SESSION

EVALUATION

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TBA

Evaluating an instant messaging protocol for digital whiteboard applications

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***Abstract*— Many whiteboard applications have been developed during the recent years. In distributed settings, where whiteboards have to be synchronized, proprietary protocols are often used for communication. Using reliable and well-known open standards can ease integration with existing systems. After evaluating existing instant messaging protocols, we look at applying those well-established standards for the problem domain of distributed whiteboards. The presented system does not only support synchronous working modes, but also asynchronous, by consequently capturing and analyzing communication data. The presented approach goes beyond existing systems in unifying those two working modes into one system. Evaluation of this in-situ capturing shows only little influence on the system's performance and therefore being an approach generally applicable to other problem domains.**

***Index Terms*— digital whiteboard, XMPP, instant messaging protocol, performance evaluation**

I. Introduction

Remote collaboration tools are well-established in today's companies. They support the communication over distances using audio/video conferencing, messaging, whiteboards, and desktop sharing. Often, this collaboration is limited to information exchange and communication can only hardly be retraced. Together with problems such as time shift over different continents, synchronous working modes are quite limited. Due to this limitations, regular face-to-face meetings are often the method of choice for successful collaboration. The result is less joint work and more reporting and documentation via e-mails and other documents about the efforts at each location. As we learned from interviews with employees of global companies, documentation is time-consuming and frustrating for all involved parties and often only done because project management demands it.

Besides these difficulties, e-mails and text documents often are not suitable to convey ideas and concepts that people have during their project work. It is very hard to communicate and understand why people took certain decisions and which were their most important concerns. Especially with creative work, which involves a lot of unforeseen ways of working, thinking about ideas or innovations and visualizations of concepts and designs, it is very difficult to write down the results of a meeting. Teams who are applying methods such as Design Thinking [1] often work with whiteboards, sketches and sticky notes and only in the end they "translate" their work to text documents or presentations.

A preferred tool for co-located work is the whiteboard. People standing in front of it can form a common ground of understanding by sketching their ideas, writing down notes, brainstorming on problem solutions, or simply working on shared todo-lists. A major benefit of a traditional whiteboard is its extremely user-centered design adopting input modes that are being used for thousands of years. Even children can easily work with this tool due to the simplicity of the used tools - pens, eraser and sticky notes. That is supposably one of the reasons for the large adoption of whiteboards in companies. On the other hand, whiteboards are still only rarely used in distributed settings. Some remote collaboration tools such as Adobe Connect support whiteboard sharing, but people refuse to use it. On a traditional whiteboard, users have to actively erase all the content from a board, whereas in known electronic whiteboard tools, whiteboards are often automatically erased, when people leave the communication channel. Moreover, the direct link between whiteboard surface and video conference is not not given, so that it is impossible to point on certain sketches on the whiteboard.

This leads to the main question of our research: How can we support people in their co-located as well as their distributed work using digital whiteboards? Can we bridge the (time)gap between continents and support collaboration more efficiently using digital communication channels?

Many whiteboard applications have been developed during the recent years. For distributed settings, where whiteboards have to be synchronized, proprietary protocols are often used for communication. Our approach uses an open standard for whiteboard data transmission, in order to benefit from existing tools as well as being able to easily integrate our system into existing solutions and reuse the generated data.

In the following, we elaborate on existing tools for whiteboard interaction as well as protocols that are being used in the domain of instant messaging. Our concrete communication protocol is introduced in order to understand the system architecture of the implemented system Tele-Board. Afterwards, we evaluate the appropriateness of an instant messaging protocol for digital whiteboards and show how the support for asynchronicity effects the system's performance.

II. Related work

Computer supported collaborative work (CSCW) is a field of research which brought up many different research projects and products on the market supporting distributed work. A typical schema to differentiate between those different solutions is the CSCW matrix shown in Fig. 1 (cf. [2], [3], [4]). The cells of this grid cannot be clearly separated, however many projects have a strong focus on one of them.

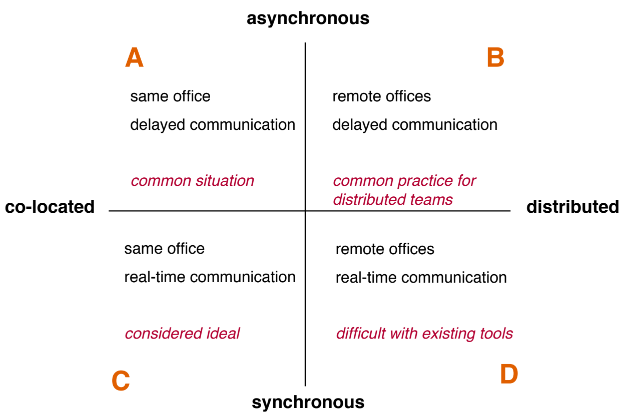


Fig. 1. Categorization matrix of working modes in CSCW

Older approaches such as *ClearBoard* [5] or *VideoWhiteboard* [6] are distributed over short ranges, but do not care about network communication over large distances. Gestures and interaction on the board are more in the focus of their work, so that it is impossible to edit remotely created artifacts.

Designer's outpost [7] is one of the more recent approaches in the area, but focusses on image vision methods and interaction concepts. A history is also provided, but users have to explicitly save certain states, which are then stored as images in the history. Editing of remote content is also not possible.

The general approach of using an instant messaging protocol for whiteboard interaction purposes is not completely new, but it still lacks tool support. There is one application called *Coccinella* [8], which essentially is an XMPP chat client with an added drawing surface. The development seams to be discontinued since two years and it also lacks any asynchronous features. The used approach here is a combination of SVG and XMPP as it is also described in [9] and [10]. There have been some proposals in order to standardize this protocol, but since 2006 there is no new progress in this area. XMPP itself is the most widely used open protocol for instant messaging applications.

Summing up, the problem domain of whiteboard interaction is a field of research, which brought up many challenges relating human computer interaction or image synthesis such as *Thor* [11], that is why the network communication infrastructure is not in the focus of the work. Often, proprietary protocols are used for communicating between the multiple locations. Recording is also done but more on a representation level (video streams or images) and not on a artifact level (cf. [7]).

Our approach should benefit from open standards in the field of internet communication protocols as well as being open for easy capturing and archiving of recent whiteboard sessions, which enables fast analyses and effortless pause and resume behavior including implicit saving of the communication history.

III. Communication protocol

There is a large variety of input devices and thus highly different requirements for the communication protocol concerning the connected devices. Generally there is a differentiation between Tele-Board-aware devices being equipped with a special client software and those coming out of the box that can participate in the design session with an existing client software that uses the communication protocol. It should be possible to write and send sticky notes from every Internet-enabled device without developing a special client software for it. When using special capabilities such as drawing, it can be acceptable to develop an adapted application for that platform.

Every single component has very special needs in terms of user interface development, data structures, and communication methods. An important decision was using the Extensible Messaging and Presence Protocol (XMPP) (defined in RFC 3921, cf. [12]) as a communication protocol in order to support a variety of input devices and different platforms. There are several implementations on almost every platform and it is supported by a many existing instant messaging clients.

XMPP is an open standard and is typically used as a chat and instant messaging protocol. Over time it has been extended to support voice, video, and file transfer. XMPP (formerly known as Jabber) is used in several instant messaging tools such as Google Talk [13] or Psi [14]. The communication is build upon a client-server model. Authorization, session and roster handling is managed by the server. People can connect with every possible client without transferring any configuration from client to client except for username and password.

The *Server-Buddy* plugin, which is deployed into the Openfire[1] server, acts as a so-called PacketInterceptor to read all messages sent between whiteboards in order to archive them in a database (see figure 2). Special packets such as a request-message for resuming a session are filtered out, directly answered and not stored in the whiteboard history. This procedure makes the collected information usable in the history component, so that every state of a Panel can easily be reconstructed.

Technically, all communication is routed over the XMPP server. In terms of XMPP, the whiteboards chat with each other. The used method for communication between any two partners is a multi user chat (MUC), as it is defined in the XMPP specification. One whiteboard session is reflected in a MUC room on the server, which is automatically created for each new session. The body of the messages is extended to

[1]Openfire is an open source XMPP server software, http://www.igniterealtime.org/

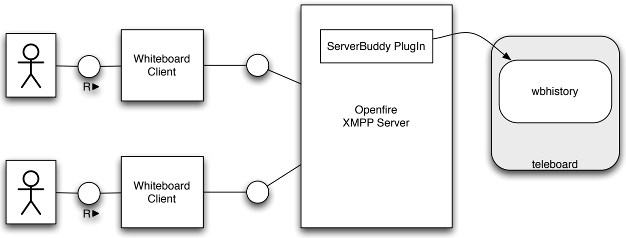


Fig. 2. Communication interception between the whiteboard clients

support the needs of the Tele-Board synchronization. XMPP-Clients producing text-based sticky notes, direct their messages to this room, the server plugin changes the packet to form a valid sticky note description.

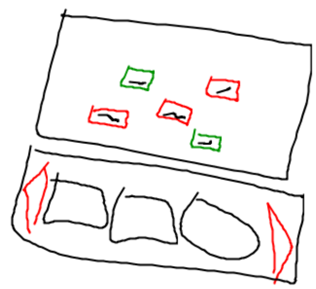
There is an operation code signaling the type of message stored in the message body. There are multiple types of operation codes (opcodes) used:

- `WHITEBOARD_SYNC_NEW` - an element is newly created
- `WHITEBOARD_SYNC_CHANGE` - an element is changed
- `WHITEBOARD_SYNC_DELETE` - an element is deleted
- `WHITEBOARD_SYNC_ALL` - request to send all whiteboard content to the communication partner
- `WHITEBOARD_SYNC_ALL_ANSWER` - answer to a sync-all request, contains the whiteboard content
- `WHITEBOARD_SYNC_PICK` - putting an element into the server-wide clipboard
- `WHITEBOARD_GET_CLIPBOARD` - request clipboard content list
- `WHITEBOARD_GET_CLIPBOARD_SET` - request clipboard content for specific id
- `WHITEBOARD_SET_META` - request to store meta-data on the server
- `WHITEBOARD_DELTA` - special operation for fast updates on single variables (e.g. position updates during drag operations)

There are multiple devices with different capabilities concerning the handling of the XMPP messages. Messages using the mentioned opcodes are sent between the whiteboard clients to synchronize the whiteboard content. The peripherals will send the message content to the local whiteboard. From the Whiteboard Client the content will be synchronized to the remote whiteboard locations. The `WHITEBOARD_SYNC_*`-messages are important for the whiteboard history and will be handled in the Server-Buddy plugin to build up the historical data. `NEW`, `CHANGE`, `DELETE` will be directly archived, `ALL` and `ALL_ANSWER` will not appear in the history.

XMPP as a communication language between the clients turned out to be very appropriate. The development of the whiteboard clients can rely on a sophisticated infrastructure e.g. for user handling and message routing. It did not have to be implemented from scratch, but could be used as an existing part of the protocol. A synchronous whiteboard session can be established without any server changes. What will not work without the server-side plugin, is the archiving of a session and the restoring of an earlier state of the whiteboard session, which means the asynchronous features of the system.

The payload of the chat messages exchanged between the whiteboard clients is an XML-encoded text representation of a single whiteboard element. The example in Fig. 3 shows a set of scribbles, having different properties; the XML extract, shows one single scribble element. There are several attributes describing the element in order to reproduce it at the remote location: `x` and `y` describe the position, `strokecolor` the color of the path, and `d` represents the path itself in SVG-notation. SVG was chosen because it is an established standard and can be directly used for screenshot rendering with only little string conversions.



```
<path id="lutz@fb10dtools_654"
    strokecolor="0.0,0.0,0.0"
    d="M 2286.0 1237.6575 L 2283.0..."
    x="119.0" y="354.0"/>
```

Fig. 3. XML representation of one single Scribble vs. graphical representation of multiple Scribbles with different colors, paths, and locations

IV. Tele-Board - a Whiteboard Application for Synchronous and Asynchronous Settings

We developed the whiteboard software suite Tele-Board. First of all, it supports synchronous working modes. That is, people having the possibility to collaborate over distances or locally at the same time. The main idea is to plug the whiteboards together. XMPP names the notion of bringing together multiple participants as a group chat or MUC (multi-user chat). One message is send to the server and from there distributed to all connected clients.

To realize the specified functionality for asynchronous features in a software system we developed three main functional units:

- interception of message flow
- storage of communication data
- enabling interaction with the history data in an appropriate user interface

The communication should be captured on-the-fly, which has influenced the selection of the technology insofar as it must be possible to analyze packets separated from the message routing. Central roles in the overall system are represented by the message server and its plugin architecture, the web-based management system, and the database management system. The history functionality is a concept that is implemented as a cross-cutting concern in all parts of the system. It can not be realized as one single component, because it enriches the functionality of the other components. A client-server architecture is used for synchronizing the participating whiteboards. A central history archive located at the server is more suitable than e.g. a peer-to-peer model as it is used in [15], because all statistical data should be kept together in order to be analyzed conveniently and enable asynchronous work.

To understand where the communication takes place and what should be captured, the existing communication infrastructure should be briefly outlined: A *Workspace Hub* is a computer system running the whiteboard client software and is located at a digital design space. This computer combines all physical components at this location. Several input devices are paired to it. Cameras make it possible to also synchronize a video stream of the people standing in front of the whiteboards in order to provide a communication channel supporting eye-contact and gestures regarding the whiteboard elements (e.g. pointing on a sticky note). How the multiple Workspace Hubs work together and how they communicate with the help of the server is shown in Fig. 4.

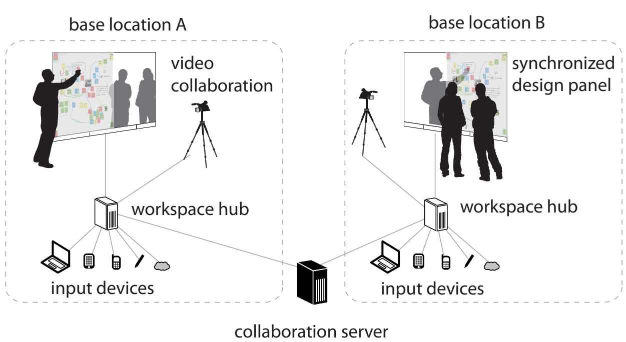


Fig. 4. Overall setup of the communication infrastructure centered around the workspace hubs that are connected via an XMPP server

Projects and *Panels* are important concepts in this context. A Panel p describes the sequence of events e_n executed on one whiteboard in temporal order of these events ($p = (e_1, e_2, e_3, ...)$). An event is a tuple of attributes describing which action has happened where, by whom, and when, to keep the temporal order of the events. Each event has an action code, which can be NEW, $CHANGE$ or $DELETE$ to describe the event type. A Project pro is the collection of multiple Panels ($pro = \{p_1, p_2, ...\}$) in order to configure them with rights to edit/view/delete.

The architecture outlined so far is reflected in three major components:

- a web portal for creating, viewing, and changing projects and panels
- a history browser for exploring earlier states of a white-board session
- a whiteboard client, for interaction with the system

The current Tele-Board system can be run on every computer equipped with a browser and a Java Runtime Environment. Nothing has to be installed on the computer. As a starting point, the user logs into the web portal using username and password. When successfully logged in, it is possible to browse through projects and panels, edit them or create new panels or projects. Every whiteboard session (panel) has a history, which is explorable also within the web portal. The tool used for this history interaction is called *history browser*. With the help of this tool, the user is able to scroll through the timeline of a session. This is possible, because every single event is stored in the history archive. Thereby, screenshots of every second in the editing process can be rendered by the server. When a user finds an interesting state he wants to continue, he can choose to start from this point by branching into a new parallel panel or resume the work from the end of a whiteboard session. Resuming the work means, starting the whiteboard client directly from the browser via Java Webstart. The client requests the latest state of a session via XMPP so that a user can continue working with the most recent information on the whiteboard.

Having this architecture in mind, the question comes up, how efficient this selection of protocols is for such an application. Extensibility, appropriateness, and performance are measures to be evaluated in the next section.

V. Evaluation

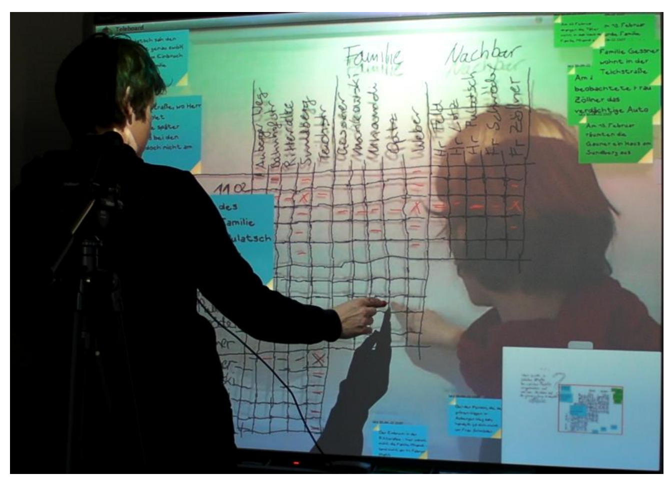


Fig. 5. Tele-Board remote setup including video conference, one participant on the local side, one participant on the remote side

In different user studies (see figure 5) we found out that performance plays an important role in user interaction. People are used to an almost-infinite performance experience with their traditional whiteboards and pens. In earlier project phases we used JavaFX to prototype a whiteboard client interface, which lacked performance at some point. A redevelopment using another UI technology solved that problem.

A more interesting question is, how does the server infrastructure behave in high-load scenarios? How does the addition of a server-wide capturing and analysis plugin influence performance? In the following, we are showing with a series of controlled measurements, how the addition of asynchronous features influences performance depending on the workload situation of the system.

A. General test setup

The test suite used of the following experiments is a small extension for the whiteboard client. We developed a command line interface which enables us to set the following parameters:

- whiteboard sessions to join in parallel
- whiteboard client threads concurrently in one session
- number of whiteboard change events per second

The purpose of the command line client is to load the whiteboard content from the server and then periodically update the content on the board. This is done to simulate human behavior. We chose 1 update per second as an change interval for all experiments. This is beyond the average workload in typical human-generated sessions, so that the absolute number of concurrently connected clients will be higher in real-world experiments.

The measurements are done directly on the server machine. This server is a Debian Linux based virtual machine limited to 1 GB of RAM - not a high-end device. The observed variables are the CPU load of the MySQL and Openfire processes on the server as well as the transfer rate from and to the server. The measurements are recorded using Unix/Linux built-in tools and averaged over a period of 2 minutes each.

As the whiteboard content used for these experiments we use a blank whiteboard with no drawings on it. Earlier observations revealed that drawings are causing only a little portion of the system load. We placed a set of 25 post-its on the board, each of them with extensive drawings on them. The clients choose randomly which sticky note to change and send the change message to the server, which then distributes it to every other client that update their whiteboard element information.

The following two scenarios will point out two major insights. First, it will be shown which limitations are in a communication model using only one session and connecting as many clients as possible to it and keep it in sync. This scenario is presented to reveal boundaries of the network infrastructure. The second scenario aims more at real-world usage and evaluates the scalability of the system serving multiple parallel sessions each with a constant amount of participants. The idea of both scenarios is to see the influence of the asynchronous in-situ capturing approach on the overall performance.

B. Scenario 1 - many clients in one session

Looking at the concrete scenario 1 of multiple clients in one whiteboard session, it becomes clear that there is potentially a large number of packets to be examined in order to be archived. Figure 6 shows how the message distribution is achieved by the XMPP server. One can see that communication between clients and server is still the major duty of the server infrastructure, even though there is a new server-side plugin being responsible for asynchronous features.

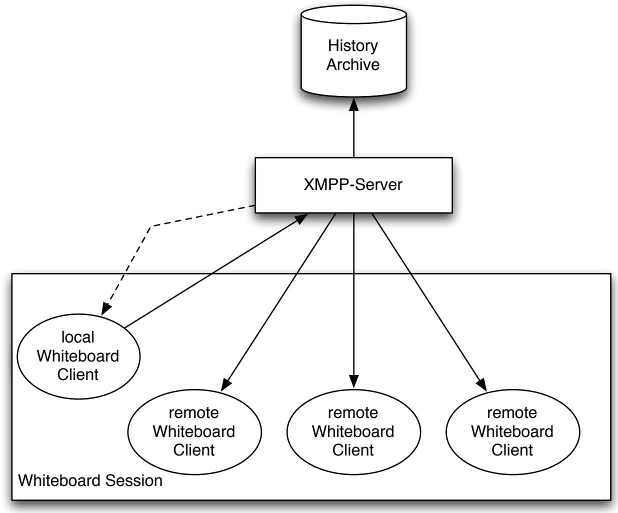


Fig. 6. Conceptual model of a group chat session used for distributed whiteboard interaction

Our evaluation now aims at showing that the server capacity is not limited by the load of the capturing feature. It is more important to look at the transfer rate plotted against the number of clients in one session. In figure 7 right you can see that there is a bottleneck in the number of clients when reaching a limit of about 10MB/s. About 40 to 50 clients seems to be the maximum of concurrent clients in one session. This is due to a 100MBit connection of the virtual machine server used in this setup. The predominant number of messages send in our system (in real-world use cases) are move messages, which are only a few bytes in size. This is also one of the reasons that real-world system load would allow more clients than this experiment.

This high bandwidth requirements are caused by the high number of active participants in the group chat. Thinking of presentation scenarios with few persons being active and many persons only receiving, the numbers would lower dramatically. An example: We have 50 clients producing 1 update per second and the server synchronizes each operation to 49 other clients, we have about $50 * 49 = 2450$ packets to send per second. $10MB/2450 \approx 4kB$ results as an average packet size, which matches the sticky note description size including a path string for the drawing on it. With the same bandwidth in a presentation scenario, theoretically about 2500 receiving clients could be handled with one presenter (1 update/s, 2500 receivers, and packet size of $4k$ lead to $10MB/s$).

Nevertheless, the tendency shown in scenario 1 is clear: The limiting factor is not the CPU load of the database for storing the archived communication channel, but it is more the transfer rate from the server, when change requests are propagated to the connected whiteboard clients. The hypothesis, the database operations for enabling asynchronous features really effects the system, cannot hold true in this experiment. MySQL uses a significant amount of CPU time, but the exponential growth

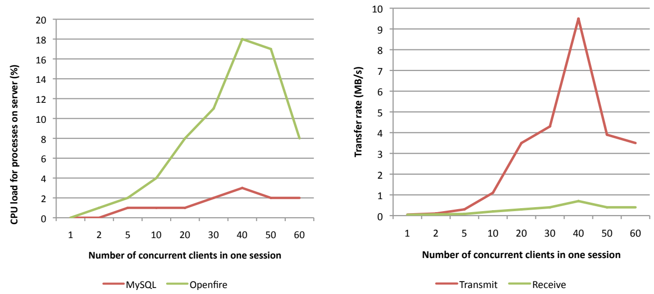


Fig. 7. Scenario 1 - many clients in one session; left: Mysql and Openfire CPU load in relation to the number of clients in one channel, right: the data transfer rate on the server differentiating transmit and receive direction

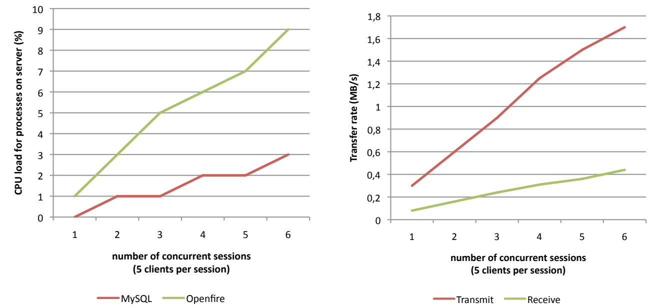


Fig. 9. Scenario 2 - many sessions with equal load; left: MySQL and Openfire CPU load in relation to the number of concurrent sessions each with 5 participating clients, right: data transfer rate on the server vs. the number of concurrent sessions

in transfer rate from the server is much more effecting system performance.

C. Scenario 2 - many session with equal load

This scenario is more realistic for the typical workload on our system. Figure 8 shows that there are multiple sessions running in parallel and few users are logged into the system. For this experiment we chose an equal number of 5 participants in one session. Even for this low number, typical real-world applications will stay under this value, because they often use only two locations (which means two whiteboard clients) at the same time.

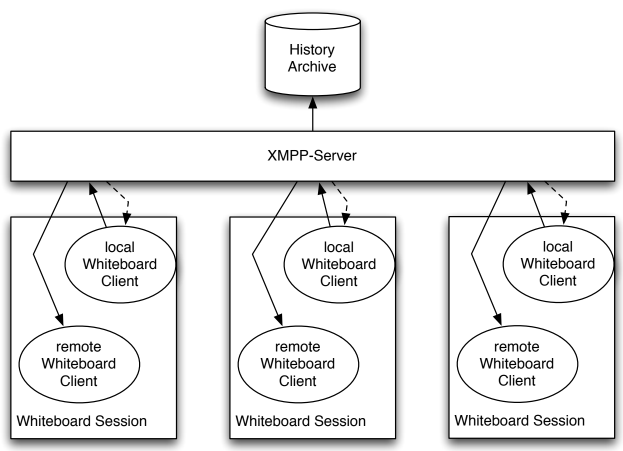


Fig. 8. Conceptual model of multiple group chat sessions used for distributed whiteboard interaction

Figure 9 shows an almost linear development of bandwidth usage and cpu load, the more clients connect. The absolute values - especially for the bandwidth - stay much lower depending on the number of clients (number of concurrent sessions $*$ 5 clients per session) compared to scenario 1. You can see that the system scales very well for this kind of workload. One can also see that the database load is also a fraction of the XMPP server load.

Therefore we can state that the implementation of asynchronous aspects directly on the server does influence all-over performance only little. It adds a proportionate load on the system, but the predominant computation time still is used for message routing to the connected clients. As we have shown in [16], the simple data storing architecture is very appropriate for the typical workload on the system: There are many small write operations and only little longer running read operations. A long and de-normalized table that is used here, optimally supports this setting as it also can be seen in the performance evaluation results.

VI. Conclusion and Outlook

We presented and evaluated the concept and the implementation of an instant messaging protocol that is used for distributed digital whiteboards. Many other tools in this sector focus on one working mode of the CSCW matrix. We try to pursue the goal to enable all modes of interaction equally, so that users can work together co-located as well as distributed and also synchronous as well as asynchronous. The additional workload to be dealt with by the system has shown to be sufficiently low.

The method of capturing and storing complete communication channel traffic in order to rebuild the session afterwards turned out to be highly beneficial for two reasons: It is an efficient way to rebuild what has happened, including the reconstruction of every point in time of the whiteboard history as well as the ability to branch into new parallel sessions. The second advantage is that analyses of the communication scenario - for example for researching on design teams - are much easier to achieve, because nothing related to the communication itself has to be recorded actively, but it is stored implicitly.

Limitations that are shown in the evaluation section are not distracting the usual usage scenarios with our system. Typical setups, when people work together in smaller groups of only a few participants, are well-performing. The assumption of having a large number of participants in one room is strongly hypothetical for our kind of usage scenarios. Even in large presentation scenarios, only few active clients would exist and a potentially large number of receivers. Typical whiteboard usage scenarios tend not to involve larger numbers than 5-10 participants.

Future applications based on the archive we develop over time aim in the direct of better analyzing the gathered communication data. For example it is unnecessary for a person who wants to understand what has happened in past sessions, to watch at every single event. Possible analyses and visualizations are based on the assumption that there are states which are more important than others and should be highlighted. Based on observation and user feedback we have to come to a common understanding of an important state.

The generalization of this method can be an approach for other applications as well and is not limited to whiteboard use. Overall, XMPP as an instant messaging protocol turned out to be well-suited for synchronizing digital whiteboards and the archived communication data carries a large potential for further research. The results of our evaluations play an important role also for gathering insights on deployment of the Tele-Board suite into business environments with high demands on reliability.

Acknowledgment

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Evaluation of the Placement of Network Services

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Abstract— *Network services are used in the Internet today for a variety of functionality, from Voice Over IP (VoIP) relays to network game servers. Determining the placement of these network services and measuring the quality of the placement in realistic scenarios is a challenging problem. This paper explores different methodologies for evaluating the placement of network services. The strategies include: simulations of with thousands of nodes, emulation of different topologies with hundreds of nodes, and Internet deployment for a diverse selection of nodes and network communications. In addition, comparisons to a centralized communication model are studied. This range of exploration enables a better understanding of the impact the selection of supernodes provides on a variety of applications and platforms.*

1. Introduction

Network researchers are faced with a range of challenges when developing a new service or application. One of these challenges is determining the proper set of evaluation techniques for the new service. In order to fully evaluate any new service, the process requires evaluation from different perspectives and implementations. This process varies from exploring the essential algorithms at the heart of the service to deploying a full implementation running on the Internet. Evaluating a new service across this field of implementation options provides a deeper understanding along with validation of ideals and assumptions at different levels.

We break the assessment process down into three categories. First we have simulation. Here we consider discrete event simulation models where the functional and performance properties of a candidate Internet service are being investigated. Second we have emulation. In emulation physical (or virtual) nodes along with a physical (or virtual) network exist in some constrained topology (such as Emulab [1]) to provide a virtual sandbox for our service. Lastly we consider experimentation as physical (or virtual) nodes and networks on an Internet topology (such as PlanetLab [2]).

Typical computer science research first investigates a simulation model, then builds a prototype to evaluate in emulation and finally a full deployment through experimentation. We are turning this model around. Here, we present an application that can utilize our proposed service and demonstrate improvement. Next, we evaluate the service itself through experimentation with Planetlab. We then evaluate the service on a constrained topology through emulation on Emulab. Finally a variety of network topologies that we could not deploy on are created and evaluated through simulation.

The network service we are evaluating is the Supernode Placement in Overlay Topologies (SPOT) [3]. This research leverages our initial work that proposed several placement algorithms. In this paper we explore the service in more detail and investigate it's behavior using different techniques. First, we will briefly discuss SPOT's behavior. Next we demonstrate the feasibility of SPOT as a service to determine Internet game server placement. Then we investigate SPOT with emulation and finally evaluate a simulator developed for SPOT.

2. Background

Here we provide detail about the general behavior of SPOT along with an example. A more complete explanation can be found in [3]. SPOT selects a subset of network nodes to be supernodes (SNs). A supernode is one that provides additional services to the network. Consider a SN as the game server node in a first person shooter game played on the Internet or a relay node to assist in Voice over IP (VoIP) communication. We now define the problem formally

2.1 Problem Statement

Consider a graph $G = (V, E)$ where V are the nodes and E represent links between the nodes. We define some subset of V as V_A, which are active sending message nodes. Of these V_A nodes we define a subset V_W that represent those nodes that are willing to become SNs. We then define V_k as the active nodes that are currently assigned as SNs, where k is the number of SNs we are interested in assigning. Therefore, $V_k \subseteq V_W \subseteq V_A \subseteq V$.

We also define a demand t for some node u as $t(u)$. The shortest distance between nodes u and v is $d(u, v)$. We are interested in finding a set of k nodes to be assigned SNs. Therefore (from [3]) we want to determine V_k such that

$$\forall S \in 2^{V_W}, |S| = k \rightarrow (\sum_{u \in V_A} t(u) d(u, V_k) \leq \sum_{u \in V_A} t(u) d(u, S)) \quad (1)$$

We also define a $TotalCost$ [3] for any S where $|S| = k$ as the following:

$$TotalCost = \sum_{u \in V_A} t(u)d(u, S) \quad (2)$$

This gives us a measurement of how well one selection of SNs compares to another.

2.2 General Behavior

As the size of the network increases, calculating the distances and demands to and from all nodes becomes expensive. In order to reduce the amount of communication required, nodes are divided into groups called *neighborhoods*. The size of the neighborhood is based on a predetermined distance metric r from the current SN. Inside each of these neighborhoods complete distance and demand information is determined. As the SN assignment changes inside a neighborhood, the center of the neighborhood refocusses on the new SN. Therefore the members inside the neighborhood may change after each assignment.

Nodes initially joining the network connect to a well known bootstrap node for authentication and initialization. Once authenticated, a node is either promoted to SN status or provided the address of an SN to connect to. Each is then notified by that SN if the node is close enough to join the neighborhood. Those nodes not joining the neighborhood locate the closest node in the neighborhood to act as a *neighborhood representative*. A neighborhood representative provides mechanisms for nodes outside the neighborhood to influence the future assignments of SNs. This is accomplished by the neighborhood representatives aggregating demand from nodes outside the neighborhood and representing it as their own. Where demand represents the desire for a node to use the service and we are assuming a demand of unity for each node. An example is now scenario is now provided.

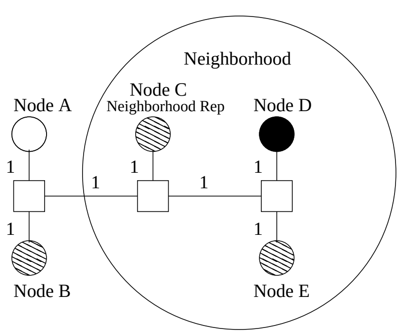

Fig. 1: Example neighborhood with SN at node D and neighborhood representative at node C with an r value of 3.

Figure 1 depicts a network with three nodes inside and two nodes outside of a neighborhood. The nodes are connected with routers (denoted as squares) and hop distances (denoted as wires between nodes and routers). A distance of two exists between Nodes A and B in Figure 1. Nodes willing to become a SN are denoted with partially filled in circles (nodes B, C, and E). The solid circle represents the current SN (node D). Nodes that are active but choose not to become the SN are represented with a plain circle (node A). In this figure, the radius metric r=3, therefore nodes nodes C and E are inside the neighborhood (with distances of 3 and 2 to the SN, node D, respectively). Nodes A and B are too far from the SN therefore placed outside the neighborhood, node A is also not willing to become an SN (it is in the active state, but not willing to become an SN) and would not join the neighborhood even if it was close enough. This is because any node not in the willing to become an SN state can not join the neighborhood because all nodes in the neighborhood must be eligible to become an SN. Nodes A and B must find their closest neighborhood representative and will select node C. With this topology, nodes A and B use node C as their neighborhood representative, and node C reports a demand of 3 to the SN, with node E reporting a demand of 1. With the node demands and topology information for nodes C, D, and E, the SN, node D, is ready to determine if it will reassign the SN to a new location. To do this, the SN solves the local k-median problem for three nodes with the specified demand and topology information using integer linear programming (ILP). In this example, the output of the k-median problem assigns node C as the SN and the neighborhood in Figure 2 is created.

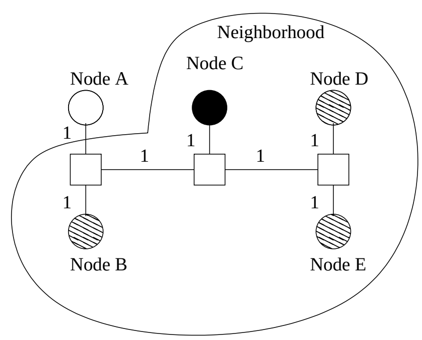

Fig. 2: Example neighborhood with SN at node C.

This SN placement strategy strikes a balance between complete global knowledge and a very limited local view. With local knowledge of the distance to each node and associated demand, the SN is able to solve the k-median or problem while not completely ignoring nodes outside of the neighborhood by considering aggregate demand information.

2.3 Software Implementation

SPOT is written in 12000 lines of multithreaded Java code. Each node initializes by listening on a well-known TCP socket and forks a thread for each incoming command. In order to solve the local k-medians problem the GNU Linear Programming Kit (GLPK) [4] is utilized.

3. Experimentation

3.1 Introduction

The first type of evaluation is via *experimentation*. Here, we are deploying the SPOT system on the Planetlab testbed. This testbed consists of over 1000 nodes distributed around the world. Each researcher is allowed access to a *slice* of every single node in the network. This is very useful with regards to the diversity of systems and networking environments. This type of environment increases the realism and quality of experimentation greatly.

The test setup involved deploying SPOT on 50 nodes in the Planetlab environment. The size of the experiment may appear small, however due to the unpredictable nature of Planetlab, hundreds of nodes are unavailable at any given time.This number is also common with other researchers working with distributed systems [5].

3.2 Planetlab Evaluation

The initial experiments on Planetlab involved collecting statistics about the nodes. In Figure 3 the cumulative distribution function is provided for the number of hops necessary to reach all of the nodes.

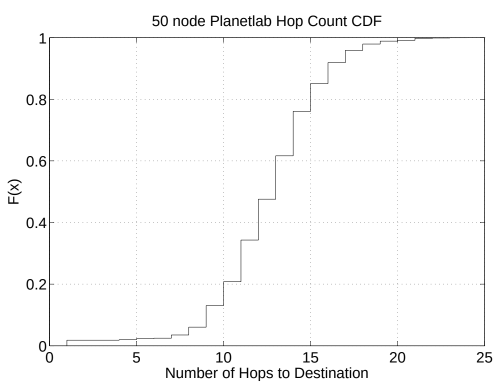

Fig. 3: CDF of the number of hops necessary for nodes to reach each other in the 50 node Planetlab experiments.

Next we deployed the SPOT system on all 50 nodes and selected an arbitrary node (not one of the 50) to operate as the bootstrap server. Once the software was deployed, 50 experiments were run with k values of 1, 2, and 3 SNs, where k represents the number of SNs deployed. The results of the experiments are shown in Figure 4. These results use the hop count metric. From the results, the total cost decreases as the number of SNs increase. This is to be expected as adding more SNs should decrease the total network cost. These results are reported using the whisker-box plot that presents the lower (0.25), median, and upper quartiles (0.75) of the data. Also included are the minimum and maximum values. The interquartile range (IQR) is defined as the upper quartile minus the lower quartile. Also, any value that is 1.5 times the upper quartile or 1.5 times the lower quartile is consider an outlier and denoted with a circle.

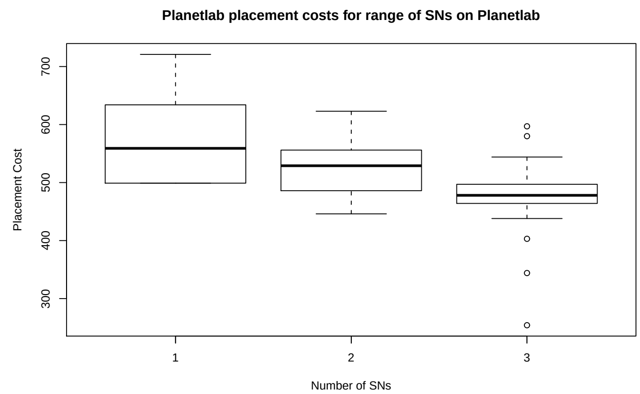

Fig. 4: 50 node Planetlab r-SPOT experiment using the number of hops distance metric illustrates the placement costs for k=1,2, and 3 SNs.

In these experiments we measured the total time to locate SNs, the number of iterations of the algorithm to assign an SN and the total amount of network traffic generated from all of the nodes in the experiment. Due to space constraints only the time to assign an SN is illustrated, Figure 5. The remaining data can be seen in [6]. From the results, as the number of SNs increase, so too does the total time necessary to assign the SNs.

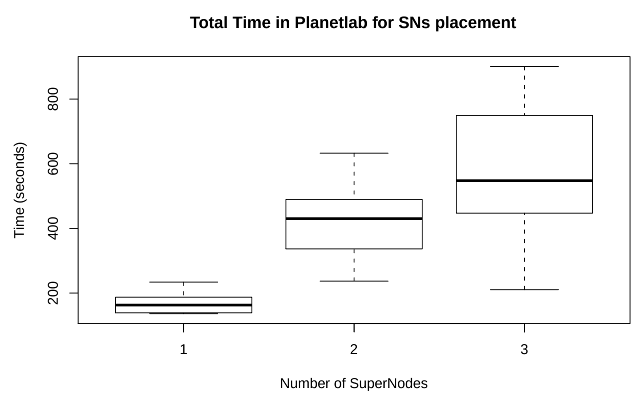

Fig. 5: 50 node Planetlab r-SPOT experiment illustrating the total time to place k=1, 2, and 3 SNs.

4. Game Servers

4.1 Introduction

The previous results have dealt with SPOT, its ability to place SNs throughout a network and measurements associated with it. Ultimately, the improvements that SPOT provides for applications utilizing its service are important. In order to evaluate that, we turn our focus to online video games, namely multi-player first-person shooters. Multi-player first person shooters (such as Quake III Arena and Half-Life [7]) are very sensitive to the latency from the client to the game server. Typically anywhere from 16 - 64 people connect to a single server or host. This host sends game updates to all players connected to the server. If the client has a RTT to the server greater than 180 - 200 ms, it can greatly reduce the quality of the experience as well as fairness in the game itself [8]. Therefore when creating a multiplayer game, it is important to choose the game server carefully.

4.2 Planetlab Evaluation

In order to evaluate the effects of server selection, the 50 node setup in Planetlab was studied. Using these 50 nodes, the ping data collected earlier was used to evaluate the RTTs letting each node become the *server* in a online game. Therefore, we are interested in the RTT from each client to that server. From this data 7 of the 50 servers or 14% of the nodes would be unable to satisfy the requirement that every node maintain a RTT under 180 ms. This demonstrates the importance of carefully selecting a SN. The RTTs are shown in Figure 6.

Next SPOT was run across all 50 nodes with k=1 and it selected node 8 as the SN, the average RTT is 60 ms to the SN and the maximum RTT is 139 ms as shown in Figure 7, from [3]. The optimal value is selecting node 6 as the SN with an average RTT of 40 ms and a maximum RTT of 118 ms. The worst case selection is node 42, with an average RTT of 127 ms, a worst case RTT of 1545 ms and three nodes over the 180 ms threshold.

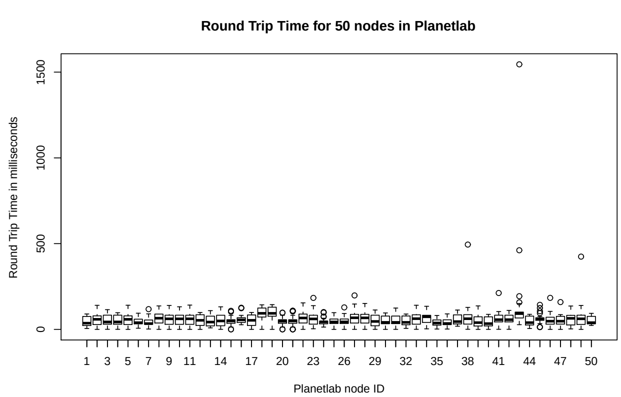

Fig. 6: Round Trip Times from each node to all 50 nodes in Planetlab.

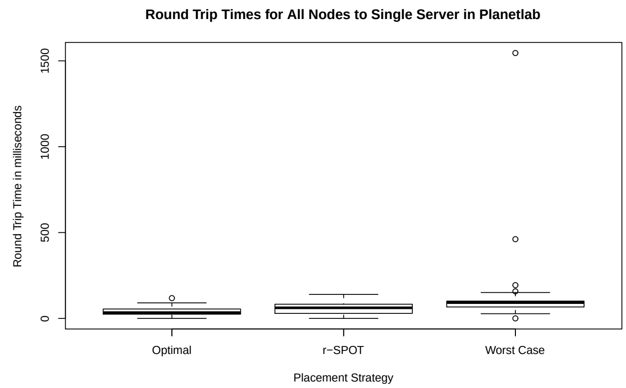

Fig. 7: Round Trip Times from each node to a single server selected based on the particular placement algorithm times.

4.3 Updating the SN as the topology changes

We are now interested in the effects of a dynamic game state where players join the game after some period of time. Here, we are interested in whether it is necessary to recalculate the location of the SN after a number of new players join. Consider a 40 player node topology taken from the original set of 50 nodes in the previous experiment. We use r-SPOT (one of the SPOT algorithms defined in [3]) to determine a location of the SN (node 7) and measure the RTT from all the players to that SN. Now suppose 10 more players join the game and the SN is not re-evaluated with all 50 players. With node 7 still serving as the SN, one of the new nodes joining is unable to play the game due to a large RTT (1545 ms to node 7). However, if the SN is re-evaluated and moved to node 8, all players are able to participate. The results of this experiment are shown in Figure 8 with a whisker-box plot of all RTTs. From the figure, when node 7 is the SN in the 40 node experiment the average RTT is 44 ms. Once the 10 additional nodes join, the average RTT jumps to 81 ms with the outlier node experiencing a large delay to SN 7. When the topology is re-evaluated the SN moves to node 8 and the average RTT drops to 60 ms. This illustrates the importance of re-evaluating the SN assignment in order to maximize the number of players in the game.

5. Emulation

Emulation experiments with SPOT are now presented. Two different placement algorithms were developed for SPOT, r-mod and r-SPOT. The r-mod algorithm takes a related approach [3] and adopts it to our problem. The r-SPOT algorithm improves upon r-mod with various optimizations [3]. The network topologies consist of hierarchical networks of size 100, 200, 300, and 400. The Emulab physical nodes were of the type pc3000. The pc3000 are 3 GHz Pentium 4 CPUs with 2 GBytes of RAM. There are 160 nodes on Emulab of this type. In order to create larger sized

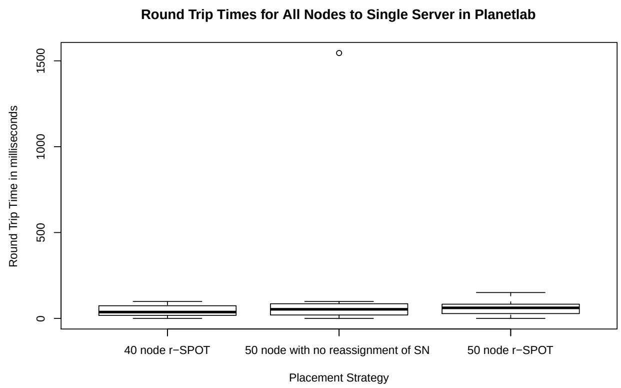

Fig. 8: Round Trip Times from each node to a single server for a 40 node topology with r-SPOT SN placement, a 50 node topology using the 40 node SN placement, and a 50 node topology with a new r-SPOT SN placement.

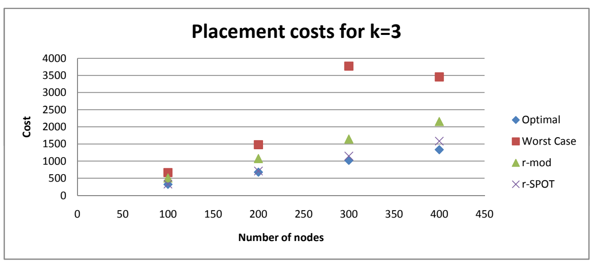

Fig. 9: Results of placement for three SNs with various sized network topologies [3].

networks, virtualization is used with Emulab. The virtual machine consisted of FreeBSD jails with an assignment of 10 virtual machines per physical node. This allowed for larger experiments (greater than 160 nodes) while not over-utilizing a single physical machine. The radius value (r) was set to 2 in all of the experiments, consistent with related work [9]. The experiments were evaluated with a k value (number of SNs) of 1 and 3. The SPOT software was loaded on each node and a special bootstrap node was also created running the bootstrap software. Each node ran a script which would execute the SPOT Java application and connect to the bootstrap node.

In Figure 9 (from [3]) the results for k=3 show improved placement for r-SPOT compared to r-mod. For example, in the 300 node experiment r-SPOT was only 13% above the optimal compared to r-mod with an average placement cost that is 61% higher than optimal.

In addition to the cost of the resulting network topology upon algorithm completion, other metrics of interest are also evaluated for r-SPOT. The total number of iterations to reach a finishing state and total system time necessary before the experiments finished are discussed next.

The first experiment uses a whisker-box plot to display the system time necessary to locate SNs in a network topology (Figure 10). From this graph we can see total system time increase as the number of SNs increase. This is to be expected as increasing the number of SNs to place increases the total amount of work and the time to complete it.

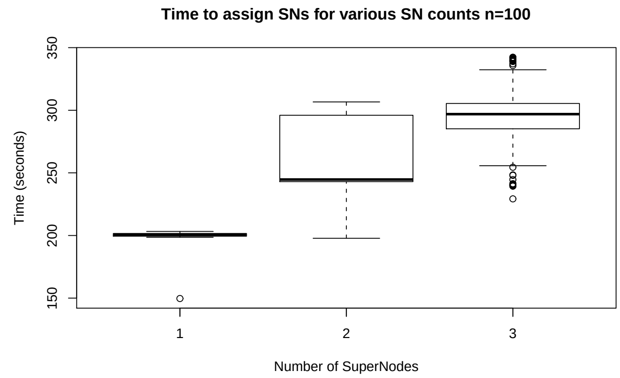

Fig. 10: Total time to place SNs in r-SPOT algorithm.

Additional experiments are run to understand some of the tradeoffs between r-SPOT and an optimal placement strategy. In Figure 11 the total time to locate three SNs is computed for various topology sizes in comparison to a global solution. The global solution requires full topological information, which is equivalent to increasing the neighborhood size to include all nodes in the network. At the 250 node size it starts to become increasingly more expensive to place nodes with the centralized optimal solution.

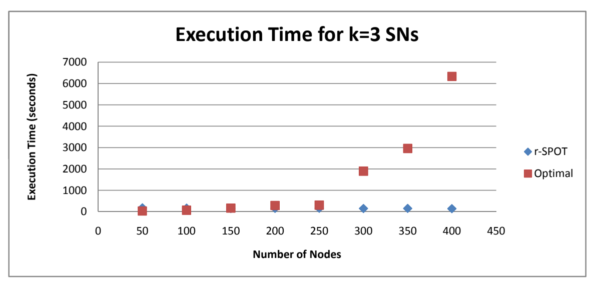

Fig. 11: Time to place 3 SNs in r-SPOT.

This section has presented the SPOT system deployed on Emulab using the r-mod and r-SPOT algorithms. The evaluation demonstrates the improved placement performance of the r-SPOT algorithm and how both approaches compare to the optimal results.

6. Simulation

6.1 Introduction

Simulation provides an excellent opportunity to extend the previous results to large, and more realistic network topologies. With research testbeds, a node size limit is reached somewhere in the hundreds of nodes. In order to evaluate systems larger than that, simulation is very useful. Also with simulation, the experiments can easily run on different topologies.

In order to provide simulation with SPOT, a discrete event simulator called SPOTSim was developed. SPOTSim was written in Java and models all of the communication between nodes running SPOT. It also interfaces with the same ILP solver (GLPK) as SPOT.

The simulator was developed after creating SPOT, therefore the true functionality of the working system was captured in the simulation environment. Typically a simulator is developed first and the final implementation ends up behaving somewhat differently due to real world constraints. This is much less the case with SPOTSim, which provides a fairly realistic model of SPOT's behavior.

6.2 SPOTSim Evaluation

In order to test the validity of the model, experiments were run on Emulab with SPOT and on SPOTSim with the topology deployed on Emulab. The first experiment illustrates the placement scores of both SPOT and SPOTSim for k=3 in Figure 12.

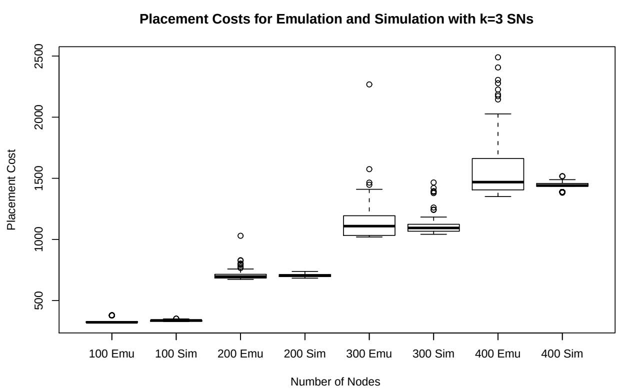

Fig. 12: Comparison of Emulation and Simulation placement results for various topologies locating three SNs.

Although the simulation and emulation results are closely related, some differences are found in the more costly emulation placement results. These results are not captured by the simulation model. The higher scores in emulation are due to the missed opportunities for SNs to join with other SNs and create larger neighborhoods. In the simulator, the merge operations occur with perfect knowledge of the other available SNs.

The strength of the simulation model is its ability to experiment on a larger number of nodes and more interesting topologies. To accomplish this a topology generator was used to aid in the design and creation of larger more realistic topologies. The topology generator used is BRITE [10]. Based on previous related work [9] the BA-2 router level topology was selected for our simulations. A range of sizes were created (500, 1000, and 1500 nodes) using the BRITE's default growth rate parameters.

Using the three topologies created with BRITE, simulations were performed and the average placement cost is presented. In Figure 13 SPOTSim and the optimal results are presented. From the figure, SPOTSim is able to place SNs with a cost of less than twice that of the optimal.

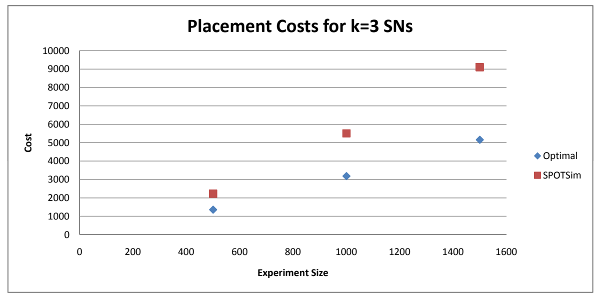

Fig. 13: Placement costs for SPOTSim simulations compared against optimal placement costs.

Simulations were also run with various neighborhood node sizes. Thus far, all simulation and emulation results were executed with a r or neighborhood size of two units, where units are some metric such as network hops. In Figure 14 three different values for the default neighborhood size are experimented with placing k=3 SNs in the 500 node router topology. From the figure, increasing the default neighborhood size r reduces the cost of placing SNs. The mean placement costs are 1045, 954, and 965 for r values of 1, 2, and 3. A default neighborhood size (r) of two provides a 7% reduction in median cost and setting r to three reduces the costs by an additional 1%.

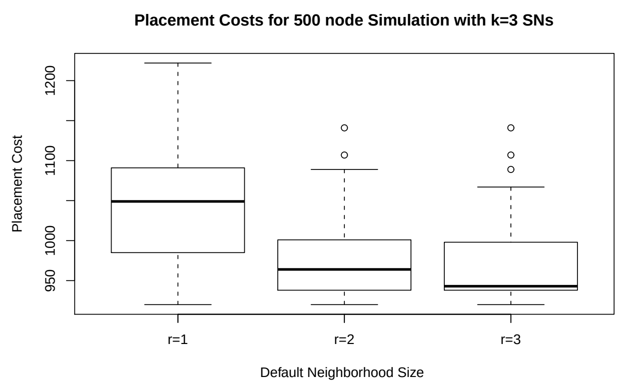

Fig. 14: Whisker-box plot of placement costs for SPOTSim for 500 nodes with varying initial neighborhood sizes.

6.3 Revisiting the Game Server with Simulation

We now return to the game server placement problem initially presented using Planetlab. With the simulator we are able to insert a Planetlab topology into the simulator to evaluate the performance of SPOT. Due to the unreliable nature the Internet and Planetlab, evaluating the same set of more than 50 nodes can be rather challenging. With this switch to simulation we are able to investigate a larger topology of 263 Planetlab nodes around the world. The RTTs were collected to and from all nodes in the evaluation. Next we evaluated the individual RTT from each node to the candidate SN. Figure 15 depicts the maximum number of players that can join the candidate SN server. A player can join the server if the RTT to that server is less than 180 ms. From the figure, 107 potential SNs can support 200 or more players in a single game (the largest is 236), also 116 potential SNs support 100 or more players. The least number of players came from the pair of nodes located in Uruguay, supporting 4 and 5 players each. Finally, a comparison is provided showing the relation between solving the k-medians problem and finding an SN that supports the most number of players. In Figure 16, a bar graph represents the total number of players that could connect to a single SN in the best and worst case. Also shown are the results of the k-medians optimal solution and the SPOTSim solution with respect to the number of players each SNs supports. From the results, the maximum number of players in a single game with the best SN placement is 236 players, while the ILP solver and SPOTSim selected nodes supporting 227 and 226 players, respectively. This helps to demonstrate the ability of SPOT to determine SN locations for a first person shooter.

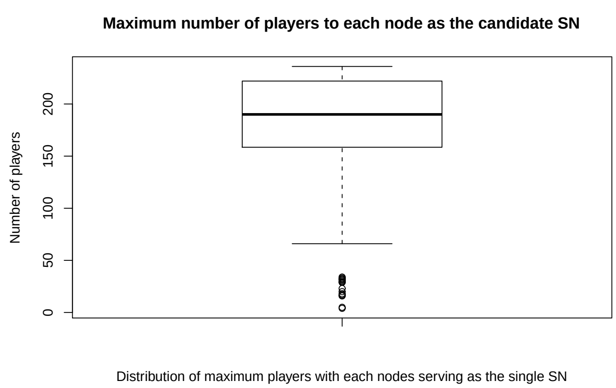

Fig. 15: Max players for each nodes as SN.

7. Conclusion

This paper has presented the SPOT system evaluated in three different environments, experimentation, emulation and simulation. This research demonstrates a fully functioning P2P system that provides SN placement for a range of applications. An example application with SPOT locating a game server showcases the benefits of this network application.

Through this variety of experiments a better understanding of the performance of SPOT is gained. This work motivates the use of different domains to evaluate a large distributed system. Finally, the creation of the SPOTSim simulator allows researches to discover the impact of different placement algorithms on a much wider set of network topologies.

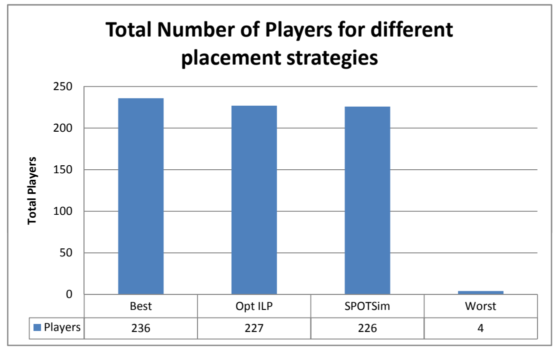

Fig. 16: Number of players supported for various SN selection strategies.

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Performance Analysis of a Model-View-DynamicViewModel Design Pattern

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ABSTRACT

The implementation of an appropriate software architecture is crucial in achieving the optimum performance from a system. Web-based applications are becoming increasingly popular in replacing previous Windows-based applications. This has lead to a growth in new technologies and architectures to cope with the new workload and performance demands that web-based applications can require.

This paper will look at a current software company that provides process-heavy web-based application systems providing grant management services to medical and academic research. The purpose of this paper is to analyze the performance of their existing Web Forms based management system against a newly developed dynamic component-based architecture using a variant of the Model View Controller (MVC) and Model-View-ViewModel (MVVM) patterns, which we have termed MVDVM (Model-View-DynamicViewModel). Accurate and meaningful results will demonstrate how applications can now benefit from this approach.

Keywords – Component based architecture, Dynamic forms, Interoperability, Model View Controller, Separation of concerns.

I. INTRODUCTION

Improvements in Internet connectivity and hardware performance within the last ten years have meant that software developers no longer have to worry as much about bandwidth, processor or memory requirements when implementing process-heavy applications. Although this has been beneficial to the software industry it has lead to many systems being implemented in ways that do not utilize best performance techniques available. This can deliver sub-optimal user experience and utilizes greater server capacity than is necessary at a time when datacenter costs are rising.

Separation of concerns (SoC) is a general problem-solving idiom that enables developers to break the complexity of a problem into loosely-coupled, easier to solve, subproblems. Underlying this idiom is the hope that the solutions to these subproblems can be composed relatively easily to yield a solution to the original problem. A concern is a distinct concept within a software system that should be considered separate during the lifetime of the system. Some examples of concerns are design patterns [1], [2], architectural rules [3], and non-functional features like persistence and distribution [4]. The benefits of SoC have been discussed elsewhere (eg. [5], [6]).

This paper will look specifically at design patterns for the user interface within the context of a real-world software product. The paper proposes a variant of the well-known Model-View-Controller (MVC) pattern, which we have termed MVDVM (Model-View-DynamicViewModel), and compares its performance in a re-architected product against an earlier version of the product that used a different technology for producing the user interface, based on Microsoft’s Web Forms. Statistics such as page load time, page size, number of database queries will be measured under varying loads and analysed to assess how these will benefit by applying latest thinking and best practice when developing high-performance software systems. In the next section, a brief overview of the product is provided followed by a discussion of related research. The subsequent sections will discuss the technology platform, the test platform and then present the performance data of the old product and the re-architected product.

II. CASE STUDY OVERVIEW

This paper analyzes a component-based dynamic solution used to manage application forms within CC Grant Tracker, a Grants Management software product. The solution builds and publishes forms, which are then used for the completion, submission and ongoing management of applications. CC Grant Tracker is a solution from CC Technology, a leading global supplier of advanced grant management software solutions, based in Glasgow, Scotland. CC Grant Tracker supports organizations such as charities, academic institutions, public funding bodies and corporate foundations who often employ a regular cycle of grant applications, where professional reviewers decide which applications are worthy of funding. For a large organization, there may be hundreds or even thousands of applications, as well as dozens of reviewers, not to mention stringent auditing requirements. In such cases, simply administering the application process is a major undertaking. CC Grant Tracker manages the full life cycle of grant administration, from initial application through evaluation, approval and ongoing management.

When CC Technology originally created the software product the emphasis of the initial analysis and design phase was based on functionality rather than the performance of the system. The application is heavily web forms-based and the initial version was delivered using the standard web development tools from Microsoft – ASP.NET Web Forms. This delivered a tightly-coupled solution with associated maintenance issues. As the number of clients and the breadth of their requirements have increased the requirement to develop the flexibility and performance of the application forms has become a priority.

CC Technology have been using a re-architected solution with a number of new large clients and the new system has been extremely well received.

III. RELATED RESEARCH

A. *Web Forms vs MVC*

Microsoft currently endorses two web architectures, Web Forms, which was released in 2002, and the Model-View-Controller (MVC), which was released in 2009. MVC is a standard design pattern that Microsoft has adopted. The MVC pattern helps deliver solutions that separate the different aspects of the application (input logic, business logic and UI logic), while providing a loose-coupling between these elements [7]. Both architectures provide benefits to the developer and end-user depending on the circumstances. For this study we examined both approaches to obtain a better understanding as to which provides the better performance benefits.

1) ***Web Forms*** **–** The major advantage of a Web Forms implementation is that it allows an application to be developed in a relatively short period of time as there are lower initial design requirements for the solution architecture. However, research and experience shows that there are a number of performance disadvantages inherent to this approach.

Web Forms allow only one form tag to be added per page. This means that in order for the page to get information to or from the server, the entire page must be posted back. A common approach to address this uses Microsoft's Ajax Control Toolkit and update panels. This reduces performance problems related to sending unnecessary data back to the server however the Ajax update methods can be just as performance heavy in bandwidth and processing on the server due to the View State that is required on all updates from a Web Form regardless of a full postback or asynchronously.

The View State holds session information for each page [8]. This adds considerably to the data size of the page and as this is exchanged in all updates impacts the bandwidth for the solution. Many of the built-in ASP.NET components make heavy use of View State, so it is not unusual for a page to have 10's of Kbytes of View State [9]. This is measured in terms of performance for end-users and bandwidth costs at the data centre. There is therefore no performance gain in using Ajax update panels throughout the application as the net effect can be the same or greater than doing a full postback.

A further serious issue for modern web pages is the abstraction from HTML, as this hinders accessibility, browser compatibility, and integration with JavaScript frameworks like JQuery and PrototypeJS [10]. Of lesser importance to this case study, the postback model makes it harder for search engines to rank ASP.NET pages high.

2) ***MVC*** **–** The Model-View-Controller (MVC) architectural pattern was first proposed by Trygve Reenskaug in the late 1970s (the earliest source available is [11] but MVC was not publicly documented until 1988 [12]). It was first used in Smalltalk [13] and is an effective approach for supporting multiple presentations of data. Users can interact with each presentation in a style that is suitable to the presentation. The data to be displayed is encapsulated in a model object and each model object may have a number of separate view objects associated with it, where each view is a different display representation of the model. Each view has an associated controller object that handles user input and device interaction. MVC was devised to target desktop applications, but because of its relatively loose formulation it was easily adapted for the Web. MVC (or Model 2 using the Sun Microsystems terminology [14]) is now used widely in web frameworks (e.g., ASP.NET, Django, Ruby on Rails, Code Igniter, Apache Struts, JavaServer Faces). In Web MVC (which differs somewhat from traditional MVC), a model encapsulates application data in a way that is independent of how that data is rendered by the application. The view accepts some number of models as input and transforms them into appropriate output that will be sent to the browser. The controller connects the models and views, typically by gathering model data and sending it to the view for rendering. The view accepts data as input and produces a string as output, which may include information about its type (e.g., HTML, XML or JSON). After being manipulated by some post-processing filters, the string is sent directly to the browser. Because the view's output is treated as an opaque string, it is difficult for the framework to reason about the structure of the content. These views are complicated by the need to provide both Ajax and non-Ajax versions of a site [15]. In ASP.NET, MVC is implemented as shown in Figure 1(a).

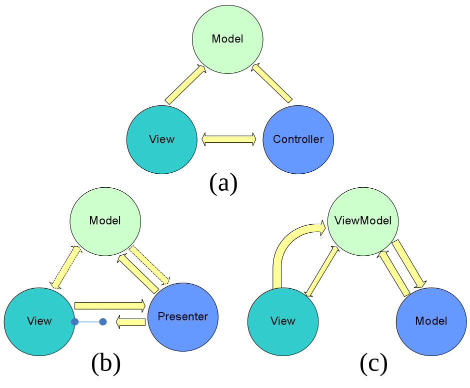

Figure 1: MVC Architectures: (a) Model-View-Controller; (b) Model-View-Presenter; (c) Model-View-ViewModel

In addition to managing complexity, the MVC pattern makes it easier to test applications than it is to test Web Forms. For example, with Web Forms a single class is used both to display output and to respond to user input, which when testing has to be instantiated along with all its child controls and additional dependent classes. Moreover, tests in Web Forms require a Web server. In contrast, the MVC framework decouples the components and makes heavy use of interfaces, which makes it possible to test individual components in isolation from the rest of the framework. The loose coupling between the three main components of an MVC application also promotes parallel development. For instance, one developer can work on the view, a second developer can work on the controller logic, and a third developer can focus on the business logic in the model.

MVC requires an initial solution design at the outset of a project prior to solution development. This design is fundamental to the model of the solution and must be correct to build the solution, however, the benefits in performance are large. First, MVC allows the developer to have full control over the rendered HTML. This means no View State and allows the user to use lower-level HTML, leading to a cleaner HTML render, avoiding standard outputs that use more bandwidth and have browser compatibility issues. Being able to render low level HTML increases the flexibility available to the developer improving the end solution and saving development time. MVC2[1] techniques were built to be interoperable with JQuery[2], this promotes the ability to deliver a richer view (i.e. client-side interface). With more functionality on the client-side there is a requirement for data exchanges to the server. Further, with MVC there can be multiple forms on each page, which means that where there are tasks on the server we can use form actions to only do a postback on that single form. As this only requires the postback of items that are specifically required there is substantial performance benefit.

Precursors to MVC were the Seeheim Model [30] (see Figure 2a) and the Arch Model [31], which was a refinement of the Seeheim Model (see Figure 2b). A variant of MVC is the Model-View-Presenter (MVP) pattern [16], (as shown in Figure 1b). In MVP, the Presenter has the same responsibilities as MVC's Controller, acting as a mediator between the view and the model. It receives user input requests from the view and evokes changes on the model in response. The Presenter is able to query the view for data but to enforce SoC, the presenter is decoupled from the view by holding a reference to the view's interface rather than its implementation [17] [18]. Fowler describes a different approach for achieving SoC called the Presentation Model [19]. This pattern is similar to MVP in that it separates a view from its behavior and state and introduces a PresentationModel that is an abstraction of a view.

A further variant of MVC is the Model-View-ViewModel (MVVM) design pattern for Microsoft's Windows Presentation Format (WPF) [20], as shown in Figure 1c. The MVVM pattern is a more specific version of the Presentation Model pattern [27]. MVVM also separates the view from the logic. However, unlike Controllers in MVC (Figure 1a) and Presenters in MVP (Figure 1b), an MVVM ViewModel has no awareness that a view even exists [21]. The ViewModel is an abstraction of the view but does not need a reference to the view like MVP does. The view uses the ViewModel as a data context and binds properties to fields in the ViewModel, providing a very loosely coupled design. This separation allows a graphics design team to focus on the view while a software development team can focus on implementing a stable and good ViewModel. As the ViewModel does not have to have a reference to the view, the logic can be tested without the view and it is also easy to make different views for GUI evaluations and compare them without changing the ViewModel. Note, Esposito and Saltarello [22] regard MVVM as the same pattern as Fowler's Presentation Model.

Zeller and Felton [15] introduce a stateless, framework-agnostic web application development style, which they call SVC (Selector-based View Composition) (SVC). With SVC, a developer defines a web page as a series of transformations on an initial state where each transformation consists of a *selector* (used to select parts of the page) and an *action* (used to modify content matched by the selector). SVC applies these transformations on either the client or the server to generate the complete web page. The authors contend this approach has two advantages: (i) SVC can automatically add Ajax support to sites, allowing developers to write interactive web applications without writing any JavaScript; (ii) developers can reason about the structure of the page and produce code to exploit that structure, increasing the power and reducing the complexity of code that manipulates the page's content.

García Izquierdo and Asensio [28] propose a double model that effectively separates the work of the web designer from the

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[1] Microsoft's second implementation of the Model-View-Controller Architecture.
[2] JQuery is a cross-browser JavaScript library designed to simplify and enhance the client-side scripting of HTML.

work of the software developer, as shown in Figure 2c. Li and Chou [29] present the design of a multimodal interface that divides the multimodal functions into layers and recursively applies the MVC decomposition to integrate these functions with interpretation. The authors argue that this leads to an extensible multimodal dialogue framework built upon only a few core constructs.

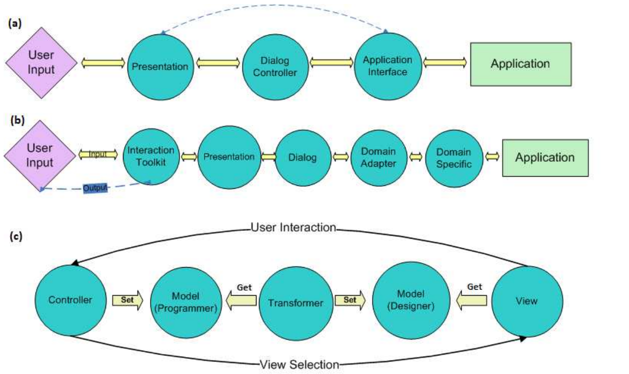

Figure 2: (a) Seehim Model; (b) Arch Model; (c) Double Model

There are many other approaches to SoC. Presentation-Abstraction-Control (PAC) [23] in which the UI is formed by a tree of agents for each of the three components (Presentation, Abstraction and Control). In Naked Objects [24], a UI is automatically generated from the model by analyzing the model interface using Java's reflection features. As a result, the model must contain all of the application domain logic. On the other hand, in Visual Proxy [25] the model/view separation is abandoned. While the model and view objects are still separate objects, they are implemented within the scope of a single class. The model and view layers are tightly coupled, rather than decoupled. Holub argues that the purpose of a model is to provide services to the view and if they are separated, the model cannot provide all the services for the view.

IV. Technology Platform

CC Technology has a proven history in the use of Microsoft development tools and therefore it was logical to deliver the original system using ASP.NET Web Forms which was the prevalent Microsoft technology in 2004 when the solution was implemented. The Microsoft Ajax Control Toolkit was implemented to limit the amount of postbacks needed to the server. For database connections a custom ADO.NET entity generator connects to a SQL Server database. This platform has worked well, the technologies interact successfully and developers can develop and maintain a web-based solution without having detailed low-level web knowledge.

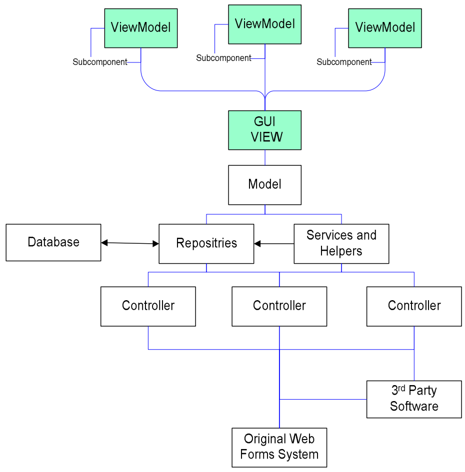

Figure 3: New MVDVM Architecture

The new component-based approach uses what we have named an MVDVM (Model-View-DynamicViewModel) architecture. The MVDVM architecture is an MVC-based architecture that is heavily influenced by the MVVM (Model View View-Model) architecture, as Figure 3 shows. It contains a set of view-models and handles each one in a similar way. The major difference between the MVVM and MVDVM architectures is that the latter model dynamically builds and persists the view-models at runtime rather than having statically designed view-models, as in MVVM. The architecture is extremely powerful because of the ability to have encapsulated components (view-models) that persist themselves, thereby requiring no code to save and load components, producing an architecture that is extremely extensible. Each component that is added has no dependency or relationship to other components, allowing a separation of concerns and removing the risk of breaking other components when introducing new ones. By having this ability the only development that has to occur is writing new view-models. The view-models' logic and behaviour are captured in the view, which means that they can be updated or created without the need to compile. This in turn allows role-based logic to be performed on smaller subcomponents than would usually be possible. The view-models can be any user control such as text boxes, drop down lists or more complex composite user controls with contained business logic. A component can also seamlessly interact with other non-MVDVM systems and other databases meaning that saving and loading is not limited to a component's own data. The architecture allows the user to

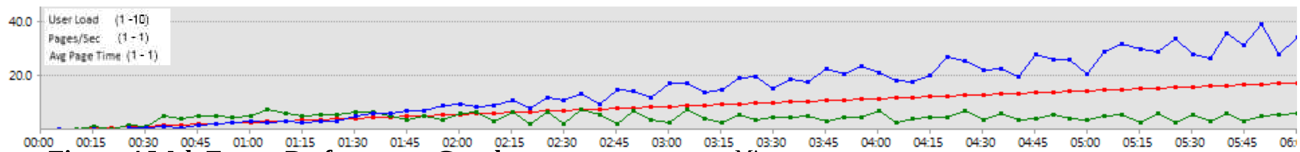

Figure 4 Web Forms Performance Graph

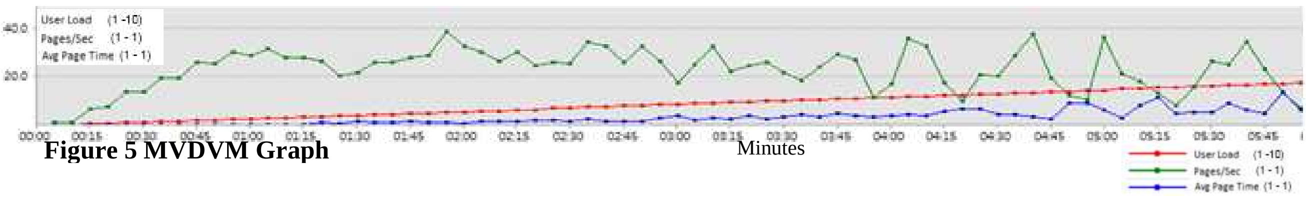

Figure 5 MVDVM Graph

specify at runtime HTML to render each individual component to multiple document types to allow hard copies of forms. The difference between what the MVDVM architecture offers from other form designer tools and architectures is that it has no bounds or limitations as to what the components can do, other than what the .NET framework offers. The entire framework is self contained, which allows seamless integration with other architectures and systems. The scalability of controls and ease of extensibility make this architecture a good alternative to the web forms implementation.

The architecture is also advanced through use of client-side logic in JQuery components. As JQuery is now part of Microsoft's release with MVC, the integration in the development environment is seamless and therefore easy to use. Having JQuery allows quicker and more efficient behaviour and postbacks than was previously possible.

The final major change to the technologies used was the move away from the company's own entity generation tool for Database access. A repository pattern was implemented into the software so that at any time we could change the database connection technique without consequence. Initially Microsoft's Entity framework was evaluated. However, due to the early version of the software it was decided that the more established and Nhibernate model should be used.

V. TEST PLATFORM

To ensure accurate measurement of the performance of the original Web Forms and the new MVDVM architectures a common set of tests was developed. The tests were performed under the same situations and with the same metrics. A number of standard web pages in each system were tested with single users and also with increasing number of users (load testing). Both tests resulted in page load speed, number of pages per second and database information being gained.

An accurate and consistent set of results were achieved by running a dedicated environment comprising a LAN, a test client containing load testing software, a web server running an instance of the forms software and a database server. To ensure that hardware performance was not compromising the results the memory and CPU usage were monitored on each computer to ensure they stayed within 60% utilization.

Visual Studio 2010 Ultimate Edition Load testing suite was used. This was chosen as it generated relevant graph and data information and was developed by Microsoft who supplied the development techniques being used

VI. SERVER PERFORMANCE

Individual tests were carried out on both architectures. The test cases used pages on each system with the same function and content, albeit with different implementations.

Area	Page Load Time (Average(seconds))	Time to receive first byte (seconds)
Opening a form	0.413	0.145
Financial page	1.361	1.113
Standard page	0.372	0.030
Saving form (including redirect)	0.635	0.588

Figure 6: MVDVM Page Times (in seconds)

Area	Page Load Time (Average)	Time to receive first byte
Opening a form	1.526	0.940
Financial page	1.556	0.599
Standard page	1.744	0.668
Saving form (including redirect)	0.656	0.114

Figure 7: Web Forms Page Times (in seconds)

The performance results in Figures 6 and 7 clearly show that the MVDVM architecture is faster on an individual load. It must be noted that there is a change in logic between the two solutions. The MVDVM application forms bind the models dynamically on every page load. Thus for every load and page exit data is read and written to the database. The Web Forms solution does not write to the database until a user hits submit, which may be after n*page visits. This is reflected in the load times for the financial page. This is a complex page where hundreds of requests are being sent to the database. Although the MVDVM is still faster for this page the number of connections is the reason the difference is not as substantial.

The single tests are a good indication of the efficiency of each architecture. However, live systems are characterized by uneven and high loads. For a production system the number of concurrent users could be in the hundreds.

Load testing was carried out on the same pages as noted in Figures 6 and 7. The resulting graphs are shown in Figures 4 and 5. The graphs show that the MVDVM out performs the Web Forms by a large margin. Each test began with one simulated user running a web test. This proceeded through each page in a defined order. Every ten seconds five more users were added to the current list of users. The results are shown in Figures 8 and 9 for each run of the test.

	Range on graph	Min	Max	Avg
User load (num users)	1000	1	176	88
Pages/sec (seconds)	10	0	7.8	4.53
Ave page load (seconds)	100	0.25	39.7	16.2

Figure 8: Web Form Graph Averages

	Range on graph	Min	Max	Avg
User load (num users)	1000	1	176	88
Pages/sec (seconds)	10	1	39	23
Ave page load (seconds)	100	0.0.35	13.9	3.04

Figure 9: MVDVM Graph Averages

Figure 5 shows that the average page load speed increases in relation to the number of users. As the number of users increases the number of pages loaded per second severely decreases.

The MVDVM results are quite different. The pages per second start very high and remain so throughout the test. At the points of lowest performance where the graph dips the response times are still acceptable for an end-user. These dips occur where a large number of users (more than 100 users) tried to call an action concurrently. This is a quirk of using a test suite and the phenomena are unlikely to occur in production at this usage level. Usually in this case it would be the hardware performance that would fail or cause performance issues. The page load speed also remains very constant and, apart from the dips, remains lower than one second for each page throughout the test.

VII. CLIENT PERFORMANCE

Client performance can be characterized by the download time for the pages and of the rendering time.

Figure 10 shows the details for an average page in the performance tests. The details emphasize the earlier comments about how large or inefficient the HTML can become due to the View State and the ASP.NET automatically generating HTML. The Javascript size is also very high as the previous forms were rendering pure Javascript. The Javascript was not compressed, which is also inefficient, causing larger page sizes.

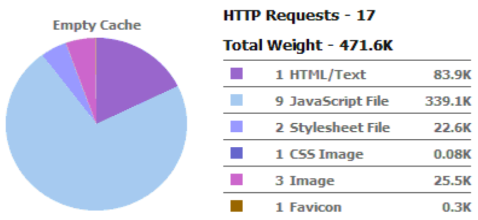

Figure 10: Web Forms Average Page Detail

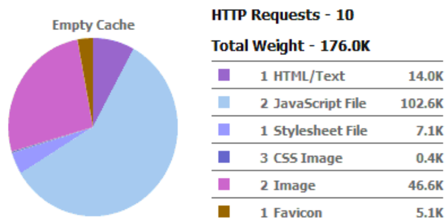

Figure 11: MVDVM Average Page Detail

The detail of an average MVDVM page in Figure 11 shows that the page size is smaller. The ability to control the HTML and use JQuery to write minimal amounts of code has resulted in a reduction of the page size from 471k to 176k.

The HTML code in the Web Forms output is controlled by ASP.NET and is not standards-compliant, whereas the MVDVM output is dependant upon and controlled by the developer. This control allows the developer to address client performance issues, which Google describe as 5 performance factors [26]: (i) optimizing caching, (ii) minimizing round-trip times, (iii) minimizing request overhead, (iv) minimizing payload size, (v) optimizing browser rendering. The MVDVM solution improves on each of these factors through the communications and page content.

VIII. DATABASE PERFORMANCE

A statistical analysis of the database transaction times compares the performance between the two architectures. The results are split into two parts: using all the data and using only the data that had duration times greater than zero. The results show every query that happened when the tests were run. Because of this there is many light weight queries that had durations of less than 1 ms, these show as 0ms durations.

A. Analysis using all of the data

Using the Web Forms architecture, reads were performed an average of 48.60 times (SD = 1061.05) with a range of 0 to 32,350 whereas using the MVDVM architecture reads were performed an average of 24.54 times (SD = 283.12) with a range of 0 to 6,440. Using the Web Forms architecture writes were performed an average of 0.0145 times (SD = 0.30) with a range of 0 to 17 whereas using the MVDVM architecture writes were performed an average of 0.0093 times (SD = 0.20) with a range of 0 to 22.

Transactions using the Web Forms architecture had an average duration of 1.01 ms (SD = 48.06) with a range of 0 to 9,157 ms where transactions using the MVDVM architecture had an average duration of 0.1223 ms (SD = 2.04) with a range of 0 to 93 ms.

Paired samples t-tests indicated that the MVDVM architecture performed significantly less read commands ($t(65,527) = 5.607$, $p < 0.000$), significantly less write commands ($t(65,527) = 3.733$, $p < 0.000$) and took significantly less time to perform read and write commands ($t(65,527) = 4.729$, $p < 0.000$) than the Web Forms architecture.

B. Analysis using only the data that had duration tims greater than zero

Using the Web Forms architecture reads were performed an average of 2,047.19 times (SD = 6,955.46) with a range of 0 to 32,350 whereas using the MVDVM architecture reads were performed an average of 990.40 times (SD = 1,664.82) with a range of 0 to 6,440. Using the Web Forms architecture writes were performed an average of 0.4087 times (SD = 1.63) with a range of 0 to 17 whereas using the MVDVM architecture writes were performed an average of 0.1095 times (SD = 0.879) with a range of 0 to 22.

Transactions using the Web Forms architecture had an average duration of 47.08 ms (SD = 324.75) with a range of 0 to 9,157 ms where transactions using the MVDVM architecture had an average duration of 5.70 ms (SD = 12.74) with a range of 0 to 93 ms.

Paired samples t-tests indicated that the MVDVM architecture performed significantly less read commands (t(1,406) = 5.339, p < 0.000), significantly less write commands (t(1,406) = 5.996, p < 0.000) and took significantly less time to perform read and write commands (t(1,406) = 4.764, p < 0.000) than the Web Forms architecture.

IX. Conclusion

This paper has presented a comparison of two web architectures used in process-heavy web applications. The first, Web Forms, is an established standard Microsoft approach and the second, MVC, is a relatively recent approach gaining some acceptance in the development community. This paper has proposed a variant of MVC, called MVDVM and has demonstrated that this architecture delivers performance benefit over the traditional Web Forms based architecture. The performance has been shown to be improved at the database, web server and client. With each of the tests it has been shown to be faster or more efficient.

The results of the case study have proven that under these circumstances the performance benefits are large and the move to an MVC architecture are justified.

Future work will involve the investigation and implementation of multilingual functionality within the MVDVM architecture.

X. Acknowledgements

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A Team Registration System to Support Tracking Training and Enhancing Performance

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Abstract - *Programming competitions are varied and range from high school to college level. In addition, competitive learning, in general, is increasingly used in courses to foster improved focus on improvement. Such competitions require management tools; however, we also wish to support tracking of individual and team performance across such competitions. We have developed a vision for a suite of tools to provide such a support system. This paper discusses the first such tool: a system for competition registration. We review the database structure and how it supports our vision for tracking individuals and teams. We then review our implementation including how it was built on top of the Joomla! content management system. Finally, some lessons learned are offered and some thoughts on our next tool are given.*

Keywords: Internet Delivery and Applications, Web Interfaces to Databases, Web-based Management Tool, Web-based Training

1 Introduction

Programming competitions have been increasing in number. While still the largest and most prestigious, the Association for Computing Machinery's International Collegiate Programming Contest (ICPC) is now joined by Google's CodeJam, TopCoder's TopCoder Open and others. In fact, the past year saw Facebook join the mix with its Hacker Cup competition. In addition, competitive learning is becoming an increasingly used educational tool [1][2]. Teachers and professors use competitions within their classes to drive performance. Indeed, the ICPC now holds a Competitive Learning Institute Workshop each year at its World Finals competition. During this event, speakers discuss how they have brought competitive learning to some area and review systems built to support such purposes.

Our university has an active student competitive programming team. The team competes in the ICPC regional competitions and, usually, in the ICPC World Finals. In addition, team members have represented the university in the IEEE SoutheastCon software competition and they have also participated on an individual basis in Google, TopCoder and Facebook competitions.

The team also holds an annual high school programming competition. Each year, area high schools come to the university's campus and participate in an ICPC-like competition. While a fun activity, it has a background goal of recruiting the best students to our university.

2 Vision

In order to support the training of our team, we have built a number of support tools over the years [3][4][5]. Whether supporting team designation, tracking training problems or program submission, each has acted as an Internet-hosted knowledge-based approach to a given task. Each tool has succeeded very well in its individual goals. However, the end result was a disjointed set of data across a series of very different tools.

Given these issues, we have spent the past year formulating a vision for training and tracking performance of individuals across their development. Our vision is actually two-fold: first, we wish to track performance of members of our competitive programming team, and, second, we wish to track performance of individuals in classes. Regarding the latter, we teach a class on “problem solving techniques” where students get to apply common computer science algorithms to solving problems (much like a programming contest, actually).

The goals for both aspects of the vision are the same. Therefore, we have planned a vision around the idea of individual training performance and team formation. A number of tools are planned to support this vision. Each member's performance will be tracked including competitions before they qualify for the team (whether in a class such as the problem solving course or as a participant in our high school programming competition) and throughout their career on the team itself.

The first of these tools, a competition registration system, is the focus of this paper. This system tracks participating teams and team members as well as the advisors and schools. In addition, payment of contest registration fees can be tracked and reports generated. Our planned reports include a list of registered teams, lists of participating and paid schools, a list of t-shirts by size (for ordering), and a list of t-shirts per team (for distributing). Other reports, of course, are also possible and are only limited by the data itself.

When creating a registration system, we had a number of goals in mind:

- Members should be tracked over time.
- It should support real competitions as well as team practices.
- It should track teams at each competition.

To address these goals, team members must have a single record that was used competition-to-competition. Otherwise, performance could not be tracked properly. In addition, we need the system to handle team-orientation competitions where the advisor would register teams (such as our high school competition) and also to handle individuals registering themselves and their team within a single competition (such as within our weekly team practices).

Previously, we chose a content management system to handle our web sites. We use the Joomla! system and run web sites for the team and the high school competition as well as an internal site to support the university team's training [6]. Accordingly, we desired to incorporate the registration system into both the high school site and the training site (to support both). We also wish to use it to assist in tracking performance within the problem solving course we teach. Therefore, in order to provide a seamless experience, we developed our approach within the Joomla! framework.

3 Design

Joomla! uses a model-view-controller paradigm and extensions can be built that fit into this paradigm. Specifically, Joomla! supports three types of extensions: components, modules and plug-ins. Modules are small units that add simple content to a web site (for example, a module might display the five most recent messages from a Twitter account). Plug-ins are non-visible extensions that provide additional functionality to the web site (such as a new authentication scheme). Components are the most complex type of extension and use the full model-view-controller paradigm to provide an enhanced add-on to the web site.

In order to provide a registration extension, we developed it as a component. An important aspect of any database driven application is the database design itself. Indeed, it is very important to get this aspect correct or one will undoubtedly need to return to this area and make adjustments in the future. Therefore, we spent ample time up-front making sure the database design was complete and held the proper relationships.

Figure 1 shows the database design used. In this design, contests refer to conceptual organizations (such as the ICPC or Google CodeJam) while competitions are specific rounds of a contest (for example, the 2011 World Finals). Competitions include all data about a specific round, including the name and date as well as information about when registration opens and closes and how payment should be handled. In addition, the competition tracks the maximum number of people allowed per team within that round. We also provide a field that stores whether teams are allowed to create their own names. Some contests allow the use of creative team names while others simply name teams after the school or organization with a numeric count added to each to distinguish the teams (e.g. “JFK High School #1”).

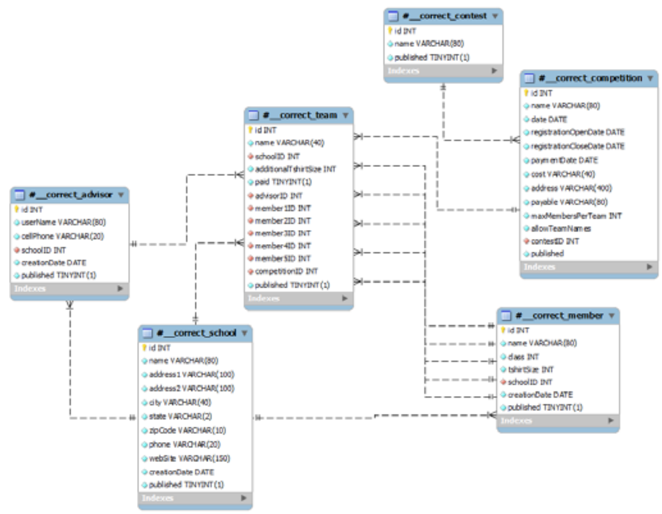

Figure 1. Database Design.

We then store team advisors and team members, each in their own table, independently from the competition of contest. This allows us to have a single record for an individual and track performance of that individual across competitions. Since the team advisors are responsible for registering their teams, we leverage our framework by providing for each a Joomla! account that is used to sign into the web site. Therefore, we chose not to duplicate basic user account data into this table; we only store additional fields in our advisor table (such as an emergency contact phone number).

Similar to the advisors and team members, school data are stored in a separate table, which allows both advisors and team members the capability to move between schools (which can occur). This also allows the tracking of all schools that have ever registered, which can be useful for future opportunities and also to be aware of a new attendee. For example, we have recognized new schools at our competitions with a special prize.

Finally, the team table stores all data about a given team within a given competition. Note that the team table can support up to a maximum of five person teams (although the actual maximum allowed by the competition is given in the competition table). Across all known programming contests, teams have consisted of at most three or four people so this table should satisfactorily address all needs.

Note that each database table has a "published" field. This is a de facto Joomla! standard that represents whether a given table row should be considered in database queries. Table rows can easily be toggled to "unpublish" its database, meaning that it should not be considered in database queries. By maintaining this standard, it provides the registration system the ability to hide data as needed or desired. For example, if a team fails to show for a competition, their table row can be unpublished as opposed to deleted. This allows interest in the competition to be tracked and potentially used for future invitations.

4 Implementation

As mentioned earlier, Joomla! uses a model-view-controller paradigm built within PHP [7] and, typically, using MySQL as a database infrastructure [8]. Each component is linked with the other two through a naming convention that includes the component name as well as the function of the component itself. For example, our system is named CORRECT (Component-based Online Registration and Reporting Environment for Contests and Tournaments). Therefore, for a list of teams within the system we have a controller named CorrectControllerTeams, a model named CorrectModelTeams, and a view named CorrectViewTeams.

The model represents the data and, typically, interacts with the database system (e.g. MySQL), performing queries and updates as necessary. The controller is responsible for coordinating the tasks operating on the data including editing data or triggering the view to display. The view, of course, coordinates rendering of the data itself and, in Joomla!, handles producing HTML for display in the web browser.

The Joomla! framework provides an administrative interface (referred to as the "back-end") in addition to the web site view itself (referred to as the "front-end"). In order to create a component, the system must provide both interfaces. The back-end will support administrative functions such as editing of the database tables and running reports of the data.

Figure 2 shows the back-end for the component we built. The default view is a control panel that allows the administrator to access all other elements of the component back-end. Tables can be edited or new rows added. In addition, filters are available within each table view to easily select a subset of the table rows. For example, the list of teams can be filtered to those in a specific competition.

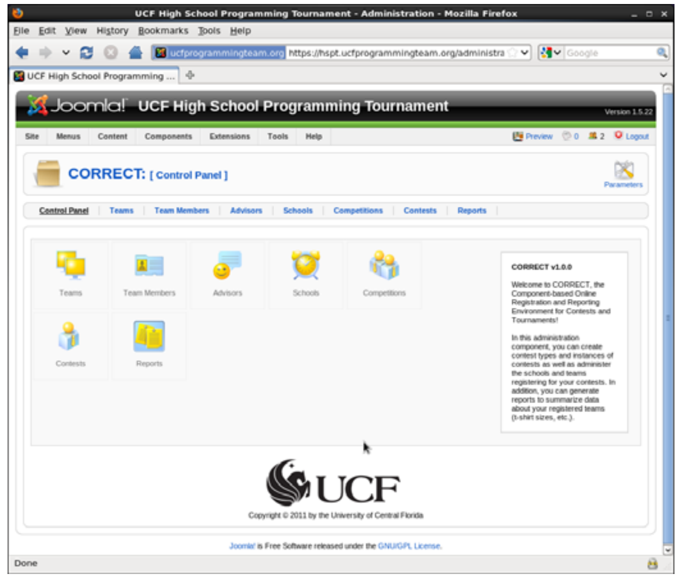

Figure 2. The CORRECT Back-end Control Panel.

Figure 3 shows the sub-panel for the reports section of the component back-end. Here, the administrator can select different reports to be run and their results displayed. For example, Figure 4 shows a t-shirt report per school. Within a competition that gives free t-shirts to teams, this report would allow somebody to collect the t-shirts necessary for a given school.

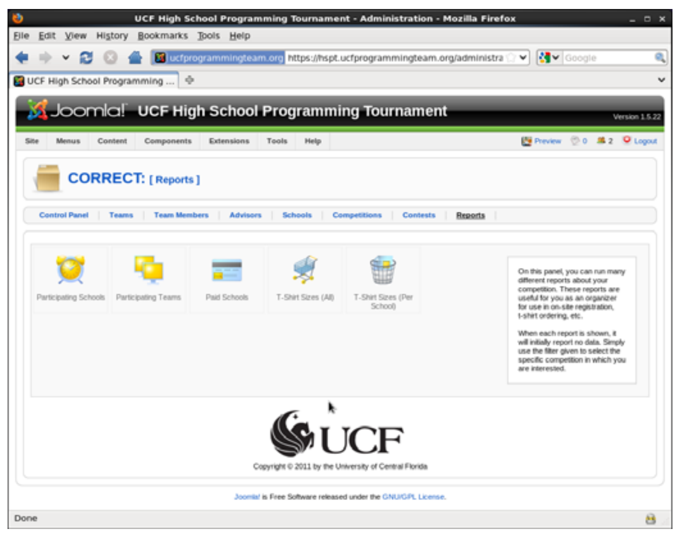

Figure 3. The CORRECT Back-end Reports Sub-panel.

While having the back-end available for data editing is beneficial, developing it also provides a method for exercising the database design before tackling the front-end and its look-and-feel and usability questions. This was a very valuable idea as we did improve the database design during this stage of development as various problems were found. For example, we initially did not separate the advisor data out into a separate table. However, this caused issues when handling an advisor that may have switched schools; therefore, we determined that this did need addressing. Exploring these issues before the front-end was completed saved a large amount of wasted effort.

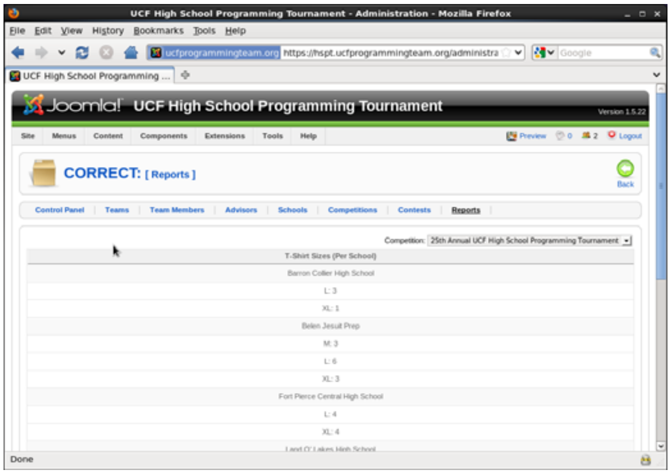

Figure 4. Example of T-shirts Per Team Report.

Once the back-end was in place and we were satisfied, only then did we approach the front-end design. This also had the benefit that it was a fairly straight-forward process. We decided to adopt the wizard interface style as opposed to any kind of menu-oriented system. We felt that the guided nature of registration would work well and provide a simpler experience for the advisors using the system. The goal of the registration system user experience was to allow a user (most often, a team advisor) to login and then easily add team members and form teams for a specific competition.

Once advisors have an account (which we use standard Joomla! capabilities to support), they proceed to register team members and teams. CORRECT provides three alternatives here. The most common use allows a web site administrator to start registration directly at Step 1 of the process (discussed next). However, in a multi-round contest that may desire parallel registration, the system supports a preliminary step where the user would select in which competition is being registered. In contrast, CORRECT also supports a mode where the competition and adivsor is already selected. This mode allows a use where teams simply directly register themselves (for example, we use this for our team practices where students can "register" how they are teamed that week; this allows us to track performance of the teams and detect trends on particular individuals working together).

For the remainder of this paper, we follow the most common, four-step registration process. The first step simply requests data about the advisor. For now, we include only an emergency cell phone number and which school he/she is connected. Figure 5 shows this interface. If an advisor is new, the school pull-down option will be left on "Select School"; however, it will be pre-filled to the existing school if the advisor is already known to the system.

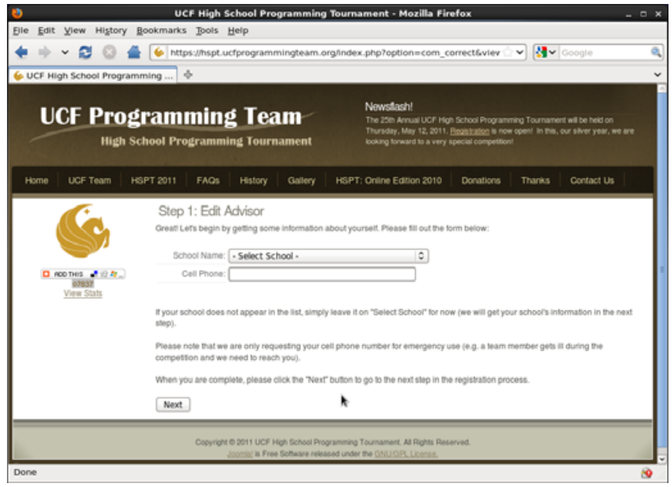

Figure 5. Advisor Registration.

Once this step is completed, the advisor then continues to Step 2, which collects data about the school itself. Figure 6 shows this interface. We include basic information about the school (such as address and phone) but also request "modern" information such as the school web site. The latter helps support us learning more about the school and we often use it to get the school's logo, which is used in other contest-related materials.

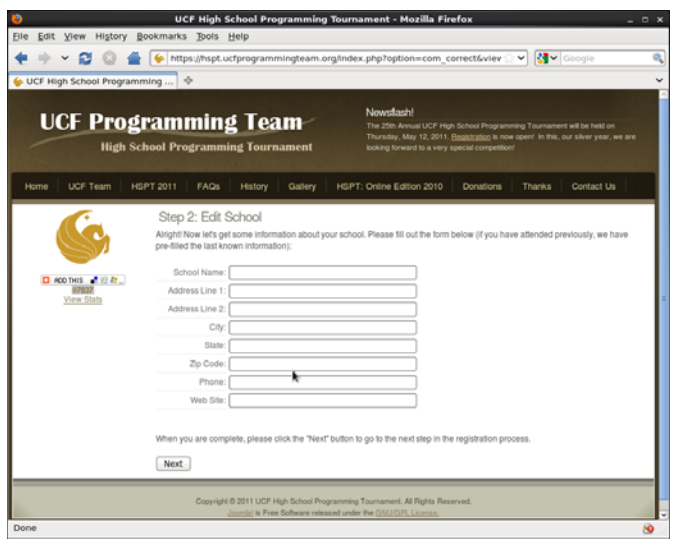

Figure 6. School Registration.

