# Integrated Model of Photovoltaic Solar System with the Sound Biometric Techniques

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***Abstract*— Photovoltaic system (PVS) is used for generating electric power by using solar cells to convert energy from the sun light into a flow of direct current electricity, which can be used to power equipment or to recharge a battery. In addition, the Sound biometric techniques can enable PVS to listen and understand their surrounding auditory environment since turning the lights on all the nights will get through a lot of energy which it might be used in other significant concerns. This paper proposed a model of combination between PVS to generate power energy (electricity) from the sunlight controlled by a sound biometric technique to reduce the consumption of the generated power energy by turning the lights on for the highways only when there are cars on the highway and only for some period of time to make the driving out of harmful ways and trouble-free; followed by an implementation of the proposed combination model between the PVS and biometric sound chip and a presentation of a discussion of the proposed system results.**

***Keywords— PVS; Energy Conversion; Energy Storage Device; Linear Predictive Analysis; Sound Recognition; Electricity Consumption.***

## I. Introduction

Generating and conserving energy is one of the most important concerns in numerous countries in view of the piece of evidence that they have inadequate resources of energy to depend on. They may import all their needs of energy from other countries. Consequently, many conventions have been held for the public to demeanor the consumption of energy. Conserving energy of highways lights using photovoltaic solar systems could be used to trim down the power invoice by generating and controlling the lights of street lamps in the highways. This will save a lot of energy.

The word photovoltaic combines two terms: photo means light and voltaic means voltage. A PVS in this paper uses photovoltaic cells to directly convert sunlight into electricity. It is controlled by a sound biometric technique including the algorithms that defines the conserving energy of street lights system that uses the database which consists of 200 sounds of cars and a lot of sounds from other domains.

There are two approaches for using photovoltaic solar systems using a sound biometric chip: First, standalone systems that require batteries to store power for the times when the sun is not shining, this approach can be used in a highway that also has utility power as long as they are completely separated. Second a grid interface systems by using the power from the central utility when needed and supplies extra generated power for a highway as a parallel system by the utility. In this paper the philosophical point is to use the first approach in the view of the second approach to reduce the loss of the transferred power.

While in the future our target is to build grid connected interface for the PV power systems for a decentralized electrical network. Power is generated closer to where it is needed such system will reduce the need to increase the capacity of transportation and distribution lines. The view is that, the grid connected interface system generates its own electricity and feeds its surplus power into the utility grid for later use such as a battery bank. The battery banks can be used to provide backup power when the grid goes down using a large inverter to convert DC power output into AC power which can handle many panels as in a standalone system.

The proposed system can be adopted for other uses such as supply power lights, televisions, pumps and other appliances at our homes. The advantages of using such systems are low-maintenance, safe, and pollution free.

This paper will introduce a system to control the lighting of street lights in highways. The system will turn the lights on only if there is a car on the highway for a pre-defined period of time, and will keep the lights off for any other sound. This paper is organized as follows. Section II presents an overview of the related techniques for combination systems. Section III presents the proposed system of PV integrated with the biometric sound chip. Section IV presents a practical implementation of using a sound chip with the PVS. A conclusion is presented in section V.

## II. An Overview of the Related Techniques

This section presents an overview of the related techniques used for the integration of our proposed system.

### *A. Photovoltaic Cells*

Semiconductor material normally composed of the silicon and is used in slim wafers or ribbons in most commercially existing cells. One side of the semiconductor material has a positive charge and the other side is negatively charged. Sunlight hitting the positive side will activate the negative side electrons and produce an electrical current [1].



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### *B. Types of cells*

Crystalline cells [2] have been in service for a long time and exhibit an outstanding longevity. Cells developed almost 40 years ago are still operating and most manufacturers offer 10-year or longer warranties on crystalline cells. There are two sub-categories of “crystalline cells – single crystal and polycrystalline”. They both perform similarly. The efficiency of the crystalline cells is around 13%.

Amorphous [2] silicon is a recent technology for solar cells. It is cheaper to produce and offers greater elasticity, but their competence is half of the crystalline cells and they will disgrace with use. These types of cells will produce power in low light situations. This technology is expected to get better application possibilities far exceeding crystalline technology. Currently, the best choice for solar cells will be the crystalline selection.

