Hearing AID System

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Abstract

Hearing impairment is a condition that affects a significant portion of the worldwide population, and it is characterised by a noticeable difference in the auditory system's perceptive capacities when compared to those of a typical ear. The current progress in digital signal processing (DSP) has led to the emergence of advanced artificial hearing aid devices, designed to improve the auditory impairments faced by those with damaged hearing. The objective of this research is to outline the process of designing and simulating a Digital Hearing Aid (DHA) system using the MATLAB programming language. The articulated DHA system has essential elements, such as a noise reduction filter, a frequency shaping function, and an amplitude compression function, all designed to enhance auditory perception. A comprehensive evaluation has been undertaken on a simulated patient model to appraise the effectiveness and operational capabilities of the proposed DHA system. Moreover, this scholarly article provides a prospective viewpoint on nascent patterns and prospective advancements in the domain of hearing aid technology, consequently advancing the discussion towards forthcoming developments in the field of auditory rehabilitation.The present study focuses on the topic of hearing loss and its potential remedy through the utilisation of Digital Hearing Aids (DHAs). DHAs employ Digital Signal Processing (DSP) techniques, which involve the use of computational algorithms to enhance auditory perception. The programming of DHAs is often carried out using software such as MATLAB. One of the key functions of DHAs is the implementation of a noise reduction filter, which aims to minimise the impact of background noise on the user's auditory experience. Additionally, DHAs incorporate a frequency shaper function that modifies the frequency response to optimise speech intelligibility. Another important feature of DHAs is the amplitude compression function, which adjusts the amplification of different sound intensities to ensure a comfortable listening experience. The ultimate goal of DHAs is to facilitate auditory rehabilitation and improve auditory perception for individuals with hearing loss.

1 Introduction

The acquisition of a hearing aid is a matter of utmost importance for individuals. These devices are small electronic gadgets designed to amplify sound and enhance speech intelligibility. The device is designed to detect and capture acoustic vibrations using a transducer, amplify attenuated signals, and transmit them audibly through a transducer. The reduction in size and improvement in quality of hearing aids may be attributed to the advancements in microprocessor technology that are now available. Hearing loss affects around 10 percentage of the global population.[KMB04a]

Nevertheless, the use of hearing aids is limited to a minority of individuals within the community. This phenomenon may be attributed to several factors, including the negative societal perception surrounding the use of hearing aids, client dissatisfaction, and the exorbitant cost of contemporary hearing aid models. Hearing loss is often measured by assessing the difference in auditory threshold for detecting a pure tone in comparison to that of a typical ear. Hearing aids are available in many forms and dimensions, including a range of functionalities and attributes that cater to a wide array of needs and requirements. The primary objective of a hearing aid is to amplify auditory information in order to facilitate perception by those with hearing impairments. Table 1 displays the various degrees of hearing loss experienced by individuals. Typically, analogue technology is used for sound processing in all hearing aids. The efficiency of hearing aids has been enhanced as a result of the introduction of digital sound processing technology.[KMB04b]

Level	Different Degrees of Hearing Loss		
	Classification	Lower limit	Upper limit
1	Normal hearing	-10 dB	26 dB
2	Mild hearing loss	27 dB	40 dB
3	Moderate hearing loss	40 dB	70 dB
4	Severe hearing loss	70 dB	90 dB
5	Profound hearing loss	90 dB	-

Table 1: Different Degrees of Hearing Loss[2]