After the school data is collected, the advisor is then requested to enter team member information. The current list of known team members for that school is displayed. The advisor can update and enter new team members. Figure 7 shows the entry for adding/updating a team member. We track current class (Freshman, Sophomore, Junior, Senior, Grad) and t-shirt size.

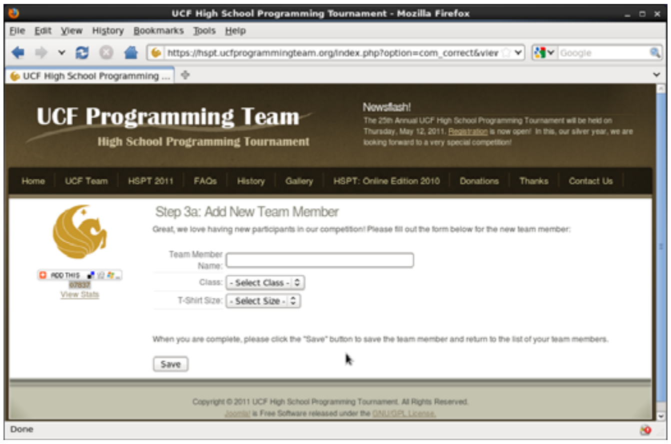

Figure 7. Team Member Registration.

As discussed earlier, tracking individual team members allows us to track performance over years. This works well for our own team members, but also provides us the opportunity to have known performance data related to new incoming freshman at the university.

Once team members are entered, the advisor can then enter teams for the competition. Figure 8 shows this interface. The form displayed is dynamically created based upon the maximum number of team members allowed per team. However, this is a maximum and only a single team member must be selected per team.

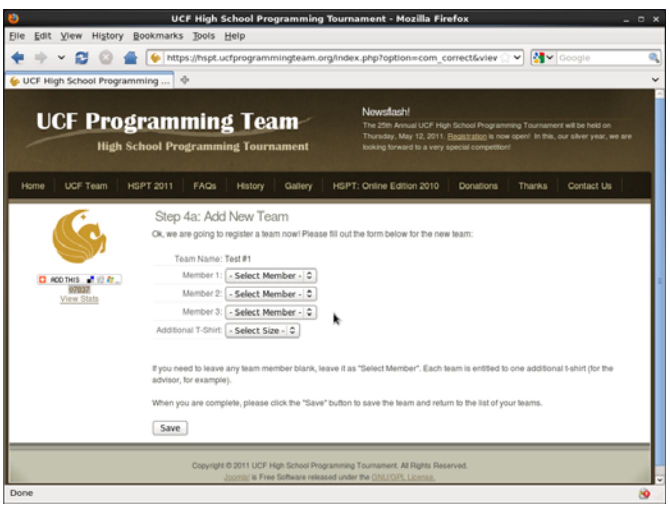

Figure 8. Team Registration.

Once the advisor completes forming teams from team members, the data is stored and a final screen is presented (not shown). This page provides a reminder about the competition's registration cost, where to send payment and also the due date for that payment.

Since each step allows both adding and editing of data, an advisor can actually step through this process multiple times if desired. This allows data to be changed (up until the registration deadline, of course) and also additional teams could be registered.

5 Discussion

During this development, we found that database design was very important and should be given due consideration. In fact, despite the design we now have, we are considering splitting the team table into a strict team formation concept (these team members make up this team) and a participation concept (this team competed in this competition). Doing so will allow us to avoid data duplication and make it easier to track a particular team across multiple competitions.

The CORRECT system has been in use for just a short while, but indications are that it is easy-to-use and works well in achieving its goals. We are particularly happy with the way it allows us to track team formation and performance. As we progress our younger team members through their competitive careers, we will find this tool invaluable for tracking them.

6 Conclusions

Having the system described in this paper satisfies the first step in our training performance vision. Team members and teams can be registered and the contest organizers have tools to help execute the competition itself. Joomla! has acted as a great vehicle for our web sites and having this first component provides us a common framework in which to operate.

In the future we hope to build upon the CORRECT registration system. Our next step will be to complete a program submission and response system that ties into the registration system. This will provide us the next piece in tracking team performance. Once we can know how a team performs against specific problems, we can truly start to see performance trends. This will aid us in future training of both the individuals involved and the teams they form.).

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An Experimental Study and Analysis of Crowds based Anonymity

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Abstract— *Crowds provides probable innocence in the face of large number of attackers. In this paper, we present the experimental results of the behavior of Crowds in a dense network. We begin by providing a brief description about Crowds followed by the experimental environment in which the simulations were carried out. We then present the results of our simulations and the inferences made out of them. We will also show that the obtained results match the predictions made by others.*

Keywords: Anonymity, Crowds

1. Introduction

Anonymous communication involves communicating without revealing the identity to each other and to the outside world. There are three types of anonymity namely sender anonymity, receiver anonymity and the unlink-ability of sender and receiver. Sender anonymity means that the information about the sender will be hidden while the receiver may not. Receiver anonymity on the other hand hides the information about the receiver. Unlink-ability of sender and receiver refers to the phenomenon that, both the sender and the receiver may be found to involve in communication but cannot be identified as communicating with each other.

Many solutions exist to achieve anonymous communication over a network. They can be broadly classified under three heads:

1) Web Proxies
2) Mix based systems
3) Other Communication systems.

In a proxy based system, additional trusted third parties called proxy remain in between the sender and receiver. Requests and responses go through this proxy, by which the identities of the communicating parties are hidden. Some of the available proxies for anonymous web browsing include: Anonymizer [1], Proxify.com [2], and Proxy.org [3].

Mix based system was introduced by David Chaum in 1981 [4]. A mix in short is an enhanced proxy employing public key cryptography to achieve anonymity. It hides the sender's identity by cryptographically altering the messages being exchanged. Mixes utilize techniques such as buffering, and circulation of dummy traffic during idle time, in order to preclude an attacker from retrieving information about the nature and the parties involved in a communication.

Other prominent communication systems include Onion Routing [5] and Crowds. In Onion Routing, the sender builds a virtual circuit by determining a path between it and the receiver using layered objects called "Onions". Every layer in the layered object contains information about the session key and next address of the node in the path. These onions that travel down the path are unwrapped using the session keys at each node. When the layers are fully removed, the session keys are destroyed.

The final system of interest and also has been the subject of analysis in this literature is the Crowd. They operate by forming a large group of users who may be geographically distributed. Crowds try to hide the actions of an individual with in that group by forwarding the requests randomly between the members before sending it to the final destination. The rest of the paper is organized as follows - section 2 provides a brief description about crowds, section 3 describes the relevant literature review with respect to crowds research, section 4 describes our simulation environment, section 5 presents the experimental results and the inferences drawn from the data, and then we end this paper with conclusion and future direction our research will take.

2. Crowds: A Brief Description

Crowds provide a mechanism for anonymous web browsing. Though there were other systems such as mix nets and DC-Nets [6] to accomplish the same, crowds were preferred because of the low latency and less computational overhead.

Every member of the crowd runs a process named jondos that registers itself with the central server called blender. Every jondos knows about every other jondos in this architecture. The blender is responsible for the distribution of symmetric keys between every pair (jondos). When a request for a web page begins at one of the nodes, the jondos running in the originator node forwards the request to one of the randomly chosen node by encrypting the message with the corresponding symmetric key. The latter node then either forward the same to another randomly selected node or to the web sever. The decision is taken based on forwarding probability with which it operates.

When a jondos receives a message, it does limited processing to preclude certain attacks and continues to transmit the message. The request and the response of the message follow the same virtual path. These paths are torn down and new paths are constructed on the regular basis whenever

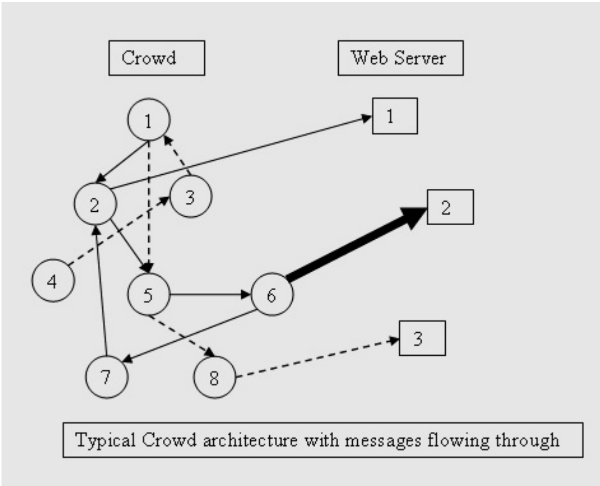

Fig. 1: A Typical Crowd Architecture

there is a change of member count with in the crowd or there is a node failure. A jondos cannot decide by itself, if the request originated from the preceding node or the one before it.

In this architecture, the adversary can observe the server receiving messages. He/she cannot determine the source of the message. Similarly, when the server transmits the response back in the same path, it is difficult to ascertain the final receiver. In addition, even if the attacker finds that there are clients and servers communicating, he /she cannot find out as to which client talks to which server. Hence the anonymity goals that the crowds achieve include sender anonymity, receiver anonymity and as well as the unlink-ability of sender and receiver. However all of these depend on the kind of attacker that we are talking about while determining the anonymity. For example, there is no sender anonymity against a local eaves dropper and, receiver anonymity against the end server. In addition, none of these anonymity schemes work against a global eavesdropper if the scope of the crowd is only with in LAN. This in turn necessitates spanning the crowd across multiple administrative domains.

3. Literature Review

In this section we will provide brief descriptions and summaries from literature significant in the domain of crowds. Reiter and Rubin [7](Crowds: Anonymity for Web Transactions), were the first to introduce the concept of Crowds. They discussed the ways by which crowds can be formed and operated. Measuring anonymity provided by Crowds by employing the concept of degree of anonymity was also discussed. Finally, it was proved that the expected length of hop count for a message to reach the end server is $\frac{1}{1-P_f}+1$, and the expected participant payload in a crowd of 'n' nodes is bounded by $O(\frac{1}{(1-P_f)^2}\times(1+\frac{1}{n}))$.Here P_f is the forwarding probability. They also provided information regarding design, implementation, security, performance and scalability of the system.

The bounds for the participant payload proposed by Reiter and Rubin was further improved in [8](The cost of becoming anonymous: on the participant payload in Crowds). This paper provides a precise formula that expected payload of a participant also tends to $\frac{1}{1-P_f}+1$. In addition, the authors also showed that participant payload in Crowds is entirely independent of its size which in turn made evident that the Crowds possess good scalability feature.

In [9] Towards measuring anonymity by Claudia, Stephen, Joris and Bart, the author discusses about measuring the degree of anonymity of systems including Crowd through entropy $H(X)$. In this literature, $H(X)$ for Crowds is measured as,

$$\frac{N-p_f(N-C-1)}{N}log_2[\frac{N}{N-p_f(N-C-1)}]+p_f\frac{N-C-1}{N}log_2[\frac{N}{p_f}]$$

Here N, p_f and C are the total number of crowd members, probability of forwarding to the another member, total number of collaborators respectively. This measure is in addition to the suggestion made in [7] where the degree of anonymity is defined as $(1-P_{sender})$ where P_{sender} is the probability assigned by the attacker to a particular user.

Trust plays a major role in deciding your forwarder. Hence Vladimiro, Ehab and Sardaouna in their paper titled [10] Trust in Crowds: probabilistic behavior in anonymity protocols, proposes a Crowds-Trust protocol that uses trust information to achieve the desired level of anonymity. They also derive expressions for different level of anonymity required.

4. Simulation Environment

Network topology consisting of 2500 nodes was generated using Georgia Tech Network Topology Generator (GT-ITM) [11].The output of the same was converted to a understandable format using the utility sgb2alt that accompanies the software. The output comprised of source,destination node and the path length between them. The path length was interpreted as delay in the simulation. This was followed by computing the shortest path between all pairs of nodes using Floyd-Warshall's Algorithm [12]. This is how we generated the network topology:

```
# <method keyword> <number of graphs> [<initial seed>]
# <n> <scale> <edgemethod> <alpha> [<beta>] [<gamma>]
geo 3
2500 2500 3 .03
```

Included here is a portion of the output that was generated by the topology generator:

```
GRAPH (#nodes #edges id uu vv ww xx yy zz):
2500 188354 geo(0,{2500,2500,3,0.030,0.000,0.000}) 2500

VERTICES (index name u v w x y z):
0 0 805 682
```

```
1 1 1134 268
2 2 2181 925
3 3 1670 310
4 4 793 291
5 5 1747 917
6 6 220 945
7 7 183 1775
8 8 1236 2415
```

Crowd network was an overlay network on top of this generated network. Hence although there remain direct connections between two nodes in a Crowd network, the message passes through numerous other nodes in the underlying network before reaching the destination. The delay that was precomputed at the end of topology generation was taken as the delay between the nodes due to the underlying network configuration. The simulations were run for different node count and probabilities. Sampling of nodes that are part of Crowd network for every simulation were generated from the underlying topology using a randomized algorithm.

The simulator that we used, was written in Java and ran on Linux machines. The server (blender) was started and made to wait for a predefined period (specified in the configuration file) to accept connections from the clients (members interested in becoming part of the network). At the end of this registration phase, every member was provided information about itself and every other member. The number of client processes to spawn, depending on the node count, were performed using a batch file. These client processes were ran on different machines. The parameters for the client such as the probability with which to operate, time at which to generate and send messages and as well as the delay incurred in transmitting message to its successive member were specified in the client configuration file. Every scenario was run for 100 iterations and the data (hop count, total delay) collected.

5. Experiments and Outcomes

The simulations were run for member counts ranging from 10 to 1000 with predefined intervals in between this range. The forwarding probability was allowed to vary from 0.1 to 0.95. Measurements were taken after running every scenario for 100 iterations.

The graphical outputs for some of the runs are presented below. The graphs are plotted between the following parameters:

a Count of nodes participating in Crowd and delay incurred in transferring message.
b Count of nodes participating in Crowd and the measured hop count.

Figures 2, 3, 4, 5, and 6 show the message transmission delays incurred in milliseconds between the source and destination when transmitted through the crowds network with varying transmission probability and with crowds composed of different node counts.

The next set of figures shows the hop count a message has to travel before reaching the destination if sent through the crowds network for networks composed of different number

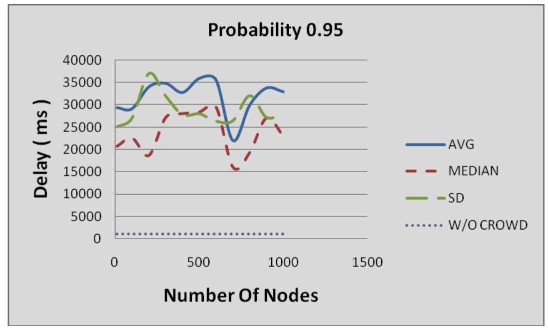

Fig. 2: Transmission delay for probability 0.95

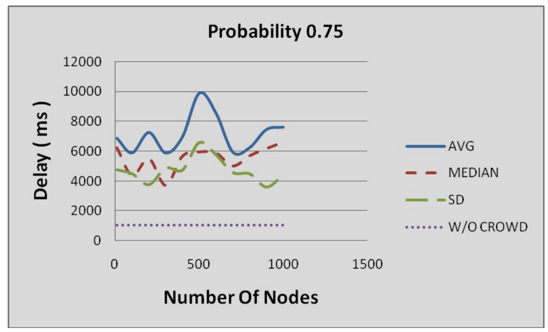

Fig. 3: Transmission delay for probability 0.75

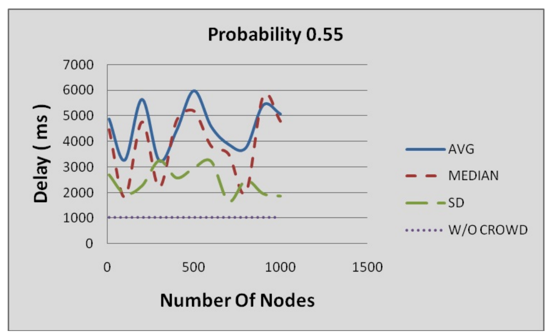

Fig. 4: Transmission delay for probability 0.55

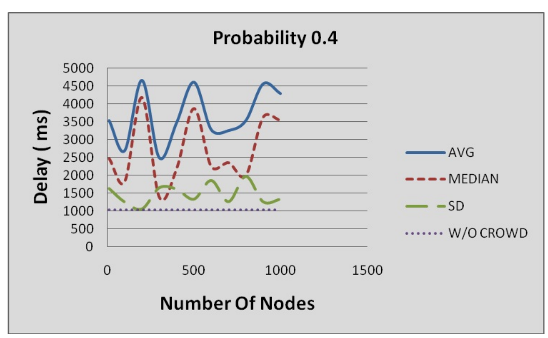

Fig. 5: Transmission delay for probability 0.40

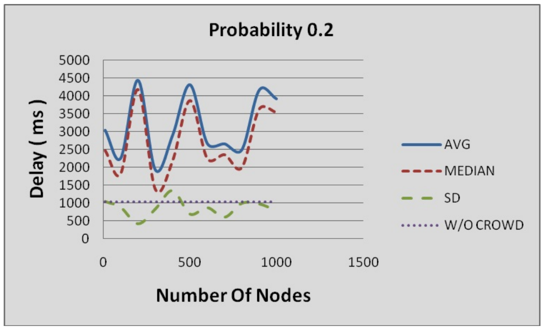

Fig. 6: Transmission delay for probability 0.20

of node counts and with crowd node varying forwarding probabilities.

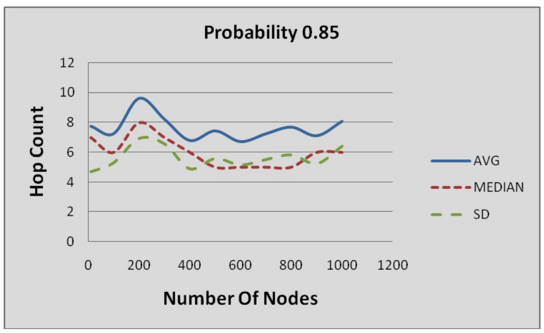

Fig. 7: Number of hops traveled for probability 0.85

Figures 7, 8, 9, 10, and 11 shows the experimental results we collected for the parameters shown. Each experiment was repeated 100 times and the average, median, and the standard deviations have been plotted.

6. Results Interpretation

When the forwarding probability associated with the nodes increased, the average hop count that the message took to reach the destination increased and mostly followed the derived entity $\frac{1}{1-P_f}+1$ except under very high probability. This is clearly evident from the fact that the lesser the likelihood of reaching the target, the more it takes to reach it. Also since the crowds neither generate cover traffic nor increase the work load of CPU by encrypting and decrypting the content, the very slow increase of hop count for huge increase in the member count is beneficial for community adopting this service.

Though the hop count is minimum,the delay is more since the hop count overlooks the underlying nodes in between the member nodes.As the result, this service cannot be adopted for systems that require faster response.The delay in the response may not be acceptable for interacting users even for queries such as the one that provides some location information.

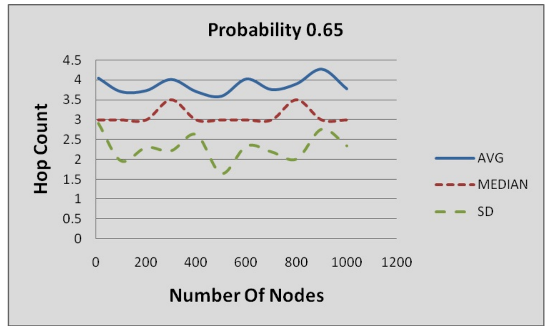

Fig. 8: Number of hops traveled for probability 0.65

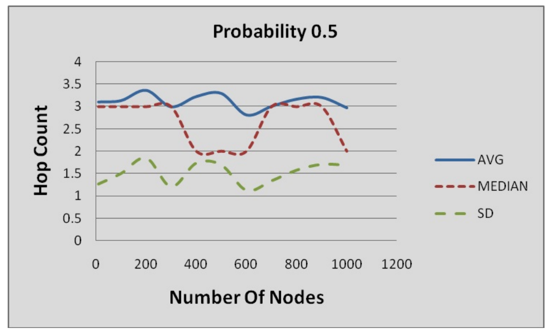

Fig. 9: Number of hops traveled for probability 0.50

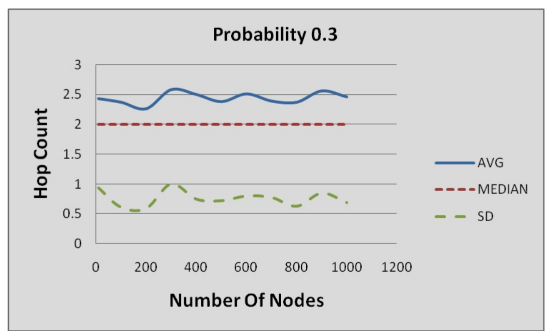

Fig. 10: Number of hops traveled for probability 0.30

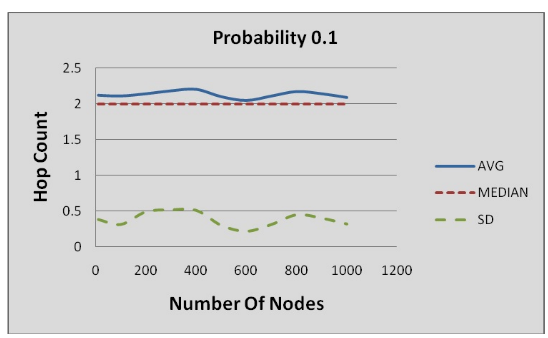

Fig. 11: Number of hops traveled for probability 0.10

The standard deviation of the hop count increased on increasing probability. The standard deviation of the hop count decreased on increasing the number of nodes for the same probability. The hop count increases gradually for every increase in probability except at above 0.9 where there is a huge increase in the hop count.

Crowds neither generate cover traffic nor increase the work load of CPU caused due to a series of encryption and decryption. The encryption and decryption performed using path key is very minimal in Crowds. Hence they are preferred over mix nets and onion routing specially in situations where security of the content is not of prime importance. From the simulations, it is observed that the hop count increases very slowly and reaches the value 8 until the probability value hit 0.85, after which it increases drastically even for the large number of nodes. This seems to be reasonable in the light of the service that it provides.

Although Crowd is meant to provide efficient service, the delay associated in sending a message to the destination is significant. This is evident from the graphs above where delay value is quite high even for minimum node count.

7. Conclusion

This paper provides the experimental results that we carried out to validate the operation of a crowd anonymity network. We have described our simulation strategy and provided the results we got. The results are in line with theoretical predictions made in several of the pioneering work in this field. In the very near future, we are planning on analyzing crowds and the degree of anonymity it provides by incorporating it within a utility function that would include the notion of cost versus degree of anonymity, transmission delay and other relevant parameters. It is still a work in progress and would take some time before we can comment on the strategy here in this paper. We are also planning on using the same utility function to compare other anonymity schemes such as onion routing, mix nets, etc.

Acknowledgment

The authors would like to thank Dr. Richard Newman for providing suggestions on experimental methodology. The CISE systems administrators were also helpful by allowing us run experiments on several of their nodes (albeit after hours) even though the study conducted was not part of any approved research work and was purely voluntary work conducted on our part.

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Controlling the Response Time of Web Servers

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Abstract - *Internet server provisioning is a very challenging problem for content providers and large server farms. In this paper we investigate the control theoretic approaches of managing servers in order to satisfy a required Quality of Service (QoS). We compare between two widely used models in the literature and our proposed simple technique. First model is a queueing based M/G/1 model with a PI controller. The second technique represents the server as a second order system and estimates the system parameters on line every sampling period. The estimated parameters are used in the design of a first order filter/controller in order to track the required QoS. Finally we present a simple technique based on the Additive Increase Multiplicative Decreases used in TCP congestion avoidance. We use simulation to compare these three techniques. Surprisingly, the AIMD performs the best among these three and it requires the least computation overhead among the three.*

Keywords: Quality of Service, web server provisioning, response time control, performance guarantees.

1 Introduction

The Internet is growing with an unprecedented rate and is infiltrating every aspect of our lives. E-commerce sites, mail servers, file servers, content servers, and search engines are few examples of applications that we use almost every day of our lives. It is difficult to imagine our lives without the *heavy* use of the Internet.

These sites are powered by powerful servers (or in many cases a large server farms where hundreds and may be thousands of servers are used) that receive users' requests, process them and send back the response. One of the major problems facing providers is how to condition the servers in order to produce the agreed-upon quality of service (QoS) and at the same time minimize their cost.

The Qos is either an agreed-upon contract between the servers owners and the content provider that must be maintained by the server owners, or a generally accepted criterion that is enforced by the server owners in order not to drive clients away. The consumers are known to be impatient, if the response is not within a specific period of time (that period varies greatly according to the application) the customer will probably terminate the session and navigate away to another site (the just-a-click-away syndrome).

Throwing hardware at the problem (also known as over provisioning) is not the optimal solution. Designing the system to work at the peak capacity wastes a lot of resources that most of the time would be unused. What is required is a policy that achieves the required QoS without wasting a lot of hardware. That is usually achieved by using admission control, where requests will be turned down if accepting the request results in a longer response time than what is required in the QoS agreement. If the system is overloaded, accepting a request not only means that this request will suffer more than usual response time, but also it means that all requests arriving after that request will suffer a longer than normal response time. By turning down one request, we lost one request but the following ones will be served according to the required QoS agreement.

Recently, there have been a lot of studies that suggests the use of classical feedback control theory in order to control access to the server and maintain the required response time. The argument is just as in the case of a controller controlling the gas rate going into a furnace in order to maintain the output (temperature) at a specific level, a controller to control the request rate delivered to the server can maintain the required output (response time) at a specific level.

However, the main differences between systems where classical control showed a lot of promise and internet servers are:

- Systems where classical control is very promising are very well understood; usually its behavior is governed by differential equations (difference equations in case of discrete systems). This lends itself very nicely to classical control theory. Where controllers are designed in the continuous time (discrete time) case using Laplace (Z) transform.
- Servers work in a highly unpredictable environment with probabilistic inputs (at best) and a lot of randomness in both arrival pattern and service time. Queuing theory has been successfully used to describe such a system. However almost all queuing theory results are based on a stable system and are valid at the steady state (average).

In this paper, we study the problem controlling the arrival rate in order to maintain the required QoS. We use 2 methods from classical control theory and one method that has proved itself to be successful in controlling congestion in TCP/IP traffic. We use simulation to compare these three methods.

The remainder of the paper is organized as follows: Section II describes our motivation and surveys previous work in this area. Section III describes the setting and the proposed solutions. Section IV shows the result of our work and compares it with previous solutions. Section V concludes the paper and describes our future work.

2 Motivation and Related Work

2.1 Motivation

Our motivation is to control the response time of an Internet server. Usually, and specifically in E-commerce applications the service is structured as 3-tier server. The first tier deals with static contents. The server gets a request to send a specific page, and it responds by sending the page. The second tier deals with dynamic contents. Requests arrive to the second tier server that fetches and calculate dynamic contents and sends it back. Third tier deals with database accesses.

Although 3-tier architecture is quite common in E-commerce applications, in this paper Our motivation is to control the response time of an Internet server. Usually, and specifically in E-commerce applications the service is structured as 3-tier server. The first tier deals with static contents. The server gets a request to send a specific page, and it responds by sending the page. The second tier deals with dynamic contents. Requests arrive to the second tier server that fetches and calculate dynamic contents and sends it back. Third tier deals with database accesses.

Although 3-tier architecture is quite common in E-commerce applications, in this paper we deal with single tier services only. The reason for that is we are concentrating on comparing the different approaches of admission control. Currently we are in the process of building a low power server with a 3-tier architecture. Once this system is built, we will test it using the three approaches mentioned here. Our objective is to implement a low power server that could achieve the same performance as bigger, more powerful, and more power hungry servers.

2.2 Previous Work

A lot of work is done in improving the performance of web servers and achieving a specific QoS. Earlier work in this area was mainly either service differentiation [3] or using data prefetching [4]. In service differentiation customers (requests) are treated differently giving a priority for one type of requests (more important customers) over the others. In prefetching, data we think will be requested soon is prefetched ahead of being actually requested. Both of these 2 techniques can improve the performance of the system (for only one group of requests in service differentiation case) but there are no guarantees that a specific level of performance is met.

Service differentiation is combined with admission control in [12]. They classified incoming requests into two categories, and admission control is based on the queue size of each category and some real time system measurements. They tested their system using Apache with static contents and some basic form of dynamic content.

A self tuning controller is proposed in [11]. They used a queuing model known as processor sharing model and a proportional integral (PI) controller to satisfy a target response time. Their queuing mode is M/G/1 where the response time is given by the equation

$$T_{RT} = \frac{E[X]}{1-\lambda E[X]} \tag{1}$$

Where λ is the arrival rate (assumed to be Poisson arrival) and E [X] is the Expected value of the service time. They also linearize the model around the operating point. Equation (1) that describes the system is valid only in the steady state and for stable queues (by stable we mean average arrival rate is less than average service rate). For short time periods and in heavy traffic the arrival rate may be greater than the service rate. Although the objective of the controller is to avoid such a case, but when it happens the equation used to model the system is not valid anymore.

The authors in [13] proposed an admission control to control the response time of the server. In their model they used Eq. (1) to represent the system. They also proposed an adaptive control scheme where the model parameters are estimated on line (using RLS technique) and is used to modify the controller parameters [2]. While in [16] the authors proposed an adaptive architecture that performs admission control. Their technique depends on using TCP SYN policer and an HTTP header-based connection control in order to maintain the required response time limit.

Malarait et al in [14] proposed a nonlinear continuous time model using fluid approximation for the server. They used this model to obtain an optimal configuration of the server in order to achieve maximum availability with performance constraints and maximum performance with availability constraints. They also validated their design using TPC-C benchmark.

Elnikety et al in [7] proposed a proxy called Gatekeeper to perform admission control as well as user-level request scheduling. Their idea depends on estimating the cost of the request, then deciding if admitting that request will exceed the system capacity or not (system capacity is determined by using offline profiling). they noted that since the proxy is external to the server, no server modification is required for their gatekeeper.

Guitart in [8] proposed a session based admission control for secure environment. In their technique, they gave preference for connections that could use existing SSL connections on the server. They also estimated the service time for incoming requests in order to prevent overloading the server.

Blanquer et al [5] proposed a software solution for QoS provisioning for large scale Internet servers. They proposed the use of traffic shaping and admission control together with monitoring response time and weighted fair queuing in order to guarantee the required QoS. For an excellent review of performance management for internet applications, the reader is referred to [9].

Almost everyone who used Control theory used the average response time as the parameter to control. One major problem with that is the QoS requested is not on the form of average response time or average delay. The QoS is usually on the form x% of requests have a response time better than y msec. Guarantees to average response time do not solve this problem.

In this paper we investigate this problem. We compare between using average response time and the actual QoS percentage of the requests that satisfied the required response time guarantees. We also present a simpler technique that does require much less overhead compared with control theoretic approaches and produces better results in our simulation.

3 System Setup

As we mentioned before, most of the work done using control theoretic approach to control quality of service considered the average response time as the parameter to control. The problem of that approach is that most of the required QoS is not about the average response time [11]. Usually, the QoS requirement is described as 90% of the requests face a processing delay of not more than 150 milliseconds. Controlling the average response time will not lead us to a specific condition such as the one described above.

The reason for that is the Internet traffic is highly volatile and unpredictable. Classic queuing theory deals with such scenarios if we know the distribution of the incoming traffic and service time (or at least know some parameters about the underlying distribution such as the average and the standard deviation). For example consider the simplest type of queues known as M/M/1. The cumulative probability distribution of the response time is [10].

$$F(\tau) = 1 - e^{-\tau\mu(1-\rho)} \tag{2}$$

Where, μ is the service rate and ρ is the server utilization. Since the average time spent in the system for M/M/1 is $1/(\mu(1-\rho)) = 1/(\mu-\lambda)$, where λ is the arrival rate. By simple substitution of τ to be the average time spent in the system, we find that the probability that any customer meets a response time more than the average to be 36.8%.

Another problem with using control theoretic approach is how to handle the overhead in calculating and adjusting the system and the controller parameters. Consider for example admission control. The system output should be monitored to collect statistics about the parameter to be controlled. Every sampling period, the collected data are used in order to calculate the admission probability and modulate the arrival with this probability to meet the required service performance measure. The major question here is how should we select the sampling period?

The requests arrivals to a server are usually in the milliseconds range, or even less for very powerful servers. Now, what should be the sampling period? If we consider the sampling period to be on order of seconds, that is good from the overhead point of view. After all we do not want to overload the server with control calculation since that time is taken from serving incoming requests. However, since the web traffic is highly volatile and unpredictable, what happened few seconds ago might not have an impact on the current operation of the system. For examples 3-5 seconds ago we received a rush of requests that resulted in prolonging the response time and not meeting the required QoS, Now we reduce the admission probability in order to slow down the arrival, but there is very little arrival now. That leads to wasting CPU cycles because of the time difference between the sampling rate and traffic variations.

The second choice is taking the sampling time in the order of milliseconds. Although the response will be much faster than the previous case, however that is too much overhead for the CPU. Even the online recursive estimator in [15] requires a number of large matrix multiplications every sample period in order to estimate the system parameters.

In this paper, we present three different techniques for QoS guarantees in a web server. First we consider a simple M/G/1 model similar to the one proposed in [13]. This model predicts the response time, so the only controllable parameter here is the average response time. Then we consider a general model for the system. We assume a second order model and we estimate the system parameters on line. Then a first order controller/filter is used to control the average response time. Our third model is a variation of the previous one where the parameter to control is the percentage of the requests that failed the QoS requirements. Finally we consider a fourth model where we used a simple variation of the Added

Increase Multiplicative Decrease AIMD that was successfully implemented in congestion avoidance for TCP/IP protocols [1].

3.1 Using PI Controller

This is the method proposed in [11]. Eq 1 is considered to represent the system where T_{RT} represents the response time. The schematic diagram of the controller is shown in Figure 1 where the server is represented by Eq. 1

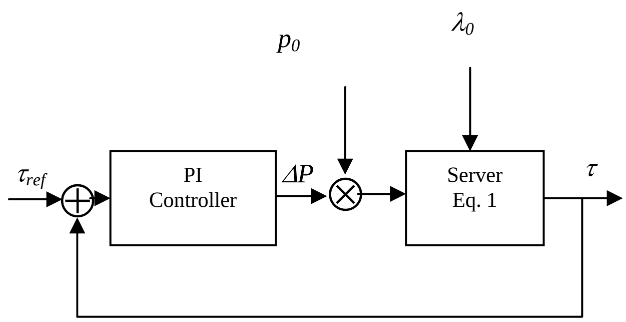

Fig. 1 Schematic diagram of the controller

In Fi.g 1 τ represents the response time, τ_{ref} is the required average response time, P_0 is the admission probability derived from Eq. 1 in order to make $\tau = \tau_{ref}$. λ_0 is the unmodulated arrival rate, and Δp is the correction produced by the controller to p in order to guarantee the required τref.

Linearizing Eq. 1 using Taylor series around the operating point λ_0 we can solve for the PI controller. The results of this scheme together with some comments about the scheme is discussed in the next section.

3.2 Estimating the System Parameters

Similar to [13] we assume no knowledge of the system under control (the server). By monitoring the input and output of the server we derive the model parameters. Here the system under control is the server with input λ_0 and output either the average response time or the percentage of the requests that confirm to the required QoS. We tried several models for the system and found out that the best fit is a second order system. In this part, we consider two solutions one that adjusts the average response time and one that adjusts the percentage of requests conforming to QoS.
In this case, we assume that the system output *y* (no matter what the output is, it could be response time, or the percentage of packets that missed the service time threshold) can be represented as a second degree system where the input *u* is the arrival rate as follows.

$$\frac{y}{u} = \frac{1 + a_1 Z^{-1} + a_2 Z^{-2}}{b_0 + b_1 Z^{-1} + b_2 Z^{-2}} \qquad (3)$$

Where Z^{-1} is the delay operator. The parameters a_i and b_i are estimated on line by measuring the output *y* and the input *u* and averaging them over the sampling period. We use a well known recursive least square estimator [15].

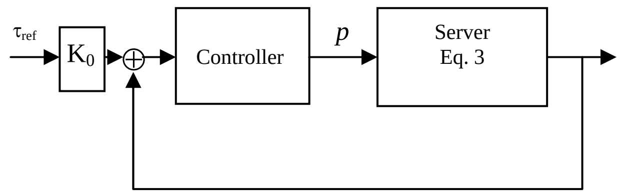

Fig. 2 Schematic diagram of the controller where we estimate the system parameters.

The controller was designed as a PI controller on the form

$$G_c = \frac{\beta_0 Z^{-1} + \beta_1}{\alpha_0 Z^{-1} + \alpha_1} \qquad (4)$$

α_0, α_1, β_0, β_1, and K_0 are chosen in order to achieve a reasonable overshot, settling time within the sampling period and the proper tracking of the output for K0

3.3 Additive Increase Multiplicative Decrease

This idea came from the sliding window control in TCP [1]. In TCP the window size is decreased (by a multiplicative factor) if there is a lost packet and is increased (by an additive factor) for successful transmission of a packet [6].

The proposed scheme works as follows. If the queue size grows beyond a high threshold N_{high} the probability of accepting a new packet is multiplied by γ, where $0 < \gamma < 1$. If the queue size drops below N_{low}, the admission probability is increased by η, where $0 < \eta < 1$. The probability is bounded in the interval [0.1, 1] for practical purposes.

The obvious question is how to choose the values of N_{high}, N_{low}, γ, and η, The proper choice of these parameters depend on the service time and inter-arrival time distributions. In our simulation, we tried different values for the parameters and found that the best choice for N_{high} is the target delay divided by the average service time, while $N_{low} = N_{high}/2$. We also found the best values for $\eta = 0.4$, and $\gamma = 0.8$-0.9. Clearly the optimal values for these parameters depend on the arrival and service distribution and can be fine tuned online.

4 Results and Discussion

In this section we show the results of our simulation using Matlab for the four cases proposed in Section III.

For all the experiment we ran the simulation for 1600 seconds using Matlab. We collected the percentage of the requests accepted, the percentage of the requests that required less than 150 msec. and the average response time. For every experiment we considered two traffic scenarios.

- **Traffic A**: This is the baseline system, we assumed an average exponential interarrival time of 55 *msec*. and an average exponential service time of 35 *msec*. (64% utilization). The target response time is 150 *msec*.
- **Traffic B**: In this scenario, we start as in Traffic A. At the simulation midpoint (after 800 seconds) we increase the arrival rate by decreasing the interarrival time to 45 *msec.* (utilization of 78%). The objective here is to see how the controller reacts to increasing the arrival rate in order to satisfy the QoS requirements.

Fig 3 shows that response time for a 20 minutes simulated run under traffic B. We can see that after 800 msec. the average response time increases. The main function of any controller is to adjust the admitting probability in order to avoid such a scenario.

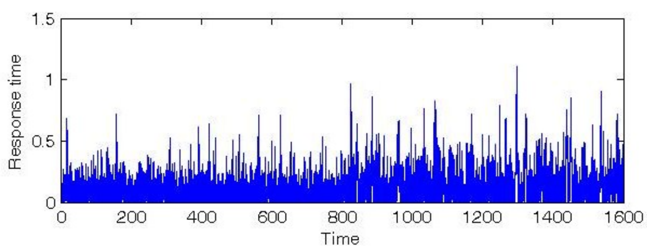

Figure 3. Response time without a PI controller under traffic B

Fig. 4 (same setting as Fig. 3) shows the response time and the acceptance probability as a function of time. It is obvious that the controller suppressed the input in the second half of the simulation leading to a more consistent (equal) response time. One thing that is very noticeable here is the rapid changes in the admitting probability compared to the other method we used. Although the probability changes very rapidly, the performance is the worse compared to the other techniques. One possible explaination this is that Eq. (1) does not describe the system when the traffic increases beyond stability even for a short period of time

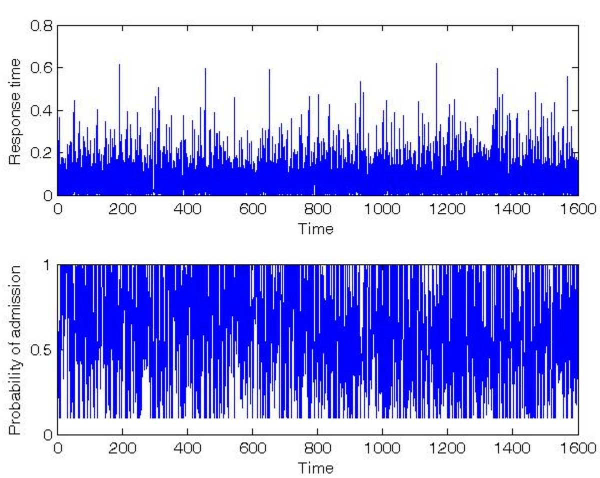

Fig. 4. Response time and admitting probability under traffic B for a PI controller

Then we consider our technique where we assume a second order system and estimate system parameters on line. Once the parameters are estimated on line, the parameters are used to choose the parameters of a first order controller/filter in order to track the required QoS criterion either directly by controlling the percentage of conforming packets, or indirectly through controlling the average response time.

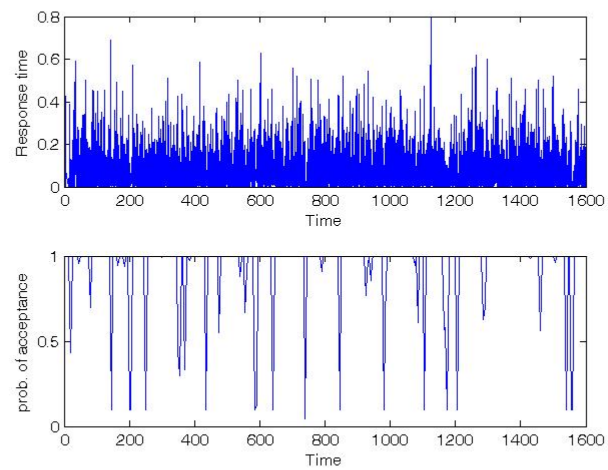

Fig 5. Response time and admitting probability assuming a second order system and a first order filter under traffic A (c0ntrolling averge response time).

Figure 5 shows the response time and admitting probability for our system under traffic A assuming a 2nd order system with on-line parameters estimation. While Fig. 6 shows the same system under traffic B scenario. The parameter to be controlled in this case is the average response time.

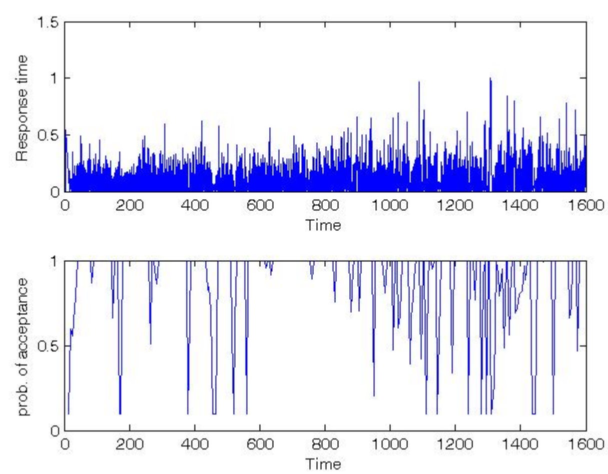

Fig 6. Response time and admitting probability assuming a second order system and a first order filter under traffic B (controlling average response time).

The changes in the admitting probability for this system is much less than the case of a PI controller. That is by itself is not an advantage (however it might give an indication that the system is stable and does not oscillate) but as we will see in Table 1, the performance here is much better than the PI controller case.

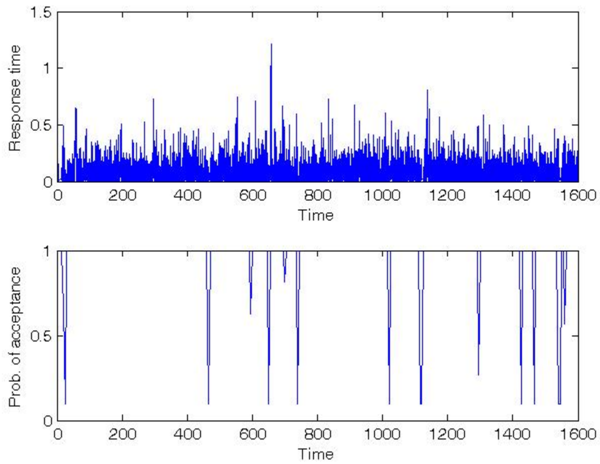

Fig 7. Response time and admitting probability assuming a second order system and a first order filter under traffic A (controlling QoS).

Figure 7 and 8 shows the same results as Figures 5 and 6 but in this case we use the percentage of the conforming requests as the parameter to control. Although it is difficult to see, directly from the Figures, which one is better in maintaining the required QoS, controlling the response time, or the percentage of conforming packets directly from the Figures, Table 1 shows that in fact controlling the percentage of conforming packets produces better results.

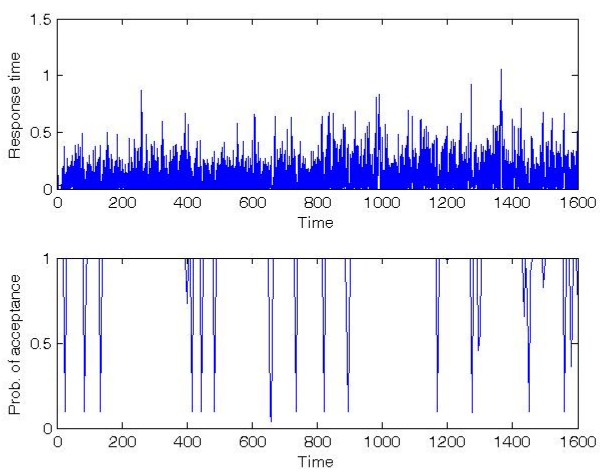

Fig 8. Response time and admitting probability assuming a second order system and a first order filter under traffic B (controlling QoS).

Figures 9 shows the system using AIMD with Nhigh $=T_{target}/\tau_{av}$, $N_{low} = N_{high}/2$. η =0.4, and γ=0.8. Where Ttarget is the target delay and is set to 150 msec. and τ_{av} is the average response time under traffic A. Figure 10 shows the same setting under traffic B. The admitting probability varies very quickly compared to Fig 5,6,7,8. However that is expected since the changes in the admitting probability is calculated every time the queue size grows beyond a specific threshold, or decreases below another threshold. However as we will see in Table 1, this is the best performance among the three techniques.

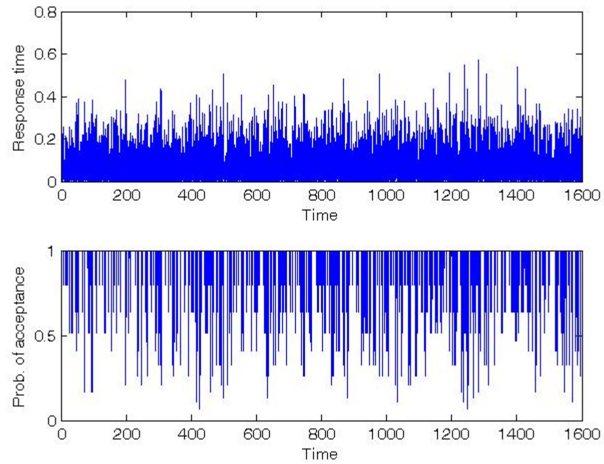

Fig 9 Response time and admitting probability using Additive Increase Multiplicative Decrease AIMD under traffic A

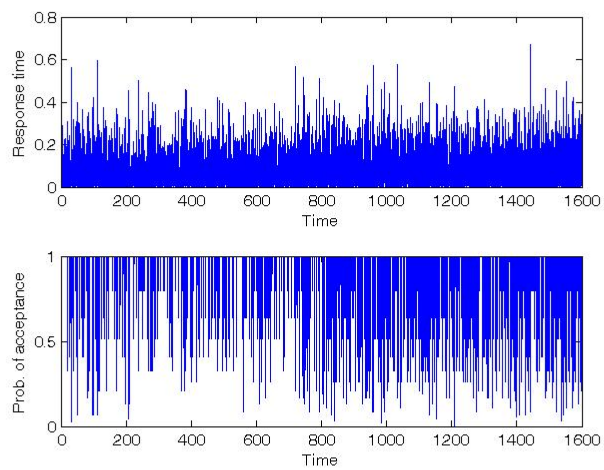

Fig 10 Response time and admitting probability using Additive Increase Multiplicative Decrease AIMD under traffic B

We summarize the results in Table 1. The first column shows the 4 different techniques we used. The second column shows for every technique the results under traffic A and traffic B.

The actual results are shown in columns 3, 4, and 5. Column 3 shows the percentage of admitted requests. Column 4 shows the percentage of the requests that is conforming to the required QoS. The first number shows the percentage of conforming requests to all arrived requests, while the number in parenthesis shows the percentage of conforming requests to admitted requests only. Finally column 5 shows the average response time for all admitted packets.

From Table 1 we can also see that AIMD has the highest admitting policy under traffic A (97%), while PI has the lowest (61%). It also shows that AMD has the highest conforming percentage under traffic A. Under traffic B assuming a 2nd order system with online parameters estimation has slightly higher admitting probability than AIMD (94% vs. 93%), however the percentage of conforming requests is much higher for AIMD (compared to all arriving requests or admitted requests). basically that states that the AIMD rejects a very small percentage of incoming requests, but it rejects the tight ones.

TABLE 1 COMPARISON BETWEEN THE FOUR RPOPOSED METHODS

Technique		% accepted	% conforming	Av. delay msec.
PI	A	61%	53%(86%)	74
	B	57%	48%(84%)	80
Est.(t_{resp})	A	92%	73%(79%)	90
	B	88%	63%(72%)	119
Est. (T_{resp})	A	95%	75%(79%)	95
	B	94%	66%(70%)	125
AIMD	A	97%	83%(86%)	78
	B	93%	77%(83%)	85

5 Conclusions

In this paper we investigated 3 different techniques for controlling admitting probability in an internet server in order to conform to a required QoS. Using simulation we show that a simple AIMD technique outperforms more complicated control-theoretic approaches, and it requires much less overhead compared to the control-theoretic approaches.

For future work, we are building a low power server using small embedded microprocessors. We will be testing these proposed methods under realistic traffic when the server is up and running

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A Future Internet Testbed in Korea

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Abstract - *Future Internet is a new concept Internet to solve inherent problems of the existing Internet and ultimately aims to provide new services such as mobility, security, scalability and manageability by replacing the present Internet. These new services can be expected to be available by Virtualization, Programmability and Federation which have not been realized on the present Internet. Already many developed countries are performing Future Internet research and constructing Future Internet testbeds, a foundation of the research. In order to keep pace with them, Future Internet's testbed deployment in Korea is necessary to create new Internet technology, a variety of services, and interconnection of resources internationally. Therefore the requirement of Korea Future Internet testbed is increasing. In order to satisfy the need, Korea Future Internet testbed is deployed between Daejeon(Korea) and GENI(USA).*

Keywords: Future Internet, KREONET[1], GENI, Federation, openflow

1 Introduction

Future Internet is a new concept Internet to solve inherent problems of the existing Internet and ultimately aims to provide new services such as mobility, security, scalability and manageability by replacing the present Internet. These new services can be expected to be available by three core technology (Virtualization, Programmability and Federation) which have not been realized on the present Internet. Additionally not only network resources in countries will be used independently, but also they will be shared together for international co-operation in the Future Internet.

Currently although research about Future Internet testbed is in an early stage, developed countries such as USA, Europe and Japan are progressing Future Internet testbed projects with the investment of human resources and budget. Among them, GENI (Global Environment for Network Innovation), FIRE (Future Internet Research and Experimentation), CORE (Collaborative Overlay Research Environment) are the leading and representative projects.

Already Future Internet testbeds in USA, Europe and Japan are being constructed and interconnected one another based on their representative national research and development networks. In order to keep pace with them, it is necessary for Korea to deploy Future Internet testbeds and to federate internationally for future science research. In doing so, Korea is expected to be the advanced country in the Future Internet areas and can lead the next generation network technology. Among them, the most important task is to deploy international Future Internet testbed and in particular federate with GENI, which is the leading project in USA. For those, Korea Institute of Science and Technology Information (KISTI) has participated in GENI project since a few years ago and was selected as the official international research partner in October, 2009. Thus, KISTI obtained a chance to promote joint technology development with Indiana University, one of the GENI project's participants, until 2012. (http://www.geni.net/?p=1480)

In this paper, we explained international Future Internet projects' trend in section 2. Then we described deployment of Future Internet testbed between Korea and GENI(USA) in section3, and openflow test through the testbed in section 4. Finally, we summarized the deployment of the Future Internet in Korea.

2 The Future Internet Trend

Developed countries around the world, as well as USA are investing a lot of budget for the Future Internet research. In this chapter, we would like to describe trend of the Future Internet.

2.1 USA

NSF (National Science Foundation) has been investing 700M$ since 2004 through GENI and FIND projects to deploy the Future Internet testbed. Particularly GENI is implementing Programmability, Virtualization, Resource Sharing, Federation, Slice-based experiment and Clean-state in order to solve problems of the existing Internet. Thus, GENI aims to construct a totally differentiated testbed from

[1] KREONET(Korea Research Environment Open NETwork) is the national R&D network supported by MEST (Ministry of Education, Science and Technology), and has been managed and operated by KISTI (Korea Institute of Science and Technology Information) since 1988. KREONET has a high performance network infrastructure that provides R&D resources such as supercomputing, GRID, and e-science applications, to about 200 key R&D centers in the industrial, academic, and corporate sectors.

existing TCP/IP networks by new technology. For building practical testbeds, GENI uses Internet2 and NLR (National Lambda Rail) as the Future Internet testbed. And now GENI is in Sprial 3 and is constructing meso-scale infrastructure.

2.2 Europe

Europe is performing the Future Internet research program by FP7 (7th Framework Programme) and FIRE (Future Internet Research and Experimentation) projects based on Future Internet service and optical network technology. All of projects are also developing federation technology to use each nation's network resources jointly. In order to build useful testbeds, Europe uses GEANT 1, 2 and 3, PanLan and OneLab as a testbed. Additionally Europe is investing 650M€ to build the Future Internet testbeds on FIRE project and expanding the testbeds to construct large-scale experiment environment across Europe. And also Europe is trying to provide a variety of experiment environment similar to GENI.

2.3 Japan

Japan is performing AKARI project which is led by NICT (National Institute of Information and Communications Technology) and is focusing on ubiquitous, mobility and service convergence. For those Japan uses JGN2plus as the Future Internet testbed. On the other hand, Japan is developing the CORE testbed to perform research of the Future Internet and is trying to expand the CORE project toward national research network. Major research areas are federation, network architecture, mobile communications and bio/nano technology. In the meantime Japan has focused on technology related to IPv6, but recently Japan is concentrating on federation and virtualization research fields.

2.4 China

China is pushing ahead the Future Internet by CNGI (China Next Generation Internet) project. China is devoting to enhance scalability by IPv6 and uses CERNET2/6iX as a testbed.

2.5 Korea

Korea is progressing various projects in KISTI, FIF (Future Internet Forum), FN2020, ETRI and KT. Generally Korea is focusing on completely new structure of Future Internet and improving the structure of existing Internet simultaneously. For example, academic areas such as FIF concentrate on core technology development and standardization such as research plan and architecture design, and research and industrial areas focus on programmable platform development for virtualization. Additionally KISTI and NIA(National Information Society Agency) are deploying the Future Internet testbeds using KREONET and KOREN.

3 Deployment of the Future Internet testbed in Korea and GENI

The Future Internet testbeds can be classified into two categories. One is a small and local-sized testbed in research labs and campus. The other is a large-scale testbed to interconnect local-sized testbeds. From this point of view, the testbed which is deployed between Korea and GENI is backbone network level testbed. Additionally this testbed was deployed by core nodes which can provide virtualization and programmability function, thus Future Internet's researcher in Korea and USA can collaborate with one another using the testbed.

Generally core nodes of the Future Internet testbeds have to satisfy switching and routing functions, as well as have to include flow control functions. Therefore HP Procurve 5412zl switch and FIRST (Future Internet Research for Sustainable Testbed) switch which is developed by ETRI are used in Korea Future Internet testbed.

Figure 1 shows the configuration of the Korea Future testbed. HP Procurve switch is set up in Seattle and FIRST switch is installed in Daejeon. Two nodes were connected by 1Gbps VLAN of SONET/SDH gigabit Ethernet in KREONET.

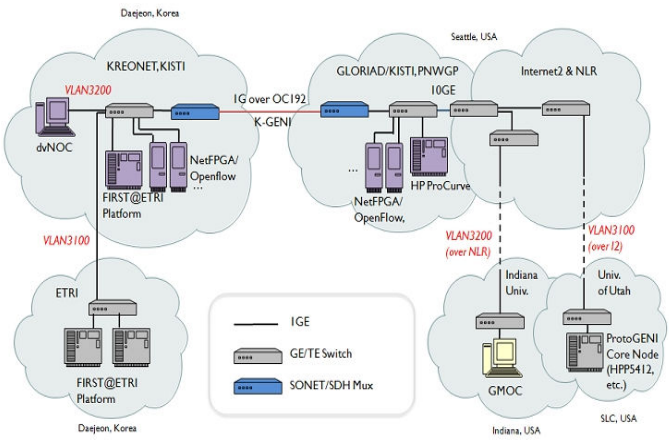

Figure 1 Future Internet testbed between Korea and GENI

Internationally VLAN which is set up between Seattle and Daejeon is connected with a variety of Future Internet backbone networks in USA, thus Future Internet researchers can use this testbed for their research. Furthermore researchers in Korea also are able to use Future Internet testbed in Daejeon to collaborate with foreign researchers. Especially since core nodes which are deployed the Korea Future Internet testbed are compatible with those of GENI, collaborative research of Korea and USA will be possible.

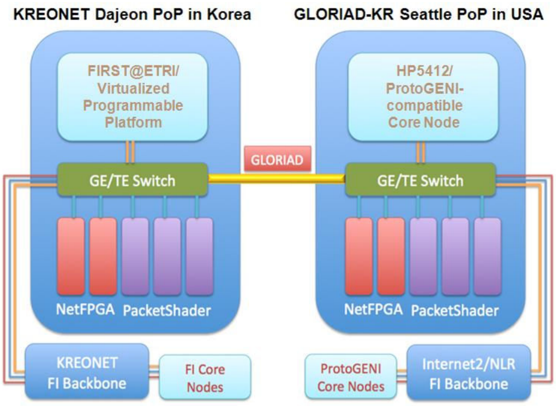

Figure 2 Components of core nodes

As we mentioned before, core nodes of Korea Future Internet testbed consist of HP Procurve 5412zl switch and FIRST switch developed by ETRI. In case of FIRST switch, because it supports virtual programmablity, users can create appropriate virtual network slice freely for research purposes. In case of a Procurve switch, it is also deployed in GENI testbed and also provides programmability functions.

Although high performance core nodes are necessary for building large-scale testbeds, programmable PC nodes are used to construct medium or small-scale testbeds and they also can be utilized as auxiliary nodes to support large-scale networks. In the Korea Future testbed, small programmable nodes also deployed. Therefore, researchers could choose appropriate nodes according to their research purposes. Figure 3 and 4 shows actually deployed Future Internet nodes of Korea Future Internet testbed.

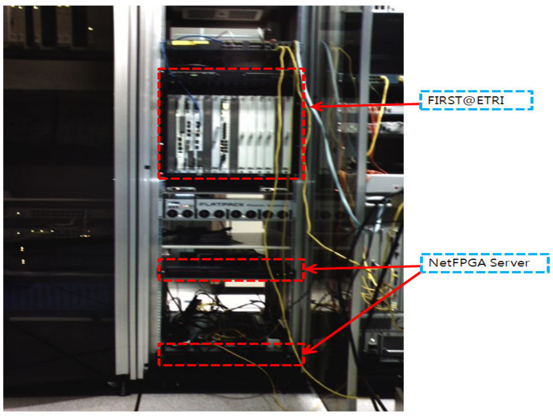

Figure 3 Korea Future Internet nodes (Daejeon)

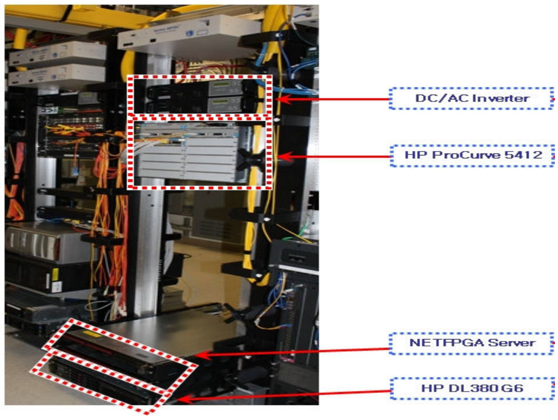

Figure 4 Korea Future Internet nodes (Seattle)

4 Openflow Test

As Openflow is considered as one of important technology in the Future Internet, we conducted a test in order for Openflow functionality of core nodes. This test was done by FIRST switch and Procurve switch in Daejeon and Seattle on August 4, 2010. The test scenario is as follows.

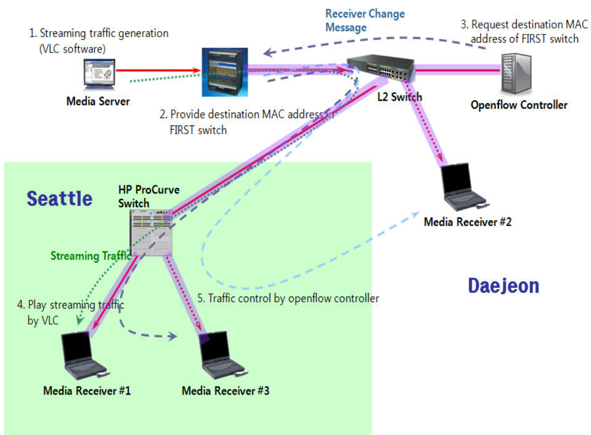

Figure 5 Openflow test configuration

4.1 Test Scenario

1) Media Server in Daejeon generates streaming traffic to FIRST switch.
2) Check that streaming traffic is normally transmitted to Media Receiver #1 in Seattle.
3) Execute commands which is developed by dpctl in Openflow Controller.
4) Check that transmission of streaming traffic to Media Server #1 in Seattle is stopped.
5) Check that streaming traffic to Media Receiver #2 in Daejeon is transmitted.

4.2 Test Results

Before executing commands, streaming traffic is transmitted to Media Receiver #1 in Seattle. After executing commands, transmission of streaming traffic is stopped to Media Receiver #1, and then traffic began to transmitted to Media Receiver #2 in Daejeon continuously. From this test, we were able to check that openflow function is operated normally in all core nodes.

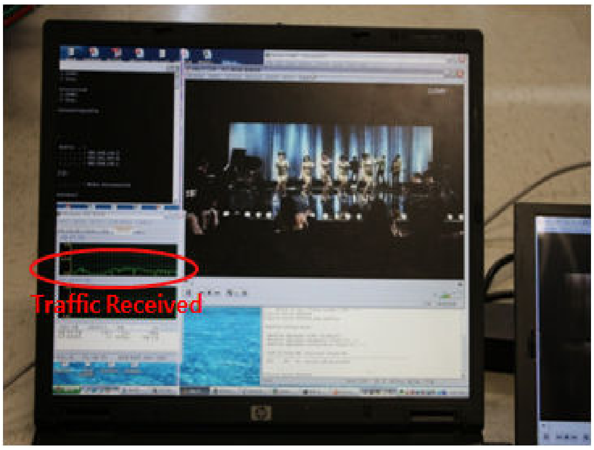

Figure 6 Traffic received in Media Receiver #1 in Daejeon

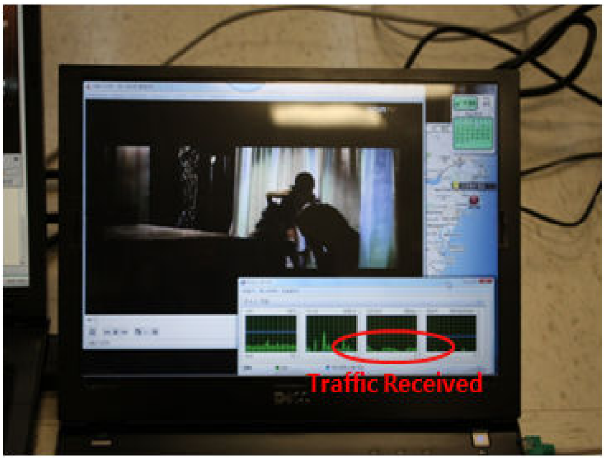

Figure 7 Traffic path is changed from Media Receiver #1 to Receiver #2

5 Conclusion

Future Internet is a new Internet to overcome inherent problems on the present Internet. Already many countries such as USA, Europe and Japan are trying to develop Future Internet technology, and several Future Internet projects are being performed to preoccupy new unexplored areas. Thus deploying Future Internet testbeds is very important, because they can be foundation of Future Internet. This Korea Future Internet testbed, built between Korea and GENI, will support experimental environment and satisfy a variety of needs of researcher of USA and Korea. Therefore it will be representative Korea's Future research network.

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CAN KEYWORD LENGTH INDICATE WEB USERS' READINESS TO PURCHASE?

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Abstract: *Over the last ten years, the internet has become an important marketing tool and a profitable selling channel. The biggest challenge for most online business is converting Web users into customers effectively and at a high rate. Understanding the audience of a website is essential for achieving high conversion rates. This paper describes the research carried out in online search behaviour. The research looks at whether the length of a Web user's search keyword can provide insight into their intent and proposes an initial model for using keyword length to predict Web users' readiness to convert when they land on a website.*

Keywords: online searching, search keyword, search keyword length, experiential users, goal-oriented users.

1 INTRODUCTION

Over the last ten years, the internet has become an important marketing tool and a profitable selling channel. According to Ofcom [1]:

"UK leads the way with the internet accounting for nearly a quarter (23 per cent) of total advertising spend. Outside of Europe, the internet accounts for 15% of advertising spend in the US and 12% in Japan. The internet's share of total advertising spend is growing everywhere."

It is estimated that the internet population will grow from 1.83 billion in 2010 to 2.10 billion in 2012 [2]. This represents a considerable increase in the number of potential customer for any online business. The biggest challenge for most online business has been about converting Web users into customers. Understanding the audience of a website is essential for achieving high conversion rates [3, 4]. When a website "it is important to keep in mind who your audience is and [to] make sure that the information you provide is relevant to them." [5].

There have been a large number of studies that have focused on understanding the goal behind a Web user's search query (also commonly called search phrase or search keyword) with a view to improve the quality of search engines' results [6]. However, few studies have investigated whether by understanding the goal behind a Web user's search query, their experience on a website could be personalised and improved or whether their likelihood to purchase or convert could be predicted.

In this paper, we first look at the two main types of Web users and how previous studies have linked Web users' search query to motivation and intent. This paper then proposes a search-conversion model to infer the likelihood of online conversion based on the length of the search query.

2 LITERATURE REVIEW

2.1 Online Search Behaviour

Web users arrive at websites with different motives and goals in mind. There are two generally recognised types of Web users [7, 8]:

- Goal-oriented or Seekers.
- Experiential or Surfers.

Goal directed Web users are motivated by external factors (extrinsic motivation), task oriented, and influenced by interests or concerns brought about by the particular situation or context that they are in. They use directed searches, characterised by work- like thoughts, to reach their goals [9].

Experiential Web users are motivated by fun which leads to browsing. Their searches were non-directed and they usually do not have an explicit goal in mind when carrying out searches. They also spend a lot of time browsing sites usually in an ad-hoc fashion. "Relatively unstructured recreational use [of the web or websites] is experiential behaviour." [10, p. 26].

One of the characteristics differentiating goal oriented Web users from experiential Web users is the type of search that they carry out. Goal-oriented Web users carry out "directed (pre-purchase)" search while experiential Web users carry out "non-directed (ongoing) search" [9]. Usually Web users start their search with a generic search phase and gradually refined their search terms until they find a search phrase that leads them to a website that satisfies their needs.

Web users can adopt two types of search strategies namely browsing and analytical strategies [11]. According to Zhang and von Dran [12]:

> browsing [is] an informal and natural information seeking approach that [depends] heavily on the information environment and the user's recognition of relevant information. Analytical strategies, in contrast, [depends] on careful planning, recall of query terms, iterative query reformulation, and examination of results. (p. 1254)

Pavlou and Fygenson [13] found that the intention of buying a product occurred before the intention of acquiring information on a product. Following this reasoning, an individual who has decided to buy a product will go online and try to express this decision in terms of a search query. Pirolli [14] made a similar distinction between task and need and referred to a query as an external representation of need.

Jansen, Booth and Spink [15] suggested that the search query was not the only expression of intent and that other "aspects of the interaction including number of query reformulations, selection of vertical, use of system feedback, and result page viewed" (p. 1255) were also expressions of intent.

2.2 Long tail search phrases

Long tail is a phrase that became popular following the publication of a best-selling book by Anderson [16]. Long tail keywords are low-volume, obscure, infrequently searched-for keywords. While long tail searches are individually insignificant compared to generic searches, added together long tail searches can provide significant search volume. They can produce higher conversion rates as it is thought that Web users who carry out long tail searches are likely to be further along the buying cycle and therefore more ready to buy than users who make generic searches [17].

Because long tail keywords tend to be specific they are usually longer than generic keywords. Online marketing agencies and experts believe that long tail keywords can boost conversion rates especially when keyword relevancy in the website content is good and elements of good landing page design are present [Search Engine 17, 18].

However, Ghose and Yang [19], found that the "length of a keyword negatively impacts the performance on all three metrics [conversion rate, order value and profit] for natural search listings but only affects the order value in paid search." (p. 2)

It was not clear whether the websites used in the research carried by Ghose and Yang [19] and Skiera [20], were well designed and whether the content was relevant to long tail keywords. These could have affected their results.

3 SEARCH-CONVERSION MODEL

Based the literature review, the research hypothesised that:

H1: Web users' search terms could indicate motive and intent.

H2: Longer search terms indicated that Web users were more ready to convert.

H3: Shorter search terms indicated that Web users were less ready to convert.

H4: Goal-oriented Web users were more likely to convert than experiential users.

A search-conversion model based on these hypotheses is shown in Figure 1. When Web users search for information online they usually go through the process of typing a search term into a search engine and browsing sites returned by the search engine. If they do not find what they are looking for, Web users refine their search term, usually by adding words to their original keywords. Web users go through these steps a number of times before they find the information that they are looking for and are ready to purchase or carry out an action that represents a conversion.

This research theorised that during the search process, Web users go through an experiential stage where they look for information and try to refine their search terms. During this stage, Web users are focused on research and are not ready to buy. Once Web users feel that they have enough information to make a decision and/or are confident that they are

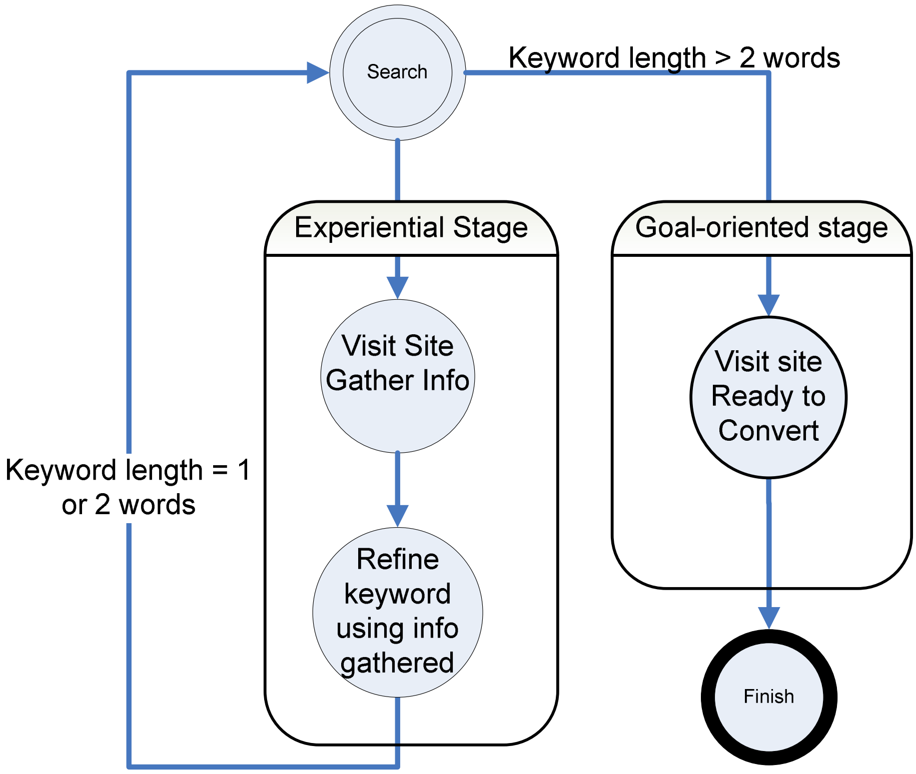


Figure 1: Search-conversion model.

using the right search terms to find what they are searching for, they enter the goal-oriented stage.

At this stage users are ready to convert and are actively searching for websites or browsing websites with the intention to buy or complete a action that represents a conversion.

It is assumed that Web users at the experiential stage are at the beginning of the search process. At this stage, Web users use general search queries that are short and contain one or two terms. Web users who have reached the goal oriented stage are likely to use more specific/detailed search queries for example search queries with more than 2 words. Therefore, it is hypothesised that when Web users arrive on a website, a longer search query could indicate a higher likelihood to convert. Initial data analysis was carried out to test H2 and H3.

4 RESULTS

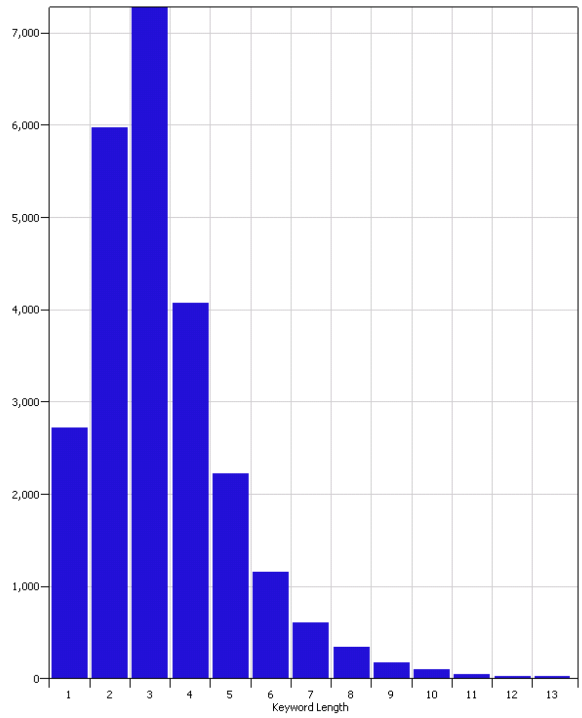
4.1 Data Collection

The research used data from a commercial website that offered manufacturing and design services online. The aim of the website was to encourage visitors to enquire about these services. Enquiries were then turned into sales offline.

An Online Tracking Module (OTM) was created to monitor visitors' behaviour while they browsed the website. Pay Per Click (PPC) advertising was used to generate traffic. The website's content and design were optimised to promoted conversion. In the case of the website investigated in this research a conversion consisted of completing and sending an enquiry form from the website.

4.2 Data Analysis

Graph 1 shows the distribution of the length of search queries in the data collected. It can be seen from the graph that search queries that were 2 and 3 words long were most popular followed by search queries that were 4 words long. The length of search queries decayed exponentially from that point onwards.



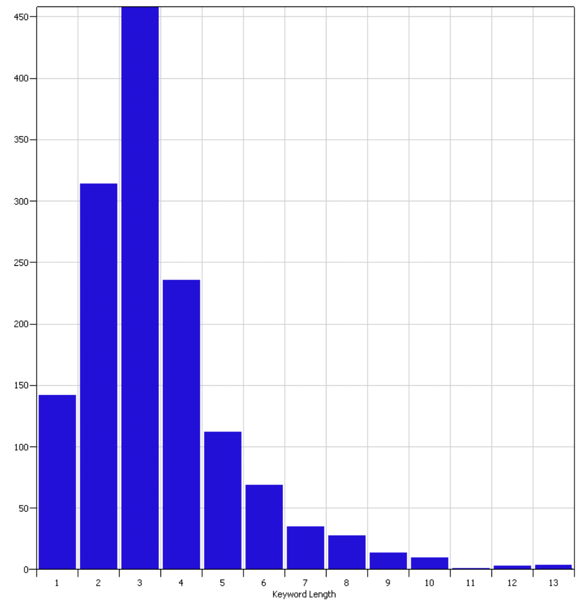
Graph 1: Distribution of search query length vs frequency of occurrence (count).

Graph 2 shows the length of search queries that generated a conversion. It can be seen that Graph 2 follows the same kind of distribution as that shown in Graph 1. Table 1 shows a breakdown of Graph 2.

Most conversions (31.98%) were generated from three-word search queries. The distribution peaked at search query length of three then decayed exponentially. There were more conversions originating from search queries containing four words (16.48%) than from search queries containing one word. If the distribution is examined on either side of the highest point on the graph (search query length = 3), it can be seen that 31.85% of conversions were generated by search queries that contained less than three words, while 36.17% of conversions were generated by search queries that contained more than three words.

This represented 13.56% more conversions by search queries that contained more than three words. This suggested that longer keywords could indicate higher probability of conversion. These conclusions supported H2 and H3.

In order to confirm these observations, the distribution of keywords that did not produce conversions was examined. This is shown in Graph 3.

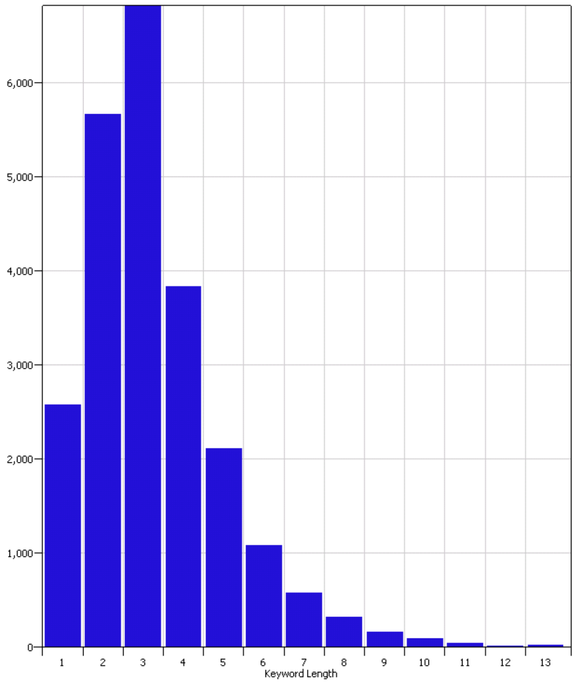


Graph 2: Search query length vs frequency of occurrence (count) for visits that converted.

Table 1: Breakdown of Graph 2.

Search query length	Count	%
1	142	9.92
2	314	21.93
3	458	31.98
4	236	16.48
5	112	7.82
6	69	4.82
7	35	2.44
8	28	1.96
9	14	0.98
10	10	0.70
> 10	14	0.98

It can be seen that the shape of the distribution shown in Graph 3 is similar to that of Graph 1 and Graph 2. Table 2 shows a breakdown of Graph 3. It was observed that three-word search queries accounted for 29.13% of visits that did not convert. The distribution peaked at length three then decayed exponentially. There were more non-conversions originating from search queries containing four words (16.40%) than from search queries containing one word (11.02%).



Graph 3: Graph 3: Keyword length vs frequency of occurrence (count) for visits that did not convert.

Table 2: Breakdown of Graph 3.

Search query length	Count	%
1	2,581	11.02
2	5,667	24.21
3	6,820	29.13
4	3,840	16.40
5	2,117	9.04
6	1,090	4.66
7	579	2.47
8	325	1.39
9	165	0.70
10	95	0.41
> 10	132	0.56

When the distribution was examined on either side of the highest point of the graph (length = 3), it was observed that 35.23% of non-conversions were generated by search queries that contained less than three words, while 35.64% of non-conversions were generated by search queries that contained more than three words.

Contrary to what was expected, longer search queries generated almost the same percentage of non-conversion as shorter keywords. These observations did not support the conclusions drawn from the data shown in Graph 2 and Table 1. If longer search queries were more likely to generate conversions, then the percentage of non-conversions from shorter search queries (less than 3 words) were expected to be higher than that of longer keywords (greater than 3 words).

4.2 Limitations

The study had some limitations that affected the results presented in Section 4.1.The relevancy of a Web user's search query compared to the content of a website can affect the likeliness that the Web user will browse or convert. The relevancy of a search query does not depend on its length but rather on the words that make up the search phrase. Therefore, the search-conversion model needs to take search query relevancy into consideration.

The results of the analysis were based on data obtained from one website only. It is possible that better results would have been obtained by analysing data from various websites and comparing the results. Also the website used in this study offered different services and it is likely that the average search query length for finding these services vary. Better results might be obtained by segmenting the data.

5 CONCLUSION

Previous studies have shown that search queries are an expression of intent and need [6, 9, 13-15]. By understanding the intent of Web users when they arrive at a website, it could be possible to infer whether they are ready to buy or convert. The length of a search query could be measure of a Web user's readiness to buy or convert. Short, generic search queries are usually associated with experiential users who are more interested in browsing than in making a purchase or enquiring about a product or service. Longer search queries are associated with Web users who are further along the buying cycle and who are ready to buy or complete an action that represents a conversion.

This paper proposed a model for predicting the likelihood that a Web user would purchase or complete an action that represents a conversion based on the length of their search query. Initial data analysis neither validated nor invalidated the proposed model. Further work has to be carried out to test the model.

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Fusing Virtual and Physical Networks – Breaking the Barrier

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Abstract - *The Internet is now synonymous with our very existence. Over the years, we have managed to internalize this global phenomenon. It has provided a base for many new ideas to emerge and one such idea is the concept of Social Networks. Social Networks are virtual entities that help provide a platform that facilitates in bringing people closer. Currently many social networks exist, each serving a different purpose. More importantly, they have helped in partially dissolving the barrier between our physical (real-life) and virtual social lives. But why stop there? We could do better and it would be to our advantage if the barrier in question could be completely removed, so as to facilitate an unparalleled service that fuses physical social networks of our everyday lives with a virtual environment thus providing a one stop social network for all purposes.*

Keywords: Social networks, social, circles, virtual

1 Introduction

A social network, when online is a virtual environment in which friends and family can interact with one another however far across the world they may be. Today with the various social networks put together roughly 800,000 people are part of some social network or the other.

In the present scenario a user logs into his/her account on a network and is able to see what his/her friends have been up to and is also able to update what he/she has been doing. This is the structure of the most commonly used networks "Facebook" and "Twitter." Then there are few other networks which have a slightly different structure, but in the end they all boil down to the same basic idea – "know what your friends are upto" and "be in touch with them."

This obviously has many benefits which are obvious, but the underlying connectivity is what strengthens the backbone of the system. Social Networks today are being used to mobilize people for various fruitful causes. Twitter has a record for very fast transmission of information, and especially during tragedies or times of need this has proven to be very useful.

Everything has its pros and cons. Social Networks are no exceptions to this rule. Every network has spam registrations which include people taking someone else's identity. This paves for way for stalking and harassing. As a result, identity theft and similar crimes have become easier. With all the information about a person being available in public impersonation has become a walk in the park

2 Proposed System

We would like to propose a system which integrates a user's daily life. The basic idea here is the fact that a social network should not be a virtual environment alone but should capable of interacting with the physical world as well.

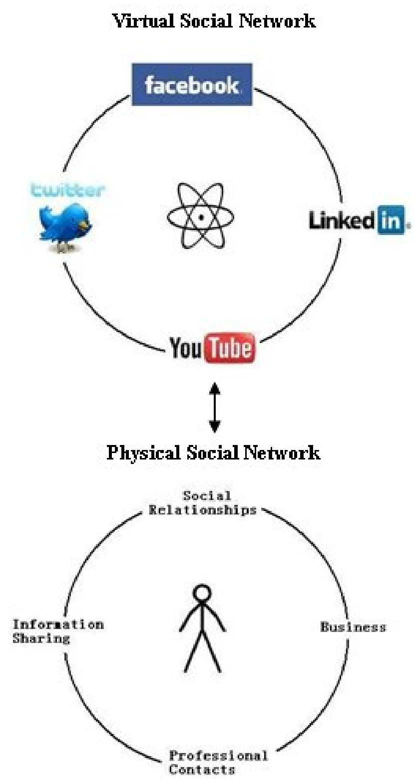

Figure 1: Mapping of Physical to Virtual Social Networks

3 Principle Idea

Our model emphasizes on categorizing contacts into what we call social circles. A social circle is a sub-environment within which you are willing to share certain information that will be visible to everyone who belongs to that particular circle only. This way a person can categorize his contacts in such a way as to share his information in such a way that it is private, yet socially public. A contact may thus be a part of more than a circle and circle could have circles within it.

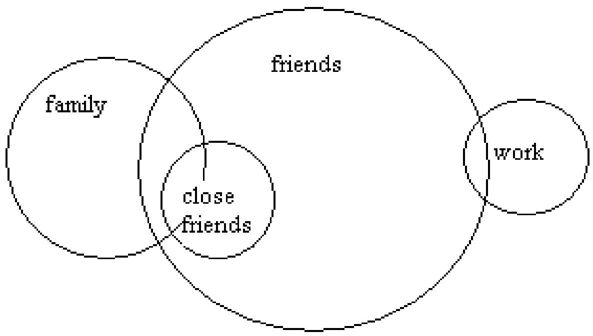

Figure 2 : Circle Concept Illustration

As mentioned previously the current social networks provide only one basic feature in a variety of ways. i.e. keeping in touch with people. As social networks grow they should be modeled in such a way to interact with one's physical life as well. Today a person may do n things in his/her day to day life out of which 2 to 3 three things may be known to his family and friends around the world. Whereas the interaction between him and the people in his house, work place etc is much more. This is the level of interaction social networks should be developed to bring about. Seamless integration to every aspect of one's life is the goal.

Although existing social networks achieve a similar end effect like social circles, there lies a stark yet subtle difference between them. Social Networks like Facebook provide facilities for creating groups and sharing information but the information is still visible to other 'friends'. The concept of social circles renders information regarding one's friends that does not pertain to one, as inaccessible. This would pave way for the one-stop social network for all purposes.

4 Improvisations

Apart from linking with people whom we already know social networks should bring us to interact with people of similar interests. This way people can form small groups with similar interests and this would help in both entertainment and productivity.

Existing social networks do rely on certain AI techniques for finding contacts but they are primarily restricted to primitive methods like keyword matching which do not accurately represent a person's interests.

A new algorithm has to be implemented which finds people with similar interests, and does not do just key word matching. This algorithm should take various factors into consideration such as level of experience, theoretical or practical knowledge etc. This should in effect be an artificially intelligent system with an ability to use data at its disposal to find people with similar interests.

Building even more on the above mentioned model can give rise to the next generation of social networks. In the future, social networks can be envisaged as a social relationship manager. They no longer just provide a platform for us to organize our social relationships but rather assist us in the management of social relationships more actively.

Each social relationship manager is an Artificial Intelligent Agent that helps us in managing our activities on a daily basis. Apart from the obvious tasks of helping in planning and scheduling, the bot can be programmed to learn about our characteristics and suggest activities or people to meet based on our inherent likes and dislikes. A sophisticated learner algorithm has to be implemented to cater towards an accurate mapping of our traits and characteristics. This algorithm would in a way create a model to discover our character profile based on our actions.

To make such a vast system implementable one of the major concerns would be the security aspect. This is where the idea of open source system architecture (like Diaspora) should be brought about where each person can manage his/her own data. The chance of a total data loss or theft then becomes a small probability.

5 Impact

The further dissolution of the barrier between virtual and physical social networks can open new avenues for social networks to pervade.

5.1 Work Life

Companies today are slowly understanding the importance of social interaction in a person's life and slowly bringing in an idea of social intranets within their company. These intranets need to play a bigger role in the assessment of an employee allowing performance rating based on activity, communication and social behavior. Today diligence in work alone does not prove enough and the winning over of clients requires finesse in all fields.

5.2 Medical Field

Today the healthcare industry focuses on treating sick or people requiring medication immediately and though stated the knowledge of prevention is not prevalent everywhere. A very usable approach would be to create a Social Network where people can join by paying some amount. They would be able to post all their activities, eating behaviors, blood sugar etc. which would be visible to medical experts who would be able to post comments/advices on the same. Such a system would be conducive in providing emergency advice to patients in case of any complications at times when medical attention is not readily available. It would also bring together people with similar issues etc enabling them to see how other people cope with these issues and how they work toward healthier lifestyle.

5.3 Education Industry

With the current boom in technology and the at the speed at which research is being conducted, students today in many developing nations and developed nations as well end up reading about older out of date technology. Bringing together research analysts into an education network would prove to be very fruitful.

Identifying students with a research potential would be easier. It would also help cultivate a research oriented approach in many students. Further the students would be up to date in current technologies and ideas when they are fit to work thus giving them a significant head start.

5.4 News Industry

The internet is pretty much the fastest updates resource in the world. Twitter a completely common man generated database of information in the form of “tweets” report news (both critical and otherwise) well before anything else. The issue in this case is that mining of this data is not yet smart enough to sieve out important information and normal human interaction.

Here a system which finds news and classifies based on locality, importance etc should be brought about in such a way as to have an almost real time flow of information categorized for easy access. This would in effect make the best possible resource of news worldwide – the common man’s outlook to the world.

6 Security

When a system houses details of almost your entire life, security becomes a major issue. It is of utmost importance that all the data is secure and cannot be broken into or stolen easily. Some of the features that could be added in order to make the level of security high are:

6.1 Self Hosted Content

If the architecture is developed allowing each user to host their own content on their server, similar to the Diaspora architecture, the security is increased to a great extent.

6.2 Personal Query System

Since the system knows pretty much everything about the user, during each login the user will be presented with a question about himself that only he/she will able to answer.

6.3 Spam Prevention Measures (SPM)

A user while registering can create an account not only by giving email id and mobile number, but every new user will be added to a temporary list. Any user on the temporary list will be have limited functionality, and this would become full functionality only if he/she is deemed a real person by a 100(say) of his her friends votes.

6.4 Content Monitoring System

The content on the system will be monitored, and in case of users below 18, all unwanted content will be filtered irrespective of messages, posts or any such communication.

7 Advantages

7.1 Making the world a smaller place

Such a system would benefit users in many ways. Currently people interact in a very small way with a limited number of people and this mostly includes people in their day to day life. This makes their outlook very narrow minded. A network such as this would strive to bring people from various backgrounds and across the world together.

7.2 Dynamic

The entire system is dynamic in nature. Information should be shared on the fly. Seamless integration as mentioned if achieved will make the Social network for all practical purposes a real time system.

7.3 Models real world behavior

The concept of social circles helps in removing the problem of information transparency, further dissolving the barrier between virtual and physical social networks.

7.4 Easier Decision Making

The system alleviates the problems we face in making decisions through its informed and intelligent suggestions derived from mapping our personality and other internal characteristics essential for the same.

8 Conclusions

The system which would bring about this structure, would not only break the barrier between the physical and virtual worlds, but also completely change human life. It would play an important, if not a major role in everyone's life. Not only would this be a great way to keep in touch with loved ones but it would be a person's personal assistant to the outside world – the people and the places he does not interact with on a daily basis.

9 Future Work

A great way to improve this system would be to improve the networks ability to relate to humans. That is an artificially intelligent system which would be able to not only monitor and find information but to understand what it means and correspondingly set the method of communication etc.

Additionally the deployment of a well structured artificial neural network and "learner" algorithms that accurately map personalities should be used to increase the quality of the functioning of the system.

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SESSION

SOCIAL NETWORKS

Chair(s)

TBA

Twitter Hash Tag Prediction Algorithm

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ABSTRACT

Social media has demonstrated quick growth, in both directions of becoming the most popular activities in internet and of attracting scientific researchers to get better insights into the understanding into the underlying sociology. Real time micro-blogging sites such as Twitter, Flickr and Delicious use tags as an alternative to traditional forms of navigation and hypertext browsing. The tag system of those micro-blogging sites has unique features in that they change so frequently that it is hard to identify the number of clusters and so effectively carry out classification when new tags can come out at any time. In this paper, we propose to use Euclidean distance between points as the measurement of their similarity. Our method has advantages in easy data storage and easy accommodation to personal settings. In experiment, we compare our model with other classification functions and show that our model maintains a false positive rate lower than 15%. Our work is relevant for researchers interested in navigating of emergent hypertext structures, and for engineers seeking to improve the navigability of social tagging systems.

KEYWORDS

Social Networks, Twitter, Hash Tags, Social Tag Prediction, Ontology Based Distance

1 INTRODUCTION

Social media has demonstrated exponential growth, making it the most popular activity on the internet [1]. Real-time micro-blogging services such as Twitter, Flickr and Delicious are widely recognized for their social dynamics – how they both encapsulate a social setting propagate information across it [2, 3, 13, 14, 15, 21].

Social tagging [26, 29] is a method for Internet users to organize, store, manage and search for resources online [9, 12]. Trant [28] categorizes the existing works on social tagging into three broad topics: (a) on the *folksonomy* that results from the collective wisdom of users of the social tagging system; (b) on the *tagging behavior* of users, such as the incentives and motivation for tagging; (c) on the software aspects of the *social tagging systems*, for improving system performance and enhancing user satisfaction.

Social tag prediction belongs to the third category. In particular, it aims at enriching tags for Web resources that are untagged or inadequately tagged [12]. The Internet users are benefited by this technique because it make search in webs become easy [17]. The current research in this direction can be classified into three categories: (1) determining topics from hypertext content [23], (2) predict new trend on topics based on existing tags [5, 10, 17, 18], and (3) enriching tags from other similar or linked resources [4, 22, 27.

Twitter is one of popular web applications nowadays [19, 20, 25]. Twitter allows users to use "Hash tags" to classify their tweets. In this research project, we propose an algorithm to predict tags, by utilizing machine learning and network relatedness methods.

Hash tag prediction is different from normal texts classification mentioned in the above. In a real time micro-blogging site, we don't know how many clusters needed to be found. In addition, the tag set changes so frequently that it is almost impossible to effectively carry out classification or clustering, since a new tag would force us to establish a new class and a new classification rule. Our intuition is: if we can measure the correlation between various tweets as the mathematical metric we can treat the collected tweets as points in a high dimensional space, and construct a network by the latent space model. We show that simple techniques are sufficient to extract key semantic content from tags and also filter out extraneous noise. We demonstrate the efficacy of this approach by comparing it with other classification functions and show that our model maintains a false positive rate lower than 15%.

The paper is structured as follows: In Section 2 we briefly introduce Twitter and its hash tag system. Section 3 presents our theoretic approach to assessing distance of tagging systems. We propose to use the ontology based distance between points as the measurement of their similarity. Our method has advantages in easy data storage and easy accommodation to personal settings. Section 4 presents and discusses the analysis results. Section 5 concludes the paper.

2 TWITTER

Twitter is one of the fastest growing Web 2.0 services. It is called a micro-blog because people can post short, quasi-public messages up to 140 characters in length. People create lists of others and are shown a list of all of the posts of those people. The substantive nature of the social tie on Twitter is attention-based [7, 8]. In addition to paying attention to one another by "following," Twitter users can address tweets to other users and can mention others obliquely in their tweets [11]. Another common practice is "retweeting," or rebroadcasting someone else's message (with attribution) so as to direct attention toward that person's tweets [1].

Twitter differs from other online social networking services in that ties are asymmetric [7, 8]. Consider friendship ties in LinkedIn, Facebook, or MySpace; in these services, when two people share a friendship tie, the tie is symmetrical; A being friends with B implies B is friends with A. This is not the case in Twitter; A can "follow" B, but B needs not follow A. People who are popular, such as basketball players or actors, can be followed by millions of people, but can barely pay attention to all of those who follow them.

The hash tag (the # sign followed by a phrase to a tweet, for example #superbowl) is probably the most important function of Twitter search, and the most used. The hash tag enables Twitter users to create searchable subject groups and so to be able to navigate the hypertext structures of the whole site. The power of the hash tag is that it creates very specific sets of content. If you want to know what other people think of the superbowl that just came on you can find it easier by searching for the hash tag than by searching for something similar in a normal search engine. Every day, many new hash tags are formed and this process can happen right before your eyes-heck. The frequent creation of new tags makes the prediction of tags challenging. This motivates us to develop the following method.

3 METHOD

3.1 Theory

An intuitive way to solve this problem is to use Euclidean distance between points as the measurement of their similarity. We developed our theory based on this distance. Since in a Euclidean Space, the distance is equivalent to the norm of a vector, we will focus our discussion on norms.

Let $\mathbf{u}_1, \mathbf{u}_2 \cdots \mathbf{u}_p$ be the standard bases (with unit norm) of a p-dimensional Euclidean Space. Then for any vector $\mathbf{v}$ with coordinates (x_1, x_2, …, x_{p-1}, x_p), we have $\mathbf{v} = \sum_{i=1}^{p} x_i \mathbf{u}_i$. Then the Euclidean norm of vector $\mathbf{v}$ is given by:

$$\|\mathbf{v}\|^2 = \mathbf{v}\cdot\mathbf{v} = \sum_{i=1}^{p} x_i \mathbf{u}_i \cdot \sum_{i=1}^{p} x_i \mathbf{u}_i = \sum_{i,j=1}^{p} x_i x_j \mathbf{u}_i \cdot \mathbf{u}_j \quad (1)$$

where · represents the inner product operation defined in the Euclidean Space. Clearly, if we assume $\mathbf{u}_i \cdot \mathbf{u}_j = 0$, that is, $\mathbf{u}_i$ and $\mathbf{u}_j$ are orthogonal, whenever $i \neq j$, the Euclidean norm equals to $\|\mathbf{v}\|^2 = \sum x_i^2$. In our problem, the bases are the words in the dictionary. The preliminary assumption for Euclidean distance is that the bases are orthogonal to each other, that is, the words in dictionary are uncorrelated, which is against common sense. Therefore, we need to perform some transformation to capture this correlation.

In Equation (1), as $\mathbf{u}_i$ and $\mathbf{u}_j$ are unit vectors, their inner product is actually the cosine of the angle between them. Thus we can rewrite (1) in a matrix form as:

$$\|\mathbf{v}\|^2 = \begin{pmatrix} x_1 & \cdots & x_p \end{pmatrix} \begin{pmatrix} \cos\theta_{11} & \dots & \cos\theta_{1p} \\ \vdots & \ddots & \vdots \\ \cos\theta_{p1} & \cdots & \cos\theta_{pp} \end{pmatrix} \begin{pmatrix} x_1 \\ \vdots \\ x_p \end{pmatrix} = XMX^T \quad (2)$$

where $\cos\theta_{ii} = 1$, i=1, …, p, and x = (x_1, x_2, …, x_{p-1}, x_p).

Now we try to find the angle between each pair of terms in the dictionary and then calculate the matrix M . Notice that M is clearly a symmetric and non-negative definite matrix. If we decompose M in the way

$$M = CC^T \quad (3)$$

then (2) becomes

$$\|\mathbf{v}\|^2 = XCC^T X^T = \tilde{X}\tilde{X}^T \quad (4)$$

where $\tilde{X} = XC$. So the norm can be seen as the Euclidean norm of the transformed coordinates. Here we take (3) as the Eigen value decomposition of M, so $\tilde{X}$ could be the coordinates of vector $\mathbf{v}$ in a new coordinate system where axes are orthogonal to each other. Please note that we can use any other decomposition in the form of (3) to get the same norm in computation, even when C is not a square matrix. With this property, the computation becomes applicable.

