# HOME SECURITY SYSTEM USING ML

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## ABSTRACT

Face recognition system is broadly used for human identification because of its capacity to measure the facial points and recognize the identity in an unobtrusive way. The application of face recognition systems can be applied to surveillance at home, workplaces, and campuses, accordingly. The problem with existing face recognition systems is that they either rely on the facial key points and landmarks or the face embeddings from FaceNet for the recognition process. In this paper, we propose a hierarchical network (HN) framework which uses pre-trained architecture for recognizing faces followed by the validation from face embeddings using FaceNet. We also designed a real-time face recognition security door lock system connected with raspberry pi as an implication of the proposed method. The evaluation of the proposed work has been conducted on the dataset collected from 12 students from Faculty of Engineering and Technology, University of Sindh. The experimental results show that the proposed method achieves better results over existing works. We also carried out a comparison on random faces acquired from the Internet to perform face recognition and results shows that the proposed HN framework is resilient to the randomly acquired faces

## INTRODUCTION

With the evolution of systems getting smarter through the integration of artificial intelligence technologies, the ways to sabotage those systems are also gauging, simultaneously. Specifically, in the security and surveillance applications, relying on a uni-modal system for reliable monitoring is not recommended [1]. Security problems are given high priority because every business owner strives to keep their organizations, assets, and workplace as sheltered as could and same goes for homes [2]. In this way, the security does matter in everyday life. One of the main reasons for compromise of security is the unauthorized access to strangers [3]. The old door security systems made use of keys, locks and chains. However, the locks can be easily broken. The use of keys to unlock the doors is not efficient sometimes, because the keys may be sometimes used by the wrong person or keys can get stolen or can be duplicated [4]. Then came the era of shallow learning algorithms with uni-modal systems which could accommodate a single biometric trait at a time to ensure the authorized access [1]. The most significant aspect of any door security framework is recognizing accurately the people who want to access the entryway; however, uni-modal systems fail to achieve that benchmark [5]. With the evolution of devices, the one thing which needs to be adapted is the pervasiveness and unobtrusive nature of acquiring biometric trait suggesting that the user should not be fatigued when requesting the authorization. Using iris recognition, and complicated traits such as Gait and signature, require the user to perform some tasks which are specific to the authorization system. Even with fingerprint recognition one has to place the finger on the device to request the access. The face modality is the only trait which can be used for a security system that complies with ubiquitous characteristics [6]. Another aspect which needs to be explored is the single-tier recognition system which results in false positives and can be spoofed with the evolutionary methods [7]. Such as deep fakes [8], however, these methods are designed to fool the single-tier systems which does not comprehend to the diversified information. A hierarchical multi-tier system needs to be proposed which is trained on diversified data to cope with the aforementioned issue [7,9,10,11,12]. There have been a lot of face recognition studies using deep learning techniques which can be considered as state-of-theart methods. However, these methods heavily rely on a singletier of recognition using the features projected in embedding space. Randomly obtained faces from Internet sometimes are able to make the networks believe that they reside in the pool of authorized users. It is therefore, necessary, to add a secondtier of authorization to improve the performance and add resilience to the recognition system. In this regard, we propose a two-tier hierarchical network (HN) architecture which uses a discriminative learning method followed by FaceNet to perform the face recognition. Furthermore, we also built a prototype which takes into account the multi-mode of recognition, i.e. recognition from embedded system followed by the authorization from the home owner via e-mail. In this way, our system reduces the probability of false positives, accordingly. Moreover, the implemented embedded system provides a realization to the proposed study. The contribution of our proposed work are as follows: 1. We propose a hierarchical network (HN) framework to improve the face recognition performance. 2. We built a prototype to show the implication of proposed work. 3. We show that the HN framework is resilient to the randomly acquired facial images from the Internet.