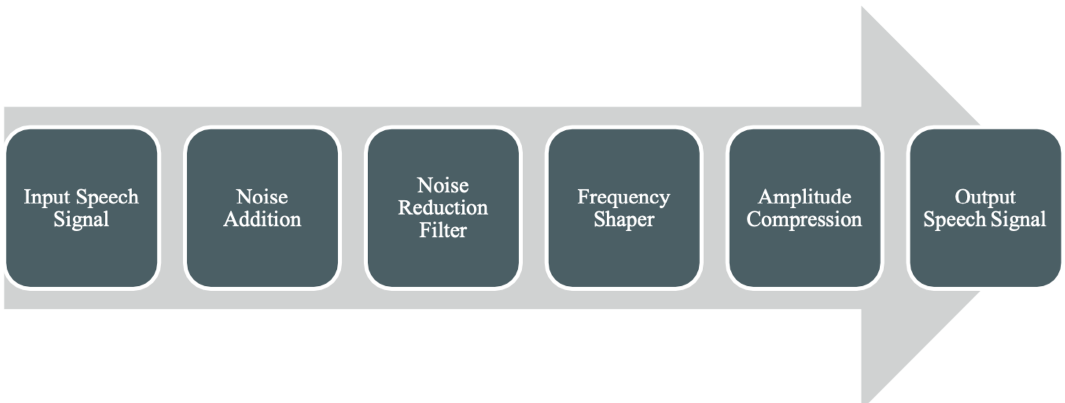


Figure 1: Block diagram of the Digital Hearing Aid system

2 Methodology

Figure 1 is the block diagram illustrating the MATLAB implementation of the Digital Hearing Aid System. The input speech signal consists of the human voice. Prior to generating a modified output speech signal that is perceptible to those with hearing impairments, the input speech signal will undergo a series of processing operations including noise addition, noise reduction filtering, frequency shaping, and amplitude compression.

2.1 Noise Addition

In order to simulate a realistic scenario, noise is intentionally injected into the speech signal used by this system, since it is inherently clean. The system used MATLAB to include Additive White Gaussian Noise (AWGN) and random noise into the input voice stream. The additive white Gaussian noise (AWGN) exhibits a frequency spectrum that is continuous and uniformly distributed throughout a certain frequency range, with equal power allocation per Hertz.[BS08] The probability density function of the distribution is Gaussian, and all frequencies exhibit similar intensity.

2.2 Noise Reduction Filter

The incorporation of a reduction filter feature signifies a pivotal advancement in tackling the primary challenge encountered by those contending with auditory impairment. In settings characterised by excessive noise, the ability to accurately perceive intended spoken communication may provide a difficult challenge, further exacerbating the everyday obstacles encountered by those with hearing disabilities.

In this novel architecture, the wavelet filter function assumes a prominent role, serving as an effective tool for extracting significant speech signals from a complex background noise environment. The advanced filtering approach demonstrates proficiency in the isolation and amplification of frequencies related to human voice, hence providing a more enhanced and discernible auditory encounter.

The successful incorporation of this noise reduction mechanism serves as evidence of the persistent endeavour to improve the overall well-being of those with hearing impairment. By integrating advanced technology and using empathic design principles, this system aims to enhance the autonomy of humans, enabling them to interact with the surrounding environment in a more seamless and significant manner. By engaging in this practise, individuals not only enhance their hearing perception but also cultivate a deep feeling of inclusiveness and interconnectedness.

2.3 Frequency Sharper

Individuals who use hearing aids sometimes express dissatisfaction with the fact that their devices indiscriminately enhance all auditory signals, rather than selectively amplifying the specific signal of interest. The majority of individuals with hearing impairments find it challenging to perceive high-frequency noises. Consequently, the frequency sharper is designed to mitigate the effects of auditory impairment at certain frequencies. The device employs a high gain setting to amplify higher frequencies and a low gain setting to amplify lower frequencies.[KMB04b]

2.4 AmplitudeCompression

Amplitude compression is a crucial factor in ensuring the efficient functioning of a voice amplification system. By implementing gain regulation, it guarantees that the amplified signal stays within the limitations of the system's capability. The purpose of this measure is to mitigate any adverse consequences, such as signal distortion or clipping, that might arise from beyond the saturation power level. The primary aim of the amplitude compression function is to serve as a protective measure, ensuring that the audio output remains clear and constant, particularly in scenarios where the strength of the input signal may vary. The aforementioned crucial feature enables the voice amplification system to effectively provide consistent and comprehensible sound reinforcement in various situations and circumstances.