3.2 Estimate the Cosine Matrix

First, we construct the preliminary weighted matrix, say H, by using the WordNet to initialize the semantic correlation among words from the dictionary. If two words t_i, t_j are similar to each other, and they both appear in one Tweet, we add positive weights for both words. This process can be expressed as

$$\hat{x}_i = x_i + \sum_{j \neq i}^{p} \rho_{ij} x_j \qquad (5)$$

where $\rho_{ij} \in (0,1)$, equals to one when t_i, t_j are similar words and zero otherwise. Here we take the same positive number ρ for all $\rho_{ij} \in (0,1)$, and if $\rho_{ij} > 0$, so is ρ_{ji}. Then we can construct the symmetric matrix H as:

$$\mathrm{H} = \begin{pmatrix} 1 & \cdots & \rho_{1p} \\ \vdots & \ddots & \vdots \\ \rho_{p1} & \cdots & 1 \end{pmatrix} \qquad \hat{X} = X\mathrm{H} \quad (6)$$

In the second step, we get m tweets, say $X_1 \cdots X_m$, and transform them by (4) to get $\hat{X}_1 \cdots \hat{X}_m$. Then by these data, we use cosine similarity in variable analysis to construct matrix M. Set the text matrix as the $m \times p$ matrix :

$$\Omega = (\hat{X}_1^T \quad \cdots \quad \hat{X}_m^T)^T = \begin{pmatrix} \hat{x}_{11} & \cdots & \hat{x}_{1p} \\ \vdots & \ddots & \vdots \\ \hat{x}_{m1} & \cdots & \hat{x}_{mp} \end{pmatrix}. \qquad (7)$$

We would estimate the cosine between the i[th] and j[th] terms as

$$\cos\theta_{ij} = \hat{X}_{\bullet i}^T \hat{X}_{\bullet j} \Big/ \left\| \hat{X}_{\bullet i} \right\| \left\| \hat{X}_{\bullet j} \right\| = \frac{\sum_{k=1}^{m} \hat{x}_{ki} \hat{x}_{kj}}{\sqrt{\sum_{k=1}^{m} \hat{x}_{ki}^2 \sum_{k=1}^{m} \hat{x}_{kj}^2}} \qquad (8)$$

The distance estimate obtained from formula (5) is equivalent to what proposed by [16], but with a better mathematical explanation. Note that since our data is represented as frequency, all the elements of the matrix Ω would be non-negative. So the cosine estimated in this way can only be non-negative. Therefore, all angles between words are cute or right angles. In this way, all words tend to be similar to each other in some degree. This may well incorporate the similarity elements, but might also be vulnerable to noise. In the following, we give a modified estimate which also includes the possibility of obtuse angle and takes dissimilarity into consideration, which is also the sample correlation in statistics,

$$\cos\theta_{ij} = \frac{\sum_{k=1}^{m} (\hat{x}_{ki} - \hat{x}_{\bullet i})(\hat{x}_{kj} - \hat{x}_{\bullet j})}{\sqrt{\sum_{k=1}^{m} (\hat{x}_{ki} - \hat{x}_{\bullet i})^2 \sum_{k=1}^{m} (\hat{x}_{kj} - \hat{x}_{\bullet j})^2}} \qquad (9)$$

Since the distance from formula (8) was named as Ontology Based Distance (OBD) in [16], here we call the distance in formula (9) centralized Ontology Based Distance (COBD). We will discuss the pros and cons of the two methods in experiments. In the following sub section, we will make another adjustment to the method.

3.3 Normalization

Note that the various scales of vectors may still cause us some problem. Consider a special case where $\tilde{X}_1 = (1,0,0)$, $\tilde{X}_2 = (10,0,0)$, $\tilde{X}_3 = (0,0,1)$. Obviously, $\tilde{X}_1$ and $\tilde{X}_2$ should have high similarity value between them. But in this case, the distance between $\tilde{X}_1$ and $\tilde{X}_3$ is much smaller.

To make our method more reasonable, before we compute the distance between transformed points, we need to rescale their distances to the original point as 1. And then we measure the Euclidean distance between normalized points.

3.4 Prediction of Tags

Finally, we predict tags based on the distance. One intuitive way is to simply select the tag of the closest tweet. In this case, it may be unwise to simply pick the closest tweet's tag, since that is not resistant to noise. To increase the accuracy, we collect a few closest tweets, and make the prediction based on tag ratios. Specifically, we will collect *n* initial closest tweets at first (*n* usually ranges from 4 to 6). Then from this point, we will keep adding tweets while check a certain tag has become dominate. If there is a tag with a ratio higher than 50%, we will choose this tag as our primary predicted tag. Since in some cases tags have very similar meanings (such as #government vs. #election), sometimes we will also pick a secondary tag to predict.

4 EVALUATIONS

To compare the performances of various distances discussed above, we use a test dataset consist of 400 tweets that are not included in the sample set we used to estimate matrix M. There are 4 different tags. We first process the OBD on a dataset with 665 tweets that are not in our test set, choose the best performance ρ (=0.2) and use it for both OBD and COBD. The table below shows the test result for Euclidean Distance (EucD), OBD and COBD.

	Test Error Rate	Type II Error
EucD	16.25%	5.1%
COBD	13.5%	4.6%
OBD	12.75%	4.2%

Table1: The test error rate and type II error for three distances. Type II error is the rate we assign a wrong tag to a particular tweet.

Both OBD and COBD outperform EucD, and OBD is the best one. If we see the data for different tags (not provided here for concise), we would find COBD is the most stable one, while EucD is far more unstable. But the disadvantage of COBD lies in computation. We need to estimate the cosine matrix M to construct the distance, which involves computation for matrices with tens of thousands rows and columns. It won't be a big problem for OBD since the matrices are sparse. But in COBD, the matrix becomes non-sparse, so we need many decompositions and transformations of matrices to make the computation applicable. Given their close performances, OBD is more practical in application, while the COBD is a better model theoretically.

The top picture in Figure 1 shows the COBD from other tweets to a random selected tweet. Different colors represent tweets with different tags. It can be seen that most of the tweets are very close to the 1.4142 distance boundary, and the majority of points falling in the circle are from the correct tag group. This indicates that tweets with different topics are projected onto orthogonal axes. The right plot illustrates the distance distribution. The lighter the color is, the shorter the corresponding distance is. Since the tweets are sorted by tags, we see that the distance within each group appears to be shorter, as shown by the light rectangles along the diagonal.

In Figure2, different colors represent what tag cluster the tweets belong to. A link will be added between a pair of nodes when they are near enough. In addition, the deeper color the line is, the higher the similarity value is. As we can see, the lines appear to be very dense among each tag cluster, and sparse between tweets with different tags. It indicates that tweets with the same tag cluster are near on average.

Due to the vagueness of many tweets, the correct rate of more than 86% is actually very high. Apart from the accuracy, our method has other advantages:

(1) The whole system is easy to store (we only need to store the C matrix in Equation (3)).

(2) It is easy to update when dictionary changes (only needs to compute an extra column and add it back to original matrix).

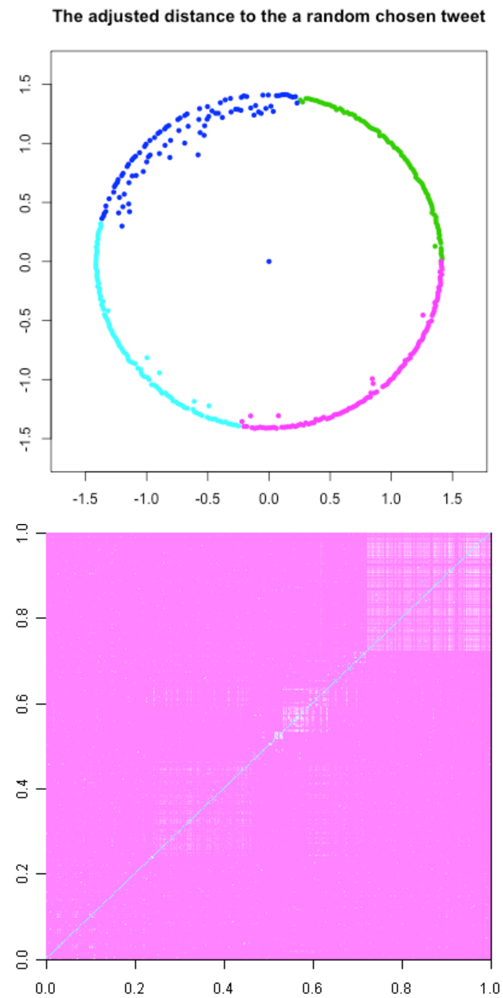

Figure 1 Distance to one point and the distribution of sample distance matrix

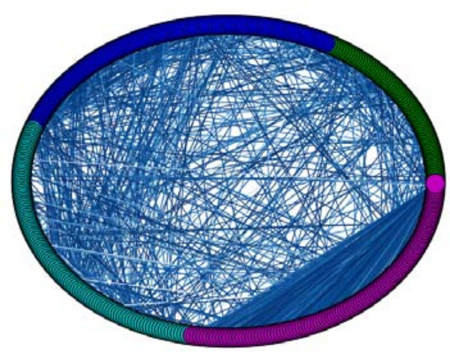

Figure 2 Prediction Visualization

(3) It won't lose power when the topics trend changes with time, and it can work with personal elements and settings, which makes it more flexible (since we can set the algorithm to only consider the distance of the objective tweet to certain subset of

other tweets, so elements like location, time, etc can be incorporated.)

(4) In addition, the distance provides us with the possibility to transform the twitter system and even other text systems into social networks by latent space approach. So we can use traditional social network methods to discuss the properties of such systems.

5 CONCLUSION

In this paper, we have presented a distance function to classify hash tags in Twitter. A major challenge to the social tag prediction problem with a micro-blog like Twitter is that the underlying dataset is updated frequently by millions of the Twitter online users. We propose a distance function that utilizes machine learning technology and latent space models. We map the collected tweets to a high dimensional space and construct a latent network to predict the similarity of these tags. Our model is general in terms of that it allows the flexibility of adapting users' personal settings. We show that simple techniques are sufficient to extract key semantic content from tags and also filter out extraneous noise.

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Credible User Identification using Social Network Analysis in a Q&A Site

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Abstract - *Although recent commercial search engines use the information concerned with Question and Answering (Q&A), it is still difficult to acquire an appropriate content from numerous user-described answers in Q&A sites. In order to identify the credible users to help people find relevant answer, in this paper, we propose a ranking algorithm, InfluenceRank, which is basis of analyzing relationship in terms of users' activities and their mutual trusts. Our experimental studies show that the proposed algorithm significantly outperforms the baseline algorithms.*

Keywords: Knowledge Sharing, Question and Answering, Social Network Analysis, Identifying Influencers

1 Introduction

A Q&A site (e.g., Navers Knowledge iN[1], Yahoo! Answers[2], Answers.com[3], eHow[4] and Baidu Zhidao[5]) can be considered as an information system between people in terms of knowledge sharing, and the knowledge sharing site is recently gaining popularity on the Web. Each Q&A site usually has a large pool of questions and their answers which are related to every thing or matter user interested. This mechanism can reduce the time and effort to find the most relevant answer from the pool, because it is higher priority that there already exists related answers provided by other users. Unfortunately, the above approach suffers from answer quality problem, i.e., the answers in the system may be irrelevant and/or poorly written even if their associated questions are relevant. Users give poor quality answers due to several reasons including limited knowledge about the question domain, bad intentions (e.g., spam, making fun of others), limited time to prepare good answers, etc. Therefore, identifying credible users became a very hot issue in Q&A sites.

Askers usually want the answer(s) by referring to authoritative users' expertise. So identifying prominent actors in Q&A sites is important to make finding valuable knowledge possible by estimating the authority of Q&A site users. Therefore, enhancing the visibility of authoritative users on a Q&A site and connecting askers with experts will be important function in fostering communities around shared interests. Web 2.0 communities are needed to encourage the collaboration and sharing knowledge between their users, while authoritative users can actively participate and select useful information by answering and voting, and then saliently improve the quality of Q&A site contents. Various approaches can be employed to find experts in Q&A sites. One way is to adopt the link analysis approaches such as HITS **[12]** and PageRank **[19]**. Jurczyk et al. **[10]** used HITS algorithm for author ranking of a Q&A site. They represent the relationship of the asker and the answerer as a SN and calculate each user's hub and authority value in order to subsequently rank users according to their authority values. Liu et al. **[18]** use expert profile, built from the contents of expert's questions and answers, to find experts without considering the reputations of experts and their authority values derived from link analysis.

In this paper, we propose the *InfluenceRank* algorithm to identify credible users, called *influencers*, to answer a given target question in a Q&A site. Influence is defined as 'the act or power of producing an effect without apparent exertion of force or direct exercise of command' or 'the power or capacity of causing an effect in indirect or intangible ways'.[6] Our assumption is that the users who have many positive choices in a Q&A site network can represent the credible users. InfluenceRank algorithm is based on users' interactions in a Q&A site, and we assess *Activity* and *Trust* which are considered as main factors to measure influence value. Basically, a person can trust another person by the degree of perceived reliability or the belief in their concerned intentions. This belief will flow through online user relations that reflect paths along which users share their experiences with friends based on closeness and trustfulness **[22]**. Users might trust a given person more than others. In other words, they might be influenced by him/her more than by others **[26]**. Therefore, trust is one of the most important factors when sharing knowledge on the Web, especially within Q&A-style knowledge networks

The reminder of this paper is organized as follows: In Section 2, we introduce the related works. Section 3 demonstrates the factors to measure influence value, and then

[1] http://kin.naver.com, Includes more than 80 million Q&A information in January 2008,
and Yahoo! Answers bench-marked this for their launching
[2] http://answers.yahoo.com
[3] http://www.answers.com
[4] http://www.ehow.com
[5] http://zhidao.baidu.com

[6] Merriam-Webster Online Dictionary, http://www.merriam-webster.com/dictionaly/influence

describes our proposed *InfluenceRank* algorithm to identify credible users. In Section 4, we analyze the proposed algorithm based on the experimental studies. Finally, Section 5 concludes this paper.

2 Related Works and Background

2.1 Finding Experts

For finding the experts in community-driven Q&A sites, the most widely used method is SNA. The general way is building a user SN first, and then using some kind of propagated algorithm to measure each user's reputation. Chen et al. **[4]** considered five types of relations between users that may influence reputation to build a SN graph in which each type of edge has different weight and proposed a reputation computation mechanism to compute the influence value when the graph is changed. Jurczyk et al. **[10]** built a SN graph based on asker-answerer relation between users and adopted the HITS algorithm to measure each user's authority value. Zhang et al. **[16]** proposed the ExpertiseRank to rank experts in an expertise network considering how many people are involved and who has helped whom. Liu et al. **[18]** proposed a method that uses the content of Q&A pairs to build expert profiles, and then found the experts by comparing the similarities between the contents of questions and expert profiles. Zhang et al. **[27]** not only compared the similarity of questions and user profiles but also considered the differences of expertise level, posting time of query, and the number of replies to questions. Bouguessa et al. **[3]** addressed the drawbacks of link analysis and proposed an approach which aggregates a mixture of gamma distributions, the Bayesian Information Criterion, and the Expectation-Maximization algorithm.

2.2 Finding High Quality Answers

According to Su et al. **[23]**, the average quality of answers in Q&A sites is good, but the quality of specific answers varies significantly. In particular, in a study of the answers to a set of questions in Yahoo! Answers, the authors found that the fraction of correct answers to specific questions asked by the authors of the study varied from 17% to 45%. The fraction of questions in their sample with at least one good answer was much higher, varying from 65% to 90%, meaning that a method for finding high-quality answers can have a significant impact in the user's satisfaction with the system.

Jeon et al. **[9]** extracted a set of features from a sample of answers in Naver's Knowledge iN, a Korean Q&A site similar to Yahoo! Answers. They built a model for answer quality based on features derived from the particular answer being analyzed, such as answer length, number of points received, etc., as well as user features, such as fraction of best answers, number of answers given, etc. Agichtein et al. **[2]** expanded on the prior work **[9]** by exploring a substantially larger range of features including both structural, textual, and community features, and by identifying quality of questions in addition to answer quality.

3 Proposed Algorithm

Widener **[17]** suggested that influence is the human capacity educing ability and making an aura of positive change of members in a given space. Under the theme of SNS, the influence can be described as 'the power that makes an individual's thoughts, feelings, attitudes or behaviors change, which results from interaction with another individual or a group' **[5, 8]**. Social influence can be found when a person affects other people's thoughts and/or actions. Deutsch et al. **[6]** described two psychological needs that lead humans to conform to the expectations of others: informational social influence and normative social influence. There are various forms of social influence in conformity, socialization, peer pressure, obedience, leadership, persuasion, sales and marketing. Kelman **[11]** identified three broad (e.g. compliance, identification, internalization) varieties of social influence. In brief, we can consider social influence as the influence in a society composed of interacting members.

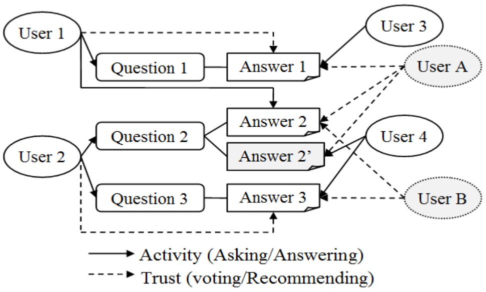
In Q&A sites, influence can be regarded as the force that lets users choose an answer as credible and relevant one among other answers of a given question. By analyzing node-edge connection, number of interactions (edges) of a user (node) can show her influence in a Q&A site. In other words, a user can be denoted as an influencer if he/she has frequent interconnections with others and these connections are believed to be trustful. For example, if a user posts many answers to questions and many of his/her answers are selected as the best answers by voting, we can consider this person as an influencer – an expert in his/her domain of knowledge (represented by the set of questions/answers).

User relationships can be regarded as the activities such as questioning and/or answering. The basic reliability of a user's activities can be measured by the number of activities, such as the number of questions/answers posted, number of answers in own questions or his/her selected answers of other users' questions. In communications, trust is a specific context that regulates degree of belief. For example, the trustful station can serve the credible communication **[14]** and it can obtain the perceived reliability and competence **[24]**. Therefore, Activity and Trust is chosen as influence factors to measure influence value.

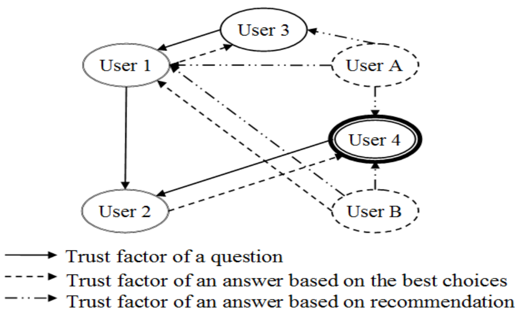
3.1 Q&A Network Construction

In online help-seeking communities SN is an expertise network. Because the way links are constructed, the prestige measure of the network is highly correlated with a user's expertise **[25]**. Thus, this hints that there are opportunities to make use of such network structures to rank people's expertise in Q&A sites, and to build related applications/ systems that further improve the expertise sharing in Q&A sites. To measure influence value, therefore, we analyze characteristics of Q&A networks by using Social Network Analysis (SNA).

In the Q&A sites, a user posts a question and other users give answers or user feedback. As shown in **Fig. 1(a)**, we regard users in the Q&A site as nodes, and their actions as edges: user1 and user2 post questions and select a best answer among answers, user 3 and user 4 give answers, while userA and userB post the feedback (e.g., comment, thumb up/down, and voting). Note that user1 also posts an answer to the question of user2. We can express the asker-answerer-recommender network like **Fig. 1(b)** based on the user interactions. We consider the directed graph $G(V, E)$. Nodes in V represent the users, and each directed edge in E indicates relationship. As an example, user3 answers the question of user 1, and user A comments on or votes the answer from user3.



(a) Link structure of Q&A sites



(b) User network of Q&A sites

Fig. 1 Structure of the Q&A sites according to activity and trust.

For finding the experts in community-driven Q&A sites, the most widely used method is SNA. The general way is building a user SN first, and then using some kind of propagated algorithm to measure each user's reputation. Chen et al. **[4]** considered five types of relations be

3.2 Activity Measurement

We define a user's activity in an asker-answerer network as the frequency of producing and consuming this knowledge. The activity value can be measured by the number of user behavior, such as the number of questions/answers posted, number of answers in own questions or his/her selected answers of other users' questions. The activity value of an i-th user, $ACT(u_i)$, is given by

$$ACT(u_i) = m + n \tag{1}$$

where m and n are the total number of questions and answers posted by i-th user, respectively. If a user asks and answers many times in the Q&A site, the user gets a high activity value.

3.3 Trust Measurement

We define a user's trust in a Q&A site as the trustworthiness of the knowledge based on user interactions (e.g., 'selecting' the best answer). For measuring trust, we use the analysis of degree centrality which is frequently used when measuring authority and influence value **[7, 16, 21, 25]**. In a Q&A site, the degree of links can be shown using questioning, answering, and recommending. In view of the asker, the question is an outbound link to the answer, and an answer is an inbound link from the answerer. With answerer's view, the question and recommendation are inbound links from the asker and from the neighbor user, respectively. As shown in **Fig. 1(a)**, we define a link formed by answering a question as an inbound link, and a link formed by selecting and recommending a best/good-answer as an outbound link in view of an asker. Considering a graph $G := (V, E_A)$ and E_A the set of the directed connections $E = (e_{1,1}, e_{1,2}, \ldots, e_{i,j})$ between the users of the set V then the indegree centality K_i^{in} and outdegree centrality K_i^{out} of a vertex V_i are the sum of the inbound connections and outbound connections to that vertex, respectively.

$$K_i^{in} = \sum_{j=1} e_{j,i} \text{ and } K_i^{out} = \sum_{j=1} e_{i,j} \tag{2}$$

$e_{i,j} = 1$ if there is an inbound link between i-th user and j-th user, and $e_{j,i} = 1$ if there is an outbound link between i-th user and j-th user with i-th user in the center. As a result, $e_{i,j}$ and $e_{j,i}$ increase if an answer is posted on the i-th asker's question by the j-th answerer, and if the i-th answerer as a best/good answer, respectively. Indegree centrality is the number of times that j-th answer gives an answer to a question of i-th asker. Outdegree centrality is the number of times that i-th asker selects of recommends a j-th answer as a best/good answer. The indegree centrality and the outdegree centrality of a vertex make sense only in cases where a directional relationship is available (the connection is non-reciprocal). However, there exist reciprocal connections (or non directed connections) in the user network of Q&A site. In that case, the degree centrality is computed by the influence domain of the vertex. For a non directed graph $G := (V, E, \overline{E})$ the influence domain of a vertex V_i is the number or proportion of all other vertices which are connected by a path to that particular vertex.

$$\overline{d}_i = \frac{1}{N-1} \sum_{j=1} \overline{e}_{j,i} \tag{3}$$

On the above measure E represents the set of paths between the vertices V_i and V_j and $N - 1$ is the number of all available nodes in the user network of Q&A network (The total number of nodes $N = |V|$ minus the node that is subject to the metric). According to these centralities in terms of prestige, we define indegree centrality and outdegree centrality of *i*-th users as a combination of the above two metrics. The trust indicator of a question of *i*-th user, $TQ(u_i)$, the indegree centrality measured from the number of answers, and the trust indicator of an answer, $TA(u_i)$, the outdegree centrality, measured by the frequency of best-answer choice or good-answer recommendation. The degree centrality of *i*-th user encompasses the normalization of the indegree and outdegree of the user by its degree of influence, which are given by

$$TQ(u_i) = \frac{k_i^{in}}{d_i} \quad \text{and} \quad TA(u_i) = \frac{k_i^{out}}{d_i} \qquad (4)$$

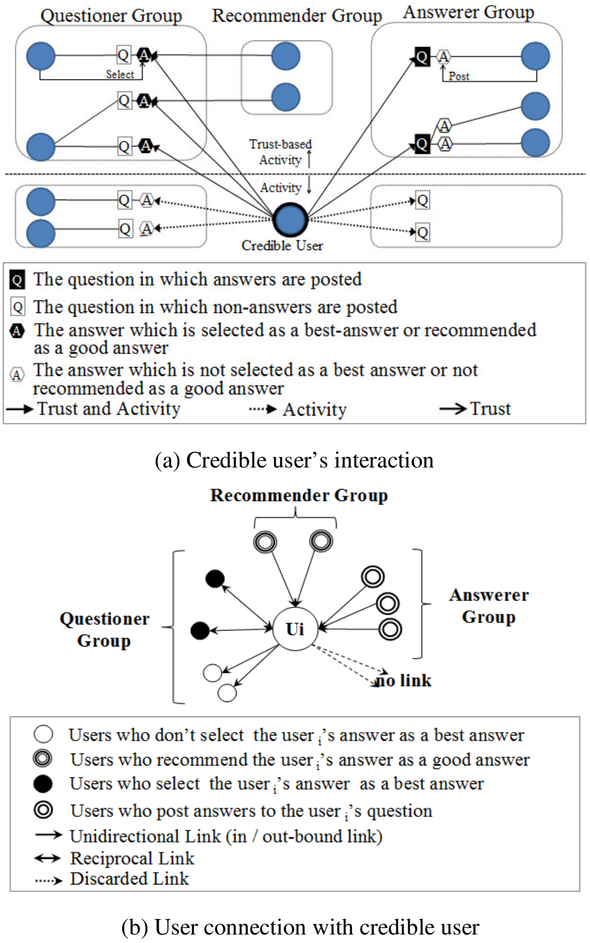


Fig. 2 Identifying Credible Users based on InfluenceRank algorithm

3.4 Credible User Identification

We focus on the relationships between users, i.e., that of asker-answerer whose answer is selected as a best answer by an asker, and an answerer whose answer is recommended as a good or best answer by other users-recommenders. A user can post multiple questions simultaneously, and there can be one or more answers in a question. The asker can choose the most intended answer as the best one, or can use voting result of other users. We assume that there are trust networks among askers, answerers and recommenders following this procedure, which is the source of discovering credible users. **Fig. 2** shows the scenario of our approach to identify the credible user in Q&A site.

The more answerers give an answer to a *i*-th asker's question and the more askers select or neighbor users recommend an answer as a best or good answer, the more credibility increases in terms of activity and trust. As shown in **Fig. 2(b)**, the degree centrality in trust network constructed in Q&A site can be used as a very meaningful factor in terms of credibility. The *InfluenceRank* algorithm of *i*-th user, $IRank(u_i)$, can be defined as follows:

$$IRank(u_i) = \alpha \times \left[\left(\frac{\sum_{p=1}^{m} |A_p|}{m} \right) \times TQ(u_i) \right] + (1-\alpha) \times \left[\frac{s}{n} \times \frac{\sum_{q=1}^{n} |R_q|}{n} \times TA(u_i) \right] \qquad (5)$$

where m and n are the total number of questions and answers posted by *i*-th user in terms of activity. A_p is the number of answers posted by other users per question, s is the total number of answers selected as a best-answer by corresponding user and R_q is the number of times the answer is recommended as a good answer by other users per answer. $TQ(u_i)$ and $TA(u_i)$ are indegree centrality and outdegree centrality in terms of question and answer, respectively (see Eq. 2 and Eq. 4). α is balance parameter ($0 \leq \alpha \leq 1$).

3.5 Optimizing Parameter

Parameter α in Equation (5) is the balance parameter considering questions and answers in each category. For optimizing α, we use the Backpropagation Neural Network (BPNN) learning algorithm which showed good performance in Artificial Neural Network (ANN) analysis. Details about the ANN and network architecture are found elsewhere (see **[13, 15, 20]**)

4 Experimental Analysis

4.1 Evaluation Metrics

We evaluate our algorithm's effectiveness by using Normalized Discounted Cumulative Gain (NDCG) **[8]**. NDCG is the accuracy checking method of Web search results' ranking. Unlike Precision/Recall or F-score, NDCG uses the observation that most of Web users refer to search results in top ranks, then counts relevance score (*rel*) of each ranks to discriminate their differences and make better ranking function. In order to apply NDCG accurately, we need to check *rel* points of each answer. 3 point measurement for quality score is used to evaluate answers posted by users. 30 editors[7] are carefully chosen, and they keep the topic range in mind before grading answers. The relevancy of given answer is divided into three scales – 'Suitable', 'Common', and

[7] 3 instructors, 1 postdoctoral research fellow and 26 graduate students (6 Ph.D. and 20 master) from Columbia Univ., Lehigh Univ., Kyoto Univ., Korea National Defense Univ., and Yonsei Univ.

'Unsuitable' shown in **Table 1**. For measuring the trustfulness, questions are divided into two groups – intellectual questions (questions requiring intellectual knowledge) and trivial questions (questions requiring small knowledge such as hints/tips in daily life). Then the trustfulness score is graded into three scales – 'High', 'Normal', 'Low', depending on needed answer levels of the question. **Table 2** shows the trustfulness assessment guidelines.

Table 1. Evaluation Criteria of Answer's Relevancy

Relevancy	Score	Guidelines	Questioner's Confidence Index
Suitable	2	• All matters in the question are answered and have sufficient data • Some matters of the question are not answered, but have sufficient data to support the answer	More than 60%
Common	1	• Some matters of the question are answered.	30 ~ 60 %
Unsuitable	0	• All matters in the question are not in the answer and have insufficient data to support the answer • Partially useful to support the answer, but matters in the question do not match with the answer	Less than 30 %

Table 2. Evaluation Criteria of Answer's Trustfulness

Trustful-ness	Score	for Intellectual Questions	for Trivial Questions
High	2	• Accurate source corresponded with public trust • Firm objectives and basis (theoretic/ scholarly source) • Logical organization (pictures, graphs) • Affirmative answer to the question	• Logical, personal opinion • Proverb, maxim, Hints / tips in daily life • Not scholarly, but practically make sense • Affirmative answer to the question
Normal	1	• Answered but insufficient source	• answered logically but insufficient source
Low	0	• Slanders, curses, lascivious comments • Disrespectful comments • Assuming comments • Useless opinions • Answer which is not concerned with question • Advertisements	

4.2 Baseline Methods

We set three baseline methods: HITS **[12]**, PageRank **[19]** and Point System **[17]** mentioned in the Section 3.4. To compare the performance of our approach with that of PageRank and HITS, we calculate authority score.

4.3 Data Sets

We gathered a large portion of the Naver's Knowledge iN, retrieved questions and corresponding answers. We choose 'Sports' category as the sample, due to its 'Forum' type network which is useful to explain mutual activities among users **[1]**. To obtain objective data set, we select users by using systematic sampling method in this category. User interactions were collected from March to August 2009, and total 997 users and 15,108 Q&A pairs were gathered. We use 10,576 Q&A pairs (70%) as training set to adjust parameter α, and 4,532 pairs (30%) as test set to show our algorithm's effectiveness.

4.4 Parameter Estimation

We use SPSS Clementine[8] tool to determine α. 627 users are selected as training set. α was adjusted to 0.12.

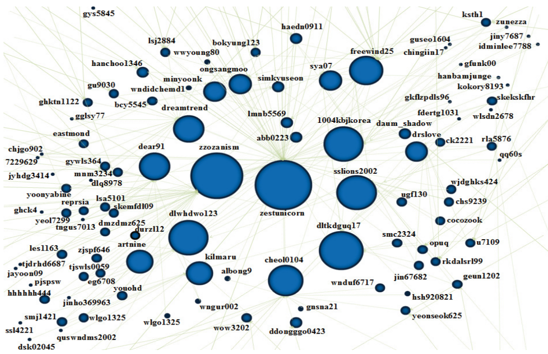


Fig. 3 Visualization of Influencers in sports category

4.5 Credible Users Identification

We use our approach to identify top-k credible users, called influencers, in 'Sports' category. We analyzed and visualized the network structure of users for discovering influencers by SNA tool. **Fig. 3** shows an example network structure over the top-k credible users. The size of circle means the degree of influence. The influencer(s) in a Q&A site are the user(s) who is in the center of given user network. We can observe that ego network of users and influencers based on influence value measured by InfluenceRank algorithm.

4.6 Evaluation Results

The InfluenceRank outperforms over all baseline methods in this case. **Fig. 4** shows the evaluation results. Higher NDCG@k score means the proposed algorithm shows better performance through discovering more influential user.

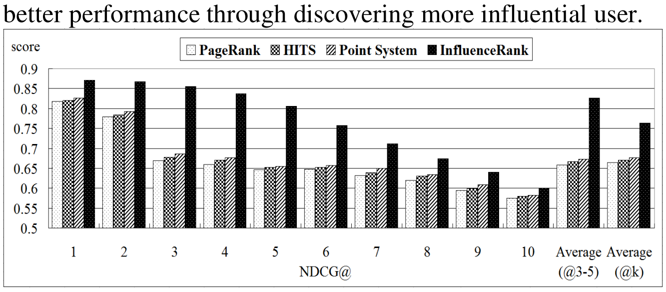


Fig. 4 Evaluation results by Average NDCG@k

[8] SPSS Clementine is a data mining software tool by SPSS Inc. We use 10.1 version for our experiment, and it was renamed PASW Modeler 13 on April 6, 2009 by SPSS.

As a result, InfluenceRank cases significantly outperform all baseline methods in all cases. *Average count of NDCG@k* shows maximum 17.11% improvement over PageRank, 16.49% improvement over HITS and 15.29% over Point System. *Maximum improvements are shown in NDCG@3* (Note that the 3-answer case is about the half of the whole data), 21.26 ~ 34.32% over PageRank, 19.52 ~ 33.50% over HITS and 18.84 ~ 31.64% over Point System. *Average improvements of NDCG@3 ~ 5* (83.3% ~ 95.4% of data set), are 19.15% ~ 32.27% over PageRank, 17.85% ~ 31.25% over HITS and 16.82 ~ 29.82 over Point System.

Discussion. It is significant that NDCG@3 ~ 5 cases show better performance by InfluenceRank compared with other answer cases, which have major portions in test data sets. This means our assumption of the characteristic of 'Forum' category is meaningful, and has efficiency when analyzing Q&A contents.

• Three baselines (PageRank, HITS and Point System) use activity factor as an important judgment metric for finding credible users. Though Point System uses trust factor, which relies on user voting results based on its popularity, but it still cannot use the weight of each voting. InfluenceRank algorithm strengthens the trust factor by using SN structure within a Q&A site, and it shows better performance.

The intuition behind PageRank is that a user is the authoritative one if several other authoritative users have out-links pointing to her. Following our observation, this is not always true in Q&A sites, because there may be no connection among experts who have different interests. For instance, users in our data set post their questions related to different subjects within 'Sports' category such as 'Baseball', 'Basketball', 'Football', etc. In general, there are few experts in all aspects of sports category, but many ones who have their specialties in particular subjects. With this reason, our InfluenceRank approach, based on the influence of each questioner/answerer, shows better performance than PageRank.

In addition, in view of HITS, askers can be regarded as hubs, and answers who were chosen as the best can be authorities. Note that HITS is originally designed to rank nodes from directed non-weighted graphs. However, interactions between a Q&A site participants can be assumed as weighted graph structure. Using HITS or PageRank on a Q&A site data may ignore a crucial information type: the magnitude of interaction between users. Therefore, HITS and PageRank could not be used to accurately identify authoritative users.

• There is plenty of knowledge by public ownership in 'Forum' category – not many answers are posted after a credible answer was already issued. In case of 6 ~ 10 answers, the questions as well as answers are organized by comparably long sentences that sometimes contain emotional comments or advices, which get lower *rel* counts by editors. Another interesting characteristic is that answer postings continue until the questioner selects an answer among them, which showed similar experimental results in **[1]**.

5 Conclusions and Future Directions

In this paper, we try to model the user authority in Q&A sites. Our *InfluenceRank* model is useful because it combines SNA to judge user authority. By applying SNA method, we compute the *Activity* and *Trust* in Q&A network. And we show a novel approach to find authoritative users using their influential information.

Questions, answers and the weights of answers are used as features of interactions on user network, and are used as Activity and Trust factor to discover influential users. Through InfluenceRank algorithm, we can measure the user's influence and eliminate the negative effect of wrong activities. In our experiment, effectiveness of InfluenceRank by using NDCG@*k* based on editors' judgments shows significant improvements in comparison with baseline methods – PageRank (15.03% average, 27.80% max), HITS (13.89% average, 26.27% max) and Point System which is now used in Naver's Knowledge in (12.87% average, 24.80% max) according to average NDCG@*k* score of three test data sets. Especially our method shows the highest performance in the case of questions with 3 to 5 answers which constitute major part of test data set 25.63% over PageRank, 24.00% over HITS and 23.03% over Point System. We confirm that influencers' answers are more relevant that those of ordinary users' through the experiment. Therefore, it is possible to enhance search effectiveness by discovering and then recommending influencers to users or letting rank answers of influencers higher than other ordinary users by our algorithm.

There are several possible directions in the future. We need to expand our experimental boundary to assess our InfluenceRank algorithm precisely and search for other factors which can affect user's influence. Activity can be measured by the frequency of interactions, but its trustfulness may have different judgment criteria. Although we pointed out method to find user's idea within Q&A site, trust factor, which is referred by just the best-answer choosing scheme by askers in this study, must be analyzed more deeply. For example, if the voting results by other users are matched with the choices of best answer by askers, it will be useful to assess answerer's trust as well as the credibility of askers because of their proper choices. Moreover the mutual trust should be also counted differently with simple summation, because two people are more trustful than strangers if they trust mutually, which is the same with answering mutually in Q&A site.

The Q&A site analyzed by network structure has an interesting characteristic. It is not intentionally built by its users for the purpose of forming ties. In this study, we focused on social relationships between users, but Q&A site also reflects members' shared interests. Whether the question is important or not, the reason of replies is usually not only because of the questioner starting the information-sharing thread but also due to the common interest in that topic. Therefore, we need to consider other factors of social relationship such as profile similarity, hop distance or actual friendship among users. User network in Q&A sites could be then weighted according to these social relationships in the future.

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Social Web Reviews

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Abstract -*In this paper we propose a new way of writing web reviews. The problem lies in the static structure of web reviews. Traditionally speaking, a site administrator or user writes up a review and enters his scores onto a review website. Once the content is ready to be publicly viewed the database just retrieves the information and displays it as it is. This means a review will have the same total score displayed to all users regardless of their preferences. Lack of updates may cause information to become obsolete and more critically misleading to users. Using PHP, HTML and MySql a web review engine was built from scratch taking into consideration the preferences of users for every category. Hence, weights associated with the preferences are introduced to the total scores viewed. A 'social rank' system was implemented to give a bias towards good users to amplify the importance of their content; ranks are based on our social web reviews. The site was able to demonstrate the potential in Social Web Reviews, as users were able to almost instantly share their opinions and receive valid information relative to their interest. Good user content was amplified while bad user content was suppressed. Similarly user preferred criteria were prioritized correctly to give a meaningful rating.*

Keywords: Social Web Reviews, Social Networking, Dynamic web sites.

1. Introduction

Web 2.0 is about democracy and freedom of opinion. Web 2.0 provides a collection of services in an application allowing user contribution and exchange of knowledge whether in form of scientific data or social information. In [1] the authors summarized it as encouraging user contribution, creating collective intelligence, making it easy to reuse and remix content, focusing on customer self-service, and finally creating a sense of belonging to a community as well as a sense of empowerment and ownership.

In our everyday life we are sharing many things between our family members and friends. Hopefully nowadays we are living a new era in Information Technology (IT) in which the social aspect has embraced all the IT's fields. The social aspect has transformed the IT sphere. In fact millions of people have embraced the social aspect which has transformed their day to day life.

The social concept has created a big virtual social network in which millions of users have adopted it. If we look to Facebook [3] as social network no media since the first communication media has reached a million of users in a short period of time. These networks such as MySpace [9], Facebook, Twitter [12], and Flickr [5] just to name a few, attract today members from diverse cultures, ages, and backgrounds. Moreover, Web 2.0 allowed people to share information on places where they did not have to create an account and register such as reviews and discussion forums.

The social aspect has been deeply involved into games, nowadays we see social games and even a transformation of current games into social games; this demonstrates the importance of this concept. Research papers have been written for that purpose. In [6], the authors try to identify hardcore players through the analysis of their behavior inside the game. The approach is based on the network analysis to do this identification. Ben Kirman and al. [7] have developed a social game dedicated to Facebook which collects data from the network analysis in order to add social data to the game player's context inside the game. The social network analysis is important for such applications such as games [13].

In health, adding social aspect is benefit since it will add a value to health libraries [1].

Regarding the review, many attempts have been made, essentially in the film industry and in the online book reviews. For the first one, a predictive rating recommendation for movies are issued and as it is mentioned that this rating is more significant than the one issued by a human-being [8]. Concerning the online book reviews, a model has been created in order to check how the social aspect influences the user review [11].

Social web reviews are our focus in this paper where we introduce a new perspective on how to rate, prioritize, and personalize them for a specific user or customer according to his/her preferences.

This paper is organized as follows: in section 2 a short overview of traditional web reviews. Section 3 presents the details about our new concept of social web reviews: 'RadRevs'. The results of our experiments are discussed in Section 4 and Section 5 concludes this paper.

2. Traditional web reviews

Most of web users when they are going to buy an item whether it is a car or an electronic device tend to look for reviews from customers on products on specialized web sites named web reviews. And based on the rating of the product, the user decides whether to buy the product or not. In this section, we will review the strengths and the weaknesses of two major review websites with two different approaches to web reviews. The first being Cnet [2], a popular more traditional review site and the second is ProductWiki [10] a fast growing review website highly influenced by the Web 2.0 movement.

2.1 Cnet

Cnet is one of the world's largest review sites currently on the internet; it currently hosts a huge database of products and reviews ranging from cars to mobile phones. The basic structure of Cnet is a product being reviewed and rated by an 'editor' and then users may register to post their own reviews as illustrated in figure 1. Currently this is a fairly traditional review system that embraces a few of Web 2.0's concepts by allowing users to share their own reviews and rate products. Though products can only be created by Cnet administrators. A fairly simple score engine has been implemented which is a 5 star rating system, allowing figures such as 5 out of 5 and 2.5 out of 5 to be displayed. Each category has x criteria that accumulate to one score. This has become a standard that many sites have cloned and created modifications for.

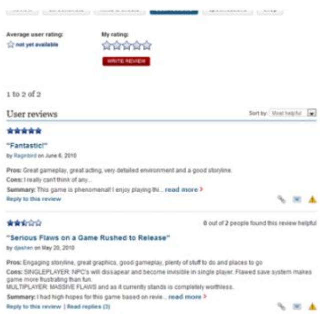

Figure 1. User reviews on Cnet.

This structure on surface seems like a good idea, but once it's implemented in practice many fundamental problems arise. This is mainly due to a set of old web standards that Cnet holds on to. Therefore, there is a need for a new structure for web reviews in order to fully utilize user participation and exploit the capabilities of computers. One of the first issues in such a system is rounding; converting a 100% (10 out of 10 scale, 2 digits with decimals allowed) scale to a 5-star rating system which means that there is a precision loss. For example, the scores 7, 8 and 8, would give an overall score of 7.6 (rounded to one decimal point according to their system). Secondly, once converted to a score value 'out of 5' scale, we get the number 3.8 which means we lose accuracy in two places. Then a third compromise occurs into the score since most 5-star systems only have a full star or a half of a star. At this point, systems are either programmed to round down or round up. In our example the 3.8 was rounded down to a 3.5, this shows how inaccurate a 5-star rating system can be. This makes the choice between two products very difficult as the rating system loses scale (3.8=>3.5 and 3.3=>3.5, it will seem as if both are equal while in reality one is better than the other).

Cnet has implemented a weights-system to introduce some 'fairness' to how products are rated. In the case of TVs, Cnet has chosen to give higher importance to the criteria 'Performance' as they see it as the most important criteria in TVs. This may hold true to a wide range of audience but may not be the case for all users. Although this may not seem like a big issue as users may still view each score individually, it makes it almost impossible to compare a large number of products, as sorting works only with cumulative scores. This pre-selection of criteria weights by administrators is extremely counter intuitive to users, and is an old web standard that needs to be dropped.

In an attempt to please the masses, Cnet has created a page dedicated for the "best" categories. This means that the products selected on the page are 100% static, and updates are not done real time. This raises a few questions for example; why not simply use the search functions available to dynamically find the same results? All the data required to create similar pages dynamically via a search engine is in the individual ratings editors enter.

Cnet's search engine fails to provide relevant results for a number of reasons:

1. You may only sort aggregate scores and not individual scores
2. Filters don't work very well, and sometimes not at all
3. Outdated reviews are mixed with newer reviews
4. Scores are almost never revised for older products

2.2 ProductWiki

ProductWiki is a review site strongly influenced by the Web 2.0 movement, especially the collaborative side of Web 2.0. It is slowly growing into the arena of reviews web sites. The first thing you will notice is that while the site still has "certified reviewers", their reviews are mixed with other user reviews but are prioritized, which is a very good thing. The process of becoming a certified reviewer is a lot easier and simpler than in Cnet since it's a collaborative site. Users are allowed to add a product which is a huge time saver for administrators, as they only have to approve and manage products rather than do all the research and write up the pages every time a new product is out, something physically impossible for a small team (or even a large one in some cases). Adding products is a simple but time consuming process, perfect for a large mass of users, which allows administrators now to have more time to deal with real work such as updates and real admin work(banning, 'keeping it clean', managing…).

Each product has 'Pros' and 'Cons' and a rating. The rating is a cumulative score of all the Pros and Cons with weights. Pros and Cons feature is also collaborative as users rate if each Pro and Con in a review is valid or otherwise, and the most valid points are added on the main page along with the weighted score based on the ratio of Pros to Cons, 'round((NumPros/(NumPros+NumCons))*100,0)' seems to be the most accurate representation as illustrated in figure 2, the problem of rounding here is significantly more accurate than the one found in Cnet.

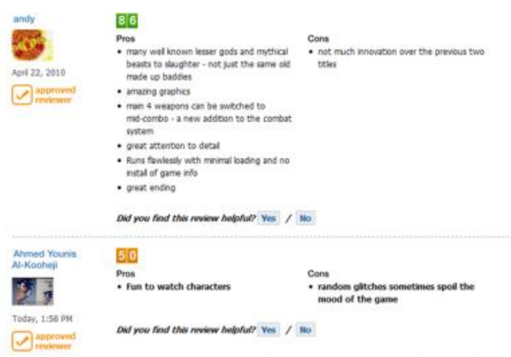

Figure 2. Basic structure of ProductWiki.

Pros and Cons that have been highly rated by users seem to give very accurate random facts regarding the product. The amounts of information and user experiences are very valuable. The same problem of unequal performing products being rated the same exists in ProductWiki; this is more predominant here since the way of calculating the rating is purely ratio based. This means there is no way to know which graphics card would be best performing, but only which ones have more Pros submitted than Cons. Although there has been a system to differentiate good users from bad ones, it has no effect on the content what-so-ever. We take note that the ranking is not based on helpfulness ratio alone, which should not be the case since what makes a good reviewer is the helpfulness of his review regardless of the amount of submitted reviews.

2.3 Findings

To summarize our finding we will firstly list common problems found between the two sites:-

1. Ratings are not affected with time, i.e. no aging factor in the rating scale
2. Current methods of preferring users are incorrect
3. Rating scales have major errors
4. Administrator(s) responsibilities are sometimes overwhelming
5. Reviews may not be objective
6. Submitting opinions should be an easy process, meaning there should be no rigorous registration process
7. Specifications must be consisting of facts and not descriptions
8. Users have no say in what they would like in a product
9. Social network integration is generally weak: both sites use Facebook connect [4] which is a cross platform connector created by Facebook in order to allow developers to use the benefits and features of Facebook on their personal/business sites, but are naturally unable to fully utilize the functionalities of Facebook as many users are registered though the traditional registration process

3. Social Web Reviews

Now that we have identified the problems and strong points of traditional web reviews. We will move on to define Social Web Reviews to maximize the pros of both sites and minimize (or even remove) the weaknesses. The following is a list of original concepts that will define Social Reviews and solve most of the listed problems. This is the part that we will actually implement and test:

1. Social Network integration via Facebook Connect
   a. Integration is ground up
   b. The site users will consist of unregistered or Facebook users, which means the site itself does not have a registration process
   c. Easy to expand in the future with new Facebook features
   d. Free outsourcing of social aspect of the site
2. Users can rate the validity of a review
   a. Valid reviews are prioritized, authors of the reviews climb up the social ladder
   b. Junk reviews (if uncaught by administrators) are ignored
3. Social Rank system
   a. 0-10 scale
   b. Rank affects content directly
   c. Users with rank 10 are 10 times more important(prioritized) than new users(rank 1)
   d. Bad users with a rank 0 are ignored from all calculations
   e. Multiple users may have the same rank
   f. Rank is based on the total quality of content submitted and not a relation between Quality and Quantity
4. Users enter their preferences
   a. Preferences for each criteria under a category
   b. Preferences act as weights for each user
   c. Unique and meaningful results are ensured
5. Administrators are users selected by the site owner
   a. Easy to add administrators
   b. Administrators may
      i. Ban
      ii. Approve reviews
      iii. Approve products
      iv. Submit new categories
   c. Site administrator approves categories submitted by administrators

An important thing to mention is that due to the nature of Social Web Reviews (Size and complexity) the time needed to generate a page may be exponentially longer than traditional reviews, but with some optimization this may be resolved.

3.1 RadRevs

After building the concepts of Social Web Reviews, it is necessary to test these concepts in a real life situation therefore we have created and tested a site named 'RadRevs' with real data to study the behavior of Social Web Reviews.

The following is a list of definitions (and redefinitions) we have created for RadRevs, our Social Website Review:

- Social Review: the concept of integrating social networks as the basis of a review website, furthermore introducing new concepts such as personalized reviews, Social Rank, Review Validity...
- RadRevs: the name of the first fully functional Social Website Review, it was created to prove the effectiveness of the concept.
- RadRating: a figure to reflect the suitability of a product relatively to the user.
- RevRating: a figure to reflect the validity of a review, the higher the RevRating the more accurate and correct the review is.
- Social Rank: a way of implementing a bias for users, good users will have their content amplified (in search and in algorithms) while bad users will have their content in less priority and may even be totally ignored if the user is bad enough.
- Review: a well structured subjective opinion of a product which includes a group of.
- Category: a group of products that may be compared to each other directly.
- Product: it is anything which has a customer base. This can be what is traditionally known as a product for example "computer, car, shoe, etc.", a service, or anything that may be reviewed without losing ground on consistency. If something can not be fully reviewed then it is not a product, for example an animal, but if the animal is categorized as a pet it will become an acceptable product.

Social Web Reviews will make use of ProductWiki's collaborative environment, and Cnet's detailed reviews and finally add new concepts to create a new and unique approach to web reviews.

3.2 RadRevs Structures

We have designed the logical structure of RadRevs as follows: the abstract structure of Social Web Reviews, looks like any other review website, the basic structure is that each category consists of products and each product has (or may not) reviews as illustrated in figure 3.

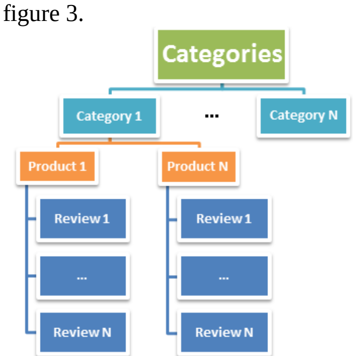

Figure 3. RadRevs logical structure.

A more detailed logical structure is shown in figure 4; here the differences between traditional web review and social web review structures are shown. To break it down simply, a product's rating is based on the cumulative score of the reviews, each review has two major factors, firstly the review score (total) and the validity weight. The review score is based on each criteria score set by the reviewer weighted by the preferences of the reader (user). While the validity weight is based on each review's RevRating, this latter is composed of the rank of the person who rated and the rating itself. The result is a large mathematical calculation to keep in mind the review scores, user preferences and validity of each review to give one number called a RevRating.

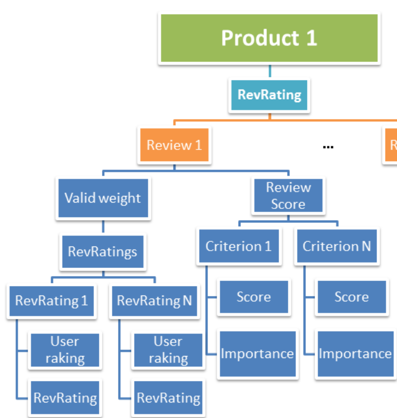

Figure 4. Detailed RadRevs structure.

Figure 5 shows the logical structure of how RadRevs ranks works; in short it is the cumulative quality of work of each user, taking into consideration the importance of every other user's opinion. There is an obvious circular reference. In order to solve this: first we copy the mysql data in php and then process it there without updating the table on each user rank calculation, but rather only updating once all the data is ready.

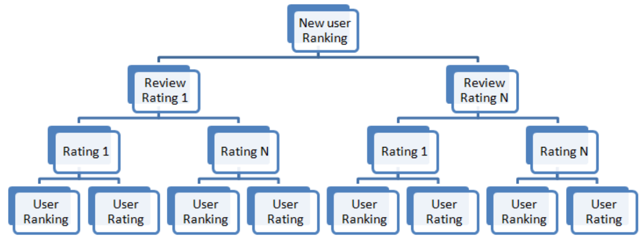

Figure 5. Social Ranking structure.

3.3 RadRevs Walkthrough

RadRevs is implemented using HTML, PHP and MySQL. For the authentication of users, we have used Facebook Connect. Figure 6 illustrates the welcome page of RadRevs which includes the Facebook Connect for the authentication.

Figure 6. RadRevs Homepage.

In the categories page, we will find a list of all the approved categories with two main details other than their name, a brief description of what the category is and the validity date as illustrated in figure 7. This is to make it clear to users that depending on the category there will be an expiration date to all reviews no exceptions. The validity date is based on the nature of the category, for example the 'Keyboard' category should be valid for a three to four years span, since it is a relatively slow market. While a category such as CPUs would most probably have a one or two year(s) expiration date.

Figure 7. Categories page.

Before a user is able to view products or reviews, he will be referred to the category preferences page. Once the form is completed it will never appear again unless the user decides to change his preferences. Users may not find it very intuitive as it is uncommon to see such pages, but with more research into designing such a page it could become a more user friendly experience. Once a user enters his preferences, the server will store all of the data entered in the database. Then, all pages of the category will now automatically calculate everything.

The next page is the best example to show how Social Reviews are different from traditional reviews, the product page in figure 8 integrates the users preferences with the review scores and most importantly the RevRating of each review, to give one figure (RadRading) reflecting the quality of the product. The site is displaying correct behavior by taking the importance rating for each criterion against other criteria and merging them with the final scores. This figure is also the best for the color coding system, as each color was correctly matched with the text to make it easier for users to understand the meaning of each figure. A gauge (going with the theme of 'revs' as in 'car revs') was set into each product and review page with the speedometer pointing towards the score of the product or review. It was added just for the user and to make easy to quickly know how good a product or review is for each user.

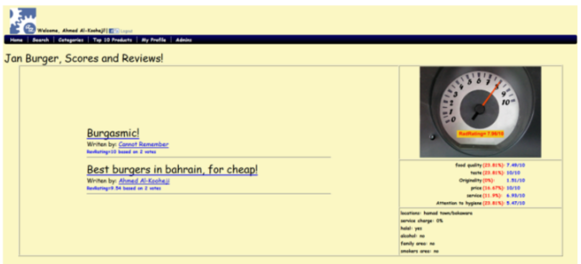

Figure 8. Product page.

The review page in figure 9 contains the reviews created by users; it will display basic information like the review itself and the author, linking the review to the reviewer. Users may also rate each review to signify its validity, as in how correct is the information displayed in front of them, keeping in mind the scores and review elements. The score breakdown is similar to the product page; this has eliminated the problem of merging scores into an unclear value, as it is very clear how we concluded the final score.

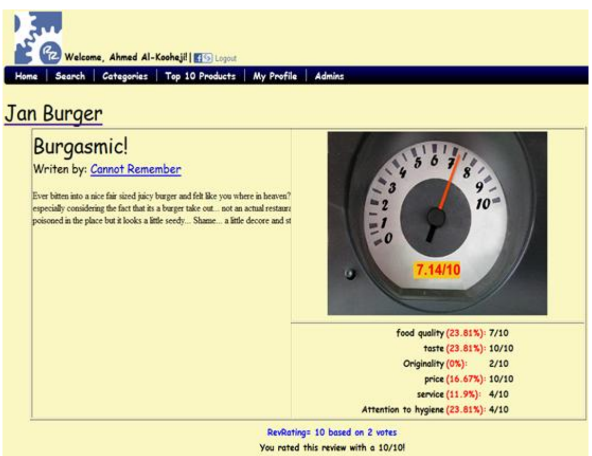

Figure 9. Review page.

Regarding how the user will add a review, we have made it very easy. The user just needs to enter a title for his review, write the review in his own style and most importantly enter the scores according to the guidelines as illustrated in figure 10. Once submitted it will become pending for approval, and only administrator may view the review until it has been approved.

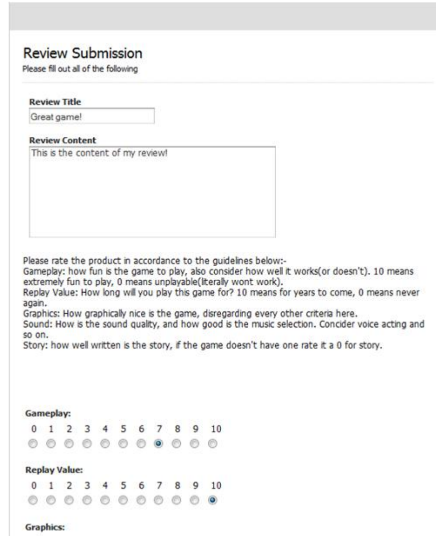

Figure 10. Review submission page.

Moving on to the next section of the site, we will find the administrator's page. Depending on the privileges, each administrator will have a different level of involvement in the site. One of the tasks that the administrators will do is to review the approval page of the user's review (figure 11). Once the administrator is inside this page, the system will search his privileges and fetch all disapproved reviews of the administrator's categories. For example, if an administrator has been assigned to 'Video Games' and 'Restaurants', he will get a sorted list of reviews broken down under his categories where he will have the option to approve or disapprove reviews, or view them in greater detail if required (via clicking the link in the title of each review). Administrators who have been granted to ban users will also have the option to ban a user that submits junk data. This page allows an infinite (theoretical) amount of reviews to be approved/disapproved and have multiple users banned at the same time without any conflicts or limitations, making it simple to visit this page as necessary to approve reviews.

Figure 11. Review approval page.

4. Experiments and results

We have conducted a survey to see the impact of RadRevs on the users. In fact, we have invited fifty users to the lab and asked them to visit "RadRev". All the fifty users filled out a survey of

10 questions to describe their experience using the website and the potential of returning to use it in the future.
The table below shows the scores for each category. Our sample included a group of users from different age categories (students, faculty and staff) with a majority of females. The overall rating of the website was 80% satisfaction with the ease of using it and its friendliness. 74% of participants confirmed return to use the website in the future for more reviews on different topics with a 70% rate of trust in the content of the reviews. 58% were willing to post reviews themselves and share their opinions with others.
These results show an overall success of the website and its purpose.

Table 1. RadRevs' survey results

Sample Data		Frequency	Percent
Gender	Male	21	42
	Female	29	58
Age	18-23 years	21	42
	24-29 years	11	22
	30-35 years	12	24
	≥ 35 years	6	12
Used the website at least once	Yes	50	100
	No	0	0
User Experience	Satisfied	40	80
	Unsatisfied	5	10
	Neutral	5	10
Trustworthiness	Trusted the review and will use it	35	70
	Didn't trust the review	7	14
	Neutral	8	16
Returning User	Yes	37	74
	No	7	14
	Don't know	6	12
User will input review	Yes	29	58
	No	10	20
	Don't know	11	22

5. Conclusion

This paper presents our new way of writing reviews. By integrating the social component, the quality of reviews and accuracy of product scores have indeed improved as it takes a unique view on web reviews. While there is always room for improvement, the site has shown the strong promise in Social Reviews. It has proved successful and correct functionality in addition to good survey results in terms of rating trust and overall satisfaction when using RadRevs. We have noticed some limitations in our model: there still headroom for improving rating scale and there may be better ways to rank users.

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Profile-Based Mood Extraction from a Social Network

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Abstract - *Within the realm of social networks, users have personal profiles that can be used to notify contacts about various events and information. Users can use messages to communicate privately with friends, upload photos, publish notes and links, as well as view and comment on friend updates. In this paper, Facebook, a very prominent social network, is used to detect user mood. Being able to detect mood from social network sites is a very vital piece of contextual information, especially in the domain of mobile pervasive applications, where cameras cannot be typically used to detect user mood without explicit user intervention. We hereby propose a method for detecting the profile owner's mood by extracting and analyzing the aforementioned information. Using such context, many applications can be made context aware, including but not limited to, the personalization of devices in reaction to user mood changes, and the personalization of the results of mobile user queries to search engines. Our mood detection classifier was able to closely match the classification made by human subjects asked to take part in our experimentation. We demonstrate our results as we vary the social networking features used in classification.*

Keywords: Profile, Social Network, Image, Text

1. Introduction and Related Work

Many research studies have been aimed at inferring emotional states from textual information, speech, images, video, and audio. The terms mood and emotion are used in research interchangeably, where mood is typically a longer state of mind than emotion. To be able to detect mood, there has to be a way to classify and measure it. Ekman [13] categorized emotions into six families of affective states, commonly known as the six basic emotions. The six basic emotions are: happiness, sadness, anger, fear, disgust and surprise. Several other ways to classify emotions appear in the literature. The most popular provides a set of emotional ratings in terms of pleasure (valence), arousal and dominance. In this model, emotion is defined by values in a three-dimensional system: the valence dimension ranging from pleasant to unpleasant; the arousal dimension ranging from calm to excited; and dominance or control [14].

The NIMH Center for Emotion and Attention (CSEA) provides affective ratings for a standardized set of English words, the Affective Norms for English Words (ANEW); a database of images, the International Affective Picture System (IAPS); and a database of acoustic stimuli, the International Affective Digital Sounds (IADS) to help researchers assess and compare their results [14].

For textual affect sensing, semantic knowledge is needed to make sense of text. A knowledge base of commonsense is needed to fully understand and analyze textual information. The Open Mind Common Sense (OMCS) project [15] is a web based initiative launched in 2000 by the MIT Media Lab that gathers common sense knowledge from the general public over the web in a simple format. ConceptNet is a commonsense knowledge base and a natural language processing tool that is automatically generated from the sentences in the OMCS Project. It contains over 300,000 nodes connected by 20 relation types. Relations are in form of binary assertions between concepts with a score based on the frequency in which the relation appears in the OMCS database, or how well it can be inferred from the other facts. From the Open Mind sentences, assertions for 17 relation types such as Is-A relations, PropertyOf, and affective relations such as MotivationOf and DesireOf are generated. Three k-line relations (SuperThematicKLine, ThematicKLine, and ConceptuallyRelatedTo) are also generated. K-line relations increase the connectivity of the nodes. They represent contextual information (ThematikKLine and SuperThematicKLine), and are also used to link nodes for which no specific relation exists (ConceptuallyRelatedTo) [15]. ConceptNet provides functions that use the generated knowledge-base for cognitive tasks such as analogy making, text summarization and affect sensing. ConceptNet's function for affect sensing, GuessMood, classifies sentences using the six basic emotions given above.

Liu [18], [19] presents an email client called EmpathyBuddy, that gives the author of the email automatic affective feedback in the form of a facial emoticon, as a reaction to the typed sentences. It is based on the Open Mind Common Sense corpus, and classifies affect into the six basic emotions, giving a scalar value to each of the emotions representing the magnitude of the valence of the entry with respect to a particular emotion. Jung [17] introduces a hybrid approach that uses ConceptNet's common sense knowledge base, ConceptNet's GuessMood function for affect sensing, and the Affective Norms for English Words (ANEW) list to detect mood in blog texts. Blog affect is classified into four moods: happiness, sadness, anger and fear. Leshed [20] also identifies affect from blog posts. Mood categories are based on the mood tags given by the bloggers themselves, and an SVM classifier is used for detecting the mood.

For mood analysis of images, studies show that low-level image features such as brightness, contrast, edges, and color contribute to the mood, as well as some mid-level features, such as texture, layout and shape [1]. Psychology studies show that there is a strong relation between color and emotion. Wexner [2] reports the results of his investigation to associate eight color hues (red, yellow, orange, green, purple, black, blue and brown) with a set of 12 mood tones, represented by a list of adjectives (examples are exciting, secure, distressed, unhappy, and powerful). It shows that certain colors were chosen to go with certain mood tones significantly more often than others. Terwogt and Hoeksma conducted a study [16] that links the primary colors (red, blue, yellow and green) along with black and white to Ekman's six basic emotions, and calculated the frequency that each color is tied to a specific emotion for different age groups. Using these color to mood mappings, we can extract mood from an image based on its color content.

It is worthy to note that the same color can have different symbolisms and meanings in different cultures. Detecting mood based on color is culturally dependent, age dependent, and may also depend on educational background as several studies show [9], [16], [21]. Moreover, color preferences are subjective and do not necessarily invoke the same emotions in all people even within the same culture. One more thing worth noting is that color alone is not an indication of mood. Psychology studies show that the color black for example indicates sadness. However, the image of a black dog does not necessarily induce the feelings of sadness in the perceiver. This leads to the conclusion that shape, texture and other higher level features have an added effect on the mood classification of images, and that it would be best to take such features into account when tagging images with mood.

To analyze the color content of an image, a suitable color model must be chosen. A color model, also called a color space, is a representation system for colors. The most commonly used color systems are the RGB (red, green, blue) system for display devices and the CMY/CMYK (cyan, magenta, yellow, black) system for printing devices. For color content analysis, color models such as the HSI (hue, saturation, intensity) model, and chromaticity-based color models such as the CIE LCH, the CIE Lab and CIE LUV models are very popular, where CIE is short for "*Commission Internationale de l'Eclairage*" and the three axes represent different color properties. The HSI Color Model is a representation that takes into account how humans interpret and describe colors. It describes color by three features, hue, saturation and brightness. Hue is an attribute that describes a pure color, for example pure red, or blue. Saturation is the degree to which the color is diluted by white light (i.e. the degree of purity). Brightness of a color is described by an intensity value. Variations in both the saturation and intensity values of a color offer different shades of the same color with pure white on the one end and pure black on the other. Chromaticity is a term that describes both hue and saturation of a color. The chromaticity-based color models are not displayable color formats. However, they have three features that make them attractive for color image analysis and manipulation. They are device independent unlike the popular RGB model. Colors are perceptually uniform which means that the same distance between two different points makes equal perceived color difference. They are colorimetric, which means that colors perceived as identical are encoded identically.

Wei [3] analyzes the mood of films based on the color content of each shot. Shots are converted to the CIELUV color space, to detect the colors in each shot. The three most dominant colors in each shot are extracted and are mapped to a set of eight primary moods (anger, fear, joy, sorrow, acceptance, rejection, surprise and expectancy). Wei-ning [7] introduces a scheme to automatically annotate images with emotions for an emotion-based image retrieval system. Twelve emotional word pairs are selected as the most significant emotions evoked by images. Three features that affect the emotional word pairs are extracted from the images in the CIE LCH color space. The extracted features are: image lightness, saturation, warm-cool description and contrast description as well as image sharpness. Support Vector Machine of Regression (SVR) are used for training and classification.

Fonseca [8] presents a technique to categorize and search images based on their emotional content. The study attempts to find the optimal relationship between characteristics of an image such as color, compression and resolution, distance and screen resolution and the emotional information that it reflects. It focuses on the user's personal data such as age, sex and education in order to create a more personalized experience, and uses preexisting databases such as the IAPS system, and other web resources for obtaining image metadata for the feature extraction process.

Dunker [9] presents a generic multi-modal mood classification framework for music and photos. The two-dimensional Reisenzein model for mood classification is used. The two dimensions are valence and arousal, which account for most independent affective states. Arousal is obtained from features such as color, saturation, lightness, orientation and character of lines. Valence is expressed by the lightness of colors. Four discreet moods (aggressive, melancholic, euphoric and calm) are proposed for mapping the selected two-dimensional space.

2. Facebook Mood Analysis

Three mood categories have been defined, recent mood, mid-term mood and long-term mood. Recent mood is extracted from posts done by the profile owner in the last three days in reference to a starting current date. Mid-term mood is extracted from posts made in the last 10 days, including the posts for the recent mood. Long-term mood is extracted from all posts made in the last 30 days, which in turn includes the posts used in the Mid-term mood calculation.

Moods within each category are classified into the six basic emotions introduced by Ekman [13]. The six moods are: Happy, Sad, Angry, Fearful, Disgusted and Surprised. Each of the six emotions is tagged with a score representing the magnitude of that particular emotion for the extracted entries. Scores are ordered highest to lowest, and the highest ranked mood is the prevailing mood for that particular time category (recent, mid-term or long-term mood).

The six basic emotions have been chosen as the representation method for the extracted mood for more than one reason. It is a simple representation and more intuitive than the alternative three-dimensional representation of valence, arousal and control. Ready-made tool-kits exist that already provide textual analysis in this format [15], and a color to mood mapping for the six basic emotions is also available in the literature [16].

Textual Mood Analysis

For the textual analysis component, the ConceptNet tool-kit is used. ConceptNet provides a GuessMood procedure that works on a sentence or a set of sentences, and produces scores for the six basic emotions. As explained in Section 1, the common-sense knowledge base of ConceptNet consists of two types of binary assertions (predicates), k-lines and non k-lines. The non k-line predicates contain relation types that describe things (Is-A, PropertyOf), events (PrerequisiteEventOf, SubEventOf), spatial (LoacationOf), functional (UsedFor), causal (EffectOf) as well as affective (DesireOf, MotivationOf) relations between concepts. K-line relations increase the connectivity of the network and represent inferred knowledge and generalization of concepts. Preliminary experiments show that using the non k-line predicates alone produces much better results than with the k-line predicates included. Therefore, this setting has been used for all reported experiments. For details on the conducted experiments and their results, see the following section.

Using the profile feed of the profile owner, status updates, comments on status updates and wall posts, posted notes, comments on posted notes and links, and inbox messages sent by the user are extracted and sent to the textual mood analysis component. Comments on photos and photo captions are also extracted and analyzed.

Image Mood Analysis

Each profile owner has a profile picture. Profile pictures can be changed as often as the user wants. A user can also upload photos and create photo albums. Profile pictures generally indicate the mood of the profile owner at the time the image has been uploaded. Based on psychological findings that color and emotions are strongly associated, it is assumed that the color tones of the selected image indicate the general mood of the user at the time the image was uploaded. Hence, the focus in this component is to extract the dominant colors in an image and map them to a corresponding mood.

The dominant colors of an image are the colors that contribute to at least 80% of the total image pixel count. To determine the dominant colors of an image, the image is first converted to the HSI color space, and the colors classified into the following set of colors: red, yellow, orange, green, purple, blue, black, white and gray. As mentioned in Section 1, the HSI color model describes color in a way that is intuitive to human viewers. A pure color is described by one attribute, the hue attribute. This fact is particularly useful for classifying colors in an image. By evaluating the value of the hue attribute a color can be classified into the following color categories: red, blue, green, yellow, orange, and purple. Color hues can be visualized as a color wheel, where the primary colors (red, green, blue) are 60 degrees apart, and secondary colors (cyan, magenta, yellow) are 120 degrees apart. The value of the hue is measured as an angle from a reference point. Most commonly, red is used as the 0 degree reference point. Values range from 0 to 360. By selecting appropriate threshold values for the hue, colors can be successfully categorized. For detecting black, white, and the different shades of gray, the saturation and brightness values have to be taken into account.

After classifying the colors into their categories and extracting the most dominant colors, the selected color to mood mapping (Table 1) is used to detect the mood. The final image mood is an average value for each of the six basic emotions corresponding to the image's dominant colors.

Color/Mood	Happy	Sad	Angry	Fearful	Disgusted	Surprised
Red	0	0.35	0.12	0.06	0.23	0.24
Yellow	0.33	0.04	0.36	0.04	0.19	0.04
Orange	0.16	0.2	0.24	0.05	0.21	0.14
Green	0.4	0.22	0.07	0.03	0.18	0.1
Purple	0.1	0.27	0.06	0.1	0.15	0.32
Blue	0.2	0.2	0	0.13	0.07	0.4
Black	0	0.22	0.22	0.34	0.22	0
White	0.06	0.2	0	0.06	0.06	0.62
Gray	0	0	0	0	0	0

Table 1: Color to mood mapping

The frequencies for the colors red, blue, yellow, green, black, and white have been obtained from the study on colors in [16] for adults. Values for orange have been calculated as averages from the values of red and yellow, whereas values for purple are average values of red and blue. Gray values have been set to zero as it is considered a neutral color in some studies.

3. Experiments and Results

For validating the proposed method, five profiles have been specifically created and maintained for a month by five different people forming a small community to simulate activity on Facebook. Surveys for the five profiles have been distributed to collect mood as observed by human viewers of the profile. Ten responses per profile have been collected.

Table 2 illustrates the results of the survey. For each profile, three mood categories are listed (R: recent mood, M: mid-term mood, L: long term mood). Percentages for each of the six basic emotions (H: happy, S: sad, A: angry, F: fearful, D: disgusted, Su: surprised) are listed next to each one. Moods are ordered by percentage values, from highest to lowest. Moods that are not clearly distinguishable, i.e. they have close values in two emotions, are marked by asterisks.

Profile 1			**Profile 2**			**Profile 3**			**Profile 4**			**Profile 5**		
R	M	L	R	M	L	R	M	L	R	M	L	R	M	L
H 83	H 47.5	H 62	H 44.5	H 38	H 47	H 69	H 43.5	S* 36	H 39	S 40	S* 28	H 81	H 67.7	H 80.1
Su 10	F 16	S 19	S 27	S 27.5	S 22	S 15	S 24.5	H* 32.5	D 19	A 19	H* 24.5	Su 11	Su 15.6	Su 11.6
S 5	S 14	A 6.5	A 9	A 19	A 13	Su 6.5	A 13	A 14.5	S 16	D 15	D 17	F 4.2	S 6.5	F 5.2
F 2	Su 10	Su 4.5	Su 9	F 7.5	F 10	A 5	D 9.5	F 7	A 16	H 14.5	A 16	S 2.5	F 6.2	D 1.1
A 0	A 8	F 4	F 8.5	Su 4	Su 4	D 3.5	Su 8.5	D 7	F 5	Su 6.5	F 8.5	A 0.5	A 3	S 1
D 0	D 4.5	D 4	D 2	D 4	D 4	F 1	F 1	Su 3	Su 5	F 5	Su 6	D 0	D 1	A 1

Table 2: Survey Results

The first experiment has been set up in two parts. The first part works on the textual entries only; the second part on images only. The textual part generates the mood for each textual entry individually, and then calculates the averages for all entries in each mood category (recent mood, mid-term mood, and long-term mood). Included textual entries are status updates, comments on wall posts and pictures, image captions, notes and all inbox messages sent by the profile owner. The second part generates the mood for all uploaded images individually, and then calculates the averages for all entries in each mood category. Included entries are profile pictures as well as uploaded albums. The results of the experiment are reported in Table 3 and Table 4. Only moods that have been given a score higher that zero are reported, moods that have been given a score of zero are left out of the table. Categories in which no text or images are present are left empty in the table.

Profile 1			**Profile 2**			**Profile 3**			**Profile 4**			**Profile 5**		
R	M	L	R	M	L	R	M	L	R	M	L	R	M	L
H	***H***	***H***	Su	***H***	***H***	***H***	***H***	H*	***H***	H	H*	***H***	***H***	***H***
S	S	S	***H***	Su	F	S	S	S*	A	***S***	A	A	S	S
F	F	F	A	F	S	F	F	F	S	A	F	S	A	A
A	A	A	S	S	A	Su	Su	Su	Su	F	S	Su	F	F
D	D	D	F	A	Su	A	A	A	D	Su	Su	F	D	Su
Su	Su	Su		D	D	D	D	D	F	D	D		Su	D

Table 3: Textual Entries Only, Simple Average

As observed from Table 3, the highest ranked moods in all categories coincide with the survey in most cases. In three of the cases, Profile 2 recent mood, Profile 3 long term mood and Profile 4 mid-range mood, the second highest ranked mood matches the highest ranked mood in the survey results. For the long term mood in Profile 4, the highest ranked mood is Happy. In this case, the survey gives close values for the moods Happy and Sad, specifically it gives 28% for the mood Sad and 24.4% for the mood Happy. The same observation can be made for Profile 3's long term mood. Therefore, it is safe to conclude that the results for the most evident moods in the survey as observed by the profile raters is detected quite accurately by the application. However, moods assigned percentages less than 20% by the profile raters, are harder to detect. Thus, the order of moods starting from the second highest rated moods does not coincide with the survey results.

Profile 1			**Profile 2**			**Profile 3**			**Profile 4**			**Profile 5**		
R	M	L (1)	R	M	L (6/1)	R (1)	M (3)	L (4)	R	M	L (1)	R	M	L (1)
		S			H/H	S	Su	S			H			Su
		D			S/A	Su	S	A			A			S
		Su			D/D	F	D	D			D			A
		A			A/S	D	A	Su			S			D
		H			F/F	A	H	H			Su			H
		F			Su/Su	H	F	F			F			F

Table 4: Images Only, Simple Average

In Table 4, the numbers in parentheses indicate the number of images that were uploaded in each category. The displayed moods are the average moods for all of the images in each corresponding category. Most images are profile pictures. Only Profile 2 contains five pictures among the six uploaded pictures, that have been uploaded as part of an album. The second set of moods in the Recent Mood column of Profile 2 corresponds to the Profile Picture mood only. It is worthy to note that the number of images in each category is very small compared to the number of textual entries. As we will see in the next experiment, combining the images with the text, therefore, does not have much effect on the final moods.

The results of the second experiment are shown in Table 5. It uses the same averaging technique, but includes textual entries as well as profile pictures and all uploaded pictures.

Profile 1			**Profile 2**			**Profile 3**			**Profile 4**			**Profile 5**		
R	M	L	R	M	L	R	M	L	R	M	L	R	M	L
H	***H***	***H***	Su	***H***	***H***	***H***	***H***	H*	***H***	H	H*	***H***	***H***	***H***
S	S	S	***H***	Su	F	S	S	S*	A	***S***	A	A	S	S
F	F	F	A	F	S	F	Su	F	S	A	F	S	A	A
A	A	A	S	S	A	Su	F	Su	Su	F	S	Su	F	F
D	D	D	F	A	Su	A	A	A	D	Su	Su	F	D	Su
Su	Su	Su		D	D	D	D	D	F	D	D		Su	D

Table 5: Textual Entries and Images, Simple Average

Profile 2 contains one profile picture and 5 photos, which have been uploaded early on in the experiment and coincides with the long-term mood category. Profile 3 has uploaded 4 different profile pictures, one of which is classified in the recent mood category, two in the mid-term mood category and one in the long-term mood category. All other profiles have uploaded only one profile picture that is classified in the long-term mood category. The categories are highlighted in Table 5. As most categories contain only one image this does not affect the mood scores much. The mid-term mood in Profile 3 shows some change in the order of moods since it contains two pictures. Profile 2's long-term mood contains around 6 pictures. The images contain mostly the colors green, black, white and yellow. By looking at Table 1, we can observe that these colors in average increase the score by the same value in all the six basic emotions. Therefore, the order in that category does not change.

The last prevailing mood of the profile owner is mostly evident in the latest post made to the profile. By giving more weight to later posts, the mood may be detected more accurately. Table 6 lists the results for mood scores calculated by giving higher weights to more recent posts. As noted, the highest ranked mood does not get affected in this experiment. For Profile 4, the second highest rated mood for the mid-term category, which in the previous experiment was "Sad" has been replaced by "Angry" and moved to the third place.

Profile 1			**Profile 2**			**Profile 3**			**Profile 4**			**Profile 5**		
R	M	L	R	M	L	R	M	L	R	M	L	R	M	L
H	***H***	***H***	Su	***H***	***H***	***H***	***H***	H*	***H***	H	H*	***H***	***H***	***H***
S	S	S	***H***	Su	F	F	S	S*	A	A	A	A	A	S
F	F	F	S	S	S	S	F	F	D	***S***	S	S	S	A
A	A	A	A	F	Su	A	Su	Su	S	F	F	Su	F	F
D	D	D	F	A	A	Su	A	A	F	D	Su	F	Su	Su
Su	Su	Su		D	D	D	D	D	Su	Su	D		D	D

Table 6: Textual Entries and Images, Weights according to Time of Post

When taking a closer look at the individual textual entries, one notices that it is mostly status updates and profile pictures that directly reflect the mood of the profile owner. Comments on posts or pictures as well as inbox messages are usually responses and reactions to something that someone said. Therefore, it does not necessarily indicate the mood. By adjusting the weights according to the type of post, giving more weight to posts directly made by the user such as status updates and notes, less weight to images, and the lowest weights to comments and messages, we get the results depicted in Table 7.

Profile 1			**Profile 2**			**Profile 3**			**Profile 4**			**Profile 5**		
R	M	L	R	M	L	R	M	L	R	M	L	R	M	L
S*	***H***	***H***	Su	***H***	***H***	Su*	Su*	H*	***H***	H	H	***H***	***H***	***H***
H*	S	S	***H***	Su	F	H*	H*	S*	A	A	A	A	S	A
F	F	F	A	F	S	F	A	Su	S	***S***	F	S	A	S
A	A	D	S	S	A	A	S	F	Su	F	S	Su	F	F
D	D	A	F	A	Su	S	F	A	D	Su	Su	F	D	Su
Su	Su	Su		D	D	D	D	D	F	D	D		Su	D

Table 7: Textual Entries and Images, Weights According to Type of Post

As observed, the order of the moods changes in some of the categories, and does not coincide with the survey ratings any more. When taking a closer look at the profile contents, it can be noted that the profiles that are not considerably affected do not include many messages or comments (Profile 1 mid-term and long-term moods, Profile 2, Profile 4 and Profile 5), whereas the mood categories that do change include messages and comments (Profile 1 recent mood and Profile 3).This is expected since the survey participants have been asked to take into account all entries, which includes messages and comments.

As earlier mentioned, five different sets of parameters were used in our experimentation to automatically classify user mood. The first set of parameters involved the usage of textual entries only with simple average, the second set involved the usage of images only with simple average, the third set involved using textual entries and images with simple average, the fourth set involved using both textual entries and images with weights according to the time of post, and finally, the fifth set involved using both textual entries and images with weights according to the type of post. For each one of the aforementioned parameters used in automatic classification, we developed a corresponding experiment to observe how well they perform in classification.

The following graph attempts to visualize the accuracy of automatic classification made in each of the five experiments compared to the classification made by human subjects. The graph illustrates what percentage of the top classification made by the experiments across all profiles and mood types matched one of the first N classifications made by the human subjects. For example, the data where N = 1 indicates the percentage of automated classification that matched the top classification made by humans. When N = 2, that indicates the percentage of automated classification which matched either the first or second classification made by humans, and so on.

As we can observe from the chart below, most experiments give an accuracy of more than 80% when N = 2, which means the automatic classification matches either the first or second human classification by an accuracy of around 85%. Experiment 2 is concerned with images only, therefore with N = 1, it had an accuracy of 28%, and then jumped to 85% with N = 2. Experiment 5 gave different weights to different types of posts, which proved less accurate overall. This may be due to the fact that human subjects were asked to evaluate the profile mood as a whole considering all posts equally.

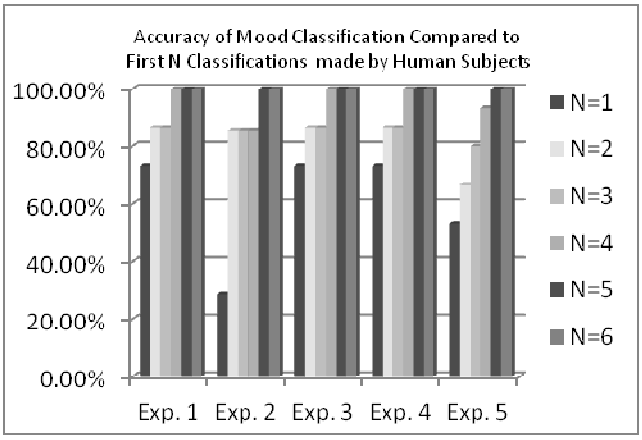

Figure 1: Accuracy of Mood Classification Compared to first n Classifications made by Human Subjects

4. Conclusion and Future Work

In this paper, we presented an approach for detecting user mood from a social networking site. We presented our approach based on the analysis of text and image postings on user profiles. We experimented our approach using text alone, images alone, and subsequently combined both text and images to detect mood. We also conducted experimentation of our approach taking into consideration how recent the posts made by users were, as well as the types of posts themselves. Our experimental results indicate accurate mood detection in the long, medium, and short term, in comparison to the mood concluded by human subjects taking part in this study. In almost all of the cases, the automated mood detection performed by our approach fell into the first two mood classifications made by human subjects taking part in this study. However, it is worthy to note that detecting mood and emotion from written text is very subjective and as reported by survey participants is not that easy, especially when the observers do not personally know the people they are rating. Furthermore, different text in various languages, and colors in various cultures have very diverse meanings, thus leading to the need make mood detection more culturally sensitive also, and probably also gender dependent.

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Subject-Driven Community Mining in Online Social Networks

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Abstract— *Discovering communities in online social networks (OSNs) has attracted much attention in recent years. Most of existing research in this area focuses on the "friend" relationship among members in OSNs, but ignores the fact that a community is subject-centric: people interested in a subject form a community by interacting with each other. In this paper we introduce the concept of* subject-driven communities *and propose to discover such communities by 1) modeling a community using a posting/commenting* interaction graph *instead of a friendship network, 2) applying text classification on the content of blogs when constructing the interaction graph to keep it subject-relevant, and 3) applying link analysis on the interaction graph to locate the core members of a community. Our initial experimental results on LiveJournal demonstrate the effectiveness of our method.*

Keywords: Online Social Network, Community Mining, Text Categorization, Link Analysis, HITS

1. Introduction

The Internet has dramatically changed the way in which people communicate with each other. In the past few years *Online Social Networks* (OSNs) have gained significant popularity. They have become pervasive platforms of communication, and they have attracted a large amount of attention from computer scientists, the intelligence community, and sociologists, among others. One of the most important issues, and most active research areas in this field, is to discover cyber communities in OSNs. We define here a *cyber community* in an OSN as a group of people who share common interests such as hobbies, occupations, etc, and interact with each other through the OSN.

There are several reasons why researchers may be interested in finding communities in OSNs. For example, these communities provide valuable information resources: They represent the sociology of the Web and their study provides insights into the structure, organization, and interests of the different sectors of society. Identifying these communities can, among other things, help intelligence departments or advertising companies to find particular targets.

We have observed that one of the most common activities in an OSN is *posting* and *commenting*. A member of an OSN usually has some *space* on which the user can post content (articles, pictures, or video clips) that s/he considers interesting; if the space is open to the public, other members can read the content and leave comments. These posting and commenting activities compose the essential interactions among members of an OSN.

Since people join OSNs for different reasons, and people have different interests, they may interact with others on various topics. The "friend" relationships provided by most OSNs do not capture the variety of interactions among OSN members. We propose in this paper a *subject-driven community* discovery approach, which focuses merely on the posting and commenting activities on a given subject, and the *interaction graph* to model these activities. In an interaction graph a node represents a member who posts blogs on a subject, and an edge represents a commenting activity by another member to the blog. Note that the interaction graph captures two essential properties of a cyber community: shared interests and interactions among members. We use Kleinberg's HITS algorithm [4] to locate members on an OSN who are well connected with respect to a given subject, and this collection of members forms the core of a cyber community.

Our contribution lies in being the first work, to our knowledge, to focus on subject-driven communities on OSNs, and to mine such communities by modeling posting/commenting activities with an interaction graph. In particular, the concept of subject-driven community and our approach to model a community are novel.

2. Related Work

Because of the popularity of OSNs and the diversity of cyber communities, it is not an easy task to mine them. Many OSNs provide services for members to create or join *groups* and to list *interests* in their profiles. Sometimes it is easy to identify communities formed in this way by simply listing the members of a group and checking all members' profiles. However, these explicit communities do not take members' activities into account and relay on the assumption that the profiles of members are accurate and reliable; this assumption might not be true because a member who joins a group may not necessarily be active in the group, or a member may list too many or too few interests in their profile. Furthermore, not all cyber communities have explicit groups; thus, mining implicit communities based on interactions among members is an important task when studying OSNs.

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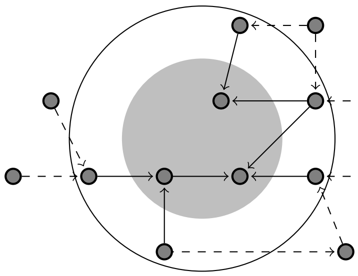


Fig. 1: Initialization and expansion of the interaction graph.

Another important feature that most OSNs provide is a *friend* list for each member. Some recent research on mining cyber communities analyze the friend link structure among OSN members, and model an OSN as a *friendship graph* [2], [6], in which a node represents a member and an edge represents the fact that a member lists another member as a friend. Although the friendship graph provides information about which members know each other, it does not reflect the shared interests and the interactions among these members [9], [10]. The friendship model also implies a simple relation between members, which might be much more complex in reality [1]: Two members may become "friends" in an OSN if, for example, they are colleagues, they have common friends, they share an interest, and so on.

Some researchers have been aware of the drawbacks of friendship graphs and have proposed methods to mine cyber communities by taking members' interactions into account. In [10], a graph based on members' activities on Facebook[1] was introduced; however, the fact that the subject of the interaction was not captured when building the graph makes it only a general representation of an OSN's social activities, and not appropriate for identifying communities related to different subjects. Methods involving both content and linkage analysis were employed in [2] to locate hate groups in blogospheres; however, modeling *group co-memberships* and *subscriptions* as the links in the graph, and using *group descriptions* as the only input for content analysis, ignores OSN members' real activities: Posting and commenting. In [7], [13] different methods were proposed to discover groups inside explicit communities. Such approaches cannot be applied to general OSNs to mine implicit communities because the structure of a general OSN is much more complex than that of an explicit community. For example, when an expertise forum was studied in [13], a simple statistical method outperformed all other "advanced" link analysis methods; this situation does not hold when dealing with a general OSN.

3. Community Mining

A member of an OSN may interact with other member on a variety of subjects. For each subject, the same member may play different roles. For example, a member may be very active in some topics, may occasionally join discussions in others, but may not take part in some other topics at all. The friendship graph does not reflect the degrees of involvement of a member on different subjects. We introduce the concept of *subject-driven community*, which is a community on a given topic, to accurately model people's behavior on OSNs. To our knowledge, no research has been published which focuses on subject-driven communities, although it is usually suggested that communities are based on shared interests, i.e., subject-related.

In this section, we introduce interaction graphs to model communities, and apply the HITS algorithm to locate the core, or most active members of a community.

3.1 Interaction Graph

As the friendship graph does not capture the interactions among OSN members, we formally define the *interaction graph* on a given subject as follows.

Interaction Graph: Given a subject t and an OSN, an interaction graph with respect to t is defined as a directed graph $G = (V, E)$, where V is a collection of nodes in which each node u denotes a member of the OSN who posts blogs, or comments on blogs relevant to t; E is a collection of edges, where edge (u, v) denotes that the member represented by u comments on the blogs of the member v. The weight of (u, v) represents the number of comments that u made on v's blogs, as described below.

It is worth to mention that 1) the term *blog* refers to the content a member posts on their space, which could be, for example, an article, a picture, or a video clip, as mentioned earlier; 2) the blog must be relevant to the subject t; and 3) the interaction graph is a weighted directed graph, which captures the posting and commenting activities of members with respect to the subject t.

We construct an interaction graph by using the following procedure.

Step 1: Given a subject t, we first look for blogs related to this subject, for example, by employing a search engine or the search service provided by the OSN, using the subject as the query. These blogs form the starting set for the interaction graph: A node is created for each member who posts or comments on these blogs and we add an edge (u, v) if u comments on v's blog. We call this step *initialization*.

Step 2: For each node u in the graph, we examine all blogs that the member represented by u has posted. If a blog b is relevant to the subject t, we add to the graph nodes for all members who comment on b and we also add edges representing the comments. Then, we update the weights of the edges and repeat Step 2. We call this step *expansion*.

[1] http://www.facebook.com

Step 3: When no more nodes are added in Step 2, we output the constructed graph as the interaction graph.

Figure 1 shows Steps 1 and 2 of the above procedure. Nodes represent members and arrows represent comments. In Step 1 a search engine returns a collection of blogs related to a given subject. The members who have written these blogs are represented by the nodes inside the shadowed area in the figure; these nodes along with the nodes for members who comment on these blogs compose the starting set, shown inside the outer circle in the figure. By following the comments that other members make on any blogs posted by the starting set (dashed arrows in the figure), more nodes are added to the graph.

In Step 2, the weight of an edge is assigned as follows: Assume that the member represented by node u comments on the blogs posted by v; for each blog b posted by v we increment the weight of edge (u, v) by 1 if u posts exactly one comment on b, or by 2 if u posts more than one comments on this blog. We adopted this weighting scheme to avoid assigning too much weight to one edge when members u and v have a long conversation by commenting on each other's comments on one blog.

3.2 Link Analysis

According to the HITS algorithm [4], there are two types of valuable pages on the Web: *Authority* pages, which are those pages that contain highly relevant information about a topic, and *hub* pages, which are those pages that contain links to the *authority* pages on a topic. HITS uses an iterative procedure to compute authority scores and hub scores for Web pages. An iteration of this procedure updates these two scores by using this intuition: A good authority is known by good hubs, and a good hub knows good authorities. In our context a good authority is a member of an OSN whose blogs are considered influential or highly relevant on certain subject by other members (i.e., these blogs receive a large number of comments), and a good hub is a member who comments a lot on blogs that are highly relevant on a subject.

The original HITS algorithm [4] was designed to work on an unweighted graph, but the interaction graph we have constructed in Section 3.1 is a weighted one. We propose below a modified version of HITS that works on weighted graphs.

1) Consider a weighted directed graph $G = (V, E)$ represented by its adjacent matrix W, where entry $W[i, j]$ is the weight of the edge (i, j) (which denotes "member i comments on member j's blogs"), or zero if there is no such an edge.
2) Let $\vec{a}$ denote the authority vector (a vector giving the authority score of each node of G), and $\vec{h}$ denote the hub vector (a vector giving the hub score of each node of G). We initialized $\vec{h}$ so all its components are 1. We update $\vec{a}$ and $\vec{h}$ iteratively by using the following equations until convergence is achieved:

$$\vec{a} = W\vec{h} \tag{1}$$

$$\vec{h} = W^T\vec{a} \tag{2}$$

where W^T is the transpose of matrix W.

It has been proven [3] that, given that the entries of W are non-negative real numbers, with appropriate renormalization, this iterative process converges after a polynomial number iterations of steps (1) and (2). We renormalize $\vec{a}$ and $\vec{h}$ after each iteration so that the sum of components of each vector is equal to 1.

After converging, $\vec{a}[i]$ and $\vec{h}[i]$, the ith entries in $\vec{a}$ and $\vec{h}$, are the authority and hub score of the ith node in the interaction graph, respectively. We consider those members who have high authority and/or hub scores as the core of the community with respect to the given subject.

4. Experimental Results

4.1 Data Set

LiveJournal[2] is one of the largest OSNs in the world and one of the most popular data sets when studying OSNs [6], [7], [12]. We chose LiveJournal as our experimental data set not only because of its availability, but also because of its nature as a blog service that contains rich features for subject distillation (compared to other OSNs like Facebook and Flickr). LiveJournal allows members to create, join and leave explicit communities freely, which provides us with a base for comparison with the implicit communities that we are interested in discovering.

We implemented four crawlers to perform data collection: one to crawl the friendship network, one to fetch each member's interest list, one to crawl the explicit communities, and one to fetch and parse blogs and comments. Obviously our crawlers have to obey LiveJournal's bot policy, so that data not available to the public or protected from our crawlers are not collected. Also our crawlers ignore the blogs and comments not written in English.

Initially, we used 200 random LiveJournal members as the seeds. The friendship crawler performs a breadth first search on LiveJournal and it fetched the information of more than 3 million members. The posts crawler fetched all blogs and comments that these members posted in the year 2010; in total we have collected more than 11 million blogs and 49 million comments. All collected data are stored locally in a MySQL database.

When constructing the interaction graph, we use the Bow classifier [5] to determine whether a blog is relevant to a given subject. The training set for Bow was collected from the Open Directory Project [8] and manually checked for relevance on the selected subject.

[2] http://www.livejournal.com/

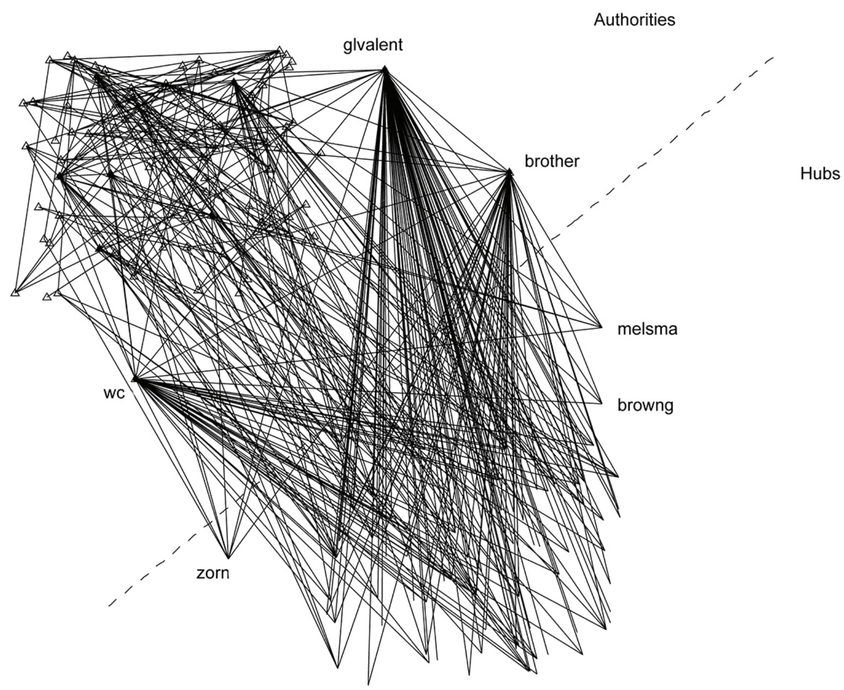


Fig. 2: The core of the interaction graph representing the "astronomy" community in LiveJournal; the core is composed by the top 100 authorities and hubs. Authorities, marked as triangles, and hubs, are separated by the dashed line. The names of the top 3 authorities and hubs are also shown.

Table 1: Members with top ten authority and hub scores in the astronomy community.

User ID	Authority Score	User ID	Hub Score
glvalent	0.25279300	melsma	0.00951980
brother	0.19863059	browng	0.00852836
wc	0.16937396	zorn	0.00847445
dr_neb	0.08875633	janetmi	0.00841167
davidke	0.04641591	ysabetwordsm	0.00799496
invader	0.04329613	pamelad	0.00762778
beamjoc	0.03607167	dd	0.00684412
bobdel	0.02450088	hh	0.00655707
dewl	0.02252651	alank	0.00650713
11011	0.01648967	luscious_pur	0.00644369

4.2 Results and Discussion

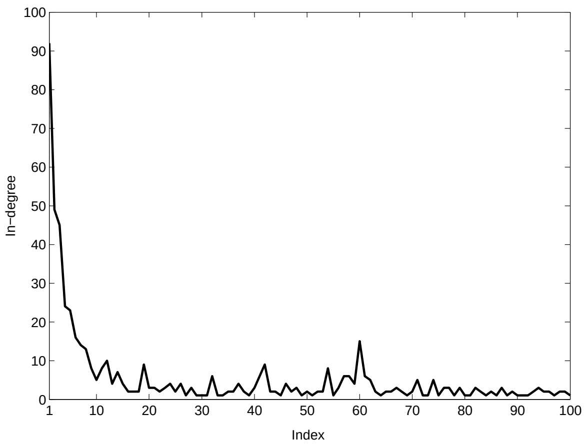
To test our approach for community discovery we first needed to select a suitable subject. We read blogs and comments from a number of LiveJournal members and discovered that a fair number of them were interested in astronomy. Hence, we decided to use "astronomy" as the first subject for our study.

We used Google[3] Web Search to obtain the initial set of members in the astronomy community by using "astronomy" as the query. Google returned a collection C of documents containing blogs from LiveJournal and we used Bow to select from C a subset C' of documents highly relevant to astronomy. The starting set of our interaction graph includes those members who posted or commented on the documents in C'; we then expanded the interaction graph as described in Section 3.1. Our program selected $4,978$ blogs and $13,493$ comments from the database, and constructed a $4,739$-node interaction graph. Finally we computed the authority and hub scores for each node.

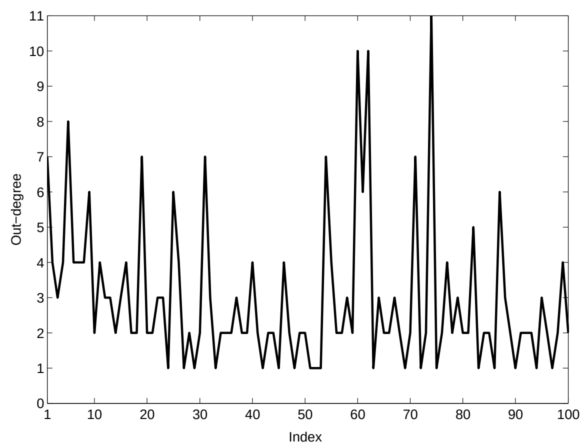
[3] http://www.google.com

Figure 2 illustrates the core of the interaction graph, which consists of the members with the top 100 authority and hub scores (henceforth these members are called authorities and hubs, respectively). One point to note is that the top authorities have a large number of links from the top hubs, and the top hubs point to many top authorities, which closely follows the definition of authorities and hubs. Another interesting observation is that there are few links between hubs, and the interconnection between authorities is not as dense as the interconnection between hubs and authorities, which implies that some members mainly limit themselves to posting blogs while some other members mainly read and comment on others' blogs, and thus the authority/hub model fits this pattern of behavior appropriately.

Table 1 lists the members with the top 10 authority and



(a) In-degrees of top 100 authorities.



(b) Out-degrees of top 100 hubs.

Fig. 3: In and out-degrees of the top 100 authorities and hubs. Authorities and hubs are indexed in decreasing order of authority and hub score.

hub scores [4]. We notice that the authority scores decrease rapidly, while the hub scores do not vary as much. This tendency can be observed in Figure 2 as well: A few authorities (the top 3) have numerous links, while the others reside in a relatively sparse portion of the graph; on the other hand, all hubs have a similar number of links. The in-degrees of the top 100 authorities (the in-degree of an authority a is the number of members who comment on a's blogs) and out-degrees of the top 100 hubs (the out-degree of a hub is the number of authorities on whose blogs the hub comments), are shown in Figure 3, and they also reflect the same fact: That the number of influential authorities is small, but they are known by many hubs.