### *C. A voltage Regulator*

A voltage regulator [3] is designed to automatically maintain a constant voltage level. A voltage regulator may be a simple "feed-forward" design or may includes a negative feedback control loops. It may use an electromechanical mechanism, or electronic components. Depending on the design, it may be used to normalize one or more AC or DC voltages.

### *D. Energy storage*

Energy storage [4] is skilled by devices or physical media that store energy to carry out useful operations at a later time. A device that stores energy is sometimes called an accumulator. An early solution to the problem of storing energy for electrical purposes was the development of the battery as an electrochemical storage device. Batteries have previously been of limited use in electric power systems due to their relatively small capacity and high cost. However, since about the last decade of the 21st century newer batteries have been developed that can now provide important utility scale load-leveling capabilities.

### *E. Shunt*

A shunt [5] is a device which allows electric current to pass about a new point in the circuit by creating a low resistance path. The term is also extensively used in photovoltaics to describe a surplus short circuit between the fronts and back outside contacts of a solar cell, typically caused by wafer damage. The origin of the term is in the verb 'to shunt' meaning to turn away or follow a different path.

### *F. Conditioning Circuit*

A conditioning circuit [6] includes a list of sensor types to modify in an electrical property to indicate in its environment. The electrical properties sheltered the voltage and it changes from the analog to digital, current, resistance, capacities and charge.

### *G. Microcontroller*

There are four types of microcontrollers devices [7] (PIC16F873, PIC16F874, PIC16F876 and PIC16F877). The PIC16F876/873 devices come in 28-pin packages and the PIC16F877/874 devices come in 40-pin packages. The parallel slave port is not implemented on the 28-pin devices. In the proposed model the use of PIC 16F876 is recommended.

### *H. Voice Recognition*

Voice recognition [8] consists of two main tasks, that is, Feature Extraction and Pattern Recognition. Feature extraction attempts to determine personality of the sound signal, while pattern recognition refers to the corresponding of features in such a way as to determine, within probabilistic limits, whether two sets of features are from the same or different domains [9]. In general, speaker recognition can be subdivided into speaker identification, and speaker verification. Speaker verification will be used in this paper to recognize the sound of cars.

### *I.* Linear Predictive Coding (LPC)

Linear predictive coding (LPC) [8] is defined as a digital method for encoding an analogue signal in which a particular value is predicted by a linear function of the past values of the signal. It was first proposed as a method for encoding human speech by the United States Department of Defense (DoD) in federal standard, published in 1984 [10]. The LPC model is based on a mathematical approximation of the vocal tract. The most important aspect of LPC is the linear predictive filter which allows determining the current sample by a linear combination of the previous samples. Where, the linear combination weights are the linear prediction coefficient.

The LPC based feature extraction [10] is the most widely used method by developers of speech recognition. The main reason is that speech production can be modelled completely by using linear predictive analysis, beside, LPC based feature extraction can also be used in speaker recognition system where the main purpose is to extract the vocal tract.

### *J. Vector Quantization (VQ)*

The quantization is the process of representing a large set of values with a much smaller set [11]. While, the vector quantization (VQ) is the process of taking a large set of feature vectors, and producing a smaller set of feature vectors, that represent the centroids of the distribution, i.e. points spaced so as to minimize the average distance to every other point [8].

However, optimization of the system is achieved by using vector quantization [12] to compress and subsequently reduce the variability amongst the feature vectors derived from the frames. In vector quantization, a reproduction vector in a pre-designed set of K vectors approximates each feature vector of the input signal. The feature vector space is divided into K regions, and all subsequent features vectors are classified into one of the corresponding codebook-elements, according to the least distance criterion (Euclidian distance) [12].

### *K. Digital Signal Processing (DSP)*

The digital signal processing (DSP) [13] is the study of signals in a digital representation and the processing methods of these signals. The DSP and analogue signals processing are subfields of signal processing. Furthermore, the DSP includes subfields such as audio signal processing, control engineering, digital image processing, and speech processing [14].

### *L. Frequency Domain*

The signals are converted from time or space domain to the frequency domain typically through the Fourier transform. The Fourier transform [15] converts the signal information to a magnitude and stage component of each frequency. Normally the Fourier transform is converted to the power spectrum, which is the magnitude of each frequency component squared. This is one of the features that we have depended on in our analysis.