3 Psuedo Code

- The input voice signal is sent to the MATLAB model.
- In order to imitate real-life scenarios, it is necessary to introduce noise into the input voice signal, since the system under consideration assumes a clean signal as its input.
- The Wavelet filter is used in order to mitigate the presence of noise. The frequency shaper is employed to rectify the auditory deficiency pertaining to certain frequencies.
- Amplitude compression is used as a means to enhance the signal's gain.

4 Implementation Simulation

The execution of the MATLAB code starts with the loading of the input speech signal, which is an essential stage in its preparation for further processing. The aforementioned signal, which serves the purpose of conveying the desired message to those with hearing impairments, has the potential to significantly improve their auditory perception. As the code advances, it systematically computes the sample frequency, which is an essential measure for precise processing. This guarantees that the succeeding algorithms function at the suitable pace, according to the attributes of the initial signal.

The last step involves the precise enumeration of individual bits included within the signal. The numerical evaluation presented herein offers crucial insights into the digital portrayal of audio, facilitating the implementation of accurate modifications. After acquiring this fundamental understanding, the signal is then enhanced by the inclusion of Additive White Gaussian Noise (AWGN), a method used to replicate the probabilistic characteristics of real-life settings. The use of this purposeful augmentation enhances the individual's auditory experience by creating a more authentic and engaging environment.

The trajectory of the vocal signal persists while it crosses a sequence of advanced MATLAB algorithms, each meticulously crafted to address the distinct requirements of individuals with hearing impairments. These algorithms function in synergy to convert the processed input into an audio output that is meticulously adjusted to match the individual's own auditory profile.This specific section enables comprehensive assessment and enhancement, hence assuring the attainment of ideal outcomes.[HATK13]

In the quest for achieving optimal results, the use of Additive White Gaussian Noise (AWGN) assumes a significant role by enhancing the selected segment to more accurately replicate real-world circumstances. This particular stage, which is of utmost importance in ensuring correctness, serves as evidence of the meticulousness inherent in this procedure.

In order to effectively end this thorough presentation, it is essential to include all the necessary parameters. These factors include the optimisation of the gain parameter to ensure that the output is customised to meet the unique requirements of the person. Moreover, the calibration of the saturation power, which is a crucial element in mitigating distortion, requires meticulous attention. Ultimately, a total of four discrete frequency values are discerned, whereby intentional fluctuations in gain occur. The degree of personalization guarantees that the resulting output is precisely adjusted to accommodate the distinct auditory profile of the individual.

Given the particular state of the patient, characterised by moderate hearing loss, it is necessary to examine additional factors. The threshold of pain, which is established at 50dB, serves as a pivotal point of reference. This parameter is designed to maintain the amplified signal within a range that is comfortable for the user, therefore mitigating any possible pain or injury. Moreover, it has been shown that elevated frequencies provide a specific difficulty for the individual's auditory perception. This essential perspective informs the meticulous adjustment procedure, guaranteeing that these frequencies are approached with the highest level of attention and accuracy.

