Hearing Aid

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Introduction

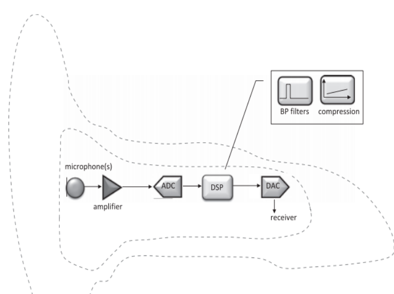
Hearing is one of the most precious senses humans can have. Hearing helps one to sense the environment and plays a major role in communication with others. Human beings are able to hear sound ranging from 20 Hz to 20 kHz but most sensitive to 1kHz to 4kHz. Sound pressure level are used to measure the loudness of sound for human ears (2×10^{-5}Pa is the minimum threshold for human hearing) and is defined in decibel (dB), which ranges from 0 dB to 120 dB (dynamic range - 40dB - 80dB). Unfortunately, a person used to lose hearing abilities because of accidents, hereditary and ageing. There are two types of hearing loss, conductive loss and sensorineural loss. One can suffer from either one or both. In the conduction loss the middle ear is damaged which prevents the conduction of the sound energy from the outer ear to the middle ear. In sensorineural loss, some auditory nerves are no longer able to send electrical pulse to the brain and this is caused by natural ageing of an individual. The sensorineural loss is characterised by following:

- Sinking of dynamic range between the soft and loud sound speech
- Loss of certain frequency components (mostly high frequencies)
- Hinders the ability to detect sound of different frequencies occurring one after the other.

Sensorineural loss of hearing can be mitigated by using hearing aids. Miller Reese Hutchison made the first ever electric hearing aid in the year 1893. First commercial hearing aids were launched in 1913. The Basic function of hearing aids is to control loudness, frequency and background noise to aid the patients hear speech or desired sounds without discomfort.

Theoretical background

The first part of a hearing aid is the microphone. This converts the sound wave into a voltage signal, which is further processed. The processing of this voltage signal is most important in the design of a hearing aid. The accuracy of this processing technology determines the overall effectiveness of the device. So, the voltage analog signal is converted to digital signal by using ADC. This digital signal is then processed using a microcontroller. Then the final, clean and amplified signal is then sent to DAC to convert it again to analog. Finally this analog signal is transferred to the receiver (speaker) in the ear.



Working of a Microphone

Majority of microphones are based on a technology called electret. On the basic level, a microphone can be of two types. First, omnidirectional, these microphones detect audio from all directions equally. Second, Directional microphones, these microphones are sensitive to one direction. This can be said that omnidirectional microphones are frequency independent sensitive. To enhance directional quality, more than one microphones are used. However, this comes with a cost of more hardware complexity and more power consumption by the device.

Single Directional Microphones

In most situations, we only need to hear sound from one direction. This direction is from front in most of the conversation. Hence, we increase the sensitivity of the microphone for sound coming from in front of the microphone. This also helps in reducing the noise.

