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The Enhanced Speech Recognition in Automated Home Lighting System using Adaptive Time-Frequency Domain Noise Removal Algorithm Filter

Sk Abu Baker Siddik*, Wan Nor Al-Ashekin Wan Husin and Thagirarani Muniandy

Abstract – **Numerous studies have explored speech recognition performance in Smart Home environments. However speech recognition accuracy diminishes when voice commands are captured in noisy areas of the home. This study aims to enhance speech recognition performance in such noisy environments. Instead of relying on remote control signals, a Bluetooth system is employed for short-range wireless communication to identify speech commands. Various sound levels are measured in decibels (dB) at different distances using the Smart Noise Application. A filter algorithm with Adaptive Filtering is used to minimize unwanted noise. The algorithm uses Adaptive Time-Frequency Domain Noise Removal (TFDNR) to mitigate background noise. Overall, the integrated system comprising Smartphone, Bluetooth, Arduino microcontroller, and noise detection software exhibits improved performance compared to previous studies, highlighting its potential for seamless smart home automation.**

Keywords- Speech Recognition, Bluetooth, Decibels, Noise, Adaptive Filter.

II. Introduction

In the present day, speech recognition technologies are developing in the area of Smart Home (SH) environments. The rationale of these standardized automation systems are to provide comfort to physically disabled and elderly people. With regard to comfort, the control elements of building automation can be adapted to the SH user's requirements [1] [2]. Home computerization and mechanization by using computer and phone technologies manage systems to reduce human labor. The rapid growth of technologies influences people to use Smartphone to control their home devices remotely. An automated device has the capacity to work with flexibility, determination and with the most minimal error rate. Voice or speech control for home automation systems not only reduces the need for human labor but also saves time and energy.

Speech control is one possible facilitator in the home automation domain. For a particular population, in particular elderly and disabled people, this voice control offers a significant advantage. People are able to turn the light on and off easily using voice or speech commands [4].

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Normally, every room in the house object has its acoustics which essentially distorts the spoken command. There can be different type of noise may occur in different room, which substantially reduce the success rate of recognition of spoken commands [3][4].

In the speech recognition process, background audio noise causes problems in getting actual commands in audio signal transmission. An audio noise reduction system is used to skip and remove the background ambient noise from the sound audio signals [4]. The voice or speech signal frequency range is 300-3400 Hz. On the other hand, Human hearing audible frequency range is 20Hz to 20000Hz. Sound recorded audio signal process frequently experience noise problems or troubles [5]. Voice background recorded noise is an unpleasant sound signal that is joined with voice signal at the time of transmission or at the time of creation of speech signal [6]. Although, this system is totally depending on the user to give the command, this is considered as a resource-type problem. The noise can be occurred any places in the room [7].

The Active noise control (ANC) which is also called noise reduction or Active noise reduction (ANR). ANR is a way or on the other hand strategy for keeping away from superfluous surrounding endless commotion that has not been handled by growing an ensuing sound that is explicitly intended to drop the ongoing one [8]. Sound is a simple and analog signal that works or operates on frequency, with compression or pressure phase and rarefaction phase.

In this study, the system uses the speech recognition command method to process the speech command within system. The system takes the command through the voice recognition method into Bluetooth-based Android phone then the method converts the command into voice-to-text form and sends it to the microcontroller using a wireless communication connection. Smart noise application will find the decibel level of the recorded sound. Another application from MATLAB will process the noise and speech using the Time-Frequency Domain Noise Removal (TFDNR) algorithm. The application will be able to capture the noise and command from the recorded file and give the best output. The TFDNR method that uses to develop this system can be helped to enhance speech command and performance of speech recognition.

II. RELATED WORKS

A. Bluetooth and Android Based Speech Controlled Wireless Smart Home

The system operates a few control factors to turn on or off the home appliances. This is a client-server-based speech control system for home automation where a voice command is captured by the client and the server system converts the command into the form that is used for home appliances to control. This system sets up costs is medium-low. The overall system has three different models or designs that operate home appliances such as Liquid Crystal Display (LCD), Motor driver, Light Emitting Diode (LED), Fan, and Emergency switch. The three designs are System design, Hardware design, and software design.