## LITERATURE REVIEW

Sound is a biometric feature used to distinguish people or species from one another. By processing an existing audio signal, it is possible to detect whether this sound comes from a human or another object. Since voices differ from one person to another, they can be used for voice recognition purposes. It is also possible to control or direct various devices through words obtained from sound signals. Therefore, the processing and use of an audio signal is very important [1]. Hearing is a very important sensory function for people. Developing a device that can perceive and classify various voices at home, and thereby improve the quality of life for hearing impaired people, is regarded as a basic requirement. A fire alarm or a phone alert that warns against danger are some of the sounds that should be perceived to encourage an urgent action. In recent years, there have been many developments in the field of wearable technology, both academically and industrially [2]. Due to factors including the high cost of modern health care, development of microcontroller technology, reduction in sensor dimensions with respect to the development of technology, and significant developments in wearable processing, applications of wearable technology products in the healthcare field are becoming more frequent. Recently, health practices have been widely used in civilian life in both patient follow-up and treatment. Glove-operated surgical robots that facilitate doctors' operations are a current example of wearable technology in the field of health care [1,3]. Using wearable objects and specialized signal processing methods presents new opportunities and visualizations for applications. Development of technology has made it possible to produce low cost, portable, remote-access patient monitoring systems. Real-time data analysis can be made by these systems, and they can be worn directly on the human body [4]. In recent years, research, especially on speech data, has gradually increased to meet the demands of developing world. This is because speech data such as audio and voice data are those that are closest to human life, and that they can best express daily life. Recently, research on audio-based systems and deep-learning technologies has increased rapidly [5]. The first studies in this area predominantly addressed the content-based general sound classification problem [6–8]. Various methods have started to be used in communication between people and machines in parallel with developments in technology. One of the methods developed for communicating between human and machine is voice recognition. A general structure of sound recognition systems is given in

Figure 1. First, speech/voice is exposed to the step of preprocessing. Generally, sound is made more understandable in this step. In the step of feature extraction, distinctive features of sound data are determined. Which sound data are new sounds is determined

- Ma et al. [9] presented an adaptable system for classification of environmental sounds, as well as experimental results of this system. They used the Mel Scale Cepstrum Coefficients (MFCC) method in feature extraction from audio files of environmental sounds, and used the Hidden Markov Model (HMM) algorithm as a classifier in model training. Additionally, they performed a separate test, where people listened to audio files. The overall success rate was reported at 92%. When these tests were applied to people, the correct detection rate of the classes remained at 35%. Eronen et al. [10] conducted a study on the identification of environmental sounds. In the proposed system, HMM and K Nearest Neighbors (KNN) algorithms were used as classifiers for model training. While the performance of the system of environmental sound recognition was 69%, the success rate declined to 58% when the test was performed on real people. Su et al. [11] designed a system for ambient sound detection. They used the Local Discriminant Bases technique in their study. With this technique, the distinction of time–frequency subspaces for ambient sounds is defined. Evaluation of their study indicated that they obtained 74.3% and 81% accuracy rates. However, when audio with background noise in the dataset was used, the overall accuracy percentage declined to 28.6%. Wang et al. [12] proposed a system classifying 12 ambient sound voices. They applied Support Vector Machine (SVM) and KNN methods as classifiers in model training. Reed and Delhome [13] researched whether environmental sounds can be recognized with the help of a tactile wearable. Chu et al. [14] studied identification of ambient sounds. By separating the sound signals into different levels, this study showed different representations of the same signal. They reported that sounds can be better defined in this manner.
- Alias et al. [15] presented an up-to-date review of the techniques for revealing features of developed speaking, music and environmental sounds to analyze the most common sound signals in their studies. Shin et al. [16] studied coughing for the detection of a patient's unusual health conditions. This application, designed in real-time, can follow a medical condition and detect abnormalities using acoustic information. A mixed model consisting of artificial neural networks (ANN) and HMM was used during classification to distinguish the cough sound from other voices in the environment. This proposed mixed model yielded better results than conventional systems designed using the HMM classifier and the MFCC feature. Schröder et al. [17] proposed a system for the detection of ambient sounds. Their study consisted of a two-layer GMM classifier. Niessen et al. [18] proposed methods for classifying ambient sounds. They used different methods, such as MFCC and Zero Crossing Rate (ZC), for feature extraction from audio files. They reached an accuracy of 34.51% resulting from evaluations based on a sound clip. Kugler et al. [19] focused on grounded applications, and proposed a voice recognition method designed for ambient voices. Jeyalakshmi et al. [20] offered a speech recognition system for normal hearing and hearingimpaired children. This system was developed using MFCC feature extraction and HMM. Its accuracy was determined to be 92.4% for hearing impaired speech, and 98.4% for normal speech. This system can be used by others to recognize speech from those who are hearing-impaired. Sakajiri et al. [21] developed a voice step control system with the aim of helping their songs via a touch screen for deaf in their studies. Kingsbury et al. [22] studied a robust speech recognition system, free of noise, using a modulation spectrogram. They classified the cleaned sound samples with a modulation spectrogram using ANN and HMM. Reynolds et al. [23] developed a speaker-recognition system that makes authentication. The Gaussian method was used for recognition in the system. Lozano et al. [24] developed an application to resolve daily issues that could lead to serious problems for hearing-impaired people. Oberle and Kaelin [25] submitted a signal-recognition system for touch-sensitive hearing instruments using HMM. Beskow et al. [26] developed a Home Hearing (HAH) project in their study. This HAH is an innovative media center solution that provides real-time speech and reading support with a variety of integrated features including audio raise, audio reduction, audio classification and event perception. These features support speech and audio perception for hearing support. The average accuracy of this classifier was found to be 82%. Seoane et al. [27] developed a project to conduct speech processing with wearable biomedical measurement systems. Shull and Damian [28] in their review determined that a wearable