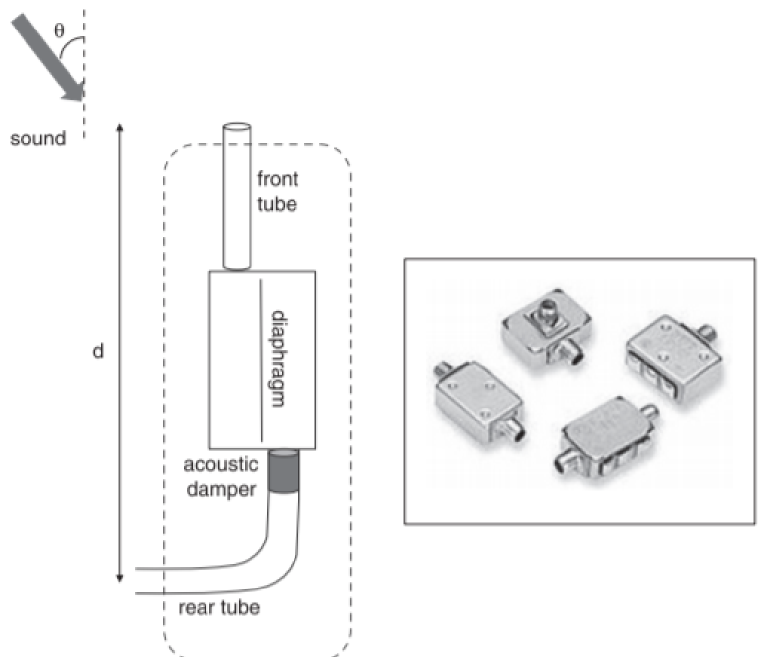


Figure : Schematic and images of omnidirectional microphone

In this model, there are two tubes fitted on the opposite sides of a diaphragm. Sound enters from both the plastic tubes. The tubes are separated by a distance of *d*. We assume that the wavelength of the sound wave is λ. Now, the formula for calculation the external phase delay of the extreme phase difference is given by

$$\phi_{ext}\,(in\ degrees) = 360\frac{d}{\lambda}$$

Dual Directional Microphones

The main problem with the Single Directional Microphone is the fixation of polar pattern during the manufacturing process. The fix to this problem and keeping the directionality, the adding of more than one microphones is being followed in the industry. Such assemblies are called microphone-arrays.

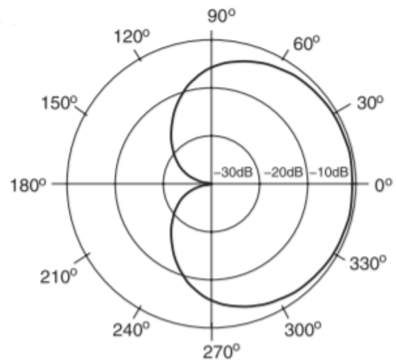


Figure: Plot of a directional microphone with sensitivity highest in front of the microphone.

Functions of Hearing Aid

Following are the basic functions of a hearing aid

- Pre-Amplification to reduce noise before processing.
- Digital processing to remove noise
- Compression of the audio signal to a suitable range for the patient

Pre-Amplification:

Signals coming from the microphone are weak and they are not feasible for transferring to ADC(Digital to analog converter. So , the signals are amplified using pre-amplifiers for further processing. We can use simple op amp or instrumental amplifiers.

Analog to Digital Converter (ADC):

Analog Signals from the microphone are sampled under a specific sampling frequency . For proper working , frequencies present in the signal must be less than half the sampling frequency. Sampled values are digitized and converted to binary numbers .

Digital Signal Processing (DSP):

The unwanted high and low frequencies noises are removed from the signal using a digital band pass filter. It can amplify the signal easily as compared to analog systems. The clean digital signal is sent to DAC for further amplification using a compression amplifier.
There are many advantages to using a digital processing system.

- It reduces the internal noise.

- The hearing aids can be customised for individual choice very easily by changing the programs instead of changing the circuit in case of analog systems.

In our project , we consider the real world situation as we can't use the matlab inbuilt functions for signal processing like fft, wavelet transform ,etc at the micro controller . So we use the difference equation for the filters.

The transfer function of the first order low pass filter is $H(s) = wo/(s+wo)$. The transfer function is converted into discrete form using matlab. The coefficients from the discrete form are used to construct the difference equation for the filter output.
The transfer function of the high pass filter is $H(s) = s/(s+wo)$. Similarly the difference equation for the high pass filter is also obtained.

Using the low and high pass filters , a band pass filter for a range of frequencies 80 to 140 Hz is formed. The clean signal is then converted into analog for further processing using DAC.

Digital to Analog Converter(DAC):

The digital signal is again converted into analog signal using DAC for further processing through a compression amplifier and then the signal is transferred to the speaker.

Compression

To understand compression, we first have to understand the need for compression. Sensorineural loss of hearing causes shortening of the dynamic range of the patient in such a manner that he/she is able to hear loud speech (desirable signal) as fine as before but lose the ability to hear soft speech as indicated in the Fig.1. Using a linear amplifier here would be less effective as it will amplify all signals given as input irrespective of the signal strength and hence may cause loud speech to be amplified to uncomfortably loud levels. There was another solution to this problem i.e. use of clipper circuit to clip the sound above 80 dB SPL, but this comes with different sets of problems like the the sound above certain level will be saturated to the high loudness permissible by the hearing aid and cause saturation-induced distortion .Peak clipping has low signal to noise ratio , hence not preferred much.
Instead we can use compression to solve such a problem. In simple words compression refers to the change in the intensity relationship among different segments of acoustic stimulus. In practise this is realised by using an amplifier to amplify the soft speech (low intensity signal) and using gain reduction for loud speeches so that the output does not become uncomfortably loud.

