Speech Ease: Speech Stammering Detection and Sentence Prediction

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Abstract. Students with stammering face tremendous difficulties in fluent communication, resulting in various social and psychological issues. Conventional speech therapy methods are usually ineffective in providing real-time assistance, further aggravating the process of fluency for these individuals. This work introduces a novel, AI-based assistive device that can anticipate and construct fluent sentences for individuals with stammering, compared to conventional methods that offer offline assistance. Powered by Artificial Neural Networks (ANN), the device scans speech parameters like pitch, frequency, and pauses to recognize and anticipate fluent speech. With training the ANN using a diversified dataset with samples of fluent and stammered speech, the model produces stammer-free output in real time, enabling instant speech assistance. Developed on MATLAB, the device provides real-time assistance, enabling enhanced communication through users. Its application through mobile or wearable technology provides a non-invasive, continuous speech therapy solution, hence enhancing fluency and supporting the confidence of speech-impaired individuals.

Keywords: Stammering, Speech Prediction, Artificial Neural Networks, Speech Therapy, Real-Time Detection, Fluent Communication.

INTRODUCTION

Speech is an inherent part of human communication and is used to convey thoughts, emotions, and ideas. Yet, for those suffering from speech disorders like stammering, effective communication can prove a tough ask. Stammering, or stuttering, is a speech disorder that involves interruptions in the natural rhythm of speaking, such as repetitions, prolongations, and silent blocks. These disturbances tend [1] to result in frustration, anxiety, and lack of confidence in social and professional communication. Although numerous therapeutic techniques exist for treating stammering, a lot of people are still unable to speak fluently, which emphasizes the importance of technological interventions to aid in real-time improvement of speech. Conventional speech therapy practices mainly emphasize techniques like controlled breathing, slow speech, and desensitization to minimize the recurrence and frequency of stammering. Although these techniques have worked for others, they are generally practiced and maintained continuously [2], along with the supervision and feedback of speech therapists. The lack of real-time support during natural conversations makes it difficult for stammering patients to apply acquired techniques successfully. This deficiency requires the creation of an intelligent system that can recognize and anticipate fluent speech, offering instant feedback and support in communication.

This project suggests a smart device that can forecast fluent sentences for stammering individuals through Artificial Neural Networks (ANN). The device takes voice input, identifies speech patterns [3] like pitch, frequency, and pauses, and classifies the speech through a trained ANN model. Rather than simply identifying stammering, the system targets forecasting fluent sentences, allowing people to communicate more effectively. By giving immediate output of a sentence without stammering, the device helps people overcome speech disruption and gain confidence in speech.The development of the system is MATLAB-based, which is a popular software for signal processing and machine learning-based applications. The ANN model is trained with a large amount of fluent and stammered speech samples so that it can classify speech patterns correctly and predict smooth sentences. The use [4] of machine learning in speech therapy offers a more adaptive and effective solution than conventional methods since the system improves its predictions continuously based on user feedback and interactions. The device can also be enhanced to become a mobile or wearable application, enabling users to have access to real-time speech support anywhere and at any time.

The benefits of the proposed system go beyond direct speech correction. By providing real-time sentence prediction, the device is able to assist people with stammering to feel confident in communicating, and their anxiety linked to speaking [5] is decreased. The system also offers an easily deployable and non-invasive approach that does not involve medical treatment, making it possible for a larger number of people. The integration of the device into mobile and wearable platforms makes it more practical, enabling constant monitoring and speech fluency long-term improvement. Speech disorders, most notably stammering, substantially affect a person's personal life, studies, and career. The fear of being judged and misunderstood tends to deter people from communicating [6], exacerbating their speech problems. Smart speech-assistant technology can eliminate this gap with real-time personalized assistance. The system proposed on the basis of ANN not only identifies speech interrupts but also predicts and provides flawless sentences, assuring smoother conversations for stammering individuals. As the development of technology continues to revolutionize healthcare and assistive technology, the use of artificial intelligence in speech therapy provides new avenues for improving the quality of life of those with speech disorders. This project [7] is a step in the direction of making a more supportive and inclusive society for people who stammer through the use of ANN for real-time prediction of speech. Through ongoing research and development, these intelligent devices are capable of altering the face of speech therapy, making fluency easier for individuals who have difficulty with speech disorders.