The average degree of the interaction graph for the "astronomy community" is 2.7, while previous research [11], [12] and our collected data show that the average degree of a node in LiveJournal's friendship graph is around 15. The considerable difference between these two degrees implies that members do not interact with all their fiends on a particular subject; this difference also indicates that the friendship graph may not be suitable to capture the interactions among members on a given subject.

We carefully examined the explicit astronomy community in LiveJournal[5], consisting of $2,425$ members, and found out that only 44 of these members appear in the implicit community that we discovered; furthermore, only 7 of them are among the top 100 authorities and hubs in the interaction graph. In addition, we analyzed the activity of this explicit astronomy community and discovered that only 186 members were active in the community in 2010 and they posted only 42 blogs and 318 comments, while the implicit astronomy community discovered by our approach consists of $4,739$ members who posted $4,978$ blogs and $13,493$ comments relevant to astronomy in 2010.

We also checked the profiles of all $4,739$ members in the interaction graph. Surprisingly only 369 of them list "astronomy" as one of their interests, although the total number of members in LiveJournal who claim "astronomy" as one of their interests is $8,624$.

Besides "astronomy", we also used our approach to identify the LiveJournal community related to "martial arts". There are $6,090$ members in LiveJournal who list martial arts as one of their interests, and, furthermore, there are 4 explicit LiveJournal communities related to martial arts: "_martial_arts", "martial artists", "fighting martial arts" and "teaching martial arts". We found out that only 49 members in these communities were active in discussions related to "martial arts" and they posted only 26 blogs and 105 comments in 2010. In contrast, our approach discovered an implicit "martial arts" community with 758 members who posted 297 blogs and $1,020$ comments in 2010.

In our experimental results we noticed more than one order of magnitude difference between the activity of the implicit communities and LiveJournal's explicit communities, and about one order of magnitude difference between the number of members who list a subject as an interest and the number of members who are actually active in discussions related to that subject. This data strongly suggests that LiveJournal's explicit communities and user's profiles are not really useful to discover subject-related communities.

5. Conclusion

In this paper we studied two properties of cyber communities on online social networks: Shared interests and user interactions. We then introduced a novel concept in community mining: Subject-driven communities, based on

[4]User IDs appeared in Figure 2 and Table 1 have been changed for privacy reasons.

[5]http://community.livejournal.com/astronomy/

the premise that a community should be defined by the interactions among members, instead of friendship relations. We proposed the interaction graph to model a community, and we proposed an approach to keep a community subject-relevant by using text classification technology. We also designed a modification to Kleinberg's HITS algorithm to locate the core of a cyber community.

Our initial experiments on LiveJournal illustrate the inadequacy of explicit communities and members' profiles to mine cyber communities. Our results show more than one order of magnitude difference between the activity of the implicit communities discovered by our approach and LiveJournal's explicit communities. It is worth to mention that this difference supports what we claimed in Section 2: A member who joins a group may not necessarily be active in the group, and not all cyber communities have explicit groups. Our analysis of members' profiles shows a small overlap between members who list interests on a subject and members who are active in discussions related to that subject; this supports our second claim in Section 2 that a member who lists an interest may not interact with other members on that topic, and members who are active on a topic may not list the topic as one of their interests. We believe that our approach of using interaction graphs and the authority/hub model is effective in discovering communities.

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How Social Networks Can Help us to Improve Tourism Business

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Abstract - *This paper focuses on different aspects of social networks. It tries to realize that how tourism can be developed by the use of social networks or social media in the other word. For the scope of this research, a region in the southern of Sweden is selected which provide a very beautiful nature for camping and they have lots of fantastic activities, historical places, crystal exhibitions of hand-made crystals and a cute hotel which is made of crystals for tourists. This paper suggests some definitions of tourism are explained to get a better understanding of subject in order to know how social media can affect on tourism. Also, tourist attractions of this area are analyzed according to usability of social media to find a suitable Information system solution to attract more tourists in this area.*

Keywords: Social networks, social media, tourism, Information.

1 Introduction

In this paper, the authors try to give a very short description of tourist and why tourism business is important. In this regard, they focus on Social Networks to evaluate the effects of social networks on tourism business. The methods of research and data gathering phases are described briefly for readers to get a better sense of an academic research steps. The main aim of this research is the research question that: "How social networks can help to improve tourism business?". In analysis section, different possibilities of social networks are assessed and results are conducted to conclusion part.

1.1 Background of Social Media

There is a nice definition of what the social media is by Kaplan and Haenlein (2009): "*The era of Social Media as we understand it today probably started about 20 years earlier, when Bruce and Susan Abelson founded ''Open Diary,'' an early social networking site that brought together online diary writers into one community. The term ''weblog'' was first used at the same time, and truncated as ''blog'' a year later when one blogger jokingly transformed the noun ''weblog'' into the sentence ''we blog.'' The growing availability of high-speed Internet access further added to the popularity of the concept, leading to the creation of social networking sites such as MySpace (in 2003) and Facebook (in 2004). This, in turn, coined the term ''Social Media,'' and contributed to the prominence it has today.*"

1.2 Problem

The main research problem which is focused is lack of tourists in the specific area of research and solving or improving this problem with the help hand of information system is our aim. For this purpose, we can design this sun research question that:

Is "Social Media" an appropriate tool to share their information in a more influential way?

1.3 Limitations and Delimitation

This research is delimited by the scope of research which is defined in Municipality of Kronoberg contains these counties: Alvesta, Lessebo, Ljungby, Vaxjo, Markaryd, Tingsryd, Uppvidinge, Almhult, and the region of Kingdom of Crystal "Glasriket" contains: Emmaboda and Nybro. All these areas are located in southern Sweden.

The other delimitation for this research is Social Networks which authors try to focus on.

One of the limitations of this research was the season. This research started on the beginning of winter and in the winter there is no tourist in this area. Tourists come here mostly because of camping and fishing which is possible in the summer time. Thus, the data collection phase was difficult and it takes a long time until spring that there are some tourists.

Another limitation of this project was distances between these cities. Setting an appointment to interview with the responsible person of tourist centre in each city was so difficult and time consuming. Also, getting information by phone or email would not have that much effect in compare with meeting because in the face to face meeting they can be aware of benefits of using social media by interviewer while they are explaining the reason of why they are not using the new technologies.

2 Tourism

The first step of promoting a business is to know the fundamental definitions of the study area. This paper wants to improve tourism business so the basic definitions of tourism are important to clarify the thing which is needed to investigate. Unfortunately the definition of tourism is challenging. For instance, Gunn (1998, cited in Shaw and Williams, 2002) deliberates that all travelling modes can be considered in a tourism definition except commuting. This definition includes out-of-home activities and travelling for any reason such as a medical appointment. Thus, it does not differentiate any form of movement from tourism.

Another definition emphasizes that the tourism includes travelling away from home for recreation aims. Kelly (1998, cited in Shaw and Williams, 2002) mentions that tourism is "recreation on the move engaging in activity away from home in which the travel is at least part of the satisfaction sought."

Another definition which is bring randomly, is the practical definition chose by international bodies such as the World Tourism Organizations, through tourism embraces all travel that includes a stay away from home for at least one night but less than one year.

Most definitions highlight that tourism is described by non-permanent moves, a purpose to return home in a rather short time period without initiating an employment or permanent residence. The realistic interpret which is adopted by both academics and many statistical bodies, is to be away from home at least one night but for less than one year (Williams and Hall 2002).

Now we should know the meaning of tourism business or tourism industry. Which themes it contains? In its broadest sense, the tourism industry is the total of all businesses that directly provide goods or services to facilitate business, pleasure and leisure activities away from the home environment. So it can contain conferences and meetings, exhibitions and trade fairs, corporate events, incentive travel, outdoor events, business or individual corporate travels (Rogers, 2003). These activities can change with considering the place and time.

In this respect, we can say that hospitality and tourism are highly labor-intensive industries (Nickson, 2007). Since, such organizations concern people as one of their valuable assets. Thus, it can be resulted that one of the important potentials for economy of each country, is Tourism. Therefore, because of this matter, tourist attracting can be always one of the main concerns of each government. Nowadays the process of economic globalization is continuous and made possible by the campaign of capitalism to develop, and with the occurrence of new technologies in communications and also transportations (Appadurai, 1996; Hannerz, 1996), international tourism is one of the main inheritors of its expression (Meethan, 2001; Wahab and Cooper, 2001; Ahmadi Parssa and HosseiniKalahroodi, 2011).

Tourism affects on almost every industry. In this respect, World Travel &Tourism Council (WTTC) declares that one of the world's most important and fastest growing economic sectors is travel and tourism. Moreover, generating quality jobs and significant wealth for economies around the world is related to tourism (WTTC, 2006). WTTC believes that tourism has become the world's largest industry. Also, WTTC mentions that the industry generates more than 10% of global economic output and employment (WTTC, 2003).

Moreover, globalization is one of the common topics which are tied to tourism nowadays. Some of definitions of globalization are little more than a classic ideology. Other definitions extend in a big range and some of them are rooted in a whole theory of social change (Robinson, 1998; Wood, 2000). MacCannell (1976) notes that to build ethnography of modernity we should "following the tourists". Also, international tourism signifies a potentially plentiful field and the forces which are responded in specific location for exploring the various ways are globalizing forces (Crick, 1994; Ahmadi Parssa and HosseiniKalahroodi, 2011).

Recently, the socio-cultural aspects of globalization are conceived by sociologists and anthropologists (Appadurai, 1996; Bauman, 1998; Inda and Rosaldo, 2002; Tomlinson, 1999; Waters, 2001). Respectively, some believe globalization has some socio-aspects which are speculates by human scientists (Arizpe, 1996; Eriksen, 2003; Hannerz, 1996; Lewellen, 2002). Also, Giddens (1990) emphasizes on globalization as a process that events, decisions, and activities regardless the place in world can converge to make significant results for other communities or individuals.

We believe that one of the important areas that should be invested for this competition is information system section. Information system investments have the fundamental roles of success. If they do not improve their information technologies, so they probably lose most of their market and competitors will win the game and it will affect their economic cycle (Ahmadi Parssa and HosseiniKalahroodi, 2011).

In this regard, governments, put their investment on different sections such as city beatification, hotels, shop centers and activities (WTTC, 2003). Thus, when we are talking about the tourism, we should consider finding a relation between tourist attractions, hotels, shops, see sights, historical places, especial activities and maybe national and international conferences. Among these different sections and tourists, there is a hidden relation of data flow which can be fulfilled by information system or information technology (Ahmadi Parssa and HosseiniKalahroodi, 2011).

Furthermore, with the usage of information technology not only can relate together these different parts, but also people can have access to all information before planning easier. What people need when they arrange a trip, is information about the place, hotels availability and prices, sightseeing, maps, historical places, etc. Therefore, one of the next important things is that how we can give them this information. Now a day, tourism business like any other kinds of business has website and use different technologies to have communication with its customer (WTTC, 2006; Ahmadi Parssa and HosseiniKalahroodi, 2011)

3 Social Networks and Social Media

What are social Networks? Fowler et al (2008) claim that social networks are differentiated by wide variation at the individual level. They confess that some people have few friends whereas others have many. Also, some people are derived in strongly-knit groups that people in the group knows each other, while others are member of many different groups that they are not very friend or do not know each other well. For more explanation, scholars have required simple models of network formation that generate an experimentally realistic distribution of network attributes as an endogenous outcome of a self-organizing process (Fowler et al, 2008).

Respectfully, we want to know what social media is. Kaplan and Haenlein (2009, cited in Ahmadi Parssa and HosseiniKalahroodi, 2011) claim that Social Media is a set of internet-base applications that they are inspired of Web 2.0 technology and ideology which are based on user creation content. Also, Wasko & Faraj (2005) note that the web has been increasingly focus to further developments manifested by the evolution of social media and the appearance of the second generation of web technologies such as Web 2.0 (Ahmadi Parssa & hosseiniKalahroodi, 2011). Therefore, it can be resulted that social media is a kind of social networks which is tight to internet and Web 2.0 technology.

There are a wide range of authors which are believed most of social media technologies, such as wikis, blogs, micro-blogs, and social networks, are progressively more gaining popularity on the web with a rising number of users embracing social media in their everyday lives (Stenmark, 2008; Hasan & Pfaff, 2006; Ahmadi Parssa & HosseiniKalahroodi, 2011).

Furthermore, social media has a significant impact in organizations and it is believed that these media can facilitate and hold up new forms of mutual and knowledge management practices in growing agreements (Yates et al., 2010; Stenmark, 2008; Wagner & Majchrzak, 2007; Majchrzak et al, 2006)

Stenmark (2008, cited in Ahmadi Parssa & HosseiniKalahroodi, 2011) confesses that there is necessity to a new generation of literature on the development of social media in business backgrounds. Also, he claims that such literature would bring description sympathetic that are principally different from our previous sympathetic of mutual tools such as intranets. Hence, the development of social media involves transformative technological advancements (Yates et al., 2010) which arouse the need to understand how we really interact, collaborate, and share with each other on the web (Ahmadi Parssa & HosseiniKalahroodi, 2011).

4 Method

We chose Mixed-Method for this research because according to Creswell (2009) in mix method author use one approach for better understanding, better explanation and building the results of the other approach. Mix method is more than simply collecting and analyzing both qualitative and quantitative data because these data affect each other and overall result and study will be affected by both of them and it can be said that mix method is greater than either qualitative or quantitative research (Creswell & Plano Clark, 2007, cited in Ahmadi Parssa & HosseiniKalahroodi, 2011).

4.1 Strategy of inquiry

Strategies of inquiry on in the other hand approaches to inquiry (Creswell, 2007) or research methodologies (Mertens, 1998 cited in Creswell 2009) are divided to different ways of achievement such as qualitative, quantitative or mixed method models which design specific direction of procedure.

Moreover, Creswell (2009) says that the strategies of Mixed Method divide to Sequential, Concurrent and transformative. In this respect we chose “Sequential Mixed Method” strategy for this research since we had to collect qualitative data before quantitative data collection. The reason is discussed in the limitation and delimitation section of this paper and briefly I can mention to the season which we faced with lack of tourist in this area and we had to wait for spring that we can find some tourists for this purpose.

Correspondingly, Creswell (2009) notes that in sequential mixed method investigators try to elaborate one method based on findings of the other. This may start with each of qualitative or quantitative interviews and follow by the other one. According to this start point, sequential strategy for mixed method segregates to *Explanatory* which start with quantitative data collection, *Exploratory* which start with qualitative data collection and Transformative which has two distinct data collection phases.

Mores (1991, cited in Creswell 2009) mentions that those researchers who want to determine the distribution of a phenomenon surrounded by a chosen population choose exploratory strategy. Also Creswell (2009) notices that exploratory strategy is appropriate to explore a phenomenon and in continue to expand on the qualitative findings. Respectively, he adds researchers in exploratory strategy have to make decisions in the subsequent quantitative phase based on findings of initial qualitative phase.

In this respect, Exploratory Strategy was chosen as the best fit for this research since the quantitative phase of my research was based on the results of qualitative phase. Moreover, according to Creswell (2009), weight is usually based on the first phase. It means that in my research, weight of both research and results is based on qualitative phase.

4.2 Data Collection Methods

We choose several kinds of data collection mix together according to the research strategy of inquiry and requirements of theoretical part sequentially. Consequently, following methods were chosen to collect data:

Interviews, Observations, questionnaires, Documentation, Call their Telephone, E-mails.

4.3 Empirical Study

In first step, we started to collect data about the subject and problem. This step contains documentation data collection. It means that we collected our data by reading books, published papers and proposals, searching on the internet and websites and any documentary sources. We believed that this is the first and foremost appropriate step of each research since to be surrounded by the subject of research there are no other ways of data collection.

Next, after cognition of subject we collected the data by phone and email that this step was the step of finding the right persons who are in charge of our research area. As an illustration, we should say that tourist centres or tourist offices of the counties in the scope of our research should be recognized in order to contact them and set appointment for the next phase. Thus, the name of responsible persons and their contact information were the most important data of this step. The table of contacts were the outline of this step.

Afterward, we made a telephone conversation with all tourist centres of this region and asked them some primary questions regarding different existed information technologies in this business to know that which of these information technologies are using by them since by analyzing this data, we could drive the questions of interview for next phase. The pre-table of information was the result of this step.

Then, we wanted to discover the reason of this data. For instance we were curious to know: Why they do not use social media while it is free of charge? Why they do not offer destination cards or some other facilities to attract tourist for this area? What information they share on their website? Who are their target groups? Who visit their websites? Do they present other languages in their websites? How they give information to the tourists those who do not have any websites? Thus, we started to call them and set appointments for interviews.

In this step, before setting any appointment for interviews, the informed consent form should be prepared to present them in advanced. Also, the questions of interview sent to all of them by email in advanced because it was their request for setting this interview session.

Then we started the next step which was interviews and analysis the data rise up from these interviews. In this regard, limitations and delimitations of practical process of data collection are explained in above.

Apart from the interviews, we decided to start a new phase which was Observation in order to strengthen our knowledge about the problem. This phase before concluding the qualitative section, helped us to realize the points which were hidden from tourist centres' point of view.

Moreover, after observation we started a deep analysis of qualitative section. The results will discuss in analysis part. However these results were somehow enough to analyze according to theoretical framework, something was needed to prove the result. This reason was exactly the answer of this question that why we chose mixed method. Thus to finalize the research, we made an online questionnaire form. The questions are design so simple and short in order to be comfort for tourists to answer them.

Before explaining the distribution of questionnaire, I would like to explain about the receivers of this questionnaire and the questions. Although the focus of qualitative part was on the information system of tourist centers, we chose tourists to fulfill the results. It means that we spread out our questionnaire between tourists. One of the reasons was that we wanted to know the other party's view. We wanted to do not decide merely according to information which is gathering from business developer side. Moreover, we wanted to know whether the results from combination of both business side and theory are applicable for customers or they are interested to use it.

Respectfully, as social media was one of the focuses of theoretical work, we wanted to know whether social media itself is interesting for people. Thus, two questions designed to reach this aim before becoming deep in the tourism business in the questions.

Apart from these two questions, it should be said that first three questions were general questions. Because of the season which there were not a lot of tourist in this area to answer the questionnaire, we thought about exchange and one year free-mover students that based on definitions of Gunn (1998), Kelly (1998), Shaw & Willams (2002), Hall (2002), Higham (2005), Williams & Hall (2002) which are explain in above, they can be counted as tourism. Although based on these definitions these kinds of students are tourists, but I designed the first question to ask they are students of Linnaeus University or they are only tourist of this region in order to categorize them in analyzing part. This separation was because of students mostly using internet and social media more than a tourist itself.

Age and gender are the other first three general questions. The next three questions focused on combination of tourism and social media.

For distribution of questionnaire, first we had to choose a random sample or cluster to spread out the questionnaire that we selected clustering method for this purpose. The basic reason was lack of contact of tourists. For Clustering we decided to choose two small cities among seven counties of this region.

In this respect, we asked the tourist centers of selected cluster to send questionnaire to their customers if they have any database of them. Unfortunately, they mostly did not have any database of their customers or refused our request. Then we asked the Student Union to send our questionnaire to exchange and free-mover students. Furthermore, we went to tourist center in the sunny days of April and closed to Easter that there was probability of coming tourists. We could reach 23 tourists in those days which accepted to fill the questionnaire. There were some more tourists that refused to fill the form or even have a glance on it. Also 41 students of above condition which are counted as tourists filled the questionnaire.

5 Analysis

The results from qualitative parts show that most of the tourist offices in this region do not use social media and social networks for giving the right information to the right customers. The traditional ways of spreading the brochures are preferred by them.

The data of quantitative part illustrates that more than 85 percent of tourists check their favorite social media every day. This statistic can motivate tourist centers to use social media for their daily news and marketing. Among these 85 percent, a few aged people are seen and it is so interesting that people in range of 45-60 years old or over 60 check social media every day.

More than 43 percent of tourists think that social media is an appropriate tool to spread out information about tourist attractions while 37.5 percent of them were somewhat agree with this idea and only one out of 64 tourists which participated in the research was strongly disagree with this idea.

Furthermore this statistic shows that about 32 percent of participants have often planned their trip by attractions that they have ever seen in social media.

6 Discussion and Conclusions

Based on results raised up within the research it can be concluded that social media and in wider meaning social networks are appropriate tools for giving information for tourist business in Kronoberg county and Kingdom of Crystal.

Although most of the tourists of this area choose these places for camping and fishing activities based on tourist offices' claims, they are interested in getting the update information with social media. It was applicable for even families with some children who participated in the research. It can be resulted that social media affected on wide variety range of people.

Additionally we can discuss that in the families with young generation, no matter that old people do not or rarely use social media, younger always affect on the other members of a family. Also a tourist attraction is one of the interesting subjects of younger. Among 64 participants in the questionnaire, 46 of them are between 18-30 years old. It shows that 64 percent of them are young people. We can result in somehow that younger are more interested in tourism subject and find information about their trips in social media.

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A Probabilistic Model To Support Web Crawling For Social Media Based Sites

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ABSTRACT
Web crawling is a popular technique for data collection among researchers, practitioners and scholars. One of the major challenges web crawling techniques often face is the inefficient utilization of bandwidth, which results in unbalanced crawl between different sites. In this paper, we described a probabilistic model to improve the crawling distribution of the overall sample. The probabilistic model is designed considering both small and large social media based sites. The proposed model is designed to achieve efficient utilization of bandwidth and greater coverage of crawling samples. The initial pilot study shows promising results.

Categories and Subject Descriptors
H.5.4 [Hypertext/Hypermedia],User issues, K.4.1 [Public Policy issues]. Transborder data flow. H5.m. Information interfaces and presentation (e.g., HCI): Miscellaneous.

General Terms
Performance, Design, Reliability, Experimentation.

Keywords
Social Media, Communities, Automated Data Collection, Probabilistic Model, Web Crawling.

1. INTRODUCTION
In recent years emergence of social media based communities have gained massive popularity all over the globe. Facebook, YouTube, Flickr, Orkut etc. are a few prominent examples of such communities. These communities depend on members' participation and contribution in order to be sustainable, though the nature of those contributions can vary widely by the actual community. The contribution can be comments in discussion systems, writing opinions or factual articles in online encyclopedias, or posting pictures in user content sites, and sharing video links; most of these social media based communities support socialization and collaboration among members [9, 12]. These social media based web communities have become a focal point of data collection for practitioners, researchers, and scholars.

Automated crawling is one of the prominent techniques researchers have employed to collect and analyze data from social media based communities [2, 6, 8]. Web crawling is an instrumented technique in which a web crawler--a Spider program-- is usually employed for data collection. The web crawler seeks out and indexes web-pages within publicly visible spaces on the web (internet) in most of the cases [5]. It usually starts with a seed-list of URLs, searches for and indexes key words or all words within the body of a document, and looks for certain HTTP tokens or HTML constructs and patterns [1,5, 14]. In summary web crawling is a technique which can automate the data collection for tracking and detecting online practices, policies, user behaviors etc.

Studies have shown that web crawling methodology has been widely used by practitioners, researchers and scholars for research and data collection purposes in the past [3, 5, 6, 14, 2]. For example, to examine the effects of 'tagging' on the photo sharing social network site Flickr, researchers have used automated crawling to detect the uploaded content by the users within the network. The efforts resulted in collections of 662066 photos, 3953314 tags and 5977891 comments from 9371 users [6].

In a separate study to explore the nature and utility of social network sites, scholars also gathered data from Flickr. The web crawling technique was employed to collect the data to understand the dynamics of the social network. The crawl extracted a sub-graph of 340,000 users with 3.3 million connections between different members of Flickr [8].

These examples demonstrate that web crawling is a popular mechanism to collect data for social media sites. Within a single domain, a crawler can search deep to extract information. However, when multiple domains are needed to crawl it often becomes a challenge. If a crawler crawls very deep within a single domain it is likely that it will utilize substantial amount of resources and bandwidth in that process. This may lead to an unbalanced crawl--where a few domains will be crawled to a large extent and others will be crawled in a small proportion.

Previous studies employing web crawlers have encountered the problem of unbalanced crawl to an extent. For example, researchers from the University of Washington had examined the presence of spyware on the web using a web crawler [10]. The seed list for the crawl was selected using a category specific keyword search from the popular search engine Google. A depth of three within each domain was specified and set for this crawl. Despite limiting to a depth of three and using a balance seed-list, an average of 6,577 pages from each domain was crawled, which was expensive in terms of bandwidths and utilization of resources [10].

Jensen et al. 2007 used a web crawler to find the privacy practices for websites across the globe. The study collected two separate samples. A total of 240,340 web pages, from 24,990 unique domains were crawled [5]. One of the major limitations of their study was the under-representations of certain market segments (educational, financial and healthcare domains were under represented compared to ecommerce domains), the crawler did not reach all the segments equally. Obtaining a right proportion of data from different market segments and sites was one of the major challenges for the researchers. The crawler searched very deep within certain sites while collecting only few pages from other sites [5].

In a separate study in 2007, researchers from Google used web crawling technique to analyze the presence of malware in the web. Using their corporate infrastructure and resources the web was crawled for a stretch of 12 months. A substantial number of pages were crawled from each domain to examine the existence of malware in the web [11]. Although for corporations like Google, crawling of such depth may not be a challenge as they have the necessary infrastructure and resources to crawl, within a limited academic setting of universities this becomes a challenge sometimes for researchers and scholars.

Researchers and scholars have tried to overcome this challenge previously, by setting arbitrary depths of 1, 2, 3 to limit the crawl within a specific domain [10]. The problem with this approach was the crawler reached more pages for a larger site than for a smaller site when a depth of 3 was specified. This led to an unbalanced crawl. Moreover, crawling a greater number of pages from a large website is a waste of resources and bandwidths, which could be utilized to reach more domains.

Another approach researchers have used in the past was setting a maximum limit of crawled number pages arbitrarily equal for all domains [5,1]. The challenge with this approach is that arbitrarily setting a threshold based on researchers' intuition may fail to achieve the purpose of the data collection for certain domains. Hence, more strategic design support is required to overcome these challenges.

This unbalanced crawling is a research gap which we intend to address in this paper. Within our knowledge no study has yet looked into how to estimate the number of pages needed to crawl for different domains which can effectively support the data collection purpose and achieve maximum coverage.

We therefore considered the following research question for this study.

RQ1: How many pages a crawler needs to crawl to detect the x% probability of occurrence of certain vulnerabilities (malware, spyware etc) or web elements (such as Photos, videos etc)?

For example, if the researchers and scholars are interested in tracking the presence of web-bugs --which tracks users' activities -- with a 99% probability of occurrence within a specific site, how many pages should the crawler crawl? The model can also be used to detect 99% probability of occurrence of web-bugs involving multiple domains. We designed a probabilistic model as a solution which can support the researchers and practitioners to determine such estimations.

2. Design of a Probabilistic Model

This research demonstrated an initial proof of concept of a probabilistic model to achieve more efficient distribution of sample for web crawling technology. For simplicity and clarity we described specific examples and then extended the model in a generalized form. We also provided examples to show how this model works.

2.1 A Simple specific example to determine the Inputs for the model

Suppose a web crawler crawled 24 pages within a popular social media site which contains photos, text, links, videos etc. In this effort the crawler found videos' shared by a user in 2 pages overall.

Hence, the initial probability of randomly finding a video, in this case is 2/24= 0.083

So probability of not finding a video is (1-0.083) = 0.917

If the web crawler has sampled 5 pages then probability of not finding a video will be (0.917) ^ 5= 0.6484

Hence the probability of finding a video in 5 pages will be = (1-0.6468) = .3532

So in that domain there is only a 35% chance of finding a video if the crawler samples only 5 pages.

3. Proposed Probabilistic Mode

A more meaningful interpretation of the above example can be described if we state the problem as follows:

How many pages would need to be sampled/crawled in order to find probability of V% (70%) chance of finding a video within a social media site?

Assume the crawler has initially crawled N (20) number of pages and number of videos in those N pages were found =K (2)

Let us consider, the probability of finding a video in x number of pages is P(C) = K/N

The probability of not finding a video will be P(C')= (1-K/N)

Clearly, the probability of not finding a video after n^{th} trials

= {P(C') ^n }= {(1-K/N) ^ n}

Let n be the number of trials required to find a video with 70% probability.

Case- 1: Assume the number of pages in the social media site is infinite (appropriate for very large web sites such as Facebook).

After n trials/samples probability of not finding of a video can be represented by (1-K/N)*(1-K/N)* (1-K/N)* (1-K/N)….n trials

Hence, U^n = E

n= logE/logU where (1- K/N)= U

Hence, the probability of finding a video is E= 70%=0.7

U in this case is 0.9

So the number of pages needed to be sampled for this page will be log(0.7)/log(0.9)=3.3= 4 (approximately)

Clearly 4 pages need to be sampled in that domain to find a probability of 70% chance of finding a video. For large domains such as Facebook etc. the number of pages within that domain can be treated infinite, which holds this model true.

Case-2: When the number of pages is finite and small. This is valid for mom and pop based small social media sites. The case follows a Hyper-geometric Distribution [7, 15]. Hyper-geometric distribution is a discrete distribution which measures the probability of success in a sequence of definite number (L) draws from a finite population without replacement [15]. Hence for smaller social media websites like FledgeWing.com with finite number of pages this model holds true.

To estimate how many trials we need for this case, let us consider the total number of pages for a small social media based site is N. If the crawler has crawled y pages and was able to detect x number of videos,

Clearly the initial sampling probability of occurrence of a video in this case is x/y.

This, x/y needs to be accurate in order to estimate the number of pages correctly. If, n is the number of pages need to be sampled to find one cookie of V% (70%) of probability, then according to Hyper-geometric distribution model [7]

The probability of i successful selection can be estimated by

P(x=i) = [# ways for i successes]*{[#ways for N-i failures]/[total number of ways to select]}

$$= \frac{\binom{n}{i}\binom{m}{N-i}}{\binom{m+n}{N}}$$

$$= \frac{m!\,n!\,N!\,(m+n-N)!}{i!\,(n-i)!\,(m+i-N)!\,(N-i)!\,(m+n)!}.$$

From our case, the experimental probability can be determined from the Sampling Probability

P=x/y= Sampling Probability

For N pages total in the domain, the

Experimental probability= (Sampling probability* N)

Hence, the Experimental Probability in this case is N*x/y=V'

Let us assume we need to scan n pages to find at least 1 cookie with 70% probability.

Let A be the number of pages that contains videos out of the t pages.

$P(A=x) = (Cx)\ ({}^{N-V'}C_{n-x})\ /({}^{N}Cn)$

Now

$P(A=0) = (C0)\ ({}^{N-V'}Cn)/({}^{N}Cn)$

Now $P(X >= 1) = 1-P(X=0)$

$= 1-\{(C0)\ ({}^{N-V'}Cn)/({}^{N}Cn)\}$

Now in order to determine the number of pages needed to be sampled it should satisfy the inequality of

$P(X >= 1) >0.70$

$1- (C0)\ ({}^{N-V'}Cn)/({}^{N}Cn) > 0.70$…..i)

Since N is known, we solved for the unknown n which can be derived solving the inequality from the above equation i).

Let us consider a specific example to illustrate this model for Case 2. Consider a small media site which contains N= 400 total pages.

Assume the of occurrence of a video (x) sampled within (y) pages within this social media site is x/y=2/20=.1

This x/y must be accurate in order to estimate the number of pages we need to sample for this trial to get one video with 70% probability.

According to Hyper-geometric distribution

Here P=1/10=0.1

For N pages we should have

experimental probability=Sampling probability* N

Experimental Probability V' = 0.1*400= 40.

Let us assume we need to scan t pages to find at least 1 video of 70% probability.

Let A be the number of pages that has videos out of the t pages.

$P(A=x) = ({}^{40}Cx)({}^{400-40}Cn-x)/({}^{400}Cn)$

For x=0

Now, $P(A=0) = ({}^{40}C_0)\ ({}^{400-40}Cn)/({}^{400}Cn)$

Now $P(X >= 1) = 1-P(X=0)$

$= 1- ({}^{40}C_0)\ ({}^{360}Cn)\ /({}^{400}Cn)$……ii)

The inequality ii) is solved for n.

$1- ({}^{40}C_0)\ ({}^{360}Cn)/({}^{400}Cn) > 0.70$…..ii)

Solving the inequality, we get nearly 18 pages that needed to be sampled in this case.

The model supports crawling of multiple domains efficiently. Instead of relying on arbitrary intuition, researchers can specify the probability of occurrence of any vulnerability or web elements (cookies etc) within the internet. The crawler can detect occurrences of vulnerabilities based on the probability which researchers/scholars have specified. The model can work in conjunction with depth of set of crawling and page limit if implemented correctly.

4. Discussion and Implication

The goal of this paper is to show a mechanism of how to estimate the number of pages needed to crawl to extract certain practices within the web. The model is designed with the intention to achieve a balance distribution for crawled samples, when a web crawler is employed to crawl data from multiple domains. Our model estimates precisely the number of pages needed to be crawled for data collection to detect practices of certain probability for social media sites. Additionally, the model may help to effectively use the bandwidth and channel capacity to achieve a greater spread of domains for web crawlers. Hence, in this study we have extended and enhanced the efficiency of automated web crawling techniques in terms of coverage and precision control.

We designed two separate probabilistic models for social media based websites. For large social media based sites like Facebook or Orkut, where number of pages can be considered infinite, the estimation is a simple probabilistic approach (Case-1). For mom and pop based small social media based communities like FledgeWing (www.FledgeWing.com) or Navy For Moms (www.navyformoms.com) etc, the probabilistic model followed Hyper-geometric distribution which deals with the finite population without replacing the population in each sequence for prediction (Case-2).

We developed this model to target a more balance crawl, involving multiple domains based on our previous experience using automated data collection to detect practices [5]. When a web crawler is employed for data collection random sampling is hard to obtain, web crawlers need seed-lists to initiate the crawling. The seed-list needs to be provided first externally, it is likely that the crawler will achieve bias in terms of sampling. Bias due to the seed-list is a hard problem to tackle. However sampling distribution within a web crawling technique can be optimized based on the requirements, specifying the probability of occurrences of vulnerabilities/elements (high or low) as per requirements.

We believe that our proposed probabilistic models would support the decision-making of how many pages need to be sampled to detect the occurrence of certain probability of any threats or elements. This will allow practitioners and researchers to utilize resources and bandwidth in an efficient way to achieve greater spread of domains. The probabilistic model is targeted to improve the coverage of the sample. For example, if the researchers want to find out the presence of probability of existence of web-bugs in various social media sites of a 99% chance, this model will support that efficiently without crawling huge amount of pages within a single domain. We conducted some initial pilot tests within social media based communities. Our initial results show that the existence of web-bugs within social media based communities is consistent with the previous research findings [13]. However the models may need further fine tunings based on the requirements of the data collection. This opens up the idea of the more rigorous testing of this model and further research. In summary, this research has demonstrated a theoretical general approach, which we believe can be useful for data collection and practices.

5. Conclusions and Future Work

In this research, we have designed a probabilistic model which can support decision making for researchers, scholars and practitioners to determine how many pages need to be crawled to detect the probability of occurrence of certain practices or web threats/element. Both of these cases need an initial probability, and finding it for each seed from a seed list may take a lot of (human) effort. Indeed, the matter of seed-list bias, is definitely a hard problem. The design and efficiency of this model should be considered with caution, due to the dynamic complex nature of the internet [5]. Moreover, sometimes the internet acts a disconnected island [4]. In the future months, we aim to conduct more distributed crawls involving multiple social media based communities to test the performance of our probabilistic model. We also aim to test the model for different web segments of social media based communities targeting healthcare, government, ecommerce etc. separately to measure the accuracy of this model.

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Mum's the Wordpress: A Comparative Analysis of Political and Mommy Bloggers

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Abstract: *This research paper presents findings into the differences between two types of popular bloggers: the political blogger and the mommy blogger. These terms are recent entries to the lexicon of online communities, but are soon becoming distinct concepts. This paper shows that mommy bloggers rarely discuss the issues mainly associated with political bloggers, although the reverse is not always true. While political bloggers talk about family issues, this often has little to do with calling for their rights, but echoing sentiments relating to the family life of political public figures.*

Keywords: weblogs, blogging, mommy blogger, political blogger, genre, sub-genre, sysop prerogative, online communities

1 Introduction

A weblog, or "blog," is a genre of online community, where the posts comprise hyperlinks to articles, news releases, discussions and comments that vary in length and are typically presented in chronological order [1, 2]. These blogs are a source of human activity knowledge comprising valuable information such as facts, opinions and personal experiences [3]. Weblogs are a relatively new type of online community, and the community element of this technology exists when the owner, who is referred to as a *blogger*, invites others to comment on what they have written [4]. While the terms *weblog* and *blog* are used synonymously, the difference between the two is that a blog is intrinsically motivated based on the owners' own thoughts and emotions whereas the weblog is extrinsically motivated based on media that the actor has consumed and experiences the author has had with others. A weblog allows individuals to express their identities through its name and title, user profiles and about pages, posted content, and the visual design of the website [5].

2 Weblogs and the Public Square

The public sphere was once the preserve of wealthy businessmen, meeting in cafes, before mass media made information accessible to the masses. However, the emergence of the *public square* where only the elites could have access to publishing outlets is also now open up to the masses [6]. Weblogs have been the means by which many people have been able to express themselves online, in many cases accessible to the whole cyberspace. Weblogs have also been talked about as a means of increasing literacy of the writers and facilitating communication between politicians and their constituents. Two prominent groups of weblog writers are referred to as the *political blogger* and the *mommy blogger.* The political bloggers' posts are usually expressions of their opinions and views or comments on topical issues. The mommy bloggers are characterised by their regular posts about family and community related issues important to many mothers, mums, moms and mams. Bloggers are always identifying against that which they do not identify [7]. Political bloggers are active in all parts of the world. Australian political bloggers and citizen journalists appear to have played an important role in the campaign leading up to the November 24, 2007 federal election [8]. Some have commented that despite the impact that influential American political bloggers have had on public policies and the mainstream media agenda in recent years, very little research is available on widely read political bloggers [9]. Mommy bloggers are known to be political at times when talking about their family. For example, one mommy blogger wrote on her site that while going through an Atlanta airport security checkpoint, federal transport agents separated her from her baby [10]. Some authors have argued that mommy bloggers are highly influential and in some cases beyond criticism because they use their children and betterment as their point of reference [11].

3 An Investigation

A total of 40 web-based communities from the genre of Weblog were selected for analysis. Twenty were selected from the BlogExplosion directory in the mommy blog sub-genre. Twenty were selected from Iain Dale's Top 100 Political Blogs [12] in the political blogger sub-genre. Noticeable from the Weblogs of the two sub-genres was the use of photographic artefacts. The mommy blogs tended to have pictures of their children, either what could be termed baby pictures or pictures of older children at parties and receiving presents. On the other hand, many of the political bloggers are photographed with politicians. These photographs could possibly act as character codes [13] for the dialogue as often the photographic artefacts are anchored with textual artefacts relating to them. A total of 16373 posts from the mommy blogger category and 173268 posts from the political blogger category were analysed. The posts were analysed to assess whether they contained 52 terms, 26 taken from the *Handbook of Parenting* [14] that were attributed to the mommy blog category and 26 taken from

Politics: An Introduction [15], which were assigned to the political blog category. Each of the blogs were assigned a score between 0 and 1 based on how much they kept to the 26 terms assigned to their sub-genre without deviating by using the 26 terms assigned to the other sub-genre. A summary of the results is presented in Table 1.

Table 1. Mean Number Posts and Scores for political and mommy blog sub-genres

Blogger Type	N	M Own	M Other	M Score
Mommy Blog	20	794.55	24.10	0.98
Political Blog	20	5576.45	3086.95	0.59

Performing an independent samples t-test of the data showed a significant difference between the two sub-genres ($t=9.44$, $p=<0.001$). It was observable that most Web-based communities assigned to the mommy blog category did not deviate into using the terms assigned to the political blog category with a mean score of 0.98 (SD=0.03) and posting a mean of 794.55 posts using their own terms and only a mean of 24.10 posts using the terms assigned to the political bloggers whereas those web-based communities assigned to the political blog category did deviate into using the terms assigned to the mommy blog category, scoring a mean of 0.59 (SD=0.18) and posting a mean of 5576.45 posts using their own terms and a mean of 3086.95 posts using the terms assigned to the mommy blog category. A discourse analysis of the use of the 52 terms across the two sub-genres was carried out to discover whether there were observable differences in the cognitions of the actors that posted in the two sub-genres. A number of discoveries were made.

4 Discussion

Weblogs are a genre of online community where the posters present links to various other pieces of content or simply present their own opinions in articles which are typically ordered chronologically. While there is a lot known about this genre of online community, there is little published research on the sub-genres of weblogs, such as the political blogger and the mommy blogger. These are two distinct types of blogger, where the former expresses their opinion and shows their affiliation to political events and causes through pictures and text. The latter, on the other hand, describes events that occur within their family and often posts pictures of them and their community. This paper has presented solid evidence for the existence of these two genres by comparing the content of various political and mommy blogs. The data shows that political bloggers are less likely to post on the same topics as mommy bloggers, and mommy bloggers are the same with regards to political blogging. It suggests that even when they overlap, the context is different and clear to the defining characteristics of those sub-genres of weblog.

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Dominant Student Modeling for Web Personalization

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Abstract

The large amount of information on the Web can play a significant role in building an efficient repository. To overcome this challenge, we create a considerable model called Dominant-Model based on Naïve-Bayes classifier. An incoming document will be reformulated to take the same shape of our model and then this considerable model will be used as a measure to classify it to suitable class. We investigated the effect of using this model on classifying Web information. We notice some promising experimental results showing that the use of our model improves the classification accuracy in most of the cases.

1. Introduction

A user model is represented by one or more of the following elements [Kobsa 01]: 1) representation of goals, plans, preferences, tasks, and/or abilities about one or more types of users; 2) representation of relevant common characteristics of users pertaining to specific user subgroups or stereotypes; 3) the classification of a user in one or more of these subgroups or stereotypes; 4) the recording of user behavior; 5) the formation of assumptions about the user based on the interaction history; and/or 6) the generalization of the interaction histories of many users into groups.

With the fast expansion of the Web, we desperately need a new framework that can classify documents faster than previous frameworks. Our framework is to create a mostly considerable model for each class called Dominant-Model (DM). Accordingly, this model will be used as a measure to filter relevant and irrelevant documents. In general, the incoming document will be reformulated to take the same framework; therefore, it is compared with DM.

Web classification is the problem of gathering Web documents into different fixed classes built in advance. The classification problem has been felt for a long time and many approaches have been tried to solve it. Accordingly a well-organized method for classifying specific fragments of these documents is a promising way for building updateable Web libraries. Machine learning framework tries to create a function that will map new documents into their classes using a set of labeled and unlabeled examples.

We have derived DM technique from the design of a perfect human body. The human body consists of three parts: head, chest and abdomen, and legs. We can not say that a body is a perfect unless i) its parts are individually perfect, ii) head and legs are compatible with chest and abdomen, and vice versa. Accordingly, the DM consists of three parts, each part has to be relevant individually, and compatible with the others.

Therefore, each Web page is divided into consists of three fragments; title, keywords and body. Each of those contains some dominant meanings [Razek 03a]. The dominant meanings definition is known as "the set of keywords that best fit an intended meaning of a target word" [Razek 03b].

This paper is organized as follows. Section 2 gives an introduction to related work. Section 3 discusses the role of the technique used to solve the classification problem. Section 4 presents the results of experiments conducted to test our classification methods. And section 5 concludes the paper and future work.

2. Related Works

Bayesian assessment is considered as a comfortable technique for estimating the probability distribution of classes over feature data inputs [Friedman 97]. In the Text Classification domain, this is often referred to as Naïve Bayes text classification.

There are two models of the Naïve Bayes classifiers: multivariate Bernoulli and multinomial mixture model. A document at multivariate Bernoulli is represented as a vector of binary attributes. However, this vector signifies which words occur and do not occur in the document, it does not take into account the number of times a word happens in a document. This approach is more traditional in the field of Bayesian networks, and has been used for text classification by several researchers [Kalt 96], [Larkey 96], and [Sahami 98].

The basic idea of it is to use the joint probabilities of words and categories to assess the probabilities of categories given a document. The advantage of Naïve Bayes technique over other techniques is based on the assumption of word independence. On other words, the conditional probability of a word given a category is assumed to be independent from the conditional probability of other wards given that category. Consequently, this way makes the computation of the Naïve Bayes classifiers more accurate than the exponential complexity of non-Naïve Bayes techniques because it does not use word combinations as predictors [Yang 99].

On the other hand, a document at multinomial technique is represented by the set of word occurrences from the document. However, the order of the words is lost, the number of occurrences of each word in the document is taken into account. This approach has also been used for text classification by several researchers [Nigam 98].

Using the hierarchical decomposition of a classification problem allows for competence in both learning and representation. Basically, each sub-problem is smaller than the original problem, and it is sometimes possible to use a much smaller set of features for each [Koller 97].

Razek et al. [Razek 4] suggests a new filtering framework, called a pyramid collaborative filtering model, to trickle the number of helpers down to just one. The proposed pyramid has four levels. Moving from one level to another depends on three filtering techniques: domain model filtering; user model filtering; and credibility model filtering.

In the big-bang approach, given a document, the classifier allocates it to one or more categories in the category tree. The allocated categories can be internal or leaf categories depending on the category structure supported by the methods. This approach is more traditional in the field of hierarchical classification, and has been used by several researchers such as Rocchio-like classifier [Labrou 99], and rule-based classifier [Sasaki 98]. In the top-down level-based approach, a document will primary is classified by the classifier at the root level into one or more lower level categories. The classifiers of the lower level categories will then further classify it until it reaches a final category, which could be a leaf category or an internal category. This approach has also been used for text classification by several researchers [Koller 97] and [Dumais 00].

In the next section, we explain in more details our technique included the construction of the proposed model.

3. Dominant-Model Approach

Machine learning offers a lot of diverse methods to perform classifying several items into their corresponding categories. Generally, text classification considers that a document consists of a set of words and tries to categorize the document among a specified set of classes. For sound classification, we must restrict our domain knowledge to the problem of text classification. We thus classify each given document as belonging to a particular concept and then restoring it to a specific dominant meaning. We must specify a concept that is closely related to the document context.

Recently, Most of the text classification algorithms proposed is based on flat vectorial representations of Web documents. This is like representing each document as a feature-vector, where features are words in the vocabulary and components of the feature-vector are statistics such as word counts in the document. However, in this way, important information is lost, since the bag–of–words representation does not consider the difference between meanings of words. The existing hierarchical classification methods have basically two approaches [Sun 01]: the big-bang approach and the top-down level-based approach.

In the big-bang approach, given a document, the classifier allocates it to one or more categories in the category tree. The allocated categories can be internal or leaf categories depending on the category structure supported by the methods. This approach is more traditional in the field of hierarchical classification, and has been used by several researchers such as Rocchio-like classifier [Labrou 99], and rule-based classifier [Sasaki 98]. In the top-down level-based approach, a document will primary is classified by the classifier at the root level into one or more lower level categories. The classifiers of the lower level categories will then further classify it until it reaches a final category, which could be a leaf category or an internal category.

3.1 Web Page Structure

In our approach, each Web page consists of three fragments; head, title and body. Each of those contains some dominant meanings. The more dominant meanings; the better a concept relates to its fragment context.

Suppose an example for representing a Web page of List concept, in Data Structure course, Computer science program. The page consists of one top-level category ("List"). Within three subcategories, there are combinations of the following dominant words ("list", "Array-based", "Array-Based", "Linked"). Next subsection outline feature extraction from fragments.

3.2 Feature Extraction using Dominant Meaning

Feature selection techniques may increase the accuracy of classification if important features are included. In contrast they may decrease the accuracy if those important features are excluded. For example, when a feature is important in one document, but is avoided by a system that is used to extract features. This example leads us to think about the problem of which features should be extract, and how to extract them. Essentially, for text features, features consist of words that have highest average common information with the class variable [Craven 98]. In other words, this is evaluated by calculating the average common information between the class of a document and the absence or presence of a word in the document.

In this work, we experiment with alternative techniques for feature selection that are dependent on the context of features. Our techniques have the advantage that the features that are used from each document are chosen solely on the context of that document. We use the dominant meanings of the class concept at each training document as features. These meanings will be rather than bag-of-words. They are evaluated by measuring the dominant meaning space between the class of a document and the frequency of those meanings in the document [Razek 03a]. Formally, we represent a set of classes in our hierarchy classification as $C = (c_1, c_2, ..., c_{|C|})$, where $|C|$ represents the number of classes in C, and $\mathrm{X} = \{x_1, x_2, ..., x_n\}$ represents a set of dominant meanings (features) of those classes.

3.3 Building Dominant Model

Suppose that D be generic Web documents, and let D_t denote the t-th page within these documents. The classification task consists of building from examples perfect dominant model that maps each page D_t into one out of $|C|$ classes. A considerable model is constructed by learning from a training set, which includes a set of documents relevant with respect to the topic.

$$\mathrm{P}(S_i \mid S_j) = \frac{1}{T}\left[\sum_{k=1}^{k=T} \frac{F(S_i \mid g_k)}{F(S_j \mid g_k)}\right] \qquad (2)$$

A document D_t divides into three differently fragments: S_1, S_2 and S_3. Accordingly, each part has to be relevant individually, and compatible with the others.

Basically, the DM classifier views fragments as set of conditionally independent features (dominant meanings) $\mathrm{X} = \{x_1, x_2, ..., x_n\}$. As follows, we will compute the value that represents relevance of each fragment.

Each fragment of D_t contains a subset of the words in X. We would like to calculate the probability that this document D_t is in belonging to a concept c_k.

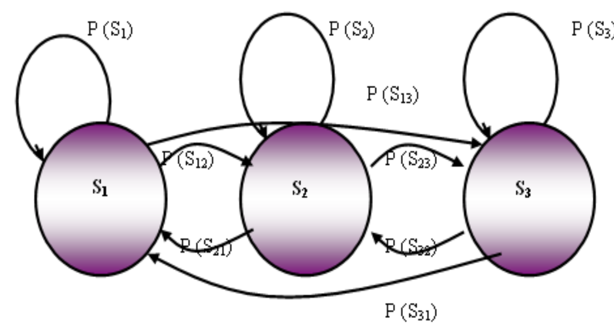

Figure 1 Dominant Model Topology

Suppose that C_h is a concept S_i that is accoicated with a document D_t and S is a fragment. And suppose also that a set of dominant meanings of the concept C_h is $\{g_1, ..., g_T\}$ which is a subset of X (more details about how these meanings determined at [Razek 2003a]). Suppose that word w_h symbolizes

$$\mathrm{P}(S) = \frac{1}{T}\left[\sum_{m=1}^{m=T} \frac{F(S \mid g_m)}{F(S \mid w_h)}\right] \qquad (1)$$

concept C_h Based on these, we compute the distance space between the fragment S and the concept C_h as follows:

Where the functions $F(S \mid g_m)$ and $F(S \mid w_h)$ represent the frequency of occurrence of the two words g_m and w_h in the fragment S.

We can compute the functions $F(S \mid g_m)$ and $F(S \mid w_h)$ from the equation (10) in section 3.4 using Naïve-Bayes classifier. The distance space between the two fragments S_i and S_j as follows:

Where the functions $F(S_i \mid g_j)$ and $F(S_i \mid g_j)$ represent the frequency of occurrence of the two words g_k in the fragment S_i and S_j.

Next section shows our experiments comparing the two feature selection methods on the data structure data set collected from the Web with the multivariate Bernoulli model.

3.4 Naïve Bayes Text Classification

Basically, the Naïve-Bayes classifier views documents as set of conditionally independent features (dominant meanings) $\mathrm{X} = \{x_1, x_2, ..., x_n\}$. We can see it as a function that maps an input feature set X to the corresponding class c_k, where $c_k \in C = (c_1, c_2, ..., c_{|C|})$, and $|C|$ represents the number of classes in C.

As discussed in the previous section, for each document (chunk) D, we only test the existence of currently dominant meanings of class to apply as features. This set is X, and can be updated over time. The document D thus contains a subset of the words in X. We would like to calculate the probability that this document D is in each of the different classes c_k. We use Bayes theorem to estimate this probability:

$$P(C = c_k \mid D) = \frac{P(D \mid C = c_k) P(C = c_k)}{P(D)} \tag{3}$$

Where $P(C = c_k)$ is the prior probability that any random document belongs to the class c_k, and $P(D)$ is the likelihood of document D. Since $P(D)$ is constant for any particular document D, it can be dropped. If we suppose that the features of the document D consists of $\mathrm{X}' = (x_1, x_2, ..., x_m)$ which are conditionally independent, given the category variable C, this simplifies the computation of $P(D \mid C = c_k)$ as follows:

$$P(D \mid C = c_k) = P(\mathrm{X}' \mid c_k) = \prod_{i=1}^{m} P(x_i \mid c_k) \tag{4}$$

Since the Bayesian approach is only using $P(D \mid C = c_k)$ for classifying a new document, thus to predict the best class $\mathrm{B}(\mathrm{X}')$ given the document vector $\mathrm{X}' = (x_1, x_2, ..., x_m)$, we simply need to compute the maximum a posteriori value of $\mathrm{B}(\mathrm{X}')$:

$$\mathrm{B}(\mathrm{X}') = \arg\max_{c_k \in C} P(C = c_k \mid D) \tag{5}$$

$$= \arg\max_{c_k \in C} \frac{P(D \mid c_k) P(c_k)}{P(D)} \tag{6}$$

$$= \arg\max_{c_k \in C} P(c_k) \prod_{i=1}^{m} P(x_i \mid c_k) \tag{7}$$

Computing $P(c_k)$ is straightforward. Following [Nigam 00], given the training data, the prior probability of a class is typically determined by the maximum likelihood estimate as the fraction of documents in it, giving

$$P(c_k) = \frac{n_{c_k}}{N} \tag{8}$$

where N represents the total number of training documents in all classes and n_{c_k} is the number of examples belonging to class c_k.

As mentioned above, there are two distinct classifiers which build on the Naïve Bayes Text Classification: Multivariate Bernoulli and multinomial mixture model. The difference lies in calculating $P(D \mid c_k)$. Whilst the multivariate Bernoulli model considers binary features which points to the appearance or non-appearance of a term x_j in a particular document D, the multinomial mixture model uses plain term frequencies in a document. This research is dealing with the multivariate Bernoulli model.

According to [Schurmann 1996], the likelihood of a document given the class is calculated

As follows:

$$P(D \mid c_k) = \qquad (9)$$

$$\prod_{i=1}^{m} P(x_i \mid c_k)^{I(x_i,D)} (1 - P(x_i \mid c_k))^{(1-I(x_i,D))},$$

where

$$I(x_i, D) = \begin{cases} 1 & x_i \in D \\ 0 & x_i \notin D \end{cases},$$

and according to the training data, we can compute $P(x_i \mid c_k)$ by:

$$P(x_i \mid c_k) = \frac{1 + N_{c_k}(x_i, D)}{2 + N_{c_k}(D)}$$, where $N_{c_k}(x_i, D)$

represents the number of documents belonging to class c_k in which term x_j appears at least once, and $N_{c_k}(D)$ is the total number of documents in class c_k.

Simply, to decide the class c_k to which a new document D belongs, here we briefly outline the steps from formula (1) to (9):

$$P(C = c_k \mid D) =$$

$$\arg\max_{c_k \in C} \begin{cases} \dfrac{n_{c_k}}{N} \prod_{i=1}^{m} \dfrac{1+N_{c_k}(x_i,D)}{2+N_{c_k}(D)} & x_i \in D \\ \dfrac{n_{c_k}}{N} \prod_{i=1}^{m} \dfrac{1+N_{c_k}(D)-N_{c_k}(x_i,D)}{2+N_{c_k}(D)} & x_i \notin D \end{cases} \qquad (10)$$

Next section shows our experiments comparing the two feature selection methods on the data structure data set collected from the Web with the multivariate Bernoulli model.

4 Experimental Results

In this section we describe the experiments in order to observe the influence of using the previous model. To show convincingly that improvements in accuracy are derived completely from the use of dominant meanings, two experiments using dominant model were performed using the following representations of date sets of training documents:

- Bag-of-words using only nouns and verbs without stop list, and
- Dominant-words of the class concept at each training document.

The first experiment represents a document as a bag-of-words, while the second one represents it as a subset of the dominant meanings of the class concept at each training document. These meanings will be in place of the bag-of-words. They are evaluated by measuring the dominant meaning space between the class of a document and the frequency of those meanings in the document (at [Razek 03a] you will find details about how we construct the dominant meanings of documents) (8)

As far as we know, there are no freely available data sets for Data Structure course. For this reason, we used WebZip[1] spider to accumulate a data set of data structure course from the Web. We divided the collection into four categories: queue, stack, list, and searching all together containing 1300 pages. The resulting vocabulary has 156000 words before erasing Stop Words List. As we mentioned before, we did not use stemming but we removed Stop Words List. Table 1 gives information regarding the data sets on which experiments were performed.

Table 1: Collection used for experiment

Number of documents in collection	1300
Number of Terms	156000
Number of categories	4
Number of documents used as training	520
Number of documents used as training in List category	50
Number of documents used as training in Queue category (9)	100
Number of documents used as training in Stack category	150
Number of documents used as training in Searching category	220

We used 40% of documents in the collection as training set for building the four categories. We used formulas (1) and (2) to compute the perfect models. Therefore, we build a model for each D_t of the collection and compared it with the considerable

[1] http://www.spidersoft.com/webzip/

model in order to decide for which categories D_t belongs.

Table 2: Experiments Results

	Measure	Classification with Dominant Meanings	Classification with Bag-of-Words
List	Pr	0.77	0.44
	Re	0.74	0.46
Queue	Pr	0.83	0.55
	Re	0.98	0.65
Stack	Pr	0.86	0.65
	Re	0.83	0.55
Searching	Pr	0.83	0.67
	Re	0.85	0.69

Table 2 shows the comparison of the results of the two experiments: dominant meanings approach against the bag-of-words approach. Comparing the precision and recall, obviously, Bag-of-words is a loser compared with Dominant Meanings. However, some enhancements were viewed for all categories; the List category had the lowest improvement. Maybe this refers to the little training data used to build it comparing with others (see Table 2).

5 Conclusions

In this paper, we presented a new classification model called dominate model approach. We examined the influence of using dominant meaning features in fragment classification using DM classifiers. In relation to using bag-of-words features, our experiments have shown that dominant model along with dominant meaning did very well for all measures precision, and recall

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SESSION

METHODOLOGIES

Chair(s)

TBA

QoE Modeling and MPEG-21 DIA Multimedia Framework in Telecommunication and Broadcasting Convergence

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Abstract - *With the increased consumption of multimedia content, users have become more eager to access multimedia content anytime, anywhere, and via any device. In this paper, we present a quality of experience (QoE) model for a broadcasting and telecommunication convergent environment and propose applying this QoE model to the MPEG-21 DIA (Digital Item Adaptation) Framework. We propose setting up a "Media Decision Taking Engine" (MDTE) in the MPEG-21 DIA Framework, and present a design for the MDTE and an explanation of how it works. As such, by making use of the proposed MDTE, an optimal multimedia content adaptation/distribution system can be made available even in a diverse network environment.*

Keywords: Quality of Experience, MPEG-21, Multimedia, Telecommunication, Broadcasting

1 Introduction

Access to multimedia content such as video and audio is becoming essential in everyday life. People are eager to access multimedia content anytime, anywhere, and via any device. Consequently, universal multimedia access (UMA) research and the standardization effort have gained serious momentum. The goals of UMA are two-fold: (1) any content should be available anytime, anywhere; and (2) access to multimedia content should be interoperable and transparent. This may require converting the multimedia content available from the content provider into a format that can be consumed by the user, while at the same time maximizing the user's quality of experience (QoE). This could also require an efficient solution to enable content variation, under different network/device conditions such as a broadcast network and the Internet.

In a previous work, Zahariadis et al. [5] proposed a cross-layer adaptation framework to achieve sustainable end-to-end quality as indicated by the user. Angelides et al. [7] reported their conceptual model design, an implementation of AQoS (Adaptation Quality of Service), and an evaluation of their design. Xu et al. [6] introduced their AQoS and decision-making algorithm/implementation in their video adaptation system, while Kim et al. [8] introduced their implementation of the objective perceptual quality-based ADTE (Adaptation Decision Taking Engine). Timmerere et al. [9] proposed an integrated management supervisor that takes into account the requirements of the stakeholders and provides an end-to-end management solution providing the end user with QoS. Tong et al. [10] proposed a model for the ADTE, which is based on a decision tree, called an adaptation decision tree (ADT). Kanellopoulos [11] presented research on ADTEs and studied how intelligent ADTEs can be designed and developed. However, the above-mentioned reports do not cover the optimal distribution balance issue in telecommunication and broadcast convergence.

In this paper, we propose a QoE model in a hybrid broadcast and communication environment and apply this model to the MPEG-21 DIA (Digital Item Adaptation) framework [1].

The remainder of this paper is organized as follows. Section 2 introduces the hybrid broadcast and communication service model, while Section 3 describes our QoE evaluation model. In Section 4, the results of an integrated QoE simulation are presented. Section 5 describes the MPEG-21 DIA multimedia framework. Section 6 introduces the proposal of applying our QoE model to the DIA architecture by extending the adaptation engine. Finally, our conclusions and future work are outlined in Section 7.

2 Hybrid broadcast and communication service model

Our model is based on digital data broadcasting as implemented in Japan and some of the results reported by us in [4][12]. It is assumed that the opening screen data is sent via the broadcast system, while the content of subsequent screens is sent via either the broadcast system or a communication network based on BML (Broadcast Markup Language), which is the digital broadcasting standard in Japan. Figure 2 shows how multimedia content is divided into modules as BML element modules and how each module is transmitted via the broadcast or telecommunication network. As shown in Fig. 1, programs first pass through a smart media server where the data is separated into a content part that is transmitted via the broadcast system and another part sent via the communication network. Communication providers feedback information on user attributes, connectivity conditions (e.g., network speeds), content popularity, and so on to the smart media server to optimize the balance of broadcast and communication services.

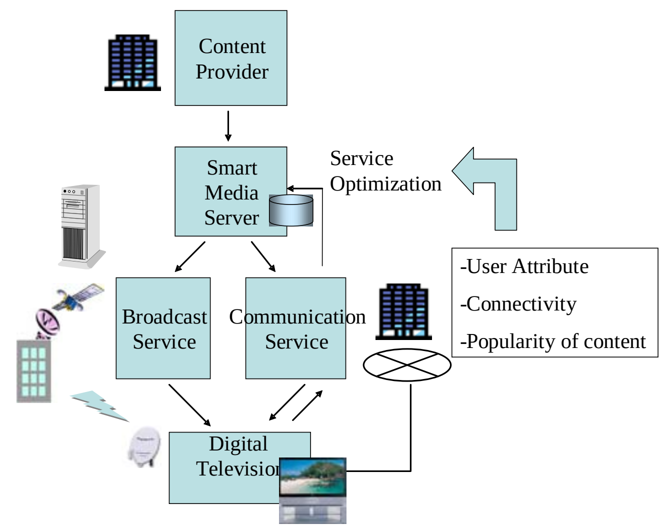

Fig. 1 Service model for broadcast and telecommunication

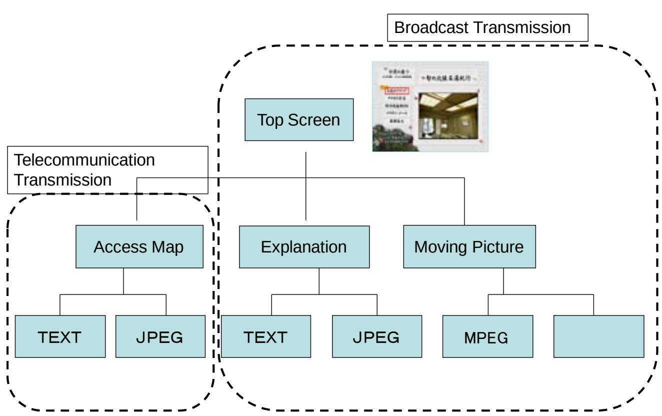

Fig. 2 Multimedia content model

3 QoE evaluation method

3.1 QoE evaluation method

To find the level of end-user utility versus delay time, it has been reported in [2] that a measurement function can be formulated using an exponential function, such as that given by Eq. (1). To measure QoE, delay time is adopted as the main factor in the QoE simulation.

$$QOE(t)=e^{-kt} \qquad (1)$$

3.2 Communication system model

For TCP file transfer times, we established a model by estimating the transfer time by means of the M/G/R/PS model in the network configuration depicted in Fig. 3.

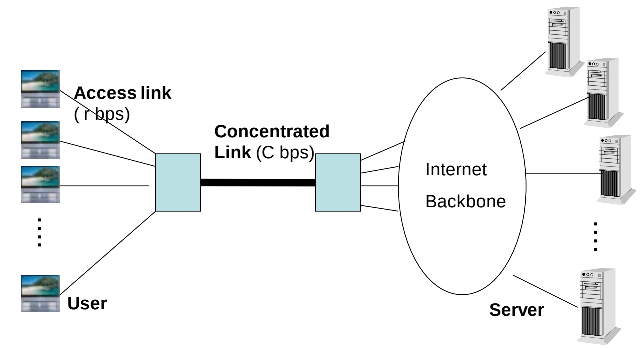

Fig. 3 Communication system model

According to [3], the average file transfer time $T(\rho)$ for a concentrated link utilization rate of ρ is derived as:

$$T(\rho)=\frac{s}{r}\left[1+\frac{E_{2,R}(\rho)}{(1-\rho)C/r}\{1-(1-\rho)(C/r-R)\}\right] \qquad (2)$$

where s is the average file size, r is the access line speed, C is the concentrated link speed, R is INT[C/r], and $E_{2,R}$ is the Erlang-C formula with the number of servers equal to R.

The end-user satisfaction (QoE) of the content received via the communication can be represented by the mean values for all users, derived by subtracting the necessary communication costs from the satisfaction rate for each user, as given in Eq. (3).

$$\sum_{i}^{Uc}(QOE(T\rho(i)))-COST(ri))/Uc, \qquad (3)$$

where *Uc* is the number of users receiving content through the communication network, and *COST(ri)* is the cost of the access line for user *i*. In this case, the relationship between link speed and COST() is assumed to be:

with link speed < 128 Kbps: COST()=0.01
with link speed < 2 Mbps : COST()=0.02
with link speed > 2 Mbps : COST()=0.03

If there is congestion on the concentrated link, $T(\rho)$ will reach an extremely high value because of the nature of Eq. (2). This means that the QoE() value for these users will become almost zero. Therefore, we do not need to consider users subject to congestion, as shown in Eq. (4).

$$\sum_{i}^{Ucs}(QOE(Sc/r)-COST(r))/Uc, \qquad (4)$$

where *Ucs* is the number of users receiving content under stable conditions (i.e., without congestion) and *Sc* is the size of the content.

Since *Sc/r* is constant, Eq. (5) is equivalent to Eq. (4).

$$(QoE(Sc/r)-cost(r)) * Ucs/Uc \qquad (5)$$

3.3 Broadcast system evaluation model

In data broadcasts, the data are sent by a carousel transmission system that repeatedly transmits the same content (a series of program data) as shown in Fig. 4.

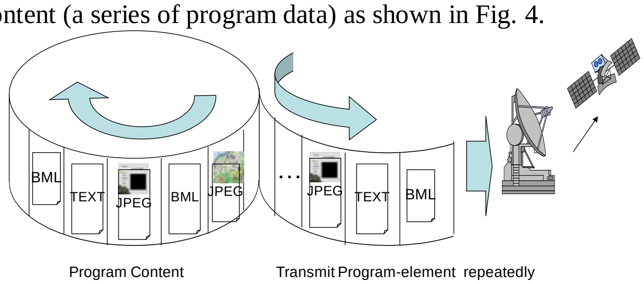

Fig. 4 Carousel transmission system

A viewer can see the program by waiting no longer than one carousel cycle. However, since the broadcast is multiplexed, it needs to be multiplied by a multiplex coefficient (*m*).

Carousel cycle = (Sb/Cb) * m,

where *Sb* is the size of the content that is sent via the broadcast system, and *Cb* is the speed of the broadcast system. Therefore, the utility of the broadcast part is evaluated using:

$$QoE((Sb/Cb)*m) \qquad (6)$$

4 Results of hybrid broadcast and communication system QoE evaluation

The model for the study in this paper, as shown in Fig. 2, is assumed to be a BML content model. In this model, content modules in the lower level are transmitted using both telecommunication and broadcast systems. Therefore, the entire content of these modules is considered subordinate to one broadcast BML program.

In this discussion, the weight of each content module is assumed to be equivalent. The average broadcast and communication convergence QoE is considered to be the product of the average QoE sent by broadcasts and that sent by the communication network.

The hybrid broadcast and communication system QoE is expressed as in Eq. (7) below, using the previously introduced Eqs. (5) and (6).

$$((QoE(Sc/r)-cost(r)) * Ucs/Uc) * QoE((Sb/Cb)*m) \qquad (7)$$

The results of a simulation we conducted, whereby the size of content provided via the communication network was continuously increased from 0% to 100%, are shown in Figs. 5, 6, and 7. In these figures, the Server Capacity coefficient is Ucs/Uc, where Ucs is the total number of users that can issue requests before congestion arises, Cb (broadcast speed) is assumed to be 1 Mbps, and *m* (the multiplex coefficient) is assumed to be 1.

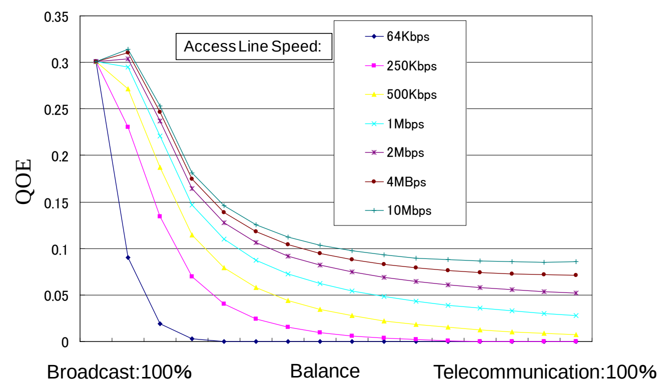

Fig. 5 With a server capacity coefficient of 0.15

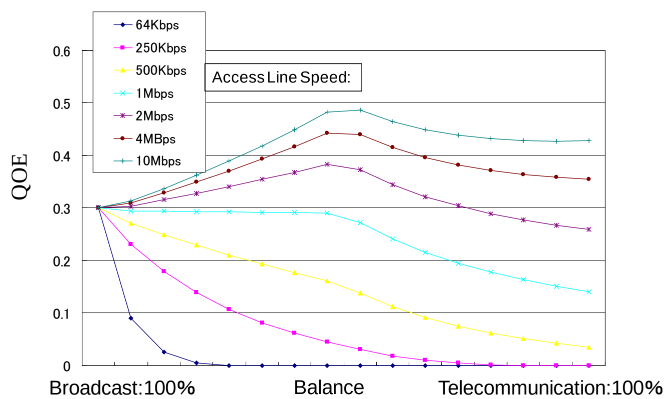

Fig. 6 With a server capacity coefficient of 0.4

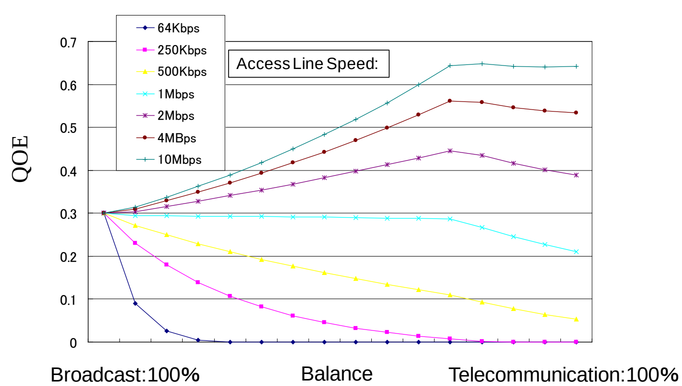

Fig. 7 With a server capacity coefficient of 0.7

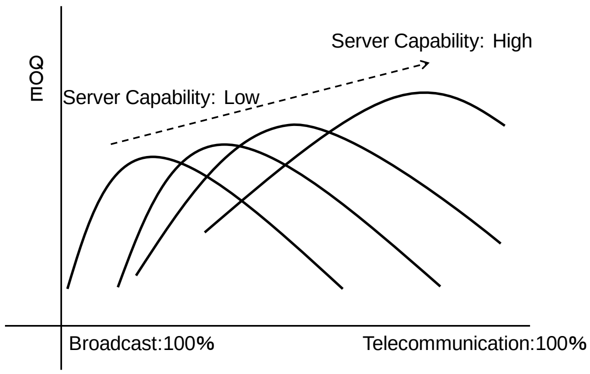

Fig. 8 QoE and server capability

Figures 5, 6, 7, and 8 can be interpreted as follows:

(a) When the access line speed is equivalent to or greater than that of the broadcast, an optimum solution for allocating the level of content to be output to maximize the QoE is derived as shown in Figs. 5, 6, and 7. Figure 5 shows that when server capabilities are low, the broadcast system provides the optimum solution. Conversely, in the case of high capabilities, as shown in Fig. 7, the communication network provides the optimum solution.

(b) As shown in Figs. 5, 6, and 7, as server capabilities increase, so too does the QoE peak value. This implies that investing in servers to increase their capabilities results in a higher rate of customer satisfaction as shown in Fig. 8. It also shows that there is an optimal balance for achieving peak values.

5 MPEG-21 DIA Multimedia Framework

MPEG-21 is a comprehensive framework that deals with multimedia. The various parts of MPEG-21 address many different requirements, allowing implementers of standards to design and implement a system or application that goes beyond simple multimedia content delivery in an interoperable way. The MPEG-21 standard provides the ability for users to carry out transactions of digital items (DIs). A DI is a structured digital object with a standard representation and metadata and can be thought of as a virtual structured digital container for media metadata and resources such as audio, video, text, and so on. Metadata is the related information for a DI that provides semantic support. Besides interoperability, digital content has to adapt to various transmission channels and terminal devices. DIA (digital item adaptation) can be achieved through various approaches including adaptation at the server, at the intermediate proxy, or at the terminal.

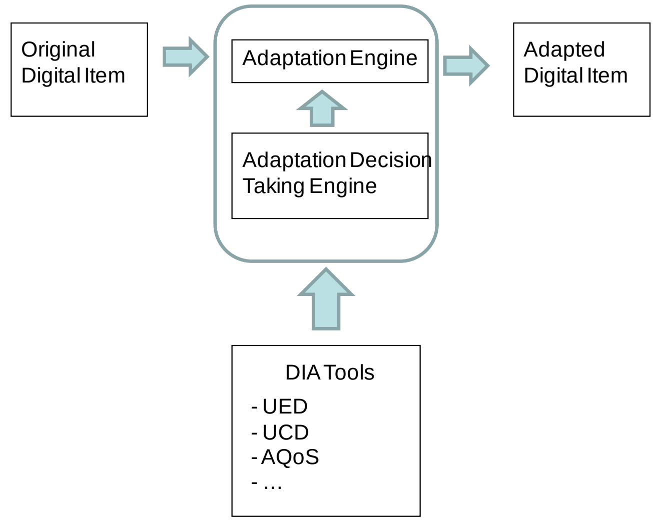

Fig. 9 Digital Item Adaptation (DIA) framework

In the MPEG-21 DIA Framework shown in Fig. 9, a strict separation is maintained between the actual adaptation engine and intelligence that determines what kind of adaptation is necessary to satisfy the constraints.

The inputs of the Adaptation Decision Taking Engine (ADTE) are as follows:

- Adaptation QoS (AQoS) or content related metadata: To achieve optimal settings under certain constraints imposed by terminals and/or the network for QoS management, AQoS is provided to assist the adaptation engine with decision making. The AQoS specifies the relationship among various constraints (e.g., bandwidth), feasible adaptation operations satisfying these constraints (e.g., frame rate reduction), and associated qualities.
- Usage Environment Description (UED): This includes a detailed description of the conditions under which media is consumed, the terminal description (e.g., display resolution) and network conditions, and the user information.
- Universal Constraints Description (UCD): A description of constraints to be applied to the adaptation process. It is desirable to constrain UEDs by explicitly providing the adaptation engine with a range of adaptation possibilities. The UCD provides a means for describing limitations and optimization constraints.

The output of the ADTE is a set of parameters that are provided to the adaptation engine (AE).

6 Media Decision Taking Engine

As discussed in Section 2, in a heterogeneous network environment, it is better to consider a hybrid network solution such as broadcasting, the Internet, wireless networks, and so on. In such an environment, an optimal content distribution architecture is required.

This section introduces the architecture for an efficient Media Decision Taking Engine (MDTE) embedded in the MPEG-21 DIA Multimedia Framework as shown in Fig. 10. Its task is to process DIA tools (AQoS, UCD, UED, etc.) to find the optimal distribution balance for each media for multimedia content delivery. The Smart Media Server in Fig. 10 corresponds to that shown in Fig. 1.

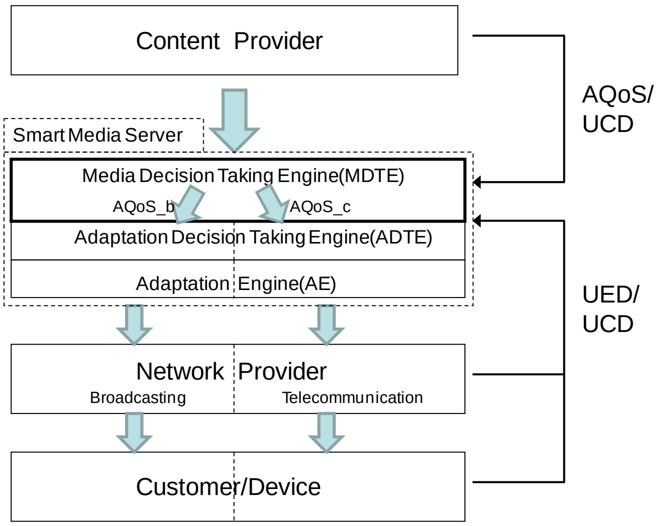

Fig. 10 Media Decision Taking Engine (MDTE) in DIA

Here, the operation of an MDTE based on the MPEG-21 DIA (as shown in Fig. 10) is described in more detail.

(1) The MDTE gathers and summarizes both the content provider side information and the user device/terminal side information. The content provider provides AQoS as well as UCD to the MDTE, while the network provider and customer (Device/Terminal) provide the UED/UCD.

(2) The MDTE processes the provided information, and extracts and summarizes regional environmental parameters (e.g., server capacity and average access network speed).

(3) The MDTE simulates QoE by using the above-mentioned parameters to ascertain the optimal distribution balance among the media (channels). As defined in Section 4, the QoE() function is used for simulation purposes. In this case, file size, server capacity and average access line speed are the main parameters in this simulation. These parameters are derived from a variety of inputs such as AQoS, UED, and UCD.

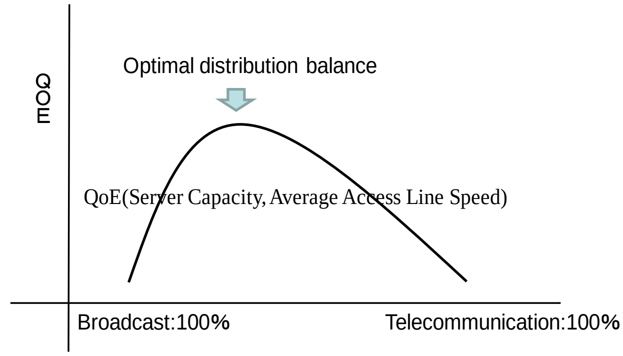

Fig. 11 Optimal distribution balance

(4) The MDTE decides the optimal distribution balance as shown in Fig. 11, and based on this delivers multimedia content on each of the channels. The file is divided accordingly between the channels. Furthermore, the AQoS, which contains content metadata, has to be regenerated for each channel, e.g., broadcast AQoS (AQoS_b) and communication AQoS (AQoS_c).

After the MDTE has made a decision, the multimedia content is divided into separate media, e.g., broadcast and network media. The divided data streams and metadata are then transferred to the Adaptation Decision Engine, and an adaptation strategy is constructed by this engine. The regenerated AQoS (AQoS_b and AQoS_c) is given to the AE and the actual adaptation (i.e., transcoding) is executed by the AE.

7 Conclusion

With the increase in consumption of multimedia content, users are becoming more eager to access multimedia content anytime, anywhere, and via any device. The vision of MPEG-21 is to define multimedia standards that enable transparent access to multimedia content across heterogeneous network environments.

In this paper, we presented a QoE model for a broadcast and telecommunication convergent environment and proposed the application of this QoE model to the MPEG-21 DIA Framework. We developed a QoE simulation model, and the results of the simulation show that an optimal distribution balance can be obtained through this simulation method. This paves the way for implementing a smart media server that can decide the optimal media usage balance for distributing multimedia content using multiple media (channels).

We also proposed setting up the MDTE within the MPEG-21 DIA Framework, and presented a design for the MDTE and an explanation of how it would work. As such, by making use of the proposed MDTE, an optimal multimedia content adaptation/distribution system could be made available even in diverse network environments. Future work includes designing the MDTE in more detail and prototyping the MDTE, ADTE, and AE to verify the appropriateness of the concept and design.

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Two Models for Inferring Network Structure from Cascades

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ABSTRACT

In many real-world scenarios, the underlying network over which the diffusions and propagations spread is unobserved, i.e. the edges of the network are invisible. In such cases, we can only infer the network structure from underlying observations. The goal of this paper is to find a model that generates realistic cascades with observed data, so that it can help us with link prediction and outlier detection. For this purpose, we investigate two cascade models. The first model is a naive two-class cascades that includes one class of positive (infected) nodes and one class of negative (uninfected) nodes. In this model, we use the sparse logistic regression method to infer network edges. In the second model, we discard all negative training nodes and treat the whole network as a single class. In this model, we use the one-class Support Vector Machines to predict underlying edges. Experiments show that even if we discarded all negative training instances, we can still infer network edges accurately.

KEYWORDS

Social Network Structure, Information Cascades, Networks of Diffusion

1 INTRODUCTION

Processes that form cascades in a social network have been studied in a number of domains [12]. Examples include the spread of infectious diseases [6], the spread of new ideas and technologies [14], marketing [5], technology transfers [3, 4], computer virus transmission [2], and power systems [17]. Edges are very fundamental parts of a network to investigate the network structure. However, in many real world situations, the edges of a network are invisible and we can observe partial information about the edges. This information can help us infer the latent edges in the network. As a fundamental process in social network, cascade is a useful information source to infer edges. In general, a cascade describes how a contagion spreads or propagates across a network [16]. Research on cascade will enable us to gain insights about how people influence each other. However, sometimes it is very difficult to track a cascade because of its complexity. Even worse, we may not know the structure of the network over which the cascade happens. In most cases, the available information is very limited. We may only know when the cascade propagates to a given node. For instance, in study of virus contagion, tracking the time when a person was infected may be much easier than tracking the source of the infection. In viral marketing, we can easily record the time when a customer made a purchase, but it is hard to know who had influenced him to make the decision. Therefore, the challenging task is to infer the latent network structure from the limited given information. Specially, we may only know the nodes, and sets of time data from a few cascades, but not the edges in the network. Our goal is to infer the hidden edges based on these limited information.

There are a few recent works focusing on inferring networks from cascades. Leskovec et al. [10] monitor one of the largest available collections of blog information and find that almost any metric they examined, such as size of cascades, size of blogs, and in- and out-degrees, follows a power law distribution.

Gomez-Rodriguez, Leskovec and Krause [11] convert the edge inference problem to combinatorial optimization problem [7]. They find that choosing the best set of *k* edges maximizing the likelihood of the data is NP-hard. They then develop NETINF, a greedy algorithm for inferring a near-optimal set of *k* directed edges. NETINF is able to scale to large real data sets by introducing some speed-up heuristics based on the submodularity of their evaluation function.

Myers and Leskovec [9] formulate the edge inference problem to an equivalent convex problem. This guarantees the optimality of the solution. In addition, they observe that social networks are sparse in a sense that on average nodes are connected to a constant number rather than a constant fraction of other nodes in the network. To enforce a sparse solution, they add the L_1 penalty regulation to the evaluation function.

Sadikov, Medina, Leskovec, and Garciamolina [13] address the problem of missing data in information cascades [8]. They propose a *k*-tree model of cascade (*k* is the number of parents for a node) to predict the missing data such as the size or depth of the tree. By given a fraction of a complete cascade C, they present a selection algorithm to select a *proxy* *k*-tree that best approximates C.

Yang and Leskovec [15] develop a Linear Influence Model to infer the influence of a node from cascades. Different from predicting which node will influence which other nodes in the network, they focus on modeling the global influence of a node on the rate of diffusion through the (implicit) network. For each node, they estimate an influence function of how many subsequent infections can be attributed to the influence of that node over time. Therefore the number of newly infected nodes is a function of which other nodes got infected in the past.

In this paper, we model the edge inference problem to the traditional classification problem. We investigate two models for inferring edges from cascades. The first model is a two-class cascades that includes one class of positive (infected) nodes and one class of negative (uninfected) nodes. Here, we use the sparse logistic regression method to infer network edges. The second model discards all negative training nodes and treats the whole network as a single class. We use the one-class Support Vector Machines to predict underlying edges. Experiments show that the second model (the single class model) can still infer network edges accurately.

The paper is structured as follows: Section 2 formulates our problem and defines our assumptions. Section 3 presents the two models in theoretic approach. Section 4 presents and discusses the analysis results. Section 5 concludes the paper.

2 PROBLEM FORMULATION

Given a graph of *n* nodes $G = (V, E)$. The edge set E is unknown. Also given a set of cascades, each cascade is represented by a vector $\{\tau_1, \ldots, \tau_n\}$ where τ_i indicates that node *i* is infected at time τ_i. If $\tau_i = \infty$, node *i* is not infected in the cascade. The problem is to infer the edges E from these cascades. Our idea is based on two assumptions: 1) earlier infected nodes may have edges to nodes infected later, and 2) infected nodes are very unlikely to have edges to uninfected nodes.

3 MODELS

3.1 Model A

Before proposing the model, let's present our assumptions first. [9] assumes the time period each infected node takes to transmit the disease follows a random distribution. However, we notice there is some inconsistency between the model assumption and the objective function. Inspired by the maximum likelihood estimation in [9], we propose another assumption that an uninfected person *i* is not always susceptible to infection, and will only be infected when he shows up in the network. For instance, for cascades in blog sphere, one will not be influenced by other persons' blog-posts if he constantly stays off-line. The already infected person, say *j*, will decide whether or not to infect person *i* with probability A_{ij}. If *j* chooses to infect *i*, he will wait until person *i* shows up and then try to infect him. The probability of *i* being infected when he shows up can be defined in various ways to finish model assumption.

There are two points about our assumption:

1) Before getting infected at time τ_i, *i* can also show up at $\tau_i` < \tau_i$ (e.g., access the Internet in the blog case). It is not clear whether he was infected at that time. Either infected or not can be possible. It may be the case that he was infected, but we did not observe his infection at that time. Therefore to avoid ambiguity, we do not take a person's history before his infection into account.
2) For node *i*, which is never infected, we assume that he will not get infected whenever he shows up. To satisfy this, other infected node must not choose to infect node *i*, otherwise there will be a probability of *i* being infected when shows up. Hence the probability of *i* never infected is

$$\prod_{j\ infected}(1 - A_{ij}) \qquad (1)$$

Moreover, instead of assuming infected people transmit the virus to a susceptible one independently, we assume they impose a collective influence. This assumption is very common in real world. For instance, when one wants to purchase a product online, he may first check its online reviews and then make the decision. In other words, the other people who have already bought the product influence the person collectively through reviews. We still maintain the assumption that a susceptible person will only get infected when he shows up. Different from the above model, we do not keep the restriction that A_{ij} ranges between 0 and 1, but assume it can take any real value. Large positive value for A_{ij} means that the person *j* is very likely to infect person *i*, while negative value indicates that the person *j* may prevent the person *i* from being infected (can be interpreted as *i* distrusts *j* very much). When *i* shows up at time τ_i, the influence of *j* on *i* is $w(\tau_i - \tau_j)A_{ij}$, where $w(t)$ is the distribution defined in [9]. The collective influence on person *i* at τ_i is

$$\sum_{\tau_j<\tau_i}(w(\tau_i-\tau_j)A_{ij}) \tag{2}$$

Finally, we assume that the collective influence on person *i* who has never been infected is

$$\prod_{j\ infected} A_{i`j} \tag{3}$$

as explained in the above section.

We now try to understand the above definition in another way. Each node *i* is associated with a set of cascades. In some cascades, node *i* was infected at τ_i, while in others *i* never got infected. For a cascade in which an node *i* infected at τ_i, the vector contains non-zero entry $w(\tau_i - \tau_j)$ for every *j* such that $\tau_j < \tau_i$. For a cascade in which node *i* never infected, the vector contains entry 1 for every already infected node. The i^{th} row of A, denoted A_i, is the weight vector characterizing how much node *i* trust other nodes. The probability of node *i* infected at τ_i or never infected is a function of the dot product (•) between feature and weight vector. Then we reduce the cascade problem to a classification problem: we have some positive and negative instances, which are respectively different cascades in which node *i* got infected or not. For each instance we have a feature vector. And we would like to infer the weight vector from the provided training data. Then we may adopt some efficient discriminate model such as logistic regression, SVM (Support Vector Machines). If we want to guarantee sparsity of $|A|$, we may use sparse logistic regression. One possible advantage of this model over [9] is that we do not have an inconvenient product of terms like

$$1-\prod_{\tau_j<\tau_i}(1-w(\tau_i-\tau_j)A_{ij}) \tag{4}$$

To summarize, we now mathematically define the above idea. For a certain node *i*, we have a set of n_i instances $\{x^{(j)}, y^{(j)}\}_{j=1}^{n_i}$. Here $y^{(j)} \in \{0, 1\}$ represents whether node *i* is infected in cascade *j*. And $x^{(j)}$ is a vector $(x_0^j, \ldots, x_k^j, \ldots, x_n^j)$ with entry x_k^j to be

$$x_k^j = \begin{cases} w(\tau_i^j - \tau_k^j) & \tau_k^j < \tau_i^j \\ 0 & else \end{cases} \tag{5}$$

for $y_i = 1$, and

$$x_k^j = \begin{cases} 1 & j\ is\ infected \\ 0 & else \end{cases} \tag{6}$$

for $y_i = 0$. We would like to learn from the training set our transmission weight $A_i = (A_{1i}, \ldots, A_{ni})$ and a hypothesis h_{Ai}: x → y. We use logistic function to

$$h_A(x) = \frac{1}{1+e^{-(A_i)^T x}} \tag{7}$$

And assume that

$$p(y = 1 \mid x, A_i) = h_A(x)$$
$$p(y = 0 \mid x, A_i) = 1 - h_A(x) \tag{8}$$

The objective function we would like to maximize is

$$F = \sum_{i=1}^{n_i} y^i \log\left(h_A(x^i)\right) + (1-y^i)\log(1-h_A(x^i))) + \gamma||A_i||_1 \tag{9}$$

Finally we use the large scale sparse logistic regression package [1] to solve the above objective function.

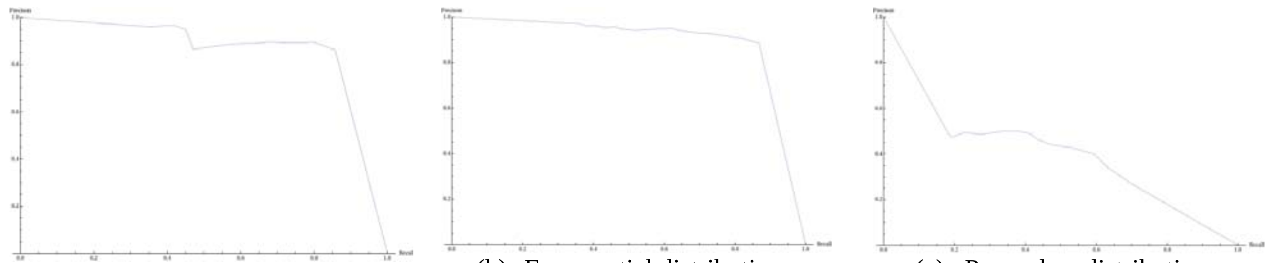
3.2 Model B

In the above section, we cast the network inference problem to be a binary classification problem and treat the cascades containing node *i* to be positive instances and those not containing *i* to be negative instances. We implicitly assume that the cascades not containing *i* indicate that there are no edges between *i* and previously infected edges. However, we claim that second assumption is presumptuous. There are lots of examples against this assumption. We take hashtag propagation as an example. Here each hashtag is a cascade. There are many cascades that one user does not adopt a hashtag and some of the users he follows adopt that hashtag. These cascades are not evidences that there are no edges between node *i* and already infected nodes. Thus these negative instances are not truly negative. Treating them to be negative maybe harmful for inferring edges. On the other side, those positive instances (cascades containing node *i*) do indicate that there may be some edges between node *i* and some previously infected nodes. Those we are facing a problem of only having confident positive instances. It is lucky that there have been many works dealing with problems which only contain only positive instances. Such problems are called one-class classification problem. By definition, one class classification aims distinguish one class of objects from all other possible objects, by learning from a training set containing only the objects of that class. This is exactly the problem we want to solve. It also worth mentions that there are many provided easy-to-use software packages for one-class classification.

The other advantage of modeling network inference to be a one-class classification problem is that training the model will be more efficient. Indeed, in many situations, the proportion of positive instances in which node *i* is infected is very small. Thus our training data size will be significantly reduced if we only consider the positive instances.

The one-class classification problem can be formulated as follows:

$$\text{Min } \frac{1}{2}|w|^2 + \frac{1}{\nu l}\sum_{i=1}^{n} \epsilon_i - \rho$$
$$\text{subject to } w \cdot x_i \geq \rho - \epsilon_i \tag{10}$$



(a) Weillbull distribution (b) Exponential distribution (c) Power law distribution

Fig. 1: PR curves for Erdos Random Graph Different Transmission Time Configurations

4 EXPERIMENT

4.1 Datasets

As in [11, 9], there are two categories of datasets we are considering using: synthetic datasets and real world datasets. The synthetic dataset is mainly used for debugging programs. For this milestone, we only use synthetic datasets. The dataset we used is similar to the one in [9]. That is, we generate an Erdos random graph. For each generated edge, we randomly generate the transmission weight A_{ij}. Then we try to generate a set of cascades. For each cascade, we randomly generate a start node and take it as the newly infected node. Then we iteratively simulate the cascading process. Pick the node *i* with the earliest infection time from the set of newly infected node. For the picked node, decide whether to infect the other uninfected node based on the transmission weight A_{ij}. If node *i* choose to infect *j*, then randomly generate a time t_{ij} according to a specific distribution. If node *j* is not in the newly infected node set, add it to the set. If it is already in the set and t_{ij} is smaller than its infection time, update the infection time. Finally we remove node *i* from the newly infected node. In this way we can generate a cascade. As in [9], we also generate enough cascades for an informational training dataset.

4.2 Model A

We test whether a latent link can be inferred correctly. We compare the links inferred from the cascades with the links in real network and get the precision and recall of the model. For different choice of the sparsity parameter, precision is different. Thus we can plot a precision-recall curve. Note that we have connected the first point on the curve to (0; 1) and the last point to (1, 0).

4.2.1 Results on Synthetical Datasets

We tested our algorithm on synthetic datasets. As in [9], we generate an Erdos random graph consisting of 512 nodes and 1024 edges. In the graph, the transmission probability A_{ij} is sampled between 0.05 and 1.

For the time generation function *w(t)*, we also tried three probability distributions detailed as follows:

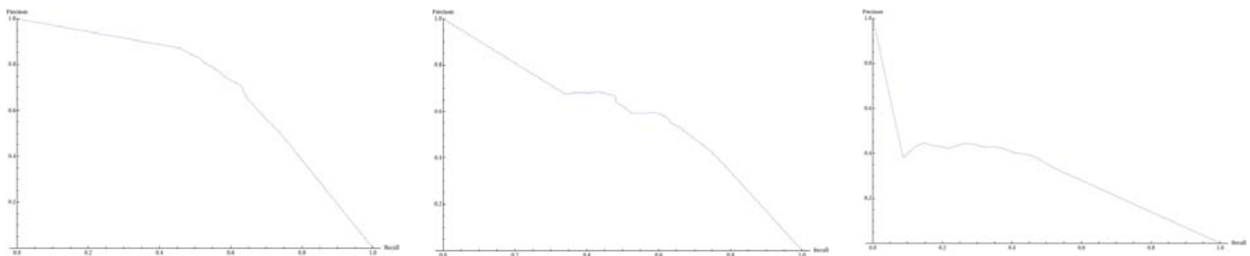
- Power law distribution $(\alpha-1)t^{-\alpha}$, where $\alpha = 9.5$.
- Exponential distribution $\frac{1}{\alpha}e^{-\frac{1}{\alpha}t}$, where $\alpha = 9.5$.
- Weillbull distribution $\frac{k}{\alpha}(\frac{x}{\alpha})^{k-1}e^{(-\frac{x}{\alpha})^k}$, where $\alpha = 9.5$ and $k = 2.3$.

Each cascade is generated as specified in 5.1. We generate enough cascades such that at least 95% edges are used to transmit a disease.

We show the experimental results in Figure 1. We plot three precision-recall curves for the three generated networks. Each point on the curve corresponds the model trained by setting the sparsity parameter to a certain value. Different points on the curve reflect models with different sparsity parameters. When sparsity parameter decreases, the recall increases and the precision increases.

Our model achieves very good performance comparable with, and even better than that obtained in [9]. However, it is quite weird that our model's performance is bad on the graph whose remission time is generated from a power law distribution. The reason is probably related to that our value of α is not the same as that set in [9]. We will try to find the reason in future work.

Apart from the Erdos random graph, we also test our model on another random graph, which is generated from the preferential attachment mechanism as taught in class. We report our results in Figure 2. As in Erdos random graph, we also tried three different probability density functions for sampling transmission time.



(c) Weillbull distribution (d) Exponential distribution (c) Power law distribution

Fig. 2: PR curves for Preferential Attachment Graph of Different Transmission Time

Fig. 2: PR curves for Preferential Attachment Graph of Different Transmission Time Configurations

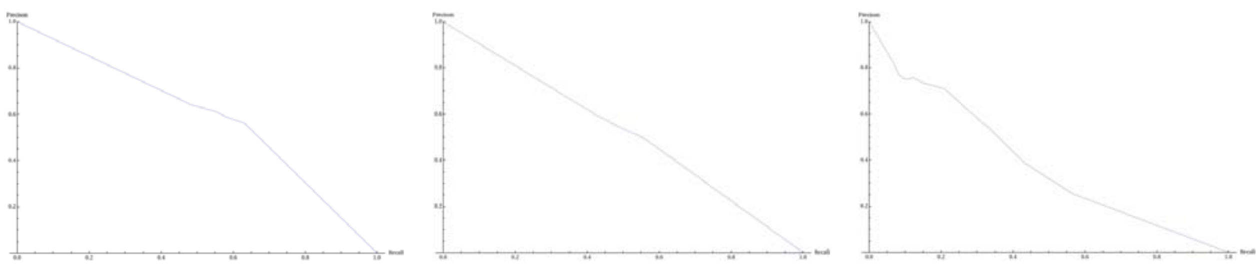
4.2.2 Results on Real World Datasets

Apart from synthetic dataset, we also tested our model on two real world datasets, specified as below:

- An email network in a research institution consisted of 437 nodes and 2805 edges.
- A collaboration network. We extract the largest component of the network. The number of nodes and edges are 379 and 608 respectively.

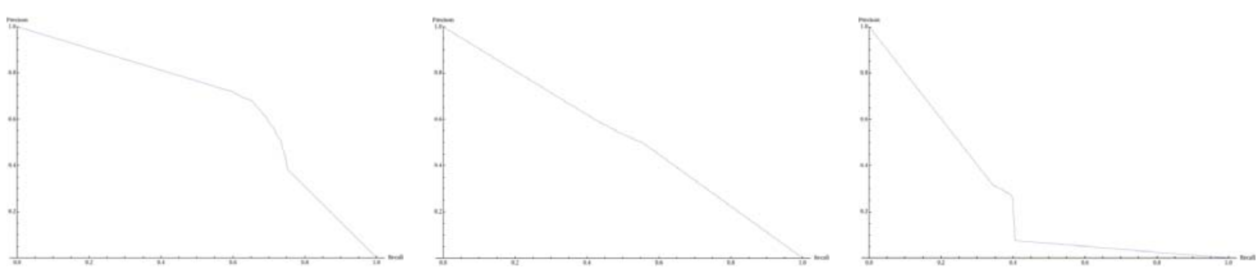
For the collaboration network, we sampled the edge transmission probability A_{ij} uniformly. For the email network, we set $A_{ij} = 1 - (1 - \phi)(1 - \epsilon)^{m_{ij}}$, where ϕ = 0:05, ϵ = 0:0001 and m_{ij} is the number of emails from i to j. This parameter set was suggested in [9]. For each network, we also adopted three different probability density functions for transmission time as in synthetic datasets. The parameters for those probability density functions are:

- Power law distribution $\alpha = 2.5$.
- Exponential distribution $\mu = 9.5$.
- Weillbull distribution $\alpha = 3.6$ and $k = 2.5$.



