

SPECIAL FEATURE

Restaurant
Recommendation System

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An added dimension to Climate Change ...

Climate change is one of the most widely discussed topics. Typically, the phrase 'climate change' refers to Global Warming as a result of increased greenhouse gases (mainly CO₂). This has resulted in polar ice-caps melting and ocean levels rising. However, another less talked-about scourge which can further exacerbate the process is acidification of our oceans. The oceans cover about 71% of the earth's surface area and are generally considered to have been the place where life-forms originated. It is also the reason the earth appears as a 'blue watery planet' from space, which gives us our signature look.

Tragically, the approximately 500 billion metric tons of Carbon, primarily in the form of CO₂, that humans have purged out to the environment over the past three centuries since the industrial revolution is ultimately taking a serious toll. About 30% of the CO₂ released by fossil fuel burning get absorbed in the oceans; consequently, the carbonate ion concentration increases, decreasing the pH and increasing acidity. This process is referred to as 'Ocean Acidification'.

As the oceans acidify, the delicate pH balance shifts from alkalinity to acidity. Various satellite images tracking ocean acidification in the Greater Caribbean, the Amazon Plume and the Bay of Bengal all unequivocally point to decreasing pH values. This could lead to serious and catastrophic changes in the ecosystems of the fragile deltas, lagoons and coral reefs that abound in these regions.

Other related troubles may also lie ahead if this relentless onslaught of dumping CO₂ is left unchecked. A lesser known fact is that the major source of oxygen on this planet are phytoplankton – tiny, little green migratory

creatures that wander around in swarms in the ocean; ironically these innocuous organisms may account for up to 75% of the atmospheric oxygen, contrary to popular notion that plants and trees supply the major portion. Drastic changes in the chemistry of the waters could alter plankton population dynamics, resulting in fisheries being adversely affected and causing a major imbalance to the entire aquatic ecosystem and food chain. Socio-economic woes to the millions of humans involved with fisheries along the coastline are almost guaranteed to follow.

The overall ocean health can be gauged using SST (Sea Surface Temperature) and salinity, along with dissolved inorganic carbon, alkalinity and pH. Some of these parameters are delicately interlinked - for example, salinity and temperature changes can affect ocean carbonate chemistry, which in turn is tied to acidification. ESA's SMOS (Soil Moisture-Ocean Salinity) mission and NASA's Aquarius mission, both provide information on ocean salinity. Some areas are more difficult to 'map' like coastal Bay of Bengal due to incessant cloud cover. However, in-situ measurements of alkalinity and other carbon content parameters can complement satellite images to complete the picture - these images portray a major decline in pH and alkalinity.

The Paris Agreement limits the global increase in temperature to between 1.5 and 2° C for this century. Even this rise coupled with Ocean Acidification would entail serious loss of ecosystems and habitat, decline in fisheries, decline in coral reefs, extremely hot days and cold nights and the arctic free of ice in summer. These are cataclysmic transformations which may alter our 'blue pearl' of a planet into a dreary mess, unless immediate and drastic measures are adopted.

Dr. Jaideep Talukdar

Dept. of BSH.

Restaurant Recommendation System

Abstract : Recommendation systems are widely used system to predict the preferences of users to products/items. It uses algorithms to provide users with product or service recommendations. Recently, these systems have been using machine learning algorithms from the field of artificial intelligence. However, choosing a suitable machine learning algorithm for a recommendation system is one of the difficult task because of the number of algorithms. In today's world, every customer is faced with multiple choices. For example, if a user looking for a book to read without any specific idea of what he wants, there's a wide range of possibilities how the search might turn out. The user might waste a lot of time browsing around on the internet and trawling through various sites hoping to strike gold. Here, comes the idea and need of a proper recommendation system to make people choice easy by helping them to find products relevant to their interest. Therefore to save time, people are highly dependent on various kinds of online recommendations to utilize their precious time. This work focuses on development of a Restaurant Recommendation System using Matrix Factorization and Latent Factor Collaborative Filtering. For this purpose, 49 restaurants were considered and for each restaurant 100 reviews were taken. The recommendation system returns top 5 recommendations of restaurant names of Bhubaneswar which is based on the ratings and text reviews by user food preferences.

Keywords: Collaborative Filtering, Matrix Factorization, Recommendation, Machine Learning

I. INTRODUCTION

Recommendation systems are information filtering systems that deal with the problem of information overload by filtering vital information fragment out of large amount of dynamically generated information according to user's preferences, interest, or observed behaviour about item. The most common way of getting recommendation for restaurants is by asking our friends. It is not necessarily true that our friends have similar taste to restaurants like we do. Our taste may differ from others and our preferences may vary. This way is not accurate and abundant. The recommendation process can be automated by a web application so that the users would get recommendations for restaurant according to their preferences. This saves user from trouble of asking which restaurant to visit, its location etc. Recommendation system is a specific type of information filtering technique that attempts to present information items (such as movies, music, web sites, news) that are likely of interest to the user. Recommendation has great importance for the success of the e-commerce sites like Flipkart, Amazon, Alibaba, etc. Restaurant Recommender is a web based recommendation application that allows user to get recommendation for restaurants. The users can rate restaurants and review them. The users are recommended on the basis of ratings they gave to restaurants. Recommender system has the ability to predict whether a particular user would prefer an item or not based on the user's profile. Here, we will be using a collaborative approach based recommendation system to provide restaurant recommendations to people based on their preferences and past history. The proposed algorithm for Restaurant Recommendation is

Matrix Factorization which is a class of collaborative filtering algorithm it works by decomposing the user-item interaction matrix into the product of two lower dimensionality rectangular matrix.

Recommendation systems are one of the most popular application of machine learning that gained increasing importance in recent years. At a very high level, recommendation systems are algorithm that makes use of machine learning techniques to mimic the psychology and personality of humans, in order to predict their needs and desires. Recommender systems are algorithms aimed at suggesting relevant items to users (items being movies to watch, text to read, products to buy or anything else depending on industries). Restaurant recommendation system is a very popular service whose accuracy and sophistication keeps increasing every day. With the advent of smartphones, web 2.0 and internet services like 3G, this has become accessible by every consumer [1]. Here, we present a personalized location based restaurant recommendation system integrated with Graphical User Interface. It ubiquitously studies the user's reviews and ratings by convertings them into vectors of matrix using Tf-Idf Vectorizer to analyze each restaurarnts and recommends on the basis of relevancy of user's search.

II. RELATED WORK

In this section, we provide a brief introduction to previous related work. Recommendation tasks can be divided into two categories: rating prediction and item recommendation. The majority of existing

recommendation algorithms are designed for one of these tasks; hence, we introduce these two algorithm types separately.

Rating rediction: In the rating prediction task, Collaborative Filtering (CF) algorithms attempt to predict user ratings for items they have not yet rated.

Item recommendation: The item recommendation task refers to generating a ranking list for each user over as yet unrated items. Traditional algorithms for item recommendations rely mainly on implicit feedback. For a situation in which only implicit feedback exists, formulated one-class Collaborative Filtering problems and proposed two different solutions: weighted low-rank approximation and negative example sampling proposed considering the rating data as indications of positive or negative preference, associated with a varying confidence level. By assigning uniform weights to the missing data, the computational complexity could be reduced significantly using the alternating least squares (ALS) algorithm. Rendle et al.[2] presented a sample-based model that optimizes the pair-wise ranking between positive and negative samples. He et al. [3] proposed weighting the missing data based on item popularity, and designed a learning algorithm based on element-wise ALS (eALS). Chen et al. [4] proposed a fast-ranking algorithm known as RankSVM. The above methods were designed only for item ranking tasks in which only implicit feedback is available. However, in a typical situation, both explicit and implicit feedbacks are available. For example, in the Netflix Prize competition, the dataset contains explicit feedback such as user star ratings for movies. Furthermore, from the rating data, we can determine user watching history, which is a type of implicit feedback.

Recommendation systems have been broadly used by the overall public of various age groups. Due to their potential of attracting peoples, recommenders are associated by means of impressive data sources e.g. incentive based recommendation in case of Epinions. com. On the other hand, impact in recommenders on social paradigm establishes emerging possibilities of crowd sourcing to promote the usage [5]. Research in the context of the recommendation system, several types of research has been tended in some extend. The methodologies utilize novel models and additionally modified rendition of existing algorithms to improvise the solution. In this section, a brief literature review is presented. When using a collaborative or a hybrid filtering approach, RSs must gather information about the user in order to develop recommendations. This activity can be done explicitly or implicitly. Explicit user data gathering happens when users are aware they are providing their information. For instance, when registering for a new online service, users usually fill in a form that asks their name, age, and email. Other forms of

explicit user data gathering (Gemmis et al., 2011; Longo, Barrett, & Dondio, 2009) are when users express their preferences by rating items using a numerical value or a preference such as a Facebook “like.” Implicit user data gathering accesses information about the user indirectly.

In order to overcome the defects of prior works, we proposed a latent factor model based on Matrix Factorization that can exploit both explicit and implicit feedback. A further difference between the proposed model and previous models is that the proposed model is suitable for both the rating prediction and item recommendation tasks.

III. OVERVIEW

In this section, we provide process model of the system, then preliminary knowledge regarding Collaborative Filtering, Matrix Factorization and Stochastic gradient descent. The process model is shown in Fig. -1.

A. Process Model

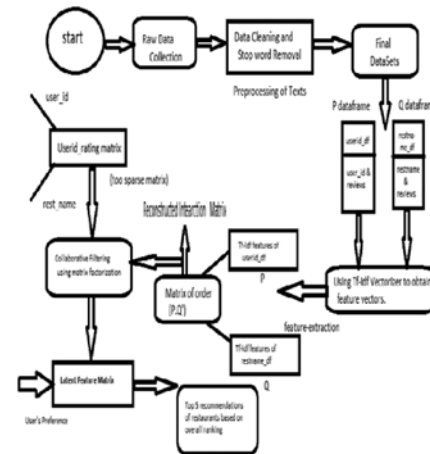


Fig.-1: Process Model

B. Collaborative Filtering(CF)

Collaborative Filtering: This algorithm uses “User Behavior” for recommending items as shown in Fig -2. This is one of the most commonly used algorithms in the industry as it is not dependent on any additional information.

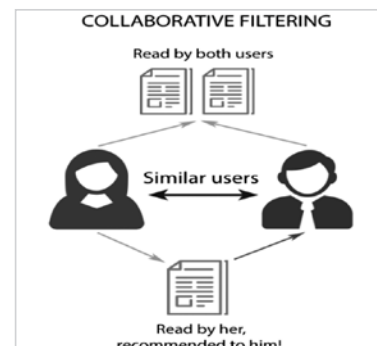


Fig.-2: Collaborative Filtering

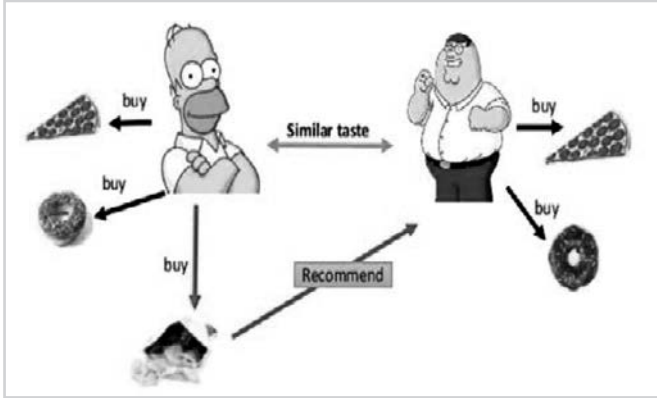


Fig-3: Collaborative Filtering Process

C. Matrix Factorization(MF)

MF is a popular approach in recommendation systems. Based on the assumption that the rating matrix has a low rank, MF maps users and items to a shared subspace(subspace is a vector space that is contained within another vector space) of dimensionality, so that a user-item rating can be modeled as the inner products of the user and item latent factor vectors in that subspace.

The concept of matrix factorization can be written mathematically as:

$$\hat{r}_{ui} = \mathbf{q}_i^T \mathbf{p}_u \quad (1)$$

Where each item i is associated with a vector $\mathbf{q}_i \in \mathbb{R}^k$, and each user u is associated with a vector $\mathbf{p}_u \in \mathbb{R}^k$. For a given item i , the elements of \mathbf{q}_i measure the extent to which the item possesses those factors, positive or negative. For a given user u , the elements of \mathbf{p}_u measure the extent of interest the user has in items that are high on the corresponding factors, again, positive or negative. The resulting dot product, $\mathbf{q}_i^T \mathbf{p}_u$, captures the interaction between user u and item i -the user's overall interest in the item's characteristics. This approximates user u 's rating of item i , which is denoted by r_{ui} , leading to the estimate.