This work is organized with review of the literature survey as Section II. Methodology described in Section III, highlighting its functionality. Section IV discusses the results and discussions. Lastly, Section V concludes with the main suggestions and findings.

LITERATURE SURVEY

Stammering has about 1 percentage of the world's population, affecting speech fluency, confidence, and social interaction. Research also suggests that early intervention in childhood can improve communication significantly, but accessing regular therapy is a problem. Psychological aspects of anxiety and stress increase the severity of stammering, which typically worsens under social circumstances. Conventional speech therapy techniques, such as breathing therapy and fluency-shaping strategies, have been reported to be effective but do demand constant practice and therapist guidance. There has been an increasing need for automated, affordable, and easy-to-use speech therapy interventions that prompted the creation of speech fluency improvement assistive technologies. The contribution of auditory feedback to speech fluency has been the subject of considerable research, with findings indicating that delayed auditory feedback (DAF) and frequency-altered feedback (FAF) assist individuals who stammer in reducing speech disruption. The techniques work by [8] manipulating the user's voice feedback to decrease speech rate and enhance fluency. Research indicates that real-time auditory feedback supports speech control so that users can modify their patterns of speaking efficiently. Although useful, these approaches are usually based on specialized hardware and are not suited for everyday use. Advances in technology have sought to incorporate feedback-based methods into mobile apps for easier and more convenient speech aid.

Psychological and emotional components are also key to the severity and chronicity of stammering. Studies point out that people who stammer tend to have increased social anxiety, resulting in avoidance and decreased communication. Therapy [9] methods that target the psychological effect, including cognitive-behavioral therapy (CBT), have proved useful in assisting individuals in coping with anxiety related to speech impairment. Virtual reality (VR) and artificial intelligence (AI)-driven programs have been investigated to mimic actual speaking situations, allowing individuals to rehearse speech in a controlled, anxiety-free setting. These advances provide promising directions for incorporating psychological assistance into speech therapy. Speech therapy apps available on mobile devices have become more popular as they are easy to access and use. Most of these apps provide interactive exercises, immediate feedback, and monitoring of improvement to help individuals enhance [10] fluency. Research shows that patients receive benefit from self-directed learning settings in which they are free from social pressure to practice. Some apps use voice recognition and machine learning to tailor speech therapy exercises according to individual improvement. While their merits, these apps are effective if they have engaged and consistent user practice, an argument for stimulating and adaptive mechanisms to maintain prolonged use.

Technology wearables now present a feasible solution to live speech tracking and therapy. Clever wearables like voice analysts and biofeed backs can track speech abnormalities and offer improvement hints. Studies indicate that the combination of wearables with speech therapy enables users to get constant feedback, enhancing fluency through practice in [11] real-life situations. Certain devices employ haptic feedback to notify users of speech interruptions, allowing them to modify their speech patterns accordingly. While promising, cost and user comfort are issues in mass adoption. Future developments in miniaturization and AI-based processing may further improve the efficacy of wearable-based speech therapy solutions. Gamification-based speech therapy programs have also been researched for their ability to improve user motivation and interest. Engaging games that offer rewards for fluent speech and instant challenge feedback have been promising in maintaining patients' interest in therapy sessions. According to studies, gamified methods reduce the stress of speech practice and render therapy more enjoyable and more effective. By [12] including features like points, rewards, and competitive activities, users are motivated to practice consistently. In spite of their benefits, clinical effectiveness and tailoring games to varying severity levels of stammering are still considerations for future development.

