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An Alternative Analysis of Two Circular Variables via Graphical Representation: An Application to the Malaysian Wind Data

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Abstract

The relationship between variables is vital in data analysis. The scatter plot, for instance, gives an easy preliminary exploratory analysis for finding relationship between two variables, if any. Statistical method such as correlation and linear relationship are standard tools in most statistical packages. For circular variables that take value on the circumference of a circle, the analysis however is different from those of the Euclidean type variables because circumference is a bounded closed space. Unlike linear variable, standard statistical packages for circular variables are limited. This paper proposes a graphical representation of two circular variables as a preliminary analysis using the MATLAB environment. A plot called Spoke plot is developed to visually display relationship between two circular variables and linear correlation. As an illustration, the Malaysian wind data is used in the analysis. This new type of representation promises an alternative approach in the preliminary analysis of circular data.

Keywords: Spoke plot, Wind data, Circular, Correlation, Linear Relationship

1. Introduction

Circular variables take values on the circumference of a circle and thus are bounded in a closed space. Unlike the usual Euclidean type variables, the analysis of directional data is different. Further readings in circular variables can be found in Fisher (1993) and Mardia (1999, 1972). A number of environmental data are circular in nature such as wave, wind direction, compass bearing, clock and others. Statistical softwares devoted to the analysis of circular variable are limited. At present, there are only two softwares in the market namely AXIS (Handerson et el., 2002) and ORIANA (Oriana Software, 2007), that offer some statistical analysis of circular variable on a window-based environment. Although these packages offer a range of graphical and analytical techniques required for statistical analysis of circular data, they have some limitations. For example, further analysis such as statistical inference, analyzing grouped data sets, circular plots of the corresponding probability density functions as well as regression and correlation of circular data are not available in the existing packages. In this study, an alternative diagrammatical representation of correlation and linear relationship analysis between two circular variables are developed. By using Matlab (Matlab Software, 2004), the programs generate graphical and calculated outputs of the analysis with a window-based environment. By interfacing with the existing softwares, this analysis could enhance the exploratory analysis of circular variable with respect to software development.

2. Theoretical formulation of the analysis of circular data

2.1 Circular data

Data on the angular displacements, directional propagations and in general periodic occurrence can be casted within the framework of directional or circular data. In other words, circular data is one which takes values on the circumference of a circle, i.e. they are angles in the range of $(0,2\pi)$ radians or $(0^0,360^0)$. To analyze this type of data, we must use techniques differing from those of the usual Euclidean type variables because the circumference is a bounded closed space, for which the concept of origin is arbitrary or undefined. Thus, the techniques that have been used for continuous linear data cannot be applied for circular data. Furthermore, continuous linear data are realized on the straight line or real line which may be analyzed straightforwardly by usual statistical techniques.

2.2 Analysis between two circular variables

As mentioned earlier, unlike linear variables, standard statistical package for analysing circular variables is limited (Friendly, 2002) and evidence on the utilization of such packages in published works is also few. For example Hussin et el. (2006) evaluated the performance of AXIS by running several exploratory analysis of the Malaysian Wind data. This paper will focus on the analysis of two circular variables. The correlation measure and linear relationship will be used in the development of the preliminary analysis of two circular variables in addition to a graphical representation.

2.3 Correlation between circular variables

Correlation or also known as a measure of a correlation coefficient indicates the strength and direction of a linear relationship between two random variables. In general statistical usage, correlation or co-relation refers to the departure of two variables from independence. When the data are linear there are several coefficients, measuring the degree of correlation, adapted to the nature of data.

A similar measure of association between two circular variables is not well known. Given n pairs of circular data $(\theta_1,\varphi_1),...,(\theta_n,\varphi_n)$, where $0 \le \theta_i,\varphi_i < 2\pi$ of circular variables θ and φ, the circular correlation coefficient given by Fisher is defined by

$$\hat{\rho}_T = \frac{\sum_{1\le i\le j\le n} \sin(\theta_i-\theta_j)\sin(\varphi_i-\varphi_j)}{\sqrt{\sum_{1\le i\le j\le n} \sin^2(\theta_i-\theta_j) \sum_{1\le i\le j\le n} \sin^2(\varphi_i-\varphi_j)}} \tag{1}$$

Alternatively, one can transform equation (1) to

$$\hat{\rho}_T = \frac{4(AB-CD)}{\sqrt{(n^2-E^2-F^2)(n^2-G^2-H^2)}} \tag{2}$$

where

$A = \sum(\cos\theta_i \cos\varphi_i)$ $\quad$ $B = \sum(\sin\theta_i \sin\varphi_i)$

$C = \sum(\cos\theta_i \sin\varphi_i)$ $\quad$ $D = \sum(\sin\theta_i \cos\varphi_i)$

$E = \sum(\cos^2\theta_i)$ $\quad$ $F = \sum(\cos^2\theta_i)$

$G = \sum(\cos^2\varphi_i)$ $\quad$ $H = \sum(\sin^2\varphi_i)$

2.4 Linear Association between two circular variables

The regression model when both variables are circular produces very interesting form. Hussin (2007) present the hypothesis testing of parameters for ordinary linear circular regression model assuming the circular random error distributed as von Mises distribution. The model is given by

$$\varphi = \alpha + \beta\theta + \varepsilon \pmod{2\pi} \tag{3}$$

where ε is a circular random error having a von Mises distribution with mean circular 0, and concentration parameter κ, which can be written as $\varepsilon \sim VM(0,\kappa)$. We can estimate α and β by maximum likelihood estimation. Based on von Mises density function, the log likelihood function for model (3) is given by

$$\log L(\alpha,\beta,\kappa;\theta_1,...,\theta_n,\varphi_1,...,\varphi_n) = -n\log(2\pi) - n\log I_0(\kappa) + \kappa\sum\cos(\varphi_i - \alpha - \beta\theta_i) \tag{4}$$

By differentiating $\log L$ with respect to α,β and κ the estimates of α and β namely $\hat{\alpha}$, $\hat{\beta}$ and $\hat{\kappa}$ are given by

$$\hat{\alpha} = \begin{cases} \tan^{-1}\left(\frac{S}{C}\right), & S > 0, C > 0 \\ \tan^{-1}\left(\frac{S}{C}\right) + \pi, & C < 0 \\ \tan^{-1}\left(\frac{S}{C}\right) + 2\pi, & S < 0, C > 0 \end{cases} \quad (5)$$

where $S = \sum \sin(\varphi_i - \hat{\beta}_{i-1}\theta_i)$ and $C = \sum \cos(\varphi_i - \hat{\beta}_{i-1}\theta_i)$ and

$$\hat{\beta}_i \approx \hat{\beta}_{i-1} + \frac{\sum \theta_i \sin(\varphi_i - \hat{\alpha} - \hat{\beta}_{i-1}\theta_i)}{\sum \theta_i^2 \cos(\varphi_i - \hat{\alpha} - \hat{\beta}_{i-1}\theta_i)} \quad (6)$$

respectively. Further, the maximum likelihood estimate of κ is

$$\hat{\kappa} = A^{-1}\left(\frac{1}{n}\sum \cos(\varphi_i - \hat{\alpha} - \hat{\beta}\theta_i)\right). \quad (7)$$

The approximation given by Dobson for the function A (ratio of the modified Bessel functions for the first kind of order one, and the first kind of order zero) for the von Mises concentration parameter, κ is

$$A^{-1}(w) \approx \frac{9 - 8w + 3w^2}{8(1-w)}. \quad (8)$$

These expressions may be solved iteratively given some suitable "initial guesses" at the estimate. The estimated $\hat{\beta}$ is obtained by iterative procedure at some predetermined stopping rules.

More often than not, an investigation on the relationship between two circular variables is required. As mentioned earlier, the strength of the correlation between two circular variables can be numerically computed using the correlation coefficient $\hat{\rho}_T$ as shown in (1). The $\hat{\alpha}$ and $\hat{\beta}$ for linear association between two circular variables can be calculated as shown in (5) and (6). In this analysis, however, the von Mises concentration parameter, κ is not include in the calculation.

2.5 Diagrammatical representation of two circular variables

Many statistical tools exist for analyzing their structure, but, surprisingly, there are few techniques for exploratory visual analyses, and for depicting the patterns of relations among variables (Friendly, 2002). The concepts of "one picture is worth a thousand words" have been used by some researchers such as Linden (2005) who developed visual display in the disease management program, Vinnakota (1988) designs charts to understand composite beam design problem and Friendly (2001) wrote macro programs for graphical analysis to reveal features of categorical data that are not apparent in traditional numerical summaries.

In most exploratory data analysis of two variables, visual representation of the correlation can provide better understanding of the association between the two variables. In the analysis of two circular variables, a diagrammatical representation to describe the relationship is developed and known as the Spoke plot. The Spoke plot consists of inner and outer rings in which lines are used to connect the pair of points (θ_i, φ_i,). Together with the Spoke plot, the program calculates correlation and parameters of the linear association between two variables.

3. Source of data

In this study, the Malaysian wind data that has been obtained from the Malaysian Meteorological Services Department is used. The data was collected daily and measured by anemometer at several airport locations throughout Malaysia over period of one year in 2005 and at a telecommunication tower in Seberang Jaya in April 2002.

4. Result and findings

A call function is developed using MATLAB version 7.0, in the analysis of two directional data. After running the call function "spokecorrelation" for the data, an output window is generated that gives the:

i. calculated correlation value of two circular variables.

ii. calculated linear association measure of two circular variables.

iii. relationship of two circular variables using Spoke plot.

Thus one can easily make comparison of the three analyses in one output window. For illustration, we run an analysis for three datasets of wind data. They are the:

i. wind direction data at maximum speed in January 2005 between KLIA and Ipoh, with the objective to compare two sets of wind data at two different locations.

ii. wind direction data in January between KLIA (time = 0000, pressure = 1000hpa) and KLIA (time = 0000, pressure = 500hpa), with the objective to compare two sets of wind data at different pressures.

iii. wind direction data recorded at Telecommunication tower, Seberang Jaya in April 2002 between the heights 45.72m and 75.28m, with the objective to compare two sets of wind data at different level.

The results are shown in Figure 1, Figure 2 and Figure 3, respectively.

5. Conclusion

The need of software development in the analysis of two circular variables is necessary. This is because a variety of environmental data are circular in nature such as wave, wind direction, frequency, compass bearing, clock and others. In this study, a program that evaluates statistical functions as well as diagrammatical representation is presented. Using MATLAB, the output window gives all the three analyses namely Spoke plot, correlation and linear association in one diagram. This research could be improved by developing further the Graphical User Interface (GUI) into the packages for the ease of application by the user.

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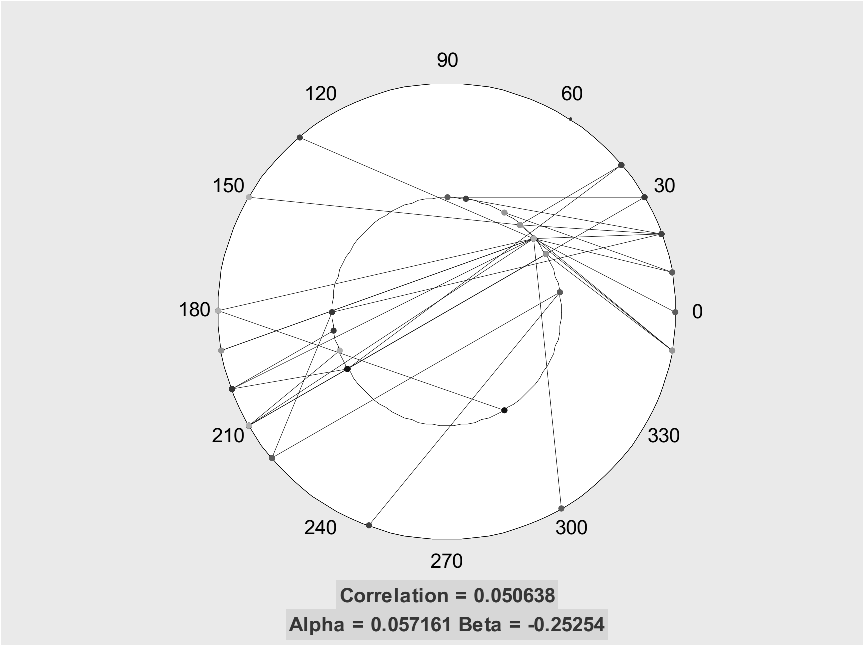

Figure 1. Spoke Plot of wind direction data at maximum speed

in January 2005 between KLIA and Ipoh.

From the Spoke plot in Figure 1, it can be seen that a number of lines crossing the inner ring implies that there is no correlation between the variables. To support the finding, the calculated correlation value of Equation (1) is 0.0506 which indicates no correlation. The linear association value also shows the absence of one to one linear relationship between the two circular variables.

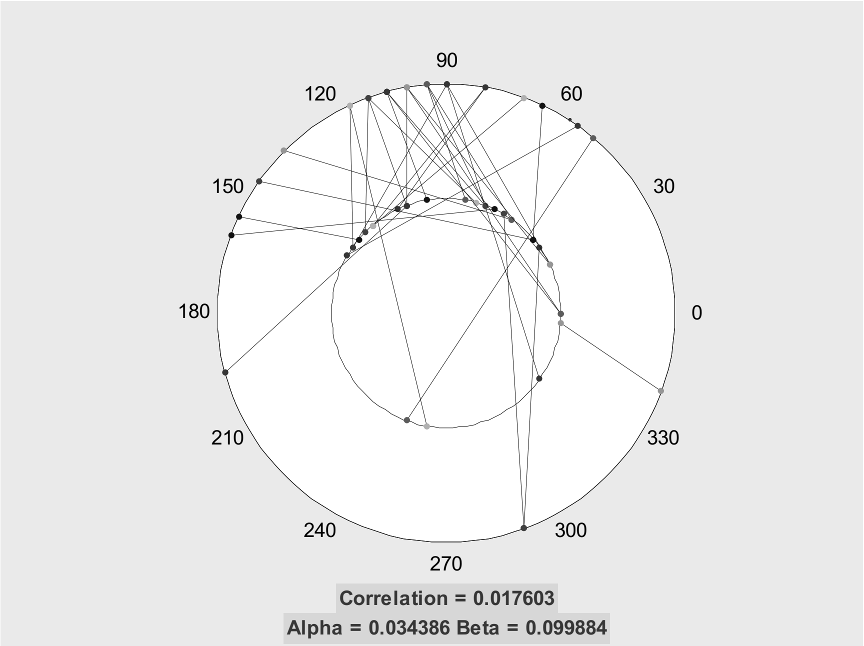

Figure 2. Spoke Plot of Wind direction data in January between KLIA

(time = 0000, pressure = 1000hpa) and KLIA (time = 0000, pressure = 500hpa).

From the Spoke plot in Figure 2, it can be seen again a number of lines crosses the inner ring indicates no correlation between the variables. To support the finding, the calculated correlation value is 0.0176 which implies no correlation. On the linear association, there seems to be evidence of a small one to one relationship.

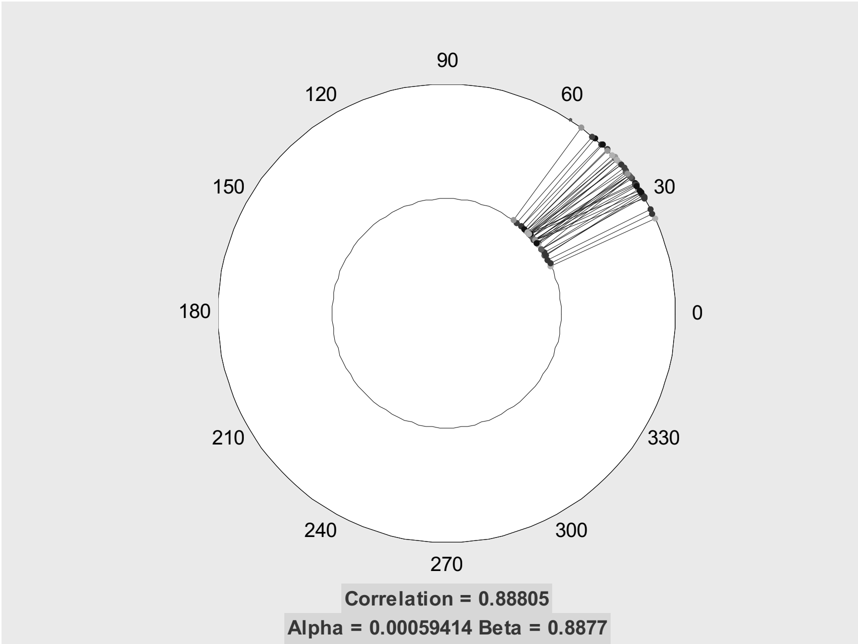

Figure 3. Spoke Plot of Wind direction data recorded at Telecommunication tower,

Seberang Jaya in April 2002 between height 45.72m and height 75.28m.

From the Spoke plot in Figure 3, it can be seen that none of the line crosses the inner ring and this suggests the presence of a strong correlation between the two variables. To support the finding, the calculated correlation value of Equation (1) is 0.8880 which indicates a strong correlation. On the linear association, there seems to be some evidence of a strong one to one relationship.

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A Text Image Segmentation Method Based on Spectral Clustering

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Abstract

We present a novel approach for solving the text segmentation problem in natural scene images. The proposed algorithm uses the normalized graph cut(Ncut) as the measure for spectral clustering, and the weighted matrices used in evaluating the graph cuts are based on the gray levels of an image, rather than the commonly used image pixels. Thus, the proposed algorithm requires much smaller spatial costs and much lower computation complexity. Experiments show the superior performance of the proposed method compared to the typical thresholding algorithms.

Keywords: Text segmentation, Graph cut, Spectral clustering

1. Introduction

Images generally contain rich messages from textual information, such as street name, construction identification, public transport stops and a variety of signal boards. The textual information assists the understanding the essential content of the images. If computers can automatically recognize the textual information from an image, it will be highly valuable to improve the existing technology in image and video retrieval from high-level semantics (Lienhart, 2002, pp.256-268). For instance, road signs and construction identification in a natural environment can be captured into images by cameras and the textual information will be detected, segmented, and recognized automatically by machines. These messages then can be synchronized as human voice to be used as instructions for visually impaired person. In addition to the example, textual information extraction plays a major role in images retrieval based on contents, cars auto-drive, vehicle plate recognition and automatics.

In general, automatic textual extraction consists of text detection, localization, binarization and recognition etc. In a natural scene texts could have different backgrounds and characters in the text message can also have variety of forms.

And, existing OCR (Optical Character Recognition)engine can only deal with printed characters against clean backgrounds and can not handle characters embedded in shaded, textured or complex backgrounds. So that characters are separated from the text in the detected region accurately is very necessary. Currently, many researchers have done a lot of work in the text detection and a lot of methods of text detection and location have been proposed. (Mariano, 2000; D. Chen,2004; Zhong,2000; X.L Chen,2004;X. Chen,2004) Compared to the text detection in natural scenes, specialized study of the characters extraction from natural environment is not more. The purpose of this paper is to extract accurate binary characters from the localize text regions so that the traditional OCR can work directly.

Most of the existing approaches are to use thresholding for binarization either global thresholds(T. Tsai,2007; Pan,2007)or local thresholds(Lienhart,2002;Wu,1999). T.Tsai(2007,pp.113-116) adopt a thresholding method suitable for segmenting the potential videotext character and modified seed-fill algorithm to extract the videotext. In Pan's(2007,pp.412-416), a simple global threshold achieved by using Otsu's(Otsu,1979,pp.62-66) thresholding technique. Lienhart(2002,pp.256-268) performed the binarization using the intensity value halfway between the intensity of the text colors and the background color as a threshold. Wu(1999,pp.1224-1229) proposed a simple histogram-based algorithm to automatically find the threshold value for each text region, making the text segmentation process more efficient. Due to its simplicity and efficiency, thresholding is a widely used method for solving this problem. But, it could not handle the cases when backgrounds have the similar color or intensity to that of the text strokes. Meanwhile, besides the changing backgrounds, texts are also changing slightly due to edge blur, image quality degrading due to video compression.

Spectral clustering has gradually gained attention from research on text classification, images segmentation and information retrieval (Shi,2000; Tao,2007). In Tao's (2007,110-118) the proposed algorithm uses the normalized graph cut(Shi,2000,pp.888-905) measure as the thresholding principle to distinguish an object from the background and a large number of examples are presented to show the superior performance of the method. But it is still a thresholding algorithm that has limitations to deal with the scene text. In this paper, we propose a new text segmentation method based on spectral clustering. In our approach, the histogram of intensity is used for the object of grouping, we partition the image into two parts using the gray levels of an image rather than the image pixels. For most images, the number of gray levels is much smaller than the number of pixels. Therefore, the proposed algorithm occupies much smaller storage space and requires much lower computational costs and implementation complexity than other similar algorithms.

The rest of the paper is organized as follows. Section 2 introduces the theory of spectral graph partition briefly. Section 3 presents the individual steps of our approach. The experimental results are given in section 4. Finally, section 5 concludes the paper.

2. Theory of Spectral Graph

The basic method used by image segmentation is to view an image as a weighted undirected graph $G=(V,E)$, where the nodes of the graph are the points in the feature space, and an edge is formed between every pair of nodes. The weight on each edge, $w(i,j)$, is a function of the similarity between nodes *i* and *j*. A graph $G=(V,E)$ can be partitioned into two disjoint subsets A and B, subject to $A \cap B = \phi, A \cup B = V$, by simply removing edges connecting the two parts. The degree of dissimilarity between these two pieces can be computed as total weight of edges that have been removed. In graph theoretic language, it is called the cut(Shi,2000,pp.888-905) :

$$cut(A,B) = \sum_{u \in A, v \in B} w(u,v) \tag{1}$$

The optimal bipartitioning of a graph is the one that minimizes this cut value. There are many criterions to measure the quality of the final partition results. Then the *Normalized Cut* value of a bipartition result can be defined as follows (Shi,2000,pp.888-905) :

$$Ncut(A,B) = \frac{cut(A,B)}{assoc(A,V)} + \frac{cut(A,B)}{assoc(B,V)} \tag{2}$$

where $assoc(A,V) = \sum_{u \in A, v \in V} w(u,v)$, $assoc(B,V) = \sum_{u \in B, v \in V} w(u,v)$ respectively, is the total connection from nodes in A or B to all nodes in the graph. And now, the minimal *Ncut* value is just corresponding to the optimal bipartition of the graph. In order to minimize (2), we can transform the optimization problem into solving the eigenvalue system,

$$D^{-\frac{1}{2}}(D-W)D^{-\frac{1}{2}} = \lambda z \tag{3}$$

where $D_{ii} = \sum_{j \in V} w(i,j)$,W is a symmetric matrix with size of $N \times N$, λ is the eigenvalue and z is the corresponding eigenvector. Shi and Malik(2000,pp.888-905) have proved that the second smallest eigenvector of the eigensystem (3) is the real value solution to the normalized cut problem of (2).

When the size of an image is too big, it is difficult to solve the above eigensystem, especially if the affinity matrix W is constructed by taking each pixel as a node, the size of eigensystem would be $N \times N$ (N is the total number of pixels in an image).

3. Our method

Suppose

$V = \{(i,j): i = 0,1,\cdots,n_h - 1; j = 0,1,\cdots,n_w - 1\}$, $H = \{H_0, H_1,\cdots,H_L\}$, $LL = \{0,1,\cdots,L\}$, where n_h and n_w is the height and the width of the image, respectively. H represents the histogram of gray, $f(x,y)$ is the gray value of position (x,y) .Then, V,H and $f(x,y)$ satisfy the following formulas.

$$(x,y) \in H_l, l \in \{0,1,\cdots L\}, \qquad \forall (x,y) \in V \tag{4}$$

$$H_l = \{(x,y): f(x,y) = l, (x,y) \in V\}, l \in LL \tag{5}$$

$$\bigcup_{l=0}^{L} H_l = V, H_i \cap H_j = \varnothing, i \neq j, \; i,j \in LL \tag{6}$$

Using just the intensity value of the pixels and their spatial location, we can define the graph edge weight connecting the two nodes i and j as:

$$w_{ij} = e^{\frac{-\|F(i)-F(j)\|_2^2}{\sigma_I}} * \begin{cases} e^{\frac{-\|X(i)-X(j)\|_2^2}{\sigma_X}}, & if\ \|X(i)-X(j)\|_2 < r \\ 0, & otherwise. \end{cases} \tag{7}$$

where $F(i)$ is a feature vector based on intensity of node i ,and $X(i)$ is the spatial location at that node, σ_I and σ_X are scale factors used to adjust the variation of gray or spatial location between nodes, r is used to decide the number of nodes from node i to j .

And then, we can get a bipartition $V = \{A,B\}$ corresponding to the graph $G = (V,E)$, where

$A = \bigcup_{k \in L_A} H_k$, $B = \bigcup_{k \in L_B} H_k$, and $L_A \cap L_B = \varnothing, L_A \cup L_B = LL$. Let $cut(H_i, H_j) = \sum_{u \in H_i, v \in H_j} w(u,v)$ be the total connection weights from nodes in H_i with gray level i to all nodes in H_j with gray level j , we can rewrite the above formulas as:

$$cut(A,B) = \sum_{i \in L_A} \sum_{j \in L_B} cut(H_i, H_j) \tag{8}$$

$$asso(A,A) = \sum_{i \in L_A} \sum_{j \in L_A} cut(H_i, H_j) \tag{9}$$

$$asso(B,B) = \sum_{i \in L_B} \sum_{j \in L_B} cut(H_i, H_j) \tag{10}$$

Since $asso(A,V) = asso(A,A) + cut(A,B)$, $asso(B,V) = asso(B,B) + cut(A,B)$, we can rewrite (2) as:

$$Ncut(A,B) = \frac{cut(A,B)}{asso(A,A) + cut(A,B)} + \frac{cut(A,B)}{asso(B,B) + cut(A,B)} \tag{11}$$

Given an image, we can construct a histogram-based matrix M by computing the all weights of nodes in the corresponding graph. $M = [m_{i,j}]$ is an $L \times L$ symmetrical matrix with $m_{i,j} = cut(H_i, H_j)$ and $m_{i,j} = m_{j,i}$,where L is the number of gray level of histogram. Now, let M be the affinity matrix, we can get a complete approach of image segmentation using spectral clustering (Shi,2000,pp.888-905) . Figure 1 shows the workflow.

Note that the size of the affinity matrix M depends on the number of gray-level L , rather than the number of all pixels N in an image. Meanwhile, the size of eigensystem to solve is $L \times L$, rather than $N \times N$, and usually, L with a fixed size is much smaller than N . Hence, the complexity of computation and spatial cost reduce greatly.

4. Experimental Results

We perform a series of experiments to test the performance of this method. The samples used are gray images with the text of characters, and they are the real images from the natural environment. For illumination and other reasons, there is a clear gray difference of pixels from the same region in many images. In the following experiments the parameter settings in formula (7) are $\sigma_I = 50$, $\sigma_X = 5$, $r = 5$, $L = 100$. Our method is compared with two other methods: the *Otsu* thresholding method (Otsu,1979, pp.62-66) and the *Ncut-based* thresholding method (Tao,2007,pp.110-118). We choose them because the *Otsu* method is a simple but classic solution employed by many text segmentation schemes, while the latter is an *Ncut-based* but thresholding solution proposed recently. The aim of the three algorithms is to separate the "foreground (texts)" from the "background (non-texts)".

We present the detailed results in Figure 2~Figure 4. By the way, the actual images are much greater than that in the paper. The images are easy to separate relatively in Figure 2, we can see the latter two methods both based on *Ncut* criterion are superior or close to the *Otsu* method from the experimental results. In Figure 3, the proposed method can be the right segmentation at a reflective white spots within the part of black spots on the letter '*B*', which is difficult to achieve for the conventional thresholding methods.

It is hard to segment the images in Figure 4 because the foreground of them is not clear enough. The segmentation results of our approach look better than the others'. From the above experiments, we can see the proposed method is far superior to *Otsu* method, also to the thresholding method based on *Ncut*.

Next, we choose 150 images with clear differences and 50 images with blur text respectively to test the three methods. The precision p and recall r are used to evaluate the methods, which are defined as follows.

$$p = \frac{com\_\mathrm{Im}age_O \cap base\_\mathrm{Im}age_O}{com\_\mathrm{Im}age_o} \tag{12}$$

$$r = \frac{com\_\mathrm{Im}age_O \cap base\_\mathrm{Im}age_O}{base\_\mathrm{Im}age_O} \tag{13}$$

where $com\_\mathrm{Im}age_O$ is a points set of objects including texts; $base\_\mathrm{Im}age_O$ is the ground-truth text region. And then, the comprehensive assessment indicator f is defined as follows (Lucas,2003,pp.682-687):

$$f = \frac{1}{a/p + (1-a)/r} \tag{14}$$

where a is a relative rate between precision and recall, letting $a = 0.5$.

The result of 150 images shows in table 1, and the result of 50 images shows in table 2. From the results, we can see the proposed method has a good performance for both the two kinds of images. And our method is superior to other two methods from the point of comprehensive indicator f .

5. Conclusion

Accurate retrieval of textual information from images that exist in the real environment is critical to understand images. The key part of the research is to retrieve the characters from the text image area. And because of the complexity of backgrounds that text built in, the conventional thresholding methods often cannot separate the characters from natural backgrounds effectively. Spectral clustering can resolve the issue by using spectral graph theory. And this method controls the complexity of algorithm effectively by changing the clustering objects from pixels to gray levels. The experiment results have proved its superiority to the traditional thresholding method.

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Table 1. The segmentation result of three methods for normal images

Method	p	r	f
Ostu method	0.8089	0.8361	0.8223
Ncut-based thresholding method	0.8364	0.8011	0.8184
Our method	0.8058	0.8802	0.8413

Table 2. The segmentation result of three methods for abnormal images

Method	p	r	f
Ostu method	0.5307	0.7734	0.6295
Ncut-based thresholding method	0.5815	0.8316	0.6844
Our method	0.6842	0.8015	0.7382

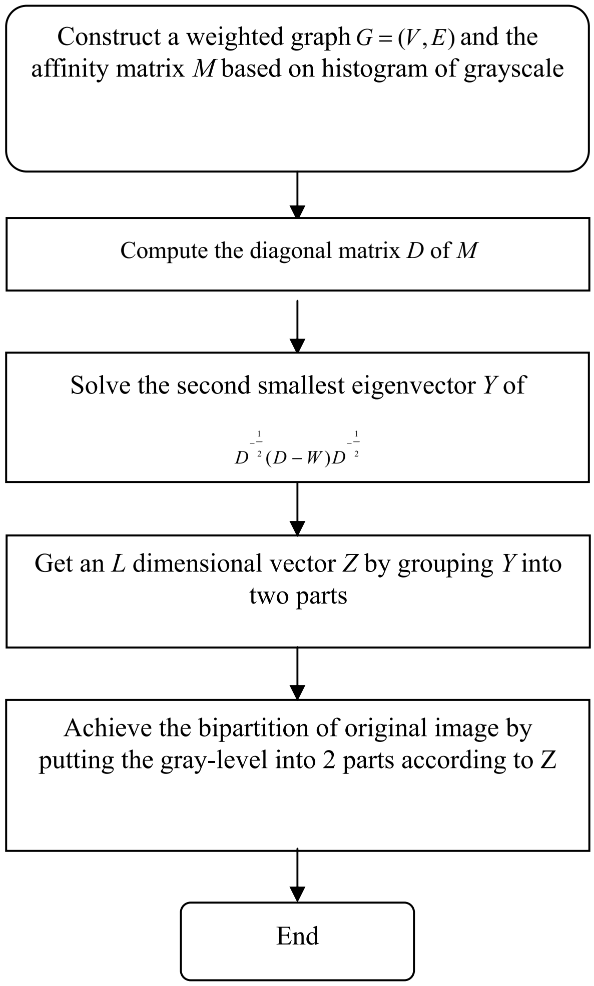
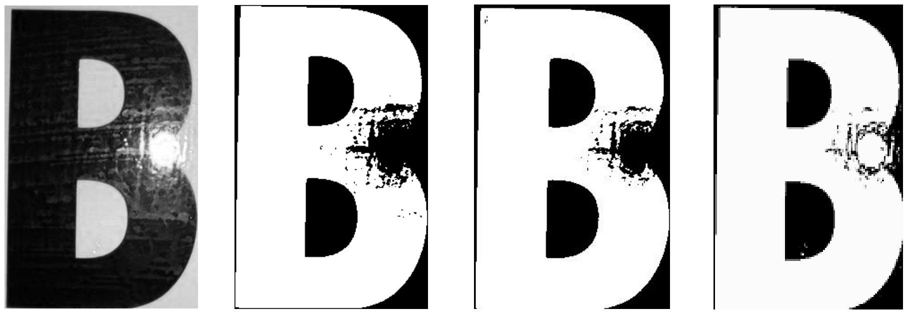


Figure 1. Workflow of Image Segmentation Based on Spectral Clustering

Figure 2. Comparison of Three Text Segmentation Methods for Normal Images



(a) (b) (c) (d)

Figure 3. Comparison of Three Text Segmentation Methods for an Illuminated Image ((a)Original image (b) Ostu result (c) Ncut-based result (d) Our result)

Original image

Ostu result

Ncut-based reslut

Our result

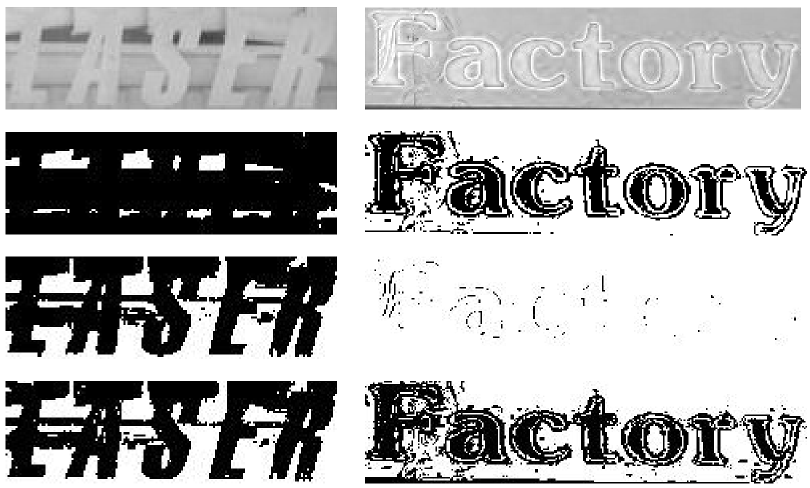


Figure 4. Comparison of Three text Segmentation Methods for Abnormal Images

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Analysis of Telephone System of a University Campus and Design of a Converged VoIP System

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Abstract

VoIP(Voice over IP), so called Internet phone, provides several unique advantages, for instance, inexpensive long distance call service than the existing wire telephone network and various multimedia services from Internet network added to voice call service. This has been widely discussed and has been a research topic throughout the developed countries in the recent years. The paper is intended to describe and analyze the feasibility of the results of the VoIP system designed if implemented in a campus network like Dhaka University telephone network. Thus the convergence strategy of an IP PABX system with the existing wire telephone network has been discussed. While taking this into consideration, upgrading method of the existing system using modern network appliances has been described and at the same time new network topology connecting the present IP network with the telephone network has been designed.

Keywords: VoIP, IP telephony / PABX system, Network convergence, Dhaka University

1. Introduction

Voice over Internet Protocol (VoIP) is a multifaceted topic. There is no single template that can be followed when contemplating the use of this technology to supplant a traditional phone system. The technology itself is fairly generic and somewhat mature, however the impact to the external environment, business conditions surrounding the actual requirements, capital funding plan and other integral factors complicate the analysis. Implementing VoIP at Dhaka University, is much the same as elsewhere except for unique characteristics endemic to the University. It is however, these unique characteristics that warrant a closer look since capital investment will be significant and once a solution is installed, it will most likely remain long beyond its' intended useful life.

There has been much discussion about VoIP phone systems within commercial and public environments. *"Although progressing rapidly, Internet telephony still has some problems with reliability and sound quality, due primarily to limitations bothin Internet bandwidth and current compression technology".* Some companies have adopted use of VoIP systems as a replacement for traditional phone systems, Public Branch eXchanges (PBXs) and Centrex type services. Most of this however, is on private intranets where bandwidth is available and predictable. Many larger organizations have adopted hybrid solutions due to geographic vagaries, economics or administrative manageability.

Within the confines of the University, there are several factors that differentiate this

environment from that of a corporate one. Although these don't affect the technology trend or the economics associated with VoIP deployment, they do influence the nature and type of deployment campus wide. This document highlights many key areas that need to be addressed when considering large-scale deployment of VoIP as well as some special considerations relative to the University.

The results are not surprising when all influences are considered. If we analyze the business need and then factor in the appropriate variables, the strategy seems straightforward. Some of the variables to consider are cost, the existing infrastructure, implementation practicality, maintenance, operability and the University topology.

2. Background

2.1 Telephone System

In order to get a telephone call to travel from one place to another, it must pass through the telephone network. This network consists of many different parts, operated by many different companies, but are inter-connected using common signaling methods. Physical components required for telephone networks (figure-1.1)are: Transmission Facilities, Local Loop, IOF - Interoffice facilities, Switching Systems, Customer Premise Equipment (CPE).

Central Offices, signaling between different telephone systems, different methods of transmission, and the use of tandems (transfer) in the network are discussed below:

On each telephone call, a talking path must be set up between the calling and the called telephone. The method of making this connection, known as switching, has progressed from the simplest of hand operated switches through the more complex manual systems to the present switching systems. In telephone switching systems, phone calls are "switched" meaning they cross through a switching matrix to route calls from an origination point to a destination point.

Local "central offices" are where the end users are located. This is the most common switching system in the telephone network. Long Distance "tandem" switches are where long-haul long distance calls are switched to connect local central office switches throughout the world.

PSTN (Public Switched Telephone Network) is the traditional telephone network that provides POTS (Plain Old Telephone Service); that is, the network that anyone could access by circuit- switch connection. This connection is the dedicated service that can guarantee the reliable, accessible and quality. PSTN converts voice into electrical signals and transmitted through the circuit-switched network.

2.2 Existing PABX System of Dhaka University Campus

Currently, Dhaka University is using Mitel SX-2000 LIGHT PABX system The Mitel SX-2000 LIGHT system is an advanced, fiber-distributed telephone system that is designed for larger organizations or for networked telecommunications environments. The distributed architecture separates the control node from peripheral, application, and network access nodes and links them by multimode fiber optic cable.

The SX-2000 LIGHT system can be configured as a multi-cabinet, control redundant system.

The redundant main control cabinet can support up to eleven expanded peripheral nodes located up to 8.7 miles (14 kilometers) away. The redundant main control cabinet also supports up to five DSU cabinets or Network Services Units (NSU). So, this installation is used in Dhaka University campus network.

The SX-2000 LIGHT system (figure 1) consists of a redundant main control cabinet and associated peripheral cabinets. Fiber optic cables connect the peripheral cabinets to the main control cabinet. Copper cables from the extensions terminate at the peripheral cabinet. Installers do not have to route the extension cables between many floors or run the cables off-premises to a centrally-located system.

Depending on the Fiber Interface Module (FIM) that is used, the peripheral cabinets can be located up to 0.62 miles (1 km), 1.9 miles (3 km), or 8.7 miles (14 km) from the main control cabinet. This versatility allows system resources such as lines, trunks, and digital service applications to be physically distributed among several remote locations.

The topology consists of the local loop, circuit-switched telephone network with the central node being the local central office. In this topology, 11 peripheral nodes have a dedicated point-to-point link to a central node. And another 4 peripheral nodes are extended through extenders with 4 of the main peripheral nodes. If one node wants to send data to another, it sends to the central node, which then relays the data to the destination node.

In this network, optical fibers are used to connect the main 11 peripheral nodes to the control node and copper wires have been implemented for all the other connections.

The positions of the nodes and the distribution boxes are given below:

1. Control node : 1 – Administrative Building
2. Peripheral nodes : Total – 15

a. Business Studies –3 nodes

2 distribution boxes

b. ISWR – 1 node

1 distribution box

c. Curzon Hall – 2 nodes

1 distribution box

d. Khondokar Mokarram Hussain Building – 2 nodes

1 distribution box

e. Arts Building – 2 nodes

1 distribution box

f. Register Building – 4 nodes

3 distribution boxes

g. Science Annex – 1 node

1 distribution box

The schematic diagram of the campus PABX system topology is given in figure 2.

3. Literature Review

3.1 What is IP PABX?

Over time, the PABX has grown to incorporate all sorts of advanced features such as voicemail, unified messaging, auto attendant (IVR), automatic call distribution (ACD), call queuing, branch office support, telecommuters, softphones, CTI (integration with the PC), and more. With the advent of IP, the acronym PBX morphed into its latest incarnation, the IP PABX. An IP PABX is a PABX that supports packet-based transport protocols - commonly referred to as "VoIP". The most popular current protocol is SIP, which stands for "Session Initiation Protocol". Then, as the IP PABX began to rise in market share an even new label appeared called the "Hybrid IP PABX" A Hybrid IP PABX is an IP PABX that, along with VoIP, also supports legacy and analog switching protocols such as TDM. A Hybrid IP PABX will typically also support analog phones to complement its support of IP Phones.

That is, with VoIP, a PABX is still used to distribute calls throughout an organization, but the PABX must support the Internet Protocol being used. In actual practice, such an IP PABX is a computer server. Once received by the IP PABX and inside the organization, the call may be carried through a data network or via digital or analog communication lines to recipient endpoints. If calls are distributed inside the organization on a data network, the need for a dedicated communications circuit is eliminated and the calls can be carried on the internal LAN as data packets.

3.2 What is VoIP?

Voice over Internet Protocol (VoIP) is a new way of communicating. It requires the use of the internet and the technology that allows information to travel between users. Before I delve into how VoIP works it is important to understand how the plain old telephone system (POTS) works.When you make a telephone call your voice travels in its analog form from your telephone to a telephone company switch. At the switch it is determined if your call is passed to another switch or if it can be routed to its destination. Using the POTS system your phone call travels the same path along physical wires based upon where the phone call will terminate.

VoIP requires an individual to have broadband Internet access. The most common of which is either through the use of a cable modem or DSL (direct subscriber line). Data travels the Internet in packets. In order for your voice to travel the Internet, it must be converted from analog to digital. Once your voice has been converted to digital packets it can be transferred via the Internet. But before your voice can enter the Internet and travel to its destination some information has to be added to it so that it meets IP (Internet Protocol) standards.

It mentioned earlier that data travels in packets, that is your voice is broken up into several different packets that must reach your destination and be reassembled in the right order and converted from its digital form back to an analog form so that the recipient can hear your voice. So each packet of data, which contains a portion of your phone call, has a minimum of 160 bits added to it so it can reach its destination [30]. In addition to the IP protocol it also requires the use of real-time protocol (RTP). "It provides timing information that allows the receiver to reconstruct the original timing of the transmitted material in a way that identifies the content being sent, provides security, and notifies the overriding application of lost data." RTP is usually used in conjunction with unreliable datagram protocol (UDP). UDP allows for information such as voice and video to be sent without waiting for acknowledgement of it being received. "It is useful in cases where one sender wants to send the same information to multiple receivers and is not too worried if some pieces get lost along the way." VoIP requires the use of these three protocols in order to function. For this reason broadband access is needed. The fastest modem dial up connection using POTS can only achieve 53Kbps (kilobits per second) whereas cable modems and DSL can achieve speeds of 10 – 100 Mbps (megabits per second). The higher speed is needed in order for an actual conversation to take place over the Internet without significant delay.

3.3 What are the technology requirements?

To get started with VoIP service there are a few basic items you need: Broadband Internet access, computer and software if you plan to use the computer as your phone, adapter box if you plan on using your regular house phone. However, for a call center there are more things to consider.

You still need broadband Internet access but supporting hardware becomes more complex. First, a decision is need to determine how much bandwidth is required to support adequate VoIP telephone service. Then you need to determine the amount of bandwidth you have on your Internet access. Then you can determine how many phones can be supported by a single broadband connection. In order to share the broadband connection an intelligent router will be needed to split the connection between all of the phones.

Westminster college in Salt Lake City, Utah changed their phone system over to VoIP. Based upon their article, Implementing campus-wide voice over Internet protocol (VoIP) phone systems at small colleges, they installed servers, switches, routers, phone hardware and power switches. They upgraded to a 1-gigabit backbone for Internet. They set over 1000 phones on this system.

Washington University in St.Louis, changed their phone system over to VoIP. Based upon their article, Designing VOIP in Campus Network, they installed servers, switches and routers had been recommended to use instead of hub and fiber optic instead of copper wire in backbone network. In the backbone network, Gigabit Ethernet had been implemented and mostly Fast Ethernet for other networks. On one link, since 80 Mbps maximum capacity was used at that time, another 12,166 lines could be used in Campus with G.729.

Rutgers, The State University of New Jersey, VoIP technology was implemented. Their environment was distributed and decentralized which subsequently fostered the growth of various systems. Each department or school within the University was given money from the Central Administration for telephone services. In some cases the department or school used those funds to purchase a more state-of-art system that enables them to pool telephone numbers thus reducing their monthly expenditures. However, this approach had resulted in over 350 different types of telephone systems distributed across the University. This model was becoming almost impossible to manage given the many different types of systems, voice mail, system software, etc.

3.4 Quality of Service

Quality of service (QoS) is something that can be different for each and every individual. For some quality can refer to how well the voice sounds over the phone, for others it may refer to the amount of noise they hear in the background, echoes, reaching an individual are all issues that go into quality of service. Another side of quality can refer to security of a conversation as well reliability. A study in India showed that the majority of people preferred the lower cost of service over responsiveness, value added services, reliability and voice quality.

Security or privacy of phone calls is another issue for QoS. This becomes exceptionally important for law enforcement officials. There are many differences between security of measures of public switched telephone networks (PSTN) and VoIP. Sicker and Lookabaugh [28]in their paper titled VoIP Security: Not an Afterthought used table 1 to show the differences between security measures.

3.5 Comparison of Costs

Which type of phone service is cheaper? PSTN or VoIP. Costs are more involved than a simple phone bill at the end of the month. Costs include hardware requirements, training costs, switch over costs, potential and loss of business in transition. Different companies will have different costs for telephone service based upon whether they are working on the international, national or local level. The company CISCO provides a lot of hardware for VoIP phone service and claims on their web site that companies have saved millions of dollars by using their technology.

4. Methodology

4.1 Analysis of Telephone System

Present telephone system in University of Dhaka, Mitel SX-2000 LIGHT system is analyzed by it's control cabinet, peripheral cabinet, links and power system as follows:

4.1.1 Control Cabinets

The Control Redundant SX-2000 LIGHT system supports applications that require up to 3000 lines. The redundant main control cabinet provides full back-up, including independent power supplies, so that system operation will not be affected if a main control component fails. When the system switches to the alternate main control, calls in progress are not dropped and callers are unaware of the system event. The redundant main control cabinet of this network has supported eleven expanded peripheral nodes.

Control Cards used in control cabinet

Main Controller Card

The standard Main Controller card for the SX-2000 LIGHT is the MC III E. In many features of this card, it is more important that it provides circuit switch matrix to establish voice and data paths from one peripheral device to another.

Circuit Switch Matrix Card

The Circuit Switch Matrix card is required for the control redundant SX-2000 LIGHT system. The Circuit Switch Matrix card increases the Main Controller card's circuit switch matrix size from 24 X 24 circuit-switched links to a 48 X 48 non-blocking link matrix.

Control Resource Card (CRC)

The Control Resource card (CRC) provides additional circuitry in the control node to support distributed system architecture. The CRC supports functionality between the Fiber Interface Modules and the Main Controller card. The CRC is non-redundant in all configurations; if power is removed from a CRC the System Fail Transfer becomes active at all peripheral nodes.

4.1.2 Peripheral Cabinets

Each peripheral cabinet holds up to 12 Peripheral Interface Cards and provides up to 192 ONS or DNI ports. By purchasing the Peripheral Node Expansion feature package, a slave cabinet can be added that expands the node up to a total of 384 ports and 22 Peripheral Interface cards (the number of voice channels remains the same).

Control Cards Used in Peripheral cabinet

Peripheral Resource Card (PRC)

The Peripheral Resource card (PRC) provides miscellaneous circuitry for distributed systems. The PRC is installed in all peripheral nodes. The PRC provides:

- System Fail Transfer contact closure
- Single ended to balanced conversion of FIM to DSU signals
- Terminal port multiplexer.

Peripheral Switch Controller (PSC)

The Peripheral Switch Controller (PSC) card is installed in all peripheral nodes. The PSC card provides control for all Peripheral Interface cards, and a fiber optic cable connects the FIM to the main control.

Peripheral Interface Cards

Peripheral Interface cards join telephone trunks and peripheral devices (such as SUPERSET telephones) to the system. Peripheral interface cards include line cards and trunk cards.

Line Cards

Line cards connect to single line sets, SUPERSETs, attendant consoles, and DATASETs. They include

- **Digital Network Interface (DNI) Line Card --** supports music-on-hold and paging and interfaces with MITEL digital network devices. (including SUPERSET telephones, attendant consoles, and DATASETs). The DNI line card provides 16 voice and data lines and has 16 circuits.
- **On-Premise (ONS) Line Card --** has 16 circuits that connect up to 16 standard telephones with line loop resistance usually not exceeding 400 ohms. It also supports modems and fax machines.

Trunk Cards

The system can connect to the public switched network or to private networks over both digital and analog trunks. Trunk cards provide an interface from the system to the public switched network and leased lines. Trunk card which is used in this system is:

- **Loop Start/Ground Start (LS/GS) trunk card --** interfaces to the analog LS/GS Central Office (CO) trunks, and is used to terminate eight CO trunks (non-dial-in trunks).

4.1.3 Fiber Interface Module

The FIM connects the control node to a peripheral unit. At the transmitting end, the FIM converts electrical signals into pulses of light to be transmitted over the cable. At the receiving end, the FIM converts the pulses of light back into electrical signals usable by the node.

Links

- single mode and for short distance multimode fiber is used between nodes
- 6/12 core fiber optic cable is use
- max. loss – 6 dB using 62.5/125 µm cable with N.A.-0.275 inch

4.1.4 Power System

Two redundant power modules in the redundant control node, one power distribution unit (PDU) in each peripheral cabinet (AC or DC) and one power converter in each peripheral cabinet (AC or DC) are used .In a DC powered peripheral ,the -48 V power is used directly. 21/19 plate battery backup is given in parallel, to keep the network active when the power system of any node fails.

4.1.5 Users handling capability of this network

From the brief description of Dhaka University telephone network the total output lines which can operate different types of peripherals, can be easily obtained as follow:

No. of peripheral nodes: 11

No. of slots in each peripheral node: 12

No. of circuits in each ONS line card: 16

So, the total voice and data lines for 11 peripheral nodes: 11*12*16 = 2112

Through extender these 11 peripheral nodes can be extended up to 22. That way, this network has the capability of handling 22*12*16 = 4224 voice and data lines. But this network is using 2100 voice and

data lines and the no. of trunk lines= 180 i.e., the no. of users of these network is- 2280.

4.2 Convergence of Existing PABX System & IP PABX System

At Mitel Networks convergence of two networks is possible by using Mitel networks 3300 Integrated Communication platform. The 3300 ICP is a resilient network appliance that adds feature rich IP telephony and advanced user applications to the corporate LAN/WAN. The 3300 ICP offers the ability to migrate from an existing SX-2000 LIGHT PBX system to an IP PBX system. The database is converted and restored onto the 3300 ICP by using the Mitel Networks 3300 Configuration Tool. Mitel Networks' architecture uses the IP network for connecting IP telephony devices and provides a supplementary TDM (Time Division Multiplexing) subsystem for switching calls between traditional telephone devices. The 3300 ICP has the advantage of being able to optimally switch all types of traffic, IP or TDM.

4.3 VoIP Perspective

VoIP is the routing of voice conversation over the internet. As packet switched networks were designed to carry data, while carrying voice, calls might experience delays and distortion. While making a regular call, subscribers are charged against distance and duration of the call, in this case these factors do not matter, rather a low fixed rate price is charged for the internet bandwidth. A simple model of VoIP is given in figure 3.

4.4 Upgrading of SX 2000 LIGHT

To upgrade the SX 2000 LIGHT, two 3300 MXe Expander is used to replace the control cabinet. The suggestive system topology of Dhaka University campus is shown in figure-4.2

4.5 Planning and Designing VoIP

The issues, related to designing an IP telephony or voice over IP (VoIP) network for transporting voice and data over a common LAN or WAN infrastructure are covered in this section. Understanding the underlying technology used to transport voice traffic is important in designing an IP telephony network. Design principles used to deploy a successful LAN-based VoIP network will not necessarily work when you apply them to a WAN configuration. This document discusses the major hurdles that need to be addressed when designing either a LAN or WAN based VoIP network.

4.5.1 Bandwidth Management

Where bandwidth is at a premium voice compression is a requirement and now widely available voice coding algorithms/compressors are G series codec. Each voice codec has a benchmark score (MOS) based on- speed of the conversion, speech quality, and data loss characteristics. To get the maximum MOS value and to achieve the best quality for voice traffic G.729 is chosen. Table 2 shows standard Codec Compression for G.729.

The capacity calculation using G.729 series is given below:

Header = Ethernet Header(18)+IP Header(20)+UDP(8)+RTP(12)

= 58 bytes

Datagram Payload Size :

$$\frac{\text{Codec Speed(bits/sec)*Datagram Delay(ms)}}{\text{(8 bits/byte)*1000(ms/sec)}}$$

= 8000*20/(8*1000) = 20 bytes

Packet Size = 78 bytes

Overall Capacity =(1/20ms)*78*8 = 31.2kbps

Using G.729 encoding series on a 100 Mbps Ethernet network, each voice call takes up to 31,2 kbps

in each direction supporting up to 3205 calls on full duplex link. On a Gigabit backbone, up to 32050 simultaneous calls can be handled. The 2 MXe expander ICP 3300 controller can support upto 2800 IP phones which will need the capacity of 31.2 * 2800 = 87.36 Mbps.

4.5.2 QoS Design

The major factors to tune QoS – Congestion control, Reliability , Throughput, Delay , Jitter

To ensure all the factors -

- Traffic conditioning is mandatory.
- Differentiated Services (DiffServ) is to be provided.
- Delay guideline is to be followed

Table 3 shows the mechanisms for traffic conditioning and the respective network effects.

4.5.3 Delay Guide Line

To make better quality following conditions are must be fulfilled:

- End to End Delay: < 150 ms
- Jitter: < 40 ms
- Lost Packet :< = 0.5 %
- reduce the -TCP/IP window size and MTU size
- Finally - the settings of the QoS on the router is done by configuring it accordingly

4.5.4 Queuing

In case of that there are so many packet waiting the Router queue, WFQ (Weighted fair queuing), CBWFQ (Class-based weighted fair queuing), LLQ (Low-latency queuing), and WRED (Weighted random early detection) could be configured to the latency of voice packet. Figure 4 shows an example of CBWFQ router configuration.

4.5.5 Traffic Shapers

This method could be applied for the inter-network connection usually because the network link is limited by ISP (external link) (There are too many users required these connection in the same time) For example, in TCP connection, whenever the packet gets lost, retransmission process will be applied again and again. As a result of this, it may need to shape the connection in order not to waste the capacity for retransmission. Figure 5 shows the traffic shaping configuration example.

4.5.6 TCP/IP Tuning

- TCP window size: It is better to reduce the TCP/IP window size in busy network.
- MTU size and low latency: due to the smaller VOIP packet, it is better idea to set MTU path along the path to fit in the packet in slower-speed link Also, Routers have no need to pay for fragment delay.

Figure 7 shows MTU (maximum transfer unit) configuration example.

4.6 Topology Design

Topology is designed as it has a 100 Mbps Ethernet backbone with equipments- 7204 VXR router, 1900 series switches and links - Cat 5 cables, DSL, ADSL, optical fibers. The IP network topology is given in figure 8.

4.7 Convergence Strategy

The 3300 MXe Expander located at the Register Building and a 1900 switch located at the IIT has been connected. Since the distance between the two location is nearly 1 km, the interconnectivity by UTP cable directly not possible. So media converters are used which supports network extensions upto kilometer range. LH1706A-ST-US media converter uses fiber optic cable and can extend upto 2 km.

Manager and the application server are connected using their Ethernet ports via local switches using UTP cables. This topology consists of three parts : the telephone network, the IP network and their interconnection. The positions of the nodes and the distribution boxes are given below:

1. Control node- 3300 ICP Controllers

(MXe Expander) : 2 – Administrative Building

2. Peripheral nodes : Total – 15

a. Business Studies –3 nodes

2 distribution boxes

b. ISWR –1 node

1 distribution box

c. Curzon Hall – 2 nodes

1 distribution box

d. Khondokar Mokarram Hussain Building – 2 nodes

1 distribution box

e. Arts Building – 2 nodes

1 distribution box

f. Register Building – 4 nodes

3 distribution boxes

g. Science Annex – 1 node

1 distribution box

Figure 9 shows the IP PABX system topology i.e. the topology of the converged network.

5. Findings

➢**Cost effectiveness-**Subscribers are charged a low fixed rate price for the bandwidth, so long distance calls are very cheap.

➢**Capacity enhancements-**By connecting more ICP 3300 controllers, IP phones and computers, the capacity can be increased without further network congestion.

➢**Redundancy-**In fail over mode the users can use the analog line by pressing a programmed key, when the IP connection has failed.

➢**Plug and Work solution-**This solution enables access to the voice network of DU from any location with a broadband internet connection.

➢**Improved media integration-**IP phones can be enabled to add media to an ongoing call as required, e.g., viewing a picture or drawing on a whiteboard. Using workstations themselves as IP phones can facilitate providing this function, whereas the standards are not yet there for coupling traditional phones and workstations.

➢**New services-**As IP Telephony evolves, it can be used to provide new services (like user-defined call processing) or to integrate existing concepts, e.g., Presence, Location Awareness or Instant Messaging. Because of the open standards available for these services, they need not to be limited to vendor-specific solutions. In other words, it can be much easier to deal with issues such as CTI (Computer Telephony Integration) and so pave the way to a completely new way of understanding telephony.

➢**Research-**The protocols and standards used for IP Telephony are open and publicly available. This allows research institutions to work on their own services and solutions. It is important to point out that before introducing IP Telephony into the network of an organisation; several issues unknown to the old telephone system have to be taken into account. A rough, non-exhaustive list may include addressing (special subnet/VLAN for phones), Quality of Service (QoS), security, positioning of gateways, interfacing of firewalls and, last but not least, maintenance of the system (backups, spares, etc., - something not very common in the legacy PBX world).

6. Recommendations

At present Dhaka University telephone network can support upto 4224 telephone lines and the designed IP PABX system has a capacity of 2800 lines. And the value to an organization in unifying communications is to improve workflow and efficiency, ultimately leading to better service delivery and an improved bottom line. So, the ultimate aim of unifying communication can be the complete migration of the legacy telephony to IP telephony in the coming days.

In this network very high capacity fibers have been used whereas 2 or 3 core fibers are adequate for the system. The rest of these unused fibers (dark fibers) can be utilized by using them as links for other purposes. Such as a very high speed internet connection at low cost can be provided by connecting these fibers to the 2 layer switches of our network and a SEA-ME-WE cable.

Again, if wave division multiplexing is used in fiber optic communication, the transmission rate can be increased to 1.1 trillion bits per second (1.1 Tbs) over 150 km and 2.6 Tbs over 120 km using 132 different wavelengths in the interval 1529.03-1563.86 nm (European Conference on Optical Communication). Thus, in this case independent signals are carried by a single fiber multiplying the capacity of an individual fiber making it capable of carrying enormous rates of information.

7. Conclusion

This report is meant to serve as a catalyst for discussion relative to VoIP applicability and deployment at the University. It is not exhaustive in content but certainly inclusive of major considerations when contemplating an implementation. As with every new technology, there are caveats, special considerations and different skills required to manage them. Some of these, such as shared administration, power considerations, disaster recovery, repair and troubleshooting, localized bandwidth consumption and monitoring capabilities are quite significant in scope.

The difference in a straightforward or “vanilla” installation and one that might occur in a mixed assets environment is dramatic. Clearly the interoperability of all environments, the re-use of existing technology and the economics of deployment are the decision variables. VoIP is not free, or cost effective in all cases. It is most efficient in plain vanilla cases when traditional systems are displaced for a variety of reasons, including obsolescence or high maintenance cost. At the University, we are in the throes of trying to establish unified voice architecture, while leveraging viable assets so that we could achieve a technologically sound solution, which is also economically feasible. Hopefully this report will help spur useful discussion in our quest to deliver high quality, maintainable phone service throughout the University.

