RESEARCH PAPER

# AI Based Smart Hearing Aid for Improved Speech Intelligibility and Tinnitus Masking

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## ABSTRACT

People with hearing problems are unable to differentiate speech from noise, especially in noisy environments. The reason for this is that currently available hearing aids amplify both speech and background noise. Tinnitus is another form of high-pitched noise such as ringing, buzzing and hissing, which can be prevented by masking. This project proposes a deep learning based digital hearing aid system that enhances speech clarity by removing unwanted noise and also reduces tinnitus by masking it using a neural network model. The system converts audio signals into spectrogram features and applies a deep learning model such as an autoencoder or CNN to separate speech from noise. A controlled masking sound is added to reduce tinnitus discomfort. Performance is evaluated using SNR, MSE, and PSNR metrics, demonstrating improved speech intelligibility and listening comfort.

**Keywords:** Hearing aid, speech enhancement, tinnitus relief, CNN, SNR

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## *I. INTRODUCTION*

Hearing impairment is one of the most common sensory disorders worldwide, affecting millions of people of all ages. Individuals with hearing loss frequently fail to perceive speech effectively, especially in noisy settings like packed streets, classrooms, offices, and public meetings. Conventional hearing aids increase incoming sound signals without efficiently separating speech from background noise. As a result, both speech and undesirable noise are amplified concurrently, reducing listening comfort and speech intelligibility. In addition to hearing loss, many people suffer with tinnitus, a disorder defined by a constant ringing or buzzing sound in the ears that becomes more evident in quiet surroundings and reduces quality of life.Among the various medical problems faced by people,here comes hearing loss,which is a crucial problem faced now a days.The major reason for hearing loss are due to the age factor and prolonged exposure to loud noise.According to the world health Organization about 5% of the world population,i.e 430 million people are suffering from hearing loss.and by 2050 its expected to reach about 25% due to the unsafe listening habits,which is targeting young adults.Cochlear implants (CIs) do not provide same level of hearing intensity in noisy environment and it should be trained using ML models to reduce the noise to reach the output. [1].

Speech recognition in noisy and multi-talker contexts is actually a prominent concern among people with HL, including those who wear hearing aids. It is frequently mentioned as a hindrance to social interactions, education, work performance, and general quality of life.l.[2]

Despite their ongoing improvement, assistive listening devices and speech processing approaches still do not perform well enough in noisy multi-talker environments, as they may fail to restore the intelligibility of a speaker of interest among competing sound sources[2]. Since the days of using animal horns and ear trumpets to convey sounds to the ear, ear aids have seen substantial advancements. Digital signal processing (DSP) was incorporated into hearing aid processing algorithms as digital technology emerged in 1996. Since then, improvements in digital technology and the DSP chip's processing speed have made it possible to increase and enhance the functionalities. Aid users are still complaining about telephone performance and trouble hearing voice over background noise. According to the MarkeTrak V statistics,
just 1% of users reported having trouble using the phone, and 25.3% of hearing aid users said they had trouble listening when there was a lot of background noise. Recent developments in directional microphone technology and improvements in access to and use of wireless technology are improving speech recognition in noise, or the signal-to- noise ratio (SNR)[3]

This article predicts which areas of hearing aid technology will see more progress in the coming decade. Research advancements in wireless technology, digital chip technology, hearing science, and cognitive science will drive innovation in digital hearing aids, both gradual and radical. We will address the opportunities and constraints in each of these areas. New technology will be driven by developing trends like connection and
individualization [4]
The various environmental context in hearing aid personalization studies involves Environmental noise signal classification, Customization of tuning parameters with an emphasis on noise reduction, Modeling the environment and suggesting ideal configurations in various situations, Recognizing daily routines to customize the setups of hearing aids, Improving the recognition of daily routines through sequence learning methods, User intentions regarding hearing preferences are represented with context obtained from mobile devices, Developing a paradigm for online contextual customisation of hearing aids using simulation and Finding daily trends and fluctuations in the use of hearing aids to learn more about user behavior [5]