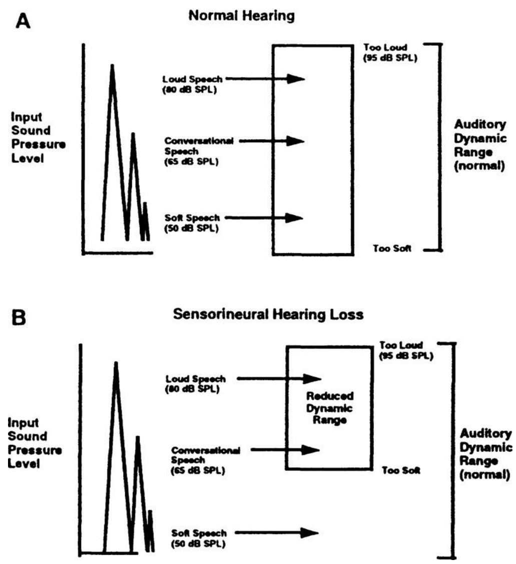


Fig.1

There are following approaches to realise of compression (Fig.2):

- Compression limiting (CL)
- Automatic volume control (AVC)
- Wide dynamic range compression (WDRC)

In our project we are planning to make a compression limiting (CL) circuit i.e the soft speech () will have high gain , conventional speech signal have moderate gain and loud speech signal have low gain .If a patient is suffering from sensorineural hearing loss which caused the dynamic range to sink from 40dB - 80dB SPlL to 50dB-80dB SPL then it is soft speech signal (40dB - 45dB) should have a gain of 10bB ,conventional speech (45db - 60dB) should have a gain of 5dB and loud speech(65dB - 75dB) should have as little gain as possible i.e 1-0.5dB SPL .For very low amplitude signal we can use squelch circuit to prevent the amplification of such signals .

