spite of all, the experimental results from the setup one were more similar to the simulations than those from setup two.

In summary, this chapter comprises all the results of this investigation compared to the ones obtained by computer simulations. The frequency behavior of the different membranes in two specific setups were analyzed and compared with the coefficient transmission filter in order to determine the factors that affected the differences between the theoric and experimental values.

Chapter 5

Conclusions and Recommendations

From the results obtained in this thesis, we concluded that membrane thickness and density were the parameters that affected sound transmission the most. As membrane thickness and density became smaller, the membrane behaved as it was acoustically transparent. Increased membrane thickness and density caused the membrane to reflect more sound, and therefore, sound transmission diminished. Membrane sound speed had minimal effect on sound transmission.

If an impermeable membrane was considered to act as an acoustic filter, the overall membrane acoustic behavior was similar to that of a low-pass filter. This behavior was evident from the experimental and theoretical transmission coefficient curves shown in Figure 61. The magnitude of the transmission coefficients were close to unity (0dB) at low frequencies ($f<1.2$kHz), but then decreased as frequency increased. The polypropylene membrane exhibited the best acoustic response as compared to the latex membranes, since its acoustic response exhibited less dependency on sound frequency. This finding was expected since the polypropylene membrane was the thinnest of all (all had similar densities), and it had already been shown that thickness greatly affects sound propagation across the membrane.

The behavior of a sound wave as it traveled through a plastic tube (½" of diameters) in the presence of different types of membrane was simulated using a coefficient transmission filter as the membrane. It suggested the tendency for the general behavior of a sound pulse propagating through a membrane and the plastic tube. The effect of the membrane on the sound wave, considering all the parameters that enclose the membrane (thickness, diameter, density, sound velocity and tension), could be related since they affect quantitatively the frequency response of the acoustic pulse. This feature was very important since the coefficient transmission filter method may be applied to design impermeable membranes made with specific characteristics. It could also be used to analyze their behavior in frequency depending on its intended use. One specific application where they can be used was in hearing aids and in endotracheal tubes.

The transmission coefficient filter method was tested in audible ranges of frequencies, between 800 – 12000 Hz. At this range, the acoustic pulse was generated to be detected by the system, but without considering its limitations. The simulated results were compared with the experimental results obtained from two different setups (of one and two microphones), both constituted by a plastic tube of ½" of diameter, one speaker (Ref. XL-9457) and microphones (Ref. EM-3046). Due to the speaker limitations, the system attenuation and the noise floor in the laboratory, the effective range in frequency was reduced between 1800-12000 Hz. This results were not the same, but the trend observed between the simulations and the experimental results were similar.

We had expected that the behavior of the polypropylene and Latex membranes would be best in terms of less influence in the output once the sound wave went through it since they were thinner. Through this research, we could confirm this by the experimental results and, at the same time, it supports the theory of the transmission coefficient filter. Unfortunately, the tension of the membrane could not be measured in the laboratory.

In the future, it would be best to use a speaker with better frequency response in order to improve the working range of frequencies and the setup. Also, it would be interesting to search for a method to decrease the amplitude of the noise floor completely using acoustic isolation in the laboratory and to make simulations that take in consideration the factors affecting the system so the results are closer to the experimental ones. In addition, we could test this model with other types of membranes and improve it by including the attenuation effect.

Moreover, as an alternative thesis research, it would be realize auditory test with people with hearing problems who wear hearing aids to try these devices with and without the designed impermeable membrane, to analyze the effect of the membrane on the quality of the sound.

Chapter 6

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