IMPLEMENTATION OF A SIGNAL PROCESSOR WITH INTELLEGIBILITY FOR A HEARING AID DEVICE

A Senior Scholars Thesis

by

SHARATH N. PRASAD

Submitted to the Office of Undergraduate Research
Texas A&M University
in partial fulfillment of the requirements for the designation as

UNDERGRADUATE RESEARCH SCHOLAR

April 2011

Major: Computer Engineering

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Approved by:

Research Advisor: Jose Martinez
Director for Honors and Undergraduate Research: Sumana Datta

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ABSTRACT

Implementation of a Signal Processor with Intelligibility for Hearing Aid Devices.
(April 2011)

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Hearing loss victims suffer from loss of hearing at specific bands of the frequency spectrum. We designed a programmable filter that can compensate for hearing losses at different frequency bands. We design the circuit using operational amplifiers and resistor-capacitor banks. The circuit has programmable parameters that control the gain at different frequency bandwidths. The design is minimalistic and optimized on silicon area and power dissipation. The filter developed could be programmed to provide gains on three bands of the hearing spectrum. The circuit was realized from the transfer function and experimentally simulated for its frequency response. The results indicated that the filter behaved as expected, and could be customized using the programmable parameters. However this does not solve the eminent “cocktail-party effect”, hence the continuum of this project will include the introduction of multiple microphones to improve the signal to noise ratio by taking advantage of the spatial locality of noise.

DEDICATION

This project is dedicated to our parents who have never failed to give us financial and moral support. To my father, who taught me that the best kind of knowledge to have is that which is learned for its own sake. It is also dedicated to my mother, who taught me that even the largest task can be accomplished if it is done one step at a time.

ACKNOWLEDGMENTS

Dr. Jose Silva Martinez has been an ideal research advisor. His stage advice and patient encouragement aided my learning and research in innumerable ways. Special thanks are owed to Raghavendra, Marvin and Mohan from the Analog and Mixed Signal Laboratory.

NOMENCLATURE

K1	High frequency parameter
K2	Medium frequency parameter
K3	Low frequency parameter
BW	Bandwidth
ω_0	Pole frequency

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CHAPTER I

INTRODUCTION

Hearing loss is a widespread problem. Since 1989, Knowles Electronics has conducted six MarkeTrak surveys of the US hearing loss population following the landmark 1984 Hearing Industries Association (HIA) study [1]. The article indicates that nearly 33.4 million individuals in United States alone have some form of hearing impairment. There has also been a sharp increase in the adoption of hearing aid devices. Fig 1 shows the growth in population over the years.

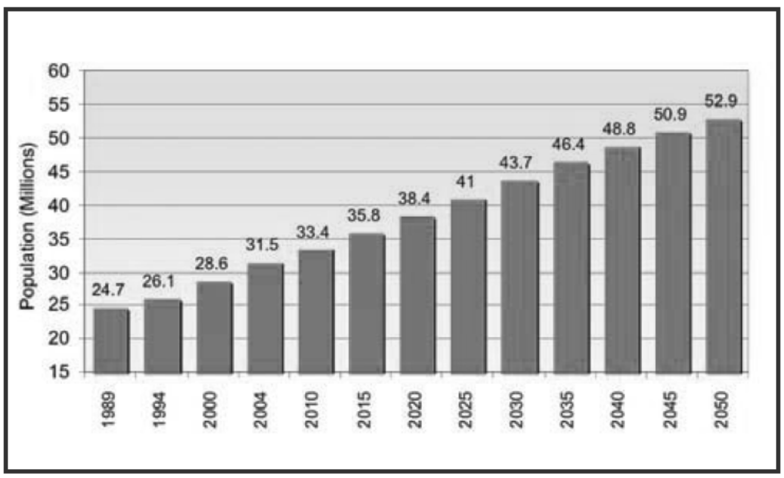

FIGURE 1. Hearing loss population (1989-2004) in millions with projections through the year 2050 based on MarkeTrak incidence of hearing loss by age group applied to US Bureau of Census age population projections.