Then we can create an objective function (that we want to minimize) with respect to \mathbf{q} and \mathbf{p} , which are (m, k) and (k, n) matrices.[10]

$$\min_{\mathbf{q}, \mathbf{p}} \sum_{(u,i) \in K} (r_{ui} - \mathbf{q}_i^T \mathbf{p}_u)^2 + \gamma (||\mathbf{q}_i||^2 + ||\mathbf{p}_u||^2) \quad (2)$$

The term on the right is the regularization term, this is added since we do not want our decomposed matrix \mathbf{q} and \mathbf{p} to over-fit to the original matrix.

D. Stochastic Gradient Descent

Simon Funk popularized a stochastic gradient descent optimization of Equation 2 (<http://sifter.org/~simon/>

journal/20061211.html) where in the algorithm loops through all ratings in the training set. For each given training case, the system predicts r_{ui} and computes the associated prediction error.

$$e_{ui} = (r_{ui} - \mathbf{q}_i^T \mathbf{p}_u) \quad (3)$$

Then it modifies the parameters by a magnitude proportional to g in the opposite direction of the gradient, yielding:-

$$\begin{aligned} \mathbf{q}_i &\leftarrow \mathbf{q}_i + \lambda \cdot (e_{ui} \cdot \mathbf{p}_u - \gamma \cdot \mathbf{q}_i) \\ \mathbf{p}_u &\leftarrow \mathbf{p}_u + \lambda \cdot (e_{ui} \cdot \mathbf{q}_i - \gamma \cdot \mathbf{p}_u) \end{aligned} \quad (4)$$

IV. CONCLUSIONS AND FUTURE WORK

In this paper we introduce a Matrix Factorization based method to implement collaborative filtering for constructing the recommendation lists. We have predicted the most relevant restaurant based on the user search i.e. simply the inner product of the feature vector of the plain text and feature vectors of the Restaurant name. Out of all, the System recommends top 5 Restaurant to the user. Fig. 4 and Fig. 5 show the outputs obtained by the system. We have also design the Graphical User Interface shown in fig. 6 to the user so that they can search their food preferences easily and get their best recommendation.

Rating Restaurant Name

Yummyis	0.341847
The Zaika	0.303724
Dakh in 9	0.299978
SMOG Resto Cafe	0.299059
Hi Bhubaneswar	0.291450

Fig.-4: Output (Top 5 recommendation)

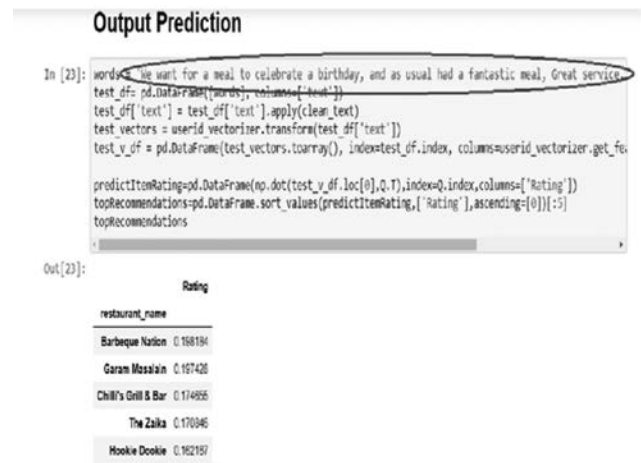


Fig-5: Output based on other preferences



Fig.-6: Graphical User Interface (GUI)

The system may further be enhanced to find the accuracy of the Restaurant Recommendation System and compare with the accuracy obtained from different models and validate a Graphical User Interface(GUI) with actual code which will be used by the user to search the plain text (food preferences).

ACKNOWLEDGEMENT

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**Akshay Kumar, Anchal Kri. Singh,
Arpita Mohanty, Pallavi Priyadarshini**

8th Sem. CSE

Stress Speech Analysis Using MFCC And SVM

Abstract: With the advancement of technology in the field of communication there have been notable advancement in speech signal processing. It is one of the largest growing areas in signal processing. Mel Frequency Cepstral Coefficient (MFCC) has been used for feature extraction from the stressed speech signal. Support Vector Machine is used for classification of normal and stressed speech signal. Challenges in the SVM based approach include selection of representative training segments, selection of features, normalization of the features, and post-processing of the frame-level decisions. We propose to construct a SVM using MFCC features because they capture the most relevant information of speech, and they are widely used in speech. The ultimate aim is to detect stress speech and provide a distinct classification of normal speech from stressed speech signal.

I. INTRODUCTION

During the past few years, vast number of research and development have taken place in the field of speech processing. Stress speech analysis is a recent research topic in the field of Human Computer Interaction Intelligence and mostly used to develop wide range of applications such as stress management for call centre employee, and learning & analyzing stress conditions of patients in medical centers. In E-learning field, identifying students stress level timely and making appropriate treatment can enhance the quality of teaching.

Speech: Speech is the most effective and natural medium to exchange information. Speech serves to communicate information from a speaker to one or more listeners. Each spoken word is created out of the phonetic combination of a limited set of vowel and consonant speech sound units (phonemes). These vocabularies, the syntax that structures them, and their sets of speech sound units differ, creating many thousands of different, and mutually unintelligible, human languages.

Stress: Stress is defined as the human response to a particular stimulus otherwise called as ‘stressor’ which may be physical, mental and emotional. It is the adoption/coping feedback that helps the body to equip in challenging positions. Also stress can be defined as any condition that causes a speaker to vary speech production from neutral conditions. If a speaker is in a ‘quiet room’ with no task obligations, then the speech produced is considered neutral. For example, if a person starts doing his work, the work itself is not the stress stimulus rather the deadline.

Perceptually induced stress results when a speaker perceives his environment to be different from ‘normal’ such that his intention to produce speech varies from neutral conditions. Physiologically induced stress is the result of a physical impact on the human body that results in deviations from neutral speech production despite intentions.

Mel-frequency cepstral coefficients (MFCCs) are coefficients that collectively make up an MFC. They are derived from a type of cepstral representation of the audio clip (a nonlinear “spectrum-of-a-spectrum”). The difference between the cepstrum and the mel-frequency cepstrum is that in the MFC, the frequency bands are equally spaced on the mel scale, which approximates the human auditory system’s response more closely than the linearly-spaced frequency bands used in the normal cepstrum. This frequency warping can allow for better representation of sound, for example, in audio compression.

SVM: A Support Vector Machine (SVM) is a promising machine learning technique that has generated a lot of interest in the pattern recognition community in recent years. The greatest asset of an SVM is its ability to construct nonlinear decision regions in a discriminative fashion.

Support Vector Machine is one of the most efficient classifiers. It is a frontier which best segregates the two classes. It looks at the extremes of the data sets and draws a decision boundary known as a hyperplane. If we do not have an optimal boundary, we could incorrectly classify the objects. And we should even maximize the

width between the two classes of support vectors for efficient classification.

II. DATABASE

In this project we only focus on the psychological stress. Here, we will be creating a database named “Exam Stress”. The database consists of 50 voice recording of people under stress and normal condition for the same sentence “A PIECE OF PAPER CANNOT DECIDE YOUR DESTINY”. Voices of the people are recorded in the morning after waking up and before giving exam. The voices are recorded using a device named ‘audacity’. The recordings are being analyzed by using a method called SVM. Finally the output is obtained which gives the number of people under stress and the number of people in normal condition. The output of the methods are compared and the best method among both of them is obtained. The samples are sampled at 16 KHz and the sampling frequency is 44 KHz.

III. METHODOLOGY

The first step in any speech detection system is to extract features i.e. identify the components of the audio signal that are good for identification of the normal and stressed content and discarding all the other stuff which carries information like background noise, etc. The block diagram for MFCC is shown in Fig-1.

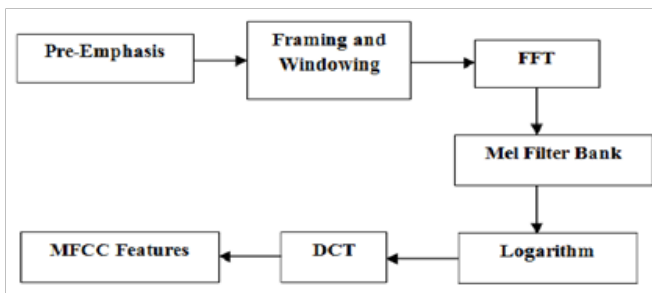


Fig.-1: Block diagram for MFCC

MFCCs are commonly derived as follows:

1. Take the Fourier transform of (a windowed excerpt of) a signal.
2. Map the powers of the spectrum obtained above onto the mel scale, using triangular overlapping windows.
3. Take the logs of the powers at each of the mel frequencies.
4. Take the discrete cosine transform of the list of mel log powers, as if it were a signal.
5. The MFCCs are the amplitudes of the resulting spectrum.

Mel frequency Cepstral coefficients algorithm is basically a technique which takes voice sample as inputs. After processing, it calculates coefficients unique to a particular sample. In this project, simulation software called MATLAB R2013a is used to perform MFCC. The simplicity of the procedure for implementation of MFCC makes it most preferred technique for stressed speech detection. After obtaining the coefficients, the mean of these coefficients are fed into the classifier to obtain two distinct classes. For two-class, separable training data sets, there are lots of possible linear separators. Intuitively, a decision boundary drawn in the middle of the void between data items of the two classes seems better than one which approaches very close to examples of one or both classes.

An SVM classifier insists on a large margin around the decision boundary. Compared to a decision hyper plane, if you have to place a fat separator between classes, you have fewer choices of where it can be put. As a result of this, the memory capacity of the model has been decreased, and hence we expect that its ability to correctly generalize to test data is increased. After starting the system, the speech signal is taken from the database and then we divide them to training and testing database.

Training Dataset: It is the set of database to train a model. We use 70% of the dataset as the training dataset so that it is large enough to yield statistically meaningful results.

Test Dataset: It is a subset to test the trained model. We use 30% of the database as the testing samples as it is

the representative of the data set as a whole. Our test set serves as a proxy for new data. This simple model does not overfit the training data.

Feature Selection: After splitting the dataset into training and testing dataset, we classify our normal and stress speech regarding parameters such as energy and pitch of the signal.

Distributions of spectral energy have wide variations across different stress conditions. The excitation spectra may also be a major representation of stress, which can be modeled by the reliable trends in the energy distribution in frequency domain. Pitch, in speech, the relative highness or lowness of a tone as perceived by the ear which depends on the number of vibrations per second produced by the vocal cords.

SVM Parameter Selection: Parameter selection in SVM algorithm affects the performance of the machine learning. At present, the research revolves around v-support vector machine. v-SVM is a deformation of SVM algorithm, it uses the parameter v instead of others. Parameter v has some meaning in values and its value range in between 0 and 1, so its choice is relatively easier. The graph of SVM is shown in Fig-2.

SVM Classification: SVM is used for classification providing SVM formulations for when the input space is linearly separable or linearly inseparable. So we classify our samples in normal and stress samples by the help of

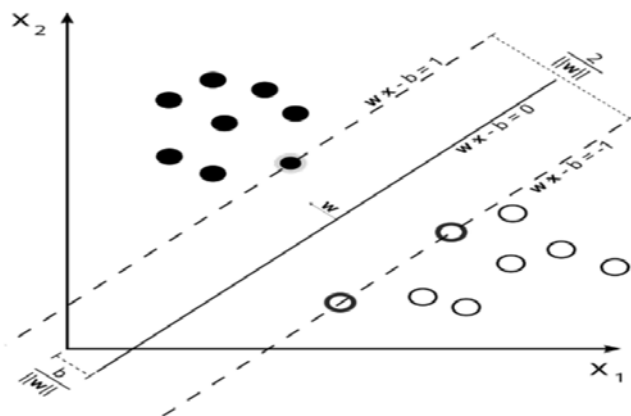


Fig.-2: Graph of SVM

a hyperplane.

To make a decision rule that would use the decision boundary. Let's take a vector w of any length constrained to be perpendicular to the median line of the street. Then we take some unknown vector u that may be present on any sides of the street. We project the vector perpendicular to the street because then we will have the distance in that direction.

So, we take the dot product, $w \cdot u \geq c$ where c is some constant

The dot product has taken the projection onto w and the bigger that projection is, the further out along street the projection will lie and eventually it will extend out so much that it will cross the median line of the street and we recognize the sample based on wherever the line extends onto [10].

$$\text{For generality, we can write, } \bar{w} \cdot \bar{u} + b \geq 0 \quad \text{and} \quad c = -b \quad (1)$$

This is our decision rule. And if this condition satisfies, we get our stress signals or black dot samples.