### *M. Time Domain*

The most common processing approach in the time or space domain is enhancement of the input signal through a method called filtering. Filtering [16] usually consists of some transformation of a number of surrounding samples around the current sample of the input or output signals. There are various ways to characterize filters.

Most filters [8] can be described in Z-domain (a superset of the frequency domain) by their transfer functions. A filter may also be described as a difference equation, a collection of zeroes and poles or, if it is an FIR filter, an impulse response or step response. The output of an FIR filter [17] to any given input may be calculated by the input signal with the impulse response. Filters can also be represented by block diagrams which can then be used to derive a sample processing algorithm to implement the filter using hardware instructions.

## III. The Proposed Photovoltaic Solar System with Biometric Chip

A PVS in this paper uses photovoltaic cells to directly convert sunlight into electricity controlled by a sound biometric technique including the algorithms that define the conserving energy of street lights system that uses the database which consists of 200 sounds of vehicles and a lot of sounds from other domains.

There are two approaches for using photovoltaic solar systems using a sound biometric chip: stand alone system that requires batteries to store power for the times when the sun is not shining, this approach can be used in a highway that also has utility power as long as they are completely separated. Grid interface system by using the power from the central utility when needed and supplies extra generated power for a highway as a parallel system by the utility. In this paper one philosophy is to use the first approach in the view of the second approach to reduce the loss of the transferred power.

The energy generated by the proposed model can be used immediately or stored in batteries for later use See-Fig. 1. In general, the surplus energy generated in autonomous PV systems during sunny periods is stored in batteries. The batteries then provide electricity at night during overcast periods or when there is not enough solar radiation controlled by the sound technique used in the system.

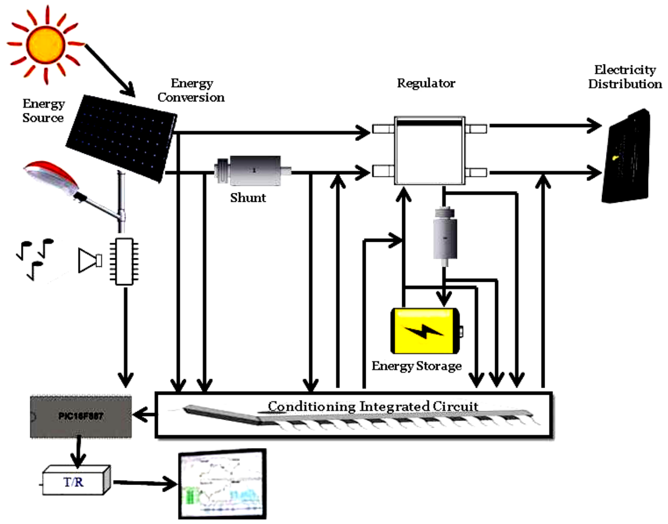


Fig. 1. The Proposed System of Integrated PVS and Biometric Sound Chip

In this paper also, many samples for different types of vehicles sound is gathered from several areas. Next, the feature extraction is applied on the collected sound. The collected sound is passed through a high and low-pass filter to eliminate the noise. The extraction of the LPC coefficient, the magnitude of the signals, and the pitch of the signals were concluded. These features are normalized and clustered into codebooks using vector quantization and the linde-buzo-gray (LBG) algorithm for clustering which based on the k-mean algorithm. In conclusion a comparison with the template database that we have built before was made.

### *A. Database*

The database that is used in this system was built from the recorded vehicles sound. They are recorded from different places, also from sounds of the rain, thunder, and air planes. These are found on internet, as well as recorded sound from different humans. For the cars, rain, and air planes group, vector quantization method is used for clustering based on LBG algorithm and k-mean algorithm, and the Euclidian distance for matching. A statistical analysis is used for the human group, since the sound of the human is very different and cannot be bounded. A statistical analysis is based on the power spectrum of the sound then the mean and standard deviation was taken to make the comparison.