Amplification is the main method of treating hearing impairment since it is frequently characterized by a loss of audibility. But the easiest part is getting sounds loud; most people with mild to moderate hearing loss can hear sound with even low-cost devices. The two main issues with sensorineural hearing loss area decrease in the residual dynamic range, which some refer to as loudness recruitment, and distortion at the auditory periphery, which lowers the signal-to-noise ratio (SNR) of the sensory code before it ascends to the auditory cortex. The remedy for the above problems are wide dynamic range compression (WDRC) and reducing spectral intensity to reduce SNR. With all the technology currently available to improve speech is an exciting time to develop, research, fit and use hearing aid [6]

A signal processing strategy combining beamforming and mask-informed speech augmentation was evaluated by evaluating sentence identification in listeners with mild-to- moderate hearing impairment under adverse listening settings that mimicked the output of behind-the-ear hearing aids in a loud classroom. Two kinds of beamforming were compared: binaural, which regarded each aid's two microphones as a single array, and bilateral, which produced independent left and right beamformers. This shows that, at least for optimally adapted beamformers, employing all available microphones to produce a narrower beam provides a higher improvement in SNR than any benefit from preserving spatial clues in residual noise [7]. For audiologists working in clinical settings, standard machine learning techniques to forecast hearing aid fitting could be a viable solution. Deep learning has recently gained popularity as a solution for resolving significant problems that the artificial intelligence (AI) field has been facing for a long time. It has proven to be quite effective in identifying complex high-dimensional data structures. In addition to surpassing previous records in speech and picture recognition. Because deep machine learning takes relatively little programming and can readily take use of the growing amount of data and processing available, researchers predict that deep machine learning will have many more triumphs in the near future. The development of new deep neural network learning algorithms will only significantly speed up this advancement [8]. The UK will have 15 million hearing- impaired individuals by 2035, costing the economy 30 billion pounds a year. Individuals who have hearing loss are more vulnerable to interference from background noise compared to listeners without impairments. However, one of the biggest issues with hearing aid technology is still speech in noise.When the signal-to-noise ratio (SNR) is poor, current hearing aids are frequently ineffective. Poor speech intelligibility is a typical complaint among those who wear hearing aids, and it is a common cause of non- use . Conventional devices often increase both the target speech and the background noise.Speech technology has seen significant advancements in machine learning during the past couple decades. For instance, performance in automated speech recognition (ASR) is lacking comparable to what was feasible ten years ago. Researchers have advanced quickly in the CHiME, REVERB, Blizzard, and

Hurricane challenges by building on open-source baseline software that is enhanced with each iteration.This is a potential method for processing speech signals from hearing aids, according to recent advances in machine learning applied to noise reduction and speech enhancement. As a result, machine learning problems usually presume normal hearing, and there is little access to hearing-impaired listeners for thorough algorithm validation [9]. The investigation focuses on sophisticated models for hearing impairments, including Support Vector Machines (SVM), Random Forests (RF), and Multi-Layer Perceptrons (MLP), and looks at how well they work in auditory assistive applications. When it comes to visual impairments, cutting- edge. The ability of object detection frameworks such as You Only Look Once (YOLO), Single Shot Multi Box Detector (SSD), and RetinaNet to facilitate real-time object recognition and navigation aids is assessed. Additionally, the study examines applications based on generative artificial intelligence for use cases involving vision and hearing impairments. Innovative solutions offered by machine learning enhance the quality of life, independence, and accessibility for people with sensory deficits through the development of sophisticated algorithms and innovativetechnology. Several machine learning models and their possible uses in helping the blind and hard of hearing have been examined in this study. These tools have shown a lot of promise in helping people with sensory impairments communicate, navigate, and access information. They include image recognition algorithms like YOLO and SSD, as well as voice recognition models like TensorFlow Voice Recognition and DeepSpeech. However, in addition to the potential benefits, it is crucial to identify and address the moral dilemmas and challenges that come up during the development and application of machine learning innovations for the benefit of the blind and deaf [10]