(a) Weillbull distribution (b) Exponential distribution (c) Power law distribution

Fig. 3: PR Curves for Email Network of Different Transmission Time Configurations



(a) Weillbull distribution (b) Exponential distribution (c) Power law distribution

Fig. 4: PR Curves for Collaboration Network of Different Transmission Time

We reported our results in Figure 3 and Figure 4. Our model can also handle edge prediction for real world datasets. It seems that that our model performance is worse than that in [9]. However, we want to point out that our model has large improvement space because we do not use enough training data. Indeed, cascades we sampled only covered 90% edges, while those in [9] covered 99% edges. The reason that we did not sample as many cascades as in [9] is that in order to sample cascades to cover 95% edges, we need to sample much more cascades since for some uncovered edges, the transmission probability is very small. Since each cascade will be one additional instance for each node. Our laptop memory cannot hold that many instances. However, sampling more cascades for getting more training instances can improve the performance significantly. We tried sampling cascades to cover 85% edges and 90% edges and tested each setting's performance. And the PR curve in the later setting lifted by 15 points. Thus we expect our model to be much better than its current state if our laptop memory can hold enough training instances.

4.3 Model B

In this section, we conducted experiments to test our model B. The data were generated as in the above experiments. However, we discarded all negative instances and only kept those positive instances for each node. We used one-class SVM component in libSVM to infer the edge weight. The libSVM tool does not provide the functionality of learning a sparse weight vector. Therefore we at this stage cannot get a PR curve with respect to the sparsity parameter as in the experiments in the above section. Thus in this experiment we evaluated our experiment from another perspective: we want to check whether the weights inferred by our model satisfy that the weights of the edges are bigger than the weights of the non-edges. For instance, consider a graph with n nodes. We now want to investigate what nodes have links to node s. Without loss of generality, we suppose that nodes $i_1^s, i_{21}^s, \ldots, i_m^s$ have links to node s. Then we rank all the nodes by our inferred weight vector w_s. Suppose the top ranked m nodes are $j_1^s, j_2^s, \ldots, j_m^s$. Then we count how many of our top ranked nodes have edges to node i, or we calculate the cardinality of the set $\{i_1^s, i_{21}^s, \ldots, i_m^s\} \cap \{j_1^s, j_2^s, \ldots, j_m^s\}$. Suppose that:

$$tt_s = \# \{i_1^s, i_{21}^s, \ldots, i_m^s\} \cap \{j_1^s, j_2^s, \ldots, j_m^s\},$$

$$tf_s = \{i_1^s, i_{21}^s, \ldots, i_m^s\} - \{j_1^s, j_2^s, \ldots, j_m^s\},$$

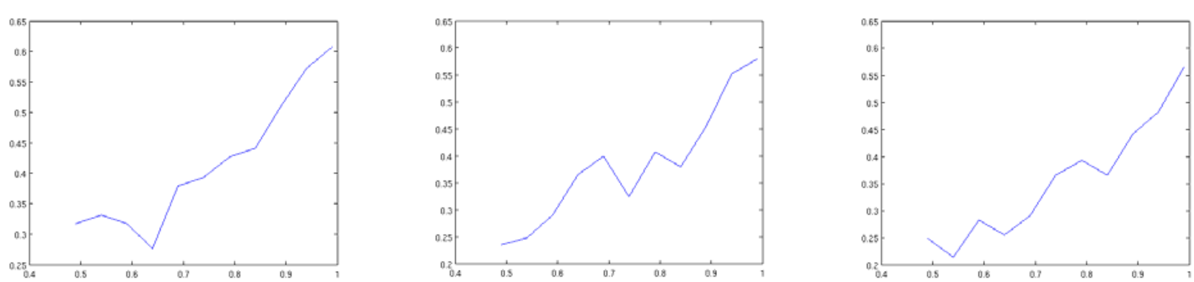
$$ft_s = \{j_1^s, j_2^s, \ldots, j_m^s\} - \{i_1^s, i_{21}^s, \ldots, i_m^s\}. \quad (11)$$

We use the following precision and recall to evaluate our model

$$P = \frac{tt_s}{tt_{s+ft_s}}$$

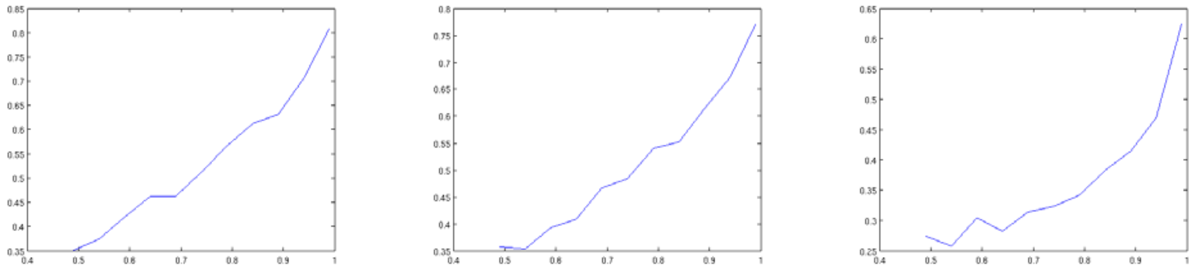
$$R = \frac{tt_s}{tt_{s+tf_s}} \quad (12)$$

Notice that in this case, $tf_s = ft_s$, leading to $P = R$. We only use precision to evaluate our model. The goal of the experiments is to show that when we have observed enough cascades, the edges in the network can also be inferred even if we discard all the negative instances. We investigated how the performance of our model change with respect to edge cover ratio. Here edge cover ratio means the proportion of edges covered by the generated cascades. The larger the cover ratio, the more cascades we generated. We show in Figure 5 and 6 that the precision-cover ratio curve for two networks for different probability density function of transmission time. From the Figures, we can see that if the edge cover ratio is high enough, or number of cascades is large, then we may get satisfactory performance even if we do not consider negative instances.



(a) Weillbull distribution (b) Exponential distribution (c) Power law distribution

Fig. 5: Precision-Cover Ratio of Preferential Attachment Network Configurations



(a) Weillbull distribution (b) Exponential distribution (c) Power law distribution

Fig. 6: Precision-Cover Ratio of Email Network Configurations

5 CONCLUSION

We investigated two models for inferring network edges from cascades. The first model is based on sparse logistic regression. It is easy to understand and both efficient and effective. Our second model is based on one-class SVM. Experiments showed that even if we discarded all negative training instances, we can also infer network edges accurately.

In the future, we will test our model on larger scale data. One of the interesting data is the twitter dataset. Indeed, we have explored the twitter dataset. We once wanted to use hashtag propagation to infer the who following whom links. However, after a careful data analysis, we found that the hashtag propagation may not provide very useful information for inferring who following whom links. If possible, we may try some other evaluations. One possible way is to test whether our inferred latent network structure conveys information that is consistent with our intuition. For instance, from the inferred weight between pairs of nodes, we may discover several communities. Then we may check whether different communities have different interests (e.g., one community is interested in sports, while the other is interested in entertainment).

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Service Area Optimization For Application Virtualization Using UMTS Mobility Model

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Abstract— *In this paper, we first introduce recent developments of the mobile computing devices and virtualization technologies available to deploy cross-platform application software. In our previous work, we proposed a distributed server arrangement and the corresponding hand-off protocol to provide smoother and more responsive user experience for application virtualization on mobile devices. Based on out previous work, we present a series of simulation results that establish quantitative relations between the performance improvement or impact and the infrastructure related parameters. The simulations are based on the proposed modified UMTS mobility model, which allows them to run for an indefinite period of time without presuming any boundary condition.*

Keywords: telecommunication and wireless networks, computer networks, information technology, UMTS mobility model.

1. Introduction

As computing devices are getting smaller, lighter, and more ergonomic and wireless communication technology becomes more and more widely available, the recent development of computing technologies is more on mobilization than driving computational capability into redline. Therefore, we can expect more and more application software deployed on mobile computing devices for years in response to the trend.

Deploying application software on mobile computing devices can be a challenge for several reasons. First of all, various mobile operating systems are still competing with each other and none is expected to dominate the market in 2 or 3 years. Releasing one application software product for all mobile operating systems on market requires extra cost and effort to work on different SDKs and maintain identical, or at least similar, user experience on different ports.

Even if an application software developer decides to focus on one single mobile operating system regime, platform fragmentation may still be experienced. Although application-platform compatibility issues exist on both general purpose computers and mobile computing devices, it is far more difficult to be worked around on mobile computing devices. Unlike operating systems for general purpose computers, mobile operating systems are highly customized per product, packed into ROM images, and secured against user access. Therefore, ordinary end-users can not upgrade or patch their mobile operating systems to address application-platform compatibility issues by themselves, as they do on PC for years. Instead, they have to wait until the ISPs or device vendors release operating system update or patch packages *specific* for their devices. Mobile application software developers, therefore, have become more responsible on making their products compatible to everyone.

Fortunately, virtualization technologies can work around the difficulties of deploying mobile application software on various mobile platforms. Over the years, application virtualization technologies have fallen into two major paradigms: one is creating a compatible runtime platform, i.e., virtual machine, and publish well managed application software packages to each client's device [1][2], and the other is executing application software on a well managed server while each client's device only deals with user inputs, such as keystrokes, and outputs, such as display updates from the server [3][4][5]. We generally refer to the later paradigm as the *browser-based approach* since web browsers provide a very ideal framework for it.

Although being technically sensible, deploying a virtual machine running on top of a mobile operating system to execute downloaded common codes facilitates the distribution of apps which circumvents the official marketplace and generally considered a violation of the “Non-Compete” policy [6][7]. Therefore, the browser-based approach becomes the remained legitimate way to provide application virtualization services on mobile computing devices, unless marketplace operators enforce the “Non-Compete” policy against interactive web contents.

The conventional solution of web-based application virtualization is setting up a server or a group of servers at a colocation center provided by an Internet service provider (ISP) and providing the application virtualization service through established Internet infrastructure. Though this configuration is very simple and straightforward, the long response latency could significantly prevent the clients from enjoying the service since every input must travel through a series of routers and bridges to the colocation center and the corresponding update has to traverse through the nodes backward. Each node along the route induces processing delay, queuing delay, and transmission delay, and each link comprises the route induces propagation delay. Generally speaking, network delay is highly related to the geographical distance between two end points given similar network infrastructure technologies.

To alleviate this issue, we have proposed an alternative configuration that geographically partitions the service area into multiple smaller service areas and each one has a smaller scale data center to provide the service locally in our previous work [8]. The proposed configurations should significantly reduce propagation delay since each server is geographically closer to its user. The proposed configuration, however, has to handle hand-off cases, i.e., mobile stations in use moving from one service area to another. Therefore, we also propose a hand-off protocol offering seamless user experience.

The proposed configuration comes with a price, such as inducing longer response latency during hand-off periods in addition to higher overall system complexity. We used an analytical and simplified approach to evaluate the performance as a result of infrastructure arrangement and application software's properties in our previous work [8]. In this paper, we set up a simulation environment referring to one of the UMTS mobility models which reflects the reality and use the empirical approach to establish the correlations between the performance and the size of each local service area and the capabilities of the network infrastructure.

The rest of this paper is organized as follows. In Section 2, we describe the proposed configuration aim to improve the user experience of application virtualization services. In Section 3, we propose a VM-level hand-off protocol to handle the additional information exchange brought by the proposed server configuration. We specify our experiment design, settings, and cost metrics in Section 4. Then the simulation results given different parameter adjustments are presented in Section 5. Finally, we conclude our work and outline some future works we expect to do in Section 6.

2. Proposed Configuration

Running application software on a remote server while creating an illusion that the client has full control of the software in hand is conceptually similar to the usage model of time-sharing mainframe computers in the 1960s [9]. Although the communication bandwidth between terminals and mainframe servers at that time was very low by modern standards, it didn't affect the user experience thanks to the text-only display and short traverse distance. However, in recent application virtualization technologies which follow the same concept, such as Virtual Desktop Infrastructure (VDI) proposed by VMWare [10], much more complex and bloated content must be exchanged over much longer distances between clients and servers than their predecessors, especially for mobile users.

An infrastructure ready to offer mobile users application virtualization services includes base stations (BSs) covering the whole service area, a core network connecting base stations and servers together, and a server hosting the services. A command sent by a mobile station (MS) has to travel over the wireless channel to the BS, go through the backhaul network to the server, and then make some changes on the server. Should any update corresponding to the command be sent to the MS, the information has to travel all the way backward. In order to reduce the network delay generated by long transmission distances among the backhaul network, we geographically deploy multiple servers among a wide area to serve their nearby MSs in the proposed configuration, instead of setting up a group of servers located at one data center serving all MSs.

In the proposed configuration, each server connects to several nearby BSs which form a *local service group* (LSG). The area covered by the BSs of the same LSG is defined as the *local service area* (LSA). Every BS should belong to one LSG in order to provide the service all over the wireless network's coverage area. When a user demands a virtual application program, the server of the LSG, based on VDI [10] paradigm, starts a virtual machine (VM) dedicated to the user and launches the application software on top of it. The MS only handles inputs and outputs that interact with the VM at the server.

As long as the MS stays in the same LSA, the user can enjoy using application software with low response latency. If the MS moves from the original LSA to a nearby one, a hand-off at the VM level, which transfers the runtime environment to the server of the next LSG, is triggered. The detail of the hand-off protocol will be proposed in the next section.

3. Hand-off Protocol

The purpose of the proposed hand-off protocol is to transfer minimum information required to recreate the runtime environment on a remote server, i.e., the *snapshot*, without interrupting the service. No matter how small the snapshot is, it still takes a period of time before the next server receives the complete snapshot and is ready to take over the service. In order to provide a seamless user experience during this period, the next server has to record all inputs from the MS, relay all inputs to the previous server, and relay all output from the previous server to the MS, until the runtime environment resumes locally. The proposed hand-off protocol is described as below:

1) When an MS moves from Server A's to Server B's LSA and sends an input command, Server B notices a newcomer within its LSA.
2) Server B broadcasts the newcomer's identification to all geographically nearby servers.
3) Server A, which hosts the MS's runtime environment, i.e., its VM server, responds Server B's inquiry. Now Server B knows the newcomer's VM server is Server A.
4) Server B records and relays the user's input commands to Server A, signals Server A to transfer the runtime environment, and relays display updates from Server A to the newcomer.
5) Once Server A is signaled to transfer the runtime environment, it takes a snapshot.
6) Besides continually responding to the input commands relayed from Server B as the MS is still in its LSA, Server A also sends the snapshot to Server B in the background.
7) Once Server B receives the complete snapshot and recreates the runtime environment from the snapshot and base data, it internally feeds the input queue, which was recorded during the transition period, to the runtime

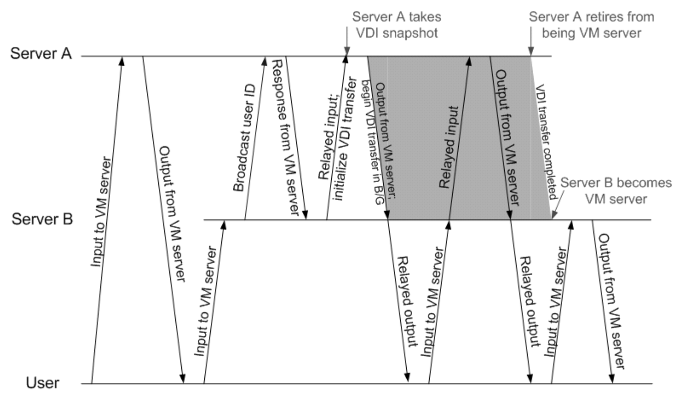

Fig. 1: Protocol timeline for an MS moving from Server A to Server B.

environment. Therefore, the runtime environment state on Server B is synchronous with that on Server A after the snapshot was transferred.

8) Server A completely stops serving the MS, the MS's VM server is now Server B instead.

The timeline of the proposed hand-off protocol is illustrated in Fig. 1.

If the MS turned around and reentered Server A's LSA before the hand-off was completed, Server A can preempt the snapshot transmission and resume serving the MS as if the hand-off never happened. Since Server B relays all inputs to Server A while the MS is absent from Server A's LSA, aborting the hand-off procedure would not generate any glitch noticed by the user. This hand-off abortion mechanism can prevent unnecessary data transmission from moving VM servers back and forth if an MS were moving around the edge of an LSA. However, we don't apply the hand-off abortion in our simulations since the scenario has very low occurrence rate in the mobility model we use.

On the other hand, if the MS moves to Server C's LSA before the hand-off was completed, Server C initializes another hand-off procedure with Server B. In addition to the snapshot, Server B has to transfer the input record before Server C joins the hand-off chain. We allow pipelining transmission to reduce hand-off periods and shorten subsequent hand-off chains in this scenario.

4. Performance Evaluation

The proposed service architecture is designed to reduce interaction latency and thus provide more responsive user experience on remote controlled application virtualization services. However, due to the involvement of the hand-off protocol, the performance of the proposed service architecture depends highly on the probability of hand-offs, the geographical deployment of the BSs, and the configuration and capability of the backhaul network. The former two factors can be modeled by the test environments of existing communication systems,

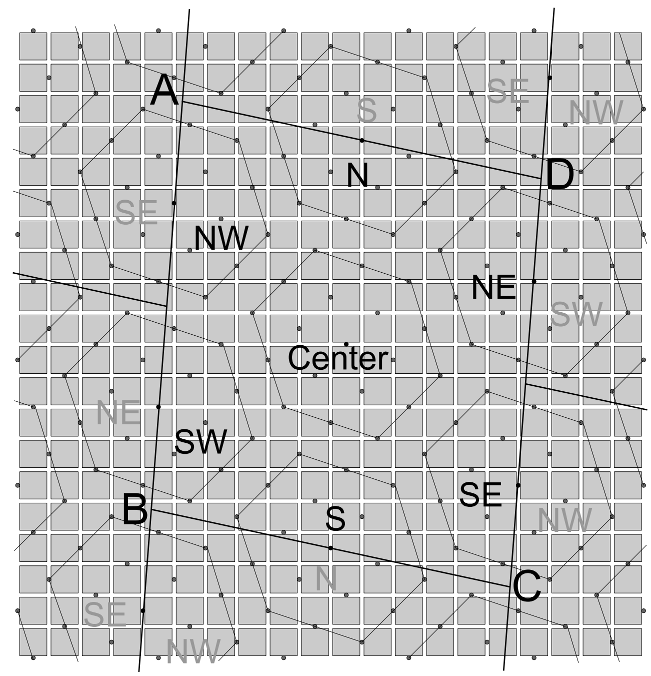

Fig. 2: UMTS outdoor to Indoor and Pedestrian test environment and LSA arrangement.

such as well published UMTS benchmark [11]. On the other hand, we can only make some assumptions on the configuration and capability of the backhaul network based on reasonable technical and cost considerations.

4.1 UMTS Urban Mobility Model

The UMTS document [11] provided three different test environments, which are Indoor Office, Outdoor to Indoor and Pedestrian, and Vehicular ones, for technology selection and evaluation. Although Jugl and Boche [12] have extended the mobility model to improve the reality, the original UMTS models, however, still provide a fair reference for mobility related performance evaluation. In this paper, we focus on deploying application virtualization services on the Outdoor to Indoor and Pedestrian test environment specified in the UMTS document.

As shown in Fig. 2, The UMTS Outdoor to Indoor and Pedestrian test environment is basically a Manhattan-like street structure where MSs move along 30 meters wide streets and are only allowed to change directions with half chance at the intersections, which are 200 meters apart. Each MS's moving speed can be updated every 5 meters with 20% chance, and the new speed is generated by a truncated Gaussian distribution whose mean equals 3 km/h, standard deviation equals 0.3 km/h, and minimum speed equals 0 km/h. All MSs are initially uniformly distributed on the Manhattan-like streets.

The UMTS document, however, does not explicitly specify where an MS turns within the intersection area. Therefore we make a reasonable assumption to overcome the ambiguity. If an MS is supposed to turn in an intersection, it has six points, which are 5 meters apart along the crosswalk, to change its direction before reaching the other side. We assume an MS picks one out of the six points with equal chances as its turning point, which keeps MSs uniformly distributed on the streets rather than be concentrated on a certain part of the streets over time.

The BSs in the UMTS Outdoor to Indoor and Pedestrian test environment are located at the dark grey dots in Fig. 2. Although the placement of the BSs is not optimal, it is not far from that. Considering an actual city could be preoccupied by tall private buildings on each block, deploying BSs along the streets makes sense both technically and politically.

One of the shortcomings of the UMTS mobility model is the bounded test area which generates ambiguities on setting boundary conditions. We consequently add some special traffic rules, known as *portals*, to eliminate the boundary discontinuities and allow the interaction among LSAs to be simulated and observed for indefinite period of time. These portals will be described in the next section.

4.2 Möbius City

What interests us is the geographical relation between the service facilities and the MSs' moving space. Once we group the BSs in Fig. 2 to form hexagon-shaped LSAs that optimize in both coverage and average transmission distance by deploying servers at the centers, we can find a regular repetitive pattern of streets and service groups, we can find a regular repetitive pattern of streets and service groups, which depends on N, the number of the BSs per LSA's edge. If we align the origin to a BS, the parallelogram ABCD surrounded by four straight lines, which are:

1) $(3N-1)x + (9N+5)y = 920(6N^2 + 6N + 2)$ on the north,
2) $(3N-1)x + (9N+5)y = -920(6N^2 + 6N + 2)$ on the south,
3) $(5N+3)x - (N-1)y = -920(3N^2 + 3N + 1)$ on the west,
4) and $(5N+3)x - (N-1)y = 920(3N^2 + 3N + 1)$ on the east,

can be regarded as the element of the repetitive pattern and represent sufficient geographical information we need. We can, therefore, crop out parallelogram ABCD in Fig. 2 as our new test area, where we call *Möbius City* as shown in Fig. 3, to represent every identical piece comprises the indefinite large test area.

Möbius City only has four LSGs. The center one is the only complete LSA. The north half (N) and the south half (S), the northwest half (NW) and the southeast half (SE), and the northeast half (NE) and the southwest half (SW), comprise the three other LSAs. The latter three LSAs' allocation emulates six complete LSAs around the center one in the original test area. Since we are only interested in when, where, and how frequently an MS moves from one LSA to another rather than specifically identifying which one it moves from and to, assigning only four LSGs is sufficient for our work.

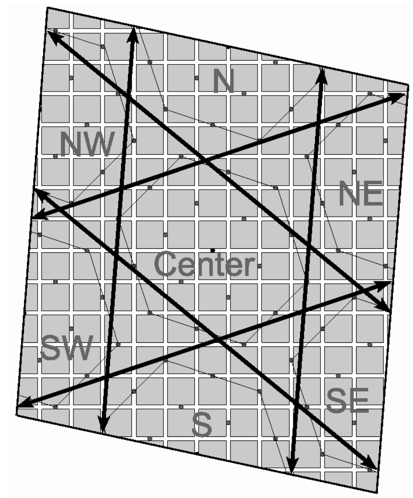

Fig. 3: Möbius City map with teleporting directions.

Möbius City is comprised by the area cropped from the original street structure and portals at the boundaries. Just like moving through the tunnels in Pac-Man's maze, whenever an MS moving among the streets reaches a boundary and is about to escape from Möbius City, the portal teleports it to a proper location at the opposite side and reenter Möbius City. The rules of the portals are:

1) For MSs about crossing north boundary, teleport them to $(-230(N-1), -230(5N+3))$ from their current locations.
2) For MSs about crossing south boundary, teleport them to $(230(N-1), 230(5N+3))$ from their current locations.
3) For MSs about crossing west boundary and their current locations satisfy $3(N-1)x + (9N+5)y > 0$, teleport them to $(230(4N+3), -230(4N+1))$ from their current locations.
4) For MSs about crossing west boundary and their current locations satisfy $3(N-1)x + (9N+5)y \leq 0$, teleport them to $(230(5N+2), 230(N+2))$ from their current locations.
5) For MSs about crossing east boundary, and their current locations satisfy $3(N-1)x + (9N+5)y > 0$, teleport them to $(-230(5N+2), -230(N+2))$ from their current locations.
6) For MSs about crossing east boundary, and their current locations satisfy $3(N-1)x + (9N+5)y \leq 0$, teleport them to $(-230(4N+3), 230(4N+1))$ from their current locations.

The teleport directions are shown in Fig. 3 as well.

An MS moving through a portal doesn't encounter any discontinuity except its coordinates: its direction and speed are the same, it associates with the same LSG, and the geographical parameters relative to the service group's facilities remain.

Thus, everything interests us is equivalent as the MS moving into an adjacent parallelogram area in an indefinite large test area.

4.3 Configuration of Backhaul Network

Although connecting every BS to the corresponding server through a line-of-sight and high-speed direct link offers the lowest transmission latency, constructing such a backhaul network is impractically expensive. Therefore, we assume each BS only has direct connections to its six neighboring BSs to form a mesh network as the core network. In mesh-styled backhaul network, network latency between a BS and the server depends on the number of nodes along the shortest path, the total length of the path, and the relay latency per node. The former two factors are related to the coordinates of the BS and the server, which will be simulated as well.

4.4 Traverse Delay

We define the response time as the average time interval between when a user sends an input and gets an expected output update. The proposed server configuration is meant to improve the response time by reducing traverse delay along the communication route between each BS to the server which is hosting the service. Factors other than the traverse delay, such as computational capabilities provided by servers, would affect the user experience and the quality of our service. Most of them, however, either affect different configurations equally, or can be overcome with reasonable cost.

Traverse delay is defined as:

$$T_{tv} = 2 \cdot \left\{ \frac{L_r}{V_r} + \frac{L_l}{V_l} + N_{rt} \cdot T_{rt} + N_{rl} \cdot T_{rl} \right\} \qquad (1)$$

where L_r is the distance of radio transmission, which is the distance between the MS and the BS it currently uses, V_r is the propagation speed of radio, which equals to the speed of light, L_l is the total length of wireline transmission in the mesh network, V_l is the propagation speed in wireline, which is approximately two thirds of the speed of light, N_{rt} is the number of nodes along the transmission path in the mesh network, T_{rt} is the average waiting time per node in the mesh network, which includes nodal processing delay, queuing delay, and transmission delay, N_{rl} is the number of servers which are receiving the snapshot and relaying data to/from the VM server, and T_{rl} is the processing and relay time per server in the hand-off chain.

4.5 Hand-off Duration

Whenever a VM-level hand-off occurs, i.e., an MS detects that it's out of the range of the original BS and the nearest BS belongs to another LSG at the latest update, we set up an anticipated hand-off end time by adding hand-off duration to the current time. The hand-off duration is given by the following equation:

$$T_{ho} = T_x + \frac{L_s}{V_l} + N_s \cdot T_{rt} \qquad (2)$$

where T_x is the total time to deliver every bit of a snapshot to media, which is the summation of queuing delay, processing delay, and transmission delay of the snapshot, which is proportional to the size of the snapshot, L_s is the total transmission distance between the current and the next VM servers, and N_s is the number of nodes between two neighboring servers, which always equals to $2N + 1$ in this case.

4.6 Update Time Points and Cost Charging

Updates occur for two reasons: a hand-off is completed, or an MS reaches an update position. At each update time point, T_{tv} and transaction counts are updated concurrently.

Whenever a position update comes at T_{now}, all hand-off end times registered in queue earlier than T_{now} have to be treated as update time points according to the algorithm described below:

1) Define T_n as the n_{th} earliest hand-off end time in queue, L_{sn} as the total transmission distance between servers corresponding to the n_{th} earliest hand-off in queue, L_r, L_l, N_{rt}, and N_{rl} are the current cost parameters calculated by the MS's current position and hand-off status, and T_{last} as the previous update time.
2) If $T_{now} > T_0$, insert an update time point at T_0, calculate the transaction counts by the Poisson process given user input rate λ and time duration $(T_0 - T_{last})$, set $T_{last} = T_0$, subtract N_{rl} by one, subtract N_{rt} by $\{2N+1\}$, subtract L_l by L_{s0}, update T_{tv} according to the new parameters, and remove T_0 and corresponding L_{s0} from the queues.
3) Redo step 2 until $T_{now} < T_0$ or the queue is emptied.
4) Calculate the transaction counts by the Poisson process given λ and time duration $(T_{now} - T_{last})$, update T_{tv} according to the new parameters, and set new $T_{last} = T_{now}$.

As specified in UMTS urban mobility model, we update the MSs' positions every 5 meters. Since a hand-off may occur at the same time, we have to handle the extra cost brought by it as well. When a new hand-off occurs with a position update at current time T_{now} while the previous update time is T_{last}, and every hand-off end time earlier than T_{now} is already treated with the above algorithm, we use another algorithm to update cost parameters, which is described below:

1) Register the new hand-off end time and the corresponding L_s in the queue.
2) Increment N_{rl} by one.
3) N_{rt} is recalculated by the MS's current position and added by $\{N_{rl} \cdot (2N+1)\}$.
4) Let L_l equals to the summation of all L_s's in queue.
5) T_{tv} is then updated accordingly.
6) The transaction counts are calculated by the Poisson process given λ and time duration $(T_{now} - T_{last})$, and then set new $T_{last} = T_{now}$ for the next update.

Every transaction in an update interval is charged with identical T_{tv}. Note that T_{tv} updated at a time point T is applied to the transactions occur *after* T, while transaction counts calculated at T are placed in the time interval ended at T.

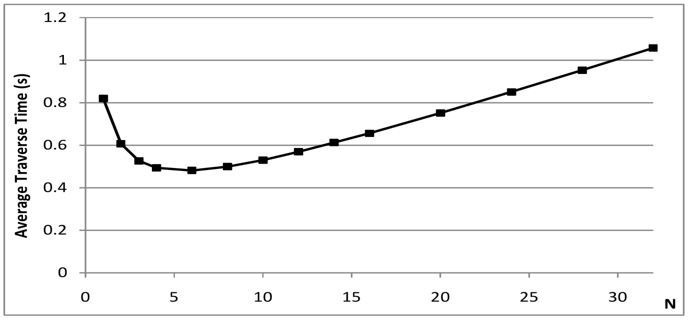

Fig. 4: Simulation result of different N given $T_{rl} = 0.5s$ $T_{rt} = 20ms$, and $T_x = 600s$, $\lambda = 1.0$.

Although technically we can create a continuous T_{tv} function and integrate it in each update interval to derive a slightly more accurate T_{tv}, it is unnecessarily complex since T_{tv} variation is negligible within the 5 meters (or less) long path.

4.7 Traverse Time Accounting

The average T_{tv} per transaction is calculated at the end of 100,000 independent simulations, each lasts 86400 seconds (one day). The simulation results of variable N, T_{rt}, T_{rl}, T_x, and λ, are presented in the following section.

5. Simulation Results

We first simulate how the size of LSAs affects T_{tv} given *nominal* parameters, which are $T_{rt} = 20ms$, $T_{rl} = 500ms$, $T_x = 600s$, and $\lambda = 1.0$. The simulation result is shown in Fig. 4.

As we can see in Fig. 4, T_{tv} is high in small LSA configurations due to the higher hand-off occurrence rate. As N increases, T_{tv} first descends, levels for a range of N's, and then linearly ascends. The descending for low N's is due to the reduction of hand-off occurrence. The smooth ascending for higher N's is caused by the higher average number of the nodes along the backhaul route and longer average transmission distance while the hand-off occurrence rate is too low to matter. The flat bottom in between is the result of the two effects competing with each other. We can conclude that setting $N = 10$ in this case is optimal in reducing average T_{tv} and keeping the total number of the servers low, which also means lower deployment and maintenance cost.

Since the above conclusion is only applicable in this set of parameters, we adjust each parameter in the nominal set to see how it affects T_{tv} as a function of N in the following subsections.

5.1 Effect of T_{rl}

T_{rl} is the cost that only applies in hand-offs. We set T_{rl} to $200ms$, $800ms$, and $1100ms$, to see how it affects T_{tv}. The simulated T_{tv} as a function of N and T_{rl} given $T_{rt} = 20ms$, $T_x = 600s$, $\lambda = 1.0$ is shown in Fig. 5.

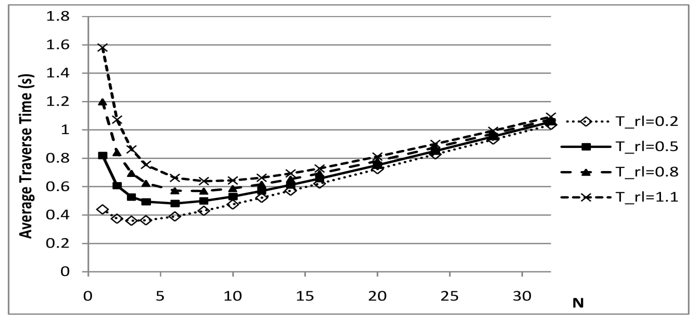

Fig. 5: Simulated T_{tv} given $T_{rl} = 0.2s$, $0.5s$, $0.8s$, $1.1s$ and $T_{rt} = 20ms$, $T_x = 600s$, $\lambda = 1.0$.

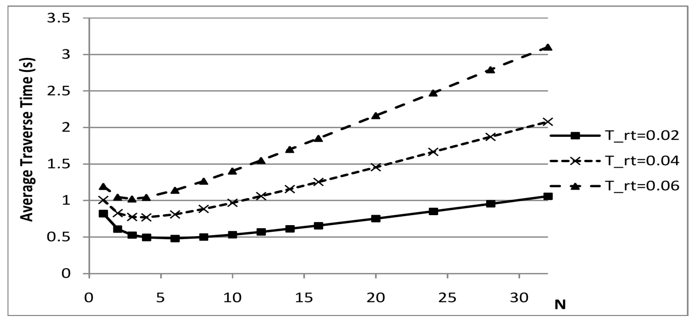

Fig. 6: Simulated T_{tv} given $T_{rt} = 20ms$, $40ms$, $60ms$ and $T_{rl} = 500ms$, $T_x = 600s$, $\lambda = 1.0$.

As we can see in Fig. 5, higher T_{rl} significantly increases T_{tv} in small LSA configurations. As N increases, T_{tv} given different T_{rl}'s has a tendency to converge together since the hand-off occurrence rate is dramatically reduced and thus renders the effect of T_{rl} insignificant.

5.2 Effect of T_{rt}

Unlike T_{rl}, T_{rt} affects both hand-offs and normal transactions since higher T_{rt} amplifies the influence of transmission distance. The simulated T_{tv} as a function of N and T_{rt} given $T_{rl} = 0.5s$, $T_x = 600s$, $\lambda = 1.0$ are shown in Fig. 6.

Fig. 6 shows the comparison of T_{tv}'s as functions of N given $T_{rt} = 20ms$, $40ms$, and $60ms$. We can easily figure out that as T_{rt} increases, not only T_{tv} increases, but it also increases more sharply for higher N and thus compresses the optimal range of N since higher T_{rt} increases the communication cost per transmission distance in the mesh network. In larger LSA configurations, although hand-offs rarely occurs and thus related cost is minimized, the inner-LSA transmission cost increases more significantly due to the higher nodal cost T_{rt}.

5.3 Effect of T_x

T_x affects the cost only in hand-offs. Higher T_x may mean larger synchronization data, longer hand-off initialization time, or longer queuing delay. How T_x affects T_{tv} is represented in Fig. 7.

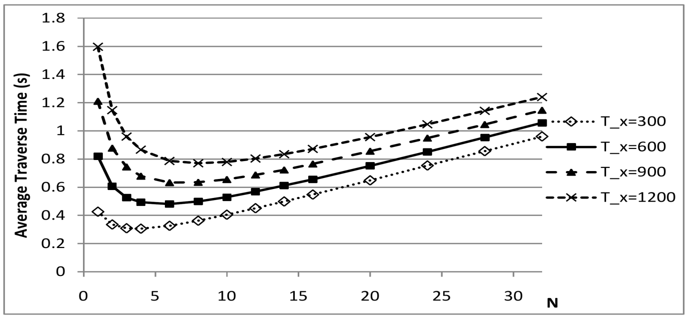

Fig. 7: Simulated T_{tv} given $T_x = 300s$, $600s$, $900s$, $1200s$ and $T_{rt} = 20ms$, $T_{rl} = 0.5s$, $\lambda = 1.0$.

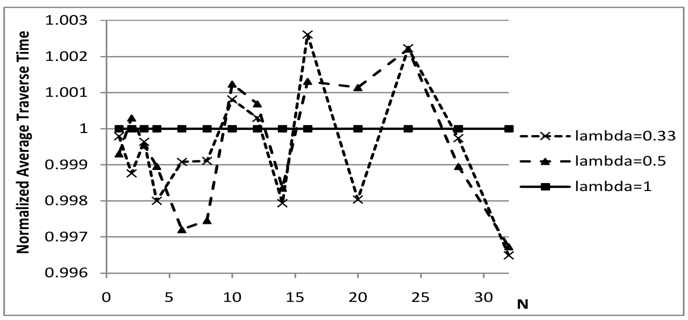

Fig. 8: Normalized simulation results given $\lambda = 0.33$, 0.5, 1.0 and $T_{rt} = 20ms$, $T_{rl} = 0.5s$, $T_x = 600s$.

Since T_x is the dominant factor of each hand-off's duration, increasing T_x fairly increases the proportion of the transactions occurred during hand-offs for every N. It is why T_{tv}'s as functions of N given different T_x's are virtually parallel to each other and show little tendency to converge as N increases.

5.4 Effect of λ

Although not being an intuitive factor, we still simulate T_{tv}'s as functions of N given different user input rates λ. The simulated T_{tv}'s given $\lambda = 0.33$, 0.5, and 1.0 inputs per second are almost identical. To visualize the differences, the normalized simulation results are compared in Fig. 8.

As we can see in Fig. 8, there is no difference induces by adjusting λ per se in statistical view. We should keep in mind, however, that the user experience and the maximum tolerable response delay depend on the interactivity of the application software.

6. Conclusions And Future Works

In this paper, we continue investigating the proposed infrastructure arrangement and hand-off protocol for application virtualization services for mobile computing devices by further evaluating performance. We have also proposed Möbius City, which is based on the original UMTS mobility model but modified to enable MSs to move in the test environment for indefinite period of time without presuming any boundary condition. We simulate the network delay as a result of MSs' movements and the occurrences of VM-level hand-offs in Möbius City given variable sizes of LSAs, server relay latencies, routing costs, and transmission delays of snapshots.

By using Möbius City as the test environment, we can evaluate the performance impact and benefit of different sizes of LSAs and infrastructure technologies and capabilities before providing an application virtualization service for mobile computing devices. Möbius City simulation can also provide a performance preview for planning network infrastructures aim to improve application virtualization services on unknown urban areas.

In this paper, we allow the MSs to move and issue commands independently but we have not addressed the resource contention among them yet. We will include the contention delay model in our nodal cost function to further improve the reality of our simulation. Furthermore, there are two other UMTS test environments specified by ESTI and we would like to evaluate the performance of application virtualization services in these test models. In fact, we can apply a similar modification on the rural vehicular test model specified in the UMTS document to enable running simulations for an indefinite period of time without involving any boundary condition.

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Intercloud Exchanges and Roots Topology and Trust Blueprint

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Abstract - *Cloud computing is a new design pattern for large, distributed datacenters. Initially, Service providers offering included cloud enabled productivity applications such as search, email, and social networks. Recently they have expanded their offerings to include compute-related capabilities such as virtual machines, storage, and complete operating system services. The cloud computing design yields breakthroughs in geographical distribution, resource utilization efficiency, and infrastructure automation. These "public clouds" have been replicated by IT vendors for corporations to build "private clouds" of their own. Public and private clouds offer their end consumers a "pay as you go" model - a powerful shift for computing, towards a utility model like the electricity system, the telephone system, or more recently the Internet. However, unlike those utilities, clouds cannot yet federate and interoperate. Such federation is called the "Intercloud". Working groups have proposed a layered set of protocols called "Intercloud Protocols" to solve this interoperability challenges. Instead of each cloud provider establishing connectivity with another in a Point-to-Point manner resulting into an n^2 complexity problem, Intercloud interoperability embodies 1-to-many and many-to-many models as opposed to mere cloud to cloud. This paper is in continuation and subsequently builds on to our earlier "Intercloud" related work. The paper proposes the overall design of decentralized, scalable, self-organizing federated "Intercloud" topology by specifically delving deep into how each "Intercloud" component fits in within the overall topology and how these components interact with each other.*

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Keywords: "Cloud Computing", "Cloud Standards", "Intercloud", "Cloud Exchange", "RDF", "Ontology"

1. Introduction

Cloud Computing has emerged recently as a new design pattern for a particular type of datacenter, or most commonly, a group of datacenters. Service providers offering applications including search, email, and social networks have pioneered this specific to their application. Recently they have expanded offerings to include compute-related capabilities such as virtual machines, storage, and complete operating system services.

Cloud Computing services as defined above are best exemplified by the Amazon Web Services (AWS) [1][2] or Google AppEngine [3][4]. Both of these systems exhibit all eight characteristics as detailed below. Various companies are beginning to offer similar services, such as the Microsoft Azure Service [5], and software companies such as VMware [6] and open source projects such as UCSB Eucalyptus [7][8] are creating software for building a cloud service.

For the purposes of this paper, we define Cloud Computing as a single logical datacenter which:

- May be hosted by anyone; an enterprise, a service provider, or a government.
- Implement a pool of computing resources and services which are shared amongst subscribers.
- Charge for resources and services using an "as used" metered and/or capacity based model.
- Are usually geographically distributed, in a manner which is transparent to the subscriber (unless they explicitly ask for visibility of that).
- Are automated in that the provisioning, upgrade, and configuration (and de-configuration and roll-back and un-provisioning) of resources and services occur on the "self service", usually programmatic request of the subscriber, occur in an automated way with no human operator assistance, and are delivered in one or two orders of seconds.
- Resources and services are delivered virtually, that is, although they may appear to be physical (servers, disks, network segments, etc) they are actually virtual implementations of those on an underlying physical infrastructure which the subscriber never sees.
- The physical infrastructure changes rarely. The virtually delivered resources and services are changing constantly.
- Resources and services may be of a physical metaphor (servers, disks, network segments, etc.; often called "Infrastructure as a Service" or IaaS) or they may be of an abstract metaphor (blob storage functions, message queue functions, email functions, multicast functions, all of which are accessed by running of code or script to a set of API's for these abstract services; often called

"Platform as a Service" or PaaS). These may be intermixed.

The terms are well accepted now [9]. Use Cases and Scenarios for Cloud IaaS and PaaS interoperability [10][11] have been detailed in the literature along with the challenges around actually implementing standards-based federation and hybrid clouds. The high level architecture for interoperability including a protocol suite and security approach was proposed where the term "Intercloud" was first coined [12].

Additional focus on security architecture was provided [13], and additional focus on how the overall architecture might be used to enable an exchange involving a marketplace was detailed and prototyped [14]. Finally, overall Intercloud technical topology and protocol blueprints have been architected [15], and implementation approaches including presence and dialog, security approach, and semantic ontology model and directory, [16][17][18] have been defined.

This paper initially briefly reviews this work and builds on that technology foundation. The paper goes on to propose detail blueprints of the Intercloud Topology describing how each component exists within the proposed topology and how these components interact with each other.

2. Review of Intercloud Technical Architecture

Cloud instances must be able to dialog with each other. One cloud must be able to find one or more other clouds, which for a particular interoperability scenario is ready, willing, and able to accept an interoperability transaction with and furthermore, exchanging whatever subscription or usage related information which might have been needed as a pre-cursor to the transaction. Thus, an Intercloud Protocol for presence and messaging needs to exist which can support the 1-to-1, 1-to-many, and many-to-many use cases. The discussion between clouds needs to encompass a variety of content, storage and computing resources.

The vision and topology for the Intercloud we will refer to is an analogy with the Internet itself: in a world of TCP/IP and the WWW, data is ubiquitous and interoperable in a network of networks known as the "Internet"; in a world of Cloud Computing, content, storage and computing is ubiquitous and interoperable in a network of Clouds known as the "Intercloud"; this is illustrated in Figure 1.

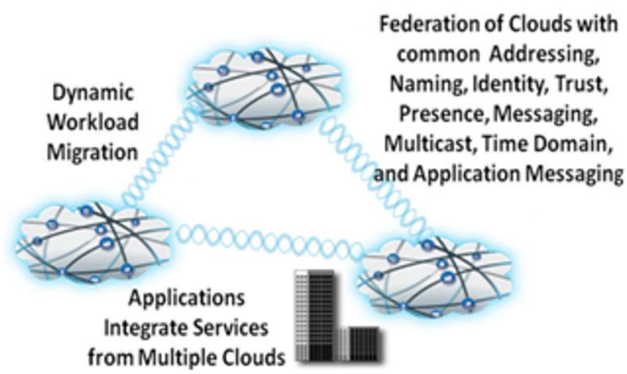

Figure 1. The Intercloud Vision

The reference topology for realizing this vision is modeled after the public Internet infrastructure. Again, using the generally accepted terminology, there are Public Clouds, which are analogous to ISP's. There are Private Clouds which is simply a Cloud which an organization builds to serve itself. There are Intercloud Exchanges (analogous to Internet Exchanges and Peering Points) where clouds can interoperate, and there is an Intercloud Root, containing services such as Naming Authority, Trust Authority, Directory Services, and other "root" capabilities. It is envisioned that the Intercloud root is of course physically not a single entity, a global replicating and hierarchical system similar to DNS [19] would be utilized.

All elements in the Intercloud topology contain some gateway capability analogous to an Internet Router, implementing Intercloud protocols in order to participate in Intercloud interoperability. We call these Intercloud Gateways. The entire topology is detailed in Figure 2.

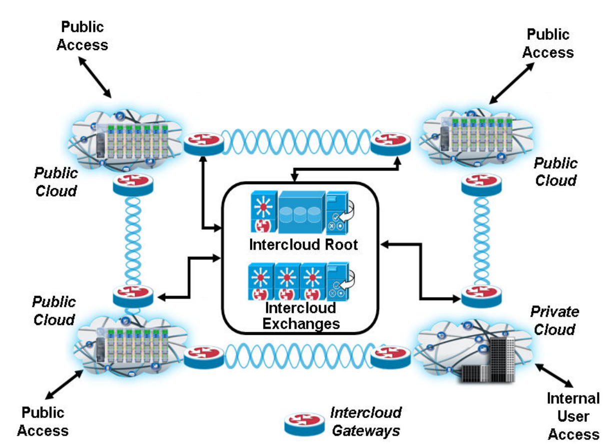

Figure 2. Reference Network Intercloud topology and elements

The Intercloud Gateways would provide mechanism for supporting the entire profile of Intercloud protocols and standards. The Intercloud Root and Intercloud Exchanges would facilitate and mediate the initial Intercloud negotiating process among Clouds.

Once the initial negotiating process is completed, each of these Cloud instance would collaborate directly with each other via a protocol and transport appropriate for the interoperability action at hand; for example, a reliable protocol might be needed for transaction integrity, or a high speed streaming protocol might be needed optimized for data movement over a particular link.

3. Intercloud Topology – Resources Directory Deployment

As described earlier that various providers will emerge in the enablement of the Intercloud. We first envision a community governed set of Intercloud Root providers who will act as brokers and host the Cloud Computing Resource Catalogs for the Intercloud computing resources. They would be governed in a similar way in which DNS, Top Level Domains [20] or Certificate Authorities [21]: by an organization such as ISOC [22] or ICANN [23]. They would also be responsible for mediating the trust based federated security among disparate clouds by acting as Security Trust Service providers using standards such as SASL [24] and SAML [25].

As part of the proposed topology, we propose that Intercloud Root providers would be federated in nature. Each of these federated nodded in the overall Intercloud topology will independently manage the "root" capabilities such as Cloud Resources Directory Services, Trust Authority, Presence Information etc. Additionally, each Intercloud Root instance will be associated with its affiliated Exchanges by defining the affiliation relationship as part of the Intercloud "root" instance.

In order for the Intercloud capable Cloud instances to federate or otherwise interoperate resources, a Cloud Computing Resources Catalog system is necessary infrastructure. This catalog is the holistic and abstracted view of the computing resources across disparate cloud environments. Individual clouds will, in turn, will utilize this catalog in order to identify matching cloud resources by applying certain Preferences and Constraints to the resources in the computing resources catalog.

The technologies to use for this are based on the Semantic Web [26] which provides for a way to add "meaning and relatedness" to objects on the Web. To accomplish this, one defines a system for normalizing meaning across terminology, or Properties. This normalization is called Ontology. Our earlier work [17] outlined approach for how Cloud Computing resources can be described, cataloged, and mediated using Semantic Web Ontologies, implemented using RDF techniques [27].

Due to the sheer size of global resources ontology information, a centralized approach for hosting the repository is not a viable solution due to the fact that one single entity can not be solely responsible and burdened with this humongous and globally dispersed task:

- Single-point-of-failure
- Scalability
- Security ramifications
- Lack of autonomy as well as arguments related to trust and the authority on data
- etc …

Instead, Intercloud Roots will host the globally dispersed computing resources catalog in a federated manner.

Intercloud Exchanges, in turn, will leverage the globally dispersed resources catalog information hosted by federated Intercloud Roots in order to match cloud resources by applying certain Preferences and Constraints to the resources. From overall topology perspectives, Intercloud Exchanges will provide processing nodes in a peer-to-peer manner on the lines of Distributed Hash Table (DHT) overlay based approach in order to facilitate optimized resources match-making queries. Ontology information would be replicated to the Intercloud Exchanges (DHT overlay nodes) from their affiliated Intercloud Roots using a "Hash" function.

There has already been lot of work done on Semantic Peer-to-Peer based systems – GridVine[28], RDFPeers[29], Piazza[30], PIER[31], and "Distributed Overlay for Federation of Enterprise Clouds" [32].

The basic idea of DHT overlay system is to map a key space to a set of peers such that each peer is responsible for a given region of this space and storing data whose hash keys pertain to the peer's region. The advantage of such systems is their deterministic behavior and the fair balancing of load among the peers (assuming an appropriate hash function).

Furthermore, DHT overlay system provides location transparency: queries can be issued at any peer without knowing the actual placement of the data. Essentially, the DHT peer-to-peer overlay is a self-organizing, distributed access structure, which associates logical peers representing the machines in the network with keys from a key space representing the underlying data structure.

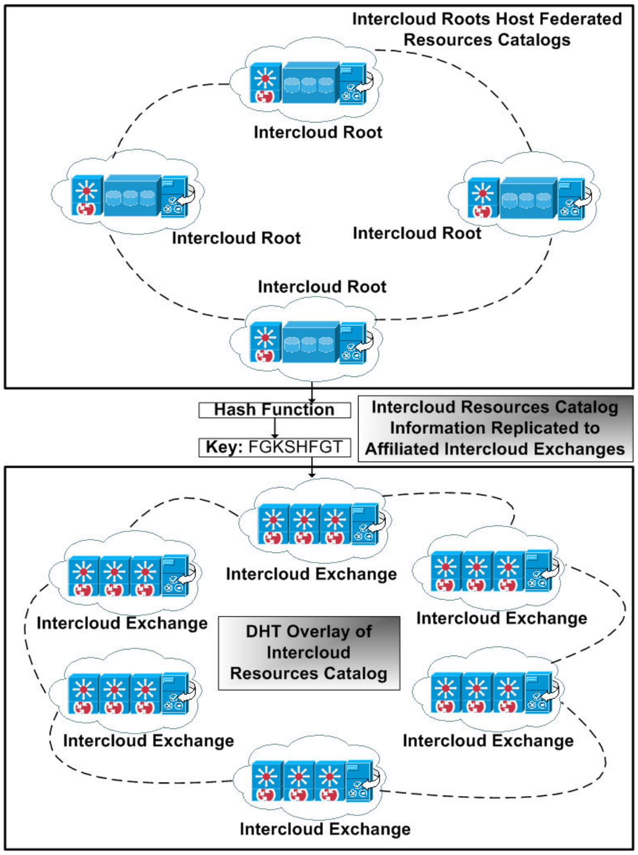

Figure 3. Intercloud Topology – Resources Directory Deployment

Nodes within the DHT overlay system are uniformly distributed across key space and maintain list of neighbors in the routing table. Each peer in the DHT overlay system is responsible for some part of the overall key space and maintains additional routing information to forward queries to neighboring peers. As the number of machines taking part in the network and the amount of shared information evolve, peers opportunistically organize their routing tables according to a dynamic and distributed binary search tree.

4. Intercloud Topology – Collaboration Details

Part of interoperability is that cloud instances must be able to conduct dialog with each other, one cloud must be able to find another cloud, which for a particular interoperability scenarios, is ready, willing, and able to accept an interoperability transaction with and furthermore, exchanging whatever subscription or usage related information which might have been needed as a precursor to the transaction. Thus, an Intercloud Protocol for presence and messaging needs to exist.

Extensible Messaging and Presence Protocol (XMPP) is exactly such a protocol. XMPP is a set of open XML technologies for presence and real-time communication developed by the Jabber open-source community in 1999, formalized by the IETF in 2002-2004, continuously extended through the standards process of the XMPP Standards Foundation. XMPP supports presence and structured conversation of XML data. Our earlier work [18] explains in great detail as far as feasibility of XMPP as control plane operations protocol for Intercloud.

Instead of each cloud provider establishing connectivity with another cloud provider in a Point-to-Point manner resulting into n2 complexity problem, as part of the Intercloud topology we propose that Intercloud Exchanges will help facilitate as mediators for enabling connectivity and collaboration among disparate cloud providers. As stated earlier that Intercloud Exchanges will leverage XMPP as control plane operations protocol for such collaboration and host the XMPP servers in a **Trusted Federated** manner to facilitate the end-to-end collaboration.

In order to establish collaboration with another cloud, an Intercloud enabled cloud will simply send a XMPP message to its affiliated Intercloud Exchange which hosts the XMPP server. If the recipient cloud is affiliated to the same Intercloud Exchange, the XMPP server will send the message directly to the recipient cloud.

On the other hand, if the recipient cloud is affiliated to another Intercloud Exchange, the XMPP server will send the message to the recipient's XMPP server hosted by the affiliated Intercloud Exchange. This is essentially termed as XMPP federation — the ability of two deployed XMPP servers to communicate over a dynamically-established link between the servers. In the Intercloud topology, a server accepts a connection from a peer only if the peer supports TLS and presents a digital certificate issued by a root certification authority (CA) that is trusted by the server — **Trusted Federation**.

In a typical federated identity model, in order for a cloud provider to establish secure communication with another cloud provider, it asks the trust provider service for a trust token. The trust provider service sends two copies of secret keys, the encrypted proof token of the trust service along with the encrypted requested token.

For scenarios where collaboration between initiating cloud provider and recipient cloud provider is across Intercloud Root or Intercloud Exchange, Intercloud Root systems will serve as a Trust Authority and act as the identity providers to mediate trust relationship as part of the **Trusted Federation**. The detail flow for this scenario is illustrated in Figure 4.

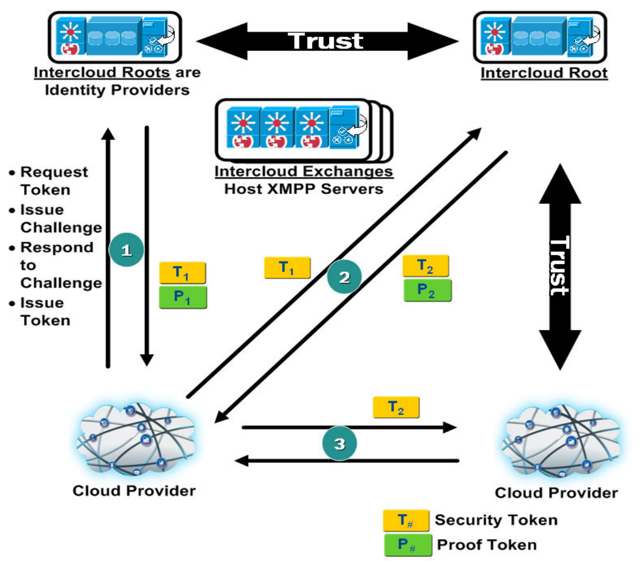

Figure 4. Inter "Intercloud Root" and Inter "Intercloud Exchange" Collaboration Scenario

For scenarios where collaboration between initiating cloud provider and recipient cloud provider is within the same Intercloud Exchange, Intercloud Exchanges will themselves will serve as a Trust Authority and act as the identity providers to mediate the trust relationship as part of the **Trusted Federation**. The detail flow for this scenario is illustrated in Figure 5.

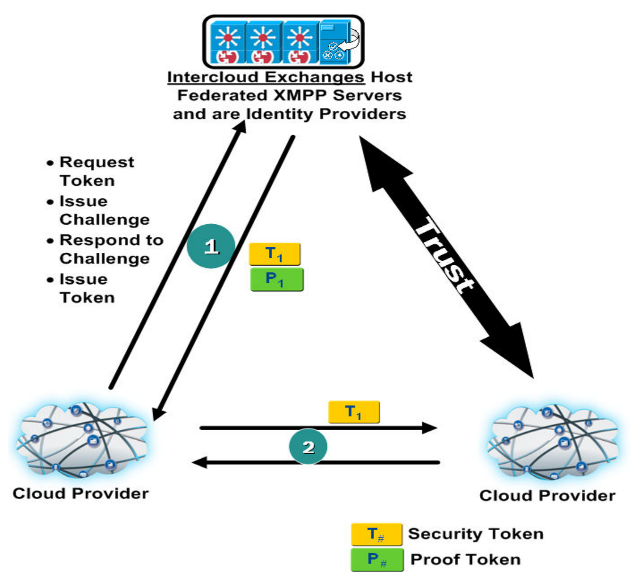

Figure 5. Intra "Intercloud Exchange" Collaboration Scenario

5. Intercloud Topology – PKI Certificates Deployment

In an Intercloud cross-clouds federated environment, security concerns are even more important and complex. Intercloud paradigm or cloud computing paradigm, in general, will only be adopted by the users, if they are confident that their data and privacy are secured. Trust is one of the most fundamental means for improving security across heterogeneous independent cloud environments.

Currently, Public Key Infrastructure (PKI) based trust model is the most prevalent one. PKI trust model depends on a few leader nodes to secure the whole system. The leaders' validity certifications are signed by well established Certificate Authorities ("CA"s).

At a basic level, proposed Intercloud topology subscribes to the PKI based trust model. In accordance to the PKI trust model, the Intercloud Root systems will serve as the Root Certificate Authority (CA) [33] and issue certificates to their affiliated Intercloud Exchange systems.

PKI Certificates not only need to identify the clouds, but the resources the clouds offer, and the workloads that the cloud wishes federation with other clouds, to work upon. Where web sites are somewhat static, and a certificate can be generated to trust the identity of that web site, cloud objects such as resources and workloads are dynamic, and the certificates will have to be generated by a CA. As per the proposed Intercloud topology, the Intercloud Exchange will serve as the intermediate "CA"s, issue temporary PKI certificates to their affiliated cloud providers acting in a just-in-time fashion to provide limited lifetime trust to the transaction at hand.

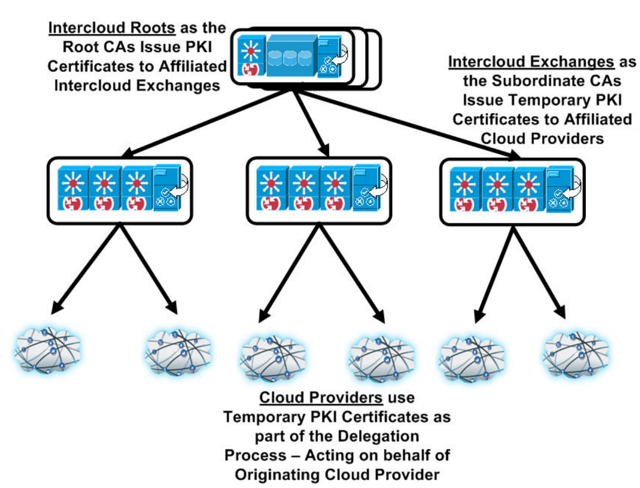

Figure 6. Intercloud PKI Certificates Topology

Cloud Providers, in turn, will use the temporary PKI certificates as part of the delegation process – acting on behalf of the originating cloud provider.

6. Intercloud Topology – Trust Management

From Intercloud topology perspectives, Intercloud Roots will provide PKI CA root like functionality. According to the current PKI based trust model, once the CA authorizes the certificate for an entity, the entity is either trusted or non-trusted. However, in the cloud computing environment, especially in the Intercloud environment, this model needs to be extended to have "Trust Zone" to go along with the existing PKI based trust model. Intercloud exchanges will be responsible for the "Trust Zone" based trust model layered on top of the PKI certificate based trust model.

The overall trust model is more of a "Domain based Trust" model. It divides the cloud provider computing environment into several trust domains. Nodes in the same domain usually are much more familiar with each other; they have a higher degree of trust for each other.

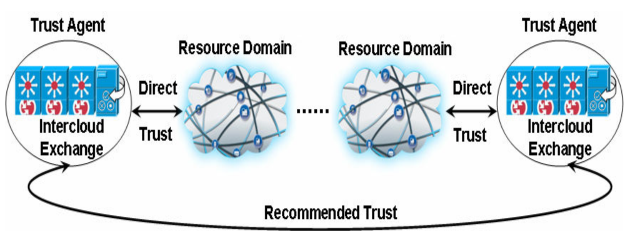

Figure 7. Intercloud Trust Management Model

Exchanges are the custodians/brokers of "Domain based Trust" systems environment for their affiliated cloud providers. Cloud providers rely on the Intercloud exchanges to manage trust. As part of the identification process for matching desired cloud resources, individual consumer cloud provider will signify the required "Trust Zone" value such as "Local Intercloud Exchange" domain or "Foreign Intercloud Exchange". Depending on the desired "Trust Zone" value, for example, one Intercloud provider might trust another provider to use its storage resources but not to execute programs using these resources. Intercloud Exchanges, in turn, will utilize the desired "Trust Zone" value as part of the matching Preferences and Constraints in order to identify matching cloud resources.

At present, as mentioned in the previous paragraph, we are considering a static "Trust Zone" as an extended trust model layered on top of the PKI certificate based trust model. However, in the future, we will also evaluate a dynamic "Trust Index" value assigned to each and every cloud resource type. "Trust Index" is essentially a level of trust demonstrated by cloud providers. Depending on the level of trust (40%, 50%, 60%, or 100%), for example, one Intercloud provider might trust another provider to use its storage resources but not to execute programs using these resources. The *trust level* is specified within a given time because the *trust level* today between two entities is not necessarily the same *trust level* a year ago. Unlike static PKI certificates and "Trust Zone" model, Trust Level is something dynamic in nature.

"Trust Level" is something that is computed in a real time basis by utilizing a Trust algorithm. The algorithm will evaluate the underlying security attributes of a cloud provider such as "Firewall Capabilities", "Intrusion Detection and Anti-Virus Capabilities" and so on. Additionally, cloud provider reputation parameters such as "Prior Success Rate", "Turnaround Time" and so on would be considered as part of the overall determination of "Trust Index". Accordingly, the fuzzy logic based aggregation algorithm will establish the "Trust Index" of a cloud provider. Once calculated, the "Trust Level" of every Intercloud provider could be cached at the affiliated Intercloud Exchange for performance reasons.

7. Conclusions and Future Work

This paper proposes the overall design of decentralized, scalable, self-organizing federated "Intercloud" topology by specifically delving deep into how each "Intercloud" component fits in within the overall topology and how these components interact with each other. In this context, the paper describes various aspects of Intercloud topology components such as Intercloud Resources Directory, Intercloud Collaboration and last but not the least, Intercloud Security.

In order to make this a reality, an operational Intercloud topology must be experimented with in a live public trial. To that regard, we are working towards establishing the Intercloud "Testbed" by collaborating with various well known academic institutions and industry leaders.

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Controlling Link Congestion on Complex Network

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Abstract - *We studied the impact of bandwidth utilization factor on converged network of Zain Contact Centre which is a complex network environment in Nigeria. Some congestion control techniques were reviewed. Experiments were carried out on the real network, a legacy network and an integrated converged network considering the same number of users. The corresponding packets were compared. As a result, higher throughput and minimal packet loss were achieved at lower bandwidth utilization and better than what was obtained at higher utilization using the same parameters.*

Keywords: Network convergence, Packet Switching, Bandwidth Utilization, Congestion control.

1 Introduction

Enterprises across the globe are continuously investigating ways to implement voice, video and data over a single network for various reasons and usage. Today's communication networks need robust and compatible platforms to meet our global society's needs. Businesses and telecommunication companies have used diverse equipments, wiring and personnel to operate and maintain separate voice, video and data networks. Traditionally, data signals are transported over dedicated Local Area Network (LAN) or (Wide Area Network (WAN) networks. Voice signals are also carried over the dedicated voice devices like Private Branch Exchange and voice gateways via circuit switched techniques. Live video contents are also sent via terrestrial broadcasting to televisions in various homes; via wireless radio and microwave technologies [1]. Network convergence is enabling enterprises and service providers to deploy a wide range of services such as Voice over Internet Protocol (VoIP), Internet Protocol Television (IPTV). A lot of benefits are accruable from the implementation of converged networks some of which are: lower cost and increased access, enhanced capacity for innovation, positive impact on the society. The drawback associated with converged networks is the contention for the limited bandwidth which brings about network congestion [2, 3]. Due to the packetization of the signals in a converged network, the random access and the contention techniques accessing a shared medium often results in collision of packets on the common link [4]. Over the years, several efforts have been invested in the study of assessing the impact of congestion on such a converge network [5-7]. This research was channeled into alleviating the challenges posed by contributing factors that leads to congestion, thus, to obtain low latency and higher throughput in the converged network environment. This paper investigates the impact of link congestion on the converged network of Zain Contact Centre.

2 Materials and Methods

In order to achieve accurate results, the converged environment was first isolated, where the conventional legacy data network was first examined and also behavioral characteristics of the network was observed. The network monitoring tool used for parameter measurement was the Orion Solarwind Software®. The software was modified to measure the latency, throughput, retransmission time (RTT), and packet loss first for a single user and then up to 30 users. Results from the link between Lagos Contact Centre and Abuja Contact Centre were obtained from the Cisco® 7201 border router at the Lagos contact centre. All convergent-possible devices on the network (like IP Phones, IP Video conferencing devices, wireless networks, streaming of video on the network and internet) were first disabled, so that the measurement for the legacy network could be taken before upgrading to the converged network, which was the objective of this work. Then, for the converged network, sources and destination of nodes were identified; the idea was to identify the origin and destination of probe packets for analyzing the network performance. The Tanderberg® video conferencing devices for both Lagos and Abuja were connected to the network via the Ethernet ports on a Cisco® Catalyst 2950 switch.

3 Analysis of Results and discussions

There are several types of delays in a complex network which differ from each other as to where they are created. The delay components are classified based on the place of their creation, mechanism or some other attributes. The causes of delay in the network could be traced to delay induced by multiple of devices and protocols responsible for the transmission of packet on the network collectively called device latency. Delay occurred (Network or Server side delay) at a particular device when the device cannot process requests arriving at its node as soon as they arrived resulting in a backlog of request or queues, thus resulting to network congestion.
The bandwidth for the LAN network was 100Mbps. The test was first carried out for a single user, and was later extended to 30 users each with average file size of 4Mbps. A legacy network was first put in place and converged packets were eliminated from the network. Measurements and results were taken and evaluated. Later, the converged packets were introduced and comparisons of the generated results were considered. Table 1 shows the file size against throughput in a legacy network, while Figure 1 depicts the graph of the table.

Table 1: File size and throughput results for a single user in the legacy network

File size (Mb)	Throughput (Mbps)
2	1.96
2.5	2.36
3	2.96
3.5	3.44
4	3.88

Figure 1 showed that as the file sizes were increased the throughput of the network increased exponentially toward its threshold.

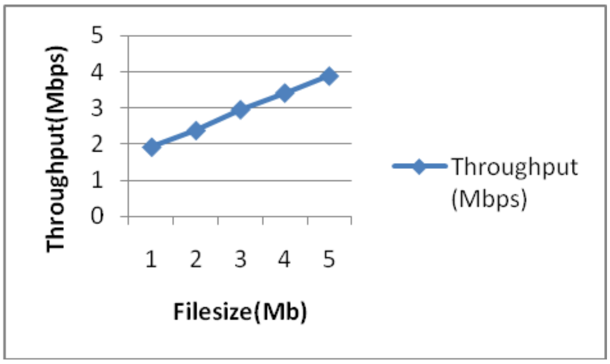

Figure 1: Throughput against file size for a single user in the legacy network.

Table 2 shows file size against throughput in a converged network with a single user. In Table 3 and 4, the number of users on the link was increased to 10, and gradually to 30 for the legacy network, and then for the converged network.

Table 2: File size and throughput results for a single user in the converged network.

File Size (Mb)	Throughput(Mbps)
2	1.92
2.5	2.38
3	2.96
3.5	3.42
4	3.89

The situation was the same as when for a single user in the converged network, Figure 2

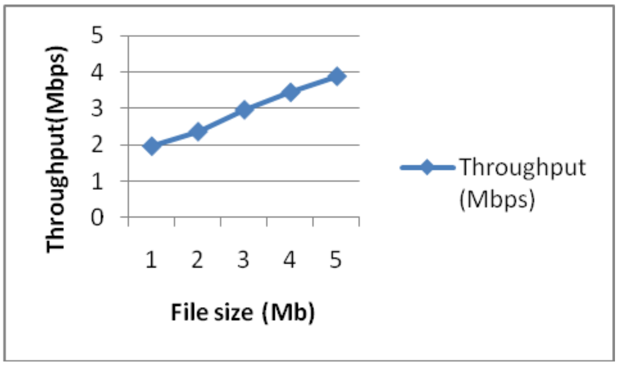

Figure 2: Throughput against file size for a single user in the converged network.

There was a gradual increase in the throughput, though the throughput appeared to respond sharply to the increase in the number of users, most noticeable between 10 and 25 users, as depicted in Figure 3 and Figure 4.

Table 3: Result of network throughput for 10-30 users in the Legacy Network.

File Size (Mb)	10 Users	15 Users	20 Users	25 Users	30 Users
	Throughput (Mbps)				
2	19.4	28.8	38.8	46.4	57.4
2.5	24.3	37.1	48.6	61	72.4
3	28.3	43.4	58.7	73.4	81.2
3.5	34.4	51.1	68.4	86.2	94.6
4	38.9	58.6	77.6	93.2	94.7

Table 4: Result of network throughput for 10-30 users in the Converged Network.

File Size (Mb)	10 Users	15 Users	20 Users	25 Users	30 Users
	Throughput (Mbps)				
2	18.8	27.6	36.2	42.6	62.1
2.5	23.8	33.8	47.6	46.3	68.4
3	26.6	41.3	53.4	71.9	82.7
3.5	32.3	49.3	64.8	82.6	82.8
4	37.6	56.2	73.4	82.8	82.6

In Figure 3 and Figure 4, the throughput increased for both networks, until when the number of users got to 30. As the file size began to increase for the 30 users, a trend began to emerge, where the throughput linear increment reduced, and then, remain constant. This suggests that a limit of saturation on the network was fast approaching, with the converged network most noticeable.

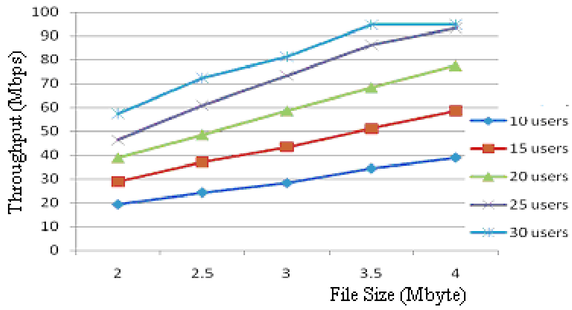

Figure 3: Graph of file size against throughput for 10-30 users in the Legacy Network.

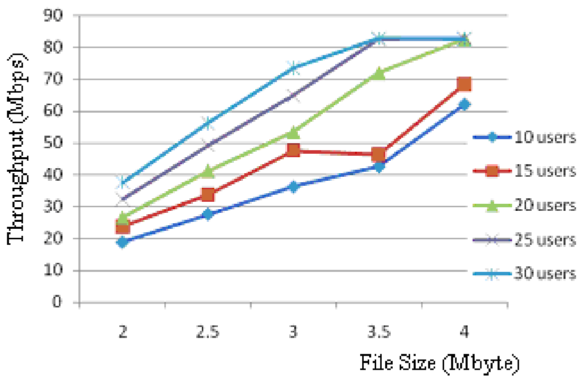

Figure 4: Graph of file size against throughput for 10-30 users in the converged Network

The graphical analysis has shown that the linear relationship between packet size, latency and round trip time (RTT) affects throughput and packet loss. Additional increase in the packet size for a given network decreases the throughput for respective users. This explains the degradation in bandwidth performance as evidenced in the loss in throughput value (around 40%).
It is also shown that the burst nature of the converged packets tend to contribute to the high latency and congestion encountered on the network as against when it was just a legacy network.
The upward linear nature of throughput in a 10-user environment was complemented when the number of users increased; this however became altered when the number of users got to 30. As more packets traversed the network, and the certainty of high packet retransmission caused by the high-bandwidth usage by the converged network, congestion occurred on the network as depicted in Figure 3 and 4.
Comparing the throughput for the legacy and converged network shows that both networks behave very much alike, that is, their response to increased packet size is similar, until where the number of users increased to a mid-point. At this stage, when throughput on the legacy network was still ascending in the linear direction, on the contrary, the throughput of the converged network was already nose-diving, and packet drops were becoming imminent, this is seen in Figure 3 and 4.
The quality of the bandwidth improves at low bandwidth utilization factor as compared to when the utilization was high. At the initial stage, the high utilization tends to improve the network performance, but this later proved to be the contrary when the amount of traffic on the network became high. Whereas when the low utilization factor was still sustaining the traffic flow and maintaining network balance, the high utilization had already saturated the traffic via a high round trip time (RTT). This translates to high latency on the network. The throughput at higher utilization factor $(\rho \geq 1)$ was found to be lower than the corresponding throughput at lower utilization factor $(\rho < 1)$. It was observed that the results from the real experimental network had packet loss, jitter in delays and that the network bandwidth was not sufficient for all requirements. There was delay on the network due to network overloads or congestions. Thus, the network bandwidth was limited and was not enough for all applications and users at the same time and there were packet loss which affects the network performance. This further substantiates the discovery as earlier pointed out, that the bursts nature of converged network traffic defies simple analytical estimation. It also indicates that a very precise latency cannot be evaluated from the simple combination of serialization, propagation and queue delay [8]. Increasing the efficiency of data exchange in computer networks based on the TCP/IP protocol requires solving complex problems which are related to the following: choice and optimization of network topology, optimizing the bandwidth capacity of the channel, choice of routes, choice of methods of data flow streams and verifying the parameters in control, analyzing the buffer sizes of switches and routers, and choosing strategy for congestion control.

4 Conclusions

The real experimental network showed that analytical model was not very good at describing packet loss rate and its relation to throughput and latency when the number of simultaneous converged packet - flow increases across the network. It should be noted that the model for evaluating the latency on the network is not exhaustive; other inherent sources of delay on the network which were not considered in this paper can be very useful in further work on the subject. In addition, the methodology used in the real experimental network, using simplex communication may also be reviewed to accommodate for duplex communication and multi stream of convergent packet data in further study.

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Differentiated Services Fuzzy Assured Forward Queuing for Congestion Control in Intermediate Routers

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Abstract – *Todays, computer networks play a very important role in our daily activities. In order to have a better network management, data packet problems, should be considerably reduced. One of the most important problems that happens in networks and is not recognizable easily, is congestion, That encouraged experts to research more about this topic. Congestion avoidance is a QOS[1] method that prevents considerable packet loss in networks and assures successful packet forwarding to destination. In this article we have implemented a new queuing mechanism that uses fuzzy logic with assured forwarding in the intermediate routers, in order to control and avoid congestion by using an AQM[2] method. We have simulated it by Matlab software and NS[3] package[4], then compared it with RED[5] and dsRED methods in order to evaluate efficiency of this new method, and discussed the results. The simulation results show a better performance compared to the previously mentioned mechanisms.*

Keywords: Assured Forwarding, Congestion control, Differentiated Services[6], Fuzzy Logic, QOS

1 Introduction

Computer networks are communication agent between workstations and in this communication, data are exchanged between workstations. So the influence of data flow control in the computer networks is very important. Routing control, traffic control, congestion control, sequence control and etc are elements to control data flow in computer networks [1]. So flow control is related to point to point traffic between a sender and a receiver, and congestion control provides mechanisms to make sure weather subnet is able to carry the traffic [2]. In recent years data packet flow has been increased, because the number of users and accessible data have been increased. Packets with different sizes are exchanged in such networks, but packet losses and delays usually occurs because of the network congestion. To avoid these unpleasant situations packet routing is used to control the flow in the network [3]. In fact congestion is the worst delay that is caused by datagram overloads in one or more nodes of a router. When congestion occurs, the delays will increase and a router tries to queue datagram until they can be sent [4].

In this article we will discuss a new method of congestion control. With this method, we control queuing in the router by using fuzzy logic and assured forwarding in order to manage queue congestion of routers so that we have minimum traffic and packet loss probability in a flow.

In this article, a method to control the congestion in intermediate routers in a network by using fuzzy logic is discussed in second section and then in the third section we will discuss the design of a fuzzy controller in this queue, and in the forth section we will explain about the conditions of simulation. Then we will compare the efficiencies of the suggested method, the RED and the dsRED queue methods in fifth section, and finally we will discuss the simulation results in the sixth section.

2 Differentiated Services Fuzzy Assured Forwarding Queue (dsFAFQ)

In this article we use distributed class-based queuing, based on fuzzy logic. We have defined 4 distributed queuing classes in differentiated nodes with its special attributes [7], and in these distributed queuing classes [5], we classify packets according to assured forwarding that is given by Heinanen and colleagues [6]. Each class takes its specific resources and produces packets classified in one of the classes willingly.Then they are marked after being placed in one of the four classes according to assured forwarding. At this level, 3 probabilities are defined to drop packet that is caused by congestion.

1- Quality Of Services
2- Active Queue Management
3- Network Simulation
4- NS 2.35
5- Random Early Detection
6- DS

Packets are marked with different drop packet probability including low drop precedence, middle drop precedence and high drop precedence. As we have four assured forwarding classes and three packet drop probabilities, so we will have 12 differentiated services class in total. Besides, a router's buffer is divided into edge and core router and can be considered as 2 seperated routers. In edge part, router is devided into 4 classes in order to separate output flow and flow that can not send data quickly, so buffer space will be occupid and other flows will face to starvation [8, 9].

As it's shown in fig.1, a physical queue is assigned to each assured forwarding class. As packets with 3 different packet drop probabilities enter to each queue, we should assign a virtual queue to any single packet drop probability. On the other hand, we must dedicate 3 virtual queue to each one of physical queues.

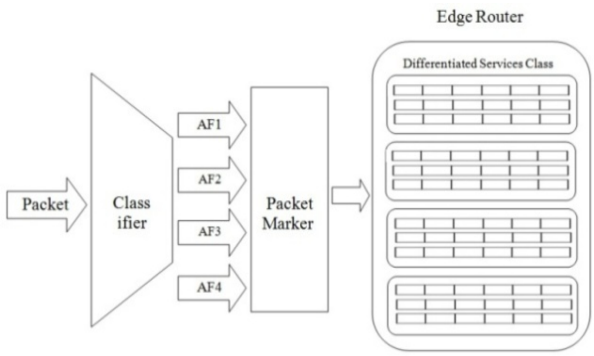

Fig.1: Classifying packets in the nodes based on differentiated services

The edge part is connectd to core part by a fuzzy controller that is shown in fig.2. This would mean that the fuzzy controller checks the queue status existing in the core at specific times, and identifies all those packets that should be transmited from edge part to core part. Then packets are placed in physical and virtual queues and will be sent from core part assuredly.

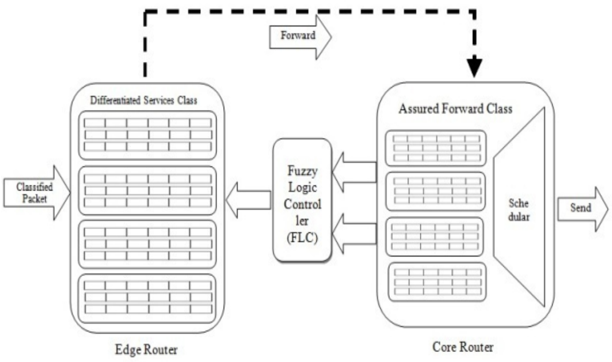

Fig2: Buffer space division into edge and core part and the connection between them .

Packets are entered into physical queue according to a specified class and specific drop probability in edge router part and then packet transmission percentage are calculated by a fuzzy controller. Then packets are transmitted from edge router to core router and scheduling is carried out to give service or exit the packets. Services to different queues is given by round robin method. When we don't have congestion in network any packets with any drop priorities can enter into edge queue of router and receive services by transmitting to core part, but in congestion period not only entering into queues in core part will be decreased by fuzzy logic, but also packets with high drop priorities don't have permission to enter into edge part of router.

3 Design of Fuzzy Controller for Fuzzy Assured Forward Queuing

The secret of success in fuzzy system is its simple implementation, simple maintenance, simple understanding, stability and cheapness [10]. And two weaknesses of fuzzy system are the absence of learning ability and it's not easy to define membership function for it [11]. But of course the second weakness depends on how good is our recognition of system. Fuzzy systems are suitable for estimated or vague inferences specially those systems that it's difficult to make a mathematical model for them [11]. So fuzzy controller in the router must be designed that, it can control the packets entrance with specific drop priority into the core router, so that congestion in router and network doesn't occur. That means the queue get filled quickly, if the percentage of packets entering the core queue of routers was more than specific queue threshold, the controller decreases the number of forwarding packets to that queue until the queue get to a balance status. In other words, the number of forwarded packets to any queue should relate to current queue load and average of waiting time in the queue. Control of packets entering the specified class priority in the core of router is done by fuzzy controller that is shown in fig.3. Two input parameters and one output parameter have been considered in fuzzy controller that is:

- **Queue load**

The number of existing packets in the queue core of router that are waiting to get services, is called queue load and 7 fuzzy sets are considered for it:

QueueLoad={Empty, MEmpty, LHalf, Half, MHalf, Full, VFull}

The members in the set show empty, more than empty, less than half, half, more than half, full, very full status.

- **Waiting time**

It's the period of time that packet waits to get services (this time doesn't include the time of servicing). We consider 7 input fuzzy sets for it:

Waiting Time= {Small, BSmall, SMed, Med, BMed, Large, VLarge}

The members in the set show small, bigger than small, smaller than med, med, bigger than med, large, very large status.

- **The number of forwarded packets**

It's the average number of sent packets to every class. The number of packets that are forwarded by fuzzy controller is output parameters that 7 fuzzy sets has been considered for it.

Forward= {VLow, Low, LNorm, Norm, MNorm, High, VHigh}

The members show very low, low, less than normal, normal, more than normal, high, very high status.

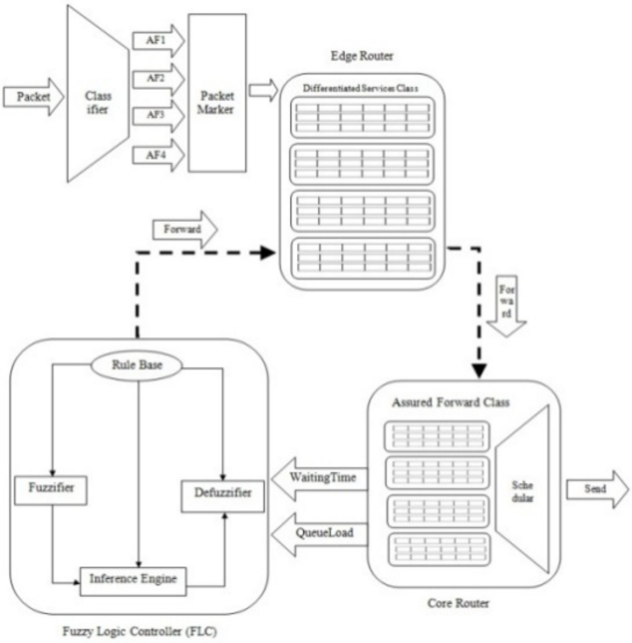

Fig.3: fuzzy controller and suggested method to manage congestion

We assign a membership function to any single output and input parameters. The most common function that are used in this method are trapmf[1] and trimf[2] that are drawn by fuzzy logic of Matlab software which are shown in fig.4, 5, 6. This software is the product of math works company in 1984 that is so practical for different field of engineering and fuzzy toolbox is used to edit and create fuzzy inference systems and we can use it's result in object Oriented programming or simulations [12].

Hence we use Matlab software to create membership function of input & output parameters and the detected result that is produced as a matrix will be used in the next simulation scenario in the following section.

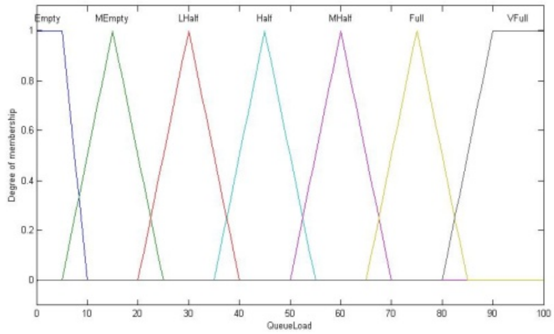

Fig.4: Membership function for queue load input parameter

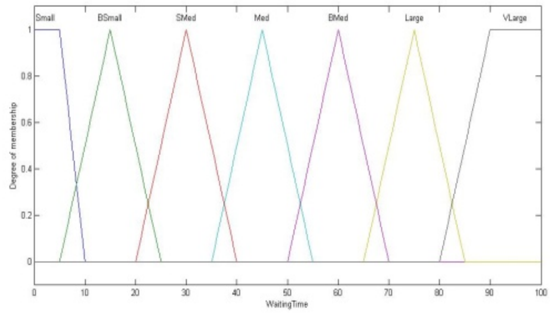

Fig.5: Membership function for waiting time input parameter

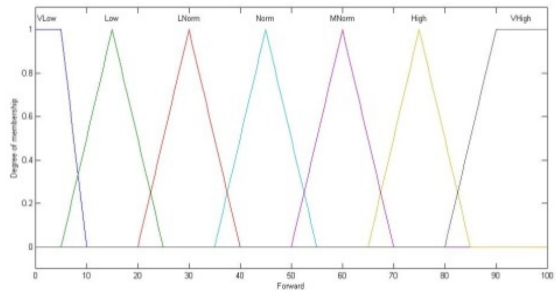

Fig.6: Membership function for number of packet forward output parameter

4 The simulation condition of dsFAFQ

The method of packet processing for assured forwarding is shown in fig.7. In the first level data packets are produced with specific rate by source and then are classified into one of the four classes according to their priority. In the second level packets that are placed in the priority classes, will be marked according to types of drop packets. So we use an 8 bits field called TOS in IP packets, and 6 bits out of 8 bits are used for identifying the packet differentiated services of classes according to suggested code point that is given by Heinanen and colleagues [6] and is explained in [7, 13]. These two levels are done by sender machine. As there

1- Trapezoidal Membership Function (trapmf)
2- Triangular Membership Function (trimf)

are 4 class and 3 drop probabilities for each packet, all in all 12 differentiated services class or DS are created. So any differentiated services of class is shown with AF_{ij} and i shows physical class or queue and j shows the type of drop packets or virtual queue. Low drop precedence is shown by number 0 and high drop precedence is shown by number 2. For example ,AF_{22} means a packet in third virtual queue and second physical queue and high drop precedence.

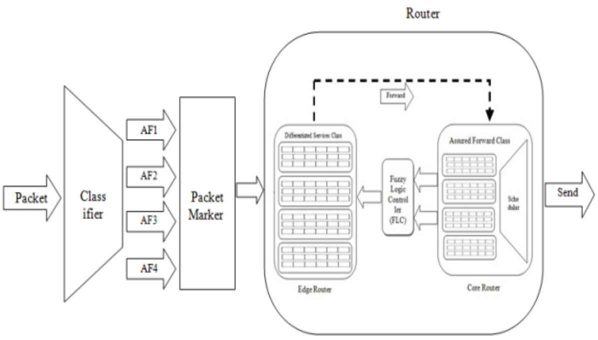

Fig.7: Packet classification based on assured forwarding mechanism for an input flow.

After separating packets in sender, output flow enters into third level (router machine). In this level buffer of router that is divided into 2 parts, edge router and core router, separated into 4 classes in each part and a physical queue is assigned to each class as can be seen in fig.7. Classified packets enter to edge router with a specific rate. Percentage of transmitted packets from the same queue in edge router to core router are determined by fuzzy controller. The fuzzy controller checks status of existing queue in core router using waiting time and current queue load parameters. The packets are forwarded to the determined physical and virtual queues in core router and then the packets are sent from core router assuredly.

The processing and servicing of packets to different classes in core router is done by round robin method. This would mean after that packets are placed in the queue by a scheduler that manage packets departure by round robin, servicing or packet sending will be done. So the suggested queue that has been made in router has the following characteristics:

- Packets are produced by 2 CBR[1] one with rate of 4Mb/s and the other with rate of 6 Mb/s in UDP protocol.
- Physical queue limit in edge router is 20 packets and total system limit is 80 packets.
- Packets are sent with output bandwidth 7 Mb and input packets enter into router queue by 2 separated 10 Mb bandwidth link.
- The size of all packets are fixed to 1,000 bytes and the number of produced packets by the source, are about 100,000 packets. These packets enter into system by a fixed rate, specified class and specified drop priority.
- Every queue in edge router has a maxthresh about 10 packets so there might be 2 state in the system.
  - If the number of packets in a queue is less than the maxthresh, packets will enter into that queue with any drop priority.
  - If the number of packets in a queue is more than the maxthresh, just LD packets enter into that queue until the queue overflows which in this case we use tail drop policy.

The results that are given in table 1, are related to the suggested queue simulation that in 20th , 40th , 60th and 80th is registered and also total simulation time is 85th.

Table 1: The number of total packets, sent packets and drop packets in router queues.

Packets Statistics

CP	TotPkts	EnPkts	TxPkts	ldrops
All	24994	15535	15128	9459
10	2147	198	198	1949
11	2047	1879	1826	168
12	2079	1870	1821	209
20	2157	102	102	2055
21	2096	1849	1824	247
22	2149	1903	1824	246
30	2014	135	135	1879
31	2077	1871	1821	206
32	2066	1870	1818	196
40	2080	179	179	1901
41	2057	1859	1825	198
42	2025	1820	1755	205

Packets Statistics

CP	TotPkts	EnPkts	TxPkts	ldrops
All	74994	45353	44962	29641
10	6371	399	399	5972
11	6189	5592	5543	597
12	6158	5522	5469	636
20	6341	225	225	6116
21	6168	5404	5358	764
22	6319	5595	5541	724
30	6212	178	178	6034
31	6223	5524	5518	699
32	6336	5615	5535	721
40	6090	183	183	5907
41	6329	5603	5541	726
42	6258	5513	5472	745

Packets Statistics

CP	TotPkts	EnPkts	TxPkts	ldrops
All	49994	30460	30087	19534
10	4238	200	198	4038
11	4065	3743	3699	322
12	4089	3729	3694	360
20	4273	106	106	4167
21	4141	3634	3621	507
22	4245	3772	3697	473
30	4148	177	177	3971
31	4145	3748	3694	397
32	4165	3739	3691	426
40	4094	183	183	3911
41	4224	3748	3698	476
42	4167	3681	3629	486

Packets Statistics

CP	TotPkts	EnPkts	TxPkts	ldrops
All	99994	60310	59941	39684
10	8506	401	401	8105
11	8290	7467	7415	823
12	8208	7347	7317	861
20	8418	225	225	8193
21	8211	7280	7229	931
22	8372	7464	7412	908
30	8309	223	223	8086
31	8323	7395	7373	928
32	8417	7468	7406	949
40	8191	183	183	8008
41	8404	7460	7413	944
42	8345	7397	7344	948

Table 1 shows that in simulation period 60, 310 packets out of 99,994 packets have been entered into router queues and 39,684 packets have been dropped because the capacity of connection link is limited. It means 59,941 packets have been sent assuredly. In table 1, the code points, are those 12 separated AF classes that is produced by senders which enter into physical and virtual router queues. In table 1 the CPs[2] of 10, 11, 12 are

1- constant bit rate traffic

2- Code Point

related to virtual queues of the first physical queue. The CPs of 20, 21, 22 are related to virtual queues of the second physical queue. CPs of 30,31,32 are related to virtual queues of the third physical queue and CPs of 40,41,42 are related to virtual queues of the fourth physical queue.

5 Performance Evaluation

In this article all simulations and their produced results are done by NS2.35 software [14]. In this section the efficiency of the three types of queues, RED, dsRED that are described in detail in [15, 16, 17] and our suggested dsFAFQ queue will be compared in terms of packets sent rate, packet drop rate and bandwidth rate.

The comparisons have been done with the same traffic rate and same simulation period of time. Input traffic enter into each three algorithms from two separated links with 10Mb bandwidth and 5ms link delay. The traffic rate is 4Mb/s in the first link and 6Mb/s in second link and simulation period for all three algorithms is 85 second. Total number of packets that are produced by sender is 100,000 packets and the size of packets are fixed to 1,000 Bytes. We ignore the servicing delay time. If we focus on the output of dsRED algorithm [7, 14, 17] in table 2 and compare it with the results of suggested algorithm shown in table 1 under the same conditions, we will find out that 50,025 packets out of a total 99,986 packets have been sent and 48,118 packets have been dropped.

Table 2 shows that dsRED algorithm with 4 differentiated services class for classifying input packets has weaker management in case of sending and dropping input packets in congestion, in comparison with our newly suggested algorithm, dsFAQ, that has 12 differentiated services class for classifying packets. As the registered result in table 2 shows, the number of sent and dropped packets in 80th simulation time are almost equal but at the same period of time, the number of sent and dropped packets in our suggested method shows a remarkable improvement comparing the difference in number.

Table 2: The number of total packets, sent packets and drop packets

Packets Statistics

CP	TotPkts	TxPkts	ldrops	edrops
All	24986	12526	11992	468
10	2508	2508	0	0
11	7486	5174	2312	0
20	2508	2322	186	0
21	12484	2522	9494	468

Packets Statistics

CP	TotPkts	TxPkts	ldrops	edrops
All	74986	37526	36084	1376
10	7508	7508	0	0
11	22487	15174	7313	0
20	7508	7322	186	0
21	37483	7522	28585	1376

Packets Statistics

CP	TotPkts	TxPkts	ldrops	edrops
All	49986	25030	24050	906
10	5008	5008	0	0
11	14987	10175	4812	0
20	5008	4822	186	0
21	24983	5025	19052	906

Packets Statistics

CP	TotPkts	TxPkts	ldrops	edrops
All	99986	50025	48118	1843
10	10008	10008	0	0
11	29987	20173	9814	0
20	10008	9822	186	0
21	49983	10022	38118	1843

As can be observed in figs 8, 9, 10, 11 the efficiency of our suggested method has been compared with RED and dsRED methods and the simulation results show that the suggested method that is based on fuzzy logic works better and has a better performance in comparison with the other 2 methods.

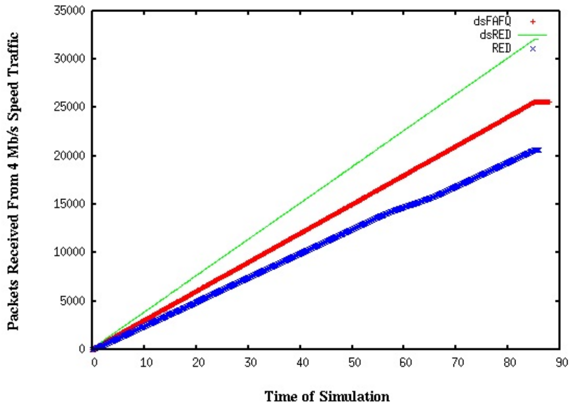

Fig.8: Comparison between our suggested method, RED & dsRED queues with 4Mb/s rate traffic agent.

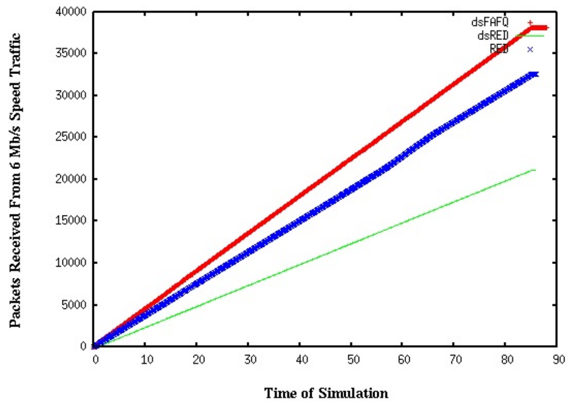

Fig.9: Comparison between our suggested method, RED & dsRED queues with 6Mb/s rate traffic agent.

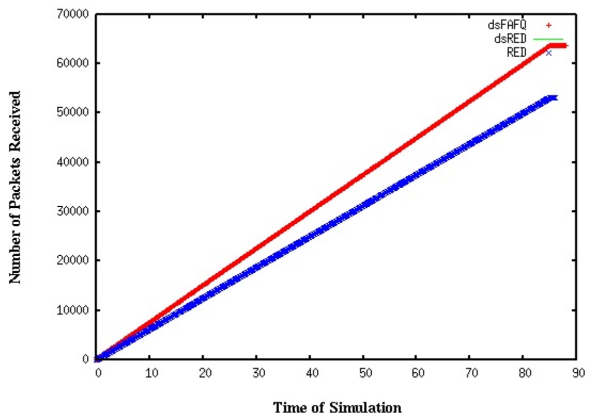

Fig.10: Total sent rate comparison between our suggested method, RED and dsRED queues.

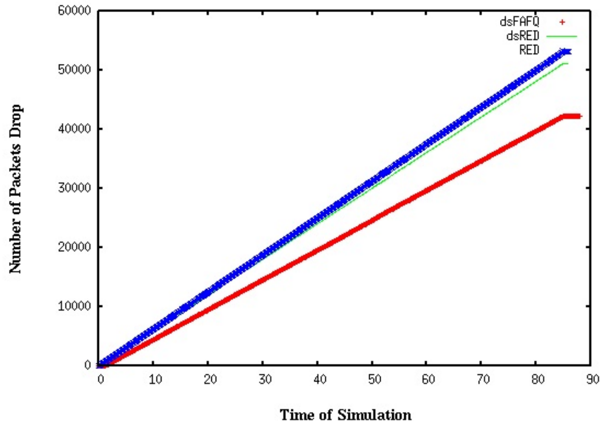

Fig.11: Drop packet rate comparison between our suggested method, RED & dsRED queues.

In fig.8 the rate of received packets from first sender with 4Mb/s traffic rate and in fig.9 sent packet rate from second sender with 6Mb/s traffic rate and in fig.10 total received packets that have been sent by two senders show preference of our suggested method in comparison with the other two methods. And in fig.11 drop rate of received packets from two senders has been shown which indicates that, the number of packets loss in our suggested method has been decreased in congestion and it has better efficiency. In fig.12 bandwidth rate usage in output channel of router is shown for our suggested method and the other two methods. It shows that bandwidth rate that is used in our suggested method is average rate of the other two methods.

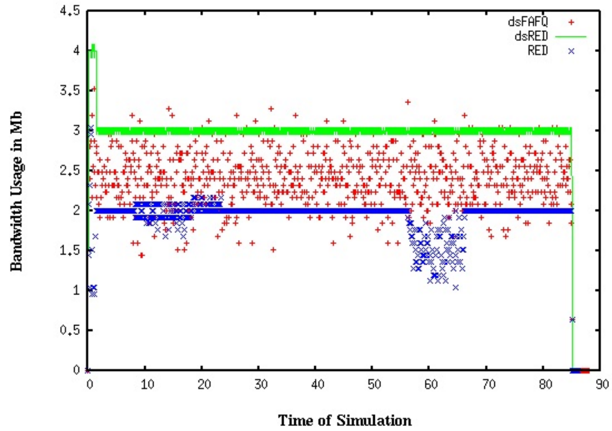

Fig.12: Bandwidth comparison between our suggested method, RED & dsRED queues.

6 Conclusions

In our suggested differentiated services fuzzy assured forward method that is described in this article and according to the requirements of each class or flow, packets are forwarded to different queues that have specific quality of service. The reason that fuzzy method is used is that we can implement unclear concepts easily and control unstable systems to make an exact decision.

The advantages of fuzzy assured forward queuing are as follows:

- Probability of starvation decreases remarkably in comparison with the RED method, because packets are forwarded into queue according to membership functions and servicing is done by round robin method.
- If the queue limit of a router increases, the number of packet drop decreases and the result is most better.
- If we assign more bandwidth to output channel of a router, this algorithm send all entered packets to router queues, assuredly.
- As the buffer space is classified and the entering into the classes is controlled by a fuzzy controller according to queue status of router, hence almost all buffer spaces are used and it avoids wasting buffer space.

Disadvantage of fuzzy assured forward queuing are as follows:

- In this method the buffer space is divided in to 4 equal parts so that any one of these parts has the same resources. So, any packets with any priority will enter into queues before reaching the threshold. And in cases when queues are overloaded, packets with high priority, can not enter into queues and will be dropped because we used tail drop policy.

