SPECIAL FEATURE

Redundancy Allocation and Optimization of Parallel-Series Systems

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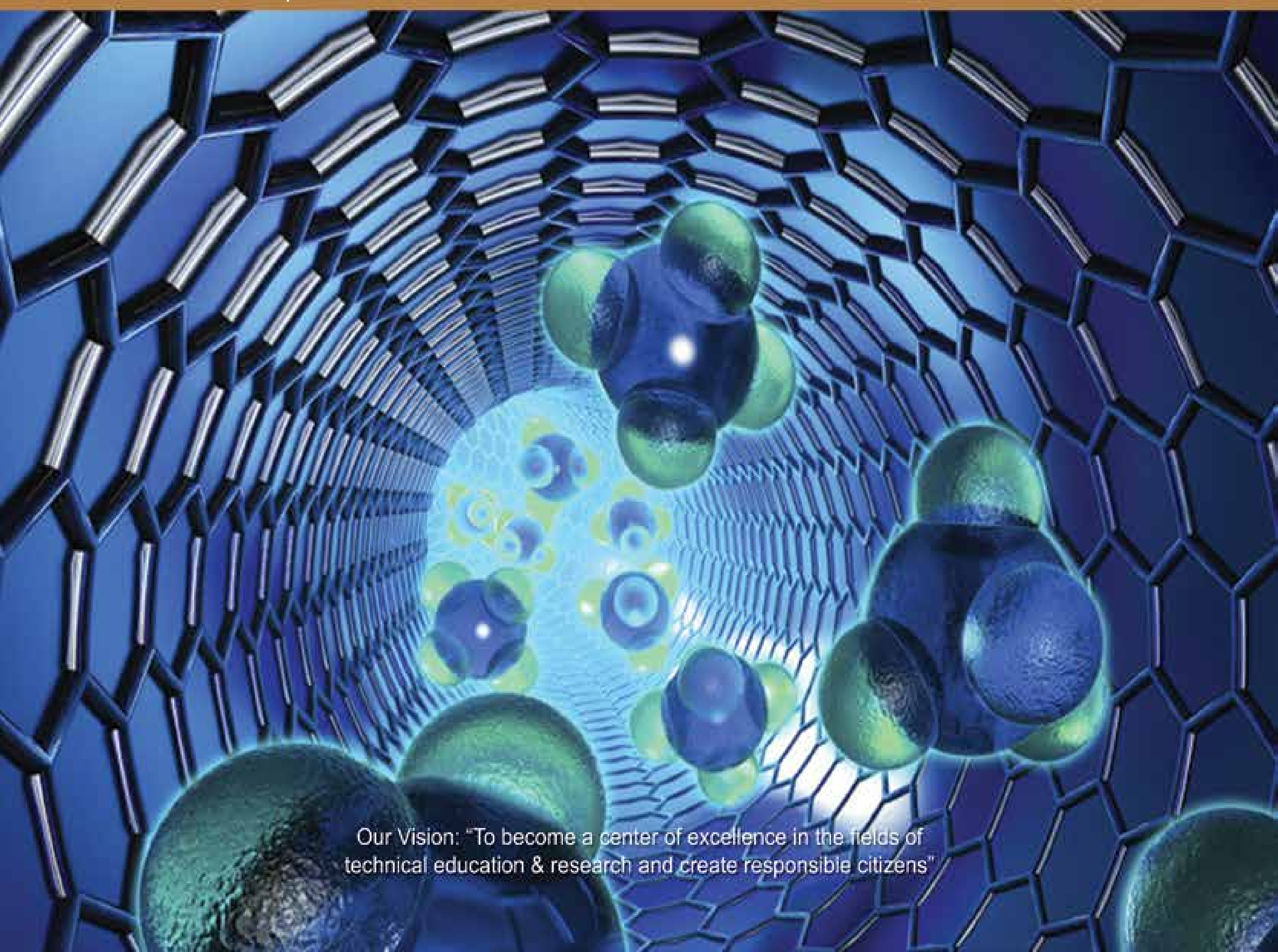


Nanotechnology

Good things come in small packages and so does nanotechnology. Nanotechnology is about 30 years old and yet it has made significant progress. It has revolutionized different industry sectors including medicine, electronics, materials science and chemistry. Nanotechnology involves study and use of structures between one nanometer and a hundred nanometers in size, where one nanometer is a billionth of a meter. For comparison, a sheet of a newspaper is one million nanometers thick!!

The development of microscopes has allowed scientists to work with nano particles. Nanotechnology allows one to manipulate the properties of particle at a minute level. This has opened a world full of application possibilities in different domains.

Nanotechnology is used in medicine to deliver drugs accurately to the diseased cells. The nano medicines/ drugs are of the size of tissue cells. This is being used by oncologists to minimize the effects of chemotherapy.

Nanotechnology is used in the fabrication of powerful computing devices. Electronic circuits are made from carbon nanotubes help in developing smaller computing devices with long lasting batteries.

Nanotechnology improves a vehicle's fuel efficiency. This is possible due to reduced corrosion. Vehicle parts are made of nano materials that are lighter, stronger and yet chemically less reactive than metals. Nano filters are used to filter out dust particles in air from reaching the combustion chambers.

Nanotechnology could be used to create cleaner and safer environment. Scientists at IIT Madras have developed a water purification system which can provide affordable and clean drinking water using a nano-filter. These water purifiers have huge potential in providing clean drinking water to rural communities in the world. University of New South Wales has fabricated nano-filters which are capable of filtering out heavy metals and oil from water at extremely high speed. The liquid metal chemistry used in these filters can be used to clean up the water bodies of oil spills.

Nanotechnology has helped in the development of nano-sized particles of carbon like nanotubes and Buckyballs which are extremely strong materials. Their strength can be attributed to special characteristics of the bonds between the carbon atoms. One of the applications of the nano tubes is in the development of t-shirts concealing bullet proof vests.

Researchers are developing wires containing carbon nanotubes that will have much lower resistance than the high-tension wires currently used in the electric grid, thus reducing transmission power losses. Nanotechnology is already being used to develop many new kinds of batteries that are quicker-charging, more efficient, lighter weight, have a higher power density, and hold electrical charge longer.

Nanotechnology is increasingly being used in diverse domains. The gadgets and equipment are becoming more sophisticated and improving the quality of our lives substantially.

Dr. Pamela Chaudhury
Asst. Prof. Dept. of CSE

Redundancy Allocation and Optimization of Parallel-Series Systems

Abstract : *The redundancy allocation problem involves simultaneous selection of components and a system level design configuration can collectively meet all design constraints in order to optimize some objective function. Reliability has been considered as one of the important design measures in various industries. The reliability evaluation and optimization of parallel-series system is found to be NP-hard in nature. Therefore, it has attracted many researchers to work in this field. Researchers from the ancient past reveal that there are basically two different approaches that are employed for optimization of parallel-series system. They are namely deterministic approach and heuristic approach. The heuristic approach has ability to obtain the near optimal solution but they do not guarantee for exact solution. Therefore, it is highly desirable to optimize large scale parallel-series system considering its cost and reliability. This work focuses on study of different redundancy allocation strategies to present a new method based on redundancy allocation for optimization of a multi-component parallel-series system.*

Keywords: Convolutional Neural Network, Recurrent Neural Network, HSV, Bhattacharyya Distance, Beam Search

I. INTRODUCTION

Designing of a hardware system involves numerous discrete choices among available component types based on cost, reliability, performance, weight, etc. If the design objective is to minimize cost for a certain reliability requirement then a strategy is required to identify the optimal combination of components and/or design configuration. When there are many functionally similar components to choose from, it becomes difficult to identify the optimal solution, particularly when redundancy can be used to improve reliability [1].

Redundancy is defined as the use of functionally similar (but not necessarily identical) components together in a design such that if one component fails, the redundant part will be available to perform the required function without the system experiencing a failure [2]. In computer science, there are four major forms of redundancy: the hardware redundancy, such as dual modular redundancy and triple modular redundancy; Information redundancy, such as error detection and correction methods; time redundancy, performing the same operation multiple times such as multiple executions of a program or multiple copies of data transmitted; Software redundancy such as N-version programming [3]. Hot standby is a redundant method in which one system runs simultaneously with an identical primary system. Upon failure of the primary system, the hot standby system immediately takes over, replacing the primary system. A cold standby is a redundancy method that involves having one system as a backup for another identical primary system. The cold standby system is called upon only on failure of the primary system [4].

In life data analysis and accelerated life testing data analysis, as well as other testing activities, one of the primary objectives is to obtain a life distribution that describes the times-to-failure of a component, sub-assembly, assembly or system. This analysis is based on the time of successful operation or time-to-failure data of the item (component), either under use conditions or from accelerated life tests. To accomplish this, the relationships between components are considered and decisions about the choice of components can be made to improve or optimize the overall system reliability, maintainability and/or availability [5]. There are many specific reasons for looking at component data to estimate the overall system reliability. One of the most important is that in many situations it is easier and less expensive to test components/subsystems rather than entire systems. Many other benefits of the system reliability analysis approach also exist. Reliability engineering is a sub-discipline of systems engineering that emphasizes dependability in the lifecycle management of a product. Dependability, or reliability, describes the ability of a system or component to function under stated conditions for a specified period of time [6].

Reliability is closely related to availability, which is typically described as the ability of a component or system to function at a specified moment or interval of time. Reliability is theoretically defined as the probability of success (1-probability of failure) as the frequency of failures; or in terms of availability, as a probability derived from reliability, testability and maintainability and maintenance are often defined as a part of “reliability engineering” in Reliability Programs. Reliability plays a key role in the cost-effectiveness of systems. Reliability engineering relates closely to safety engineering and to system safety, in that they use common methods for their analysis and may require input from each other. Reliability engineering focuses on costs of failure caused by system downtime, cost of spares, repair equipment, personnel, and cost of warranty claims. Safety engineering normally focuses more on preserving life and nature than on cost, and therefore deals only with particularly dangerous system-failure modes. High reliability (safety factor) levels also result from good engineering and from attention to detail, and almost never from only reactive failure management (using reliability accounting and statistics).

System optimization is the term of system science and now it is usually defined as the term of computer technology. System optimization requires reducing running processes in computer, changing work mode, deleting unnecessary break off for more efficient computer performance, optimizing file location for faster data write & read, freeing more system resources for computer use and reducing unnecessary system boot processes, etc. This will increase the stability and speed of computers, no harm to hardware. Computer system optimization includes many fields, it can clean Windows Temp Files in temp folder, free disk space, clean registry, reduce the possibility of system errors. It can also speed up computer boot, hold back auto start processes, seed up internet speed and computer shutdown. Or even personalize the theme of Windows. There are basically two different approaches that are employed for optimization of parallel-series system. They are namely Deterministic approach and Heuristic approach.

Deterministic approach is an algorithm which, given a particular input, will always produce the same output, with the underlying machine always passing through the same sequence of states. Deterministic algorithms are by far the most studied and familiar kind of algorithm, as well as one of the most practical, since they can be run on real machines efficiently. Deterministic algorithms can be defined in terms of a state machine: a state describes what a machine is doing at a particular instant in time. State machines pass in a discrete manner from one state to another. Just after we enter the input, the machine is in its initial state or start state. If the machine is deterministic, this means that from this point onwards, its current state determines what its next state will be; its course through the set of states is predetermined.

A heuristic algorithm is one that is designed to solve a problem in a faster and more efficient fashion than traditional methods by sacrificing optimality, accuracy, precision, or completeness for speed. Heuristic algorithms often times used to solve NP-complete problems, a class of decision problems. In these problems, there is no known efficient way to find a solution quickly and accurately although solutions can be verified when given. Heuristics can produce a solution individually or be used to provide a good baseline and are supplemented with optimization algorithms. Heuristic algorithms are most often employed when approximate solutions are sufficient and exact solutions are necessarily computationally expensive.

The heuristic approaches, always may not reach the near optimal solution. Chances of pre-mature convergence, reaching local optimal solution and suitable selection of candidates are the common problems in heuristic approaches. However, the deterministic approaches always results in correct solutions. These limitations motivated us to design an efficient deterministic algorithm for optimization of parallel-series system within some predefined constraints.

Considering a series parallel system composed of sub-systems having actively redundant components in parallel, we have to simultaneously select components and system level configuration which can collectively meet all the design constraints in order to optimize the reliability of the whole system. Given cost, reliability of each component of each sub-system of a system and maximal cost constraint Cmax, we have to determine the components optimally such that the maximum reliability is attained considering the cost constraint. Here, the optimal component reliabilities and redundancy level of components in a system is determined to maximize the system reliability based on cost. Let there be s sub-systems and for each sub-system i,($1 \leq i \leq s$), consider k components for each sub-system. Then, cost of considering the components is given by,

$$C = \sum_{i=1}^{s} \sum_{j=1}^{k} (c_{ij} \times x_{ij} \times d_{ij})$$

The considered system’s reliability is given by,

$$SR = \prod_{i=1}^{s} \left[1 - \prod_{j=1}^{k} (1 - r_{ij} \times d_{ij})^{x_{ij}} \right]$$

A. Objective Function:

Given a parallel series system with known cost and reliability of each components, the objective is to maximize the system reliability (*SR*) under the cost constraint

$$Max\ SR$$
$$s.t\ constraint\ that\ \ C \leq C_{max}\ .$$

B. Notations:

x_{ij}: number of j^{th} component in i^{th} sub- system

c_{ij}: cost of j^{th} component of i^{th} sub- system

r_{ij}: reliability of j^{th} component of i^{th} sub- system

s: number of sub-systems

k: number of different components

r: reliability array for different components

c: cost array for different components

n: total number of components in the system

d: array containing decision parameter of different components

cmin: minimum cost after selecting one component of each type

PS_{mat}: parallel-series system matrix

cmax: maximum cost

R_{mat}: reliability matrix of different components

C_{mat}: cost matrix of different components

R_{max}: maximum reliability returned by selecting different components

D_{mat}: matrix consisting of decision parameter of different components

d_{ij}: decision variable where ,

$$= \begin{cases} 1\text{ , if } j^{th} \text{ component of } i^{th} \text{ sub-system exists} \\ 0\text{, if only one } j^{th} \text{ component of } i^{th} \\ \text{sub-system exists} \end{cases}$$

Component []: array of components

II. PROPOSED 0/1 KNAPSACK ALGORITHM

Reliability()

Input: (PS_{mat},R_{mat}, d,s,k)

Function to calculate and return the system's reliability.