## II. Related Works

Courtney Kolberg et al.,has introduced bimodal cochlear implant (CI) users with a CI in one ear and a hearing aid (HA) in the other. Deep neural net work (DNN)-based noise reduction have improved speech understanding for hearing aid users in noisy environments with boosting of speech intensity. Method used was Phonak Audré Sphere was fitted to eleven bimodal CI patients, ages 71 to 89. They received an Infinio 90 HA in their non-implanted ear along with two programs: Spheric Speech in Loud Noise, which employs DNN-based noise reduction, and Calm Situation. AzBio sentences in both calm and noisy environments with the CI alone, hearing aid alone, and bimodally with the Calm Situation and DNN HA programs were used to test sentence recognition scores.DNN programming had increased the bimodal performance in noise, scoring 79% compared to 60% with Calm Situation [11].

Tao Zhang et al in 2016, Concluded Nonlinear amplification, which offers adequate correction for both mild and loud noises, was the driving force behind the first revolution in hearing aids. The focus of the second revolution in digital signal processing is on algorithms and programmability. Wireless technology, which connects a pair of hearing aids to external devices, is the third revolution in hearing aids. In recent years, machine learning has drawn a lot of attention and has been used in a variety of other fields, including crowdsourcing, robotics, speech recognition, and genetics. Applications of machine learning in voice improvement, individualization and customization of signal processing methods, and increasing the efficacy and efficiency of clinical examinations are the three main fields [12]

**J**ehad Feras et al, 2024 made a revolution stating that diagonsis and treatment for hearing loss was ruled by artificial intelligence (AI), particularly machine learning (ML) which affects over 5% of the global population. AI uses audiology by finding early detection, rehabilitation plans and integrating electronic health records .A new terminology "computational audiology," combining audiology and ML ,allows for automated, accurate hearing tests. Continued research and development will enhance AI applications in audiology, improving patient outcomes and quality of life worldwide.

AI addresses these issues by producing more complete and accurate audiograms, allowing patients to better their health results. Furthermore, AI and connections with Electronic Health Records (EHR) provide ongoing and coordinated care based on entire historical information. This method acknowledges that a diverse set of elements contribute to patient care, resulting in a more comprehensive approach to healing. Applying AI to audiology might be viewed as a significant step toward enhancing several parts of hearing aid processes. Future breakthroughs in this field will only boost the use of AI in evaluation, resulting in better patient experiences and transforming global audiology techniques.[13]

Vahid Ashkanichenarlogh et al 2025, demonstrates DNN - based model improves sound quality and speech clarity. Using objective auditory-model metrics (HASPI for intelligibility, HASQI for quality), the DNN approach consistently outperformed an excellent output. The integration of DNN processing with beamforming yielded the highest predicted intelligibility scores at SNRs of 0 and +5dB. It is important to note that HASPI estimates are based on a better-ear metric. The parameter such as as noise type, venting configuration, and azimuth indicate that the magnitude of DNN benefit. A comparative table for speech quality and intelligibility represents the combination beamforming plus DNN consistently enhances both HASQI and HASPI scores across all conditions [14]

Designing, creating, and evaluating a wristband monitoring device for blood leak detection during haemodialysis is the aim of this research. A photointerrupter, a Bluetooth 4 wireless module, power, and alert components are all included in the design. According to the validation results, it just requires a very tiny volume of blood (0.01 ml) and detects blood leaks in 1.6 seconds. Additionally, this device's battery lasts longer than those of the commercial goods that are already on the market. It can monitor constantly for up to 41 hours and continuously sound an alert for 18 hours. Furthermore, the Bluetooth wireless signal's transmission range can be increased to 23 meters. The created blood leakage monitoring system in this study is an independent system that can be easily employed in conjunction with the haemodialysis equipment now in use. Furthermore, it is a non-invasive monitoring tool that makes it simple to mount the detector on a person's arm. The primary function of the developed blood leakage monitoring system is that, upon detection of the blood leak, a warning light and alert sound are activated. Additionally, the alert signal is transmitted via Bluetooth 4.0 wireless transmission to a monitoring computer that has the developed user interface software installed, such as a healthcare station. As a result, the proper medical personnel could start providing treatment right once [14].

