Hearing Aid System

Seyedkhashayar Hashemi

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Abstract:

In this project, the level-dependent amplification block of the hearing aid system is implemented. Hearing aids should be able to provide suitable audibility of sounds while avoiding intolerable loudness. In other words, weak input signals should be amplified more than stronger ones. Therefore, different amounts of amplification would be required for different input signal levels. Ideally, the gain should continuously decrease with increasing input signal level. This means that the function between input and output is compressive. Due to hardware limitations, the compressive function can not be applied to the whole input level range. The input level range can be divided into multiple ranges with different functions to tackle this issue. The goal is to design, implement, and test an amplifier with such function characteristics.

Table of Contents

List of Abbreviations:

HA: hearing aid

CF: center frequency

RMS: root mean square

SPL: sound pressure level

Multichannel Compression

Background And Design Specifications

Individuals with hearing impairment experience different amounts and patterns of hearing loss [1]. Therefore, the parameters used in the design of the HA blocks should match the frequency-specific needs of the person with hearing impairment [2]. HAs typically process the sound in 8-20 frequency channels (8 in our case). The width of the channels expands as the center frequency increases as shown in part 1. A level-dependent gain is applied to each frequency channel. This process is called multichannel compression. The goal of multichannel compression is to provide suitable audibility of sound while preventing excessive loudness. In this project, we will design and test a single channel of the multichannel HA using Simulink. The CF of this channel is assumed to be 1 KHz. The specifications are provided in Figure 1.

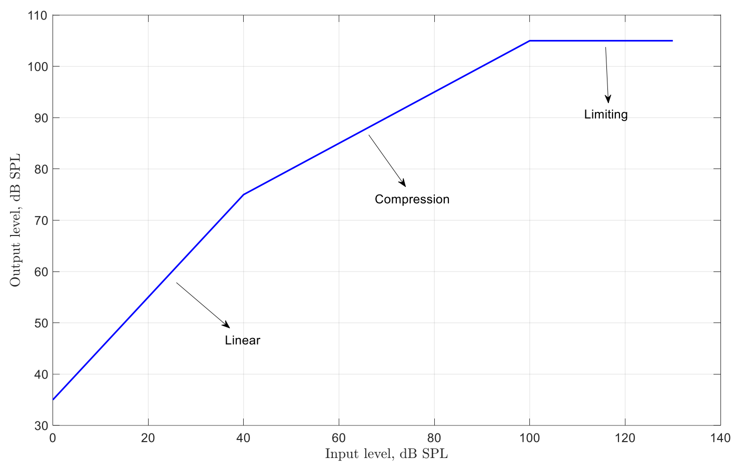


Figure 1: Output level vs. input level in dB SPL for a single channel of a multichannel hearing aid. The function consists of three regions: Linear, Compression, and Limiting.

As seen in Figure 1, three regions are illustrated. For input levels less than 40 dB Sound Pressure Level (SPL), a linear gain of 35 dB is applied to the input. (Linear) For input levels between 40 to 100 dB SPL, compression is applied. (Compression) Compression means the applied gain should be decreased with increasing input level. For example, the applied gain is 35 dB when the input level is 40 dB SPL but when the input level becomes 100 dB SPL, the applied gain is reduced to 5 dB. For input levels higher than 100 dB SPL, the output level should be kept at 105 dB SPL. (Limiting)

Design Procedure

First, we will briefly discuss what is meant by SPL. Then, we will discuss how each of the three regions can be implemented in Simulink.

1) Sound Pressure Level (SPL) measurement

To design the system illustrated in Figure 1, we need to measure a parameter called Sound Pressure Level (SPL). Sound Pressure (SP) refers to the measure of air or gas pressure resulting from the vibration induced by sound within the medium [3]. The unit of this parameter is $\frac{N}{m^2}$ or Pa.

Sound Pressure Level (SPL) is defined as:

$$SPL = 20 \log \left(\frac{p}{p_{ref}} \right) \tag{1}$$

Where p is the sound pressure and p_{ref} is the reference pressure value taken as $2 \times 10^{-5} \frac{N}{m^2}$ [4].

In hearing aids, sound pressure is measured by a microphone and converted to an electric signal [5]. These microphones also apply a filter to the sound called A-weighting [4]. Since the human ear does not perceive all frequencies equally, it is less sensitive to lower and higher frequencies. Thus, the A-weighting mimics the frequency response of the human ear. This frequency response can be seen in Figure 6.

For our system, we decided to measure the root mean square (RMS) sound pressure level. So, in (1), instead of p we use $rms(p)$. The block diagram of the sound pressure level meter can be seen in Figure 2.

