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A REVIEW PAPER ON THE PRESENT AUDIO PROCESSING TECHNIQUES

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ABSTRACT

In this review paper, different advancement in the speech filtering methods has been discussed. Modern audio processing techniques have developed significantly, partly as a result of developments in digital signal processing and machine learning. Deep learning has completely changed the discipline, enabling cutting-edge outcomes in automatic voice recognition, music creation, and noise reduction. Convolutional and recurrent neural networks in particular have revolutionized the subject. For applications like video conferences, gaming, and virtual reality, real-time audio enhancement techniques are essential because they give better audio quality through noise cancellation, echo suppression, and spatial audio rendering. The use of voice assistants has become ubiquitous, and they rely on cutting-edge audio processing to recognize speech and comprehend natural language. Various fields can benefit from audio content analysis, which includes speech emotion recognition and sound event identification. AI-driven solutions are advantageous for music and audio creation, making duties for artists and engineers easier. Techniques for audio restoration and preservation are also essential for preserving and improving old recordings. Ethics related to privacy and appropriate data usage are becoming more important. Overall, these advancements indicate that the area of audio processing will continue to expand and innovate, improving audio experiences in a variety of contexts.

Keywords: Audio Processing, MATLAB, Etc.

I. INTRODUCTION

Audio processing is a specialized field within digital signal processing (DSP) that involves manipulating and enhancing audio signals, such as speech or music, using various algorithms and techniques. It plays a crucial role in numerous applications, including telecommunications, multimedia content creation, speech recognition, music production, and even medical diagnostics.

The benefits of audio processing are multifaceted.

1. It allows for the improvement of audio quality. By removing noise, equalizing audio frequencies, and enhancing overall clarity, audio processing ensures that sound recordings and transmissions are of high quality, enhancing the user experience in various contexts.
2. Audio processing enables the extraction of meaningful information from audio data. In speech recognition systems, for instance, it helps convert spoken language into text. In music analysis, it can identify and categorize musical elements, facilitating music recommendation systems and genre classification Furthermore, audio processing contributes to data compression, which is essential for efficient storage and transmission of audio content. By reducing the data size without significant loss of quality, it enables streaming services, internet radio, and audio file formats like MP3.
3. Audio processing is beneficial as it enhances audio quality, extracts valuable information, and enables efficient data compression. Its applications span from entertainment and communication to healthcare and scientific research, making it an essential component of modern technology and everyday life.

1.1 Audio signal processing using MATLAB

Audio processing using MATLAB offers unparalleled advantages over other software solutions. Its superiority lies in its comprehensive toolset, extensive libraries, user-friendly interface, robust scripting capabilities, and widespread industry adoption.

1. MATLAB provides a rich array of functions and toolboxes specifically designed for audio processing. These built-in features streamline complex tasks like filtering, spectral analysis, and waveform manipulation, saving users valuable time and effort. Moreover, MATLAB's extensive library of audio processing functions ensures that users have access to cutting-edge techniques and algorithms.
2. MATLAB boasts an intuitive and user-friendly interface that simplifies the audio processing workflow. Its graphical user interface (GUI) allows both beginners and experts to interact with audio data effortlessly. The

interactive environment facilitates real-time experimentation, enabling users to make rapid adjustments and see immediate results.

3. MATLAB's scripting capabilities are unmatched. Users can write custom scripts and functions tailored to their specific audio processing needs. This flexibility empowers researchers, engineers, and musicians to create custom algorithms and automate repetitive tasks efficiently. MATLAB's scripting capabilities also foster collaboration and knowledge sharing within the audio processing community.

4. MATLAB's widespread adoption in academia, research, and industry makes it a standard choice for audio processing. This prevalence ensures that users can easily find resources, support, and expertise when working with MATLAB for audio-related tasks. It also promotes compatibility and interoperability across various domains.

5. MATLAB's commitment to innovation ensures that it stays at the forefront of audio processing technology. With regular updates and enhancements, users can expect continuous improvements in performance, accuracy, and functionality. MATLAB's strong community of users and developers contributes to a dynamic ecosystem where ideas and best practices are shared and refined.