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Table 1. Differences between security measures

Security Concerns	**Wired PSTN Measures**	**VoIP Measures**
Confidentiality	Physical security	Encryption techniques
Integrity	Physical security	Encryption techniques
Availability	Physical access control	Network/Service access control
Authentication	Physical connectivity, voice recognition, caller ID	Login, password
Authorization	Caller ID, access control	Access control role-based authorization
User Expectation	Assumed and static	Variable
Implementation and Design Concerns		
Software design	Large, monolithic, complex	Variable, distributed, complex
Interoperability	Centralized and tested	Distributed and potentially ad hoc
Software implementation	Centralized and tested	Distributed and potentially ad hoc

Table 2. Standard Codec Compression for G.729

Codec	Data rate	Packetization delay	Jitter Buffer Delay	Datagram spacing	MOS	Total capacity required
G.729	8 kbps	25 ms	40ms(2)	20 ms	4.07	31.2kbps

Table 3. Traffic conditioning and mechanism

Traffic conditioner	**Mechanism**	**Network effect**
Marking	IP Precedence, DSCP, CoS	• Sets IP precedence/ DSCP • By apps, protocol, address, etc
Policing	CAR, Class Based	• Enforce a maximum transmission rate • Conform or exceed threshold
Scheduling	PQ, CQ, WFQ, LLQ, WRR, MDRR	• Bandwidth management: traffic priority • Set service sequence
Shaping	GTS, FRTS	• Conforms traffic to commited bandwidth • Interwork L2 notification, BECN
Drop	RED, WRED, Flow RED	• Avoid congestion by notifying source • Prioritize which traffic is told to reduce
Compress	CRTP	• Reduce the volume of traffic sent
Fragment	LFI, FRF.12	•Reduce delay on slower-speed links • Split, recombine larger frames

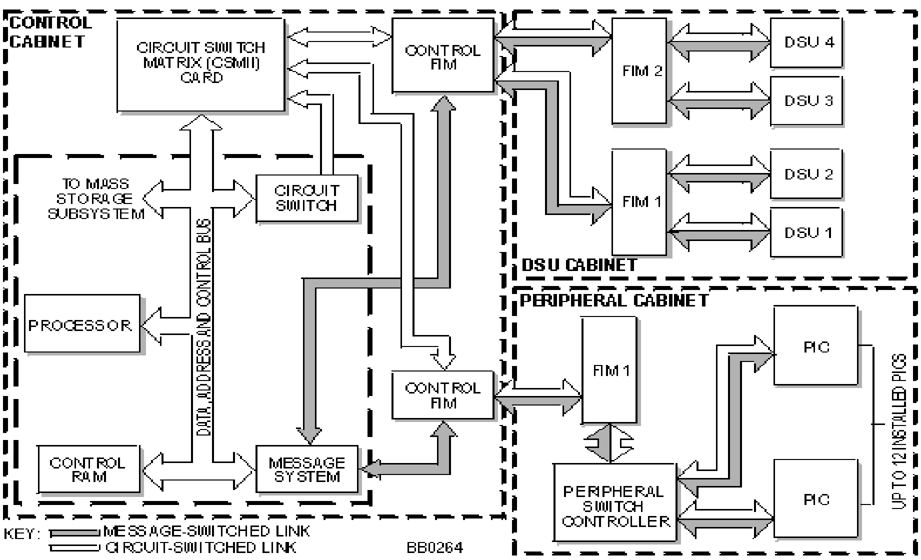

Figure 1. Basic System Architecture

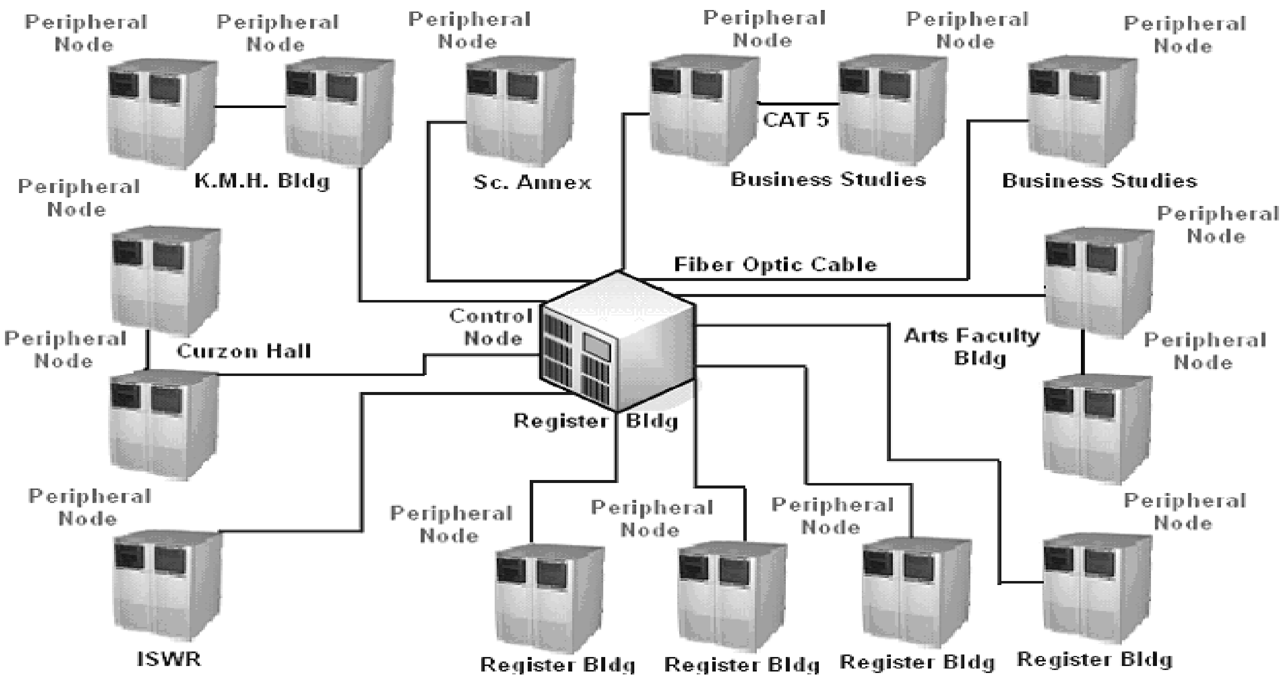

Figure 2. PABX system topology in Dhaka University campus

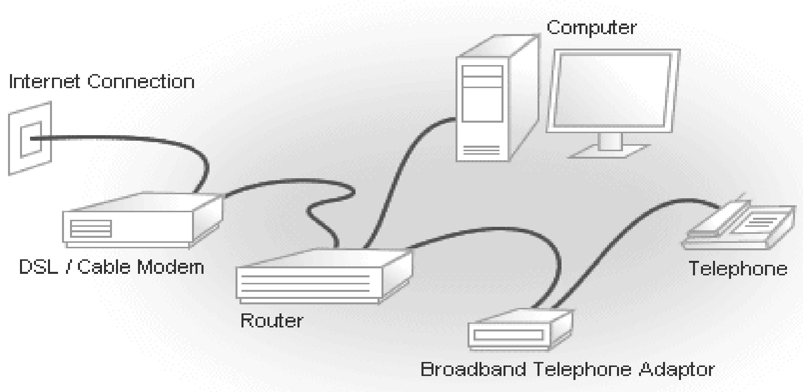

Figure 3. A simple VoIP model

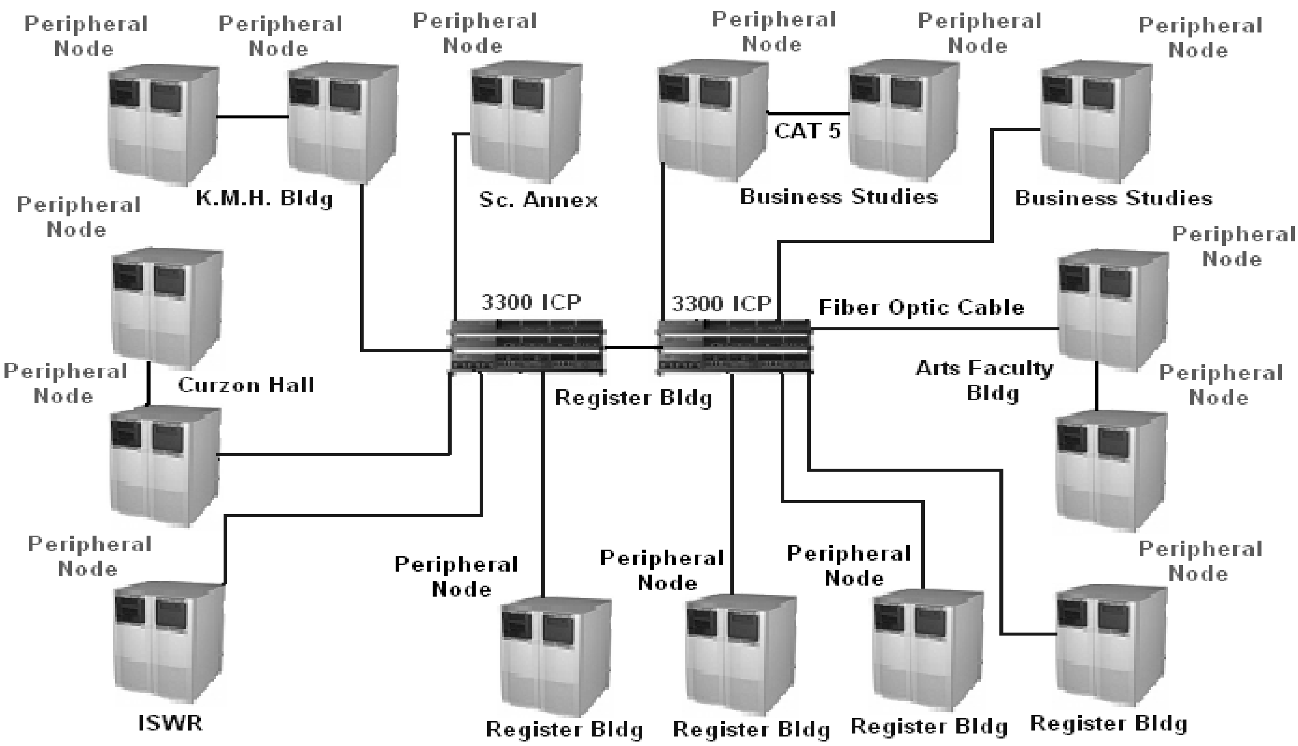

Figure 4. Suggestive system topology in the campus

```
Router(config)#   policy-map   shape-cbwfq
Router(config-pmap)#   class   cust1
Router(config-pmap-c)#   shape   average   384000
Router(config-pmap-c)#   capacity   256
Router(config-pmap)#   class   cust2
Router(config-pmap-c)#   shape   peak   512000
Router(config-pmap-c)#   capacity   384
Router(config-pmap-c)#   configure   terminal
Router(config)#   interface   Serial   3/3
Router(config-if)#   service   out   shape-cbwfq
```

Figure 5. CBWFQ Configuration Example (Cisco Network)

```
 interface   <serial interface or sub-interface>
        traffic-shape rate   64000   8000
46320000
interface   <LAN interface>
traffic-shape rate   64000   8000   46320000
```

Figure 6. Traffic Shaping Configuration Example (Cisco Network)

```
interface pos 0/0
mtu 1500
interface pos 0/0
ip mtu 1500
```

Figure 7. MTU configuration example

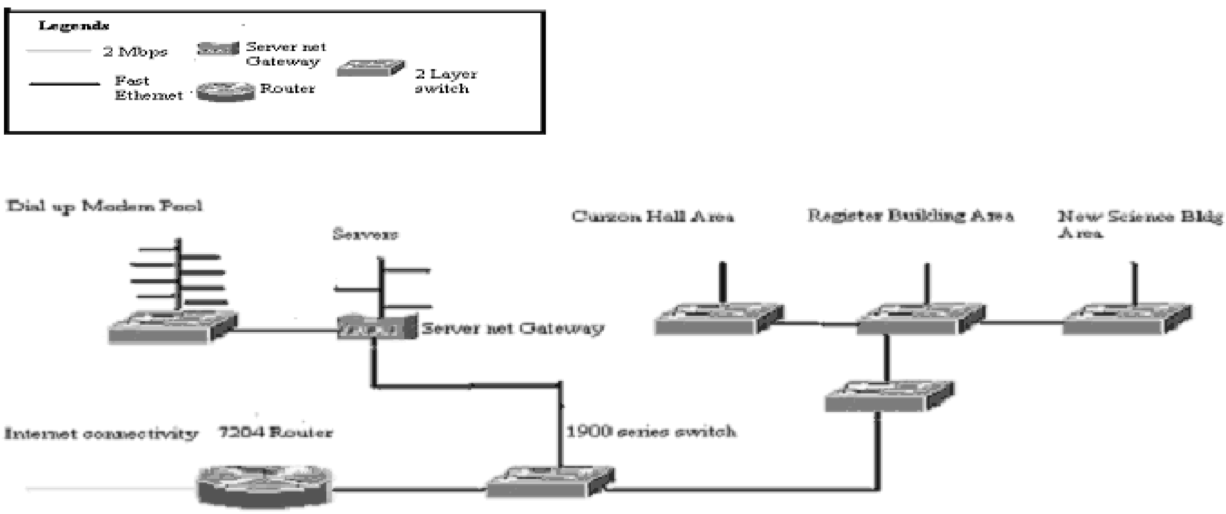

Figure 8. IP network topology

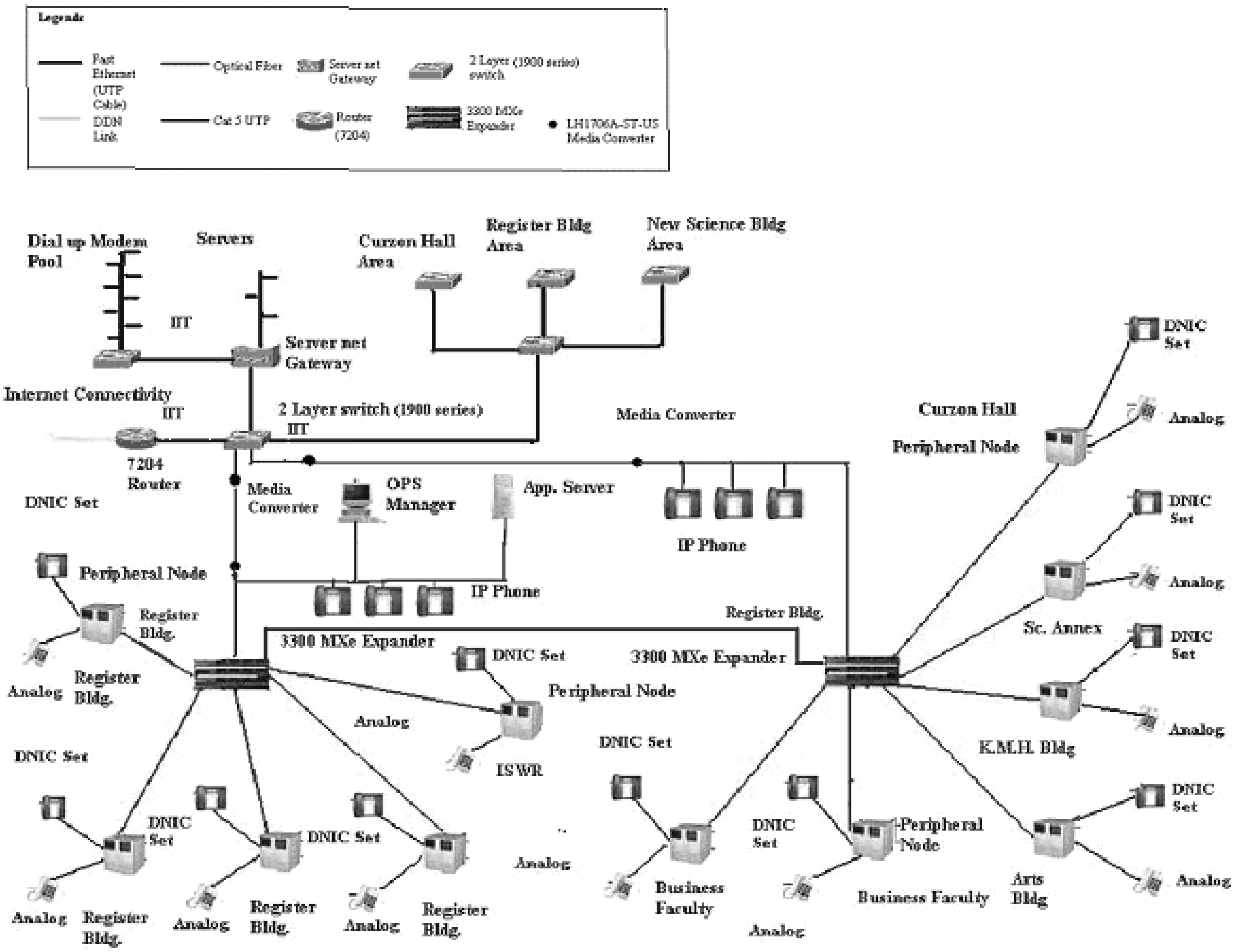

Figure 9. Converged network topology

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Design of Temperature Monitor Device for DC Source Based on 1-Wire Bus

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Abstract

The structure of X6000 DC source system is introduced. A temperature monitor device for its storage pile based on 1-wire bus is designed. Its central processing unit is single chip microcomputer 80296SA in the monitor unit of this DC source. The DS1822 chip, a kind of digital thermometer chip based on 1-wire bus, is used to measure the temperature of the X6000 DC source system storage pile. The hardware, interface and software are designed and the condition description of the interface routine and realization are given. The system possesses of many advantages, including its novel structure, simple circuit and expedient control.

Keywords: 1-Wire bus, Temperature, DS1822, 80296SA

1. Introduction

X6000 DC power supply system is extensively applied in transformer substations and power plants. It can not only offer work electrical sources for apparatus, meters, relay protection and failure illumination in the secondary loop, but also offer impulse current for breaker switch brake loop. For the system management, various sorts of information need to be monitored by the monitor module which also should control the system and make it in the best status. The temperature monitor module needs to monitor the temperatures of many batteries, send alarms when it finds errors, and the system will continually run after human solves the problems, and accordingly the power supply system can normally work. Thus it can be seen that the temperature monitor equipment possesses important meanings for the system.

The 1-Wire bus temperature sensor DS1822 is the series product made by US Dallas Semiconductor Company. It is a sort of integrate circuit chip only using one signal line with another one return line to realize the interlink communication, and it has characters such as high speed and low cost, and it is very fit to be used in locale application (Zhang, 2007, p.183-185 & Zhang, 2005, p.44-47). It can realize the real time presence monitor to the temperature of the pile when combining the DS1822 with the monitor module of X6000 DC power supply system.

2. Introduction of X6000 DC power supply system

X6000 DC power supply system is extensively applied in transformer substations and power plants. In these application situations, it can offer 220V DC operation power supply for the first equipment and offer work power supply for the secondary intelligent equipment. The classic frame of X6000 is composed by storage battery, charger, monitor and other assistant equipments. The system chart of X6000 is shown in Figure 1.

X6100 is the monitor equipment in the X6000 system, and it assumes many core tasks such as collecting data of various parts, timely sending sound and light alarm when failures happen to the system, and implementing various controls according to the requirements of system management and battery management, and its structure is seen in Figure 2. X6100 can monitor the temperatures of many storage batteries. It supports the temperature testing of 18 storage batteries, i.e. 18 Ds1822 chips are needed to compose the 1-Wire bus network. The system adopts the topological structure, and one single main line starts from the host computer and extends to the furthest DS1822 in the slave computer, and other slave computer parts are linked with the main line through the spur track or branch line with less than 3m.

3. Design of hardware

The system is linked by one host computer and many slave computers through one line. The host computer completes the addressing, control, data transmission and power supply through the line. The host computer is composed of micro controllers, and slave computers are composed of 1-Wire bus parts offered by Maxim/Dallas. Every 1-Wire bus part has unique address code to realize the addressing of host computer to different slave computers. And the host computer can link with the computer system through various modes.

The X6100 mainboard adopts Intel 80296SA as the core controller, and it uses EPA (Event Processing Array) part to monitor and control events and enhance the efficiency of the host computer. The EPA is very flexible, and it can be used to produce PWM output. The 1-Wire bus interface occupies the EPA of CPU, and EPA works in the comparison mode to realize the time interval of 1-Wire bus. In order to restrain the transient over-voltage on the bus, the transient voltage suppressor (TVS) P6KE6.8A is added on the data line to absorb surge impulse and other transient over-voltages occurred in the bus. The CPU system chart that 80296SA is used in the electric power monitor system is seen in Figure 3.

The system adopts the three-wire system (seen in Figure 4), i.e. power supply VDD, electric wire GND and data line. The X6000 battery screen closes with the monitor screen, and the creepage of parasitic capacitance in the part make the batteries can not work normally when the batteries are centralized to let and the temperature is over 100 centigrade degree. The reason to adopt the three-wire system is that the software structure of X6100 makes the bus reset impulse must continue a sample halt 1.667ms, or else, the presence signal on the slave computers can not be collected. What's more, in the two-wire mode, such a long reset time will be thought as the power-on reset, and one advantage is that whether the transformation ends through inquire can be judged when the start temperature is transformed.

To ensure the electromagnetic compatibility and the safety of mainboard when the most accidental situation occurs, the high-speed light electrical coupling element 6N137 to insulate the 1-Wire bus and the system board. Two I/O ports of single chip microcomputer are occupied to realize this interface. If the parasitic power supply is used, one added I/O port is needed.

4. Design of software

The software of X6100 monitor equipment adopts the layer system structure which is seen in Figure 5. The lowest layer in the system is the hardware platform, then BIO (rooting program) which answers for hardware initialization, self test and hardware drives. The application program is implemented through the transfer of system monitor, and visits the system services offered by the hardware or the rooting program through the interface of API.

When the system powers on or resets, the BIO runs firstly, and the rooting program initializes CPU and circumambient parts, implements the system self-diagnosis program, judges whether the system program and user course are effective. When the system rooting is successful, the control right is gave to the system program and the user course begins to run.

After deadly errors (such as ROM error and RAM error) occur in the self diagnosis, the rooting program will not run the system program and the user course any longer, and when the human solves the problem, the system can continually run (seen in Figure 6).

The fist step of all 1-Wire bus communication needs that the bus controller sends a reset signal to make the bus in-phase, and then selecting a controlled part to implement the succeeding communication. We can select the controlled part through selecting all controlled parts or selecting one special controlled part to use the series number of this part to implement selection. Once the special part is selected, all other parts are hung on to ignore succeeding communication before the next reset signal is sent out.

Because one sample interval of X6100 is 1.667ms and CPU must deal with other super tasks, 1-Wire bus drive program can not make CPU still wait for the accomplishments of other operations. The task transfer mechanism of X6100 decides the drive program can only realize the read-write of 1 bit in one sample halt. The subprogram produces time interval position on the bus and implements the read-write operation at the same time. One read time interval is just one write time interval, and the salve computer changes 1 into 0, and returns anticipant data according to the requirements.

There is no search of ROM series number of DS1822 in X6100, but it offers the function of Read Rom. In the actual using, one chip is linked in the network, and the series number of the chip is read and memorized in Flash Rom. After reading every DS1822, the system can be run normally. The system sets up 12 bits transformation mode for DS1822, 1LSB corresponds with 0.0625 centigrade degree, and the arithmetic adopt appointed decimal fraction (holding 2-digit decimal fraction) is 100*T=100*Code*0.0625°C=(Code*25/4) centigrade degree. The real temperature can be obtained when the data read from the X6100 in the upper computer divides 100. Because on the 12-digit transformation mode, DS1822 needs 750ms to complete one time transformation, so here the method that the broadcast order sends the transformation to read the real temperature tested by every chip is adopted to realize the temperature circular test of the storage battery.

5. Conclusions

The technology of 1-Wire bus has incomparable application foreground because of its advantages such as simple circuitry, cheap spending of hardware, low costs and simple software design. The X6000 DC source temperature monitor equipment based on the intelligent temperature sensor DS1822 of 1-Wire bus technology is designed in this article. It can directly export the temperature value of the tested point in the digital form, and the equipment has characters such as small temperature error, high differentiation, strong anti-jamming ability and low costs, and it can also transmit data in long distance.

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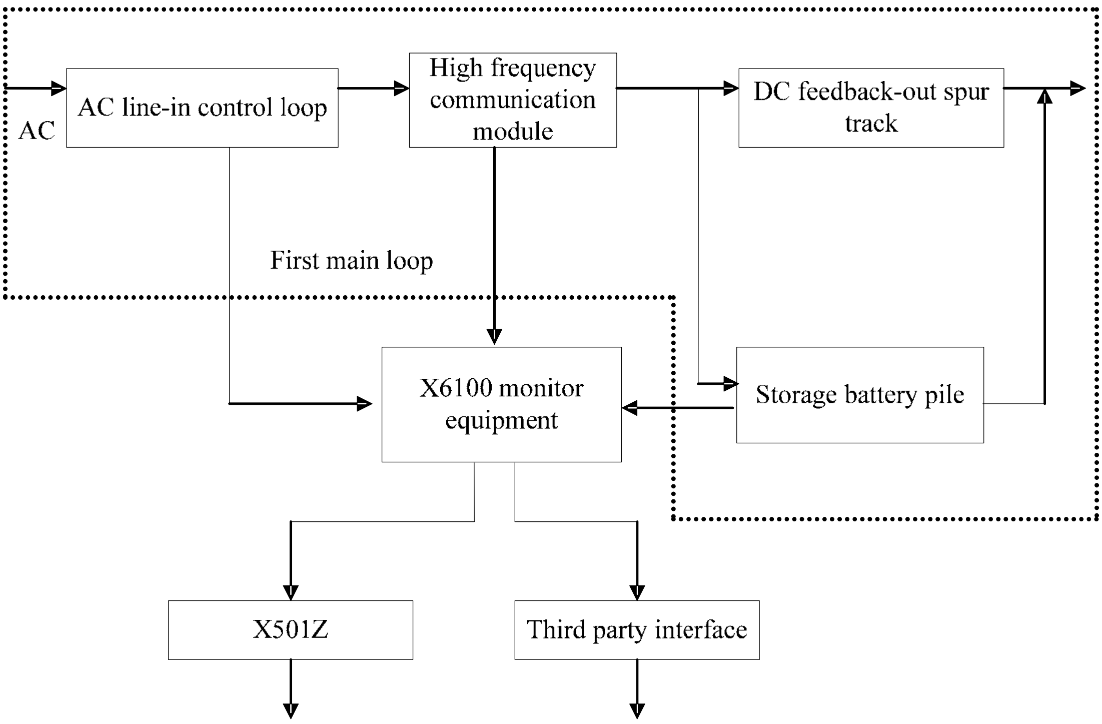

Figure 1. X6000 System Chart

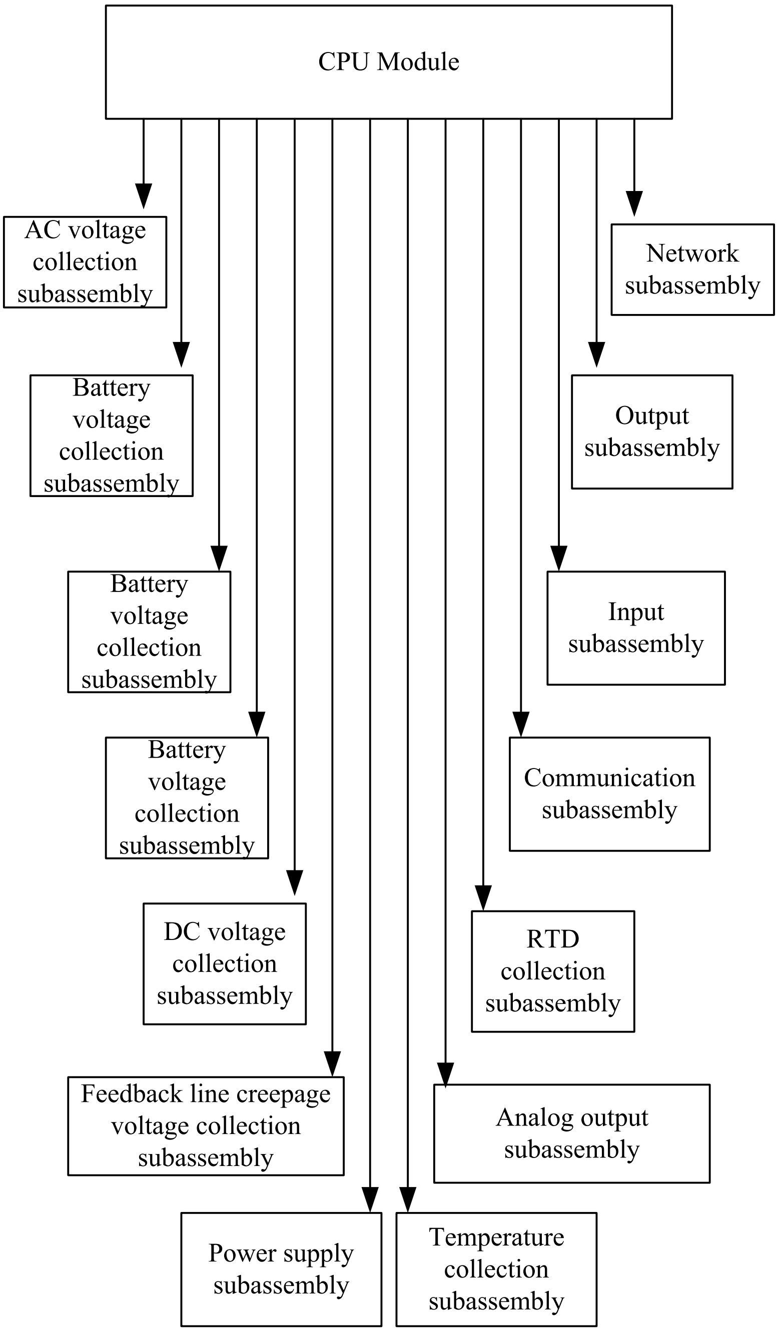

Figure 2. Structure of X6100

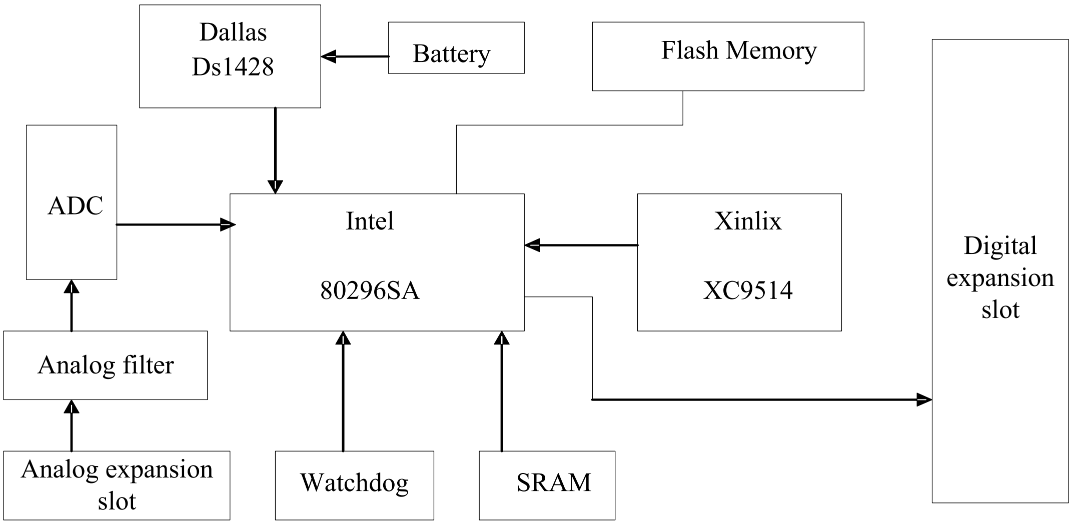

Figure 3. CPU System Chart of 80296SA Used in the Monitor of Electric Power System

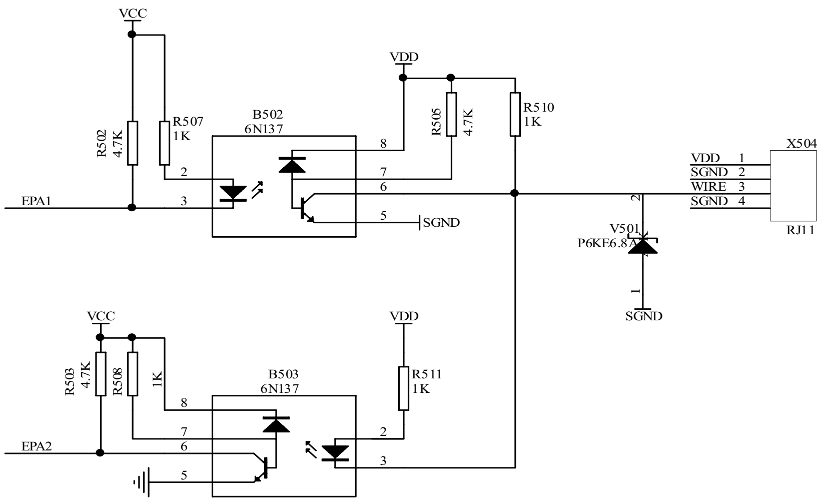

Figure 4. The Interface of Host Computer Port for X6100 1-Wire bus

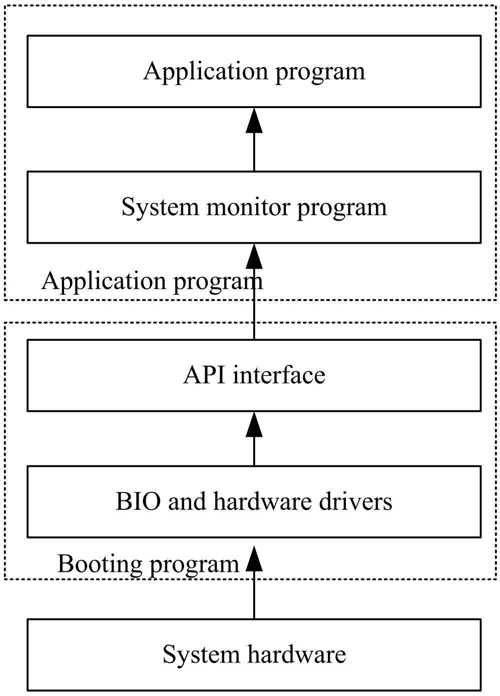

Figure 5. Software System of X6100

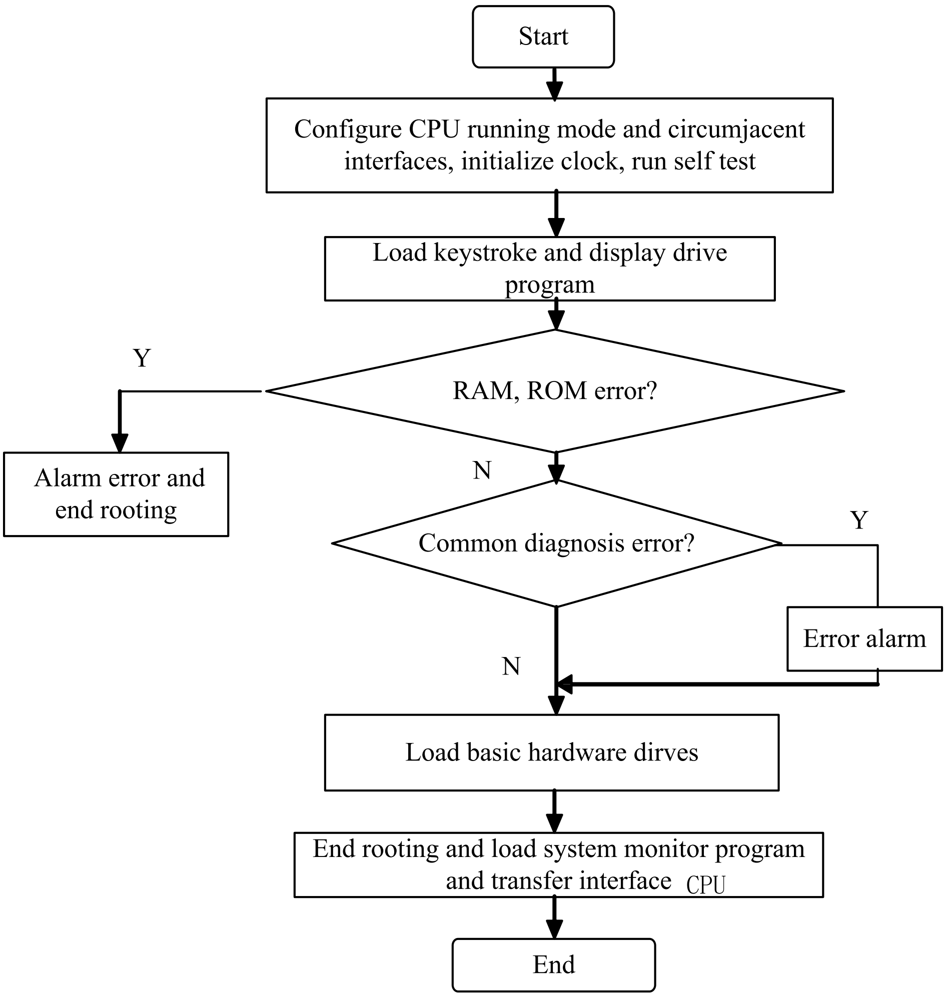

Figure 6. Power on Self Test of X6100

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The Integration of 3D GIS and Virtual Technology in the Design and Development of Residential Property Marketing Information System (GRPMIS)

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Abstract

This paper discusses about a research with the aim of investigating the potential integration of 3D GIS and virtual technology in designing and developing residential property marketing information system. The method adopted in this research is a standard system development lifecycle; commencing with the user requirements study, followed by the system design, the system development, the system implementation and finally the system evaluation. This research uses an informal method i.e. semi-structured interview, survey questionnaire and review of the existing information system to establish the user requirements. Ten user requirements were outlined alongside with the examination of four 3D integration and three virtual reality methods. Three out four methods of 3D features integration are selected for the system development. The developed system is tested using the black box and white box testing methods. The prototype system can be used by the real estate agents and property developer as the concept, framework and references for future development of a better conducive property marketing information system as well as simplifying the traditional flow of housing selection which gives positive impacts in the marketing transaction.

1. Introduction

Property Management is a term that has many disciplines related in it. According to Narains Corporation (1971) a famous property consultant and realtor in India, property management can be defined as the function of looking after buildings. These functions comprise of rental collection, payment, building maintenance, services condition, insurance policy and many more. Selling and marketing are also one of property management functions which have become the main focus in this research. Marketing involves several parties such as estate agents, home owners, developers, lawyers and legal companies. Housing estates and property industries have become significantly active industry in Malaysia.

This has led to a large number of investments from the property developers, contractors, professionals and local authorities. The rapidity of property development results in more choices of houses available. Consumers have certainly become more indecisive towards making a choice.

Traditionally, house buying activity involves site visiting and verbal communication between the property sellers and buyers. Even though the practice has been successfully implemented for many years back, this method is rationally inefficient as it requires more time, energy and money. Having known that, some of the property developers and estate agents have taken one step ahead by putting and advertising properties on the road billboard, newspapers, television, posters and even on the web for selling and renting purposes. These methods only succeed in getting the information reach the consumer. Consumers still need to go to the site to obtain details such as the ground state and architectural design. Considering these issues, an investigation on the potential technology to be embedded in designing a marketing system to facilitate the needs of such activities has to be established to improve the traditional activity of choosing, buying and selling a property.

Currently there are a number of property marketing systems available in Malaysia. Examples of such systems are *Fullhouse.com.my* and *Propertycity.com.my* websites. *Fullhouse.com.my* was developed by *Fullhouse Sdn.Bhd* and Real Estate and Housing Developers' Association Malaysia (REHDA). The system only provides the database information of the house and images for some of the listed property. was created by *LCT Info Enterprise.* This system is a one stop property search engine for users to search for potential house/properties for sale or rent and it also provides a Real-Time Property Alert through email to their subscribers.

The computerised property systems have moved from database-oriented to the location-based system using GIS technology. GIS technology is particularly beneficial in integrating data from a wide range of sources using location as a common point of reference. The applications of GIS are generally set out to fulfil the *five Ms* i.e. mapping, measurement, monitoring, modelling and management (Longley *et al.*, 2001). These capabilities can be seen in the *GIS-based Property Database*. It was developed to permit the geographical representation of property information at the individual property level (Wyatt, 1997). Despite the realisation of the importance of using location-based system using GIS technology, the development of such a system has not been the focus of many property marketing systems. Thus the objectives of this research are:

i) To analyse the user requirements for the design and development of Property Marketing Information System (PMIS) and its current practices,

ii) To design and develop the GIS-based Residential Property Marketing Information System (GRPMIS).

In view of this, the next section provides a brief review of the 3D GIS and virtual technology in the property and marketing disciplines.

2. Literature Review

The introduction of GIS technology in computerised property systems has enable users to handle large spatial data and improve the system efficiency. Evidences and justifications of insufficiencies in the existing information systems to handle spatial information were identified by several authors (Bodum *et al.*, 1998; Croswell *et al.*, 1994; Doyle *et al.*, 1998; Gruber *et al.,* 1997 and Tempfli, 1998). GIS technology is typically implemented in the property field when the property systems are designed to support geographical information (Wyatt and Ralphs, 2003). Recent technological advances in the delivery of spatial data across the web have facilitated the development of open and scaleable land management solutions, with interfaces to centralized data sets from the internet, intranet, and local and wide area networks.

According to Zlatanova (2000), commonly established systems dealing with spatial data are 2 Dimensional (2D) GIS and 3 Dimensional (3D) Computer Aided Design (CAD) system. The CAD system was found to have deficiencies in modelling large models of 3D visualization. These deficiencies can be overcome with the technology such as Virtual Reality (VR) and the improvements of hardware and libraries. VR is a form of human-computer interface (HCI) that involves the creation based on a reality (Brodlie *et al*., 2002). VR is a computer generated simulation that allows user to interact with data that gives the appearance of a three-dimensional environment, while 3D is a view that provides depth perceptions. Fisher and Unwin (2002), have demonstrated numerous applications of virtual reality in geography (see also Brodlie *et al*., 2002; Kluitmans and Collin, 1991; Gillings, 2002; Haklay, 2002; Kraak, 2002; Batty and Smith, 2002; Ogleby, 2002 and Cheeseman and Perkins, 2002). The VR technology such as animation can provide a walk through model of building and construction sites. Animation enables the 3D view to become alive. This can be done by recording actions and replay as desired.

Earlier studies (Md Sadek *et al*., 2005, Ljungvist, 2003, Dogan *et al.*, 2004; Kolbe and Groger, 2003; Smith and Friedman, 2004, Zhou et al, 2004 and Zlatanova, 2000) have identified the uses of 3D and VR for city modelling and planning. The result has shown a significant difference in the degree of efficiency throughout the study (Marcus, 2005 and Dogan *et al*., 2004). The application of 3D and VR technologies in property systems could improve the management efficiency. According to Kolbe and Groger (2003), to reach interoperability on the system level, multiplatform compatibility has to be achieved. Hack and Sides (1994) stated that VR or simulated worlds will be used for GIS modelling in three-dimensions in the future. The animated GIS objects will enable us to visualize spatio-temporal objects in the environment (Davis and Williams, 1989). The key element in utilizing 3D GIS is the conceptual model. Zlatanova (2000) stated that the conceptual 3D model integrates information about semantics, 3D geometry and 3D spatial relationships (3D topology). The methods for describing real world are provided through the conceptual model (Zlatanova, 2000).

Having known that technologies such as GIS, 3D and virtual reality have the capabilities to improve system efficiency, the establishment of the integration of such technologies in the property marketing system could improve and simplify the process of selecting, buying and selling a property. The time requirements may be shortened and information retrieval could be done effectively. These technologies have not yet been utilised in any of the existing property marketing system. In due respect to the above concerns, this thesis particularly investigates and explores the potential integration of these technologies in the design and development of property marketing system and how GIS technology helps to manage many projects and to produce more effective, more equitable and more predictable results. Property management decisions need to be made on the basis of accurate information and therefore GIS applications often entail a relatively low level of analysis but high degree of accuracy (Wyatt and Ralphs, 2003). Therefore, investment in GIS technology and data collection can be justified.

3. Methodology

Based on standard System Development Life-Cycle (SDLC) five stages i.e. the study of user requirement, system design, system development, system implementation and system testing and evaluation were applied in the development of the system. With some alteration, these stages are further divided into three phases (refer Figure 1). The study of user requirements is conducted by using semi structured interview, survey questionnaires and review of existing system. Triangulated research model for this study is adapted from Sauer (1993). This model described the need for the system engineer and the potential user to work together in order to establish the user requirement.

Two groups were chosen in the study of user requirements are the global perspectives and individual perspectives groups. The global perspectives group provides experiences of using the existing property marketing system. This is usually done by going through the technical ability of the system, system report, reviewing current practices and functions available within the existing system. The information is gathered using interviews and conducting a survey. The individual perspectives group on the other hand supply information regarding the users' perception and expectation through interviews, observations and questionnaire. This group provide useful information for the design of the proposed system user interface.

The core of the system design phase is the establishment of conceptual, logical and physical design for the proposed system. Two approaches used to create the conceptual, logical and physical design are the database approach and object-based approach.

The proposed system needs to undergo testing and evaluation. The testing methods that are used in this study are the Black Box and White Box testing.

4. The Result of the User Requirements Analysis

There are three methods used for the user requirements analysis in this study. First, literature review and observation of the existing property marketing systems were conducted. Second, semi-structured interview to the property developers, real estate agents and potential consumers and third is the questionnaires to the potential users of the proposed system integration. The observation of current practices and existing system is carried out by looking at the current activity of buying/selling property, observation on the existing property marketing systems in Malaysia and analyzing existing study performed to identify user requirements for building information system. Several existing property marketing system have been looked at provide insight into the overall system architecture of the existing property marketing information system. The semi-structured interviews were undertaken to selected individuals from different organizations.

The inferential method was applied for the identification of user requirement in the survey questionnaire option. The inferential statistics is used to infer and conclude the population judgement and opinion of the development of the proposed information system based on the responded questionnaire. As the name implies, the inferential statistics are used to draw conclusions, to make inferences, or to make predictions about a given population from sample data from the population. Based on the population of the Seksyen 7, Shah Alam, Selangor, Malaysia observed in year 2000, the number of population is 5440 and number of sample is calculated based on margin of error of 10 percent which is the amount of error that can be tolerated, 90 percent (90% = ±1.645) tolerated amount of uncertainty (confidence level). Five hundred questionnaires were distributed; only 260 samples are used for the study. The result from the study undertaken reveals 10 user requirements to be used in the development of the proposed system as listed below:

1. The system should enable users to perform database query and view tabulated result.
2. The system should allow users to perform location query and view projected map on the screen.
3. The system should allow users to perform visual query and visualize query result on-screen and report.
4. The system should provide users with the facilities information for the housing area.
5. The system should allow users to obtain information regarding the housing property e.g. price, built in area, address, owner/developer, etc.
6. The system should provide users with the topographic information of the housing area.
7. The system should allow users to view inside and outside the house in 3D and perform virtual analysis (walk through/fly through and shadow).
8. The system should allow advertisement (permitted to certain users).
9. The system should enable users to print out the report.
10. The system should be user friendly, conducive and easily access.

5. The Development of GRPMIS

The system development of the GRPMIS involves several stages. First is data preparation which will be described in Section 5.1. Second is the animation development. The animation development comprises of the exploration of three methods. These methods are described in detail in Section 5.2. Third is the system integration. Four methods are examined for the proposed system integration is described in Section 5.3. Figure 2 shows the proposed system development workflow.

5.1 Data Preparation

Figure 3 shows the workflow of the constructions of basemap for the GRPMIS. The data preparation workflow illustrates that two main data types are generated. First is vector data generation. The vector data for the proposed system is generated from digital map and topographic map. Second is 3D data generation. The 3D data is constructed by using digital photograph and house floor plan. Three major steps involved are data acquisition, extract layer and post-processing. The output of the data preparation is the basemap of the test area for the proposed system development.

5.1.1 Vector Data Preparation

The process of vector data preparation involves 4 sub-stages i.e. data acquisition, extract layer, post processing and output. Data pre-processing consist of vector data registration and calibration. Vector data registration is constructed by assigning the coordinate system to the unsigned data. The uniformity of the coordinate system and projection is crucial. Vector data calibration is done by performing some topological testing. At this stage all the layers are converted into geodatabase and the topological rules are applied. This step is important in order to ensure that the topological property of the data is valid. Topological rules can be applied within the ArcCATALOG environment.

The existing digital data is then reproduced, edited, updated and corrected. Four layers that were extracted from the topographic map are *street, land lot, contour* and *building* layers. The data produced from the topographic map and existing digital map are then merged. The database of the topographic map and the digital map need to be connected by unique identifier so the attributes table can be related. With that, the GIS spatial data and attributes data are now integrated. Using the 3D analyst extension, a Digital Elevation Model (DEM) is constructed to give perspective view of the test area. All the designated test layers were draped onto the DEM giving a new evolution of visualization. The test area is then represented in model view where the topographic features can now be identified. The completions of vector data preparation provide the basemap of the test area.

5.1.2 3D Data Generation

The 3D data of the test area do not need to be constructed after the vector data are prepared. 3D data can be generated at anytime or alongside the vector data preparation. This is because some of the 3D such as trees, play-ground, lamp or street furniture data do not need the base-map as the point of reference. The need for the reference point is dependent on the method chosen to integrate 3D features into GIS data. The method tested for integration is elaborated in Section 5.3. As in vector data preparation, four stages are involved in 3D data generation i.e. data acquisition, extract layer, post processing and output.

The textures for the 3D features generated for the test area are acquired based on the digital photograph captured at the test area. The photographes are edited using the selected graphic software and implanted into the 3D features of the test area. The floorplan is used to generate 3D house of the test area. The floorplan is scaled and projected based on the spatial data imported into the selected 3D software. The coordinate system is maintained according to the spatial properties of the GIS data. The construction of the 3D data is performed by using Sketch-Up 5 and 3D ModelBuilder software. The 3D features are then imported into the geodatabase or directly imported to ArcScene through the 3D marker symbol option.

5.2 Animation Development

Animations consist of tracks that are bound to objects. These objects may consist of layers, the map view (ArcMap), the camera (ArcScene and ArcGlobe), or the scene (ArcScene) whose properties can be animated. The animation scheme is developed from two softwares; ArcGIS and Sketch-Up 5. Figure 4 shows the animation development workflow. It involves five format of 3D features and these data are examined using three methods of generating animation. The animation files are them imported into VRML and .AVI format. VRML format require virtual files player such as Cosmo Player.

Scene animation for the whole study area is created using ArcScene while the residential property animation and facilities are created within the Sketch-Up program. The animation toolbox is activated via the standard pulldown menu in ArcScene. There are three methods of creating animation in ArcGIS 3D Analyst; animation capture by camera, by scene and by layer.

5.2.1 Method 1: Motion capture by Layer

Layers in ArcScene can be animated in various ways. The proposed system is chosen to animate moving layer such as a car track at the study scene. One way of creating the layer track is to move a layer along the path and create flyby from path. Using the existing layer such as road or street layer, a path is selected and the options to move layer along the path and create flyby from path will be activated. Subsequently, the move layer along path options and the scene importer will be prompted.

The motion animation can be displayed in various formats (vrml, .avi and .asa). Viewing animation files in format .avi require exporting animation to video step. When the file types format to be saved is selected as .avi, the menu for video compression option will be prompted. There are several choices of video codec compression options available. However the cinepak compressor is recommended by ESRI. The 'Compression Quality' setting indicates how high a resolution will be used when rendering the video. The higher the setting, the sharper is the video resolution. However, it results in a larger file size and will require more sophisticated and faster computer for the processing.

5.2.2 Method 2: Scene Animation capture by Camera

Motion capture by scene is done by manipulating the scene view and uses camera tools to save the scene. The animation will move from sequential captured keyframes. These keyframes are captured by using the navigation tools in ArcScene to select the prefered perspective view of the study area and camera from the Animation Tools for scene capturing.

Each scene, layer or frame captured using ArcScene Animation Tools is managed by using the Animation Manager. Animation Manager control the timeframe of the animation. When the animation files are saved, these files are saved into two format (.AVI and VRML). File format .avi is used for the scene playback. Using windows media player this animation files can be viewed without users control. The VRML format will offers 3D Scene user navigation power.

5.2.3 Method 3: Sketch-Up 5 Animation Developer

Creating animation in Sketch-Up 5 is fairly straighforward. Theoretically, animation in Sketch-Up is the same as Scene animation in ArcScene. The difference is that the scene is save in separated window and played according to the window arrangement. Every scene of the house is stored in different windows tab, and the sequential scene is saved in another window tab simply by adding new tab. The preview of the animation can be displayed within the Sketch-Up windows itself by selecting the slideshow option. When the desired sequential is saved and completed, these house scenes can be exported using export to animation option.

5.3 3D System Integration

The data preparation stages have to be completed in order for the 3D system integration to take place. The output from data preparation stages is used at this stage. Figure 5 shows the integration of 3D features into spatial data. Four methods are examined. Each method is described in detail in the following sections.

The 3D model library output is represented in two different formats; the OpenFlight (*.flt*) and Sketch-Up format (*.skp*). Both formats are compatible with ArcScene, this is important in order to be able to visualize and retrieve the symbols in ArcScene. The basemap created within data preparation is used as the reference subject for placement of the 3D models and features created.

The process of bringing the 3D models into the base map fulfills the integration of 3D GIS. To retrieve these 3D features from Sketch-up, ESRI GIS Plug-Ins is needed. This extension will enable the 3D model to be imported into ArcMAP Personal Geodatabase. Customization is started when all the 3D models are fully retrieved into the base map. From this point, the development of the system begins by using Microsoft Visual Basic and MapObjects as the GIS enabling tools. Four Methods are examined to investigate the capabilities of the 3D model integration into GIS data.

5.3.1 Method 1: Importing TIN into Sketch-Up and Drape Lotpoly layer onto TIN

The perspective view map displayed in ArcScene can be imported into Sketch-Up. The first method is by selecting the features and clicking the import to Sketch-Up button provided the GIS plug-Ins is installed. The import window is prompted and it will ask whether TIN should be included as the import features or not. TIN is crucial to maintain the height value of each land lot. These features need to be included. In some cases where the height value is stored within the land lot attributes table, TIN can be excluded. This step will not affect the projection and coordinate system that has been assigned to the spatial data in the earlier process. The Lotpoly layer is brought up onto the TIN surface. Soften edges tool is used to smooth the TIN surface. This is done to simplify the TIN and to minimize the processing time. A complex triangulated network contained in TIN layer may delay or even fail the draping process. Although this method can be used for the integration, it requires more time to complete due to the TIN complexity that also affects the visual memory and capacity. The performance of the machine tends to get affected and it is impossible to complete the 3D features placement of only one house on an average of ten minutes. Only if this problem could be solved, a perfect 3D GIS integration method is found.

5.3.2 Method 2: Convert layer into 3D and import using Shapefile Importer

The second method uses the shapefiles importer. This options is easier but it might take awhile if the shapefile size is big. Digital Elevation Model (DEM) is created from the layer contour. Within ArcToolBox, using 3D Analyst extension in ArcGIS, this DEM is interpolated into polygon multipatch with the *lotpoly* layer. This is done in order to preserve the high value within DEM into the specified layer. The result of this method is that the layer output in shapefiles (.shp) data format instead of multipatch. The value of height is stored within the type of polygon as '*Polygon ZM*'. Although the layer displays well in ArcScene, the features from interpolation appeared to be in multipatch format and cannot be imported into Sketch-Up and as for the shapefiles (.shp) output, it failed to import a complete layer of lotpoly layer. Some contains in lotpoly layer appeared to be missing. Although, the layer appears to be elevated in Sketch-Up but the missing pieces interrupt the process of 3D features placement and retrieval.

5.3.3 Method 3: Using Plane-view layer and import using Import to Sketch-Up tools

Method 3 is examined by using the land lot GIS layer. This method only requires the Import to Sketch-Up plug-Ins. The import process of GIS layer into 3D software is easy. The step is straighforward by selecting the layers and using Import to Sketch-Up button in ArcMAP or ArcSCENE, the layer is available in Sketch-Up. The method of import GIS layer into 3D software used in method 3 is the same with the one used in method 1. This time, the option of bringing TIN (elevation model) into 3D software is not selected (uncheck). This is to examine how is the complexity of the elevation model may affects the performance of the memory of the machine capacity and the time require to model the 3D features created. As a result, the placement of 3D features using this method is easy. The only drawback of this method is that all the 3D features is placed onto a plane land lot instead of the lot with an elevation. That is why the process did not require longer time.

5.3.4 Method 4: Importing 3D features by creating point layers

This method is examined to determine the performance of 3D software and GIS software by eliminating the large and complex value of layers such as the large files to export 3D site from Sketch-Up to ArcScene. This method is performed by creating point layers for each lot, where the original polygon layer for lot is converted into point features. For other features such as trees and road facilities, the point features are digitized and the type of trees and facilities are stored.

6. The GRPMIS

This section described the GRPMIS functions and flow according to the three levels of user. The elaboration starts with the guest user environment, followed by the registered user environment and finally the admin user environment. The detail about how GRPMIS runs within each environment is illustrated. The GRPMIS will start with the window shows in Figure 6.

This window introduces the research and GRPMIS in brief. Three links presented in this window are entering the system (GRPMIS), looking up for system info (SYSTEM INFO) and exit starting window (EXIT). Clicking at the GRPMIS link will lead to the Log In window of the system (refer Figure 7).

6.1 The Guest User

The user will have three different ways to Log In depending to the type of user browsing the system. Guest user is a system browser. Guest user can enter to the system without typing any username and password. However, the functions available to the user will be restricted. The guest user can click the Login button and that action will get them to the GRPMIS State Selection Menu (refer to Figure 8). Clicking at the Selangor link will take user to the GRPMIS Main Menu as shown in Figure 9.

As can be seen in Figure 9, the function Manage Property and Edit Propety are not active. These functions are meant for the registered and admin users. The guest user will only be allowed to perform the following functions:

i. Layer Activation functions (1)
ii. General function (2)
iii. Standard Navigation Tool (3)
iv. Viewing Window (4).

Layer activation function enables the user to select specific layer to be viewed in the viewing window. The standard navigation tool offer functions such as zoom in/out, interactive zoom in/out, pan and pointer. The general function allow guest user to perform buffer search, radius search, 3D GIS view, Search Advertisement and View Advertisement. Snapshot of the Search Advertisement window is shown in Figure 10.

From here, a guest user can access to the detail property information. User can highlight the search result by clicking on the *View House Details* button. A new House Details Window will appear on the screen (refer to Figure 11). This window has four connecting tabs (House Information, Picture, Floor Plan and 3D Floor Plan). All four tabs are available for the guest user to view. The House Information, Picture, Floor Plan and 3D Floor Plan tab is shown in figures 11 and 12 respectively.

The Floor Plan 1, Floor Plan 2 and Floor Plan 3 buttons will lead the user to 3D animation Plan as shown in the following Figure 13. Finally, guest user is able to print out the report of the search result information.

A Guest user can also access to the property information through the viewing window. This can be done by clicking at the pointer function at the navigation tool. Then click at the lot or land parcel at the map presented in the viewing window. A small window will appeared on top of the viewing window as presented in Figure 14.

Using the Pointer, a Guest user can choose any land parcel and from the pop up window user can click at the view button to display the House Details Window (same as Figure 11). However, the Edit button will not be activated in the pop up window, as the Guest user is restricted to use this function. This function will only be activated within the Registered User environment.

In additional, a Guest user can also perform radius and buffer search. Using *Pointer*, the chosen land parcel is selected (yellow dot become apparent indicating the selected land parcel). Clicking the Buffer or Radius button will lead user to the function menu. Figures 15a), b), c) and d) illustrate the results and the menus of these functions.

The 3D View function enables user to visualize the area in perspectives view. A user can access to the Perspective View window and the CosmoPlayer (VRML) window subsequently. In the Perspective View window, the navigation function is available with some limitations. The navigation function is operated using the mouse pointer. This window also allows user to visualize the area without the 3D model features. Only land parcel, road and other selected layers will be draped onto the elevation model. The Cosmo Player (VRML) window is able to view the elevation model of the test site. Further investigation is still needed as there still some problem in the rendering process of the 3D features. Figures 16 show the generated perspective view and CosmoPlayer(VRML) window.

6.2 The Registered User

The Registered user is the user with personal account or registered as the formal user of the system. Registered user will be provided with the username and password upon their registration to use the system (refer Figure 17). This type of user usually has specific purposes for joining as the system member such as to advertise and sell their property.

This username and password will allow them to access extra function called Edit Property function. A screen-chop of the Main Menu for the Registered user is shown in Figure 18.

As shown in Figure 18, the Registered user can access to all the functions available within the Guest User environment with an extra function known as Edit Property function. Furthermore, Registered user can access to the editing environment by clicking at the pop up pointer menu where the function Edit will be activated or from the Edit Property function. From these functions, the user will have access to the House Details window where another Edit button is activated (refer to Figure 19).

Access to the Edit function is undertaken by clicking at the highlighted Edit button (refer to Figure 20). Once the Registered user clicked at the Edit button, the new window called Advertisement (Edit) window will be prompted on the screen (refer to Figure 20).

A Registered user is able to perform editing task for the property database information only. Information to be entered are location, price, residential type, status, number of rooms, facilities, pictures and the contact number. To change the price, the user can select price range from the drop-down menu as shown in Figure 21. The property status can also be changed from the selections provided in the drop-down menu as shown in Figure 22.

Changing and updating the facilities information can be performed by clicking on the box given in the Advertisement (Edit) Menu (refer to Figure 20) while updating the house pictures can be done by selecting the pictures at the

Registered User local computer, where the system will download the pictures information into GRPMIS. Uploading pictures from the user local computer can be performed according to the steps given in Figure 23.

The Registered user can click at the triple dot button (as shown in 1) and window explorer will

open file menu as shown in 2 in figure above. Specific file is chosen to be uploaded into GRPMIS.

Finally, a registered user can also update the contact information by selecting the contact references available in the GRPMIS. This contact information is entered during the GRPMIS registration. This function is illustrated in Figure 24.

Once the Registered user has completed the editing task, Save button should be clicked to store the updated information and a window will be prompted stating the information have been updated (refer to Figure 25).

Besides these functions, a Registered user can also perform other tasks and functions available for the Guest User.

6.3 Admin User

The Admin User is the system author. The Admin user has the ability to reconstruct the GRPMIS. Within the Admin User Environment, all the functions are available including the Manage Property function. The Admin user will also have the Admin username and password. Figure 26 shows the Login window for the Admin User.

The Admin user functions enable the registered user to acquire the 3D model and virtual room (model) for their property. The Admin User will be the one who construct the 3D model and upload it in the specific property database. The Admin User is able to access Manage Property function (refer to Figure 27).

Once the 3D model construction is completed, Admin user will update the property information by

clicking at the Manage Property button 1 within the Search/View Advertisement window as

highlighted in Figure 27. Then, a new window titled Manage Property window 2 will become apparent. This

window will lead the Admin User to the local machine storage where the completed 3D model is stored. Then the chosen 3D model and Virtual Room (model) are uploaded into GRPMIS for the Registered User and Guest User viewing. Figure 28 shows the open file explorer result when the triple dot button is clicked at the Manage Property window.

After the related files are chosen, the Admin user will update the uploaded information. When the information is successfully updated, a window will appear informing the Admin user that the information have been updated (refer to Figure 29).

Other functions that are available for the Guest and Registered User are also available for the Admin User. The Admin User has the freedom on how to manage the GRPMIS.

7. Testing

The objectives of system testing and evaluation is to enable the system in its entirety to be analyzed and to ensure that the business function for which it was originally intended is being met. These proposed functions will be examined in order to verify the reliability and effectiveness based on the users' perception and technical perception. One could view System testing as the final destructive testing phase before Acceptance testing. It was conducted using the Black Box. The term *black box* indicates that the internal implementation of the program being executed is not examined by the tester. The term *white box* (or *glass box*) indicates that testing is done with knowledge of the code used to execute certain functionality (Tonella and Ricca, 2004). For this reason, a programmer is usually required to perform white box tests. The following sections will elaborate on how these methods are adopted in this research.

7.1 The Black Box Testing

The Black Box Testing is also known as *functional testing*. This is a system testing technique whereby the internal workings of the item being tested are not known to the tester. The system testers for this method have no idea with the internal environment of the system. These people only look at the output of certain functions without knowing the rules and logic used in order to get that kind of output. The tester does not examine the coding and does not require having the background knowledge about the programming languages and so on. The focus of the Black Box testing is to ensure the functions are available and able to give outcomes. The advantages of this type of testing include:

i) The test is unbiased because the designer and the tester are independent of each other.

ii) The tester does not need knowledge of any specific programming languages.

iii) The test is done from the point of view of the user, not the designer.

iv) Test cases can be designed as soon as the specifications are complete.

The disadvantages of this type of testing include:

i) The test results can be redundant due to the various tester backgrounds and if the testing has already been conducted.

ii) The test cases are difficult to design.

Under the black box testing, there are a number of sub-testing aspects. Three components were tested for GRPMIS are integration testing, system testing and user interface testing.

7.1.1 Component 1: Integration Testing

Integration testing is required to ensure the related sub-system and software is working correctly. The following list is the step taken to perform integration testing:

i) Determine the sub-system to be tested.

ii) Install system into testing machine.

iii) Perform functional testing against the system.

iv) Record test results.

v) Verify the test results.

7.1.2 Component 2: System Testing

System testing is required in order to verify all the system functionality and making sure that these functions is working correctly and conforms to specifications. It is done when all the system functions had been implemented. Steps below are taken during the system testing:

i) Determine all functions to be tested.

ii) Deploy the system into testing machine.

iii) A checklist of all functions and requirements is used.

iv) Record test results

v) Verify the test results

7.1.3 Component 3: User Interface Testing

This testing is conducted to rate the usability of GRPMIS. This testing is performed by selected system users or clients in order to find the true ends users experience and perceptions. Below are the steps taken to perform user interface testing:

i) Install system to the test machine.

ii) Guide the end user to use the product.

iii) Give the tester User Interface Test Result.

iv) Records user comments, suggestion, difficulties encountered and experience during the session.

v) Report test result.

vi) Verify the test result.

7.2 The White Box Testing

The White Box Testing is also known as *glass box testing*. This is a system testing technique whereby explicit knowledge of the internal workings of the item being tested is used to select the test data. Unlike black box testing, white box testing uses specific knowledge of programming language to examine the outputs. The focus of this test is to examine the robustness of the system construction and looking into the logic, rules and parameters used in order to get certain output. The test is accurate only if the tester knows what the program is supposed to do. White box testing does not account for errors caused by omission, and all visible code must also be readable. The advantages of White box testing are:

i) As the knowledge of internal coding structure is prerequisite, it becomes very easy to find out which type of input/data can help in testing the application effectively.

ii) The other advantage of white box testing is that it helps in optimizing the code.

iii) It helps in removing the extra lines of code, which can bring in hidden defects.

The disadvantages of white box testing are:

i) As knowledge of code and internal structure is a prerequisite, a skilled tester is needed to carry out this type of testing, which increases the cost.

ii) It is nearly impossible to look into every bit of code to find out hidden errors, which may create problems, resulting in failure of the application.

White box testing is conducted by using security testing and database testing.

7.2.1 Components 1: Security Testing

Security testing is performed to determine the safety measures of user login activity. Information that is uploaded within this activity is unique and redundancy should not exist as its will affects the system performance. The security testing is performed as follows:

i) System developer choose an anonymous user ID and password

ii) Perform standard procedure of user registration

iii) Check the data uploaded

iv) Repeat the task using unique name and same name to imitate data redundancy

v) Check the result and determine whether the error is detected

vi) Verify the error detected

vii) Test results are reported and verified.

7.2.2 Component 2: Database Testing

Database testing is required to determine the accuracy and reliability of the database design. The steps below are taken for the testing purpose:

i) The data for testing is determined

ii) The data server is connected.

iii) The functional test is performed to ensure that update and edit function is correct.

iv) Check the test result output is correct.

v) Verify result with the information stored in database.

vi) Test result is reported and verified.

Overall, the black box and white box tests have solved three problems i.e. idle codes elimination, display and loading layer problem and finally the login/logout/exit session from GRPMIS. The designed system can be successfully launched with several arrangements, corrections and improvements. The user requirements were then validated and the results are shown in Table 1.