As the adoption of hearing aid devices increase, the technology behind these devices

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This thesis follows the style of *IEEE Transactions on Speech and Audio Processing.*

improves drastically. The biggest push in hearing aid devices is to make them smaller and improve their signal-to-noise ratio [2]. Electronics have played a significant role in shaping the architecture and functionality of hearing aids since the transistor and integrated circuits era.

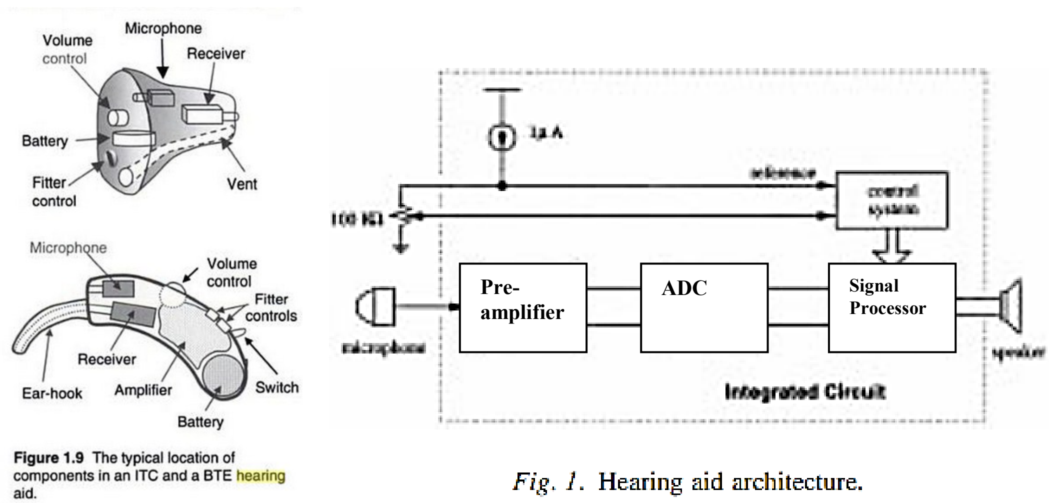

Fig 2: Hearing aid architecture [3]

Key components of a modern hearing aid device are the microphone, the signal processing circuit and the speaker. Digital hearing aids that perform amplification on digital filters have an analog-to-digital convertor (ADC) and a Digital-to-Analog convertor (DAC). Fig 2 represents typical hearing aid architecture.

Hearing loss among people are not the same. Hearing impairment could be in the form of decreased audibility, decreased dynamic range, decreased frequency resolution and decreased temporal resolution to name a few. Hearing impaired individuals have trouble hearing certain frequencies of sound. Sounds are recognized by noting which frequency has the most energy. Individuals with impairment will have trouble understanding speech, since they cannot hear some phonemes very well.

Hearing aids compensate for the losses in a person's ear by providing gain at frequencies that the person is hard at hearing. Hearing aids face a range of challenges that include low power consumption, high noise environments, dynamically changing noise conditions, size consideration, etc. In the system described below, we attempt to develop a low power hearing aid using Operational Amplifiers that can be programmed to provide gain at different frequency bands to compensate for the hearing losses [4], as illustrated in Figure 3.