The system is directly attached to the wall and has a Renesas Microcontroller that takes the commands through Bluetooth devices. An Android phone is used to connect the device of the Renesas Bluetooth device with the Graphical User Interface (GUI) wirelessly. The remote control input is taken by the Android phone and passes the digitalized command into Renesas through a Bluetooth device. Once the commands are given, the microcontroller drives the motor driver and changes the state of the fan and LED [9].

The hardware system uses a Renesas microcontroller, R5F100 to establish Bluetooth connectivity with its serial interface features. Renesas microcontroller auto-generates code for the chosen ports, which is useful in coding, and no need to write extra code. Renesas Microcontroller manages at 32MHz. this microcontroller consists of 16 bits. The Random Access Memory (RAM) capacity is 4kb only. The input voltage of the Renesas is 12 Volts (V) but for the home appliances for 5V is enough to run the system. It is a 64-pin microcontroller but the uses only 58 pins, the other 6 pins are reserved. Here, the HC-04 Bluetooth module is used to establish connectivity between the GUI and the main control board. To turn on or off the light and fan, an integrated circuit Motor Driver L293D is used. The Motor is used to vary and control the brightness of the LED light and the speed of the fan. It requires 5V and 12V power supply that has a dual control two motors i.e. led light and fan with a single Integrated Circuit. It uses ‘1’ and ‘0’ logics i.e. if it is 1 (high), fan and led light is enabled, if it is 0 (low), fans and led light is disabled. LCD is used to display the command result executed by the controller.

At the software design smart Android app provides a user login page that can be accessed by a particular user. The Main Page appears after the login is successful. Then the user needs to connect the Bluetooth. If the connection is not enabled then it will show a message to turn on the Bluetooth. Once the Bluetooth is on, the user has to pair it with the HC-04 Bluetooth module circuit. The user can give the input button command after completing the Bluetooth connection. When the command is given to the phone, it is passed to the Renesas microcontroller which executes the command immediately [9].

B. Noise Annoying and Loudness: Acoustic Sound Performance of Residential Homes and Buildings

Acoustic issues connect with the view of noise. Individuals can hear frequencies of around 20Hz to 20,000Hz. Sound noise is close to subjective and one individual's sound music could be someone else's sound noise. Noise or Commotion is one of the most striking normal issues for individuals [10]. The major focus on acoustic comfort is influenced by factors, for instance, human sources (developments, steps, voice, TV, or radio), aggregate hardware (lifts, radiators, climate control systems or transformers), individual gear (condo warmers and clothes washers), homegrown gear, and open-air commotion (transports, cars, airplane clamor, rail route, or modern commotions The transmission of sound waves through walls, windows, channels, shafts, and openings, and the transmission of vibrations through the arrangement will finish up the sound decibel pressure level, accomplishing a room piled up with sounds from both the indoor and outside sources. Sound commotion could influence occupants' productivity and prosperity.

The sound noise loudness is unbiasedly quantifiable with reasonable fitting hardware anyway the disturbance or irritation to inhabitants can't be estimated clearly or straightforwardly. Much of the time alluded to as a Sound Strain Level (SPL) meter, commotion meter or noise meter, decibel (dB) meter, mobile application decibel understanding meter, and a sound level meter that utilizes a receiver to catch sound and noise. From their exploration, traffic noises are the most undesired sound noise which is 49.11%, followed by the sound noise from neighbors. Even more inquisitively, the aggravation of traffic commotions and neighbor's noise are connected. Yet, there's actually a research hole in this review. The paper just shows the catching sounds from a specific close distance. It is not cleared that the sound noise captured device was located where in the room or how much distance it has to capture the sound [11].

C. Noise Cancellation Using Wiener Time and Frequency Filter & Adaptive Time and Frequency Filter Technique in MATLAB

Wiener Time-Frequency Filter for noise reduction- The spectral-subtractive algorithms depend generally on instinctive and heuristically based standards. All the more explicitly, these calculations took advantage of the way that noise is added substance, and one can acquire an estimate of the clean perfect signal spectrum essentially by taking away or subtracting the noise spectrum from the noisy speech spectrum. The improved signal range or spectrum was not determined in an ideal way. The Wiener filters move toward inferring the improved signal by enhancing a mathematically tractable error criterion, the mean-square error [12].

In voice enhancement programs, the given signal y (n) is the speech signal that is combined with noise:

$$y\,(n) = x(n) + n(n) \qquad (1)$$

Here, x(n) means the clean Speech signal and n(n) remains the noise signal while d(n) is the clean (noise-free) desired signal x(n), that is, x(n)=d(n).