1. If no component is selected, return 0
2. x[k][s], y=1
3. for i=1 to s:
4. for i=1 to k
5. *x[j][i]=d[y++]*
6. rel=1
7. for i=1 to s:
8. p=1
9. for j=1 to k
10. $p=p*(1-R_j*x_{ji})^{PS}{}_{ji}$
11. rel = rel*[1-p]
12. return (rel)

Output: This function returns the calculated reliability using the formula.

Knapsack ()

Input: (cmax, c, r, n)

Function to calculate and return the optimized system reliability and the deciding parameter for the components.

1. *d[n]*
2. *for i= 0 to n*
3. d[i]=0
4. if n==0 or C_{max}==0
5. return d[]
6. if C[n]>C_{max}
7. return (Knapsack (cmax, c[], r[], n-1))
8. else
9. d_1 = Knapsack(cmax,c[], r[], n-1)
10. d_2 = Knapsack(cmax-C[n], c[], r[], n-1)
11. d_2[n] = 1
12. R_1 = Reliability (PS[][], R[], d_1,s,k)
13. R_2= Reliability (PS[][], R[], d_2,s,k)
14. if(R_1> R_2)
15. *return* d_1
16. else
17. *return* d_2

Output: This function returns decision array of the selected components.

Optimization( )

Input: (PS_{mat}, R_{mat}, C_{mat}, s, k)

Algorithm to optimize the system reliability subject to the cost constraint.

1. r=[], c=[]
2. for i=1 to s
3. for j= 1 to k
4. *r.append(R[j])*
5. *c.append(PS[j][i]*C[j])*
6. n=s*k
7. d[] = Knapsack(C_{max}, c[], r[],n)

8. R_{max} = Reliability (PS[][], R[], d,s,k)
9. D[k][s], y=1
10. for i=1 to s:
11. for j=1 to k
12. D[j][i]=d[y++]
13.Return R_{max}

Output: This function returns the maximum reliability obtained after selection of certain components under the given cost constraint.

III. EXPLANATION OF 0/1 KNAPSACK ALGORITHM

The proposed algorithm takes parallel-series system, reliability and cost matrices and maximum cost values as input.

Explanation of optimization function:

This function is used to optimize the system reliability subject to the cost constraint. From step 2-5: For the different components in each sub-system, the reliability matrix is converted to a one-dimensional array. The cost matrix is converted to a one- dimensional array by multiplying the cost of each component to the number of components of each type of a particular sub-system. n is the total number of components in the entire system which is the product of s and k. In step-7: Knapsack function is called by passing the maximum cost, cost array, reliability array, n and the returned value is stored in the decision array. From step 9-11: For the different components in each sub-system, the decision array is converted into matrix. And at last this function returns the maximum reliability.

Explanation of Knapsack function:

This function is used to calculate and return the optimized system reliability and the deciding parameter for the components. From step 2-3: In decision array all the values of components are initialized as 0. From step 4-5: If total length (n) is 0 or the maximum cost is 0 then we return all 0's for the decision parameter as for not considering any elements. From step 6-11: If cost of n^{th} component is greater than maximum cost then we find the optimum system components with the leftover components. Else we find out the reliability1 by calling the knapsack without considering that element and reliability2 by calling the knapsack with consideration of that element i.e. subtracting the cost of the nth item.

From step 14-17: Then, if reliability (R_1) is greater than reliability (R_2) then we return the decision parameter as returned by the knapsack while finding out the reliability1. Else we add the decision parameter of the nth item as 1 with the decision parameter returned by the knapsack while finding out reliability2.

Explanation of Reliability function:

This function is used to calculate and return the system's reliability. In the first step, if none of the components are selected, then the reliability is returned as 0. From step 3-5: For the different components in each sub-system, the decision array is converted into a matrix. From step 7-11: The reliability for the entire system is calculated using the formula stated in the algorithm. In the last step, the reliability is returned.

IV. PROPOSED MODIFIED KNAPSACK ALGORITHM

Although the 0/1 knapsack problem gives us optimum solution for resource allocation, however the maximum cost constraint is not fully utilized. So, the system can be further optimized by utilizing the remaining cost. Therefore, a modified version of the knapsack problem is proposed here.

Structure used:

Component:
- index: stores the index of the components
- n: number of components
- rel: stores the reliability of the components
- cost: stores the cost of the components

Functions used:

sort():
Input: (component[],n)
Function to sort component[] based on reliability per unit cost in descending order.

convert():
Input: (PS_{mat},C_{mat},R_{mat},s,k)
This function converts two dimensional PS matrix to one dimensional components array.

sortindex():
Input: (component,n,d)

This function sorts the components index wise as well as sorts the d array.

Reliability()
Input: (PS_{mat},R_{mat}, d,s,k)

Function to calculate and return the system's reliability.

1. If no component is selected, return 0
2. x[k][s], y=1
3. for i=1 to s:
4. for i=1 to k
5. x[j][i]=d[y++]
6. rel=1
7. for i=1 to s:
8. p=1
9. for j=1 to k
10. if x[j][i]==0
11. var=1
12. else
13. var=PS[j][i]+1
14. p=p*(1-Rj)var
15. rel = rel*[1-p]
16. return (rel)

Output: This function returns the calculated reliability using the formula.

Knapsack ()
Input: (cmax, c, r, n)
Function to calculate and return the optimized system reliability and the deciding parameter for the components.

1. d[n]
2. for i= 0 to n
3. d[i]=0
4. if n==0 or C_{max}==0
5. return d[]
6. if C[n]>C_{max}
7. return (Knapsack (C_{max}, c[], r[], n-1))
8. else
9. d_1 = Knapsack(C_{max},c[], r[], n-1)
10. d_2 = Knapsack(C_{max}-C[n], c[], r[], n-1)
11. d_2[n] = 1
12. R_1 = Reliability (PS[][], R[], d_1,s,k)
13. R_2= Reliability (PS[][], R[], d_2,s,k)
14. if(R_1> R_2)
15. return d_1
16. else
17. return d_2

Output: This function returns decision array of the selected components.

KnapsackFract()
Input: (component[], cmax, n,d[])
This function greedily selects the components and returns the d[] based on fractional selection.

1. for all components[] as i:
2. if (component[i].cost*component[i].n <= cmax
3. cmax= cmax – componet[i].cost*
   C omponent[i].n
4. d[i] = d [i] + component[i].n
5. component[i].n -= component[i].n
6. else:
7. J= (int)cmax/component[i].cost
8. cmax - = component[i].cost*j
9. d[i] = d[i]+j

Output: This function gives decision array of the fractionally selected components.

Algorithm Steps:

1. Input: PS_{mat}, R_{mat}, C_{mat}, s, k
2. for all component as i:
3. cmin=cmin+C[i]
4. cmin=cmin*s
5. Input: cmax
6. cmax=cmax-cmin
7. for allcomponents as i:
7. for allsubsystem as j:
9. PS[i][j]- -
10. for all subsystem as i:
11. for all components as j:
12. l=0
13. r[l]=R[j]
14. c[l]=PS[j][i]*C[j]
15. l++

16. n=s*k
17. d=knapsack(cmax, c, r, n)
18. for all subsystem as i:
19. for all components as j:
20. y=0
21. if(d[y])
22. cost=cost+PS[j][i]*C[j]; 23.cmax=cmax-cost
24. for all components as i:
25. for all subsystem as j:
26. a=i*s+j
27. z=PS[i][j]
28. PS[i][j]=d[a]? 0: PS[i][j]
29. d[a]=d[a]? z: 0
30. d[a]++
31. Component[] = convert(PS, R,C,s,k)
32. .sort(component, k*s)
33. KnapsackFract(component[], cmax,n,d[])
34. sortindex(component[], n, d[])
35. for all subsystem as i:
36. val=1
f37. or all components as j:
38. val=val*(1-R[j])$^{d[s*j+i]}$
39. if(val!=1)
40. rel=rel*(1-val)

V. EXPLANATION OF MODIFIED KNAPSACK ALGORITHM

The proposed algorithm takes parallel-series system, reliability and cost matrices and maximum cost values as input.

Explanation of Reliability function:

This function is used to calculate and return the system's reliability. In the first step, if none of the components are selected, then the reliability is returned as 0. From step 3-5: For the different components in each sub-system, the decision array is converted into a matrix. From step 7-15: The reliability for the entire system is calculated using the formula stated in the algorithm. Here, the lower bound is kept as one and the upper bound is that the entire component will be selected. So, either the entire component will be selected or only one will be selected. In the last step, the reliability is returned.

Explanation of Knapsack function:

This function is used to calculate and return the optimized system reliability and the deciding parameter for the components. From step 2-3: In decision array all the values of components are initialized as 0 From step 4-5: If total length (n) is 0 or the maximum cost is 0 then we return all 0's for the decision parameter as for not considering any elements. From step 6-11: If cost of n^{th} component is greater than maximum cost then we find the optimum system components with the leftover components. Else we find out the reliability1 by calling the knapsack without considering that element and reliability2 by calling the knapsack with consideration of that element i.e. subtracting the cost of the nth item. From step 14-17: Then, if reliability (R_1) is greater than reliability (R_2) then we return the decision parameter as returned by the knapsack while finding out the reliability1. Else we add the decision parameter of the nth item as 1 with the decision parameter returned by the knapsack while finding out reliability2.

Explanation of Fractional Knapsack function:

This function greedily selects the components and returns the matrix d based on fractional selection. From step 1-9: For all components, if the product of the cost of a component and the number of components is less than or equal to the maximum cost left then the component is selected as a whole and the cost is subtracted from the maximum cost. If not then a fraction of the component is selected.

Explanation of algorithm steps:

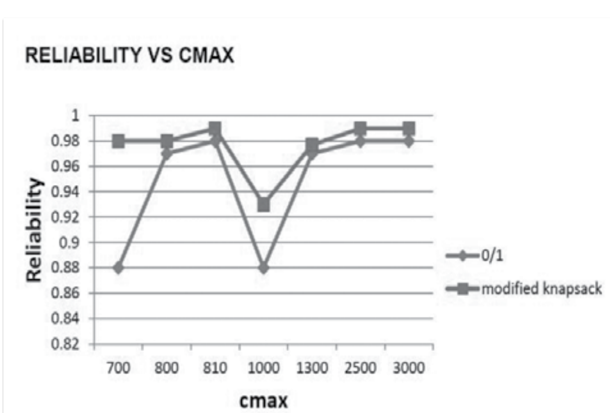
The input is taken of the number of subsystems, number of components, PS matrix, R matrix and C matrix. Step 2-4: The lower limit for the selection of components is taken as 1. Minimum 1 component of each type in every subsystem has to be selected. In this step the minimum cost is calculated by electing one component of each type. Step 5-6: Input for cmax is taken and the new value for cmax is calculated by subtracting cmin from cmax. From step 7-9: The value of the components in each subsystem is decreased by 1 as already 1 component from each has been selected. From step 10-15: The reliability matrix and the cost matrix are converted to a one dimensional array. In step 17, the knapsack function is called which returns the decision array as to which components are selected. From step 18-23: The cost left after applying 0/1 knapsack is calculated. From step 24-30: The PS matrix is updated which will represent the components which are left to be selected using fractional knapsack. From steps 31-34: The convert, sort, KnapsackFract and sortindex functions are called respectively. From steps 35-40: The reliability is calculated according to the formula used in the algorithm which includes the components selected after applying 0/1 knapsack and then applying fractional knapsack to the left over components.

Table I: Comparison of the results of 0/1 knapsack and modified knapsack

SI. No.	PS_{mat}	C_{mat}	R_{mat}	Original Cost	Original Reliability	C_{mat}	0/1 knapack		Modified 0/1 knapak	
							Optimized cost	Optimized reliability	Optimized cost	Optimized reliability
1	4 2	3D	0.42							
	2 3	20	0.29	440	0.934	400	330	0.89	330	098
	3 1	40	051							
2	S 2	50	ass							
	1 4	30	032	360	0.99	810	660	0.98	810	099
	4 2	60	0.61							
3	6 4	50	0.50							
	2 1	40	0.40	775	0.98	1000	485	0.88	685	0.98
	3 2	31	0.32							
4	5 2	72	0.71							
	8 4	35	0.36	1194	0.99	800	744	0.98	789	0.98
	3 3	45	0.47							
5	4 2 2	61	0.60							
	2 3 5	39	0.31	1163	0.96	1000	824	0.38	946	0.93
	1 1 3	57	0.54							
6	5 3 2	25	0.15							
	4 6 3	71	0.65	1506	0.98	1300	1295	0.97	1295	0.977
	2 4 3	37	0.25							
7	2 5 1 1	ao	0.41							
	4 3 7 2	60	0.61	2770	0.99	2500	2430	0.99	2500	099
	3 6 1 1	20	0.20							
	1 2 6 7	70	0.80							
8	2 4 1 5 3	15	0.10							
	5 1 4 3 2	43	0.39							
	6 2 5 3 1	67	0.65	366S	0.98	3000	2967	09S	2932	0.99
	2 1 3 5 7	27	0.23							
	5 1 4 6 7	51	0.50							

VI. RESULTS AND DISCUSSIONS

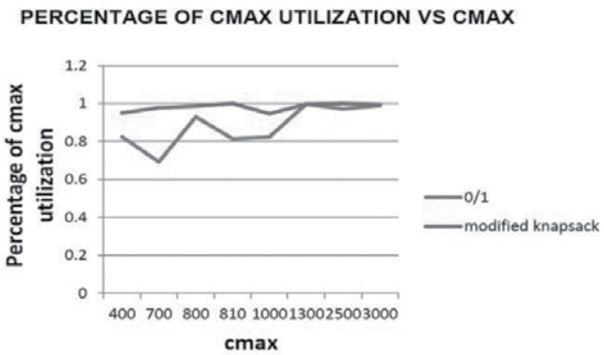
Different configuration of parallel-series system with different sub-systems and components are taken into consideration for applying the proposed algorithm. On a parallel - series system by taking s sub-systems and k types of components by taking different values of the maximum cost. The proposed algorithm is developed using C language and executed in Intel i5 processor with 4gb RAM. And the computed results are recorded in Table I. Some sample parallel-series systems are taken and the optimized cost and optimized reliability of 0/1 knapsack and modified knapsack are compared. From the results recorded in the table it is seen that the modified knapsack approach utilizes more cost and gives better reliability than 0/1 knapsack.



Cmax	0/1 Knapsack	Modified Knapsack
400	0.89	0 98
700	0.88	0.98
800	0.97	0.98
810	0.98	0 99
1000	0.88	0.93
1300	0.97	0.977
2500	0.98	0.99
3000	0.98	0.99

Fig.1: *Comparison between reliability of 0/1 and modified knapsack*

A graph has been plotted between reliability and cmax as presented in Fig.1. From the graph it can be observed that the reliability in modified knapsack is better as compared to 0/1 knapsack under the same cost constraint. A graph is also plotted between percentage of cmax utilization and cmax and presented in Fig. 2. From the graph it can be observed that the percentage of utilization in modified knapsack is better as compared to 0/1 knapsack.