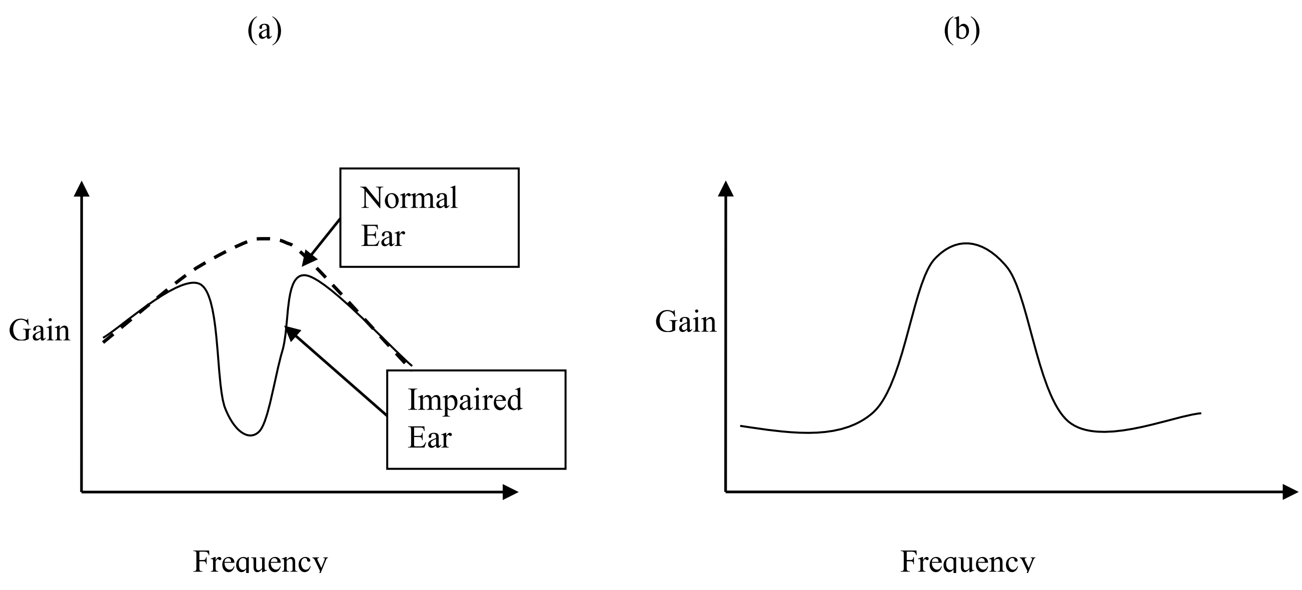

Fig 3: (a) Human ear, and (b) Compensation from hearing aid

CHAPTER II

METHODOLOGY

Transfer function

The main goal of the project is to design and implement an electric circuit that emulates a transfer function with three variables that vary the gain.

A conventional approach would be to implement a low pass, a band pass and a high pass filter in parallel with programmable peak gain. For a second order system, the implementation needs at least six operational amplifiers, three capacitor banks, and the implementation of six poles. This approach is expensive on silicon area, power dissipation and the number of switches [4]. Therefore, we implement a system that is not heavy on these resources.

One of the methods to achieve the result is described below. We realize a single transfer function that controls the three bands (low, high and medium). The values of K1, K2 and K3 alter the gain on higher, middle and lower frequency bands respectively [4].

Transfer Function:

$$H(s) = \frac{K_1\left(s^2+K_2K_3BWs+K_3\omega_0^2\right)}{s^2+K_1BWs+K_1\omega_0^2} \qquad (1)$$

K1 = Parameter that can be varied from 1 to 10. Controls the gain for the high frequency band

K2 = Parameter that can be varied from 0.5 to 2. Controls the gain for the medium frequency band

K3 = Parameter that can be varied from 1 to 10. Controls the gain for the low frequency band

BW = Bandwidth of the poles respectively

ω_0 = The frequency of the poles

This transfer function can also be implemented in digital hearing aids.