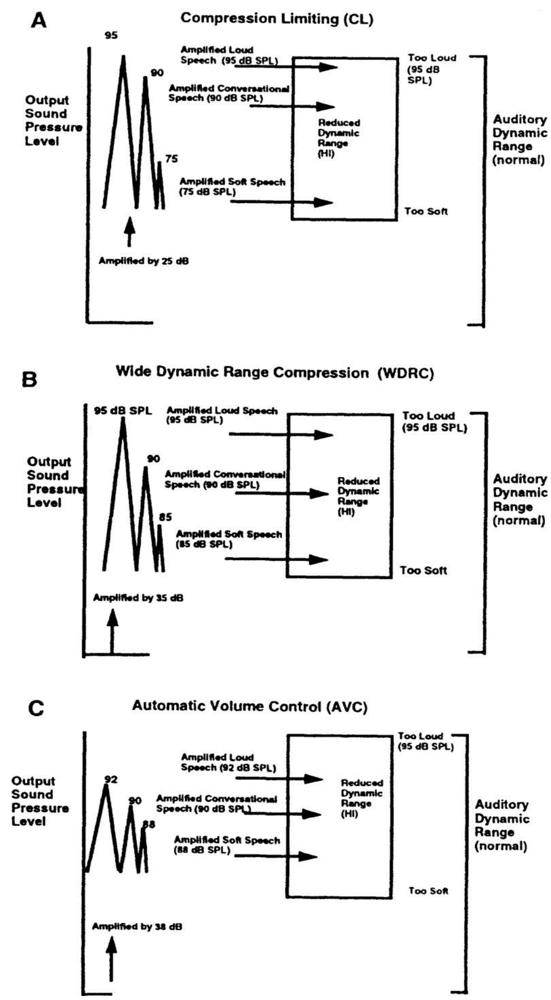


Fig. 2 - Different approaches to realise of compression

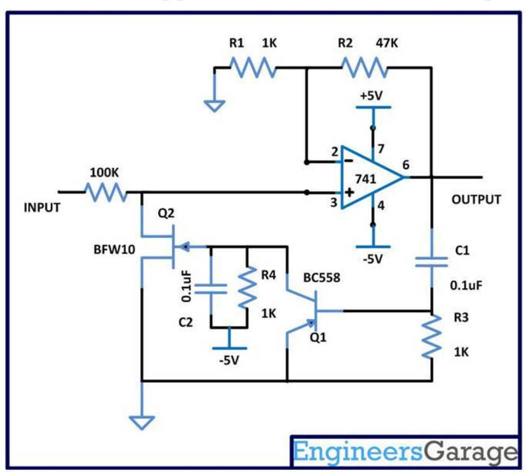


Fig. 3 - Circuit to realise compression limiter

The circuit in Fig.3 will work as automatic gain control (AGC). At a signal lower than the upper threshold (65dB) than the circuit act as a simple negative feedback op amp amplifier the source and the drain terminals of the MOSFET are not connected . If the input signal exceeds the threshold the source and the drain terminals of the MOSFET are not connected and the gain reduces accordingly.

Speaker:

Speaker generates the sound wave according to the signal given to it. This helps the patient who has hearing problems.

Design of the model:

Digital Signal Processing (DSP):

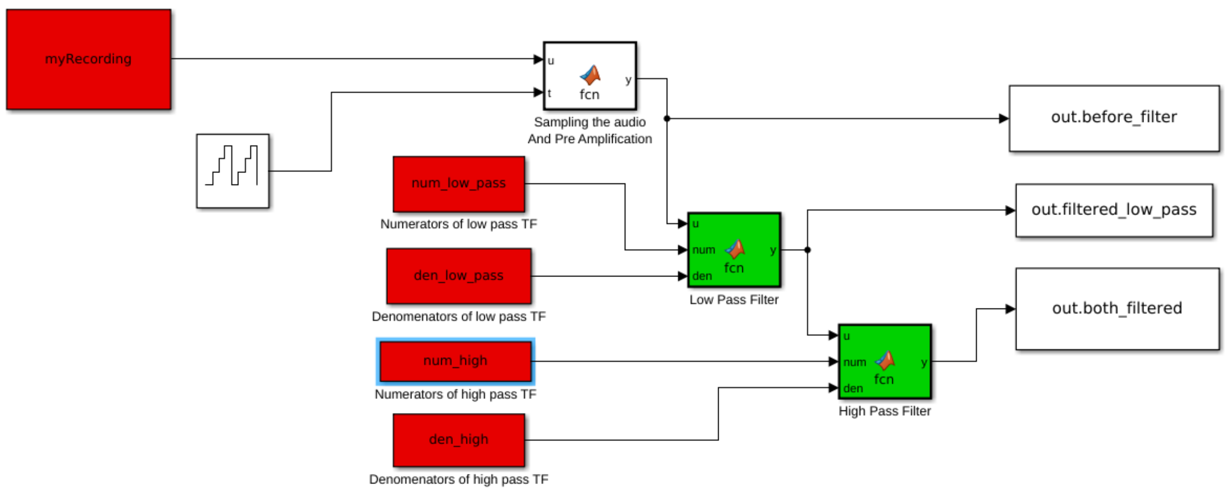


Fig: Simulink model for the Digital signal processing unit.

Details for various blocks:

1. myRecording Block ; It is a variable which represents the real time signal coming from the microphone through ADC .
2. Counter block : Counter block is used to transfer the data points from myRecording block in a discrete manner during the simulation .
3. Num_low_pass and den_low_pass : They are the numerators and denominators of the transfer function of the low pass filter.
   In this case, the cut-off frequency is 140Hz.

```
Gz_low =

  0.05211 + 0.05211 z^-1
  ----------------------
      1 - 0.8958 z^-1

Sample time: 0.000125 seconds
Discrete-time transfer function.
```

4. num_high and den_high: They are the numerators and denominators of the transfer function of the high pass filter.
   In this case, the cut-off frequency is 80Hz.

```
Gz_high =

  0.9695 - 0.9695 z^-1
  --------------------
    1 - 0.9391 z^-1

Sample time: 0.000125 seconds
Discrete-time transfer function.
```

5. Sampling the audio and Pre- Amplification: This block is to sample the audio data discreetly during simulation and the pre-amplification can be done assuming that the signal is just coming from the microphone .
6. Low pass filter : Using the numerators and denominators, a difference equation can be formed to filter the higher frequencies than 140 Hz.