According to Eric W. Healy et al., monaural (single-microphone) algorithms can improve the clarity of speech. To distinguish speech from noise, a binary masking-based technique was created. The current study's masks for separating speech from noise were approximated by training the algorithm on audio that wasn't used during testing. At different signal-to-noise ratios (SNRs), sentences were combined with babble and speech-shaped noise. Intelligibility improved after processing under all circumstances, according to tests conducted with normal- hearing and HI listeners. These gains were greater for the least-favorable SNRs, the modulated backdrop, and HI listeners. Additionally, they were frequently significant, enabling a number of HI listeners to raise their intelligibility scores from close to zero to over 70%.[15]

The study by Nada and Moaty (2025) explores the transformative role of artificial intelligence (AI) in redefining hearing aid technologies beyond conventional amplification. The authors highlight that traditional hearing aids often fail to address individual user preferences, leading to dissatisfaction among nearly half of users. The integration of AI, particularly machine learning, enables adaptive sound processing, personalized tuning, and improved speech clarity in complex acoustic environments. The paper emphasizes the capability of AI to learn user behavior and environmental patterns, thereby optimizing hearing aid performance in real time. Additionally, features such as remote adjustments, smartphone connectivity, and data-driven customization significantly enhance user experience. The study also discusses how AI reduces dependency on frequent audiologist visits by enabling automated calibration. Overall, the paper concludes that AI-driven hearing aids have strong potential to improve communication effectiveness and quality of life for individuals with hearing loss [16]

The research on the Hazebot system presents a comprehensive AI-driven tinnitus management platform integrating sound therapy, mindfulness, and symptom tracking. The system leverages machine learning algorithms such as Random Forest and K-means clustering to analyze user-reported data and identify patterns in tinnitus severity and triggers. It also incorporates natural language processing techniques and chatbot functionality to provide personalized guidance and support. The inclusion of validated clinical tools like the Tinnitus Handicap Inventory (THI) and Tinnitus Functional Index (TFI) enhances the reliability of self- assessment. Furthermore, the system promotes user engagement through daily monitoring and therapeutic interventions such as breathing exercises and meditation. The authors emphasize the importance of combining data-driven insights with holistic care approaches. The results indicate improved user engagement and better symptom awareness. This study demonstrates how AI can bridge gaps in traditional tinnitus care by offering scalable, personalized, and continuous support [17].

Searchfield and Sanders (2022) conducted a randomized single-blind controlled trial to evaluate a digital polytherapeutic intervention for tinnitus management. The system integrates personalized counseling, passive sound therapy, and interactive game-based auditory stimulation. The study compares this approach with conventional white noise therapy over a 12-week period. Results indicate that the experimental group experienced clinically meaningful improvements in Tinnitus Functional Index (TFI) scores, suggesting enhanced symptom relief. The use of mobile applications and wearable audio devices improves accessibility and usability for patients. Additionally, the study highlights the importance of personalized and engaging interventions in achieving better outcomes. Although statistical differences between groups were not always significant, user satisfaction and engagement were higher in the intervention group. The research supports the effectiveness of combining multiple therapeutic strategies in a single digital platform. It also underscores the growing importance of digital therapeutics in audiological care.[18]

The HUSH trial investigates the effectiveness and cost-efficiency of digital hearing aids in managing tinnitus among patients with hearing loss. The study adopts a randomized feasibility design to assess clinical outcomes and practicality of large-scale trials. Results show that a significant proportion of participants experienced improvements in tinnitus symptoms, as measured by the Tinnitus Functional Index (TFI). The study also highlights the variability in patient responses, indicating the need for personalized treatment approaches. Although health-related quality of life improvements were limited, hearing aids were found to be beneficial in reducing tinnitus perception for many users. The research emphasizes the importance of standardized clinical guidelines for tinnitus management. Additionally, cost-effectiveness analysis suggests potential long-term benefits despite higher initial intervention costs. The findings support the integration of hearing aids into tinnitus treatment plans while recommending further large-scale studies for validation.[19]