Cmax	0/1 Knapsack	Modified Knapsack
400	0.825	0.95
700	0.692	0.978
800	0.93	0.986
810	0.814	1
1000	0.824	0.946
1300	0.996	0.996
2500	0.972	1
3000	0.989	0.994

Fig.-2: *Comparison between percentage of cmax utilization of 0/1 and modified knapsack*

VII. CONCLUSIONS

The present work addresses an optimization problem of parallel-series system to optimize its system reliability under the permissible cost constraint. The proposed approach is efficient enough to find the optimized combination of components which can be selected to give the maximum reliability within the given cost constraint. A dynamic programming based approach is proposed here for optimization of parallel-series system. The problem is mapped into a modified improved knapsack problem and then the operations are carried out to get the optimum result. The result so obtained from the proposed modified Knapsack problem yields better result than its counterparts. As the dynamic programming approach is deterministic in nature therefore, it always produces the correct result for the problem. The proposed approach is simulated on some sample benchmark parallel-series systems which prove the correctness and efficiency of the algorithm. In short, the proposed approach uses encoding of the inputted parallel series system optimizes the system considering its system reliability under the permissible cost constraint.

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Sanjana Sabat, Namrata Agrawal
Swagatika Paricha
Dept. of CSE

Voice-controlled Smart Home Appliance Automation and Security Systems

Abstract: *With the advancement of technology in the field of automation there has been a lot of changes observed in the living style. This article aims to provide automatic control of the home appliances through voice recognition system. Moreover, this article also includes security system which detects unauthorized motion, vibration and intrusion and alerts using a buzzer with status display on the LCD screen..*

I. INTRODUCTION

Home, it is the place where one fancies or desires to be after a long tiring day. People come home exhausted after a long hard working day. Some are too tired and they find it hard to move once they land on their couch, sofa or bed. So any small device/technology that would help them switch theirs lights on or off, or play their favorite music etc. on a go with their voice with the aid of their smart phones would make their home more comfortable.

Moreover, it would be better if everything such as warming bath water and adjusting the room temperature were already done before they reach their home just by giving a voice command. So, when people would arrive home, they would find the room temperature, the bath water adjusted to their suitable preferences, and they could relax right away and feel cozier and rather, feel homelier.

Human assistants like housekeepers were a way for millionaires to keep up their homes in the past. Even now when technology is handy enough only the well to do people of the society are blessed with theses new smart home devices, as these devices costs are a bit high. However, not everyone is wealthy enough to be able to afford a human assistant, or some smart home kit. Hence, the need for finding an inexpensive and smart assistant for families keeps is rising.

This article proposes such inexpensive system. It uses the Google Assistant, the IFTTT [1] application, the Blynk [2] application and the NodeMCU [3] microcontroller as the major components along with a relay board comprising of 4/8 relays along with ULN 2803 IC. Natural language voice is used to give commands to the Google Assistant [4]. All of the components are connected over the internet using Wi-Fi which puts this system under the IoT.

The past decade has seen significant advancement in the field of consumer electronics. Various 'intelligent' appliances such as cellular phones, air-conditioners, home security devices, home theatres, etc. are set to realize the concept of a smart home. They have given rise to a Personal Area Network in home environment, where all these appliances can be interconnected and monitored using a single controller. [5]

Busy families and individuals with physical limitation represent an attractive market for home automation and networking. A wireless home network that does not incur additional costs of wiring would be desirable. Bluetooth technology, which has emerged in late 1990s, is an ideal solution for this purpose. [6]

Home automation involves introducing a degree of computerized or automatic control to certain electrical and electronic systems in a building. These include lighting, temperature control etc.

This work demonstrates a simple home automation system which contains a remote mobile host controller and several client modules (home appliances). The client modules communicate with the host controller through a wireless device such as a Bluetooth enabled mobile phone, in this case, an android based Smart phone.

II. METHODOLOGY

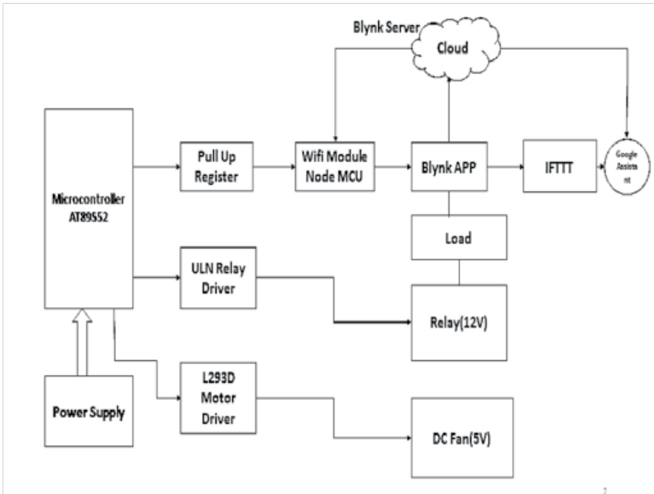

Fig1: Voice Controlled Home Automation

The working of the project starts with the microcontroller AT89S52 as shown in figure - 1. The loads are connected to the microcontroller through relay and the relay is connected with the microcontroller through relay driver. To control the speed of the fan, the fan is connected to the microcontroller through motor driver which gives proper voltage required by the fan motor. All these functions are to be controlled through voice commands for which Wifi interface has been used. We will feed our voice commands into the Wifi module using android smartphone.

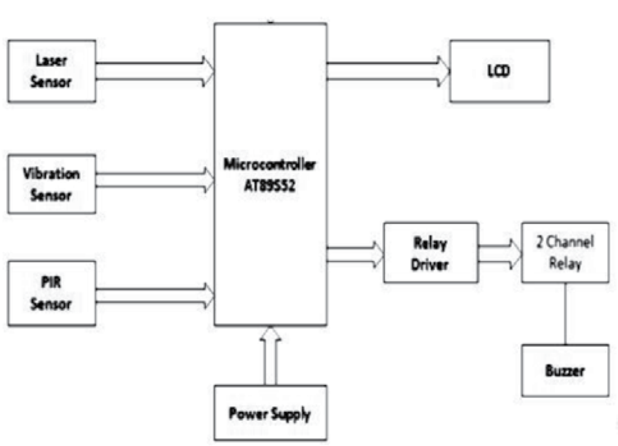

Fig2: Block Diagram of Security Systems

In Security systems as shown in figure - 2, three different sensors are used to sense the movement of living body, objects and vibration sensing. PIR sensor is used to detect motion of live bodies, LDR can recognize the objects entering through windows and vibration sensor sense the vibration of the wall. All the unauthorized activities detected will activate the alarm and status will be displayed on the LCD screen.

RESULTS:

The prototype of home automation is shown in figure – 3 & 4.

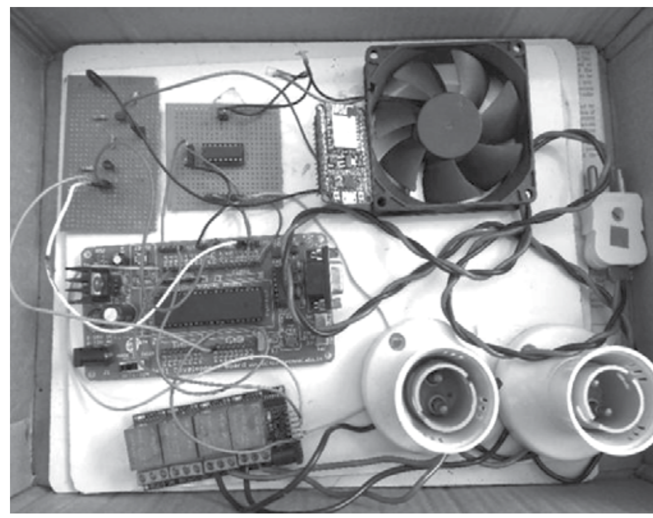

Fig. 3: Home Automation

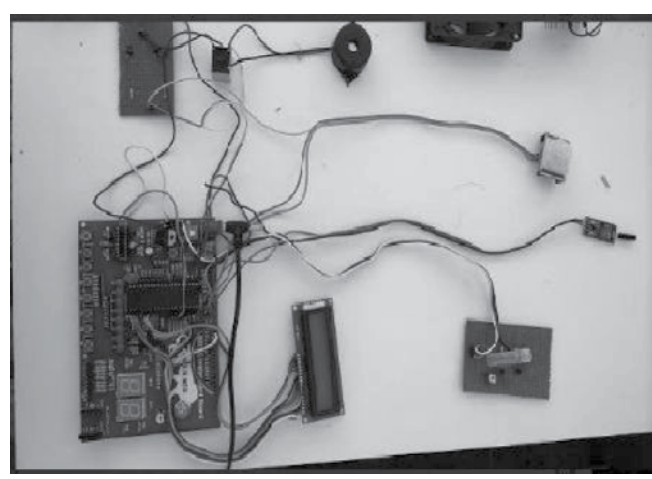

Fig.4: Home Security System

i. Appliances turned ON/OFF based on voice commands.

ii. Speed of the fan is controlled using voice commands.

iii. PIR sensor detects motion of living objects.

iv. LDR will detect objects entering from windows.

v. Vibration sensor detects breakage from wall.

vi. Buzzer activates on unauthorized entry.

vii. Status is displayed on the LCD.

APPLICATION

i. Differently abled person will be able to control the basic appliances by their voice commands.

ii. Senior Citizens will be benefited.

iii. Theft alarm enhances security of the home.

iv. Security System can also be used in industries

CONCLUSION

This article discussed the development of the voice recognition home automation system which can be used to replace the old and conventional way to switch on the power of an electrical device. This system consists of a voice recognition circuit, a microcontroller circuit and a set of relay. The voice recognition home automation system has been successfully developed and through this project we have gained much experience especially in the field of applying the technique of troubleshooting an electrical circuit and also in programming the microcontroller. This project is a very simple project compare to any of those who are already in the industry and commercialized but yet we hope that this project can be research on further to create a better design that can be applied to a larger scale of controlling. Besides that, we also hope that this project can be jumping stone for the application as one of the smart home necessity. Besides the achieving of the main objective, by using this system, it can help reduce any occurrence of getting shock due to the failure of the switch and it offer a more safety way to turn on the switch. Moreover, if this system is fully equipped in a house it can reduce the addition of the wall switch and what left is only the plug point for user to plug in their devices only.

In this article it has been observed that the prototype model works without any basic error. So it can be implemented in practical field. Beside the cost of the project is not too much. Here it has provided closured security so it is quite impossible for any burglar to enter the room. It could be implemented by GSM based home security system. For this when a burglar enters the room without the concern of owner a sms will be sent to the user. Then he will take precautionary measure. It may be used another technique called biometrics which is more prominent and a recognized means of positive identification. Some new technologies such as fingerprint scanning, retinal scanning and iris scanning, and voiceprint identification also can be incorporated.

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Chaitanya Surya Gupta
K Ravi Shankar
Saswati Panda
Subham Sahoo
Dept. of ECE

Kailasavadivoo Sivan

ISRO's ROCKET MAN

A Successful India is the dream of 133.92 crore Indians. Dr. Kailasavadivoo Sivan is the name who has brought success in the scientific technology. The 62-year-old ISRO chief hails from a farmer's family in Sarakkalvilai in Kanniyakumari district is devoted to his work and his country.

K Sivan's journey, from working barefoot on his father's farm to heading one of the biggest Space agencies in the world, has been inspiring. Sivan is the first graduate in his family. His brother and two sisters were unable to complete higher education due to poverty.

His early education was at the government primary school in Sarakkalvilai. He later studied at the Vallankumaravilai government school and graduated from the ST Hindu College in Nagercoil. The primary school bears his name on the entrance. Not just the people of the village but the whole country is proud of him for the achievement in Chandrayaan-2. In addition to this, he enrolled for his Engineering Course at the Madras Institute of technology. Then he successfully joined the Indian Institute of Science (IISC), Bengaluru for his Master's Degree in Engineering. He then obtained his Doctoral degree in the Aerospace Engineering in IIT, Bombay.

In his career, ISRO's 'Rocket Man' had contributed immensely to the development of cryogenic engine, PSLV, GSLV and reusable launch vehicle (RLV) programmes. He had also played a key role in the launch of 104 satellites in one go on February 15, 2017, which is a world record.

Through his great perseverance and commitment to this Chandrayaan mission, he has made the whole world to look up to our nation. He is capable of shaking the world in his own gentle way. His dedication level was so high for country that he rejected the job offer from NASA.

ISRO will attain more heights in future because of the hard work and dedication of scientists like K. Sivan.

Tanmaya Bal
Dept. of ECE

Design of a DC-DC Converter Using Sepic Topology

Abstract: *In this project modeling and simulation of an integrated SEPIC converter with a multiplier cell for standalone photovoltaic application is designed. An integrated SEPIC converter with a multiplier cell is proposed for the use as a high gain step-up converter and its semiconductor switch operates in continuous conduction mode (CCM). This work describes the operational principles of the proposed converter and it's performance in photovoltaic application. The system performance was verified through simulation using MATLAB software. The simulation results showed that the proposed converter has performed successfully as a high gain boost converter for standalone PV application. The single-ended primary-inductance converter (SEPIC) is a DC/DC-converter.*

I. INTRODUCTION

Topology that provides a positive regulated output voltage from an input voltage that varies from above to below the output voltage. This type of conversion is handy when the designer uses voltages(e.g.,12 V) from an unregulated input power supply such as a low-cost wall wart. Recently, several inductor manufacturers began selling off-the-shelf coupled inductorsin a singlepackage at a cost only slightly higher than that of the comparable single inductor. The coupled inductor not only provides a smaller footprint but also, to get the same inductor ripple current, requires only half the inductance required for a SEPIC with two separate inductor.

II. SOLAR PV CELL

Photovoltaic (PV)cell is an energy harvesting technology that converts solar energy into useful electricity through a process called the photovoltaic effect. There are several different types of PV cells which all use semiconductors to interact with incoming photons from the Sun in order to generate an electric current.

III. TYPES OF CONVERTER

A. Buck Converter

A buck converter as shown in Fig. 1, steps a voltage down, producing a voltage lower than the input voltage. A buck converter could be used to charge a lithium ion battery to 4.2 V, from a 5 V source.