- If the capacity of router queues increases, efficiency of our algorithm will decrease, because of processing of fuzzy logic information in the overloaded in the router.
- As we use fuzzy logic and need to make decisions in this algorithm, hence the implementation of this algorithm will be costly in intermediate routers.

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Session Initiation Protocol and Services

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Abstract – *This paper discusses about the Session Initiation Protocol and the call setup between the user agents with scenarios like single proxy server and multiple proxy servers of different domains. The messages and the parameters of those calls are analyzed in detail. Scenarios for a call flow with and without record are discussed, explaining the various system entities of a SIP network. The session description protocol is elaborated as to how it is related to SIP. The concept of Instant Messaging and Presence is studied and the pros and cons of SIMPLE and Jabber protocols are indicated. The global unique numbering concept, ENUM is explained with a real time example.*

Keywords: SIP, SDP, Proxy Servers, IMP, SIMPLE, XMPP

I. Introduction

Session Initiation Protocol (SIP) [1] is a general purpose application-layer control protocol for setting up, altering, and tearing down sessions. Session is by which the source and destination communicate as in Internet multimedia conferences, Internet telephone calls, and multimedia distribution. SIP makes use of the proxy servers to forward (route) the requests to the present location of the users. Such servers allow the users to inform their current locations by registering and are also responsible for call-routing. SIP works on top of various transport layer protocols. SIP works with various other protocols to carry the session descriptions and the media streams. Session descriptions allows the endpoints to agree upon a set of capable media types to communicate and it is governed by Session Description Protocol (SDP). Real-Time Protocol (RTP) is used for actually carrying the media of various types like voice-data, video and text messages.

II. Call Flow between Two User Agents and a Proxy Server

In this section, we will see in detail the call establishment between two user agents g.harish (sip:g.harish@iptel.org) and g.gokul (sip:g.gokul@iptel.org) connected to the same proxy server iptel.org. They use the SIP application SJPhone to communicate. Figure 1 shows the successful call flow involving the basic SIP functions such as the initial signaling, negotiation of session parameters to establish the session, exchange of media information in the form of SDP payloads, establishment of the media session and the termination of the session once established as in [1] and [2].

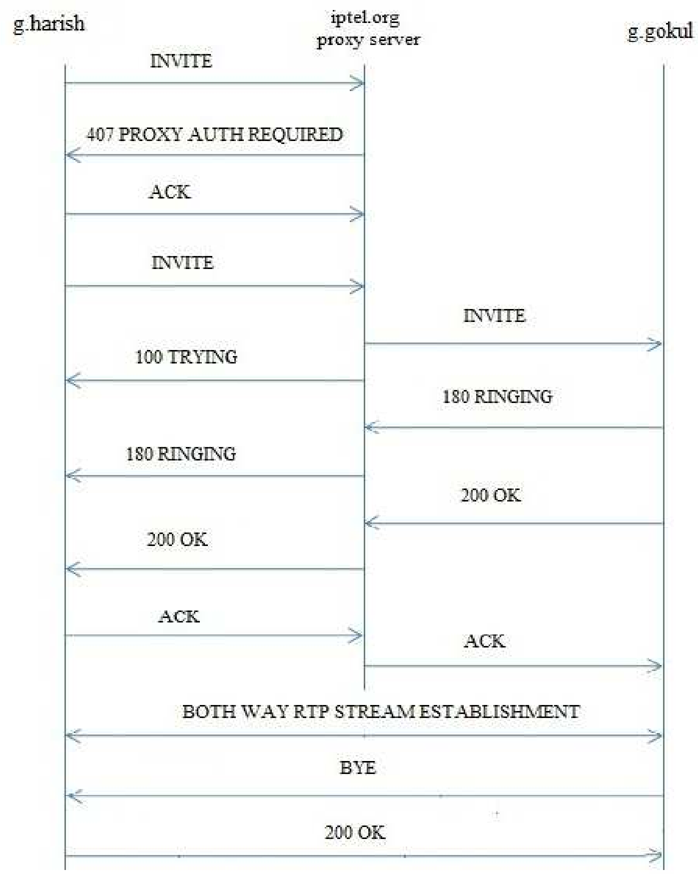

Figure 1: *SIP call flow between two user agents via proxy server*

A. Dissection of an INVITE Message

Wireshark packet analyzer is used to capture and study the packets in detail. A SIP packet consists of the Request-Line, Message Header and the Message Body. The type of the request made can be found in the 'Request-Line' section of a SIP packet.

INVITE sip:g.gokul@iptel.org SIP/2.0
Via:SIP/2.0/UDP
193.10.39.148;branch=z9hG4bKc10a2794000000764cd125fd000024d300000036;rport
From:"Harish" <sip:g.harish@iptel.org>;tag=2cb2479100c
To: <sip:g.gokul@iptel.org>
Contact: <sip:g.harish@193.10.39.148>

Call-ID:
5CFE60C3F1AF4F939DAAF44A09BBD9100xc10a2794
CSeq: 1 INVITE
Max-Forwards: 70
User-Agent: SJphone/1.65.377a (SJ Labs)
Content-Length: 368
Content-Type: application/sdp
Supported: replaces,norefersub,timer

Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 3497763965 3497763965 IN IP4 193.10.39.148
Session Name (s): SJphone
Connection Information (c): IN IP4 193.10.39.148
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 49198 RTP/AVP 3 97 98 8 0 101
Media Attribute (a): rtpmap:3 GSM/8000

We could see that the transaction starts with the user g.harish sending an INVITE request addressed to g.gokul's URI (sip:g.gokul@iptel.org).

B. Message Header

The header fields are described below:

Via: has the address (193.10.39.148) at which g.harish waits to receive responses to the request. The branch parameter in this field identifies this transaction.
From: consists of a display name (g.harish) and a SIP or SIPS URI (sip:g.harish@iptel.org), which indicates the sender of the request. These display names are described in [3] This header field also contains a tag parameter containing a random string (2cb2479100c) added to the URI by the SIP phone for identification functions.
To: consists of the SIP or SIPS URI (sip:g.gokul@iptel.org) to which the request is actually to be sent.
Contact: specifies a SIP or SIPS URI for a direct way of contacting g.harish, mostly made of a username at a fully qualified domain name (FQDN). As per RFC 3261 [1], though FQDN is preferred, most systems user IP addresses since they don't have registered domain names.
Call-ID: has a unique global id for this call, which is formed by combining a stochastic string and the SIP application's host name or IP address.
Dialog: The combination of To and From tags, and Call-ID defines a peer-to-peer SIP relationship between g.harish and g.gokul.
Command Sequence: referred to as Cseq contains an integer and a method name. This number is incremented for every new request inside a dialog.
Max-Forwards: puts forth the limitation on the number of hops a request can make on its way to the destination. It's an integer, which is decremented by one at every hop.
User-Agent: field consists of the information about the user agent client where the request originates.
Content-Length: is the size of the message body represented in bytes.
Content-Type: consists of a description about the message body.

The whole set of SIP header fields are explained in [1].

C. Message Body

The message body comprises of the Session Description Protocol. It contains a description of the audio/ video channel that needs to be established and the SDP fields, which are generally categorized as mandatory and optional fields.
Mandatory Fields:
v – refers to the protocol (SDP) version number, which is '0' in our case.
o – depicts the owner/ creator and an identifier of the session. In our example, 3497763965 is the session ID where 'IN' and 'IP4' refers to Internet and IP version 4 addresses.
s – shows the session name (SJphone).
t – represents the active session time.
m – is the media type, format and the transport address. In our example, "audio 49198 RTP/AVP 3 97 98 8 0 101", the media type is audio. 49198 is the port of RTP (Real-Time Protocol), which must always be an even number. The next odd port is used by RTCP (Real-Time Transport Control Protocol). RTP/AVP denotes the RTP protocol with the profile for "Audio and Video Conferences with Minimal Control" (refer [4]). The numbers denote the codecs and their preferences. For example, 3 denote the GSM codec.
Optional Fields:
c – connection information.
a – represent session attributes. For example, a=sendrecv means that the media is both sent and received, which is the case mostly with SIP communication.

D. Call Trace Analysis

Referring to Figure 1, we will assume g.harish as UA1 and g.gokul as UA2, iptel.org proxy server as PS.

- UA1 sends an INVITE request with the SIP URI of UA2 to the proxy server PS. This is a SIP/ SDP request, which means that it's a signaling as well as a message indicating the description of the session that it wants to establish with UA2.
- PS sends a "407 Proxy Authentication Required" message back to UA1 requesting it to provide its credentials to be authenticated by the server.
- Now UA1 sends an 'ACK' to the server indicating that it has received its request for providing the login credentials.
- UA1 again sends an INVITE along with the authentication credentials to PS.
- On successful authentication of the UA1, PS sends a '100 Trying' message indicating that it will work on behalf of UA1 to route the INVITE message to the destination.
- UA2 receives an INVITE from PS, which is actually a request from UA1.
- UA2 responds with "180 Ringing" to indicate that the phone is actually ringing for it to decide whether to accept or reject.
- PS routes this "180 ringing" to UA1.
- After deciding to accept the call, UA2 picks up his/her phone and thus indicating a successful "200 OK". If it

was decided to reject the call, an error message would have been sent as response.

- This message is again routed to UA1 via PS from UA2.
- UA1 now sends an "ACK" to this as response, which the UA2 receives via PS.
- At this juncture, the media session has begun between UA1 and UA2 and they start transferring the media packets using the format to which they agreed upon by exchanging SDP.
- In our example, UA2 wants to end the call and hence sends a "BYE" to UA1 directly as it no longer needs the PS to route as it has learnt about its location. It can also be sent via the PS and this concept is known as "Record-Routing", which we will be discussing later.
- To this, UA1 responds with a "200 OK" agreeing to UA2's request to terminate the connection.
- On receiving this message by UA2, the call between the endpoints is successfully terminated.

III Signaling In Two User Agents and Two Proxy Servers of Different Domains Model

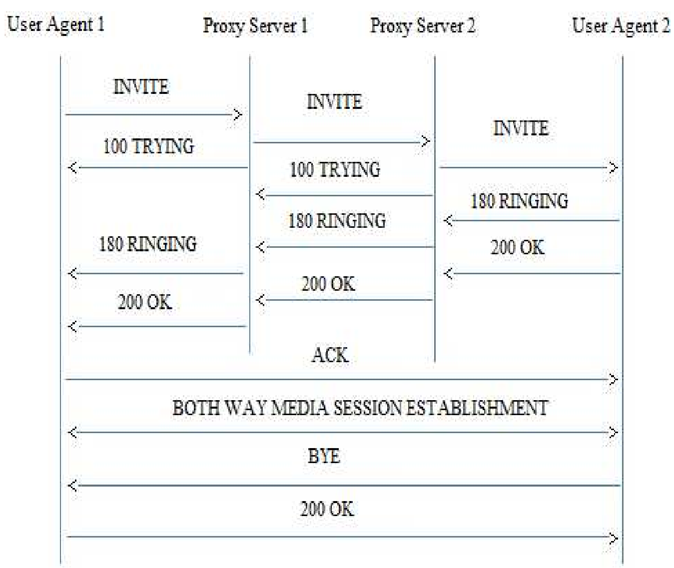

Figure 2.1: *SIP session establishment through two different proxy servers – SIP Trapezoid*

SIP response messages are classified as:

- Informational – 1xx
- Success – 2xx
- Redirection – 3xx
- Client error – 4xx
- Server failure – 5xx
- Global failure – 6xx

For the complete list of response codes and information about them, refer section 21 of [1].

A. Call Trace Analysis

- The call transaction begins with UA1 making an INVITE request for UA2. But its not aware of the location of UA2 in the IP network. Hence it passes the request to Proxy Server 1, which forwards it to PS2 for User Agent 2. And it sends a 100 TRYING response to UA1 informing that it is trying to reach UA2. PS1 knows that it has to forward the request to PS2 through the registration process of SIP.
- Similar to PS1, PS2 works similarly as PS1 On receiving INVITE. It forwards the INVITE request to UA2 (Assuming that PS2 knows the location of UA2. If not, the INVITE request would have got forwarded to another proxy server) and then issues a 100 TRYING response to PS1.
- On receiving the INVITE request, the SIP phone at UA2 starts ringing informing to inform the call request. And it issues a 180 RINGING response back to PS2 which reaches UA1 through PS1.
- UA2 now can choose to either accept or decline the call. In our example, we would like to keep it the accept way as seen in Figure 2. A 200 OK response is sent to PS2 when the call is accepted. Similar to the route of INVITE, this reaches UA1. UA1 sends an ACK message to confirm the call setup. This 3-way-handshaking (INVITE+OK+ACK) is used for reliable call setup. Its important to note that the ACK message doesn't use proxy servers to reach UA2 as by now UA1 is aware of the exact location of UA2.
- At this point, the connection is set and media flow happens between the two User Agents using the format that was agreed upon by exchanging session description using SDP.
- When UA2 wants to terminate the call it sends a BYE message to UA1 for which it responds with a 200 OK message to confirm the teardown of the session. Refer [2] for detailed description.

B. SIP Trapezoid with Record Routing

According to [1], record routing is the process in which the request traverses the proxy or list of proxies. This route set can be learned through message header Record-Route or it can be configured.

While comparing with Figure 2.1, we could see that all the messages go through the proxy servers in Figure 2.2. This method of passing through proxy servers is called as Record Routing. It is achieved by informing the endpoints about record routing with a record-route header field.

A sample packet with record-route header field is as follows:

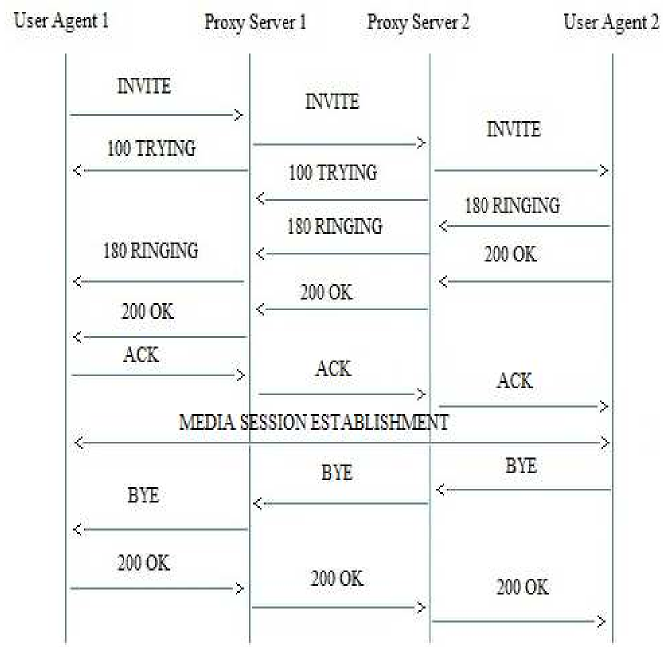

Figure 2.2: *With Record Routing - SIP Trapezoid*

Via:SIP/2.0/UDP
193.10.39.148;branch=z9hG4bKc10a27940000007a4cd12603000045330000003e;rport=5060
From:"Harish" <sip:g.harish@iptel.org>;tag=2cb2479100c
To:"Gokul" <sip:g.gokul@iptel.org>;tag=61f3478fbd8
Contact: <sip:g.gokul@193.10.39.147>
Call-ID:
5CFE60C3F1AF4F939DAAF44A09BBD9100xc10a2794
CSeq: 3 BYE
Content-Length: 0
Record-Route:
<sip:213.192.59.75;ftag=2cb2479100c;avp=orUDBwBhY2NvdW50AwB5ZXMDCQBkaWFsb2dfaWQWADViOTQtNGNjODc5ZjYtMWM3YWE2NDMDBgBzdGltZXIEADE4MDA;lr=on>
Server: SJphone/1.65.377a (SJ Labs)

IV System Entities

The system entities comprise of two important components clients and servers namely.

Clients:

As per [1], a client is any network element that sends SIP requests and receives SIP responses. User agent clients and proxies are in general termed as clients. The SIP application running on a phone, computer etc is an example of a client.

Servers:

Servers are the important elements of the network that receives requests, processes them and sends back the responses. Examples of servers are proxies, user agent servers, redirect servers, and registrars etc.

User Agent is a logical entity that can act as both a client and server depending on the situation.

A. User Agent Client:

It's a logical entity that generates a request. Typically UAC's role lasts only for the duration of that transaction. i.e., on initiating a request, it acts as a UAC for the duration of that transaction.

B. User Agent Server:

On the other hand, on receiving a request, the same UAC becomes a user agent server for processing that transaction. UAS is a logical entity that receives and processes a request.

C. Proxy Server:

The most important role of Proxies are routing in the network. Generally the exact address of the destination is not known in advance to forward the request generated. Proxy servers, forwards them on behalf of the client to the destination or to the nearest proxy server.

D. Registrar:

Location of the users is very important for the SIP communication to happen. Hence users have to register their locations to a registration server by sending REGISTER requests.

E. Redirect Server:

It's an UAS that responds with 3xx messages to indicate that the client has to contact an alternate set of URIs.

F. Location Server:

It deals with the binding of the logical and physical addresses of the users registered to a Registrar.

V. Purpose of SDP in SIP

As we know, SIP works at the layer 7 to create, modify and terminate media sessions such as voice calls, multimedia exchange, IP conferences etc. The voice/ video stream communications in SIP is carried through another layer 7 protocol, Real-time Transport Protocol (RTP). SIP messages carry session descriptions, which has to be exchanged between the end devices before the actual communication starts. This allows the participants to agree on compatible media types like audio/video codec, encoding information, connection metadata etc. This information is transported inside the SIP message body. For an example on how SDP is encapsulated inside a SIP packet, see **Section II**. Also refer [6] for more information about SDP.

VI. Presence and Instant Messaging

Presence is a mechanism by which one can sense the ability of another user to communicate. Messages like 'Online', 'Offline', 'Busy' etc. are examples of Presence. It is generally used to know if the other party is ready to start a conversation via an instant message.

Instant Messages are sent when the user hits the send button. They are generally shown in sequential order that is grouped together in a window. AOL is one of the earliest IM that was in use amongst the web surfers.

A. SIP for Instant Messaging and Presence:

Several millions of users use instant messaging and presence (IMP) programs. Different IMP clients are used to access the various IMP servers (like AOL, Skype.) because they are proprietary. This made IETF to formulate the IMP Working Group (WG).
SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE add up the basic SIP protocol with instant messaging and presence capabilities. Though SIP was originally developed for voice over IP (VoIP) it has matured to support conferencing and other media streams.

As stated in [7], SIMPLE [5], is designed in a way such that it can be applied to the Session Initiation Protocol in order to register presence information, receive notifications when events occur, send short messages (SMS), handle session of real-time messages like streaming between the involved entities.

B. SIMPLE and JABBER (XMPP), a Comparison:

SIMPLE is designed to be more general purposed than XMPP. It is widely used in voice, video, push-to-talk and other communication options. It's capable to be used in more applications than just IM and presence though it was developed as a protocol for signaling purposes. This makes it great for single session traffic such as SMS or IM. But it does not cater well for the huge data like video signals. It also misses the popular IM features such as contact lists and group chats.

Whereas, the XML-based Extensible Messaging and Presence Protocol (XMPP) for real time communication powers many of the popular applications including instant messaging, presence, video/voice calls, multi-party chat etc. Jabber, the famous instant Messaging and presence technology is built based on this protocol. Jabber is known to be used by millions of people over Internet. It doesn't have many of the pitfalls that SIMPLE possesses. It is said that JABBER is the Linux of IM, where a lot of developers are contributing to its betterment. While this is the case, the popularity of XMPP continues to grow.

VII. ENUM

A method of representing the various resources of the IP and telephony world under a single unique identifier/ phone number is known as **ENUM** (t**E**lephone **NU**mber **M**apping). It's based on the Domain Name Service (DNS) design where the telephone numbers are mapped to the domain names.

Before we move on to see how ENUM works it's imperative to understand what E.164 is.

A. E.164

It's an existing global numbering system administered by the International Telecommunications Union (ITU) and is therefore suitable for use by ENUM. An E.164 number comprises of the phone number, country code and the area code. For example, a telephone number 43595670 in Chennai, India can be written as +91 44 43595670.
Refer to [7] for more information.

B. How ENUM Works

On entering a telephone number, it will be converted into E.164 format. Lets take the same example as that of E.164.

- The entered number 43595670 is converted into +91 44 43595670 (Chennai, India).
- The preceding symbols are removed and reduced to just numeric digits i.e., 4443595670.
- These digits are then reversed – 0765953444.
- Each digits are separated by dots – 0.7.6.5.9.5.3.4.4.4
- E164.arpa is the proposed domain for E164. Therefore add the domain "e164.arpa" to the end of the numbers - – 0.7.6.5.9.5.3.4.4.4.e164.arpa.
- On this domain, a DNS query is made and the definitive name server is found.
- Subsequently, NAPTR records are retrieved by ENUM and an action is performed according to the registered services for that number.

VIII. Summary

We have shown how a SIP communication happens between the user agents connected to proxy servers and analyzed the call trace for the INVITE message in detail explaining all the fields and parameters. We studied how the session description parameters are carried in SDP inside the SIP protocol. The difference between the Instant Messaging and Presence (IMP) protocols such as SIMPLE and XMPP was discussed. We have also seen how the global numbering system, ENUM works.

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Security Review of P2P Applications and Networks

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Abstract - P2P security should be of interest to anyone who wants to protect their personal information and who actively uses the Internet. P2P Security has been a popular research topic for as long as P2P computing came into existence nearly a decade ago. P2P file sharing applications still remain popular and other applications based on P2P networks have gained in popularity. How do you secure your information when using P2P applications? How do you protect yourself against malicious nodes in a P2P network? This paper will review the risks inherent to P2P computing and discuss methods for securing a P2P network.

Keywords: P2P Networks, Internet Security, Network Attacks, Client-Server Computing, P2P Security

1 Introduction

Where client/server computing laid the foundations of the Internet in the mid-1990s, P2P computing grew out of the Internet itself. The Internet has made it possible to decentralize the computing experience away from the client/server model. P2P technologies have enabled Internet applications such as file sharing and new technologies such as Voice Over Internet Protocol (VOIP). Prior to the inter-networking of computers, you needed to sit at the computer in order to break into it or maybe you could use a modem as in the movie War Games. Security was not a huge concern due to the requirement to be physically at the computer. Since the Internet, securing computers has become a discipline and its own branch of study in Computer Science.

In a client/server network, the topology is highly centralized; meaning there are multiple clients obtaining data from one or only a few servers on the network. This model is a byproduct of the pre-Internet years, where the client is dependent on the server for all communication and most data processing power including databases and computation. The clients are highly dependent on the server, and all communication is routed through it.

On the other hand, in a P2P network, all the nodes are equal and have multiple communication paths leading to several of your closest nodes. The communication paths are built upon the foundation of the Internet and comprised of "…collections of computers (i.e. nodes) that simultaneously function as both clients and servers to achieve a common purpose [1]." What this means is that each node (i.e. process) is both a "supplier" and "consumer" of data simultaneously.

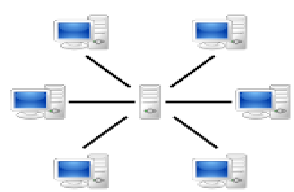

Figure 1 Client / Server - Centralized[2]

In order for peers to communicate, P2P networks use a virtual network, called an "overlay" network. The overlay network uses the structure of the Internet for basic communication, but contains its own logical links over which all nodes communicate with one another. The pure P2P paradigm calls for network topologies which are highly decentralized; though in reality, the majority of P2P overlay networks are a hybrid of both centralized and decentralized network topologies.

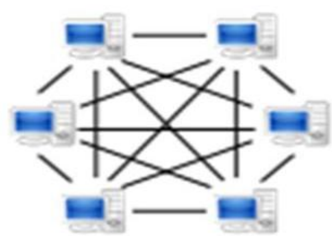

Figure 2 P2P - Decentralized [2]

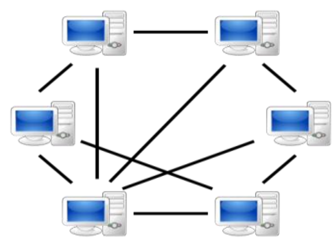

Figure 3 P2P - Hybrid [2]

1.1 P2P File Sharing

P2P networks gained much of their popularity from file sharing applications provided by companies such as Napster, Gnutella, Freenet, and BitTorrent. P2P networking made it very easy for users to exchange files – software is widely available, typically free, and easy to use. But file sharing applications are widely considered by security experts to be inherently dangerous to computer security. These are just some of the risks identified by The National Cyber Alert System [7] and "Why file Sharing Networks are Dangerous [8]":

- Installation of malicious code: it is difficult, if not impossible, to verify that the source of the files is trustworthy.
- Exposure of sensitive or personal information: you may be giving other users access to personal information: passwords and family pictures might be kept in the same folder, or a file is placed accidentally into the wrong folder.
- Susceptibility to attack: certain ports on your firewall might be open by default and allowed to transmit the files.
- Confusing interface design: in a user study mentioned in [8], Good and Krekelberg found that the KaZaA interface design contributed to user confusion over what files were being shared.
- Incentives to share a large number of files: due to the general laziness on the part of the user, the user may allow access to the My Documents folder.
- Software wizards: automatically determine your folders that contain media. Folders that normally contain media might contain important documents. These folders could be exposed by these wizards.

P2P file sharing is still very popular. "From 2009 to 2010, the major changes to the peak period composition of North America's fixed networks is the increasing presence of Real-Time Entertainment, and a slight rebound in the levels of P2P File sharing traffic (which has increased to 19.2%)." As of 2010, BitTorrent is the dominant P2P Files sharing protocol (just about everywhere except Latin America), representing almost 30% of upstream peak period traffic and slightly more than 8% of downstream peak period traffic [25].

1.2 Industries

P2P networks were made popular by file sharing applications, but the P2P technology has made its way into many other industries, including the Government [2]:

- Bioinformatics: Used to run large programs designed to carry out tests to identify drug candidates.
- Education and academia: Pennsylvania State University, MIT, and Simon Fraser University are working on LionShare, a secure P2P network for facilitating file sharing among educational institutions globally. Also, the sciencenet P2P search engine provides a free and open search engine for scientific knowledge.
- Military: The U.S. Department of Defense has already started research on P2P networks as part of its modern network warfare strategy.
- Mobile Peer-to-Peer (P2P) systems: Advances in wireless networking and mobile computing technologies, such as wireless LANs, wireless mesh networks and 3G cellular networks have further facilitated the migration of the P2P paradigm into wireless mobile computing [3].

2 P2P Security Concerns

P2P networks pose challenges to computer security beyond those of client/server computing and also beyond simply being connected to the Internet.

Besides the specific risks associated with sharing files, there are several reasons why P2P networks are inherently less secure than client/server networks.

2.1 “Open” networks.

In client/server computing, the malicious users are typically on the “outside.” Protecting yourself from malicious users involves implementing precautionary measures such as firewalls and antivirus software. These techniques are generally enough to keep malicious programs out. However P2P networks are traditionally open networks and are less secure since their functionality is based on the principles of decentralization. The P2P application software might allow members unfettered access to your computer. So the malicious user might be one of your peers with whom you share files. Extra vigilance is required to protect against malicious nodes on your network.

2.2 No central management

Because the structure and content of P2P networks are more varied, there is typically no centralized management of security functions. “IP and domain level security features like logging, filtering, and standard authentication methods don’t apply due to a lack of Super Servers [5]”. A Super Server is actually a process on a server. In a typical UNIX system, you can have lots of server processes running simultaneously, passively waiting until a request comes in. Instead of having several processes just waiting, a single Super-server can listen to many endpoints for each service [18, p.89].

2.3 Users are novices

Many users of P2P applications are novices. According to Bailes [6], “They do not understand the consequences of their inaction with regard to security. Configuration is only nominally supported during setup and ongoing use in P2P applications. That is a core problem with P2P deployment on even the most secure networks—the technology risk relies heavily upon the user’s level of technical knowledge and skills [6].”

2.4 Identity Theft

There is a greater risk for Identity Theft. According to an article on www.eweek.com file sharing not only opens you up to malicious programs, but can be an enabling factor for malicious users to steal your identity. Howard Schmidt (a former cyber-security adviser to the Bush administration, former chief information security officer at Microsoft and eBay, and now a principal in R&H Security Consulting) said that “...one woman’s credit-card information was found in such disparate places as Troy, Mich., Tobago, Slovenia, and a dozen other places. Why? We found that the shared folder in her music-downloading application was in fact making readily available her entire My Documents folder to that apps entire P2P audience, 24 hours per day [12].”

3 Attacks on P2P networks

What makes an attack possible on the Internet? The Internet was designed for the efficiency of moving packets, not on security. “The end-to-end paradigm pushes the complexity to end hosts, leaving the intermediate network simple and optimized for packet forwarding [10].” There are a number of attacks that thrive in the world of P2P computing.

3.1 Sybil

In a Sybil attack, a malicious node with multiple identities takes over the network. “Within a distributed environment, it is possible for the same physical entity to appear under different identities, particularly in systems with highly transient populations of nodes. This poses a security threat, especially in P2P systems that employ content replication, or fragmentation schemes over many peers for security and availability, and, therefore, rely on the existence of independent peers with different identities [9].” The only way to prevent this attack is to use some form of centralized authentication management.

3.2 Distributed Denial of service (DDOS)

DDOS or DOS attacks can use Botnets. Botnets are tiny nodes with malicious data that are hard to find. What makes this type of exploit particularly damaging on P2P networks is the popularity and ease of file sharing on P2P networks. File sharing and careless downloads enable the Botnets to spread. The Botnets flood a network, a node, or group of nodes that have data to share.

3.3 P2P attacks

“The attacker instructs clients of large P2P file sharing hubs to disconnect from their P2P network and to connect to the victim's website instead. As a result, several thousand computers may aggressively try to connect to a target website. This method of attack can be prevented by specifying in the P2P protocol which ports are allowed or not. If port 80 is not allowed, the possibilities for attack on websites can be very limited [21].”

3.4 Eclipse

In an eclipse attack, malicious nodes conspire to fool correct nodes into adopting the malicious nodes as their peers, with the goal of dominating the neighbor sets of all correct nodes.

3.5 File Poisoning

The following is the sequence of events in a file poisoning attack on a P2P network [11]:

1. Corrupt the target file
2. Inject the poisoned files into the P2P system,
3. Unsuspecting P2P users download the poisoned file into their own shared folders

4 Securing P2P Networks

What makes a P2P network secure? There are three major pillars of P2P security [15, p.105]:

- Ensure Secure Communication (between peers)
- Trust Management (know your peers)
- Access Control (secure admission to the P2P network)

4.1 Ensure Secure Communication

Peers within a P2P network are formed into overlay networks. An overlay network allows peers to communicate without specifying an IP address. By far the most researched security issue in overlay networks is in secure routing between peers.

To understand the requirements of secure routing, it is necessary to review the structure of overlay networks. In an overlay network, each node is running the P2P software, has its own node identifier, and maintains a routing table. This routing table is the key to finding its peers. The overlay network is based on TCP/IP networking since it is built upon the foundation of the Internet. A Distributed Hash Table (DHT) is a common form of routing table.

Each node maintains a DHT which contains key/value pairs of its neighbors. Values can be any type of object. When a piece of data is requested, the data value is hashed which returns a unique key. With that key, a DHT can locate the node that has the requested resource.

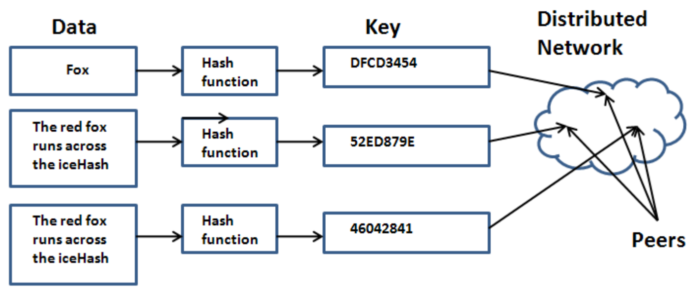

Figure 4 Distributed Hash Table [2]

According to Tannenbaum and Van Steen[18], secure routing for DHT-based P2P networks requires that:

a) Nodes are assigned identifiers in a secure way
b) Routing tables are securely maintained
c) Lookup requests are securely forwarded between nodes

First, if nodes are not assigned in a secure way, a malicious node can assign itself multiple personalities (Sybil attack). If a malicious node controls many non-faulty neighbors, you have an eclipse attack. Solutions do exist, but they require some form of centralized authority for handing out identifiers, which goes against a pure P2P paradigm. The more decentralized the network the harder to assign node identifiers in a secure way. Without a secure way to assign node identifiers, an attacker could "…arrange to control all replicas of a given object, or to mediate all traffic to and from a victim node [19]."

Second, maintaining secure routing tables ensures that "…the fraction of faulty nodes that appear in the routing tables of correct nodes does not exceed, on average, the fraction of faulty nodes in the entire overlay. Without it, an attacker could prevent correct message delivery, given only a relatively small number of faulty nodes [19]."

Finally, lookup requests should be securely forwarded between nodes. When a node in a P2P network issues a request, typically that request is sent out to possibly dozens of other known peers. Confidentiality and integrity of this communication are generally built into the overlay network. The solution proposed by Berket [20] suggests using Secure Group Layer (SGL) for secure group communication and using a shared group key for securing the messages. In existing P2P file sharing networks, any authenticated peer has unfettered access to the information at all other authenticated peers. Berket [20] "…provides mechanisms that allow each end user to autonomously specify the authentication and authorization requirements for each information item."

4.2 Trust Management

How do you know the good from the bad? Use Trust Management or Reputation Systems. The goal is to achieve a "good" reputation once you are already a peer. According to Jumppanen[16], "…the reputation system rewards peers that cooperate with other peers. Secondly it punishes peers that cheat or behave maliciously. Finally, it motivates or even forces network peers to cooperate with each other."

The authors of [17] have proposed a reputation management scheme that builds trust among members. "The proposed algorithms can detect malicious peers sending inauthentic files. The Malicious Detector Algorithm is also proposed to detect liar peers that send the wrong feedback to subvert the reputation system. Simulation results confirm the capability of the proposed algorithms to effectively detect malicious peers and isolate them from the system, hence reducing the amount of inauthentic uploads, increasing peers' satisfaction, and preserving network resources."

4.3 Access Control

Saxena[15] argues that there is no point in maintaining key management (secure communication) or trust management (trust your peers) unless there is some form of access control (secure admission). They suggest that once inside, a malicious peer could easily generate false identities which could impact trust mechanisms.

How do you become a peer in a secure manner? The hallmark of pure P2P networking is that the *peers* are in control. If there is no single entity making admission decisions, and you still want access control, then some form of consensus is required to determine who can be admitted to the P2P network. In order to determine even a limited consensus, information about the present membership of the network must be maintained somewhere. For example, in order to achieve 75% consensus regarding the admission of a peer, you'll need to know the current total number of peers.

Saxena[15] describes a generic admission process:

1. A prospective peer M_{new} obtains the group charter out of band and then the information of current group size from either the Group Authority(GAUTH) or some bootstrap node; which is a designated node that has information on the membership of the group.
2. M_{new}, initiates the protocol by sending a join request message to the group. This message, signed by M_{new}, includes M_{new}'s public key certificate and the target group name.
3. Upon receipt of a join request, a group member first extracts the sender's public key certificate and verifies the signature. If a voting peer approves of admission it replies with a signed message. Several signature schemes can be used for this purpose.
4. Exactly who issues the Group Membership Certificate(GMC) for M_{new} depends on the security policy. If the policy stipulates using an existing GAUTH, once enough votes are collected (according to the group charter), M_{new} sends GAUTH a group membership certificate request message. It contains PKC_{new}, group name, and the set of votes collected in Step 3.

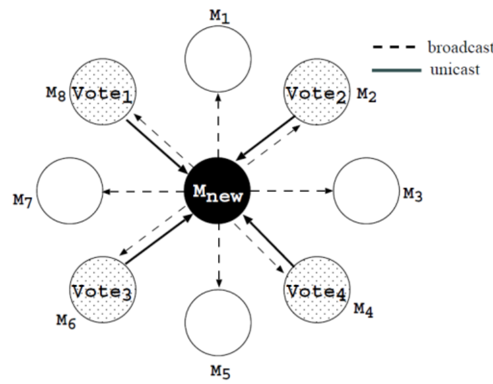

Figure 5 Secure Admissions Process [15]

The new member can then prove membership to another party by signing a message.

5 Anonymity

The authors found references [14][24][26] that suggest anonymity is a desirable characteristic of P2P networks. Anonymity supports both privacy as well as freedom of speech. To what extent does it support P2P Security? According to Lee [26], "Anonymous P2P provides enough anonymity such that it is extremely difficult to find the source or destination of a data stream." Using this definition, anonymity impacts Trust Management; which encourages peers to behave for the good of the overall network. The Trust Management paradigm suggests rewarding peers for good behavior and punishing them for bad behavior. If your peers remain anonymous, there is no mechanism to build reputations and "know your peers." Anonymity might also defeat the secure admission process for the same reason because a new peer is admitted by consensus. Anonymity can also make prosecution difficult in copyright and pirating cases. Even though anonymity supports freedom of speech and can prevent certain attacks, it interferes with other mechanisms used for the overall security of the P2P network.

6 Conclusion

The same security practices that apply to client/server computing also apply to P2P computing. If you use the Internet, you should follow computer security "best practices" in order to protect yourself. These practices include, but are not limited to:

- Don't open suspicious or unrecognizable emails
- Don't click on links within emails unless you're absolutely sure the source is legitimate;
- Use anti-virus software that is up-to-date.

However, because of the structure of the overlay network and the general ease of becoming a peer, anyone who uses P2P networks must be extra vigilant. All the security precautions discussed here will play a pivotal role in P2P computing security. Technologies that enable users to collaborate so easily will provide efficiency and convenience and at the same time introduce new challenges to secure computing.

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SESSION

DATA PRESENTATION

Chair(s)

TBA

Trustworthy Distributed Search and Retrieval over the Internet

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Abstract— *This paper describes iTrust, a novel distributed search and retrieval system that provides trustworthy access to information over the Internet. Nodes with information to distribute transmit their metadata to nodes that are selected at random from a set of participating nodes. Similarly, nodes seeking information distribute their requests to nodes that are selected at random from the set of participating nodes. When a node receives a request, the node tries to match the metadata in the request with the metadata that it holds. If the node has a match, it supplies a URL for the information to the requesting node, which then retrieves the information from the source node. The paper describes our implementation of iTrust, and provides a performance evaluation of iTrust, based on both analysis and simulation using our implementation. Distribution of metadata and requests to relatively few nodes suffices to achieve a high probability of a match.*

Keywords: trustworthy distributed Internet search retrieval

1. Introduction

Our modern world relies heavily on the ability to publish, search for, and retrieve information over the Internet, which has created a highly distributed information society, distributed in both the sources of information and the uses of information. For reasons of efficiency and scalability, conventional search and retrieval over the Internet employs centralized search engines.

Unfortunately, centralized Internet search engines can be tampered with easily by their administrators to bias the results, concealing or censoring information. The experience of history, and even of today, indicates that we cannot rely on centralized Internet search to remain unbiased forever. Perhaps, the moment at which we are most dependent on our ability to communicate over the Internet is also the moment at which centralized Internet search is most likely to be compromised. It is important to ensure that a trustworthy distributed search and retrieval system for the Internet is available when it is needed, even though a user normally uses a conventional centralized search engine.

The iTrust system, described in this paper, is a novel distributed search and retrieval system that provides access to information over the Internet. The iTrust system involves distribution of metadata and requests, matching of requests and metadata, and retrieval of information corresponding to metadata. iTrust has no centralized mechanisms that can be tampered with easily by a small group of administrators. iTrust is inevitably more costly in bandwidth, processing and storage than a centralized search engine. Individuals who are concerned about a risk of censorship ought to find that cost acceptable.

The iTrust system is deployed on a set of participating nodes in the Internet (also referred to as the membership). iTrust distributes both metadata that describes information, and requests for information, to a random subset of the participating nodes in the Internet. Because the metadata and the requests are distributed to nodes that are chosen at random from among all of the participating nodes, no one node or small group of nodes can suppress or censor information.

In the iTrust system, source nodes produce information and publish that information to make it available to other participating nodes. The source nodes create metadata keywords for their information, and communicate that metadata, together with a URL, to a subset of the participating nodes that are chosen at random, as shown in Figure 1.

Requesting (querying) nodes generate requests (queries), containing metadata keywords for information that they seek to retrieve. The requesting nodes distribute their requests to a subset of the participating nodes that are chosen at random, as shown in Figure 2.

If a participating node receives a request, it compares the metadata in the request with the metadata that it holds. If the metadata match, which we call an encounter or a match, the matching node returns to the requesting node the URL that the source node included with the metadata, as shown in Figure 3. The requesting node then uses the URL to retrieve the information from the source node.

The random distribution of the metadata and the requests achieves a high probability of a match, even when the metadata and the requests are distributed to relatively few nodes. Moreover, the probability of a match remains high even when some of the participating nodes (even some of the randomly chosen nodes) are subverted or non-operational.

The rest of this paper is organized as follows. Section 2 describes the implementation of the iTrust system. Performance evaluation results, based on both analysis and simulation using the iTrust implementation, are presented in Section 3. Section 4 presents related work, and Section 5 presents conclusions and future work.

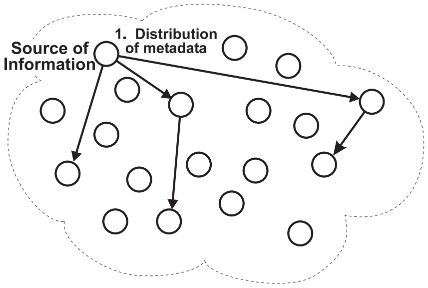

Fig. 1: A source node distributes metadata, describing its information, to randomly selected nodes in the membership.

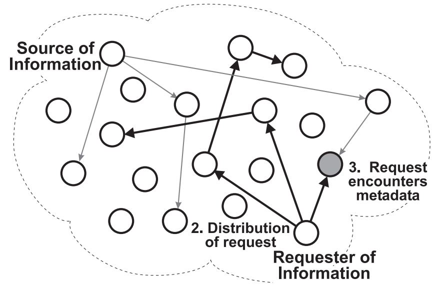

Fig. 2: A requesting node distributes its request to randomly selected nodes in the membership. One of the nodes has both the metadata and the request and, thus, an encounter occurs.

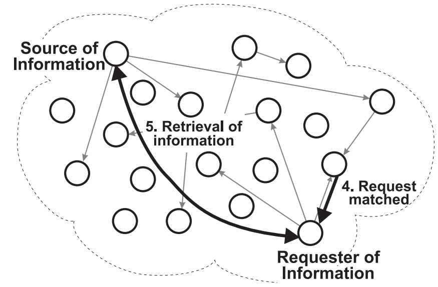

Fig. 3: A participating node matches the metadata and the request and reports the match to the requester, which then retrieves the information from the source node.

2. The iTrust System

The iTrust system on a node consists of three distinct components that interact with each other to distribute metadata and requests and to retrieve information (resources). Figure 4 shows the three components: the Web server foundation, the application infrastructure, and the public interface. Arrows on connecting lines indicate the direction of information flow. The following subsections describe these three components and their interactions.

2.1 Web Server Foundation

The basis of the current implementation of iTrust is the Apache Web server compiled with several PHP standard modules and library extensions. The Web server foundation component contains no custom code; all software is used as is, which enables rapid node deployment. iTrust utilizes various standard modules, including the session and logging modules described below.

The session module allows tracking of users on each node, so that multiple users can interact with the same node at the same time in a convenient manner (*i.e.*, without having to re-enter the same data on each Web page load). For example, session variables persist between multiple Web page fetches and between multiple resource retrievals. However, all session variables are purely for the convenience of the user, and a careful user may safely turn off session tracking (with only a minor inconvenience of re-entering certain data occasionally). In either case, all session data are deleted when the session (the Web browser window) is closed; there is no ability to identify a given user in subsequent sessions.

The logging module is enabled only for debugging and simulation, and can be disabled at any time by the node administrator. There is no direct relationship between the logging and session functions, *i.e.*, a user's actions cannot be tracked simply by viewing access logs (unless, of course, only one individual ever uses the node). The log file is written to disk but, optionally, may be automatically emailed to the node administrator. In the case where there are multiple nodes on the same computer, all of the nodes share the same log file and prefix each log entry with a unique node identifier.

iTrust also utilizes compiled-in modules, including cURL, SQLite, and the PHP Extension Community Library (PECL) for HTTP, as described below.

The cURL functions are used primarily for inter-node communication and resource-specific actions. When a resource is added to a node, a call may be made to that resource's URL to scan for metadata automatically. cURL automatically follows HTTP redirects and resolves file dependencies (such as HTML frame sources and image sources). Both the fetched text and the fetched images are accessible to the Java jar files, as described below.

SQLite is used for all administrative information such as node, metadata and resource information. For example, the node membership is stored in a database table, and the relationships between the metadata and the resources are stored in a normalized table. SQL constraints enforce several fundamental iTrust features, such as non-duplicate node addresses in the membership and unique resource URLs. Use of SQLite as a PHP module, instead of MySQL or PostgreSQL servers, aids with the rapid deployment of iTrust nodes. iTrust works on any reasonably modern Web host, because the file-as-a-database model of SQLite requires only minimal local write privileges.

The PHP Extension Community Library (PECL) for HTTP is an external compiled-in module used for inter-node search and metadata queries. A requesting node may use PECL HTTP to send a POST statement to a potential source node to search for the metadata that match the user's metadata query.

2.2 Application Infrastructure

The key iTrust methods reside in the application infrastructure; indeed, all of the node- and resource-related functions exist in this component. The infrastructure is divided into three parts: metadata-related functions, node- and resource-related functions, and Java jar files. All parts interact with the Web server foundation, whereas only some functions are exposed to the public interface component.

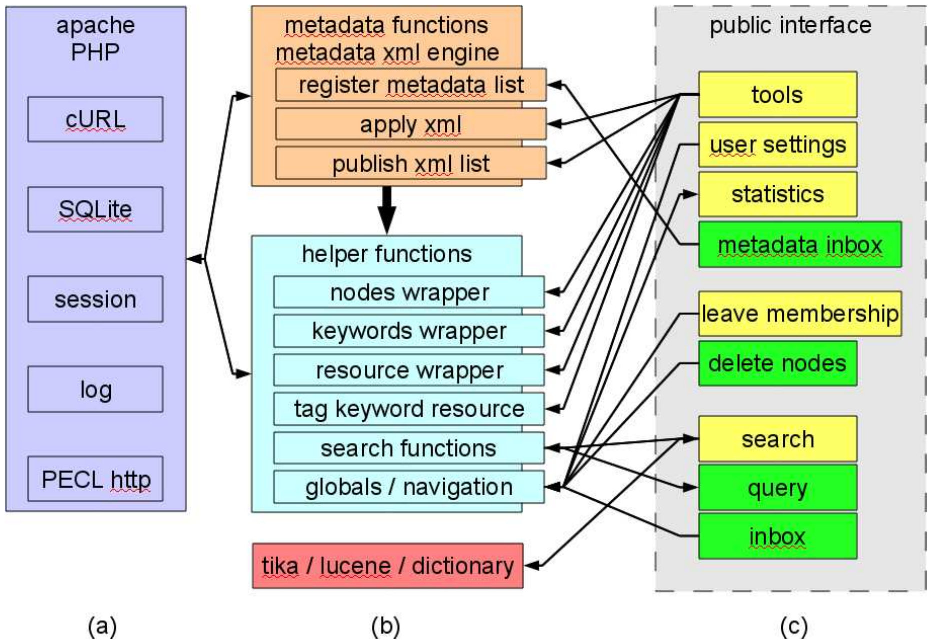

Fig. 4: The iTrust system, which comprises (a) the Web server foundation, (b) the application infrastructure, and (c) the public interface.

The creation and distribution of metadata, both internal and inter-node, are handled by the metadata-related functions. A node generates metadata from existing resources by invoking the metadata XML engine, which exhaustively scans all resources and creates an XML list describing the relationship between the metadata and the resource. Other metadata-related functions deal with the distribution of the XML list to other nodes, or with the receipt of XML lists distributed by other nodes. In the latter case, the received XML lists are scanned, and the metadata are inserted into the current node. In this way, the metadata are replicated among participating nodes.

Node- and resource-related functions, also known as helper functions, deal with bookkeeping tasks. These functions include functions that insert nodes into the membership, insert keywords into the database, and upload or fetch resources. Resources can be tagged with metadata manually by the user, or they can be automatically scanned for metadata, depending on the user's preferences. Node querying and query relaying are also handled by the helper functions (mostly through the use of PECL HTTP). All user variables (per session) and global administrative variables are stored.

Java jar files are used to generate metadata quickly and easily, and to provide the user with many conveniences. Apache's Tika and Lucene packages are used to generate metadata from resources automatically and efficiently, in the case where the user chooses not to generate metadata manually. The WordNet dictionary is used to provide the user with functions, such as spell checking and synonym suggestions.

2.3 Public Interface

The public interface, through which the users and the system administrator interact with iTrust, is divided between human and computer interfaces. Computer interfaces (dark boxes on the right in Figure 4) handle all inter-node communication such as queries, resource distribution, and metadata list distribution. All of the other interfaces (clear boxes on the right in Figure 4) are human-oriented and consist of PHP driven HTML Web pages; in fact, all human interaction with iTrust is through Web pages.

Administration is performed through the tools Web pages and other Web pages. Tools allow an administrator to add nodes or metdata keywords using simple HTML form text boxes. Adding resources requires uploading a file (form file input) or providing a URL (form text box input). User settings and statistics Web pages provide feedback to the administrator about the membership size, resource count, *etc*. An administrator may generate and distribute metadata XML lists or update the participating nodes' metadata lists. An administrator may also request that a node be removed from a node's membership. In this case, the request is activated through a human interface, and the request is distributed through the iTrust network using computer interfaces.

The most used feature of iTrust is the human interface for searching, where a user can enter a search query to request a resource. The query is sent from the current node to participating nodes using computer interfaces in a simple inbox-type fashion. Participating nodes read their inbox for queries, send back a response if there is a match, and independently decide whether to relay the query.

3. Performance Evaluation

In the performance evaluation, we consider the probability of a match, using both analysis and simulation based on our implementation of iTrust. We assume that all of the participating nodes have the same membership set. In addition, we assume that the Internet is reliable and that all of the participating nodes have enough memory to store the source files and the metadata. We randomly select nodes without repetition from the membership set for distribution of the metadata and requests. If a node receives a request and it holds the metadata that matches the metadata in the request, we say that the node has a match.

3.1 Probabilistic Analysis

First, we consider the probability that a node has a match, when all of the participating nodes are operational. Then, we consider the probability that a node has a match, when some of the participating nodes are not operational.

3.1.1 Probability that a node has a match when all of the nodes are operational

In an iTrust network with a membership of n nodes, we distribute the metadata to m nodes and the requests to r nodes. The probability p that a node has a match then is:

$$p = 1 - \frac{n-m}{n}\frac{n-1-m}{n-1}\dots\frac{n-r+1-m}{n-r+1} \qquad (1)$$

Equation (1) holds for $n \geq m+r$. If $m+r > n$, then $p = 1$. The formula is obtained as follows.

If $n \geq m+r$, first we find the probability q of no match on any of the r trials at the r nodes to which the requests are delivered. The probability of no match on the first trial is $\frac{n-m}{n}$. The probability of no match on the second trial is $\frac{n-1-m}{n-1}$, and so on. The probability of no match on the rth trial is $\frac{n-r+1-m}{n-r+1}$. Thus, the probability q of no match on any of the r trials is:

$$q = \frac{n-m}{n}\frac{n-1-m}{n-1}\dots\frac{n-r+1-m}{n-r+1} \qquad (2)$$

and the probability p of a match on one or more of the r trials is:

$$\begin{aligned} p &= 1-q \\ &= 1 - \frac{n-m}{n}\frac{n-1-m}{n-1}\dots\frac{n-r+1-m}{n-r+1} \qquad (3) \end{aligned}$$

If $m+r > n$, then the subset of nodes to which the request is delivered and the subset of nodes to which the metadata are delivered intersect in at least one node and, thus, $p = 1$.

3.1.2 Probability that a node has a match when not all of the nodes are operational

If x represents the proportion of the n nodes that are operational (and, thus, $1-x$ represents the proportion of the n nodes that are not operational), then the probability p that a node has a match is:

$$p = 1 - \frac{n-mx}{n}\frac{n-1-mx}{n-1}\dots\frac{n-r+1-mx}{n-r+1} \qquad (4)$$

Equation (4) holds for $n \geq mx+r$. If $mx+r > n$, then $p = 1$. This formula is obtained as follows.

If $n \geq mx+r$, then the probability of no match on the first trial is $\frac{n(1-x)+(n-m)x}{n} = \frac{n-mx}{n}$. The probability of no match on the second trial is $\frac{(n-1)(1-x)+(n-1-m)x}{n-1} = \frac{n-1-mx}{n-1}$, and so on. The probability of no match on the rth trial is $\frac{(n-r+1)(1-x)+(n-r+1-m)x}{n-r+1} = \frac{n-r+1-mx}{n-r+1}$. Thus, the probability q of no match on any of the r trials because all of the r nodes that receive the request are not operational or do not hold the metadata is:

$$q = \frac{n-mx}{n}\frac{n-1-mx}{n-1}\dots\frac{n-r+1-mx}{n-r+1} \qquad (5)$$

and the probability p that one or more of the r nodes that receives the request is operational and has a match is:

$$\begin{aligned} p &= 1-q \\ &= 1 - \frac{n-mx}{n}\frac{n-1-mx}{n-1}\dots\frac{n-r+1-mx}{n-r+1} \qquad (6) \end{aligned}$$

If $mx+r > n$, then the subset of nodes to which the request is delivered and the subset of nodes to which the metadata are delivered intersect in at least one node and, thus, $p = 1$.

3.2 Simulation Based on Implementation

Using our implementation of iTrust described in Section 2, we performed simulation experiments to validate Equations (1) and (4). In our simulation, we used libCURL (which is a free client-side URL transfer library for transferring data using various protocols) to collect the match probabilities.

Before we run our simulation program, we provide the following input to the program: the number n of nodes in the membership, the number m of nodes for metadata distribution, the number r of nodes for request distribution, and the proportion x of operational nodes.

First, the simulation program clears the data from the SQLite databases. Next, the program adds the nodes to the membership. Once all of the nodes are added to the membership, we call the source node to upload a file and the program then creates the corresponding metadata. Then, the simulation program randomly selects nodes for metdata distribution, and distributes the metadata to those nodes. Next, the program randomly selects the nodes for request distribution, and distributes the requests to those nodes. Then, the simulation program waits for 5 seconds. If one or more nodes has replied back to the simulation program, it means that there is a match and the program returns 1; otherwise, there is no match and the program returns 0.

We repeat the same process 100 times for the source nodes and correspondingly for the requesting nodes, and

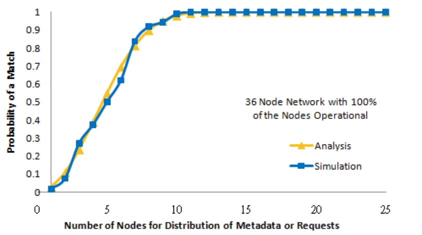

Fig. 5: Match probability vs. number of nodes for distribution of metadata and requests with 36 participating nodes, all of which are operational.

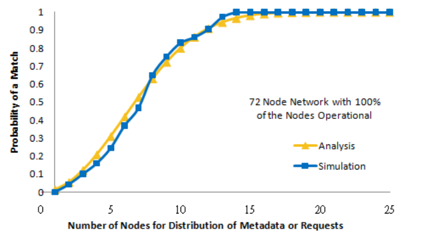

Fig. 6: Match probability vs. number of nodes for distribution of metadata and requests with 72 participating nodes, all of which are operational.

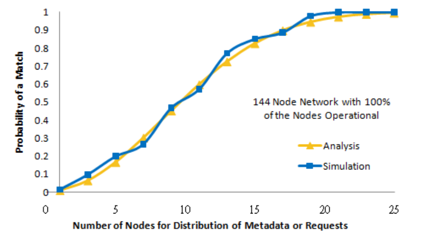

Fig. 7: Match probability vs. number of nodes for distribution of metadata and requests with 144 participating nodes, all of which are operational.

plot the mean results in our simulation graphs. We collected simulation data for 36, 72 and 144 participating nodes, when all of the nodes are operational. We also collected simulation data for 144 participating nodes when 100%, 80% and 60% of the nodes are operational.

3.3 Performance Evaluation Results

First, we consider the analytical and simulation results for the probability of a match, as the number of participating nodes increases. Then, we consider the analytical and simulation results for the probability of a match, as the proportion of non-operational nodes increases.

3.3.1 Increasing the number of participating nodes

Figures 5, 6 and 7 show both the analytical results and the simulation results for 36, 72 and 144 participating nodes, all of which are operational. The analytical curves obtained from Equation (1) are shown in the background (light curves), and the simulation curves obtained from our iTrust implementation are shown in the foreground (dark curves). We see from these figures that the simulation results are very close to the analytical results.

Figure 5 shows the match probability versus the number of nodes for distribution of metadata and requests in a network when 100% of the 36 nodes are operational. From the figure, we see that the probability increases as the number of nodes to which the metadata and requests are distributed increases. The reason is that the more nodes to which the metadata and requests are distributed, the more matches there are.

When the membership contains more nodes, the match probability asymptotically approaches 1 more slowly than for a membership with fewer nodes. That is, if we distribute the metadata and the requests to the same number of nodes, but the membership contains more nodes, the probability of a match is less than that for a membership with fewer nodes.

When we increase the membership to 72 nodes in Figure 6, the curves approach 1 more slowly than do the curves in Figure 5 for a membership containing 36 nodes. In other words, as we increase the membership, we must distribute the metadata and the requests to more nodes to obtain a higher match probability. Similarly, in Figure 7, when we increase the membership to 144 nodes, we see that the curves grow even more slowly than do the curves in the 36 node and 72 node networks.

Suppose now, for example, that we want to achieve a 0.98 match probability, in these three cases, which involve 36, 72 and 144 nodes all of which are operational. In the 36 node network, we need to distribute the metadata and the requests to only 10 nodes to achieve a 0.98 match probability. However, in the 72 node network, we need to distribute the metadata and the requests to 15 nodes to achieve a 0.98 match probability, whereas in the 144 node network, we need to distribute the metadata and the requests to 22 nodes to achieve a 0.98 match probability.

Thus, when we distribute the metadata and the requests to only a few nodes, the match probability is lower and the requester is unlikely to receive multiple responses from multiple matching nodes. When we distribute the metadata and the requests to more nodes, the match probability is higher and the requester will more likely receive multiple responses from multiple matching nodes. For a network with more participating nodes, the match probability grows more slowly than the match probability for a network with fewer participating nodes.

3.3.2 Increasing the number of non-operational nodes

Figures 8, 9 and 10 show both the analytical results and the simulation results for 144 nodes, when 100%, 80% and 60% of the participating nodes are operational, *i.e.*, when 0%, 20% and 40% of the participating nodes are non-operational. The analytical curves obtained from Equation (4) are shown in the background (light curves), and the simulation curves obtained from our iTrust implementation are shown in the foreground (dark curves). Again, we see from these figures that the simulation results are very close to the analytical results.

In Figures 8, 9 and 10, we see that the match probability curves increase as the number of nodes for distribution of metadata and requests increases. However, if we compare Figure 8 with Figure 9, we notice that the curves in Figure 8 asymptotically approach 1 faster than the curves in Figure 9. The reason is that in Figure 8 every node is operational, whereas in Figure 9 only 80% of the nodes are operational. Therefore, for distribution of metadata and requests to the

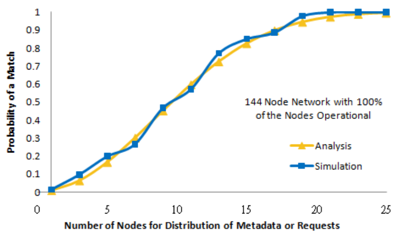

Fig. 8: Match probability vs. number of nodes for distribution of metadata and requests with 144 participating nodes where 100% of the nodes are operational.

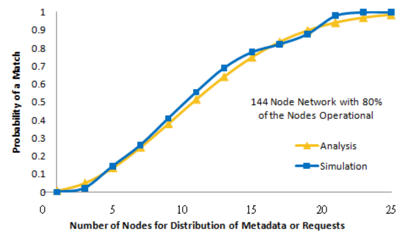

Fig. 9: Match probability vs. number of nodes for distribution of metadata and requests with 144 participating nodes where 80% of the nodes are operational.

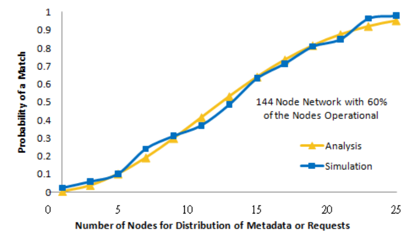

Fig. 10: Match probability vs. number of nodes for distribution of metadata and requests with 144 participating nodes where 60% of the nodes are operational.

same number of nodes, the probability of a match in Figure 8 is generally higher than it is Figure 9. Similarly, in Figure 10, where only 60% of the nodes are operational, the curves asymptotically approach 1 more slowly than the curves in Figures 8 and 9.

Suppose now, for example, that the metadata and the requests are distributed to 20 nodes, in these three cases, all of which involve 144 nodes. If 100% of the 144 nodes are operational, the probability of a match is 0.96. But, if 80% of the 144 nodes are operational, the probability of a match is 0.92, whereas if 60% of the 144 nodes are operational, the probability of a match is is 0.85, which is still quite good.

Thus, when all of the participating nodes are operational, the match probability is higher and the requester will likely receive multiple responses from multiple matching nodes. When there are fewer operational nodes, the match probability is lower and the requester is less likely to receive multiple responses from multiple matching nodes. Consequently, we must distribute the metadata and the requests to more nodes as the number of non-operational nodes increases, to obtain higher match probabilities. Nonetheless, iTrust retains significant utility even when not all of the nodes are operational, demonstrating that iTrust is quite robust.

4. Related Work

Centralized search engines for Internet search, such as Google [7], store metadata for information in a centralized index, and match queries containing keywords against the metadata at the central site. Centralized search engines are used commercially for Internet search, because they are efficient and scalable; however, they are vulnerable to manipulation by administrators. Centralized publish/subscribe systems also use a centralized index [4], against which queries are matched, raising the same issues of trust of the centralized site.

Risson and Moors [14] provide a survey of search in peer-to-peer networks, and Mischke and Stiller [12] provide a taxonomy of distributed search in such networks. Distributed publish/subscribe strategies are categorized as either structured based on managed overlay networks, or as unstructured based on gossiping and randomization. The structured approach is more efficient than the unstructured approach, but it involves administrative control with a consequent risk of manipulation. iTrust falls within the unstructured distributed search category.

Gnutella [8], one of the first unstructured networks, uses flooding of requests to find information. An extension of Gnutella involves supernodes [19], which improves efficiency but incurs some of the trust risks of centralized strategies. Freenet [2] is more sophisticated and efficient than Gnutella, because it learns from previous requests. In Freenet, nodes that successfully respond to requests subsequently receive more metadata and more requests. Thus, it is easy for a group of untrustworthy nodes to conspire together to gather most of the searches into their group, rendering Freenet vulnerable to subversion.

Sarshar *et al.* [15] combine random walks and data replication with a two-phase query scheme in a Gnutella-like network. BubbleStorm [16] replicates both queries and data, and combines random walks with flooding. GIA [1] combines biased random walks with one-hop data replication. Lv *et al.* [10] show that path replication and random replication are near-optimal in unstructured peer-to-peer networks. Like these systems, iTrust exploits randomization and replication.

Ferreira *et al.* [6] use a random-walk strategy to replicate both queries and data to the square root of the number of nodes in the network. Zhong and Shen [20] use random walks for requests, where the number of nodes visited by a request is proportional to the square root of the request popularity. Cooper [3] exploits search trees whose node degrees approximate the square root of the size of the network. Like these researchers, we can also exploit the square root function in iTrust.

Pub-2-Sub [17] is a publish/subscribe service for unstructured peer-to-peer networks of cooperative nodes. Instead of gossiping, Pub-2-Sub uses directed routing to distribute subscription and publication messages to the nodes. None of the above unstructured systems is particularly concerned with trust, as is iTrust. Rather, their objective is efficiency, which is not the primary concern of iTrust.

Systems for social networks [11], [13] exploit the trust that members have in other members, and route information and requests based on relationships among members. Social networks, like Facebook [5], are centrally administered and, thus, depend on benign administrators.

There exist a few systems for social networks that, like iTrust, are concerned with trust. OneSwarm [9] is a peer-to-peer system that allows data to be shared either publicly or anonymously, using a combination of trusted and untrusted nodes. OneSwarm is part of an effort to provide an alternative to cloud computing that does not depend on centralized trust. Its initial goal is to protect the privacy of the users, which iTrust does not aim to do. Quasar [18] is a probabilistic publish/subscribe system for social networks. The authors note that “an unwarranted amount of trust is placed on these centralized systems to not reveal or take advantage of sensitive information.” Thus, the trust objective of Quasar is quite different from that of iTrust.

5. Conclusions and Future Work

We have described the iTrust system, a distributed search and retrieval system for the Internet with no centralized mechanisms and no centralized control. iTrust is particularly valuable for individuals who fear that the conventional centralized Internet search mechanisms might be subverted or censored. The very existence of iTrust can help to deter attempts to subvert the conventional Internet search mechanisms, and can provide assurances to individuals that the information they seek will be available to them.

In the future, we plan to evaluate the effectiveness, efficiency, scalability, and reliability of the iTrust system in PlanetLab. In addition, we plan to conduct further probabilistic analyses and to investigate a range of possible attacks on iTrust and countermeasures to such attacks. Our objective for iTrust is a network in which individual nodes can detect a potential attack, and can adapt to an attack to maintain trustworthy distributed search and retrieval even when the network is under attack.

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User-friendly interfaces for Web GIS

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Abstract— *This paper describes a geospatial information system (GIS) that can be used for several mission-critical activities, such as management of risky activities, and urban planning. The main novelty of this system, stands on the ability to analyze and present data inside a 3D media-rich geo-referenced environment. The 3D user interface suggested is based on the idea of interoperable processing units that can be directly manipulated and linked to create complex process chains; thus, freeing the operator from unnecessary information or constraints on the analytic reasoning. Additionally, users can use this interface to visually program and deploy web accessible algorithms. This is an important aspect within collaborative activities, where is often necessary to create new bridges between different knowledge areas in response to unexpected situations.*

Keywords: geovisualisation, interactive 3D graphics, service oriented architectures, user-interface paradigms

1. Introduction

Today's society is becoming everyday more rapidly vulnerable to natural disasters due to growing urban populations, environmental degradation, lack of planning, management and preparedness. This scenario is bound to worsen in the coming years, since extreme weather-related events are increasing, particularly floods and droughts [1], [2].

Spatial planning and simulation, land management and monitoring provide the missing basis for taking precautions against catastrophes or hazards. In order to face this problem we can make use of a new generation of geospatial technologies that allow experts to capture, store, process and display an unprecedented amount of information about the environment and a wide variety of phenomena [3], [4]. However, the extremely wide range of data available from large and heterogeneous datasets (e.g. high-resolution satellite imagery and digital maps that can exceed several pentabytes; and economic, social and demographic information) poses a critical challenge in operators that need to move from data to information, to awareness, to pieces of knowledge [3].

1.1 Interactive Tools For Environmental Management

Environmental control is largely based on management and processing of very complex databases providing a variety of different spatial information on the environment [4]. This information has to be improved with information retrieved from monitoring systems. An ideal system should support operators with the largest possible set of automatic or semi-automatic processes which should be capable to support them in interpreting specific data patterns, potentially triggering early warnings.

When dealing with geospatial information, be this a physical map, a web-based mapping applications, or a more interactive 3D system, the first priority is to determine where the user is looking at and where the information is located. A user-friendly framework is essential when dealing with mapping applications [4] and the latest generation of 3D interactive applications are the ideal candidates to provide the most user-friendly experience. As consequence, in the last few years we have witnessed a large market response to this demand resulting in the development of 3D applications, engineered for use by the large public, such as Google Earth [5], Microsoft Virtual Earth [6], NASA World Wind [7] as well as several other 3D geospatial solutions designed for professional use. Those applications, often referred to as Virtual Globes or 3D Geobrowsers, have been designed to provide high usability in terms of navigation and data visualization.

The ease with which the user can interact with the data and the environment they are immersed in, is essential to facilitate exchange and dissemination of spatial information among stakeholders and government agencies.

1.2 From Data to Knowledge Through Interaction

Navigation and data access must be complemented with functions that enable operators to analyse interrelations between spatial information, data patterns and environmental effects. 3D interactive interfaces can provide a fundamental support to effectively understand complex spatial relationships among information and dataset, for instance to define the most appropriate evacuation routes or to plot an emergency plan [4].

However the very nature of this process, which is intrinsically cooperative -as diverse competences are involved- underlines the importance of developing user-friendly universal interfaces, essential to achieve short training time, ease of use and fast response. Further a user-friendly interface is essential to avoid programming of simulation procedures, which notably requires extensive training and time. The use of visual programming languages can ease the definition of complex simulation process.

The main contribution of this work is to show how 3D interfaces can be used to sustain abstract visual metaphors in combination with a human information discourse in order to enable detection of unexpected events within massive, dynamically changing information spaces.

The system presented provides interactive capabilities that allow intuitive control of the data available and, most importantly, of the various processing tools required by the operator. Furthermore, the system supports visual programming and deployment of complex simulation processes. Programming is performed through the use of visual expressions, spatial arrangements of text and graphic symbols, used both as syntactic elements as well as secondary components. Visual components can be manipulated by users in an interactive way thus making simulations easier and faster to build and debug. New algorithms can be visually created without any knowledge about the service and server architecture.

2. Related Work

A vast body of work on general principles in 3D navigation, interaction and visualization can be found in literature. With specific regards to the field of environmental control, 3D GIS have received considerable attention and the number of works found in the literature on this topic is constantly growing.

Card et al [8] have a vast work showing that when information is presented visually, efficient innate human capabilities can be used to perceive and process data. Information visualization techniques that amplify cognition by the increase human mental resources, can reduce search times, improve recognition of patterns, increase inference making, and increase monitoring scope.

Thorndyke and Hayes-Roth [9], as well as many others authors [10], [11], [12], [13], [14], have studied the differences in spatial knowledge acquired from maps and from their exploration. Darken and other authors [15], [11], [16], [13] have explored cognitive and design principles and how these are applied to large virtual worlds. Furnas explored view traversability and navigability for effective navigation through large data structures [17], [13].

In 2002, Zlatanova et al. [18] edited a survey of mainstream GIS software, where they reviewed a number of systems including: ArcGIS, Imagine VirtualGIS [19], PAMAP GIS Topographer [20] and GeoMedia Terrain [21]. The authors found that some initial steps forward had been made in terms of visualization of 3D spatial data, mainly through the extension of the map metaphor to 3D [22], [23]. Further advancements found by the authors comprise the inclusion of some main requirements such as multi-resolution and multi-view representations, real-time rendering, interactivity, and high visual quality [24]. This has been possible through the adoption of algorithms and data structures specifically developed for efficient visualization of terrains which had been extensively studied in the past (e.g., multi-resolution geometry modelling [25], [26] and multi-resolution texture modelling [27], [28]).

Further examples of existing 3D research prototypes include Terrafly [29], GeoVR [30] and TerraVisionII [31]. A noteworthy system, called GeoTime [32] proposes an interesting solution to the problem of integrating timeline events into interactive GIS. The Adelaide 3D GIS planning model [33] is an example of a 3D GIS which provides 3D visualizations of social and environmental data within a 3D cityscape [34]. Two other interesting applications are GeoZui3D [35] and VGIS [36]. GeoZui3D is a 3D marine GIS that supports real-time input and draped imagery. VGIS [36] is an integrated global GIS and visual simulation system [34] and finally Heidelberg 3D which is a client that allows the user to explore and analyse the 3D city and landscape models which are streamed by the W3DS server [37], [38].

These applications, besides showing clear bottlenecks in the manipulation of 3D data and generic 3D analysis [18], often lack even simple GIS function [34] such as querying, usually performed through the adoption of standard WIMP (Windows, Icons, Menus and Pointers) metaphor [3], [39]. This clearly produces a cognitive barrier that does not allow to visualize and manipulate geographical information in a realistic and natural manner [39].

3. Application Overview

The system presented in this paper allows users to interact with large 3D geo-referenced environments. The software has been developed on top of Java World Wind [7] APIs and it can be deployed as standalone application, applet or as Java Web Start [40].

Furthermore, the system provides support for the access and automation of multiple-step procedures known as workflows by providing a rich set of tools and mechanisms to combine a series of tools into a sequence of operations using models [41].

The types of tasks to be automated can be very diverse in their nature, ranging for example from being able to predict the path of wildfire, to analysing and finding patterns in crime locations, to predicting which areas are prone to landslides, to predicting flooding effects of a storm event. Workflows can be potentially very complex sequences of elementary operations necessary to model and analyse complex spatial relationships. For that reason, the interface allows performing operations on different datasets (such as a feature classes, raster images, or database tables) and to produce a new dataset as result. Additionally, the interface also supports common geoprocessing operations, including geographic feature overlay, feature selection and analysis, topology processing, raster processing, data conversion, projecting a dataset from one map projection to another, adding a field to a table, or creating a buffer zone around features. All these functionalities, which are provided as Web Processing Services (WPS), are interactively available to the user through the user interface.

Additionally, operators can create new WPS components through the system UI without having to worry about all the issues related with the creation or deployment of a web process service. The WPS component will be compiled and deployed, at the server side, in a completely transparent way.

4. 3D Interface for Web Processing

In a previous work [42], we described a server architecture using interoperable protocols such as Web Processing Service (WPS), Web Map Service (WMS) and Web Feature Service (WFS). In this paper, we describe how to benefit from that interoperability. The only information required is the address of a web service supporting geo-processes across the web (WPS). The system will automatically update the menu with all the new geoprocessing services discovered at that service address. Their category and name is automatically retrieved from the WPS metadata and are used to structure the menu layout.

4.1 Creating Complex Process Chains

Once the user clicks on one process, the corresponding 3D process can be placed on the desired position and dragged over the terrain. To maximize usability and readability, the interface automatically sticks to the nearest terrain point, and process icons are rendered as billboards - always facing the user. Besides that, interface occlusion rarely happens due to the level of detail and to the nature of the navigation. A process can be composed up to three distinct types of components: the process controller, input and output slots.

As visible in Figure 1, the process controller has four distinct buttons. The first button (starting from the left side) is used to remove the process from the 3D environment. The remaining buttons are respectively used to execute or stop the process, as detailed in the following sections, and to obtain a process description.

Fig. 1: Process representation

Every process operates as a black box that can receive input and transmit results to a further process via its output slots. Each input and output slots is automatically created from the process descriptor exposed by the WPS, and represented using distinct icons. In this way, it is immediate the identification of its name and mime type. Being completely automatic, this architecture scales very well in case of availability of a large number of services.

In the example of Figure 1 the slot at the left represents an output (double) that can be connected to a compatible input of any other processing unit. Input slots (right) can be connected to some output (left) from another process or connected to results of user inputs (for instance when the user selects an area within the virtual scene).

The icon associated with each slot turns green if there is some data available on it (see for instance the slot on right in Figure 1). An additional button on the right side of the slot can be used, depending on the data type, to save the XML content as a file, to visualize text in a window, to represent a 3D shape within the environment, to render an image on the terrain, etc.

To create a connection between two processing units, the user has to drag the input slot over the output slot (or vice-versa) as illustrated in Figure 2.

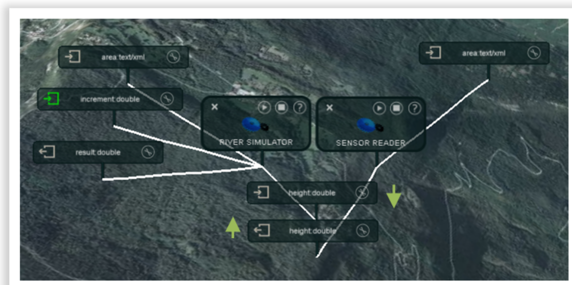

Fig. 2: Creating a connection

When the user drags an in-slot close to an out-slot they snap together and are replaced by the resulting slot shown in Figure 3. This last component is logically analogous to a bridge (both visually and practically). This way the user can create complex processing structures by connecting in- and out- bound connectors. If the user needs to break a connection, the task can be simply archived by clicking on the button indicated by the arrow on Figure 3. When this is done the process chain is broken and the original in- and out- slots become available to the user. As result, the system gets back to the state illustrated in Figure 2.

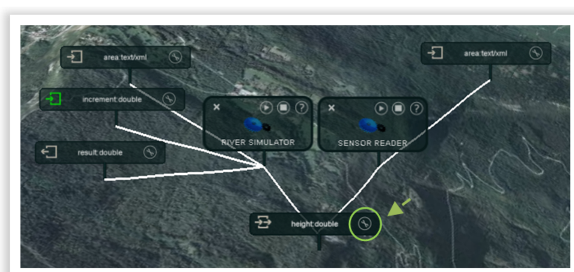

Fig. 3: Breaking a connection

As each block of the chain represents a processing service, the operators can perform very articulated web-service orchestration tasks in a completely transparent way, by just interacting with icons directly within the 3D representation of the scene. This approach has proven to be extremely user-friendly and it allows performing complex processing tasks with very little training.

Furthermore, processing functionalities can be constrained to specific areas of the territory. Therefore, users can interactively select within the 3D scene, a specific area containing the area of interest (see Figure 4). Once the process chain is created, it can be activated by pressing the relevant icon on the process controller. Most importantly, as the processing takes place at the server level, this is asynchronous by its very nature, meaning that the 3D client continues to respond in a fully interactive manner also when complex simulations are being executed.

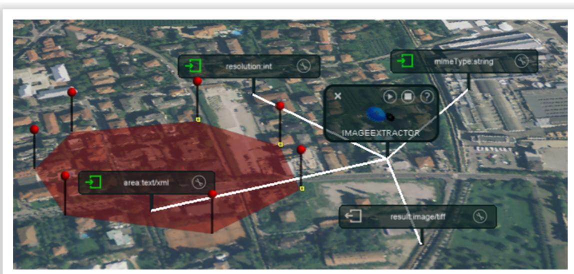

Fig. 4: Process chainning

4.2 Web Process Suggestion

The system has also the ability to suggest possible processes that can be used within the working scenario. This not only reduces search-time (supposing the process is known by the operator) but also informs the operator about new related processes, which may not have been available before. By using this feature, users can learn about new potential-useful algorithms without affecting their normal routine (Note that WPS protocol provides mechanisms to get information about processes). Besides that, the suggestion interface is only displayed when the user selects an input or output slot (see figure 5).

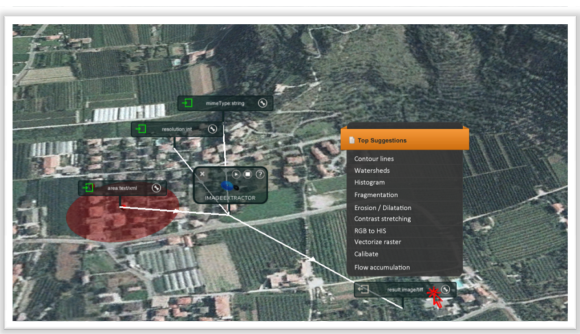

Fig. 5: Process suggestion

The algorithm used to suggest a list of processes is based on the following formula:

$$\omega(X) = \max_{10} \left\{ \frac{1}{g(\alpha) f(\alpha)} + f(\alpha)\chi(\alpha, X) \mid \forall \alpha \in \Omega(X) \right\} \quad (1)$$

where X represents the set of processes currently in use, Ω, the set of compatible processes (using type matching), $g(\alpha) \in [0, \ldots, 1]$, the usage's frequency of the process by the operator, $f(\alpha)$, the matching parameters percentage, and $\chi(\alpha, X)$, the semantic relation evaluator, which is described by the following equation:

$$\chi(\alpha, X) = \sum_{\beta \in X} 0.35\rho(\alpha, \beta) + 0.55\varphi(\alpha, \beta) + 0.10\phi(\alpha, \beta) \quad (2)$$

Where $\rho(\alpha, \beta)$ evaluates the relevance of data to be processed (e.g. complex geometry as advantage over numeric data), $\varphi(\alpha, \beta)$ gives a score to questions such as *Where*? *When*? and *What?* (In the example illustrated in Section 7, *Where* is the user selection, *When* is given by the simulation process and finally *What* is who is living there) and $\phi(\alpha, \beta)$ is a score for sharing the same computation group (e.g. simulation, image analysis).

5. Extending the WPS protocol

Currently, the WPS standard does not provide any support for hot deployment or source code retrieval. Yet, these are relevant features within collaborative activities. For that reason we will introduce and describe three new operations. The first operation is called *StoreNewAlgorithm* and it allows the user to deploy new algorithms into a web server. For this operation we have to specify a XML document that describes the process in the standard way (*ProcessDescription* element), and provides its respective source code. Eventually, the request can be associated to an expire time, after which the algorithm will be removed.

A parameter *IDGEN* should also be provided to describe how the process ID is defined, or how the response from the server is handled when the ID is duplicated (See Table 1). Note that the algorithm identifier should be unique because it is used to identify the process.

IDGEN value	Action
GenerateNew (default)	The WPS will generate a unique identifier for this algorithm.
UseExisting	In response to an operation *StoreNewAlgorithm* a WPS uses the value of the IDENTIFIER attribute as identifier for the algorithm. If the IDENTIFIER is already been used, the WPS respond with an exception.
ReplaceWhenDuplicated	A WPS client can request to the service to generate an IDENTIFIER in order to replace the identifier if duplicated. In this case, the WPS server will not respond with a duplicated IDENTIFIER exception.

Table 1: Possible IDGEN Values

The source code has to be included in the XML file, delimited by an element called *Source*, which is composed by a list of one or more sub-elements *File*. Each element *File* stores a source code block, as well as the programming language name, the original file name, and if it contains a main function.

The following example illustrates an example of a *StoreNewAlgorithm* request.

```
<?xml version="1.0"?>
  <wps:StoreNewAlgorithm
  version="1.3.0"
  service="WPS"
  idgen="ReplaceWhenDuplicated"
  xmlns="http://www.someserver.com/myns"
  xmlns:ows="http://www.opengeospatial.net/ows"
  xmlns:wps="http://www.opengis.net/wps"
  xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  xsi:schemaLocation="http://www.someserver.com/myns
    http://www.opengis.net/wps ../wps/1.3.0/WPS.xsd">

    <ProcessDescription
      processVersion="2"
      sourceAvaliable="true"
      statusSupported="false">
        ...
    </ProcessDescription>
    <Source>
      <File
        name="TestAlgorithm.Java"
        language="Java"
        main="true">
          R0lGODlhZABq ... base-64 ...
      </File>
      <File name="sample.bin">
        R0lGODlhZABq ... base-64 ...
      </File>
    <Source>
</wps:StoreNewAlgorithm>
```

The second operation proposed, extends the WPS protocol in order to provide access to source code. If users can have access to the source code, then it can easily be extended, fixed or improved.

However this feature may not be desired for all processes due to multiple reasons. Thus, it should only be possible to obtain the source code of those algorithms that specify the attribute *sourceAvaliable* as *true*, within *ProcessDescription*.

The user will receive an error message when isn't possible to retrieve the source code. The format in which is possible to receive the source code can also be specified through the parameter FORMAT.

An example of a request would be:

```
HTTP://www.someserver.com/path/wps?SERVICE=WPS&VERSION↩
    =1.3.0&REQUEST=GetAlgorithm&IDENTIFIER=TestProcess&↩
    FORMAT=XML
```

The last addition will be the operation to remove a process. This operation can be combined with *StoreNewAlgorithm* to perform update operations.

```
HTTP://www.someserver.com/path/wps?SERVICE=WPS&VERSION↩
    =1.3.0&REQUEST=RemoveAlgorithm&IDENTIFIER=↩
    TestProcess
```

Additionally servers can implement security measures through regular security mechanisms such as user authentication, in order to control who can deploy or remove new algorithms.

5.1 Protecting server from malicious code

Protecting servers with authentication mechanisms it is simply not enough, since they can't block users from uploading malicious algorithms. Therefore, new security mechanisms are required to address these new security flaws.

One of the mechanisms that can be used and ensure some good level of security is the use of the Java Security Manager. *SecurityManager* was mechanisms to monitor and control all the permissions of the running processes. In the most drastic case, it can deny access to any resource (be it local or remote) or the access to other objects/code.

6. Hot algorithm deployment

To facilitate the creation of new algorithms, a 2D widget is rendered in overlay. This strategy allows the operator to work simultaneously in 2D and 3D. Initially, the widget contains only one icon that represents the core algorithm. By clicking on it, the operator is able to write the algorithm according to one of the supported programming languages. When required, new input or output parameters can be added by pressing the button "plus". Whenever a new parameter is added, a new icon will become visible. This new icon can be used to define specific parameters such as name, type (input or output), data type and if applicable the default value. Depending on slot type, the icon will have different representations.

The user can inspect if the process has default values associated by checking the icon color, as previously mentioned - it will be green when some value is assigned (See example in Figure 6 - icon on the left).

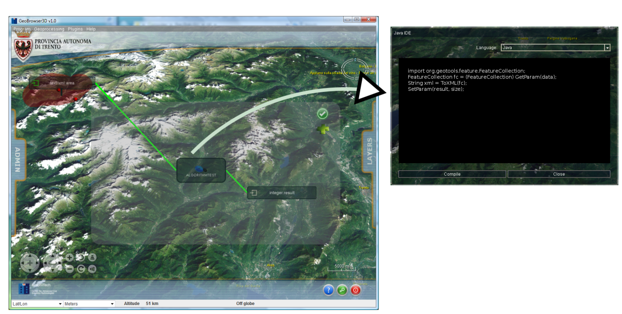

Fig. 6: Process Creation

These slots will became part of the 3D context when dragged outside the 2D widget. So, they can be connected to any selection or object, and that will automatically become the default value. As illustrated in Figure 6, the input was associated with an area on the ground, dynamically selected by the user.

A new window will be visible if the user selects the icon that represents the algorithm core. Within this new window it is possible to select the programming language, write the algorithm code and have a run-time diagnostic of programming errors. Java and Javascript are among the supported languages.

During the programming phase, the programmer has only to care about the algorithm and does not need to be aware of how it will work at the server side as a web service. Furthermore, to facilitate this abstraction, there are some macros that can dramatically simplify the code. Some examples are *FromXML(Object)* and *ToXML(Object)*, which can be used to convert XML into native objects and vice-versa. *GetParam(String)* on the other hand, can be used to obtain the value of one input parameter through its name, while *SetParam(String, String)* can be used to set the value for an output parameter.

When the entire process structure is ready, the user can write an abstract for each component (algorithm, inputs and outputs) as well as keywords that later will support catalogues against client's searches.

If the algorithm passes all tests, then it will be automatically published online, becoming at the same time immediately available to everyone.

When a newly added algorithm is selected from the menu, it will appear inside the 3D environment with all its predefined values (e.g. associated to a selection of some particular area).

7. Use Case

This section describes a practical use case and illustrates the potential of the interface. The use case consists in a simulation, where users have to raise the current river level and identify people potentially affected by the flooding.

As shown in Figure 7, a first process is used to identify and retrieve water levels from sensors, accessible through web services and within a given region. The relevant sensor data is passed as output onto a second processing unit (2nd). The second processing unit will create a flood using a detailed terrain model available at the server side, and delimited by a selected area passed as further input. The output of this process, is passed as input together with the affected area to a final processing unit (3rd), which retrieves from a spatial database the list of streets within the affected areas. Then invokes a public white pages web service to identify the telephone numbers of those potentially affected by the flood. Their address is in turn passed to Yahoo Geocoding API that returns the corresponding spatial location, used to render a green pin in the scene. Further, operators can elaborate evacuation plans taking in account attributes such as people distribution, roadworks, optimal paths, etc. using similar processes.

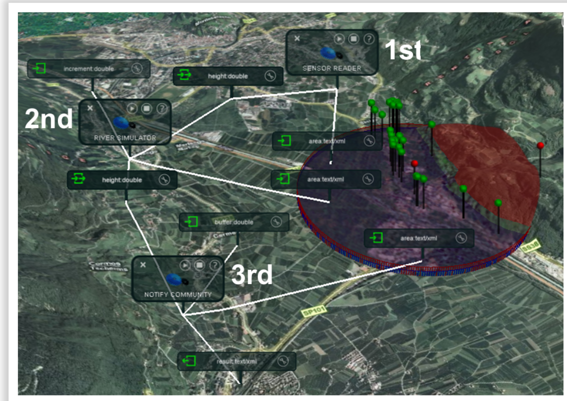

Fig. 7: Flooding Simulation

7.1 Testing And Validation

The usability test, based on the ISOMETRIC test [43], has involved 10 users with different expertise. One was younger than 25, six aged between 25 and 35, two between 35 and 45 and the last one older than 45. 70% of the sample were males, seven were students and the remaining post-graduates with no specific experience with 3D user interfaces. Only one of the users had previous knowledge in GIS systems.

Generally speaking users have clearly indicated that the functions implemented in the software support them performing their work (arithmetic mean of 5.0/5.0) and they are not forced to perform any task not related (arithmetic mean of 5.0/5.0).

In terms of self descriptiveness the test has shown that the users were satisfied with the interface. However it is clear that future effort is required to further improve the system by the possibility to easily retrieve information about a certain entry field (arithmetic mean of 3.2/5.0). The interface has proved quite clear to use (arithmetic mean of 4.1/5.0) with the user perceiving clearly the concept transmitted by the interface (arithmetic mean of 4.0/5.0).

The application can be considered appropriated for learning, since users have indicated that they did not need long time to use the software (arithmetic mean of 3.5/5.0), and it is supposedly easy to relearn to use after a lengthy interruption (arithmetic mean of 4.4/5.0). Finally, users stated that there isn't much need to remember many instructions to properly use the software (arithmetic mean of 2.5/5.0).

8. Conclusion

Today interoperability in the domain of geo-visualisation and geovisual analytics is starting to become a reality thanks to several international harmonization efforts trying to solve the problem by offering standards for management of geographical information. However, several issues regarding data access and management, 3D visualization and analysis need still to be tackled.

The work described in this paper provides an example of 3D interfaces applied to the context of visualization, analysis and processing of geographical information, and illustrates that future developments should maximize usability to ensure a profound understanding of the respective information space. Specifically, this framework gives the user the opportunity to work with any information available within the system by simply drag-and-drop connectors. This avoids not only the need to memorize object ids, which are not always known but also the need to perform unnecessary searches. Another useful contribution is an extension of the WPS protocol in a way that allows users to build new processes from scratch and share them on-fly with other users.

Besides that, our experiments have indicated that this user interface can be considered suitable and self-descriptive for a given task, breaking the complexity of most standard GIS systems and allowing any decision maker to make use of it, with virtually no training.

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A System of Adaptive Keyword Search in XML

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Abstract— *Existing LCA-based approaches for XML keyword search have several fundamental flaws, among which, the most important one is that the search results are eternally determined nonadjustable. In this paper, we present a system of adaptive keyword search in XML, called AdaptiveXKS. It employs a novel and flexible result model and a corresponding scoring function, within which, four critical metrics are weighted and the associated parameters can also be modified. Through the interface the system administrator or the users can adjust some parameters according to their search intentions. One of three searching algorithms could also be chosen freely in order to catch specific querying requirements, which reveals the adaptiveness as the biggest merit of this system. Extensive experiments show that AdaptiveXKS outperforms existing keyword search systems in both precision and recall.*

Keywords: Keyword Search; XML; LCA

1. Introduction

XML Keyword Search is a user-friendly information discovery technique, which attracts many interests these years. Different with keyword search over flat documents, the search object is a single XML document/database with structure information inside and the results are supposed to be fragments of it containing keywords. Since it is difficult and sometimes impossible to identify users' intentions through keywords, it is really a hard task to determine which fragments should be returned. As a matter of fact, this is still an open issue of keyword search on structured and semi-structured data.

To define the results of XML keyword search many valuable models are proposed, and the most popular ones are the Smallest Lowest Common Ancestor (SLCA) model [1] and its variants ([2][3][4][5][6][7][8][9]), all of which are called as the *LCA-based models* in this paper. Many work [4][5][9] mention that former LCA-based result models have *false positive* or *false negative* problems. Interestingly, their remedy approaches conflict with each other and counterexamples can always be found to testify that the two problems still happen.

We employ the original examples from former researches [4][5][9], which are illustrated in Figure 1 as three separate XML trees, to explain the issues about LCA-based approaches. In these trees keywords are marked in bold and only important nodes are identified by numbers. Besides, we use $node_i$ to denote the node with the number i in any of the trees. Consider keywords {"XML", "David"} issued on the XML document in Figure 1(a), apparently SLCA method can find two SLCA nodes: $node_7$ and $node_{17}$, and the subtrees rooted in them will be returned as the final results.

MLCA [6] method and GDMCT [3] method should agree with the answers because they both define a result model very similar to SLCA. But, ELCA [2][9] method will claim that several reasonable results are missed. For instance $\{node_3, node_{25}\}$ and $\{node_{16}, node_{23}\}$, which indicate a conference and one of its sessions respectively. It seems ELCA model has fixed the false negative issue of SLCA model and thus can find results perfect enough. However, based on the example illustrated in Figure 1(b) Li et al. [5] claim that both SLCA and ELCA models are suffering from the false positive problem. Suppose {"XML", "John"} are the keywords, either SLCA or ELCA method returns $\{node_6, node_{15}\}$ as the only result which is thought as meaningless in [5]. Actually, more examples can be found to support this point of view because in some cases the keyword nodes in a result could be really far from each other in the tree. Rather than implement semantics inference to improve the results as XSeek [10] does, Li et al. introduce a simple rule to filter all the ELCA nodes. For any two keyword nodes n_i and n_j in an ELCA result, if two nodes n'_i and n'_i which are from the paths "$lca(n_i, n_j) \to n_i$" and "$lca(n_i, n_j) \to n_j$" respectively satisfy that n'_i and n'_j have the same elementary type, then the result is unqualified. Accordingly, $\{node_6, node_{15}\}$ is not a qualified result because $node_4$ and $node_9$ have the same elementary type. After the filtering, the left ELCA nodes are called the Valuable Lowest Common Ancestor (VLCA) nodes.

The rule of VLCA method is actually kind of overstrict, and more relaxed criteria could be used such as the LCA have to be low or the compactness should be high. Kong et al. present a counterexample in [4]. To search the keywords {"Liu", "Chen", "XSeek"} in the tree from Figure 1(c), $\{node_4, node_6, node_8\}$ is a reasonable answer yet will be eliminated by VLCA method because $node_3$ and $node_5$ have the same elementary type "author". Kong et al. [4] also propose a concept called Related Tightest Fragments (RTF) as final search results, which is equal to representing the results of ELCA method in MCTs. Obviously, it keeps the vague problem of false positive results being existed.

After further analysis we explore several defects of all previous LCA-based models. The most important one is that the search results are eternally determined and nonadjustable, though different users probably have distinct intentions with

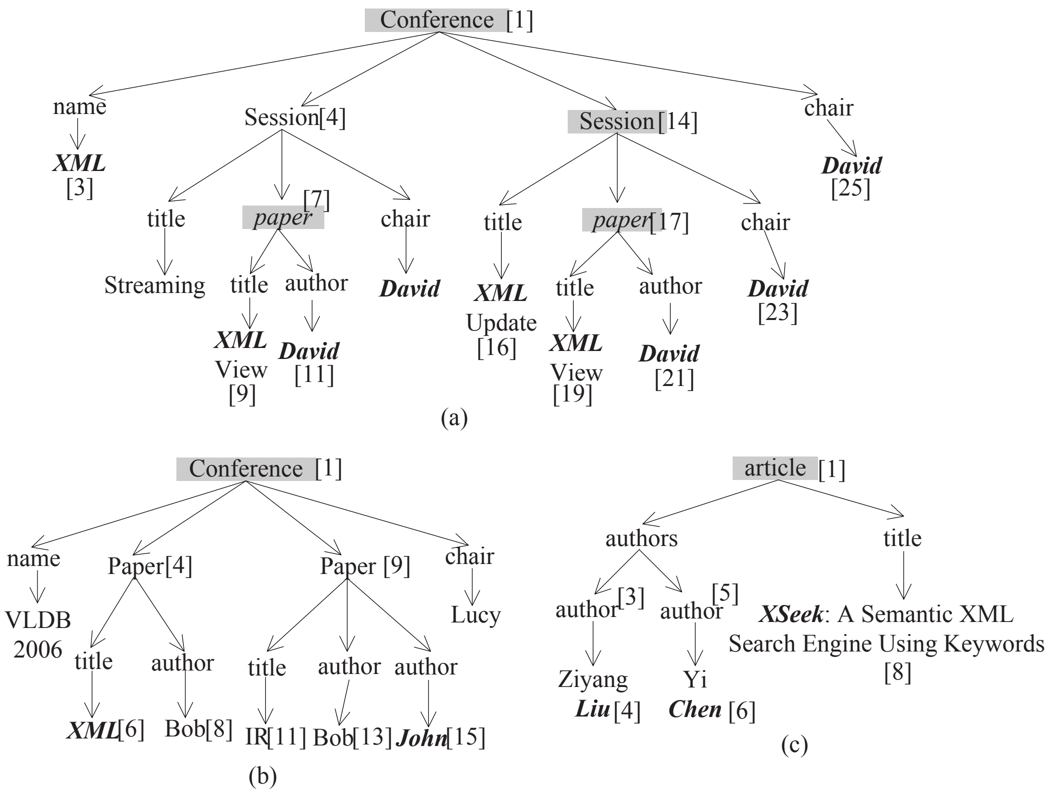

Fig. 1: Original examples from LCA-based approaches

the same keywords. With any LCA-based approach neither the administrator nor a user has a choice to improve their search results. Secondly, there is a lack of universal criteria to judge whether a result is qualified or not. After SLCA [1] is proposed, arguments have been aroused due to its inherent *false positive* and *false negative* problems. Later researches try to fix these issues by introducing other variations of SLCA, or by providing some rules. However, counterexamples can always be found to testify that two problems still happen. Thirdly, in most LCA-based models, some important features are missed, for example, the compactness and the size. Lastly, some useful information is omitted in their results. Actually all the LCA-based models only serve the best results (in which all the specified keywords have to be contained) and refuse to organize and return the second-best ones.

In this paper, we present an adaptive and effective XML keyword search system AdaptiveXKS that can avoid all the aforementioned defects. Search results are evaluated according to a pre-defined scoring function which considers several critical metrics. Moreover, all the results including those second-best ones with relatively lower ranking scores are regarded as useful to users. Furthermore, through the interface, the administrator or the users can arbitrarily modify the parameter values employed in the scoring function in order to catch specific requirements.

The contributions of our work include:

- We propose a flexible evaluation approach based on an adjustable scoring function, which considers sufficient features, each of which is weighted and so that can be adjusted as necessary.
- We have designed and developed an XML keyword search system AdaptiveXKS that is adaptive and effective. For each query, users can select one of the three algorithms developed to search results, based on their specific querying requirements.
- Extensive experiments show that AdaptiveXKS generates results with improved precision and recall over those LCA-based approaches.

The remainder of the paper is organized as follows. Section 2 discusses the related work. Section 3 introduces the result model and scoring function of the AdaptiveXKS system. The system implementation is presented in Section 4. Section 5 studies the experiments and Section 6 summarizes the paper.

2. Related Work

Extensive research has been conducted in XML keyword search, which can be done in either tree data model or the digraph data model. And the ranking of query results is another important issue in this area.

Keyword Search on XML Tree Databases. We have briefly reviewed the most important LCA-based result models for XML keyword search, like SLCA[1], ELCA[9], VLCA[5], RTF[4], GDMCT[3], MLCA[6] in the Section 1. All of these works model the underlying XML data as a rooted, labeled, unordered tree. The most important one of them is SLCA. In SLCA model a result is defined as a subtree that: (1) the labels of whose nodes contain all the keywords, (2) none of its subtree satisfies the first condition except itself. The root of such a subtree is called a SLCA node. It's recognized that SLCA model is definitely not a perfect one. All the ELCA nodes can be retrieved through a straightforward two-step process: (1) find all the SLCA nodes, halt the process if if there isn't any; (2) remove all the SLCA nodes along with the subtrees rooted in them, then turn to the first step. It can be easily proved that the union of all the SLCA nodes obtained each time in the first step is indeed the set of ELCA nodes. For the sake of space, we will not cover other LCA-based models in detail. In addition, in other tree data model, XSeek[7] implements semantic inference to improve the results through the adoption of entities and keyword match pattern. It also works without schema information.

Keyword Search on XML Graph Databases. XKeyword[11], which is built on a relational database, is a typical one implementing XML keyword search on XML graph databases. It builds a set of keyword indices along with indexed path relations that describe particular patterns of paths in the graph. During query processing, plans that use a near optimal set of path relations are developed to efficiently locate the keyword query results. Besides, XKeyword also use the graph's schema for optimization. BLINKS[12] adopts a completely different approach. It partitions a data graph into blocks and then builds a bi-level index, which stores summary information at the block level to initiate and guide search among blocks, and more detailed information for each block to accelerate search within each block. BANKS [13] uses bidirectional expansion heuristic algorithms to search as small portion of graph as possible. Different from the above approaches, EASE[14] models not only semi-structured data, but also unstructured and structured data as graphs, thus can deal with indexing and querying large collections of heterogeneous data.