Meng et al. (2022) provide an editorial overview of digital hearing healthcare, emphasizing the growing role of advanced technologies in addressing hearing-related disorders. The paper highlights the increasing global burden of hearing loss and tinnitus, affecting billions worldwide. Digital healthcare solutions, including tele-audiology, AI- driven diagnostics, and wearable devices, are identified as key enablers of accessible and cost-effective care. The authors discuss how computational audiology and machine learning techniques improve diagnosis, treatment, and rehabilitation processes. Furthermore, digital platforms facilitate remote monitoring and personalized intervention strategies. The study also emphasizes interdisciplinary collaboration across fields such as neuroscience, engineering, and clinical medicine. Despite the advancements, challenges such as data privacy, standardization, and clinical validation remain. Overall, the paper concludes that digital hearing healthcare has the potential to revolutionize traditional clinical practices and significantly enhance patient outcomes [20]

## III. PROPOSED METHODOLOGY

The proposed system plans to develop a deep learning-based digital hearing aid which will act as user -friendly device for hearing impaired people to overcome the disadvantage faced by them in the exciting system. It separates noise from speech and at the same time enhancing speech intensity and includes tinnitus masking. It effectively improves both comfort and communication. The artificial intelligence dominates by providing an efficient and more natural hearing experience. This work is not only academically strong but also an useful innovation in health care applications, where we can develop a real time hearing aid machine which will be a major break through in the medical field. Therefore it would be the best solution for people suffering from hearing impairments and highlight the importance of deep learning in recent technologies.

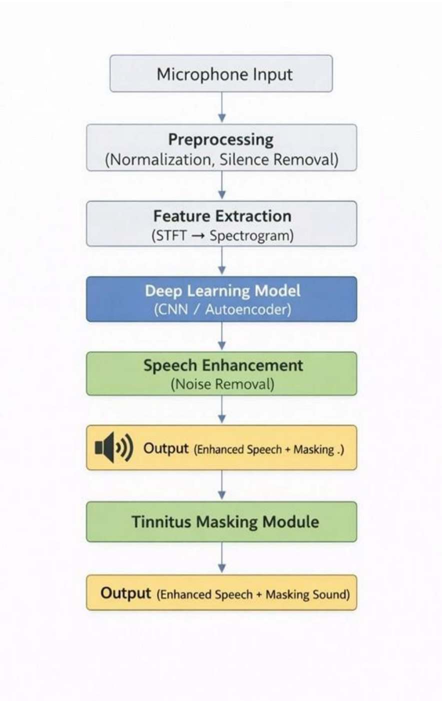


**Fig.1 Architecture of the Hearing Aid**

Fig 1 represents the entire architecture of the proposed system. The system has a microphone input, which consists of audio and environmental noise. Then it moves to the preprocessing stage, maintains constant amplitude and silent portions are removed. After preprocessing, the signal undergoes feature extraction using the Short-Time Fourier Transform (STFT). This process converts the time-domain audio signal into a spectrogram, which represents the frequency components of the signal over time. Since speech and noise have different frequency patterns, this transformation helps in distinguishing between them effectively.

*A. Overall Architecture of the Proposed System*

The spectrogram is allowed to undergo deep learning model algorithms, where the input is analyzed and separates speech from noise-based patterns learned during training. A clear version of voice is reconstructed by removing the unwanted background noise.

Next, the tinnitus masking module is applied. In this stage, a controlled low-level background sound, such as pink noise, is generated to reduce the perception of tinnitus. This masking sound helps in minimizing the discomfort caused by ringing in the ears.

Finally, the system produces the output, which is a combination of enhanced speech and masking sound. The output provides improved speech intelligibility along with increased listening comfort, making the system effective for individuals with hearing impairment. The other parameter such as signal -to-noise Ratio which measures the amount of noise reduced. Mean Square Error describe the distortion of the speech signal. Peak Signal -to-Noise Ratio which evaluates the overall signal quality.