Circuit stage

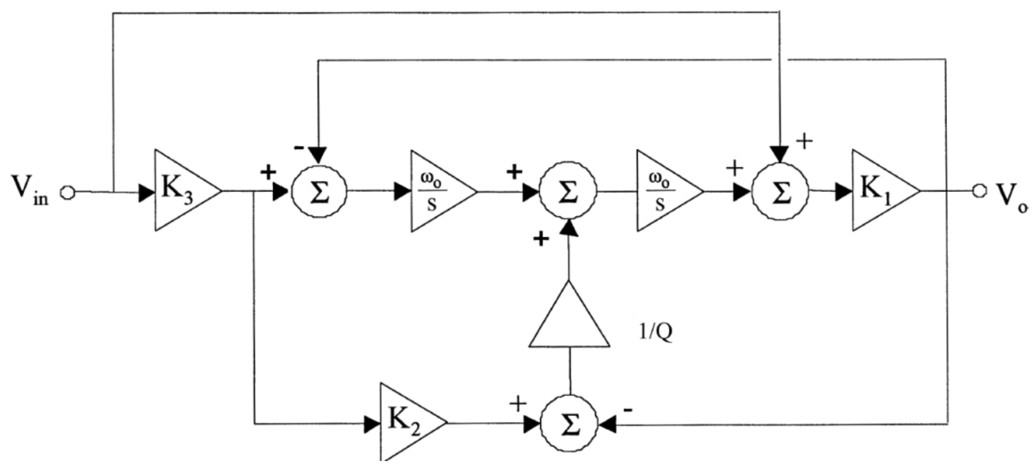

Fig 4: The transfer function flow chart

Fig 4 represents the behavior of the transfer function. In this figure, the filter bandwidth BW is equal to 1/Q, with Q equal to the quality factor of the filter. The control parameters are the gain factors K1, K2, K3 [4]. The gain of the circuit at the different bands varies from 0dB to 20dB. The bands are defined in regions 100-350Hz(low), 350-1000Hz(medium) and 1000-5000Hz(high).

A switched capacitor circuit that implements the block diagram requires only three operational amplifiers as shown in Fig 5.

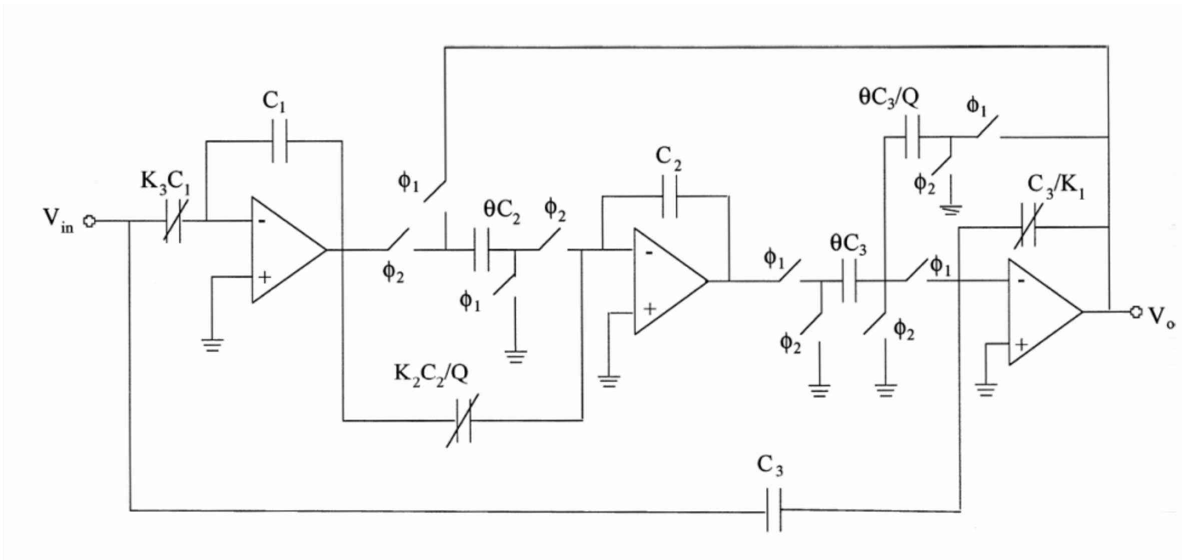

Fig 5: Op-Amp and switch implementation of transfer function

The proposed circuit does not have switched capacitor circuits. We implement resistors and capacitors to mimic the same behavior. The figures below (Fig. 6, Fig. 7 and Fig. 8) is the PSpice simulation of our circuit.