5 MATLAB Code

```
clc;
clear all;
close all;
input = 'sinewave.wav';
[y,fs] = audioread(input);
figure;
plot(y);
title('Input signal');
xlabel('samples');
ylabel('amplitude');
disp('Original signal')
sound(y,fs);
figure;
freqz(y)
title('Frequency spectrum of input signal')

% Adding Noise
y = awgn(y,40);
noise = y;
figure,plot(y);
xlabel('Samples');
ylabel('Amplitude');
title('Signal + AWGN');
disp('Added noise');
sound(y,fs);
pause(10)
figure;
freqz(y)
title('Frequency spectrum of noisy signal')

% Removing Noise
%'Fp,Fst,Ap,Ast'(passband frequency, stopband frequency, passband ripple, stopband attenuation)
hlpf = fdesign.lowpass('Fp,Fst,Ap,Ast',3.0e3,3.5e3,0.5,50,fs);
D = design(hlpf);
```

```
freqz(D);
x = filter(D,y);
disp('Denoised sound');
figure,plot(x);
title('Denoise');
sound(x,fs);
pause(10)
xlabel('Samples');
ylabel('Amplitude');
figure;
freqz(x)
title('Frequency spectrum of denoised signal')

% Frequency Shaper
T = 1/fs;
len = length(x);
p = log2(len);
p = ceil(p);
N1 = 2^p;
f1 = fdesign.bandpass('Fst1,Fp1,Fp2,Fst2,Ast1,Ap,Ast2',2000,3000,4000,5000,60,2,60,2*fs);
hd = design(f1,'equiripple');
y = filter(hd,x);
freqz(hd);
y = y*100;
disp('Frequency shaped');
sound(y,fs);
pause(10);
figure;
plot(y);
xlabel('Samples');
ylabel('Amplitude');
title('Frequency shaped signal')
figure;
freqz(y)
title('Frequency spectrum of frequency shaped signal')

% Amplitude Shaper
disp('Amplitude shaper')
out1=fft(y);
phse=angle(out1);
mag=abs(out1)/N1;
size(mag);
[magsig,~]=size(mag);
threshold_dB=40;
saturation_dB=60;
threshold=10^(threshold_dB/10);
saturation=10^(saturation_dB/10);
out=zeros(magsig,1);
T=min(mag);
T_dB=10*log10(T);
for i=1:magsig/2
    if(mag(i)>saturation)
       mag(i)=saturation;mag(magsig-i)=saturation;
    end
    out(i)=mag(i)*exp(j*phse(i));
    out(magsig-i)=out(i);
end
```

```
outfinal=real(ifft(out))*10000;
disp('Amplitude shaped');
sound(outfinal,fs)
pause(10);
figure;
plot(outfinal);
xlabel('Samples');
ylabel('Amplitude');
title('Amplitude shaped signal')
figure;
freqz(outfinal)
title('Frequency spectrum of Amplitude shaped signal')
figure;
subplot(2,1,1);
specgram(noise);
title('Spectrogram of Original Signal');
subplot(2,1,2);
specgram(outfinal);
title('Spectrogram of Adjusted Signal')
```

6 Result

The implementation of this notion in MATLAB involves the first step of giving an input voice signal. Figure 2 displays the temporal response of the input signal, while Figure 3 illustrates its frequency response.

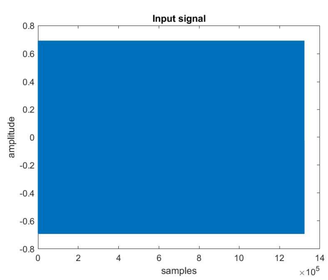


Figure 2: Time response of Input speech signal

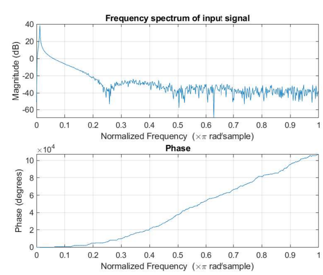
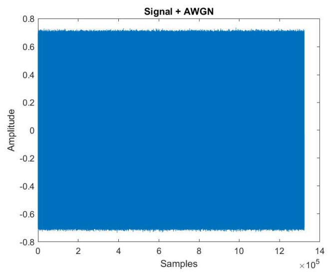


Figure 3: Frequency response of input speech signal

In order to examine this scenario as a practical issue, Gaussian noise with a power level of 40dB is introduced to the input voice.

The voice signal exhibits its highest strength within the frequency range of up to 3.5kHz. To eliminate sounds beyond this range, a low pass filter is used, with a stopband frequency set at 3.5kHz. To some degree, the removal of high-frequency noise is achieved using this process.



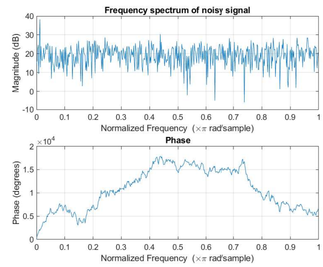
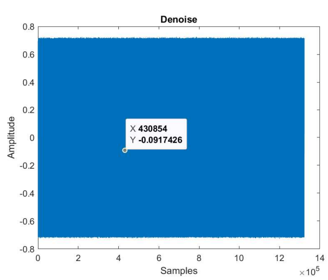


Figure 4: Time response of input + AWGN signal

Figure 5: Frequency response of input + AWGN signal

The voice signal exhibits its highest strength within the frequency range of 0 to 3.5kHz. To eliminate sounds beyond this range, a low pass filter is used, with a stopband frequency set at 3.5kHz. To a certain degree, the removal of high-frequency noise is achieved by this process.