8. The Integration Issues and Problems

There are several issues and problems identified during the integration process. These issues and problems relate to the technical and hardware capacity, software availability and timeframe. For the creation of complex three-dimensional models and scenes, time is always a major concern. Time is required for preparing data, modeling buildings, and importing models into a scene. For larger datasets, computer processing and refreshing time may become unacceptably long. These considerations make the use of high-end machines critical. For example. the scenes used in one of the previous study were generated using an Intel Xeon (dual) 3.2 GHz processor with 2 GB RAM.

The hardware requirements are considered when the investigations of the software component were performed. The machine that is used during the investigation consists of a capacity of 80 GBytes hard disk storage, 1GBytes memory (RAM) and Intel Centrino Duo Core Processor 1.66 GHz. The storage, memory and processor capacity does implicate the effectiveness of the system developments. The main challenge occurs during the graphical processing, 3D design and also performing virtual interactive. As a result the higher memory and processor capacity of desktop are needed.

Large TINs demand a high amount of computer resource. Without a sophisticated computer the 3D integration will be sluggish or jerky during the rendering process in ArcScene. ArcScene does not support files greater than 300mb, so large areas of high resolution or very high resolution imagery are not always possible to view as drapes. Again, a maximum of 300,000 nodes are supported which have caused limitation for handling large extents of complex data such as road networks, hydrology networks or property parcels. Any large dataset will cause a significant performance issues and subsequently restricted some of the functions proposed for the system constructions mainly the application of 3D GIS. Table 2 summarized the advantages and disadvantages of the four methods of 3D integration discussed above.

Other issues that arose during the integration and development process was the availability of the ArcGIS Designer License and ArcGIS Developer Kit. License tools such as the TOC Control, Scene Control and Globe Control require the designer license and developer kit in order for them to be available for the execution in Microsoft Visual Basic and MapObject. This control carries a complete 3D data manipulation functions. The unavailability of these controls result in the limitations of the 3D GIS functions in the GRPMIS. Only SceneViewer Control is available to be used as the 3D GIS function in the developed system. Then again, there are also barriers identified as the programming language used in the developer kit is Visual Basic Application (VBA) and ArcObject is needed instead of MapObject as the companion module. ArcObject on the other hands still require the final user to have ArcGIS, designer license and developer in order to implement the system in their local machine/computer.

9. Conclusion

Generally, the work presented in this research shows that the integration of GIS technology into a property marketing information system can be performed. The integration of GIS technology also enables the house sellers and potential buyers to be independent in obtaining information about the house. Embedding 3D and virtual technology can be undertaken with the availability of the software needed for such integration. GIS is highly relevant to property marketing information system as real property is mainly about location. The new technologies, expansion in science, philosophical ideas and increasing number of populations drive to the complexity of the living environment. Business technology as well as other entities must be able to adapt constantly to the modernisation or be left in the dust of the society changes. The information system should intensify and strengthen the communication within the business activities. Internet has become a major communication tool used by commercial enterprise, and government agencies to support trade, operations, and interaction with customers and suppliers and in the property world by and large and property marketing in particular are no exception.

Today 3D GIS (Geographic Information Systems) applications are very useful not only for reducing the hardware costs and increased graphics processing capabilities but also for creating real-time 3D scenes directly from common geospatial and GIS data sources. In addition, there is an unprecedented amount of GIS data being accumulated and available from vendors, government agencies, data suppliers and users and much of this data is in the public domain. Transforming 2D map data into real-time 3D scenes allows users to extract more information from their GIS data. Although the integration of 3D and virtual reality to the GPMIS is not conclusive and contains some limitations, the initial results in this research can be used by the related agencies especially the real estate agents and property developers as the framework and references for future development of a better and conducive property marketing information system.

The development of GRPMIS underwent the standard system development lifecycle (SDLC). The black box and white box testing and evaluation are performed and they show that to complete the testing process is somewhat infeasible. Testing is complex and expensive as it requires expertise and longer time-frame. However, knowing that at some point the developed system has to be delivered, the testing process need to stop. The stopping point can be decided based on time-frame and budget or the developed system reliability meets the requirements. Besides the conventional system testing, the alternative method, such as system or software inspections and clean-room engineering are worth to be evaluated for testing purposes. This is an are for further research.

Finally, GRPMIS design and construction have established and proved that GIS, 3D and virtual technology can be integrated into a property marketing system. Although, the findings have stated some limitations, the research has found that there are technologies available to be integrated and to improve the efficiency of the existing property marketing system. The integration requires four main softwares i.e MapObject 3.2, Microsoft Visual Basic 6, ArcGIS 9 and Sketch-Up 5.

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Table 1. Succeeded User Requirements

No.	User requirements	Succeed
1.	The system should enable users to perform database query and view tabulated result.	/
2.	The system should allow user to perform location query and view projected map on the screen.	/
3.	The system should allow user to perform visual query and visualize query result on-screen and report.	/
4.	The system should provide user with the facilities information for the housing area.	/
5.	The system should allow user to obtain information regarding the housing property e.g. price, built in area, address, owner/developer, etc.	/
6.	The system should provide user with the topographic information of the housing area.	/
7.	The system should allow user to view inside and outside the house in 3D and perform virtual analysis (walk through/fly through and shadow).	The function works with some limitations
8.	The system should allow advertisement (permitted to certain users).	/
9.	The system should enable users to print out the report.	/

Table 2. Comparison of 3D Integration Methods

No.	Advantages/Disadvantages	Method 1	Method 2	Method 3	Method 4
1.	Elevation Model	/	/	X	/
2.	Complete layer properties	/	X	/	/
3.	Simple 3D Features Placement	X	Inconclusive	/	/
4.	Time-consuming	/	/	X	/
5.	Import Process meet Interruption	X	/	X	X
6.	Poor Visualization Performance	/	/	X	X
7.	Affects by machine capacity and memory	/	/	X	X
8.	Dragging Processes	/	/	X	X
9.	Orientation Issue	/	/	X	/

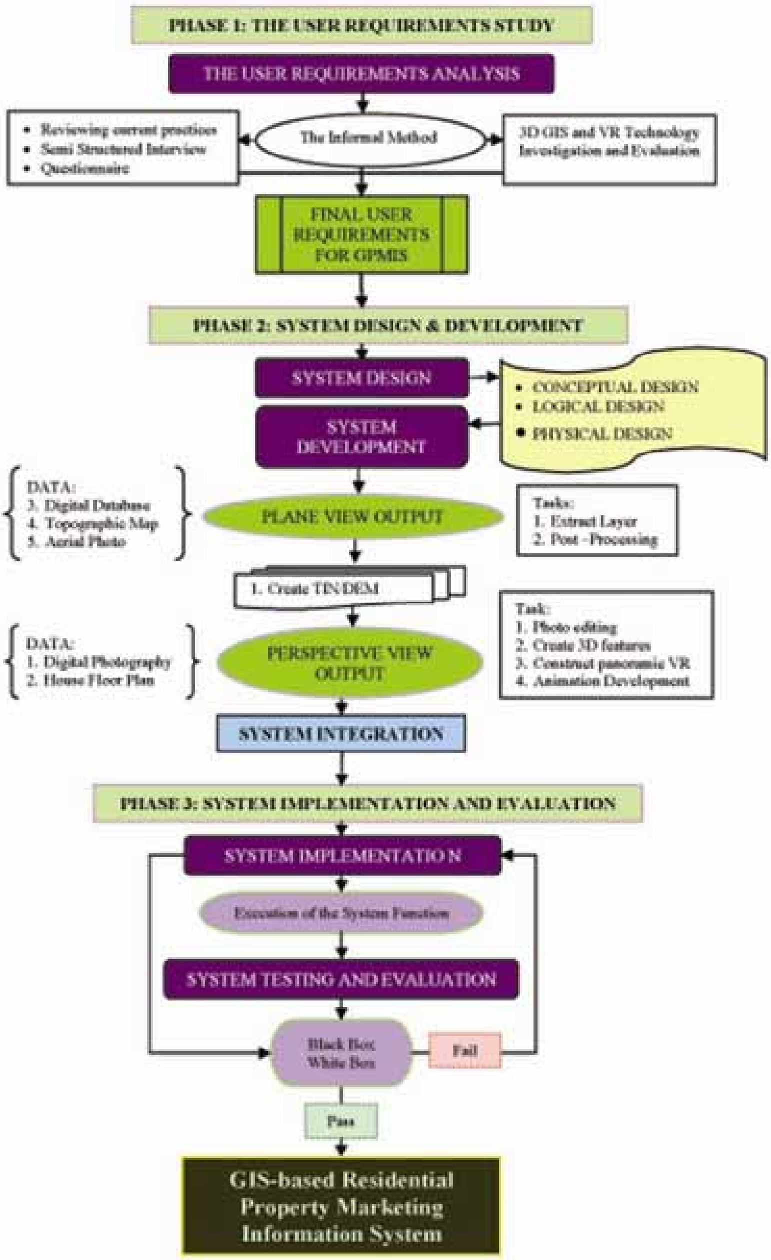

Figure 1. Research Methodology

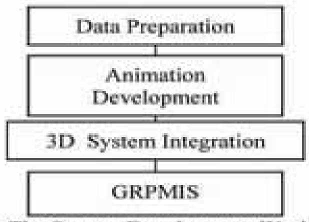

Figure 2. The System Development Workflow

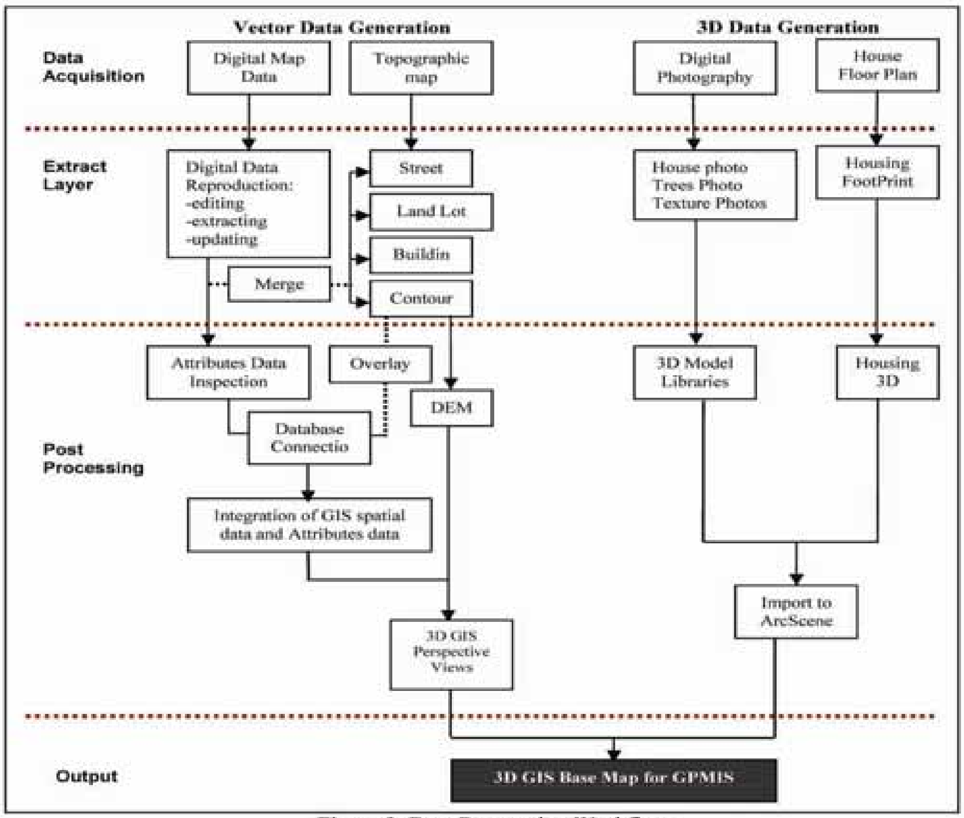

Figure 3. Data Preparation Workflow

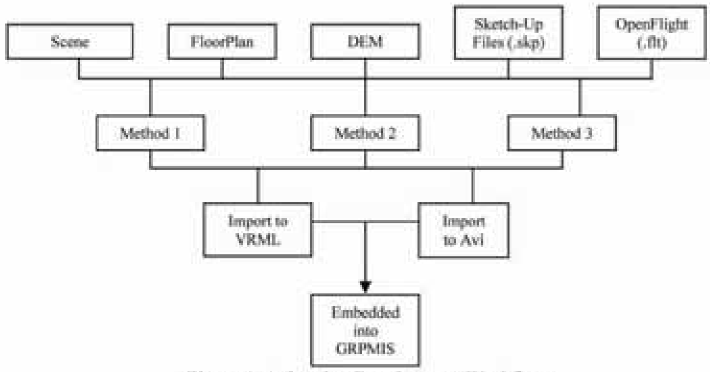

Figure 4. Animation Development Workflow

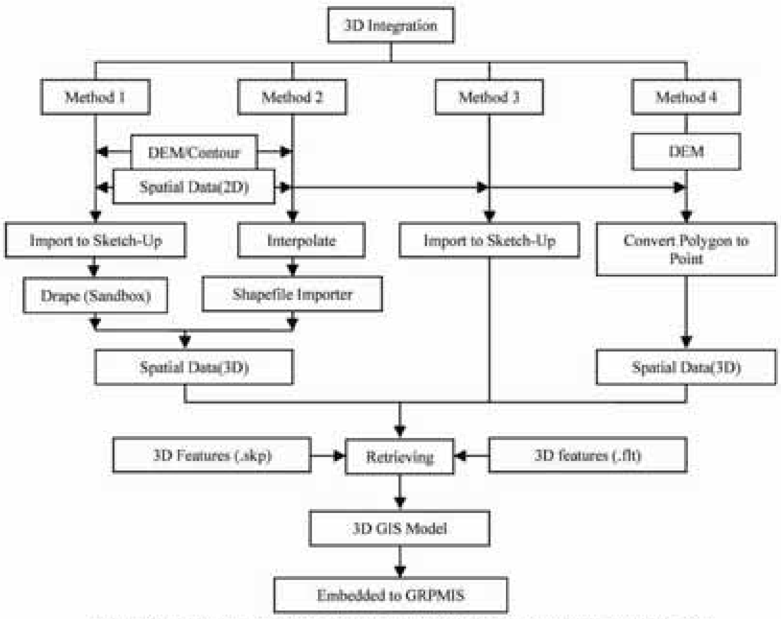

Figure 5. The Four Proposed 3D Integration Methods Examination Workflow

Figure 6. Starting Window

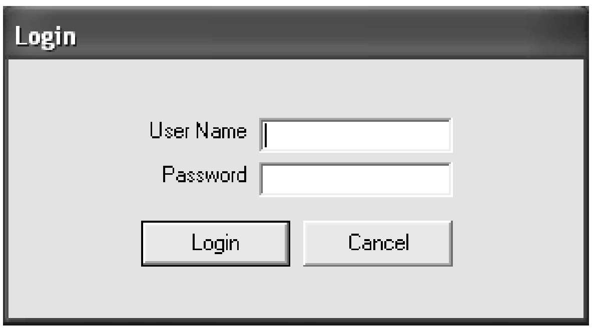

Figure 7. Log In Window

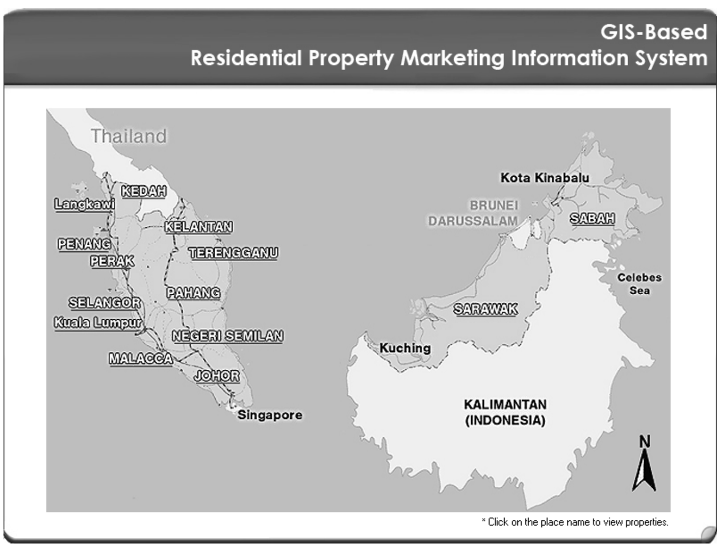

Figure 8. State Selection Menu

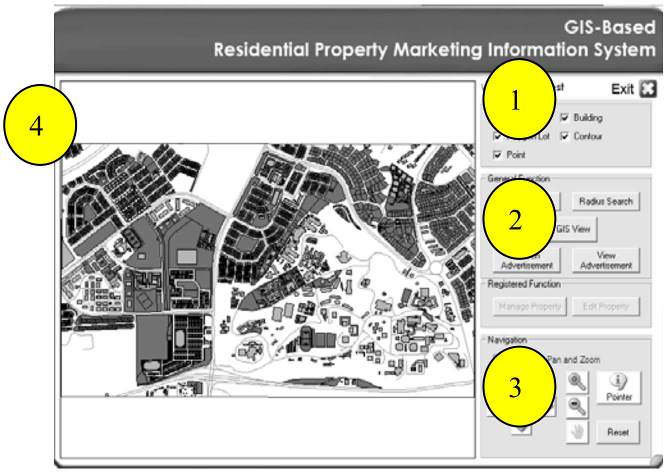

Figure 9. GRPMIS Main Menu

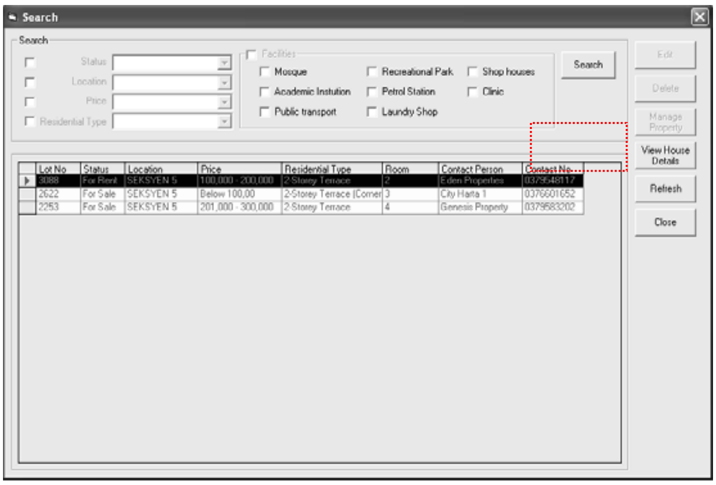

Figure 10. Search/View Advertisement Window

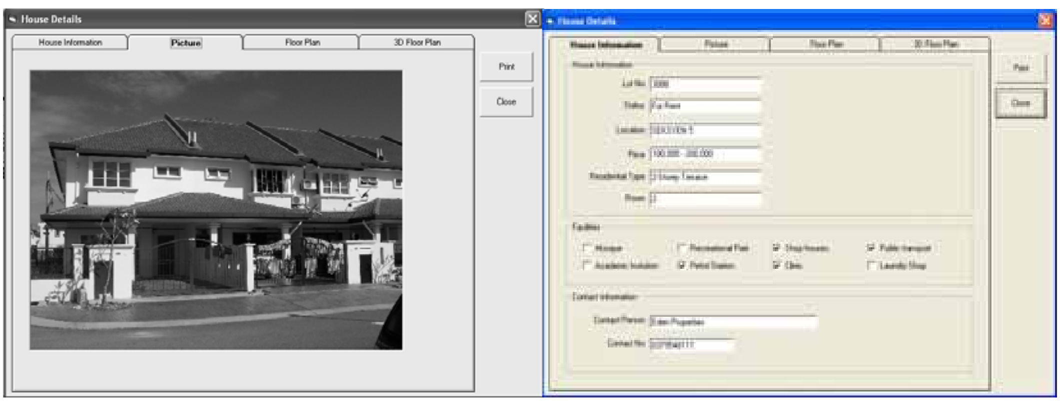

Figure 11. House Details Window (House Information Tab)

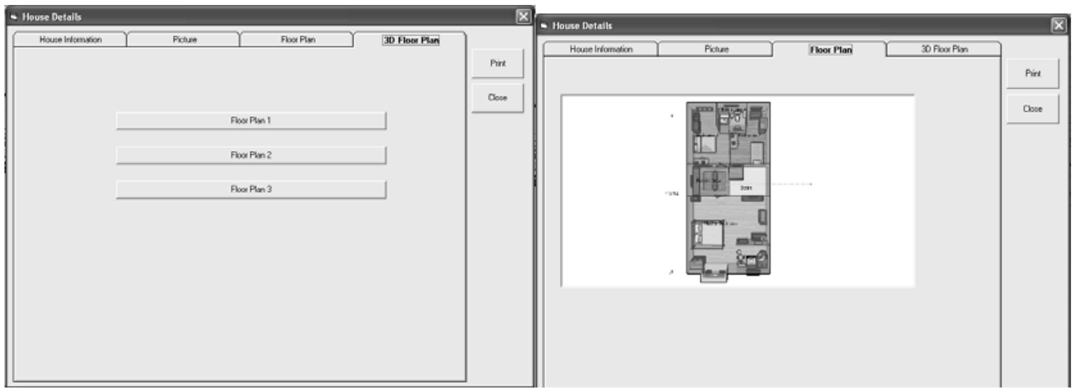

Figure 12. House Details Window (Floor Plan Tab)

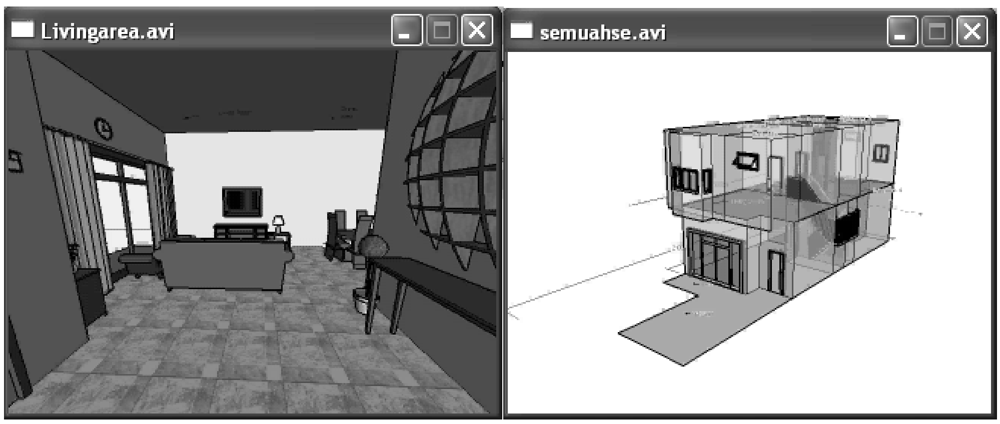

Figure 13. 3D Layout Plan

Figure 14. Access to Property Info from Pointer Menu

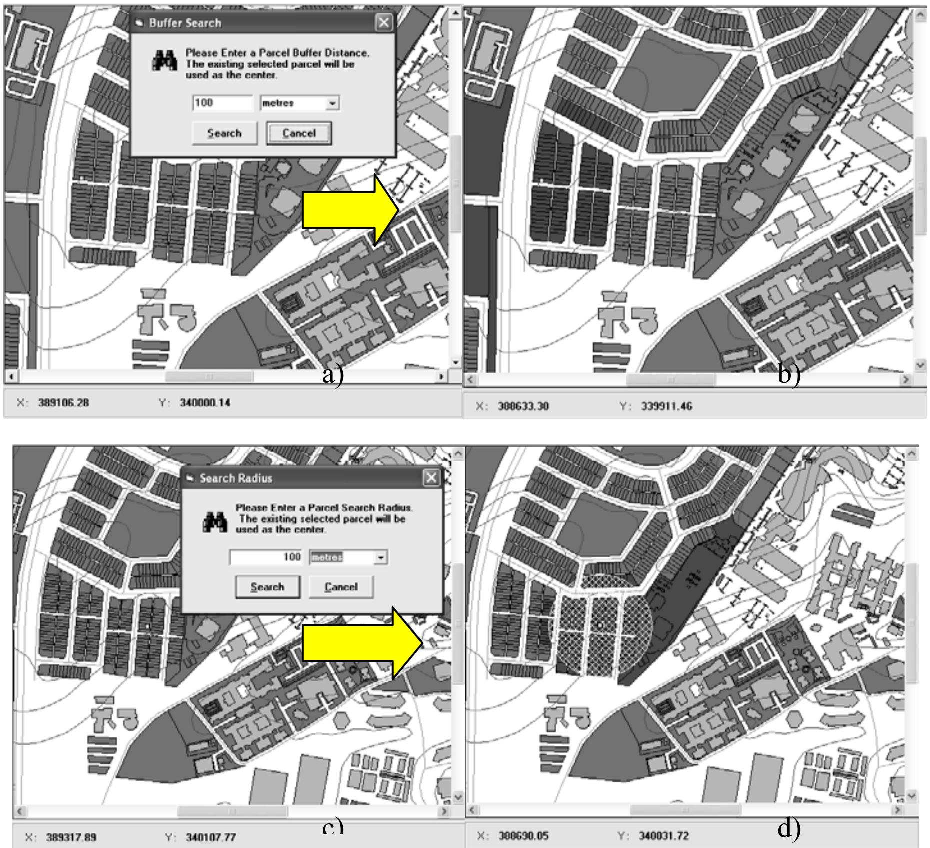

Figure 15. Buffer and Radius Function

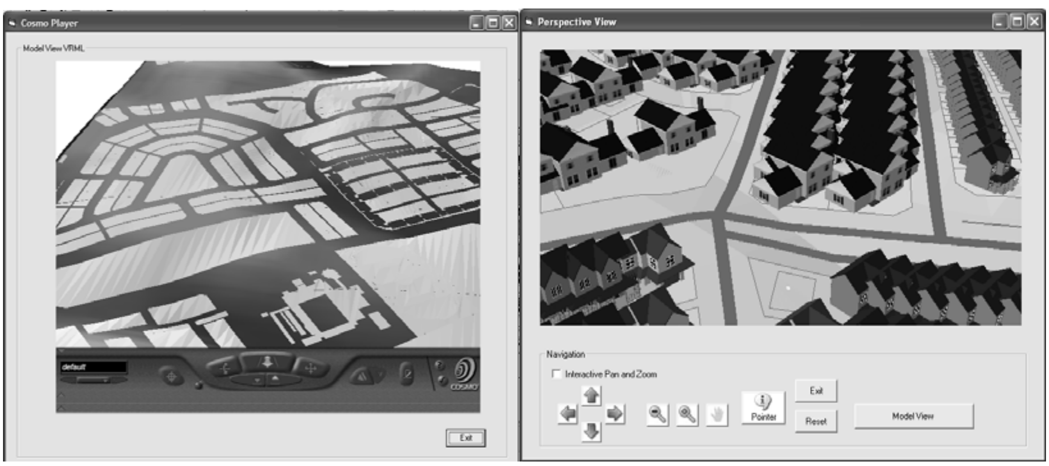

Figure 16. The Perspective View Window

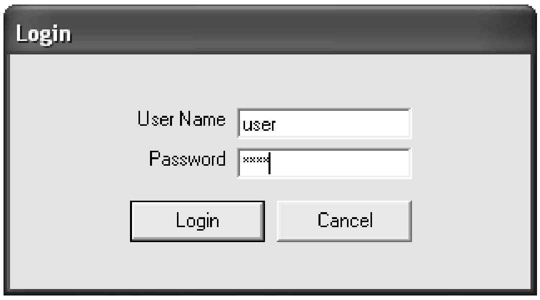

Figure 17. Registered User Login

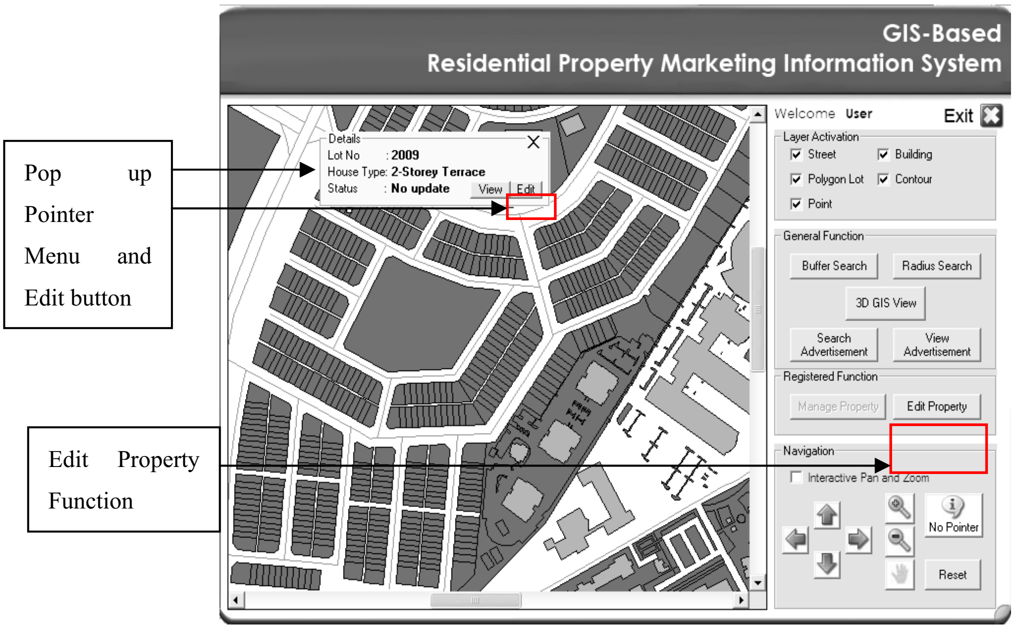

Figure 18. Registered User Main Menu

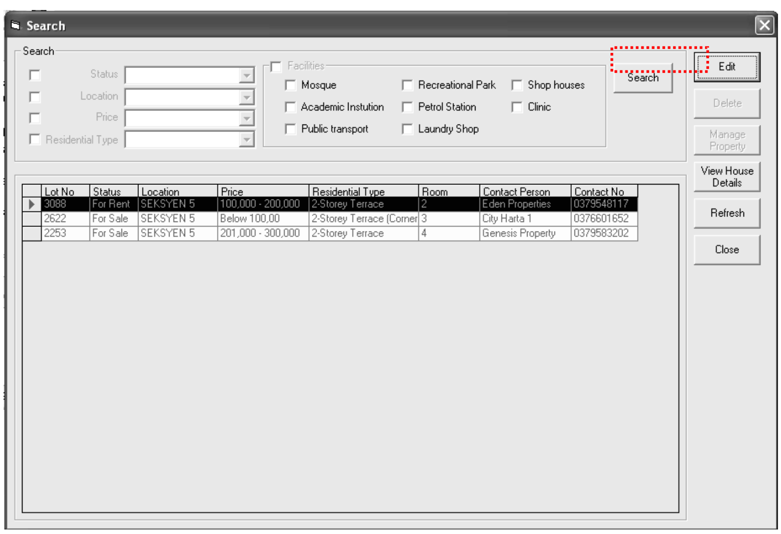

Figure 19. House Details Window/ Edit Button Activated

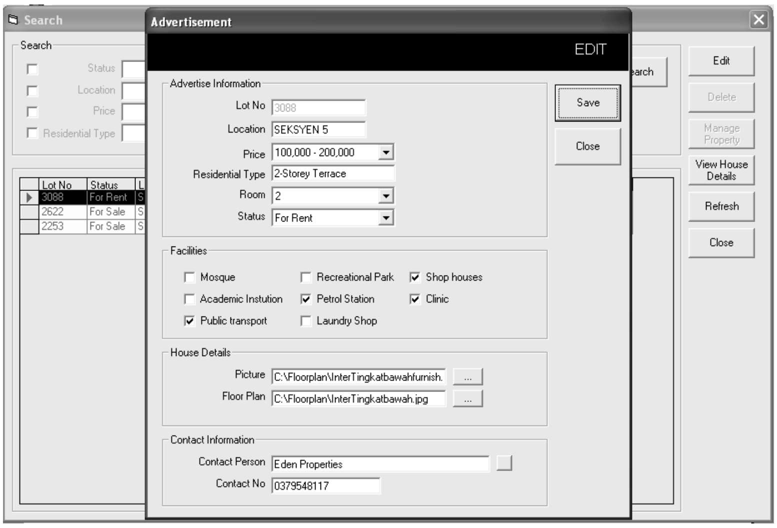

Figure 20. Advertisement (Edit) Window

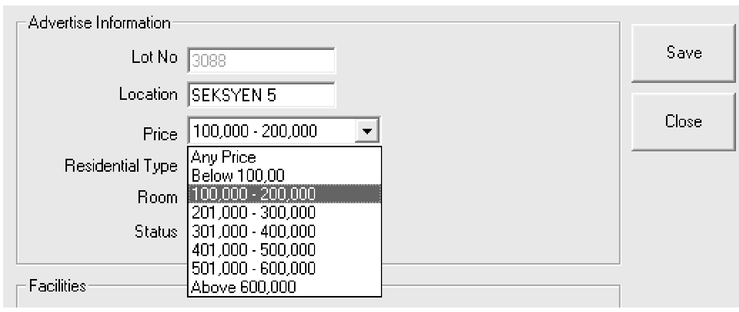

Figure 21. Price Range Drop-down Menu

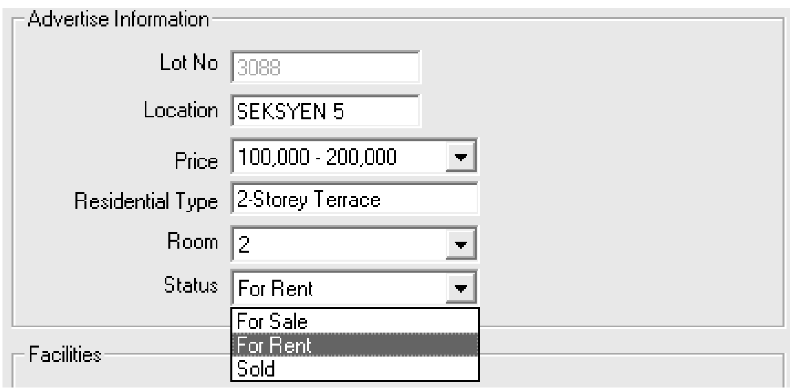

Figure 22. Status Drop-down Menu

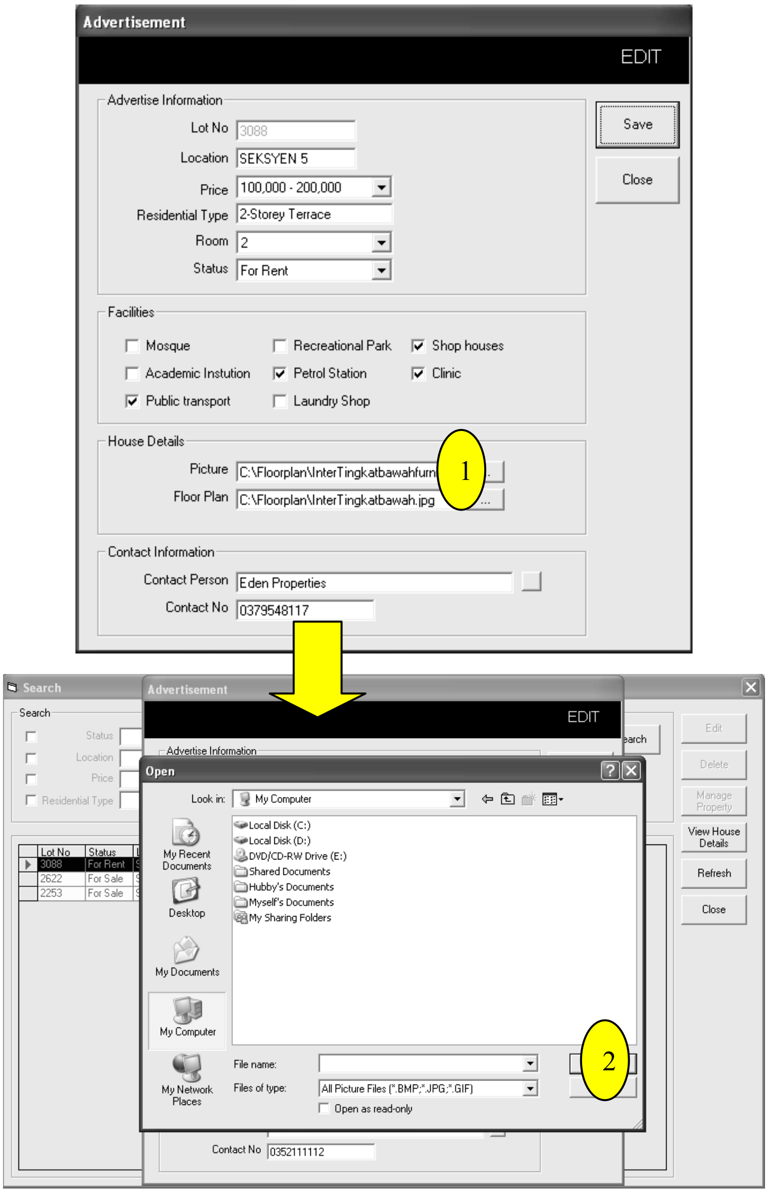

Figure 23. Housing Pictures Uploading Step

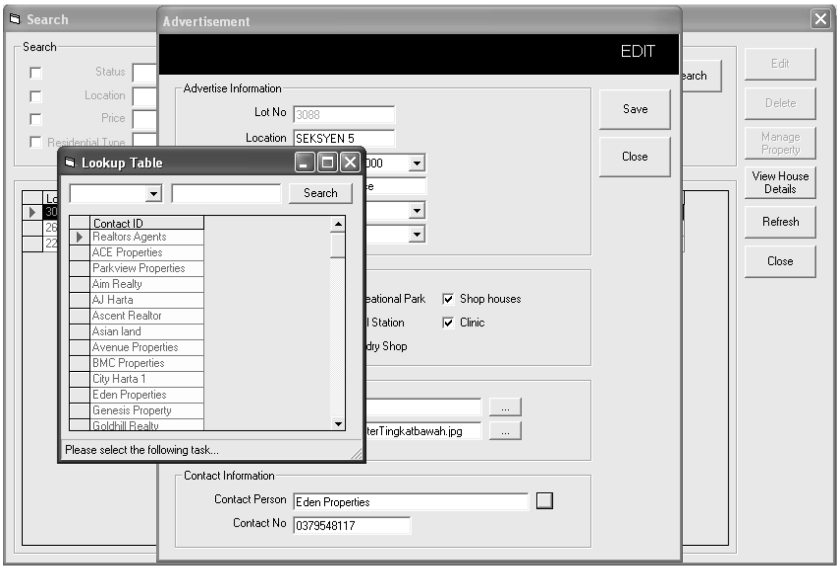

Figure 24. Contact Information Lookup Table

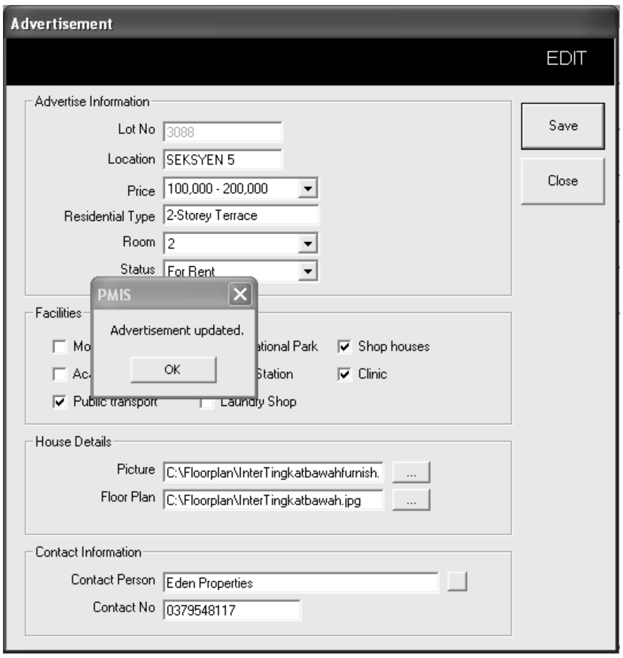

Figure 25. Information Updated Message

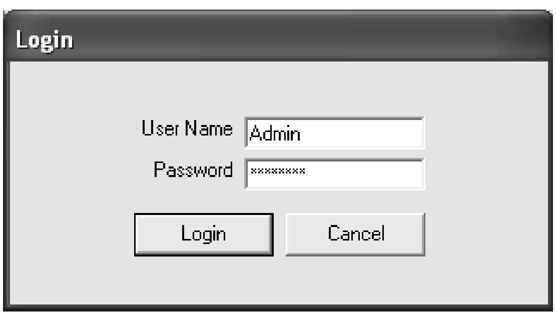

Figure 26. Admin User Login

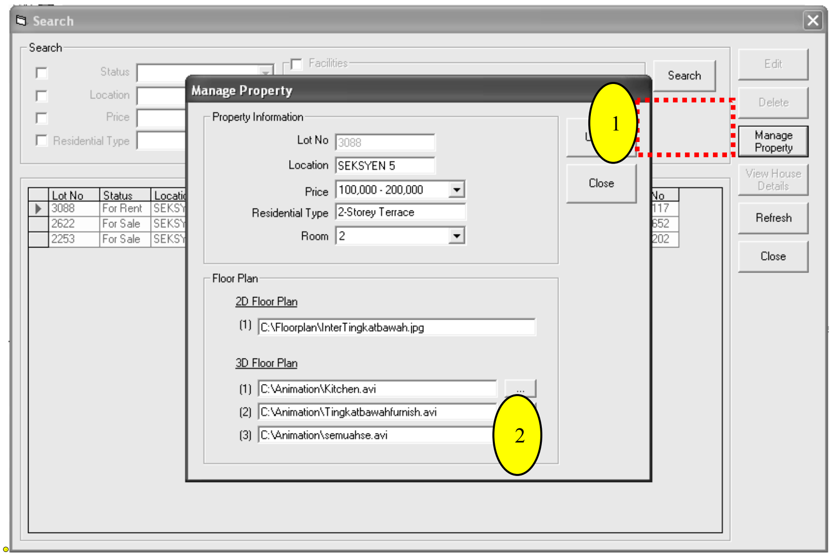

Figure 27. Manage Property Window

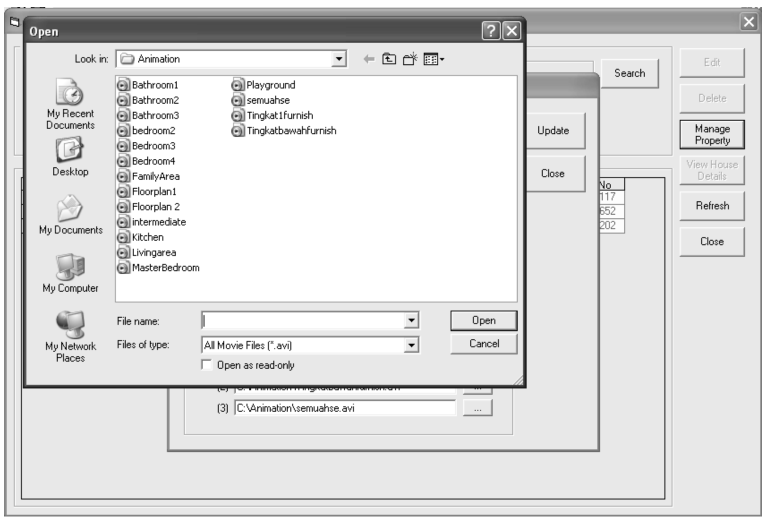

Figure 28. Open File Explorer

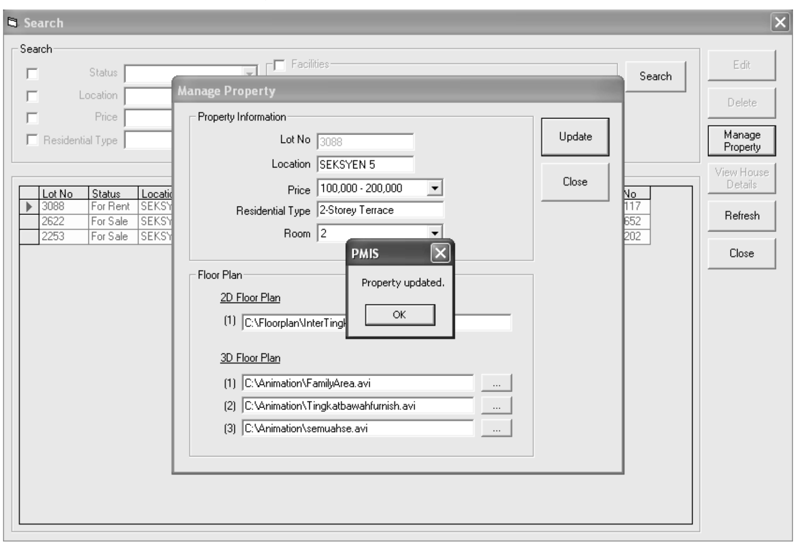

Figure 29. Property Updated Message

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Accelerated Routing Strategy of Application Service in the Multi-addressing Mode of College Campus Network

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Abstract

According to the situation that the multi-addressing universally exists in present domestic college campus network, taking the network of Henan University of Technology as the model, we put forward the accelerated routing strategy of campus application service interview under the conditions of multi-campus, multi-network supplier and multi-addressing.

Keywords: Bilingual teaching, Bilingual teaching material, Bilingual teachers, Circuit

1. Introduction

The construction of new school area and the combination of multiple school areas have been the multiple-school areas schooling mode universally adopted by various colleges, and the network construction of multiple school areas is implementing subsequently. Because of further extension and complexity of network size under the mode of multiple school areas, the information points increase quickly and the network flux multiples. And the interlink speed between CERNET and public net is the problem which should be solved for a long time, so the single network address only depending on CERNET becomes into the main bottleneck to limit the campus network flux. Therefore, when we construct the new school area for domestic colleges, we should introduce the support of the third party network except for CERNET such as CNC or Telecom. In the network environment with multiple school areas and multi-addressing, the former application service interview mode with single address has been broken, many colleges adopted different methods such as multiple domain, multiple service images and multiple chains DNS analysis to realize the application service utilization with higher speed and higher efficiency in the complex environment. Taking the network of Henan University of Technology as the model, we put forward a sort of simple and highly efficient application service routing interlink method based on generalized routing concept, which would further enhance the interview speed of application system, strengthen the security of application system, and optimize the network structure.

2. Description of network structure

Actual network structures of various colleges may be different, and the actual application mainly includes address position, bandwidth and the interview rationality of application service. The network structure based on multiple school areas and multi-addressing is seen in Figure 1.

Many school areas are interlinked by kilomega fibers. The addressing of old school area is the address of CERNET, and it is connected with CERNET through firewall. The new school area is linked with CNC by the router. The DNS address of old school area and the application service address of new school area are addresses identified by CERNET, and some addresses are unchangeable. Because the interview objects include campus network, CERNET and Telecom network, so the speed that users interview the application service in the school can be found directly by OSPF and be not influenced by the router. The problem we should solve is to try to let CNC users and CERNET users rapidly interview application service. Simple service offering only depending on the address of CERNET or CNC would limit the interview speed of another part of interview objects.

3. Concrete implementation strategy

To further save sources of software and hardware, the implementation method of concrete routing strategy is implemented based on following conditions such as the application service doesn't apply for multiple domains, the service image of hardware is not offered to the application service, the CNC IP addresses in the new school area are limited, so the application server is used to offer services through private network address.

3.1 Establishment of multi-link DNS service

Multiple chains DNS service is also called intelligent DNS service which is mainly used to distinguish the network types of interview users such as CERNET or Telecom network, and offer different IP analysis to same one domain according to user network type. There are many construction methods for the service and there are many products with mature hardware in the society. The intelligent DNS software used in the article is developed by us. Whether for hardware or for software, the service is mainly to distinguish different network types, and the network IP address segment with different types could be found in CERNET and Telecom and kept updating at any time. The explanation of intelligent analysis taking WEB service as the example is seen in Figure 2.

In the Figure 2, the networks with different types are distinguished strictly, and the interior network IP address is used for campus network users. From Figure 1, the addresses in the article only include CNC address, but for colleges with Telecom address, they can fill in the address of Telecom such as the address in Figure 2. The construction of concrete intelligent DNS also includes many contents.

3.2 Routing collocation

The routing collocation mainly aims at the operation of the router in the new school area of Figure 1, and it mainly solves the interview of public network (non-CERNET) to application service. The CNC addressing router of Henan University of Technology is the NE40 ten thousands high end router made by Huawei Company, and the address pool for exterior NAT conversion is 123.15.55.1/29. As seen in Figure 1, the routing conversion principle taking WEB service as the example is to take an effective address such as 123.15.55.10 to be the conversion address of users' interview. When interview users acquire effective CNC address through intelligent DNS analysis, the router NE40 convert the effective address in the address pool into the private network address such as 172.18.22.11 of WEB server through NAT conversion from exterior to interior. The concrete implementation process includes following approaches.

(1) Establishing the address pool

nat address-group cncmain 123.15.55.8 123.15.55.16 mask 255.255.255.240 slo3

(2) Establishing the routing control strategy and setting up the address gateway

nat-policy number test ip 123.15.55.31 nat address-group cncmain

(3) Establishing interview control list and associating them

rule-map intervlan test ip 172.18.22.11.220 0.0.0.0 any

flow-action test nat test service-class 4

(4) Reversing NAT

nat server protocol tcp global 123.15.55.10 inside 172.18.22.11 80

The reversing NAT is the core approach for routing conversion, and the port 80 is the service port from reverse conversion to application service, and we can add necessary service ports according to actual application when implementing above approaches, i.e. above orders can be repeated.

For the college with Telecom address, we can add the address of Telecom according to above setups, but the WEB server should be required to collocate double network card and double private network address, and we can take out an effective address in the address pool as the conversion address and point to another private network address. For campus users who interview application service, we can point to any one private network address.

3.3 Port image of firewall

In 3.2, we solve the problem of non-CERNET network interview application service. Because some special addresses of application system in the interior of CERNET must be used after enrollment on CERNET, so the address that CERNET interviews WEB service is fixed CERNET addresses such as DNS or WEB, and the intelligent DNS must be required to return an effective CERNET address such as 202.196.110.3. The concrete implementation is actualized through the port image of firewall.

(1) Bidirectional conversion between CERNET address with private network address on the firewall

202.196.110.3→172.18.22.11

(2) Setting up interview control on the firewall, and opening the interview limitation of any address to the private network address 172.18.22.11

Because present anti-firewalls have status supervision system with the character of anti-attack, so the interview user after conversion should be ensured to obtain correct answer of private network address, which is very important. The effective IP 202.196.110.3 of CERNET is only taken as a dummy address actually, and it is only used in address conversion, and the address is analyzed through intelligent DNS analysis. The answer routing should be appointed on the router through this approach, for example,

rule-map intervlan test ip any 202.196.110.3 0.0.0.0

flow-action to111 redirect ip 202.196.111.1 GigabitEthernet2/0/0

Where, the address of 202.196.111.1 is the port address of router and firewall on the firewall.

The port image is the special function of firewall, and it can be converted correspondingly according to actual situation in the process of implementation.

4. Conclusions

We solve the problem of application service interview speedup for users coming from different network types under the conditions of multiple school area, multiple network supplier and multi-addressing, and convert the address of application service to the private network address through NAT and port image, which further ensures the security and stability of application service and strengthen the anti-attack ability of application server. The strategy is being applied in the campus network of Henan University of Technology, and we achieve anticipated effect from the views of speed and security through half years' observation. But under the condition that the network structures are largely different, the routing strategy may not be the optimal choice, so we only put forward the method to offer references for further researches.

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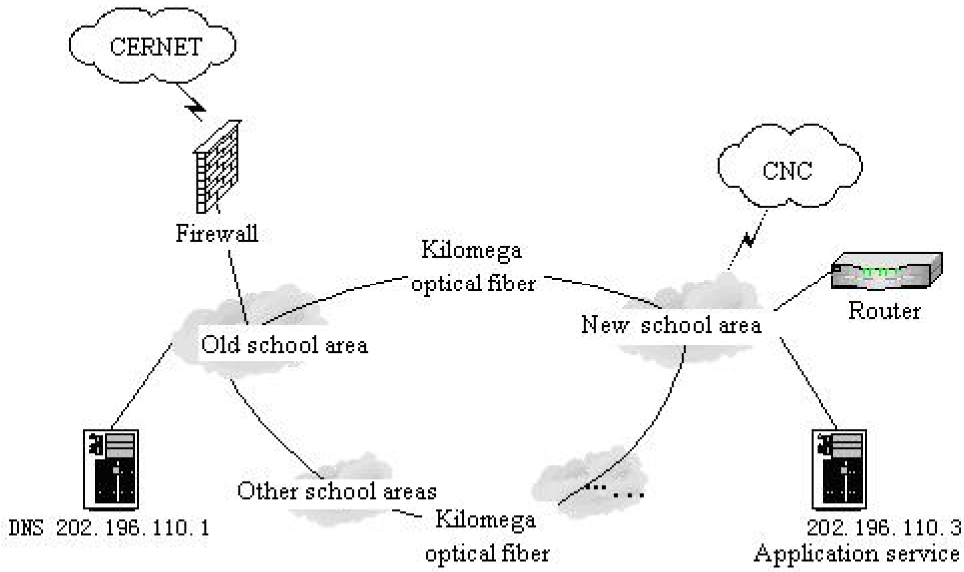

Figure 1. Network Structure Based on Multi-campus and Multi-addressing

Analyzed host computer	Campus IP	CERNET IP	CNC IP	Telecom IP
www.test.edu.cn	172.18.22.11	202.196.110.5	123.15.55.10	59.64.128.1

Figure 2. Intelligent Analysis

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Artificial Neural Networks for the Prediction of Thermo Physical Properties of Diacetone Alcohol Mixtures

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Abstract

A predictive method based on Artificial networks has been developed for the thermophysical properties of binary liquid mixtures of diacetone alcohol with benzene, chlorobenzene and bromobenzene at (303.15,313.15 and 323.15) K. In method 1, a committee ANN was trained using 5 physical properties combined with absolute temperature as its input to predict thermo physical properties of liquid mixtures. Using these data we found out the predicted data for intermediate mole fraction of different systems without conducting experiments. ANN with back-propagation algorithm is proposed, for Multi-pass Turning Operation and developed in MATLAB. Compared to other prediction techniques, the proposed ANN approach is highly accurate and error is <1%.

Keywords: Artificial neural network, Neurons (nodes), Perceptron, Network training, Diacetone alcohol

1. Introduction

In many real world applications, we want our computer to perform complex recognition problems. Since our conventional computers are obviously not suited to this type of field, we therefore borrow features from the physiology has come to be known as Artificial Neural Systems (ANS) Technology or Neural Networks. Artificial neural network is a branch of artificial intelligence (AI) that attempts to achieve human brain like capability. Traditional approaches of solving chemical engineering problems frequently have their limitations, as for example in the modeling of highly complex and nonlinear systems. Artificial neural networks (ANN) have proved to be able to solve complex tasks in a number of practical applications. The utility of artificial neural network models lies in the fact that they can be used to infer a function from observations. This is particularly useful in applications where the complexity of the data or task makes the design of such a function by hand impractical. Because ANN are nets of basis functions, they can provide good empirical models of complex nonlinear processes useful for a wide variety of purposes. The applications of ANN include detection of medical phenomena, stock market prediction, credit assignment, monitoring the condition of machinery and engine management. There are only few reports of using ANN in the prediction of physiochemical properties, these reports have generally been restricted to equilibrium rather than transport properties(John.et.al 1999). It is believed that so far there has been no attempt to truly predict the properties of liquid mixtures across their wide temperature range using ANNs. The present paper presents the findings of a programme of work devoted to the application of ANNs to thermo physical properties of binary mixtures. The thermo physical properties of binary mixtures are found by experimental work done by us. A predictive method based on Artificial Neural Networks (ANN) has been developed for ultrasonic velocity, density, kinematic viscosity, surface tension and refractive index for diacetone alcohol with benzene, chlorobenzene and bromobenzene at (303.15,313.15 and 323.15) K a wide range of mole fraction. The study of various properties like viscosity, refractive index, density, surface tension and ultrasonic velocity for different systems are very important to understand the molecular interaction, purity of compounds. Using these data we found out predicted data for intermediate mole fraction of different systems without conducting experiments.

2. Working procedure

A Neural Network is an interconnected assembly of simple processing elements, *units* or *nodes*, whose functionality is loosely based on the animal neuron. The processing ability of the network is stored in the inter-unit connection strengths, or *weights*, obtained by a process of adaptation to, or *learning* from, a set of training patterns. It has been shown that non linear feed forward neural networks are capable of universal functional approximation and that a single hidden layer is sufficient to uniformly approximate any continuous function Hornic.et.al.(1989). The neurons in a single hidden layer tends to interact globally but in complex functions this interaction makes it difficult to improve the approximation Heykin (1994), Maren .et.al (1990). The brain is principally composed of a very large number (circa 10,000,000,000) of *neurons*, massively interconnected (with an average of several thousand interconnects per neuron, although this varies enormously).

2.1 Artificial neurons

To capture the essence of biological neural systems, an artificial *neuron* is defined as follows:

- It receives a number of inputs (either from original data, or from the output of other neurons in the neural network). Each input comes via a connection that has a strength (or *weight*); these weights correspond to synaptic efficacy in a biological neuron. Each neuron also has a single threshold value. The weighted sum of the inputs is formed, and the threshold subtracted, to compose the *activation* of the neuron
- The activation signal is passed through an activation function (also known as a transfer function) to produce the output of the neuron.

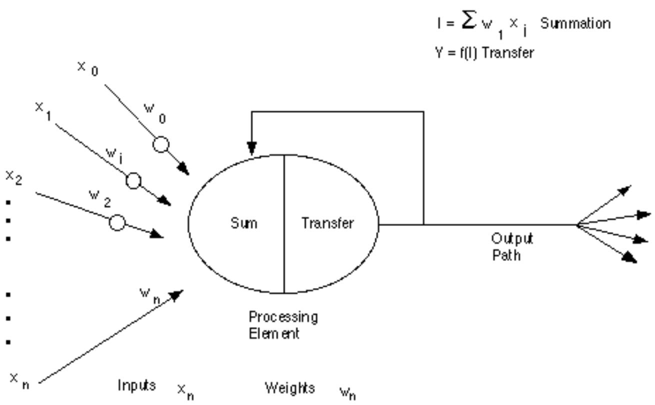

Figure 2.1 A Basic Artificial Neuron.

In Figure 2.1, various inputs to the network are represented by the mathematical symbol, x(n). Each of these inputs is multiplied by a connection weight. These weights are represented by w(n). In the simplest case, these products are simply summed, fed through a transfer function to generate a result, and then output. This process lends itself to physical implementation on a large scale in a small package. The commonest type of artificial neural network consists of three groups or layers of units: input, hidden, and output. The layer of input neurons receives the data either from input files or directly from electronic sensors in real-time applications. The output layer sends information directly to the outside world, to a secondary computer process, or to other devices such as a mechanical control system Necat(2006). Between these two layers can be many hidden layers. These internal layers contain many of the neurons in various interconnected structures. The inputs and outputs of each of these hidden neurons simply go to other neurons.

2.1.1Prediction Of Thermo physical Properties

Step 1: Collect data:

Things to be kept in mind while choosing the input data .

- ✓ The variables that are influential must be chosen
- ✓ Numeric and nominal variables can be handled. Convert other variables to one of these forms, or discard.
- ✓ Hundreds or thousands of cases are required; the more variables, the more cases.

Step 2: Define a network structure

An appropriate network topology is selected. Here for prediction problem, the neural network developed is fully connected feed forward multilayer perceptron. For this problem, the input variables are Temperature and mole fraction.

So, the number of Input nodes is 2. Here, we have to predict the density, kinematic viscosity, ultrasonic velocity, surface tension and refractive index of the systems. Hence, the number of output nodes is 5. After a number of experiments, the transfer function of the hidden layer and the number of epochs are set, the details of which are given below. For example consider a feed forward net with 30 hidden nodes in a single layer.

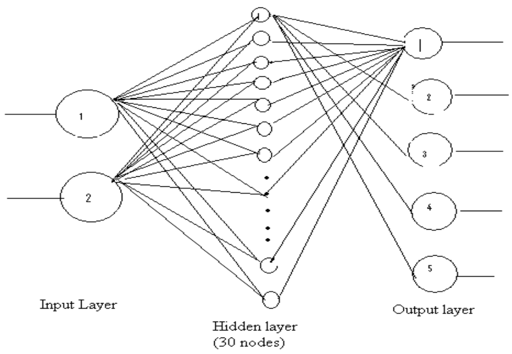

Figure 2.1.1 Feed forward net work

Step 3: Random initialization of weights and biases for the defined network

Step 4: Back propagation learning Mandic (2001), Reilly (1982) algorithm is selected for training the network

Step 5: Training parameters are set

Number of iterations: 600

Performance goal : 0

Step 6: Start training

During training, the input and output data are presented to the network.

Step 7: Termination

The error for the network is calculated. The error calculation and weight updating process continues until the neural network reaches the performance goal or number of iteration.

3. Results and Discussion

Experimental thermo physical property values were extracted from the data base for all the mixtures studied. The list of binary liquid mixtures studied were displayed in Table 1. Input and output variables are listed in Table 2. Random experimental verification of ANN predicted thermo physical properties (not included in the training data) of mixtures with percent standard deviation are listed in Table 3. The thermo

physical properties were density, viscosity, refractive index, surface tension and ultrasonic velocity. The input to the ANN consisted of five thermo physical properties of three binary mixtures at (303.15,313.15 and 323.15) K . The out put from the ANN consisted of five predicted thermo physical properties of three binary mixtures. A BPN simulator is designed and the input data were fed in. BPN simulator is trained several times using selected data from the collected data, which consists of normal as well as abnormal data. During training, the simulator is presented with both input and output pairs and the error is generated which is the difference between actual and desired output. The error is minimized using the steepest descent technique. When the error obtained is of acceptable value, then the simulator is said to be trained. Then, the data for prediction is presented to the neural network after training. The mean square error is calculated using BP algorithm and the learning curve is plotted between the mean square error and the number of generations (iterations) i.e. epochs. Performance and error graph of the trained committee ANN for the binary systems are presented in Figure 3.1- 3.3 While examining the results of these various neural networks, they were found to be in agreement with the desired results and within permissible error range. The algorithms traingdm or traingd are not producing satisfactory results for the 2 X 20 X 10 X 5 neural network for the maximum epochs of 800 and for a performance goal of 0. The trained algorithm with a single hidden layer with 30 nodes i.e 2 X 30 X 5 neural network.

(Figure2.1.1) is producing the desired result for the prediction of these properties. The predicted values of other new mole fractions of the mixtures were taken and verified experimentally. The percentage of error is <1.

