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MOBILE APPLICATION FOR HEARING IMPAIRMENT PEOPLE

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Abstract - Applications that turn mobile computing platforms into hearing aides are known as hearing aid applications (HAAs). The fundamental operating concept of hearing aid applications is the same as the fundamental operating principle of conventional hearing aids: the microphone picks up an acoustic signal and converts it to a digital form. Conventional hearing aids come with a number of limitations, including price, size, and capability. This study suggests a smart phone app that would record sound using the microphone and then process it using digital signal processing methods. The smart phone's speaker or a tethered or wireless headphone will then play back the processed audio.

Keywords - Hearing Aid Application (HAA), User Interface, Graphical User Inetrface.

I. INTRODUCTION

A prevalent issue affecting millions of individuals globally is hearing loss. This issue has a solution in hearing aids. Traditional hearing aids, however, have a number of disadvantages, including price, size, and capability limitations. The integrated development environment for Android apps, Unity, will be used to create the hearing aid application. The programme will record sound using the smart phone's microphone and then manipulate it using techniques for digital signal processing. The smart phone's speaker or a tethered or wireless headphone will then play back the processed audio. We will carry out a number of subjective and objective tests to gauge how well the hearing aid application is working. Users with hearing loss will be asked to use the hearing aid application and provide comments on how well it works as part of the subjective test. Using standardised test signals, the objective test will gauge the application's frequency response, noise reduction, and distortion. For various Android smart phones, the hearing aid app was created and put through testing. The software delivered good amplification, noise reduction, and voice intelligibility results. The findings of the arbitrary test indicated that users thought the programme was simple to use and helpful in enhancing their hearing. The application's frequency response was within the permitted range, and the objective test results demonstrated that it offered sufficient noise and distortion reduction.

Physical hearing aids are tools that those who have hearing loss wear in or behind the ear to increase sound. In addition to helping millions of individuals enhance their hearing, these devices have been around for a long time and do have some drawbacks. Physical hearing aids can cost hundreds to thousands of dollars, making them an expensive option. Those who cannot afford to buy them may find this price to be a barrier, especiallythose with little financial resources. Many physical hearing aids are noticeable and large, which can cause wearers to feel self- conscious and uneasy when using them in public. For younger people, who might feel stigmatised or ashamed by donning a conspicuous hearing aid, this might be especially difficult.Regularmaintenance for physical hearing aids includes cleaning, battery replacement, and audiologist adjustment as needed. For people whoreside in locations with a dearth of audiology services, thiscan be time-consuming and expensive. Hearing aids that are worn physically may cause feedback or distortion, especially in noisy settings or when the device is not fitted properly. Users may find it challenging to interpret speech in this situation, which can be unpleasant and disconcerting. There is a chance that all audio sources, including smart phones and other digital devices, won't work with traditional hearing aids. As a result, their utility may be limited, and users may find it difficult to maintain relationships with friends and family. Despite certain drawbacks, physical hearing aids are nevertheless a valuable and practical option for many people with hearing damage. Additionally, new technologies are developing, such as implantable devices and hearing aids that can be used with smart phones, which may provide alternatives with less restrictions.The usage of smart phones makes it uncomplicated to evaluate several algorithms for every hearing aid module. A thorough analysis of the user requirements, programming languages, software development environments, and testing and optimisation methods is necessary for the implementation and optimisation of a mobile hearing aid application. The signal processing algorithms utilised in the application, as well as

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the noise and distortion levels as well as the frequency response, must all be optimised for a hearing aid mobile application. The user interface can be improved to make it simpler to use and more accessible for people who have hearing loss. Also, by reducing CPU utilisation and utilising power-saving capabilities, the programme can be made to run more efficiently on batteries.