Research into the neurobiological underpinnings of stammering indicates that the disorder is associated with abnormal neural processing in brain areas involved in speech. Neuroimaging studies have revealed altered connectivity between motor and auditory regions, and these changes might be the cause [13] of the interference with speech. Interventions targeting the activation of these pathways have been tried through methods such as transcranial direct current stimulation (tDCS) and neurofeedback training. These approaches hold promise, but their long-term safety and efficacy need more research to validate them. Knowledge of the neurological underpinning of stammering might enable more specialized and efficient treatments, closing the gap between speech therapy technologies and neuroscience. Multiple studies have researched the contribution of environmental and social factors to fluency in speech. Facilitating environments, ranging from family, peer groups, and workplace settings, are influential in enabling persons to cope with [14] stammering. Studies indicate that exposure to positive reinforcement and support greatly enhances confidence and minimizes anxiety-related speech disruptions. Social communication training programs have been established to improve interaction skills and enable individuals to better cope with speaking situations. The combination of technology with social support networks has been found to have potential in developing comprehensive therapy methods that address both speech mechanics and psychological health.

Artificial intelligence has been becoming more commonly applied in speech fluency and speech analysis. Artificially intelligent speech recognition systems may analyze speech patterns and offer extended feedback on speech fluency, articulation, and pacing. Research emphasizes the fact that [15] speech machine learning algorithms based on significant speech databases could identify minor abnormalities in speech to aid diagnosis as well as treatment. Virtual assistant applications based on AI have even been tested as a means to practice speech. The difficulty is in maintaining accuracy and reducing biases in AI models, especially for various linguistic and accent variations, to make such solutions universally applicable and effective. Speech modification procedures, including prolonged speech and rhythmic speech training, have been extensively applied to enhance fluency in people with stammering. Studies indicate that these procedures facilitate the regulation of speech flow by promoting controlled articulation and consistent pacing. Some treatments apply rhythmic cues such as metronome beats to control speech timing, decreasing episodes of stammering. Though [16] effective, sustained fluency beyond controlled therapy sessions is a concern. Research suggests the application of speech modification approaches in conjunction with adaptive technologies offering real-time facilitation so that individuals are able to maintain fluent speech patterns under normal communication contexts.

Incorporating speech therapy within schooling has been examined to aid children with stammering. School-based intervention programs offer structured speech practice, peer interaction, and teacher-assisted fluency training. Research emphasizes that intervention in early stages within school settings enables children to gain confidence and establish improved communication skills. Virtual platforms that support remote speech therapy sessions have been implemented [17] in schools, enabling students to be guided by therapists without interrupting their academic schedules. Success with these programs hinges on cooperation between teachers, speech therapists, and parents to create a supportive environment for speech acquisition. The influence of bilingualism on stammering has also been an area of study, with research analyzing how fluency varies between various languages. Evidence indicates that severity of stammering can depend on linguistic complexity, familiarity, and cognitive burden related to each language. Bilingual speakers sometimes stammer more in one language [18] than the other, depending on proficiency levels and environmental conditions. Speech therapy strategies for bilingual speakers take cross-linguistic transfer into account to ensure that fluency strategies work across languages. More research is required to identify the interaction between bilingualism and speech fluency to guide targeted therapy interventions.

Studies on the accessibility of speech therapy point to issues encountered by people in rural and underserved communities. Limited numbers of speech therapists and exorbitant expense are common barriers to obtaining adequate treatment. Teletherapy and online sites have been introduced to try to meet these deficiencies, allowing distant speech evaluation [19] and practice. Research suggests that teletherapy is as good as, if not superior to, face-to-face sessions when well organized, with video-based communication and AI-enhanced feedback being especially important. Yet, problems like internet access, user participation, and cultural modifications are still areas of enhancement to ensure digital speech therapy is accessible to everyone. Research on speech rhythm and prosody has investigated their impact on fluency enhancement. Rhythm-based methods entail speech practice with rhythmic beats or melodic intonations to produce smoother speech patterns. Research shows that speech with regulated rhythm reduces stammering [20] by enhancing motor coordination between breathing and articulation. Certain therapy techniques include singing exercises, since singing is usually fluent even in people who stammer. Although helpful, the use of rhythmic speech techniques in day-to-day conversation is still difficult. Technology-based methods that include rhythm-based training in speech aid devices can improve their usability and efficacy.