Adaptive Time-Frequency Filter- The Time Frequency Representation (TFR) is a perspective on a signal that is taken to be an element of time addressed throughout both time and frequency. Time–frequency analysis implies investigation into the time-frequency domain provided by a TFR. The primary issue with the decent time-frequency resolution Short Time Fourier Transform (STFT) is the spreading of signal energy. Spreading in frequency can forestall recognizing firmly separated sounds and spreading in time can influence assessment of positions and terms of drifters. At the point when utilized with regards to source partition, this spreading precludes exact altering in the time-frequency space and can block execution. [13].

When an over-complete set of L basis functions:

$$gk[n] = h[n]Wkn\,L\,, 0 \le k \le L - 1 \qquad (2)$$

Where h[n] is a reasonable genuine esteemed window with a low-pass trademark and with length N (N<L) that confines the fundamental capabilities in time and WL = e2πj/L. A time-domain signal x[n] is planned to change space by the convolution among x[n] and gk[n].

Adaptive Time-Frequency filter is often considered superior to Wiener time and frequency filter in certain situations due to its ability to adapt to changing noise conditions and better preserve important signal features. Adaptive filter excels in scenarios where the noise characteristics change over time and frequency. This adaptability allows it to effectively track and reduce non-stationary noise, which can be challenging for Wiener filter that assumes stationary noise.

Adaptive filter is designed to maintain or enhance speech intelligibility by selectively attenuating noise while preserving essential speech features. This is particularly important in applications where clear communication is crucial, such as speech recognition, telephony, and broadcasting. In environments with high levels of background noise, echoes, reverberation, and other complex acoustic factors, adaptive filters can outperform Wiener filter by focusing on the specific characteristics of the noise and signal components.

Research has shown that Adaptive filter can handle a wider range of noise types and levels, making them versatile for different environments without requiring manual adjustments.

III. METHODOLOGY

The MIT App Inventor is used to create an Android application to operate the speech command method whereas the Arduino IDE is used to handle the microcontroller. The Time and Frequency Algorithm is used in MATLAB program to capture and filter the audio noise.

A. Android and Arduino

Android and Arduino are used to capture speech and process speech commands. The methods of Android and Arduino consist of the following:

- The User has to open the installed "Siddik_speech.apk" application from the Android Smartphone.
- With the help of the phone's Bluetooth enable option the users need to search Bluetooth device HC – 05.
- The user needs to connect the Arduino UNO to the Personal Computer (PC) or laptop with a USB to enable the power supply and sketch the code to the Arduino UNO microcontroller.
- The Arduino microcontroller and Bluetooth module need to connect each other with the four normal copper wires. There are another three wires connected with lights and Arduino UNO microcontroller.
- The user needs to compile and upload or sketch the Arduino code "Siddik_light" into the Arduino microcontroller board UNO.
- Once the code is uploaded, the user needs to establish or connect the Bluetooth within the Smartphone and Bluetooth HC – 05 module device to communicate.
- The user has to open the "say something" option from the phone app and give some commands to operate. Whatever command is given to the system that captured command will appear in the application in the form of text.
- That captured command will pass into the Arduino through wireless Bluetooth. At the end the Arduino system will operate and execute the signal to the light and the light will be on or off according to the command signal.
- Whenever the user gives a command the light will change its state because this is a continuous process.
- On the other hand, MATLAB takes the input of the speech command recorded file and checks the noise from the audio file.
- Once the audio file is operated the method of noise reduction, that method reduces the noise from the audio file and produces a new audio file.

B. Smarter-noise

The data testing and collecting of this section is the critical part of all. The sound-noise pressure level below 25 decibels causes hearing loss for a person and the system. Pressure levels between 25 to 85 decibels are the standard level for a person and system to understand and above 85 decibels cause hearing loss and sound pollution. The author has tested and collected the sound noise data for normal conversation, vacuum machine, scream noise, motorcycle, train, jack hammer and jet engine. The data is measured by smarter noise application near different noisy devices in different situations or distances.

C. MATLAB System

MATLAB's Adaptive Time-Frequency Domain Noise Removal (TFDNR) filter excels in processing signals in both the time and frequency domains simultaneously. By analyzing the signal in the time-frequency domain, it can effectively capture non-stationary noise components, transient noise, and time-varying noise sources. The key advantage of the adaptive TFDNR technique is its ability to dynamically adjust its filtering parameters based on the input signal's characteristics. This adaptability allows the filter to continuously update its coefficients, making it highly effective in tracking changes in the noise environment.