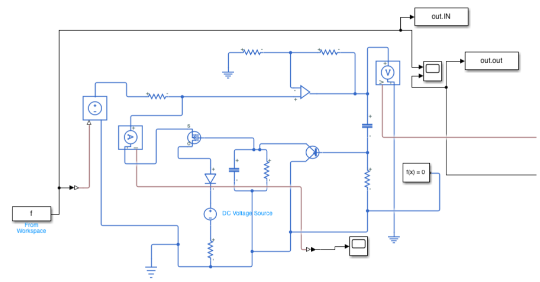
```

function y= fcn(u,num,den)
persistent x_1 y_1
if isempty(x_1)
   x_1=0;
end
if isempty(y_1)
   y_1=0;
end

y = num(1)*u+num(2)*x_1-den(2)*y_1;
x_1=u;
y_1 = y;
end

```

7.High pass filter : Using the numerators and denominators, a difference equation can be formed to filter the lower frequencies than 80 Hz.
Function is the same as a low pass filter . The only difference is the parameters.

8. Out.before_filter , out.filtered_low_pass and out.both_filtered : They are 'To the workspace blocks' to bring the values to the work spaces in matlab for analysis .

<u>Compression Littering</u> :

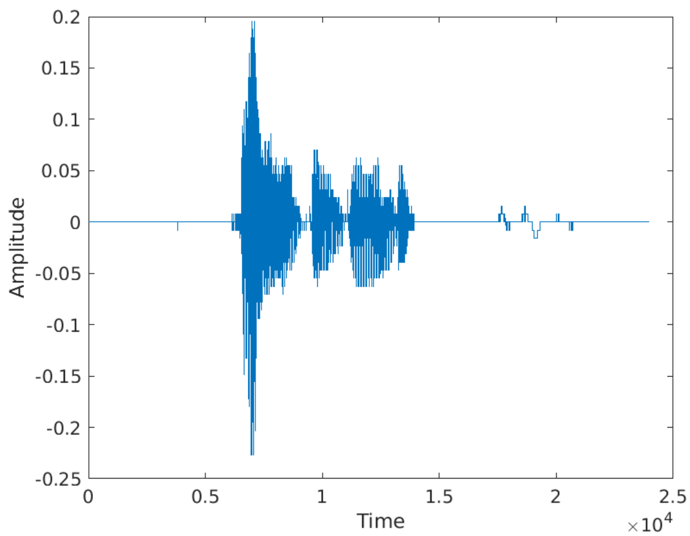
Name of blocks required	Circuit diagram
• Solver configuration • Controller voltage source • PS-simulink converter • P-channel MOSFET • BJT (PNP)	

● Diode , resistor and capacitor ● Electrical reference ● Opamp ● To workspace and from workspace ● Scope ● Voltage sensor and current sensor ● Bode plot block	 Note- similar circuit is connected in series to this to realise the functioning of the compression block of the hearing aid .