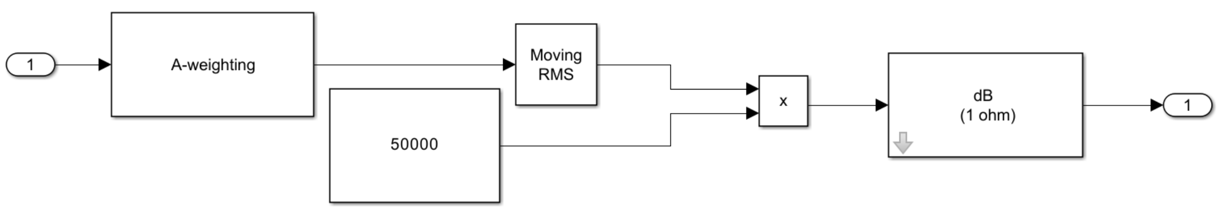


Figure 2: SPL measurement block diagram. At the beginning, the input goes through an A-weighting filter. Then, the output is fed to a moving RMS block which calculates the RMS of its input in the past second. After that, sound pressure level is calculated based on (1).

Now that we have a system for measuring the sound pressure level, we can proceed with our design. As illustrated in Figure 1, a single channel of a multichannel hearing aid consists of three regions: Linear, Compression, and Limiting. To implement our system, we used three blocks as shown in Figure 3. The compressor and limiter blocks are predefined blocks from the Audio Toolbox of Simulink. The third block is the dB Gain block from the DSP System Toolbox.

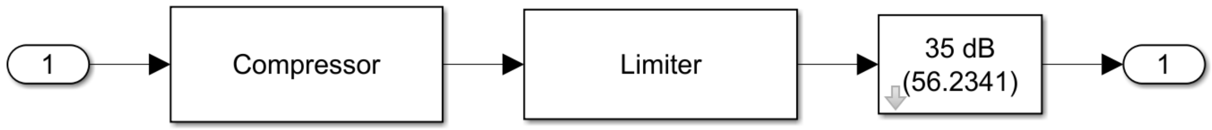


Figure 3: Block diagram of the single-channel compression system. The input gets compressed after the compressor block. The limiter block ensures the level of its input does not exceed certain level. At the end, a constant gain is applied.

2) Compressor

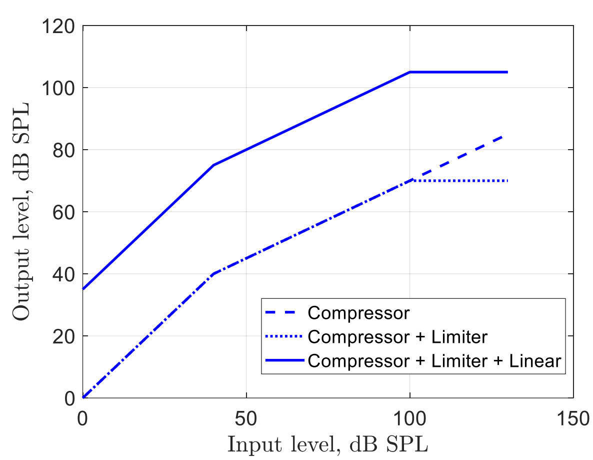
The compressor block requires two parameters to function: Threshold (dB) and Ratio. The threshold is the input level above which the compression starts. The ratio is the inverse of the slope of the curve in the compression region. For example, if the slope is 0.5, the ratio would be 2. According to Figure 4, we set the threshold to 40 dB SPL and the ratio to 2.

3) Limiter

The limiter block requires one parameter to function: Threshold (dB). The threshold is the input level above which the limiting starts. According to Figure 4, we set the threshold to 70 dB SPL. Thus, the output level in the dotted-line curve never exceeds 70 dB SPL.

4) Linear

So far, the characteristic of our system is the dotted-line curve in Figure 4. To meet the specifications described in Figure 1, all we have to do is elevate our curve by 35 dB. This is done by adding a linear 35 dB gain as the final block. The final characteristic of the system would be the solid-line curve in.



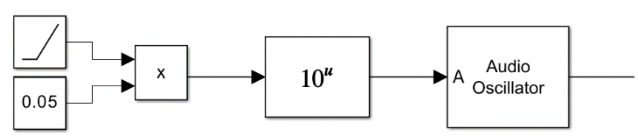


Figure 4: Step by step characteristic of system blocks. (left) The design procedure starts from the compressor block. By applying the appropriate threshold and ratio, the dashed-line curve can be implemented. Next, the limiter block is applied to the output of compressor (dotted-line curve). At the end, a constant gain is applied to elevate the curve so it would match the specifications defined in Figure 1. Block diagram of input signal generator. (right) A ramp function

starting from -90 dB SPL with slope of 1 is used. Then the output of the ramp is converted from dB to amplitude and fed to the audio oscillator. The audio oscillator generates a 1 KHz sine wave with amplitude defined from its input port.

Testing And Simulation Results

To verify the performance of our system blocks, we use the block diagram in Figure 4 to generate the input signal to our single channel compressor in Figure 3. In Figure 4, the audio oscillator block generates a 1 KHz sine wave.