But there can be lots of w vector that are perpendicular to the median of the street because it could be of any length. So we don't have enough constraints to fix a particular b or particular w. So, we are going to lay on some additional constraint so that we can calculate b and w.

$$\text{Let's take ou } \bar{w} \cdot \bar{x(s)} + b \geq 1 \quad (2)$$

$$\bar{w} \cdot \bar{x(n)} + b \leq -1 \quad (3)$$

Here, $(x(s))$ is the stress sample vector and $(x(n))$ is the normal sample vector.

Introducing another variable for mathematical convenience i.e. y.

$y = +1$ for normal speech samples and $y = -1$ for stress speech samples

$$y(\bar{w} \cdot \bar{x(s)} + b) \geq 1 \quad (4)$$

$$y(\bar{w} \cdot \bar{x(n)} + b) \geq 1 \quad (5)$$

$$\text{So, we get } y_i (\bar{w} \cdot \bar{x(i)} + b) - 1 \geq 0 \quad \text{or} \quad y_i (\bar{w} \cdot \bar{x(i)} + b) - 1 = 0 \quad (6)$$

To find the width of the street, we take a unit vector from the support vectors present in the street(the dotted line on sides of hyperplane).

$$\text{Width} = (\overline{x(n)} - \overline{x(s)}) \cdot \frac{\bar{w}}{|w|} \quad (7)$$

And after computation, we get the maximum width between the support vectors to be $= \max 2/(|w|)$, since we want to maximize 2 over the magnitude of w if we want to get the widest street under the constraints that we are going to work with.

If we want to find the extreme of a function with constraints, then we are going to have to use Lagrange multipliers which will give us a new expression that we can minimize and maximize without thinking of constraints anymore.

$$L = \frac{1}{2} |w|^2 - \sum \alpha_i [y_i (\bar{w} \cdot \overline{x(i)} + b) - 1] \quad (8)$$

This gives us the maximum width of the optimal boundary.

IV. RESULTS

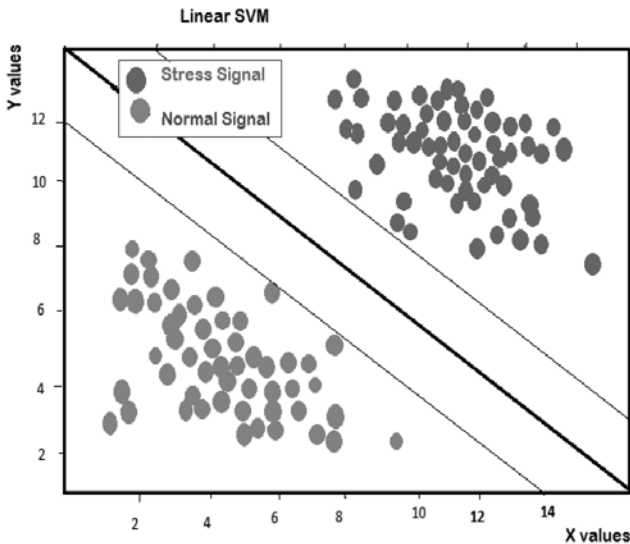


Fig.-3: Graphical representation of speech signal in SVM

Fig - 3 is the graphical representation of the support vector machine classification. Here, we have shown the stress and normal dataset using red and blue dots respectively. This classification by the hyperplane is

commonly known as wide street approach. And using this approach and some kernel functions, we have distinguished the stress signal from the normal signal. What we have done is a linear approach but nonlinear distribution can also be solved as simply as the linear one in support vector machine just by adding another dimension.

Since SVMs are inherently binary classifiers, application to a multi-class problem involves using a mixture-of-experts paradigm. We use a simple strategy to handle this that involves representing a given test vector classified as a positive example with the decision boundary that is furthest from the sample for the right classification

V. CONCLUSION AND FUTURE WORK

In this study, “Stress Speech Analysis using MFCC and SVM”, we used MFCC technique to extract features and to determine the psychological stress portion in a speech. we applied these techniques and methods to the database “Exam Stress” or “stress speech during exams” that we created for this project. After the appropriate analysis we have reached to the following conclusion. MFCC (Mel frequency Cepstral coefficients) seems to be the best method available as of now to get the desired result in a MATLAB based approach. After feature extraction, it is observed that the classification of normal and stress speech signal is best done by Support Vector Machine. It has higher accuracy and larger optimal boundary separation than any other algorithms studied. Support Vector Machine (SVM) has been profoundly successful in the area of pattern recognition. In the recent years there has been use of SVM for speech recognition. Many kinds of kernel functions are available for SVM to map an input space problem to high dimensional spaces. We lack guidelines on choosing a better kernel with optimized parameters of SVM. Some kernels are better for some questions, but worse for other questions. Which is better is unknown for speech emotion recognition, thus the thesis studies the SVM classifier and proposes methods used to select a better kernel with optimized parameters. The new method we proposed in this project can more efficiently gain optimized parameters than common methods. In order to improve recognition accuracy rate of the speech emotion recognition system, a Non-linear

Support Vector Machine was introduced.

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**Sumit Kumar,
Akankshya Jena,
Ipsita Pattnaik,
Ayush Tiwari**
Dept. of ECE

The most beautiful equation comprising of five fundamental mathematical constants, is shown below.

Euler's identity:

$$e^{i\pi} + 1 = 0$$

e : Euler's constant (≈ 2.71828)

i : imaginary unit of complex numbers

π : pi (≈ 3.14159)

Fog Accident Detection System

Abstract: the number of accident due to heavy fog/smog are steadily on the rise. This is due to the lack of visibility during the fog, so we have designed a smart system that would be installed on the light poles of the highways and expressways to minimize the accidents. When there is a blockage on the road, the ultrasonic sensor will detect it and send the message to the poles prior to the blockage through the Arduino. When the blockage is detected the message of blockage ahead will be displayed on the LCD and high intensity red lights will glow and the automatic speed breakers prior to the blockage will be activated to the upcoming vehicles. When there is no blockage the yellow lights will glow and the message of foggy weather and drive slow will be displayed. The foggy condition is detected by using a laser and LDR module.

I. INTRODUCTION

We know that an accident is an unplanned, uncontrolled event which may lead to the injury of any person or some other damage or loss to the equipments. Accidents basically mean the road accidents for which the major reasons can be:

- Over speeding
- Not using helmets or seat belts
- Using mobile phones during driving
- Disobeying the traffic rules

Including all these there is another cause of accidents that is most common in the foggy areas are the accidents related to low visibility. Moreover this fog related accidents are increasing day by day. To overcome this problem there are some existing technology such as optical fog sensor, which would detect the intensity of the fog using a LASER beam. This system can be used to detect any blockage on the road using an ultrasonic sensor, which is mounted on a servomotor that is fed by Arduino program. Moreover to make the system cost effective an equivalent principle of fog sensor is used to detect the intensity of fog using the LASER beam and LDR module. This system is going to warn the upcoming vehicles about the accident that has occurred and prevent the cascading of accidents that happens due to low visibility.

II. LITERATURE REVIEW

Literature survey reveals that the main problem is associated with the detection of fog. Many scholars have proposed their way of detection of fog such as Georgii and Schmitt, 1985, their instrument measure both the air and the wet bulb temperature. If these temperatures are equal, and a separate rain detector is not activated, a fog collector is switched on. It is obvious that this technique does not give a positive and

unambiguous proof of the occurrence of fog. Prior to the development of the optical fog detector (OFD), another type of fog detector was developed. This instrument collects fog droplets by aspirating air through a round jet impact or. The collected water runs down a vertically oriented impaction plate,

Thereby creating an electrical contact between two platinum electrodes. This aspirating instrument suffers from poor response characteristics. In general, aspirating devices tend to have poor sampling efficiencies for large droplets. In addition, the sampling rate is low. As a result, the typical response time is about 15 min. The need for a pump makes the device less suitable for unattended continuous operation. For these reasons, a new device was built. The objective was to have a low-cost, reliable, fast-responding instrument without sampling biases. The best possible approach to meet these requirements is to make use of an optical detection method.

The next problem associated with the proposed system is the detection of moving vehicles. Moving object counting usually goes with moving object tracking. Intuitively, if we can track each of the moving objects, certainly we can count them. Moving object tracking, especially people tracking, is a hot area in the field of computer vision. For years, many researchers have focused on this topic. The basic idea is to track moving people by applying computer vision algorithms on sequences of images from a camera or synchronized images captured from multiple cameras (Cai and Aggarwal, 1998). These video-based algorithms focus on distinguishing individuals rather than counting them, so that they always require huge computation and communication resources. Moreover, such algorithms usually fail when there is a crowd. Yang has proposed an alternative approach which directly estimates the number of people. In this system, groups of image sensors segment

foreground objects from the background, aggregate the resulting silhouettes over a network, and compute a planar projection of the scene's visual hull. Then, a geometric algorithm is employed to calculate the bounds on the number of people in each region of the projection. Although the authors claim that the image sensors rather than cameras can be attached on wireless sensor nodes, no results have been reported yet.

Commercial products on moving object counting have been released recently. Acorel, a French company, provides people counting systems in several areas. Their system provides a fully automatic solution to record how many people enter or exit a restricted area, such as the door of a bus. They adopt both active and passive infrared sensors. Active infrared sensors are used to determine the presence of a person, while passive infrared sensors are used to detect the infrared generated by the human body in order to differentiate the direction. A Swiss company, Dilax, also has similar products. These systems, both in Dilax and Acorel, are based on infrared technology, which requires higher voltage and consumes more power than ultrasound sensors in order to achieve the same coverage. Furthermore, many different devices such as centralized data analysis device, data transmitting devices and cables are required in their architectures. Recently, wireless sensor networks have a wide range of applications most of which are aimed at environment monitoring, object localization and tracking [3]. Most of vehicle monitoring systems that are available nowadays use camera as its detection component. Camera capture an image by recording the light waves reflected by object's surface. Camera can give very clear images if the light intensity captured by the camera is sufficient. However, in a condition with low light intensity or there is a medium interferes with the propagation of light, camera will give poor quality images. This problem of camera based vehicle monitoring system can be solved with ultrasonic sensors.

III. GLANCE TO THE CONCEPT

A. Servomotor

A servomotor is a rotary or linear actuator that allows for precise control of angular or linear position, velocity and acceleration. It consists of a suitable motor coupled to a sensor for position feedback. It also requires a relatively sophisticated controller, often a dedicated module designed specifically for use with servomotors. Servomotors are used in applications such as robotics, CNC machinery or automated manufacturing .

B. Arduino Nano

Arduino is an open source, computer hardware and software company, project and user community that designs and

manufactures microcontroller kits for building digital devices and interactive objects that can sense and control objects in the physical world. The Arduino Nano is a small, complete, and breadboard-friendly board based on the ATmega328P. It has more or less the same functionality of the Arduino Duemilanove, but in a different package. It lacks only a DC power jack, and works with a Mini-B USB cable instead of a standard one.

C. Ultrasonic Sensor

An ultrasonic sensor is an instrument that measures the distance to an object using ultrasonic sound waves. An ultrasonic sensor uses a transducer to send and receive ultrasonic pulses that relay back information about an object's proximity. High-frequency sound waves reflect from boundaries to produce distinct echo patterns.

Ultrasonic sound vibrates at a frequency above the range of human hearing. Transducers are the microphones used to receive and send the ultrasonic sound. Ultrasonic sensors use a single transducer to send a pulse and to receive the echo. The sensor determines the distance to a target by measuring time lapses between the sending and receiving of the ultrasonic pulse. The working principle of this module is simple. It sends an ultrasonic pulse out at 40 KHz which travels through the air and if there is an obstacle or object, it will bounce back to the sensor. By calculating the travel time and the speed of sound, the distance can be calculated.

D. Laser Diode Module

Modules containing diode lasers, and possibly also some optics, electrical elements, etc. are called LASER diode modules. Some optics (consisting of collimating lenses, micro-optics, and anamorphic prism pairs) can be used to shape the output beam, in order to obtain an approximately circular beam with small divergence. Such a collimated beam can be more easily transmitted over a long distance.

E. LDR Module

A LDR (light dependent resistor), photo resistor, photoconductor or photocell, is a resistor whose resistance increases or decreases depending on the amount of light intensity. LDR sensor module used to detect the intensity of light. The greater the intensity of light, the lower the resistance of LDR. The sensor has a potentiometer knob that can be adjusted to change the sensitivity of LDR towards light.

F. Fog Sensor

The meteorological use of the word "fog" is to indicate those atmospheric conditions in which the presence of water droplets, which are not rain, reduces the visibility to less than 1 km. Hence, the response of a fog detector should

have a relationship to visibility reduction caused by fog droplets. The instrument must therefore have a maximum sensitivity for water droplets in the size range of about 2-20 μm . Visibility reduction is caused by extinction, the combination of scattering and absorption of light.