4. Conclusion

Due to high speed of processing, low consumption of memory, great robustness, possibility of self learning and simple incorporation into chips the approach ensures prediction condition in real time. It provides robust representation clue to the fault-tolerant nature of neural networks. Our future proposal is Extending the proposed idea of ANN prediction to systems at extremely high or low temperatures where conducting of experiments are difficult

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Table 1. Details of the systems studied

Sl.no	**systems**	**Temperatures**	**Net work topology**	**Training algorithm**
1	Diacetone alcohol + benzene	303.15, 313.15 and 323.15 K	Feed forward multilayer perceptron	Back propagation learning algorithm
2	Diacetone alcohol + chloro benzene			
3	Diacetone alcohol + bromobenzene			

Systems with temperatures, network topology and training algorithm used

Table 2. Input and output variables

Method	
Input	**Output**
Mole fraction Experimental five Thermo physical properties	Predicted five thermo physical properties

Input and output of this method

Table 3. Random experimental verification of ANN predicted thermo physical properties (not included in the training data) of mixtures with percent standard deviation.

x_1	T/ K	ρ (pre)	S %	η (pre)	S %	n_D (pre)	S %	σ (pre)	S %	U (pre)	S %
Diacetone alcohol + benzene											
0.1500	303.15	0.8885	0.15	1.1111	0.50	1.4876	0.62	30.10	0.11	622	0.87
0.5000	313.15	0.8821	0.21	1.6724	0.97	1.4517	0.84	24.74	0.45	785	0.99
0.9000	323.15	0.8944	0.71	2.4444	0.82	1.4266	0.41	26.82	0.13	1000	0.21
Diacetone alcohol + chloro benzene											
0.1500	303.15	1.0621	0.14	1.1225	0.29	1.5331	0.65	33.10	0.68	1228	0.99
0.5000	313.15	0.9887	0.24	1.8111	0.88	1.4872	0.98	30.80	0.14	1004	0.00
0.9000	323.15	0.9200	0.47	2.4889	0.68	1.4299	0.67	28.29	0.47	1078	0.14
Diacetone alcohol + bromobenzene											
0.1500	303.15	1.4001	0.54	1.3221	0.21	1.5321	0.18	33.31	0.64	1188	0.31
0.5000	313.15	1.1844	0.11	2.1254	0.31	1.4999	0.00	32.01	0.00	1149	0.22
0.9000	323.15	0.9520	0.44	2.6221	0.12	1.4384	0.00	29.11	0.98	1118	0.31

Diacetone alcohol with benzene, chlorobenzene and bromobenzene mixtures

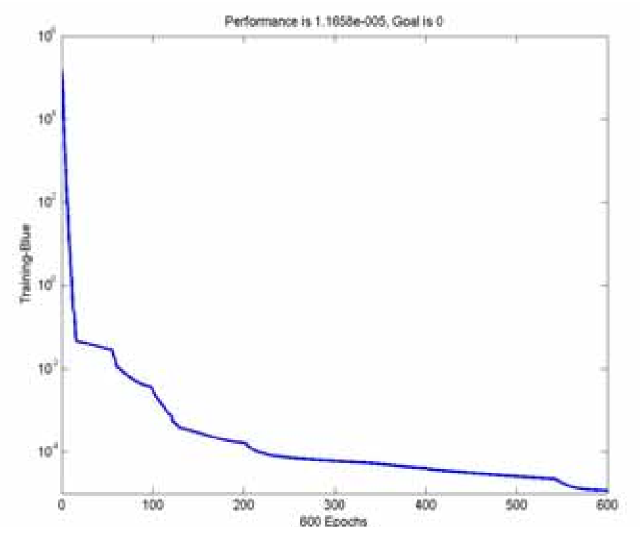

Figure 3.1 Plot of error and number of iterations

Error performance for Diacetone alcohol + benzene mixture

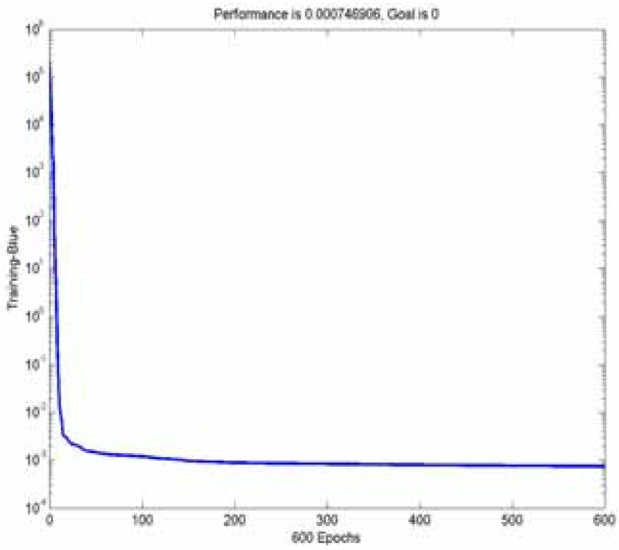

Figure 3.2 Plot of error and number of iterations

Error performance for Diacetone alcohol + chlorobenzene system

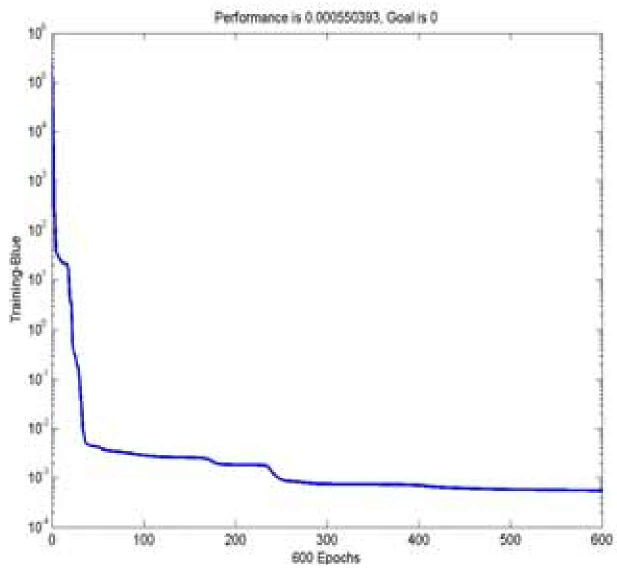

Figure 3.3 Plot of error and number of iterations

Error performance for Diacetone alcohol + bromobenzene system

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Design of the Automatic Spreader Control System Based on Embedded System

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Abstract

The control system of the traditional automatic spreader with complex structure and inconvenient servicing is designed based on MCU, and the new control system is designed based on the embedded system. Based on the analysis of the work process and work principle for the automatic spreader, we put forward the new scheme to improve the design of the control system for traditional spreader and design the hardware structure and relative software. The new control system can not only actualize the automatic control for the spreader and possess many functions such as spreading tier setup and automatic cloth edge alignment, but also possess the functions including network and system extension and effectively reduce the price of automatic spreader, and the improved spreader will possess stronger functions, more convenient operation and simpler maintenance.

Keywords: Computer application, Clothing, Control, Embedded system, Automatic spreader

1. Introduction

In the garment production, the sewing processing is implemented after spreading and cutting. The spreading needs 50~200 tiers, and its length is about 5~13 meters, and the human operation needs 4~6 people, and the time is about 1~2 hours, and the labor intension is large and the production efficiency is low. It only needs one people to complete the spreading work by the automatic spreader, so the automatic spreader can significantly enhance the production efficiency, effectively reduce the production cost and increase the competition force for clothing enterprises. At present, most automatic spreaders used in China are introduced from foreign countries, and because the price of the equipment is higher, so only few large-sized enterprises can use automatic spreaders. Therefore, it is very meaningful to develop and popularize automatic spreader for such large-sized clothing countries as China.

Because of the limitation of technology, the control system of traditional automatic spreader is actualized by MCU and middle and small sized logic circuit, and the system design is complex, and failures often occur, and the operation speed is low, and the equipment function and the extended ability are worse. In recent years, the quick development of electric technology brings large convenience for the research and development of product, and for example, the price of the 32 bits embedded system only corresponds to the price traditional 8 bits MCU, but the operation ability can achieve thousands MIPS, and the development of the ASIC technology represented by CPLD and FPGA technology (Jia, 2005, P.270-281) largely simplifies the circuit structure, and the operation speed and product performance can be largely enhanced, and at the same time, the ISP technology of the embedded system and ASCI technology can actualize the translation from the hardware design of product to the software design, and largely reduce the research and development cycle of product. In addition, the popularization of the AC servo motor represented by vector control can offer stable base for the exact control of the equipment, and at the same time, with the enhancement of system operation ability, many complex control algorithms such as Kalman filter, Fast Fourier Transform (FFT) and fuzzy control have entered into the application stage, which can offer the reliable operation for the system.

2. Function analysis for control system

The work of spreader must accord with the requirement of cutting technology, and implement spreading according to the layout length and the planned cutting quantity (Sun, 1994, P.1-3). The automatic spreader should possess functions such as treading, cloth coiling, work state display, cloth edge automatic alignment, automatic rising of cutting knife, cutting, automatic counting and terminal checking, and realize the tension free spreading. So the control system function requirements of spreader should include following aspects.

2.1 Setup of spreading parameters

Before spreading, the spreader should set up spreading length, spreading tiers, spreading mode (such as face to face spreading and one way spreading which are seen in Figure 1) and other work parameters according to production requirement, and when the actual spreading length is set up, the length should increase 1~2cm than the plate length obtained through the layout length, and it should be taken as the remnant increment. In the operation, the automatic spreader can automatically compute the spreading length and remnant wastage through the control system, and automatically add up to the needed spreading tiers.

2.2 Automatic adjustment of tension and edge alignment

According to the technical requirement, the tension can not occur in the cloth when spreading and the spreader could automatically adjust the tension induced in cloth feeding. In addition, to convenient for cutting, the cloth edges should be automatically realigned.

2.3 Automatic rise of knife rest

With the continual increase of spreading tier, the spreading position is higher and higher, and the bracket of the round-bladed cutting knife to support the break could automatic rise.

2.4 Cloth edge checking

After one reel cloth is spread, the spreader could check the cloth edge to replace the cloth in time. The spreader could install one reel new tubular cloth on the cloth bracket from the cloth table, and the spreader restarts.

2.5 Display of work state

In the spreading process, the spreader could display the evolvements and states of the work such as spreading tier, spreading mode and spreading length.

3. Hardware design of control system

The automatic spreader generally includes treading device, cloth loosing device, cloth feeding device, press cloth bracket and other devices, and it is a sort of highly automatic garment equipment, and its hardware structure includes the mechanic part and the control part, and its work principle is seen in Figure 2.

The control system checks the operation state of the equipment through sensors, and controls the electric execution machines of the equipment, and actualizes the automatic operation of the whole equipment.

3.1 Hardware structure

To actualize the work requirement of the spreader, we should first implement the structure design of the control system, which mainly control various motors and sensors in the spreader through microprocessor (Chen, 2005, P.39-42). The motors used in the spreader include treading drive motor, cloth loosing drive motor, cloth feeding drive motor, round-bladed cutting drive motor, cutting position drive motor, knife rest lifter driver motor and cloth alignment drive motor, and the sensors include cloth edge alignment sensor, terminal checking sensor and tension checking sensor, and the hardware structure of the control system is seen in Figure 3.

3.2 Processor selection

There are many kinds of processor, and their performances and prices have large differences, and proper processor decides the hardware structure of the whole product. At present, ARM9 series embedded processors possess 32 bits operation ability, powerful function and abundant interfaces, and they can realize the functions of the traditional large-sized control system, even workstation based on lower costs (Wayne, 2002, P.30-37). Therefore, the control system of the spreader adopts S3C2410 embedded processor with the kernel of ARM90T. S3C2410 not only has quick operation speed, but offer abundant human-machine interface and conveniently actualize the extension and maintenance interfaces of the equipment.

3.3 Structure of peripheral circuit

The main control circuit needs processor and abundant peripheral circuits to realize the whole function (Zhuge, 2003, P.104-105), and S3C2410 integrates TFT display interface, touch screen interface, serial interface, USB interface and A/D interface used to sensor sampling. With the development of electric apparatus technology in recent years, the function of CPLD is highly enhanced and the cost reduces with large extents, and the CPLD adopted in the design largely simplifies the circuit design, and both the CPLD and the S3C2410 processor could offer functions such as online diagnosis and JTAC interface, the pin definition of the apparatus, and interior logic, and these functions could be modified in time through the computer, and the circuit design can be largely simplified, and the development process of the circuit will be evolved as software transfer process to overcome the repeat board-projects in the traditional circuit design, and the once design and once modeling of the circuit become possible, which can not only save the research and development costs, but effectively reduce the product research and development circle. The execution motor drive, various relative protections of the equipment, the logical disposals of various circuits and the operation with high timing and reliability in the main circuit of the spreader are realized by CPLD, so the operation speed and reliability of the system can be fully ensured.

3.4 Equipment interface

Except for automatic spreading, the spreader also offers convenient human-machine interface and abundant equipment interfaces. The human-machine interface mainly includes 15 inch colorful liquid crystal (TFT), operation keyboard and touch screen. The extension and maintenance interfaces mainly include network interface (ISO 8802.3), serial interface, USB interface and JTAG interface (IEEE 1149.1).

The network interface offers strong extended ability for the equipment, and it can supervise the operation of the equipment and actualize the automatic management in the production through the network, and offer reliable base for the equipment and subsequent extension and update.

The JTAG interface could offer hardware maintenance and updating interface for the equipment, and the JTAG interface could quickly judge the operation state of the hardware system and diagnose the failure of the equipment, and at the same time, the JTAG interface could update the supervision software of the embedded system and the circuit logic of CPLD could complete the function update of the equipment.

3.5 Selection of execution motor

The common control motor includes DC motor, stepper motor and AC servo motor. Because of the limitation of brush structure, the DC motor has serious electromagnetic interference and bad reliability. The stepper motor controls simply and could realize certain precision, but the step loss will occur when the opening-loop control of the stepper motor has heavy load, and the mechanic vibration will occur when the speed is low, but the deficient moment will occur when the speed is high. The closed loop control of AC motor has high precision, quick start speed, stable operation and stable interior moment output with rating rev which are ideal performances of the control motor. In the traditional design, the servo motors are rarely adopted because of high costs, and with the development of the control technology, the cost of AC servo motor quickly reduces, and the AC servo motor tends to replace other control motors, so the motor in the spreader selects AC permanent magnet servo motor which can effectively improve the operation status of the equipment and enhance the equipment efficiency and control precision.

4. Software design of control system

The software design of the main control board composed by embedded system CPLD mainly includes operation system collocation, application development and driver development.

4.1 Collocation of operating system

The prevalent embedded operating system mainly includes Linux, μC/OS-Ⅱand Windows CE (Wang, 2002, P.81-124, Li, 2003, P.20-23 & Li, 2005, P.59-61). The software developed in the Windows CE system has bad stability and complex structure. The Linux system has high opening degree and stable and tidy kernel, so it is the prevalent operating system in present embedded system. According to the demand of spreader control system, we simplify the Linux operating system, leave useful models and cut down useless models. The operating system and application software are stored in the ROM of main control board.

4.2 Drivers

The drivers are the extensions of the operating system aiming at concrete hardware, and the hardware is disposed and the standard transfer interface could be offered through drivers. The equipment drivers are integrated in the kernel of the operating system, and they have higher privilege level, memory-resident and sharing. According to the characters of the spreader, the drivers mainly include sensor driver, motor driver, display driver, touch screen driver, network driver, USB driver, serial port driver and electric disk driver.

4.3 Application

The application includes equipment operating interface, operation parameter management module, real time control module, network management module and communication module (Wang, 2002, P.81-124 & Huang, 2005, P.44-46). To better organize the software system, we adopt the multiple task work mode, and the spreader is divided into equipment operating management course, control course, network management course and communication course according to concrete application characters. Various courses operate independently and harmonize each other through the information structure. The key technology in the application includes two aspects.

(1) Sensor driver and signal processing

The sensor translates the physical parameters actually inspected into simulative electric parameters, and translated them into corresponding digital signals through A/D transform. Because the collected signals not only include useful status parameters and various interferences enter into the system together, and these interferences are very harmful to the precision and reliability of the equipment, so we should adopt various measures to eliminate these interferences when the date are collected. The Kalman filter has good filter performance and few signal delay. Various mechanistic resonances and periodic electric interface will occur in the running of spreader, and through the system stability analysis of FFT, we can eliminate system interferences through corresponding measures and ensure that the signals collected by sensors are stable and reliable.

(2) Running control of the equipment

The running control of the equipment includes parameter setup, status display, servo motor driver and dynamic inspection and other modules. We first should set up spreading parameters (including spreading length, spreading speed, spreading tier, spreading mode and remnant increment) according to the requirement of the production technology before spreading. When the tubular cloth loading is completed, the spreader begins to tread, and in the spreading process, the spreader will inspect the cloth edge, cloth tension and cloth edge alignment and adjust the situation in time, and when the tubular cloth uses out, the spreader will automatically stop to replace until to the required spreading tiers. In the spreading process, the spreader could automatically account, and display the spreading tiers and spreading mode real time. The driver of servo motor needs considering the design of the lifter speed and port output, for example, when every tier of cloth begins and ends, the spreader should slowly start and slowly pause, and avoid over leading and lagging.

4.4 System extended interface

Because various garment enterprises have different production sizes and products, and they require different equipment functions. To make the equipment better serve for the production, the spreader offers various all-purpose extended interfaces, and through these interfaces, we can conveniently use computer and network resource to extend the equipment functions (Fu, 2005, P.93-97). Except for offering compatible 10BASET standard RJ45 hardware interface, the network interface also could deploy standard network communication agreement and user communication agreement, so users could deploy the system network of production line according to actual situation, and establish the base for the automatic management for the garment enterprises.

The software flow of the control system is seen in Figure 4.

5. Conclusions

The control system of automatic spreader using embedded system possesses many advantages such as simple structure, low cost, strong function, high control precision and good extension, and the system has stronger function, more convenient operation and simpler maintenance, and reduces labors' work intension, enhances the rapid reaction ability and the automatization degree of production for the garment enterprises and enhances the competitive ability for enterprises.

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Figure 1. Spreading Mode

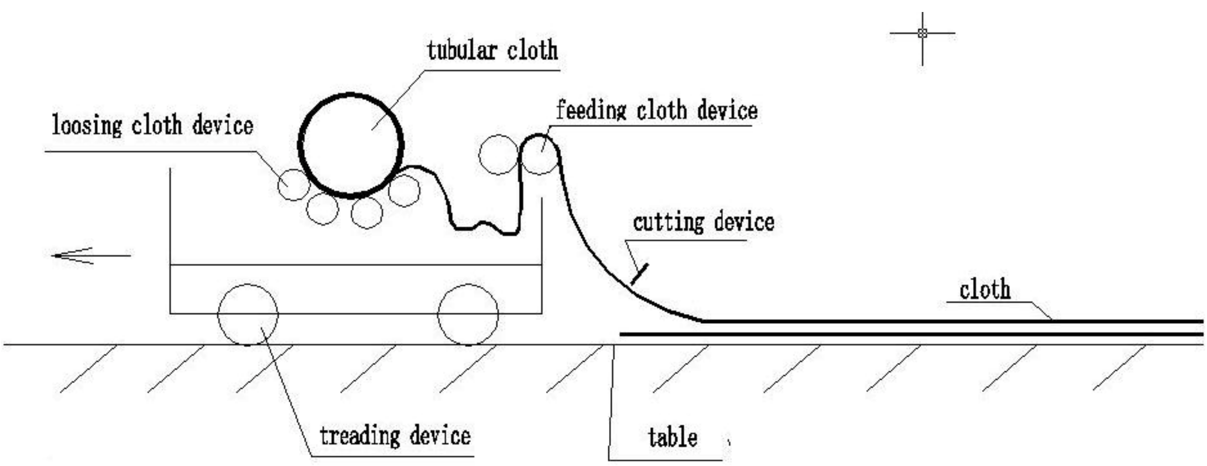

Figure 2. Work Principle of Spreader

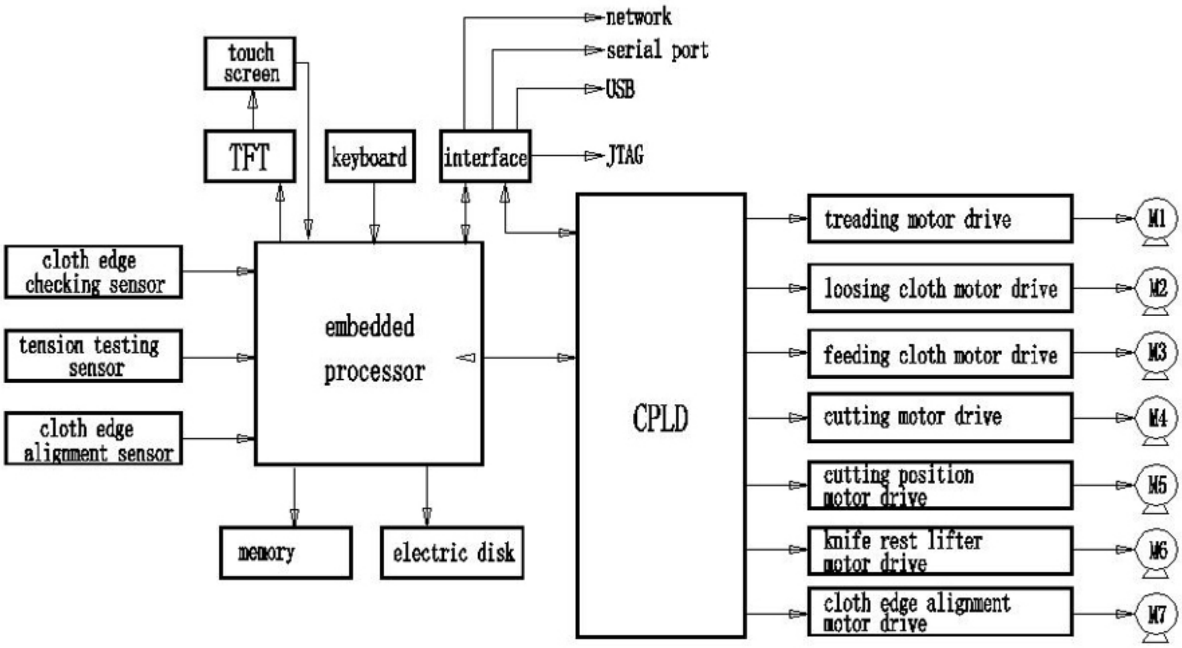

Figure 3. Hardware Structure Chart of Control System

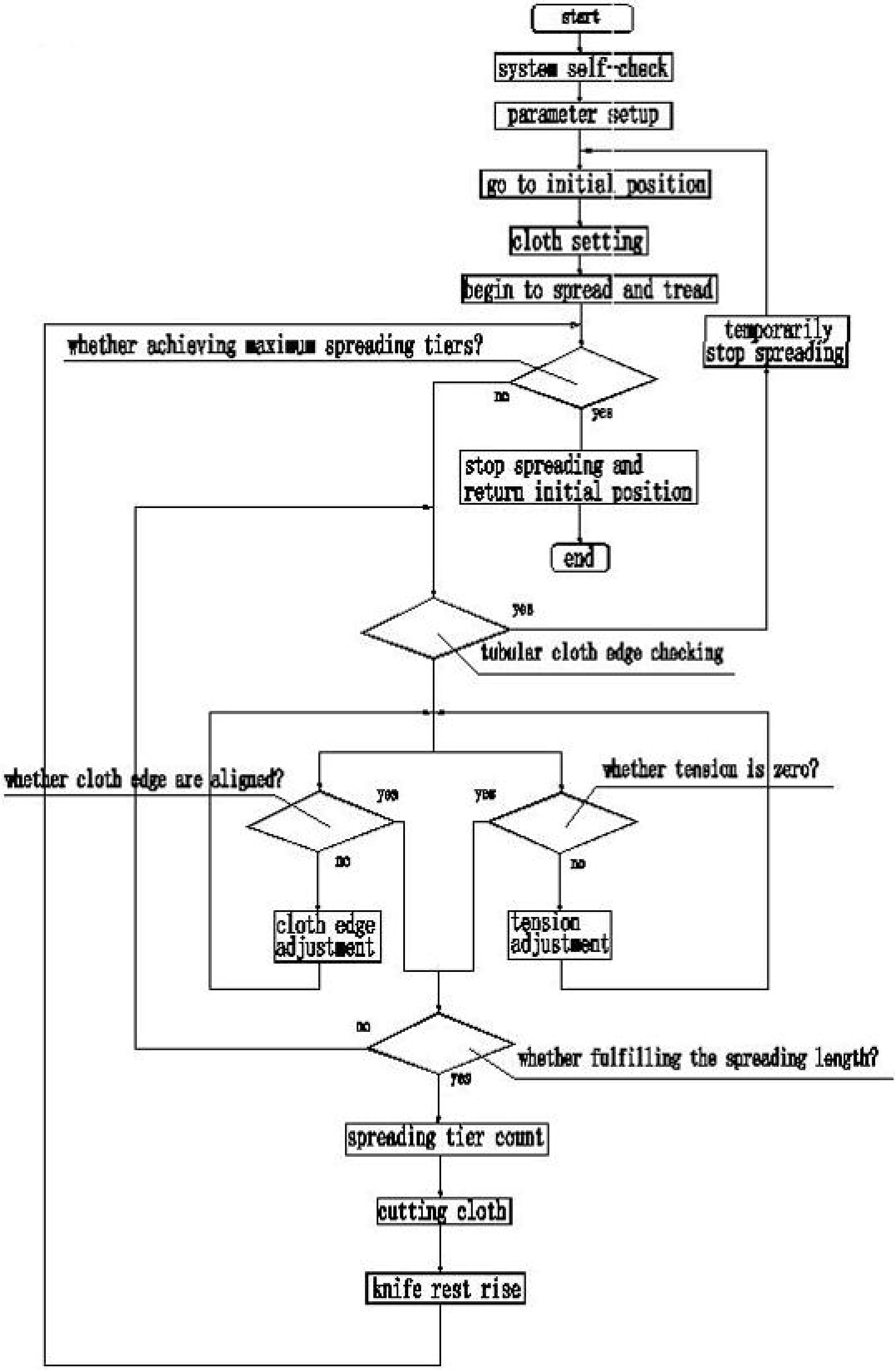

Figure 4. Software Flow Chart of Control System

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Text Document Pre-Processing Using the Bayes Formula for Classification Based on the Vector Space Model

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Abstract

This work utilizes the Bayes formula to vectorize a document according to a probability distribution based on keywords reflecting the probable categories that the document may belong to. The Bayes formula gives a range of probabilities to which the document can be assigned according to a pre determined set of topics (categories). Using this probability distribution as the vectors to represent the document, the text classification algorithms based on the vector space model, such as the Support Vector Machine (SVM) and Self-Organizing Map (SOM) can then be used to classify the documents on a multi-dimensional level, thus improving on the results obtained using only the highest probability to classify the document, such as that achieved by implementing the naïve Bayes classifier by itself. The effects of an inadvertent dimensionality reduction can be overcome using these algorithms. We compare the performance of these classifiers for high dimensional data.

Keywords: Document Classification, Naïve Bayes, Support Vector Machines, Self-Organizing Map

1. Introduction

Document classification can be defined as the task of automatically categorizing collections of electronic documents into their annotated classes, based on their contents. In recent years this has become important due to the advent of large amounts of data in digital form. For several decades now, document classification in the form of text classification systems have been widely implemented in numerous applications such as spam filtering [5], [6], [7], [23], [25], [27], [31],, e-mails categorizing [16], [32], knowledge repositories [11], and ontology mapping [28], contributed by the extensive and active researches. An increasing number of statistical and computational approaches have been developed for document classification, including decision tree [26], rule induction [26], k-nearest-neighbor classification [10], naïve Bayes classification [20], and support vector machines [15].

Each of the above mentioned document classification schemes have their own unique set of properties. The decision tree induction algorithm and rule induction algorithm are simple to understand and interpret after a brief explanation. However, these algorithms do not work well when the number of distinguishing features is large [26]. *k*-nearest neighbor algorithm is easy to implement and shows its effectiveness in a variety of problem domains [10]. A major drawback of the k-NN algorithm is computationally intensive, especially when the size of the training set grows [10]. Support vector machines (SVM) can be used as a discriminative document classifier and has been shown to be more accurate than most other techniques for classification tasks [4], [15]. The main problem associated with using the support vector machine for document classification is the effort needed to transform text data to numerical data. We call this essential step, the "vectorization" step. Techniques such as TF-IDF vectorize the data easily enough, however, dimensionality then becomes an issue since each vectorized word is used to represent the document. This leads to the number of dimensions being equal to the number of words. For linear classifiers based on the principle of Empirical Risk Minimization, capacity is equal to data dimensionality. For these learning machines, high dimensionality leads to "over-fitting" and the hypothesis becomes too complicated to implement computationally. On the other hand, over fitting does not occur in the SVM since its capacity is equal to the margin of separation between support vectors and the optimal hyper-plane instead of the dimensionality of the data. What is still of concern, however, is the high training time and classification time associated with high dimension data.

The self-organizing map (SOM) is a clustering method which clusters data, based on a similarity measure related to the calculation of Euclidean distances. The idea of this principle is to find a winner-takes-all neuron to find the most closely matching case. The SOM was proposed by Kohonen, and is based on the idea that systems can be designed to emulate the collective co-operation of the neurons in the human brain. Collectivism can be realized by feedback and thus can also be realized in the network, where many neighboring neurons react collectively upon being activated by events. If neurons are activated in the learning process, the neighboring neurons are also affected. The network structure is defined by synapses and has a similar total arrangement after a phase of self organization as the input data of the event space [33]. Consequently, the SOM is an established paradigm in AI and cognitive modeling being the basis of unsupervised learning. This unsupervised machine learning method is widely used in data mining, visualization of complex data, image processing, speech recognition, process control, diagnostics in industry and medicine, and natural language processing [34]. As similar to the SVM, the SOM needs a pre-processing step to transform the text documents into a suitable format for input into the classifier. Hence, the same problem is faced by the SOM, the high dimensionality of vectorized data that represents a text document requires a computational extensive processes during the training and classifying stages.

In this work, we reduce the dimensions from thousands (equal to the number of words in the document, such as when using TF-IDF) to typically less than 20 (number of categories the document may be classified to) through the use of the Bayes formula and then feed this probability distribution to the SVM and SOM for training and classification purposes. With this, we hope to reduce training and classification time to a feasible level (reduce dimensions from thousands to 10 or 20) while maintaining acceptable generalization (using the optimal hyper-plane) and accuracy (10 or 20 dimensions used for classification instead of 1 as is the case with the naïve Bayes classifier).

By itself, the naïve Bayes is a relatively accurate classifier if trained using a large data set. However, as in many other linear classifiers, capacity control and generalization remains an issue. In our case, however, the naïve Bayes is used as a pre-processor in the front end of the classification algorithms based on the vector space model, in our case, the SVM and the SOM, to vectorize text documents before the training and classifying stages are carried out. This is done to improve the generalization of the overall system, while still maintaining a comparatively feasible training and categorization time afforded through the dimensionality reduction imposed by the Bayes formula.

In the context of document classification, the Bayes theorem uses the fact that the probability of a particular document being annotated to a particular category, given that the document contains certain words in it, is equal to the probability of finding those certain words in that particular category, times the probability that any document is annotated to that category, divided by the probability of finding those words in any document, as illustrated in equation (1).

$$\Pr(Category \mid Word) = \frac{\Pr(Word \mid Category).\Pr(Category)}{\Pr(Word)} \tag{1}$$

Each document contains words which are given probability values based on the number of its occurrence within that particular document. Naïve Bayes classification is predicated on the idea that electronic documents can be classified based on the probability that certain keywords will correctly identify a piece of text document to its annotated category.

At the basic level, a naïve Bayes classifier examines a set of text documents that have been well organized and categorized, and then compares the content of all categories in order to build a database of words and their occurrences. The database is used to identify or predict the membership of future documents to their right category, according to the probability of certain word occurring more frequently for certain categories. It overcomes the obstacles faced by more static technologies, such as blacklist checking, and word to word comparisons using static databases filled with pre-defined keywords.

In other words, the overall classifier is robust enough to ignore serious deficiencies in its underlying naïve probability model [12], ie, the assumption of the independence of probable events. This robustness is encapsulated in the relative magnitudes of the probabilities (relatively constant for the same category) which is important for the SVM as opposed to an accurate "absolute" value of probability for each category.

In this paper we emphasize the hybrid classification approach which utilizes the simplicity of the Bayes formula as a vectorizer and the good generalization capability of the text classification algorithm based on the vector space model, such as the SVM and the SOM as a classifier. The results of the naïve Bayes-SVM hybrid approach and the naïve Bayes-SOM hybrid approach have been compared to distinguish the performance of both the hybrid classification approaches in terms of the classification effectiveness and efficiency.

2. The Hybrid Classification Approach

We propose, design, implement and evaluate a hybrid classification method by integrating the naïve Bayes vectorizer and text classifiers based on the vector space model to take advantage of the simplicity of the Bayes technique and the accuracy of the SVM and the SOM classification approaches. These hybrid approaches shows an improved classification performance as compared to the traditional naïve Bayes's performance by itself. This has overcome the major drawback of using the traditional naïve Bayes technique alone as a classifier.

The text classification algorithms based on the vector space model are unable to handle the data in text format. An electronic text document may contain a huge number of features such as keywords, which are very useful in terms of the classification task. Therefore, to classify text documents by using the SVM or the SOM, keywords need to be transformed into vectors which are suitable for input to the classifiers.

Generally, the words of a text documents are directly vectorized thereby transforming the text documents into a numerical format. With TF-IDF for example, the huge number of features transformed from the text data of a single document makes running the classifiers a long affair as it takes time to process vectorized data with large dimensions.

Many methods have been carried out to overcome this problem by reducing the dimension of the features. In order to reduce the number of features, we introduce the naïve Bayes as the vectorizer for the text documents by using the probability distribution described earlier, where the dimension of the features is based on the number of available categories in the classification task.

Since the naïve Bayes classifier is able to handle raw text data via the probability distribution calculated from key word occurrence, and the text classifiers based on the vector space model such as the SVM and the SOM typically requires preprocessing to vectorize the raw text documents into numerical values, it is natural to use the naïve Bayes as the vectorizer for the classifiers based on the vector space model. The structure of the proposed classification approach is illustrated as Figure 1.

For the purposes of using the SVM or the SOM as the classifier, after the naïve Bayes vectorizer has been trained, each training document is vectorized by the trained naïve Bayes classifier through the calculation of the posterior probability of the training documents for each existing category based on the Bayes formula. For example, the probability value for a document X to be annotated to a category C is computed as $Pr(C|X)$. As an assumption that we have a category list as $Cat1$, $Cat2$, $Cat3$, $Cat4$, $Cat5$, ……., $CatN$, thus, each document has N associated probability values, where document X will have $Pr(Cat1|X)$, $Pr(Cat2|X)$, $Pr(Cat3|X)$, $Pr(Cat4|X)$, $Pr(Cat5|X)$, ……, $Pr(CatN|X)$.

All the probability values of a document are combined to construct a multi-dimensional array, which represents the probability distribution of the document in the feature space. In this way, all the training documents are vectorized by their probability distribution in feature space, in the format of numerical multi-dimensional arrays, with the number of dimensions depending on the number of categories. With this transformation, the training documents are suitable for use in constructing the vectorized training dataset for the classifiers based on the vector space model. Our approach uses the same training dataset for both the naïve Bayes vectorizer and the classifier, where the naïve Bayes vectorizer uses the raw text document for training purposes, and the classifier uses the vectorized training data supplied by the naïve Bayes vectorizer.

As for the classification session, the input to the trained naïve Bayes system is now replaced by the unknown text documents. The output from the naïve Bayes vectorizer, which is the vectorized data of the text documents, in the format of multi-dimensional numerical probability values, is used as the input for the SVM or the SOM for the final classification steps.

2.1 The Naïve Bayes Classification Approach for Vectorization

Our proposed naïve Bayes classifier [13] performs its classification tasks starting with the initial step of analyzing the text document by extracting words which are contain in the document. To perform this analysis, a simple word extraction algorithm is used to extract each individual word from the document to generate a list of words. This list is always helpful when the probabilistic classifier calculating the probability of each word being annotated to each category. The list of words is constructed with the assumption that input document contains words $w\ 1, w\ 2, w\ 3, \ldots\ldots\ldots, w\ n\text{-}1, w\ n,$ where the length of the document (in terms of number of words) is n.

The list of words is then used to generate a table, containing the probabilities of the word in each category. The column of "Word" is filled with words which are extracted from the input document. For the columns of probabilities of the particular word for each category, the values to be filled will be calculated by the probability classifier in the following stage. The tables below illustrate the use of this method for the input document in Figure 2.

Before the probabilistic classifier performs the calculation of words' probability for each category, it needs to be trained with a set of well-categorized training dataset. Each individual word from all training documents in the same category are extracted and listed in a list of words occurrence for the particular category, by using a simple data structure algorithm.

Based on the list of word occurrence, the trained probabilistic classifier calculates the posterior probability of the particular word of the document being annotated to particular category by using the formula which is shown as equation (2), since each word in the input document contributes to the document's categorical probability.

$$\Pr(\mathit{Category} \mid \mathit{Word}) = \frac{\Pr(\mathit{Word} \mid \mathit{Category}).\Pr(\mathit{Category})}{\Pr(\mathit{Word})} \tag{2}$$

The derived equation above shows that by observing the value of a particular word, wj, the prior probability of a particular category, Ci, $\Pr(Ci)$ can be converted to the posterior probability, $\Pr(Ci|wj)$, which represents the probability of a particular word, wj being a particular category, Ci. The prior probability, $\Pr(Ci)$ can be computed from equation (3) or equation (4):

$$\Pr(\mathit{Category}) = \frac{Total\_of\_Words\_in\_Category}{Total\_of\_Words\_in\_Training\_Dataset} \tag{3}$$

$$= \frac{Size\_of\_Category}{Size\_of\_Training\_Dataset} \tag{4}$$

Meanwhile, the evidence, which also known as the normalizing constant of a particular word, wj, $\Pr(wj)$ is calculated by using equation (5):

$$\Pr(\mathit{Word}) = \frac{\sum occurrence\_of\_Word\_in\_every\_category}{\sum occurrence\_of\_all\_words\_in\_every\_category} \tag{5}$$

The total occurrence of a particular word in every category can be calculated by searching the training data base, which is composed from the lists of word occurrences for every category. As previously mentioned, the list of words occurrence for a category is generated from the analysis of all training documents in that particular category during the initial training stage. The same method can be used to retrieve the sum of occurrence of all words in every category in the training data base.

To calculate the likelihood of a particular category, Ci with respect to a particular word, wj, the lists of words occurrence from the training data base is searched to retrieve the occurrence of wj in Ci, and the sum of all words in Ci. These information will contribute to the value of $\Pr(wj \mid Ci)$ given in equation (6).

$$\Pr(\mathit{Word}|\mathit{Category}) = \frac{occurrence\_of\_Word\_in\_Category}{\sum occurrence\_of\_all\_words\_in\_Category} \tag{6}$$

Based on the derived Bayes' formula for text classification, with the value of the prior probability Pr(*Category*), the likelihood Pr(*Word*|*Category*), and the evidence Pr(*Word*), the posterior probability, Pr(*Category* | *Word*) of each word in the input document annotated to each category can be measured.

The posterior probability of each word to be annotated to each category is then filled to the appointed cells in the table as illustrated in Figure2. After all the cells of "Probability" have been filled, the overall probability for an input document to be annotated to a particular category, Ci is calculated by dividing the sum of each of the "Probability" column with the length of the query, ***n***, which is shown in equation (7).

$$\Pr(\mathit{Category} \mid \mathit{Document}) = \frac{\Pr(\text{Category} \mid \text{w 1, w 2, w 3,......, w n - 1, w n})}{n} \tag{7}$$

where ***w** 1, **w** 2, **w** 3,………, **w** n-1, **w** n*, are the words which are extracted from the input document.

Typically, the ordinary naïve Bayes classifier is able to determine the right category of an input document by referring to the associated probability values calculated by the trained classifier based on the Bayes formula. The right category is represented by the category which has the highest posterior probability value, Pr(*Category*|*Document*), as stated in the Bayes Classification Rule. As the ordinary naïve Bayes classification algorithm has been proven as one of the poor performing classifier, we have extended the classification step for input documents by performing the SVM, for the purpose of increasing the classification accuracy.

2.2 Support Vector Machines for Text Classification

The classification problem here can be restricted to the consideration of the two-class problem without loss of generality. The goal is to produce a classifier that will work well on unseen examples, i.e. it generalizes well. Consider the example in Figure 3. Here there are many possible linear classifiers that can separate the data, but there is only one that maximizes the margin (maximizes the distance between it and the nearest data point of each class). This linear classifier is termed the optimal separating hyperplane.

We will start with the simplest case: linear machines trained on linearly separable data. Therefore consider the problem of separating a set of training vectors $\mathbf{x}_i{}^d$ belonging to different classes y_i {-1, 1}. We wish to separate this training set with a hyperplane [12]:

$$\mathbf{w}.\mathbf{x} + b = 0 \quad (8)$$

There are actually an infinite number of hyperplanes that could partition the data into two sets (dashed lines on Figure 3). According to the SVM principle, there will just be one optimal hyperplane: the hyperplane lying half-way in between the maximal margin (we define the margin as the sum of distances of the hyperplane to the closest training points of each class). The solid line on Figure 3 represents this optimal separating hyperplane, the margin in this case is d1+d2.

Note that only the closest points of each class determine the Optimal Separating Hyperplane. These points are called Support Vectors (SV). As only the Support vectors determine the Optimal Separating Hyperplane they is a certain way to represent them for a given set of training points. It has been shown that the maximal margin can be found by minimizing ½ $\|\mathbf{w}\|^2$ [12].

$$\min \{½ \|\mathbf{w}\|^2 \} \quad (9)$$

The Optimal Separating Hyperplane can thus be found by minimizing (9) under the constraint (10) that the training data is correctly separated.

$$y_i.(x_i.w + b) \quad 1 \;, \; i \quad (10)$$

The concept of the Optimal Separating Hyperplane can be generalized for the non-separable case by introducing a cost for violating the separation constraints (10). This can be done by introducing positive slack variables $_i$ in constraints (10), which then become:

$$yi.(x_i.w + b) \quad 1 - \quad _i \;, \; I \quad (11)$$

If an error occurs, the corresponding $_i$ must exceed unity, so $_i$ $_i$ is an upper bound for the number of classification errors. Hence a logical way to assign an extra cost for errors is to change the objective function (9) to be minimized into:

$$\min \{ ½ \|\mathbf{w}\|^2 + C. (\quad _i \quad _i) \} \quad (12)$$

where C is a chosen parameter. A larger C corresponds to assigning a higher penalty to classification errors. Minimizing (12) under constraint (11) gives the *Generalized Optimal Separating Hyperplane.* This is a Quadratic Programming (QP) problem which can be solved here using the method of Lagrange multipliers.

After performing the required calculations [12], the QP problem can be solved by finding the LaGrange multipliers, α_i, that maximizes the objective function in (13),

$$W(\alpha) = \sum_{i=1}^{n} \alpha_i - \frac{1}{2} \sum_{i,j=1}^{n} \alpha_i \alpha_j y_i y_j \left(\mathrm{x}_\mathrm{i}^\mathrm{T} \mathrm{x}_\mathrm{j} \right) \quad (13)$$

subject to the constraints,

i=1,…..,n, $\quad 0 \le \alpha_i \le C,$ and $\quad \sum_{i=1}^{n} \alpha_i y_i = 0.$

The new objective function is in terms of the Lagrange multipliers, α_i only. It is known as the dual problem: if we know **w**, we know all α_i. if we know all α_i, we know **w**. Many of the α_i are zero and so w is a linear combination of a small number of data points. x_i with non-zero α_i are called the support vectors [9]. The decision boundary is determined only by the SV. Let t_j (j=1, ..., s) be the indices of the s support vectors. We can write,

$$w = \sum_{j=1}^{s} \alpha_{t_j} y_{t_j} x_{t_j} \quad (14)$$

So far we used a linear separating decision surface. In the case where decision function is not a linear function of the data, the data will be mapped from the input space (i.e. space in which the data lives) into a high dimensional space (feature space) trough a non-linear transformation. In this (high dimensional) feature space, the (Generalized) Optimal Separating Hyperplane is constructed. This is illustrated on Figure 4.

By introducing the kernel function,

$$K(x_i, x_j) = \langle \Phi(x_i), \Phi(x_j) \rangle, \tag{15}$$

it is not necessary to explicitly know Φ (.) [3]. So that the optimization problem (13) can be translated directly to the more general kernel version,

$$W(\alpha) = \sum_{i=1}^{n} \alpha_i - \frac{1}{2} \sum_{i=1, j=1}^{n} \alpha_i \alpha_j y_i y_j K(x_i, x_j)$$

subject to
$$C \geq \alpha_i \geq 0, \sum_{i=1}^{n} \alpha_i y_i = 0 \tag{16}$$

After the α_i variables are calculated, the equation of the hyperplane, $d(\mathbf{x})$ is determined by,

$$d(x) = \sum_{i=1}^{l} y_i \alpha_i K(\mathbf{x}, \mathbf{x_i}) + b \tag{17}$$

The equation for the indicator function, used to classify new data is given below where the new data **z** is classified as class 1 if $i>0$, and as class 2 if $i<0$ [17].

$$i_F(\mathbf{x}) = \text{sign}\,[d(\mathbf{x})] = \text{sign}\left[\sum_{i=1}^{l} y_i \alpha_i K(\mathbf{x}, \mathbf{x_i}) + b\right] \tag{18}$$

Note that the summation is not actually performed over all training data but rather over the support vectors, because only for them do the Lagrange multipliers differ from zero. This particular characteristic of the SVM makes it ideal for applications like text classification, where there can be millions of documents. The SVM uses these documents for training and only retains a smaller portion of the entire set as its support vectors. This makes SVM computationally more efficient, needing less memory resources as compared to other techniques like neural networks. However, in a system with the SVM in combination with the naïve Bayes vectorizer multi-dimensional categorization of the documents is possible leading to more accurate results. This is achieved thru the mechanism by which an optimal hyperplane is constructed using only the closest data points or “support vectors”.

The SVM uses only some of these documents (represented by feature vectors) for training and retains a small proportion of the entire set as its support vectors. This makes the SVM computationally more efficient, needing less memory resources as compared to other techniques like neural networks which need relatively larger training data sets in order to generalize accurately. This good generalization performance of the SVM is one of the main reasons for using it in text classification. Although millions of documents may be available for training, the burden will be on the high training time. Since the SVM needs a comparatively small training set to isolate the support vectors which form the margins of the optimal hyper plane, training time is reduced compared to other margin classifiers. However, the higher the data dimensionality, the longer the SVM training time and we have addressed this be using the Bayes formula as a vectorizer.

As mentioned previously, another factor which sets the SVM apart is the fact that through its implementation of Structural Risk Minimization, capacity is now independent of data dimensionality. The time to train the SVM is however still proportional to data dimensionality. However, in our case the vectorization process reduces the dimensions of the data to the number of available categories. In vectorization techniques such as TF-IDF however, the number of dimensions is equal to the number of words in that document. As will be shown later, this sacrifice does not detrimentally affect classification accuracy, but buys us a further improved SVM training time which can be significant for large documents. Using this method we find the middle ground between the single dimension classification technique afforded by the naïve Bayes classifier (SVM is multi dimensional) and the high training time needed for the techniques like TF-IDF SVM hybrids (we reduce data dimensionality and therefore training and classification time).

2.3 The Self-Organizing Map for Text Classification

Knowledge discovery tasks can be broken down into two general steps: pre-processing and classification. In the pre-processing step, data is transformed into a format which can be processed by a classifier. The self organizing map (SOM) can be used to carry out the classification tasks effectively, especially for the analysis and visualization of a variety of economical, financial, scientific, and manufacturing data sets [24] [30]. The first step in designing the SOM is to decide on what prominent features are to be used in order to effectively cluster the data into groups. The criterion for selecting the main features plays an important role in ensuring that the SOM clusters properly and thus supports goal based decision making. Traditionally statistical cluster analysis is an important step in improving feature extraction and is done iteratively. An alternative to these statistical methods is the SOM. Kohonen's principle of topographic map formation, states that the spatial location of an output neuron in the topographic map corresponds to a particular feature of the input pattern. The SOM model, which is shown in Figure 5, provides a map which places a fixed number input patterns from an input layer into the so called Kohonen layer [30]. The system learns through self organization of random neurons whose weights are attached to the layers of neurons. These weights are altered at every epoch during the training session. The change depends upon the similarity or neighborhood between the input pattern and the map pattern [34]. The topographic feature maps reduce the dimensions of data to two dimensions simplifying viewing and interpretation.

In the SOM, certain trends in clustering can be observed by changing some of the training parameters. After the incremental training of the map, the application saves the weight vectors of the map and these weights can be used as the starting weights. Once the training is over, the output mean and variance of each cluster is reported. Furthermore, the location of each cluster is also reported. Speed is a big concern in SOM clustering. By reducing dimensionality thru the use of a probability distribution of categories in feature space, instead of a raw word occurrence count we reduce computation time in both training and classification. The concerns associated with data pre-processing before the training starts and final drawing of the map once the training is over is also addressed by our hybrid system.

The SOM is trained iteratively. In each training step, a sample vector, *x* from the input data set is chosen randomly and the distance between *x* and all the weight vectors of the SOM, is calculated by using a Euclidean distance measure. The neuron with the weight vector which is closest to the input vector *x* is called the Best Matching Unit (BMU). The distance between *x* and weight vectors, is computed by using the equation below:

$$\| x - m_c \| = \min \{\| x_i - m_i \|\}$$

where ||.|| is the distance measure, typically Euclidean distance. After the BMU is found, the weight vectors of the SOM are updated so that the BMU is moved closer to the input vector in the input space. The topological neighbors of the BMU are treated similarly. The update rule for the weight vector of *i* is:

$$x_i(t+1) = m_i(t) + \alpha(t)\, h_{ci}(t)\, [\, x(t) - m_i(t)]$$

where x(t) is a vector which is randomly drawn from the input data set, and function $\alpha(t)$ is the learning rate and t denotes time. The function $h_{ci}(t)$ is the neighborhood kernel around the winner unit *c*. The dataset of manufacturing details are fed into the input layer of SOM. Learning parameter is selected between 0.0- 0.9, and the SOM is trained. The training steps will be in the range of 100000 epochs in order to obtain a trained map. These training datasets are coded with reference to their prominent features [30].

3. The Evaluations and Experimental Results

The objective of this evaluation is to determine whether our proposed hybrid approach result in better classification accuracy and performance compared to the ordinary naïve Bayes version. As mentioned in the sections above, the hybrid document classification approach utilizes the simplicity and low requirements of the naïve Bayes classifier as a vectorizer, and the superb generalization performance of the SVM as a classifier. The evaluations are made by comparing the classification accuracy of the ordinary naïve Bayes classifier (along with some specialized techniques with the naïve Bayes vectorizer) and the SVM classification approach which is proposed in this paper.

Initially we performed the experiment by implementing the naïve Bayes algorithm with different ranking schemes: the naïve Bayes with flat ranking algorithm, the naïve Bayes with single elimination tournament ranking algorithm and both the above with the High Relevance Keywords Extraction (HRKE) algorithm. These classification algorithms are discussed briefly below and also in our previous work [13]. In particular the naïve Bayes with flat ranking computes the probability distribution by considering all categories in a single round of competition. The single elimination method entails finding a winner that has not lost even once within the competition. The HRKE algorithm culls out words such as "a", "the" etc which have a low effect on the classification task because it appears in every document. The algorithms mentioned above determine the right category for input documents by referring to the associated probability values calculated by the trained classifier based on the Bayes formula. The right category is represented by the category which has the highest posterior probability value, ***Pr(Category|Document)***.

To evaluate the hybrid classification approach which is proposed in this paper, we implement the naïve Bayes classification algorithms mentioned above in the front end to vectorize raw text data using the associated probability values calculated using the Bayes formula. SVM is used to perform the rest of the classification tasks using the RBF (Radial Basis Function) and the Sigmoid kernels and the results of the classification for the hybrid approach are based on the highest accuracy values.

A dataset of vehicle characteristics extracted from Wikipedia is tested in the prototype system for the evaluation of classification performance in handling a database with four categories which have low degrees of similarity. Our selected dataset contains four categories of vehicles: Aircrafts, Boats, Cars, and Trains. All the four categories are easily differentiated and every category has a set of unique keywords. We have collected 110 documents for each category, with the total of 440 documents in the entire dataset. 50 documents from each category are extracted randomly to build the training dataset for the classifier. The other 60 documents for each category are used as the testing dataset to test the classifier. The results here were seen to be worse than those obtained using cross validation. As such we only present the worse case here. Figure 6 illustrates the table of comparison of the performance between the pure naïve Bayes classification algorithms mentioned above and the hybrid algorithms perform in conjunction with the SVM or the SOM in the back end while classifying the Vehicles dataset.

Looking at the SVM results in comparison with the Bayes classifier, the tournament structure based ranking algorithms show poor performance in terms of classification results. With the proposed hybrid approach, however, significant improvement in classification accuracy is seen since a 20.84% gain is achieved with the hybrid approach measurement compared with the pure naïve Bayes model, where both the hybrid and ordinary approaches were implemented with the single elimination tournament ranking technique.

In this experiment, we also present the classification by using SOM on the Vehicles dataset with 200 training documents. The dataset has four dimensional of information, which is categorized by different methods as shown in Figure 6. The listing of results shows the performances are carried in conjunction with SOM. The vectorized data by the front end naïve Bayes vectorizer are considered as the input data to the SOM for further clustering purposes. The results are tested for 30000 epochs of training cycle. Initial radius of 3 is considered in rough training and radius of 1 is considered in the final training.

The comparison table in Figure 6 illustrates that the recognition rate is poor, when the SOM is trained in combination with the single elimination tournament ranking. The recognition rate is only 56.66%. A similar situation was found when the hybrid approach is used with the flat ranking naïve Bayes classifier. The recognition rate is only 58.00%. The hybrid approach with the naïve Bayes vectorizer when enhanced by High Relevance Keywords Extraction Facility shows good results, with 98.33% recognition rate, a slight improvement over the pure naïve Bayes classifier classification rate.

We have also tested our proposed hybrid classification approaches on the 20 Newsgroups dataset. The 20 Newsgroups dataset is one of the most common datasets used by many text classification research groups to evaluate the performance of their presented classification approaches. The 20 Newsgroups dataset is a collection of 20,000 Usenet articles from 20 different newsgroups with 1,000 articles per newsgroup. In our experiments using this dataset, every category was divided into two subsets. 300 documents from each category were divided for training while the remaining 700 documents were used for testing purposes. Figure 7 illustrates the table of comparison of the performance between the pure naïve Bayes classification algorithms mentioned above and the hybrid algorithms perform in conjunction with the SVM or the SOM in the back end while classifying the 20 Newsgroups dataset.

The comparison table illustrated in Figure 7 has proven again our prediction that the naïve Bayes – SVM hybrid classification approach has increased performance as compared to the naïve Bayes classifiers. Similar to the experiment using the Vehicles dataset, significant improvement of classification accuracy is gained by implementing the naïve Bayes – SVM hybrid approach against the pure naïve Bayes model implemented with tournament ranking methods. An approximately 10% gain is achieved with the naïve Bayes – SVM hybrid approach measurement compared with the pure naïve Bayes model with the implementation of the single elimination tournament ranking method. As for the pure naïve Bayes model which is implemented with the single elimination tournament ranking plus the HRKE facility, a 14.36% gain is achieved by the naïve Bayes – SVM hybrid approach as compared to the pure naïve Bayes model.

The experimental results for the naïve Bayes – SOM hybrid classification approach illustrated in the comparison table shown in Figure 7 have gone against our prediction that the implementation of the hybrid classification approach should show an improvement in the classification performance as compared to the ordinary naïve Bayes classifiers. The naïve Bayes – SOM hybrid approach has greatly reduced the classification accuracy as compared to the naïve Bayes classifiers, in the classification task on the high dimensionality 20 newsgroups dataset.

The main reason why the naïve Bayes – SOM hybrid approach shows worse results with the 20 Newsgroups dataset as compared to the naïve Bayes classifier is that the number of categories for this dataset is high. The SOM algorithm which uses a Euclidean distance measure in determining the best matching units is accurate only for a maximum of four dimensions.

4. Conclusion and Future Works

The hybrid text document classification approach through the implementation of the naïve Bayes method at the front-end for raw text data vectorization, in conjunction with a classifier based on the vector space model, in our case, we have used the SVM and the SOM at the back-end to determine the right category for the input documents, has been proposed and developed by our research group. This hybrid approach takes advantages of both the naïve Bayes system and the classifiers based on the vector space model which greatly enhances the performance of ordinary Bayes classification algorithms. The results from our experiments show that the proposed hybrid approach of naïve Bayes vectorizer and the classifiers based on the vector space model has an improved classification accuracy compared to the pure naïve Bayes classification approach. However, in the cases where the number of categories is high (such as the 20 Newsgroups dataset), the implementation of the classifiers based on the unsupervised learning, such as the SOM is impractical. This is due to the reason that due to the fact that the Euclidean distance measure is inaccurate for dimensions above four as previously stated.

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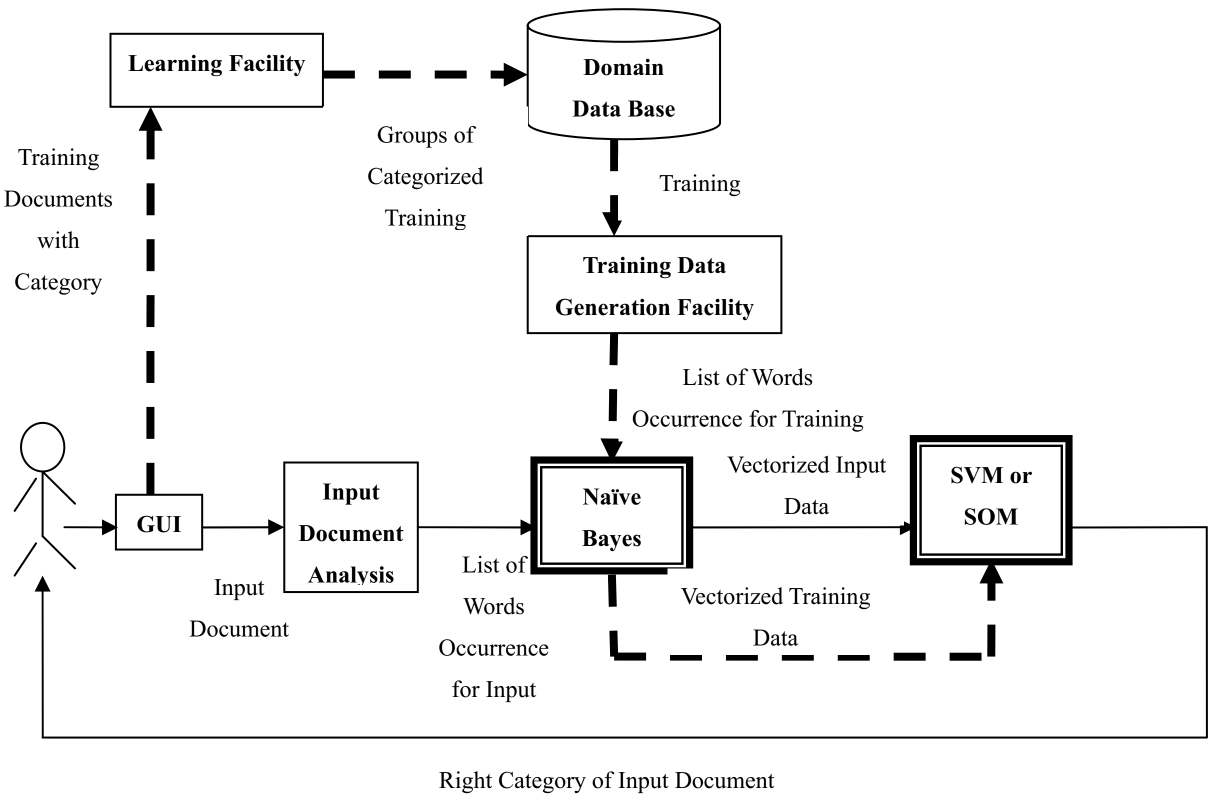

Figure 1. Block diagram of proposed hybrid classification approach.

Word	*Probability* *Category 1*	*Probability* *Category 2*	*Probability* *Category 3*	… …	*Probability* *Category k-1*	*Probability* *Category k*
w 1	Pr(*C1*|*w1*)	Pr(*C2*|*w1*)	Pr(*C3*|*w1*)	…	Pr(*C k*-1|*w1*)	Pr(*C k*|*w1*)
w 2	Pr(*C1*|*w2*)	Pr(*C2*|*w2*)	Pr(*C3*|*w2*)	…	Pr(*C k*-1|*w2*)	Pr(*C k*|*w2*)
w 3	Pr(*C1*|*w3*)	Pr(*C2*|*w3*)	Pr(*C3*|*w3*)	…	Pr(*C k*-1|*w3*)	Pr(*C k*|*w3*)
w n-1	Pr(*C1*|*w n-1*)	Pr(*C2*|*w n-1*)	Pr(*C3*|*w n-1*)	…	Pr(*C k*-1|*w n-1*)	Pr(*C k*|*w n-1*)
w n	Pr(*C1*|*wn*)	Pr(*C2*|*wn*)	Pr(*C3*|*wn*)	…	Pr(*C k*-1|*wn*)	Pr(*C k*|*wn*)
Total	∑Pr(*C1*|*W*)	∑Pr(*C2*|*W*)	∑Pr(*C3*|*W*)	…	∑Pr(*C k*-1|*W*)	∑Pr(*C k*|*W*)

Figure 2. Table of words occurrence and probabilities.

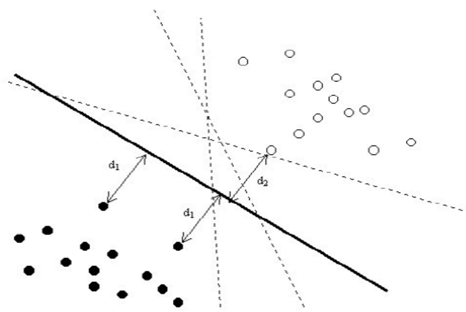

Figure 3. Optimal Separating Hyper-plane.

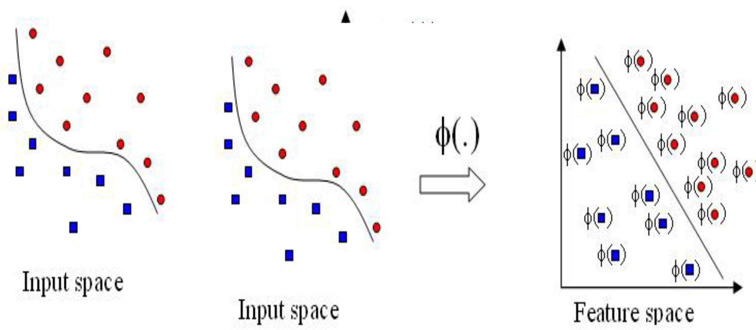

Figure 4. Mapping onto higher dimensional feature space.

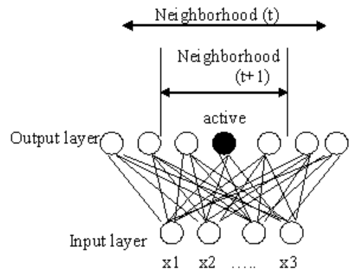

Figure 5. The SOM model.

Classification Optimization Techniques	Naïve Bayes (Accuracy)	Naïve Bayes – SVM Hybrid (Accuracy)	Naïve Bayes – SOM Hybrid (Accuracy)
Flat Ranking	81.25%	85.42%	58.00%
Flat Ranking with HRKE Facility	96.25%	94.58%	98.33%
Single Elimination Tournament Ranking	64.58%	85.42%	56.66%
Single Elimination Tournament Ranking with HRKE Facility	76.66%	85.83%	57.00%

Figure 6. Comparison table for classification accuracy using different methods

(Dataset: Vehicles-Wikipedia).

Classification Optimization Techniques	Naïve Bayes (Accuracy)	Naïve Bayes – SVM Hybrid (Accuracy)	Naïve Bayes – SOM Hybrid (Accuracy)
Flat Ranking	70.13%	78.99%	20%
Flat Ranking with HRKE Facility	77.82%	79.55%	34%
Single Elimination Tournament Ranking	40.68%	50.55%	27%
Single Elimination Tournament Ranking with HRKE Facility	65.05%	79.41%	14%

Figure 7. Comparison table for classification accuracy using different methods (Dataset: 20 Newsgroups).

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Multi-object Segmentation
Based on Improved Pulse Coupled Neural Network

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Abstract

This paper introduces an approach for image segmentation by using pulse coupled neural network (PCNN), based on the phenomena of synchronous pulse bursts in the animal visual cortexes. The synchronous bursts of neurons with different input were generated in the proposed PCNN model to realize the multi-object segmentation. The criterion to automatically choose the dominant parameter (the linking strength β), which determines the synchronous-burst stimulus range, was described in order to stimulate its application in automatic image segmentation. Segmentations on several types of image are implemented with the proposed method and the experimental results demonstrate its validity.

Keywords: The Pulse-Coupled Neural Network (PCNN), Automatically Image Segmentation, Parameter Determination

1. Introduction

Pulse coupled neural networks (PCNN) were introduced as a simple model for the cortical neurons in the visual area of the cat's brain. Important research in the 80's and 90's led to the establishment of a general model for PCNN . Such models are proved to be highly applicable in the field of image processing, a series of optimal procedures being developed for contour detection and especially image segmentation. Taking PCNN abilities into consideration, we propose an extended PCNN using fast linking, and make some improvements to extend PCNN to work effectively. Finally, we apply this PCNN to actual images under various conditions of illuminations and demonstrate the effectiveness of this model through experiments.

In the second section of this paper, the PCNN's basic model and the fast-linking method are introduced. In the third section, the new approach for multi-object image segmentation based on fast linking is brought forward. In the fourth section, results of multi-object segmentation based on the proposed PCNN model are shown.