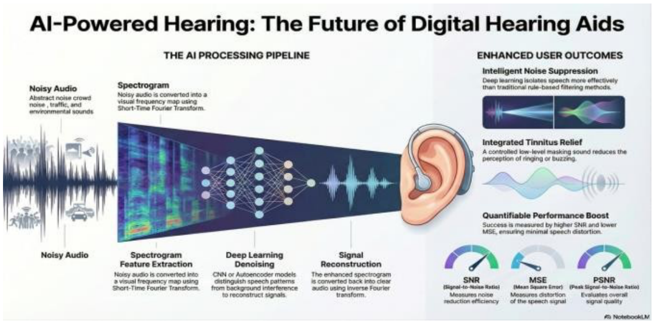


**Fig 2.Schematic overview of the project** Fig 2 explains the overview project ,when an input noisy audio is given,its is processed using spectrogram feature extraction

,where is converted into visual frequency map using short time Fourier Transform,then followed by deep learning denoising ,where the CNN model distinguish speech pattern from background noise and reconstruct the signal.Finally the enhanced spectrogram is converted back into clear audio using inverse Fourier transform.The expected output are intelligent noise suppression ,along with it pink masking is used to integrate tinnitus relief.The various parameters obtained are SNR,MSE and PSNR.

The proposed system, AI-Based Smart Hearing Aid for Improved Speech Intelligibility and Tinnitus Masking, is designed to enhance hearing support for users affected by speech perception difficulties and tinnitus. The system combines advanced signal processing, artificial intelligence, and adaptive tinnitus masking in a compact smart hearing aid architecture. The main objective is to improve speech clarity in noisy environments while simultaneously providing personalized masking signals to reduce the perceptual impact of tinnitus.

The proposed model operates in two parallel paths. The first path focuses on speech enhancement and intelligibility improvement, where the incoming acoustic signal is captured through a microphone array, preprocessed, denoised, and then enhanced using an AI-based speech enhancement module. The second path is dedicated to tinnitus masking, where the system analyzes the tinnitus profile of the user and generates an adaptive masking sound that minimizes tinnitus perception without interfering with speech comprehension. These two outputs are then fused and delivered to the user through the hearing aid receiver.

### *B. Acoustic Signal Acquisition*

The hearing aid continuously captures environmental audio and converts it into a digital signal. After preprocessing, the signal is fed into an AI engine trained to separate speech from background noise. Simultaneously, a tinnitus masking module generates a personalized sound pattern based on the user's tinnitus frequency and loudness characteristics. The system then adaptively mixes the enhanced speech and masking signal according to real-time listening conditions.

B. Acoustic Signal Acquisition

The first stage of the proposed system is the acquisition of the surrounding sound using miniature low-power microphones. Let the input acoustic signal be represented as:

$$x(t) = s(t) + n(t)$$

where:

- $x(t)$ is the observed noisy signal.
- $s(t)$ is the clean speech signal,
- $n(t)$ is the environmental noise.

This mixed signal includes speech, ambient disturbances, and other unwanted components. The goal of the proposed system is to estimate the clean speech signal and generate an appropriate masking signal for tinnitus relief.

### *C. Preprocessing and Feature Extraction*

The acquired signal is first passed through a preprocessing block for normalization, framing, and spectral analysis. The signal is divided into short overlapping frames to preserve local temporal information. A Hamming window is applied to each frame before frequency-domain transformation.

The acquired signal is first passed through a preprocessing block for normalization, framing, and spectral analysis. The signal is divided into short overlapping frames to preserve local temporal information. A Hamming window is applied to each frame before frequency-domain transformation.

The short-time Fourier transform (STFT) of the input signal is expressed as:

$$X(m,k) = \sum_{n=0}^{N-1} x(n)w(n-m)e^{-j2\pi kn/N}$$

where:

- $X(m,k)$ is the spectral coefficient,
- $w(n)$ is the window function,
- $m$ is the frame index,
- $k$ is the frequency bin.

From the spectral representation, relevant features such as Mel-frequency cepstral coefficients (MFCC), spectral entropy, zero crossing rate, and energy measures are extracted. These features are used by the AI model for speech enhancement and environmental classification.