Exploiting the optical properties of droplets, the presence of fog can be detected by two methods: (a) by measuring the light extinction, or (b) by detecting the scattered light. Method a is employed in many commercially available instruments that measure visibility. The major disadvantage of the method is that a long optical path is required to obtain a good signal-to-noise ratio. This can be accomplished by creating a large distance between the light source and the light receiver, or by multiple folding of the beam. In the first case, the instrument cannot be compact, and it needs sturdy supports. In the latter case, collimating optics and the use of mirrors increase the instrument's price. In both cases, the instrument is very sensitive to dirt accumulation on the optical parts. Method b is probably the more attractive. Fog and cloud droplet diameters have a range of ~ 2-20 μm . Theory predicts that most of the light scattered by these droplets is confined in a narrow cone (angle $< 10^\circ$). The amount of light that reaches a receiver therefore increases drastically if this receiver is capable of measuring the near-forward scattering light.

G. Upward Motion Smart SpeedBreaker

In this assembly shown in fig.1 the bump rises few centimetre above the road surface and give physical remainder to driver [9]. The upward motion to the bumps is provided by various mechanism like Rack and Pinion mechanism, Scissor Jack mechanism

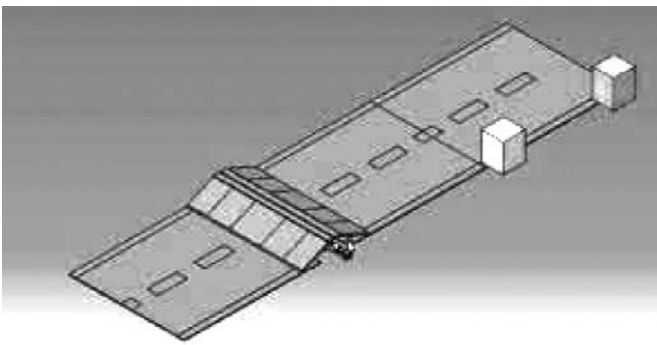


Fig.-1: Upward movement of speed breaker

H. Downward Motion Smart SpeedBreaker

In this assembly the bumps of smart speed breaker lower into the road surface producing a notch in the road surface thus giving physical remainder to driver. The downward motion of the bump is provided by a roller mechanism.

I. Detecting Objects Using Ultrasonic Sensors

In industrial applications, ultrasonic sensors are characterized by their reliability and outstanding versatility. Ultrasonic

sensors can be used to solve even the most complex tasks involving object detection or level measurement with millimetre precision, because their measuring method works reliably under almost all conditions. Infrared sensors too, find applications in many everyday products. Their low power requirements, their simple circuitry and their portable features make them desirable.

IV. WORKING

In highways/expressways fog causes a serious problem by decreasing the visibility which leads to the cascading of accidents and loss of many lives. Our proposed system is going to prevent the cascading of accidents by detecting any blockage (we are treating an accident occurred on the highway as blockage) on the highway and by alarming and giving physical warning to the upcoming vehicles. The whole system is going to be mounted on the light poles of the highways [10]. At first our system is going to detect the visibility of the area. We are using a LASER beam and LDR assembly to detect the visibility. The LDR is mounted on one of the light poles and the LASER module is mounted on the opposite light pole at a certain distance from the LDR-mounted light pole. When the LASER beam falls on the LDR, the resistance value decreases and the current flows but when the intensity of the LASER beam is less the resistance is more and less current flows. We have kept a threshold value for the LDR module, when the intensity of the LASER beam is below the threshold value it indicates that the visibility is less and when the intensity of the LASER beam is more than the threshold value it indicates that the visibility is more [11]. Both the LDR and LASER module are connected to the Arduino. Once the Arduino detects the presence of fog in the environment (i.e. visibility is less than the threshold value) the accident blockage detection system starts working. The accident blockage detection system consists of an ultrasonic sensor mounted on a servo motor. The servomotors have low torque and are of plastic gear type. The servo motor is programmed to rotate from 50 to 1750 to cover most of the portion of the roadway. The ultrasonic sensor is mounted to the servo motor and detects any obstacles on the road it covers. It does that by sending ultrasonic waves and if there were any obstacles the waves reflect back to the ultrasonic sensor module which consists of a receiver to pick up the reflected signals. We have written a program such that the ultrasonic system is going to detect any blockage in two revolutions of the servo motor to improve the accuracy of detection. Both the ultrasonic sensor and servo motor are connected to the Arduino. Once the ultrasonic sensor detects a blockage it sends the data to the Arduino, which then triggers the automatic speed breakers, turns on the warning lights and displays a warning message on the display. The automatic speed breakers consist of two servo motors which lift a piece of hard rubber up to 2.5 cm. The lifting of the speed breaker is done by connecting a rack and pinion assembly to the servo motors. The servo motors which are going to lift the speed breaker have high torque

and are of metal gear types. Whenever a vehicle passes over the speed breaker, the driver gets a physical signal that there is a blockage ahead.

When visibility reappears the intensity of the LASER beam at the LDR side increases and the whole system turns off.

V. RESULTS AND CONCLUSIONS

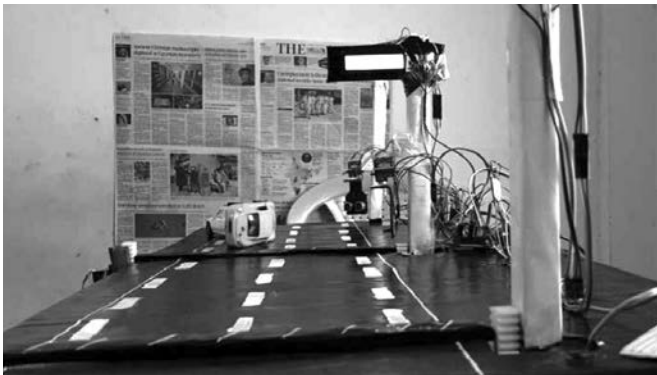


Fig.-: Hardware setup.

Accident is a major problem of India and is also a worldwide problem that affects millions of people worldwide every year. Due to the alarming rate of the increase of accidents, most of the countries are implementing new strategies to minimize the rate of accident. So our country also needs some type of system to be implemented. Self awareness and alertness of the individual and following the traffic rules is the effective way in which an individual can avoid accidents. Moreover our system can be helpful to minimize the accident due to foggy condition. The hardware setup is shown in fig.2. As this system is economical, it can be adopted by any organization on trial basis and if it works as expected then it may be a boon for the people who travel during the foggy condition.

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**Adyasha Pradhan,
Badri Narayan Sahu,
Satyabrata Nayak,
Situparna Sahu,
Snehasis Panda,
Soumya Ranjan Das**

Dept. of EEE

Solar Tracking System

Abstract: Residents that use solar power as their alternative power supply will bring benefits to them. The main objective of this paper is to develop a microcontroller-based solar panel tracking system which will keep the solar panels aligned with the Sun in order to maximize in harvesting solar power. When the intensity of light is decreasing, this system automatically changes its direction to get maximum intensity of light. Light dependent photo resistors are used as the sensors of the solar tracker. For rotating the appropriate position of the panel, a stepper motor is used. This design is covered for a single axis and is designed for residential use.

Keywords: LDR, Solar panel, Microcontroller, Stepper motor

I. INTRODUCTION

This is a solar tracking system which can be used as a power generating method from sunlight. This method of power generation is simple and is taken from natural resource. This needs only maximum sunlight to generate power [1]. This project helps for power generation by setting the equipment to get maximum sunlight automatically. This system is tracking for maximum intensity of light. When there is decrease in intensity of light, this system automatically changes its direction to get maximum intensity of light. We are using two sensors in two directions to sense the direction of maximum intensity of light. The difference between the outputs of the sensors is given to the micro-controller unit. Here we are using the micro-controller for tracking and generating power from sunlight. It will process the input voltage from the oscillator's circuit and control the direction in which the motor has to be rotated so that it will receive maximum intensity of light from the sun [2].

General background: Solar energy is clean and available in abundance. Solar technologies use the sun for provision of heat, light and electricity [3]. These are for industrial and domestic applications. With the alarming rate of depletion of major conventional energy sources like petroleum, coal and natural gas, coupled with environmental caused by the process of harnessing these energy sources, it has become an urgent necessity to invest in renewable energy sources that can

power the future sufficiently. The energy potential of the sun is immense. Despite the unlimited resource however, harvesting it presents a challenge because of the limited efficiency of the array cells. The best efficiency of the majority of commercially available solar cells ranges between 10 and 20 percent. This shows that there is still room for improvement. This project seeks to identify a way of improving efficiency of solar panels. Solar tracking is used. The tracking mechanism moves and positions the solar array such that it is positioned for maximum power output. Other ways include identifying sources of losses and finding ways to mitigate them [4].

Scope of the project: The solar project was implemented using a servo motor. The choice was informed by the fact that the motor is fast, can sustain high torque, has precise rotation within limited angle and does not produce any noise. There is the embedded software section where the Atmega 328P is programmed using the C language before the chip removed from the Arduino board. The Arduino IDE was used for the coding. It is then used as a standalone unit on a PCB during fabrication and display. The design is limited to Single Axis tracking because the use of a dual axis tracking system would not add much value. There is the design of an input stage that facilitates conversion of light into a voltage by the Light Dependent Resistors, LDRs. There is comparison of the two voltages, then the microcontroller uses the

difference as the error. The servo motor uses this error to rotate through a corresponding angle for the adjustment of the position of the solar panel until such a time that the voltage outputs in the LDRs are equal. The difference between the voltages of the LDRs is gotten as analog readings. The difference is transmitted to the servo motor and it thus moves to ensure the two LDRs are an equal inclination. This means they will be receiving the same amount of light. The procedure is repeated throughout the day.

II. METHODOLOGY

The main impulsion is to design a high quality solar tracker. This paper is divided into two parts; hardware and software. It consists of three main constituent which are the inputs, controller photo resistor or Light dependent resistor (LDR) or photocell is a light-controlled variable resistor [6]. LDRs or LDR are very useful especially in light/dark sensor circuits. Normally the resistance of an LDR is very high, sometimes as high as 1000 000 ohms, but when they are illuminated with light resistance drops dramatically. LDR's have low cost and simple structure. The Servo motor can turn either clockwise or anticlockwise direction depending upon the sequence of the logic signals. The sequence of the logic signals depends on the difference of light intensity of the LDR sensors. The principle of the solartracking system is done by Light DependentResistor (LDR). Two LDR's are connected to Arduino analog pin AO to A1 that acts as the input for the system. The built-in Analog-to-Digital Converter will convert the analog value of LDR and convert it into digital. The inputs are from analog value of LDR, Arduino as the controller and the Servo motor will be the output. LDR1 and LDR2 are taken as pair .If one of the LDR gets more light intensity than the other, a difference will occur on node voltages sent to the respective Arduino channel to take necessary action. The Servo motor will move the solar panel to the position of the high intensity LDR that was in the programming.

III. HARDWARE AND SOFTWARE DESCRIPTION

Solar panel:Photovoltaic solar panels absorb sunlight as a source of energy to generate electricity. A photovoltaic (PV) module is a packaged, connected assembly of typically 6x10 photovoltaic solar cells. Photovoltaic

modules constitute the photovoltaic array of a photovoltaic system that generates and supplies solar electricity in commercial and residential applications as shown in Fig-1.

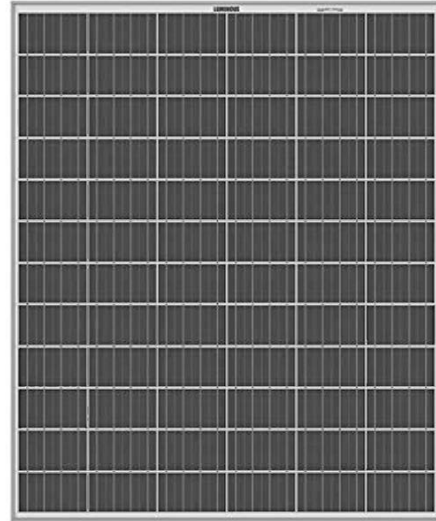


Fig.-1: Solar Panel.

A PV junction box is attached to the back of the solar panel and it is its output interface. Externally, most of photovoltaic modules use MC4 connector's type to facilitate easy weatherproof connections to the rest of the system. Also, USB power interface can be used. Module electrical connections are made in series to achieve a desired output voltage or in parallel to provide a desired current capability (amperes). The conducting wires that take the current off the modules may contain silver, copper or other non-magnetic conductive transition metals. Bypass diodes may be incorporated or used externally, in case of partial module shading, to maximize the output of module sections still illuminated. Some special solar PV modules include concentrators in which light is focused by lenses or mirrors onto smaller cells. This enables the use of cells with a high cost per unit area (such as gallium arsenide) in a cost-effective way. Solar panels also use metal frames consisting of racking components, brackets, reflector shapes, and troughs to better support the panel structure.