MATLAB noise-checking part takes a speech input and checks or filters the noise that is captured in the command record. It will compute the noise in the given command time and show the result after reducing noise using the graph. The TFDNR algorithm method subtracts the static normal sound from the noisy combined sound and gets the noise.

$$\mathrm{L\_clear} = \mathrm{L\_combo} - \mathrm{L\_static}; \qquad (3)$$

Where L_clear is the noise, L_combo is the combined noisy sound and L_static is the normal sound.

There are two different recorded files are tested for each conversation or situation. One audio file is combined with sound and noise which is called a combo file. Another audio file is the normal record which is called a static file.

- First of all, the user needs to open the MATLAB software in his or her computer.
- Once the MATLAB is open the user has to launch the program named "noiseremoval.m".
- In the program a combo recording and a static recording need to be added manually.
- After adding the recording, the program will run and process the noise removal subtraction algorithm calculation.

- Last off all, the program generates the results with five visual graphs named “original recording with noise”, “recording after noise removal” with amplitude and time (in seconds) and also “original recording with noise”, “background noise recording” and “recording after noise removal” with file’s amplitude and frequency. The figure 1 gives the clear image of process.

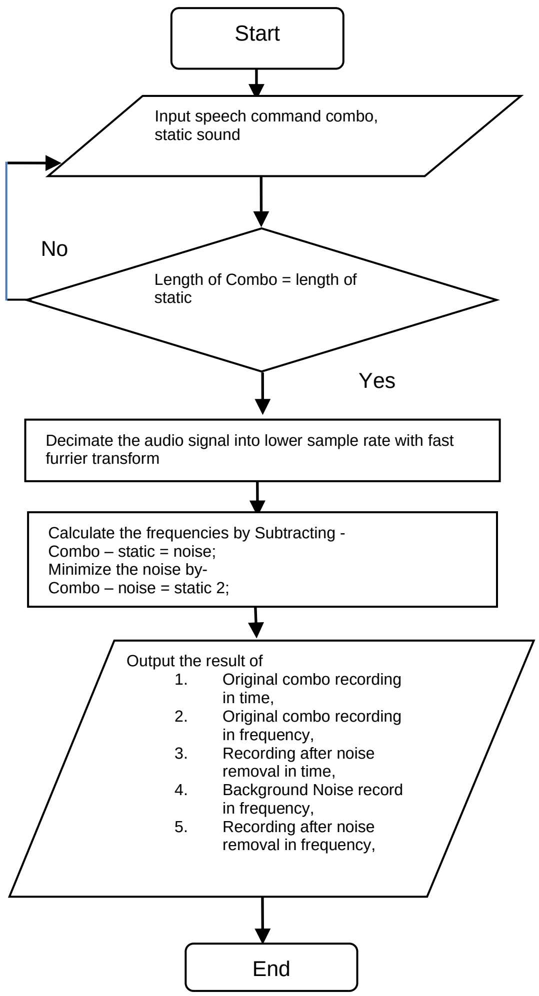


FIGURE 1. Flow chart for MATLAB sounds noise checking.

D. Overall system

The user should launch the Android Application which helps the user to connect the phone with HC-05 module Bluetooth device. The HC-05 module must have to connect with the microcontroller board which is wired to the Arduino Uno device. However, the Arduino UNO needs to be programmed with the Arduino IDE. Once the Bluetooth connection is established between the Android phone and the Bluetooth Module, the Android phone will be considered as a Client and the Bluetooth module and Arduino system will be considered as a server. The user must have to give a command through the phone using the record button shown in Figure 3.

The recorded file has to be with time and frequency filtering. The MATLAB program calculates the audio file’s signal at a lower rate through the Fast Fourier transform. It will subtract the static audio file from the combo audio file. Once it gets the noise it will reduce the noise from the combo file and make the file to use in the Android phone.

The Android phone client always gives the command and the Arduino server system detects the command and produces the command into signal. The programmed Arduino UNO board processes the digital command and converts it to an analog signal. The Arduino-programmed algorithm method helps to match the text command into the system. After that, the system sends the actual signal to the appliances. If the appliances receive the right signal, then the LED light is turned on or off.