II. RELATED WORKS

The size, price, and functional limitations of conventional hearing aids are only a few of their flaws. Using Unity, an integrated development environment for Android apps, the hearing aid application will be created. The programme will record sound using the smart phone's microphone and then process it using digital signal processing methods. The smart phone's speaker, a wired or wireless headphone, or both will play back the processed sound after that. [1] The adaptive noise reduction module is a crucial part of contemporary hearing aid technology. By reducing background noise, it is intended to enhance speech recognition in noisy contexts. Adaptive noise reduction algorithms are often implemented in the hearing aid's signal processing unit and are based on digital signal processing techniques. The adaptive noise reduction module operates by constantly assessing the environment and modifying the level of noise reduction in accordance with the current noise level. [2] An essential part of contemporary hearing aid systems is the compression module. It is made to compress loud noises and enhance delicate sounds, expanding the range of dynamic sound that a hearing aid can process. When a sound signal is supplied, the compression module first analyses it and then applies a gain to the signal based on its amplitude. [3] One of the most crucial parts of a hearing aid system is the amplification module. It is in charge of boosting the input sound signal to an audible volume for the hearing-impaired user. The amplification module increases the amplitude of the incoming sound signal in accordance with the user's hearing loss and preferences.

A prevalent issue affecting millions of individuals globally is hearing loss. The programme will record sound using the smart phone's microphone and then manipulate it using techniques for digital signal processing. The usage of smart phones makes it smooth to evaluate several algorithms for every hearing aid module. Strong ARM processors found in smart phones make it possible to implement computationally demanding signal processing techniques in real-time. The majority of people currently own and use smart phones, thereby making them a free mobile platform. Smart phone manufacturers provide free software development tools that are well-maintained. The digital signal processing (DSP) activities required for creating algorithms for hearing impairment are carried out using the C programming language. In hearing aid systems, DSP is used to analyse audio signals in order to minimise background noise, increase speech intelligibility, and digitally filter out unwanted sounds. The fast Fourier transform (FFT) technique, which is used to examine the frequency content of audio signals, is one example of an algorithm for processing audio data that was created using the C programming language. The buttons, sliders, and touch displays that make up the user interface of hearing aids were created using the C programming language. The user interface of the hearing aid device enables users to customise the volume, tone, and other parameters to suit their unique requirements. Due to its potent signal processing capabilities and integrated functions for analysing and manipulating audio data, MATLAB is a well-liked programming language and toolset utilised for constructing hearing enhancement methods. For creating hearing enhancement algorithms, MATLAB offers a large variety of signal processing tools. Filtering, equalisation, dynamic range compression, and noise reduction are some of these operations. Digital signal processing (DSP) activities required for creating algorithms for hearing enhancement are performed using MATLAB. DSP involves modifying audio signals to reduce background noise, improve voice quality, and carry out other audio processing tasks. Algorithms for hearing enhancement can be optimised using MATLAB's parallel processing, GPU acceleration, and compiler optimisation tools. A wide variety of hardware and operating systems are supported by MATLAB. The Objective-C based software development environment for iOS mobile devices includes a number of elements that enable programmers to produce top-notchapps for Apple's mobile operating system. Xcode is the name of the Objective-C software engineering environment for iOS mobile devices. With iOS, macOS, watchOS, and tvOS, developers can create apps using this integrated development environment (IDE). When creating Android applications, Java is typically utilised as the programming language. To create desktop, mobile, and web apps, developers frequently utilise Java, an object-oriented language. Among the characteristics offered by Java are platform independence, garbage collection, and exception handling.

III. PROPOSED WORK

A hearing aid device is desired by those who have hearing loss in order to hear their surroundings. With the help of their own earbuds, we suggest they use an application that can run on an Android phone to hear the sounds around them. Software that is placed on mobile computational platforms and turns them into hearing aids is known as a hearing aid application (HAA). The fundamental operating principles of conventional hearing aids are the same as those of hearing aid applications, the microphone receives an acoustic signal and translates it into a digital form. According to the kind and severity of the user's hearing loss, sound is amplified using a mobile computing platform.

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There are specified prerequisites for the mobile hearing aid application. This entails comprehending the intended user base, determining the necessary features, and specifying the performance metrics. The hearing aid mobile application's design is made based on the specifications. This comprises defining the app architecture, coming up with a feature list, and designing the user interface. Create a mobile application with the features you need and a user-friendly UI. Those with hearing difficulties should be able to use the design with ease. Moreover, controls for volume, frequency response, noise reduction, and other parameters should be available in the mobile application. To make sure it satisfies the specified performance metrics and performs as intended, the mobile hearing aid application is tested. The testing encompasses user acceptance testing, functional testing and performance testing.