Arduino UNO: The ArduinoUno is a microcontroller board based on the ATmega328. It has 14 digital input/output pins (of which 6 can be used as PWM outputs), 6 analog inputs, a 16 MHz ceramic resonator, a USB connection, a power jack, an ICSP header, and a reset button. It contains everything needed to support the microcontroller; simply connect it to a computer with

a USB cable or power it with an AC-to-DC adapter or battery to get started, as shown in Fig-2..

Specifications: Microcontroller : ATmega328 Operating Voltage : 5V, Input Voltage (recommended) : 7-12V Input Voltage (limits): 6-20V Digital I/O Pins: 14 (of which 6 provide PWM output), Analog Input Pins :6, DC Current per I/O Pin:40 mA, DC Current for 3.3V Pin :50 mA Flash Memory :32 KB of which 0.5 KB used by bootloader, SRAM:2 KB EEPROM :1 KB Clock Speed:16 MHz

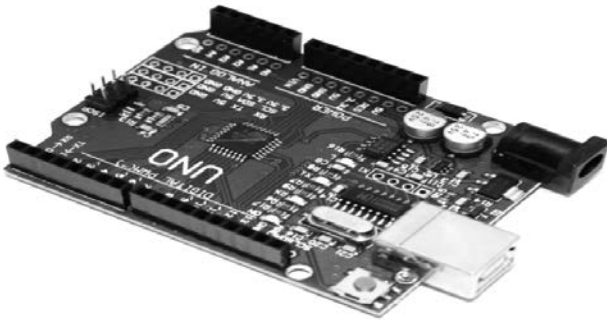


Fig.-2: Arduino UNO microcontroller.

LDR-light dependent resistor: A Light Dependent Resistor (LDR) is also called a photo resistor or a cadmium sulfide (CdS) cell. It is also called a photoconductor. It is basically a photocell that works on the principle of photoconductivity. The passive component is basically a resistor whose resistance value decreases when the intensity of light decreases. This optoelectronic device is mostly used in light varying sensor circuit, and light and dark activated switching circuits. Some of its applications include camera light meters, street lights. **LDR Structure and Working:** The basic structure of an LDR is shown in Fig-3.

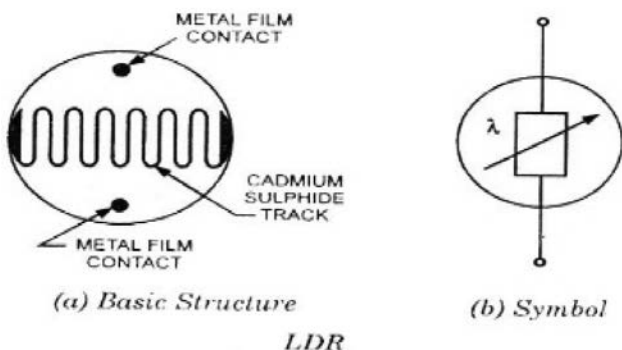


Fig.-3: Light dependent resistor

The snake like track shown below is the Cadmium Sulphide (CdS) film which also passes through the sides. On the top and bottom are metal films which are connected to the terminal leads. It is designed in such a way as to provide maximum possible contact area with the two metal films. The structure is housed in a clear plastic or resin case, to provide free access to external light. As explained above, the main component for the construction of LDR is cadmium sulphide (CdS), which is used as the photoconductor and contains no or very few electrons when not illuminated. In the absence of light it is designed to have a high resistance in the range of megaohms. As soon as light falls on the sensor, the electrons are liberated and the conductivity of the material increases. When the light intensity exceeds a certain frequency, the photons absorbed by the semiconductor give band electrons the energy required to jump into the conduction band. This causes the free electrons or holes to conduct electricity and thus dropping the resistance dramatically (< 1 Kiloohm) [1, 2].

Stepper Motor: A stepper motor is an electro-mechanical device which converts electrical pulses into discrete mechanical movements as shown in Fig-4. The shaft or spindle of a stepper motor rotates in discrete step increments when electrical command pulses are applied to it in the proper sequence. The motors rotation has several direct relationships to these applied input pulses. The sequence of the applied pulses is directly related to the direction of motor shafts rotation. The speed of the motor shafts rotation is directly related to the frequency of the input pulses and the length of rotation is directly related to the number of input pulses applied. Stepper motor is brushless, load independent, has open loop positioning capability, good holding torque and excellent response characteristics. A typical controller for a hybrid stepper motor includes:- (a) Logic Sequence Generator: - Generates programmed logic sequence required for operation of stepper motor. (b) Power Drivers: - These are power switching circuits which ensure a fast rise of current through the phase windings which are to be turned on at a particular step in the logic sequence. Stepper motor driver has been used in the prototype. (c) Current limiting circuits: - These are meant to ensure a rapid decay of current in phase winding that is turned off at a particular step in the logic sequence [7, 9].



Fig.-4: Stepper motor

ALGORITHM

Step 1

Read all analog voltages from analog channels

Step 2

If all voltages are equal then Servo motor will be in stop position.

Step 3

If $LDR1 > LDR2$ Then the Servo motor will rotate clockwise.

Step 4

If $LDR2 > LDR1$ Then the down motor will rotate anti clockwise.

IV. APPLICATION

The industrial development over the past few decades greatly improved the food, clothing, shelters and transportation in the world. However, the advancement in industries accompanies ever-increasing energy consumption and pollution, and therefore, using renewable energy and reducing energy waste are the primary issues. The solar energy is one of the renewable energy, Making good use of it not only will solve the energy shortage problem, but also greatly reduce pollution and ease the effects of global warming.. There are currently two ways to generate electricity from solar energy:

- 1) Converting solar energy to electricity
- 2) Converting solar energy to heat, then heat to electricity

Converting solar energy to electricity: The solar energy converts the energy from sun to electricity. The medium for conversion are PV (Photovoltaic) and CPV

(Concentrated Photovoltaic). The fixed angle is primary to the installation of PV. The installation of CPV has to work with 2-axis devices due to the area of power generation medium. Following the earth rotation, CPV adjusts the angle of the solar panel to the best one according to the incident angle of sunlight.

PV: Most PV solar panels face the sky from a fixed west). Electricity can be generated by direct or indirect (scattered) sunlight.

CPV: Most CPV solar panels are driven by 2 axes (horizontal angle and pitch angle). The angle to catch sunlight is calculated by the host controller, bringing forth the best vertical angle between the sunlight and the panel and further driving the 2 axes to rotate and enhance the efficiency of electricity generation.

The big reflecting mirror gathers beams of light into one light tower where liquid sodium is stored. When the temperature in the tower rises, steam will be generated to push the turbine inside it to further generate electricity.

V. CONCLUSION

Solar trackers generate more electricity than their stationary counterparts due to an increased direct exposure to solar rays. There are many different kinds of solar tracker, such as single-axis and dual-axis trackers, which can help us find the perfect fit for our unique jobsite. Installation size, local weather, degree of latitude, and electrical requirements are all important considerations that can influence the type of solar tracker that's best for us. Solar trackers generate more electricity in roughly the same amount of space needed for fixed tilt systems, making them ideal optimizing land usage. Solar trackers are slightly more expensive than their stationary counterparts, due to more complex technology and moving parts necessary for their operation. Some ongoing maintenance is generally required, though the quality of the solar tracker can play a role in how much and how often this maintenance is needed.

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Harsh Srivastava
Ronit Kumar
PrabirRoul
J.Krithika
Dept. of ECE

Green Process Promises Pristine Graphene in Bulk Using Waste Food, Plastic and Other Materials

Scientists are using high-energy pulses of electricity to turn any source of carbon into turbostratic graphene in an instant. The process promises environmental benefits by turning waste into valuable graphene that can then strengthen concrete and other composite materials. A new process introduced by the Rice University lab of chemist James Tour can turn bulk quantities of just about any carbon source into valuable graphene flakes. The process is quick and cheap; Tour said the “flash graphene” technique can convert a ton of coal, food waste or plastic into graphene for a fraction of the cost used by other bulk graphene-producing methods.”This is a big deal,” Tour said. “The world throws out 30% to 40% of all food, because it goes bad, and plastic waste is of worldwide concern. We’ve already proven that any solid carbon-based matter, including mixed plastic waste and rubber tires, can be turned into graphene.”

As reported in Nature, flash graphene is made in 10 milliseconds by heating carbon-containing materials to 3,000 Kelvin (about 5,000 degrees Fahrenheit). The source material can be nearly anything with carbon content. Food waste, plastic waste, petroleum coke, coal, wood clippings and biochar are all prime candidates.



Noam Chomsky

THE FATHER OF MODERN LINGUISTICS

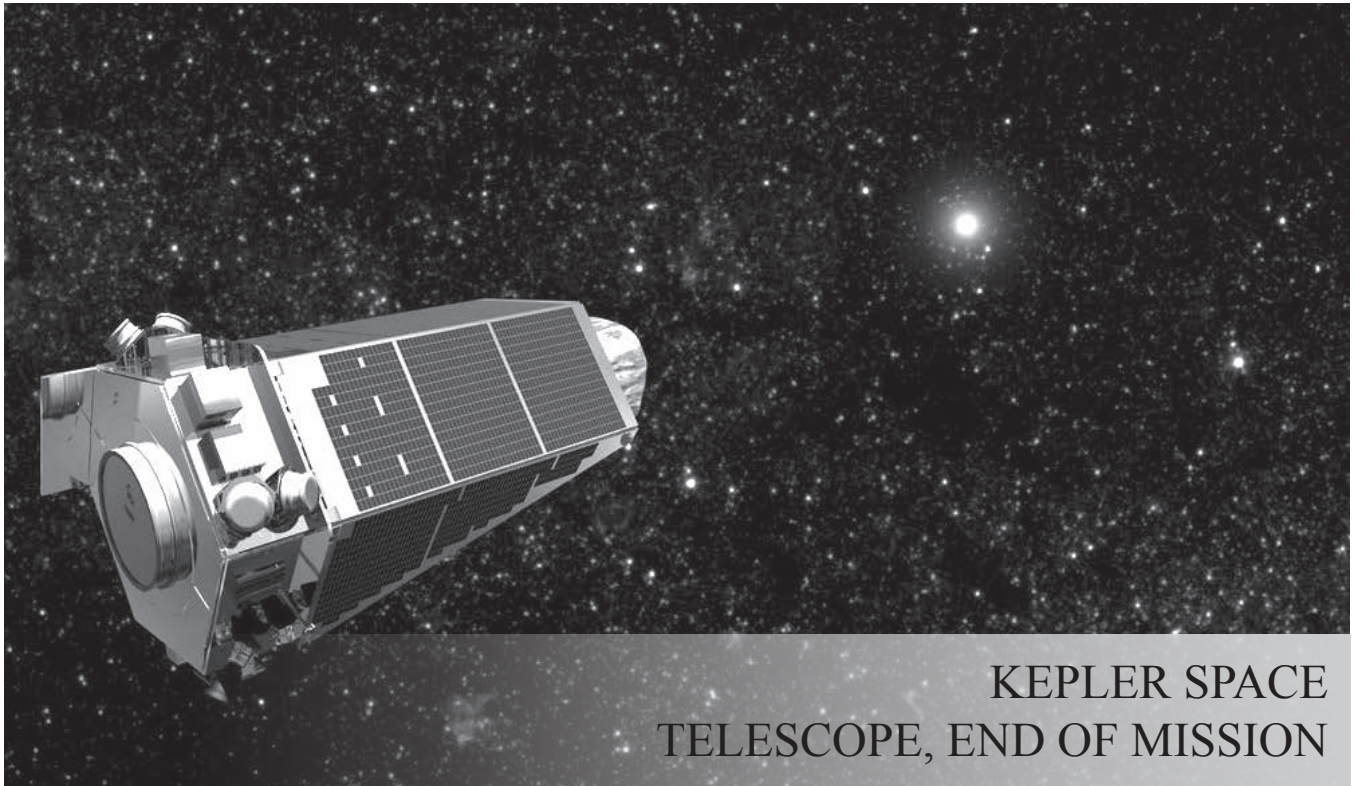
Language is the foundation of civilization. It is the glue that holds the people together and is the best weapon, to be honed. Avram Noam Chomsky, is an American theoretical linguist whose work from the 1950s revolutionized the field of linguistics by treating language as a uniquely human cognitive capacity. Chomsky initiated and sustained what came to be known as the “cognitive revolution.”

This well-versed gentleman born on December 7, 1928 is an American. Born into a middle class Jewish family, at an age of 10 he developed his own interest for writing editorials and self- directed learning. He began taking trips of New York and found books for his voracious reading. During his High School, he excelled academically and joined various clubs and societies. In 1945, Chomsky started studying at the University of Pennsylvania but soon frustrated by his experience at the university, he dropped out and moved to Palestine, where his intellectual curiosity reawakened through conversation with Zellig Harris. Harris introduced Chomsky to the field of theoretical linguistics and convinced him to major in the subject. The Morphophonemic of Modern Hebrew, and especially in The Logical Structure of Linguistic Theory (LSLT), was written while he was a junior fellow.