Audio processing using MATLAB stands out as the superior choice due to its comprehensive toolset, user-friendly interface, scripting capabilities, widespread adoption, and commitment to innovation. Whether you are a researcher, engineer, musician, or audio enthusiast, MATLAB provides the ideal platform to explore, analyze, and manipulate audio data with unmatched precision and efficiency.

II. LITERATURE REVIEW

Rakesh kumar et al [1], in their research work has mentioned that in a tropical rainforest, hearing a species is considerably simpler than seeing one. Even if a person in the forest might not be able to see every kind of bird or frog there, he can hear them. If a common person visits a rain forest for an adventure, he may not even know how to recognise these species, let alone taking appropriate action against them. A forest ranger may know what to do in these situations and may be an expert in identifying the various types of insects and dangerous species that are out there in the forest. This study develops a model that can take an audio signal as input, analyse the signal intelligently to extract patterns and characteristics, and then output the species contained in the audio signal. The model is end-to-end functional and can handle raw input; moreover, a pipeline is developed to carry out all preprocessing operations on the raw input. This study tests several neural network designs based on convolutional This technology is becoming more widely available to consumers thanks to the expansion of the semiconductor industry and the spread of wireless communication, enabling engineers to carry out ever-darker projects like Smart Cities and Smart Farms equipped with mini unmanned aerial vehicles (UAVs) or drones for security and monitoring [2],neural networks (CNN) and long short-term memories (LSTM).

Jao paolo et al [2], in their research work has mentioned One of the most exciting areas of study and industry in the field of communications engineering right now is the Internet of Things (IoT) .This technology is becoming more widely available to consumers thanks to the expansion of the semiconductor industry and the spread of wireless communication, enabling engineers to carry out ever-darker projects like Smart Cities and Smart Farms equipped with mini unmanned aerial vehicles (UAVs) or drones for security and monitoring . In this study, a mesh network algorithm prototype for digital audio signal processing applications using IoT devices in Smart Farm was described. The test findings demonstrated the technique's viability and provided good outcomes for the intended use.

Arun Anoop [3] in his research work has mentioned that for the researchers, removing noise from the images may be a fun task. Edge detection is a picture-processing method for identifying the edges of objects in images. Edge find filters search for boundaries between completely dissimilar colours to identify object outlines. There are several image processing approaches that may be used to apply and eliminate signal- and noise-based issues in a picture. He looked at noise filters and edge filters in concept view and used MATLAB to do some analysis. The filter name, analysis output, and pseudocode are the foundations for analysis work. In his early study, he examined fundamental noise and edge filtering methods. he experimented with Gaussian noise, salt-and-pepper noise, speckle noise, poisson, median, wiener, sobel, prewitt, Robert, and Laplacian filters before calculating the PSNR (Peak Signal to Noise Ratio) number.

Archit Prakash [4], in his research paper has mentioned that adaptive filtering has garnered significant attention from researchers in the current decade, especially in the realm of communication. Adaptive noise cancellation represents an approach employed to reduce noise in speech signals. Since the received signal is persistently subject to noise interference, and both the received and noise signals undergo continuous changes, the necessity for adaptive filtering becomes apparent. This research paper focuses on the mitigation of noise in speech signals through the utilization of two adaptive algorithms: the Least Mean Square (LMS) algorithm and the Normalized Least Mean Square (NLMS) Algorithm. The selection of these algorithms is driven by their ability to deliver efficient performance while minimizing computational complexity.

Jayshree K.C. et. al [5], this work examines real-time audio signal analysis with the goal of reducing the noise in the message signal under analysis. The fundamental disadvantage of noise in an audio signal is that it lowers the quality of the signal being conveyed through the communication system. White gaussian noise (awgn) is added to the audio signal under consideration for analytical purposes, and the resulting noisy audio signal is then processed using various filtering techniques, including IIR, FIR, and Wavelet transform methods. When the aforementioned approaches are used, the noisy audio signal is analysed in relation to the various filter responses.