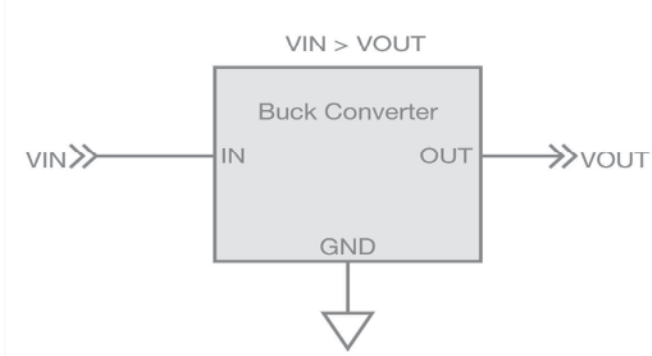

Fig.1:*Buck converter*

B. Boost Converter

Aboost converter shown in Fig. 2, steps a voltage up, producing a voltage higher than the input voltage. A boost converter could be used to drive a string of LEDs from a lithium cell, or provide a 5 V USB output from a lithium cell.

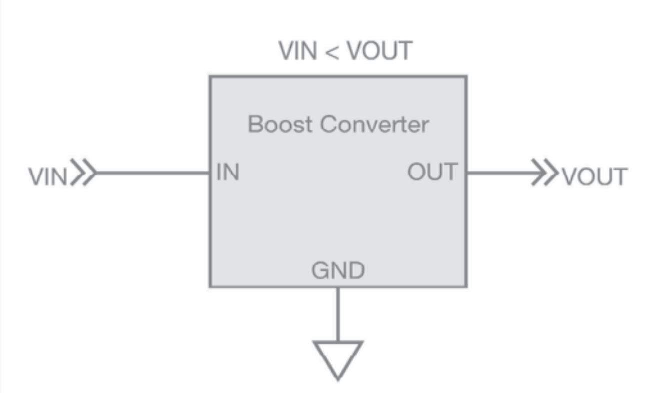

Fig. 2: *Boost converter*

C. Buck-Boost Converter

A buck-boost converter as shown in fig. 3, steps a voltage up or down, producing a voltage equal to or higher or lower than the input voltage. A buck boost could be used to provide a 12 V output from a 12 V battery. A 12V battery's voltage can vary between 10 V and 14.7 V.A buck boost could also power an LED from a single cell. An LED forward drop is ashigh as 3 V. Alithium battery cell can vary between 2.5and 4.2V. There are buck-boosts that produce positive and negative voltages.

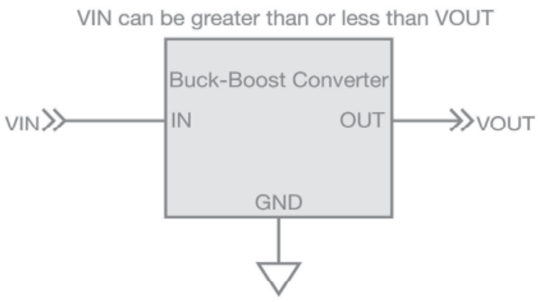

Fig. 3: *BuckBoost converter*

D. Sepic Converter

A SEPIC converter as shown in fig.4, also steps a voltage up or down, producing a voltage equal to or higher or lower than the input voltage. A SEPIC is used for similar applications as the buck-boost, but provides some advantages in some applications.

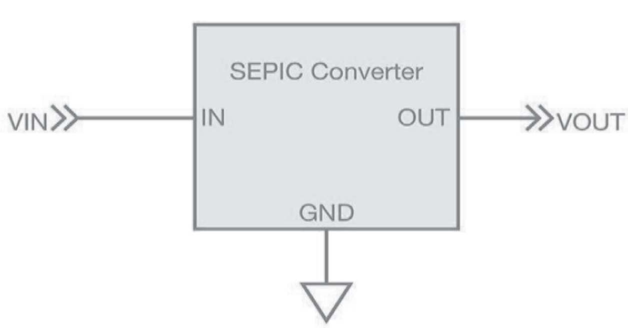

Fig. 4: *SEPIC converter*

V. SIMULINK MODEL AND RESULT

The Simulink model is shown in fig. 9 and the results are shown in fig. 9, 10, 11.

IV. DC –DC CONVERTER USING SEPIC TOPOLOGY

The single-ended primary-inductor converter (SEPIC) is a type of DC/DC converter that allows theelectrical potential (voltage) at its output to be greater than, less than, or equal to that at its input. The output of the SEPIC is controlled by the duty cycle of the control transistor.A SEPIC is essentially a boost converter followed by a buck-boost converter,therefore it is similar toa traditional buck-boost converter, but hasadvantages of having non-inverted output (the output has the same voltage polarity as the input), using a series capacitor to couple energy from the input to the output (and thus can respond more gracefully to a short-circuit output), and being capable of true shutdown: when the switch is turned off, its output drops to 0 V, following a fairly hefty transient dump of charge. SEPICs are useful in applications in which a battery voltage can be above and below that of the regulator's intended output. For example, a single lithium ion battery typically discharges from 4.2 volts to 3 volts; if other components require 3.3 volts, then the SEPIC would be effective.Fig. 6 shows a simple circuit diagram of a SEPIC converter, consisting of an input capacitor, CIN; an output capacitor, COUT; coupled inductors L1a and L1b; an AC coupling capacitor, CP; a power FET, Q1; and a diode, D1.

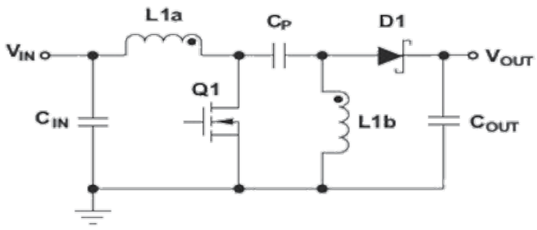

Fig.5: *Circuit diagram of SEPIC converter*

SEPIC operating in continuous conduction mode (CCM). Q1 is on in the top circuit and fig. 7 shows off in the bottom circuit. To understand the voltages at the various circuit nodes, it is important to analyze the circuit at DC when Q1 is off and notswitching. When switchS1 is turned on, current IL1 increases and the current IL2 goes more negative. (Mathematically, it decreases due to arrow direction.) The energy to increase the current IL1 comes from the input source. Since S1 is a short while closed, and the instantaneous voltage VL1 is approximately VIN, the voltage VL2 is approximately −VC1. Therefore, the capacitor C1 supplies the energy to increase the magnitude of the current in IL2 and thus increase the energy stored in L2. The easiest way to visualize this is to consider the bias voltages of the circuit in a dc state, then close S1.

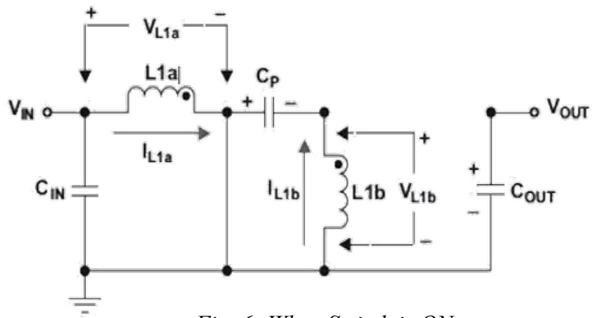

Fig. 6: When Switch is ON

When switch S1 is turned off, shown in fig. 8, the current IC1 becomes the same as the current IL1, since inductors do not allow instantaneous changes in current. The current IL2 will continue in the negative direction, in fact it never reverses direction. It can be seen from the diagram that a negative IL2 will add to the current IL1 to increase the current delivered to the load. Using Kirchhoff's Current Law, it can be shown that ID1 = IC1 - IL2. It can then be concluded, that while S1is off, power is delivered to the load from both L2 and L1. C1, however is being charged byL1 during this off cycle, and will in turn recharge L2 during the on cycle.

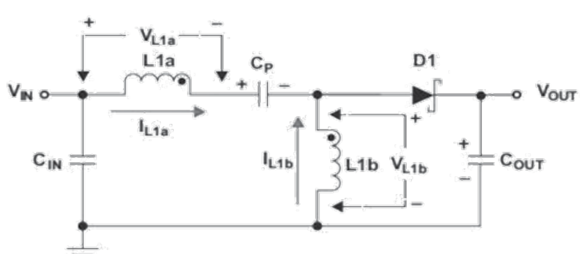

Fig.7: *When Switch is OFF*

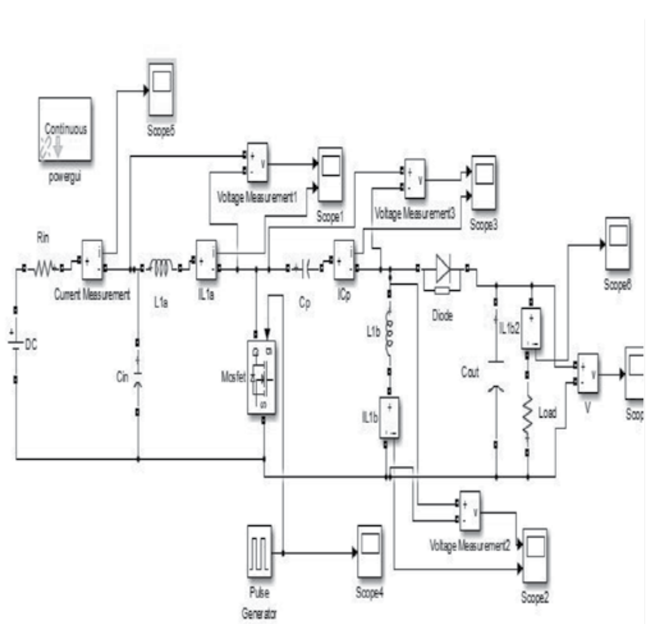

Fig. 8: *Simulink model*

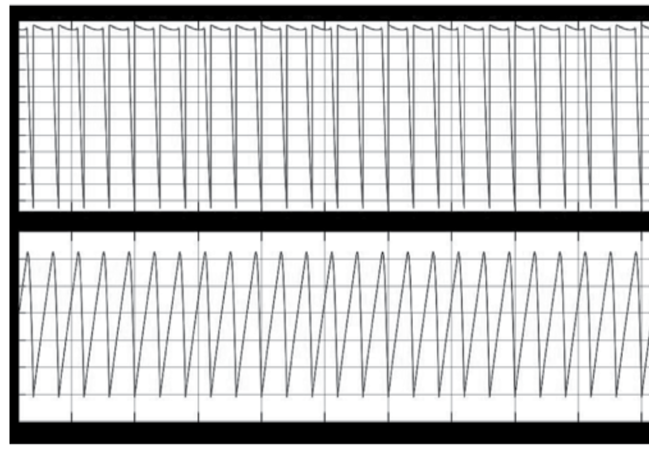

Fig. 9: *Voltage and Current Waveform of Inductor L_a*

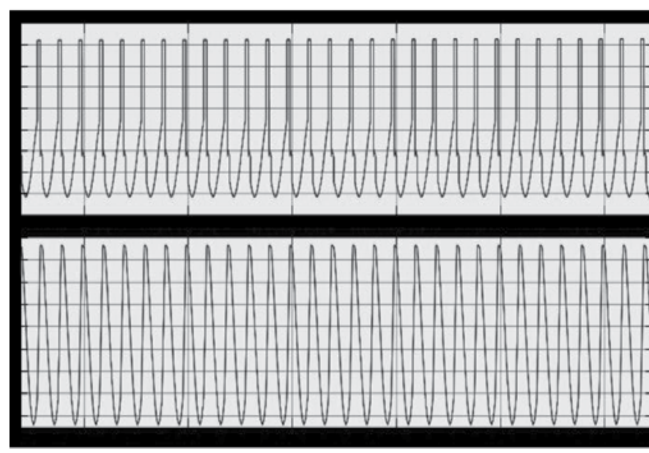

Fig. 10: *Voltage and current waveform of inductor l_b*

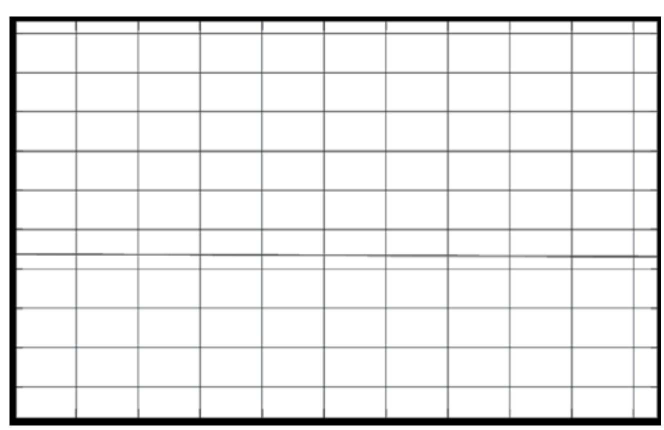

Fig. 11 : *Output voltage waveform*

VI. HARDWARE IMPLEMENTATION OF SEPIC CONVERTER

The hardware setup is shown in Fig. 12 & 13.

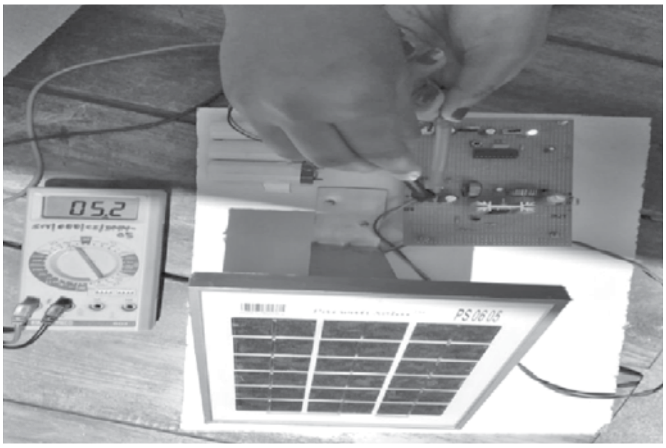

Fig. 12 : *Input of Hardware*

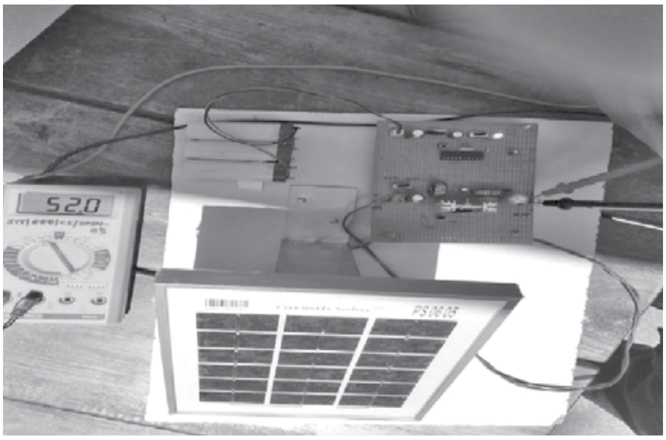

Fig. 13 : *output of Hardware*

VI.CONCLUSIONS

In this work, a SEPIC converter with a multiplier cell was designed for the photovoltaic application. The switching of the proposed converter is easy to control, because of the semiconductor switch is controlled with a complementary switching pattern in continuous conduction mode (CCM).The SEPIC converter is simulated using MATLAB and was also implemented in hardware and a high voltage gain was also obtained hence this converter can be coupled with intermittent source like PV modules.In future this work can be extended to a closed loop system using MPPT controllers or other controllers which can be used as standalone systems or grid connected systems.