In 1959 Chomsky published a review of B. F. Skinner’s 1957 book Verbal Behavior in the academic journal language, in which he argued against Skinner’s view of language as learned behavior. The review argued that Skinner ignored the role of human creativity in linguistics and helped to establish Chomsky as an intellectual. In 1961 he was awarded tenure, becoming a full professor in the Department of Modern Languages and Linguistics. Noam Chomsky is known as The Father of modern linguistics. Back in 1957, Chomsky with his revolutionary book “Syntactic structures”, laid the foundation of his non-empiricist theory of language. Chomsky continued to publish his linguistic ideas throughout the decade, including in Aspects of the Theory of Syntax (1965), Topics in the Theory of Generative Grammar (1966), and Cartesian Linguistics: A Chapter in the History of Rationalist Thought (1966).

In the late 1970s and 1980s, Chomsky’s linguistic publications expanded and clarified his earlier work, addressing his critics and updating his grammatical theory. His theory helped in evolution of grammar and vocabulary system. He made absolutely crucial contribution to linguistics, and is extremely well regarded in the field because of that.

Tanmaya Bal
Dept. of ECE



KEPLER SPACE TELESCOPE, END OF MISSION

The Kepler space telescope is a retired space telescope launched by NASA. Named after astronomer Johannes Kepler, the spacecraft was launched on March 7, 2009 into an Earth-trailing heliocentric orbit. The Kepler Mission is specifically designed to survey our region of the Milky Way galaxy to discover hundreds of Earth-size and smaller planets in or near the habitable zone and to determine the fraction of the hundreds of billions of stars in our galaxy. The scientific objective of the Kepler Mission is to explore the structure and diversity of planetary systems. This can be achieved by surveying a large sample of stars, determine the percentage of terrestrial and larger planets that are in or near the habitable zone of a wide variety of stars. The Kepler Spacecraft can detect planets by the telescope, at the heart of which is an array of 42 camera sensors specifically designed to detect alien planets passing in front of their stars. Kepler telescope located is in an earth-trailing orbit around the Sun. Kepler telescope uses infrared and visible light for the visibility. The Kepler telescope sees for 600 light years. The distance to most of the stars for which Earth-size planets can be detected by

Kepler is from 600 to 3,000 light years. Till date, the Kepler space telescope had observed 5,30,506 stars and detected 2,662 planets.

When Kepler was launched in 2009, it had spotted just 300 planets outside of our solar system. It has since identified some 2300 more and thousands await confirmation. Of all the exoplanets we have found, roughly 70 per cent were spotted by Kepler, which searched a small patch of sky for tell-tale dips in the brightness of stars as alien worlds passed in front them, known as transits. Exoplanet is a planet that does not orbit around Sun and instead orbits different star, stellar remnant, or brown dwarf. It is also termed as extra solar planet. The first such planet orbiting star similar to our own Sun was detected only in 1995. Today some 3,600 exoplanets have been found, ranging from rocky Earth sized planets to large gas giants like Jupiter.

NASA has already launched Kepler's successor, the Transiting Exoplanet Survey Satellite, which will scan most of the sky, taking in at least 200,000 nearby stars. Then there are several ground-based observatories, including SPECULOOS in Chile's

Atacama Desert, which has had its sights trained on 500 ultra-cool dwarfs since December 2017. Its prototype discovered seven rocky planets around a star called TRAPPIST-1, three of which sit in the habitable zone.

The transit method of detecting exoplanets looks for dips in the visible light of stars, and requires that planets cross in front of stars along our line of sight to them. Repetitive, periodic dips can reveal a planet or planets orbiting a star.

TESS scientists expect the mission will catalogue thousands of planet candidates and vastly increase

the current number of known exoplanets. Of these, approximately 300 are expected to be Earth-sized and super-Earth-sized exoplanets, which are worlds no larger than twice the size of Earth.

On 30 October, 2019 NASA confirmed that, after almost a decade of service, the Kepler Space Telescope had finally run out of fuel. This means it can no longer reorient itself to point at stars or beam data back to Earth. It will spend its retirement in eternal orbit around the sun, gradually moving further away from home. But Kepler's legacy isn't likely to fade.

Rohit kumar Nayak

"The universe is a pretty big place. If it's just us, seems like an awful waste of space." Carl Sagan

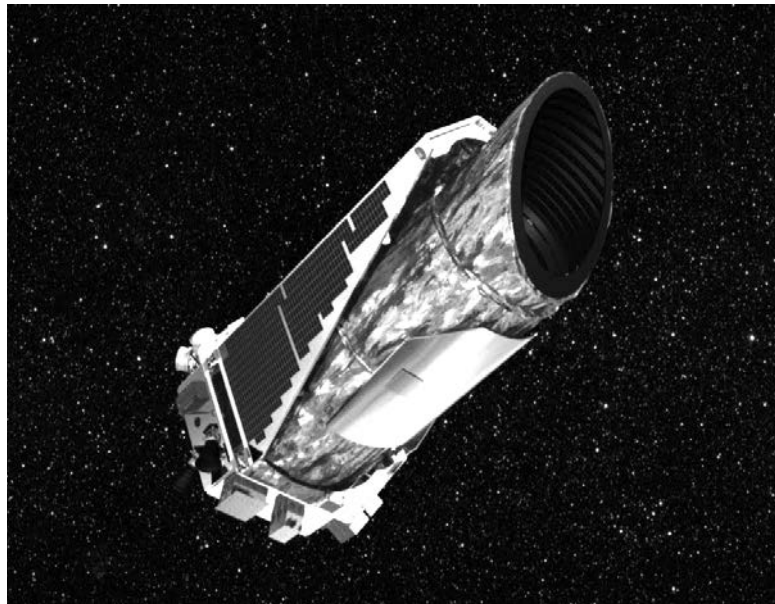


Fig.: Kepler space telescope

Iot Based Real Time Weather Monitoring Station

Abstract: The concept of weather plays important part in our daily life, hence designing wireless system to monitor weather conditions which in turn it can be used as useful tool to impact the human life daily. In this system we design a system to monitor the current environmental conditions such as humidity, temperature, air quality, rainfall levels instantaneously in addition to that storing the collected data and comparing it data with the past gathered data to predict the future changes in the weather conditions. According to our design we will set a clear recommendations and precautions that help in escalating the adverse effects of changes in weather conditions and helping to sustain healthy and hygienic environment. To develop such project, we will use ArduinoUNO as the heart of the system and other supportive sensors. Finally achieving high degree of reliability, compactness, modularity, and cost effectiveness for our design is the aim and final goals targeted.

Keywords: Internet of Things (IoT), Arduino, carbon oxide (CO), Light Emitting diode (LED).

I. INTRODUCTION

The internet of Things (IoT) is viewed as an innovation and financial wave in the worldwide data industry after the Internet. The IoT is a wise system which associates all things to the Internet with the end goal of trading data and conveying through the data detecting gadgets as per concurred conventions. It accomplishes the objective of keen recognizing, finding, following, observing, and overseeing things. It is an augmentation and extension of Internet-based system, which grows the correspondence from human and human to human and things or things and things. In the IoT worldview, many articles encompassing us will be associated into systems in some shape. It is a current correspondence paradigm that envisions a near future, in which the objects of regular day to day existence will be outfitted with microcontrollers, handsets for computerized correspondence, and reasonable convention stacks that will make them ready to speak with each other and with the clients, turning into a vital piece of the Internet. The IoT idea, consequently, goes for making the Internet much more immersive and unavoidable. Present innovations in technology mainly focus on controlling and monitoring of different activities. These are increasingly emerging to reach the human needs. Most of this technology is focused on efficient monitoring and controlling different activities. An efficient environmental monitoring system is required to monitor and assess the conditions in case of exceeding the prescribed level of parameters (e.g., noise, CO and

radiation levels). When the objects like environment equipped with sensor devices, microcontroller and various software applications becomes a self-protecting and self-monitoring environment and it is also called as smart environment. In such environment when some event occurs the alarm or LED alerts automatically. The effects due to the environmental changes on animals, plants and human beings can be monitored and controlled by smart environmental monitoring system. By using embedded intelligence into the environment makes the environment interactive with other objectives, this is one of the applications that smart environment targets [1]. Human needs demand different types of monitoring systems these are depends on the type of data gathered by the sensor devices. Sensor devices are placed at different locations to collect the data to predict the behavior of a particular area of interest. The main aim of this paper is to design and implement an efficient monitoring system through which the required parameters are monitored [2]. A solution for monitoring the temperature, humidity and CO levels i.e., any parameter value crossing its threshold value ranges, for example CO levels in air in a particular area exceeding the normal levels etc., in the environment using wireless embedded computing system is proposed in this paper. The solution also provides an intelligent remote monitoring for a particular area of interest.

II. SYSTEM ARCHITECTURE

The implemented system consists of a microcontroller (ATmega328) as a main processing unit for the entire system and all the sensor and devices can be connected with the microcontroller [3]. The sensors can be operated by the microcontroller to retrieve the data from them and it processes the analysis with sensor data and updates it.

A. Arduino UNO



Fig.-1: Arduino UNO.

Fig.1 shows an Arduino board which use a variety of microprocessors and controllers. The boards are equipped with sets of digital and analog input/output (I/O) pins that may be interfaced to various expansion boards or breadboards (shields) and other circuits. The boards feature serial communications interfaces, including Universal Serial Bus (USB) on some models, which are also used for loading programs from personal computers [4]. The microcontrollers can be programmed using C and C++ programming languages. In addition to using traditional compiler toolchains, the Arduino project provides an integrated development environment (IDE) based on the Processing language project. The board is equipped with series of digital input/output pins that may be interfaced to various expansion boards [6]. The board is equipped with series of digital input/output pins that may be interfaced to various expansion boards. The board has 14 digital pins, 6 analog pins and programmed with Arduino IDE via type B USB cable.

B. Temperature and humidity sensor (DHT11)



Fig.-2: DHT11

The DHT11 shown in fig. 2 is an essential, ultra-minimal effort computerized temperature and humidity sensor. It utilizes a capacitive humidity sensor and a thermistor to gauge the surrounding air and releases a digital data on the data pin (no analog information pins required). The main genuine drawback of this sensor is you can just get new information from it once every 2 seconds, so when utilizing our library, sensor readings can be up to 2 seconds old. It works on 3V to 5V power supply. Good for 20% - 80% humidity readings with 5% accuracy and for 0-50°C temperature readings $\pm 2^\circ\text{C}$ accuracy. Small size & low consumption & long transmission distance (20m) enable DHT11 to be suited in all kinds of harsh application occasions. Single row packaged with four pins, making the connection very convenient.

C. Air Quality Sensor (MQ135)

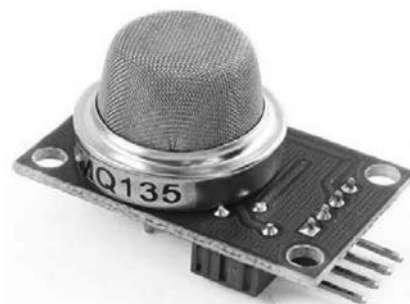


Fig.-3: MQ135

This is suitable for sensing CO concentrations in the air. Carbon monoxide sensor, suitable for sensing CO concentration in air is shown in fig. 3. The MQ-6 can sense CO-gas concentration somewhere in the range of 20 to 2000ppm. This sensor has a high affectability and quick

reaction time. The sensor's yield is analog resistance [7]. The drive circuit is exceptionally straightforward; you should simply control the heater curl with 5V, include a load resistance, and associate the output to an ADC. The standard reference strategy for the estimation of carbon monoxide concentration in air depends on the ingestion of infrared radiation by the gas in a no dispersive photometer. This technique is reasonable for stable establishments at fixed site monitoring stations. All the more as of late, convenient carbon monoxide analyzers with datalogging have turned out to be accessible for individual presentation observing. These estimations depend on the electrochemical responses between carbon monoxide and deionized water, which are detected by exceptionally planned sensors.

D. Rain Sensor Module

The rain sensor module is an easy tool for rain detection. It can be used as a switch when raindrop falls through the raining board and also for measuring rainfall intensity. The module features, a rain board and the control board that is separate for more convenience, power indicator LED and an adjustable sensitivity through a potentiometer as in fig. 4.



Fig.-4: Rain Sensor

The analog output is used in detection of drops in the amount of rainfall. Connected to 5V power supply, the LED will turn on when induction board has no rain drop, and DO output is high [8]. When dropping a little amount water, DO output is low, the switch indicator will turn on. Brush off the water droplets, and when restored to the initial state, outputs high level.