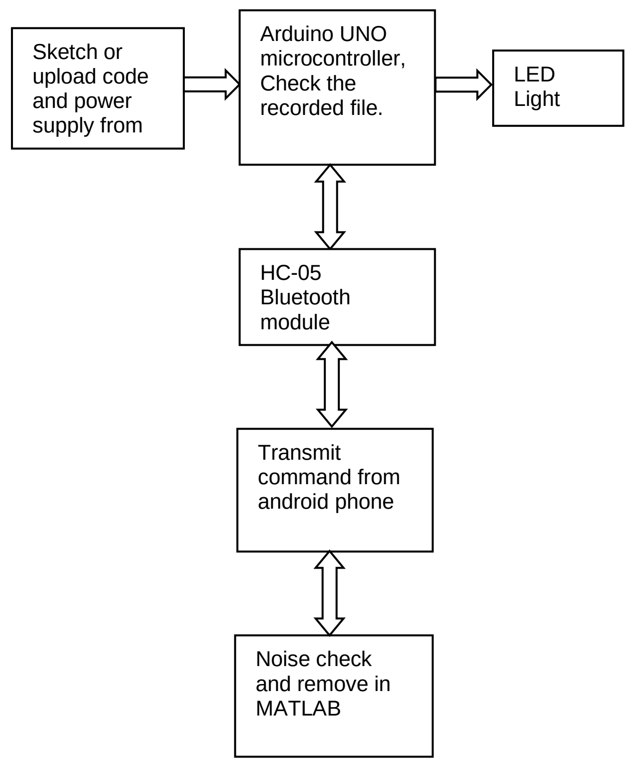


FIGURE 2. Overall Block diagram for the home automation system.

Figure 2 is the overall system’s block diagram where the Arduino UNO microcontroller connects with the computer using a USB cable that provides the chips to upload code and power supply. On the other side, there is an Android phone that takes the recorded filtered command and transfers the command to the microcontroller using HC-05 Bluetooth module. The microcontroller also connects with the LED light which turns on or off after getting a signal.

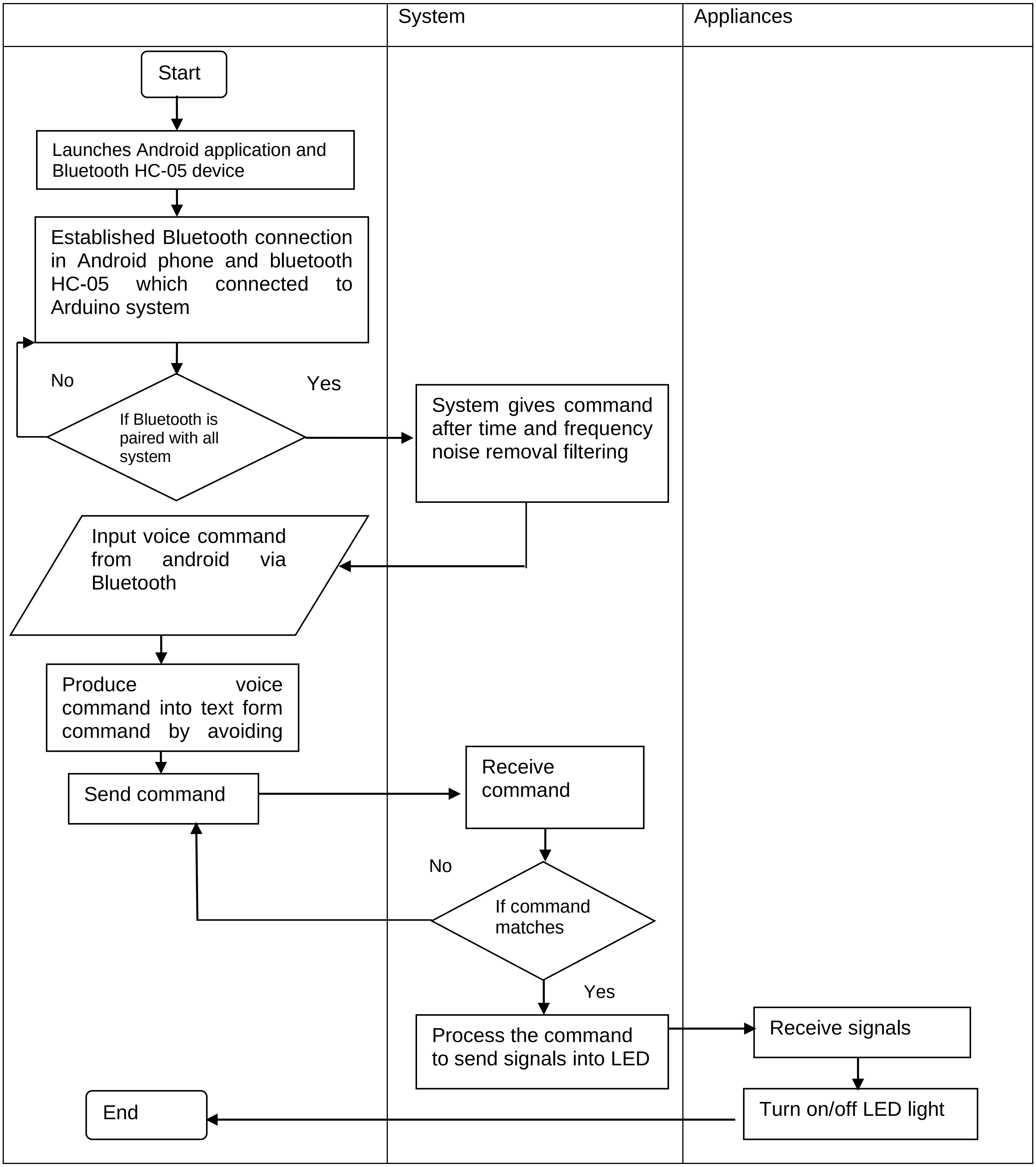
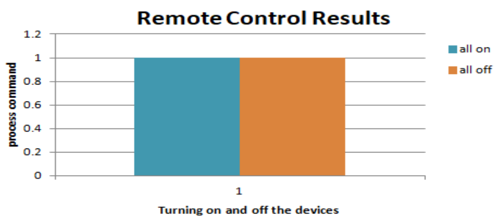