Ranking of Search Results. Ranking schemes have been studied for keyword search on XML documents. XRank[2] employs such a rank model that extends Googles's PageRank to XML element level to rank all LCA results, which takes into account result specificity, keyword proximity and hyperlink awareness together. In addition, extended tf-idf techniques stemming from traditional information retrieval have been adopted in many ranking schemes. However, tf-idf-based ranking models can easily ignore the inherent hierarchical structural information of XML data, which may always indicate some important semantics. RACE[15] is a ranking mechanism to rank compact connected trees, by taking into consideration both the structural similarity and the textual similarity, thus leverage the efficiency and effectiveness of keyword proximity serach over XML documents. In XSEarch[16], the XSEarch Ranker ranks the results by giving a score to each result, considering both its structure and its content. Moreover, in this paper, we present a proper scoring function for each potential result, involving four critical metrics. Besides, our approach differs from previous ones in that our scoring function is more likely a well-designed evaluation model during the query processing phase, instead of an isolated ranking component after all the search results have been produced. Thus our work can generate results with improved precision.

In addition, most recently, much work has been done in new problems in this area. For example, eXtract[17] system can generate self-contained result snippets within a given size bound which effectively summarize the query results and differentiate them from one another, according to which users can quickly assess the relevance of the query results. XReal[18] adopts a novel idea to implement XML keyword search with relevance oriented ranking. it develops a formulae to identify the search for nodes and searches via nodes of a query, and then adopts a novel XML TF*IDF ranking strategy, which basically utilizes the statistics of underlying XML data, to rank the individual matches of all possible search intentions. On the issue of result semantics, [19] is the first work that reasons about keyword search strategies from a formal perspective. It proposes the desirable properties that an XML keyword search strategy should ideally satisfy, including *data monotonicity*, *query monotonicity*, *data consistency* and *query consistency*. And then MaxMatch, which also satisfies all those properties, is implemented to identity relevant matches.

3. Result Model & Scoring Function

Defining search results. In AdaptiveXKS, XML keyword search can be regarded as a problem that dividing the set of all keyword nodes into groups that are meaningful. For convenience a search result is considered as a group of keyword nodes instead of a document fragment. Different from LCA-based models, we think two results can share some common nodes as long as none of them is a keyword node, which means our model is more relaxed to the results. Besides, rather than applying some restrictions upon the results, we give each result a score to evaluate its quality(how meaningful it is). Such a score of a keyword node set R is calculated through a function $score(R)$. In our opinion any R satisfying $score(R) > 0$ can be regarded as a result. Since users always prefer the best results, in our model the best results are those results with the largest scores.

Defining a scoring function. A proper scoring function represents a well-designed evaluation model, in which at least four metrics need to be involved: (1) the content information, which mainly refers to the keywords it contains; (2) the structure information, which specifically means the hierarchical position of the root (the LCA of the keyword node set); (3) the compactness, which usually can be calculated through dividing the number of keyword nodes by the number of all nodes in the fragment; (4) the size, which must not be too large. More importantly, the evaluation model should be adjustable to suit different contexts. As a matter of fact, in the evaluation model we can use as many features as possible when they are reasonable. Meanwhile, the LCA-based models only consider the first two features, the content and the structure information, and that's why they suffer from those defects aforementioned.

Next, we provide a universal scoring function for the result model. It is obvious that the four metrics have various priorities, and undoubtedly the content information possesses the highest one because users would most like to get the results containing as many keywords as possible. The structure information should be the second important one. Because the hierarchical position of the LCA node represents the semantics of the result, and a lower LCA indicates a more refined and compact result. This is widely accepted by existing researches and is indeed the essential idea of LCA-based models. As for the compactness and the size, users are always reluctant to see the results with large size and few keyword nodes. Therefore, a result is considered better if the compactness is higher. Moreover, if some result have a really large size then it won't be appropriate to be returned to users.

Some extra notations have to be defined as follows before the scoring function is proposed:

- K is the set of all the keyword nodes;
- for any node v, $dpt(v)$ returns the depth of v in the tree (the depth of the root is 1, and the height of the tree is h);
- for any set of keyword nodes N, $kn(N)$ is the function to get the exact number of keywords N contains.
- for any set of keyword nodes N, $mct(N)$ is the set of all the nodes contained by the MCT of N.

For any result R we provide separate formulas to evaluate the four kinds of features a result possesses: $\mathscr{T}(R)$, $\mathscr{H}(R)$, $\mathscr{C}(R)$ and $\mathscr{S}(R)$. Each of the functions returns a real number between [0, 1], and the greater the value is R is better in one respect.

- Content Information:

$$\mathscr{T}(R) = kn(R)/t \tag{1}$$

- Structure Information:

$$\mathscr{H}(R) = dpt(lca(R))/h \tag{2}$$

- Compactness:

$$\mathscr{C}(R) = |R|/|mct(R)| \tag{3}$$

- Size:

$$\mathscr{S}(R) = \begin{cases} 1 & |mct(R)| \leq st \\ 0 & |mct(R)| > st \end{cases} \tag{4}$$

In the formula of $\mathscr{S}(R)$, st is an integer which is the size threshold of R's MCT. When the size exceeds the threshold, the result is considered inappropriate to be returned. Finally, the scoring function is defined as follows.

$$score(R) = \frac{(1+\mathscr{T}(R))^{\alpha} \times (1+\mathscr{H}(R))^{\beta} \times (1+\mathscr{C}(R))^{\gamma} \times \mathscr{S}(R)}{2^{(\alpha+\beta+\gamma)}} \tag{5}$$

In formula (1) α, β, and γ are three adjustable parameters, each of which should be a positive real number greater than or equal to 1. Furthermore, according to former discussion under normal circumstances α should be set much greater than β, at the same time β is usually larger than γ. Hence we have $\alpha > \beta > \gamma \geq 1$. It is easy to find that any $score(R)$ is either equal to zero or between $(1/2^{(\alpha+\beta+\gamma)}$, 1].

It should be noticed that a result won't be always better than another, so to get a best result we should recalculate the scores when the values of the parameters are changes. Besides, in many cases the results with high scores cannot coexist because of sharing common keyword nodes. That's why we are looking for a partition rather than a coverage of all the keyword nodes.

4. System Implementation

Figure 2 shows our proposed system architecture. We implement AdaptiveXKS by using Java. It takes keywords as input, and returns the information in the XML documents/datasets that matches the user's needs. AdaptiveXKS has a user-friendly interface which allows users to specify an XML document for retrieval, to set the values of parameters and to select a specific processing algorithm to implement the user-submitted keyword queries. In detail, we provide several sample XML documents/datasets, including Synthetic DBLP, DBLP, TreeBank and XMark. AdaptiveXKS returns such subtrees as search results that they root in the LCA of a certain keyword node set meeting user-specified requirements. AdaptiveXKS also allows user to select a particular searching algorithm from the Clustering Algorithm, the Content-Information-First Algorithm, and the Structure-Information-First Algorithm (all of them will be discussed later), according to their needs. As a most important merit, in order to refine search results, AdaptiveXKS allows users to update the values of parameters.

In the remainder of this section, each critical component of our proposed system will be discussed in detail.

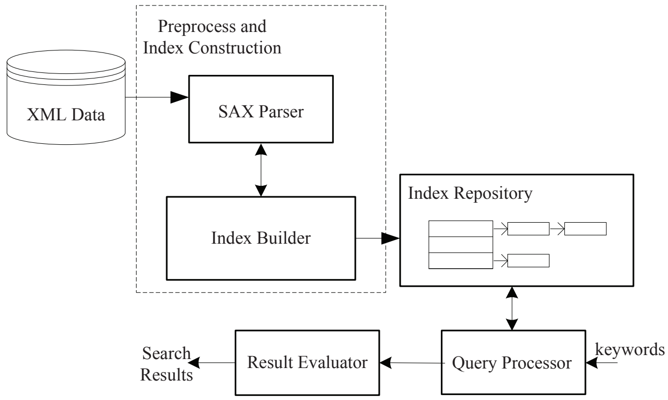

Fig. 2: the AdaptiveXKS architecture

4.1 the Preprocess & Index Construction component

In order to execute user queries efficiently, the parsing, storing and indexing of XML data is an important part of our proposed system. We now provide some necessary background on these techniques. In AdaptiveXKS, we use Apache Xerces XML SAX Parser to parse XML document. Rather than building a complete internal, tree-shaped representation of the XML document, like what DOM model does, SAX is an event-driven model for processing XML. A SAX parser fires off a series of events as it reads the document from beginning to end. Those events are passed to event handlers, which provide access to the contents of the document.

During the parsing of XML document, it is essential that each element is assigned with an ID, which is a way to uniquely identify an element. The AdaptiveXKS implementation uses Dewey numbers as the element IDs, since it is commonly used to label an XML document to facilitate XML query processing by recording information on the path of an element. One interesting feature of Dewey numbers is that it is a hierarchical numbering scheme where the ID of an element contains the ID of its parent element as a prefix. At the same time, the Index Builder is employed to build the inverted index, which typically store for each term in the document/dataset, the list of XML elements that directly contain that term. In addition, an index with Dewey IDs as the keys, such as a B+-tree, is built on top of the inverted list so that we can efficiently check whether a given element contains a keyword. Lastly, the XML indices are stored in the Index Repository, which will be accessed once a keyword query is issued.

4.2 the Query Processor component

The Query Processing component is the core of AdaptiveXKS system. Once a keyword query is issued, the system accesses the Index Repository and retrieves element list (These elements are so-called keyword nodes in this paper) matches to each keyword efficiently.

As is discussed in Section 3, since the four metrics involved in the scoring function may have various priorities, three heuristic searching algorithms are developed in the Query Processing component. In order to meet specific requirements, users are allowed to choose one of them each time to work coordinately with the scoring function to generate search results.

In particular, Strategy Pattern, which is a popular software design pattern, is adopted in the implementation of the Query Processor component. It provides a means to define these three algorithms, encapsulate each one as an object, and make them interchangeable. What's more, it lets the algorithms vary independently from users. Users can also develop other searching algorithms and combine them into the system freely.

In the rest of this subsection, we'll briefly discuss the three algorithms. The details was given in [20].

4.2.1 Matrix Algorithm

The Matrix Algorithm we present in this subsection is a basic agglomerative hierarchical clustering algorithm to obtain the result set $\mathbb{R}$. The input is the set of all the keyword nodes K which is then transformed to a candidate set $\mathbb{C}$ such that each entry $C \in \mathbb{C}$ is a node set and originally contains an individual keyword node. Afterwards, a *Score Matrix* of $\mathbb{C}$ is built. Suppose $|K| = n$ and $\mathbb{C} = \{C_1, ..., C_n\}$, then the score matrix of $\mathbb{C}$ is an n-by-n matrix M that each item $m_{i\,j}$ is set to be $score(C_i \cup C_j)$ if $score(C_i \cup C_j)$ is greater than both $score(C_i)$ and $score(C_j)$, otherwise $m_{i\,j}$ is 0. At each step we find the highest $m_{i\,j}$ in the matrix and merge C_i and C_j to be a new result in $\mathbb{C}$. Then, the matrix is updated to be a $|\mathbb{C}| \times |\mathbb{C}|$ one for current $\mathbb{C}$. The program stops when there is no positive value left in the matrix, then $\mathbb{C}$ is the final result.

From [20], we know that the space complexity of this algorithm is $O(n^2)$, and in the worst case the time complexity is $O(n^3)$. If the scores are stored as sorted lists (or heaps), the time complexity is reduced to $O(n^2 logn)$. Still it is not as good as the performance of the algorithms from the LCA-based approaches. However, theoretically it generates much better results because it calculates possible scores as many as possible and always chooses the largest one.

4.2.2 Content-Information-First (CIF) Algorithm

Many believe that the Content information should be an overwhelming criterion to evaluate a result and thus in our scoring function they would prefer α to be much larger than β and γ. Under this circumstance, the results containing all the keywords should be returned as many as possible and a results with insufficient content information R will only be generated in two cases. First, there is no result R' can be found that satisfies: (1) $R \cup R'$ contains all the keywords; (2)

$score(R \cup R') > score(R')$. Second, such a R' can be found however the size of $R \cup R'$ exceeds the limit. Among these results with insufficient content information one definitely dominates another when it contains more keywords. In this subsection, we present an algorithm called the Content-Information-First (CIF) algorithm to retrieve such results.

Given a set of t keywords $\{w_1, ..., w_t\}$, there are t sets of keyword nodes $K_1, ..., K_t$ possessed through the inverted index and each K_i stores all the keyword nodes containing the keyword w_i. Let K_1 be the one with the smallest size. In line with the principle that a keyword node only exists in a single result, we can only obtain $|K_1|$ results containing all the keywords at most. For each set from K_2 to K_t, the CIF algorithm distributes one or more keyword nodes in it to every node in K_1 which can form a result with highest score. The pseudo code of this algorithm can be referred in [20].

In the best case (when the sizes of the keyword node sets are even), the time complexity of CIF is $O(|K_1| \times \sum_{2 \leq i \leq t}(|K_i|))$, and in the worst case the time complexity is $O(|K_1| \times \sum_{2 \leq i \leq t}(|K_i|^2))$.

4.2.3 Structure-Information-First (SIF) Algorithm

In some specific cases, we concern what the structure of a result much more than how much keyword information inside. For example, it is quite reasonable to assume that no matter how many keywords a result contains only those papers are qualified when proceeding keyword search on DBLP data set. In this case, we can set α close to or even smaller than β and γ and then generate results based on some restrictions built on the structure. Here we provide an algorithm called the Structure-Information-First Algorithm (SIF) which actually comes from another work of us [21] in which it is called the Core-driven Clustering Algorithm. To save space we don't explain the details here. Normally the time complexity of Algorithm SIF is O(n), and O(n^2) in the worst case.

4.3 the Result Evaluator component

After the candidate results are produced in the Query Processor component, they are propagated to the Result Evaluator, which is responsible for the sorting of searching results based on their scores. Also, if necessary, only the top-K of them will be returned to the user. It should be noted that AdaptiveXKS has provided the users ways to modify searching results by tuning the values of parameters of the scoring function. Search results and their scores have to be recalculated each time the underlying values of parameters are updated.

5. Experiments

To evaluate the effectiveness of AdaptiveXKS, we compare its performance with SLCA and ELCA as introduced in Section 1. The Indexed Lookup Eager Algorithm [1] and the Indexed Stack Algorithm [9] are implemented for the SLCA and ELCA query semantics respectively. In addition, the contrasting of three algorithms in AdaptiveXKS is also tested.

Two XML data sets are tested. They are DBLP (size 127M, 6332225 nodes) and TreeBank (size 82M, 3289511 nodes). We have evaluated the quality of the search results to compare these approaches, by measuring precision and recall.

Experimental results indicate that the results generated by our approach have a overwhelming recall value comparing with SLCA and ELCA. Our approach always has the largest recall since any keyword node is considered meaningful and included in a result. In contrast, SLCA always gets a poor value because lots of keyword nodes are abandoned. In case that the parameters are set to static values, and for any approach only the scores of top-10 results are considered, Matrix and CIF overcome the other three as we vary the number of keywords. SIF doesn't act quite good since in the scoring function α is set much larger than β and γ. Similarly, when we give the parameters some static values and use three keywords to search each of the data sets, there is a dramatic decline while k is enlarged for SLCA and ELCA. This can be explained that they usually only find a few best results and omit those second-best ones. Lastly, if four keywords are issued, when we change the values of parameters, Matrix has a stable and excellent performance which is much better than SLCA and ELCA do. The most interesting thing is, for either CIF or SIF the proximity precision changes severely with different parameter values. That's why it is so important to select the appropriate algorithms according to them.

6. Conclusion

In this paper, we present an adaptive and effective XML keyword search system, named AdaptiveXKS. It can avoid several significant defects embedded in LCA-based methods. We devise a novel result model and a scoring function, based on which three heuristic searching algorithms are developed. Most importantly, in order to catch users' specific querying requirements, they are allowed to manually tune the values of parameters weighted in the scoring function, and then choose a most adaptive processing algorithm before submitting the keyword queries. Extensive experiments show how AdaptiveXKS dominates the previous LCA-based approaches and how it overcomes those defects we explored.

7. Acknowledgement

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Teleassistance in Accessible Shopping for the Blind

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Abstract— *In this paper, we present TeleShop, the teleassistance module of ShopMobile 2, our mobile accessible shopping system for visually impaired (VI) and blind individuals that we have been developing for the past several years. TeleShop enables its users to obtain help from remote sighted guides by transmitting images and voice from their smartphones to the guides' computers or phones. We have successfully tested TeleShop in a laboratory study in which a married couple (a blind husband and a sighted wife) used it to retrieve grocery products and read nutrition facts from product packages.*

Keywords: accessible shopping, teleassistance, mobile computing, eyes-free computing, assistive technology, rehabilitation engineering

1. Introduction

We are developing a mobile shopping solution, ShopMobile 2, which allows VI shoppers to shop independently using only a smartphone [1], [2]. ShopMobile 2 has three software modules: an eyes-free barcode scanner, an optical character recognition (OCR) engine, and a teleassistance module called TeleShop. The eyes-free barcode scanner allows VI shoppers to scan UPC barcodes on products and MSI barcodes on shelves. The OCR engine will allow them to read barcode labels and nutrition facts on products. TeleShop allows VI users to obtain assistance from remote sighted caregivers by transmitting images and voice from their smartphones to the guides' computers or phones. TeleShop provides a backup in situations when the barcode scanner and OCR engine fail or malfunction. There is research evidence that having sighted guidance reduces the psychological stress on VI individuals [3]. TeleShop can provide the equivalence of sighted guidance without requiring the guide to be physically present.

The remainder of the paper is organized as follows. Section 2 presents an overview of existing assistive grocery shopping and navigation systems. Section 3 describes the TeleShop module of ShopMobile 2. Section 4 describes our laboratory study. Section 5 offers our conclusion and thoughts on future work.

2. Related Work

2.1 Assistive Grocery Shopping Systems

RoboCart [4], [5], ShopTalk [6], ShopMobile 1 [7], GroZi [8], iCare [9], [10] and Trinetra [11] are some examples of assistive shopping systems. RoboCart was developed by researchers at our laboratory at Utah State University. Shoppers followed a Pioneer 2DX robot equipped with a laser range finder and RFID reader to arrive in the vicinity of products where they used a hand-held barcode scanner for product identification. ShopTalk and ShopMobile 1 were developed at our laboratory. ShopTalk used an OQO computer connected to a wireless handheld barcode scanner and keypad; ShopMobile 1 used a smartphone connected wirelessly to a Baracoda pen barcode reader. In both systems, shoppers would scan MSI barcodes on shelves to obtain directions to target products and UPC barcodes on products to verify that they have picked the correct product.

GroZi was developed at UCSD. The system employs a custom device known as a MoZi box that contains a camera and a haptic feedback mechanism. To use the system, the VI shopper enters the aisle and points the MoZi box towards the products. The MoZi box collects images of products and compares them with images in two databases to locate target products and guide users towards them. iCare was developed at Arizona State University. The system is based on the assumption that products are tagged with RFID tags. Shoppers use a RFID reader embedded within a glove to locate products. Trinetra is a CMU system that uses a Baracoda pen barcode reader and a RFID reader connected to a Nokia smartphone. Shoppers can use the barcode reader to scan barcodes and the RFID reader to scan RFID tags on products when and if those tags become available on products.

2.2 Assistive Navigation Systems

Human navigation can be classified in to two categories - *micro-navigation* and *macro-navigation* [12]. Micro-navigation involves tasks in immediate vicinity of the traveler like obstacle avoidance. Macro-navigation involves tasks outside of the immediate perceptible environment. Planning a path between two points, looking for landmarks and way-points are examples of macro-navigation tasks. VI travelers perform both tasks continuously.

VI travelers typically use long canes or guide dogs to handle micro-navigation tasks. However, sophisticated devices such as sonar canes and optical systems such as the Tom Pouce [13] or the TeleTact [13] system may also be used. Long canes can detect obstacles in front of the traveler from the ground up to waist height but are unable to detect overhanging obstacles or obstacles at head height. Sonar based systems cannot detect small obstacles while optical based systems do not perform well in areas glass surfaces.

GPS based systems [14], [15] are broadly used to assist VI travelers with macro-navigation. However, since GPS solutions do not work well indoors, some researchers resort to other methods, such as RFID [16] for indoor navigation. Vision based systems can also be used for indoor navigation. The system described in [17] places fiducials next to barcodes, which can be decoded with a cell phone camera. Another vision based system is Google Goggles [18]. Using this system, the VI traveler can capture an image using her cell phone and Google Goggles can automatically decode text from it or match it with other images in its database. While this approach may be the right way to go in the long term, the system is currently not too reliable.

2.3 Teleassistance

The term *teleassistance* covers a wide range of technologies to enable VI individuals to transmit video and voice to remote locations to obtain assistance which is typically given through voice. The systems developed by Bujacz et. al. [19] and by Garaj et. al. [12] are but two examples of such systems. The system developed by Bujacz et. al. uses two notebook computers - one is carried by the VI traveler in a backpack and the other used by the sighted guide. The VI traveler transmits video through a USB camera mounted on the chest and connected to the computer. A earphone and microphone headset are used for communicating with the guide. The authors conducted indoor navigation trials and found that VI travelers walked faster, at a steadier pace, and were able to navigate easily when assisted by remote guides. The system developed by Garaj et. al. uses a GPS receiver in addition to the camera and notebook computer. Communication is established by using two GSM cell phones - one for voice and one for transmitting GPS data and a UHF link for transmitting video. The sighted guide can view the VI traveler's position on a map obtained from a GIS database in addition to the images from the camera. They conducted an outdoor trial and tested both the micro-navigation and macro-navigation functionality of the system. They found that mobility levels for VI travelers increased when they were aided by sighted guides as compared to traveling unguided.

3. TeleShop

The TeleShop module of ShopMobile 2 consists of a server running on the VI shopper's smartphone and a client running on the caregiver's computer. As shown in Figure 2, images from the phone's camera are continuously transmitted by the server to the client and subsequently displayed on the GUI shown in Figure 1. The client allows the user to start, stop, and pause the incoming image stream and to change image parameters like resolution and quality. The pause option allows the caregiver to hold the current image on the screen when she wants to read something in the image. Changing the image parameters allows the caregiver to choose between the level of detail in the image and the smoothness of the incoming image stream. Images of high resolution and quality provide very good detail but may cause the resulting video stream to be choppy. On the other hand, images of lower resolution and quality result in a smoother video stream but do not provide much detail. The remote guide is given the option to choose the settings that suit her best.

All communication occurs over UDP. The VI shopper inputs the IP address and port number of the client to the server, which uses it to transmit images to the client. The client can retrieve the IP address and port number of the server from the incoming packets and uses it to transmit image parameters to the server. The client's information was input on the server because the client's IP address stays the same whereas the server's IP address can change if it is on a 3G network. TeleShop can operate with WiFi or 3G.

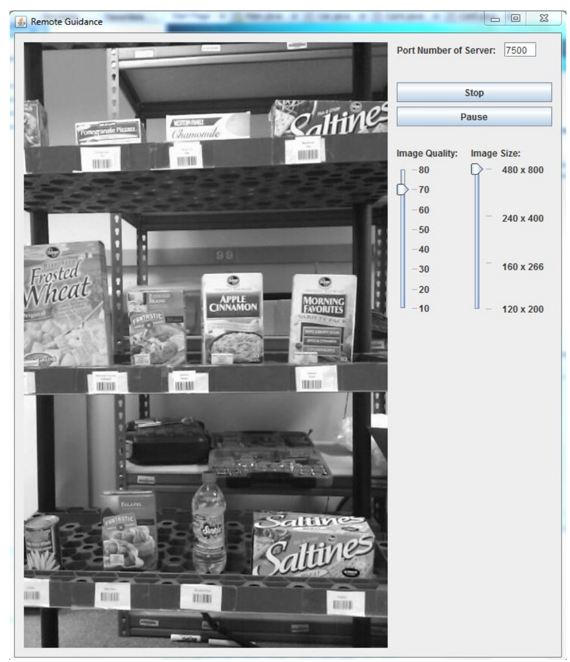

Fig. 1: Screenshot of the Client.

4. Laboratory Study

Two laboratory studies TeleShop were conducted. The first study was done with two sighted students, Alice and Bob. The second study was done with a married couple:

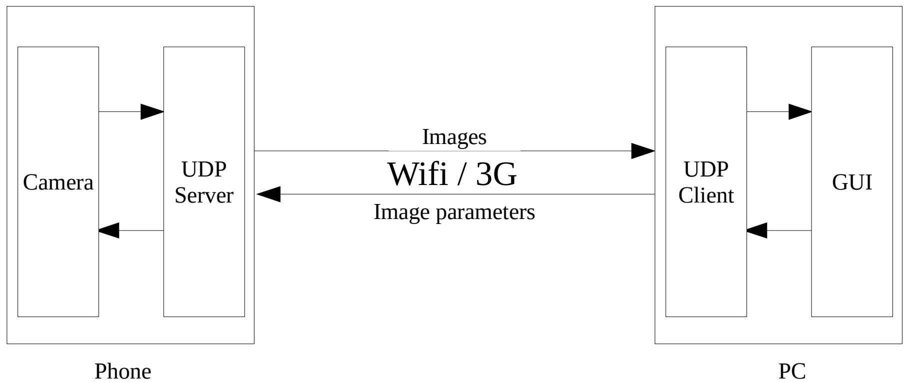

Fig. 2: Overview of Communication Between the Server and the Client.

a completely blind person (Carl) and his wife (Diana). All names have been changed to protect privacy. For both studies, we stocked four plastic shelves with empty boxes, cans, and bottles to simulate an aisle in a grocery store. In both studies, a Google Nexus One smartphone ran the TeleShop server and transmitted images and voice over WiFi to the remote guide's laptop with the client software in a different room.

In the first study, we blindfolded Bob so that he could assume the role of a VI shopper, and Alice assumed the role of the sighted guide. Alice was trained to use the client GUI, and Bob was trained to use the cell phone. A voice link was established between the two by making a regular call. Once both of them were comfortable with the system, Alice was given a list of nine products (three sets containing three products each), which she had to help Bob shop for. Bob used the smartphone to transmit images of the shelf and Alice helped him pick the target products. When a target product was found, Alice would help Bob align the product with the camera so that she could read the nutrition facts from the product's package. She would then read out the nutritional facts on the product to Bob before moving on to the next product on the list. The second laboratory with Diana and Carl used the same training and settings.

Both teams were able to retrieve and read nutrition facts from all the nine products successfully. Figure 3 shows the times taken to retrieve products and to read the nutrition facts of each product for both teams. It must be noted that product six did not have any nutrition facts on it and so the times taken to read its nutrition facts are zeros. Alice and Bob took an average of 57.22 and 86.5 seconds to retrieve a product from the shelf and to read its nutrition facts, respectively. The corresponding times for Carl and Diana were 19.33 and 74.8 seconds respectively [20]. The times taken to read the nutrition facts are greater than the times taken to retrieve products. To read the nutrition facts, the VI shopper had to align the product so that its nutrition table faced the camera, which took considerable time for both teams. It was observed that communication between the VI shopper and the sighted guide were key for quick retrieval and alignment of products. This may be the reason why Carl and Diana, being a married couple, were able to retrieve products and read nutrition facts faster than Alice and Bob. It was also observed that Alice did not change the resolution and quality settings at all whereas Diana changed it several times.

During the post-experiment informal interviews, Alice said that she was comfortable with the default resolution and size settings and did not need to change them. Both teams also said that they were comfortable with the system and did not have any problems with it. Diana suggested that allowing her to rotate the paused image would help with reading the nutrition facts. When asked about using this system in real-life, Carl said that he would find this system very helpful. He mentioned that when he travels, he uses Skype from his laptop to video call his wife to get information about the layout of his hotel rooms. The TeleShop module would allow him to get the same assistance more easily.

5. Conclusions and Future Work

We have presented Teleshop, the teleassistance module of ShopMobile 2, our accessible shopping system for VI and blind individuals. TeleShop enables VI individuals to obtain help from remote sighted guides by transmitting images and voice over wireless connections. Two laboratory studies conducted with the system have demonstrated that it is possible for blind and blindfolded shoppers to retrieve products and obtain their nutrition information. Currently, this system simulates a video feed by transmitting images. In future, we would like to replace it with a real-time video streaming protocol such as RTSP. We would also like to develop client applications that can run on smartphones in addition to laptops and desktops.

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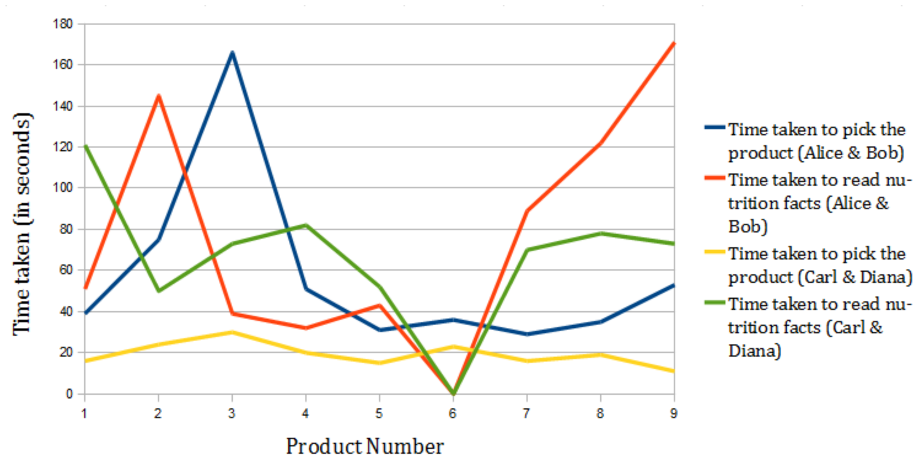

Fig. 3: Times Taken to Retrieve Products and Read Nutrition Facts on Products.

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Integrating CMS Features Into The HTML Markup Language

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Abstract - *This paper looks at the concept idea for further developing the HTML markup language by introducing attributes and tags which would enable content editing and management by final web page users while making it easier for designers and developers to build Content Management System features into their codes. One editor page would work with a corresponding web page and CSS file to display editable content to the user. The web designer/developer would code pages as usual setting out editable portions within the HTML to be accessed by the user merely calling up the editor versions of the same pages on his/her browser.*

Keywords - HTML, content management, CMS, CSS, web browser, code.

1 Introduction

Web designing has come a long way over the past decade with more dynamic and easy-to-use web sites being designed and developed for the World Wide Web. The HTML scripting language has provided the key framework upon which a majority of pages for web sites are built [1]. The latest incarnation, HTML5 has been revamped to give web designers even richer tools for coding their web pages.

The basic way to build a web page is simply via a text editor such as Microsoft's Notepad. The relevant HTML markup syntax is typed out and saved with the .htm or .html file extension and the resulting web page can then be viewed in a browser. Modern web pages have HTML working with CSS (Cascading Style Sheets) which help share out the task of laying out page framework (HTML) and handling the display and appearance of page contents like text, pictures, colours and so on (CSS). Thus in the portion of code below, you can have HTML specifying the display of a paragraph of text while the CSS aspect sets styling for the font size and colour of the text:.

```
<p style=”font-size: 14pt; color:
     red;”>Roses are red</p>
```

Coupled with the use of other scripting languages such as JavaScript, PHP and the .NET framework, web designers and developers now can produce a wider variety of web sites for their clients be it for personal or organisational use, e-commerce, social networking or even gaming.

However, while more tools are being made for the web coders, another trend gained popularity in recent times. The creation of editing tools for the web site owners and users themselves who generally have little or no knowledge of the HTML and CSS aspects of their web pages has been booming.

This class of people aren't interested in the codes for their web pages – they are interested in the content of their web pages. So emerged the era of *Content Management Systems* (CMS) allowing them to edit the web pages produced for them by others or even use set templates to produce the pages on their own.

2 The CMS Challenge

Web designers and developers worldwide today have to meet the demands of their clients to come up web pages which can be edited and updated through some form of CMS or the other, ever more frequently. They have had to build pages in such a way that allows content to be accessed, edited and updated on other pages. A variety of methods exists for web coders to give users these CMS control features and a number of them will be examined in this paper.

The question, however, is: why not have CMS features integrated into the HTML language itself so designers and developers can code web pages and corresponding editing pages without the need for scripting separate CMS bundles or using third-party CMS packages?

HTML can work with CSS internally within its codes or externally linked to a CSS file to produce the visual appearance of the web page displayed on a browser. HTML5 comes with additional features for the coder to manipulate graphics and media directly from within HTML.

The concept being proposed here would, thus, enable the coder to build a web page as usual then set editable sections of the HTML codes. Another HTML file (on the same web server) containing only a link to the first corresponding page would use CSS to build up its own content and display those identified portions in editable form to the user on a browser. This way an authorised user can access this *editor* page with content displayed as on the other public page and when the user clicks on any part of the page which is editable, he/she can go on to modify the content [2].

3 Basic CMS Structure

An understanding of the basic features of a typical CMS would help us visualise the kinds of tags and attributes we would be needing to add to the HTML markup language to produce simple controls for both coders and end users. While the CRUD functions [3] - *Copy, Retrieve, Update* and *Delete* - detail the basic editing capabilities available to a CMS user, it would be helpful to study the steps involved in using the CMS as well and then design controls based around a typical user experience [4].

Common steps involved in using a CMS can be summarised as follows:
i) user logs into secure CMS editing page/environment
ii) user selects web pages and content to be edited
iii) user edits or updates selected web page content
iv) user can preview changes made
v) user saves changes to page(s)
vi) user logs out of secure CMS editing page/environment

4 Methods of Providing Users with CMS Capabilities for Their Web Pages

4.1 Use of CMS packages

Web pages are built using CMS software or application packages such as *WordPress*, *Joomla* and *Drupal*, amongst others. These CMS packages come with preset themes and features for web pages which can be customised by web designers and developers and also later edited by the user who is the authorised web administrator.

Pros:
i) Pre-packaged scripts and templates ready for quick use and customization.
ii) No need for the designer or developer to build complex codes for editing features from scratch.

Cons:
i) One may be limited to the set of templates and features available unless you can build up your own codes then integrate as new theme templates.
ii) Usually these packages need to be installed and set up first on a local system and/or server.
iii) There may be less room for creativity than when one has total control over code manipulation for pages.

4.2 Custom scripting the CMS package

Separate CMS packages are built by the developer for editing the web pages via scripting languages such as XML, PHP and ASP.NET in association with databases. Rich text editor scripts, such as *TinyMCE* and *Aloha*, can be incorporated into the finished packages.

Pros:
i) Customizable codes with direct manipulation by the developer.
ii) Re-useable solutions like code libraries can be incorporated into the product.
iii) CMS codes can be scripted directly by the developer without the need for special software or applications being installed.

Cons:
i) Long periods of coding, testing and debugging might be required for developing the CMS package in addition to the web pages.
ii) The database to be used will need to be installed.
iii) The scripted codes, like those for PHP and ASP.NET, cannot be run and used except on a server or within a framework environment.

4.3 Use of web editing software

Web editing software such as Adobe's Dreamweaver can be used to design, develop and edit web pages.

Pros:
i) Professional tools and features are provided for building the website.
ii) A graphical user interface makes building pages easy with or without direct code editing.

Cons:
i) The software package needs to be bought and installed.
ii) Pages need to be re-uploaded each time they are edited.
iii) Not all users are savvy enough to use such software and master techniques involved.

4.4 Use of online CMS editors

Another option would be to build a web page then mark out editable sections of code which can be accessed online using web CMS services like *CushyCMS* (http://www.cushycms.com) and *SurrealCMS* (http://www.surrealcms.com).

Pros:
i) Free versions available for use.
ii) Web pages can be custom coded using a text-editor then editable sections simply marked out.
iii) No installations required.

Cons:
i) The CMS account exists on one server while the user's web pages are hosted on another server which can sometimes lead to communication problems between them.
ii) The CMS web account requires FTP access to the webhosting account in order to access the codes which may sometimes be restricted.
ii) These CMS tools rely on rich-text editors built on JavaScript which may be switched off on some users' web browsers.
iii) Content editing can sometimes alter the web pages' codes in a way unintended by the designer or developer.

5 Proposed Concept for CMS Features to Be Added to HTML and CSS

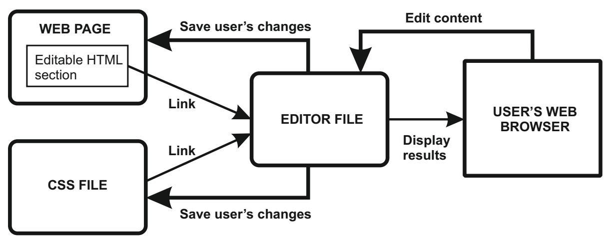

Figure 1: Illustration of how the concept would work

Figure 1 above shows the interaction between web files and the user's browser within the proposed concept. A web page created has sections in its HTML codes marked out as editable. Another web page file acts as the editor file which is linked to both the first web page and a CSS file. Contents from the first web page are displayed on the user's web browser as editable text, images and other media on the page, when the user accesses the editor file. The user can click on these contents, edit and update them.

The editor file would be writing changes directly into the web page and related CSS file.

This system would enable web designers and developers to:

i) focus primarily on building their HTML and CSS files as usual, merely marking out editable elements within the web pages' content

ii) have editor files not requiring complicated coding and linked to the related files

iii) have the web files and editor files stored together in the same location

iv) give users a means of directly editing their web pages' content without the need for databases storing users' inputted values first

v) have a much lighter CMS setup involving just the files created at the time of coding (no complex setup procedures or installations requiring numerous other additional CMS files) [5].

Everything would be taking place basically within the HTML codes. One could conceptualise a web page and its editor file codes to look something as in the following examples.

```
<html>
<head>
<title>Welcome to Webpage 1</title>
<link href="cssfile.css"
rel="stylesheet" type="text/css"/>
</head>
<body>
<p id=”main_content_section”
editable=”on”>Zaafirah and the Ibrahim
kids shouting ‘Hello World!’</p>
<img id=”Aberdeen_Hybrid”
src=”abhybrid.jpg” width=”400”
height=”100” title=”Aberdeen’s Hybrid
celebrity” editable=”on”/>
</body>
</html>
```

Figure 2: Proposed HTML codes for the web page

The codes in Figure 2 are for a typical web page called ‘webpage1.htm’. Content on this page consists of text in the paragraph block marked by the id *main_content_section* and an image marked by the id *Aberdeen_Hybrid*. Both have the proposed *editable* function switched on and so can be edited. We assume that the CSS file, *cssfile.css*, controls the display styles for the page's content (fonts, font colour, image positioning and so on).

Publicly viewing webpage1.htm displays a normal web page and reveals none of the editing features.

```
<html>
<head>
<link href="cssfile.css"
rel="stylesheet" type="text/css"/>
<link href="webpage1.htm"
rel="webpage" type="text/html"/>
</head>
<body>
<edit>
<text id="main_content_section"
update="font-family; color; align;" />
<image id="Aberdeen_Hybrid"
update="size" />
<input type="submit" value="Save" />
</edit>
</body>
</html>
```

Figure 3: Proposed HTML codes for the editor file

The editor file accessing webpage1.htm could be called webpage1_edit.htm and the codes in Figure 3 above show it being linked to both webpage1.htm and *cssfile.css*. While the same paragraph block and image from the first web page are displayed in the same format in editor file, one can notice the new tags now surrounding them.

The layout and display of content on webpage1_edit.htm would be controlled by the webpage1.htm and *cssfile.css* files leaving only the editing options on the page for the user. Thus, a hypothetical <*edit*> tag is added to the HTML codes surrounding other elements to be edited, namely the text and image from webpage1.htm.

Given that we would not need to repeat the same HTML tags from the original web page file (as this file would already have some control over the display of its contents in the editor file), the content to be edited can be represented by another set of hypothetical tags <*text*> and <*image*> for the text and image respectively. Both and any others to used would be contained within the <*edit*> environment.

In this example, the text content from the paragraph block to be edited would be represented as:

```
<text id="main_content_section"
update="font-family; color; align;" />
```

where the text is identified by the id, *main_content_section*, marking the paragraph block in webpage1.htm. An *update* attribute allows the web coder to set the editing options that would be available to the user in updating this block of text content. In this instance, the user can change the font type (that is, font family), the text colour and alignment.

Just as with regular form elements in HTML , these editing options would be displayed in graphical mode for the user. So, say the user were to click on the editable text on the page displayed on his/her browser, drop-down options, buttons or the like would be displayed. The user can then view, click on and implement the options desired for font, colour and alignment of the text.

For the editable image, we have:

```
<image id="Aberdeen_Hybrid"
update="size" />
```

where the image bearing the id *Aberdeen_Hybrid* can have it's dimensions modified by the user as set in the *update* attribute. The *size* option would allow this. Other plausible options definable within the image's *update* attribute could include *upload* (so when the user clicks on the image, there is an option to upload a new image), *title* (so the user can change the title attached to the image) or *quality* (so the user can adjust the tone, shade, brightness, contrast and so on).

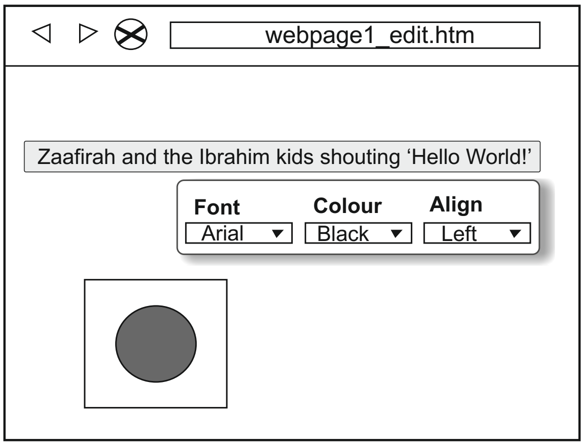

Figure 4: User browser view for editing text in the editor file

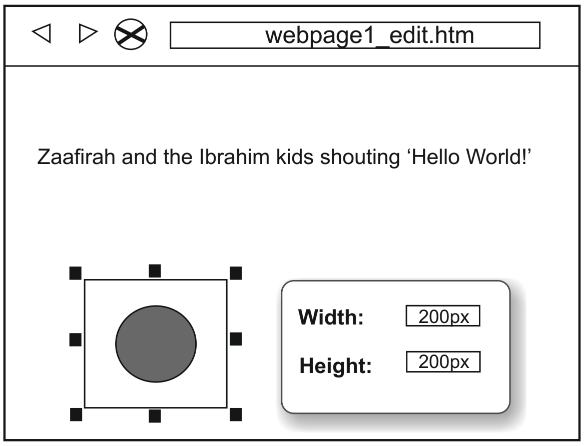

Figure 5: User browser view for editing the image in the editor file

Figures 4 and 5 above illustrate what the editor page could look like when viewed on the user's browser

6 Requirements for the Proposed Concept

i) HTML codes will need to be able to specify editable attributes for sections of code to be identified for editing.

ii) Editable code sections need to be capable of being displayed on web browsers for manipulation by users via HTML coding using a minimal number of files (that is, not building or using an entire CMS package in addition).

iii) Edited pages need be capable of being saved.

iv) The editing process needs to focus on just the editable portions of the page alone and not have to keep re-writing the entire page each time the user makes changes. The system needs to offer similar efficiency levels or more as with one working with a database [6].

v) Access to web pages for editing needs to be secure.

vi) Adequate web browser support and compatibility will be needed on the different browser platforms available to users.

7 'localStorage' and 'contenteditable' Features in HTML5

Two exciting developments in HTML5 are those of the *localStorage* object [7] and *contenteditable* attribute [8] which allow HTML to store and recall data as well as define aspects of code which are identified as editable. No database is used nor cookies.

The HTML5 code for a sample editable web page below uses Google's jQuery API (online access needed) to highlight how *contenteditable* and *localStorage* work [9]:

```
<!DOCTYPE HTML>
<html>
<head>
<script
src="https://ajax.googleapis.com/ajax/
libs/jquery/1.4.2/jquery.min.js"></scr
ipt>
</head>
<body>
<p id="editsection"
contenteditable="true">Hi, edit this
text!!!</p>
<script type="text/javascript">
$(function() {
var editsection =
document.getElementById('editsection')
;
$(editsection).blur(function() {
localStorage.setItem('user_edit',
this.innerHTML);
});
if (localStorage.getItem('user_edit'))
{
  editsection.innerHTML =
localStorage.getItem('user_edit');
}
});
</script>
</body>
</html>
```

Figure 6: HTML5 example with *contenteditable* and *localStorage*

Saving this code as an HTML file from a text-editor produces an editable web page where the text *"Hi, edit this text!!!"* has been identified using the id *editcontent*. *Contenteditable* is set to "true" so on the resulting page the user can edit the text by clicking on it.

The modified text is then passed as a variable also called *editcontent* unto a JavaScript function making use of *localStorage* to store the data locally on the user's computer while updating the web page on the browser.

With the changes now stored locally, the user can refresh the web page and still get to see the modified text.

The modified text remains intact even when the user returns to view the page at a later date after turning off his/her computer.

There are currently limitations to using *contenteditable* and *localStorage*, however.

i) The HTML5 example in figure 6 relies on JavaScript which is a client-side script – it executes functions on the user's browser on a local system and does not apply changes to the web page itself on the web server. For the proposed CMS model to work, a means of getting content edited via HTML and being saved on the web pages is needed. This may require new functions and attributes being developed for HTML allowing secure server-side storage of data. Using XML, PHP or ASP.NET scripted pages amongst others on the web servers would work but the goal is integrate CMS features in HTML itself.

ii) The user has to be using an HTML5 compatible web browser in order for these functions to work. Fortunately a lot of the modern browsers now, including Explorer 9, support HTML5.

iii) On its on so far, the user cannot use *contenteditable* to carry out formatting tasks like setting fonts, font size, colour and so on. Such capabilities could be added using JavaScript but that would mean more coding by the designer or developer.

iv) Using *contenteditable* along with *localStorage* as demonstrated would also mean if the user changed his/her computer, the changes made to the web page on the previous computer would no longer be reflected. The same is the case if the web page file is transferred on to a different computer. The actual HTML codes remain unaltered and changes are not saved within them.

Nevertheless, even with these highlighted limitations of features available in HTML5, the possibilities of an improved method for an HTML/CSS-based CMS are highlighted.

8 Conclusion

Having a simplified HTML/CSS-based CMS coding structure works to the benefit of both web designers and developers and well as the users of their products. Recent achievements with HTML5 show such a structure is attainable with further research and development on the markup language.

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SESSION

POLICIES

Chair(s)

TBA

Private Distributed Cloud Deployment in a Limited Networking Environment

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***Abstract*—The focus of our research began with the deployment of a cloud computing system to offer resources similar to commercial vendors. Realizing the implications of deploying such a system in an environment with limited networking resources (IP addresses), we decided to find a solution that would work giving our cloud only one public IP address. Users will be able to access cloud resources through a simple web interface and maintenance of the cloud will be contained with private networking resources. Users needing resources in the local cloud will rely on a NAT router, with other functionalities, to gain access these resources. We will also demonstrate, if the need arises to have multiple geographically distributed clusters in the local cloud, how to scale in this sense with only one IP address per cluster.**

***Index Terms*— Cloud Computing, IP Networking, Resource Distribution.**

I. Introduction

THE cloud computing architecture is becoming a dominant contender in the distributed systems paradigm. Its differences from the client/server architecture are based in its heavy use of resource elasticity. The cloud architecture uses virtualization technology in distributed data centers to allocate resources to users as they need them. Cloud computing has emerged from the previous industry standards, such as grid and cluster computing. Although cloud architectures are similar to these distributed systems, the resources are usually maintained by a single entity and might not be used by customers to complete similar goals. Depending on the level of control the user has, Infrastructure-as-a-Service (IaaS), Platform-as-a-Service (PaaS), or Software-as-a-Service (SaaS), jobs can differ greatly from customer to customer [7].

There are many implementations of the cloud architecture by various commercial vendors and open source communities. The Amazon Cloud, or collectively known as Amazon Web Services (AWS) is arguably the leader when it comes to offering customers access to hardware in the cloud (IaaS). To demonstrate this commanding lead in the field, the vendor, Netflix, Inc. (the largest on-demand movie and television streaming company) uses AWS to run mission critical customer-facing and backend applications [2]. Microsoft has entered the cloud computing paradigm with its Azure platform (PaaS) [3]. This platform allows customers to create and run applications hosted by Microsoft's datacenters. Customers using Azure have access to Microsoft's cloud operating system and all of the services needed to develop applications remotely. Unlike AWS, Azure does not allow the user lower level access to hardware. Google's Docs application [4] allows users to store and edit office document related data. Users of this application have no permissions to access underlying hardware, or change the software applications presented to them (SaaS). The user consumes this resource with little or no care of how it's implemented. In this paper, the cloud architecture will refer specifically to Infrastructure as a Service (IaaS), providing users the ability to create virtual machines.

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Following in the footsteps of cloud leaders such as Amazon and Google, open source communities have provided software packages that allow individuals to deploy their own local cloud. As customers become more and more dependent on retrieving data whenever they want it, they become heavily dependent on the cloud vendor's reliability for data access. There could be many reasons for a user or organization to deploy a local cloud. One example would be the user wanting a higher utilization of their physical resources. Another example could be that some data created by local organizations cannot be stored by a public cloud vendor. Whatever the reason for a local cloud, users still need access to the data stored, which requires expensive networking resources. When a user needs to connect to a local cloud resource, each instance of that resource has to have a unique private IP address. In the case of our experiments, the resources are stateless virtual machines. The user is given a list of operating systems to choose from, and if persistent storage is needed, they have the ability to mount space located on the storage controller.

The open source community is led by three prominent distributions: Eucalyptus, OpenNebula, and Nimbus [1]. Our local cloud implementation uses Eucalyptus. An overview of the Eucalyptus architecture will be given in section three. The topic of this paper is concerned with deployment of privately maintained clouds using limited networking resources.

The rest of the paper is organized as follows. Section two will present the problem statement of deploying a private cloud into an environment with limited networking capabilities. Section three explains how to build the cloud using Eucalyptus. Section four gives details on deploying cloud resources with minimal interactions from users. Section five shows how to deploy a distributed multi-cluster setup using VPN. Section six presents our conclusions.

II. Problem Statement

Customers may be reluctant to move their data to off-site cloud vendors due to access regulations, cost, and trust issues [9]. Instead, new software has become available for them to build their own personal cloud computing system. This gives the customer the ability to understand the issues and benefits of using cloud technology for distribution of resources before making the move to an enterprise vendor.

There is a major problem with deploying a cloud locally; network resources in terms of IP addresses are usually expensive, and therefore, limited for smaller organizations. We are proposing a solution that needs only one public IP address for resource distribution to users of the local cloud. This approach will provide benefits of more than just efficient use of network resources. By having only one public address associated with the cloud system, security vulnerabilities are decreased. This is due to the fact that all incoming and outgoing traffic to the cloud will be associated with one IP address. This address can be monitored more efficiently than if the entire cloud system were implemented on public IP addresses. In the case of our cloud construct proposition, no new network infrastructure will be needed in the organization.

In some cases, organizations may have resources geographically distributed. The idea of implementing a cloud architecture across these resources may be a possibility if a minimum of networking resources are used to connect them. It is possible to scale an organization's local cloud to other sites using VPN and a single IP address per distributed cluster. We understand that as demands rise to access resources not located in the local cloud, a bandwidth bottleneck may arise. Implementation of this architecture should give careful attention to the bandwidth requirements of the local cloud WAN IP address, and the bandwidth needed to connect each distributed resource to the local cloud.

III. Local Eucalyptus Cloud

Currently, our local cloud, fluffy, consists of seven Dell OptiPlex desktop computers, one network attached storage device, one 1Gbps Ethernet switch, and one Linksys WRT54GL NAT router. The underlying operating system of our cloud is Ubuntu Server 10.10 64-bit, which includes the Eucalyptus 2.0 open-source cloud architecture software [5]. Following the current specifications of deploying Ubuntu Enterprise Cloud (UEC) [8], we have configured the following machines to make our local cloud environment. Our goal is to deploy this cloud environment into a working environment with a single IP address. The following sections outline the main components of our local cloud. We make note here to show that our cloud consists of nonhomogeneous commodity hardware that could easily be found in office environments. The only equipment that requires specific consideration are the servers used to run node controller software. These servers should have CPU's that are capable of hosting virtual machines.

A. Cloud Controller

This machine is the front end user interface to our cloud setup. It is hosted on a Dell OptiPlex 745 desktop (Core 2 Duo 2.8 GHz, 2GB RAM, 80GB hard drive). Its purpose is to provide the web interface to users while interacting with the rest of the components in the cloud architecture. The cloud controller monitors the availability of resources of various components in the local cloud and monitors the running instances currently deployed on the node controllers. This device is fully customizable in the fact that we can install additional software packages to meet our needs. For example, we have apache, MySQL, Java, and PHP-CLI installed to handle various processes of deploying our cloud resources.

B. Cluster Controller/Storage Controller

The current deployment of UEC requires the cluster controller and storage controller to be on the same physical machine. These are hosted on a Dell OptiPlex 755 (Core 2 Duo 2.8 GHz, 2GB RAM, 160GB hard drive).

The cluster controller determines on which node controller a deployed virtual machine will run. It also allows network access to the virtual machines running on the node controllers. DHCP and DNS processes for the virtual machines are maintained on this controller. The cluster controller is responsible for load balancing virtual machines across all node controllers. In the case that there are multiple node controller clusters, each cluster will be led by an individual local cluster controller. All of the cluster controllers report back to the cloud controller on the status of the node controllers that it controls.

The storage controller gives persistent storage access to instance users. This service is similar to the elastic block storage service from Amazon Web Services. Storage blocks can be mounted to running instances since the virtual machine concept in Eucalyptus is stateless.

Our current setup has only one cluster, therefore we have only one cluster controller. Alternative cloud configurations will show later that it is possible to connect many resources at different locations back to the cloud controller using only one IP address per cluster.

C. Walrus Storage Controller

Our walrus controller is hosted on a Dell OptiPlex 620 (Pentium D 2.93 GHz, 2GB RAM, 120GB hard drive). The walrus controller stores machine images that can be launched as virtual machines in the local cloud. We are currently hosting two versions of Ubuntu, 9.10 and 10.04 for users to access. The walrus controller will hold any machine image that we wish to make available to users. When the user decides to launch a virtual machine, the machine image has to be transferred from the walrus controller to the node controller (Section III.D) before being booted. If a specific image from the walrus is being used relatively often, that image can be cached on the node controller to decrease the amount of traffic in the cloud's private network.

D. Node Controller

The node controller is a machine with VT (Intel) extensions on the CPU(s) used to host the running virtual machines in the local cloud. Our current setup has three node controllers. The first node controller is hosted on a Dell OptiPlex 755 (Core 2 Duo 2.8GHz, 2GB RAM, 160GB hard drive). Our second node controller is hosted on a Dell OptiPlex 960 (Core 2 Duo 2.93GHz, 2GB RAM, 160GB hard drive). The third node controller is a custom build (Core 2 Quad 2.5GHz, 4GB RAM, 320GB hard drive).

The default hypervisor used by Eucalyptus is KVM. We are satisfied using this hypervisor because it is a full virtualization solution for Linux operating systems on x86 hardware containing virtualization extensions (Intel VT or AMD-V). The default configuration settings located in: /etc/eucalyptus/eucalyptus.conf defines a variable named MAX_CORES. This variable is initially set to the number of physical cores located on the specific node controller. Administrators can change this initial value to allow for a higher number of virtual machines per node controller. The equation below gives the maximum number of virtual machines KVM will allow on any single physical node controller:

# of VM's = 8*(actual number of physical CPU cores) (1)

The total number of virtual machines in our setup allows:

8*(actual number of physical CPU cores/2) (2)

Equation (2) ensures that no single CPU will be overloaded (at most 4 virtual machines per physical core) and virtual machines will execute with sufficient processing resources. Other resources also play a role into the number of virtual machines available for launch. These include the total amount of RAM and available hard disk space. We are able to handle this amount of virtual machines per core because the specific Linux operating systems we offer are without graphical user interfaces. Figure 1 shows the information from the cloud controller as it keeps track of resources while virtual machines are instantiated and terminated.

AVAILABILITYZONE	fluffy 192.168.1.100				
AVAILABILITYZONE	|- vm types	free / max	cpu	ram	disk
AVAILABILITYZONE	|- m1.small	0024 / 0032	1	192	5
AVAILABILITYZONE	|- c1.medium	0022 / 0028	1	256	5
AVAILABILITYZONE	|- m1.large	0011 / 0014	2	512	10
AVAILABILITYZONE	|- m1.xlarge	0006 / 0007	2	1024	20
AVAILABILITYZONE	|- c1.xlarge	0002 / 0002	4	2048	20

Figure 1: Available resources on the local cloud.

E. Backup Device

The backup solution for our local cloud is a Dell OptiPlex 270 (Pentium 4 3.2GHz, 2GB RAM, 120GB hard drive). We use this device to run a scheduled backup on all other devices in our setup. The device mounts folders to our NAS (Netgear Stora 1TB), which stores our nightly backups.

F. Client Device

This device is a Dell OptiPlex 280 (Pentium 4 2.8GHz, 2GB RAM, 120GB hard drive). This machine is used for testing the cloud. Instances can be launched, and cloud components can be accessed by SSH.

The router used in our setup is the Linksys WRT54GL. This device has been flashed with version 24 of DD-WRT [6]. This gives us more control over our network through the manipulation of ports through SSH connections, VPN access, and DHCP services. The open nature of the router will become important in the deployment of our local cloud.

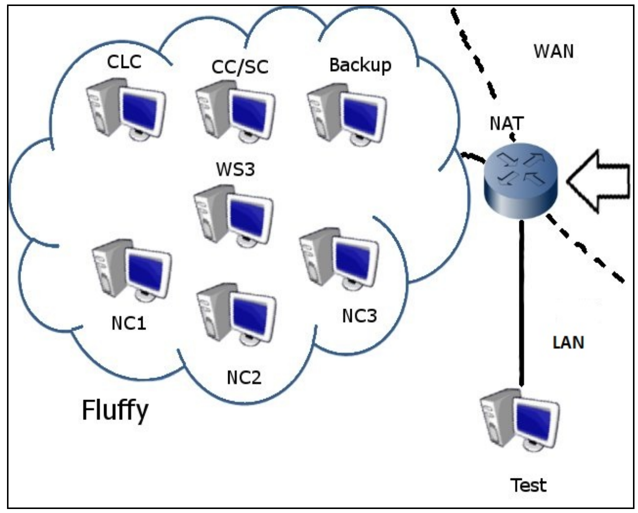

Figure 2: Local Cloud Architecture.

As of now, users with private addresses have the ability to launch virtual machines in our cloud setup. This is an inconvenience since either the user has to be physically connected to our private network, or they need VPN accounts to access our cloud from a public address. A solution for making resources available to users outside of the private cloud network is given in the next section.

IV. CLOUD INTERFACE WITH NETWORK LIMITATIONS

In this section, we are proposing our cloud deployment scheme which allows an organization to relatively simply introduce a cloud infrastructure into their environment using a limited amount of networking resources. This deployment technique will take some initiative on the IT staff members depending on the amount of availability and scalability needed. Some things to take into consideration before deploying this type of cloud architecture (Figure 2) are WAN port bandwidth for single cluster setups and bandwidth between NAT routers while running the VPN protocol when deploying a distributed cluster setup.

As previously stated, the resources given to users in our setup are virtual machines hosted on the node controllers. Users do not have physical access to the cloud's hardware resources, but can spawn virtual machines of different configurations to meet their demands. Users also have full root access to their instantiated virtual machines, and are responsible for any security precautions presented by having resources available from potentially unsecured networks (WAN).

The cloud architecture will be hosted completely within its own private network. A NAT router using DD-WRT firmware will be used to manage ports opened to the virtual machines created by users outside of the cloud's private network. The physical resources for the cloud would only demand the multiple servers needed, switches, and a single router. Users will access the cloud resources using commands to launch the virtual machines, and opened forwarded ports handled by the router.

A. *Router Setup and Log Information*

Our NAT router was loaded with DD-WRT firmware,giving us a great deal of control over its packet forwarding features. When connecting to these cloud resources, users shouldn't worry about the networking responsibilities of providing these resources. We have the ability to dynamically open and close ports and manipulate the iptables of our private network. This depends on the current number of virtual machines in execution. Port forwarding is necessary simply because we are depending on one WAN address to connect many users to many virtual machines that could be running similar protocols.

The cloud controller keeps a list of the currently running virtual machines, as shown in Figure 3. This information will be used to maintain a log.txt file also located on the cloud controller. Each row in the log file is associated with an individual virtual machine. The only information required is the private IP address assigned to the virtual machine by the cluster controller, and the port forwarding information decided by the specific protocols the user chooses during launch.

```
192.168.1.151; protocol; port #; protocol; port #; protocol; port #, …
192.168.1.152; protocol; port #; protocol; port #; protocol; port #, …
192.168.1.200; protocol; port #; protocol; port #; protocol; port #; …
```

Figure 3: Log.txt file located on Cloud Controller.

When a virtual machine is launched, the cluster controller informs the cloud controller of the pending network details. During the instantiation of the virtual machine(s) the graphical interface allows the user to select their preferred protocol. The default protocol used is SSH, which will initially only open port 22 on the virtual machine. All other ports will be closed, and therefore unreachable. Once the command has been sent to the cloud controller to boot a virtual machine, a process writes the private IP address of the new virtual machine to the log file. It then reads the log for protocols that are already used on other virtual machines, updates the log with the forwarded port for the user to utilize for each protocol, and notifies the user of these forwarded ports. Figure 4 shows the current status of the log file after a user instantiates a new virtual machine (151) with other virtual machines already running in the cloud.

```
192.168.1.151; SSH; 1200; HTTP; 1500;
192.168.1.152; SSH; 1201; HTTP; 1501; FTP1; 1800; FTP2; 2000
192.168.1.153; SSH; 1202; FTP1; 1801; FTP2; 2001; SMTP; 2400
```

Figure 4: Example Log.txt file with instantiated vm's.

Once the log file has been updated for a specific virtual machine, the process communicates with the NAT router via SSH and updates the iptables. The following code shows how the cloud controller can update the iptables:

$ iptables –t nat –I PREROUTING –P tcp –d $(nvram get wan_ipaddr) –dport 1200 –j DNAT –to 192.168.1.151:22
$ iptables –I FORWARD –p tcp –d 192.168.1.151 –dport 22 –j ACCEPT

After the execution of these two commands, the router will forward incoming WAN packets from port 1200 to port 22 of the virtual machine with the IP address of 192.168.1.151. All protocols will be forwarded in this manner by the NAT router.

When the user decides to terminate their virtual machine, a process on the cloud controller updates the log file by removing the record with the terminating virtual machine's IP address. Once this record is removed from the log, the cloud controller updates the iptables of the router by executing the following commands. The variable 1 is the record in the iptable specific to the terminating virtual machine's IP address:

$ iptables –D FORWARD 1
$ iptables –D PREROUTING 1 –t nat

B. *User Interface*

The user interface for connection to our cloud resources is relatively simple. The user should first create an account to use on the cloud. This is done by using the default Eucalyptus web interface. Once the user has an account, they must login and download the Credentials.zip file to use when connecting to virtual machines they instantiate. This file contains an RSA key, used to authenticate the user whenever they attempt to use resources on the cloud.

As shown in Figure 5, the user navigates to the custom web interface that gives them the choice of operating systems, size of virtual machine (CPU, RAM, disk space), and protocols needed. Once their choices are made, they launch the virtual machine. The interface returns the IP address of that virtual machine and port forwarding values for the protocols they chose. The user then has the ability to connect to the virtual machine (typically by SSH) using their RSA key. From this point, the user has root access to perform any tasks they desire.

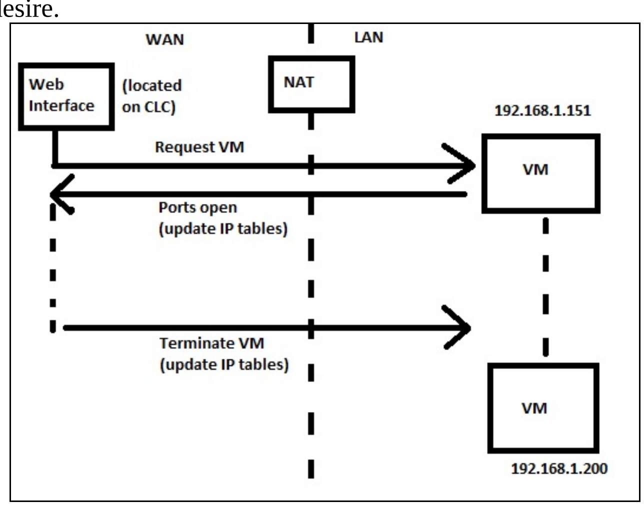

Figure 5: User interface creation/deletion of vm's.

C. Network Considerations

Deploying a scalable private cloud in the manner described in this paper requires careful planning to ensure availability to the cloud resources. The WAN connection will be the bottleneck when deploying this architecture. IT administrators should make sure to allocate as much bandwidth to the WAN port as possible. Generally, users will at least have SSH sessions open on each virtual machine. SSH, by protocol design, delivers constant data transmissions between the client and server processes to minimize attacks on traffic patterns. In our design, we used a 1Gbps copper Ethernet link to supply traffic to the WAN port. This provided sufficient bandwidth to host all of our available resources (32 virtual machines) to users.

Another consideration that should be made is the selection of the NAT router. Our choice of the WRT54GL was one of low cost and ease of manipulating the firmware to allow root access. Even though we had a 1Gbps WAN link, our maximum sustainable throughput from WAN to LAN address was around 50Mbps. These speeds are typical of the 10/100Mbps Ethernet switch used by this router. This is enough to satisfy the current scale of our cloud, but would become a factor if a major increase in cloud resources were needed. Gigabit NAT routers such as the Linksys E4200 or Cisco RV220W provide excellent sustainable WAN to LAN throughput of 686Mbps and 720Mbps respectively. The implementation of the distributed multi-cluster setup described in the next section brings even more demanding requirements from the organizations available networking resources.

V. Distributed Multi-Cluster Setup

Depending on an organization's geographical distribution, a multiple cluster setup may be desired. Each cluster in the Eucalyptus architecture contains the cluster and storage controller, and multiple node controllers. The DHCP service hosted on the cluster controller can be configured to accommodate unique IP subnets for each cluster. Using Eucalyptus' default of IPv4, the private distributed cloud could potentially have on the order of $\sim 2^{32}$ virtual machines. IP addressing is clearly not a scalability factor when deploying this type of cloud.

A. Router and VPN Setup

The VPN protocol [10] will be used to deploy this cloud architecture. Each cluster is assumed to be geographically separate from all other clusters and from the main cloud controller cluster, as shown in Figure 6. In this architecture, we assume a main cloud cluster which contains the cloud controller, and auxiliary clusters used for hosting additional virtual machines. Each auxiliary cluster will require only one WAN IP address to connect back to the main cloud cluster. Users will still contact the single cloud controller to gain accounts and launch virtual machines.

A VPN is an authenticated, encrypted connection between two networked devices. The WRT54GL NAT routers (and others) have the ability to form VPN gateway to gateway tunnels. This connection between NAT routers allows data to be transmitted over the internet as if it were in a single private network. The details of setting up VPN gateway tunnels to each router will not be covered as it is out of scope of this paper. Each cluster will have its own IP subnet, relaying this information back to the cloud controller. For example, the main cluster hosting the cloud controller would use 192.168.0.0/16 giving $\sim 2^{16}$ potential virtual machine IP addresses. Cluster 1 would use 192.169.0.0/16 giving the same amount of IP addresses for use as the main cluster. Each cluster would use this concept to provide a unique IP address across the entire geographically distributed cloud.

Load balancing in a single cluster topology is executed by the cluster controller, but in this distributed setup, an additional layer of load balancing will be performed by the cloud controller. As the cloud controller is the only element that knows the state of the entire cloud, it will determine which cluster controller to contract when a user decides to launch a new virtual machine.

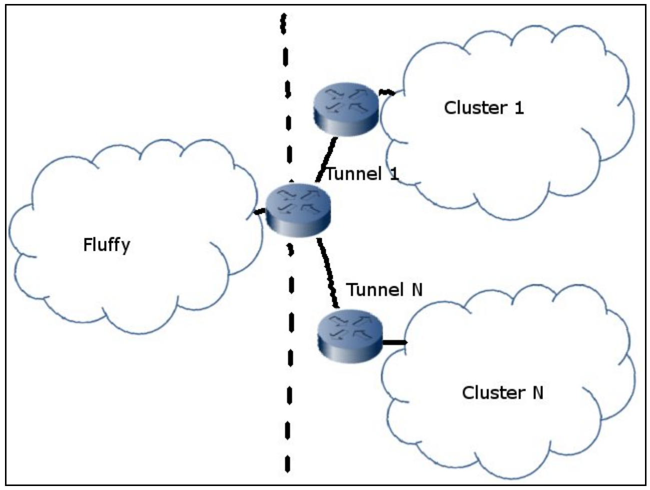

Figure 6: VPN tunneling to connect distributed clusters.

B. Scalability Considerations with a Distributed Architecture

The potential for scaling using a distributed architecture ideally should be unlimited. Careful consideration should again be given to the available bandwidth to the WAN ports of each NAT router. The traffic generated in this setup is substantially greater than in the single cluster architecture. Organizations may consider using protocols such as ATM or others that guarantee a minimum bandwidth between NAT routers. The NAT router selected for this architecture is also more important than the one used in the single cluster architecture due to the increased demands on networking resources.

Users will still use the single point of interface with the main cloud controlled cluster. There is no cloud controller in the auxiliary clusters, and therefore, no interface to use those resources directly. A future work direction to reduce the traffic and processing requirements of the single WAN port on the main cluster would be to propagate iptable updates to the other NAT routers of the auxiliary clusters. This way, users could connect directly to the resources using the single IP address of those auxiliary clusters.

VI. Conclusion

This paper presents the idea of deploying a private cloud using a limited amount of networking resources. Organizations have the ability to create a private cloud and introduce it into their network without greatly affecting their current infrastructure. With the introduction of the single cluster cloud, users are able to request resources through a simple interface. Once the user has access to these resources, they can use it to perform any task they desire. When the user no longer needs the resources, they are released and can be used by others.

We also introduced the concept of a distributed cluster architecture. In this architecture, VPN is used to connect the auxiliary cloud clusters back to the main cloud cluster. Each cluster is hosted on its own subnet of IP address, giving each virtual machine the ability to have a unique address.

Scalability needs careful consideration while implementing the single or distributed cloud architecture. Administrators should be aware that the single WAN ports could be heavily utilized and should be allocated as much bandwidth as possible. This is especially true in the case of the distributed cloud topology. Administrators should also carefully consider the NAT router used when deploying this cloud. A few NAT router suggestions were given, and if more performance is needed, a dedicated NAT routing server could be used.

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Converged Services Orchestration vs. Choreography Revised

A Scenario-based Approach for an Effective Enterprise Management

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Abstract - *Converged services alignment (i.e., orchestration and choreography) has been a dominant topic of several regulatory initiatives. These initiatives were successful to some extent, but had faced also various difficulties, that prevented any final specification grade. Thus, this paper revises the converged services orchestration and choreography concerns that apply for an effective enterprise management and develops a framework that enables service composition considering multiple interconnected perceptions. Furthermore, since the regulatory initiatives in this area have not been built on top of a defined and all-encompassing conceptual foundation, we present a viable alternative to make up for this drawback. The paper also shapes a service provisioning arrangement intended to identify requirements and concepts to be addressed by and integrated into the specification framework.*

Keywords: Converged Services, Services Orchestration and Choreography, Services Composition, Web Services, Virtual Enterprise, Enterprise and B2B Management

1 Introduction

Contemporary network technologies allow developing the new collaboration business paradigms, such as virtual enterprises, where different companies pool together their services to offer more complex, added-value products and services. Besides, network technologies and Internet make services easily accessible and thus they allow composing virtual enterprises in very flexible ways [1, 2].

However, B2B (Business-to-Business) interaction requires new modes of coordination between participating enterprises. Unquestionably, one main requirement is preserving the autonomy of each participating partner during the interaction, without restricting the efforts to reach the overall goals of the common process. Thus, mechanisms regulating distributed service workflows (or business processes), when the business process is composed of the invocation of different organizations, are needed [1, 3].

In turn, service-oriented computing provides technologies that enable multiple organizations to integrate their businesses over the Internet. Typical execution behavior in this type of distributed systems involves a set of autonomous peers/enterprises interacting with each other through messages. Modeling and analyzing interactions among the enterprises is a crucial problem in this domain due to the following reasons:

- Organizations may not want to share internal details of the services they provide to other organizations (e.g., for reasons of competition). In order to achieve decoupling among different enterprises, it is necessary to specify the interactions among different services without referring to the details of their local implementations.

- Modeling and analyzing the global behavior of this type of distributed systems is particularly challenging since no single party has access to the internal states of all the participating enterprises. Desired behaviors have to be specified as constraints on the interactions among different enterprises since the interactions are the only observable global behavior. Moreover, for this type of distributed systems, it might be meaningful to specify the interactions among different enterprises before the services are implemented. Such a top-down design strategy may help involved organizations to improve the coordination of their development efforts.

These distributed systems can be modeled as composite (Web) services which consist of a set of peers interacting with each other via synchronous and/or asynchronous messages. A conversation is the global sequence of messages exchanged among the enterprises participating to a composite service(s). Thus, the service choreography specification identifies the set of allowable conversations for composite services. An orchestration, on the other hand, is an executable specification that identifies the steps of execution for the enterprises [4 ÷ 6].

2 Behavioral Models for Service Configuration

2.1 Service Orchestration vs. Service Choreography – Some Definitions

Service choreography is a form of service composition in which the interaction protocol between several partner services is defined from a global perspective [7].

Thus, during run-time each participant in service the service value chain executes its part (i.e., its role) according to the other participants' behavior [8]. The choreography's role identifies the expected messaging behavior of the participants in terms of sequencing and timing of the messages that they may consume and produce [9].

Service choreography is better understood through the comparison with another paradigm of service composition: i.e. service orchestration. Whereas, in service choreographies the logic of the message-based interactions among the participants is specified from a global perspective. In service orchestration, however, the logic is specified from the local point of view of one single participant, called the orchestrator. In the service orchestration language BPEL, for example, the specification of the service orchestration (e.g., the BPEL process file) can be deployed on the service infrastructure (for example a BPEL execution engine like Apache ODE). The deployment of the service orchestration specification creates the composed service.

Service choreographies are not executed: they are enacted. Service choreography is enacted when its participants execute their roles [10]. That is, unlike service orchestration, service choreographies are not run by some engine on the service infrastructure, but they "happen" when their roles are executed. This is because the logic of the service choreography is specified from a global view point, and thus it is not even realized by one single service like in service orchestration [11].

2.2 Choreography Model for Service Configuration

A choreography model describes collaboration processes between collections of services to achieve a common goal. It captures the interactions in which the participating services engage to achieve this goal and the dependencies between these interactions, including: causal and/or control-flow dependencies (i.e., a given interaction must occur before another one, or an interaction triggers another one), exclusion dependencies (a given interaction excludes or replaces another one), data-flow dependencies, interaction correlation, time constraints, transactional dependencies, etc.

The choreography does not describe any internal action of a participating service that does not directly result in an externally visible effect, such as an internal computation or data transformation. Choreography captures interactions from a global perspective meaning that all participating services are treated equally. In other words, choreography encompasses all interactions between the participating services that are relevant for the choreography's goal [12, 13].

2.3 Orchestration of Network Services

The orchestration of the network services refers to the automated arrangement, coordination and management of computer systems, storage, security and networks in order to efficiently deliver application services to end users. Orchestration triggers an executable process that involves centrally controlled message exchanges among network entities. As part of the orchestration of network services, the orchestration system provides a layer of abstraction between the application services and the infrastructure. This layer of abstraction is sometimes referred to as network services virtualization (NSV).

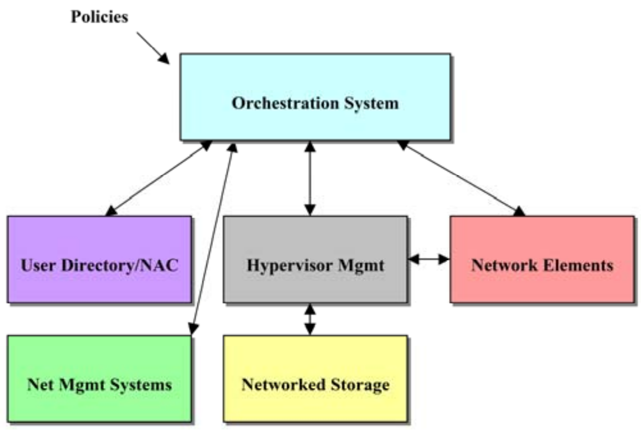

Figure 1. Orchestration System with Centralized Control [14].

A conceptual model of an orchestration system is provided in Fig. 1 including its relationships with the infrastructure. Polices define the relationship between users, computing resources, security and network services. These policies are automatically translated in real-time into device configurations that dynamically provide the necessary resources and/or modify the resource pool to bring it into alignment with the service definition. User access information is updated in user directories and network access control system databases. The virtual server requirements of the service are communicated to the hypervisor management system. The hypervisor management system provides the required services. The hypervisor manager will also be able to make the necessary configuration changes for virtual switches and virtual appliances under its control.

Network elements not under the control of the hypervisor management system (e.g., routers, switches, firewalls, and other external devices that provide secure access to networked services) are controlled on an individual basis by the orchestration system. The orchestration system also communicates with traditional network management solutions, such as fault and performance management systems, which help to monitor services and assure that type of service level specified in the policies delivered for each network service [14, 15].

In the next section, we shape the definition of the requirements to the services recognized as converged, based on the above analysis of the behavioral models for the service composition.

3 Defining the Requirements to the Services Recognized as Converged

The term "converged network" has been used primarily as a reference to integration of the traditional telephone networks with IP-based networks. However, lately the convergence has been interpreted as the more general integration of wireline and wireless networks, also referred to as fixed-mobile convergence (FMC). To merge those views, it could be stated that a converged network is a communications network that seamlessly integrates circuit, packet, wired, and wireless networks.

This characterization allows moving further on to services exploring. Without getting into a rigorous definition of a service, since this may differ from one context to another, it is a fact that each network layer in any communications network offers some services to the layer above, until eventually they are composed into an end-to-end service that is offered to an end consumer (e.g., an end user or a system operating on behalf of an end user or business entity). Among other reasons, network convergence is desirable because of the potential to offer better end-to-end services. But at the same time, service convergence cannot be reduced to services offered by converged networks simply because users expect more than just removing the technical differences between networks; users expect composed services that satisfy their increased needs for collaboration, customization, and personalization. A battle that is going on between network equipment providers and IT vendors also adds a new convergence dimension - convergence between architectures and technologies from different domains, all with the declared intent to improve the end-to-end services to the consumer.

Taking the above described pros and cons into consideration while trying to keep the definition reasonably simple, it is proposed [16] that converged services are complex services that combine media, data communications, and telecommunications services through seamless interactions of technologies, features customized by policies, and personal preferences to deliver new value-added services.

It seems reasonable to assume that converged services are services that may be offered when making use of converged networks (as defined), although that may not be the only possibility [15, 16].

Consequently, the services supported by converged networks have to fulfill the following requirements:

1. Requirement 1 - any-to-any communication - have to support communication services in any kind of configuration (e.g., any-phone-to-PC, "any IP terminal"-to-phone, and so on); in addition supported topologies must be one-to-one, one-to-many, many-to-many.
2. Requirement 2 - customer-centered - user is represented in the network by a user profile that contains and/or refers to all the relevant information concerning personal subscriptions, preferences, constrains, etc.; the user profile has to be shared by any service; additional preferences, constrains, etc. could derive from the composition of profiles into groups and business relationships (e.g., user/subscriber).
3. Requirement 3 - services should enable communication between people, machines, and applications. QoS should be negotiable in order to match the requirements of users for any single instance of communication.
4. Requirement 4 - seamless service access - users must be able to access to services by means of any terminal, according to the terminal capabilities, in the same way, with the same subscribed features, etc.
5. Requirement 5 - application-network synergy - services could be activated by applications and could interact with application/systems, possibly deployed in third party administrative domains (e.g., enterprises, added value service providers) [17 ÷ 20].

The next section presents the detailed description of the converged services taxonomy.

4 Converged Services Taxonomy

The converged service categories discussed and modeled in this paper are can be classified as follows [17, 18]:

- Internet Call Waiting (ICW) - Internet call waiting service enables a user engaged in a dial up Internet session to be alerted when an incoming call has arrived. After the Internet user has been alerted, he is given several options for handling the call e.g., forwarding it, sending a waiting announce/tone, accepting the call over telephone network suspending the Internet session, or accepting the call over IP keeping alive the Internet session.

- Virtual Second Line - allows the subscriber to answer incoming phone calls while his single telephone line is busy due to an ongoing Internet session. A vocal gateway can be used to transform the incoming telephone call into a voice over IP flow directed to the terminal interconnected to the Internet. In this way, the terminal could manage the IP flow carrying the voice along with the other IP flows originated by the web surfing.

- Click-to-Dial (Request-to-Call) - A user is able to initiate a telephone call by clicking a button during a web session. The called address (as well as the caller address) is either an IP address or a phone line number. The charging party could be either the initiator or one of the called parties.

- Unified Communication - allows users to send, retrieve and receive messages disregarding the format and the terminal where the user is connected. The user must be able to create and respond to multimedia messages from any terminal and create and send any type of message without regard to the recipient's mailbox requirements.

- Virtual Presence - allows its subscribers to be reached anywhere, anyway by using both asynchronous messages and real time communication and from any terminal independently of the type of terminal he is logged to. The aim of the service is to provide an integrated set of features that enable for example a subscriber to control the incoming calls according to a set of personal screening/routing rules.

The taxonomy of converged services performed in this section serves as a catalogue framework for our further research towards the "converged services orchestration" specification model, as we indicate it in the abstract.

5 ICW Scenario Alignment vs. Orchestration

According to the definitions from Section 2 - service choreography is not executed - it is enacted when its participants execute their roles [10].

Thus, we take typical converged service (i.e., Internet call waiting) and develop on its basis an orchestration model, scenario-based.

Internet call waiting (ICW) is a service that enables a user engaged in a dial-up Internet session to be alerted when an incoming call has arrived. After the Internet user has been alerted, he is given several options for handling the call (e.g., forwarding it, sending a waiting announce/tone, accepting the call over telephone network suspending the Internet session, or accepting the call over IP keeping alive the Internet session). In parallel the caller is announced that the callee is busy and (s)he is asked to hold the line. In order to enable the service the subscriber has to use a client software that performs the registration phase by storing the association between the telephone network line number and the IP address of his/her Internet session.

The service features those have an impact on the context, e.g., handling an incoming call, are as follows - the subscriber receives a telephone call directed to his phone, which is busy, since it is engaged in a dial-up Internet session. The IN service switching point (SSP) triggers the service logic to handle this call event. The caller is connected to an intelligent peripheral in order to send a message to inform other side to wait (call queuing). The service retrieves the IP address of the callee and then, depending on the user profile, notifies the incoming call to the Internet user with caller number or name or other call relater information. The Internet user may choose different options:

(1) accept call on PC using voice over IP;
(2) accept call on phone;
(3) suspend IP session and answer the call on the phone;
(4) reply the caller with a pre-registered message;
(5) reject the call [17, 19 ÷ 23].

5.1 Connection to Internet

The ICW subscriber connects to Internet by means of a dial-up connection (Table 1, Fig. 2) [17 ÷ 23].

Table 1. ICW – Connection to Internet

Pre-conditions	User A has subscribed to ICW service
Action 1	The dial-up connection is established.
Action 2	User A launches the Internet connection software, which dials the ISP phone number, that is an number which represents an IN service. As a result of the pre-arranged agreement between the Internet service provider and the network provider the DP3 (AnalyzedInformation) was set.
Action 3	The SSP triggers the service logic by sending an InitialDP message to the IN service control point (SCP). Then the SSP, through a gateway, notifies the ICW service logic about a network event related to an incoming call for the ISP phone number. The service logic is executed within a kind of SLEE (service logic execution environment) that provide API to interact with network functionality. The ICW service logic consequently sets TDP13 using some management interface on the SSP. The ICW service logic then subscribes to call event related to user A call link disconnection since it needs to disarm the TDP13, when the Internet connection is disconnected.
Action 4	The ICW service logic gives instruction to the network to route the call to the address of the network access server (NAS) to be connected. Furthermore it arms DP7 (OAnswer) in order to be notified about the result of the call routing.
Action 5	The connection is set up between the SSP and the NAS and consequently the network acknowledges the establishment of the connection. Since the ICW service logic needs to monitor NAS disconnection as well as user A disconnection, it subscribed to call event (DP9ODisconnect) related to NAS disconnection (this subscription could not be requested before the successful notification of the call routing to the NAS). When the network acknowledges the successful establishment

	of the call link towards the NAS, the call processing is stopped at the DP7 (OAnswer) since it has been set as an EDP-R. The ICW service logic instructs the SCP to continue the call processing (this is mapped onto the INAP operation Continue).
Action 6	The subscriber is now requested to authenticate through a login and password. This subscriber data are sent to a RADIUS server to authenticate him/her and to retrieve from a directory server user's A profile based on its home phone number.
Action 7	The association between the IP address of user A and home phone number of user A is stored in his user profile. The software in charge of handling invitations is launched automatically after the connection to Internet is established e.g. IETF SIP UAC.
Post-conditions	Trigger detection point (DP) 13 (in ETSI core INAP terminology DP13 corresponds to TCalledPartyBusy event) is armed so that when there is an incoming call for user A and his line is busy the ICW is notified.

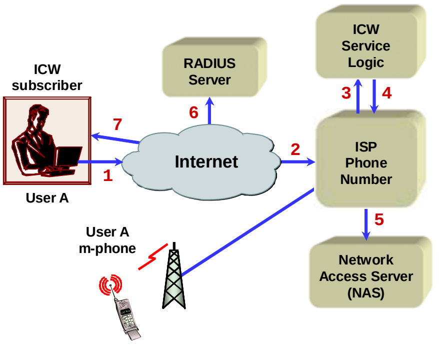

Figure 2. ICW – Connection to Internet.

5.2 Rejecting a Call

The user A is invited to a call by a user B and chooses to reject it (Table 2, Fig. 3) [17 ÷ 23].

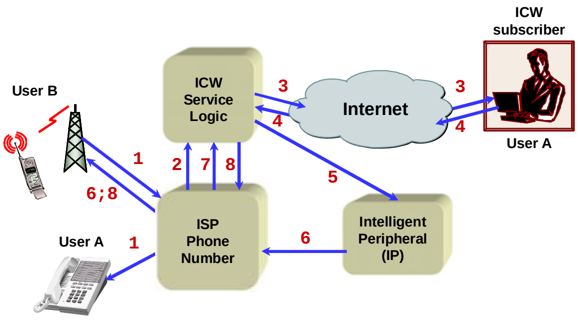

Figure 3. ICW – Rejecting the Incoming Call.

Table 2. ICW – Rejecting the Incoming Call

Pre-conditions	User A is connected to Internet via a dial-up connection, an incoming call has arrived for User A, User A chooses to reject the call.
Action 1	An user B invites the user A to a call.
Action 2, 3, 4	The user A chooses to reject it.
Action 5, 6	When the Internet user rejects the incoming call the ICW service logic requests to play an announcement to the user B saying that the callee has rejected the call. The service logic asks to play an announcement.
Action 7	When the announcement is finished the SSP notify the ICW service logic about the end of it.
Action 8	The ICW service logic then releases the call initiated by the user B.
Post-conditions	The incoming call is terminated. The dial-up connection is still active.

5.3 Accepting a Call on the Phone

The user A is invited to a call by a user B and chooses to answer on his PSTN line (Table 3, Fig. 4) [17 ÷ 23].

Table 3. ICW – Accepting the Call on the Phone

Pre-conditions	User A is connected to Internet by means of a dial-up connection, an incoming call has arrived for the User A, the User A chose to answer the call on his PSTN line.
Action 1	User A is invited to a call by an user B.
Action 2, 3	The ICW client software disconnects the dial-up connection.
Action 4	The "disconnect" signal is sent to the SSP. This is triggered by the SSP since an EDP9 (ODisconnect) related to the call between user A and the NAS was previously armed.
Action 5	Consequently the SSP sends an EventReportBCSM operation to the ICW service logic. The ICW service logic disarms TDP13 on user's A phone line by means of some management interface.
Action 6	Since a connection to an IVR was still established, the ICW service logic releases this connection. The SSP disconnect the connection to the IVR.
Action 7	And, sends to the NAS a message to disconnect it.
Action 8	As the result of the previous arming of an EDP9 (ODisconnect) related to the disconnection of the NAS the SSP sends an EventReportBCSM operation to the ICW service logic.
Action 9	The NAS detects the end of the dial-up connection and instructs the RADIUS gateway to stop the accounting.
Action 10	The RADIUS gateway asks the accounting server to stop the accounting for user's A account. The user's A profile is updated by deleting the IP address from the scope of the previous dial-up connection.
Action 11	The ICW service logic has just released the call to the NAS, therefore it tries to route the call to user's A phone line. The call finally is routed to user's A phone line.
Post-conditions	The dial-up connection is terminated and the call is established between the User A and the User B using PSTN to PSTN connection.

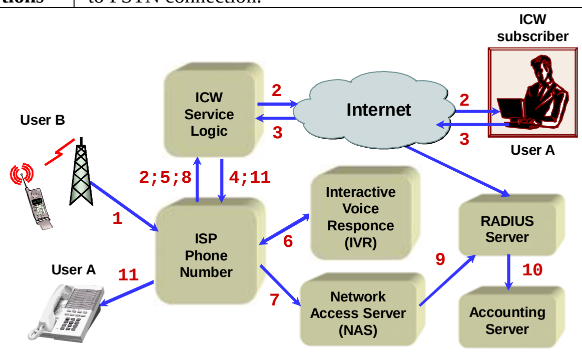

Figure 4. ICW – Accepting the Call on the Phone.

5.4 Accepting the Call on IP

The user A during his Internet session chooses to answer an incoming call using VoIP (Table 4, Figure 5) [17 ÷ 23].

Table 4. ICW – Accepting the Call on IP

Pre-conditions	The User A is connected to the Internet, an incoming call has arrived to User A, User A chooses to answer the call over IP.
Action 1, 2, 3	The user A is connected to Internet, an incoming call has arrived for user A, user A chooses to answer the call over IP.
Action 4	Since a connection to an IVR was still established, the ICW service logic release this connection.
Action 5	The SSP disconnect the connection to the IVR.
Action 6	The ICW service logic retrieves User's A IP address and instructs the network to route the call to the appropriate VoIP gateway and requests to be notified of the outcome of the call routing.
Action 7	The SSP establish the PSTN connection to the VoIP gateway. The ICW service logic controls the VoIP gateway to terminate the call to user's A IP address. Depending on the VoIP gateway used, different control interfaces are possible e.g. H323, MeGaCo.
Action 8	The VoIP gateway terminates the call to user's A IP address.
Post-conditions	The call is established between User A and the User B using a VoIP gateway.

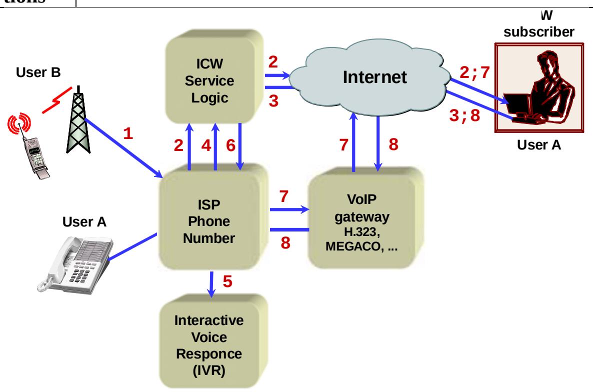

Figure 5. ICW – Accepting the Call on IP.

To complete and substantiate our approach (e.g., scenario-based service alignment and modeling), it is essential to make some statements. Thus sequentially: (1) the innovations flourish in a receptive service environment, so that service groups are commonly at odds with their parent enterprises/businesses. Service derivatives have resulted in a prospering service economy through entrepreneurship and innovation. Therefore, it is important to recognize that service is a business, and that the principles apply equally well to internal and external service organizations. (2) The services are ubiquitous so practically everyone knows what one is. However, what most people do not think about, is that a service is a process, usually a collection of activities to support that process. The activities are organized into components. A component is an organizational entity for instantiating services. Some components provide more than one service and some services are comprised of more than one component. Collectively, the arrangements of components that make up a service offering constitute its architecture. In service architecture, some components are internal people or units, some components are outsourced, and some components are business partners. (3) So, an effective management of service architecture is needed. The main aspect of service management is the choreography of components in a specific business process – that is, how information or tasks is passed between components without explicit direction. Another important aspect of service management is keeping track of the components and their attributes [24, 25].

Thus, a practical scenario of the service execution scheme examined - in order to find main imperative aspects of what and how it can influence the entire service management/control framework while applying different service technologies for different service components.

6 Related Work and Approaching Specification Efforts

As it is defined in the paper title and abstract, we study here converged services orchestration and choreography strategies and develop a scenario-based approach for an effective services management. The main idea of our research, in general, is to develop a specification model/framework for telco services orchestration based on working successfully converged services, e.g., based on best-practices.

This paper is one of the steps to build up the specification strategy and model. We plan to analyze further all converged services, mentioned in Section 4, in order to build an all-encompassing framework.

Furthermore, after examining the research efforts in this area, e.g., academic as well as industrial ones, we have concluded that: (1) it is about impossible to arrange a specification framework for the orchestration of new services without considering practical scenarios of effectively working models; (2) almost of research efforts covers certain aspects only, and does not targets towards pervasive specification.

Namely, general (without specific details) standards of architecture are analyzed in [3, 12, 13, 14 and 28]; business related issues are discussed in [15, 24, 25, 26 and 27]; single performance features of particular service without considering specification strategy are examined in [5, 6, 10, 17, 21, 23]; and the general regulatory approaches are surveyed in [1, 2, 4, 8, 9, 12, 27 and 28].

Thus, we attempt to develop in this paper a viable alternative to make up these drawbacks.

7 Conclusions

The origination of new integrative service creation and modeling technologies along with converged networks comprises unique business models for Telco service providers (SPs). Such business models allow SPs to protect their revenue streams by integrating their networks with other IT-oriented technologies. As a consequence, a central topic in recent research is the creation of technologies that contribute to convergence within heterogeneous service infrastructures.

Thus, we examined in this paper the requirements and an approach for service architecture alignment enabling an efficient provisioning of converged applications [26].

Furthermore, according to the fact that converged services alignment (e.g., orchestration and choreography) has been the main issue of numerous regulatory initiatives, and these initiatives had to face various difficulties and mixed degrees of success, but none of them has yet achieved both de facto and de jure a final specification grade, this paper has revised the converged services orchestration and choreography concerns meant for the effective enterprise management with respect to a framework where service composition is advanced from multiple interconnected perceptions.

Besides, since the regulatory initiatives in this area have not been built on top of a defined and all-encompassing conceptual foundation, we presented a viable alternative to make up for this drawback [12].

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A Cache Replacement Policy Based on Neural Networks Applied to Web Map Tile Caching

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Abstract— *Web mapping has become a popular way of distributing interactive digital maps over the Internet. Traditional web map services generated map images on the fly each time a request was received, which limited service scalability and offered a poor user experience. Most popular web map services, such as Google Maps or Microsoft Virtual Earth, have demonstrated that an optimal delivery of online mapping can be achieved by serving pre-generated map image tiles from a server-side cache. However, these caches can grow unmanageably in size, forcing administrators to use partial caches containing just a subset of the total tiles. Web map tile caching is a paging problem, and the same strategies applied to other paging problems, such as main memory management, can be applied. When the cache runs out of space for allocating incoming requests a cache replacement algorithm must determine which tiles should be replaced. This paper proposes a cache replacement policy based on neural networks to take tile replacement decisions in a Web map cache. Neural networks are trained using supervised learning with real data-sets from public web map servers. High correct classification ratios have been achieved for both training data and a completely independent validation data set, which indicates good generalization of the neural network.*

Keywords: Web mapping, Tile cache, Replacement policy, Neural networks.

1. Introduction

The Web Map Service (WMS) standard of the Open Geospatial Consortium (OGC) offers a standardized and flexible way of serving cartographic digital maps of spatially referenced data through HTTP requests [1]. However, spatial parameters in requests are not constrained, which forces images to be generated on the fly each time a request is received, limiting the scalability of these services.

A common approach to improve the cachability of requests consists of dividing the map into a discrete set of images, called tiles, and restrict user requests to that set. Several specifications have been developed to address how cacheable image tiles are advertised from server-side and how does a client request possibly cached image tiles. The Open Source Geospatial Foundation (OSGeo) developed the WMS Tile Caching (usually known as WMS-C) proposal [2], while the OGC has recently released the Web Map Tile Service Standard (WMTS) [3] inspired by the former and other similar initiatives.

Most popular commercial services, like Google Maps, Yahoo Maps or Microsoft Virtual Earth, have already shown that significant performance improvements can be achieved by adopting this methodology, using their custom tiling schemes.

Map image tiles can be cached at any intermediate location between the client and the server, reducing the latency associated to the image generation process. Tile caches are usually deployed server-side, serving map image tiles concurrently to multiple users. Moreover, many mapping clients, like Google Earth or Nasa World Wind, have embedded caches, which can also reduce network congestion and network delay.

The problem that arises when deploying these server-side caches in practical implementations is that the storage requirements are often prohibitive for many organizations, thus forcing to use partial caches containing just part of the total tiles. Even if there are enough available resources to store a complete cache of the whole map, many tiles will never be requested, so it is not worth it to cache those "unpopular" tiles because no gain will be experienced.

When the cache runs out of space it is necessary to determine which tiles should be replaced by the new ones. The cache replacement algorithm proposed in this paper uses a neural network to estimate the probability that a request of a tile occurs before a certain period of time. Those tiles that are not likely to be requested shortly are good candidates for replacement (assuming the cost to fetch a tile from the remote server is the same for all tiles).

Another problem that must be addressed is to maintain cache consistency, that is, cached copies should be updated when the originals change. A common approach consists of establishing a Time-To-Live (TTL) to cached tiles. The cache considers a stored copy out-of-date if its TTL has expired, and stale tiles are removed from the cache. Those removed

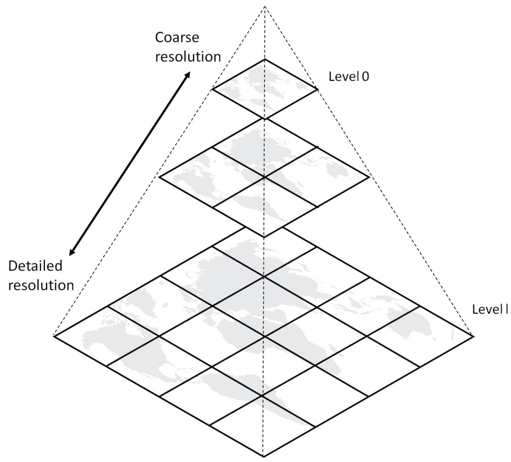

Fig. 1: Tile pyramid representation.

tiles with high probability of being requested in a near future should be requested again to receive and store a fresh copy.

The rest of the document is organized as follows. Section 2 provides a brief background about how the map is tiled in order to offer a tiled web map service. Section 3 presents related work in replacement algorithms. Section 4 describes the trace files used for training and simulation. In Section 5 the proposed replacement algorithm based on neural network is discussed. Results are presented in Section 6. Finally, the conclusions of the paper are gathered together in Section 7, and Section 8 indicates directions for future work.

2. Tiling Space

Maps have been known for a long time only as printed on paper. Those printed cartographic maps were static representations limited to a fixed visualization scale with a certain Level Of Detail (LOD). However, with the development of digital maps, users can enlarge or reduce the visualized area by zooming operations, and the LOD is expected to be updated accordingly.

The adaptation of map content is strongly scale dependent. A small-scale map contains less detailed information than a large scale map of the same area. The process of reducing the amount of data and adjusting the information to the given scale is called cartographic generalization [4].

In order to offer a tiled web map service, the web map server renders the map across a fixed set of scales through progressive generalization. Rendered map images are then divided into tiles, describing a tile pyramid as depicted in Fig. 1.

For example, Google Maps uses a tiling scheme where level 0 allows representing the whole world in a single tile of 256x256 pixels (where the first 64 and last 64 lines of the tile are left blank). The next level represents the whole world in 2x2 tiles of 256x256 pixels and so on in powers of 2. Therefore, the number of tiles n grows exponentially with the resolution level l. Using this tiling scheme, to cover the whole world with a pyramid of 20 levels, around 3.7×10^{11} tiles are required and several petabytes of disk are used.

3. Related Work

Most important characteristics of Web objects used in Web cache replacement strategies are: *recency* (time since the last reference to the object), *frequency* (number of requests to the object), *size* of the Web object and *cost* to fetch the object from its origin server. Depending on the characteristics used, replacement strategies can be classified as recency-based, frequency-based, recency/frequency-based, function-based and randomized strategies [5].