devices with a function for hearing loss has been developed. Deep Neural Network (DNN), which has manyf hidden layers and is trained using new methods, has sometimes shown better performance than Gaussian mixture models (GMMs) with a big difference in various speech recognition criteria [29]. Salomon and Bella [30] proposed a DNN architecture for the classification of ambient sounds. Dahl et al. recommended a new content-dependent model for speech recognition with broad identification. This model takes advantage of the latest developments in the use of DNN for phone recognition [31]. Deng et al. presented experimental proof that DNN with MFCC is outstanding in terms of speech spectrogram features

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**EXISTING SYSTEM:**

In existing system, we used Fingerprint, passwords and RFID Technology for opening doors of homes. In Fingerprint we need add fingerprint of the users and it should match. For RFID Tag is compulsory for opening and closing of doors. For password we need to remember password. These technologies have some drawbacks.

**Drawbacks:**

- Time Taking Process
- May not be accurate
- Passwords can hack
- Cards maybe lost
- Sometimes fingerprints cannot be taken

**PROPOSED SYSTEM:**

In this Project is very simple. A PIR sensor is used to detect the presence of any person and a web (usb) Camera is used to capture the images when the person presence it is detected.

Whenever anyone or intruder comes in range of PIR sensor, **PIR Sensor triggers the USB Camera through ESP32**. ESP32 sends commands to usb camera to click the picture and save it. After it, ESP32 is used to open door means ON the motor. Here the pictures are saved in ESP32 with the name which itself contains the time and date of entry

**IMPLEMENTATION**

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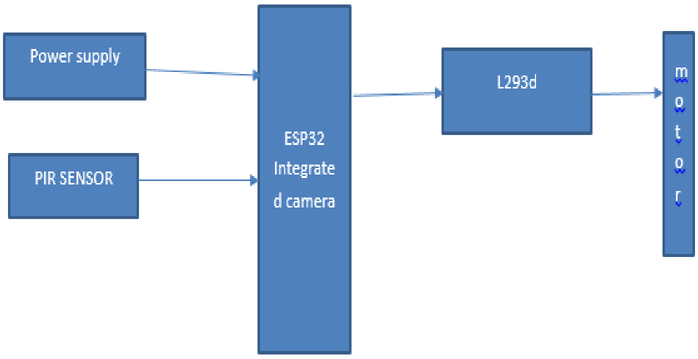


**DESCRIPTION**

**ESP32**

ESP32 is a single 2.4 GHz Wi-Fi-and-Bluetooth combo chip designed with the TSMC ultra-low-power 40 nm technology. It is designed to achieve the best power and RF performance, showing robustness, versatility and reliability in a wide variety of applications and power scenarios. The ESP32 series of chips includes ESP32-D0WD-V3, ESP32-D0WDQ6-V3, ESP32-D0WD, ESP32-D0WDQ6, ESP32-D2WD, ESP32-S0WD, and ESP32-U4WDH, among which, ESP32-D0WD-V3, ESP32-D0WDQ6-V3, and ESP32-U4WDH are based on ECO V3 wafer

**PIR SENSOR**

PIR Sensor is short for passive infrared sensor, which applies for projects that need to detect human or particle movement in a certain range, and it can also be referred as PIR(motion) sensor, or IR sensor. Since its powerful function and low-cost advantages, it has been adopted in tons of projects and widely accepted by the open-source hardware community for projects related to Arduino and raspberry pi. All this can help the beginners learn about PIR sensor more easily.