IV. CIRCUIT & BLOCK DIAGRAM

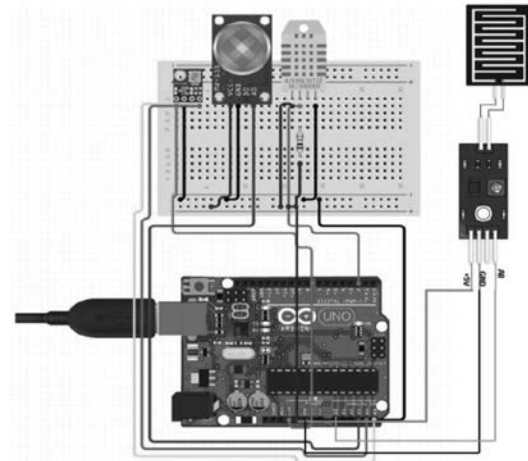


Fig.-5: Circuit Diagram

Fig. 5 shows the circuit diagram where we can read weather parameters such as temperature, air quality level, rainfall and humidity. After that, the instantaneous results and the values of the rain gauge are sent to the end user. The flowchart of overall process is shown in Fig-6

V. FLOW CHART

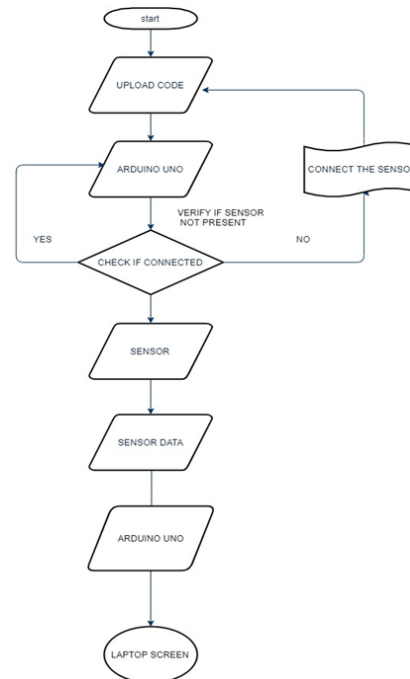


Fig.-6: Overall Process Flow

VI. CONCLUSIONS

By keeping the embedded devices in the environment for monitoring enables self-protection (i.e., smart environment) to the environment. To implement this need to deploy the sensor devices in the environment for collecting the data and analysis. By deploying sensor devices in the environment, we can bring the environment into real life i.e. it can interact with other objects through the network. In the proposed architecture functions of different modules were discussed. The noise and air pollution monitoring system with Internet of Things (IoT) concept experimentally tested for monitoring two parameters. It also sent the sensor parameters to the cloud (Google Spread Sheets). This data will be helpful for future analysis and it can be easily shared to other end users. This model can be further expanded to monitor the developing cities and industrial zones for pollution monitoring. To protect the public health from pollution, this model provides an efficient and low-cost solution for continuous monitoring of environment.

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**Saswati Panda
Amanwit Kumar
Soumen Pattnaik
Shrinivas Moharaj
Ankit Sen
Manish Haldar
Prabhudutta Ray**

Dept. of EEE

AI-Based Music Composition

Abstract: Deep learning techniques for generating music that has melody and harmony and is similar to music compositions by human beings is something that has fascinated researchers in the field of artificial intelligence. Nowadays, deep learning is being used for solving various problems in numerous artistic fields. There has been a new trend of using deep learning models for various applications in the field of music that has attracted much attention, and automated music generation has been an active area of research that lies in the cross section of artificial intelligence and audio synthesis. Previously, the work in automated music generation was solely focused on generating music, which consisted of a single melody, which is also known as monophonic music. More recently, research work related to the automated generation of polyphonic music, which consists of multiple melodies, has met partial success with the help of estimation of time series probability density (stochastic model). In this work, the Restricted Boltzmann Machine (RBM) is used for music generation by training it on a collection of Musical Instrument Digital Interface (MIDI) files. other supportive sensors. Finally achieving high degree of reliability, compactness, modularity, and cost effectiveness for our design is the aim and final goals targeted.

Keywords: Artificial Intelligence, Deep Learning, RBM, MIDI, ployphonic, monophonic, stochastic

I. INTRODUCTION

The field of music composition is inherently related to human expressions and emotions and, therefore, is not something that machines could reproduce. The researchers in the field of Artificial Intelligence (AI) have demonstrated the opposite though. AI has improved in many fields, allowing computers to learn from its own experience as humans do. Music is one of them. Creating compelling music is something unique to humans until now. In contrast there has been some great advancements in the field of music composition using machine learning algorithms. This work involves the generation of music by training the Neural Network model on a set of homogeneous instrumental music to produce new music. Applying machine learning techniques in the domain of music generation is a challenging area which yet to be explored. This method will model the melodic repetition that is found in almost all kinds of songs. Music theory (sheets) notes are related to each other. These notes are just ways to mapping out pitches or sound waves. The common problem to tackle to fit the model or make the model learn in such a way that it recognizes the relation between notes. An example music sheets and notes are presented in fig. 1.

The data sets in real life are more complex. Application of machine learning algorithm on the data set requires data cleansing. This is even more difficult when the data is in an unstructured format such as image or audio. This is so because one has to represent image/audio data in a standard way for its applicability.



Fig.-1: Music Sheets & Notes

This work involves the generation of music by training the Restricted Boltzmann Machine (RBM)[14] model using TensorFlow on a set of homogeneous instrumental music to produce new music [15]. Applying machine learning techniques in the domain of music generation is an interesting subject that has not been studied in depth. This model may be able to model the melodic repetition that is found in almost all kinds of songs.

II. MIDI (MUSICAL INSTRUMENT DIGITAL INTERFACE)

MIDI is a technical standard that describes a communications protocol, digital interface, and electrical connectors that connect a wide variety of electronic musical instruments, computers, and related audio devices for playing, editing and recording music. [2] A single MIDI link through a MIDI cable can carry up to sixteen channels of information, each of which can be routed to a separate device or instrument. This could be sixteen different digital instruments, for example.

MIDI carries event messages, data that specify the instructions for music, including a note's notation, pitch, velocity (which is heard typically as loudness or softness of volume), vibrato, panning to the right or left of stereo, and clock signals (which set tempo). When a musician plays a MIDI instrument, all of the key presses, button presses, knob turns and slider changes are converted into MIDI data. One common MIDI application is to play a MIDI keyboard or other controller and use it to trigger a digital sound module (which contains synthesized musical sounds) to generate sounds, which the audience hears produced by a keyboard amplifier. MIDI data can be transferred via MIDI cable, or recorded to a sequencer to be edited or played back.

A file format that stores and exchanges the data is also defined. Advantages of MIDI include small file size, ease of modification and manipulation and a wide choice of electronic instruments and synthesizer or digitally-sampled sounds. A MIDI recording of a performance on a keyboard could sound like a piano or other keyboard instrument; however, since MIDI records the messages and information about their notes and not the specific sounds, this recording could be changed to many other sounds, ranging from synthesized or sampled guitar or flute to full orchestra. A MIDI recording is not an audio signal, as with a sound recording made with a microphone.

Prior to the development of MIDI, electronic musical instruments from different manufacturers could generally not communicate with each other. This meant that a musician could not, for example, plug a Roland keyboard into a Yamaha synthesizer module. With MIDI, any MIDI-compatible keyboard (or other controller device) can be connected to any other MIDI-compatible sequencer, sound module, drum machine, synthesizer, or computer, even if they are made by different manufacturers.

MIDI technology was standardized in 1983 by a panel of music industry representatives, and is maintained by the MIDI Manufacturers Association (MMA). All official MIDI standards are jointly developed and published by the MMA in Los Angeles, and the MIDI Committee of the Association of Musical Electronics Industry (AMEI) in Tokyo. In 2016, the MMA established the MIDI Association (TMA) to support a global community of people who work, play, or create with MIDI. An example of a MIDI signal is presented in fig. 2.

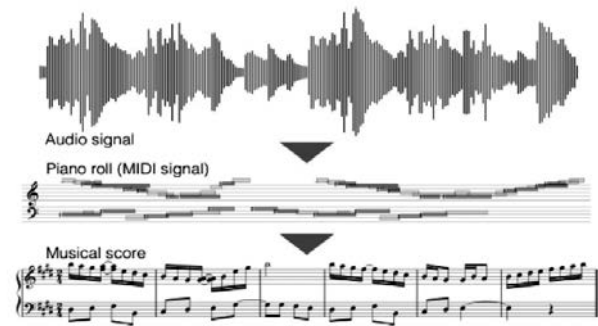


Fig.-2: MIDI Signal

It's important to realise that MIDI doesn't contain any actual sound itself, it's simply instructions for how other instruments should create sound. In this sense it works like the conductor of an orchestra. The conductor doesn't make any sound by their self, but they send signals to all of the other members of the orchestra that tell them what to play, when to play, and what expression to use while playing. MIDI is like an electronic version of that [15].

III. RESTRICTED BOLTZMANN MACHINE (RBN)

RBM is a variant of Boltzmann Machine. RBM was invented by Paul Smolensky in 1986 with name Harmonium. In the mid-2000, Geoffrey Hinton and collaborators invented fast learning algorithms which were commercially successful [14]. RBM can be use in many applications like Dimensionality reduction, Collaborative Filtering, Feature Learning, Regression Classification and Topic Modelling [3]. It can be trained in either Supervised or Unsupervised ways, depending on the task.

A. RBM's Architecture

This Restricted Boltzmann Machine (RBM) has an input layer (also referred to as the visible layer) and one single hidden layer and the connections among the neurons are restricted. Neurons are connected only to the neurons in other layers but not to neurons within the same layer. There are no connections among visible neurons to visible neurons. There are no connections among hidden neurons to hidden neurons. In RBM visible and hidden neurons connections form a bipartite graph. An RBM is considered restricted because no two nodes in the same layer share a connection.

B. How RBM works?

RBM has two phases: Forward Pass and Backward Pass or Reconstruction. RBM takes the inputs and translates

them to a set of numbers that represents them (forward pass). Then, these numbers can be translated back to reconstruct the inputs(backward pass). In the forward pass, an RBM takes the inputs and translates them into a set of numbers that encode the inputs. In the backward pass, it takes this set of numbers and translates them back to form the re-constructed input. Through several forward and backward passes, an RBM is trained to reconstruct the input data. Three steps are repeated over and over through the training process-

(a) With a forward pass, every input is combined with an individual weight and one overall bias, and the result is passed to the hidden layer which may or may not activate.

(b) Next, in a backward pass, each activation is combined with an individual weight and an overall bias, and the result is passed to the visible layer for reconstruction.

(c) At the visible layer, the reconstruction is compared against the original input to determine the quality of the result. A trained RBM can reveal which features are the most important ones when detecting patterns.

A well-trained net will be able to perform the backwards translation with a high degree of accuracy. In both steps, the weights and biases have a very important role. They allow the RBM to decipher the interrelationships among the input features, and they also help the RBM decide which input features are the most important when detecting patterns.

An important note is that an RBM is actually making decisions about which input features are important and how they should be combined to form patterns. In other words, an RBM is part of a family of feature extractor neural nets, which are all designed to recognize inherent patterns in data. These nets are also called autoencoders, because in a way, they have to encode their own structure.

1. Forward Pass: One training sample X given as a input to all the visible nodes, and pass it to all hidden nodes. Processing happens at each hidden layers node. This computation begins by making stochastic decisions about whether to transmit that input or not (determine the state of each hidden layer) [10].
2. Backward Pass: The RBM reconstructs data by making several forward and backward passes between the visible and hidden layers. So, in the Backward Pass (Reconstruction), the samples from the hidden layer play the role of input.

RBM learns a probability distribution over the input, and then, after being trained, the RBM can generate new samples from the learned probability distribution. As we know, probability distribution is a mathematical function that provides the probabilities of occurrence of different possible outcomes in an experiment [9]. The screen short of hidden nodes and visible nodes mapping in RBN is presented in fig. 3.

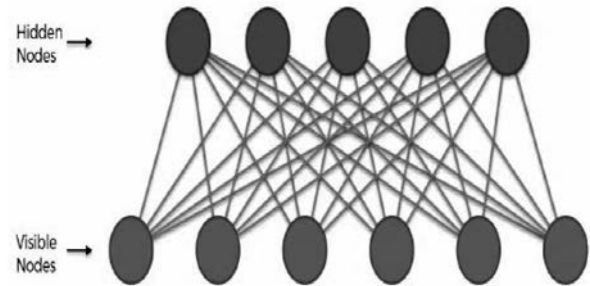


Fig.-3: Restricted Boltzmann Machine

C. Gibbs Sampling

Gibbs sampling is named after the physicist Josiah Willard Gibbs, in reference to an analogy between the sampling algorithm and statistical physics. The algorithm was described by brothers Stuart and Donald Geman in 1984, some eight decades after the death of Gibbs [4]. In its basic version, Gibbs sampling is a special case of the Metropolis Hastings algorithm. However, in its extended versions, it can be considered a general framework for sampling from a large set of variables by sampling each variable (or in some cases, each group of variables) in turn, and can incorporate the Metropolis Hastings algorithm (or more sophisticated methods such as slice sampling, adaptive rejection sampling and adaptive rejection Metropolis algorithms) to implement one or more of the sampling steps.