2. The pulse-coupled neural network

A simplied PCNN model

A simplified pulse coupled neuron(PCN) consists of three parts: the receptive field, the modulation field, and the pulse generator, see Figure 1. Compare with real neurons, this model contains some simplifications and approximations. Some factors, such as multiple ionic synaptic channels, active channels, cellular aging, temperature effects, are not considered. The two channels in the receptive field and the pulse generator are simplified too. Equations from Eq (1) to (5) describe this model.

$$F_j = S_j(n) \tag{1}$$

$$L_j(n) = \sum WY_j(n-1)$$

(2)

$$U_j(n) = F_j(n)(1+\beta L_j(n))$$

(3)

$$Y_j(n) = \mathrm{Step}(U_j(n) - \theta_j(n))$$

(4)

$$\theta_j(n) = \alpha^T \theta_j(n-1) + V^T Y_j(n)$$

(5)

The neuron receives input signals from other neurons and external sources by two channels in the receptive field. In general, the signals from other neurons are pulses; the signals from external sources are analog timing-varying signals, constants, or pulses. Each neuron has two channels. One channel called F channel is the feeding input; the other called L channel is the linking input. In this paper, F channel receives signals from external sources, namely the intensity of the image pixel. In figure 1, feeding input is the intensity of the image pixel connected to neuron and received by channel F, see (1). L channel receives pulses emitted by neighboring neurons, see (2). In modulation field, see Figure 1, the linking input is added a constant positive bias firstly. Then it is multiplied by the feeding input and the bias is taken to be unity, see (3). β_j is the linking strength. The total internal activity U_j is the result of modulation and is inputted to the pulse generator. If U_j is greater than the threshold θ_j, the neuron output Y_j turns into 1 (namely the neuron fires), see (4). Then Y_j feedbacks to make θ_j rise over U_j immediately so that Y_j turns into 0. Therefore, when U_j is greater than θ_j, neuron j outputs a pulse. Next θ_j drops with time increasing. In Figure. 1 V_j^T and α_j^T are the amplitude gain and the time constant of the threshold adjuster, respectively.

Fast Linking Method

In order to process actual images under various conditions of illuminations, we apply PCNN with "fast linking", after the first signal is input, calculate all the output, and then refresh the linking territory. At last, the internal state is calculated, the output is decided. During the calculating process, if one of the neurons is changed, the linking territory is changed correspond. The calculation will continue until all the outputs are unchanged. Such cycle process is called one iterative. During this process, in order to keep the input unchanged, the linking territory will change constantly. The input wave transmit the data after one iteration is finished, while linking territory wave send information to all the elements of image during this iterative. This method is called Fastlinking. It can decrease the effect of timing quantification. The flashings in original model are all separated because the time delay of the linking field. While adopting the Fastlinking model, the neuron can be flashed in one terrirory, shown in Figure 2.

In the model of PCNN, the linking coefficient β plays an important role. The larger β is, the further distance is transmitted. It can be obviously seen in the Fastlinking model. Figure 3 shows the segmentations of different β. It also can be seen from Figure3 (b) and (c) that, for the connected territory, the value of β influences the amount of neurons.

3. Multi-object segmentation using improved PCNN

Supplementary term

In Fastlinking model, the effect of β only is limited in the connected area that is, in the connected area, between the neurons pulsed with the highest grey value and those with the lowest the range of grey will enlarge because β increased. But this will not influence those unconnected neurons. This property can make the different results even they are in the same grey degree, the neurons connected to the higher grey degree will be fired, but those unconnected to the higher grey degree will not be fired. The segmented image can be seen in figure 4.

For those duties using PCNN separate the connected target territory from the background, Fastlinking model is more advanced. But to those which want to separate multiple targets from different background synchronously, the traditional Fastlinking has some limited, such as not making the firing synchronously.

We can describe this type segmentation as: to realize synchronous firing of the objects with different intensity as many as possible. To address this problem, we make an improvement to the model. We define a loop variable t in a step. An iteration of variable t is defined as a sub routine. From the second sub routine, a correction term defined as Eq (6) was added into internal activity:

$$L_j^{add}(t) = \frac{F_j(t-1)}{F_{j\min}(t-1)} * E_j \tag{6}$$

Where $F_j(t-1) = \begin{cases} F_j(n) & j \in (\Phi_F(t-1) \setminus \Phi_Y(t)) \\ 0 & otherwise \end{cases}$

$$\Phi_F(t-1) = \{j \mid F_{\min}(t-1) < F_j(n) < F_{\max}(t-1)\}$$

$$\Phi_Y(t-1) = \{j \mid Y(t-1) = 1\}$$

Where, $F_{\min}(t-1)$ and $F_{\max}(t-1)$ are the highest and lowest intensity pulsed in t-1 sub-step respectively. E_j is the edge data of the image. Then, the internal activation of the neuron can be modeled as follow

$$U_j^{\mathrm{add}}(t) = F_j(n)(1 + \beta L_j(t) + \gamma L_j^{\mathrm{add}}(t)) \tag{7}$$

where γ is the power index, the larger γ is, the stronger the correction is, and vice versa. The correction term is zero when γ=0 and the model is the classic PCNN model with Fastlinking.

The internal activity of neurons on the edges of the object-regions with lower highest-grayscale are raised to the threshold to fire by inducing the edge data into the proposed model, and some neurons linking with them are captured. So the object-regions with lower highest-grayscale are segmented. In the model, it is β to mainly determine the range of the intensity of synchronous pulsed neurons and correction term to expand the space of the wave propagation to generate firing seeds in unlinking region. The regions created by seeds capturing the linking neurons, separated by the edges, are each marked in order to be convenient to be segmented.

Determination of β

The linking strength β should be satisfied: 1) To ensure that the points in target area will be captured, 2) To ensure that the points in background cannot be captured. If the condition 1) is satisfied, β must be large enough; while 2) is satisfied then β will be smaller. From the principle of firing, the neuron must satisfy:

$$U_j(n) \geq \theta_j(n) \tag{8}$$

Take the first firing as an example, let $\theta_{ij}(0) = f_{\max} = \max(S)$, choose the neuron P from the fired part freely, its input is f_P then according to Eq (3) and Eq (8): $f_P(n)(1 + \beta l_P(t)) \geq f_{\max}$ make small transformation:

$$\beta \geq \frac{\Delta f}{l_P(t) \times f_P(n)} \tag{9}$$

where, $\Delta f = f_{\max} - f_P(n)$ is the distance between input and the threshold value, which denotes the impulse range of PCNN in one iterative. Let Y (t-1) be the output of t-1sub-iterative. Then there will be two uttermost conditions of neighborhood of a neuron in the t^{th} iterative linking area, all neurons are fired or the only neighbor neurons are fired, we can describe by the matrix as follows:

$$Y_{ij}^{\max}(t-1) = \begin{bmatrix} 1 & 1 & 1 \\ 1 & 0 & 1 \\ 1 & 1 & 1 \end{bmatrix} \quad \text{or} \quad Y_{ij}^{\min}(t-1) = \begin{bmatrix} 0 & 0 & 1 \\ 0 & 0 & 0 \\ 0 & 0 & 0 \end{bmatrix}$$

The central neuron is (i, j). If we adapt 1/r kernel, and let *r*=3, then combine Eq (3) and Eq (9), the range of β is:

$$\beta \geq 1.0\frac{\Delta f}{f_P(n)} \tag{10}$$

or

$$\beta \geq 11.1\frac{\Delta f}{f_P(n)} \tag{11}$$

But in practice, it is between two extremely condition. If all the neurons with higher grey value than f_P (including f_P) are expected to be fired, β defined by form Eq (11) must be chosen. Then the smallest input fired is:

$$f_{P\min}(n) = \frac{f_{\max}}{(1+11.1\frac{\Delta f}{f_P(n)})} \tag{12}$$

If form Eq (10) is selected as a rule, then f_P must be the smallest pulse value, some neurons whose value are higher than f_P will not be fired. Considering the above two rules, we adapt form Eq (11) as the basic rule, f_P will be decided by the whole grey distribution. From the segmentation example, to the above duty, when we select the first trough(12) near the highest value as f_P, the result is better.

Improved PCNN algorithm

The parameters we selected ensure that we can get the result in one iterative. The last result is saved in Y. The detailed steps are as followed:

Step1: Initialization

Let the unitary image grayscale value as the impulse signal S_j;

Make differential to the image; get the verge value of image as the outside input;

Initialize the parameter of the net;

Step2: Fastlinking processing: t is the iterative variable

a) Let t_{max}=20; t=1;

b) The first iterative t=1, from Eq (1)~(5),calculate each PCNN internal and output part, then put the result in $Y_{ij}^{(t)}(1)$;

c) From t=2 iterative, internal activation defined by Eq.(3) will be replaced by Eq.(7), then calculate each PCNN internal and output part, then put the result in $Y_{ij}^{(t)}$;

d) t=t+1;

e) If t<t_{max} then move to step c); otherwise output Y_{ij}.

Figure 5 shows the segmentation of the improved PCNN algorithm on the image of figure 4(a) The maximum grey level of each fired region are 1.0 and 0.8 respectively.

4. Experiments

We test some image with multi-target in improved model. Figure 6 and Figure 7 show the examples of the proposed PCNN and Fastlinking model. Fig6 contains the images composed of the objects include long, thin lines and the background has large homogenous areas, such as the handwriting. Fig7 contains the images composed of many grains of target objects, which are much smaller than the background such as the rice. The proposed PCNN is fit for such simple background image. Obviously, the improved PCNN model segment more target area than Fastlinking model. Figure 8 shows some segmentations for natural images. (a) is the original input image; (b) is the segment output, including the target area and non-target area. For this example, we adapt geometry characteristic method to segment the area. Firstly, the pixel number in this area must satisfy some extent. The other area is the non-target area. Then approximate the target area, for example, the shape of plane is approximated by “T” shape or cross shape; the shape of face is approximated by oval. After such filtering, the area left is the target. Column (c) shows the extracted objects. The traditional PCNN model has better robustness to noise. The paper compared the improved PCNN model and the traditional Fastlinking model. Figure 9 is the compared results. The former figure is the rate of false negative; the latter figure is the rate of false positive. From them, it can be seen that, to lowly noised image, the rate of false negative in improved PCNN model is lower than traditional Fastlinking model. To highly noised image, the rates of false negative in improved PCNN model are different for the edge detectors. Canny operator is the lowest, Sobel is lower, Prewit operator and Sobel operator are near. When considering the rate of false positive, Prewit operator and Sobel operator are almost same and the lowest; traditional Fastlinking is higher; Canny operator is the highest. Considering the above analysis, Sobel operator is more suitable for this application.

5. Conclusion

This paper improved the traditional PCNN model and added verge data at the input target area. Neurons of different grey in the non-connected area are fired simultaneity, in order to realize segmentation of multi-target. The paper defined the important parameter β which is the key factoring firing processor. The experiment proved that in the multi-object identification the improved PCNN model is superior to the traditional Fastlinking model for the non-noise image. When existing noise in the image, decreasing the noise is affected by the verge detection operator. All in all, proposed PCNN model preserved the advantages of traditional Faslinking, and realized the simultaneous segmentation of multiple objects.

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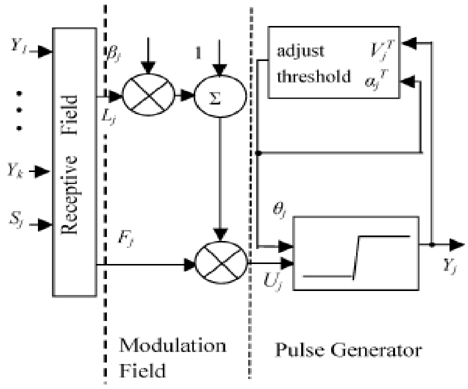


Figure 1. Simplied PCNN

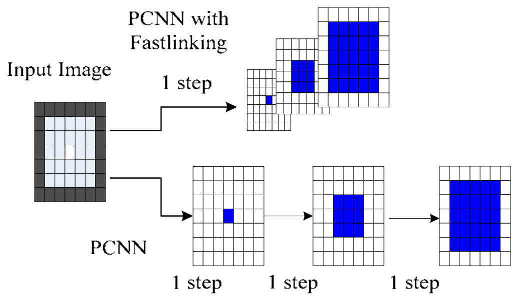
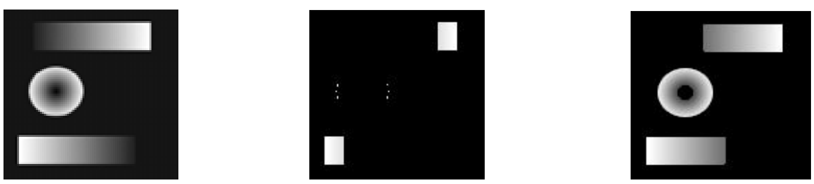
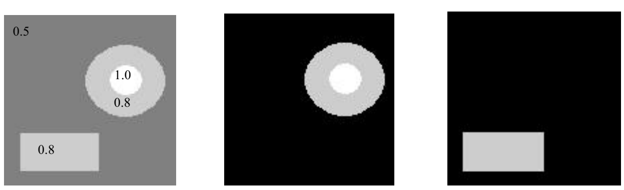


Figure 2. Fastlinking model



(a) Original image (b) $\beta = 0.2$ (c) $\beta = 5$

Figure 3. Segmentations with different β



(a) Original image (b) results image

Figure 4. Segmentation example of Fastlinking PCNN ($\beta = 1.0$)

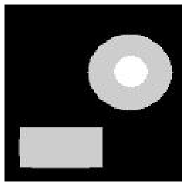
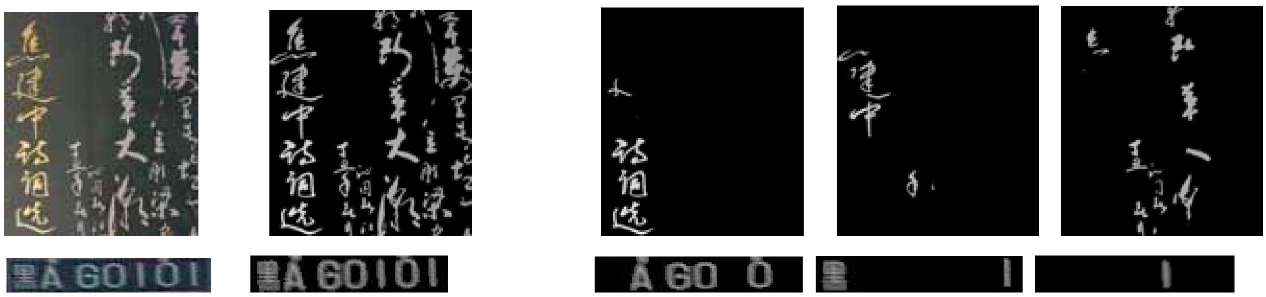


Figure 5. The segment result of improved PCNN



(a)original image (b) segment in improved PCNN (c) segment in Fastlinking model

Figure 6. The segment test between improved PCNN and Fastlinking model.

(a) original image (b) segment in improved PCNN (c) segment in Fastlinking model

Figure 7. The segment test between improved PCNN and Fastlinking model.

(a) original image (b) segment in improved PCNN (c) abstract target area

Figure 8. The segment test of improved PCNN

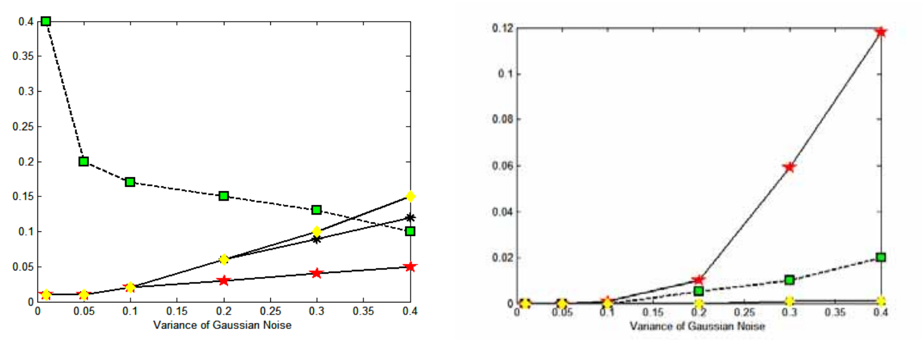


Figure 9. The segment compare between improved PCNN and Fastlinking model. Dash line is the traditional Fastlinking model; -◆-,-*-and -★-are the Prewit operator, Sobel operator and canny operator. The left figure is the rate of false negative; the right figure is the rate of false positive.

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Multi-level Frontier based Topic-specific Crawler Design with Improved URL Ordering

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Abstract

The rapid growth of World Wide Web has urged the development of retrieval tools like search engines. Topic specific crawlers are best suited for the users looking for results on a particular subject. In this paper, a novel design of a topic specific web crawler based on multi-agent system is presented. The architecture proposed employs two types of agents: retrieval and coordinator agents. Coordinator agent is responsible for disseminating URLs from crawling frontiers to individual retrieval agents. The URL frontier is modeled as multi-level queues to implement tunneling and is populated with URLs by a rule based engine. The coordinator agent dynamically assigns URLs to retrieval agents to avoid downloading non productive and duplicate Web pages. The empirical results clearly depict the advantage of using multi-level frontier queues in terms of harvest ratio, time, and downloading highly relevant Web pages.

Keywords: Topic-specific crawler, Multi-agent system, Rule based system, Multi-level frontier queue

1. Introduction

The World Wide Web (WWW) or Web can be sighted as a huge distributed database across several million numbers of hosts over the Internet where data are stored as Web pages on Web servers. The logical relationship between these Web pages is represented by hyperlinks. Using the hyperlinks, users can navigate through the Web pages with the help of browsers which is time consuming. Another way to locate information is through search engines. As the size of WWW is colossal, search engines become a vital tool to search for information. Different search tools can be compared with the number of indexed pages, quality of returned pages, and response time. Google has gathered about 25 billion Web pages during 2006, but covers only 40% of the publicly available Web pages. Most of the empirical studies say that Google outperforms in both technology of gathering Web pages and indexing methods. Because of its huge collection most of the search results are irrelevant to the users. Now-a-days, people began to use more specialized search tools which will fetch only those URLs that are more important to them.

Web spiders or crawlers or robots are the main component of every search engine. The crawlers visit Web pages across the Web, following hyperlinks from site to site, storing downloaded pages as they visit, to build a searchable index. A crawler design is termed as good if it handles certain issues in an efficient way (Cho et.al., 1998). The design criteria mentioned below endorse the fact that every crawler should posses the techniques to handle them in an appropriate manner:

- avoid overloading Web servers or network lines
- deal with large volume of data
- should have unlimited resources and time (ideal requirement)
- decide on a particular link to follow
- decide on re-crawl policy to keep the database fresh

General purpose crawlers have the disadvantage of processing large portion of the Web in a centralized manner.

Currently, some of the general purpose search engines employ distributed crawlers and distributed indexes. A collection of distributed focused crawlers can collect more specialized topics in more depth and can maintain the freshness of the indexes since the re-crawl covers only less portion.

Unlike general-purpose web crawlers, focused crawlers are designed to gather pages on specific topic. A focused crawler tries to “predict” whether a target URL is pointing to a relevant and high-quality Web page before actually fetching that page. There are many works in focused crawling. Researchers have employed machine learning approaches to improve the crawler’s functionalities in many aspects. Classification is one of the supervised learning algorithms that is used to determine the relevance of a Web page. Another important application of the above approach is to build and maintain Web directories or portals. In semi-supervised machine learning approaches like reinforcement learning, the agents learn progressively by interacting directly with the dynamic environment. Because of the dynamism, the agents are never told about the correct action, instead they are told about how good or how bad its action was.

As the scenario of information search in a large environment such as Web becomes tough, there is some assistance in the form of agents. The agents can act as autonomous computational entities for accessing, discovering and retrieving information on the Web. Unlike objects, an agent is defined in terms of its behavior. The characteristics of agents have made its use in several research areas in multi disciplinary domains. The properties like its autonomy, cooperativeness among other agents to accomplish a task, and its adaptability to the changing environment makes it suitable to Web search scenario.

Since a single agent approach may be inefficient and impractical for the large scale IR environment, most of the systems employ multi-agent systems. This paradigm has become more and more important as they represent a new way of analyzing, designing, and implementing complex software systems. In multi-agent systems, communication and organization enables the agents to cooperate and coordinate their actions. There are number of communication languages like KIF (Knowledge Interchange Format) (Genesereth et.al 1992), KQML (Knowledge Query and Manipulation Language) (Finin et.al 1994), and ACL (Agent Communication Language) (Labrou et.al 1999).

In this paper, an architectural framework is presented for crawling topic specific Web pages using multi-agent based system. The framework consists of multiple, parallel crawling agents and a master agent to coordinate them. In this prototype, the crawling agents are incorporated with intelligence which guides them in deciding the appropriate URL to download next. This feature is enabled with the help of an enhanced rule based system employing multi-level crawl frontier. The rule based system has the capability of harvesting a relevant URL after visiting some URLs with less relevance scores. This concept is called tunneling.

The remainder of this paper is organized as follows: section 2 discusses on the work related to Web crawlers. In section 3, the architectural framework of the crawler is described. In section 4, experimentation and evaluation details are discussed. Finally in section 5, conclusions are presented.

2. Related Works

The Web in many ways simulates a social network: links do not point to pages at random but endorses the page authors’ idea of what other relevant or interesting pages exists. This information can be exploited to collect more on-topic data by intelligently choosing what links to follow and what pages to discard.

One of the pioneer approaches in ordering the URLs according to relevance is FISH SEARCH (De Bra et al., 1994). The system is query driven and considers only those pages that are matching the query and links that emanate from those pages. An improvement of Fish Search is proposed as Shark search (Hersovici et al.,1998). This algorithm uses weighted term frequency (TF) and inverse document frequency (IDF) measure to determine page relevance score. Another algorithm (Cho et.al) proposed a technique to reorder the URLs in the frontier queue according to various heuristics like page rank, in links count, back link count and combination of these attributes.

A soft focused crawler is proposed by Chakrabarti et al. (1999). This technique uses classifier to obtain a score. The main shortcoming of this technique is that it will not support tunneling i.e. following a path of off-topic pages.

A context-graph based crawler (Diligenti et al. 2000) and Cora’s crawler (McCallum et al., 1999) use tunneling concept. Cora is a domain specific search engine whose spider employs reinforcement learning algorithm. There are other systems like Web Topic Management System (Mukherjea, 2000) that fetches only those pages that are parent, child or sibling to on-topic pages. Bingo! (Sizov et al., 2003) is a crawling system that eliminates the initial training step and employs progressive training of the classifier with high quality pages. Menzcer et al. (2003) presented a framework to evaluate focused crawlers and developed a crawler based on an evolutionary algorithm. An information integration framework ALII is presented by Barfouroushi et al. (2002) which uses active logic. ALII, employs a compact context representation and build a hierarchy model of query and web pages. The crawler also does limited backward crawling using general search engine indices.

An intelligent crawling architecture is presented by Aggarwal et al. (2001) based on predicates. It uses a self-learning mechanism that can dynamically adapt to the particular structure of the relevant predicate. Several factors were considered during the crawl to evaluate its effectiveness, building a composite crawler.

Page relevance to the user's need is computed using ontology-based algorithm by Ehrig et al. (2003). Entities, the words occurring in the ontology are extracted from the page and are counted. Relevance of the page is computed with regard to user selected entities using several measures on ontology graph like direct match, taxonomic and more complex relationships. This system has improved harvest rate when compared to baseline focused crawler that decides on page relevance by a simple binary keyword match. Case based BDI-agent (Olivia et al., 1999) is a domain specific search engine that uses case based reasoning (CBR) as its learning component. It uses past results and reuse that for answering future queries.

3. Architecture

The components of the crawler are shown in figure 1. The system proposed has two types of agents: coordinator agent and number of retrieval agents. The retrieval agents download Web pages and classify them as relevant or irrelevant. Coordinator agent disseminates URLs to different retrieval agents. The system should consider three key issues.

1. deciding on whether the downloaded pages are on-topic or not
2. deciding on which URL to visit next
3. avoiding multiple agents downloading same page

The first issue is considered by retrieval agents. The last two issues are considered by coordinator agent. Our system concentrates in using an intelligent multi-agent based system that mines the information contained in both hyperlinks and content. The advantages are two fold in using two different agents for collecting information. One is to reduce the network traffic load and another is to parallelize the computation. The crawling system is fed with the inputs such as seed URLs, and topic specific query terms. With these inputs the crawler must predict the next URL to download.

The coordinator agent functions as an interface between the crawler agents. The interface collects the seed URLs from a trusted search engine like Google. An expert intervention is necessary in this step to identify the most important seed pages from millions of search results from the search engine. The URLs are then placed in the frontier queue along with the relevant scores given by the expert. The coordinator agent maintains four levels of frontier queues. First level contains the URLs which are relevant to the topic of search. Second level contains the URLs that are less relevant but following a URL in that level will lead to a relevant page. Similarly the remaining levels of frontier queues are implemented to achieve tunneling.

Initially, the coordinator agent spawns number of retrieval agents equal to the number of URLs in the first level of frontier queue. The queues in the lower levels are empty. The retrieval agents start downloading pages, parse them, and are given as input to the Naïve-Bayes classifier. The classifier in each retrieval agent classifies the page and returns the relevance scores which are then updated in the frontier queue.

As in the rule based system exploiting inter-class relationships proposed by Altingövde et al. (2004), the relevance scores are computed based on associations among the classes. In this proposed system, rules are generated and normalized relevance scores are computed for each downloaded page as follows:

$$R[p] = NHyp_{i\rightarrow i} / NHyp_{i\rightarrow i}\ NHyp_{i\rightarrow j} \qquad (1)$$

where $NHyp_{i\rightarrow i}$ is number of hyperlinks in a page that belong to the same class and $NHyp_{i\rightarrow j}$ is number of hyperlinks that belong to other classes. Instead of considering the probabilities, the counts are taken into account. This computation gives 20% more efficiency in finding quality Web pages. The URLs with updated relevance scores and a value for level attribute is communicated to coordinator agent. It performs URL seen test and inserts the parsed URLs in the appropriate queues with the help of values in level attribute. Since the frontier queue is global and the assignment of URL is centralized, there are little chances for downloading duplicate pages by different crawler agents. The resultant pages that are classified as relevant are stored in the database for later processing.

4. Discussion

The architecture described above is implemented using JADE framework. JADE is one of the promising agent development frameworks supporting the deployment of multiple agents. Each of the agents can dynamically discover other agents and can communicate according to peer-to-peer paradigm (Nikraz et al., 2006). FIPA ACL is used as the communication language.

For training the Naïve-Bayes classifier, Yahoo! topic taxonomy is used. The performance of the proposed crawler design is compared with two baseline crawlers:

(i) baseline crawler without rule based system

(ii) baseline crawler with single level frontier queue

One of the parameter to evaluate the performance of the crawler design is the harvest ratio which is defined as the average relevance of all pages retrieved on a particular topic.

$$\text{Harvest ratio} = \frac{\sum_{i=1}^{N} R(p_i)}{N} \qquad (2)$$

The performance comparison of all the three crawlers is shown in Figure.2.

The graph clearly depicts that the proposed crawler initially has low harvest rate. Two reasons were observed by carrying over the experiments repeatedly. They are:

1. time consumed in initial training of the classifier
2. time consumed in initial coordination among all retrieval agents for receiving URLs to store in different levels of queues

Besides these two observations, the newly proposed crawler outperforms both.

Another observation is made on the crawler design which is the time taken by the crawlers to download upto 7000 pages. The graph is depicted in Figure. 3.

The Figure. 4 depicts the time taken for downloading 1000 pages with different levels of queues. The graph shows the advantage that is gained due to implementation of frontier queues in multiple levels. It is much reduced when using 4 levels of frontier queues. This is due to the fact that the retrieval agents need not wait for the master agent to dynamically assign URLs. Implementing tunneling in different levels also increases the number of relevant pages harvested. The observation recorded is shown in Figure. 5.

The experimentation is done on different levels of frontier queues. It was found that the relevance scores of the harvested pages increased till the number of levels was 4. Adding one more level to the queue decreases the relevance score as topic of the low quality pages drifted from the topic of search.

5. Conclusions

This paper has discussed the issues of designing a focused crawler. The fast improvements in the computational tools help in framing a novel architecture for locating relevant information in the rapidly growing Web. This paper proposed a novel framework of implementing multi-level queues as URL frontier to realize tunneling concept. The proposed system employed multiple agents communicating with each other to achieve the goal of finding quality results. The system also employs a mechanism to avoid duplicate downloading of Web pages. The empirical results suggest that the relevance scores of the pages downloaded is quiet high and the time taken to download those relevant pages is less. This system can be improved further to alleviate the bottleneck due to the centralized task allocation by the coordinator agent.

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Figures

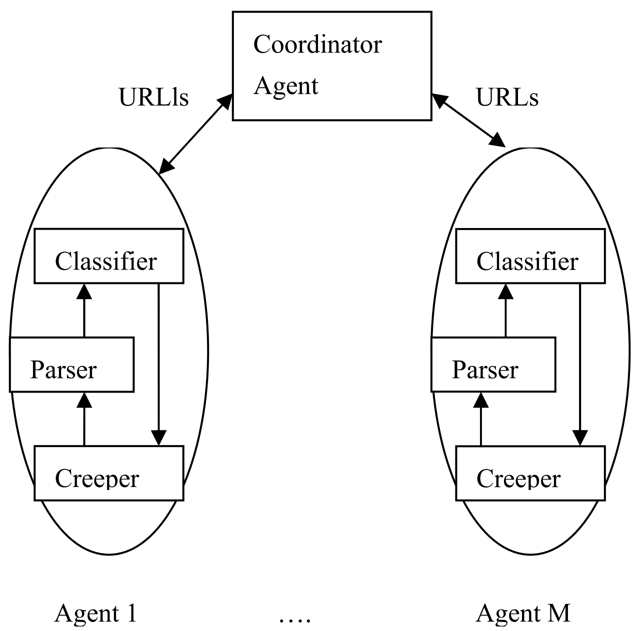

Figure 1. Architecture of the Web crawler

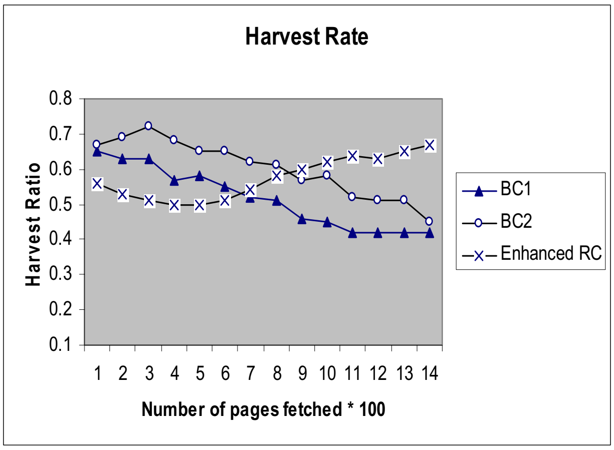

Figure 2. Harvest Ratio

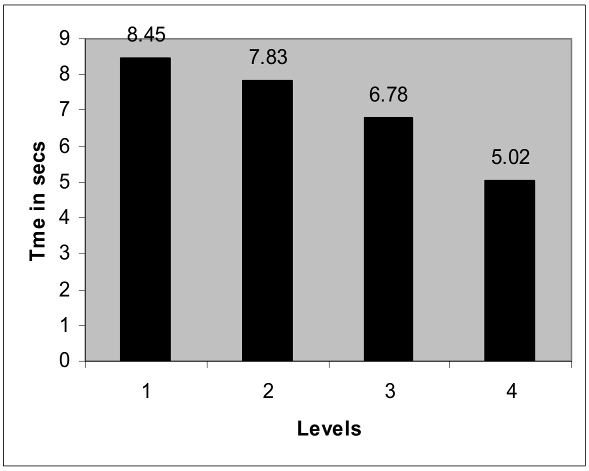

Figure 3. Time taken to download pages

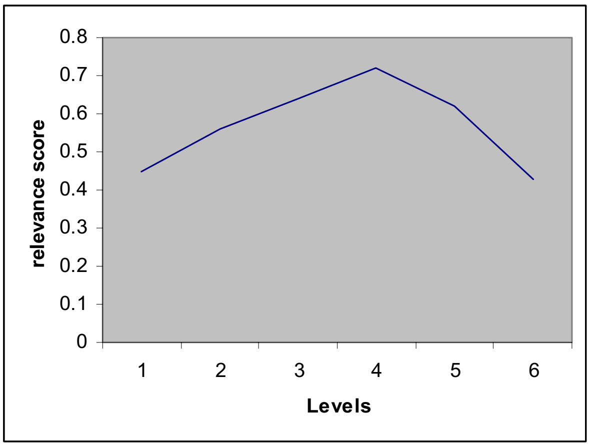

Figure 4. Number of levels in frontier Vs time taken to download 100pages

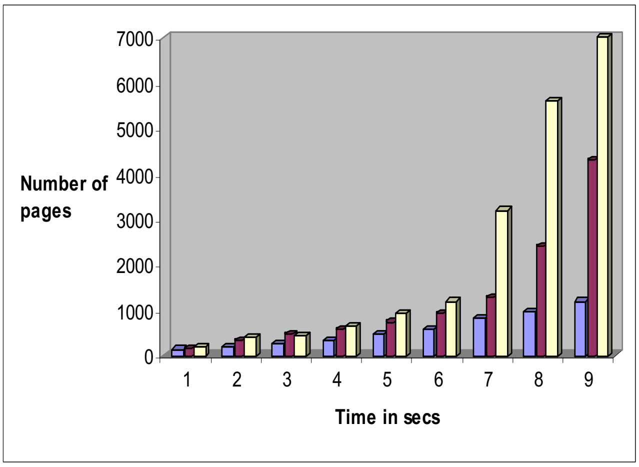

Figure 5. Number of levels Vs relevance scores

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Study on the Quality Evaluation System of College Information Network

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Abstract

At present, various colleges have established their own information networks. Aiming at actual demand to different colleges, the quality evaluation system of information network is urgent problem which should be solved. In this article, we will study the quality evaluation system of college information network from three aspects including the evaluation of the hardware system, the evaluation of the management system and the evaluation of the application system of the college information network.

Keywords: Information network, Research, Quality evaluation, Evaluation system

1. Introduction

With the quick development of computer network, domestic colleges establish the campus information network in succession, and the information networks of most colleges are continually reconstructed and gradually updated, and the campus information network has been the basic condition for teaching, scientific research and management. From network technology, network application and network management, the campus information network is the large-sized computer network with the highest network technical content, the most complete network application requirement and the most complex network management. The users facing the campus information network mainly include college teachers and students, so very high requirements are required for the campus information network whether from network technology, network application or network management, but many problems still exist in the campus information network, so it has not better fulfill college teachers and students' works and studies, and the campus information network needs to be optimized, updated and reconstructed. At present, the capital amount is the only factor to decide the layout and the construction when the campus information network is optimized, updated and reconstructed, so scientific theoretic base is deficient, and the layout is blind. For the campus information network, the hardware system is base, the management system is guarantee, and the application system is the attention. So to study the college information network is to study these three systems and the interior association among these three systems.

In this article, we will study the quality evaluation system of college information network from three aspects including the evaluation of the hardware system, the evaluation of the management system and the evaluation of the application system.

2. Quality evaluation of hardware system

When designing a campus information network, the hardware system should be first considered, and when evaluating a campus information network, the hardware system is the first factor. So we need analyze and study the hardware system to design and evaluate the campus information network.

2.1 Network system structure analysis

Figure 1 is the poetic campus information network principle sketch, and we analyze the sketch as follows.

The main composing of the network system structure is the topology structure connecting the network equipment, and the network technology adopted by the campus information network is the Ethernet technology. According to the size of college, the campus information trunk network can be divided into Ethernet of ten thousand-mega and kilomega Ethernet. The campus information network mostly adopts the three-tiers structure mode including the core tier, the cluster tier and the access tier, and the core tier is composed by one set or several sets of exchanger and these exchangers are connected by ten thousand-mega Ethernet and kilomega Ethernet, and the core tier and the cluster tier are connected by ten thousand-mega Ethernet and kilomega Ethernet, and the cluster tier and the access tier are connected by kilomega Ethernet and hundred-mega Ethernet, and the computer terminal access is accessed by hundred-mega, and according to the terminal quantity and application of computer to confirm whether the campus information trunk network is connected by kilomega or ten thousand-mega, and the router, firewall, IDS, load equilibrium and other network equipments can be updated and increased at any time according to the demand of application and management.

2.2 Server system structure analysis

The server system is the base of network service, and the performance of server should fulfill the service offered by network, and the system structure of server directly decides the quality of service offered by network, and the structure design of network server system can adopt the "three tiers/ multiple tiers computation" frame based on the Internet application service mode, and the three tiers model is seen in Figure 2. The functions of various tiers are described as follows.

(1) Client tier. It is the emitting station for user access and user request, and the typical application is the network browser.

(2) Server tier. The typical applications include the Web server and the application server to run operation code and logic.

(3) Data tier. The typical applications include relationship database and other back-end data resources such as Oracle database.

2.3 Analysis of network system exit design

At present, various college information networks are directly connected with CERNET by kilinega or hundred-mega, and most colleges possess second exit or third exit which respectively connected with Telecom or other nets. When one college information network has several exits, the key to design the exit of network system is to fully utilize the exit bandwidth and enhance the speed that college network users interview exterior nets to the maximum extents. We will analyze and study the design of network system exit from two aspects including the route technology and the bandwidth management.

(1) Route technology

The route technology can be divided into static route selection and dynamic route selection, and different route selections needs the supports of different hardware systems, so when we design the network system exit, we should first confirm whether the exit route is to select static route or dynamic route.

The selection of static route is to control the interview of different exits to different exterior nets, and the selection of dynamic route is the optimal route selection.

(2) Bandwidth management

The bandwidth management is to manage the network exit bandwidth occupied by college interior users when they interview the exterior net resource. From Figure 1, the bandwidth manager is the hardware equipment, and it is set in front of the route management equipment, and it is to manage the campus net exit width occupied by corresponding IP to users with different characters in the interior of the campus net.

3. Analysis of application system quality evaluation

The intention to construct campus information network is the application, and some basic applications such as WWW, FTP, BBS, E-mail and VOD of present college information network are mature, and what we should emphasize is the construction of college digital campus, i.e. we should establish an advanced digital campus from following aspects.

3.1 Software frame system

The application system frame design of college digital campus is mainly the total frame system of public database platform in the whole digital campus, and it is seen in Figure 3. It is composed by the operation tier, the display tier, the resource tier and the system tier, and its core is the operation tier.

(1) Operation tier

The operation tier is the core support tier to operate the public database platform, and it is composed by the operation objective tier, the public service tier and the communication gateway.

(2) Display tier

The main function of the display tier is to interview various application system function offered by the application service tier, and display the result returned by application logic to users through various technical means, and make uses could access the management information system platform through different interfaces and communication modes.

(3) Resource tier

The system resource tier could offer basic services for the running of the application service tier, and it includes the database system which can offer the storages of basic data and running data.

(4) Other system tier

For some existing operation systems, we should keep them or integrate them through EAI Server. We could adopt three cluster modes including data cluster, operation cluster and view cluster.

3.2 Performance requirements

(1) Running performance

For common colleges, there are many users, so the network should achieve that the network could support twenty thousand users, and the page interview speed could achieve three seconds/ 500 hundred people, and the maximum delay could not exceed 30 seconds, and the system can ensure the running of 7*24hours, and the system could support load equilibrium and extension.

(2) Education operation and teaching standard

The system could support the management information system construction standard of the Education Ministry of China, the SCORM standard, IMS standard, DLTS, IEEE LTSC, international current network agreement and route agreement and other opening agreement standards.

3.3 Door platform

(1) Single sign on (SSO)

The requirement that users log on the digital campus application platform through the door is that the system platform could complete users' single sign on when entering into another function application from one function according to users' role and limitation.

(2) Interview control based on strategy

The system could safely implement remote interview, and authorize that users only need one Web browser based on Java technology and one connection which could log on the internet at any time to see their individual desk contents. Once users could log on and pass the authentication, they could safely implement remote interview bag including the encrypted private interview to the data document on any interior document server.

(3) Limitation management

The system design could offer flexible limitation setups according to the authorizations of role, role+ organization, user, or organization, and the individual information must be highly secreted and only individual could inquire. According to the fixed mode, the back-stage management must strictly implement limitation control by managers, and the system information implements strictly limitation control according to the organization (college, department, specialty and class).

3.4 Selection of database

The database is the core part to establish the digital campus and teaching management system. Generally speaking, the database system of college digital campus has complex data type, huge data amount, the coherence of the data and high reliability, so the selection of database is the important content in the construction of the whole system. For the selection of database, we should consider the requirements including the mature product with maximum market share, the design based on B/S system structure, supporting SQL standard, supporting parallel technology and dual computer copy, supporting multiple CPU and multiple Cluster structures, offering database type and log image, continually completing data encrypt on media, supporting real time copy and stretch copy granularity and disaster recover, transparent and DDB technology, multithreading and multi-course technology and supporting various application development tools.

4. Quality evaluation analysis of management system

The college information network management system includes network equipment management, application management, security management and user management.

4.1 Network equipment management

As seen in Figure 1, the hardware system is the base in the college information network, and the hardware system includes not only good network equipment and scientific and reasonable network structure, but scientific and effective management to these network equipments, for example, the topology structure of the whole network and the flux analysis in the network, and the key of network equipment management is to deploy special management system aiming at various equipments in the information network, and offer perfect technical management means for network managers.

4.2 Application management

The intention of college information network construction is the application, and with the development of network technology, the applications of network are more and more complex, which needs us implement scientific classification and management to the network applications. We should differentiate which network information is especially for interior users, which network information is aiming at students, which network information is aiming at teachers and which network information is completely open. Only good classification to these network information and effective management to these network applications can fully ensure and utilize the application of the whole information network.

4.3 Security management

The network security is more and more emphasized in the college information network, and the network security management mainly includes two aspects such as anti-invasion and antivirus.

The anti-invasion is to prevent illegal operations and attacks from the interior and exterior of college information network. The antivirus is to prevent the spread of various network viruses from the interior and exterior of college information network, and eliminate existing viruses at the same time.

4.4 User management

The user management in college information network is mainly to manage interior users. To standardize the network behaviors of interior users, there are two means including technology and regulation to manage users.

The technical means is to manage the system through the access terminal authorization system and the exit user application. And the regulation is to mainly standardize network users' network behaviors through administration means.

5. Comprehensive evaluation

It is difficult to comprehensively evaluate one college information network, because there is not one fixed mode for the evaluation to different information networks, and the size, application and hardware support platform of different college information networks are different, so aiming at different college information networks, we can follow several principles.

5.1 Application

Because the intention to establish the college information network is the application, so when we evaluate the application of college information network, we should judge whether the application offered by the information network can fulfill the actual application of the whole college, and whether the application is stable, and whether there are corresponding technical application development and support.

5.2 Hardware

All applications of college information network needs the support of hardware platform, and when we evaluate the hardware system of college information network, we should judge whether the whole hardware support platform could fulfill the demand of application, and implement technical evaluation to the stability of the hardware system.

5.3 Management

The management is very important for the work of college information network. When we evaluate the management system of the college information system, we should do the works from two aspects. One is the technical means, i.e. whether we can implement good management to the hardware system, application system and network users technically. The second is whether we can standardize users' net behaviors from the system.

6. Conclusions

The comprehensive evaluation of college information network is a very huge and complex system. We should implement comprehensive evaluation for the college information network from three aspects including application system, hardware system and management system. For different colleges, if the college information system could fulfill the whole demand of college information network and possess higher stability, so it is the information network with high quality.

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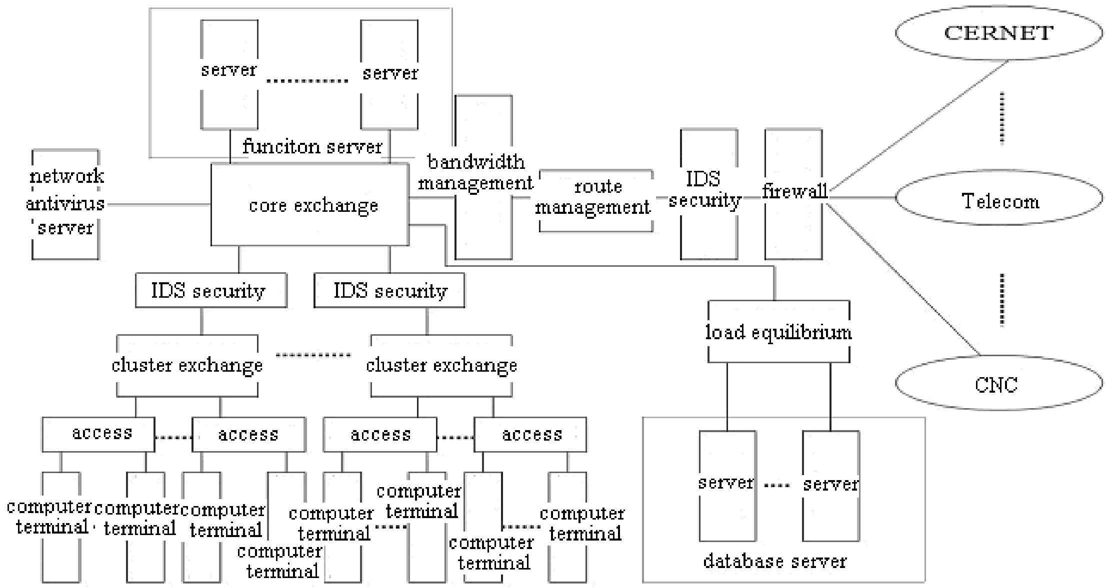

Figure 1. College Information Network Principle

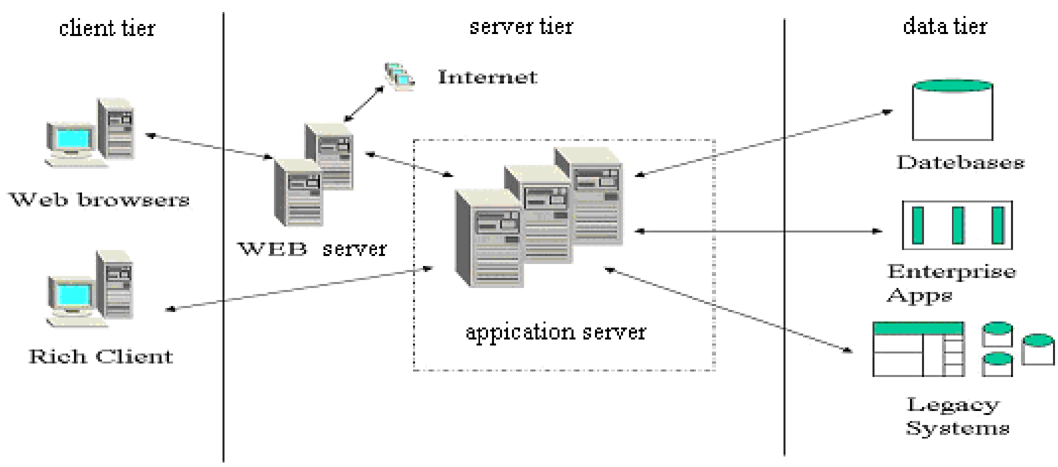

Figure 2. Server System Structure

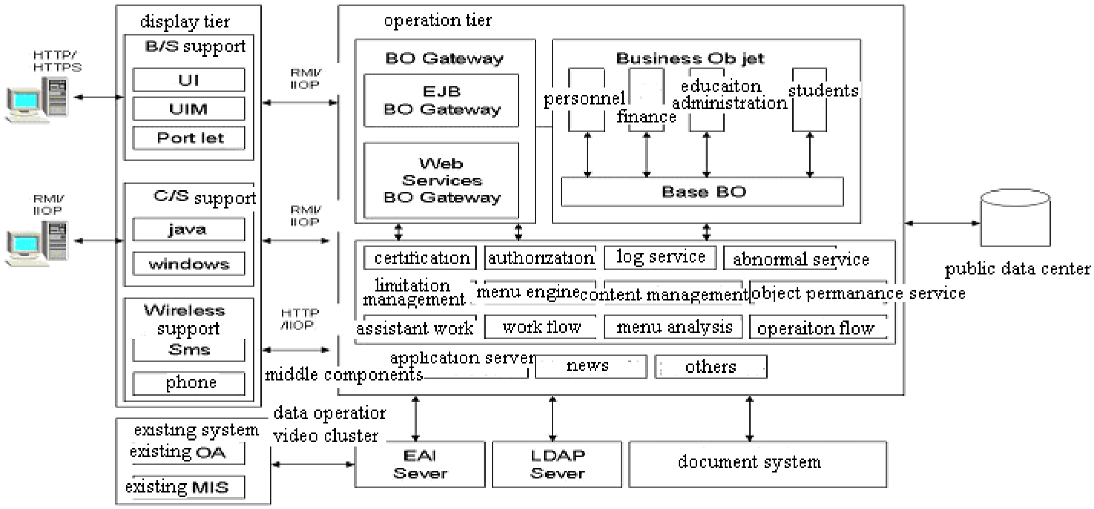

Figure 3. Structure Principle of Software System Frame

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Acoustic Pronunciation Variations Modeling for Standard Malay Speech Recognition

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Abstract

This paper presents different methods of handling pronunciation variations in Standard Malay (SM) speech recognition. Pronunciation variation can be handled by explicitly modifying the knowledge sources or improving the decoding method. Two types of pronunciation variations are defined, namely, complete or phone changes and partial or sound changes. Complete or phone change means that one phoneme is realized as another phoneme. Meanwhile, a partial or sound change happens when the acoustic realization is ambiguous between two phonemes. Complete or phone changes can be handled by constructing a pronunciation variation dictionary to include alternative pronunciations at the lexical level or dynamically expanding the search space to include those pronunciation variants. Sound or partial changes can be handled by adjusting the acoustic models through sharing or adaptation of the Gaussian mixture components. Experimental results show that the use of a pronunciation variation dictionary and the method of dynamic search space expansion can improve speech recognition performance substantially. The methods of acoustic model refinement were found to be relatively less effective in our experiments.

Keywords: Pronunciation variations, Standard Malay (SM), Complete changes, Partial Changes

1. Introduction

The tremendous growth of technology increased the need of integration of spoken language technologies into our daily applications, providing an easy and natural access to information. These applications are of different nature with different user interfaces. Besides voice enabled Internet portals or tourist information systems, Automatic Speech Recognition (ASR) systems can be used in home user experiences where TV and other appliances could be voice controlled, discarding keyboards or mouse interfaces, or in mobile phones and palm-sized computers for a hands-free and eyes-free manipulation.

Speech is a process used to communicate from a speaker to a listener. Pronunciation relates to speech, and humans have an intuitive feel for pronunciation. For instance, people chuckle when words are mispronounced and notice when foreign accent colors a speaker's pronunciations (Strik and Cucchiarini, 1999). If words were always pronounced in the same way, ASR would be relatively easy. However, for various reasons words are almost always pronounced differently and varied from one speaker to another and from one situation to another. The variability is due to co-articulation, regional accents, speaking rate, speaking style, etc. Pronunciation variation can be further classified into two types: *complete changes* and *partial changes* (Fung *et al*., 2000; Li *et al*., 2000; Saraclar and Khudanpur, 2000; Liu, 2002; Kam, 2003; Liu and Fung, 2003). Complete changes, or phone changes, are the replacement of a phoneme by another alternate phone, such as 'b' being pronounced as 'p'. Partial changes, or sound changes, are the variations within the phoneme such as nasalization, centralization, voiceless and voiced. Both complete changes and partial changes are very common in spontaneous speech. Research in ASR has gradually progressed from isolated words, via connected words and carefully read speech to conversational or spontaneous speech. Although many current application still make use of isolated word recognition example in dictation system, in ASR research the emphasis is now on spontaneous or conversational speech. It is clear that in going from isolated words to conversational or spontaneous speech the amount pronunciation increases. This is because spontaneous speech contains much more phone change (substituted, deleted, and inserted) phenomena and sound change (nasalized, centralized, voiced, voiceless, more rounded, syllabic, pharyngealized, and aspirated) phenomena because of variable speaking rates, moods, emotions, prosodies, co-articulations and so on, even when the speaker is tending to utter in canonical pronunciations (Greenberg, 1999). Other phenomena, such as lengthening, breathing, disfluency, lip smacking, murmuring, coughing, laughing, crying, modal/exclamation, silence, and noise, will also bring difficulties to ASR systems. At the linguistic level, there are a lot of spoken language phenomena, such as repetitions, ellipses, corrections, hesitations, and so on, resulting from the fact that people are often thinking while speaking in daily life. Therefore, since the presence of variation in pronunciation can cause errors in ASR, modeling pronunciation variation is seen as possible way of improving the performance of the current system.

There have been many studies on modeling pronunciation variations for improving ASR performance. They are focused mainly on two problems, namely the prediction of the pronunciation variants, and effective use of pronunciation variation information in the recognition process (Strik and Cucchiarini, 1999). Knowledge-based approaches use findings from linguistic studies, existing pronunciation dictionaries, and phonological rules to predict the pronunciation variations that could be encountered in ASR (Aubert and Dugast, 1995; Kessens *et al*., 1999). Data-driven approaches attempt to discover the pronunciation variants and the underlying rules from acoustic signals. This is done by performing automatic phone recognition and aligning the recognized phone sequences with reference transcriptions to find out the surface forms (Saraçlar *et al*., 2000; Wester, 2003). Some studies used hand-labelled corpora (Riley *et al*., 1999).

The key components of speech recognition system are the acoustic models, the pronunciation lexicon and the language models (Huang *et al*., 2001). The acoustic models are a set of hidden Markov models (HMM) that characterize the statistical variations of input speech. Each HMM represents a specific sub-word unit, e.g. a phoneme. The pronunciation lexicon and the language models are used to define and constrain the ways sub-word units can be concatenated to form words and sentences. They are used to define a search space from which the most likely word string(s) can be determined with a computationally efficient decoding algorithm. Within such a framework, pronunciation variations can be handled by modifying one or more of the knowledge sources or improving the decoding algorithm. Phone changes can be handled by replacing the baseform transcription with surface-form transcriptions, i.e. the actual pronunciations observed. This can be done by either augmenting the baseform lexicon with the additional pronunciation variants (Kessens *et al*., 1999; Liu *et al*., 2000; Byrne *et al*., 2001) or expanding the search space during the decoding process to include those variants (Kam and Lee, 2002). In order to deal with sound changes, pronunciation modeling must be applied at a lower level, for example, on the individual states of a hidden Markov model (HMM) (Saraçlar *et al*., 2000). In general, acoustic models are trained solely with baseform transcriptions. It is assumed that all training utterances follow exactly the canonical pronunciations. This convenient, but apparently unrealistic, assumption renders the acoustic models inadequate in representing the variations of speech sounds. To alleviate this problem, various methods of acoustic model refinement were proposed (Saraçlar *et al*., 2000; Venkataramani and Byrne, 2001; Liu, 2002).

In this paper, the pronunciation variations in spontaneous Standard Malay (SM) speech are studied. The linguistic and acoustic properties of spoken SM language are considered in the analysis of pronunciation variations and, subsequently, the design of pronunciation modeling techniques for the speech database. There are 500 million people that speak Bahasa Melayu or Bahasa Malaysia and it is the official language in Malaysia, Indonesia and Brunei. This language is part of Austronesian language and it is agglutinative in nature, that is the words in Bahasa Melayu are formed by joining syllables. The term “Standard Malay” (SM) is a term that is basically accepted by the speech community to be the norm or the prestige dialect, which is also the official language in Malaysia. It is widely believed that the so-called “Standard Malay” is based on the Johor-Riau Malay (JM) dialect, mainly spoken in the southern part of the peninsular Malaysia. There are other three dialects, namely Kelantan Malay (KM), Ulu Muar Malay (UMM) and Langkawi Malay (LM) which are soken in the different parts of peninsular Malaysia (Teoh, 1994).

There are some common features between the Malay language and English language. Firstly, Malay language is a phonetic language and it is written in Roman characters. Secondly, all syllables in the Malay language are pronounced almost equally and it is thus, considered as a non-tonal language. In general, there are six (6) main vowels and 29 consonants in SM. SM have a total of nineteen of the consonants, where /m/, /n/, /f/, /l/, /s/ and /y/ are pronounced almost the same way as in English. In Malay language, the syllabic structure is well-defined and can be unambiguously derived from a phone string. The basic syllable structure of the Malay language is generated by an ordered series of three syllabication rules. The linguists claimed that Malay is a Type III language (Teoh, 1994), namely Consonant-Vowel (CV) and Consonant-Vowel-Consonant (CVC) are the most common and they can be found almost in every Malay primary word. Based on the CV(C) structure, coda is optional for the syllable in Malay language and open syllable are commonly found. For Standard Malay language, the alphabets in a word itself is good enough to identify its pronunciation. However not all words, can be pronounced exactly as it written. The structure of the syllable (open or closed) and the position of the syllable (initial, middle or final) control the distribution of the SM vocalic segments. For instance the vowels /e/ and /o/ does not occur in open syllables. If this vowels occur, they will be removed by the final deletion rule (example: *"bazir"* = /b.a.z.e./ (waste)). SM also does not have /a/ occurring in the final position except in borrowed words such as a *"baba"* /b. a. b. a./ (Malaysia Straits born Chinese) and *"lawa"* /l. a. w. a./ (attractive, beautiful). This types of pronunciation variations can be classified as complete or phone changes.

Meanwhile the SM consonants are non-syllabic, the SM vowels are syllabic. Oral vowels can also be nasalized vowel, where vowels are followed by nasal sound. For example the SM word *"minggu"* (week) /m. i. ng. u/, where the vowel "i" is nasalized as it is followed by the velar nasal /n/. The length of the vowel is not distinctive and it is not a feature that differentiates one vowel phoneme from another. In addition to the vowel depending on the context in which they occur, vowels can be long or short, and these pronunciation variations could be classified as partial or sound changes. There are lots more word variations if we compare between native and non-native speakers pronunciations because not all people that speak Standard Malay use the same pronunciation (El-Imam *et al*., 2000) and it more variations during spontaneous speech.

2. Development of speech database

In building the Standard Malay speech database, the selections of utterances are derived from Buletin Utama TV3 Broadcast News that contains about 550 utterances for four hours news. All recognition experiments described in this paper use the Hidden Markov Toolkit (HTK) version 3.2 (Young *et al.*, 2001). We trained the recognition model based on syllables that formed by concatenating three types of phonological units: the Initial, the Middle and the Final that represented as a sequence of (SM) characters as shows in Table 1 and Table 2. For the purpose of this study, we focus on the Initial and Final (IF) representation. As a recognition feature, we extract 12 mel-frequency cepstral coefficients (MFCCs) with a logarithmic energy for every 10 ms analysis frame, and concatenate their first and second derivatives to obtain a 39-dimensional feature vector. During training and testing, we apply cepstral mean normalization and energy normalization to the feature vectors. The whole training procedure is divided into two stages, where monophone and triphone stages should be applied. In each stage, there are always two steps, which are repeated iteratively by estimation and realignment. The process begins with the training of the monophone models, followed by training of the triphone models. The acoustic models are based on 3-state left to right, context-dependent, 4-mixture, and cross-word triphone models, trained using the HTK toolkit (Young *et al.*, 2001).

3. Recognition modeling and implementation

3.1Modeling pronunciation variation for complete/phone changes

The pronunciation lexicon used in the baseline system provides only the baseform pronunciation for each of the word entries. In real speech, the baseform pronunciations are realized differently, depending on the speakers, speaking styles, etc. A pronunciation model (PM) is a descriptive and predictive model by which the surface-form pronunciation(s) can be derived from the baseform one. There have been three different types of models proposed by previous studies such as the phonological rules for generating pronunciation variations (Wester, 2003; Kessens *et al*., 2003), a pronunciation variation dictionary (PVD) that explicitly lists alternative pronunciations (Aubert and Dugast, 1995; Kessens *et al*. 1999; Liu *et al*., 2000], and the statistical decision trees that predict pronunciation variations according to phonetic context (Riley *et al*., 1999; Fosler-Lussier, 1999; Saraçlar *et al*., 2000].

In this study, two different approaches to handling phone changes in Standard Malay ASR are formulated and evaluated. The first approach uses a probabilistic PVD to augment the baseform lexicon. This is a straightforward and commonly used method that has been proven effective for various tasks and languages (Strik and Cucchiarini, 1999). In the second approach, pronunciation variation information is introduced during the decoding process. Decision tree based PMs are used to dynamically expand the search space. In (Saraçlar *et al*., 2000) a similar idea was presented. Decision tree based PMs were applied to a word lattice to construct a recognition network that includes surface-form realizations.

In the first approach, the PVD, each word can have multiple pronunciations, each being assigned a word-level variation probability (VP). The PVD can be obtained from the IF confusion matrix. The word-level VP is given by multiplying the phone-level VPs of all the individual phonemes in the surface-form pronunciation. With the use of the PVD, the goal of speech recognition is essentially to search for the most probable word sequence by considering all possible surface-form realizations. This can be made as

$$W^* = \arg\max_{W} P(O \mid B) P(B \mid W) P(W) \tag{1}$$

where $P(O\mid B)$ and $P(W)$ are referred to as the (sub-word level) acoustic models and the language models, respectively. $P(B\mid W)$ is given by a pronunciation lexicon. Further Eq. (1) can be modifying as

$$W^* = \arg\max_{W,k} P(O\mid S_{W,k})P(S_{W,k}\mid W)P(W) \tag{2}$$

where $S_{W,k}$ denotes one of the surface-forms realizations of W. $P(S_{W,k}\mid W)$ are obtained from the word-level VPs.

The PVD includes both context-independent and context-dependent phone changes. Since each word is treated individually, the phonetic context being considered is limited to within the word. To deal with cross-word context-dependent phone changes, we propose applying pronunciation models at the decoding level. Our baseline system uses a one-pass search algorithm (Choi, 2001).

In the second approach, a context-dependent pronunciation model is needed to predict the surface-form phoneme given the baseform phoneme and its context. It is implemented using the decision tree clustering technique, following the approaches described in (Riley *et al*., 1999; Fosler-Lussier, 1999). Each baseform phoneme is described using a decision tree. Given a baseform phoneme, as well as its left context (the right context is not available in a forward Viterbi search), the respective decision-tree pronunciation model (DTPM) gives all possible surface-form realizations and their corresponding VPs (Kam and Lee, 2002). Like the confusion matrix, the DTPM is trained with the phoneme recognition outputs for the HTK toolkit training utterances. The training involves an optimization process by which the surface-form phonemes are clustered based on phonetic context. At a particular node of the tree, a set of "yes/no" questions about the phonetic context are evaluated. Each question leads to a different partition of the training data. The question that minimizes the overall conditional entropy of the surface-form realizations is selected for that node. The node-splitting process stops when there are too few training data (Kam, 2003).

3.2 Modeling pronunciation variation for partial/sound changes

Unlike phone changes, a sound change cannot be described as a simple substitution of one phoneme for another. It is regarded as a partial change from the baseform phoneme to a surface-form phoneme (Liu and Fung, 2003). Our approaches presented below attempt to refine the acoustic models to handle the acoustic variation caused by sound changes. The acoustic models are continuous-density HMMs. The output probability density function (pdf) at each HMM state is a mixture of Gaussian distributions. The use of multiple mixture components is intended to describe complex acoustic variabilities. In this study, we investigate both the sharing and adaptation of the acoustic model parameters at the mixture level (Kam *et al*., 2003). In the first approach, sharing of mixture components is applied and the states of the baseform and surface-form models are aligned. It is assumed that both models have the same number of states. Then, state *j* of the baseform model is aligned with state *j* of the surface-form model. Consider a baseform phoneme *B*. The output pdf at state *j* is given as

$$b_j(o_t) = \sum_{m=1}^{M} w_{jm} N(o_t;\mu_{jm},\Sigma_{jm}) \tag{3}$$

where M is the number of Gaussian mixture components, and w_{jm} is the weight for the m th mixture component. The baseform output pdf can be modified to include the contributions from the surface-form states

$$bj'(o_t) = VP(B,B).b_j(o_t) + \sum_{n=1}^{N} VP(S_n,B).qs_n,j(o_t)$$

$$S_n \neq B \tag{4}$$

where S_n denotes the n th surface-form of B, N is the total number of surface-forms, $VP(S_n,B)$ is the variation probability of S_n with respect to baseform B, and $qs_n,j(o_t)$ denotes the output pdf of state j of the n th surface-form model.