FIGURE 3. Flowchart for Speech Recognition Smart Home Automation system. This chart is showing the whole operational framework of user, the system and home appliances such as LED light.

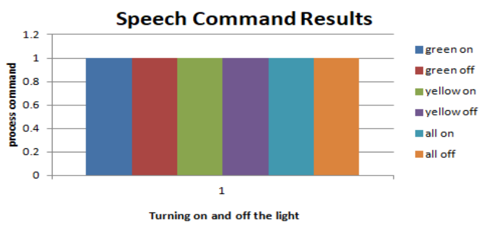
IV. Result and Discussion

A. Android and Arduino

The author's findings indicate that the existing method relies on a remote control system where all devices are commanded to turn on simultaneously. In contrast, the author's speech recognition method transforms the system into a speech recognition system. This new approach allows users to give individual or all light commands through an Android application, making the system act accordingly. Table II's noisy positive 1 sound file works smoothly in this voice recognition system. The best advantage of the speech recognition system method is that it can turn on lights without the need for a remote control, thus saving human labor and creating a smart automation system.



(a)



(b)

FIGURE 4. (a) Existing remote control method result. (b) Speech control method result.

B. Smarter-noise

For a human being, the normal decibel understanding level is 30 to 60 decibel. But at the same time devices such as phone or computer microphones hearing and understanding could be different as 30 to 150 decibels. People can have hearing problems and health issues if the sound pressure level is above 60 dB [11]. The machine can understand the decibel level up to 150. It captures all the sound noises. It is very challenging to skip the noise from the noisy recorded file. Table 1 shows some of the device's noise sound detection in decibels at different distances. The normal conversation sound, vacuum device, scream sound, motorcycle engine sound, train engine sound, and jackhammer device sounds have been captured to understand and skip the noise for the MATLAB system.

TABLE 1. Details of close range and zero, two ,four, eight, ten and twenty feet average recording for normal conversation, vacuum machine, scream sound, motor cycle engine, train engine, and jack hammer.