Gibbs sampling is applicable when the joint distribution is not known explicitly or is difficult to sample from directly, but the conditional distribution of each variable is known and is easy (or at least, easier) to sample from. The Gibbs sampling algorithm generates an instance from the distribution of each variable in turn, conditional on the current values of the other variables. It can be shown that the sequence of samples constitutes a Markov chain, and the stationary distribution of that Markov chain is just the sought-after joint distribution. [5] In statistics, Gibbs sampling or a Gibbs sampler is a Markov chain Monte Carlo (MCMC) algorithm for obtaining a sequence of observations which are approximately from a specified multivariate probability distribution, when direct sampling is difficult. This sequence can be used

to approximate the joint distribution (e.g., to generate a histogram of the distribution); to approximate the marginal distribution of one of the variables, or some subset of the variables (for example, the unknown parameters or latent variables); or to compute an integral (such as the expected value of one of the variables). Typically, some of the variables correspond to observations whose values are known, and hence do not need to be sampled. Gibbs sampling is commonly used as a means of statistical inference, especially Bayesian inference. It is a randomized algorithm (i.e. an algorithm that makes use of random numbers), and is an alternative to deterministic algorithms for statistical inference such as the expectation maximization algorithm (EM). The Gibbs sampling on example data is presented in fig. 4.

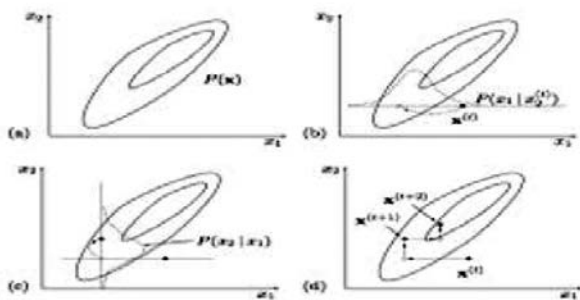


Fig.-4: Gibbs Sampling

IV. THE MODEL AND IMPLEMENTATION DETAILS

Here new music files are generated by applying the model on the music files as input data. The major task here is to take some existing music data and then train the model using this existing data. The model has to learn the patterns in music. Once it learns, the model should be able to generate new music. It understands the patterns of music to generate new music. The main focus of the work is to generate a decent quality music which may be melodious and good to hear.

The algorithm most often used to train RBMs, that is, to optimize the weight vector W , is the contrastive divergence (CD) algorithm. The algorithm performs Gibbs sampling and is used inside a gradient descent procedure (similar to the way back propagation is used inside such a procedure when training feedforward neural nets) to compute weight update. The basic contrastive divergence procedure for a single sample can be summarized as follows:

1. Take a training sample v , compute the probabilities

of the hidden units and sample a hidden activation vector h from this probability distribution.

2. Compute the outer product of v and h and call this the positive gradient.
3. From h , sample a reconstruction v' of the visible units, then resample the hidden activations h' from this. (Gibbs sampling step).
4. Compute the outer product of v' and h' and call this the negative gradient.
5. Let the update to the weight matrix W be the positive gradient minus the negative gradient, times some learning rate: $\Delta W = \epsilon (vh^T - v'h'^T)$
6. Update the biases a and b analogously: $\Delta a = \epsilon (v - v')$, $\Delta b = \epsilon (h - h')$.

The presented model is implemented in PyCharm using tensorflow and executed in command prompt. The different steps of implementation are:

1. Hyperparameters:

The hyperparameters are initially decided and assigned with required values. The hyperparameters used are:

- `Lowest_node`: The index of the lowest note on the piano roll
- `highest_node`: The index of the highest note on the piano roll
- `note_range`: Difference between the highest and lowest note
- `num_timesteps`: The number of timesteps created at a time
- `n_visible`: the size of the visible layer
- `n_hidden`: the size of the hidden layer
- `num_epochs`: The number of training epochs going to run
- `batch_size`: The number of training examples going to send through the RBM at a time,
- `lr (learning rate)`: The learning rate of the model.

2. Tensorflow Variables:

- x: The placeholder that holds the value
- W: Weight matrix that stores the edge weights
- bh: Bias vector for hidden layer
- bv: Bias vector for visible layer

3. Generative Algorithm

- Gibbs Sampling: We calculate the probabilities that neurons from the hidden layer are activated based on the input values on visible layer. The gibbs sample of x :

```
x_sample=gibbs_sample(1)
```

The sample of the hidden nodes, starting from the visible state of x:

```
hk = sample(tf.sigmoid(tf.matmul(xk, W) + bh))
[6]
```

- Contrastive Divergence: Update the value of W,bh and bv based on the difference between the samples that are drawn and the original values.

4) Training the model

The model is initialized using init and sess_run(init). Then we run through all of the training data num_epochs times. After training process is complete, a gibbs chain is run where the visible nodes are initialized to zero. At last the vector is converted and reshaped and saved as midi file. This completes the training step. The audion signal of output music is shown in Fig-5

The output screen short of a music generated by the model is shown in fig. 5.



Fig.-3: 5 Audio Signal of output music

V. CONCLUSIONS AND FUTURE SCOPE

Artificial music generation has been important for many artists as it provides new and unique music which can be used for commercial purpose. It can also be used according to the need of a person for creating beautiful

and emotional music. In this work, new music is generated by training the Restricted Boltzmann Machine (RBM) model using TensorFlow on a set of homogeneous instrumental music. However, in the presented model, a small sequence of unique harmonic music is generated by training of Restricted Boltzmann Machine (RBM). This work may be enhanced to generation of music for infinite time. Also, the model may further be enhanced to generate music corresponding to the emotion of the person predicted by the model. Different kinds of music such as guitar tones, natural melodies etc can also be included.

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Arya Kumari
Divya
Sanghamitra Routray
Shubham Nishad
8th Sem, CSE

Promising, Economical Solution to Indoor Air Pollution.

Most of the human beings spend most of our times indoors, be it home or working in office environment. Hence it is essential that indoor air quality is of high standard. Indoor air gets polluted not only from pollution outside but also from chemicals used in paints and adhesive. With rising pollution levels and poor air quality index there has been a surge in the sale of indoor air purifiers across major Indian cities. However as per NASA report indoor plants help improve the indoor air quality. They conducted a Clean Air Study, according to which some of the indoor plants are best in removing benzene, formaldehyde, trichloroethylene, xylene, and ammonia from our surroundings – chemicals that have been linked to adverse health effects. As per the report plants like peace lily, aloe-vera, spider plant, English ivy, Chinese evergreen and snake plant are some of the plants that purify the air by removing these harmful chemicals and also enhancing the oxygen level. The report further suggests having one plant per hundred square feet of home/office space.

Source: "<https://ntrs.nasa.gov/archive/nasa/casi.ntrs.nasa.gov/19930073077>."

QoS Provisioning and Achieving Security in Next Generation Wireless Networks and Cloud Computing

Keywords: Quality of Service, Cloud Computing, Software Defined Network, MANET, VANET, Vehicular Social Networking, Digital Signature.

Quality of service (QoS) is a performance level of a service offered by the network to the user. The goal is to achieve more deterministic network behavior, so that information carried by the network can be better delivered and network resources can be better utilized. The next generation wireless network is expected to support a wide range of real-time, communication-intensive applications because of the extensive demand of Cloud Computing and Internet of Things (IoT). The next generation network is seen to be the hybridization of different networks both infrastructure and infrastructure less with different networking technologies such as MANET/VANET, Sensor Networks (SN) and Software Defined Network (SDN). For a huge amount of data (Big Data), large network bandwidth and high processing power are needed to meet the future requirements. The quality-of-service (QoS) requirements for the timely delivery of the huge amount of multimedia data raise new challenges for the development of future networks. There are various issues which need to be investigated in order to guarantee QoS, such as Traffic Specification, Data Packet Routing, Call Admission Control, Resource Reservation, Packet Scheduling, Wireless Channel Characterization and Security Issues. In this work we survey and investigate the efficient and secured data dissemination issues in various scenarios including the current and future generation wireless networks with cloud computing. One of the key issues is QoS routing, which allows selecting network routes with sufficient resources for requested QoS parameters. The goal of QoS routing is two-fold: satisfying the QoS requirements for every admitted connection and achieving global efficiency in resources utilization. Many unicast/multicast QoS routing algorithms were published recently. However, there still exist a lot of unsolved problems in this area because of the new challenging scenario which is a hybrid network consisting of

MANET/VANET, Sensor Networks, Cloud Computing and Internet of Things (IoT). A few examples are (1) There lacks a simple solution with predictable performance and adjustable overhead for the NP-complete multi-constraint routing problem. (2) All existing algorithms are tailored towards specific problems, and there lacks a simple general routing framework which can be easily extended to handle new problems. (3) Most routing algorithms assume the availability of precise state information about the network, which however is impractical in the real world. (4) Further resolving security issues are an important concern for the success of the future networks. In future Vehicular Cloud Computing and Vehicular Social Network will play an important role in the information dissemination and various other applications of VANET. Achieving QoS in Vehicular cloud computing, Extracting and using useful information from the vehicular social networking is an important issue because of the rapid growth of Internet of Vehicle users and mobile devices. We address the above issues, and the goal of this dissertation is to provide simple, general and extensible solutions for achieving QoS in terms for efficient data dissemination, security, and efficient use of new applications of next generation wireless networks. We study different solution methods, compare them and outline the challenges and propose various algorithms and techniques based on different network state models, evaluate these algorithms by analysis and simulation, discuss their strengths and weaknesses of different strategies and solutions, and compare them with the existing algorithms.

Asif Uddin Khan
BK Rath

Recycling of Plastic Waste in India

According to the recent United Nations Environment Programme (UNEP) report, the world produces 400 million tonnes of single-use plastic (SUP) waste annually (47 per cent of the total plastic waste). As per CPCB reports, plastic contributes to 8% of the total solid waste, with Delhi producing the maximum quantity followed by Kolkata and Ahmedabad. Interestingly, almost 60 per cent of the total plastic waste generated in India gets recycled while the remainder makes its way into the environment. At this juncture, India needs robust and stringent waste management tools to substantially improve the situation. Households generate a lot of plastic waste, of which water and soft drink bottles form a large number. In India, around 43% of manufactured plastics are used for packaging purpose and most are of single use.

The seas near Mumbai, Kerala and the Andaman and Nicobar Islands are among the worst polluted in the world. Plastic debris affects at least 267 species worldwide, including 86% of all sea turtle species, 44% of all seabird species, and 43% of all marine mammal species. Significant amount of toxic heavy metals like copper, zinc, lead and cadmium recovered from plastic wastes from sea shores have an adverse effect on the coastal ecosystems. Like any other country, waste management is a pressing issue in India, especially with the unceasing growth of consumerism throughout the nation. Since the 1980s, a number of countries have predicted the inexorable unintended consequences of using plastics in the long run and began to address the matter by adopting legal measures, instruments or grave consequences to stem the distribution and consumption of plastics.

Plastic Waste Management Rules 2016 mandated the producers and brand owners to devise a plan in consultation with the local bodies to introduce a collect-back system. This system is known as the Extended Producers Responsibility (EPR). There is an urgent need to establish a monetized collection model for plastic waste that has economic returns for all those involved. Virgin plastics (e.g. those used in food packets, etc) should be collected separately because of the higher value it draws. Some plastics have fibers which shorten every time it is recycled. Thus, a plastic can be recycled 7-9 times before it is no longer recyclable. Recycling is good - every ton of plastic waste recycled results in a saving of approximately 3.8 barrels of petroleum. Technologies are available in India that can convert 1 Kg of plastic to 750 ml of automotive grade gasoline. Shredded plastic waste can be used in laying roads. Jambulingam Street in Chennai was one of India's first plastic roads built in 2002. In 2015-16, the National Rural Road Development Agency laid around 7,500 km of roads using plastic waste. Co-processing of plastic in cement kilns offers a sound, environmentally viable mechanism to process non- recyclable, combustible plastic waste.

However, long-term impacts and overall problem of the issue can be addressed only by establishing better waste management systems by improving source segregation, designing an effective municipal solid waste (MSW) plans, ensuring collection and transportation of segregated waste, and encouraging the country to identify and use affordable plastic alternative products.

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Contents

Editorial	2
DD Feature	3
Profile of a Scientist	21
PhD Synopsis	34
Environmental Awareness & Concerns	35

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