The amplitude of the sine wave is specified from the input port. In Figure 4, the output of a ramp function is converted from dB to amplitude and fed to the audio oscillator. Figure 5 demonstrates how the amplitude of the input signal is increased with time.

Next, we feed our generated signal to our single-channel compressor. We use the SPL measurement system defined in Figure 2 to measure the SPL of the signal before and after compression. The input sound pressure level starts from 0 dB SPL and increments until it reaches 120 dB SPL. This way, we will have an adequate number of samples in a suitable range for our simulation results. Figure 5 shows the sound pressure level of the signal before and after the single-channel compression. As we can see, a linear gain of 35 dB is applied to the signal for the first 0.66 minutes since the SPL of input is below 40 dB SPL. At minute 0.66, the compression starts and continues for a minute. At minute 1.66, the SPL of input reaches 100 dB SPL and the limiting starts.

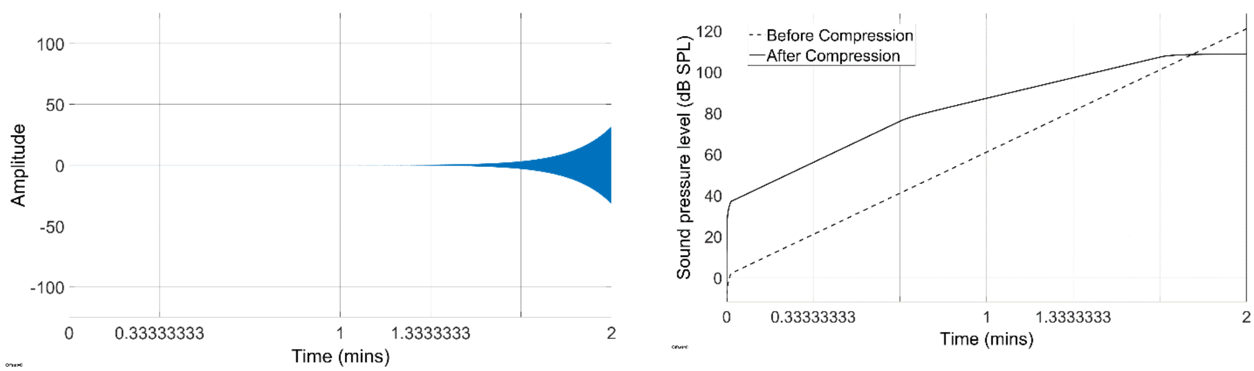
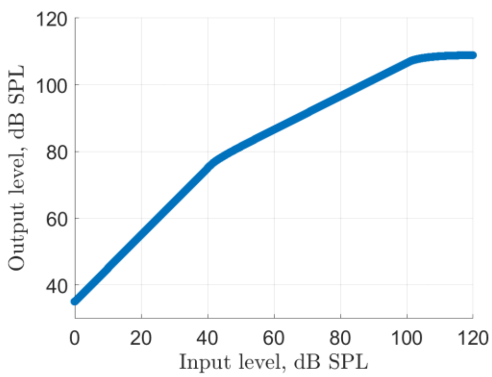


Figure 5: Amplitude of input sine wave vs. time. (left) Simulation result demonstrating SPL of the signal before and after compression vs. time. (right)

Finally, Figure 6 shows the characteristics of the single-channel compressor that we implemented. As we can see, the pattern shown in Figure 6 completely matches the specifications described in Figure 1.



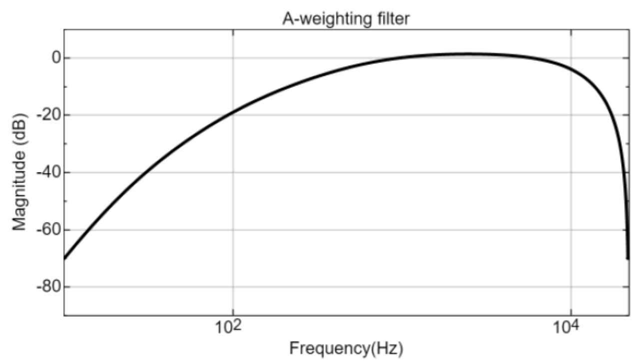


Figure 6: Simulation result for output level vs. input level in dB SPL for a channel with the center frequency of 1 KHz. (left) The curve is a scatter plot of input-output level sample pairs. Frequency response of A-weighting filter. (right) As we can see, the frequency response is almost constant from 1 KHz to 5 KHz. Thus, our implemented single-channel compressor for the channel centered at 1 KHz can also be used for any other channel in the 1 KHz to 5 KHz range. No parameter needs to be changed and the same specifications described in Figure 1 can be achieved.

Conclusion

In this section, we designed, tested, and analyzed a single-channel compressor for a particular frequency channel. To design a multichannel compressor, multiple single-channel compressors would be required. The analysis would be exactly the same for different channels only with different parameter values. This is mainly because a different set of specifications might be required in different channels.

References:

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