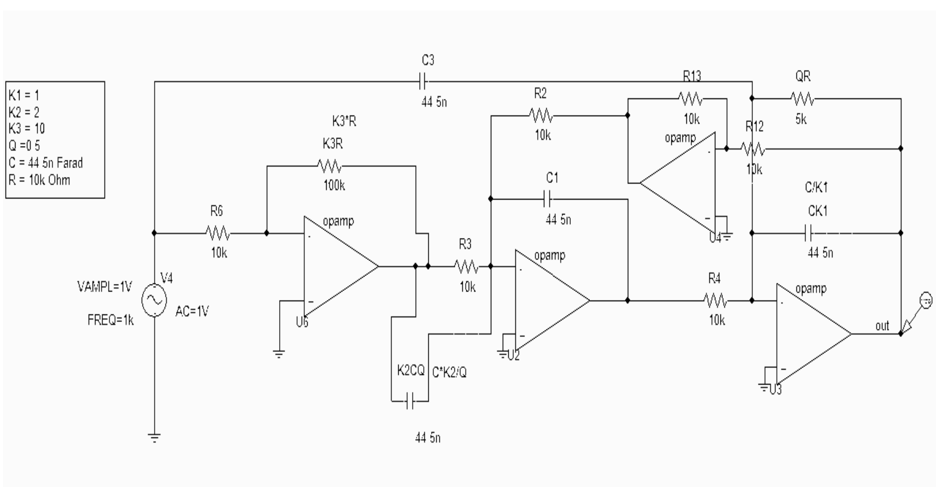

Fig 6: Pspice simulation of the function

Below are the frequency responses of the circuit above.

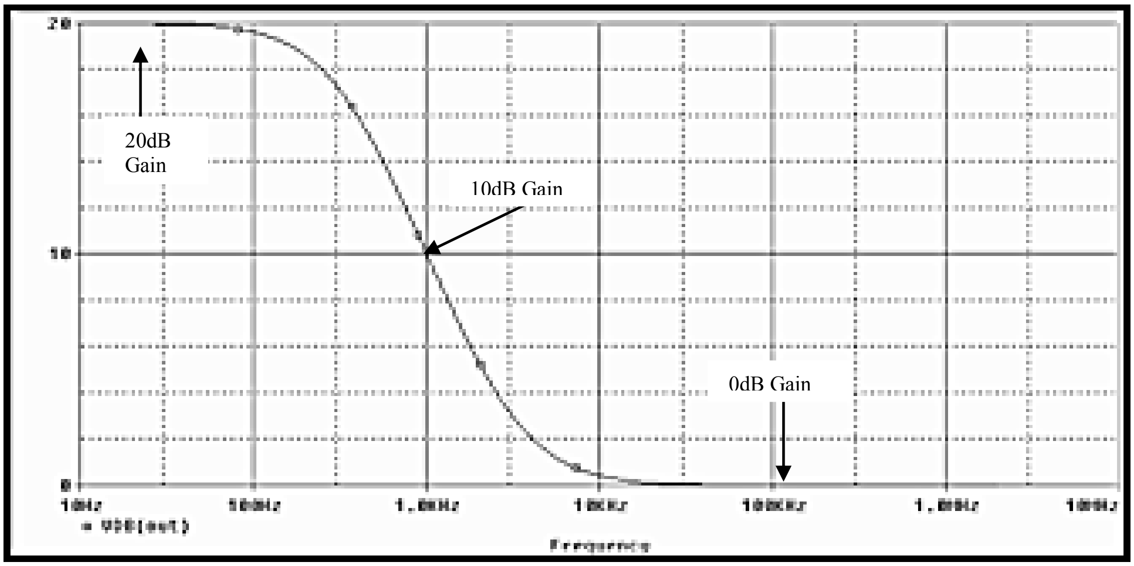

Fig 7: Frequency response of the simulation (K1=1 K2=1 K3=10 Q=0.5)