A. BLOCK DIAGRAM

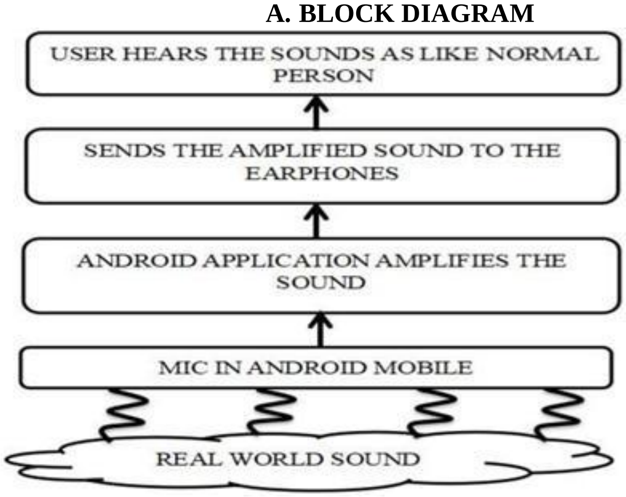

Figure 1. Block diagram

A digital signal is created from an acoustic signal that is received by the mobile device's microphone. In order to be output to the user's headphones, the processed audio signal is converted into an audio signal.

B. PROPOSED MODULE

The proposed work is divided into three modules, each the block represents the various steps involved in the hearing aid application.

I. Adaptive noise reduction.
II. Compression.
III. Amplification.

I. Adaptive noise reduction

The adaptive noise reduction module learns the characteristics of various noise situations and adapts to them using machine learning methods. It may be taught to distinguish between various noises, such as voice noise, wind noise, and traffic noise, and to change the noise reduction parameters accordingly. Many variables, including the calibre of the microphone, the precision of the noise estimation method, and the complexity of the noise reduction and speech enhancement filters, affect how well the adaptive noise reduction module performs. The processing power and battery life of the hearing aid can also have an impact on the performance of the module. An integral part of contemporary hearing aid systems is the adaptive noise reduction module. By utilising digital signal processing techniques to reduce background noise, it enhances voice recognition in noisy conditions. According to the current noise level, the module continuously analyses the environment and modifies the noise reduction parameters. The module's performance is dependent on a number of aspects and can be improved using signal processing and machine learning approaches. There are three primary steps to the adaptive noise reduction module.

- The module determines the level of background noise during the noise estimation stage. Analyzing the audio signal that the hearing aid's microphone has picked up allows for this. The module calculates the amount of voice and background noise using statistical methods.
- The module lowers the amount of background noise in the audio signal during the noise reduction step. A filtering operation used in the noise reduction procedure subtracts the estimated noise from the audio signal. The filter's goal is to lessen noise while keeping the voice signal intact.
- The module improves the speech signal during the speech enhancement stage. Applying a gain to the voice signal makes up for the signal loss brought on by the noise reduction procedure is how this is accomplished. Taking into account both the present noise level and the anticipated speech level, the gain is adjusted.

II. Compression

An essential part of contemporary hearing aid systems is the compression module. In order to increase the entire dynamic range of sound that a hearing aid might comprehend, it is made to amplify quiet sounds and compress loud noises. When a sound signal is supplied, the compression module first analyses it and then applies a gain to the signal based on its amplitude. Algorithms for linear, nonlinear, and frequency-specific compression are only a few of the various types utilised in hearing aids. Whereas nonlinear compression offers a more complicated gain control that depends on the amplitude of the input signal, linear compression offers a set gain to the input signal. The hearing aid may compress sounds at a chosen frequency while keeping other frequencies unchanged thanks to frequency-specific compression. By modifying the compression parameters to take into account the user's hearing loss and listening preferences, the performance of the compression module can be improved. The performance of the compression module in contemporary hearing aids may also be adjusted to the user's listening environment using machine learning techniques. An essential part of contemporary hearing aid systems is the compression module. It increases the dynamic range of sound that a hearing aid can process overall by compressing loud noises and amplification of gentle sounds. The module's performance may be optimised for various forms of sound using different compression methods, and via the application of machine learning algorithms, the performance can be tailored to the user's hearing loss and listening preferences. There are two basic steps to the compression module.