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Kumari Sonali, Debanshu Meher
Poonam Agarwal, Md. Aquib Manza R
Amares kumar Samal, Gaurav Kumar
Dept. of EEE

Speaker Diarization

Abstract: *Speaker indexing or diarization is an important task in audio processing and retrieval. Speaker diarization is the process of labeling a speech signal with labels corresponding to the identity of speakers. This paper includes a comprehensive review on the evolution of the technology and different approaches in speaker indexing and tries to offer a fully detailed discussion on these approaches and their contributions. This paper reviews the most common features for speaker diarization in addition to the most important approaches for speech activity detection (SAD) in diarization frameworks. Two main tasks of speaker indexing are speaker segmentation and speaker clustering. This paper includes a separate review on the approaches proposed for these subtasks. However, speaker diarization systems which combine the two tasks in a unified framework are also introduced in this paper. Another discussion concerns the approaches for online speaker indexing which has fundamental differences with traditional offline approaches. Other parts of this paper include an introduction on the most common performance measures and evaluation datasets. To conclude this paper, a complete framework for speaker indexing is proposed, which is aimed to be domain independent and parameter free and applicable for both online and offline applications*

Keywords: Speaker diarization, speaker segmentation and clustering

I. INTRODUCTION

Speaker diarization has emerged as an increasingly important and dedicated domain of speech research. Whereas speaker and speech recognition involve, respectively, the recognition of a person's identity or the transcription of their speech, speaker diarization relates to the problem of determining 'who spoke when?' Speaker diarization is a combination of speaker segmentation and speaker clustering. The first aims at finding speaker change points in an audio stream. The second aims at grouping together speech segments on the basis of speaker characteristics. With the increasing number of broadcasts, meeting recordings and voice mail collected every year, speaker diarization has received much attention by the speech community, as is manifested by the specific evaluations devoted to it under the auspices of the National Institute of Standards and Technology for telephone speech, broadcast news and meetings.

In speaker diarization one of the most popular methods is to use a Gaussian mixture model to model each of the speakers, and assign the corresponding frames for each speaker with the help of a Hidden Markov Model. There are two main kinds of clustering scenario. The first one is by far the most popular and is called Bottom-Up. The algorithm starts in splitting the full audio content in a succession of clusters and progressively tries to merge where each cluster corresponds to a real speaker. The second clustering strategy is called top-down and starts with one single cluster for all the audio data and tries to split it iteratively until reaching a number of clusters equal to the number of speakers. The redundant clusters in order to reach a situation

II. SPEAKER DIARIZATION FRAMEWORK

A. Speech detection

The aim of this step is to find the regions of speech in the audio stream. Depending on the domain data being used, nonspeech regions to be discarded can consist of many acoustic phenomena such as silence, music, room noise, background noise, or cross-talk.The general approach used is maximum-likelihood classification with Gaussian mixture models (GMMs) trained on labelled training data, although different class models can be used, such as multistate HMMs. The simplest system uses just speech/nonspeech models such as in [11], while [12] is similar but four speech models are used for the possible gender/bandwidth combinations. Noise and music are explicitly modelled in [13]–[15] which have classes for speech, music, noise, speech music, and speech noise, while use wideband speech, narrowband speech, music and speech music. The extra speech xx models are used to help minimize the false rejection of speech occurring in the presence of music or noise, and this data is subsequently reclassified as speech. The classes can also be broken down further, as in [15], which has eight models in total, five for nonspeech (music, laughter, breath, lip-smack, and silence) and three for speech (vowels and nasals, Fricatives, and obstruent). When operating on unsegmented audio, Viterbi segmentation, (single pass or iterative with optional adaptation) using the models is employed to identify speech regions. If an initial segmentation is already available (for example, the ordering of the key components may allow change point detection before nonspeech removal), each segment is individually classified. Minimum length constraints [11],

[16] and heuristic smoothing rules [12], [15] may also be applied. An alternative approach which does not use Viterbi decoding, but instead a best model search with morphological rules is described in [19]. Silence can be removed in this early stage, using a phone recognizer (as in [11]) or energy constraint, or in a final stage processing using a word recognizer (as in [14]) or energy constraint (as in the MIT system for RT-03. Regions which contain commercials and thus are of no interest for the final output can also be automatically detected and removed at this early stage [4]. For broadcast news audio, speech detection performance is typically less than 1% miss (speech in reference but not in the hypothesis) and 1%–2% false alarm (speech in the hypothesis but not in the reference), whereas for meeting audio, the figures are typically around 1% higher for both. When the speech detection phase is run early in a system, or the output is required for further processing such as for transcription, it is more important to minimize speech miss than false alarm rates, since the former are unrecoverable errors in most systems. However, the DER, used to evaluate speaker diarization performance, treats both forms of error equally. For telephone audio, typically some form of standard energy/ spectrum-based speech activity detection is used since non speech tends to be silence or noise sources, although the GMM approach has also been successful in this domain with single-channel or cross-channel classes. For meeting audio, the non-speech can be from a variety of noise sources, like paper shuffling, coughing, laughing, etc. and energy-based methods do not currently work well for distant microphones, so using a simple pretrained speech/non speech GMM is generally preferred. An interesting alternative uses a GMM, built on the normalized energy coefficients of the test data, to determine how much no speech to reject while preliminary work in [6] shows potential for the future for a new energy-based method. When supported, multiple channel meeting audio can be used to help speech activity detection. This problem is felt to be so important in the meetings domain that a separate evaluation for speech activity detection was introduced in the spring 2005 Rich Transcription meeting evolution.

B. Change detection

The aim of this step is to find points in the audio stream likely to be change points between audio sources. If the input to this stage is the unsegmented audio stream, then the change detection looks for both speaker and speech/no speech change points. If a speech detector or gender/bandwidth classifier has been run first, then the change detector looks for speaker change points within each speech segment. Two main approaches have been used for change detection. They both involve looking at adjacent windows of data and calculating a distance metric between the two, then deciding whether the windows originate from the same or a different source. The differences between them lie in the choice of distance metric and thresholding decisions. The first general approach used for change detection, used in [15], is a variation on the Bayesian information criterion (BIC) technique introduced. This technique searches for change points within a window using a penalized likelihood ratio test of whether the data in the window is better modelled by a single distribution (no change point) or two different distributions (change point). If a change is found, the window is reset to the change point and the search restarted. If no change point is found, the window is increased and the search is redone. Some of the issues in applying the BIC change detector are as follows. 1) It has high miss rates on detecting short turns (2–5 s), so an be problematic to use on fast interchange speech like conversations. 2) The full search implementation is computationally expensive (order), so most systems employ some form of computation reductions. A second technique used first in [3] and later in [13], fixed-length windows and represents each window by a Gaussian and the distance between them by the Gaussian

Divergence (symmetric KL-2 distance). The step-by-step implementation in [11] and system for telephone audio in [3] are similar but use the generalized log likelihood ratio as the distance metric. The peaks in the distance function are then found and define the change points if their absolute value exceeds a predetermined threshold chosen on development data. Smoothing the distance distribution or eliminating the smaller of neighbouring peaks within a certain minimum duration prevents the system overgenerating change points at true boundaries. Single Gaussians are generally preferred to GMMs due to the simplified distance calculations. Typical window sizes are 1–2 or 2–5 s when using a diagonal or full covariance Gaussian, respectively. As with BIC, the window length constrains the detection of short turns. Since the change point detection often only provides an initial base segmentation for diarization systems, which will be clustered and often regimented later, being able to run the change point detection very fast (typically less than 0.01 for a diagonal covariance system) is often more important than any performance degradation. In fact, [11] and [12] found no significant performance degradation when using a simple initial uniform segmentation within their systems. Both change detection techniques require a detection threshold to be empirically tuned for changes in audio type and features. Tuning the change detector is a trade-off between the desires to have long, pure segments to aid in initializing the clustering stage, and minimizing missed change points which produce contaminations in the clustering. Alternatively, or in addition, a word or phone decoding step with heuristic rules may be used to help find putative speaker change points such as in and the Cambridge 1998–2003 systems. However, this approach can over-segment the speech data and requires some additional merging or clustering to form viable speech segments, and can miss boundaries in fast speaker interchanges if relying in the presence of silence or gender changes between speakers.

1) Bottom-Up Approach: The bottom-up approach is by far the most common in the literature. Also known as agglomerative hierarchical clustering (AHC or AGHC), the bottom-up approach trains a number of clusters or models and aims at successively merging and reducing the number of clusters until only one remains for each speaker. Various initializations have been studied and, whereas some have investigated -means clustering, many systems use a uniform initialization, where the audio stream is divided into a number of equal length abutted segments. This simpler approach generally leads to equivalent performance [8]. In all cases the audio stream is initially oversegmented into a number of segments which exceeds the anticipated maximum number of speakers. The bottom-up approach then iteratively selects closely matching clusters to merge, hence reducing the number of clusters by one upon each iteration. Clusters are generally modelled with GMM and, upon merging, a single new GMM is trained on the data that was previously assigned to the two individual clusters. Standard distance metrics, such as those described, are used to identify the closest clusters. A reassignment of frames to clusters is usually performed after each cluster merging, via realignment for example, and the whole process is repeated iteratively, until some stopping criterion is reached, upon which there should remain only one cluster for each detected speaker.

2) Top-Down Approach: In contrast with the previous approach, the top-down approach first models the entire audio stream with a single speaker model and successively adds new models to it until the full number of speakers are deemed to be accounted for. A single GMM model is trained on all the speech segments available, all of which are marked as unlabelled. Using some selection procedure to identify suitable training data from the non-labelled segments, new speaker models are iteratively added to the model one-by-one, with interleaved realignment and adaptation. Segments attributed to any one of these new models are marked as labelled. Stopping criteria similar to those employed in bottom-up systems may be used to terminate the process or it can continue until no more relevant unlabelled segments with which to train new speaker models remain. Topdown approaches are far less popular than their bottom-up counterparts. Some examples include [16]. While they are generally out-performed by the best bottom-up systems, top-down approaches have performed consistently and respectably well against the broader field of other bottom-up entries. Top-down approaches are also extremely computationally efficient and can be improved through cluster purification [14].

C. Segmentation

In the literature, the term 'speaker segmentation' is sometimes used to refer to both segmentation and clustering. Whilst some systems treat each task separately many of present state-of-the-art systems tackle them simultaneously. In these cases the notion of strictly independent segmentation and clustering modules is less relevant. However, both modules are fundamental to the task of speaker diarization and some systems, such as that reported in [6], apply distinctly independent segmentation and clustering stages. Thus the segmentation and clustering models are described separately here. Speaker segmentation is core to the diarization process and aims at splitting the audio stream into speaker homogeneous segments or, alternatively, to detect changes in speakers, also known as speaker turns. The classical approach to segmentation performs a hypothesis testing using the acoustic segments in two sliding and possibly overlapping, consecutive windows. For each considered change point there are two possible hypotheses: first that both segments come from the same speaker (H0), and thus that they can be well represented by a single model; and second that there are two different speakers (H1), and thus that two different models are more appropriate. In practice, models are estimated from each of the speech windows and some criteria are used to determine whether they are best accounted for by two separate models (and hence two separate speakers), or by a single model (and hence the same speaker) by using an empirically determined or dynamically adapted threshold [10]. This is performed across the whole audio stream and a sequence of speaker turns is extracted. Many different distance metrics have appeared in the literature. Next we review the dominant approaches which have been used for the NIST RT speaker diarization evaluations during the last 4 years. The most common approach is that of the Bayesian Information Criterion (BIC) and its associated _BIC metric which has proved to be extremely popular. The approach requires the setting of an explicit penalty term which controls the trade-off between missed turns and those falsely detected. It is generally difficult to estimate the penalty term such that it gives stable performance across different meetings and thus new, more robust approaches have been devised. They either adapt the penalty term automatically, i.e. the modified BIC criterion or avoid the use of a penalty term altogether by controlling model complexity. BIC-based approaches are computationally demanding and some systems have been developed in order to use the BIC only in a second pass, while a statistical-based distance is used in a first pass. Another BIC-variant metric, referred to as cross-BIC and introduced in involves the computation of cross-likelihood: the likelihood of a first segment according to a model tuned from the second segment and vice versa. In [69], different techniques for likelihood normalization are presented and are referred to as bilateral scoring. A popular and alternative approach to BIC-based measures is the Generalized Likelihood Ratio (GLR), .In contrast to the BIC, the GLR is a likelihood-based metric. And corresponds to the ratio between the two fore mentioned hypotheses, as described in. To adapt the criterion in order to take into account the amount of training data available in the two segments, a penalized GLR was proposed in [14].

D. Clustering

Whereas the segmentation step operates on adjacent windows in order to determine whether or not they correspond to the same speaker, clustering aims at identifying and grouping together same-speaker segments which can be localized anywhere in the audio stream. Ideally, there will be one cluster for each speaker. However, with such an approach to diarization, there is no provision for splitting segments which contain more than a single speaker, and thus diarization algorithms can only work well if the initial segmentation is of sufficiently high quality. Since this is rarely the case, alternative approaches combine clustering with iterative resegmentation, hence facilitating the introduction of missing speaker turns. Most of present diarization systems thus perform segmentation and clustering simultaneously or clustering on a frame-to-cluster basis, as described. The general approach involves Viterbi realignment where the audio stream is resegmented based on the current clustering hypothesis before the models are retrained on the new segmentation. Several iterations are usually performed. In order to make the Viterbi decoding more stable, it is common to use a Viterbi buffer to smooth the state, cluster or speaker sequence to remove erroneously detected, brief speaker turns, as in [16]. Most state-of-the-art systems employ some variations on this particular issue. An alternative approach to clustering involves majority voting whereby short windows of frames are entirely assigned to the closest cluster, i.e., that which attracts the most frames during decoding. This technique leads to savings in computation but is more suited to online or live speaker diarization systems.