### *D. Al-Based Speech Enhancement Module*

The core of the proposed work is an AI-based speech enhancement model that improves speech intelligibility. A deep neural architecture such as CNN-BiLSTM or Transformer-based enhancement network is employed to distinguish speech components from background noise. The model learns complex temporal and spectral patterns from large-scale speech datasets and estimates a clean speech mask.

The enhanced speech spectrum is obtained as:

$$\hat{S}(m,k) = M(m,k) \cdot X(m,k)$$

where:

- $\hat{S}(m,k)$ is the extimatod closn cneech spectrum,
- $M(m,k)$ is the □ Ask ChatGPT equency mask,
- $X(m,k)$ is the noisy input spectrum.

The AI module dynamically adapts to different listening environments such as traffic, crowd, home, and office spaces. This intelligent adaptation significantly improves speech perception, especially in low signal-to-noise ratio conditions.

E. Hearing Loss Compensation

After speech enhancement, the processed signal is passed through a hearing compensation block. Since different users have different audiometric hearing loss patterns, the system amplifies specific frequency bands according to the user's hearing threshold profile.

## V RESULT AND DISCUSSION

The hearing aid for improved speech enhancement and tinnitus making was performed using deep learning algorithm techniques such CNN and autoencoder, along with various preprocessing and post processing steps. The output obtained are, the input audio with noise given, which can be heard and once the program coding is completed, the clear voice can be heard. The other visual outputs that are represented are a graph indicates the NR, MSE and PSNR values. Performance evaluation of the proposed system using three important metrics: Mean Square Error (MSE), Signal-to-Noise Ratio (SNR), and Peak Signal- to-Noise Ratio (PSNR) over time.is possible to design a wearable device that can specifically detect losses of body fluids such as sweat, blood, and urine, alerting the affected individual. By developing these health monitoring devices, we can significantly enhance the prediction and diagnosis of adverse health conditions in their early stages. Along with it pink masking is used to reduce the tinnitus.

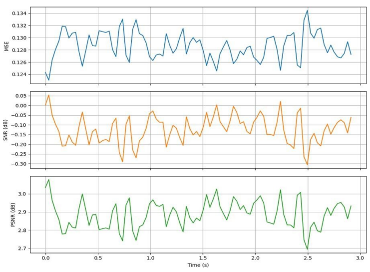

**Fig 3: Graphical representation of SNR,MSE and PSNR**

Inputs and produced enhanced speech outputs with significantly reduced noise components, which can be heard clearly using a speaker or headphones. The results demonstrate that the use of deep learning models such as Convolutional Neural Networks (CNN) and Autoencoders plays a crucial role in accurately separating speech signals from background noise. The performance of the system was evaluated using standard metrics such as Signal-to-Noise Ratio (SNR), Mean Square Error (MSE), and Peak Signal-to- Noise Ratio (PSNR). The results show a noticeable improvement in SNR values after processing, indicating that the system effectively reduces noise and enhances the desired speech signal. A lower MSE value confirms that the reconstructed speech signal is very close to the original clean signal, with minimal distortion. Similarly, higher PSNR values indicate better overall signal quality and improved clarity.

## VI CONCLUSION

The suggested solution significantly outperforms traditional hearing aids despite a number of drawbacks. For those with hearing loss, the combination of deep learning, spectrogram analysis, and tinnitus masking offers a complete solution. Future developments may concentrate on improving the model's adaptability to various situations, lowering computing complexity, and optimizing it for real-time processing.Overall, the conversation demonstrates that the suggested deep learning-based digital hearing aid system is a viable strategy for raising user comfort, lowering noise, and boosting speech intelligibility. It illustrates how artificial intelligence can revolutionize assistive technology and enhance the lives of people with hearing impairments.

Audio output clearly ensure the overall output of the project and tinnitus removal ,along with it the MSE value remains relatively low and SNR value is fluctuating and remains us about the improvement in the ratio, while PSNR is high. We can convert it into a real -time hearing aid for future enhancement

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