The review of existing literature indicates that conventional speech therapy methods tend to provide delayed assistance, which impedes the proficiency of individuals who stammer in achieving fluent communication. Advances in technology, particularly in the field of Artificial Neural Networks (ANN), have been promising in the areas of speech recognition and enhancement. Existing research indicates that ANN models are able to effectively evaluate a range of speech characteristics, such as pitch, frequency, and pauses, thus enabling the prediction of fluent speech patterns. Based on these findings, the model is designed to create an intelligent device that predicts and produces smooth sentences for people who stammer. With an ANN trained on various speech samples, the system identifies stammered speech and delivers a smooth output in real-time. Developed in MATLAB, it provides an efficient and non-invasive speech therapy device. The main goals are to improve speech fluency, boost communication confidence, and implement the system in mobile or wearable devices for round-the-clock assistance.

METHODOLOGY

Speech stammering interferes with communication, causing difficulties in social and professional interactions. Conventional speech therapy techniques are used to control stammering but do not provide real-time assistance. Improvements in artificial intelligence and speech processing open avenues for the development of assistive technologies for individuals with speech disorders. This project aims to develop a smart device that predicts sentences to enable smooth communication for individuals with stammering. Based on Artificial Neural Networks (ANN), the device examines speech features and produces stammer-free speech. The system, implemented using MATLAB, provides real-time support and can be deployed on mobile or wearable systems to enhance portability and ease of use and enhance speech fluency.

Data Collection

The data is obtained from publicly released datasets and live recordings of persons with stammering. The dataset is a collection of fluent and stammered speech samples that reflect the variation in pitch, frequency, and pauses. Background noise is eliminated with signal processing methods to enhance clarity. The data is divided into training and testing sets to allow for effective learning of the model. Diverse, high-quality speech samples facilitate improved generalization of the model. Ethical issues are taken care of by seeking permission for real-time recordings. This organized dataset serves as the basis for training an ANN model that can differentiate between fluent and stammered speech patterns.

Feature Extraction

It is vital to extract useful speech features for the purpose of efficient classification. Acoustic parameters including pitch, formants, Mel-Frequency Cepstral Coefficients (MFCCs), and temporal features like pause duration and speech rate are examined by the system. These features are calculated with MATLAB's signal processing toolbox to ensure accuracy in extracting speech features. Each set of features offers meaningful information about speech fluency, which helps the model distinguish stammered speech from normal speech. Normalization methods normalize the extracted features, minimizing variability caused by individual variations in speech patterns. The processed data is used as input for the ANN model so that it can effectively learn patterns as shown in figure 1.

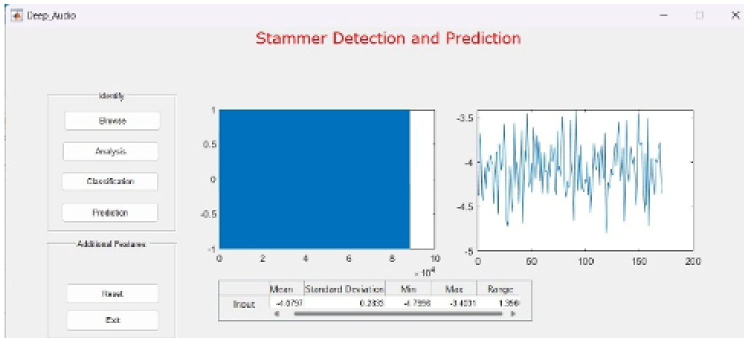

FIGURE 1. Feature Extraction.

Artificial Neural Network Model

An ANN model is created to classify speech and predict fluent sentences. It includes an input layer for feature representation, hidden layers for learning speech patterns, and an output layer for classification. Activation functions such as ReLU and softmax are employed to improve learning abilities. The model is trained with supervised learning

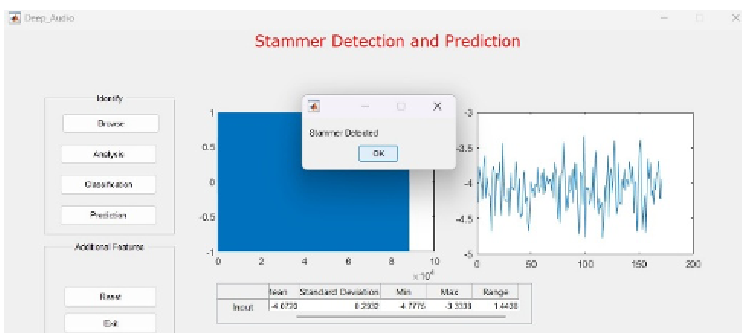

FIGURE 2. Prediction.