E. One-Step Segmentation and Clustering

Most state-of-the-art speaker diarization engines unify the Segmentation and clustering tasks into one step. In these systems, segmentation and clustering are performed hand-in-hand in one loop. Such a method was initially proposed by ICSI for a bottom-up system [11] and has subsequently been adopted by many others [9], [10]. For top-down algorithms itwas initially proposed by LIA [14] as used in their latest system [16]. In all cases the different acoustic classes are represented using HMM/GMM models. EM training or MAP adaptation is used to obtain the closest possible models given the current frame-tomodel assignments, and a Viterbi algorithm is used to reassign all the data into the closest newly-created models. Such processing is sometimes performed several times for the frame assignments to stabilize. This step is useful when a class is created/eliminated so that the resulting class distribution is allowed to adapt to the data. The one-step segmentation and clustering approach, although much slower, constitutes a clear advantage versus sequential single-pass segmentation and clustering approaches [5]–[7]. On the one hand, early errors (mostly missed speaker turns from the segmentation step) can be later corrected by the re-segmentation steps. On the other hand, most speaker segmentation algorithms use only local information to decide on a speaker change while when using speaker models and Viterbi realignment all data is taken into consideration. When performing frame assignment using Viterbi algorithm a minimum assignment duration is usually enforced to avoid an unrealistic assignment of very small consecutive segments to different speaker models. Such minimum duration is usually made according to the estimated minimum length of any given speaker turn.

F. Resegmentation

The last stage found in many diarization systems is a resegmentation of the audio via Viterbi decoding (with or without iterations) using the final cluster models and nonspeech models. The purpose of this stage is to refine the original segment boundaries and/or to fill in short segments that may have been removed for more robust processing in the clustering stage. Filtering the segment boundaries using a word or phone recognizer output can also help reduce the false alarm component of the error rate.

Experiment and analysis

In this section, we report an analysis of speaker diarization in which the political database is taken as the input database. In the database the speaker is former president of USA, Mr. Barack Obama who is talking about the sustainability, the speech input is trim down to time duration of 16seconds.

Framing- Framing is the process which is done to divide the audio signal into equal individual speech frame to make it stationary.

Windowing-when the signal is framed it is necessary to consider how to treat the edges of the frame. This result from the harmonics the edges add. Therefore it is necessary to use a window to tone down the edges. Here we have use hamming window since it is more efficient windowing technique.Pre-emphasis filter- Pre-emphasis is the process which increase the amplitude of high frequency bands and decrease the amplitude of lower bands. And thus we finally get the pre-processed signal. Speech activity detection- Speech activity detection is used to detecting silent or non-speech part of audio signal. An ideal speech activity detector should be robust to noise and performs in real time. It is useful in speaker diarization as it would be process whole audio signal if first filter remove non-speeches from the signal.Feature Extraction- The process to extract features from audio data. MFCC (Mel frequency spectral coefficient) is widely used features in automatic speech recognition system. Sampling frequency of input audio.Then we have to do segmentation which is also called acoustic change detection aims to detect the speaker change such that each contiguous segment corresponds to single speaker only. To find the two segments corresponds to same speaker we have to define some notion of distance matric. The results are shown in the Figures- 1, 2, 3, 4 & 5.

Figures :

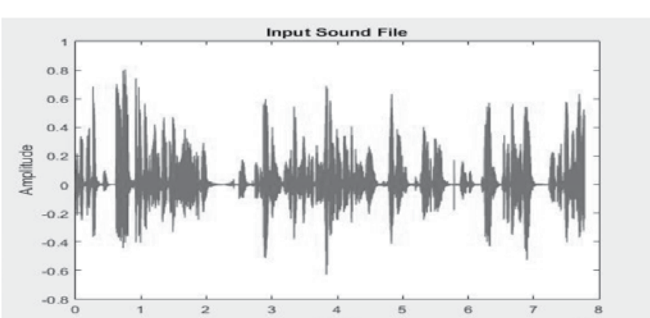

Fig.1: Input Signal

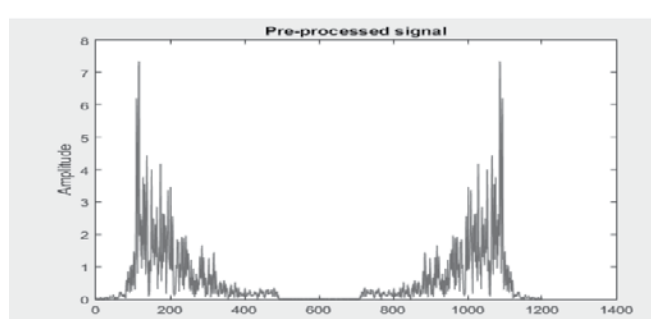

Fig.3: Pre-Processing of Signal

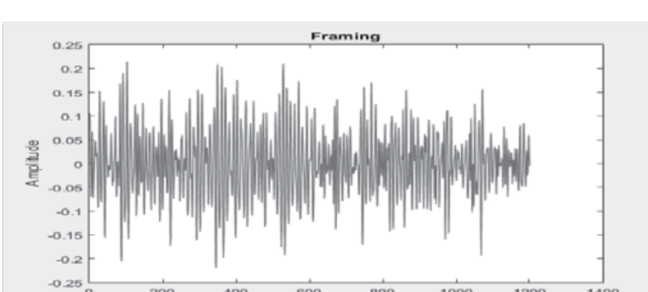

Fig.2: Framing of Speech Signal

Fig.4: Unvoiced Signal/Silent Part of Speech

Fig.5: Speech Without Silence

CONCLUSIONS

In audio processing and retrieval, speaker diarization is an important task which is the process of labeling a speech signal with labels corresponding to the identity of speakers. In this work, review related to speech diarization is represented. To conclude this paper, a complete framework for speaker indexing is proposed, which is aimed to be domain independent and parameter free and applicable for both online and offline applications

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Anshuman P Das
Sanjeev Kumar Sethi
Shishir Kumar Mohanty
Dept. of ECE

Analysis and Testing of Meters – An Internship Report

A subsidiary of Feedback Infra Ltd., FEDCO is an acronym for Feedback Energy Distribution Company. The company is currently involved in servicing nearly half a million consumers with an objective of creating greater operational efficiencies, better load management and improvement in consumer services. It serves four operational areas in the state of Odisha: Khorda, Nayagarh, Balugaon and Puri with an outreach of approximately 10000 Km2 and a service base of more than 4 Lakh consumers and growing. The territory covers a variety of geographical escapes: plain land, hilly terrain, forest and across expanses of water.

For any organization in the electricity distribution space, the key starting point is to know one's assets (technical hardware used in distribution). The legacy data received at inception was at worse archaic and at best inadequate which could not be used for the analysis and bill payment. The company aims at retrieving data in an organized and simplified manner, to increase efficiency and productivity of the company.

The biggest challenge faced by FEDCO was optimizing the revenue cycle management time to avoid any slippage in billing and collection. Alongside, the other issues were monitoring the field activities both in terms of quantity, quality, timeliness, fudging/collusion and timely data transfer for taking action. To achieve this objective GPRS (General Packet Radio Service) and Infrared enabled Spot Billing Machines (with Linux Operating System) have been deployed which apart from spot billing are also being used for collection and generate an Electronic Receipt. These SBMs are integrated with ENSERV for transmitting data to Server over the data network of telecom operators on live basis for monitoring purposes.

Based on the mandate of the Regulatory Commission, implementation by integrating Infrared (IR) enabled single phase meter with GPRS and IR enabled SBM's will lead to a reduction in human intervention. Here, the Meter Data is downloaded on to the spot-billing-machines through Infrared communication between the meter and the SBM; bills are generated without manual intervention; data is sent to the Central Server using GPRS connectivity and Electronic Payment Receipt is generated on the spot. In any utility 3Phase (Generally Industries) and HT (High Tension) consumers use the maximum electricity and consequently contribute more to the overall revenue. It therefore becomes critical to thoroughly sanitize such meters for errors and constantly analyze meter data for consumption, tampering and load declaration etc. Further it is desirable that for such consumers manual interfacing is avoided during meter reading. This is done through using Common Meter Reading Instruments (CMRI). These enable direct download of data on to these machines for later analysis and action.

In order to completely do away with human intervention and avoid consumer site visits and enable automated real-time data acquisition and analysis, the company is in the process of piloting an Automatic Meter Reading Project which in a phased manner will be widely adopted in the organization to cover Feeder and High consumption level meters. AMRs enable remote meter data acquisition through our Meter Data Acquisition System (MDAS) and are integrated with the Meter Data Management (MMD) module of 'aapt'.The AMR'S allow interaction only one way from the consumer to the energy supplier Whereas Smart Meters allow bidirectional communication between the supplier and the consumer. The amount of information they provide is also different. AMR meters only provide kWh information and possible peak kW demand for the month.

Smart Meters not only enable remote meter data acquisition but also allow discontinuing and reconnecting electricity supply without physically visiting a premise. This not only enables us to have control on defaulting parties but also allows consumers to remotely monitor misuse of electricity at their establishment. Advanced techno commercial discussions with our Principals CESU (Central Electricity Supply Utility of Odisha) are on for piloting and implementing this project, the first of its kind in India.

The detail component of an energy meter is shown in Fig. 1. The meter is constructed using 100% transparent virgin raw polycarbonate having good dielectric and mechanical strength which should be unbreakable , high grade and fire resistant with protection against dust

, moisture and vermin in combination with be non – hydroscopic, non-aging of tested quality and corrosion proof in normal conditions (most preferably in humid and tropical conditions). One of the major additions refers to the extended type cover transparent terminal cover to prevent unauthorized tampering .Push Buttons also called as scrolls are commonly used to display the various parameters .Output devices refers to the display that is present in the front so the readings can be observed during operation at the site . Test output device is to be provided in the form of blinking LED. The meter is provided with a terminal block, terminal cover to prevent the fire from spreading .The display is powered by Ni-mh and Ni-Cd batteries which are long lasting .It should remain unaffected by magnetic field of the magnitude 0.2 tesla on all sides and sustain a phase to phase voltage at least for 5 minutes The power supply provided is of micro –control type and the Current transformer is provided with magnetic shield.

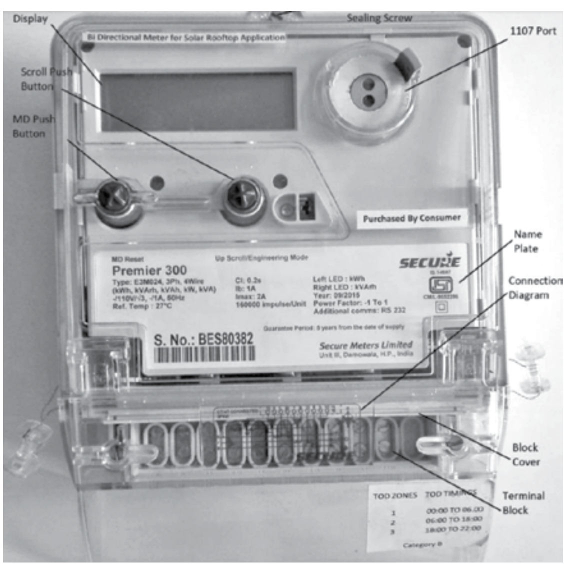

Fig. 1: Detail component of an energy meter

The synchronization is done through a Hand held unit and Remote server through suitable communication .The RTC battery for display however is separate which is made up of Ni-cd or Ni-mh. For communication capability two separate ports are provided. First, Optical port – present on the front of the meter, and a hardware port –RS232(which Is a standard introduced in1960 for serial communication ad transmission of data, generally from a computer to a modem).Detection or tamper events shall be registered plus the number of time tampering has occurred. A minimum of 200 events shall be recorded. They can be of the following types: Change in phase sequence, Reversal of load line terminals ,Drawing current through local earth and Three phase line wire without using neutral .Tampering is usually recorded through hand held units or remote access through suitable communication work .Display of measured events – a parameter identifier is present showing a minimum of 7 digits . a permanent backlit of a liquid crystal display I having a viewing angle of 120 degrees is present.It has a number of predefined parameters already entered into it – voltage rating (415V,ph-ph),auto reset mode to MD , average power factor having at least 2 decimal points , the data of a minimum of 45 days and an MD integration period with 30 minutes real based time . The composition of meters consists of current transformers as well as pressure transformers. Current transformers are used to reduce heating converting high current to low current according to the specified current ratio whereas pressure transformers are used to reduce the high voltage to low voltage according to the given Pressure ratio .The main objective of testing is to compare the load consumption with the meter reading and to observe if there is any deviation which is taken as the error (generally the comparison is done between the load current and the primary side current).

The meters can be classified on the basis of low voltage and high voltage: Whole meter current(single phase meter) , low tension 3-phase whole current static meter following DLMS protocol , high tension 3 phase hole current static meter following DLMS protocol. DLMS protocol stands for Device language message specification and COSEM- companion specification of energy metering.

Coming to the process of AMR 's as shown in Fig. 2 They are used by high valued consumers who use 3 phase meter consuming a load greater than 20KW. For example stone crushers and factories. They help in transferring data for billing, troubleshooting and analysis. It saves the expense of periodic trips to each physical location. Based on mobile and network technologies it uses radio

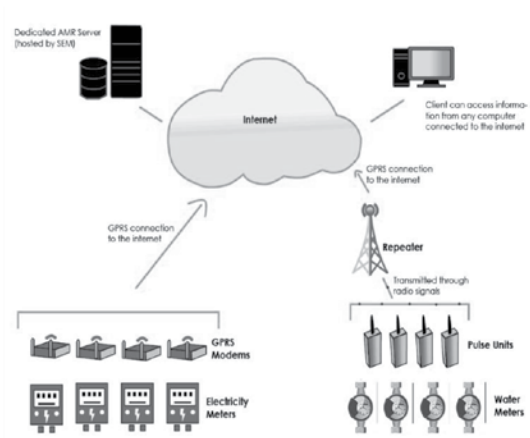

***Fig. 2:** Automatic Meter reading*

frequency and can be used in power line transmissions. In general, the GPRS modems transfer data from electrical modems to the internet where it is picked up by the AMR server (hosted by SEM).The data transfer takes place over a GPRS (general packet radio service)/ GSM (global systefor mobiles)

The radio frequency communication is of 2 types: a two way type radio communication calls Also known as "wake up call". It uses radio frequency to alert the transceiver to transmit the data. A 1 way radio communication: it sends the data continuously throughout the day which is very similar to broadband type and a hybrid communication which uses 1 way communication to do the reading and 2 way procedure for program communication. In some cases PLC (power line communication)-through this electronic data is transmitted over. Power line back to the substation relayed to a central computer in the utility main office.Depending on the number of users sharing the device, the GPRS is the best effort of throughput (maximum rate of production and rate of transfer of data)and latencyas shown in fig.3.