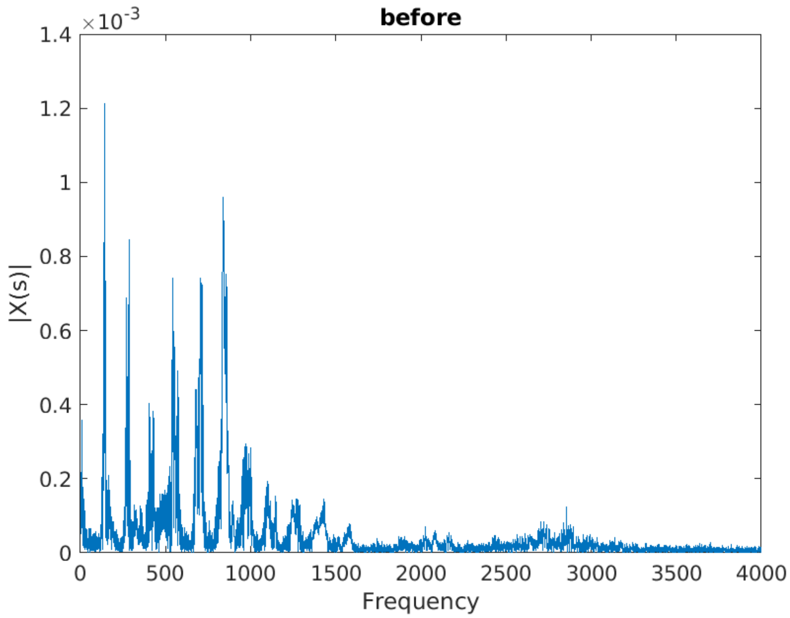
Results :

Digital Signal Processing (DSP):

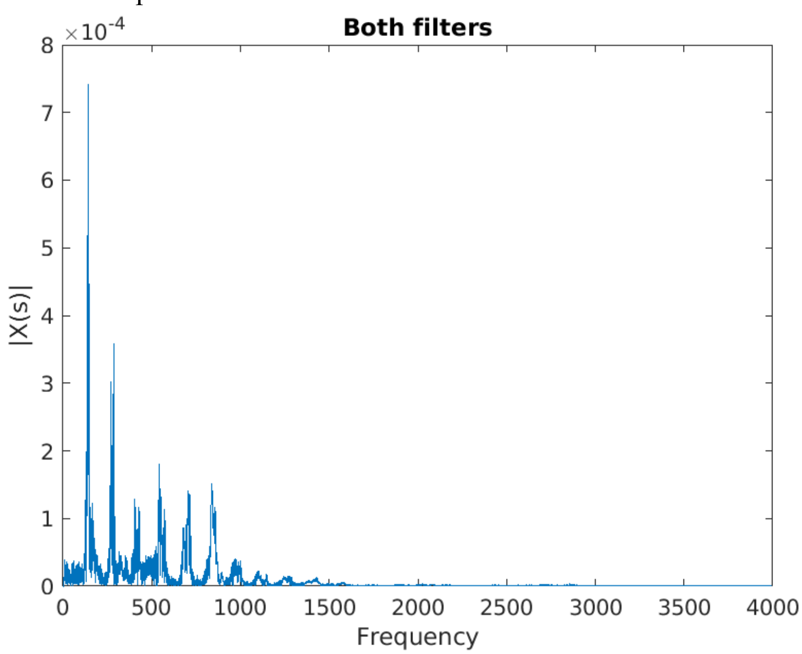
Original Audio:



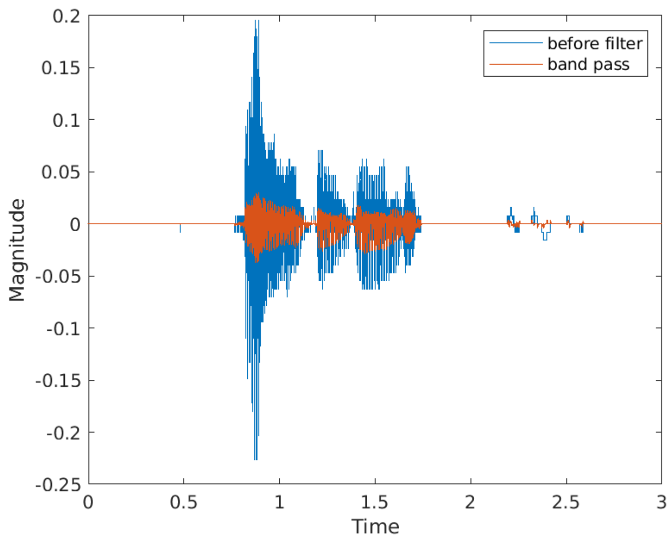
FFT for the original signal:



After band pass filter:



Comparison between the original and the filtered signals in the time domain:



After this compression amplifier is used for amplification.