The number of mixture components in the resultant baseform model depends on N. More surface-form pronunciations bring in more mixture components to the modified baseform state. As the number of mixture components is changed, re-estimation of mixture weights is required. Although sharing mixture components yields an acoustically richer model, it also greatly increases the model size for which more memory space and higher computation complexities are required. Moreover, if the baseform and surface-form mixture components are very similar, including them all in the modified baseform is unnecessarily superfluous.

For the second approach, we propose to refine the baseform acoustic models through parameters adaptation. The total number of model parameters remains unchanged. Like in the approach of mixture sharing, the states of the baseform and surface-form models are aligned. The surface-forms are generated from the IF confusion matrix. Consider the aligned states of the baseform phoneme B and one of its surface-forms S. Let $m_B(i)$ and $m_S(j)$ denote the *i*th mixture component in the baseform state and the *j*th mixture component in the surface-form state, respectively, where

$i, j = 1,2,..., M$. The distances between all pairs $(m_B(i), m_S(j))$ are computed. Then each surface-form component is paired up with the nearest baseform component. That is, for each $m_S(j)$, we find

$$\hat{i} = \arg\min_{m_B(i)} d(m_B(i), m_S(j)) \tag{5}$$

The "distance" between two Gaussian distributions is calculated using the Kullback-Leibler divergence (KLD) (Myrvoll and Soong, 2003). Given two multivariate Gaussian distributions f and g , the symmetric KLD has the following closed form

$$d(f,g) = \frac{1}{2} trace\{(\Sigma_f^{-1} + \Sigma_g^{-1})(\mu_f - \mu_g)(\mu_f - \mu_g)^T + \Sigma_f \Sigma_g^{-1} + \Sigma_g \Sigma_f^{-1} - 2I\} \tag{6}$$

where μ and Σ denote the mean vectors and the covariance matrices of the two distributions, respectively, and I is the identity matrix.

As a result, for this pair of baseform and surface-form states, each Gaussian component $m_B(i)$ is associated with k surface-form components, as illustrated in Figure 1. The centroid of these k components is computed. If the baseform B has n surface forms, there will be n such centroids. These surface-form centroids and the corresponding baseform component are weighted with the VP, and together produce a new centroid that is taken as the adapted baseform component. In this way, the adapted model is expected to shift towards the surface-form phonemes. The extent of such a shift depends on the VP. The mean and covariance of the centroid of k weighted Gaussian components can be found by minimizing the following weighted divergence

$$\{\mu_c', \Sigma_c'\} = \arg\min_{\mu_c, \Sigma_c} \sum_{n=1}^{k} a_n d(f_c, f_n) \tag{7}$$

where f_n denotes the nth component and a_n is the respective weighting coefficient. Assuming diagonal covariances, the weighted centroid is given as (Myrvoll and Soong, 2003)

$$\mu_c'(i) = \frac{\Sigma_{n=1}^{k} a_n (\Sigma_c^{-1}(i) + \Sigma_n^{-1}(i))\mu_n(i)}{\Sigma_{n=1}^{k} a_n (\Sigma_c^{-1}(i) + \Sigma_n^{-1}(i)}$$

$$\Sigma_c'(i) = \sqrt{\frac{\Sigma_{n=1}^{k} a_n \left[\Sigma_n(i) + (\mu_c(i) - \mu_n(i))^2\right]}{\Sigma_{n=1}^{k} a_n \Sigma_n^{-1}(i)}} \tag{8}$$

4. Results and discussions

4.1Pronunciation variation modeling for complete/phone changes

The recognition results with the use of PVDs that are constructed with different values of the VP threshold are shows in Table 3. The baseline system uses the basic pronunciation lexicon that contains 451 words. The size of the PVD increases as the VP threshold decreases. It is obvious that the introduction of pronunciation variants improves recognition performance. The best performance is attained with a VP threshold of 0.05. In this case, the PVD contains 568 pronunciations for the 451 words. With a very small value for the VP threshold, e.g. 0.02, the recognition performance is not good because there are too many pronunciation variants being included and some of them do not really represent pronunciation variation.

The recognition result attained by using the DTPM for dynamic search space expansion is show in Table 4. It appears that this approach is as effective as the PVD. Unlike the results for the PVD, the performance with a VP threshold of 0.2 is better than that with a threshold of 0.05. This means that the predictions made by the DTPM should be pruned more stringently than the IF confusion matrix. Because of its context-dependent nature, the DTPM has relatively less training data, and the variation probabilities cannot be reliably estimated. It is preferable not to include those unreliably predicted pronunciation variants.

By analyzing the recognition results in detail, it is observed that many errors are corrected by allowing the following pronunciation variations:

Initials: [b]→[p] or [m], [d]→[l] or [t], [g]→ [k], [s]→ [t] ,[t]→ [d]

Finals: [ang]→[an], [sy]→[sa] or [su], [ng]→[m] (syllabic nasal)

These observations match well with the findings in sociolinguistic studies on Standard Malay phonology.

4.2 Pronunciation variation modeling for partial/sound changes

The recognition results attained with the two methods of acoustic model refinement as shows in Table 5. The VP threshold for surface-form prediction is set at 0.05. Apparently, both approaches improve recognition performance. The sharing of mixture components seems to be more effective than adaptation. However, this is at the cost of a substantial increase in model complexity. The baseline acoustic models have a total of 2,251 Gaussian components. The adaptation approach retains the same number of Gaussian components. The models obtained with the sharing approach have 2,620 components, 17% more than the baseline. If we use an equal number of components in the baseline acoustic models, the baseline word error rate will be reduced to 14.34%, and the benefit of sharing mixture components is only marginal.

With the adaptation approach, the baseform pdf is shifted towards the corresponding surface forms. If a surface-form pdf is far away from the baseform one, the extent of the modification will be substantial and, consequently, the modified pdf may fail to model the original baseform. On the other hand, the sharing approach has the problem of undesirably including redundant components in the baseform models. Thus we combine these two approaches. The idea is to perform adaptation using the surface-form components that are close to the baseform, and at the same time, to use those relatively distant components for sharing.

The values of the KLD between the baseform pdf and the nearest surface-form pdf have been analyzed. As illustrative examples, the histograms of the KLD at different states between [oi] (baseform) and [o] (surface form), and between [oi] and [ou], are shown as in Figure 2. There are two main types of KLD distributions: 1) concentration around small values (e.g., states 1 and 2 of the pair "[oi]→[o]"), and 2) a wide range of values (e.g., states 3 to 5 of the pair "[oi]→[ou]"). A small KLD means that the mixture components of the baseform and surface forms are similar. In this case, the baseform components adapt to the surface form. In the case of a widely distributed KLD, the surface-form components should not be used to adapt the baseform components, but rather should be kept along with the modified baseform model in order to explicitly characterize irregular pronunciations. In this way, a combined approach to baseform model refinement is formulated.

Despite the good intentions, the combined use of sharing and adaptation does not lead to favorable experimental results. With a total of 2,441 mixture components in the refined acoustic models, the word error rate is 14.57%. The baseline performance is 14.93% with the same model complexity.

5. Conclusion

In this study, we have classified pronunciation variations into complete/phone changes and partial/sound changes. However, these are not well defined classifications, especially for the partial/sound changes. There is not a clear boundary that separates a phoneme substitution (phone change) from a phoneme modification (sound change). This may partially explain why the proposed techniques of handling sound change are not as effective as the methods for handling phone change.

The use of a PVD is intuitive and straightforward in implementation. It can reduce the word error rate noticeably. When constructing a PVD, the value of the VP threshold needs to be carefully determined. While a tight threshold obviously does not show any effect, a lax control of the PVD size leads to not only a long recognition time but also performance degradation. The method of dynamic search space expansion during decoding can bring about the same degree of performance improvement as the PVD. However, the training of context-dependent pronunciation prediction models requires a large amount of data.

The methods of acoustic model refinement do not improve recognition performance as much as we expected. Similar effect can be achieved by using more mixture components. Indeed, more mixture components can describe more complex acoustic variations, which include the variations caused by alternative pronunciations. The sharing of mixture components is equivalent to having more mixture components right at the beginning of acoustic models training. Adaptation of mixture components is not as effective as increasing the number of mixture components.

For any of the above methods to be effective, the accurate and efficient acquisition of pronunciation variation information is most critical. Manual labeling is impractical. Automatic detection of pronunciation variations is still an open problem.

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Table 1. Classification of vowel sounds in Standard Malay language

	Initial	**Middle**	**Final**
High	i		u
Medium	e	at	o
Low		a	

Table 2. Consonants in Standard Malay except those in brackets are loaned consonants

Manner of Articulation	**Place of articulation**					
	Labial	**Alveolar**	**Palate-alveoral**	**Palatal**	**Velar**	**Glottal**
Plosive-Voiceless	p	t			k	
Plosive-Voiced	b	d			g	
Fricative-Voiceless	(f)	s			(x)	h
Fricative-Voiced	(v)	(z)				
Affricate-Voiceless			c			
Affricate-Voiced			j			
Nasal	(m)	n		ny	ng, nx	
Roll		r				
Lateral		l				
Semivowel	w			y		

Table 3. Recognition results of using a PVD with different VP thresholds

		VP threshold				
	Baseline	0.02	0.05	0.10	0.15	0.20
Word error rate (%)	15.34	13.91	13.49	13.70	13.64	13.58
No. of word entries in the PVD	451	840	568	356	210	171

Table 4. Recognition results by dynamic search space expansion

	Baseline	VP 0.05	threshold 0.2
Word error rate (%)	15.34	13.53	13.27

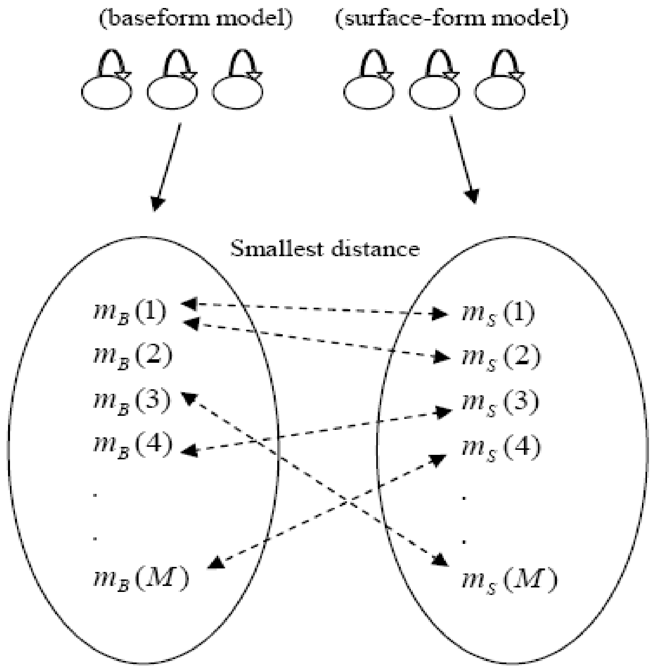

Figure 1. Mapping between baseform and surface-form mixture components

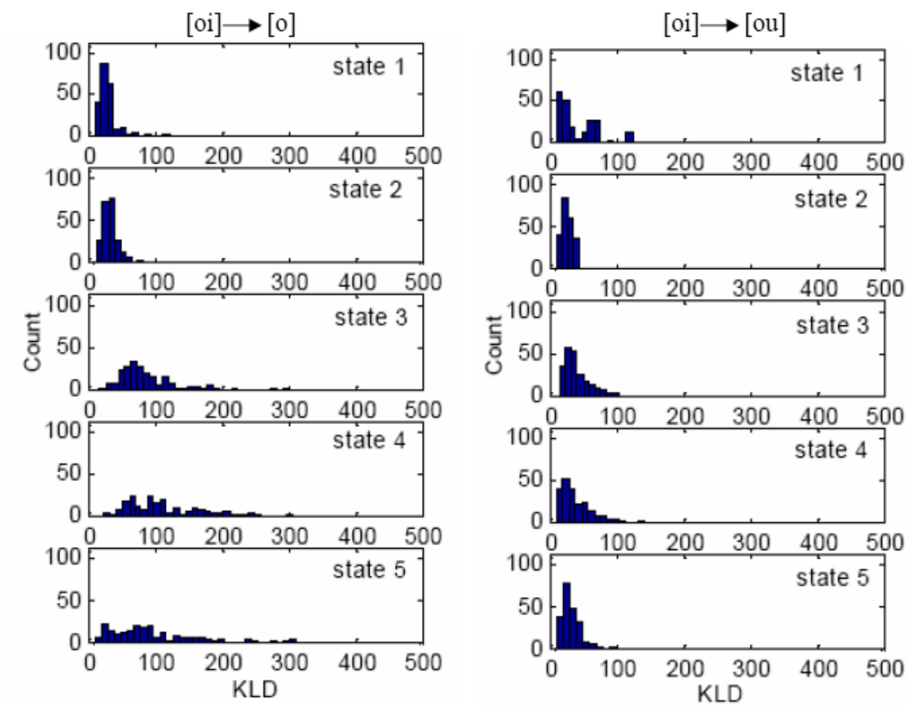

Figure 2. KLD distributions for variation pairs [oi]→[o] and [oi]→[ou]

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A Modified Adaptive Algorithm for Formant Bandwidth in Whisper Conversion

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Abstract

The whisper conversion technology is to transform undistinguished whispers with lower SNR into clear normal speech, and it has important application prospect in mobile communication. Because the whisper speech is stirred by the yawp source, its formant position shifts and its bandwidth increases, which induces the problem of formant combination occurs in the whisper conversion. By analyzing the power spectrum, in this article, we proposed a modified adaptive algorithm for formant bandwidth. Based on the rule that the pole power does not change, the algorithm has resolved the problem of formant combination by modifying the formant bandwidth of whisper before implementing formant conversion. The experimental results with six Chinese mandarin monophthong phoneme conversions proved the validity of the algorithm.

Keywords: Whisper speech, Voice conversion, Formant combination, Bandwidth

1. Introduction

The research to the whisper which is the special speech first rooted in the need for the phonation to the laryngeal excision sufferer (Liang, 1997, P.151-152). With the extensive use of mobile communication tool, whisper has been a sort of effective communication mode which could increase the secret of talk and doesn't influence the hearing environment. How to transform undistinguished whispers with lower SNR into clear normal speech and realize the whisper communication has been more and more noticed by foreign and domestic scholars (Li, Xueli, 2004 & Morris R W, 2002).

Unlike the pronunciation mode of normal speech, the whisper source is yawp and the vocal cords doesn't oscillate, so comparing with normal speech, the whisper has no pitch frequency, the sound level is lower about 20dB (Gao M, 2002). The change of track transfer function makes the formant position of the whisper change, and the corresponding formants that the first and second formants are higher than the normal speech and the phrase spectrums such as the adding bandwidth of formant occur (Jovicic S T, 1998, P.739-743).

According to above acoustics characters, the conversion from whisper to normal speech is mainly realized by adding the pitch frequency and modifying the formant spectrum (Morris R W, 2002, P.515-520). But in the experiment of Chinese mandarin monophthong phoneme conversion, we found that the traditional formant conversion algorithm would meet the problem of formant combination and influence later normal speech combination. In this article, through analyzing the pole power spectrum of formant, we pointed out that the adding whisper bandwidth is the direct reason to produce the problem of formant combination for the whisper conversion, and put forward a sort of modified adaptive formant bandwidth algorithm which could effectively reduce the bandwidth of whisper formant and solve the problem of formant combination of whisper after conversion.

2. Formant combination in whisper conversion

Figure 1 is the former three formants extracting whisper signal [o4] by the linear predictive spectrum algorithm

(McCandless S, 1974, P.135-141), and Figure 2 is the formant of the normal speech [o4] after conversion by the Gauss mixture model (Lv, 2004). From Figure 2, the first formant and the second formant produce the formant combination. And we adopt the method of pole power spectrum to analyze the reasons for above phenomena.

First, we realize the conversion from the frequency domain to the z domain. If the sampling frequency is F_s, the formant F_i and the 3dB bandwidth B_i from the LPC algorithm can be converted to the pole with the angle of ϕ_i and the radius of r_i in the z domain according to the following formulas.

The radiation angle of the pole: $\phi_i = 2\pi \dfrac{F_i}{F_s}$ (1)

The radius of the pole: $r_i = e^{-\frac{B_i * \pi}{F_s}}$ (2)

From the radiation angle and the radium, we can obtain the transfer function

$$H(z_i) = \frac{1}{1 - r_i e^{j\phi_i} z_i^{-1}} \tag{3}$$

The power spectrum of the pole z_i in the z domain is

$$\left|H(e^{j\theta})\right|^2 = \prod_{i=1}^{n} \frac{1}{1 - 2r_i \cos(\theta - \phi_i) + r_i^2} \tag{4}$$

Convenient for discussion, we first suppose two poles z_1 and z_2, so the power $\left|H(e^{j\phi_1})\right|^2$ at the radiation angle ϕ_1 is

$$\frac{1}{(1 - r_1)^2} \cdot \frac{1}{1 - 2r_2 \cos(\phi_1 - \phi_2) + r_2^2} \tag{5}$$

In the z domain, when poles z_1 and z_2 gradually close up, the difference of their radiation angle would reduce, and from formula (5), we can see that the power peak value at the radiation angles ϕ_1 and ϕ_2 will also reduce until two power peaks combines.

The traditional formant conversion algorithm only directly shifts the formant of whisper signal [o4] from *F1*=1078*Hz*, *F2*=1721*Hz* and *F3*=2718*Hz* to *F1*=670*Hz*, *F2*=1173*Hz* and *F3*=3630*Hz* and doesn't modify the bandwidth, and because of the poles close up, so the problem of formant combination occurs.

3. Modified bandwidth adaptive algorithm

According to above analysis, if we can reduce corresponding formant bandwidth when the formant is converted, so the combination of formant could be eliminated in theory. In the traditional LPC algorithm, when we abstract the formant, we directly delete the formant poles which don't accord with the requirements. Based on the rule that the pole power does not change, in this article, we propose the algorithm which automatically add the energy of the deleted formant to the reserved formant and realize the adaptive formant bandwidth change. The implementing principle of the algorithm includes the reserved formant pole with the angle of ϕ_i and the radius of r_i is z_i, and the deleted formant pole with the angle of ϕ_j and the radius of r_j is z_j. According to formula (5), the power at the angle of ϕ_i is

$$\frac{1}{(1 - r_i)^2} \prod_{j=1}^{M} \frac{1}{1 - 2r_j \cos(\phi_i - \phi_j) + r_j^2} = \frac{1}{(1 - r_i')^2} \tag{6}$$

Here, r_i' denotes the corresponding new pole radius when deleting pole z_j and reserving the unchanged pole energy, and M denotes the deleted pole amount.

In addition, we must consider the influence to other reserved poles when changing the radius of a pole. So the formula (6) could be extended as

$$\frac{1}{(1-r_i)^2} \prod_{k=1,k\neq i}^{N} \frac{1}{1-2r_k \cos(\phi_i - \phi_k) + r_k^2} \times \prod_{j=1}^{M} \frac{1}{1-2r_j \cos(\phi_i - \phi_j) + r_j^2} = \frac{1}{(1-r_i')^2} \prod_{k=1,k\neq i}^{N} \frac{1}{1-2r_k' \cos(\phi_i - \phi_k) + r_k'^2} \tag{7}$$

Here, r_k is radius of other reserved poles, $r_k^{'}$ is the corresponding pole radius after modification, and N is the amount of reserved linear predictive multinomial pole.

4. Experimental results

In the experiment, we select six monophthong whisper speeches including /a/, /o/, /e/, /i/, /u/ and /ü/, and the normal speech as the samples, and every speech possesses four pronunciations including level tone, rising tone, falling-rising tone and falling tone, and the sample amount is 24.

We take the stable speech area sample to implement pre-aggravating and window-adding processing, and the pre-aggravating coefficient μ is 0.975, and we adopt the window of Hamming. The experimental sampling rate is 8kHz, and every frame has 256 sampling points. The experimental results showed that the problem of formant combination also occurs in the whispers [o2], [i2] and [u4] except for whisper [o4].

The experiment result is seen in Figure 3. The point lineation is three formants obtained by traditional LPC algorithm, and the thin real line is the normal speech frequency spectrum curve after conversion obtained by Gauss mixture model, and from the figure, we can intuitively see the combination of formants F1 and F2. The broken line is the whisper frequency spectrum curve obtained by the modified adaptive bandwidth algorithm, and through the comparison, we can see that the new algorithm the 3dB bandwidth of three formants is smaller the bandwidth obtained by the traditional LPC algorithm. The wide real line is the frequency spectrum curve of normal speech through the whisper formant conversion obtained by the new algorithm, and comparing with the conversion result through the traditional method, it eliminates the combination of formants F1 and F2.

5. Conclusions

In the conversion of Chinese whisper, because the formant bandwidth of whisper is wider than the normal speech, it will induce the problem of formant combination when we adopt the conversion method directly shifting the formant. In this article, we put forward the method which first add the spectrum power of the deleted formant pole to the formant of the reserved pole, realize the adaptive adjustment of the formant bandwidth, and utilize the Gauss mixture model to implement the formant conversion. The experimental conversion of Chinese mandarin monophthong phoneme proved the method could better solve the problem of formant combination occurring in the speech conversion.

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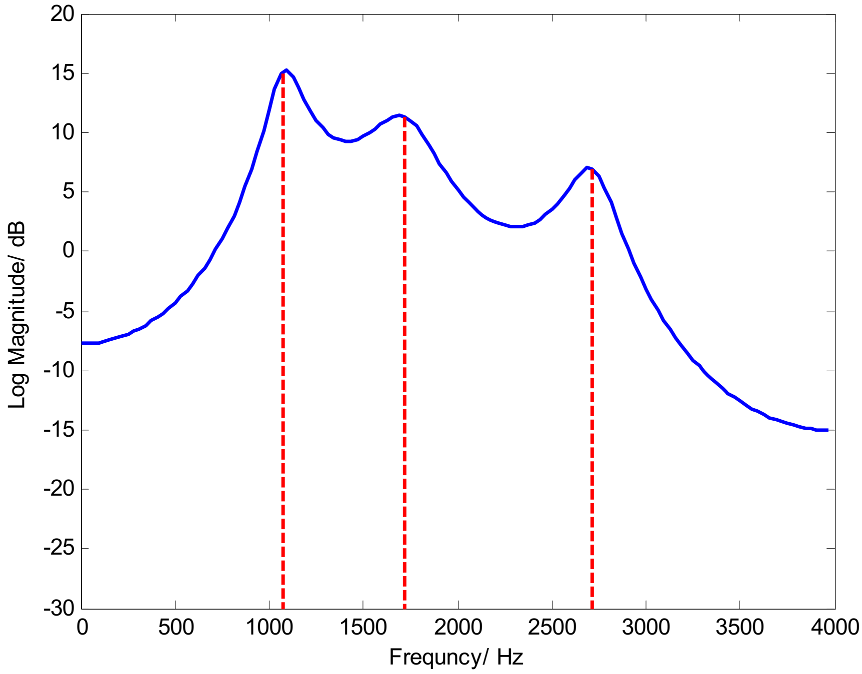


Figure 1. LPC Frequency Spectrum Envelope of Whisper Speech [o4]

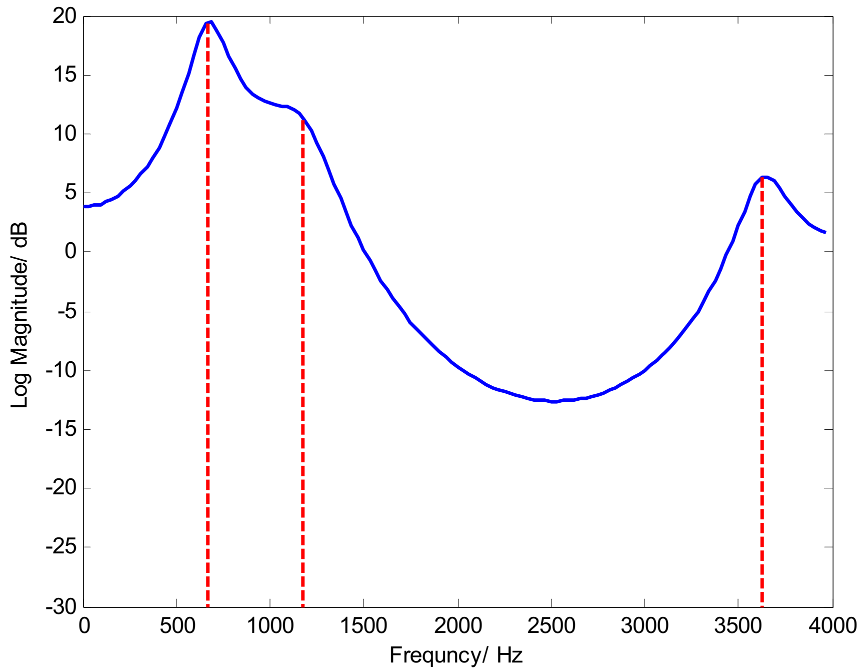
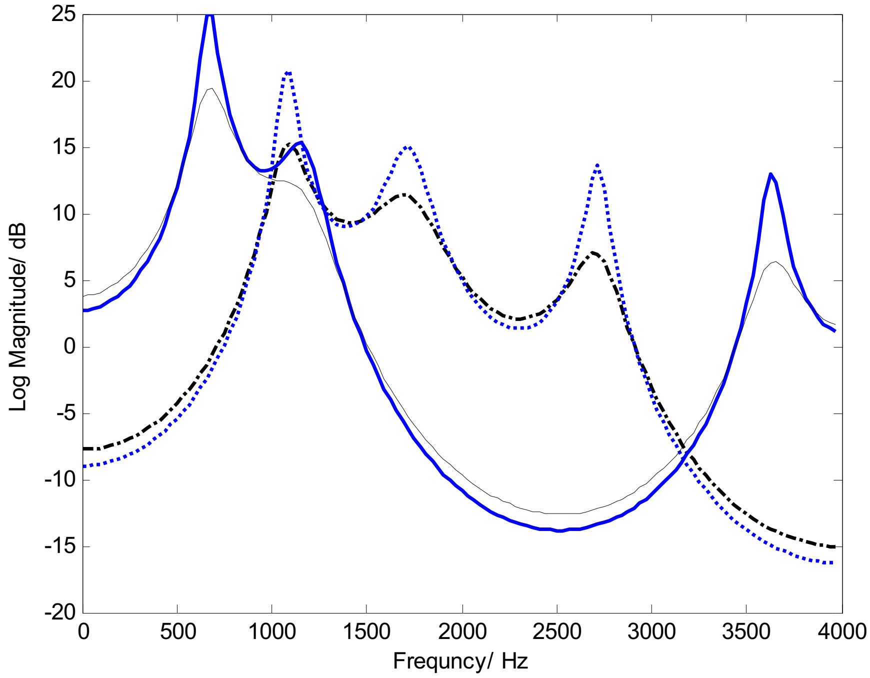
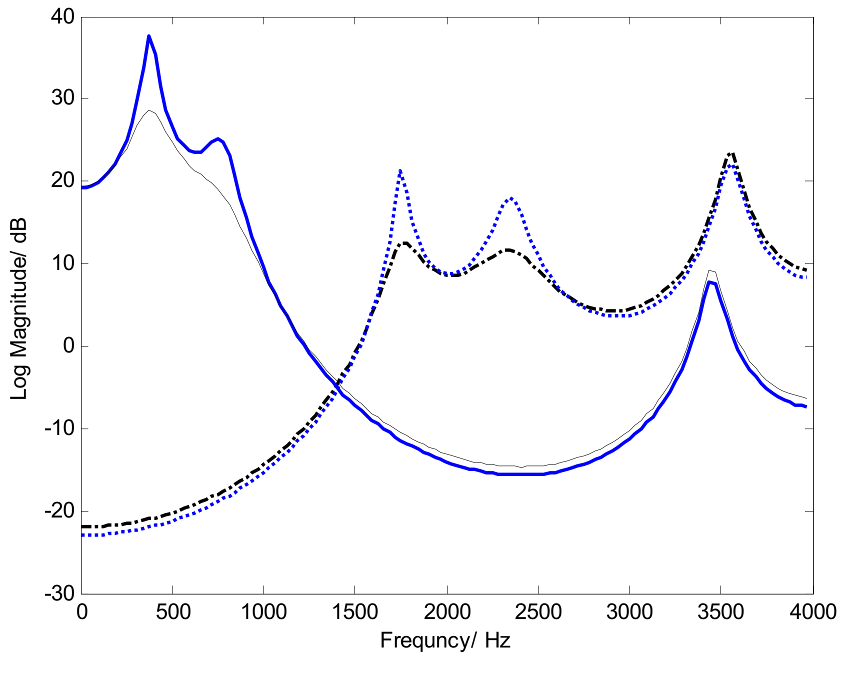


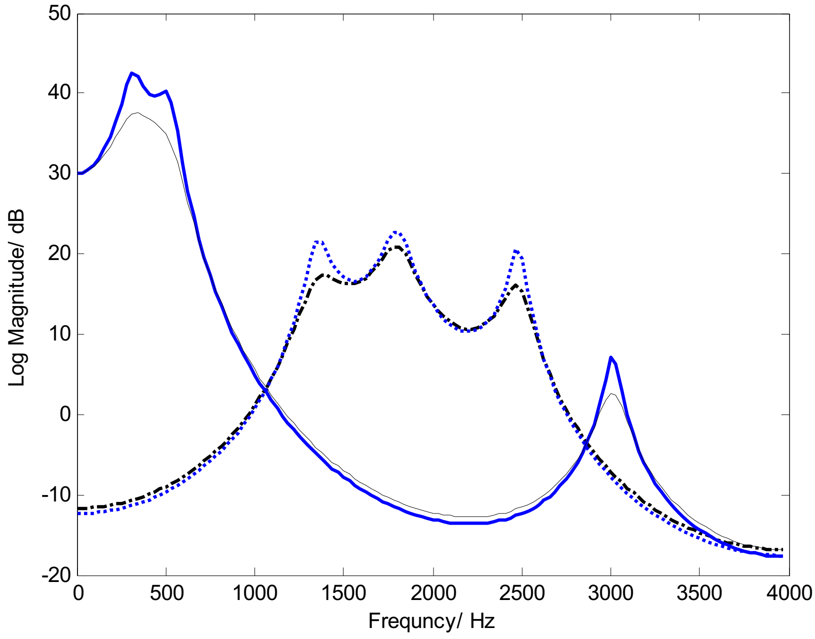
Figure 2. Frequency Spectrum Envelope of Normal Whisper Speech [o4] after Conversion


25
20
15
10
5
0
-5
-10
-15
-20
Log Magnitude/ dB
0
500
1000
1500
2000
2500
3000
3500
4000
Frequncy/ Hz

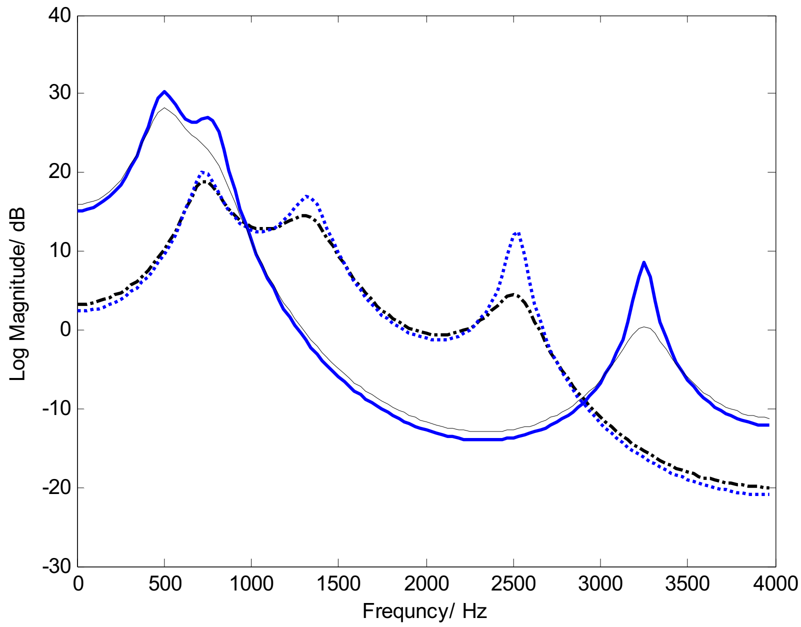
[o4]


40
30
20
10
0
-10
-20
-30
Log Magnitude/ dB
0
500
1000
1500
2000
2500
3000
3500
4000
Frequncy/ Hz

[u4]



[i2]



[o2]

Figure 3. Comparison between Proposed Algorithm and LPC Algorithm on the Effect of Frequency Spectrum Envelopes of Normal Whisper Speeches after Conversions

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Performance Evaluation of QoS by Combining Medium Access Control and Slow Start in MANETS

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Abstract

This paper evaluates the performance evaluation of interaction between Transport and the MAC layer protocols operating in a mobile adhoc network. In Adhoc networks, certain QoS parameters like error rate, delay and packet loss are increased and certain parameters like throughput and delivery ratio are decreased in Transport layer is due to MAC problems and disconnection is also possible due to mobility or power failure. So, combine the mechanisms of these two layers improve the QoS drastically. We examine the effects of two different MAC protocols— IEEE 802.11and IEEE802.11e with Slow start mechanism of TCP. Specifically, we access the impact of multiple wireless hops and node mobility on the throughput performance of TCP on each MAC protocol. Additionally the other Qos parameters of throughput, delay, Bandwidth delay product, delivery ratio and packet loss using slow start of TCP mechanism with two different MAC protocols is also investigated. Results show that in all instances, the QoS parameters 15-20% improvement in throughput, 40-45% improvement in bandwidth-delay product, 10-15% improvement in delivery ratio in IEEE 802.11e than IEEE802.11 and packet loss is reduced drastically to 40-50% in IEEE802.11e where only 3-5%delay is higher in IEEE802.11e than IEEE802.11.

Keywords: Mobile adhoc networks, Medium access control(MAC), Transport layer Protocol(TCP), Slow start and Quality of Service(QoS)

1. INTRODUCTION

In the near future, researchers envision a truly ubiquitous computing environment that will allow users to communicate from anywhere and at anytime. Mobile adhoc networks (MANETs) are part of this vision and aim to provide communication capabilities to areas where limited or no communication infrastructure exist; or, where it is simply more convenient to allow the communication devices to form a dynamic and temporary network among themselves. A "mobile adhoc network" (MANET) is an autonomous system of mobile routers (and associated hosts) connected by wireless links. In current wireless networks, WIMAX or WIFI the wireless mobile node is never more than one hop from a base station that can route data across the communication infrastructure. However, in mobile adhoc networks, there are no base stations. Instead, routing functionality is incorporated into each mobile host and, because of a limited transmission range, multiple hops may be required to allow one node to communicate with another across the adhoc network. The routers are free to move randomly and organize themselves arbitrarily; thus, the network's wireless topology may change rapidly and unpredictably. Thus, MANETs can be characterized as having a dynamic, multi hop, and constantly changing topology. While mobile adhoc networks can be used in a standalone mode—where there is no fixed infrastructure, their use is also being considered as an extension to the Internet. Much of the current research of mobile ad hoc networks has focused on the design and development of routing protocols, efficient power consumption, Energy saving techniques, Security in various layers, Enhancement in Qos and cross layer design. However, the success of wireless mobile adhoc networks will also depend significantly on controlling access to a wireless physical layer having relatively low bandwidth links {Paolo, 2006]. Thus, the effectiveness of the wireless medium access control (MAC) protocol and mechanisms will play a central role in the success of MANETS. Several MAC protocols have been developed for wireless environments (i.e. wireless LANS) such as Carrier Sense Multiple Access (CSMA), Multiple Access with Collision Avoidance (MACA), Floor Acquisition Multiple Access (FAMA), IEEE802.11 and IEEE 802.11e. Each MAC protocol is based on multiple design choices and utilizes distinct medium access mechanisms.

This research centers on investigating the performance of and interaction between TCP and two different MAC protocols— IEEE 802.11and IEEE 802.11e, operating in mobile adhoc networks. Reliable data transfer and congestion control are key requirements for any computer network. TCP, which fulfills both of these requirements, is the most widely used reliable transport protocol in today's Internet and has demonstrated its viability with respect to Internet connectivity. TCP is used to transport a significant portion of Internet traffic such as e-mail (SMTP), file transfers (FTP), and WWW (HTTP). Thus, the use of TCP in mobile adhoc networks is clearly advantageous [Perkins et al, 2005]. However, the defining characteristics (e.g., time-varying, dynamic, multihop, and constantly changing topology) of mobile adhoc networks may result in unpredictable link failures resulting in the poor performance of TCP.

The goal of this paper is, therefore, to study the effects of these characteristics on the performance of and interaction between TCP and the MAC layer protocol operating in a mobile ad hoc network. This includes examining the effects of IEEE 802.11 and IEEE 802.11e MAC protocols on the performance of TCP. Specifically, we access the Qos parameters throughput, delay, Bandwidth delay product, and delivery ratio and packet loss performance of TCP as function of node mobility.

2. RELATED WORK

TCP has been shown to have poor performance over wireless links. Thus, several studies have focused on improving TCP performance in the wireless mobile Environment. These include end-to-end mechanisms such as TCP-SACK and ELN and link-layer protocols such as AIRMAIL, Indirect-TCP, and TCP-Snoop. Such mechanisms and protocols were designed to work in the context of cellular-based networks fixed infrastructure networks. However, the aforementioned schemes have not considered the unique characteristics of adhoc networks, namely multi-hop routing and the lack of a centralized controller and manager (e.g., base stations). Recent work has begun to evaluate the performance of TCP in context of adhoc networks. This work demonstrated how the use of combining the mechanisms of both TCP and MAC protocols improve the QoS parameters. Previous work investigates that IEEE802.11e better than IEEE802.11 [Choi.S, 2003] but not combined with TCP mechanisms. Hence, evaluating the performance of TCP in a mobile adhoc environment and quantifying the effects of the unique characteristics is an open and interesting problem. These results should facilitate the development of mechanisms for improving TCP performance in adhoc networks as well as the design of efficient and scalable quality-of-service (QoS) schemes.

3. SIMULATION AND METHODOLOGY

This simulation study was conducted using NS2 to simulate adhoc network, which consist of 60 wireless/mobile nodes roaming in a 2600 x 400m area. In this dynamic topology, the radio transmission range of each node is approximately 250 meters and the channel capacity is 2Mbits/sec. The free space propagation model is used to determine if a node is reachable. This model predicts received signal strength when the transmitter and receiver have a clear, unobstructed line-of-sight path between them. Received power decays as a function of the T-R separation distance. This study investigates the performance slow start mechanism of TCP over two different MAC protocols:IEEE802.11 and IEEE 802.11e. Both protocols requires carrier sensing before transmission and operates as follows

3.1 IEEE802.11 MAC Protocol

The basic access mechanism for both MAC protocols is the Distributed Coordination Function (DCF). DCF is essentially a Carrier Sense Multiple Access (CSMA) that incorporates Collision Avoidance (CSMA/CA) and a positive acknowledge (ACK) scheme. Receipt of an ACK (from the receiving node) indicates that no collision occurred [Qixiang.P, 2005]. If the sending node does not receive an ACK, then it will retransmit the fragment until it gets acknowledged or discarded after a specified number of retransmissions. Optionally, a mobile node can utilize the virtual carrier sense mechanism, which utilizes request-to-send (RTS) and clear-to-send (CTS) exchanges for channel reservation. Using virtual carrier sensing reduces the probability of two nodes transmitting simultaneously (collisions) because they cannot hear each other (i.e. hidden terminal problem). The difference between IEEE802.11 and IEEE 802.11e is, to assign priority for user packets in IEEE 802.11e and there is no priority assignment for user packets in IEEE 802.11.

3.2 IEEE802.11e MAC Enhanced DCF (EDCF)

The DCF is supposed to provide a channel access with equal probabilities to all stations contending for the channel access in a distributed manner. However, equal access probabilities are not desirable among stations with different priority frames [Maarten et al, 2000,2002]. The emerging EDCF is designed to provide differentiated, distributed channel accesses for frames with 8 different priorities (from 0 to 7) by enhancing the DCF as shown in Table1. As distinct from the legacy DCF, the EDCF is not a separate coordination function. Rather, it is a part of a single coordination function, called the Hybrid Coordination Function (HCF), of the 802.l1 MAC. The HCF combines the aspects of both DCF and PCF.The EDCF adopts eight different priorities that are further mapped into four access categories (ACs) as shown in figure1. ACs are achieved by differentiating the arbitration inter frame space (AIFS), the initial window size and the maximum window size.

For the AC *i* (*i* = 0, 1, 2,3), the initial backoff window size is CWmin[*i*](= *Wi,*0), the maximumbackoff window size is CWmax[*i*] and the AIFS is AIFS[*i*].

For 0 = *i* <*j* = 3, CWmin[*i*] = CWmin[*j*],

CWmax[*i*] = CWmax[*j*], and

AIFS[*i*] = AIFS[*j*],

and at least one of the above inequalities must be strict. If one class has a smaller AIFS or CWmin or CWmax, the class's traffic has a better chance to access the wireless medium earlier. Four transmission queues are implemented in a station and each queue supports one AC class, behaving roughly as a single DCF entity in the original IEEE 802.11 MAC.

It is assumed that a payload from a higher layer is labeled with a priority value, and it is pushed into the corresponding queue with the same priority value [Xiao.Y 2004,2005,2006]. Each queue acts as an independent MAC entity and performs the same DCF function with a different inter frame space (AIFS[*i*]), a different initial window size (CWmin[*i*]), and a different maximum window size (CWmax[*i*]). Each queue has its own backoff counter (BO[*i*]) that acts independently the same way as the original DCF backoff counter. If there is more than one queue finishing the backoff at the same time, the highest priority frame is chosen to transmit by the virtual collision handler. Other lower priority frames whose backoff counters also reach zeros will increase their backoff counters with CWmin[*i*] (*i*= 0, 1.. . 3), accordingly. Use EDCF (enhanced distributed co ordination function) and Slow start mechanism of Transport layer enhance the MAC performance and also transport layer performance.

4. SIMULATION

The AODV (Adhoc On Demand Vector) protocol, available in NS2 uses dynamic routing in order to deliver packets to any destination in a mobile adhoc network. The random waypoint mobility model, each node is placed randomly in the simulated area and remains stationary for a specified time and then randomly selects a destination from the physical terrain. The node then moves in the direction of the destination point at a speed uniformly chosen between a minimum and maximum speed (meters/sec). To increase the performance there should be different types of priority level or traffic categories (TC) for data transmission in MAC layer and use user priority level of 0,1 and 2.For simulation produce 3 different packets of data and set priority 0(high priority) for large size packet, priority 1 (medium priority) for medium size packet and priority 2(low priority) for small size of packet. To send acknowledgements from transport layer in SIFS interval, a acknowledgement packet which contain less bytes of data is transmitted for all different types of traffic categories. The time slots for various inter frame spacing is set as SIFS=16µs, PIFS=25µs, DIFS=34µs, $AIFS_1$ (priority level=0 or TC1) >=34µs and every contention slot is equal to 9µs interval. If there is no high priority packet for the specified time interval immediately medium level packet are transmitted.

4.1 RESULTS and PERFORMANCE METRICS

To analyze the performance and interaction of TCP and MAC layer protocols, we evaluate them using the following metrics:

4.1.1 Throughput: It is the rate of successful message delivery over a communication channel. This data may be delivered over a physical or a wireless channel and it is usually measured in bits per second (bit/s or bps), and sometimes in data packets.

With 20nodes 802.11 transmitted 8406 bits, 802.11e transmitted 9234 bits successfully. With 60nodes 802.11 successfully transmitted 8286 bits, 802.11e transmitted 9412 bits. The Slow start mechanism of TCP with IEEE802.11e improves throughput 10-15% than IEEE802.11 with Slow start. Figure2 shows comparison of throughput performance of IEEE802.11 with Slow start and IEEE802.11e with Slow start.

4.1.2 Bandwidth-Delay Product: It refers to the product of a data link's capacity (in bits per second) and its end-to-end delay (in seconds). The result, an amount of data measured in bits (or bytes), is equivalent to the amount of data "on the air" at any given time, i.e. data that have been transmitted but not yet received. This product is particularly important for protocols such as TCP that guarantee reliable delivery, as it describes the amount of yet-unacknowledged data that the sender has to duplicate in a buffer memory in case the receiver requires it to re-transmit a garbled or lost packet.

With 20nodes 802.11 transmitted 214187.73 bits where 802.11e transmitted 240017.87 bits successfully. With 60nodes 802.11 successfully transmitted 172308.41 bits where 802.11e transmitted 244467.13 bits. The Slow start mechanism of TCP with IEEE802.11e drastically improves Bandwidth Delay Product 40-45% than IEEE802.11 with Slow start. Figure3 shows comparison of Bandwidth-Delay Product performance of IEEE802.11 with Slow start and IEEE802.11e with Slow start.

4.1.3 Packet Delivery Ratio: It is the ratio between total number of packets received to the total number of packets transmitted.With 20nodes 802.11 transmitted 290 packets and 802.11e transmitted 317 packets successfully. With 50nodes 802.11 transmitted 287packets where 802.11e transmitted 312 packets. The Slow start mechanism of TCP with IEEE802.11e improves delivery ratio 10-15% than IEEE802.11 with Slow start. Figure4 shows comparison of Packet Delivery Ratio performance of IEEE802.11 with Slow start and IEEE802.11e with Slow start.

4.1.4 Delay: The time taken by the packets to reach the destination successfully. With 20nodes 802.11 transmitted with a delay of 13msec, where 802.11e transmitted with a delay of 14msec. The Slow start mechanism of TCP with IEEE802.11e is only 0-5% higher than IEEE802.11 with Slow start and this is acceptable. Figure5 shows comparison of Delay performance of IEEE802.11 with Slow start and IEEE802.11e with Slow start.

4.1.5 Packet loss: The number of packets missed to reach the destination. With 20nodes 802.11 missed 60 packets and 802.11e missed 33 packets. With 50nodes 802.11 missed 63 packets where 802.11e missed 38 packets. The Slow start mechanism of TCP with IEEE802.11e reduces drastically the packet loss from 40-45% than IEEE802.11 with Slow start. Figure6 shows comparison of Packet loss performance of IEEE802.11 with Slow start and IEEE802.11e with Slow start.

5. CONCLUSION

In this paper, evaluate the performance of QoS parameters in MAC layer and its interaction with the transportation layer protocol in a mobile ad hoc network is tabulated in Table.2. This system using IEEE 802.11e and IEEE802.11 MAC mechanisms are contention based channel access function or distributed coordination function that improves quality of service in MAC layer. To improve the performance of at the transport layer will require the design of distributed medium access control scheme and proper packet transmission mechanism like slow start. A suitable MAC layer protocol and slow start algorithm improves quality of service in transport layer.

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Table 1. EDCF user priority table

User priority	Access category	Designation
0	0	Best effort
1	0	Best effort
2	0	Best effort
3	1	Video probe
4	2	Voice
5	2	Voice
6	3	Video
7	3	Video

Table 2. Comparison of QoS Parameters

S.No	Parameters	No. of Nodes	802.11 With Slow Start	802.11e With Slow Start
1	Throughput (bps)	20	8406	9234
		30	8266	9337
		40	8366	9252
		50	8196	9311
		60	8286	9412
2	Delay (sec)	20	0.1340	0.1477
		30	0.1326	0.1486
		40	0.1370	0.1480
		50	0.1310	0.1489
		60	0.1391	0.1505
3	Packet Delivery Ratio (Packets)	20	290	317
		30	290	314
		40	285	316
		50	287	312
		60	290	317
4	Bandwidth Delay Product (bits)	20	214187.73	240017.87
		30	195114.45	248245.30
		40	218268.71	240956.52
		50	208455.60	242702.57
		60	172308.41	244467.13
5	Packet loss	20	60	33
		30	60	36
		40	65	34
		50	63	38
		60	60	33

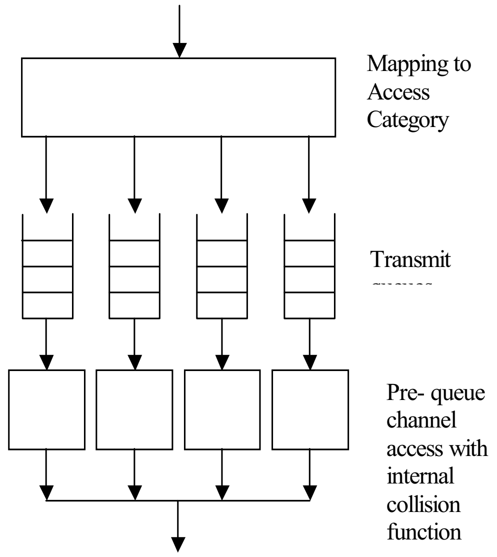

Figure 1. Reference Implementation model of IEEE 802.11e

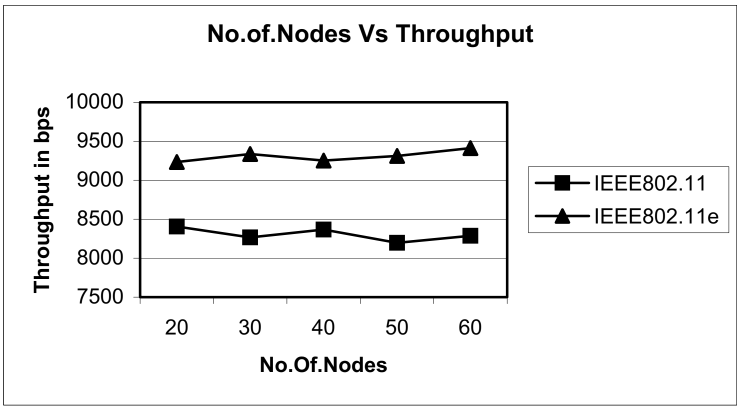

Figure 2. No. of Nodes Vs Throughput

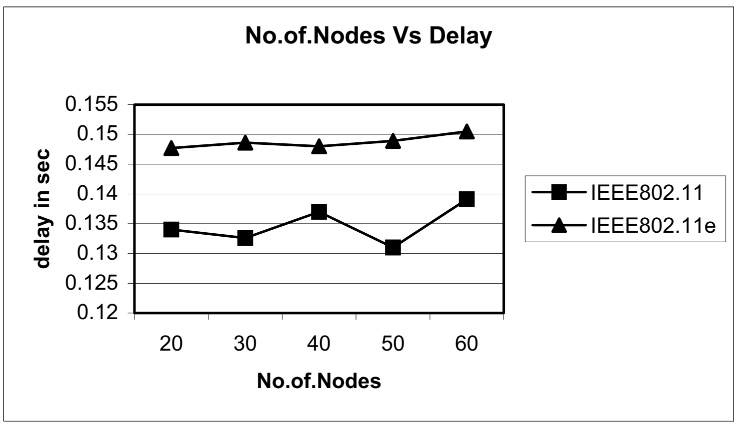

Figure 3. No. of Nodes Vs Delay

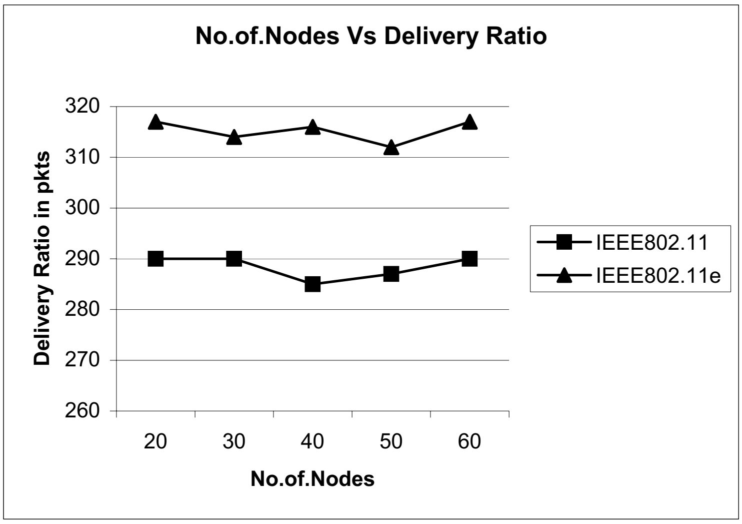

Figure 4. No. of Nodes Vs Delivery Ratio

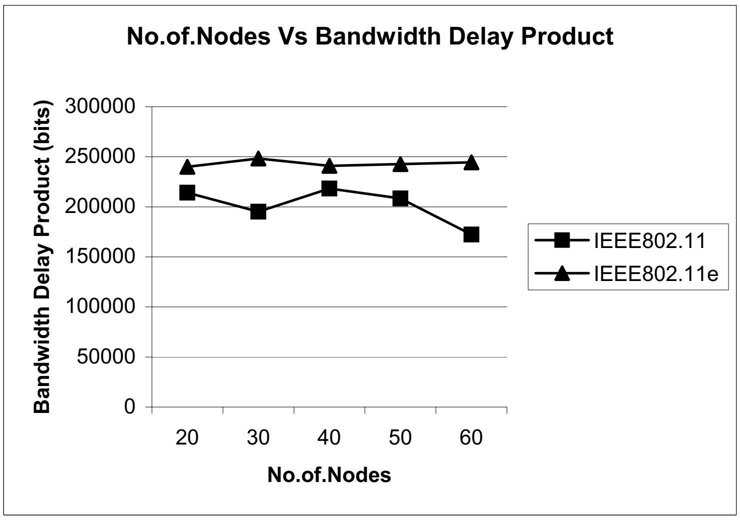

Figure 5. No. of Nodes Vs Bandwidth Delay Product

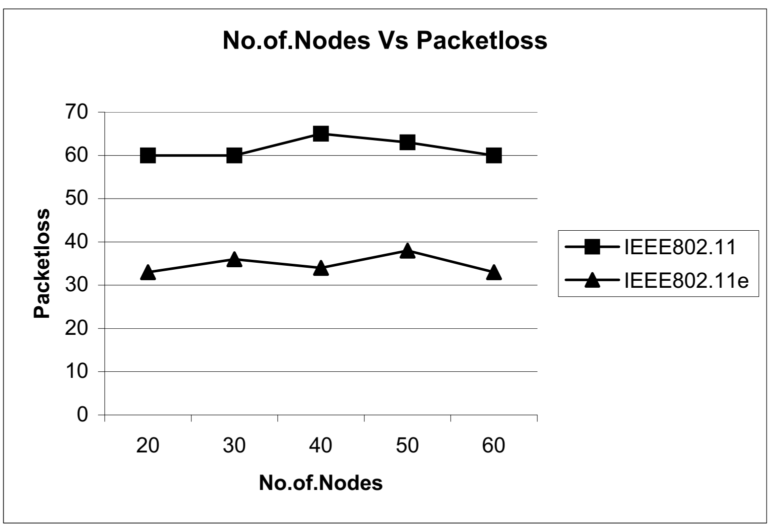

Figure 6. No. of Nodes Vs Packet loss

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Adjustment of Performance for the Application of PL/SQL

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Abstract

The oracle database system is the large relation database management system which is applied most broadly at present. The PL/SQL language is a sort of block structured language, and it is the procedural language that the Oracle extends SQL, and it allows encapsulating operation logics, and its advantage is to use inscribed SQL sentence to define and realize 3GL language structure and describe more complex query, trigger, memory process and other objectives. Therefore, to more reasonably use PL/SQL program can exert important function to implement the Oracle database performances. Through the adjustment of database shared pool and the introduction of different technologies of data access, we discussed some technologies to enhance the implementation performance of PL/SQL and optimize the performance of the database system by means of PL/SQL language.

Keywords: PL/SQL language, Performance optimization, Shared pool, Data access

PL/SQL (Procedural Language/SQL) is the extension of SQL, and it absorbs many excellent characters of programming language in recent years, such as data encapsulation, information concealment, reloading, and exception treatment. It allows the data control language and query sentence of SQL contained in the block structure and the code process language, and makes PL/SQL become into a sort of transaction processing language with powerful function. Figure 1 shows the client/server system structure of PL/SQL based on the application of Oracle.

In the process of software design and development, the implementation performance of software is an important problem which we should consider. The quick and highly effective implementation performance of software is the important limited factor to decide whether its output, input, throughput and customer satisfaction are eligible. In the application of Oracle database, this sort of limitation is more important. The size of Oracle database and the user amount using this application are important factors to decide whether the application achieve maximum implementation performance through adjustment. Therefore, we need to adjust the application of PL/SQL and make its performance implement by the optimal mode according to a great lot of user's throughput and retractility.

Before the application is adjusted, we need to know what reasons induce it operate so slowly, so Oracle offers many database supervision and diagnoses tools to analyze the performance of the application, or we can use the mechanism offered by PL/SQL to get the implementation time of centisecond, and this mechanism can be described as that it uses the function DBMS_UTILITY.GER_TIME, and usually transfer this function twice, and get the difference between two return values, and the difference is the time consumption of program operation, so we can confirm where is the bottleneck of the application.

1. The adjustment to the SGA shared pool

Both the shared memory buffer and the background course are called as an Oracle example. When one Oracle example is started, many Oracle background courses will be started, and every course takes charge the treatment to database for different aspects, and various courses communicate each other through shared memory, and this memory is the whole system SGA. SGA is divided into different areas and one of them is called as the shared pool which contains the text and PL/SQL block of SQL sentence transmitting to the database, and the presentation forms and implementation project after analysis.

Before one memory module is implemented or one memory package is cited, the codes in the coded package must be loaded into SGA. The default method to maintain the package in SGA is to let RDBMS use its arithmetic management codes which are used least recently. When one package is first cited, the coded codes will be loaded into the shared pool, so any user with right of EXECUTE to the package can use it. It will be in the shared pool until other resources based on memory need memory and that package is not used recently, and so the package is washed out from the shared pool. When this package objective is needed next time, the whole package must be loaded into the memory again. Therefore, the shared pool is larger, the probability that the program stays in the memory next time will be higher. However, too large shared pool will waste memory. So we should supervise the shared buffer to ensure it saves all analyzed SQL cursors and PL/SQL code snippets which usually are cited in the application. If too many exchanges occur, we can adjust the parameter of SHARED_POOL in the document of INIT.ORA to add the size of the shared pool (when the physical memory is allowed).

2. The adjustment to data access

2.1 Utilizing package data to implement minimized SQL access

When announcing one variable in a package or package explanation, its function area is not limited in any special process or function. So the function area of package data is the whole Oracle dialog, and the value of that data will be continually effectively in the whole term of dialog, and we can utilize this fact to minimize the times of access from the SQL layer. The structured implementation query aiming at PGA is much quickly than the query to SGA from A to Izzard, so when the application needs to implement the query that the term of multiple re-dialogs is not changed, the above method is the most convenient one. For example, suppose that one program needs to get one exclusive dialog id to avoid folding with other dialogs, one method to realize this aim is to transfer DBMS_LOCK.ALLOCATE_UNIQUE and pick up the exclusive latch appellation.

2.2 Citing the transfer parameter

The citation of transfer parameter is to use the transfer of a finger pointing to the argument to replace the parameter transfer mechanism which transfers the copy of the argument.

The call-back of subprogram parameter is decided by the parameter type and parameter mode. The argument and the parameter in the parameter type are corresponding. For parameter, the parameter modes are respectively IN, OUT and INOUT. Two transfer modes include value transfer and address transfer when the argument is transferred. PL/SQL adopts the mode of address transfer for the parameter of IN mode, and adopts the mode of value transfer for the parameter of OUT and INOUT mode to transfer them. Because the copy of value must be made, so when large-sized data result uses OUT or INOUT parameters, the transfer of value will occupy more implementation time and consume more memory space, especially transferring the result set, the recorded index-by tables or objectives. In order to reduce this sort of consumption, we should adopt the coder clew of NOCOPY. Table 1 describes the behaviors of appointed IN, OUT, INOUT and NOCOPY.

Because of avoiding copying the argument to the parameter, the mode of cited transfer makes the program implement quicker, and when large-sized data is transferred, its meaning is more obvious.

2.3 Using local dynamical SQL and batch binding

The local dynamical SQL means the announced sentence when operating and the SQL sentence implementing dynamic confirmation, which is the interior support of dynamic SQL in PL/SQL explainer, and is implemented through the mode of IMMEDIATE and OPEN FOR without relative API. The implementation performance of local dynamic SQL is relative to the using times of dialog of local dynamic SQL, and if the dynamic SQL sentence is implemented many times in one dialog, so the local dynamic SQL is implemented quickly, because the local dynamic SQL sentence will analyze the SQL sentence when it is implemented every time. Otherwise, because the local dynamic SQL announced in the expression doesn't contain API, so it can minimize the system spending of process transfer, on the other hand, because same SQL sentences are implemented by the binding variables with different values, so one cursor can be shared by many SQL sentences, thus the binding variables in the local dynamical SQL will be implemented quicker.

The batch binding means to use set input or take many rows of records from the table to one set, then only use one order to insert, renovate and delete many rows of records in the table. This process needs few switches of context between PL/SQL and SQL, and this advantage is combined with the quickness of implementation of local dynamical SQL, which can fully enhance the implementation performance of program.

2.4 Using objective type and objective set

The objective type can simulate the entity in the real world, and it and its methods operating on these types integrate data and methods into one structure, which reduces the demands to multiple tables. The introductions of objective REFS and REF line in the database are more convenient than the use of exterior keys, because it avoids complex links. Through reducing the system spending of link treatment, we can save and operate the objectives as one whole to enhance the implementation performance of program.

2.5 Using pipelined table functions (from Oracle9i)

Pipelined table function means the function that uses PIPELINED sentence in the definition of function uses PIPEROW sentence in the interior function to return the set by the pipelined way.

The steps to establish the pipelined table functions include following aspects.

(1) Establishing one objective type or PL/SQL record type.

(2) Establishing the set of this objective type or PL/SQL record type.

(3) Establishing one function to return this set.

(4) Using PIPELINED and defining this function as the pipelined function.

(5) Using PIPE sentence to return one factor of the set.

(6) Compiling one null RETURN sentence which is used to appoint the complete form of the pipelined table function.

When returning large-sized record set, the pipelined table functions can obviously enhance the implementation performance of program, and this function needs not be completely implemented, and relative sets need not be completely initialized in the memory, which can enhance the response time and reduce the memory consumption, and the function can be transferred from one table function to another common table function, which can reduce the demand of memory data in the middle table and enhance the implementation performance.

The performance optimization in the applied system has been the standard to measure the applied system. To effectively apply various optimized measures based on good database design are the base to get higher performance. In this article, we mainly discuss some technologies to enhance the implementation performance of PL/SQL and optimize the performance of the database system. In fact, the optimization of applied system in Oracle should enter into the whole applied development process including system design, database design, application design and database server deployment, and we need studying the system characters and concrete applied environment such as the resource deployment of database layer, the flux control of network layer and the whole design of operation system layer, and based on that, we can utilize the characters of PL/SQL language to implement selections from multiple selectable projects and reasonably optimize the system performance of Oracle database.