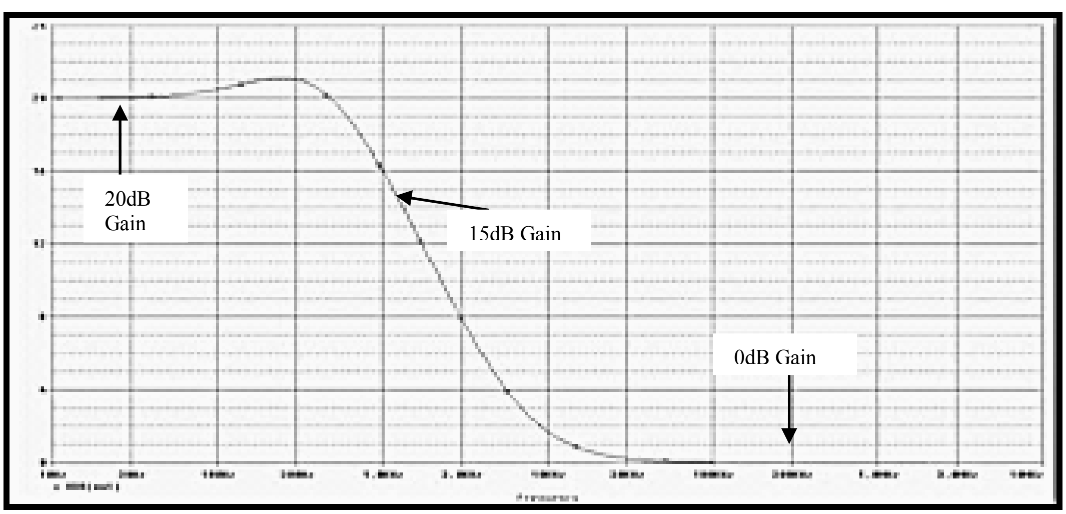

Fig 8: Frequency response of the simulation (K1=1 K2=2 K3=10 Q=0.5)

We notice from the above plots that as the K1, K2 and K3 values are varied, it varies the gain on at different frequency bands.

Hearing loss at the different bands is compensated by adjusting the gain at those bands using the variable parameters.

Multiple microphones

A radically new approach to the problem would be to try to use more than one microphone to distinguish between spatially-separated sound sources and improve the intelligibility between sound and noise to significantly increase the signal to noise ratio. Hearing impaired listeners using monaural hearing aids (single microphone) often have difficulty understanding speech in noisy and speech like noise environment. In these situations the fact that normal listeners have no difficulty due to listening from both ears indicates that multiple microphones could help in "Cocktail party environments".

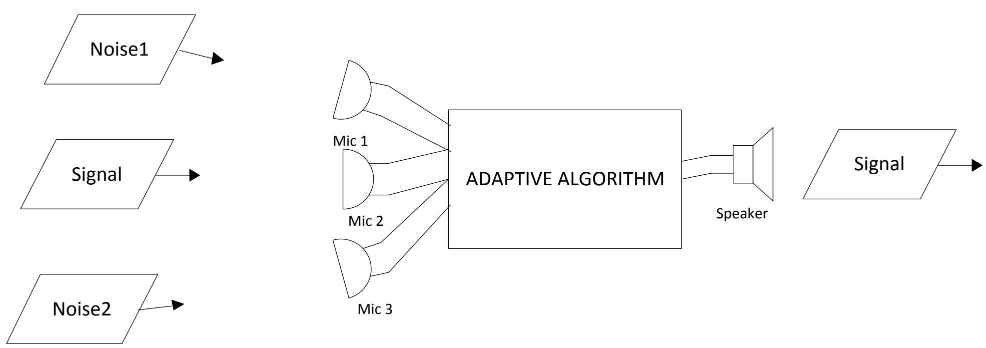

Fig 9: Multiple microphone approach with adaptive algorithm

The idea of our approach would be to try to identify the signal by comparing the signals from the three microphones. The Adaptive algorithm would be trained to perform signal processing to identify the signal in a noisy environment. Fig 9 represents the overall architecture of the device. We have three separate sound sources and we aim to enhance the signal with respect to the noise sources. We feed the signals from the three microphones and aim to capture the signals and process it using an adaptive algorithm that filters out the noise [5-8].