<u>Compression limiting</u> :

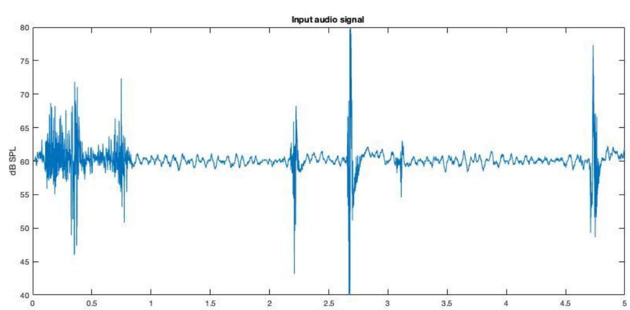


Fig. Input signal (audio)

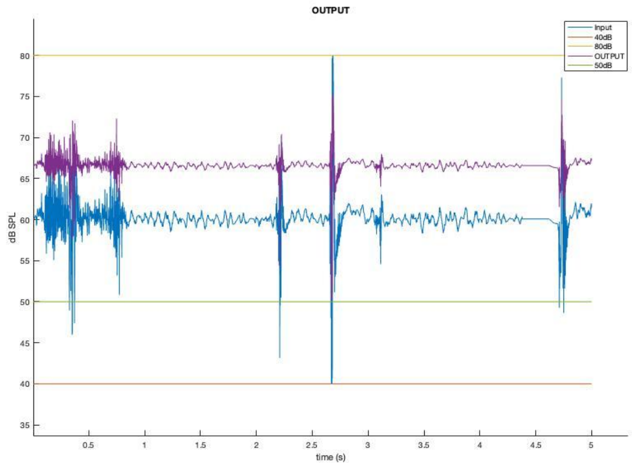
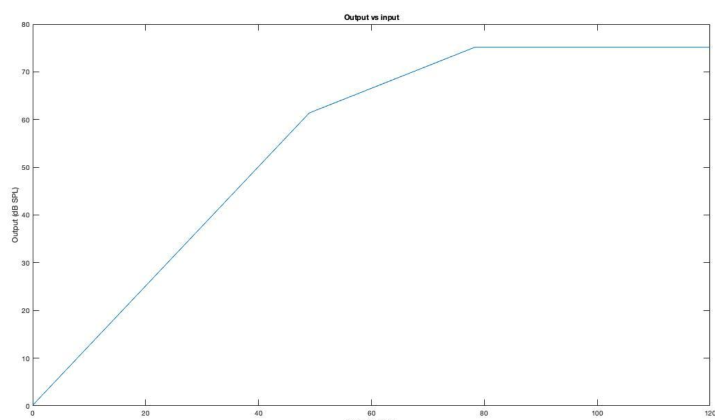
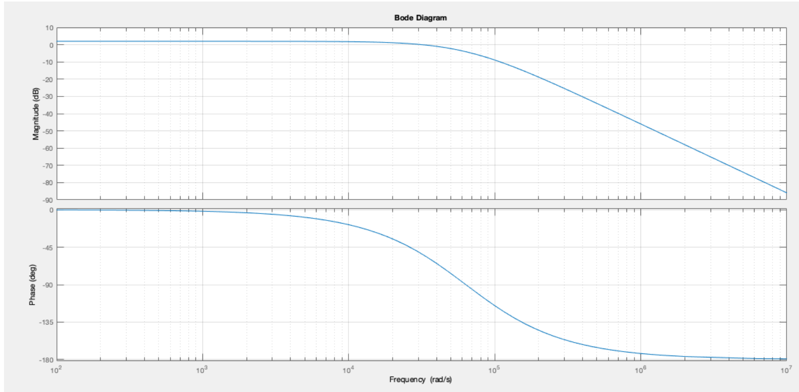


Fig. Output (using compression limiting)

If a patient is suffering from sensorineural hearing loss which causes the dynamic range to sink from 40dB - 80dB SPlL (orange and yellow line in the output plot) to 50dB-80dB SPL (green and yellow line in the output plot) . The compression limiting circuit acts as an automatic gain control circuit to compression / squeeze the audio signal into the patient's dynamic range . The output vs input plot and frequency response for the compression limiting circuit is below .



Output vs input plot



Frequency response for compression limiting

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