### *B. Collecting Samples*

200 samples of vehicles sound are collected from different places beside the highways. These samples were taken on the mid night to assure that we have taken the wholesome sound of the car with the smallest amount possible noise. A microphone connected to a laptop was used; it was at a far above the ground place to assure to collect all the sounds, since the proposed hardware should be beside the light of the highway, which is about 1 to 30 meters above vehicles. A program called sound forge is used to record the sounds. Most of the sounds were recorded at a sample frequency of 50 KHz to make sure that the sound has a high quality, and all the component of the sound will be shown when converting the sound to the frequency domain.

### *C. Feature Extraction*

To identify the sound of the vehicles among other sounds one need to extract the parameters from the sound signal; these parameters help us to distinguish the sounds domain from others (cars, airplanes, weather, and human sounds). Feature extraction consists of choosing those features which are most effective for preserving class separately. The main features chosen expressed most effectively the sounds are LPC analysis, magnitude of the signal, and pitch of the signal.

### *D. Pitch Extraction*

The harmonic peak based method [8, 18] has been used to extract pitch from the wave sound. Since harmonic peaks occur at integer multiples of the pitch frequency, then a comparison peak frequencies is made at each time (t) to locate the fundamental frequency to find the highest three magnitude peaks for each frame. Consequently, the differences between them computed.

Since the peaks should be found at multiples of the fundamental, one know that their differences should represent multiples as well. As a result, the differences should be integer multiples of one another. Using the differences, one can derive our estimate for the fundamental frequency. While, the largest three peaks in each frame are identified; then an establishment of the spectrogram of the signal is done; spectrogram computes the windowed discrete-time Fourier transform of a signal using a sliding window. The spectrogram is the magnitude of this function which shows the areas where the energy is mostly appearing, after that the largest three peaks in each frame is taken. A major advantage to this method is its very noise-resistive. Even as noise increases, the peak frequencies should still be detectable above the noise.

### *E. Feature Comparison*

Subsequent to feature extraction, the similarity between the parameters derived from the collected sound and the reference parameters need to be computed. The three most commonly encountered algorithms in the literature are hidden Markov modeling (HMM), dynamic time warping (DTW), and vector quantization (VQ). In this paper, one uses the VQ to compare the parameter matrices.

### *F. Decision Function*

There are usually three approaches to construct the decision rules [19], that is: topological or probabilistic or geometric rules. In this paper, two types of decision rules are discussed, which are based either on linear functions or on more complex functions such as support vector machines (SVM).

### *G. Microcontroller*

The decision function should take a decision if the input sound is matched with our constraints or not, if the sound matched our conditions then the sound electronic circuit chip send order to the program that found in the microcontroller to open the photo voltaic circuit to turn the light on for some predefined period of time.

The type of microcontroller used in the proposed model is PIC 16F887. The PIC16F887 is used in a wide range of applications, high quality and easy availability; it is an ideal solution in applications such as: the control of different processes, machine control devices, measurement of different values. The PIC 16F887 is used as interface between the sound chip circuit and PV circuit.

## IV. PRACTICAL IMPLEMENTATION

First of all, a collected sound is passed throughout a high-pass filtered. Then the signal is split into frames, each about 30ms long. By breaking the signal into frames, one fairly accurate these discrete sounds in this analysis. For each frame the LPC coefficient is calculated. As well as, the magnitude and the pitch of the sound are calculated. These coefficients characterize each sound domain, see-Fig. 2.

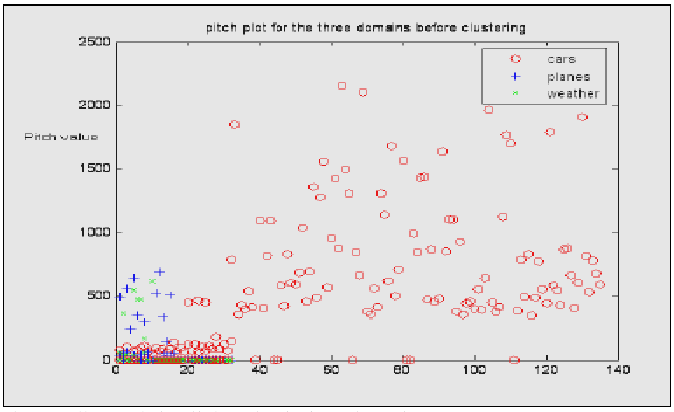


Fig. 2. All sounds in all domains before clustering

The second step is to map these data by using vector quantization (VQ) and is accomplished by a clustering algorithm. However, the clustering algorithm takes a number of random vectors and condenses the vectors that are nearest to it, iterating until the least mean error between the vectors is reached. One clustered the data into vectors, and each of these vectors is called a codeword. This set of vectors, or codewords is created for each sound. The codewords for a given sound are then stored together in a codebook for that sound domain, see Fig. 3.