**D.C. Motor:**

A dc motor uses electrical energy to produce mechanical energy, very generally through the interaction of magnetic fields and current-containing conductors. The reverse process, producing electrical energy from mechanical energy, is carried out by an alternator, source or dynamo. Many types of electric motors can be run as sources, and vice verse. The input of a DC motor is current/voltage and its output is torque (speed).

**DC Motor**

The DC motor has two basic parts: the rotating part that is called the armature and the stable part that includes coils of wire called the field coils. The stationary part is also called up the stator. Figure shows a depict of a distinctive DC motor, Figure shows a picture of a DC armature, and Figure shows a picture of a distinctive stator. From the picture you can see the armature is made of coils of wire wrapped around the core, and the core has an covered shaft that rotates on charges. You should also notice that the ends of each coil of wire on the armature are finished at one end of the armature. The outcome points are called the commutator, and this is where's brushes make electrical contact to bring electrical current from the stationary part to the rotating part of the machine

**CONCLUSIONS** The proposed system tests only the human face images resulting from the Viola-Jones algorithm. The proposed system can recognize a person who is in the database by projecting the face vector to the base vectors. The recognition of the test face image and the training prototype is calculated between the resulting projection coefficients, which are used as representation of each face image feature. If the match is high, then we have the best recognition. From the results we can conclude that the look should not be at the center of the image so that the person can be recognized, and can be laterally oriented at an angle of 45°. The access control system allows automatic opening/closing of a door immediately after the identification of the person is acknowledged, and after a preset period of time (2 seconds), the door closes automatically. The edge sensors of the door control system (optocouplers) are used to stop the engine if the door reaches one of the two extreme positions.This switch functions mainly with an algorithm that is loaded into the microcontroller and based on data received through the serial port from the PC after face detection.

**REFERENCES**

[1] Li S.Z., Jain A.K.: "Handbook of face Recognition", Springer New York, 2005, Ch. 1, pp. 1-11.
[2] Choi Y.-K., Kim K.-M., Jung J.-W., Chun S.-Y., and Park K.-S.: "Acoustic intruder detection system for home security", IEEE Trans.Consumer Electron., vol. 51, no. 1, pp: 130-138, Feb. 2005.
[3] Zuo F., and P. H. N. de With: "Real-time embedded face recognitionfor smart home", IEEE Trans. Consumer Electron., vol. 51, no. 1, pp.183-190, Feb. 2005.
[4] Ibrahim R. and Zin Z. M.: "Study of Automated Face Recognition System for Office Door Access Control Application", IEEE 3rd Conference on Communication Software and Networks, pp. 132-136, May 2011.
[5] Kim D.-J., Chung K.-W., and Hong K.-S.: "Person Authentication using Face, Teeth and Voice Modalities for Mobile Device Security", IEEE Trans. Consumer Electron., vol. 56, no. 4, pp. 2678-2685, Nov. 2010.
[6] Kim J.-M.; Kang M.-A.: "A Study of Face Recognition Using the PCA and Error BackPropagation", Intelligent Human-Machine Systems and Cybernetics (IHMSC), 2010 2nd International Conference on , vol. 2, no., pp.241-244, 26-28 Aug. 2010.
[7] Kamencay P., Breznan M., Jelsovka D., Zachariasova M.: "Improved face recognition method based on segmentation algorithm using SIFT-PCA", Telecommunications and Signal Processing (TSP), 2012 35th International Conference on , vol., no., pp.758-762, 3-4 July 2012.
[8] Mustafah Y. M., Azman A. W.: "Out-of-Plane Rotated Object Detection using Patch Feature based Classifier", Procedia Engineering, Volume 41, pp. 170- 174, ISSN 1877-7058, 2012.
**[9]** Gîlcă G., Bîzdoacă N.: "A Biometric Face Recognition System using an Algorithm Based on the Principal Component Analysis Technique", Robotica & Management, Editura Eftimie Murgu, Reşiţa, ISSN-L: 1453-2069, Vol. 20, No. 1, pp. 40-43, June 2015.