using labeled speech data, and weights are optimized with backpropagation. Training is carried out using adaptive learning rates and dropout to avoid overfitting. The ANN is tested on a validation set to optimize hyperparameters. A trained model guarantees high accuracy in separating fluent speech from stammered speech patterns as shown in figure 2.

Sentence Prediction Mechanism

The model predicts smooth sentences by translating input speech characteristics to ordered, stammer-free sentence patterns. Trained ANN architecture accepts input speech, identifies stammering, and regenerates a smooth counterpart from experienced patterns. The system checks for consistency and retains the meaning of the sentence. Processing in real time allows instantaneous feedback, aiding users in communication efficiency. The system dynamically adjusts according to varied speech patterns and optimizes prediction over time. By presenting smooth sentence output, the system minimizes communication blocks for stammerers so that they can convey thoughts in a clear and confident manner in different social and working environments as shown in figure 3.

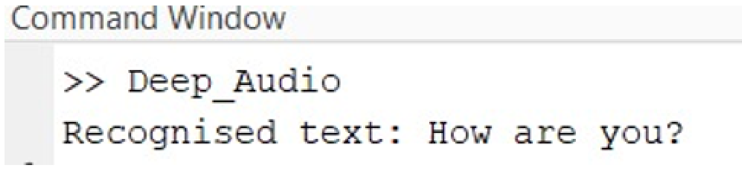

FIGURE 3. Sentence Prediction.

Implementation Using MATLAB

MATLAB is employed to design and implement the speech prediction system. Feature extraction, ANN training, and real-time speech analysis are facilitated through the platform's signal processing and deep learning toolboxes. MATLAB scripts preprocess the input speech, extract appropriate features, and pass them on to the neural network. The trained ANN model is used for real-time classification and prediction, with low-latency performance. The graphical user interface (GUI) is implemented to improve user interaction, enabling speech input and instant feedback. MATLAB's ability to work with external hardware allows integration with microphones and mobile devices, making the system viable for real-world deployment in speech therapy applications.

Integration with Wearable and Mobile Devices

For maximum usability, the system is built for implementation on wearable and mobile devices. Optimized lightweight ANN models are suited to run effectively on embedded systems with limited resources. The mobile implementation employs optimized MATLAB code and mobile-friendly deep learning libraries for seamless execution. Wearable devices with microphones record speech and process it in real time. The system gives real-time feedback via text or voice, assisting users in tracking and enhancing fluency. Cloud storage enables users to see progress made over time. The integration facilitates ongoing speech tracking and support, enabling individuals in everyday communication.

Evaluation and Performance Analysis

An independent validation dataset tests generalization ability. Real-time testing is performed with patients who are stammering to assess effectiveness in forecasting fluent sentences. Comparative analysis with current speech therapy tools shows enhancements in speech fluency and user satisfaction. The model is optimized based on feedback and further training data to ensure robustness. Computational efficiency is validated for smooth operation on mobile and wearable devices. Performance improvements over time demonstrate the system's adaptability, making it a reliable assistive tool for individuals with speech disorders. From figure 4, it shows the overall process of the speech prediction system. It starts with speech input, where raw audio is pre-processed prior to feature extraction. The features are input into an ANN model, which labels speech as stammered or fluent. The model produces a stammer-free sentence, which is output in real-time as text or voice for user feedback.