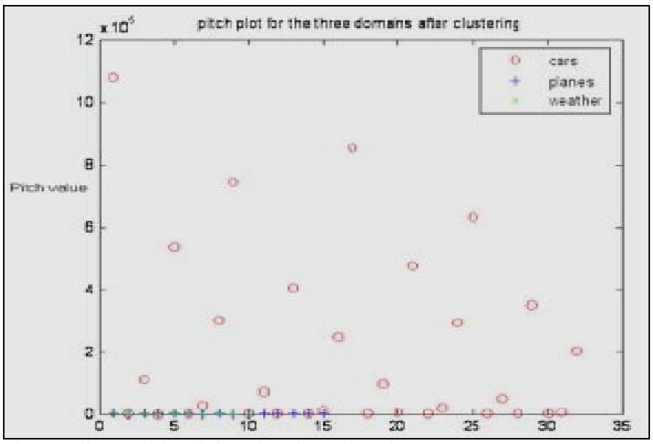


Fig. 3. All sounds in all domains after clustering.

Then, each speaker's codebook is then stored together in a master codebook which is compared to the test sample during the testing phase to determine the sound domain, see Figure 3. Suppose there is a region of space where codeword vectors from several different sounds were laid. If a test vector also falls in this region, the codewords do not help determine the identity of the sound domain because the errors between the test vector and the various codewords will be roughly equal.

The sound biometric system that is built identifies the three main domains sounds which are airplanes, cars, and weather. The recognition for these domains was 100% according to the vector quantization method but the vector quantization method divided all the sound into three main regions so any new sound will be approximated into one of these regions so to get more accuracy and to avoid any similar sound to the vehicle sound. A new code called feature extraction based on statistical analysis is developed to discard any sound that is similar to the vehicle sound but really not with car sound this method with the vector quantization method get accuracy 100%. Doing so, one can assure to make energy conserving with 100% probability reached. Since the lights will turn only for cars .the feature extraction based on statistical analysis method gain accuracy of 92.5% when working alone.

However, the results of this method are divided into three choices as follows:

- When executing the code as it is (with acceptable interval [av-2*std, av+2*std]), the results are:
  - The probability of data in this case = 95%.
  - The correct detection of car sound = 92.5%.
  - Detection sound of airplanes as car= 4%
  - Detection sound of rain as car= 2%
  - Detection sound of animals as car= 0%
- When executing the code and taking the acceptable interval to be [av-sd, av+sd], the results are:
  - The probability of data in this case = 67%.
  - The correct detection of car sound= 87%
  - Detection sound of airplanes as car= 0%
  - Detection sound of rain as car= 0%
  - Detection sound of animals as car =0%
- When executing the code and taking the FFT (sound, 11000) and summing the area under the curve from 0-5500 and interval to be [av-2*sd, av+2*sd] the results are:
  - Acceptable interval [av-2*std, av+2*std].
  - The correct recognition of car sound =91%
  - Detection sound of planes as car= 0%
  - Detection sound of rain as car= 8%
  - Detection sound of animals as car= 0%.

Based on the results of this paper, one can note that the number one is the best choice.

## V. Conclusion

The lights are available for highways to avoid accidents and to increase the safety for driving and make it easier, but turning the lights on all the nights will consume a lot of energy.

This paper proposed a new and integrated system to generate electricity using a proposed PVS and to reduce electricity consumptions for the electricity in Jordanian highways by using the sound recognition techniques in order to turn the lights on only when there are cars on the highways and only for some period of time. However, this paper made the following contributions: designing a new system for conserving energy based on the voice recognition of the cars sounds. This system is the first application of this type that concern the street lights. This paper also demonstrates that the weighted Euclidian distance with the LBG algorithm was very helpful and achieved high accuracy as well as a proposed model of using a PVS with the biometric sound chip.

This paper shows that the feature of the sound extracted is very precious and really can make a distinction between different sounds, so it makes speaker identification with high precision.

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