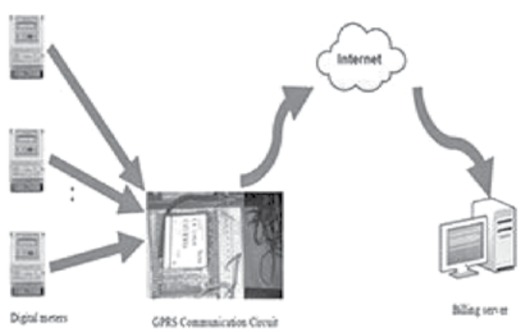

Fig. 3: A GPRS Network

It is typically charged on the basis of the volume of data that is transferred .it provides moderate data transfer using time division multiple access channel.GPRS is a packet-based wireless communication service that provides data transmission rates from 56 to 114 kbps. Due to the fast data rate and widespread usage in mobile communication worldwide, GPRS has potential to be developed for use in energy metering systems with access to the private/public network over the wireless system. The Features of GPRS Communication Technology.Fast Speed and Wide Coverage Area – It shortens the access time from the End-Customer Access Point (Domestic-Unit Meters) to the central data server, and it can serve wider coverage areas to minimize the traffic issue. Network Extensibility - GPRS can be complemented with Internet Protocol (IP), X.25, GSM EDGE, and UMTS, which are all gaining worldwide popularity.

Developed by Bell Laboratories in 1970, the GSM network is widely used in Mobile communication.It is an open/digital cellular system that transmit mobile and voice data and data service operating at 850, 900, 1800 and 1900MHz .The GSM network uses TDMA channel access method for sharing frequency by dividing it into different time slots. Multiple stations to share the same transmission medium while using part of its channel capacity. Instead of using 1 transmitter, there are several transmitter present with respect to 1 receiver(Multi-receivers to one receiver).

A GSM Network usually consists of-A Mobile Station, it is the mobile phone which consists of the transceiver, the display and the processor and is controlled by a SIM card operating over the network.Base Station Subsystem which It acts as an interface between the mobile station and the network subsystem. It consists of the Base Transceiver Station which contains the radio transceivers and handles the protocols for communication with mobiles. It also consists of the Base Station Controller which controls the Base Transceiver station and acts as an interface between the mobile station and mobile switching center.Network Subsystemwhich provides the basic network connection to the mobile stations. The basic part of the Network Subsystem is the Mobile Service Switching Centre which provides access to different networks like ISDN, PSTN etc. It also consists of the Home Location Register and the Visitor Location Register which provides the call routing and roaming capabilities of GSM. It also contains the Equipment Identity Register which maintains an account of all the mobile equipment wherein each mobile is identified by its own IMEI number. IMEI stands for International Mobile Equipment Identity.

The final stage procedure takes place as the analysis of the Instantaneous parameters –that consists of the general parameters voltage, current, power factor, load and the vector diagram, Billing / Reading – Maximum Demand, and the three powers (Active, reactive and Real), Load survey – Data of the amount of consumption is supplied to the server at particular intervals to observe any abnormalities in the consumption,Load survey – Data of the amount of consumption is supplied to the server at particular intervals to observe any abnormalities in the consumption.

K.V.Pallavi
Dept. of EEE

Smart Attendance System using Biometrics

Abstract: *Conventional attendance system followed in an educational system, where the teacher calls out the name of each student and marks the attendance, causes time wastage during lecture time. This becomes more severe especially when the number of students in a class is very large. Managing the attendance data of such a large group is also very difficult. Another disadvantage of the conventional systems is the chance for the student to mark fake attendance. The need for easy recognition of students in various activities has become very important in educational institutes. This work presents the attendance management system using fingerprint technology in an Institute environment.*

I. INTRODUCTION

Conventional attendance system followed in an educational system, where the teacher calls out the name of each student and marks the attendance, causes time wastage during lecture time. This becomes more severe especially when the number of students in a class is very large. Managing the attendance data of such a large group is also very difficult. Another disadvantage of the conventional systems is the chance for the student to mark fake attendance. The need for easy recognition of students in various activities has become very important in educational institutes. This is to check truancy and lateness to classes by students, respectively. Among the most important of these activities are Lecture and Laboratory attendance and Semester Examinations.

In recent time, there has been high level of impersonation experienced on a daily basis in private and public sectors. The ghost worker syndrome has become a menace across all tiers of government. Employer is concerned over the levels of employee absence in their workforce and has difficulty in managing attendance. Fingerprints are a form of biometric identification which is unique and does not change in one's entire lifetime. Every organization whether it be an educational institution or business organization, it has to maintain a proper record of attendance of students or employees for effective functioning of organization. Designing a better attendance management system for students so that records are maintained with ease and accuracy was an important key behind motivating this work.

This would improve accuracy of attendance records because it will remove all the hassles of roll calling and will save valuable time of the students as well as teachers. Biometric Identification Systems are widely used for unique identification of humans mainly for verification and identification. Biometrics is used as a form of identity access management and access control. So, use of biometrics in student attendance management system is a secure approach. There are many types of biometric systems like fingerprint recognition, face recognition, voice recognition, iris recognition, palm recognition etc. In this work, we used fingerprint recognition system.

This work presents the attendance management system using fingerprint technology in a university environment. It consists of two processes namely; enrolment and authentication. During enrolment, the e fingerprint of the user is captured and its unique features extracted and stored in a database along with the user's identity as a template for the subject. The unique features called minutiae points were extracted using the Crossing Number (CN) method which extracts the ridge endings and bifurcations from the skeleton image by examining the local neighborhoods of each ridge pixel using 3 x 3 window. During authentication, the fingerprint of the user is captured again and the extracted features compared with the template in the database to determine a match before attendance is made.

II. USE OF FINGERPRINT

A fingerprint is the pattern of ridges and valleys on the surface of a fingertip. The endpoints and crossing points of ridges are called minutiae. [1] It is a widely accepted assumption that the minutiae pattern of each finger is unique and does not change during one's life. Ridge endings are the points where the ridge curve terminates, and bifurcations are where a ridge splits from a single path to two paths at a Y-junction. When human fingerprint experts determine if two fingerprints are from the same finger, the matching degree between two minutiae pattern is one of the most important factors.

Fingerprints are considered to be the best and fastest method for biometric identification. They are secure to use, unique for every person and does not change in one's lifetime. Besides these, implementation of fingerprint recognition system is cheap, easy and accurate up to satisfiability. Fingerprint recognition has been widely used in both forensic and civilian applications. Compared with other biometrics features, fingerprint-based biometrics is the most proven technique and has the largest market shares. Not only it is faster than other techniques but also the energy consumption by such systems is too less.

Managing attendance records of students of an institute is a tedious task. It consumes time and paper both. To make all the attendance related work automatic and online, we have designed an attendance management system that uses a fingerprint identification system. This fingerprint identification system uses existing as well as new techniques in fingerprint recognition and matching. A new one to many matching algorithm for large databases has been introduced in this identification system.

III. MODEL

This work presents the attendance management system using fingerprint technology. There is nothing about the reporting of students' attendance, which can evaluate the rate of discipline for each student, or the monitoring of students' attendance by their guardians. This work presents online students 'attendance monitoring system using RFID (Radio Frequency Identification technology). However, it uses the ID card affixed with RFID tag. Hence, when a student or staff forgets his ID card they will not be recognized by the system.

The general structure of a fingerprint scanner is shown in Figure 1: a sensor reads the finger surface and converts the analogue reading in the digital form through an A/D (Analog to Digital) converter; an interface module is responsible for communicating (sending images, receiving commands, etc.) with external devices (e.g., a personal computer). The different technologies the sensors are based on (e.g., optical, solid-state, ultrasound, etc.). [2]

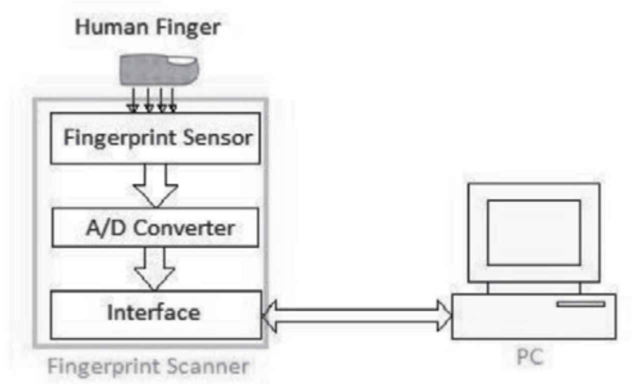

Figure.1: structure of a fingerprint scanner

During the enrollment phase, the biometric characteristic of an individual is first scanned by a biometric reader to produce a raw digital representation of the characteristic and a compact but expressive representation, called a template is generated and stored in the database. During the operation phase, the user's name or PIN (Personal Identification Number) is entered through a keyboard (or keypad); the biometric reader captures the characteristic of the individual to be recognized and converts it to a digital format, which is further processed by the feature extractor to produce a compact digital representation. The resulting representation is fed to the feature matcher, which compares it against the template of a single user (retrieved from the system DB based on the user's PIN). In the identification task, no PIN is provided and the system compares the representation of the input biometric against the templates of all the users in the system database; the output is either the identity of an enrolled user or an alert message such as "user not identified." [3]

Finger Print Sensor Module or Finger Print Scanner is a module which captures finger's print image and then converts it into the equivalent template and saves them into its memory on selected ID (location) by Arduino. Here all the process is commanded by Arduino like taking an image of finger print, convert it into templates and storing location etc. During enrolling the student is asked to lace his finger over the finger print module. Now user needs to put his finger over finger print module. Then LCD will ask to remove the finger from finger print module and again ask for placing the finger. Now user needs to put his finger again over finger print module. Now finger print module takes an image and converts it into templates and stores it by selected ID in to the finger print module's memory. Now user can open the gate by placing the same finger that he/she have added or enrolled into the system. By the same method, the user can add more fingers. The enrolment phase and authentication phase is shown in Figure.2.

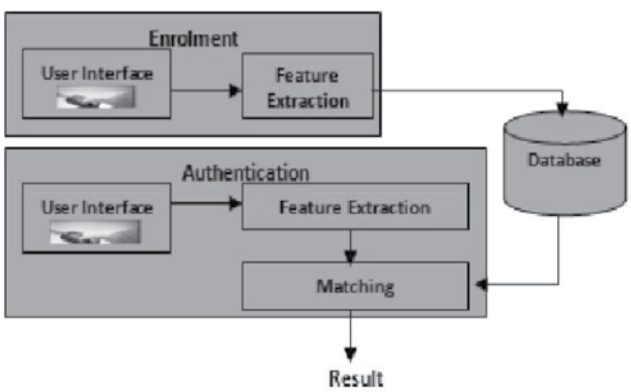

Figure.2: fingerprint enrolment and authentication process

The SD card module is a sample solution for transferring data to and from a standard SD card. It used to store the attendance and transfer to database of PC in this system. The pin out is directly compatible with Arduino, but can also be used with other microcontrollers. It allows to add the mass storage. The module has SPI interface which is compatible with any SD card and it use 5v or 3.3v power supply which is compatible with Arduino mega. [4]

IV. IMPLEMENTATION

The following aspects of biometrics system testing and evaluation have been addressed:

- The acceptance of the biometrics captures methods.
- The practicality of the biometrics captures methods.
- The cost effectiveness of the biometrics system.
- The repeatability of the biometrics scan.

The proposed e-attendance system utilizes fingerprint information, which are relatively fixed in humans and, therefore, virtually eliminating the need to repeat the enrolment process. The acceptance, practicality and cost-effectiveness aspects of the proposed e-attendance system are discussed.

Fingerprint images that are found or scanned are not of optimum quality. So we remove noises and enhance their quality. We extract features like minutiae and others for matching. If the sets of minutiae are matched with those in the database, we call it an identified fingerprint. After matching, we perform post-matching steps which may include showing details of identified candidate, marking attendance etc.

For identification process, first, it takes an image of a finger. Then finger scanner saves characteristics of every unique finger and saved in the form of biometric key. Actually, finger print scanner never saves images of a finger only series of binary code for verification purpose. No one can change the algorithm into an image so it is totally impossible to duplicate your fingerprints so no need to worry about it.

Secondly, the biometric attendance system determines whether the pattern of ridges and valleys in this image matches the pattern of ridges and valleys in pre-scanned images. When a student enters in the classroom, he/she punches his/her finger on the fingerprint scanner. If the image matches with the template stored in the database, then the student is granted access to enter into the classroom else he/she is denied access in the classroom which is shown in snapshots. The student can later on view his/her attendance by logging in into the website of the institution.

Algorithm to Enroll the Students:

Step 1: Initialize Fingerprint Scanner from Serial Port

Step 2: Get sensor information

Step 3: Start service to read a finger

Step 4: Wait that finger is read

Step 5: Checks if finger is not already enrolled go to step 8

Step 6: print message with id

Step 7: Go to step 16

Step 8: Converts read image to characteristics and stores it in charbuffer 1

Step 9: Wait that finger is read again

Step 10: Converts read image to characteristics and stores it in charbuffer 2

Step 11: Compares the charbuffers

Step 12: If fingerprint does not match raise exception

Step 13: Creates a template and store it to fingerprint scanner device

Step 14: Get Id and position and save to database

Step 15: Print success message

Step 16: Stop

Algorithm to verifying attendance:

Step 1: Initialize Fingerprint Scanner from Serial Port

Step 2: Get sensor information

Step 3: Start service to read a finger

Step 4: Create new session for attendance

Step 5: If system is exit go to step 15 Step 6: Wait that finger is read

Step 7: Search the finger and calculate hash

Step 8: Get finger position from the device Step 9: If finger match go to step 12

Step 10: Print “student not found”

Step 11: Go to step 6

Step 12: Match position with database and get id

Step 13: Mark present of that id in session

Step 14: Go to step 6

Step 15: Stop

V. CONCLUSIONS

The industry is shifting away from cumbersome passwords to more secure biometrics, and for very good reasons. From hackers hacking, to users forgetting them, to juggling and constantly changing dozens of them, passwords are archaic and change is paramount. Fingerprint authentication is leading the way, with innovation around every corner. Next steps may include adding multi factor biometrics which will once again greatly enhance the security and further improve usability. In the presented model, only few data in the local server are stored, however huge amounts of these data can be stored in different cloud storage platforms where it could be accessed easily. Further an android application can be developed for the entire system to take the attendance.