Adaptive algorithm

The adaptive algorithm provides a possibility of combining the signals from a set of microphones to produce one signal in which a particular sound source is emphasized relative to all other sources.

One of the approaches would be Linear processing, which combines signals from different microphones with different weights, can be synthesized to optimize a mathematical performance criteria, such as the signal to noise ratio (SNR) [8].

CHAPTER III

RESULTS

Operational amplifier

The key component of the signal processor is the operational amplifier. The Op-Amp must work at a low voltage of +/- 0.7 V (1.4V pk-pk), to work with battery of the hearing aid device. It must provide the required gain for the circuit. The operational amplifier chosen was the LMV951 from National Semiconductor Corporation [9].

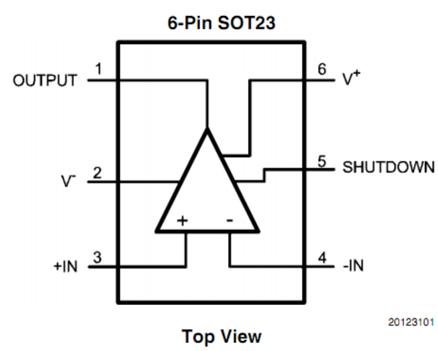

Fig 10: Connection diagram of op-amp

LMV951 is a 1V, 2.7MHz, Rail-to-Rail Input and Output Amplifier with Shutdown Option. Fig 10 shows the pin connections of the op-amp.

Features of the Op-Amp are:

1. 1 V single supply operation
2. Wide Bandwidth
3. No VOS glitch over the input CMVR

4. No input IBIAS current reversal over VCM range

5. Buffered output stage

6. High output drive capability

7. Output short circuit

   -Sink current - 35mA

   -Source current – 45mA

8. Rail-to-rail buffered output

   -@600Ω load – 32mV from either rail

   -@2kΩ load – 12mV from either rail

9. Temperature range - -400C to 1250C [9]

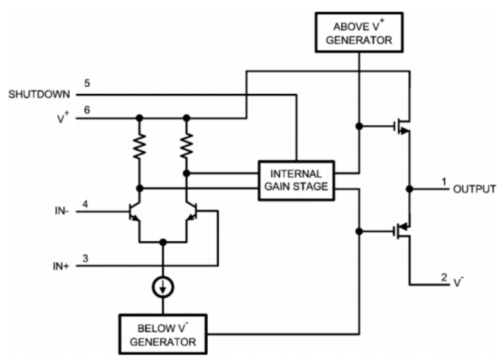

Fig 11: Internal schematic of the op-amp [9]

The model is complete with the spice macro model that describes the behavior of the Op-Amp. Simulation of the circuit by replacing the op-amps we used in spice with the macro model was possible using LTspice. Fig 11 shows the internal configuration of the operational amplifier.

LTspice

Simulation software developed by Linear Technologies that can run spice net-lists [10]. Sub-circuit of LMV951 was imported and the programmable filter was simulated. Fig 12 represents the implementation of the LMV961 op-amp for simulation and experimentation. Fig 13 represents the response from the circuit developed in figs 12-13.