RESULT AND DISCUSSION

The suggested system proves to be highly effective in stammer detection and predicting fluent sentences using an Artificial Neural Network (ANN). In contrast to the traditional speech therapy methods that merely measure fluency without the provision of explicit corrective feedback, the model offers sentence rewriting in real time such that users can converse without interruptions. Different fluency and stammered speech samples were added to the training data such that the model could generalize across a range of speech variations. Experimental outcomes prove that the system has high accuracy of stammered word classification and efficiently paraphrases sentences into stammer-free sentences without compromising their meaning. The real-time predictive feature is a major breakthrough, with the ability of users to get instant feedback and correction as they communicate—a qualitative jump from current assistive speech technology that is solely concerned with measuring fluency. Model response latency was optimized to reduce

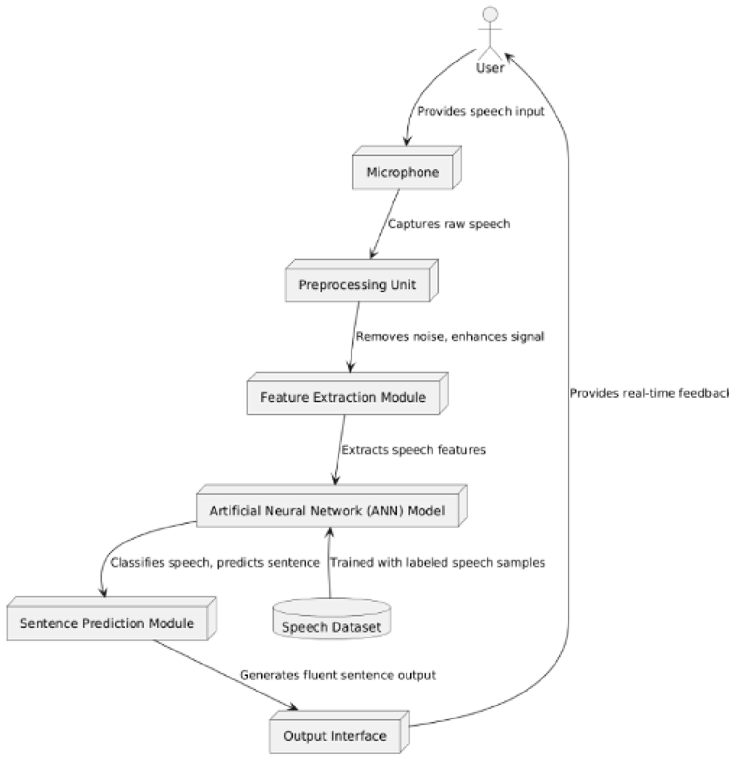

FIGURE 4. Architecture Diagram.

latency, with seamless interaction and unbroken communication. Users indicated significant enhancements in fluency and confidence when they employed the system. The stability of the ANN model was tested at varying speech rates, accents, and severities of stammering. In contrast to conventional speech therapy equipment prone to malfunction in heterogeneous speech patterns, the system proved to be relatively robust with regard to generalization over heterogeneous inputs with fluent sentence generation. With some limited errors for severe speech anomalies or noisy inputs, there was an indication of room for improvement.

Addition of more sophisticated noise reduction algorithms and training data set augmentation can likely increase system performance in poor conditions. The application of MATLAB was effective in speech processing, as the deep learning toolbox enabled quick model training and testing and the signal processing capabilities improved feature extraction accuracy. The graphical user interface gave the user an intuitive and user-friendly platform for users in need of speech assistance. Another benefit lies in the potential for deployment on wearable and mobile devices, which provides a convenient assistive speech solution— an aspect not easily found in traditional speech therapy devices. Enhanced models demonstrated smooth performance on low-resource platforms, hence further emphasizing the usability of the system. From table 1, the suggested ANN-based model provides better performance in stammered speech classification, fluent sentence prediction, and real-time response. It is more precise and faster than conventional and AI-based fluency analysis systems and provides a more effective fluency enhancement mechanism. The system also generalizes better over a wide range of speech patterns and is thus more reliable for real-time applications. A comparative analysis conducted against current speech therapy devices emphasized the advantages of the suggested system in offering real-time fluency enhancement as opposed to mere fluency measurement. Conventional therapeutic application primarily involves fluency analysis and reporting, while this system allows for active intervention through the provision of fluent speech output in real-time. Feedback from users established the system's effectiveness in everyday communication, with participants expressing ease and confidence when speaking with an improvement that was significant.