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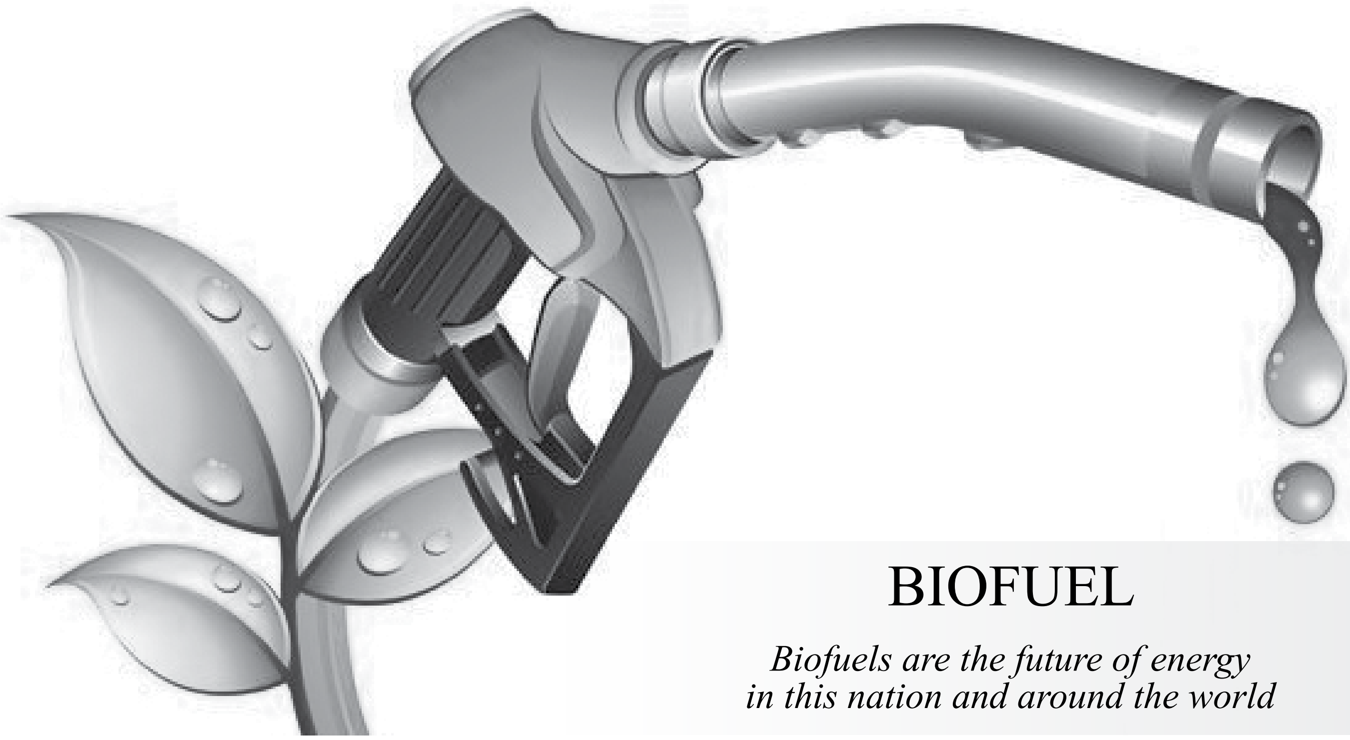
Manisha Nandi, Monalisa Routray
Rupshana Ekka
Dept. of IT



BIOFUEL

Biofuels are the future of energy in this nation and around the world

Bio-fuel is the fuel which is produced from organic products and wastes. Solid, liquid, or gaseous fuels that are produced from biomass are called biofuel i.e., any fuel that is extract from biomass like plant or algae material or animal waste. Since such feedstock material can be replenished readily, that's why biofuel is considered to be a source of renewable energy, unlike fossil fuels such as petroleum, coal, and natural gas.

The common commercially used biofuels are bioethanol, biodiesel and biomethane. Bioethanol is made from sugar, algae, wheat and sugar beet. Biodiesel is made from vegetable oil, algal lipids, and animal fats. Biomethane can be produced from waste organic material, sewage, agriculture waste and domestic wastes.

Biofuel was introduced in early eighteen century.In 1890s, Rudolf Diesel was the first person who made biodiesel from vegetable oil. In 1970s and 1980s environmental protection agency EPA situated in America suggested that fuel should be free from sulphur dioxide, carbon monoxide and nitrogen oxides. In 1998 EPA allowed the production of biofuels on commercial level which was the alternative source of the petrol. In 2010 the production of biofuels reached up to 105 billion liters worldwide. In 2011, European countries were the largest that made biodiesel almost about 53%. The international Energy Agency set a goal to reduce the usage of petroleum and coal and will be switched on to biofuels till 2050.

First-generation biofuels or conventional biofuels are biofuels made from food crops grown on arable land. Sugar, starch, or vegetable oil obtained from the crops is converted into biodiesel or ethanol. Second generation biofuels are fuels produced from various types of biomass. Biomass is plant or animal material used for energy production (electricity or heat), and used in various industrial processes as raw material for a range of products. Third-generation biofuel are extract from algae mostly marine algae. Fourth-generation biofuels are similar to third-generation biofuel, these biofuels are made using non-arable land. However, unlike third-generation biofuels, they do not require the destruction of biomass. This class of biofuels includes electro fuels and photo biological solar fuels. Some of these fuels are carbon-neutral.

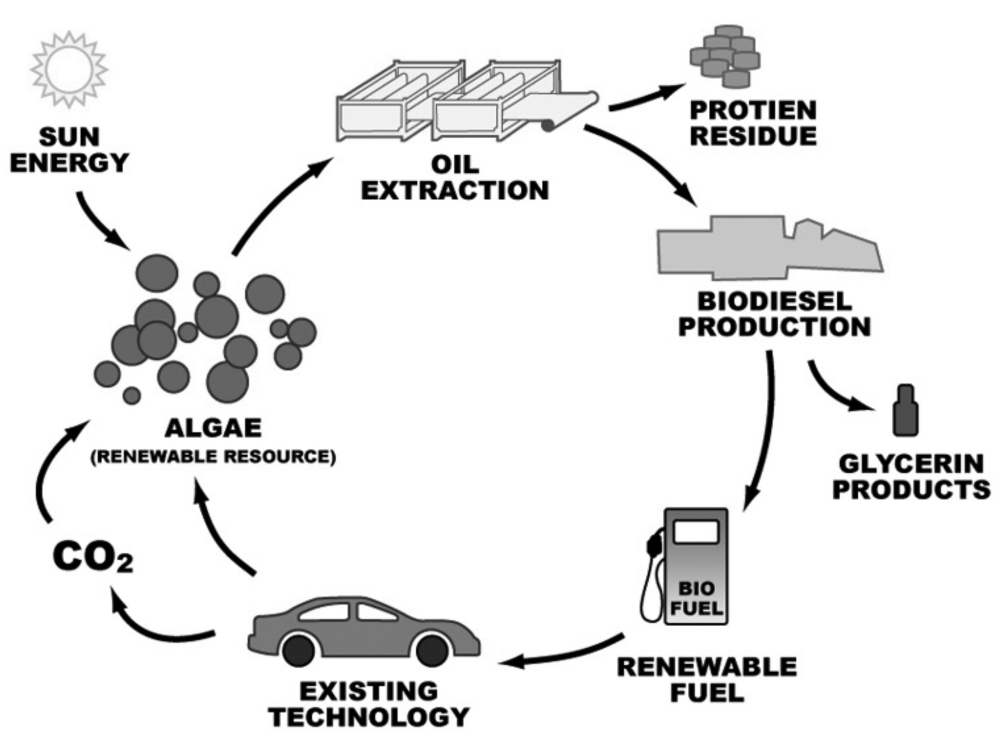
The main advantage of biofuel over fossil fuels is that burning them emits less of the greenhouse gas into the atmosphere. The are renewable, biodegradable and safer. Biofuels shouldn't be produced by starving people and destroying the already-depleting forests. Nuclear energy can completely revolutionize the energy scenario in the world. There are some disadvantages of biofuel are high cost of production, industrial pollution, future rise in price, shortage of food, increased use of Fertilizer.

Many scientists recently are doing research on biofuels. Some of them are bioethanol, jatropa, fungi and animal gut bacteria .Bioethanol is a form of renewable energy that can be produced from agricultural feedstock. Jatropha curcas, a poisonous shrub-like tree that produces seeds, is considered by many to be a viable source of biofuels feedstock oil. Most of this research focuses on improving the overall per acre oil yield of jatropa through advancements in genetics, soil science, and horticultural practices. Biofuel had isolated large amounts of lipids from single-celled fungi and turned it into biofuels in an economically efficient manner. More research on this fungal species, animal gut bacteria- Microbial gastrointestinal flora in a variety of animals has shown potential for the generation of biofuels.

The effective application of Biotechnology is to improve Renewable Fuel Production. Engineered microorganisms or plants are used to manufacture enzymes used in fuel production and to improve algae strains for Biofuel production. Selected or engineered plant species with favorable traits for use as improved Biofuel feedstock. Therefore, the future contributions of biotechnology to the energy industry are not only influenced by technical advances in biotechnology, but also by the price of fossil fuels, the development of renewable energy generally, politics, global population growth, and other factors.

The growing use of Biofuel will be an invaluable contribution to the generation of income, society, inclusion, reduction of poverty in major poor countries of the world.

Rohit kumar Nayak
Dept. of ECE



Priority Based Multi-Phase Phasor Measurement Unit Placement Using Soft-Computing Techniques for Smart Grid Implementation

The conventional grid is in the process of transforming into Smart Grid (SG) and the basic requirements for SG execution are advanced measurement infrastructure for obtaining fast and highly precise measurement data as well as transferring such data through fast and reliable communication channel. Phasor measurement unit (PMU) is the most suitable device to fulfil such above mentioned requirements. To achieve the full network observability is one of the important requirements of the power industries. A network is considered to be fully observable when all its bus voltage phasors are obtained whether by direct or pseudo measurements. However, due to high PMU installation cost, the researchers were forced to find an alternative way strategically place optimal number of PMUS at strategic locations to achieve the full observability of the entire transmission network. In the problem of optimal PMU placement (OPP), the objective is to find out the minimal number of PMUs with proper locations to obtain the full observable network.

While solving the OPP problem, the soft-computing algorithms can play significant role for providing multiple optimal solutions (MOSs) for OPP. The MOSs for OPP can provide different observability approach with same number of PMUs. However, the effectiveness and necessity of MOSs was never addressed in the earlier literatures. In practical power industry, the planning engineers (PEs) are more concerned about the redundant observability of the critical buses (CBs) such as vulnerable, generator, tie-line and transformer-connected buses. The CBs contain several critical information of the network health and loosing such crucial data can affect the power system secure operations. In this regards, the MOSs can provide the redundant measurements of the CBs without compromising the full observability criteria and with the same number of PMUs. Therefore, in the proposed work is objective is to obtain the possible MOSs for OPP using different soft-computing algorithms and choosing the best desirable solution strategically, depending on the redundant observability of CBs.

Even for higher networks, optimal PMU placement is subjected to high initial investment. Therefore, a multi-phase PMU placement approach is adopted in the thesis to avoid high investment. According to the proposed method, the observability of the CBs is prioritized at the initial and intermediate phases so that critical information can be achieved at the earliest concern.

In practical power industry, the cost of overall PMU installation at a bus is dependent on several factors such as cost of PMU device, communication channel, equipment, software, cyber security, etc. However, in most of the literatures, a uniform cost factor was assumed for all PMU installed buses. Therefore, to incorporate the variation in the PMU installation cost, a cost minimization based OPP is proposed in this work. Considering the requirement of the PEs in practical scenario, the direct observability of the CBs is also preferred along with the OPP. The direct observability of CBs can provide better decision making ability to improve the power systems operations as well as to predict future possible disturbance.

In modern power system era, the developing and developed countries have already initiated to implement their SG infrastructure to achieve the self-monitoring, self-control, self-protection and self-healing properties. To achieve those, highly reliable and precise data quality with strong data communication network is very essential that can be obtained by installing PMUs at all buses of transmission network. In this regards, a PMU placement strategy is proposed in this work for direct observability of the entire network. Since PMU placement at all buses can be an expensive proposition, a novel multi-phase PMU placement approach is proposed.

Finally, the performance of all the proposed algorithms and their results are compared with each other.

Dr. Tapas Kumar Maji

SO_2 emissions in India

As per a new report by Greenpeace released on August 19, 2019, India has become the largest emitter of sulphur dioxide (SO2) in the world, contributing more than 15 per cent of global anthropogenic emissions. Sulphur dioxide is a major contributor to acid rain; being a criteria pollutant it can cause serious damage to the human respiratory system, and hurt foliage.

Due to expansion of coal-based electricity generation, high emission of SO2 is on the rise. As a result, according to Greenpeace report five of the top 10 SO2 emission hotspots from coal/power generation industry across the world are in India.

This analysis is based on hotspots detected by NASA Ozone Monitoring Instrument (OMI) satellite data that captured more than 500 major source points of SO2 emissions across the globe including natural sources such as volcanoes, but natural sources were not considered while preparing report.

According to the study, the thermal power plants or clusters at Singrauli, Neyveli, Talcher, Jharsuguda, Korba, Kutch, Chennai, Ramagundam, Chandrapur, and Koradi are deemed to be the major emission hotspots in the country. One of the reasons is that the vast majority of plants in India lack flue-gas desulfurisation (FGD) technology to reduce sulphur oxide emissions.

The Union Ministry of Environment, Forest and Climate Change had, for the first time, introduced SO2 emission limits for coal-fired power plants in December 2015. But a Supreme Court order changed the deadline for installation of FGD technology in power plants from 2017 to December 2019 in Delhi-NCR and till 2022 for other parts of the country.

The condition is getting worse in India, as there has been an increase of SO2 emissions at already existing hotspots and new emerging sites are also adding emissions to the existing ones across the country.

When it comes to individual hotspots, the Norilsk smelter site in Russia continues to be the largest anthropogenic SO2 emission hotspot in the world, followed by the Kriel area in Mpumalanga province of South Africa, Zagroz in Iran, and Rabigh in Saudi Arabia. Singrauli in Madhya Pradesh is at number five.

According to the new target, the world will have to cut its GHG emissions by 45 per cent in the next 11 years and bring it to zero by 2050.

Clean air is a major problem and according to the World Health Organization, air pollution annually kills 7 million people all over the world, 600,000 of them being children.

Governments must take initiative to implement e-mobility and other sustainable mobility practices and should go for implementing national fuel quality standards, support the implementation of tighter vehicle emission standards and actively promote alternative fuels.

Source: https://www.downtoearth.org.in/news/air/india-emits-the-most-sulphur-dioxide-in-the-world-66230

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