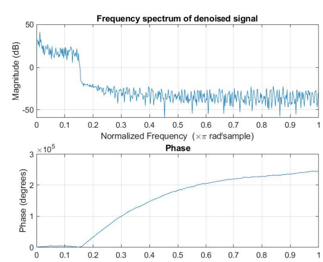


Figure 6: Time response of Denoised signal

Figure 7: Frequency response of Denoised signal

Individuals with hearing impairments sometimes struggle to perceive high-frequency sounds. To address this issue, it becomes necessary to amplify the strength of high-frequency signals. Additionally, it is worth noting that the majority of relevant speech information is contained in frequencies above 2000Hz. Consequently, a bandpass filter is used to allow just these frequencies to pass through.

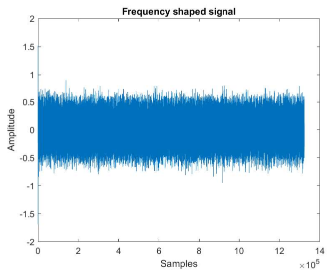


Figure 8: Time response of Frequency shaped signal

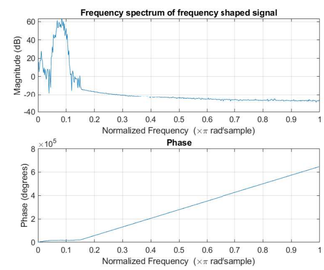
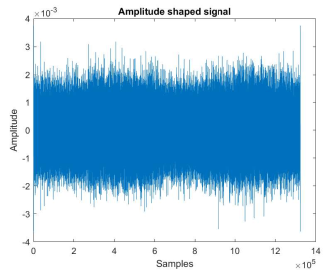


Figure 9: Frequency response of Frequency shaped signal

The process of amplitude shaping is conducted to decrease the power of frequencies that exceed the saturation point. The saturation value is 60 decibels (dB).



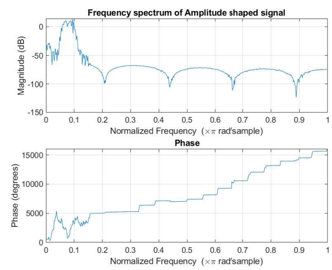


Figure 10: Time response of Frequency shaped signal

Figure 11: Frequency response of Frequency shaped signal

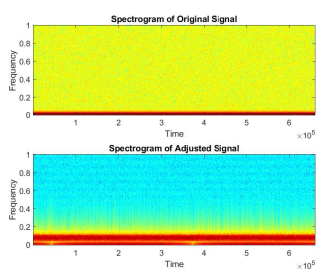


figure 12:Spectrogram of original and output signal

Figure 12 displays the spectrograme of both the original and output voice samples.

7 Conclusion

The most recent advancements in digital hearing aids demonstrate exceptional capabilities in enhancing sound while maintaining its integrity. The present implementation uses MATLAB for the purpose of digitising sound processing inside a digital hearing aid system. As a result, this technology facilitates the amplification of vocal impulses while simultaneously reducing extraneous background noise. Furthermore, the use of digital technology enables the exact enhancement of certain frequencies that are customised to the user's own preferences. This statement pertains to the limitations of conventional amplifiers, which amplify all signals without discrimination, including undesired noise. Essentially, a digital hearing aid is designed to transform incoming signals into a digital format, so enabling precise analysis and filtering of signals across several frequency channels. In the end, the digital signal undergoes a process of reconversion to its analogue condition. The use of digital aids has several advantages, with the most notable being an improvement in auditory fidelity, hence positively impacting an individual's overall well-being.

References

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