- The module measures the input sound signal's amplitude during the amplitude detection step. For this, the incoming signal is examined, and its decibel level is measured. To decide when the signal should be compressed, the module can be programmed with various thresholds.
- The module uses the input signal's amplitude to determine the gain it applies to the signal in the gain control stage. Loud noises are compressed while soft noises are amplified. To tailor the performance of the module for various sound types, the gain control can be programmed with various compression ratios, attack and release times, and knee points.

III. Amplification

One of a hearing aid system's most crucial parts is the amplification module. It is in charge of boosting the sound signal input to a volume that the hearing-impaired user can hear. The amplification module increases the amplitude of the incoming sound signal in accordance with the user's hearing deficiency. Other functions including feedback cancellation, directional microphones, and adaptive signal processing may also be included in the amplification module of contemporary hearing aids. Once that amplified sound is picked up by microphone and re- amplified, feedback or whistling noises may develop. These disturbances can be avoided by using feedback cancellation. By focusing on noises originating from in front of the user, directional microphones help speech be understood more clearly in busy circumstances. The performance of the amplification module is modified via adaptive signal processing in accordance with the user's listening situation. Gain settings, equalisation parameters, and other settings can be modified to match user's hearing in order to improve the performance of the amplification module. Moreover, contemporary hearing aids might employ machine learning algorithms that modify the performance of the amplification module over time in response to a user's listening environment as well as preferences. Advanced assistive listening systems must have the amplification module. According on the user's hearing loss and preferences, it enhances the incoming sound signal and may also include extra functions like feedback cancellation, directional microphone as well as adaptive signal processing. By modifying the gain settings, equalisation parameters, other variables, the efficiency of the amplifier module can be improved. Machine learning methods may also be used to adapt the performance over time.The amplifier module can be broken down into different stages.

- The incoming sound signal gets preprocessed at the preprocessing stage to reduce unnecessary sound and improve the signal-to- noise ratio. This can involve filtration, noise cancellation as well as other signal processing methods.
- The level of overall amplification of the incoming sound signal is chosen in the gain control stage. Several gain settings for various frequency bands can be put into the gain control, enabling the hearing aid that account for various forms of hearing loss.
- The equalisation stage is in charge of adjusting the incoming sound signal's frequency response. By doing this, it is made sure that the signal's amplification matches the user's normal hearing response.
- To stop the hearing aid from creating noises that are excessively loud or unpleasant for the user, the output limiting step is used. To restrict the output for loud noises, this may involve setting a maximal output level or utilising compression.

IV. RESULT AND DISCUSSION

Anyone with hearing loss desire a hearing aid equipment so they can hear their environment. We advise them to use an app that can operate on an Android smart phone to perceive what's happening around them while using their own earbuds. An application for hearing aids is a piece of software that is installed on portable computing devices to make those into hearing aids (HAA). Conventional hearing aids operate on the same basic principles as hearing aid applications; a microphone picks up an acoustic signal and converts it to a digital format. Sound is boosted utilising a mobile computing environment depending on the type and degree of the user's hearing impairment.

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Figure 2. Live Play and Stop Activity

Figure 3. Live Play and Stop with Amplification Activity

Figure 4. Record and Play Activity

Traditional hearing aids have a number of drawbacks, including restrictions on cost, size, and functionality. An acoustic signal is picked up by the smart phone device's microphone and converted to a digital signal. The processed acoustic signal is transformed into an audio signal and supplied to the user's headphones.

V. CONCLUSION

This study proposes a smart phone application that will capture sound with the microphone and afterwards edit it by using the digital signal processing techniques. Hearing aid programmes may be able to distinguish various speech varieties in various settings by integrating more sophisticated machine learning algorithms. For those who have hearing loss, the assistive listening application is a great tool, and there's a lot of scope for future upgrades to increase its performance and usefulness.

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