Tatsian et al [6], in their research work has mentioned that the article's subject matter revolves around the identification of musical instruments and playing techniques in music, with the primary goal of detecting and rectifying errors within a given musical excerpt. It elucidates methods to acquire the distinctive attributes of recorded sound and elaborates on the process of comparing amplitudes and frequencies within the same musical composition, albeit performed by different individuals and utilizing various instruments. To achieve this objective, the article employs signal processing algorithms readily accessible in commonly used Python libraries such as "numpy" and "scipy." The core concept behind these processing techniques lies in error detection while preserving the authentic playing technique and the unique artistic style of the musician.

Nivea Sharma et al. [7], in their research work has mentioned that In many signal processing applications, the FIR filter is a crucial tool for digital signal processing. In their study, MATLAB is used to generate sound effects such echo and Gaussian noise, and removal of those effects has been planned (analysis). One of the areas of audio signal processing that is expanding quickly is the application of digital signal processing. The creation of sound effects in real time is crucial and difficult. A band pass filter design is used to eliminate Gaussian noise. The applications of digital signal processing are the main topic of this essay. The most common use of DSPs is to modify audio signals in response to sound. The analysis of the MATLAB simulation's results shows how nois e and echo affect the audio signal. Noise may be reduced to some extent using low pass filters. The emphasis of their work is on the creation and implementation of the necessary effects for audio input.

Wulan Meiniar et al. [8], in their research work has mentioned that the filtering of human voice was carried out by reducing the frequencies of the human voice inside the musical frequencies. One of the filter designs employed in this research is the band-stop filter design. One sort of windowing, the Butterworth windowing, is used in the band-stop filter design. A band-stop filter design simulation in MATLAB is used as the approach for filtering human sound. The frequencies of the human voice are used to determine the required stop-band. The algorithm was created to remove human voices from audio files. The outcome is then tweaked to obtain the best outcome for the filtering. Once the heard voice has the best outcome, the optimisation methods are used to simulate the band-stop filter design with various stop-bands, FFT the output to demonstrate its frequency spectrum, and repeat the process. Based on the approaches, a comparison of different stop-band specifications is offered in order to determine which is the best. To determine whether the selected stop-band is appropriate for both types of voices, male and female sounds are filtered using the stop-band.

Priya Saboo et al. [9], in their research work has mentioned that technologies for audio and voice processing are developing as demand for high-quality speech increases. During audio capture, many noise kinds taint the audio. The original signal's noise content should be able to be removed using speech enhancement techniques without affecting the original signal. The human auditory system model has recently been included into a number of speech processing systems to attain excellent performance. In order to imitate the feature of signal processing carried out at the human auditory peripheral into the speech processing technology, consideration of the human auditory response and response of the basilar membrane will be made. Their proposed work focuses on the development of the auditory filter bank, a bank of filters where each filter employs the human ear response as the basis function, in MATLAB and Simulink. The quality measurement of the improved signal over noisy signal was also covered by the suggested study. Applications such as speech and music synthesis, voice feature extraction, audio coding, hearing aids, etc. would be significant.

III. CONCLUSION

In this review paper, different advancement in the speech filtering methods has been discussed. the diverse array of techniques available for audio processing underscores the dynamic and ever-evolving nature of this field. From the powerful capabilities of deep learning, which have revolutionized speech recognition and music generation, to the real-time enhancements improving audio quality in various applications, audio processing continues to push the boundaries of what is possible in the realm of sound. Content analysis and restoration techniques are enabling us to extract invaluable insights from audio data and preserve historical recordings, while spatial audio and 3D sound technologies are redefining immersive experiences. Moreover, ethical considerations surrounding data privacy and responsible use of audio data are gaining prominence as audio processing techniques become increasingly sophisticated.

As audio processing techniques continue to advance, they hold the promise of enriching our lives in countless ways, whether it's through clearer communication, enhanced music creation, or more immersive entertainment. The ongoing fusion of technology and creativity in this field assures that audio processing will remain at the forefront of innovation, shaping the auditory experiences of the future.

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