Average feet	Normal conversation	Vacuum machine	Scream sound	Motor cycle engine	Train engine	Jack hammer
Zero feet db	46.9	73.8	78.8	85.2	95.5	95.1
Two feet db	59.0	72.4	78.4	83.3	97.0	94.7
Four feet db	58.8	69.2	77.2	79.7	95.7	94.2
Six feet db	55.2	68.0	76.2	73.7	94.1	93.9
Eight feet db	53.3	67.8	75.2	72.3	91.6	93.3
Ten feet db	47.6	67.5	75.1	70.9	90.0	92.7
Twenty feet db	43.3	59.8	70.0	63.2	70.4	88.5

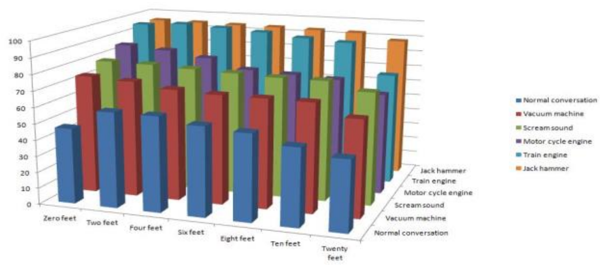


FIGURE 5. 3D formation comparison of decibel sound noise.

C. MATLAB

The amplitude and time-frequency differences and comparison are shown in the below graphs. There are two useful speech signals are recorded for checking and testing. The first speech signal is a noise speech signal and the second one is a speech signal without noise. The both signals are very high using Adaptive Normalize filters. For better understanding, the white noise is taken and analyzed [13].

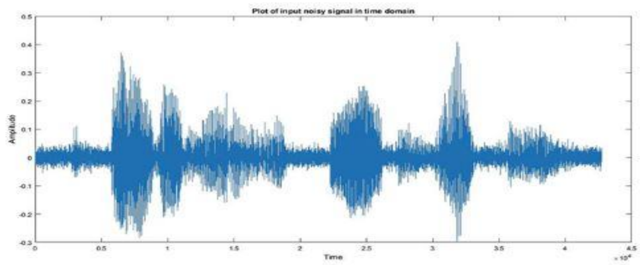


FIGURE 6. The graph is the results of Noisy signal.

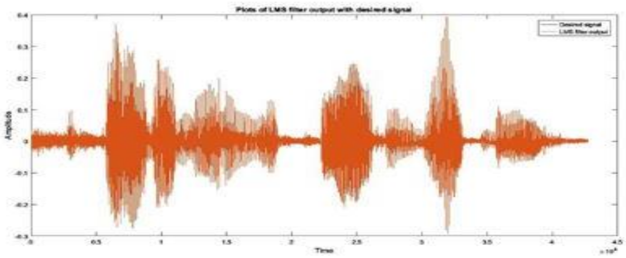


FIGURE 7. The graph is the results of filtered signal with desired signal.

This study has investigated Table I's noisy sound file with a modified Adaptive Time-Frequency Domain Noise Removal (TFDNR) filter subtraction algorithm and found impressive results. The Time-Frequency Domain Noise Reduction filter result for speech recording with noise, background captured noise, and noise removal graph for normal conversation white noise record is shown in Figures 8, 9, and 10.

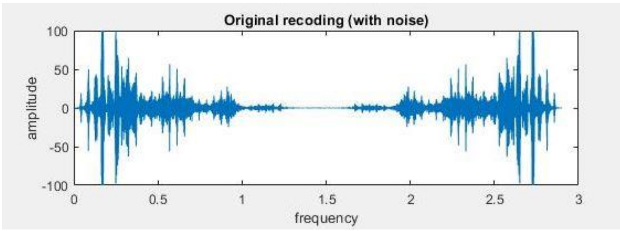


FIGURE 8. Time-Frequency Domain Noise Reduction filter result of original recording with noise.

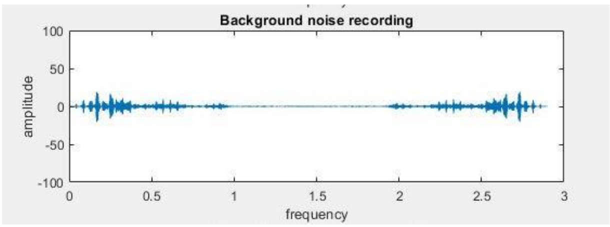


FIGURE 9. Time-Frequency Domain Noise Reduction filter's result of noise detection.

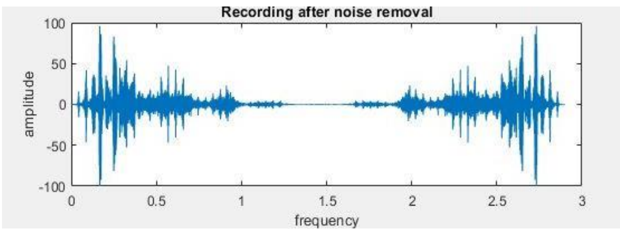


FIGURE 10. Time-Frequency Domain Noise Reduction filter's result of sound after noise removal.