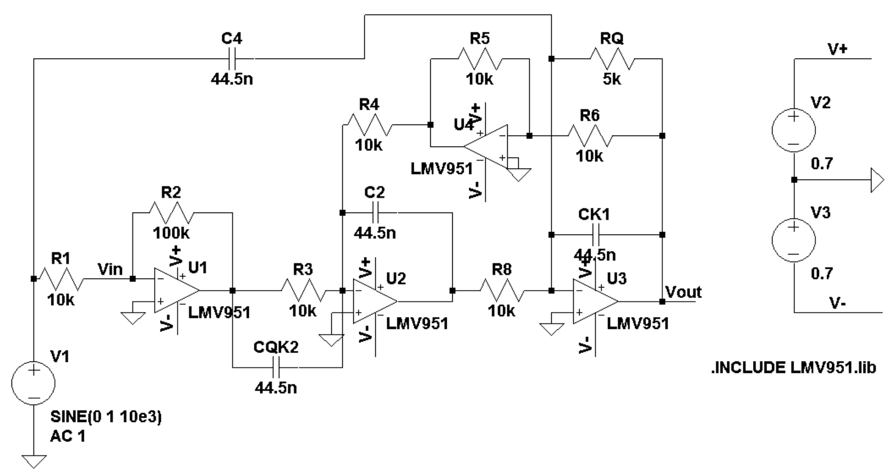

Fig 12: LTspice simulation of the filter using LMV951

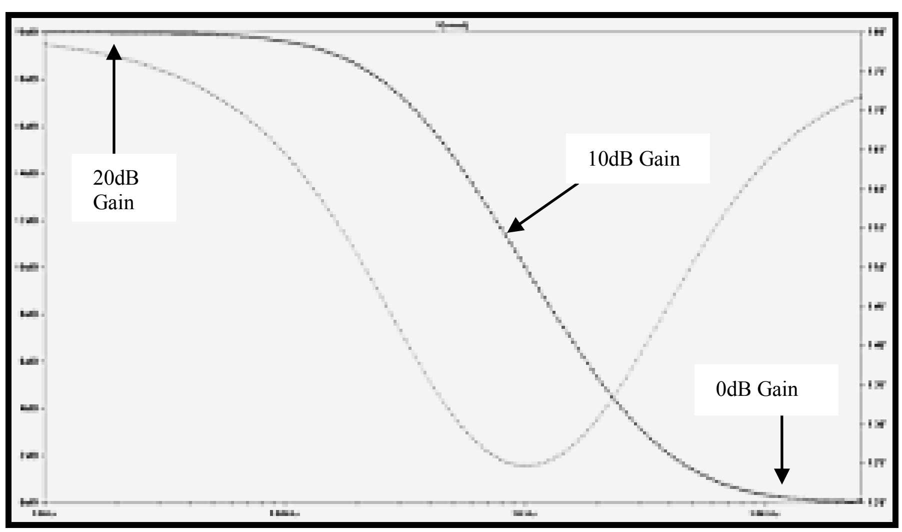

Fig 13: LTspice simulation K1=1 K2=2 K3=10 Q=0.5

CHAPTER IV

SUMMARY AND CONCLUSIONS

The project shows the implementation of a signal processor that can compensate for the losses in a human ear. The signal processor works on three parameters K1, K2 and K3. The parameters can be programmed to each individual with hearing loss at specific bandwidths of hearing. The signal process first describes the transfer function with the three parameters. It is a single input and single output type processing. The transfer function is theorized in circuit. The circuit is then idealized in PSpice (simulation software) using ideal operational amplifiers and resistor-capacitor banks. Simulation results exactly describe our prediction on the gains provided for the specific values of the parameters. During experimentation, we replaced ideal operational amplifiers with a LMV951, an off the shelf amplifier, from National Semiconductor. LTspice enabled the importing of the spice netlist for LMV951 in order to do the simulation. The results of the experiment from LTspice indicated that the design produces the results as predicted in theory.

Our current approach is to include multiple microphones in order to develop a system that can improve the signal to noise ratio in a high noise environment where the noise is more speech like. An adaptive algorithm that maximized signal and minimizes noise is being explored. We plan to develop a system that is optimized on SNR and minimalistic on resources such as power and number of microphones.

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