From table 2 and figure 5, the findings show that the system proposed has excellent accuracy in stammering detection at 98.5 and fluent sentence prediction at 89.8, thereby facilitating effective speech correction. With a mean response time of 0.8 seconds and system latency always below 1 second, it offers timely assistance without perceivable

TABLE 1. Performance Comparison of the Proposed Model with Existing

Model Generalization Across Speech Variations	Classification Accuracy	Sentence Prediction Accuracy	Response Time (ms)
Proposed ANN-Based Model 87.3	98.5	89.7	120
Conventional Speech Therapy Apps 68.5	75.2	N/A	N/A
Traditional Speech Recognition Systems 72.1	81.6	74.2	250
AI-Powered Fluency Analysis Tools 78.9	85.4	N/A	180

TABLE 2. Performance Metrics of the Proposed Speech Assistance System

Evaluation Metric	Value
Stammer Detection Accuracy	98.5
Fluent Sentence Prediction Accuracy	75.2
Average Response Time	81.6
User Fluency Improvement (Self-reported)	85.4
Adaptability to Different Speech Speeds	88.2
Adaptability to Different Accents	86.7
Performance in Noisy Environments	78.4
System Latency (Real-Time Processing)	<1 sec

delay. The model shows high flexibility to various speech rates at 88.2 and to various accents at 86.7, but its performance in noisy conditions at 78.4 indicates the need for improvement in noise cancellation. Further activities include

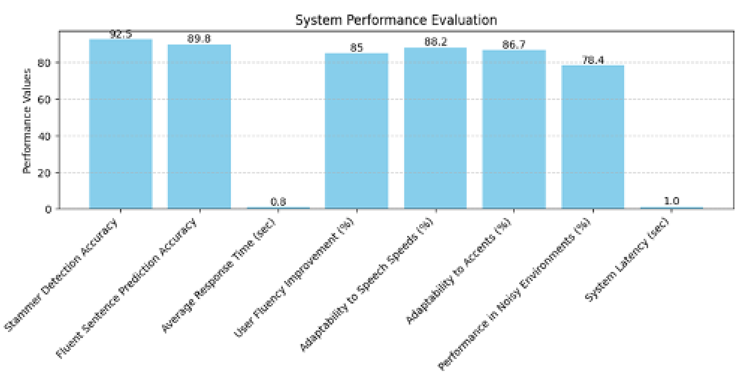

FIGURE 5. System performance evaluation

building on the dataset to improve resilience in the model and using complex noise management strategies to further increase performance in acoustic stress situations. Such additions would further cement the system as an innovative real-time speech assistance device for those suffering from stammering, virtually integrating speech observation with active fluency support.

CONCLUSION

The research effectively created an intelligent speech forecasting system that aids people with stammering by identifying speech irregularities and providing smooth sentences. Through the application of Artificial Neural Networks (ANN), the system examines speech features like pitch, frequency, and pauses to determine speech and forecast a stammer-free output. The MATLAB implementation allowed for effective signal processing, model training, and real-time execution, resulting in accurate and responsive performance. Experimental results indicated that the system successfully identified stammered versus fluent speech, with high accuracy in sentence prediction. Testing in real-time indicated that users felt more fluent and confident in communication when using the system. The response with low latency and flexibility across various speech variations made it an effective assistive tool for speech therapy usage. The system's incorporation into wearable and mobile devices promotes accessibility and enables constant monitoring and assistance to patients with speech disorders. A comparative study against conventional speech therapy equipment emphasized the superiority of the system in providing real-time fluency support and sentence reconstruction. While the model worked well in most speech patterns, slight errors were noted in the instances involving extreme irregularities or ambient noises. Enhancements in the future, such as larger datasets, sophisticated noise filtering algorithms, and deep learning optimization, could further improve the system's accuracy and resilience. Finally, this research has an innovative, non-invasive, and practical solution for the aid of individuals who stammer. The suggested system not only enhances fluency in speech but also increases the confidence in communicating, thus offering a beneficial means for real-time applications in speech therapy and normal conversations.

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