The current system's utilization of the TFDNR filter is yielding 0.89 % better results than the previous study's Adaptive filter. The TFDNR filter capitalizes on the time-frequency domain to address the inherent limitations of time-domain filtering methods. It effectively addresses issues like transient noise and time-varying noise by performing noise reduction in both the time and frequency domains simultaneously. By taking advantage of this comprehensive approach, the TFDNR filter can significantly improve the signal quality and intelligibility shown in Figure 10. Table 2 shows the significant result of normal conversation, vacuum machine, scream sound, motor cycle engine, train engine and jackhammer sound that capture in zero to twenty feet distances.

TABLE 2. MATLAB sound noise understanding in positive and negative formation.

Sound	Zero feet	Two feet	Four feet	Six feet	Eight feet	Ten feet	Twenty feet
Normal conversation	1	1	1	1	1	1	1
Vacuum machine	1	1	1	1	1	1	1
Scream sound	1	1	1	1	1	1	1
Motor cycle engine	-1	1	1	1	1	1	1
Train engine	-1	-1	-1	-1	-1	-1	1
Jack hammer	-1	-1	-1	-1	-1	-1	-1

Table 2 provides insights into the subjective preferences and impacts of different sound sources at varying distances. Here, for MATLAB's TFDNR program below 85 dB sounds are readable and editable which are considered positive values and easily understandable. But, above 85 dB are considered as negative values which are sometimes understandable but mostly not understandable. A value of "1" signifies a positive preference and impact, indicating that the sound is generally perceived favorably and has a positive effect. On the other hand, a value of "-1" denotes a negative preference, suggesting that the sound is generally perceived unfavorably and has a negative effect. Figure 11 provides the percentage graphical representation of Table 2.

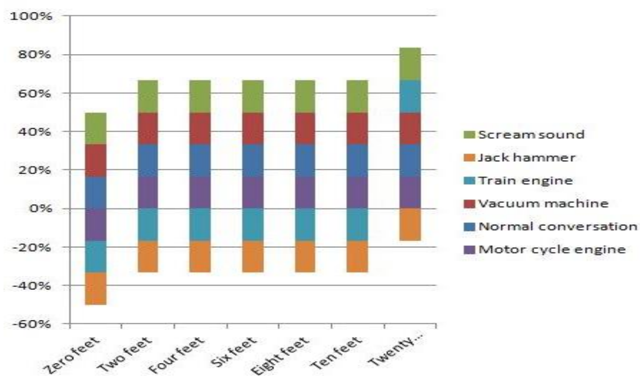


FIGURE 11. MATLAB program's utilization comparison.

The information suggests the performance of the program in terms of working fine or not fine at different distances. Here in the graph 0 percent is the neutral point where above 0 means the program outputs the expected best results, but below 0 means the program cannot provide good results in MATLAB. The previous study has shown a success rate of

49.11% in traffic noise. The noise sound is captured from a very close range of zero feet distance [11]. The proposed program's performance evaluation in a noisy environment reveals distinct outcomes at varying distances: at zero feet distance, the success rate is 50%; within the range of two to ten feet, the rate rises to 65%; and notably, for distances spanning above 10 to 20 feet, the program achieves its highest success rate of 85%. Furthermore, consistent with prior studies and existing literature, these results reinforce the trend that greater distances correlate with improved program performance, substantiating the program's effectiveness across a range of scenarios.

V. CONCLUSION

The Android speech recognition home automation is much better than the remote-control system. It is more time and work-efficient. It helps the elderly and disabled people to work without moving from their place in the home. The decibel level that is described in the results and discussion shows the best results to operate the system. The details analysis provided in the results part of MATLAB’s, the TFDNR filter enhances the understanding of the noise reduction process and establishes it as a promising solution for improving the quality of recorded audio signals, making it a key component of a smart home automation system. The overall system of Smartphone, Bluetooth, Arduino microcontroller, Smarter Noise detection, MATLAB noise removal program operates properly and outputs better results than the previous study. But for future work, a researcher may develop the system into one single program like every coding and phases are works in one application.

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