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Table 1. Behaviors of various parameter modes

IN	OUT	INOUT	INONT NOCOPY, OUT NOCOPY
Arguments are transferred by the mode of citation	Arguments are transferred by the mode of value transfer	Arguments are transferred by the mode of value transfer	Parameters are transferred by the mode of citation
The transfer points to the finger of value	Transferring the copy of value	Transferring in or out the copy of value	Transferring the address of value
N/A	When the un-disposal abnormity occurs, the value transferred is returned	When the un-disposal abnormity occurs, the value transferred is returned	When the un-disposal abnormity occurs, the value transferred is not returned, so it can not correctly predict the value of out

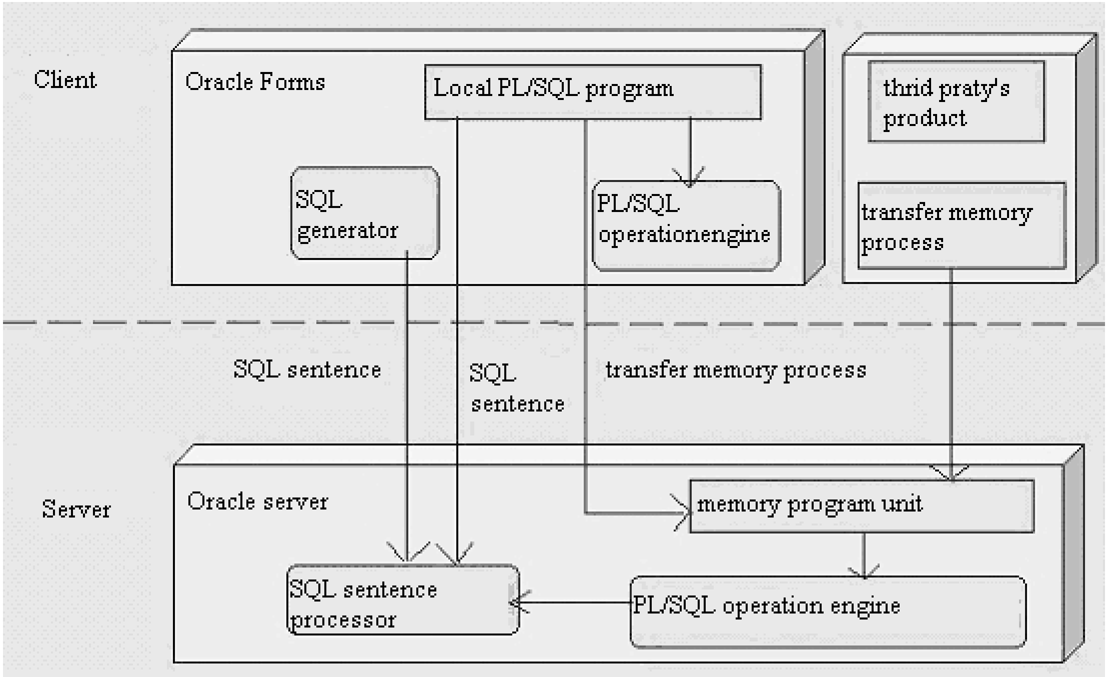

Figure 1. PL/SQL in Oracle Client/Server System Structure

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A New Approach for Data Clustering Based on PSO with Local Search

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Abstract

Data clustering is a popular approach for automatically finding classes, concepts, or groups of patterns. The term "clustering" is used in several research communities to describe methods for grouping of unlabeled data. These communities have different terminologies and assumptions for the components of the clustering process and the context in which clustering is used. This paper looks into the use of Particle Swarm Optimization (PSO) for cluster analysis. In standard PSO the non-oscillatory route can quickly cause a particle to stagnate and also it may prematurely converge on suboptimal solutions that are not even guaranteed to local optimal solution. In this paper a modification strategy is proposed for the particle swarm optimization (PSO) algorithm and applied in the data sets. This paper provides a method for particles to steer clear off from local stagnation and the local search is applied to improve the goodness of fitting. The effectiveness of this concept is demonstrated by cluster analysis. Results show that the model provides enhanced performance and maintains more diversity in the swarm and thereby allows the particles to be robust to trace the changing environment.

Keywords: PSO, Roulette-Wheel selection, K-Means, Local Search, Stagnation, Optimization

1. Introduction

Clustering algorithms can be categorized as either hierarchical or optimization. Hierarchical clustering techniques proceed by either a series of successive merges or a series of successive divisions. The result is the construction of a tree like structure or hierarchy of clustering's which can be displayed as a diagram known as a dendogram. Agglomerative hierarchical methods begin with the each observation in a separate cluster. These clusters are then merged, according to their similarity (the most similar clusters are merged at each stage), until only one cluster remains. Divisive hierarchical methods work in the opposite way. An initial cluster containing all the objects are divided into sub-groups (based on dissimilarity) until each object has its own group. Agglomerative methods are more popular than divisive methods.

Unlike hierarchical techniques, which produce a series of related clustering's, optimization techniques produce a single clustering which optimizes a predefined criterion or objective function. The number of clusters in this clustering is either specified a priori or is determined as part of the clustering method. Optimization methods start with an initial partition of objects into a specified number of groups. Objects are then reassigned to clusters according to the objective function until some terminating criterion is met. These methods differ with respect to the starting partitions, the objective functions, the reassignment processes, and the terminating criteria. Unlike hierarchical clustering techniques, optimization methods do not store similarity matrices. Thus the size of the data is not limited by storage space. However, there are a number of disadvantages affecting optimization methods:

(i) Some methods require the number of clusters a priori, and will divide the data into this number of clusters regardless of the data structure;

(ii) Certain clustering criterion are biased towards particular cluster shapes, and will impose these shapes on the data; and

(iii) The performance of optimization techniques is highly dependent on the initial partition.

In this study, a data clustering algorithm based on Simple PSO, Roulette Wheel Selection and K-Means algorithm. The remainder of this paper is organized as follows: Section 2 provides the related works in clustering. Section 3 gives a general overview of the PSO. The proposed PSO clustering algorithm is described in Section 4. Section 5 presents the detailed experimental setup and results for comparing the performance of the proposed PSO algorithm with the K-means approaches.

2. Related works

Even though there is an increasing interest in the use of clustering methods in pattern recognition [1], image processing [2] and information retrieval [4], clustering has a rich history in other disciplines [5] such as biology, psychiatry, psychology, archaeology, geology, geography, and marketing. Other terms more or less synonymous with clustering include *unsupervised learning* [5], *numerical taxonomy* [6], *vector quantization* [7], and *learning by observation* [8]. The field of spatial analysis of point patterns [9] is also related to cluster analysis. The importance and interdisciplinary nature of clustering is evident through its vast literature. A survey of the state of the art in clustering *circa* 1978 was reported in Dubes and Jain [10]. A comparison of various clustering algorithms for constructing the minimal spanning tree and the short spanning path was given in Lee [11]. Cluster analysis was also surveyed in Jain et al. [12]. A review of image segmentation by clustering was reported in Jain and Flynn [2]. Comparisons of various combinatorial optimization schemes, based on experiments, have been reported in Mishra and Raghavan [13] and Al-Sultan and Khan [16].

3. Particle Swarm Optimization

Swarm Intelligence (SI) is an innovative distributed intelligent paradigm for solving optimization problems that originally took its inspiration from the biological examples by swarming, flocking and herding phenomena in vertebrates. Particle Swarm Optimization (PSO) incorporates swarming behaviors observed in flocks of birds, schools of fish, or swarms of bees, and even human social behavior, from which the idea is emerged [17][18]. PSO is a population-based optimization tool, which could be implemented and applied easily to solve various function optimization problems. As an algorithm, the main strength of PSO is its fast convergence, which compares favorably with many global optimization algorithms like Genetic Algorithms (GA) [22], Simulated Annealing (SA) [20.] and other global optimization algorithms. For applying PSO successfully, one of the key issues is finding how to map the problem solution into the PSO particle, which directly affects its feasibility and performance.

Bird flocking optimizes a certain objective function. Each particle knows its best value so far (pbest) and its position. This information is analogy of personal experiences of each particle. Moreover, each particle knows the best value so far in the group (gbest) among pbests. This information is analogy of knowledge of how the other particles around them have performed. Namely, each particle tries to modify its position using the following information:

- current positions
- current velocities
- distance between the current position and pbest
- distance between the current position and gbest

This modification can be represented by the concept of velocity. Velocity of each particle can be modified by the following equation:

$$v_{id} = w * v_{id} + c_1 * rand() * (P_{id} - X_{id}) + c_2 * rand() * (P_{gd} - X_{id}) \quad (1)$$

Where, v_{id} : velocity of particle

x_{id} : current position of particle

w : weighting function,

c_1 & c_2 : determine the relative influence of the social and cognitive components

p_{id} : pbest of particle i,

p_{gd} : gbest of the group.

The following weighting function (2) is usually utilized in

$$w = w_{\max} - \frac{w_{\max} - w_{\min}}{iter_{\max}} x \; iter \quad (2)$$

Where, $w_{\max}$: initial weight,

$w_{\min}$: final weight,

$iter_{max}$: maximum iteration number,

iter : current iteration number.

Using the above equation, a certain velocity, which gradually gets close to pbest and gbest can be calculated. The current position (searching point in the solution space) can be modified by the following equation (3):

$$X_{id} = X_{id} + V_{id} \quad (3)$$

Fig. 1 shows the general flow chart of PSO.

The features of the searching procedure of PSO can be summarized as follows:

(a) As shown in equation (1)(2)(3), PSO can essentially handle continuous optimization problem.

(b) PSO utilizes several searching points like genetic algorithm (GA) and the searching points gradually get close to the optimal point using their pbests and the gbest.

(c) The first term of right-hand side (RHS) of (1) is corresponding to diversification in the search procedure. The second and third terms of that are corresponding to intensification in the search procedure. Namely, the method has a well-balanced mechanism to utilize diversification and intensification in the search procedure efficiently.

The above feature (c) can be explained as follows [18]. The RHS of (2) consists of three terms. The first term is the previous velocity of the particle. The second and third terms are utilized to change the velocity of the particle. Without the second and third terms, the particle will keep on "flying" in the same direction until it hits the boundary. Namely, it tries to explore new areas and, therefore, the first term is corresponding to diversification in the search procedure. On the other hand, without the first term, the velocity of the "flying" particle is only determined by using its current position and its best positions in history. Namely, the particles will try to converge to the pbests and/or gbest and, therefore, the terms are corresponding to intensification in the search procedure.

4. Proposed PSO for Data clustering

The original PSO described in section 3 is basically developed for continuous optimization problems. However, lots of practical engineering problems are formulated as combinatorial optimization problems. Kennedy and Eberhart developed a discrete binary version of PSO for the problems (Kennedy, 1997). The proposed system employs Discrete Binary PSO with gobalized and localized search.

4.1 Problem Formulation

The fitness of panicles is easily measured as the quantization error. The fitness function of the data clustering problem is given as follows:

$$f = \frac{\sum_{i=1}^{N_c} \left\{ \frac{\sum_{j=1}^{p_i} d(0_i, m_{ij})}{p_i} \right\}}{N_c} \quad (4)$$

The function f should be minimized.

where m_{ij} : jth data vector belongs to cluster i

O_i : Centroid vector of the i^{th} cluster

$d(O_i, m_{ij})$: the distance between data vector *mij* and the cluster centroid *Oi.*

Pi : stands for the number of data set, which belongs to cluster ***Ci;***

Nc : number of clusters.

4.2 Particle Representation

In the context of clustering, a single particle represents the cluster centroid vectors. That is, each particle $\mathbf{X}_{ij}$**,** is constructed as follows:

$X_{ij} = (mi_1, m_{i2}, \ldots, mi_m)$

where mij refers to the j-th cluster centroid vector of the i-th particle in cluster Cij. Therefore, a swarm represents a number of candidates clustering for the current data vectors.

4.3 Initial Population

One particle in the swarm represents one possible solution for clustering. Therefore, a swarm represents a number of candidate clustering solutions for the data set. At the initial stage, each particle randomly chooses k different data set from the collection as the initial cluster centroid vectors and the data sets are assigned to cluster based on one iteration of K-Means.

4.4 Local search

After finding the solutions of N particles, a local search is performed to further improve fitness of these solutions. Local search helps to generate better solutions, if the heuristic information can not be discovered easily. Local search is applied on all generated solutions or on a few percent N. In this work, local search is performed on 20% of the total solutions. So in the test data set of N data, local search is applied on the 20% of solutions based on roulette-wheel selection. The requirement is that the fittest individuals have a greater chance of selection than weaker ones. In the local search procedure, the objective function values selected particles are computed again. These solutions can be accepted only if there is an improvement on the fitness, namely, if the newly computed objective function value is lower than the first computed value, newly generated solution replaces the old one.

4.5 Personal best & Global best positions of particle

The personal best position of particle is calculated as follows

$$P_{id}(t+1) = \begin{cases} P_{id}(t) & if\ f(X_{id}(t+1)) \geq f(P_{id}(t)) \\ X_{id}(t+1) & if\ f(X_{id}(t+1) < f(P_{id}(t)) \end{cases}$$

The particle to be drawn toward the best particle in the swarm is the global best position of each particle. At the start, an initial position of the particle is considered as the *personal best* and the *global best* can be identified with minimum fitness function value.

4.6 Finding new solutions

According to its own experience and those of its neighbors, the particle adjusts the centroid vector position in the vector space at each generation. The new velocity is calculated based on equation (1) and changing the position based on equation(3)

Generally, in PSO algorithm, operations described above are iterated in main loop until a certain number of iterations are completed or all particles begin to generate the same result. This situation is named as stagnation behavior, because after a point, algorithm finishes to generate alternative solutions. The reason of this situation is, after a certain number of iterations, particles generate continuously the same solutions. Aiming minimizes the stagnation behavior of particles, the proposed technique follows the Quantization error of particles and if there is no change on the error after last 10 iterations, it moves the particles with the random velocities. In other words, to improve the solution, a feedback technique is applied on the algorithm. Fig 2 demonstrates the proposed Hybrid PSO for data clustering.

5. Experiment Results

In this section, results from the proposed PSO method and the K-Means on well-known test data sets are reported. The choice of the parameter values seems not to be critical for the success of the methods; it appears that faster convergence can be obtained by proper fine-tuning. The balance between the global and local exploration abilities of the proposed system is mainly controlled by the inertia weight, since the positions of the particles are updated according to the classical PSO strategy. A time decreasing inertia weight value, i.e., start from 0.9 and gradually decrease towards 0.4, proved to be superior to a constant value. The optimal solution (fitness) is determined with N=20, c1= 2.1 & c2 = 2.1. The test data sets are obtained from UCI's machine learning repository [23]. The Results obtained from test data sets by K-Means and the proposed system are shown in Table 1 & Table 2 respectively.

Iris plants database: This is a well-understood database with **4** inputs, **3** classes and 150 data vectors.

Wine: These data are the results of a chemical analysis of wines grown in the same region in Italy but derived from three different cultivars. The analysis determined the quantities of 13 constituents found in each of the three types of wines.

Glass identification: From USA Forensic Science Service; 6 types of glass; defined in terms of their oxide content.

For each data set with two different distance measures 50 runs have been performed using the proposed PSO and the performance is exhibited in terms of the Fitness value , Inter and Intra Cluster similarity. Results for all of the aforementioned datasets are reported with the conventional cluster algorithm K-Means. Table 1 illustrates the analysis of the results for K-Means and Table 2 shows for Proposed PSO system

Conclusion

The advantages of the PSO are very few parameters to deal with and the large number of processing elements, so called dimensions, which enable to fly around the solution space effectively. On the other hand, it converges to a solution very quickly which should be carefully dealt with when using it for combinatorial optimization problems. In this study, the proposed PSO algorithm developed for data-clustering problem is verified on the datasets. It is shown that it increases the performance of the clustering and the best results are derived from the proposed technique. Consequently, the proposed technique markedly increased the success of the data-clustering problem.

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Table 1. Analysis with K-Means

Data sets	Distance Measure	K-Means Clustering		
		FV	Intra	Inter
Iris	Euclidean	0.8013	0.0616	5.2805
	Chebychev	0.6873	0.1902	4.7052
Wine	Euclidean	126.14	11.4103	759.170
	Chebychev	124.68	11.0918	759.008
Glass	Euclidean	1.5968	0.49094	6.2713
	Chebychev	1.1856	0.2544	5.0068

Table 2. Analysis with Proposed PSO System

Data sets	Distance Measure	Proposed PSO System		
		FV	Intra	Inter
Iris	Euclidean	0.5439	0.0616	9.8228
	Chebychev	0.4209	0.0537	9.2193
Wine	Euclidean	83.826	5.4399	831.25
	Chebychev	83.416	3.9643	822.12
Glass	Euclidean	0.5991	0.4909	10.2561
	Chebychev	0.4209	0.1569	9.8352

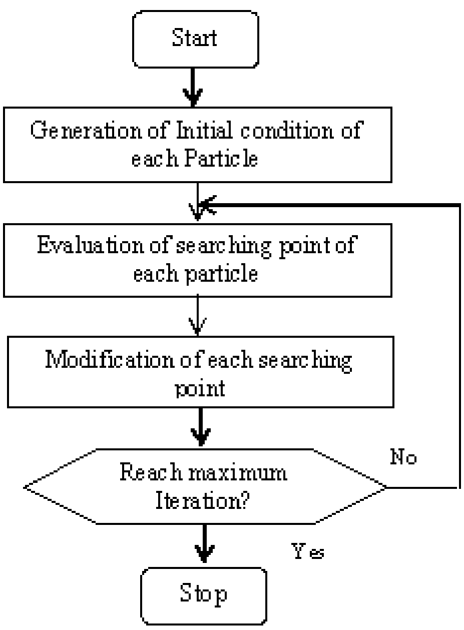

Figure 1. Simple PSO

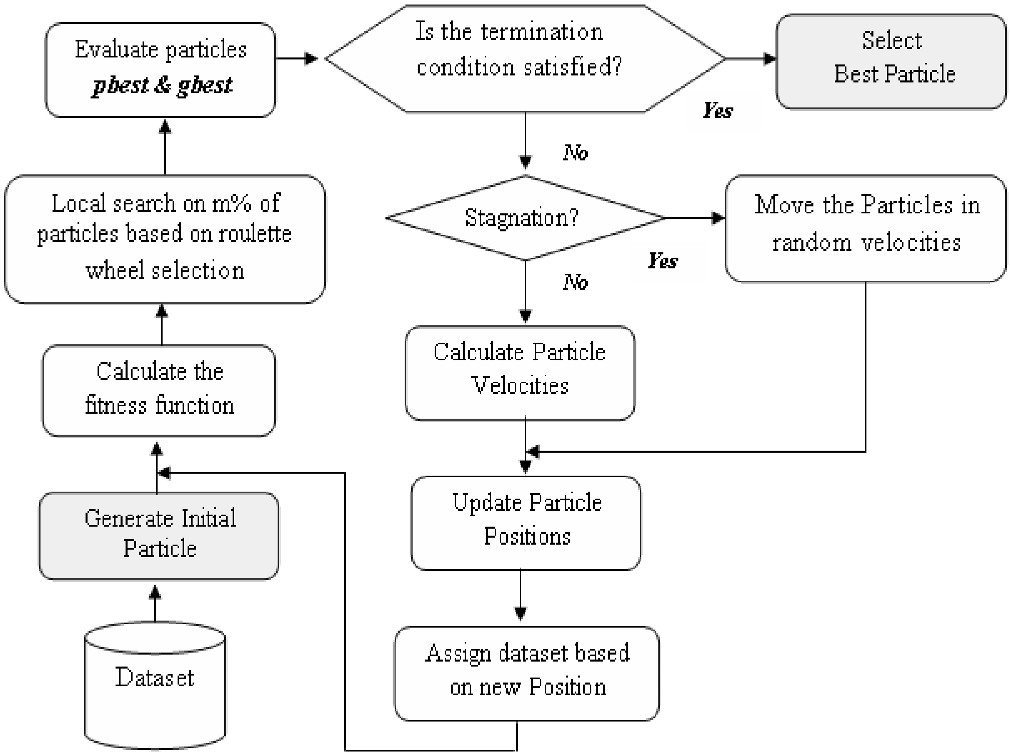

Figure 2. Hybrid PSO for Data Clustering

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Radio Propagation for Wireless Ultra-wide Band Systems

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Abstract

Wireless ultra-wide band(UWB) wave has been studied by incomplete Mathematical Method, the multiple paths of electromagnetworksic wave on UWB broadband is described from propagate to receive, and analysis mechanism of quality that is affected on the channel by received signal quality: propagation, Teflection, and Transmisson,: practice shows that the Mathematical method can explain the channel characteristics are influenced by the buildings, vegetation and terrain, Review on the crucial effect from the quantum theory on the characteristic and mechanisms of UWB signal, so there are the guidable function which mathematic models are been setted up for the channel indoor and outdoor models .

Keywords: Wireless ultra-wide band (UWB), Reflecting, Transmission, Diffraction, Uniform theory of diffraction(UTD)or geometrical theory of diffraction(GTD), Quantum, Quantum entanglement phenomenon

1. preface

There are many types of Ultra Wide Band (UWB) signal shape, polycycle pulse shape, Gaussian Doublet pulse shape, etc., for example, two characteristics of Narrow pulse signal pulse :the trans form of signal excitation is single short pulse shape leading edge, and spectrum include regions from direct current to microwave frequency line.

It is physical process that of propagation is electromagnetworksic wave makes for mutual action with medium.

Expression for Model of UWB signal:

$$p_x(t) = \frac{d^x(p(t))}{dt^x} \qquad (1\text{-}1)$$

We get Power spectral density function:

$$p(f) = e^{(-2(2\pi f\sigma)^2} \int_{-\infty}^{+\infty} \frac{1}{\sqrt{2\pi\sigma}} \exp\left(\frac{-(t + j2\pi f\sigma)^2}{2\sigma^2}\right) dt \qquad (1\text{-}2)$$

It compare with pulse types of Gaussian shape :

We have power:

$$p_r(t) = \mathrm{a}_1^2 \left\{ e^{-2(t-T_1)^2/\tau^2} + \left(\frac{a_2}{a_1}\right)^2 e^{-2(t-T_2)/\tau^2} + 2\frac{\mathrm{a}_2}{\mathrm{a}_1}[\mathrm{e}^{-(\mathrm{t}-\mathrm{T}_1)^2/\tau^2} e^{-(t-T_2)/\tau^2}] COS(\phi_2 - \phi_1 - \omega\tau) \right\} \qquad (1\text{-}3)$$ [2]

In (1-3), $T_2 - T_1 = \tau$, a_1, a_2 is constant,

It show that Beam is symmetrical, and feeding equalizer filter.

Equivalent circuit:

According to no uniformity properties of microwave, it can Equivalent effect by use networks, in principle:

$$\mathrm{p} = \frac{1}{2} \mathrm{Re} \int_s (\mathrm{E_T} \times \mathrm{H_T^*})$$

$$\mathrm{ds} = \frac{1}{2} \mathrm{Re}(\mathrm{UI}^*)$$

We get condition of normalizer:

$$\int_s (\mathrm{e_T} \times \mathrm{h_T}) ds = 1 \qquad (1\text{-}4)$$

$$\mathrm{P(f)} = \exp(-2(\pi f\sigma)^2) \int_{-\infty}^{\infty} \frac{1}{\sqrt{2\pi}\sigma} \exp(\frac{-(\mathrm{t} + \mathrm{j}2\pi \mathrm{f}\sigma)^2}{2\sigma^2}) dt$$

It compare with pulse types of Gaussian shape:

$$\mathrm{P_r}(t) = a_1^2 \{e^{-2(t-T_1)^2/\tau^2} + (\frac{a_2}{a_1})^2 e^{-2(t-T_2)/\tau^2} + 2\frac{\mathrm{a}_2}{\mathrm{a}_1}[\mathrm{e}^{-(\mathrm{t}-\mathrm{T}_1)^2/\tau^2} \mathrm{e}^{-(\mathrm{t}-\mathrm{T}_2)^2/\tau^2}]\cos(\phi_2 - \phi_1 - \omega\tau)\} (1-5)$$

It shows receiving envelope voltage that it is at different phase shift of two aims.

Network function of UWB signal needn't change into sine wave shape, in general, its spectrum of pulse sequence is made equality by pseudo Random sequence, at most case, it is propagated by code Unit in UWB system.

Considerate parameter and factor of UWB Radio Propagation environment, add electromagnetic properties and electromagnetic properties of mediun and objects, then responsible for electromagnetic wave ,search for properties on Division and frequency

Area, it is primary content.

2. Propagation mechanisms for UWB Radio

Frequencies and wavelengths of the electromagnetic spectrum regions for UWB Radio from "3.1 GHz" to "10.6 GHz", and from "C "Band Interleaved to "X" Band Interleaved.

Around the ideal- free space environment, the research on foundation for UWB Radio Propagation is given by

$$\nabla \times E = -\mu \frac{\partial \mathrm{H}}{\partial t} \ (2\text{-}1)$$

$$\nabla \times \mathrm{H} = \varepsilon \frac{\partial \mathrm{H}}{\partial t} \ (2\text{-}2)$$

And average Poynting vector formula:

$$\mathrm{S_{av}} = \frac{1}{2} \mathrm{Re} \mathrm{E} \times \mathrm{H}^* \qquad (2\text{-}3)$$

it in lossy homogeneous medium,

If σ, μ and εare complex, Equation (2-1), (2-2)

thus becomes

$$\nabla \times E = -\mu \frac{\partial \mathrm{H}}{\partial t} \qquad (2\text{-}4)$$

$$\nabla \times H = \sigma E + \mu \frac{\partial \mathrm{H}}{\partial t} \qquad (2\text{-}5)$$

and it in layered media, from the boundary conditions, cindider UWB plane wave incident from the boundary conditions,

Snell's law are given by:

$$\eta_1 \sin\theta_\mathrm{i} = \eta_1 \sin\theta_\mathrm{r} = \eta_2 \sin_\mathrm{t} \theta_t \qquad (2\text{-}6)$$

For free space propation, UWB Radio propagation, the path gain will decrease as the reflected.

In fact, Radio propagation are be effected of loss by medium or barrier, so UWB Radio propagation will received three of different loss: path propagation loss, shadpw fading loss, fast fading loss; and four main effects: shadowy effect, effects of far and wide fading, Multipath effect, Doppler effect.

Base on standard of IEEE .802.15.4a Personal area network(PAN),as fiducial physical layer, wireless USB attest to keep function of wire USB, and offer wireless connect, and testing UWB Devices ,Measurements of the spatial correlation in indoor environment indicate that the multipath arrivals come from a region about the subscriber that around about 10 metrer, the coherence bandwidth is often taken to be that for which the coherence function, normalized to its peak value,is above 0.5.

3. UWB Radio Propagation Procedure

Single period UWB pulse is nonsinusoidal, there are much difference between period sinusoidal and pulse, also we must consider fully hysteresis loss of absorption wave materials, so Maxwell equations research on transient state solution, not steady state solution to design UWB signal, it can be given modifying Maxwell equations by quote magnetic density.

$$\oint H \cdot dl = \int_s (J + \frac{\partial D}{\partial t}) \cdot ds$$

$$\nabla \times H = -\frac{\partial B}{\partial t}$$

Method for nonsinusoidal wave Reflection and Transmission in two different nonsinusoidal wave, are the conclusion, that is deal with oscillator, refraction law and snell's law are fitting, where space medium characteristics change slowy with position, electric field strength of field E are given by Geometrically optical method in uniform mediums, when incident ray encounters of scatter objects near faces, edges, corner, bare wedge-shaped and rolling terrain new diffraction ray.

Diffraction ray don't solve the Boundary problem of electromagnetic field, and don't suit Geometrically optics law, so that distribute for UWB electromagnetic wave field in space is determined by Uniform Theory of diffraction and time dispersion, and determine the receiver electric field strength, the method have been used with success to study 3D complex building environments, and computing the electromagnetic fields have been developed to predict the propagation characteristics.

On the other hand, UTD algorithm is used with every ray by 3D complex radiation field alternating and real condition of all UWB Radio Reflection, Transmission and Diffraction by determine ray azimuth and elevation angle, UTD use Go Geometrical optics principle, for example ,that Ray Tracing technology for ray shooting begin from source of rays, and follow propagating finish propagating simulation, to get UWB channel in pulse response or transfer function with geography information of Arrival.

Several Reflection or Refraction etc., thus that method of UTD can get function of UWB signal voltage

We have studied experiment of UWB wave process from propagation to receive antenna, pulse amplitude and wave traveling have observed , parameter involve normalized pulse funation, spectrum of pseudo random

made pulse sequence uniformly.

UWB radio flashy function are gotten during 25ns and 1ms times, its characteristics are quantum code convolution, and very ability of correct mistaken, so we use quantum theory for analyse its data.

4. Quantum calculate for UWB Radio propagating

Significant applications of group theory to quantum physics of crystalline solids derives from the rotational-reflectional symmetry of the entire crystal lattice under a group of rotations, reflections, and translation.

Under line Axiom does not require the commutative law,

$$\forall a,b \in G, a \circ b = b \circ a$$

When we discuss propagation, we are dealing with microwave radiation theorem, and use five hypothesis of quantum mechanics ,its foundation is the solutions of the homogeneous Helmholtz equation

$$(\nabla^2 + k^2)\psi(r) = 0$$

And the schrodinger equation

Thus, equation is given by

$$H\psi(r) = E\psi(r)$$

Discrete quantum system is an element of c^n such that the normalization condition[2].

$$\sum_{i=1}^{n} \left|\psi^i\right|^2 = 1$$

Is obeyed:

Quantum codes have shown ability strongly in UWB system, and Quantum cryptography Based on entanglement phenomenon can change into NP mathe matical problem,-----etc. so its have come true.

According to Quantum no-cloning, quantum position of unknown status travel by quantum entanglement phenomenon.

5. summary

Computer codes the directions of the rays at both ends of the links for computing the electromagnetic fields have been developed to predit the propagation characteristics for the UWB system simulations, classical is used to analys UWB Radio reflection and Transmission, but diffraction by Edges and corners can only use UTD theory ,and scatter and Radiation need use quantum theory, the simulation results show UWB Radio analysis correctly.

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General Framework of Chain Store Information System base on Supply Chain Management Theory

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Abstract

The purpose of this paper is to develop a chain store information system framework. This paper summarized information system requirements of chain store according to supply chain management theory firstly. It classified three types of information requirements. Then an information flow diagram was designed. This paper also projected concept structure, software structure and architecture of chain store information system. It proposed to use the distributing system and B/S computerizing module in system architecture.

Keywords: Chain store, Supply chain, Information system, System framework

1. Introduction

Now, because of the cruel competition, more and more retailers operate their business with the form of chain store. Following the company's development, some new problems emerge. These problems lead them to notice a full new management theory: the supply chain management theory. Different from other management theory, this theory emphasizes the cooperation and sharing information with your suppliers. So it is significant for building an efficient information system base on supply chain management theory.

2. System Requirements Analysis

The information system requirements of chain store include three aspects at least: basic information service for management, decision-making information and value-added service information.

2.1 Basic Information Service for Management

The information exchange between consumer and retailer: It includes business service information, the customer demand and complaint information.

The information exchange between retailers and corporate headquarters: It includes article of commerce information, adjust commodities information, advertisement and sale promotion activity information, marketing plan information, human resource management information, complementarity commodities information, branch vendition information, financial information, customer demand and complaint information.

The information exchange between corporate headquarters and distribute center: It includes distribution order, acceptive goods order, inventory information.

The information exchange between company headquarters and suppliers: It includes supply information and purchase information.

2.2 Decision-making Information

The information of upgrading service: Most of the information refers to customers such as popular commodities information, sale out information, the analysis results of customer complaint, suppliers' ability, amendatory product information and so on. The quality of service can be improved by analyzing the information.

The feedforward control information. The information includes enterprise internal management information, marketing information, competitor information, environment information and the forecast according the information above.

Expand information. The information belongs to investment and strategy information. There are four aspects should be considered: the population of new investment place, the culture, competition environment and the social political situation.

2.3 Value-added Service Information

The chain store provides paid information search and inquiry service for customers combining with manufacture factory and wholesale. By processing the basic management information and decision-making information, the information is benefit for profits increase and cost decrease. Indeed we can regard this as the enterprise resource mining.

3. Information Flow Diagram of Chain Store

Before designing a framework of information system, it is essential to comprehend information flow. Each function of management has to rely on logistic information flow, fanatical information flow, sales and purchase information flow.

3.1 Logistic Information Flow

Distribute centers accept supply order form headquarters and arrange transport then check and turn into warehouse. Distributes also take responsibility for putting them into different packages and distribute to different stores. At the same time, headquarters will transmit commodities' variety, serial number and price information using information system. According to the information, each store sales commodities to customer.

3.2 Purchase Information Flow

Headquarters must take sale and inventory situation into account to affirm the purchase commodities' variety, amount and price. According to headquarters' order and the negotiation between purchase department and suppliers. In the light of headquarters' order, distribute centers transport goods and turn them into warehouse.

3.3 Sale Information Flow

Each store transmits their sale report to headquarters by network. Headquarters counts up and analyses these reports to understand the sale situation which is essential for purchase inventory activity.

Headquarters also transmits commodities' variety, serial number, amounts and price information by network. After each store submits his makeup commodity report form, headquarters ask each distribute center to transport these commodities in time. Each store must report their sale situation including popular commodities and unmarketable commodities information by day. Headquarters analyses them then decides the market strategy and sale promotion event.

3.4 Financial Information Flow

The financial dynamic analysis is implemented by financial department. Chain store corporation carry on accounting management centering on headquarters. Headquarters uses financial measure to supervise and control the whole management process and analyses companies' profit to judge the cost rising factor and profit improvement factor.

To sum up, information flow diagram of chain store can be drew

4. System Structure Design

4.1 Concept Structure

Chain store information system should contain four parts at least: information fountain, information processor, information user and information manager.

It is easy to find that information witch come from information fountain, are processed and save d by information processor. At last it can be used by information users to make decision. The information manager's responsibilities are designing, running and harmony.

4.2 Function Structure

It is no doubt that every management information system can implement many functions. All of these functions connect each other for a given goal. So chain store information system is a integration system and its' function can be demonstrated below.

4.3 Software Structure

Information system has to depend on system software and application software to implement various functions. Because of the final goal of information system is sharing information, it is necessary to establish data warehouse management system witch supported by specifically operating system and can save abundance to complete multiform functions. In the structure of data warehouse, sharing database provides service for some public data which often required by every subsystem. Of course, each subsystem is accredited to construct independent database which deposit special purpose data. Application software should support diversified management activities otherwise.

Chain store information system formed by six subsystems which is illustrated in figure 4

Headquarters management information system: it not only supervises departments, but also collects distribute center data and local store data.

Distribute center management information system: it implements static and dynamic management of inventory for example: commodities circulation, transportation and warehouse management.

Branch management information system: it contains POS system and security system.

Electronic order system (EOS): order data occur from wholesales and local store is transmited to headquarters, suppliers and manufacturers.

Customer relationship management system: it is a process to compile information that increases understanding of how to manage an enterprise's relationships with it customers. CRM is a strategy that uses information technology to provide a integrated view of its customers.

MAN network systems. MAN network system is a important bridge to connect headquarters, distribute centers, branches, suppliers and banks system that integrates the whole supply chain and reduces costs. With this connection, headquarters gains distribute centers' and branches' data and feedback to them betimes.

5. System Architecture

According to the hardware and software dispose, we can partition it into centralized system and distributing system.

5.1 The Characteristics of Centralized System:

Information facilities are concentrated in somewhere; it is safe and can be used efficiently; personal manager conveniently. But there are some disadvantages: it is weak for meeting an emergency and hard to maintain reliability. It is difficult for repairing, extending and upgrade.

5.2 The Characteristics of Decentralized System:

Information facilities are not disposed in a certain location. Information facilities are distributed in several location and they share information though network. The decentralized system not only can work together, but also can use local information to work independently. It has strong reliability and expansibility. Because the chain store information system has to Cross city, province, even states to operate, it is reasonable for undertake decentralized system. The prevalent B/S computerizing module is usually used in decentralized system. Without installing specify system software, customers can use ordinary browsers (IE etc.) to inquiry data servers though WEB format. B/S is a special three layers' C/S computerizing module and his data transmission format also very multiplex, for example: multimedia etc. Universal communication protocol is independent from platform, so the whole system is very easy to development and maintenance. Its structure is shown in figure 5

MAN network is the key of chain store information system project planning duo to the distributing in different areas of headquarters, distribution centers, and filiales. Each filiale and distributer center develops his LAN network, then these LAN networks connect each other though decentralized network. Figure 6 is a vision of the whole network architecture.

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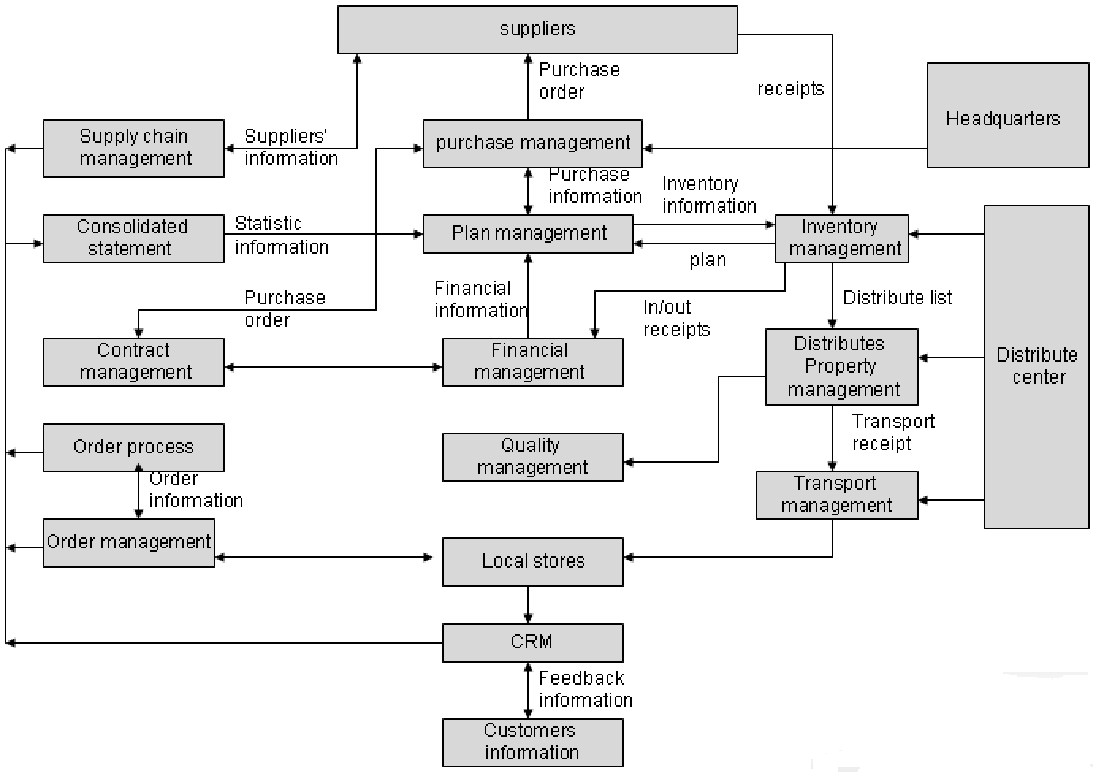

Figure 1. The information flow diagram

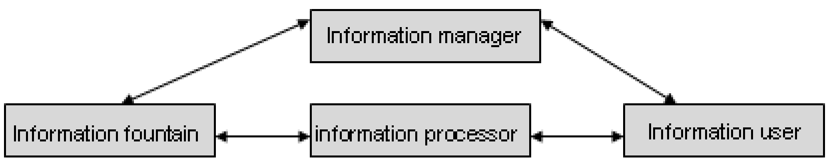

Figure 2. System concept structure

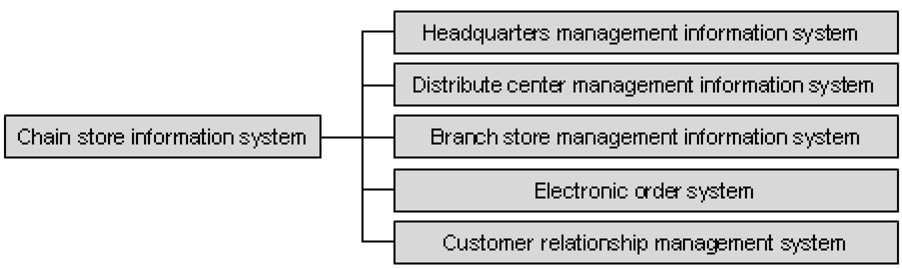

Figure 3. Function structure

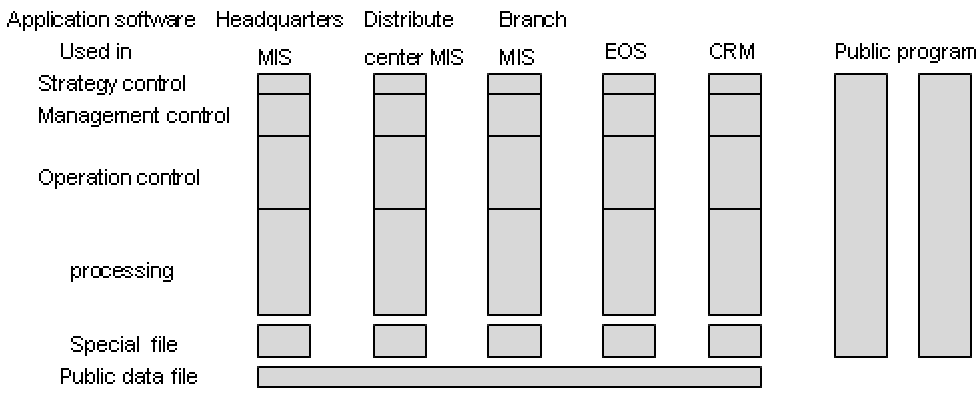

Figure 4. Software structure

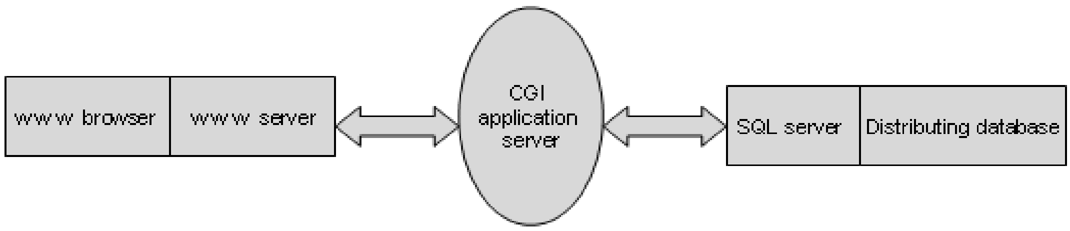

Figure 5. B/S structure

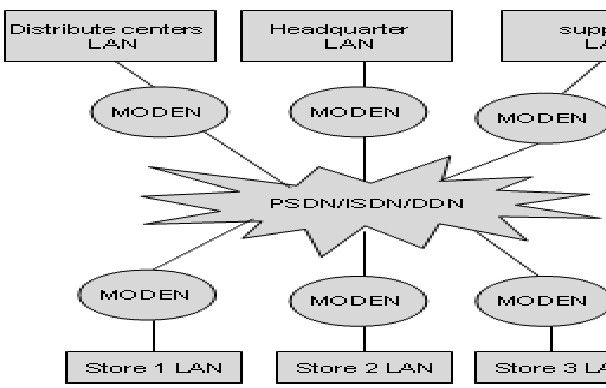

Figure 6. The architecture of whole network

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Architecture of Embedded System Software

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Abstract

The verification of real-life C/C++ code is inherently hard. Not only are there numerous challenging language constructs, but the precise semantics is often elusive or at best vague. This is even more true for systems software where non-ANSI compliant constructs are used, hardware is addressed directly and assembly code is embedded. In this work we present a lightweight solution to detect software bugs in C/C++ code. Our approach performs static checking on C/C++ code by means of model checking. While it cannot guarantee full functional correctness, it can be a valuable tool to increase the reliability and trustworthiness of real-life system code. This paper explains the general concepts of our approach, discusses its implementation in our C/C++ checking tool *Goanna,* and presents some performance results on large software packages.

Keywords: Real-life, C/C++, Non-ANSI, Hardware

1. Introduction

Showing the full functional correctness of system software written, e.g., in C/C++, is a major challenge. It requires a precise understanding of the underlying semantics, typically needs to include an abstract hardware model, and has to give a full functional proof. There are a number of projects currently undertaking this task supported by interactive theorem provers. While this is the only way to guarantee the full correctness of a program, it requires substantial resources both in time as well as in the number of highly qualified people.

On the other hand, commercial system software has a high pressure to market, needs to run on various platforms and is rewritten frequently, making the above approach even more challenging. There are a number of lightweight analysis approaches that seek to complement full verification by detecting software bugs at the coding stage and, thus, increasing the reliability and trustworthiness of the code. Those tools make a limited but practical contribution to program correctness and can support full verification by reducing property violations in early stages.

The model-checking community has made significant advances in recent years to cover realistic C/C++ programs and produced a number of powerful tools. However, they are not yet well-suited for real-life embedded system code. On the other hand, commercial static analysis tools cope well with most C/C++ code and make a valuable contribution to software correctness. In contrast to model checking tools, static analysers typically do not allow for any user-defined specifications, but rather implement a set of independent analysis heuristics or allow specification which are less expressive than the temporal logics used by model checkers.

In this work we present a static analysis approach based on model checking. While we retain the flexibility and power of temporal logics specifications, we are able to handle any parsed C/C++ code in a uniform manner. In particular, we present the underlying idea of translating C/C++ checks into model checking properties, which can then be checked by one single analyzer, instead of a set of static analysis heuristics. In our case we use the NuSMV model checker as back end. Moreover, we present some implementation details of our checker *Goanna* and its performance on the source of large, real-life open-source software packages.

Section 2 describes our underlying framework, while Section 3 presents some of our performance results and Section 4 discusses the current state of our research as well as ongoing and future work.

2. Static Analysis by Means of Model Checking

In this section we describe how to statically check properties of C/C++ source code by means of model checking. This approach has been inspired by and is also followed by.

Using a model checker for solving static analysis problems has a number of advantages. All properties can be expressed in a single, flexible analysis engine. This means that it is easy to add new checks by adding new checking properties. In addition, the analysis scales well with increasing number of properties. The details of our path-sensitive, intra-procedural analysis can be found in.

The basic idea is to annotate the control flow graph (CFG) of a program with atomic propositions of interest. In order to check, e.g., for uninitialized variables, we can identify atomic propositions declq, readq and writeq, representing program locations where a variable *q* is declared but not initialised, where it is read from or written to, respectively, and mark those locations in the CFG accordingly. The atomic propositions are identified by purely syntactic criteria on the abstract syntax tree (AST) of the program by means of a pattern language. We define patterns for each proposition, e.g., a variable is written to if it occurs on the left hand side of an assignment statement and so on. Once identified, the proposition is placed on the node in the CFG most closely corresponding to the nodes in the AST where it was identified.

This means we require that on all program paths if a variable *q* is declared it must not be read until it has been written or it will not be written at all. We use the *weak until* operator W here to include the second possibility. The latter can also point to unused variables, which is checked separately.

In the same style we can express other properties on correct pointer handling, variable usage or memory allocation and deallocation. Moreover, it allows specifying application specific properties to handle general programming guidelines, API-specific rules or even hardware/software interface rules for device drivers.

Once the patterns relevant for matching atomic propositions have been defined and the CFG has been annotated, it is straightforward to translate the annotated graph automatically into the input format of a model checker. Adding new checks only requires one to define the property to be checked and the patterns representing atomic propositions. All other steps can be fully automated.

Although this framework was developed in first instance for C/C++ it can be also extended to deal with embedded assembly code. This is important for the embedded systems space, since interaction with the hardware is frequently implemented as embedded assembly code. In particular, we take C/C++ and ARMv6 assembly interface information for our analysis into account, check for compliance of embedded assembly code with its C/C++ interface, and perform various checks on the pure assembly level. The combined analysis of C/C++ code with embedded assembly code enhances, in addition, the precision of the analysis.

3. Implementation and Evaluation

The aforementioned approach has been implemented in our program analyzer Goanna, using the open source model checker NuSMV [14] as a generic backend analysis engine. The surrounding code for pattern matching structures of interest, property definitions, CFG generation, translation into NuSMV, and representation of analysis results is written in OCaml. Moreover, Goanna can be invoked just like the gcc/g++ compiler and, therefore, integrates seamlessly into standard development environments such as Eclipse.

We evaluated Goanna on a number of open source packages ranging from highly optimized system software such as the L4 microkernel1 to large application code bases such as the 260 kLoC2 Open SSL package. For an unoptimized version of Goanna some run-time results for Open SSL are shown in Figure 3. It shows that over 80% of all files are analyzed within 1 second and that 99% of all files are analyzed within 5 seconds. The whole analysis takes less than 15 minutes. Proportionally, the time spent purely in NuSMV is mostly negligible with 98.7% of all files being analyzed in less then 2 seconds.

The run-times of Figure 3 are based on checking for 15 properties ranging from simple uninitialized variables, over potential null-pointer dereferences, to memory leaks. It is worth to mention that increasing the number of properties typically scales well in our framework as it only increases the number of labels and property specifications in the same NuSMV model, which is handled well by the model checker. For instance, increasing the number of properties from one to 15 only doubled the overall analysis time.

Moreover, we found that the analysis time is not only well within the same order of magnitude as the compile time, but that the memory requirements of the analysis fit easily in the RAM of current developer machines.

The analysis of C/C++ code with embedded assembly code was evaluated for Pistachio 0.4 implementation3 of L4, compiling for an ARM SA1100 architecture. It contains 54 C++ files, two of which have embedded assembly blocks (3.7%), and they include a total of 72 header ¯les, of which 10 have embedded assembly blocks (13.8%). The additional assembly analysis lead to a modest increase from 75.9 seconds to 77.3 seconds, which is an increase of only 1.4 seconds or 1.8%.

4. Conclusion

In this work we presented our framework and results on model checking system software by means of static analysis. We showed how to easily encode static checks as model checking properties, providing the basis for an extendable and flexible checker. Moreover, we implemented our analysis framework in Goanna, the first static checker using NuSMV as its analysis engine, and presented some run-time and scalability results. We showed that this is a viable solution that can be integrated well in the software development process.

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The Forming Theory and Computer Simulation of the Rotary Cutting Tools with Helical Teeth and Complex Surfaces

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Abstract
This paper researched the forming theory of the cutting tools with helical teeth and complex surfaces. Deduced the nonlinear equations of the movement of NC system to generate such tools, presented the way to find the solution of the equations, calculated the cross section graphics of the teeth profile and computer simulation of NC machining.

Keywords: rotary cutting tools, Helical teeth, Computer simulation

1. The cutting edges of the rotary tools with helical teeth

Fig.1 showed a rotary tool with complex surfaces. The coordinate of any point on the cutting edge:

$$x_{PA}^{(1)} = x_{PA}\,,\, y_{PA}^{(1)} = r_{PA}\cos(\psi_A + \varphi)\,,\, z_{PA}^{(1)} = r_{PA}\sin(\psi_A + \varphi) \qquad (1)$$

Where r_p : the rotary radius of any point P

ψ_A : The position angle of the radius line relate to the *xoy* plan.

From derivative geometry, we know that ψ should be satisfied by the following equation

$$\frac{d\psi_A}{dx_{PA}} = \frac{tg\beta}{r_{PA}}\sqrt{1+\left(\frac{dr_{PA}}{dx_{PA}}\right)^2} \qquad (2)$$

Where β : the helical angle of cutting edge. The tangent vector of cutting edge:

$$t_{PAx}^{(1)} = \frac{dx_{PA}^{(1)}}{dx_{PA}} = 1$$

$$t_{PAy}^{(1)} = \frac{dy_{PA}^{(1)}}{dx_{PA}} = \frac{dr_{PA}}{dx_{PA}}\cos(\psi_A + \varphi) - r_{PA}\frac{d\psi_A}{dx_{PA}}\sin(\psi_A + \varphi) \qquad (3)$$

$$t_{PAz}^{(1)} = \frac{dz_{PA}^{(1)}}{dx_{PA}} = \frac{dr_{PA}}{dx_{PA}}\sin(\psi_A + \varphi) + r_{PA}\frac{d\psi_A}{dx_{PA}}\cos(\psi_A + \varphi)$$

If x_{pA}, r_{pA}, ψ_A are substituted by $x_{pB}, r_{pB}, \psi_B + 2\pi/z$, equation (1) will be the coordinates of the two cutting edged A and B, which are in the neighborhood with each other, respectively. And so are the Eqs (3), after rotate an angle ϕ about axis x, ψ_A should be substituted by $\psi_B + 2\pi/z$. z represents the number of teeth.

2. the establishment of the coordinate system

In the Fig.2, a grinder is generating a rotary cutting tool. Two coordinate system need to be established.

1) The fixed coordinate system *Oxyz*

The *x*- axis of this system is the rotation axis of the work; the original point O of this system is at the rotation center of the larger end of the work.

2) Grinder coordinate system *Oxyz*

The original point *Os* is at the center of grinder wheel, the y_s axis is parallel to y -axis, the angle between x_s and x equal to $\pi/2-\Sigma$. The coordinates of point *Os* in fixed system are x_c, y_c, z_c. The coordinate transformation of two systems can be state as

$$\begin{bmatrix} x \\ y \\ z \end{bmatrix} = \begin{bmatrix} \sin\Sigma & 0 & \cos\Sigma \\ 0 & 1 & 0 \\ -\cos\Sigma & 0 & \sin\Sigma \end{bmatrix} \begin{bmatrix} x_s \\ y_s \\ z_s \end{bmatrix} + \begin{bmatrix} x_c \\ y_c \\ z_c \end{bmatrix} \tag{4}$$

3. The surface equation and normal vector of grinder

The surface of grinder is a circular cone, let M_b represent any point on the cone, the coordinates in local system of M_b can be expressed as following (see Fig.2)

$$x_{MB}^{(S)} = -(R - r_{mB})tg\alpha_B \,, y_{MB}^{(S)} = -r_{mB}\cos\theta_B \,, z_{MB}^{(S)} = r_{MB}\sin\theta_B \tag{5}$$

Where r_{mb} : The radius of point M_b

θ_b : The angle position parameter

α_b : The bottom angle of the grinder

The coordinates in the fixed system are

$$x_{MB}^{(0)} = -(R - r_{MB})tg\alpha_B \sin\varepsilon + r_{MB}\sin\theta_B\cos\varepsilon + x_c$$

$$y_{MB}^{(0)} = -r_{MB}\cos\theta_B + y_c \tag{6}$$

$$z_{MB}^{(0)} = (R - r_{MB})tg\alpha_B \cos\varepsilon + r_{MB}\sin\theta_B\sin\varepsilon + z_c$$

The normal vector in the local system is:

$$n_{MB}^{(S)}(-\cos\alpha_B, -\sin\alpha_B\cos\theta_B, \sin\alpha_B\sin\theta_B) \tag{7}$$

The normal vector in the fixed system is:

$$n_{MB}^{(0)} = (-\cos\alpha_B\sin\Sigma + \sin\alpha_B\sin\theta_B\cos\Sigma, -\sin\alpha_B\cos\theta_B, \cos\alpha_B\cos\Sigma + \sin\alpha_B\sin\Sigma) \tag{8}$$

Let *Ma* represent any point at the largest circle of the grinder, the coordinate of point *Ma* in local system can be expressed as following:

$$x_{MA}^{(S)} = 0 \,, y_{MA}^{(S)} = -R\cos\theta_A \,, z_{MA}^{(S)} = R\sin\theta_A \tag{9}$$

Where R: the radius of the largest circle of the grinder

θ_A : The angle position parameter

The coordinates of the same point in the fixed system are

$$x_{MA}^{(0)} = R\sin\theta_A\cos\varepsilon + x_c \,, y_{MA}^{(0)} = -R\cos\theta_A + y_c \,, z_{MA}^{(0)} = R\sin\theta_A\sin\varepsilon + z_c \tag{10}$$

4. The moving equations of NC system

The problem to be solved by this thesis is: when use a grinder with curtain shap to generate the teeth of the cutter, the two neighborhood cutting edges A and B should be made out by the two sides of the grinder in the same time, so that, the relative movement of the work and grinder should satisfy special mathematical and geometrical relationship.

Suppose that, the contact points on B-side of grinder and cutting edge B of work are M_b and P_b respectively. The coordinates in the fixed system should be the same

$$x_{mb} = x_{pb}, y_{mb} = y_{pb}, z_{mb} = z_{pb} \tag{11}$$

Following relation can be derived from Esq. (1) and Esq. (6)

$$x_{pb} + (R - r_{MB})tg\alpha_B \sin\Sigma - r_{MB}\sin\theta_B\cos\Sigma = x_c$$

$$y_{mb} + r_{MB}\cos\theta_B = y_c \tag{12}$$

$$z_{mb} - (R - r_{MB})tg\alpha_B \cos\Sigma - r_{MB}\sin\theta_B\sin\Sigma = z_c$$

The normal vector of the grinder at Mb should be vertical to the tangent vector of the cutting edge at Pb, so that

$$n_{mb} \bullet t_{pb} = 0 \tag{13}$$

From Esq.(3) and Esq.(8), we obtain:

$$(t_{pbz} \sin\Sigma + \cos\Sigma)\sin\theta_b tg\alpha - t_{pby}\cos\theta_b tg\alpha - (\sin\Sigma - t_{pbz}\cos\Sigma) \tag{14}$$

Suppose that the contacting points of the largest circle A and cutting edge *a* are *Ma* and *Pa* respectively, the coordinates of such two points in the fixed system should be the same:

$$x_{ma} = x_{pa}, y_{ma} = y_{pa}, z_{ma} = z_{pa} \tag{15}$$

From Esq.(1) and Esq.(9), we obtain:

$$x_c = x_{pa} - R\sin\theta_a \cos\Sigma, y_c = r_{pa}\cos(\psi_a + \phi) - R\cos\theta_a, \quad z_c = r_{pa}\sin(\psi_a + \phi) - R\sin\theta_a \sin\Sigma \tag{16}$$

At the same time, on the same grinder, the position of the center point C should be single. From Esq.(12) and Esq.(16), following equation can be derived.

$$x_{MB} + (R - r_{MB})tg\alpha_B \sin\varepsilon - r_{MB}\sin\theta_B \cos\varepsilon = x_{pa} - R\sin\theta_a \cos\Sigma \tag{17}$$

$$y_{MB}^{(0)} + r_{MB}\cos\theta_B = r_{pa}\cos(\psi_a + \phi) - R\cos\theta_a \tag{18}$$

While, during the generation, the coordinate Zc remain unchanged, so that:

$$z_c = r_{pa}\sin(\psi_a + \phi) - R\sin\theta_a \sin\Sigma \tag{19}$$

$$z_{mb} - (R - r_{MB})tg\alpha_B \cos\Sigma - r_{MB}\sin\theta_B \sin\Sigma = z_c \tag{20}$$

Esq. (14)(17)(18)(19), and (20) form a nonlinear equation system with five equations, In these equations there are 6 unknown parameters: $\theta_a, \theta_b, x_{pa}, x_{pb}, r_{mb}, \phi$. One of the parameters can be given before hand, so the equations are solvable.

5. Example

There is a rotary tool with the outside surface of a transected cone, as showed in Fig.4. The main parameters of the tool areas following:

The maximum diameter: d=10mm

The length of the cutter: l=24.5mm

The number of teeth: z=20

The tape angle: $\xi = 7^\circ$

The helical angle: $\beta = 20^\circ$

The parameters of the grinder are as following:

The diameter of the grinder: D=80mm

The bottom angle of the grinder: $\alpha_b = 60^\circ$

The parameters of installation are listed below:

Zc=0

Installation angle: $\Sigma = 20^\circ$

Through computer calculation, the coordinates of the grinder center during generation can be solved out and listed bellow. (Deleted)

6. The computer modeling and simulation of the generation of the rotary tools

In generation, the relative displacement of the grinder and the work is very complicate. Only by numerical control operator can realize such motion requirement. For testifying the theoretical deduction and numerical calculation, we used the imported I-DEAS software to simulate the cutting movement of the work and the grinder at the SUN-CAD workstation. This is the first time to use computer solid modeling in research of the rotary tools with complex surfaces, and we made the best use of the new achievement in computer graphics, and computer Aid Design at home and abroad.

The I-deas soft ware system contains 5 families:1)solid modeling, 2)system assembly,3)engineering analysis,4)system dynamics and 5)drafting. The research of this thesis used two of them: solid modeling (Geomod) and engineering drafting(Geodraw)

First, create the object of the work and grinder, according to their shapes, and stored them with Geomod; secondly, made grinder and work to move relate to each other; than do the Boolean operation, at every position. The generation of the work corresponds to "cut object" operator, one of the Boolean calculation. Finally, after the grinder goes through all move positions the teeth profile of the work was formed.

In the beginning, we choose hand operation, by select menu with mouse, let the grinder and work go several steps according to the required replacement, and recorded our operation into a program file. A passage of the I-deas program corresponds to one cutting step is as follows:

Return to main menu---get the stored work----orient it this time----rotate about its axes---give the value of rotate angle---store it the second time---get the stored grinder---translate it along x and y direction respectively ---give the value of the translate amount ---store the grinder once more ---return to main menu again--- select Boolean operation ---select cutting operation ---define the cutter, the grinder---define the object to be cut, the work---store the new cutter work the third time

In running this short passage of program, we can see from the sceen of the computer a series of pictures in consequence:

The original shape of the work;

The original position of the grinder;

The rotation of the work;

The translation of the grinder relate to the work;

The grinder in cutting with the work;

The shape and the size of the chip formed from cutting, it is growing larger and larger every moment;

The work cut is waiting for further cutting;

Than, let the Fortran program, with which the coordinates of grinder center were calculated, to output automatically a passage of program as mentioned above, whenever a displacement is calculated. Every movement is corresponding to one passage of I-deas format exactly. In this way, the whole generate progress can be simulated. The shape of the work with one groove showed in Fig.3. The sectional curvature of the tooth profile showed in Fig.4.

7. Conclusions

From the analysis and research mentioned above, following conclusions can be arrived at:

1) The forming theory of the rotary tools with helical teeth and complex surfaces is correct. To generate such a kind of tools, three coordinates numerical control manufacturing system is needed.

3) From computer simulation, we can testify the formulas of theoretic deduction and result of numerical calculation presented in this paper.

We can examine if the interference and other problem. Which are often caused by incorrect parameter selection, would take place and the way to prevent it before hand.

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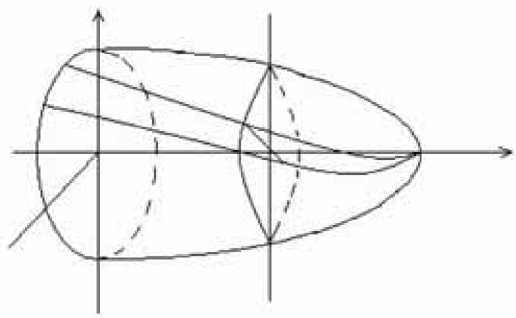

Figure 1

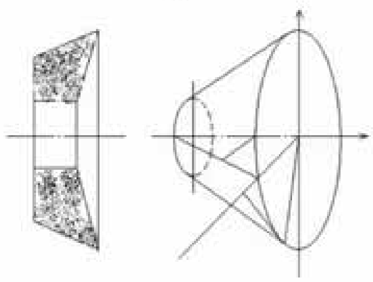

Figure 2

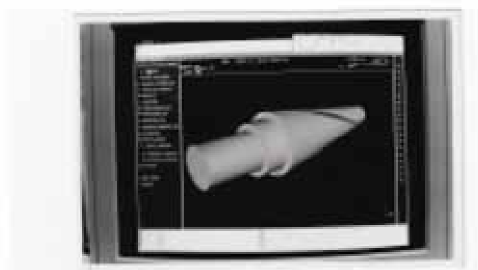

Figure 3. computer simulation of the rotary cutting tools

Figure 4. computer simulation of the rotary cutting tools

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Study on the Lane Mark Identification Method in Intelligent Vehicle Navigation

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Abstract

The intelligent vehicle navigation based on machine vision is the important part to realize Intelligent Transportation System (ITS), and it includes road detection, obstacle detection and motion control. Combining with foreign and domestic latest research trends, in this article, we mainly study the typical method of road detection, and point out the research and development tendency of intelligent vehicle navigation of technology based on the machine vision.

Keywords: Intelligent vehicle, Machine vision, Road detection, Intelligent Transportation System (ITS)

1. Introduction

When human being enters into the 21st century, with the development of the transportation industry, the problems of transportation become more and more serious, for example, the vehicles on roads are crowd, the traffic accidents are frequent, the traffic environment is deteriorated, the energy sources are intense, and the environmental pollution is continually serious. To solve these problems, it must be restricted from many aspects only depending on the extension of road establishment construction. Under this situation, in order to systematically solve the problems of road traffic, we must completely consider vehicles and road, utilize all kinds of high and new technologies, and the new scientific research and application area, i.e. Intelligent Transportation System (ITS). ITS integratively applies advanced information processing technology, computer technology, data traffic technology, sensor technology, electric control technology, GPS technology, automatic control theory, operational research, artificial intelligence and other high and new technologies and theories to transportation, service control and vehicle intelligent. The intelligent vehicle navigation based on machine vision is the important part of ITS, and it mainly completes tasks of road detection and obstacle detection and controls vehicles safely running on roads. Because the computer vision system possesses many characters such as cheap price, multiple purposes and simple structure, and conveniently integrating with other sensors, it has broad application foreground and become the main direction of present research, and the real-time, robustness and reliability are key targets to test the intelligent vehicle navigation system. In this article, we mainly study the road identification in the intelligent vehicle navigation and summarize the research and development tendency of intelligent vehicle navigation of technology based on the machine vision.

2. Road detection

In the intelligent vehicle navigation based on machine vision, the road detection is mainly to estimate the position and direction of vehicle on the road and control vehicles safely running on the road. Otherwise, it can confirm search range for the next obstacle detection, reduce the range of obstacle detection, and reduce the arithmetic complexity and error identification rate. Because the roads in practice differ in thousands ways, the road detection is a very complex problem about pattern recognition, but there is no universal arithmetic of road detection at present. To reduce the treatment difficulty of the road detection and the detection speed of arithmetic, the proper introduction of basic hypothesis about road will help to simplify the problems. In fact, the present road detection systems are to design corresponding detection arithmetic aiming at all kinds of concrete roads. The road supposes in common use at present includes following aspects. The first one is the hypothesis of road form, and the road simplifying model is a sort of effective method, and it implements the road diction by the familiar road form. The second one is the hypothesis of road width, and if the road width can be fixed or changed slowly, the problem of road detection can be simplified to be the parallel road character detection problem. The third one is the consistent hypothesis of road surface character, and usually the area of corresponding road surface in the image possesses the consistent character differing to non-road area, such as the character of gray level, the character of color and the character of texture, so the method of clustering can be adopted to detect the road area. The fourth one is the hypothesis of road evenness, and when the system obtains the image characters (such as the lane mark line), it needs transform the characters from the image coordinate system to the vehicle coordinate system in order to control the vehicle, and if the forward road is even, so we can utilize the known vidicon positioning information to implement transformation. The fifth one is the hypothesis of special interest area, and though the operation quantity of real-time treatment of road following is very large, but according to the physical restriction and the continuity restriction, it an be simplified through the time pertinence between neighbor frames, and we can only analyze and look for expected characters in some interest areas, but not analyze the whole image, and accordingly reduce the treatment time and reduce the complexity of the arithmetic. The present road detection arithmetic mainly includes two sorts, the method based on road character and the method based on road model.

2.1 Road detection based on characters

The road detection arithmetic based on character mainly includes two parts, the character abstraction and the character integration. First analyze the road image and confirm which characters should be selected, then use these characters to implement image partition, and finally compose the partition results according to certain rules to visual road expression. The road character selection can be considered respectively from area view and edge view. The character selection based on area is mainly to analyze the differences between road area and non-road area, and the both dissimilarity can be characters such as color, texture and gray level. In the colorful road image, the road color has large difference with the environment, and we can use this character to realize the partition of road area. Because of uneven illumination and shadow, the colors on the road in one image can not go all the way, and the environmental colors differ in thousands ways, and only the partitions of road area and non-road area are difficult to realize the road partition. In SCARF, the road area and the non-road area are divided into four sorts to enhance the nicety of the partition. Some articles used the method of morphologic to realize the local clustering of similar areas in the image and complete the area integration by means of the rules and obtain the complete road area. The texture is a distinct character in the road image. The road texture is relatively single and the arrangement is ordered, but the texture of the environment is disorderly and unsystematic. We can realize the road area partition according to this character. The edge of the road is the focus noticed by human all along. In usual arithmetic, we first get the edge grads in the image, then follow the edges with big grades according to the grads direction and finally obtain the whole boundary. Because of the discontinuity of the boundary, the key is to obtain the description of the whole roadside by means of desultory edge combination which is usually realized by the fitting method, and the fitting methods include the whole line fitting, the subsection linear fitting and other sorts. Under strong illumination, the influence of shadow can not be ignored, and some times the edge of shadow is stronger than the road edge, so the elimination of shadow edge is a difficulty. Usually we can utilize the scrambling of shadow edge and the color character of shadow area to solve this problem. Because various characters have their own advantages and disadvantages, the adoption of various characters such as edge and area at the same time is a sort of tendency. In KRUSL4, it used the edge information, area information and road experience knowledge to make the road detection arithmetic more exact and reliable. To reduce the computation of the arithmetic, we usually use the Kalman filter to predict the possible position of the road in next frame of the image and reduce the treatment area of the image and quicken the speed of road detection.

2.2 Road detection based on models

Both the structured road and the non-structured road have relatively regular road mark or road edge, and the establishment of corresponding curve model according to the road edge form is another difficulty in the road detection. The simplest model is the line model. In the finitely long range, the roadsides are supposed to be two parallel lines. In the forward looking image, the roadsides are two radials intersected on the vanishing point. First, we find the line existing in the image though the Hough transformation method or other methods, then we abstract the parallel lines from the image. We can also implement the reverse perspective transformation, and transform the forward looking image to the quasi-planform, and then abstract the parallel lines from the image. The parabola and multinomial model is a sort of familiar arc road description. The parameter confirmation of this model is very important. To describe border road structure, some articles used the B-spline curve model, because the B-spline curve can form curves with random forms through a group of control points. First, confirm the initial position of B-spline rhough the vanishing point, and then detect the control point of B-spline model in the whole image by means of the minimum mean energy method. This model can effectively restrain the effects of road surface stain, shadow and uneven illumination, but the complexity of arithmetic is very high. Comparing with the road detection method based on character, the road detection based on model can effectively overcome some influences of environmental factors such as road surface stain, uneven illumination and shadow. But when the road fall short of the beforehand hypothesis, the model will be not effective, so the selection of the model is very important. To enhance the robustness of the road model, we usually use the present detection results to timely update the parameters of the road model, and make the road model more accord with the description of the actual road.

3. Tendencies of research and development

From above analysis, the road tracking based on machine vision has obtained large achievements. Most experiment vehicles can basically realize independent running in most time. The research emphasis of this domain has gone to the stage of system performance enhancement and stride forwards the practicality and production from the stage of exploring experiment. In the process of road detection, the road image collected suffers following factors such as the differences of construction time, construction mode and construction material of various road sections, changes of environment, the changes of the weather, the changes of illumination, and lane line deformity induced by water stain and smirch, which also induce certain difficulties for the real-time and exact road detection. At present, the road following with high reliability under all-weather and multiple situations will the aim to be further studied in the future. Most systems use the visible light vidicon at present, which can only work in the day, and can not work at night, but the nights is the frequent periods of traffic accident, so the intelligent vehicle navigation is most needed. The IR thermal imaging technology can be used to realize the vehicle navigation at night, which is a hot spot in the present researches. However, because its price is very expensive and the imaging quality cannot be satisfied, there is a long way to achieve the practicality. The following technology of the structured road has been basically matured, and the attention begins to turn to the complex urban traffic environment with low speed and backland wild environment, and try to realize the continual and independent work under multiple road situations. In the urban traffic environment, the traffic mark and lane mark detection is one new hot spot in the research of intelligent vehicle navigation. The reliability is the aim of the road following system, which is the key to realize the practicality and production of the system. To enhance the reliability, more and more systems begin to adopt the integrated technology including multiple sensors integration for hardware and multiple algorithms integration for software. Except for visual sensor, the laser range finder, millimeter wave radar and some initiative range equipments are used together to enhance the reliability of the detection result. For the algorithm, many sorts of detection algorithms are used at the same time to decide the final detection results by the methods of voting or blur ratiocination.

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The Implementation of CAN Bus Adapter Based on CH372

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Abstract

To implement quick and reliable communication between CAN bus network and computer, we propose a sort of design method of CAN bus network adapter based on CH372, and mainly introduce the concrete implementations of hardware circuit and software program in this article. This system adopts the modularization design, and a new USB interface chip CH372 is adopted in the USB communication module, which can simplify the design of software program. SJA1000 bus controller is adopted in the CAN bus module. The system can effectively implement the data transfer between CAN bus and computer with high speed.

Keywords: USB bus, CAN bus, CH372, SJA1000

1. Introduction

In the CAN bus measurement and control network, the network adapter assumes the important task of data transfer between monitor unit and lower computer. The design of traditional adapter is generally based on ISA bus and EISA bus, which has not fulfilled the need of high-speed data transfer nowadays. Though the network card based on PCI bus has quick transfer speed, but it still has many disadvantages and deficiencies such as complex agreement, system resource occupation, nonsupport hot swap and expensive price. Though RS232 has convenient swap, but its transfer speed is too low, and its maximum communication speed is only 20Kbps. USB bus supports hot swap, and it is easy to be extended, and it has quick speed, so it is the ideal choice to design CAN bus adapter. However, past USB interface chips always adopt PDIUSBD12 and USBN903, and their software designs were complex. In this design, we adopt new USB interface chip CH372 with interior integrated USB agreement, which could make program design and implementation easier, and users could put their energies on the design of application. The adapter designed in this article is actually applied in the product updating project of Tianjin Dontai S&T Development Co. Ltd. The practice proved that the design could enhance the communication speed, its installation and maintenance were convenient and it had extensive application future in the industrial control domain.

2. The implementation of adapter hardware circuit

2.1 Total structure of hardware circuit

The hardware circuit of the adapter designed in this article adopts the modularization design including main controller module, USB bus communication module and CAN bus communication module. The main controller module adopts AT89C51, and it mainly complete the initialization works of CH372 and SJA1000, and the communication tasks of USB bus and CAN bus. The total structure of CAN bus network adapter is seen in Figure 1.

2.2 Hardware design of USB communication module

The hardware circuit of USB bus communication module adopts new-style agreement chip CH372, which could not only simplify the design of hardware circuit, but reduce the design of software program. The SCM AT89C51 mainly completes the initialization work to CH37, responds the interrupt produced by CH372, and completes the data transfer with computer. The hardware circuit of USB bus communication module is seen in Figure 2.

CH372 is a full speed USB interface chip, and it is compatible with USBV2.0, and it supports plug and play, and its surrounding electron components only include crystals and capacitances.

CH372 has all-purpose local 8bits data bus and 4 lines control including read strobe, write strobe, chip selection input and interrupt output. The WR# and RD# of CH372 could respectively connect with WR# and RD# pins of SCM. CS# connects with P2.1 of SCM. The interrupt level outputted by INT# is that the low level is effective, and it is connected to INT0 of SCM. A0 is the address line input, and it differentiates order and data port, internally installs weak pull-up resistance, and when A0=1, it could write order, and when A0=1, it could read and write data. In the design, A0 connects with pin P2.0 of SCM. So the I/O addresses on the write order port of CH372 could be defined as 0xBD00, and the I/O addresses on the read and write data port could be defined as 0xBC00. When WR# is on high level and CS#, RD# and A0 are on low level, the data in CH372 output through D0-D7, and when RD# is on high level and CS#, WR# and A0 are on low level, the data are written in CH372 through D0-D7, and when RD# is on high level and CS# and WR# are on low level and A0 is on high level, the orders are written in CH372 through D0-D7.

CH372 internally installs power supply POR, so it generally needs not reset from the exterior. CH372 needs 12MHz clock signal supplied by the exterior when it works normally. The clock signals are produced by the inverter internally installed in CH372 through crystal stabilized oscillator. The surrounding circuit only needs to connect a crystal with standard frequency of 12MHz between XI and XO, and the pin XI and pin XO respectively connect with a high frequency oscillator capacitance. CH372 supports 5V power supply voltage or 3.3V power supply voltage. When it works on 5V voltage, VCC pin of CH372 inputs exterior 5V power supply, and the V3 pin should exteriorly connect with the decoupling capacitance of about o.1μF. When it works on 3V voltage, V3 pin of CH372 should connect with VCC pin, and input exterior 3.3V power supply, and the work voltage of other circuits connected with CH372 should not exceed 3.3V.

2.3 Hardware design of CAN communication module

The hardware circuit of CAN bus communication module adopts CAN bus control chip SJA1000, bus driver PCA82C250 and high-speed photocouplers 6N137. The SCM AT89C51 completes the initialization of SJA1000, and implements the transmission and acceptance of data through controlling SJA1000. The hardware circuit of CAN bus communication module is seen in Figure 3.

CAN controller adopt SJA1000 made by PHILIPS, and it could work on BasicCAN mode or PeliCAN mode. The maximum bit speed could achieve 1Mbit/s, and it supports Intel and Motorola microcontroller. SCM could interview SJA1000 through the mode to interview exterior memorizer.

WR#, RD# and ALE of SJA1000 respectively connect with WR# and RD# of SCM. The pin INT# of SJA1000 connects with INT1 of SCM. The patch selection port of SJA1000 connects with P2.3 of CSM, and the low level is effective, and its initial address is defined as 0xBB00. TX0 and RX0 of SJA1000 connect with CAN bus transceiver PCA82C250 through high-speed light-coupler insulation 6N137. To avoid the collision of over current, CANH and CANL of PCA82C250 respectively connect with CAN bus through 5Ω current limited resistance. CANH and CANL parallel connect two 30pF capacitances with earth, which could eliminate the high frequency interference and prevent certain electromagnetic radiation on the bus.

3. The implementation of adapter software program

3.1 Total structure of software circuit

The software program in the design mainly includes USB communication module software program, CAN communication module software program and upper computer software program.

3.2 Software design of USB communication module

CH372 internally integrates the bottom agreement in USB communication, and it possesses convenient internal firmware mode and flexible exterior firmware mode. Under the internal firmware mode, it shields relative USB agreements, and automatically completes standard USB enumeration collocation process, and largely simplifies the firmware programming of SCM without any disposal to local port controller.

Because CH372 doesn't deal with complex bottom communication agreement, so it is very simple to design the software program for USB communication module. The programs mainly include the initialization of CH372 and the interrupt disposal program. The initialization program of CH372 mainly includes USB work mode setting, USB firmware mode (interior firmware or exterior firmware) setting, and exterior custom USB VID and PID setting.

CH372 is specially used to deal with USB communication, and it informs SCM to deal with the data by the interrupt mode after it receives data or transmits data. The interrupt disposal program mainly includes the disposal to patch download success, the disposal to patch upload success, and the disposal to upload interrupt data success. The interrupt disposal flow is seen in Figure 4.

The interrupt program first acquires interrupt status and cancels interrupt request and judges the types of the interrupt, which mainly include three types such as patch ports download success, patch ports upload success and interrupt data transmission success, then makes out different disposals according to the interrupts with different types, and if the interrupt is induced by the patch ports download success, it also should make out disposal after judging whether the download is data or order.

3.3 Software design of CAN communication module

The functions of the physical layer and the link layer of CAN bus are completed by SJA1000, and the software program in the design mainly includes SJA1000 initialization subprogram, message transmission subprogram and message acceptance subprogram, and the disposals to data overflow interrupts and frame errors.

The initialization of SJA1000 could be implemented only under the reset mode, and the initialization mainly includes the setting of work mode, the setting of acceptance protection register AMR and acceptance code register ACR, the setting of baud rate parameter, and the setting of interrupt allowance register IER. The basic transmission process is that AT89C51 saves data to SJA1000 transmission buffer, and then resets the transmission require TR symbol of order register and begins to transmit. The acceptance process is that SJA1000 stores the data received from the bus to the acceptance buffer, inform AT89C51 to deal with the received information through the interrupt symbol bit, empties the buffer after acceptance and waits for receiving next time. The acceptance flow is seen in Figure 5.

3.4 Program design of the upper computer

For the program design of upper computer program, CH372 offers the interface of application layer. The interface of application layer is the API facing function application offered by CH372 DLL, and all APIs will return operation status after transfer, but answer data do not certainly exist, which could largely simplify the program design of the upper computer. APIs offered by CH372 DLL include equipment management API, data transfer API and interrupt disposal API. User program could transfer corresponding API function according to actual needs.

4. Conclusions

The CAN bus network adapter designed in the article combined the advantages of CAN and USB, and it extends the function application of USB in the industrial control. The system has many advantages such as good real time performance, high reliability and easy implementation, and it could implement the high-speed data communication between computers and CAN bus in the product testing of Tianjin Dontai S&T Development Co. Ltd, so it possesses good application foreground.

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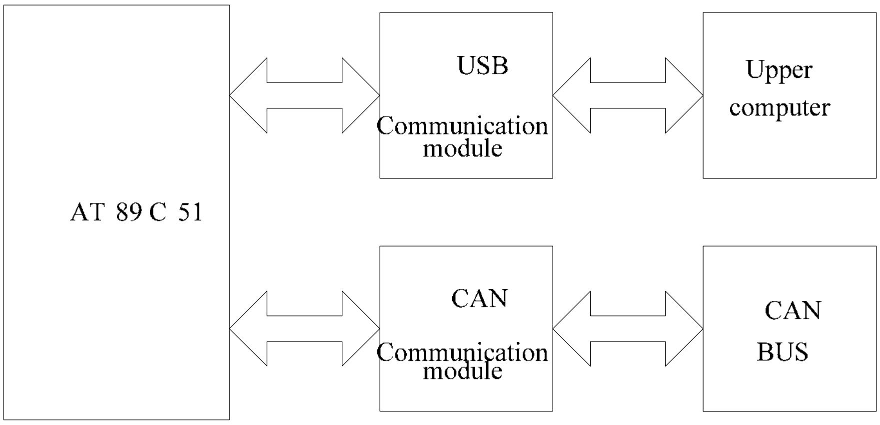

Figure 1. The Sketch of CAN Bus Network Adapter

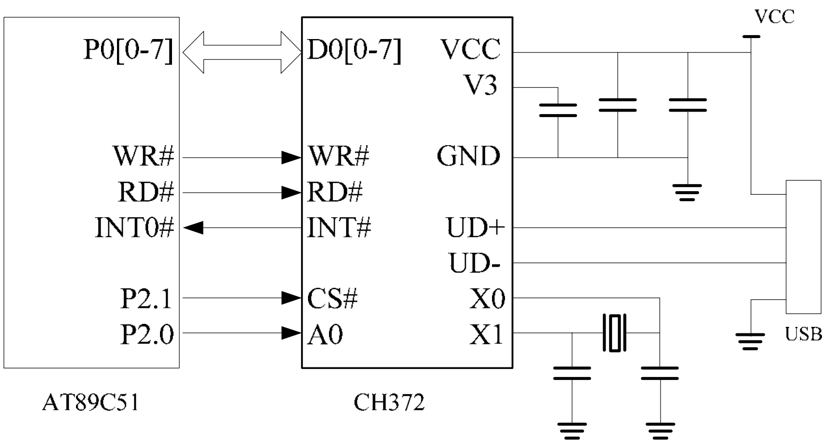

Figure 2. The Hardware Circuit of USB Bus Communication Module

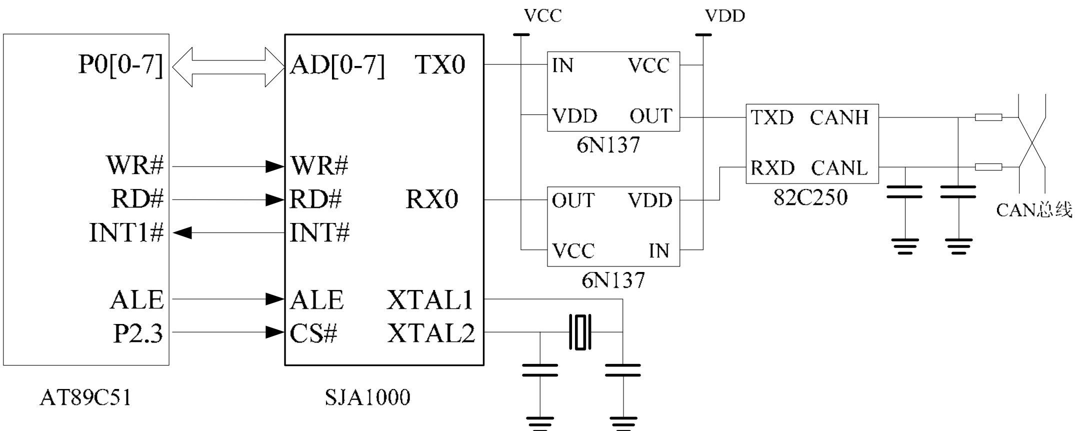

Figure 3. The Hardware Circuit of CAN Bus Communication Module

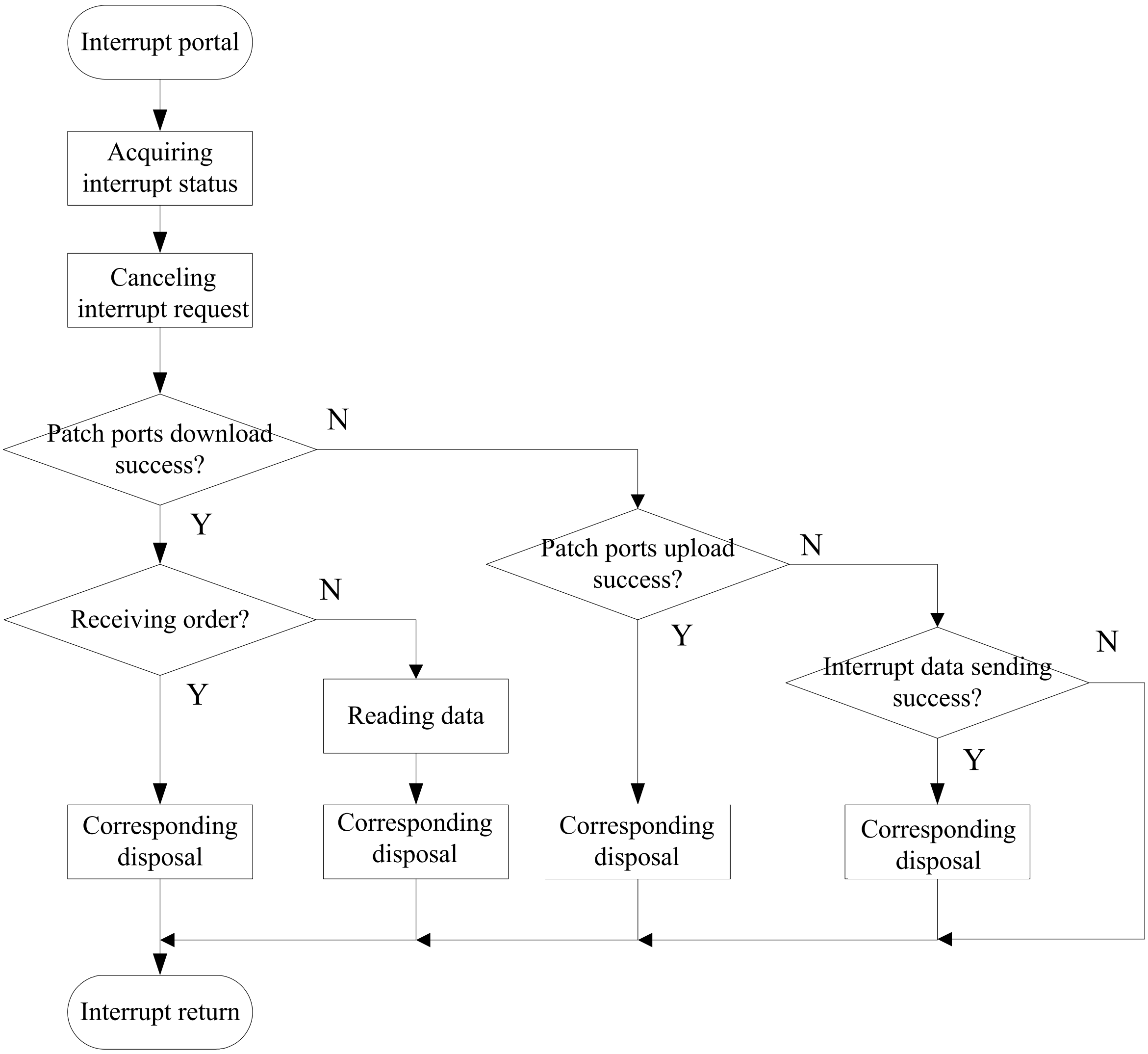

Figure 4. CH372 Interrupt Processing Flow

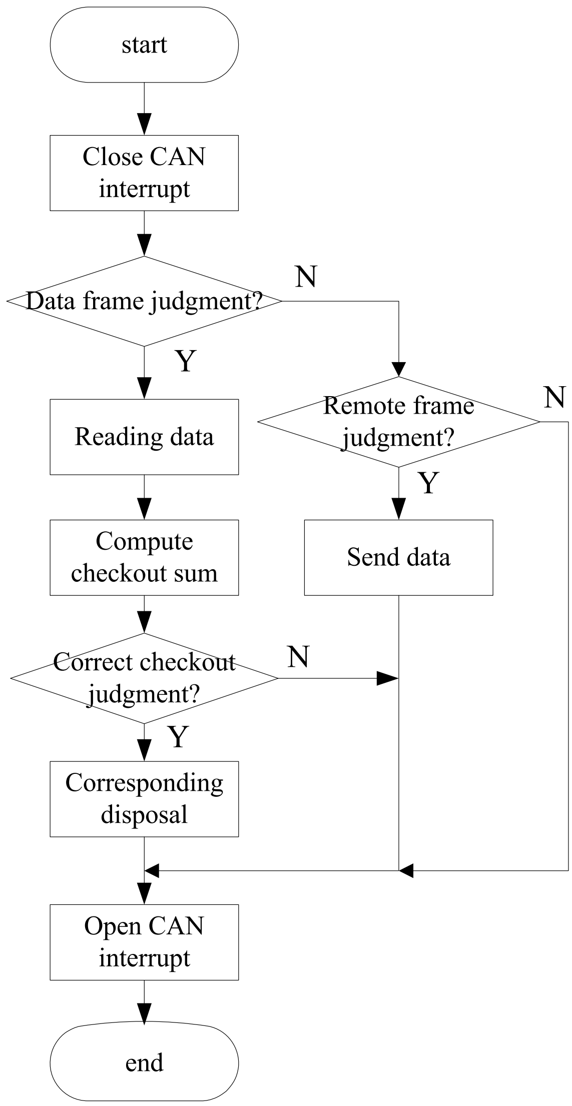

Figure 5. CAN Communication Module Acceptance Flow

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Study of Genetic Algorithms on Optimizing PI Parameters in Prime Mover Simulation System

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Abstract

This paper proposes using the genetic algorithms to optimize the PI regulator parameter in the prime mover simulation system. In this paper, we compared the step response characteristics under the conditions of the genetic algorithms and traditional method by MATLAB simulation and field test tested the dynamic characteristics of the prime mover simulation system. The results proved that genetic algorithms can optimize PI parameters quickly .With this method the prime mover simulation system can meet the requirements of dynamic performance simulation

Keywords: Prime mover simulation, PI regulate, Genetic Algorithms, MATLAB simulation, Dynamic characteristic

1. Introduction

The prime mover simulation system is one of the important equipment in power system dynamic simulation, essentially the current speed comprehensive regulator of this system is a PI regulator[1] [2](regulator for short, as shown in the dashed line frame of Fig.1) .A group of suitable PI parameter, i.e., ratio coefficient k_α and integral time constant τ_L[2] [3] [9] is required to provide the prime mover simulation system with good dynamic performance and realize self-balanced characteristic simulation of the prime mover [2].

According to the traditional trial and error process [7], the optimal values of ratio coefficient K_α and time constant τ_L depend on experience and repeated test; Simulating units of different capacity, PI parameters are required to redeploy with different rated output power of the prime mover to maintain the dynamic characteristics as before [9] [10].This is time-consuming, additionally, we must carry on complicated field test. Using genetic algorithm we can obtain optimal PI parameters quickly, which could be directly applied to field operation. Even if the rated output power of prime mover varied, we just need to input new corresponding parameters. Based on these parameters, computer calculates corresponding optimal PI parameters which ensure better dynamic characteristic simulation. This method is not only suitable for the off-line PI regulator parameters calculation, but also the on-line PI parameters calculation of computer prime mover simulation system. We will discuss the genetic algorithm and its application in the prime mover simulation system and simulation tests in the following section.

2. Principle of the genetic algorithms optimizing PI parameters

The genetic algorithm is a optimization algorithm simulating natural selection and evolution process and its basic theorem is: firstly, encode PI parameters and initialize population by certain size, thus, each individual in the population represents a possible solution. Then according to fitness function, compute fitness value of each individual, which can be used to control regeneration operation. Finally perform crossover and variation operation in a probability .Thus, the population continues evaluating till the end of the optimization process [5] [6].

2.1Parameter coding

Encode the parameters need optimizing. Solutions to the control problem are real number, which can be regarded as manifestation of genetic algorithms, therefore it is appropriate to adopt the binary coding. Since the problem is combinatorial optimization problem involves 2 parameters, we may first carry on the binary coding to obtain two sub-strings, then connect these sub-strings to former an integral chromosome, namely individual. K_α and τ_L are parameters satisfied with: $K_{\alpha min}<K_\alpha<K_{\alpha max}$, $\tau_{Lmin}<\tau_L<\tau_{Lmax}$. K_α and τ_L is respectively determined as length of the sub-strings, according to the precision request. Thereupon the coding precision of the 2 parameters, K_α and τ_L is: $\delta_{K\alpha}=\frac{K_{\alpha\max}-K_{\alpha\min}}{2^{L_{K\varepsilon}}-1}$, $\delta_{\tau_L}=\frac{\tau_{L\max}-\tau_{L\min}}{2^{L_{\tau_L}}-1}$. According to experiences, the parameter range of K_α is determined as (0, 10); the parameter rang of τ_L is (0, 1). In this paper we set the value of K_α as precise as 2 decimals. Because the length of interval of K_α is 10-0=10, the interval(0, 10) must be divided into $10*10^2$ equal parts. And furthermore, the coding binary strings of K_α at least need 10 bits, considering $2^9<10*10^2<2^{10}$. Similarly, if we set the value of τ_L as precise as 3 decimals , the coding binary strings at least need 10 bits.

2.2 Control parameters selection[5]

The procure of control parameters selection should ensure high search efficiency as well as the ability to find optimal solution. Generally, M, scale of the control parameter group takes 20~100, cross probability Pc takes 0.4~0.9 and variation probability P_m takes 0.0001~0.1. In this paper , M=50, Pc=0.5, P_m=0.01.

2.3 Initial population generation

Firstly, on the basis of experience, we could determine the possible value of the two parameters: K_α and τ_L, then generate a initial population nearby these two values. With this method, the search spaces reduce rapidly and we can also obtain optimal solution in a short time.

2.4 Determination of object function and calculation of fitness value[5]

The fitness function indicates: the ability of individual adapt to environment is related to the object function we choose. The PI parameter optimization is problem seeking for the minimum value of object function, namely minimize performance index J. Here, J adopt ITAE performance index (integration of the product of time and absolute error): $J=\int_0^\infty t\,|\,e(t)\,|\,dt$ .Since the goal of genetic algorithms is seeking for solutions of the fit max, the object function must be transformed from seeking for the maximum value into seeking for the minimum value. This paper defines fitness function $f=1/f$. Thus, when the fit max has been found, we gain the solution of object function, that is optimization success .We can compute fitness value f_n of each individual n in community M and the total fitness value $\sum_{n=1}^{M} fn$ of the whole community.

2.5 Genetic operation

2.5.1 Selection operation [6]

The core of selection operation is to determinate selection operator, whose function is to choose some quite fine individuals from the current generation of community, and replication them in the next generation. Fitness ratio method is introduced in this paper, namely, the probability of individual to be selected and be inherited to the next generation is proportional to its fitness value. Individuals of big selection probability have more descendants in next generation, others will extinct in the evolution process. Detailed operation process: first computed the total fitness of the community $\sum_{n=1}^{M} fn$, then computed ratio of each individual to be inherited to the next generation: $p_n=fn\Big/\sum_{n=1}^{M} fn$, n=1,2..., M; Finally, the times of each individual to be selected were determined according to the random number from 0 to 1.

2.5.2 Crossover operation

In this paper, we apply single-point cross method to crossover operation. Detailed process as follows: [M/2]=25 pairs of individuals are formed by pairing . For each pair, we randomly define the cross point behind a locus. Consequently, there are L-1=19 possible crossover point in all. Exchange partial chromosomes of the individual at the crossover point with defined cross probability P_c=0.5, thus generate two new individuals.

2.5.3 Variation

In this paper, the basic bit mutation method is introduced to carry on the variation operation .For individual coding string, we randomly define gene value at one or several locus with the variation probability P_m=0.01 to carry on the variation operation. Detailed process as follows: with the variation P_m=0 .01,define each individual locus as a variation point, then employ each gene value of the variation point to complementary operation, thus a new individual generated.

2.6 Generation new population

Evaluate the new population and compute the fitness value of it.

2.7 Judgment of evolution termination condition

We can obtain a new generation of population through replication, crossover and variation .We employ this new population to the fitness function after coded. If the new population satisfies the termination condition, we can get optimal solution, or else, return to the step (5) till the condition is satisfied.

3. MATLAB Simulation and field test

Using the m language of MATLAB to compile corresponding software[11]. Its flow is illustrated in Fig.2:

3.1 MATLAB simulation

Equivalently transform system diagram of Fig.1 to Fig.3. As shown at Fig.3,the transfer function of the controlled object of PI regulator can be expressed as: $G(s)=\frac{k_{scr}\beta/R_a}{(\tau s+1)(\tau_R s+1)(\tau_I s+1)}$

3.1.1 test parameters

Parameters of direct current motor: P_{N1}=38.5KW, U_N=220V,n_N=1000r/min, maximum armature current 150%I_N, its feedback voltage of the direct current motor is 1V,when flowing rated current. The voltage amplification coefficient of thruster rectifier bridge K_0=82.5, the equivalent time-constant of rectifier bridge τ=2ms, total inductance L=2mH, total resistance of Armature circuit R_a=0.055Ω. current feedback filter time constant T_I=1ms;

(1)Transfer function of controlled object by corresponding regulator can be written as: $G_1(s)=\frac{28.50}{0.0394s+1}$

(2): P_{N2}=22KW, other parameters are same as (1); $G_2(s)=\frac{14.9985}{0.0394s+1}$

By means of MATLAB/SIMULINK simulation experiment, we can observe the response waveform at U_o when inputting a step signal at U_i in chart 3 .

3.1.2Contents of experiment

Test 1: Traditional method. The corresponding step response wave form based on different load level (P_{N1}=38.5KW, P_{N2}=22KW) but the same PI parameters (K_a=0.10, τ_L=0.02s) are illustrated in Fig.4-a, b:

Test 2: Genetic algorithms, when different rated power, different PI parameters, considering the following cases (1)Ka=0.21, τ L=0.0364s,(2) Ka'=0.404, τ L'=0.0364s, the corresponding step response waveform is shown in Fig.5-a,b.

Test 3: Genetic algorithms, When different rated power but the same PI parameters (Ka=0.21, τ L=0.0364s), (1)The corresponding step response waveform is shown in Fig.5-a, Fig.5-c.

The results above are arranged in table 1.

*3.2 Dynamic performance tes***t**

The simulation prime mover has been successfully developed and put into field operation. The dynamic performance test was carried on 15KVA simulation generator units, single unit with rated load, unit inertia time constant HJ remained unchanged, taken 4 groups of different combined parameters of the simulation prime mover system, suddenly dropped of 100% load and recorded waveform. The experiment content and its results are illustrated in table 2; the recorded waveforms are shown in Fig.6. Eugene value in the table is approximate value based on the speed waveform, speed overshoot and speed steady change rate are calculated on basis of the speed digital readout. Concussion times are determined based on the degree of speed deviation from stable value. In this experiment, the speed increased and tended to be stable after load rejection, no lower than the steady-state value. Thus, the concussion time is 0.5.

According to the waveform graph and test data, the model parameters of speed control system (δ,TS, T0)had different effects on dynamic characteristic. The law of dynamic characteristic accorded with the practical prime mover system.

4. Conclusion

(1)As the rated power of the simulation prime mover varying, we must readjust PI parameters so that the system maintain the original dynamic characteristics; (Shown in Fig. 4- a, b and Fig.5-a, (c); By means of employing genetic algorithms, we can quickly optimize and adjust the PI parameters, meanwhile, maintain the original dynamic characteristics. Furthermore it could effectively save debugging time and simplify debugging process.

(2)Application of genetic algorithms to determine PI parameters could provide the regulator good dynamic response characteristics (Fig.4- a and Fig.5-a);

(3)The law of dynamic characteristics of the prime mover simulation system accorded with fact, and met the requirements of dynamic simulation test of power system. (Fig.6 and Table.2)

At present, we take off-line calculation as applying genetic algorithm to determine PI parameters. It remains further study and research to realize on-line modification and optimization of PI parameters in the prime mover simulation system controlled by microcomputer.

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Table 1. Dynamic performance of step response simulation test

		overshoot	rise time	waveform	dynamic performance
Same rated power	traditional method	8.5%	1.04	Fig.4-a	poor
	genetic algorithms	0.1%	1.02	Fig.5-a	good
Same PI parameters traditional method	P_{N1}=38.5KW	8.5%	1.04	Fig.4-a	poor
	P_{N2}=22KW	7.6%	1.05	Fig.4-b	poor
same PI parameters Genetic algorithms	P_{N1}=38.5KW	0.1%	1.02	Fig.5-a	good
	P_{N2}=22KW	8.9%	1.07	Fig.5-c	poor
different PI parameters Genetic algorithms	P_{N1}=38.5KW	0.1%	1.02	Fig.5-a	good
	P_{N2}=22KW	0.1%	1.03	Fig.5-b	good

Table 2. Major eigenvalue of system (speed) process with different model parameters

Figure number	δ	TS	T0	Transient time	Over shoot	maximum amplitude time	concussion times	stability
f	4.7	0.2	0.4	5.0	2.0	1.7	1.0	Ibid.
h	1.5	0.2	0.1	/	/	/	/	shock
e	2.6	0.2	0.3	4.0	2.4	1.2	0.5	Ibid.
g	2.6	0.2	0.4	6.4	3.0	1.5	1.0	Ibid.

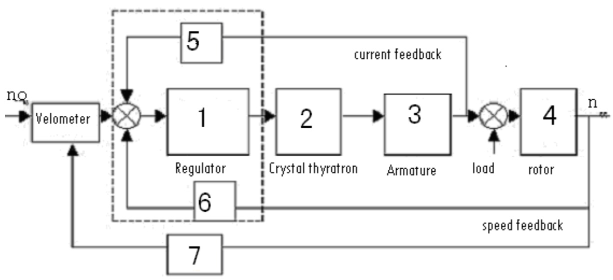


Figure 1. Principle of prime mover simulation system[2]

In Figure 1: 1—$\frac{(\tau_L s+1)k_\partial}{\tau_L s}$; 2—$\frac{k_{scr}}{\tau s+1}$; 3—$\frac{1/R}{\tau_R s+1}$; 4—$\frac{C_m}{J_n S}$; 5—$K_i = \beta/(1+T_i S)$; current feedback; 6—$k_n$ speed feedback coefficient;7—k_n' speed feedback coefficient .where: k_α—proportionality coefficient of regulator; τ_L—integration time constant; τ_r, k_{sc}—the equivalent time constant of rectifier bridge and times of voltage amplification; τ_R ,R—time constant of the armature circuit and equivalent resistance; C_m—rotor coefficient; J_n—rotational inertia; S—differential operator; β—current feedback coefficient; Ti—current feedback filter time constant.

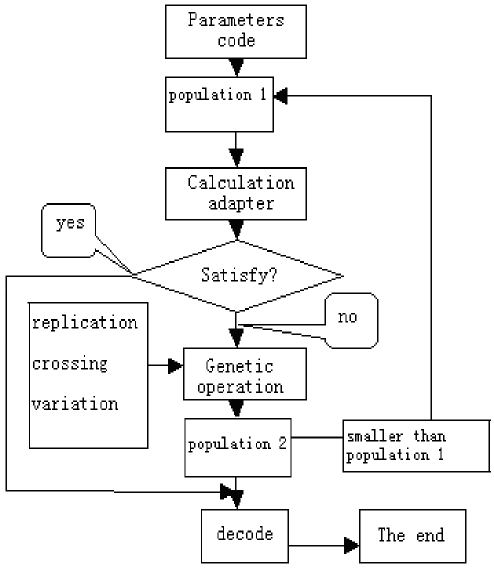


Figure 2. Genetic algorithms flow chart

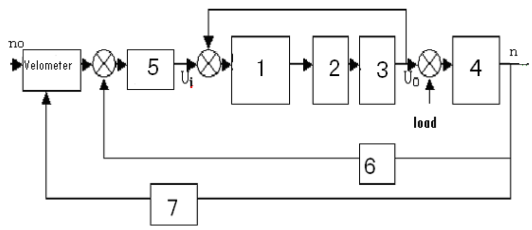
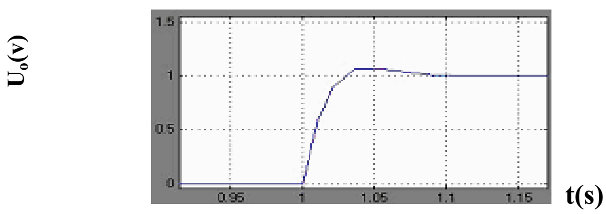
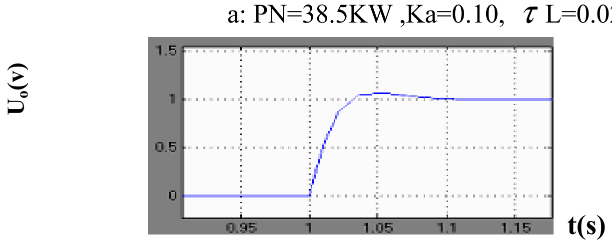


Figure 3. Prime mover simulation system equivalent diagram

In Figure 3:1—$\frac{(\tau_L s+1)k_\partial}{\tau_L s}$;2—$\frac{k_{scr}}{\tau s+1}$; 3—$\frac{k_i/R}{\tau_R s+1}$; 4—$\frac{C_m}{J_n S}$;5—$1/k_i$;6—k_n ;7—k_n';

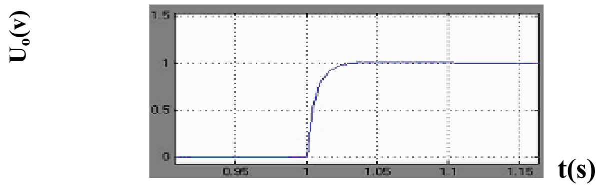


a: PN=38.5KW ,Ka=0.10, τ L=0.02s

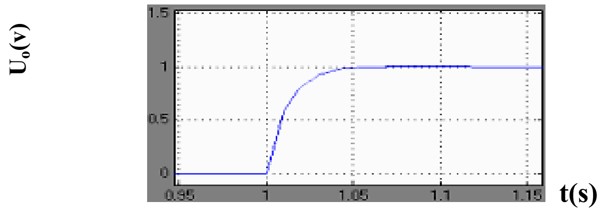


b: PN=22KW, Ka=0.10, τ L=0.02s

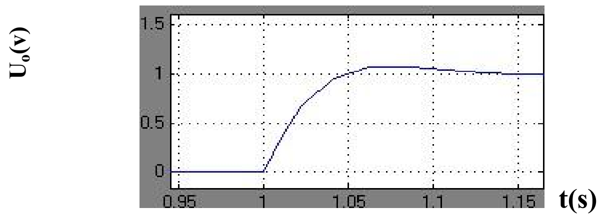
Figure 4. traditional method different rated power same PI parameters step response curve



a: PN=38.5KW, Ka=0.21, τ L=0.0364s



b: PN=22KW, Ka'=0.404, τ L'=0.0364s



c: PN=22KW,Ka=0.21, τ L=0.0364s

Figure 5-a, b Genetic algorithms different rated power different PI parameters step response curve

Figure 5-a, c Genetic algorithms different rated power same PI parameters step response curve

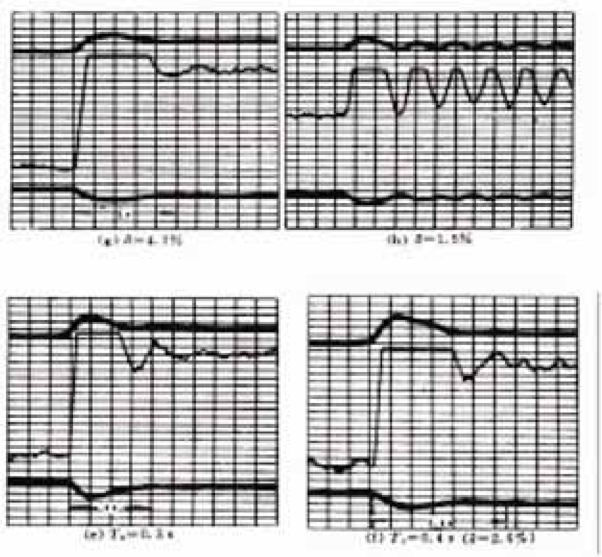


Figure 6. Wave-record chart of dynamic process of rejecting 100% load

In Figure 6. from top to bottom ,curves ordered: η speed deviation of the prime mover simulation speed control system, oil engine explanted displacement μ and speed n, where δ-difference coefficient ,T_s - oil engine time constant, T_0 -steam inertia time constant

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Panoramic Images Automatically Stitching Algorithm Introduction

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Abstract

The panorama is more effective image-based rendering one of the methods. Papers on the panorama generation process, and key technologies on one of the most panoramic picture of the stitching algorithm is classified, summarize and compare.

Keywords: Panoramic, Stitching, Mosaic

1. Introduction

Panorama is a kind of full-horizon and high-resolution images, in robotics, computer vision, virtual reality, military and commercial on a wide range of applications. Web-based Panorama in the real estate and tourism are widely Application, the user can interact homes to enjoy the exotic customs. Use the full range of imaging sensors and cameras compatible with the composition of panoramic cameras, can be a one-time access to a wide range of images - a panoramic picture, but the method of hardware devices require a higher, more expensive, and the need to better shooting skills, to be universal. Image Mosaic approach will be adopted by ordinary camera with the image sequence automatically splicing, is the structural panorama one of the most commonly used method in order to make ordinary photo mosaic to Panorama, to maintain perspective on the visual effects and the consistency necessary to Images that re-projection to a surface. That is commonly used spherical surface, the cube and cylindrical projection, cylindrical projection of the most widely used. Cylindrical panorama in the direction of the level of 360 ° to meet the look, it has the advantage of cylindrical image can be launched into a Plane, greatly simplifies the cylindrical projection and display images quickly.

2. Panoramic image generation process

2.1 Panoramic image

Based on the graphic image mapping technology is a new graphic image technology. IBR existing technology probably can be divided into four categories: Based on the panoramic view of the method, based on Morphing methods, image-based methods of the depth of information based on formation the way light field. At this stage, a relatively mature technology is the first panoramic view on the way. In fact, the panoramic view can be simply interpreted as: the perspective used in a fixed camera or video camera in accordance with a certain way (in accordance with the uniform point of view is usually around 360 rotating Degrees) image acquisition, image acquisition after the importation of computer image mosaic, integration processing, generation seamless panoramic images, after the final re-use computer display projector out and provided a partial limited roaming capabilities. Although the panoramic view has its own limitations, such as a single viewpoint, can only be realized within the scenes, such as robots, but because the technology is extremely workable, but also relatively mature, has become the most widespread application of the IBR technology 1. Panoramic view of the current application in the main: virtual environment, game design, film special effects, virtual museums, etc. In commercial areas relatively well-known Apple in the early 1990s, launched QuickTime VR system.

2.2 The general panorama generation process

The general panorama generation process including panoramic model of choice, image acquisition, image mosaic, image and suture Panorama show here five steps

2.2.1 Panorama model of choice: According to Panorama projection display of different ways, can be divided into three main modes: cube model, cylinder model, spherical model. This is the three models were the stitching has been good panorama projection to the cube / Cylinder / sphere of the inner surface. There are other display modes, such as a polyhedron is spherical approximation to the method.

2.2.2 Image Acquisition: There are two methods used to shoot panoramic photography equipment or by an ordinary camera image mosaic again. Ago a way to relatively easy to collect images, but this approach often means that the purchase of expensive photographic equipment, Impact of its generality. Latter, with ordinary cameras in fixed-point shooting pictures and then splicing generate panorama of it is more active, and the panorama generated by the core technology - image mosaic algorithm is the focus of the study.

2.2.3 Image mosaic: the existing panoramic image mosaic generation algorithm can mainly be divided into three categories: Based on the current method, based on the characteristics of the method and based on the phase approach.

2.2.4 Suture image: a good image by stitching, the images also need to deal with some overlap, in order to achieve a seamless mosaic image. Now often used in a simple image suture technology is linear interpolation.

2.2.5 Panorama show: get a 360-degree panoramic image, but also to the image projection to the model chosen by the inner surface impressions, and to provide simple browsing.

3. Panoramic image mosaic of technology

3.1 panorama of the classic generation algorithm

Panorama from the concept to the present, there are many scholars who have the panorama of the Algorithm done in-depth study, McMillan and Szeliski, who is particularly prominent. McMillan and Bishop made a panorama function model, the algorithm based on the camera around 360-degree rotating taken by the camera image sequence for each parameter, which conducted panorama stitching, but the algorithm is not suitable for cameras, because it requires between the two images have more than 2 / 3 of overlap, so If a scene with the camera to collect images on the need to take a lot of pictures, which apparently has increased the difficulty of image acquisition, but also increase the amount of computation error and the amount. McMillan and through the cylinder is trying to achieve geometric constraints automatically match point, but the very core of geometric constraints on the basis of error of over-sensitive matrix, and therefore lead to practical results do not match. Szeliski and Shum in their series of articles in the proposed use 8 parameters of the two-dimensional projection model for projection matrix M:(figure 1-2)

Set up two match points overlapping images in their respective coordinate system in the homogeneous coordinates for X (x, y, 1) and X '(x', y ', 1'), Formula One (1) has given them Spatial relationship between the mathematical expression. Assumption that X 'and X' (the brightness of a point (or color), respectively I '(x', y ') and I (x, y), then two overlapping regional counterparts Points between the brightness (or color) and the square for the poor:(figure 3)

To minimize, is actually for a non-linear least-squares problems, with L-M can be solved projection matrix M, but the method of calculation and large there are limitations. Need a good initial value, otherwise Solving the time is likely to get the wrong match. Szeliski and Shum improved the 8-parameter model for the 3-parameter model, so although to a certain extent, reduce the volume of operations, but this improvement is within the parameters of the camera correction obtained at the cost of, And the calculation method is still greater.

3.2 panoramic image mosaic algorithm classified and compared

The existing panoramic image mosaic generation algorithm can mainly be divided into three categories: Based on flow method, based on the characteristics of the method and based on the phase approach.

(1)Based on the current approach: the area also called on the way the method is by comparing the two images brightness (or color) differences, and the minimum to find the best match point. Description of the above two classic Algorithm that is part of the way, this approach were: Duffin and Barrett in Szeliski on the basis of the restoration of a nine-parameter model of the algorithm. Method based on the area of the shortcomings is that it's obviously too large amount of computation.

(2)Based on the characteristics of the method: This method is the main ideas from one image to extract certain characteristics, such as: points, lines, and so on the edge and use characteristics of this template for the match in the second image in the search. The Ways to increase computing speed, but the right image feature extraction more difficult. How to extract the appropriate image features, many scholars have done in this study, Kim and others used as the outline of objects taken from features, and the bell of overlapping images, and other people to use the brightness of a two (color) or brightness than the (color) Poor characteristics as a template, Zhang Peng, who value the use of a gray-scale information extraction characteristics of a rectangular region as a template to match. Need to pay attention to another issue that is collected in the overlapping part of the image is not a simple plane displacement transformation, and there are still Stretching deformation, and so on, so it also increases the feature extraction more difficult.

(3)Based on phase approach: the use of Fourier Transform, the first wavelet transform, and so transform the image and then use transform the image of some of the features match. But if there is space on the local changes will lead to The method have a greater error.

4. Improved algorithms

The algorithm proposed in [4] can be summarized as the follows.

Step 1: Camera orientation estimation. Using an internally precalibrated camera, the extrinsic orientation parameters of camera can be determined from an aerial instrumentation system (GPS/INS) and bundle adjustment techniques.

Step 2: Image rectification. An image rotation transformation is applied to each frame in order to eliminate the rotational components.

Step 3: Slice determination. Determine the fixed lines in the current frame *k* and the previous frame *k*-1 by the left slit window distance *dy*/2, and "ideal" straight stitching lines by their 2D scaled translational parameters. The locations of the stitching lines are in the middle of the two fixed lines. Therefore, two overlapping slices in the *k* and (*k*-1) frames are obtained. And each of them starts from the fixed line and ends a small distance away from the stitching lines in opposite directions.

Step 4: To stitch the left stereo mosaics and the right stereo mosaics respectively by match and ray interpolation, triangulation and warping mentioned.

In order to satisfy the increasing demands for the remote sensing applications, the UAVRSS was integrated by UAV platform, GPS/INS, RS sensors (including Charge Coupled Devices (CCD), CCD TV camera and Synthetic Aperture Radar (SAR)), mobile ground station, and the RS data processing center [3]. The UAV used in RS is designed according to the manned aircraft standards and RS mission initially. The CCD TV camera is mounted on the UAV to acquire the image sequence, which is transmitted to the ground receiver in real-time. We need to create the stereo mosaics in real-time to satisfy the applications. Therefore, according to the speciality of the UAVRSS, the following modified algorithm is proposed based on the parallel perspective stereo mosaics.

Step 1: Camera orientation estimation. The CCD TV camera is precalibrated in laboratory to get its intrinsic parameters. The extrinsic orientation parameters in each camera point can be determined from an aerial instrumentation system (GPS/INS) in theory. However, it is difficult and time-consuming to estimate the extrinsicorientation parameters for all images. Therefore, an interpolation approach is proposed to estimate the extrinsic orientation parameters for each image.

Step 2: Image rectification. In [4] an image rotation transformation is applied to each frame in order to eliminate the rotational components. The result sequence will be a rectified image sequence as if it was captured by a "virtual" camera undergoing 3D translation (*Tx*, *Ty*, *Tz*). However, the sequence after that rectification still remains the height deviation. In order to simplify the latter mosaics, we introduce the rectification to compensate the aircraft flying height deviation. Therefore, after eliminating the rotation and height deviation, the camera's motion can be seem as 2D translation. This will make the latter mosaics easier and faster.

Step 3: Slice determination. This step adopts the same method in [4]. And in order to get different oblique viewing directions, we suggest that to choose different *dy* to produce several pairs of stereo mosaics. So, it is possible to view the same area in different viewing directions.

Step 4: To stitch the left stereo mosaics and the right stereo mosaics respectively.

5. Conclusions

Image Stitching is an increasingly popular field of study, it could provide Realistic Panorama, is to create virtual reality scenes and virtual roaming the foundation. In this paper, the algorithm used to independently choose the baseline characteristics of the block thinking, and the image of Precise stitching, a pyramid-layered match the idea and then making the Image Stitching fast. Therefore, this algorithm is both good and robust has good practicality.

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Szeliski R. Video Mosaics for Virtual Environments. IEEE Computer Graphics and Applications (C). IEEE, 1996, 16(2): 22-30

$$\boldsymbol{X}' \sim \begin{bmatrix} u \\ v \\ w \end{bmatrix} = \boldsymbol{MX} = \begin{bmatrix} m_1 & m_2 & m_3 \\ m_4 & m_5 & m_6 \\ m_7 & m_8 & m_0 \end{bmatrix} \begin{bmatrix} x \\ y \\ 1 \end{bmatrix}$$

Figure 1

$$\boldsymbol{X}' = \begin{bmatrix} x' \\ y' \\ 1 \end{bmatrix} = \begin{bmatrix} \frac{u}{w} \\ \frac{v}{w} \\ 1 \end{bmatrix}$$

Figure 2

$$E = \sum [I'(x', y') - I(x, y)]^2 = \sum e^2$$

Figure 3

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Web Access Pattern Algorithms in Education Domain

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Abstract

Sequential pattern mining discovers frequent user access patterns from web logs. Apriori-like sequential pattern mining techniques requires expensive multiple scans of database. So, now days, WAP (Web Access Pattern) tree based algorithm is used. It is faster than traditional techniques. However, the use of conditional search strategies in WAP-tree based mining algorithms requires re-construction of large numbers of intermediate conditional WAP-trees, which is also very costly.

In this paper, Kongu Arts and Science College (KASC) web logs are taken for mining. Here, we propose an efficient sequential pattern mining techniques for KASC web log access sequences known as CS-WAP Tree. This proposed algorithm modifies the WAP tree approach for improving efficiency. The proposed algorithm totally eliminates the need to engage in numerous reconstructions of intermediate WAP trees and considerable reduces execution time. The results of experiments show the efficiency of the improved algorithm. The next key aim is to compare WAP algorithms.

Keywords: Web log file, Data mining, Web Usage Mining, Web Access Pattern

1. Introduction

Web usage mining [2] is used to identify user behavior on a particular website. It performs mining on web usage data or web logs. The mined knowledge can then be used in many practical applications [4], such as improving the design of web sites, analyzing user behaviors for personalized services, and developing adaptive web sites according to different usage scenarios.

The access and usage of information is the major functionality of the broad diversity of WWW user community. As a result, discovery of useful information from Web content is not the only important task of Web mining. Resource finding, information selection and pre-processing, generalization and analysis are sub-tasks of Web mining. Web mining is a broad domain which covers fields like database, information retrieval and artificial intelligence, machine learning, natural language processing, network analysis and information integration.

A sequential access pattern [3] is a sequential pattern in a large set of pieces of web logs, which is pursued frequently by users. Most of the previous studies for discovering sequential patterns are mainly based on the Apriori algorithm [1]. However, these algorithms encounter the same problem as most Apriori-based algorithms.

The WAP-mine algorithm [5] has been developed for mining sequential access patterns from the WAP-tree. This approach avoids the problem of generating an explosive number of candidates as encountered in Apriori-based algorithms. The WAP-mine algorithm is in general an order of magnitude faster than traditional sequential pattern mining techniques. In this paper, we propose an efficient sequential access pattern mining algorithm known as Constrained Sequence WAP mining algorithm (CS-WAP).

2. Background Study

As an important branch of data mining, sequential pattern mining, which finds high-frequency patterns with respect to time or other patterns, was first introduced by (R. Agrawal, & R. Srikant (1994)) as follows: given a sequence database where each sequence is a list of transactions ordered by transaction time and each transaction consists of a set of items, find all sequential patterns with a user specified minimum support, where the support is the number of data sequences that contain the pattern. Since the access patterns from web log take on obvious time sequence characteristic, it is natural to apply the technology of sequential pattern mining to web mining. According to the downward closure property of frequent sequences, to some extent, maximal frequent sequences have already included all frequent sequences. The space to store maximum frequent sequences is much lower than to store complete set, and web mining applications partly only depend on maximum frequent sequences rather than the complete set of frequent sequences so that mining maximum frequent access sequences is of essential practicability.

Sequential access pattern mining techniques are mainly based on two approaches: Apriori-based mining algorithms and WAP tree based mining algorithms.

2.1 Apriori-based Mining Algorithms

The AprioriAll (J. Pei, J. Han, B. Mortazavi-asl & H. Zhu (2000)) algorithm proposed a three-step approach for mining sequential patterns. It first finds all frequent itemsets. Then, it transforms the database such that each original transaction is replaced by the set of all frequent itemsets contained in the transaction. And finally, it finds the sequential patterns. However, this algorithm does not scale well due to the costly transformation step. In (J. Pei, J. Han, B. Mortazavi-asl & H. Zhu (2000)), a generalized sequential pattern mining algorithm known as GSP mining algorithm was proposed. Similar to the AprioriAll algorithm, GSP scans the database several times. In the first scan, it finds all frequent items and forms a set of frequent sequences of length one. In subsequent scans, it generates candidate sequences from a set of frequent sequences obtained from the pervious scan and checks their supports. The process terminates when no candidate is found to be frequent. So this algorithm requires multiple scans of database. So now we discuss WAP tree based mining algorithm.

2.2 WAP Mining Algorithm

The WAP-tree is a very effective compressed data structure designed for storing the data obtained from web logs. To construct a WAP-tree, we need two scans of the web access sequence database: (1) Scan database once, find all frequent individual events; (2) Scan database again, construct the WAP-tree over the sub-sequences with only frequent individual events of each sequence, which are also called frequent subsequences, by merging their common prefixes,. At the same time, all nodes that contain the same frequent event are linked into an event queue and the Header Table with all frequent events is created for this WAP-tree with the head of each event queue registered in it. Then, all the nodes labeled with the same event can be visited by following the related event queue, starting from the Header Table.

The FS-tree extends the WAP-tree structure for incremental and interactive mining. The corresponding mining algorithm FS-mine (Frequent Sequence mining) is used to analyze the FS-tree to discover frequent sequences.

J Han puts forward a web access pattern tree structure (WAP-tree) and an algorithm for mining frequent access path based on WAP-tree (WAP-mine) in (J. Pei, J. Han, B. Mortazavi-asl & H. Zhu (2000)). This algorithm and not producing candidate frequent patterns. Consequently, WAP-mine algorithm is an order of magnitude faster than Apriori algorithm (B.Y. Zhou, S.C. Hui, & A.C.M. Fong (2004)) put forward by Agrawal at earlier stage. Nevertheless, WAP-mine needs to produce a mass of conditional WAP-tree, which influences the efficiency of WAP-mine in a certain degree. In recent years, some classical algorithms applied to mine maximum patterns include MaxMiner, DethProject, MAFIA and GenMax etc.

3. Prototype Description - CS – WAP

The CS – WAP mine (Constrained Sequence Web Access Pattern mining algorithm) enhances our WAP-tree based algorithm (J.Srivastava, R.Cooley, M. Deshpande, & P.-N. Tan (2000)) by employing directly the *conditional sequence base* of each frequent event, **without the need of constructing any WAP trees**. Although, the WAP-tree is a highly compressed data structure for storing sequence data, we need recover the uncompressed sequences during mining process in practice. That is the reason why our new CS-WAP mine algorithm is not based on the WAP-tree as our previous CS-mine algorithm. The proposed CS-WAP mine algorithm can improve quite significantly on the efficiency when compared with the WAP-mine algorithm, especially when the support threshold becomes smaller and the size of database gets larger.

The Prototype Algorithm

The prototype algorithm for mining sequential access patterns using KASC web log is shown in Figure 1.

Input: Web access sequence database, Minimum Support Threshold

Output: Set of sequential access patterns

Method:

Step 1 : Construct event queues for *CSB*(*Sc*).

Step 2 : Test single sequence for *CSB*(*Sc*).

(i) If test is successful, insert all ordered combinations of items in frequent sequence *FS* = *Sc*+*SingleSeq* into *SAP*.

(ii) Otherwise, for each *ej* in Header Table of *CSB*(*Sc*), use **SubQueue** to construct *CSB*(*Sc*+*ej*). Set *Sc* = *Sc*+*ej* and recursively mine *CSB*(*Sc*) from step 3.

Step 3 : Return *SAP*.

Figure 1. The CS-WAP algorithm

4. Experimental Results

The proposed algorithm is implemented in VB.NET and all experiments were found on Intel Pentium running on Microsoft Window XP profession. The web server log file dated on Sep 2007 from KASC web server after preprocessed (Gomathi.C., Moorthi M. & Duraiswamy K. (2008)) has been selected for our experiments. This KASC log file size is 148KB. The proposed method was applied on this preprocessed web log files to prepare sequence pattern. The proposed mining algorithm has significant advantages when compared with the original WAP-mine algorithm.

From table-1 and Figure 2, It can be seen that the execution time increase as size of the database increase. As the size of the data decrease, the execution times also decrease. So the CS - WAP algorithm always use less run time than other WAP algorithms. Also the table shows that WAP tree algorithm requires more memory than proposed algorithm. It reduces storage cost. The experiments showed that the proposed methodology needs less time to find frequent sequence and needs only minimum storage area.

5. Conclusion

In this paper, CS-WAP mine tool developed using VB.NET for sequential access pattern from KASC web log files. The proposed system eliminates the need to store numerous intermediate WAP trees during mining. Since only the original tree is stored, it drastically cuts off huge memory access costs. This system also eliminates the need to store and scan intermediate conditional pattern bases for reconstructing intermediate WAP trees. This algorithm uses the pre-order linking of header nodes to store all events *ei* in the same suffix tree closely together in the linkage, making the search process more efficient.

"The proposed algorithm analyzes the student's behaviors. Based on the behaviors, the web site is restructured to get knowledge immediately. So, student's society updates their knowledge, face challenge and find the solutions for it".

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Table 1. Execution times trend with different data sizes with fixed minimum support 7%

	Different Changed transaction size (navigational path)		
Algorithms time in second	63K	126K	189K
WAP	360	600	780
Efficient-WAP	300	420	480
CS-WAP	60	180	240

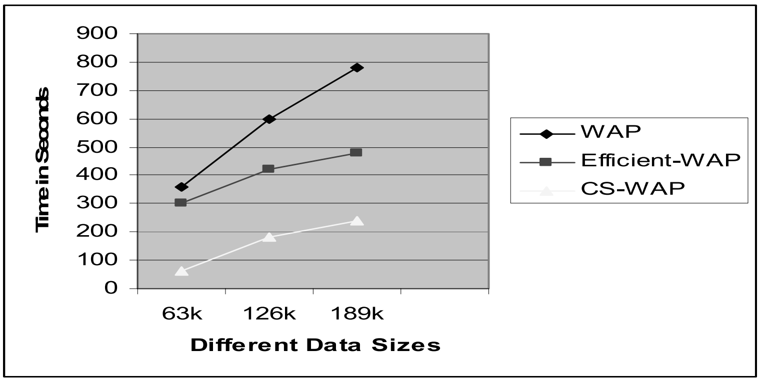

Figure 2. Execution time s trends with different data sizes.

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