Recency-based strategies exploit the temporal locality of reference observed in Web requests. These are usually extensions of the well-known LRU strategy, which removes the least recently referenced object. Pyramidal Selection Scheme (PSS) [6] algorithm uses a pyramidal classification of objects depending upon their size, and objects in each group are maintained as a separate LRU list. Only the least recently used objects in each group are compared, and the object selected for replacement which is maximizing the product of its size and the number of accesses since the last time it was requested.

Frequency-based strategies rely on the fact that popularity of Web objects is related to their frequency values. These strategies are built around the LFU strategy, which removes the least frequently requested object.

Recency/frequency-based strategies combine recency and frequency information to take replacement decisions.

Function-based strategies employ a general function of several parameters to make decisions of which object to evict from the cache. GD-Size [7] strategy uses a function of size, cost and an aging factor. GDSF [8] is an extension of GD-Size that also takes account for frequency. Least-Unified Value (LUV) [9] rates an object based on its past references to estimate the likelihood of a future request, and normalizes the value by the cost of the object per unit size.

Randomized strategies use a non-deterministic approach to randomly select a candidate object for replacement.

For a further background, a comprehensive survey of web cache replacement strategies is presented in [5]. According to that work, algorithms like GD-Size, GDSF, LUV and PSS were considered "good enough" for caching needs at the time it was published in 2003. However, the explosion of web map traffic did not happen until a few years later.

Despite the vast proliferation of web cache replacement algorithms, there is a reduced number of replacement policies specific to map tile caches that benefit of spatial correlation between requests.

Map tile caching can benefit from spatial locality principle as stated by the Tobler's first law of geography, which states that *"Everything is related to everything else, but near things are more related than distant things"* [10].

In [11] and [12], a probability-based tile pre-fetching algorithm and a collaborative cache replacement algorithm for Web geographical information systems (Web GISs) are proposed. The server collects and maintains the transition probabilities between adjacent tiles. With these probabilities the server can predict which tiles have the highest probability of access in next time than others, based on the global tile access pattern of all users and the semantics of query. The proposed cache replacement algorithm determines which tiles should be replaced based on the estimated future access probabilities.

In [13], [14], [15] the web application *Hotmap* is presented. This application analyzes log requests from the Microsoft's Live Search Maps service and visualizes the number of requests of each tile accurately on a map. These studies reflect that several features which drive most people's attention, like shorelines, roads and populated places, can be identified. [16] proposes that significant performance improvements can be achieved by caching tiles corresponding to those features known to be of interest to Web map users.

The use of neural networks for cache replacement was first introduced by Khalid [17], with the KORA algorithm. KORA uses backpropagation neural network for the purpose of guiding the line/block replacement decisions in cache. The algorithm identifies and subsequently discards the dead lines in cache memories. It is based on previous work by Pomerene et al. [18], who suggested the use of a shadow directory in order to look at a longer history when making decisions with LRU. Later, an improved version of the former, KORA-2, was proposed [19], [20]. Other algorithms based on KORA were also proposed [21], [22].

A survey on applications of neural networks and evolutionary techniques in web caching can be found in [23].

[24], [25], [26], [27], [28] proposes the use of a backpropagation neural network in a Web proxy cache for taking replacement decisions.

A predictor that learns the patterns of Web pages and predicts the future accesses is presented in [29].

[30] discusses the use of neural networks to support the adaptivity of the Class-based Least Recently Used (C-LRU) caching algorithm.

The novelty and significance of this work resides on the target scenario where it is applied. No similar studies about the application of neural networks to take replacement decisions in a Web map cache have been found in the literature.

Although the underlying methodology has already been discussed in related work on conventional web caching, web map requests' distributions and attributes are very different from those of traditional web servers. In this context, the size of the requested object is heavily related to its "popularity" so it can be used to estimate the probability of a future request. Traditional web caching replacement policies commonly use this parameter only to evaluate the trade off between the hits produced on that object and the space required to store it.

4. Training Data

Simulations are driven by trace files from three different tiled web map services, CartoCiudad, IDEE-Base and PNOA, provided by the National Geographic Institute (IGN) of Spain. CartoCiudad is the official cartographic database of the Spanish cities and villages with their streets and roads networks topologically structured. PNOA serves imagery from the Aerial Ortophotography National Plan, which updates every two years Spanish covers by aerial photography, high resolution and accuracy digital ortophotography, and high density and accuracy Digital Terrain Model. IDEE-Base allows viewing the Numeric Cartographic Base 1:25,000 and 1:200,000 of the IGN.

Trace files were filtered to contain only valid web map requests, so the neural network is trained with requests that would actually be cached. Each trace comprises requests received from the 1^{th} to 7^{th} of March in 2010. Between these dates, a total of 25.922, 94.520, and 186.672 valid map requests were received respectively for CartoCiudad, IDEE-Base and PNOA.

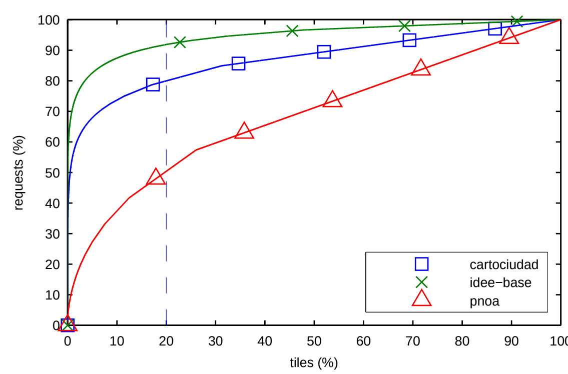

Fig. 2: Percentile of requests for the analyzed services.

It must be noted that the performance gain achieved by the use of a tile cache will vary depending on how the tile requests are distributed over the tiling space. If those were uniformly distributed, the cache gain would be proportional to the cache size. However, it has been found that tile requests describe a Pareto distribution, as shown in Figure 2. Tile requests to the CartoCiudad map service follow the 20:80 rule, which means that the 20% of tiles receive the 80% of the total number of requests. In the case of IDEE-Base, this behaviour is even more prominent, where the 10% of tiles receive almost a 90% of total requests. PNOA requests are more scattered. This happens because about the 90% of requests belong to the two higher resolution levels (19 and 20), the ones with larger number of tiles.

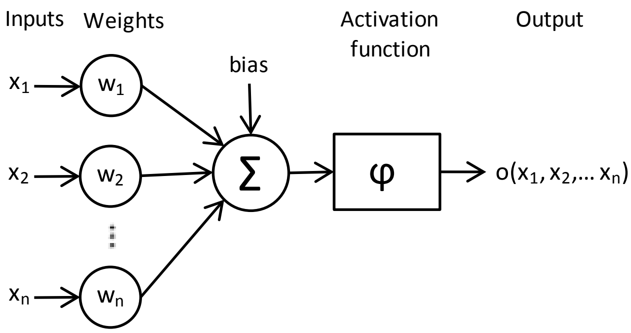

Fig. 3: Artificial neuron.

Services that show Pareto distributions are well-suited for caching, because high cache hit ratios can be found by caching a reduced fraction of the total tiles.

5. Neural Network Cache Replacement

Artificial neural networks (ANNs) are inspired by the observation that biological learning systems are composed of very complex webs of interconnected neurons. In the same way, ANNs are built out of a densely interconnected group of units. Each artificial neuron takes a number of real-valued inputs (representing the one or more dendrites) and calculates a linear combination of these inputs. The sum is then passed through a non-linear function, known as *activation function* or *transfer function*, which outputs a single real-value, as shown in Fig. 3.

In this work, a special class of layered feed-forward network known as multilayer perceptron (MLP) has been used, where units at each layer are connected to all units from the preceding layer. It has an input layer with three inputs, two-hidden layers each one comprised of 3 hidden nodes, and a single output (Fig. 4). According to the standard convention, it can be labeled as a 3/3/3/1 network. It is known that any function can be approximated to arbitrary accuracy by a network with three layers of units [31].

Learning an artificial neuron involves choosing values for the weights so the desired output is obtained for the given inputs.

Network weights are adjusted through supervised learning using subsets of the trace data sets, where the classification output of each request is known. Backpropagation with momentum is the used algorithm for training. The neural network parameters, such as the learning rate, momentum constant or network size, have been carefully chosen through experimentation. The parameters that provide the best convergence of the neural network are summarized in Table 1.

5.1 Neural Network Inputs

The neural network inputs are three properties of tile requests: recency of reference, frequency of reference, and the size of the referenced tile. These properties are known to be important in web proxy caching to determine object cachability.

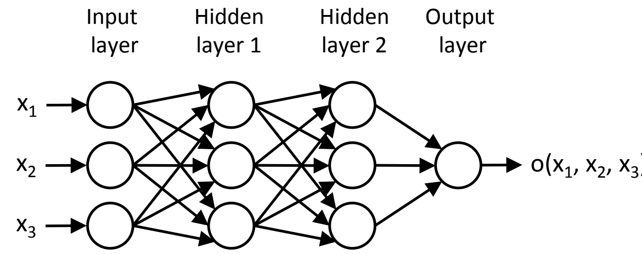

Fig. 4: Proposed two-hidden layer feed-forward artificial neural network.

Table 1: Neural network parameters

Parameter	Value
Architecture	Feed-forward Multilayer Perceptron
Hidden layers	2
Neurons per hidden layer	3
Inputs	3 (recency, frequency, size)
Output	1 (probability of a future request)
Activation functions	Log-sigmoid in hidden layers, Hyperbolic tangent sigmoid in output layer
Error function	Minimum Square Error (mse)
Training algorithm	Backpropagation with momentum
Learning method	Supervised learning
Weights update mode	Batch mode
Learning rate	0.05
Momentum constant	0.2

Inputs are normalized so that all values fall into the interval $[-1, 1]$, by using a simple linear scaling of data as shown in Equation 1, where x and y are respectively the data values before and after normalization, x_{min} and x_{max} are the minimum and maximum values found in data, and y_{max} and y_{min} define normalized interval so $y_{min} \leq y \leq y_{max}$. This can speed up learning for many networks.

$$y = y_{min} + (y_{max} - y_{min}) \times \frac{x - x_{min}}{x_{max} - x_{min}} \qquad (1)$$

Recency values for each processed tile request are computed as the amount of time since the previous request of that tile was made. Recency values calculated this way do not address the case when a tile is requested for the first time. Moreover, measured recency values could be too disparate to be reflected in a linear scale.

To address this problem, a sliding window is considered around the time when each request is made, as done in [24].

With the use of this sliding window, recency values are computed as shown in Equation 2.

$$recency = \begin{cases} max(SWL, \Delta T_i) & \textit{if object i was requested before} \\ SWL & \textit{otherwise} \end{cases} \qquad (2)$$

Table 2: Size on disk for a tile belonging to a city centre and to a sea region, for the different services.

	CartoCiudad	PNOA	IDEE-Base
city centre	101,08 KB	29,95 KB	12,56 KB
sea region	3,57 KB	5,49 KB	1,73 KB

where ΔT_i is the time since that tile was last requested and SWL is the sliding window length.

Recency values calculated that way can already be normalized as stated before in Equation 1.

Frequency values are computed as follows. For a given request, if a previous request of the same tile was received inside the window, its frequency value is incremented by 1. Otherwise, frequency value is divided by the number of windows it is away from. This is reflected in Equation 3.

$$frequency = \begin{cases} frequency + 1 & \textit{if } \Delta T_i \leq SWL \\ max\left[\frac{frequency}{\frac{\Delta T_i}{SWL}}, 1\right] & \textit{otherwise} \end{cases} \tag{3}$$

Size input is directly extracted from server logs. As opposite to conventional Web proxies where requested object sizes can be very heterogeneous, in a web map all objects are image tiles with the same dimensions (typically 256x256 pixels). Those images are usually rendered in efficient formats such as PNG, GIF or JPEG that rarely reach 100 kilobytes in size.

As discussed in [16], due to greater variation in colors and patterns, the popular areas, stored as compressed image files, use a larger proportion of disk space than the relatively empty non-cached tiles. To illustrate this, Table 2 compares the storage size of a "popular" tile corresponding to a city centre and an "unpopular" tile in the middle of the sea for the analyzed services.

Because of the dependency between the file size and the "popularity" of tiles, tile size can be a very valuable input of the neural network to correctly classify the cachability of requests.

5.2 Neural network target

During the training process, a training record corresponding to the request of a particular tile is associated with a boolean target (0 or 1) which indicates whether the same tile is requested again or not in window, as shown in Equation 4.

$$target = \begin{cases} 1 & \textit{if the tile is requested again in window} \\ 0 & \textit{otherwise} \end{cases} \tag{4}$$

Once trained, the neural network output will be a real value in the range [0,1] that must be interpreted as the probability of receiving a successive request of the same tile within the time window.

A request is classified as *cacheable* if the output of the neural network is above 0.5. Otherwise, it is classified as *non cacheable*.

5.3 Training

The neural network is trained through supervised learning using the data sets from the extracted trace files. The trace data is subdivided into training, validation, and test sets, with the 70%, 15% and 15% of the total requests, respectivelly. The first one is used for training the neural network. The second one is used to validate that the network is generalizing correctly and to identify overfitting. The final one is used as a completely independent test of network generalization.

Each training record consists of an input vector of recency, frequency and size values, and the known target. The weights are adjusted using the backpropagation algorithm, which employs the gradient descent to attempt to minimize the squared error between the network output values and the target values for these outputs [32]. The network is trained in batch mode, in which weights and biases are only updated after all the inputs and targets are presented. It is worth to note that, although training the neural network in on-line mode would allow the system to be adapted progressively, it would require to maintain a registry with the information of past accesses. Due to the exponential nature of the tile pyramid it is impractical to store any information of individual tiles.

The pocket algorithm is used, which saves the best weights found in the validation set.

6. Results

Neural network performance is measured by the correct classification ratio (CCR), which computes the percentage of correctly classified requests versus the total number of processed requests.

Table 3: Correct classification ratios (%) during training, validation and testing for the different services.

	CartoCiudad	PNOA	IDEE-Base
training	76.5952	96.5355	75.6529
validation	70.2000	97.1985	77.5333
test	72.7422	97.4026	82.7867

Figures 5, 6 and 7 show the CCRs obtained during training, validation and test phases for the different services. As can be seen, the neural network is able to correctly classify the cachability of requests, with CCR values over the testing data set ranging between 72% and 97%, as shown in Table 3.

The network is stabilized to an acceptable CCR within 100 to 500 epochs.

Best predictions are made with the PNOA service. As stated before, this service presents an anomalous request

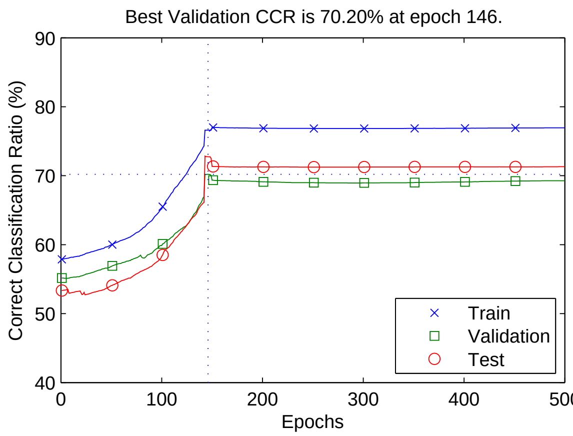

Fig. 5: Correct classification ratios for CartoCiudad.

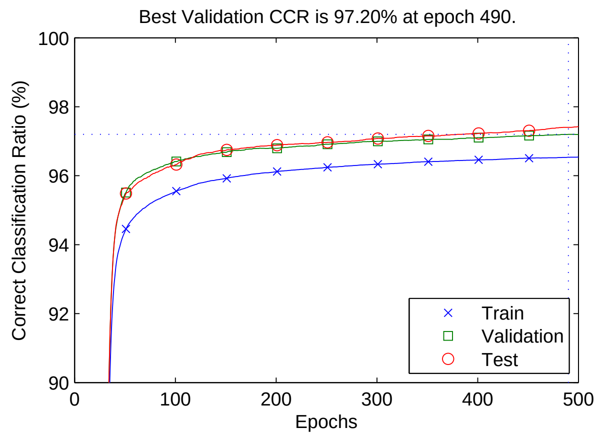

Fig. 7: Correct classification ratios for PNOA.

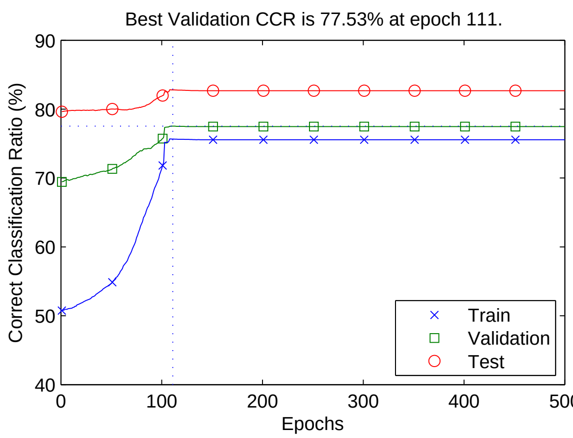

Fig. 6: Correct classification ratios for IDEE-Base.

distribution, with the 90% of requests concentrated in the two higher resolution levels (street levels). In this service, requests are very scattered and it rarely receives two requests of the same tile inside the time window. For this reason, it is easy for the neural network to learn this fact and correctly classify requests as non-cacheables, thus obtaining high CCR values.

7. Conclusions

Serving pre-generated map image tiles from a server-side cache has become a popular way of distributing map imagery on the Web. However, storage needs are often prohivitive which forces the use of partial caches. In this work, a cache replacement policy based on neural networks has been proposed. The network is a feed-forward multilayer perceptron with two-hidden layers. Inputs to the neural network are recency, frequency and size of the requested tiles, and it outputs a real value in the range [0,1] which can be interpreted as the probability of receiving a future request of the same object before a certain period of time. It has been trained with backpropagation through supervised learning using real-world trace requests from different map server logs. Trace data sets have been divided into three groups; one for training the neural network, another one for validating network generalization and a completely independent dataset used for testing. The results show that the proposed neural network is able to accurately classify cachability of requests in several scenarios from the real world with different workloads and request distributions.

8. Future work

While the current work successfully achieves the objective of classifying the cachability of requests, better results could be obtained by adding new spatial parameters as inputs of the neural network, as the resolution level, spatial density of requests or the spatial locality between tiles. As web map usage patterns are scale dependent, it would be easier for the neural network to be trained with requests of an individual resolution level. With this strategy in mind, it would be necessary to train as many neural networks as resolution levels are offered by the map service.

Acknowledgment

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A Method of Detecting Code Cloning in Open Source Software

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Abstract - *In this paper, we discuss an approach to detect code copying in open source software. This approach generates a small fingerprint that includes the main features of the class files contained in a Java Archive (Jar) file. By comparing the intrinsic and extrinsic features of the class files, the approach can find the cloned code in anonymous Jar files. This approach provides a tool to detect cloned code correctly and effectively*

Keywords: Open Source Software, Code Cloning, Intrinsic Features, Extrinsic Features

1 INTRODUCTION

Open source software (OSS) has become critical for most organizations, as there are many advantages of using it [1]: such as high-quality software, lower software costs, numerous technical support and accountability, active communities, and less dependence on vendors. OSS has impact on more than developers and IT-managers but also potentially for all the people throughout the value chain of an organization including suppliers, customers, and partners. Today, more and more OSS is developed and adopted in commercial products. Although organizations have already obtained benefits from OSS, they have to take a critical consideration of OSS and face some questions raised by using it. For example, OSS uses licenses to protect itself and its developers. However, many of these licenses are different and mutually incompatible. Before using OSS, organizations must survey the licenses of the software and obey them to develop their commercial products. However, code of unknown origin is encountered occasionally in development and its pedigree must be made clear. This action is related to the legality and success of software products.

1.1 Problem

In this paper, we look at detecting software clones that are written in the Java programming language. We use the known software code to compare with the software that is copied from an anonymous source. After comparing, we can determine whether an organization is copying another's code illegally. There are many existing clone detection methods, such as string matching. But sometimes those methods don't work when the original source code is not available. To avoid this disadvantage, our method focuses on accessible attributes of class files and calculates the similarity of their attributes.

1.2 Motivation

The Open Source community attracts very bright, very motivated developers, who although are frequently unpaid and lack disciplined [2]. For many developers, peer review and acclaim is important. They prefer to build software with clean design, reliability and maintainability that is admired by their peers. They develop software to contribute the open source community and get benefits from others' contribution. But there are some phenomena, to which we have to pay attention, destroying the balance of the open source community. Some organizations or individuals copy intellectual property, and just make it their own by modifying the code with out any legal right to do so. This behavior shows a complete lack of respect for other developers' work and creates an attack on the ecosystem of the community. We must take action to prevent these unethical behaviors and protect authors' rights and the community. This is our main motivation for detecting cloned code in OSS.

The problems of exploiting OSS to improve one's own product and avoiding infringing others' intellectual property are concerned by commercial software developers and project managers. Some issues [3] are related to this consideration:

- Some commercial software suppliers have been sued by open source advocates for downloading the open source while ignoring the license obligations.
- Customers want to know the details of the software that are provided by IT companies.
- Software engineers use some open source in their programs but they do not declare them.
- Some OSS licenses may influence the intellectual property of software products which include the corresponding open source code.

Based on those issues, organizations must take measures to protect against these legal risks of open source. They should periodically conduct a complete audit of their source code, making sure developers and managers know exactly what open source is inside and what the license obligations of that open source are. But the price of the service supported by a professional audit firm is high, and sometime there is no access to the original source code [8]. We devote our focus to audit open source software with low cost.

1.3 Goals

There are many available applications detecting code cloning with simple string-matching [10]. We pay more attention to the specific attributes of a class, such as parameters of methods and return value. We take those features from a Jar file to create a unique footprint and then calculate the similarity between the known Jar's fingerprint and an unknown Jar file. The application outputs the result to display a match percentage between the two Jars.

1.4 Objectives

Several clone detection techniques have been described and implemented [8] [9] [10]. But most of those techniques require the original source code and require much memory. Our objective is to make a small fingerprint using the particular features of the compiled classes in a Jar file and not all the source code.

Our approach does not rely upon the original source code because it does not tend to be available in the Jar files. We mainly pick up basic features from the Jar files. Although we utilize less information that describes the features of the code, it does not mean low accuracy. Following Walenstein [4], clone detection adequacy depends on applications and purposes. Intellectual property thieves may modify the name and order of methods but rarely features of the class. Class files contain distinguished features that are the key points to comparing two files. Comparing these features is the core idea of our approach.

1.5 Outline

In section 2, we will discuss relevant background work. In section 3 we concentrate on our design, algorithm and what decisions we made. In the section 4 and section 5, we compare several groups of files, present results, draw our conclusions and look at future work.

2 BACKGROUND

Much research in the field of clone code detection has been done. Most of it has a high degree of accuracy. However some disadvantages exist in that software, e.g. time-consuming, memory-waste, and especially requiring the source files. Cate Huston [10] uses winnowing to fingerprint Jar files and some potentially interesting information (e.g. filenames, size of the Jar file, the number of entities, and the Jar name) of the Jars to detect significant similarities of those Jars. From the conclusion of that paper we can learn that those potentially interesting information are less consistent. This demonstrates what can be changed in the software clone and the key point of comparison is the features or contents of methods. Some researchers have paid attention to the situation that there is no source code available but compiled code. Carson Browns and David Barrera [8] use the modification of n-gram method to generate the fingerprint of the compiled Java program. They make much improvement including detection speed, small fingerprint and good accuracy. Patrice Arruda et al [9] use graph to describe the dependency of classes and calculate the similarity with matrix. Their approach focuses on the relationship between classes. We follow this research. We also take Jar files as input, and generate the fingerprint of the Class files that are Java's compiled files. The difference between our approaches and [8] is that we do not consider all the byte code of methods in Class files but just the features of Class files, e.g. the number and type of the input parameters of methods, the type of the return value of methods and the like. These features are termed as intrinsic features in our approach. Although software clones can include many modifications, such as renaming classes, variables and methods, adding some inessential code lines, and changing the path of files, the main function of the method will not be changed. In our approach, the relationship between the classes is considered as the extrinsic features. The idea of extracting extrinsic features of a class in our approach is enlightened by [9].

3 APPROACH

3.1 Design

Considering that Java Jar files mainly comprise class files, our approach analyses the class files and uses features of class files to describe fingerprints of Jar files. We divide the features of class files into two groups: intrinsic features and extrinsic features. Intrinsic features of a class file include the description of its methods. The number and types of input parameters and the type that the method returns compose the basic features of a method in a class. The extrinsic features are composed of the relationships of classes, for example super classes, interfaces, and inner classes of a class. These features can indicate the purpose of the class file so that the approach can distinguish the behavior of cloning. Employing these intrinsic and extrinsic features we discover the cloning code.

3.2 Rules

Generally, Jar files are kinds of Zip files. Thus before extracting features from class files, we need to decompress the Jar file using the Java Class Foundation Library API [5], java.util.Jar, to finish this job. We use the JarResource class which is from a Java World article [6] to obtain the byte codes of all classes in a Jar file. The org.netbeans.modules.classfile API [7] can transform the byte codes of a class to an object in memory. Therefore our algorithm can manipulate class files to get all the features of methods in a class file. To clarify our rules of calculation, we assume that X is an original Jar file which is used to generate a fingerprint and Y is an anonymous Jar file which is possible the cloning Jar file. C_i denotes a Class file that belongs to X. D_j denotes a Class file that belongs to Y. M_k denotes a method that belongs to C_i. N_l denotes a method that belongs to D_j. In our algorithm, we use eight basic rules to demonstrate how to calculate the similarity of two Jar files. Based on the similarity, the cloning Jar file can be detected correctly.

The similarity of two methods is calculated by Rule 1:

Rule 1: If two methods have same number of parameters, same type of parameters and the same type of return value, then the similarity of two methods S_{method} is 1; otherwise, the similarity S_{method} is 0.

If the similarity of two methods is obtained, the similarity of intrinsic features of two classes can be analyzed by Rule 2:

Rule 2: the similarity of classes

$$S_{in-class} = \frac{\sum_{k=1}^{n} S_{method}}{n}$$

n is the number of methods in a class.

The similarity of extrinsic features of two classes can be calculated by Rule 3 to Rule 6:

Rule 3: the similarity

$$S_{ex-class} = \left(S_{SuperClass} + S_{Interface} + S_{innerClass}\right)/3$$

Rule 4: the similarity $S_{SuperClass}$ is 1 if the types of super classes are same; otherwise is 0.

Rule 5: the similarity $S_{Interface}$ is 1 if the number of interfaces is same and the types of interface are same; otherwise is 0.

Rule 6: the similarity $S_{InnerClass}$ is 1 if the number of inner classes is same and the types of inner classes are same; otherwise is 0.

The similarity of two classes is calculated by combining $S_{in-class}$ and $S_{ex-class}$ as Rule 7:

Rule 7: $S_{class} = 0.6S_{in-class} + 0.4S_{ex-class}$

The weight of $S_{in-classes}$ is set to sixty percentage points because our approach mainly focuses on methods in a class.

The similarity of two Jar files is calculated by the Rule 8:

Rule 8: $$S_{Jar} = \frac{\sum_{i=1}^{n} \max S_{classes\_c_i}}{n}$$

n is the number of classes in the original Jar file, $\max S_{classes\_c_i}$ indicates the maximum of similarity between C_i and $\{D_j | 1 \le j \le m\}$, m is the number of classes in the anonymous Jar file.

3.3 Algorithm

Our approach compares the original Jar file and the anonymous Jar file with two components. The first component is Fingerprint Generator that manages to generate a fingerprint from the original Jar file. Figure 1 describes the main steps of generating fingerprint. The Fingerprint Generator visits every class in the original Jar file, obtains the intrinsic features and extrinsic features of a class and finally produces the fingerprint of the Jar file. The second component is Fingerprint Detector that computes the similarity of two Jar files through eight rules mentioned. Figure 2 describes the main steps of generating the similarity.

3.4 Decisions Made

Our approach only focuses on class files in a Jar file because class files are the core of a Jar file. The similarity between two methods is simply calculated by Rule 1 and expressed by 1 or 0. There is no value of similarity between 0 and 1. As a prototype of first implementation, we intuitively keep the algorithms simple. Although this calculation losses some accuracy, the intention can be described clearly and results are reasonably.

We also only consider the basic types of Java such as int, short, long, byte, float, double, string, char and boolean, and the original objects of Java such as Integer. Except these primary types, other self-defined types are uniformly defined as Object type. Through these simple categories of types of parameter, our approach can cover most cases when comparing two methods.

In the course of implementing the algorithm, some classes in the different Jar files are very similar but the similarity calculated by our algorithm is not high. However, some classes are different but the similarity is high. Through analyzing the source code artificially, we find that these classes have several methods that have no parameters and no return value. This circumstance will influence the result of comparison. Assuming that there are three classes A, B and C, Class A has 6 methods (X1, X2, X3, X4, X5, X6) among which there are two methods (X5 and X6) have no parameters and no return value. Class B has 4 methods (Y1, Y2, Y3, Y4). Class C has 4 methods (Z1, Z2, Z3, Z4) among which there are two methods (Z3 and Z4) have no parameters and no return value. Assuming X1, X2, X3 are equal to Y1, Y2, Y3, respectively, so the similarity of Class A and B is 0.5. Assuming X1, X2 are equal to Z1, Z2, respectively, the similarity of Class A and C is 0.67 because Class C has two methods Z3 and Z4 which have no parameters and no return values, and Class A also has X5 and X6 which have no parameters and no return values. Even if Z3, Z4 and X5, X6 are entirely different, the similarity is high based on rules. Actually, Class A is more similar to Class B than Class C. Thus methods of no parameters and return value have negative effect in our algorithm. In most classes, methods of no parameter and return value tend to be less important. Considering this situation, we decided to remove methods that have no parameters and return value when calculating the similarity of methods.

4 RESULTS

To examine the correctness and effectiveness of our approach, eight selected Jar files are used as the testing set. The environment of testing and the performance are described first. Then the testing results and analysis are discussed.

4.1 Performance

The implementation runs on a laptop. Its technical parameters are shown as below:

Processor: Pentium(R) Dual-Core CPU T4500 @ 2.30GHz

Memory: 4.00 GB

System Type: 64-bit Operating System

The size of Jar files ranges from 1Kb to 2.7Mb. There are four Jar files that come from Eclipse IDE [11] plugins, two files from our own implementations, and two files from Spring Framework [12]. The worst case is the largest pair of Jar file. Our approach spends 148.513 seconds for comparison. Although the algorithm runs slowly, the results were good.

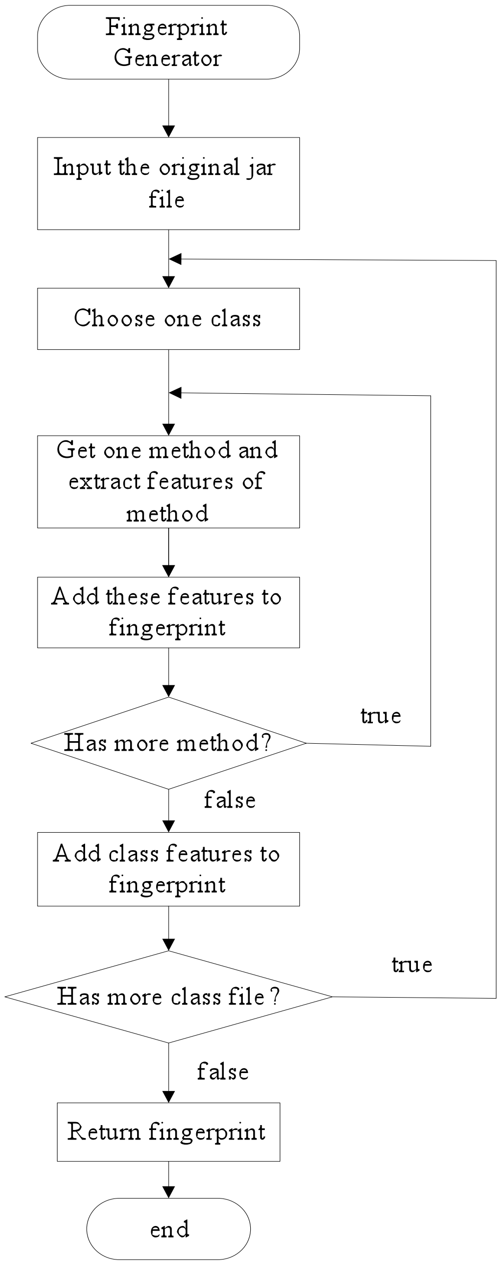

Figure 1. Fingerprint Generator

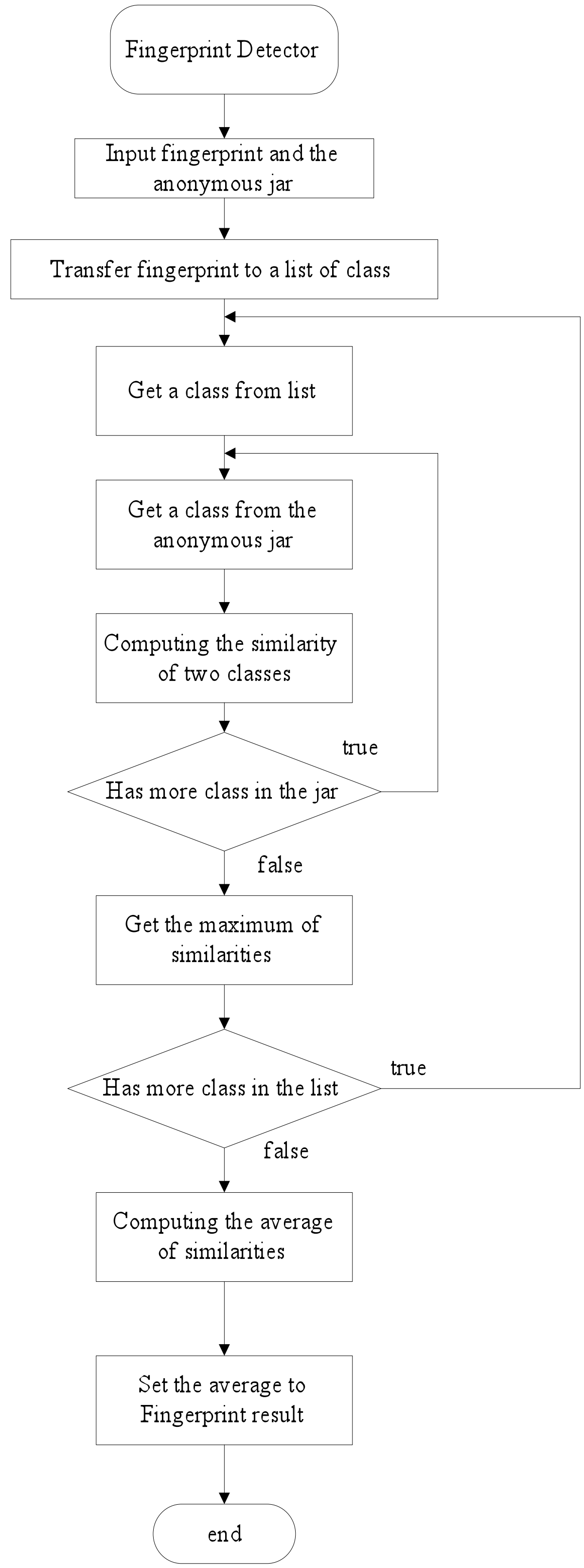

Figure 2. Fingerprint Detector

4.2 Testing Results

In the set of testing Jars, we first choose the simplest Jar file, test.Jar, which contains one class including three simple methods, to compare with this Jar itself. The result is very good as is the execution time. Next, when comparing two entirely different Jar files (test.Jar and com.zxwei.comp5900.Jar), the result computed by our approach is correct. The first two Jar files are devised to test our basic idea. The Jar files, org.eclipse.help.ui_3.2.0.v20060602.Jar and org.eclipse.help.ui_3.5.0.v20100517, were obtained from Eclipse plugins folder under Eclipse IDE. They have similar functions, but they are different versions. The similarity of org.eclipse.help.ui_3.2.0.v20060602.Jar to itself is first calculated and then the similarity to org.eclipse.help.ui_3.5.0.v20100517 is done. The result shows that the former's similarity is higher than the latter's. This data is reasonable because the difference between two versions is larger than one version itself. Working with the different versions of Jar files of JUnit [13], we discover that their similarity is low because JUnit version 4 has many modifications of structures and functions compared with JUnit version 3.8. JUnit version 4 introduces the annotation function to facilitate programming test units. Our approach exploits the number of classes in the first Jar file. Although the similarity is different, the time consumed by calculation is almost equal. Finally, we compare two larger Jar files, one from Spring Framework 2.0.8 which is 2645KB and another from Spring Framework 2.5 which is 2773KB. The result reveals that the similarity of these two Jar files is 89.07% and the calculation execution time is 148.513s. This indicates the higher version of the Spring Frame has some modification but the main part is similar to the old one. Table 1 lists the all our results of our testing set. Through the testing, our approach shows the good performance on both correctness and efficiency.

5 CONCLUSION

Our approach of detecting code cloning in open source software focuses on class files in Jar files. We believe that utilizing the intrinsic and extrinsic features of classes can effectively seek out code cloning and our approach is immune to simple refactoring. Although our approach has slow execution speed, the result were correct and accurate.

5.1 Review goal and contributions

Our algorithm and implementation fulfill the objective of detecting code cloning and provide a tool to help companies or institutions, who want to use open source software, to detect the existence of copied code in their software or software they use, thus avoiding lawsuits from distributed or using code with license incompatibilities.

5.2 Future work

Our approach has three main aspects for improvement. In the calculation of the similarity of two methods, we can set different weight to the number of parameters, types of parameters and types of return value instead of a Boolean value (0 or 1). We believe this improvement can further increase the accuracy of our algorithms. In the aspect of extrinsic features, attributes of a class and references of other objects in a class are both important properties of a class. Adding these features into the calculation of extrinsic features can promote the accuracy of calculating the similarity of two classes. In the aspect of intrinsic features, considering structures of if-else and while clauses can strengthen detailed description of a method, thereby increasing the accuracy of the algorithm.

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TABLE I. TESTING RESULTS

Fingerprint Generated From	Anonymous Java Jar File	Certainty	Time (s)
test.Jar	test.Jar	100%	0.015
test.Jar	com.zxwei.comp5900.Jar	0%	0.017
org.eclipse.help.ui_3.2.0.v20060602.Jar	org.eclipse.help.ui_3.2.0.v20060602.Jar	98.846%	3.959
org.eclipse.help.ui_3.2.0.v20060602.Jar	org.eclipse.help.ui_3.5.0.v20100517.Jar	94.313%	3.797
junit-3.8.Jar	junit-3.8.Jar	98.0%	1.01
junit-3.8.Jar	junit-4.1.Jar	25.396%	0.875
junit-4.1.Jar	junit-4.1.Jar	31.675%	0.891
spring-2.0.8.Jar	spring-2.5.Jar	89.07%	148.513

SESSION

THE 9TH INTERNATIONAL WORKSHOP ON XML TECHNOLOGY AND APPLICATIONS - XMLTech'11

Chair(s)

TBA

EXTRACTING INFORMATION SCIENCE CONCEPTS BASED ON JAPE REGULAR EXPRESSION

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Abstracts *Recently, an unstructured data on the www has generated important further interests in the extraction text, email, webpage, report, and research papers in its raw form. Far more interesting, extracting information from a specific domain using distribute corpora from World Wide Web is vital step towards creating corpus annotation. This paper describe a methods of annotation concepts of Information Science to build domain ontology using Natural Language programming NLP technology to speed up the developing ontology process as time consuming and experts in the domain has many barriers as time and loads to do. Using some NLP to reduce the domain experts work and they can be evaluated the results.*

Keywords *ontology– Regular expression- Information extraction – Natural Language Programming.*

1 Introduction

Recently Information Extracting (IE) has a great interesting in the area of emerging web pages on the internet which contains unstructured data. This amount of information available on the Internet needs a tool to extract to make it available to use in the right time. Many specialists in the field of extracting information have worked to find suitable tools, as Wrappers, that classify interesting data and mapping them to some appropriate formats XML or relational database Furthermore, some HTML aware tools can be based on inheriting constructural features of documents to achieve the extracting process. On the other hand, the natural language process (NPL) is a technique used by many tools to extract data that existing in

natural language document. These tools, such as a part of speech tagging-filtering- lexical semantic tagging to link between relevant information, identify the relationships among phrases and sentence elements within text such as GATE. In fact, each of these tools has advantages and disadvantages. It required comparative analysis of existing tools for data extracting to recognize their capabilities. For the purpose to adopt the most appropriate tool we compared between them to provide a distinct of the Information extracting tools[1,8] as illustrated in table (1).

In this paper firstly we provide a brief idea about information extracting tools to justify the reason for using NLP technique. To speed up the building process of the ontology of Information Science (OIS) and extract concepts in the field, CREOLE Plagins in GATE has been deployed into this IE system.

2 Background

Basically, the annotating concepts of IS is based on GATE developer which is a tool of architecture for text Engineering. It is a free open source developed by a team at Sheffield University which started in the early of 1990s. The first version was in 1995, the second one was in 2002 and the new version is in 2010s. GATE is running at any platform and support JAVA 5 .0. Also, it developed and tested on Linux, windows, and Mac OS X. It has user interface to enable user editing and visualization and quick application development. Furthermore, it Support for manual annotation, sime-automatic and semantic annotation beside ontology management. Moreover, GATE uses CREOLE plug-ins as objects for language engineering. All of these are packaged as Java Archive and XML configuration data[4].

Table 1: Information extracting tools

Tools	Type	Degree of automation	Based on	Easy of use	Written language	Adv. &Dis.
SHOE	Knowledge annotation	Automatic		+	Java	Allows users to mark up pages in SHOE guided by ontologies or URL
Annota	Annotation schema W3C	Automatic	RDF mark up XML,XHTML,CSS &Xpointer	+	C & is available for windows, unix,&MAC	Doesn't support IE,it is liked to ontology server. – Makes annotation publicly available
Annozilla	Email annotation	Automatic	Mozilla	++		-
MnM	Ontology editor	Semi-automatic & automatic	HTML	+		Close to malita
Ontomat		Automatic	OWL	++		Use to create & maintain ontology – Use OtoBroker as server
COHSE	integration of text processing components	Automatic	DAML+OIL	+	RDF	Use ontology server to mark up pages in DAML+OIL& reuse as RDF
Melita	annotation interface	Semi-automatic	Extensible mark up language,Java,HTML	++		To retrival structure & semi structured annotations
KIM ontotext	Semantic annotation platform	Automatic	RDF	++		Semantic annotation, indexing, and retrieval of unstructured and semi-structured content.
GATE	Annotation tool	Semi-automatic & automatic	XML,HTML,XHTML,e mails	+++		Comprises an architecture, framework. Based inNLP group

GATE is a tool of Information Extraction system (IE). Which is a method to extract unseen texts as input and produce it in fixed format as XML, HTML, these data can be displayed for users or stored in database to analysis. Before talk about GATE in more details we should clarify what is the different between Information Retrieval and Information Extraction (IE) [3] . IE helps to extract information from huge amount of text for the purpose of fact analysis. Whereas, Information retrieval (IR) just pulling the document that have relevant information according to the key word research. In contrast IE identify the query in structure methods and provides knowledge at the deep level. While IR use normal queries engine which hard to gain the accurate answer, besides providing knowledge at typical level.

For instance, If you have an enquiry about when something is happened as which airports are currently closed due to the sever condition weather in UK? Or to ask about where and who did something as where did Gordon Brown last visit before he left? [7]

IR gives just the webpage containing the relevant information and you need to search on it using terms or concepts to meet your needs, to analyze this information. IE provides specific information about your enquiry, even if the information is not accurate but you can back the text. IE is used for many applications such as; Text Mining, Semantic Annotation, Question Answering, Opinion Mining, Decision Support, Rich information retrieval and exploration.

GATE has many features of both automatic and semi-automatic semantic annotation and also manual annotation which helps you to create your own annotations, for this purpose GATE developer is used as the tool to extract terms and concepts from a specific text effectively and efficiently. For this work we annotate text belong to members of Ontocop. Ontocop is a virtual community of practice of Information Science. That helps to speed up ontology process of building a conceptual model as a life cycle of ontology of IS.

Additionally, GATE is a Module that has a comprehensive set of plug-ins as: Alignment, ANNIE, Annotation_Merging, Copy_Annots_Between_Docs, Gazetteer_LKB, Gazetteer_Ontology_Based, Information_Retrieval, Keyphrase_Extraction_Algorithm, Language_Identification, Ontology_Tools, WordNet.

GATE based on ANNE which is a new IE system has core processing resources. ANNE relies on finite state algorithm and JAPE grammar and the application combines from Tokenisor, Sentence splitter, POS tagger, Gazatteer, Name entity tagger (JAPE transducer). Orthomatcher (co-references), NP and VP chanker. Among these modules we used: Tokenisor, Sentence splitter, Gazatteer, JAPE transducer [5].

3 Methods

The process followed the method is based on creating documents-corpora and Gazetteer of Information Science, and is based on JAPE rules to extract IS concepts as well. Gate provides facilities for loading corpora for annotation from a URL and uploading from a file. The process starts by uploading the corpus to the application framework with a JAPE grammar and Gazatteer to enable annotating the concepts from the corpus. Diagram (1) illustrates the process of corpus annotation.

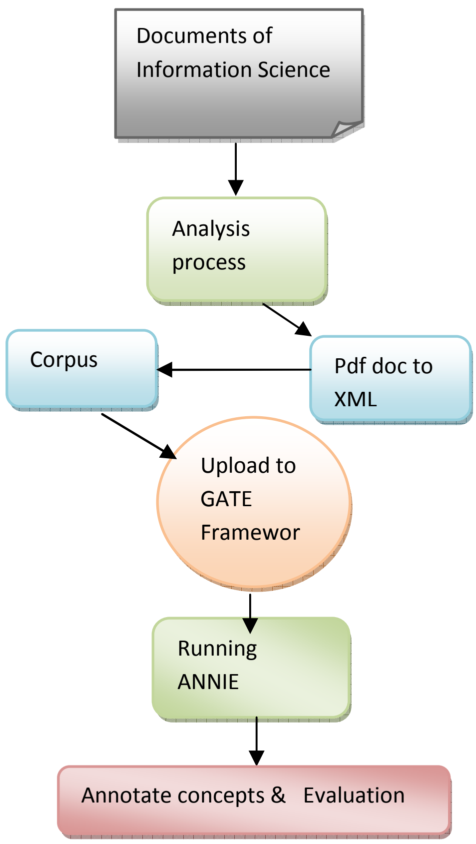


Figure (1) Annotation workflow

Corpus: Collecting the corpus contains 300 documents, all the documents are relevant to Information science field.

Gazetteer: The IS list included in Gazetteer which contains terms. These terms have value to be identified such as; MajorType and MinarType for each one, e.g.

Acquisition policy: major type= concept

Computer aided design: minor type= term

Data analysis: major type= concept

JAPE rule: Using JAPE rule extracts concepts to identify Tokens that contain the concepts in the correct order, and looking up to the concepts in the Gazatteer list.

JAPE (Java Annotation Patterns Engine) rules create a phase based on Java for creating specific grammar. Each JAPE rule consists of LHS which contains patterns to match. RHS details the annotations to be created [4].

We used JAPE grammar to support regular expression matching, as it is the way of annotation by GATE. Annotation can be made by using other CREOLE plug-ins such as Gazatteer which required to create own list of concepts to be annotated.

4 NLP technique of extract IS concepts:

We present an automatic extraction methods based on ANNE by creating JAPE grammar that extracts concepts form xml, HTML text, by creating Corpus with 300 documents in XML format.

Our JAPE rule to extract concepts shown in the following role. The first entity detected is Information service {Type=Token, start=867, end= 837, id= 4210, majorType=concept} labelled as information service.concept

Phase: one

Input: Lookup Token

Options: control = appelt

Rule: concept1

(

({Token.string == "information"})

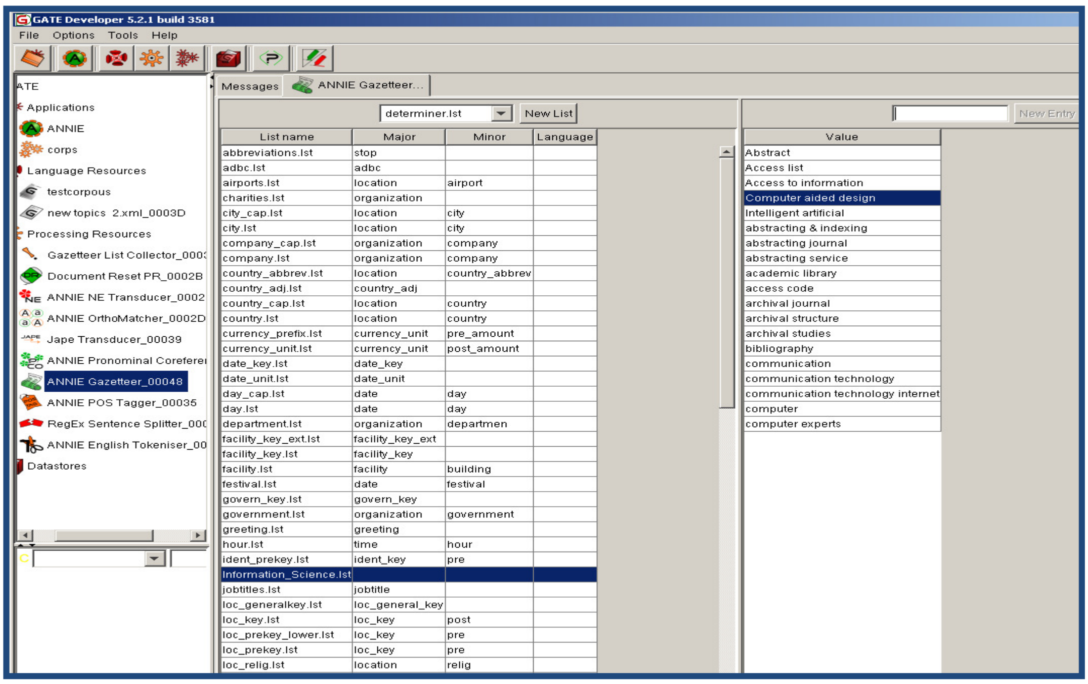


Figure (2) shows screenshot of IS Gazetteer

{Token.string == "service"}

({Lookup.minorType == region}): reginName

) : service

-->

: reginName.Location = {},

: Information service.concept = {}

({Lookup. major Type == "concept"})

) : information

-->

: Information. Concept = {Rule=concept2}

For more precise details we apply regular expression for matching strings of text, e.g

Phase: Concept

Input: Lookup Token

Options: control = appelt

Rule: Glossary

(

({Token.string == "catalog?e"})

): concept

-->

:{} .concept= {Rule= "Glossary"}

In this rule we specify a string of the text {Token.string == }string matching to specify the attributes of the annotation by using operators as "==",which provide the whole string matching. Some of these regular expressions in next example annotate concepts related to (abstract) metacharacter (dot, *, [ ], |),

1. *{Token.string == "abstract(ing)"}*

It may be abstract, abstracting, abstractor.

Also, if we want to annotate acquisition concept followed by another word as:

2. *{Token.string == "acquisition. number"}*

It could be annotate the

Acquisition. police

Phase: Two

Input: Lookup Token

Options: control = all

Rule: concept2

Priority: 20

(

({Token.string == "information"})

{Token.string == "service"}

Acquisition .service

3. *{Token.string == "archival * "}*

It will annotate archival library, archival journal, archival processing, archival software, and archival studies. All these rules are sorted in the INFCO. jape file .

5 Experiment & Evaluation

Extraction IS concepts by using JAPE grammar and Regular expression based on GATE developer for automated extracting information provides a significant output. The main idea of using JAPE and Regular Expression is to identify IS terminology as tokens, for example, Computing, Libraries and Information technology from a large text where terms are founded. The term identification relies on lookup from Gazatteer list of IS which could be matching, for instance, it could be book art, book card, book guidance or book catalogue. Also, look up at these concepts such as computer application, computer Science, computer experts, computer file, or computer image.

The corpus we used to extract information science concepts contains 300 documents which were obtained. Therefore, a total of document is analyzed. By running ANNIE application organized as document reset, Tokenisor, sentence Spliter Gazzater, POS tagger, JAPE transducer and Orthomatcher. In annotation set appeared in display pan and concepts are highlighted in the annotation default, as shown in figure (3)

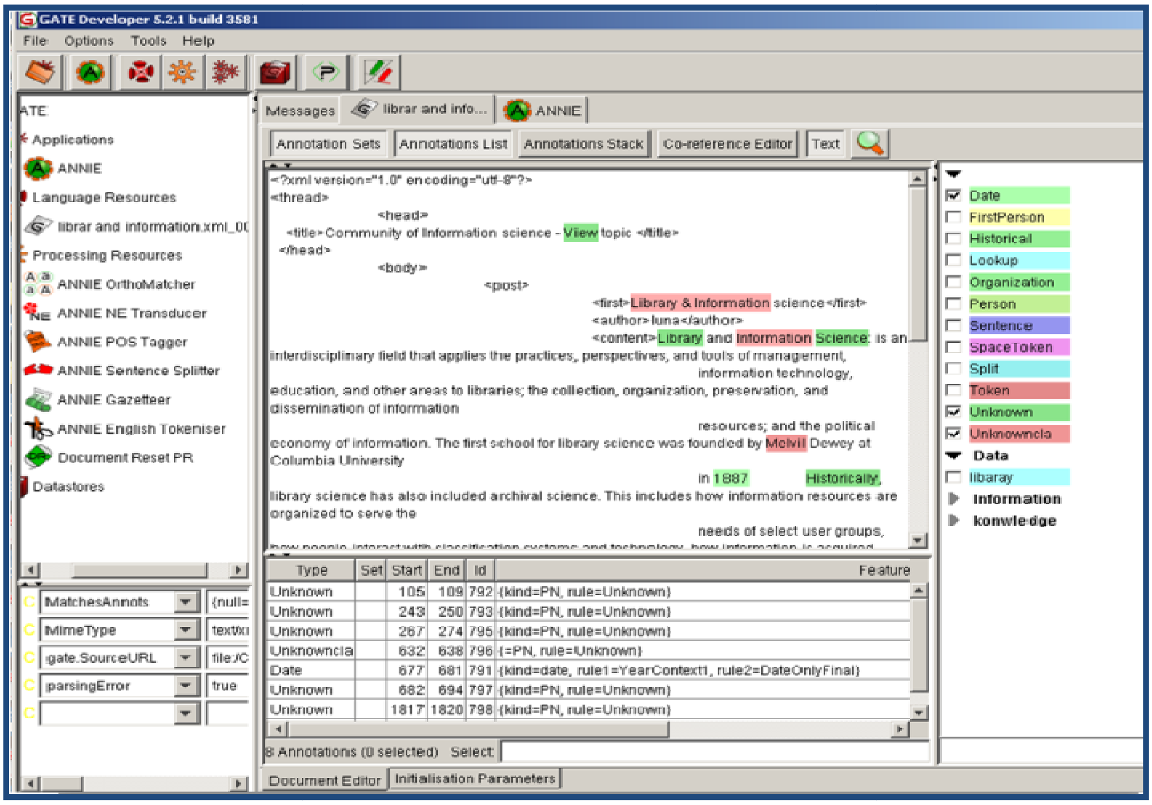


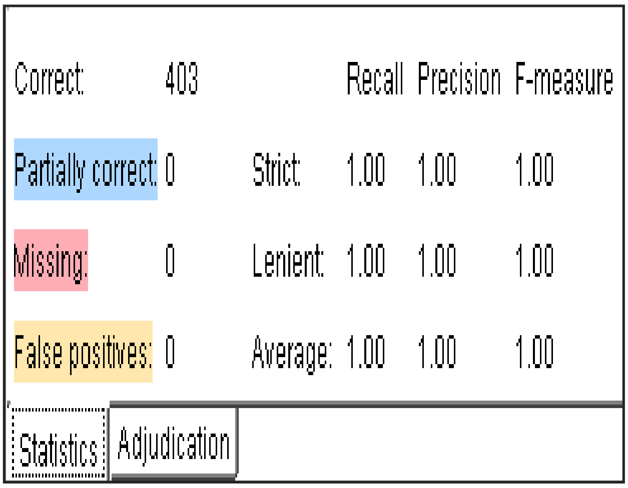
Figure (3): annotation concepts in Gate

The results show that our approach annotated concepts (see figure 4) and the annotation derives the knowledge started from (896) end in (905) while computer science concept annotated from (2008) and end at (2024), with its features {major Type=concept}. Each annotation starts from specific point and ends at different point based on how many token it has and listed each time.

Start	End	Key	Features	=?	Start	End	Response	Feature
557	566	knowledge	{majorType=concept, minorType=term}	=	557	566	knowledge	{majorType=concept, n
624	636	organization	{majorType=concept, minorType=term}	=	624	636	organization	{majorType=concept, n
751	760	knowledge	{majorType=concept, minorType=term}	=	751	760	knowledge	{majorType=concept, n
867	879	organization	{majorType=concept, minorType=term}	=	867	879	organization	{majorType=concept, n
896	905	knowledge	{majorType=concept, minorType=term}	=	896	905	knowledge	{majorType=concept, n
1023	1032	knowledge	{majorType=concept, minorType=term}	=	1023	1032	knowledge	{majorType=concept, n
1084	1096	organization	{majorType=concept, minorType=term}	=	1084	1096	organization	{majorType=concept, n
1151	1160	knowledge	{majorType=concept, minorType=term}	=	1151	1160	knowledge	{majorType=concept, n
1280	1289	knowledge	{majorType=concept, minorType=term}	=	1280	1289	knowledge	{majorType=concept, n
1323	1332	knowledge	{majorType=concept, minorType=term}	=	1323	1332	knowledge	{majorType=concept, n
1492	1501	knowledge	{majorType=concept, minorType=term}	=	1492	1501	knowledge	{majorType=concept, n
1876	1885	knowledge	{majorType=concept, minorType=term}	=	1876	1885	knowledge	{majorType=concept, n
1898	1910	organization	{majorType=concept, minorType=term}	=	1898	1910	organization	{majorType=concept, n

Figure 4: Result of the annotation IS domain

We conduct this experiment to achieve accuracy rates that equal to the manual output by IS experts for the annotating concepts. Statistics of the corpus show pattern matching of IS concepts based on the lookup IS list (402), correct concepts and accuracy were generally higher, whereas, partially correct (0) missing and false positives (0).



			Recall	Precision	F-measure
Correct:	403				
Partially correct:	0	Strict:	1.00	1.00	1.00
Missing:	0	Lenient:	1.00	1.00	1.00
False positives:	0	Average:	1.00	1.00	1.00

Figure 5: The result accuracy

However, we use GATE due to its benefits as open source and it contains multi-language NLP models which can be reused for developing other resources.

6 Conclusion

6.1 Achievement This paper described a method of using NPL technique to extract concepts for the purpose of speed up developing process of IS ontology. Furthermore, the development of IE system saved the efforts of domain experts by labelling most common concepts. In total we extract (664) concepts which is the classes of Information Science Ontology, and (650) subclasses, which is the main component of the ontology skeleton. Using IE technique can be applied to many different formats as XML, HTML documents even using URL or emails).

6.2 Future work Ontology is at the heart of the semantic web. It defined the concepts and their relations to make global interoperability possible. In future work we plan to enhance these concepts to develop IS ontology to creating the taxonomy of IS as domain. Next step is coding it by using Protégé as ontology editor. Additionally, such a generic model of the IS ontology will be evaluated.

Acknowledgment

The authors wish to thank Libyan government for its support. And for each one who provide feedback on this work.

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Opportunities of Interactive Learning Systems with Evolutions in Mobile Devices: A Case Study

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Abstract - *Rapid technical advances and proliferation of mobile devices in recent years have created chances for its application in learning activity. Its strength lies in distribution and access of multimedia content and context aware service provisioning. However, heterogeneity in hardware, software and user preference of mobile computing systems has brought forth new research barriers for cross-platform application. To demonstrate the feasibility and effectiveness, this paper presents the design and implementation of a Wireless Response System (WRS) for in-class interaction between teacher and learners based on the latest mobile phones as a case study. This research aims at utilizing the latest mobile devices and internet technologies such as XML, Ajax to improve the simplicity, friendliness, speed and multimedia capability of the traditional approaches with mobile device features. Tests and evaluations are carried out in class, and results show that the system is user friendly, fast and reliable. The performance evaluation of the system demonstrates that the application of the latest mobile technologies has greatly improved the performance of the system compared with traditional approaches, and the application of mobile devices in learning activities deserves more research effort.*

Keywords: mobile computing, mobile learning, wireless response system, cross-platform, context aware

1 Introduction

With the technological evolutions in recent years, many technical approaches have been applied in education area to improve the learning efficiency in class or for distant learning [1,2]. Traditional systems relying on the computer systems are fixed in specified position and it is space consuming. Thus, handheld devices with wireless modules are introduced to classroom participation, such as TuringPoint, ActiVote, and Qwizdom. However, most of these wireless modules are for visual distant and low bandwidth data communication [3,4,5]. The recent mobile devices (usually called smart phones) combine the strengths of the computer systems and above mentioned wireless devices, namely portability and strong processing and communication capability.

Mobile device today integrates many advanced technologies such as wireless communication, integrated circuits, human-computer interaction, multi-touch screen, and light-weight database, etc. These advances make it not only a telephony terminal, but also a potential integrated computing device. The bandwidth growth of Wi-Fi and 3G networks has facilitated multimedia content in computer aided learning to be applied in mobile systems [6]. The multi-touch screen gives the user much better interface to display information and convenient approach to feedback. The context awareness concept in mobile computing has built solid foundation for personalized service. All these novel technologies provide chances for ubiquitous mobile learning system, which is fast in speed, open in information type, unconstrained in geographical areas, and device independent with mobile device features.

An obvious trend of mobile computing is its convergence with traditional computing systems and other areas, and M-learning is an area resulted from the convergence. Lan et.al defined M-learning as a kind of learning model allowing learners to obtain learning materials anytime and anywhere using mobile communication, mobile devices and the Internet [7]. Chu presented a mobile learning behavior system detecting and learning guiding system to enhance learning motivation [8]. Tan and Liu developed an interactive learning environment for English study with mobile devices [9]. Pu et.al presented research works on device independent mobile learning in [10]. Lu et.al introduced a Student Response System (SRS) using mobile phones, and defined the term WRS for ubiquitous application scenarios [11, 12]. However, in these systems there is not enough consideration about user context and mobile device feature. Scott and Benlamri presented an intelligent learning system based on semantic web and ubiquitous computing [13]. Yin et.al presented a contextual mobile learning system, in which the system allows users to acquire contextualized learning resources depending on learning context [14]. Although a lot of progresses are made, research works in these areas have not efficiently utilized the advanced technologies and features of the mobile devices in both hardware and software platform.

The remaining of the paper is structured as follows: The purpose and goals of the research work is introduced under the aforementioned background in section 2. Then, the system design and implementation are presented in section 3 to reach the objectives, followed by the testing of the system and evaluation in section 4. Finally, conclusions are drawn and future work is suggested in section 5.

2 Purpose and Goals of the Investigation

The purpose of this research work is to verify and demonstrate the feasibility and effectiveness of the application of the latest mobile technologies to mobile learning system in enhancing the learning motivation, and improving the user experience and result management. To reach this purpose, this investigation aims at a device independent and context aware mobile learning system for the mainstream mobile phones with mobile device features, which can be used in learning and research to collect data and analysis the results data collected. The five dominant mobile operating systems (Nokia Symbian, Apple iOS, Google Android, Microsoft Windows Mobile, and RIM Blackberry) had taken 96.2% of the global smart phone market share in 2010 [15], and MSIE, Safari, Firefox, Chrome, Opera took 95.68% of global web browser market share in 2010 [16]. As the mobile browsers are derived from the abovementioned PC browsers, we aim at these five dominant browsers and five operating systems with default browsers to unfold the research work.

3 System Design and Prototype Implementation

To effectively enhance the interaction between the teacher and learners, the system should be able to distribute the teacher's questions and content, then collect, display and analysis the results data instantly. This part gives an introduction to the system architecture design, functionality and implementation approaches to reach this dynamic interaction between teacher and learners with the exploration of mobile native application sense.

3.1 System Architecture Design

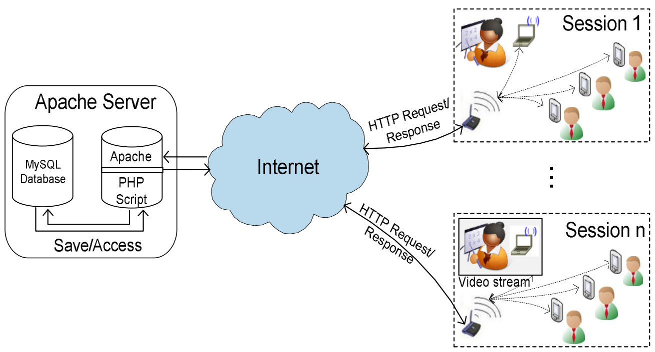


Figure 1. System Architecture of Mobile WRS

For the teacher side, the system may not only be able to control the system, it may also display the results to learners to enhance learners' impression. While, for learners, they just need to be able to get the information and make corresponding decisions and then response. Meanwhile, for both sides, it is important to consider the visual effect of interfaces to fit people's health and aesthetic requirements. Bear the above requirements in mind, the Apache+PHP+MySQL is selected as the system service framework. Because of the flexible interface and internet data communication functions of Adobe Flex Builder, it is a good choice for teacher's content distribution and result display. Figure 1 gives the system architecture of the proposed system.

This system can be divided into three layers: presentation layer, application layer, and data layer. The presentation layer forms the request operation and presents the requested information to the interface for convenient interaction with the users. The data requested may be in XML format, and data is presented to the screen in HTML/XHTML with mobile oriented style. The application layer coordinates the upper and the lower layer which realizes the main functions of the dynamic interaction. The Data layer is responsible for MySQL database runtime and history data operation.

3.2 Functionality of the System

The main function of the system is content (questions and learning materials) distribution and result collection during learning sessions. The functionality framework can be modeled by the utilization sequence of the system: before session, during session, and after session. Before the session, the teacher needs to prepare the subject domain setting, prepare questions and learning materials. During the session, the following functions are needed: system authentication, user preference setting, question state control, results data collecting, correct answer notification, results display, question results export, and results analysis. When the session is finished, the system should be able to search history results, display history results in charts, analysis history results, and export results data to files. In current stage, the above functions are incorporated into the prototype system to verify the proposed approach.

3.3 System Implementation Approaches

According to user requirements and application situation, the critical issues of this user oriented system are simplicity, friendliness, and functionality. To reach these objectives and implement it with mobile device features, we need to investigate the development approaches and principal concerns of mobile application development. We can find the main development approaches in Table 1 [17].

Table 1. Classification of Mobile Development Approaches

Approaches / Compare Items	SMS/MMS	Web-based Application	Mobile Native Application
Cross-platform	Poor	Excellent	Poor
Multimedia Capability	Good	Good	Good
User Experience	Good	Good	Excellent
Data Management	Poor	Limited	Excellent
Complexity	Medium	Medium	Complex
Device Features	Limited	Limited	Excellent

As the system is used in class without preparing the mobile phones for learners, the cross-platform performance is a critical issue. Although mobile native approach can bring the device feathers into play, which is also strong in data management, it is difficult to develop application for each model. Thus, the web-based approach which stores the data onto the backend server is a prospective method to reach the objectives.

3.3.1 User Friendly Interface with Mobile Native Sense

As mentioned above, although web-based application is weak in device feature implementation compared with mobile native application, it is also possible to make the application fit the device well with mobile native sense. For example, the layout of the interface on mobile is inconvenient to run the desktop/laptop web applications. Thus, different styles for different devices should be designed to avoid incorrect presentation and make the operation correct and convenient. For Apple iPhone/iPod, we used jQTouch to make use of its graphics animation and iPhone sense interface, thus the page slide, swipe, and curl can be available. For other models, we also made corresponding design to fit the screen size and exploit the device features using Ajax technique.

3.3.2 Automation in System Control

For learner side, the system should be as easy and simple as possible to be used as a learning tool. Thus the system should be able to request the state of the system and distribute the corresponding content to learners. Meanwhile, it should also be able to keep the state and interface before the state of the system changes. That is to say, the clients request the state frequently and keep the current content during operation. To simplify the design procedure, the concept of state machine is introduced in the system design. For a system with n question states, there are (n+2) states in the system. The states should be able to jump from one to another according to the changes made in the backend by the teacher. Thus, the quantity of the states is:

$$N = 2\times C_n^2 = n\times(n-1) \qquad (1)$$

The states in the state machine are: Home page: State before login in; Waiting: Idle state; Question 1 – Question n: Question state.

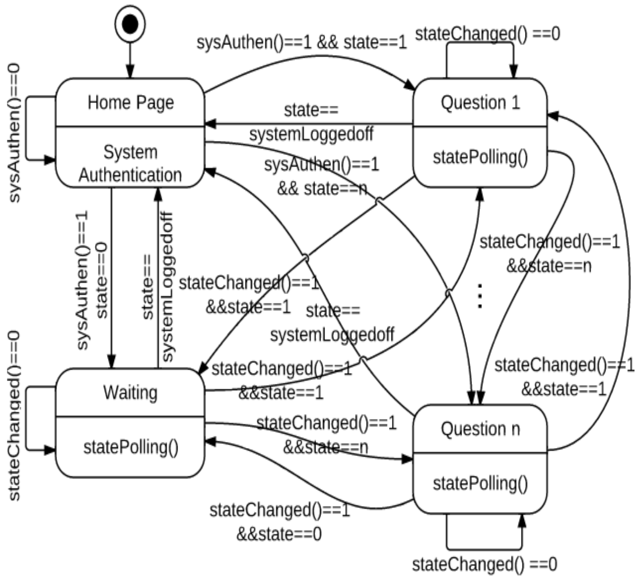


Figure 2. State Diagram of Learner Side Control Logic

For each state transfer, the next state relies on not only the current state, but also the input of the signals during the current state. The whole control logic of the system is given in Figure 2.

3.3.3 Context Aware Service for Cross-platform Application

Heterogeneity of hardware and software of mobile devices makes the development of mobile application for multiple platforms rather difficult. Although web-based application is strong in interoperability, the performance may not identical on different browsers and platforms. To make the system work well on most mobile phones, we need to make it work on the mainstream web browsers and provide services according to device context information. For example:

$$C_P = [p1, p2, p3, ..., pm]^T,$$

$$C_B = [b1, b2, b3, ..., bn],$$

$$Sc = C_P * C_B = \begin{bmatrix} S11 & S12 & ... & S1j & ... & S1n \\ S11 & S22 & ... & S2j & ... & S2n \\ ... & ... & ... & ... & ... & ... \\ Si1 & ... & ... & Sij & ... & Sin \\ ... & ... & ... & ... & ... & ... \\ Sm1 & Sm2 & ... & Smj & ... & Smn \end{bmatrix} \qquad (2)$$

C_P denotes platforms such as Symbian, iOS, Android, Windows Mobile, and Blackberry, etc. And C_B is for browsers such as MISE, Safari, Firefox, Chrome, and Opera, etc. Based on the context data, different services in equation (2) are provided. In this way, as long as it works for each element Sij, the system can cover most combinations of the browsers and platforms. Please see sample code for browser aware service below.

```
<script type="text/javascript">
  if(navigator.userAgent.toLowerCase().indexOf('safari') > -1)
      location.replace("index1.htm");
  else if(navigator.userAgent.toLowerCase().indexOf('chrome') > -1)
      location.replace("index2.htm");
  else if(navigator.userAgent.toLowerCase().indexOf('msie') > -1)
      location.replace("index3.htm");
  else if(navigator.userAgent.toLowerCase().indexOf('firefox') > -1)
      location.replace("index4.htm");
  else if (navigator.userAgent.toLowerCase().indexOf('opera') > -1)
      location.replace("index5.htm");
  else
      location.replace("indexn.htm");
</script>
```

3.3.4 Make it Faster – Optimizing the Performance of the System by Reducing Data Exchange

As the structure of the system is distributed service, the burden of the server is heavy if there are a number of sessions with many learners using the system at the same time. Although the computing burden at the mobile client side is light, the bandwidth of the wireless networks is still a constraint for large group use. From this point of view, the performance of the system can be improved by reducing the amount of data exchanging for bandwidth constrained devices. While for wideband connections, we need to transmit the data to the client at the beginning to reduce runtime data exchange. There are two approaches to distribute the content to client side by Ajax technique:

1) Clients request content for each question when state changes are determined
2) Request the whole content once and display the specified part according to current state

The main functions of the two approaches are shown in Figure 3 (1) and (2).

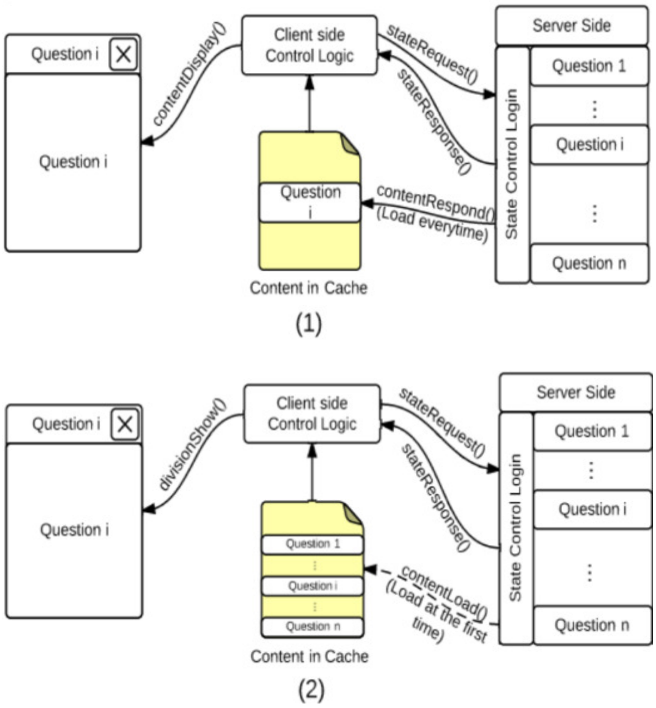


Figure 3. Diagram of Content Distribution

The first approach reduces the data exchange at the beginning, and the work is distributed to runtime loading. It is used for mobile devices with limited bandwidth. While the second approach loads the data at the first time, thus reduces the data exchange during the session. It can be used for wideband connected devices to reduce bandwidth consumption during the learning session, especially when there are a large number of users. Thus, the performance of the system can be optimized by combining the two approaches.

3.3.5 Use XML for Interoperable Information Exchange

As the results data is stored in the database, and it is saved and retrieved frequently during and after sessions. While at user end, it is graphics interface. Thus, it is a critical issue to map the information from database to user end efficiently. XML is a flexible and interoperable tool to transmit data via the internet. In this system, the PHP script reads data from database and generates XML files. Then, Flex Builder application and mobile device parses the XML documents and map the data to the interfaces. The following sample code is for PHP script to read database data and generate XML documents.

PHP Script transfers MySQL raw data to XML file:

```
...
$dom = new DOMDocument("1.0","UTF-8");
header("Content-Type: text/xml");
$root = $dom->createElement("Test");
$dom->appendChild($root);
$f = $dom->createElement("Results");
$root->appendChild($f);

$dbcon = new DBService();
$r = $dbcon->rstTable($_POST["code"],$_POST["questionid"]);

while($data = mysql_fetch_array($r)){
  $item = $dom->createElement("student");
  $f->appendChild($item);

  $studentid = $dom->createElement("studentid");
  $studentid->appendChild($dom->
createTextNode($data['stid']));
  $item->appendChild($studentid);

  $question = $dom->createElement("questiontype");
  $question->appendChild($dom-
>createTextNode($data['qntp']));
  $item->appendChild($question);

  $answer = $dom->createElement("answer");
  $answer->appendChild($dom->createTextNode($data['answ']));
  $item->appendChild($answer);

  $response = $dom->createElement("response");
  $response->appendChild($dom-
>createTextNode($data['rspn']));
  $item->appendChild($response);

  $mark = $dom->createElement("mark");
  $mark->appendChild($dom->createTextNode($data['mark']));
  $item->appendChild($mark);
}
echo $dom->saveXML();
```

XML file generated:

```
...
<Results>
    <student>
        <studentid>S001</studentid>
        <questiontype>atod</questiontype>
        <answer>cd</answer>
        <response>c</response>
        <mark>50</mark>
    </student>
    <student>
        <studentid>S002</studentid>
        ...
    </student>
    ...
</Results>
```

3.3.6 Multimedia Content Distribution

An obvious strength of the mobile approach is its multimedia capability, which provides open and flexible platform to distribute multimedia (i.e. image, audio, and video) to learners. These media types may be used for scenarios of some subjects, or distant learning. In this system we introduced image, audio, and video to meet the requirements of different cases. For video streaming, we use YouTube API, as the FLV files are small in size which requires less bandwidth. However, the preparation of the video is complex. This area needs further investigation to make it easy to use and manage, and break the constraints of wireless infrastructures with network technology such as multicast for heterogeneous mobile devices [18]. Image and video distribution sample in this system is given in Figure 4.

Figure 4. Images and Video Clips Distribution

4 Testing and Evaluation

4.1 Systematic Testing of the System

The system is tested in-house before releasing, and its beta version is also tested in class in three European universities in current stage. Based on the feedback of the users, we will make further improvements. For in house testing, in order to find existing problems and guarantee that it works on heterogeneous platforms. The system is tested on the five major browsers on different PC operating systems such as Mac OS, Windows XP, Windows 7. For mobile devices, the system is tested on the five mainstream mobile operating systems: Symbian, iOS, Android, Windows Mobile, and BlackBerry. Besides the cross-platform performance, the testing items are divided into four parts: interface, functionality, speed, and robustness. The testing matrix with the testing conditions, testing items and testing results are given in Table 2.

In Table 2, there is probably time delay when the learners get questions and response error may occur, they are both because of the state request time delay via HTTP. If the interval is set to be very short, there may only hundreds of milliseconds delay which is acceptable for non real-time application. However, it is difficult to fix this problem for hard real-time application because of the intrinsic attributes of the communication mechanism.

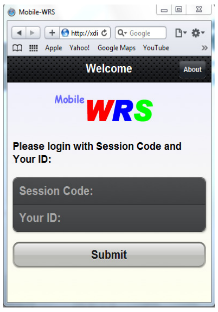
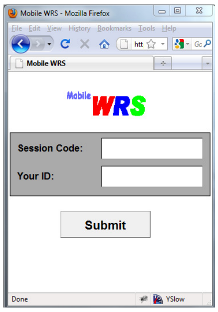
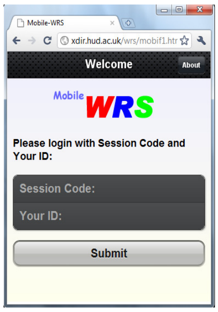
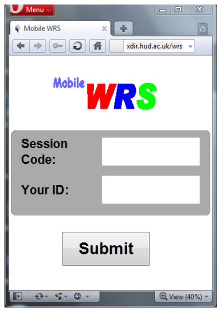
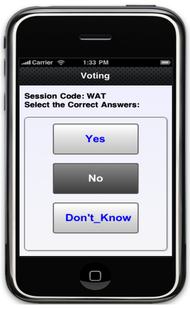
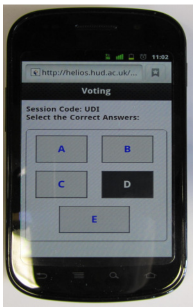
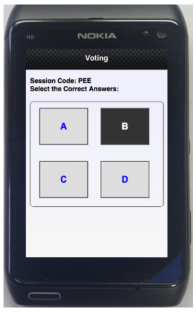
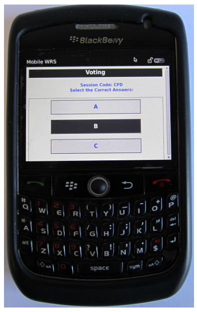
Table 2. WRS System Testing Form

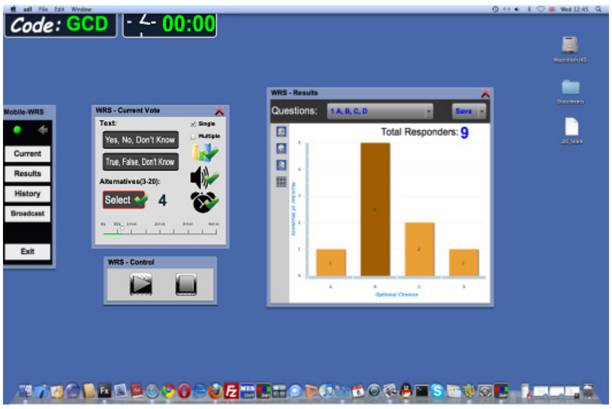
Test Condition \ Test Items		Interface			Functionality								Speed				Robustness			
		Layout	Mobile CSS	Element position	Authentication	Question trigger	Receive/response	Mark selected	Data collection	Answer setting	Result distribution	Result print	Login speed	Get question speed	Respond speed	Result distribution	Redirection errors	Response errors	Result errors	Display/save errors
Platform(Win XP, Win7, Mac OS with Wi-Fi connection) ǀ Browser																				
1	Internet Explorer (version 9.0)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
2	Safari (version 5.0.4)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
3	Firefox (version 3.6.3)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
4	Chrome(version 10.0)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
5	Opera (version 11.01)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
Default browser ǀ Dominant Mobile Operating System (3G & Wi-Fi connection)																				
1	Symbian^3 (Nokia N8)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
2	iOS 4.0/3.2 (iPod 2G, iPad)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
3	Android 2.3 (Samsung Next S)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
4	Windows Mobile 6.5 (HTC HD2)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N
5	Blackberry 5.0 (Blackberry curve 8900)	√	√	√	√	√	√	√	√	√	√	√	F	D	F	F	N	P	N	N

(Some abbreviations in Table 2: F for Fast, D for time delay, N for no error, P for probably)

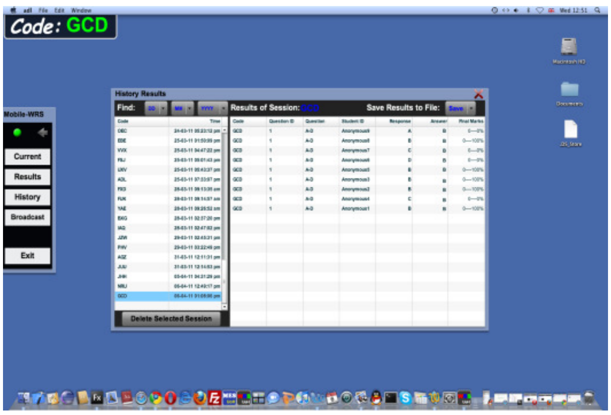
4.2 Results and Evaluation

Table 3. Screenshots of the Learner Side on Mainstream Web Browsers and Mobile Devices

MSIE	Safari	Firefox	Chrome	Opera
				
iPhone 3GS (iOS 4.1)	Samsung Next S (Android 2.3)	Nokia N8 (Symbian^3)	HTC HD2 (WinMobile 6.5)	Blackberry 8900 (Blackberry 5.0)
				



(1)



(2)

Figure 5. Teacher Side Control, Distribution, and Data Management Panels

The prototype system is implemented concerning the above mentioned points. For mobile device, the interface is given in Table 3, it is easy to find that the presentation is appropriate for the mainstream browsers and mobile phones. The information in the interface is clear, and it is easy to operate. For teacher side, the function is easy to use, data presentation is clear, and data management is successful.

To identify the strength and weakness of the proposed approach and evaluate the performance of the system, it is compared with the traditional approaches in Table 4.

Table 4.Comparison of The Proposed System with Traditional Ones

Classification Compare Items	**Infrared- based WRS**	**Existing mobile WRS**	**Proposed mobile WRS**
Application Scenario	Prepare devices	No preparation	No preparation
Operation	Press button	Press /touch	Press/touch
Interaction	Question/answer	Question /answer Distribution /feedback	Question /answer Distribution /feedback
Distance	Visual distance	Unconstrained	Unconstrained
Network Dependence	Dependent on self-constructed network	Depend on the internet	Depend on the internet
Learning material	Text only	Text, image, audio, and video	Text, image, audio, and video
User Interface	Poor	Poor	Good
Mobile Feature	/	Poor	Good

5 Conclusions & Future Work

5.1 Conclusions

The prototype system is successfully implemented with the functionality and mobile specialized considerations mentioned above. Based on the tests, analysis, and feedbacks from our partners and users, we can draw the following conclusions:

1) The mobile devices in the system expanded the information content form text to multimedia. Therefore, it improves the openness of the platforms for learning materials distribution as well as quantity and complexity of information interaction in learning.

2) The internet access with Wi-Fi or 3G in the system breaks the geographical constrains of learning activity, thus the interaction can be in-class, on campus, and international. However, the dependence on internet connection decreases the reliability and simplicity of communication mechanism.

3) The web-based application is an effective approach to simplify the development and achieve cross-platform interoperability, and it is also possible to keep the device features used to be explored by mobile native application only.

4) Mobile devices become competent platforms for information interaction for the learning activities, and it is a prospective area which will attract more research interest.

5.2 Future Work

It can be said that mobile WRS is a successful case study to apply mobile devices into learning activity. Some technical and utilization assumptions are verified in the prototype design, implementation, and test. However, it still needs to pay attention to the following issues for further investigation:

1) Web-based application with mobile native sense is good choice to balance the system complexity and user experience, while it needs further research to meet the heterogeneity of mobile operating systems.

2) System performance for multimedia content communication under Wi-Fi and 3G networks needs systematic evaluation and improvement for occasions with large number of users.

3) Results data management and analysis after sessions is a potential area, which can be combined with related disciplines to evaluate the learning efficiency or be used as a research tool mining the underlying theories from the results data.

4) Subject domain oriented design for different disciplines to meet users' requirements of specific areas.

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Innovative Evaluation System – IESM

An Architecture for the Database Management System for Mobile Application

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Abstract - *As the mobile applications are constantly facing a rapid development in the recent years especially in the academic environment such as student response system [1-8] used in universities and other educational institutions; there has not been reported an effective and scalable Database Management System to support fast and reliable data storage and retrieval. This paper presents Database Management Architecture for an Innovative Evaluation System based on Mobile Learning Applications. The need for a relatively stable, independent and extensible data model for faster data storage and retrieval is analyzed and investigated. It concludes by emphasizing further investigation for high throughput so as to support multimedia data such as video clips, images and documents.*

Keywords: Mobile Computing, Mobile Exam System, Database Management System, XML Schema

1 Introduction

Ever since Mobile Technologies emerged in 1980s, enormous amount of development has been made and improvements are constantly evolving everyday that almost any given Internet application can now be utilized to its maximum usability on a handheld device such as PDA, iPhone, iPod, Android etc [16]. Further developments are being made to enhance higher data rate, effective use of smaller screen size and the ability to handle multimedia data formats such as images, documents and video clips as if they are used on a personal computer or a work station [17]. Currently the academic bodies are making good use of mobile applications for e.g. student response system [1-8] particularly mobile based response system which can be evolved to support a large number of concurrent users to access resources [18].

This objective of this investigation is to propose a suitable architecture for database management to support ‘Innovative Evaluation System in mobile learning.

2 Background

2.1 Wireless Response System

The XDIR research group proposed a new prototype to improve operational efficiency of wireless response system based on pure mobile devices [9]. Objective of this research was to identity a simple and relatively faster solution for using the mobile based response system simultaneously for e.g. in a class room.

It has been found in this system:

- Mobile based application could be simple and faster.
- Users does not have to always rely on internet browsers, therefore cost effective by using alternate methods such as Wi-Fi connection.
- Importantly, mobile devices can potentially reduce a large amount of hardware, memory, storage and maintenance in comparison with workstation.

2.2 Database Management System for Mobile Applications

Because of the fact the data collected by the response system [9] are short codes of limited character sets; the need for a data model seems inevitable and hence structured data for storage in data management systems such as relational databases in considered as the starting stage of the database architecture and hence the idea of using RDBMS arises [12].

2.2.1 RDBMS in Mobile Applications

Ideally the data model has to evolve in such a way that whenever the application interfaces are changed, the database mechanism acts in a relatively stable manner. This will strongly pave way to scale the data model to accommodate multiple application support.

Table: 1 RDBMS Advantages and Disadvantages

RDBMS	Advantages	Disadvantages
1	Relational theory	Increase resources – increase performance linearly
2	Transactions: atomicity, consistency, isolation levels	Throughput
3	Multiple indexes, auto increments/sequences and triggers	Vertical scalability (scaling up)
4		Horizontal scalability (scaling out)
5		Performance(sub-queries/correlation, joins, aggregates)

Many successful and effective RDBMS are available in the market such as MS SQL Server, Oracle, MySQL, MS Access, SyBase etc. For the initial investigation, the database architecture will implement MySQL and analyze the results.

There has been constant increase in the number of companies using MySQL for instance Youtube video, Adobe, Virgin Media, McGraw-Hill Education, iStockphoto, social networks such as Wikipedia have currently benefited and effectively using MySQL [13].

Relational database servers generally provide:

- Data Management
- Data backup and recovery
- Data Integrity
- Data Security
- Transaction processing

2.2.2 ORDBMS

Object-Relational Database Management System is very similar to relational database management but with object-oriented database model like classes, objects, inheritance, polymorphism and other object-oriented concepts that are directly supported in database schemas and query language. When an application uses this type of database, it will generally consider the data that is stored as objects. Similarly, for data retrieval, it must be reconstructed from simple data to complex objects.

The aim of ORDBMS is bridge the gap between conceptual data modeling methods like ER diagrams with Object-Relational Mapping (ORM).

Table: 2 ORDBMS Benefits and Performance Constraints

ORDBMS	Benefits	Performance Constraints
1	The main benefit of this type of database is that the software to convert the object data between RDBMS format and object database format is already provided and therefore not necessary to write a code for conversion between two formats	ORDBMS converts the data between object oriented format and RDBMS format and hence the speed and performance of the database is degraded substantially
2	Database access is easy and simpler when accessing from an object oriented computer language	Additional conversion work for the database

2.2.3 ODBMS

Object-Oriented databases are also referred to as *Object Database Management System (ODBMS).* These type of databases store objects instead of data such as numbers, strings and other integers.

Table: 3 ODBMS Advantages and Disadvantages over RDMBS

ODBMS	Advantages over RDBMS	Disadvantages over RDBMS
1	Easier Navigation	Lower efficiency when the data and its relationships are simple

2	Better concurrency control	Relational tables are simpler
3	Less code required when applications are object-oriented	More user tools exist for RDBMS
4	Data model is based on the real world	Standards for RDBMS are more stable
5	Works well for distributed architectures	Late binding may slow access speed
6	Reduced paging	Support for RDBMS is more certain and changes are less likely to be required

2.2.4 XDBMS

An XML Database Management System is also called as XDBMS, innovative database technology software that allows the data to be stored in an XML format. The data thus stored can be queried, exported and serialized into any desired format.

There are two types of XML database that exist. They are:

- XML – Enabled
- Native XML (NXD)

2.2.4.1 Advantages of XML Databases

- Efficient - Eliminates redundancy; generates consistent and cost-effective workflow.
- Sturdy - Fast, stable, traceable and comes with a sophisticated authorization system.
- Simple - Automatically takes care of complex tasks and relatively simple.
- Connected - Effortlessly assimilates existing data collections.
- Object-oriented and relational.
- Embedded security and high speed access to even the complex data models.
- Reduced maintenance and flexible interfaces.
- No more redundancy.

Table: 4 XDBMS Vs RDBMS

	XDBMS	RDBMS
1	An XML based database management system does not worry about relations between data. It just stores the data in the database.	A relational database management system stores the data in such a way that it explicitly shows the relation between the data.

3 Problems Identified

However, the system has also identified that there are a few major limitations as follows:

- Traditional data management system could pose bottle neck because the response system is based on ‘many to many’ relationship at any given time and will scale to a large number of simultaneous responses.
- The system has to serve a large number of concurrent users accessing or responding to the system and hence the need for a suitable database management system arises.
- The response collected was in an unorganized format and concern for storing this data and retrieve readily is inevitable.

4 Aims and Objectives

Based on previous investigation, the student response system can be scaled to an ‘Innovative Evaluation System’ and the paper proposes:

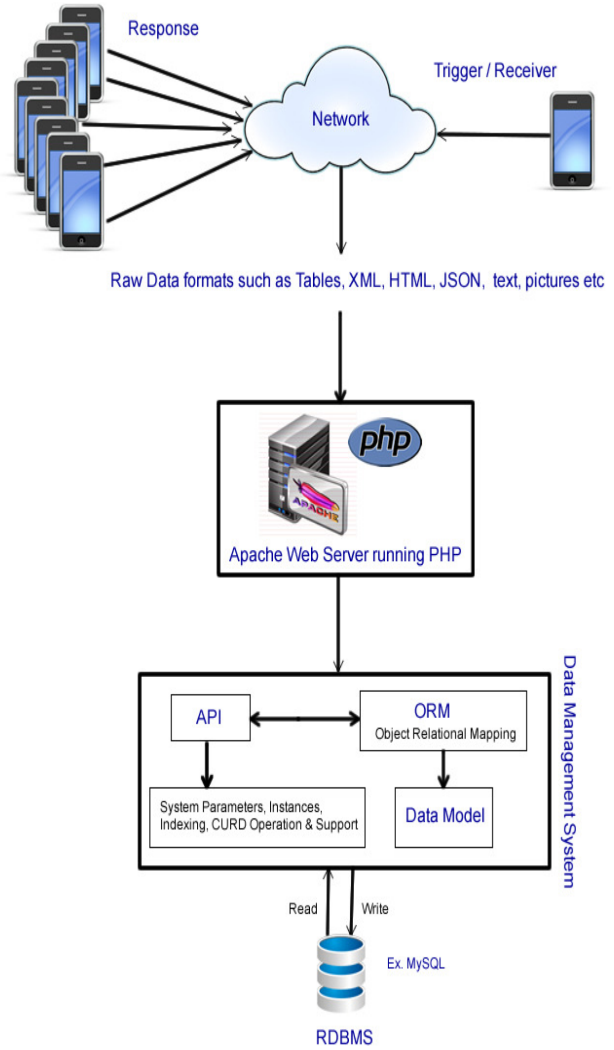
- To design a suitable architecture for database management system to support ‘Innovative Evaluation System’.
- The architecture to support an independent data model that is relatively stable and scalable.
- Effectively convert raw data or responses that are in the form of tables into well defined XML schema

so that it can be stored as data objects into database and to be able to retrieve readily when required.

5 Proposed Database Architecture

The data or response collected in raw format such as tables, texts are parsed and converted into an XML schema. This well documented XML schema is then set as data feed through an API access over a server scripting language like PHP run on an apache server as shown in figure 1.

Figure 1. Database Architecture



XML schema is then converted into data objects by the use of Object Relational Mapping (ORM) open source tools such as Doctrine [10] or Propel [11] based on Objected Oriented Programming concepts and data model. This mechanism makes it possible to address the access and manipulate objects without having to consider how those objects relate to their data sources. Thereby lets the system maintain a consistent view of objects over time, even as the sources that deliver them, the sinks that receive them and the applications that access them change. The API then adds system parameters, deploys indexing techniques and automatically generates the code to create, insert, read, update, and delete (CIRUD) records from the database systems.

For the initial system architecture, RDBMS is used as the initial attempt but not limited to it. Keeping in mind they are powerful because they require few assumptions about how data is related or how it will be extracted from the database. As a result, the same database can be viewed in many different ways. Due to the fact that RDBMS is used, the data models describe structured data for storage at this stage. The API layer is designed to perform all the data storage and retrieval mechanisms associated with the system.

5.1 Data Indexing Algorithms

Even though there exist many ways of improving the performance of the database system, the most effective and efficient method should effectively implement the data indexing mechanism. The most used indexing mechanism in nowadays Database Request Module System (DBRMS) is *B+ Tree and Bitmap.*

5.1.1 Bitmap Indexing

Bitmap index is a unique structure of database indexing technique that uses bitmaps. This type of indexing has a significant advantage of space and performance over other data structures. Bitmap indexing generally uses bit arrays and functions by performing bitwise logical operations on these bitmaps.

5.1.1.1 History of Bitmaps

The concept of bitmap index was first introduced by Professor Israel Spiegler and Rafi Maayan in their research "Storage and Retrieval Considerations of Binary Data Bases", published in 1985 [14].

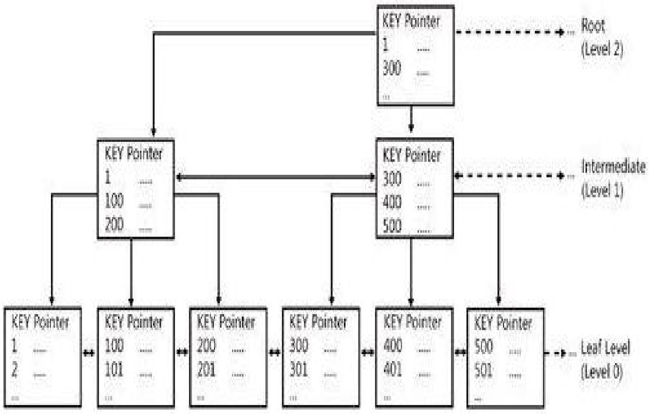
5.1.2 B+ Tree Indexing

B+ Tree is also called as a *Balanced Tree.* This tree represents sorted data that allows basic operations like insertion and deletion of records, each of those identified by their unique *key*. It is dynamic and multi-level index that has the block or node with maximum and minimum bounds on the number of keys in each segment. Its main objective is to store data for efficient retrieval of records in block-oriented storage structure, mainly file systems.

5.1.2.1 History of B Tree

The B tree was first described in the paper *Organization and Maintenance of Large Ordered Indices Acta Informatica* 1: 173–189 (1972) by *Rudolf Bayer* and *Edward M. McCreight* [15].

Figure 2: Architecture of B+ Tree [19].



6 Conclusions and Future Work

6.1 Achievements

- A scalable, stable and rationale database management system is required to build an 'Innovative Evaluation System' for mobile based application.
- Data model to effectively handle and manipulate objects or instances of data.
- Simple to use and fast data storage and retrieval for a mobile based application requiring a large number of concurrent users.

6.2 Future Work

The initial investigation of the database architecture has to evolve into the following:

- Database model establishment.
- Authentication techniques.
- Integration with mobile based application and possibly multiple interfaces.

Although the initial research is in the area of structured data formats collected from the student response system, further investigation is highly desirable to build a scalable data model that will enable multimedia support for unstructured data such as texts, images, documents, video clips on handheld devices without affecting the throughput or performance of the system.

7 Acknowledgement

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SESSION

APPLICATIONS

Chair(s)

TBA

Time zone considerations for web-hosting plans

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Abstract — *It is well accepted among the web community that local servers out-perform distant ones. This common knowledge dictates buying a local web-hosting plan for one's business. In this paper we show that, all other factors being equal, time zone influence on web traffic peak times might dictate migrating web-hosting plans. We performed simulation which shows that for entertainment, auctions and similar evening traffic sites - sites who's main clientele access is between 4:00PM and 2:00AM, distant servers actually outperform local ones and it's beneficial to migrate to remote hosts. The simulation suggests that moving web-hosted sites ±3 time zones away from local clientele decreases the total response times experienced by clients.*

Keywords: Internet, Internet topology

1. Introduction and motivation

Common web developer's wisdom when setting up a website is to physically place the web server on a geographically local location to the population it services[1]. This intuition is based on the assumption that the latency encountered during the *transition* of data from the server to the client should be minimized. Payload from a geographically local server would traverse fewer router nodes to reach client machines than a geographically remote server.

This common belief is obviously correct if one is setting up a dedicated server that only services the specific site in question. Thus, processing time for the client's request isn't influenced by the server's location; rather, that value can be taken as a constant. And with the minimization of transition time accomplished by a locally stationed server, the total request response cycle times will decrease as well.

In a shared web-hosting environment, however, the question arises as to whether this practice is really wise. Here, there are a number of other factors that influence response times of servers. Allocation of processing time (CPU cycles) to other domains, cap on bandwidth and memory constraints will also influence response times – to name just a few. In this scenario, it might be wiser to buy a hosting plan on a server that mostly services its local constituents, but is geographically placed so as to utilize the "off peak" hours to minimize the total request-response cycle. This must also add the increase in transition time incurred, as payload of a non-local server would experience longer transition times than local counterparts.

*Listed in alphabetical order.

[1] This behavior is what leads Google's search engines to take into account in their point system the server location when generating answers to a localized query (such as one's submitted by www.google.co.fr). See [10]

This paper is organized as follows: In section 2 we formulate the problem domain so as to set the ground work for further research. Section 3 provides a mathematical model to capture the relationship between variables influencing the question at hand. We identify known (easily accessible) and un-known (requiring extensive research and testing) information, so as to allow for an online tool to be built upon the information collected. In addition results of a MATLAB simulation[2] are presented. Sections 5 and 6 provide future research venues and concluding remarks, respectively.

2. Problem formulation and related work

There are many factors that may influence the time it takes from the instant a client posts a request to a server until the response is displayed by the browser. Some of these factors include (in a web hosting server):

- CPU speed – the speed at which the server CPU can execute commands
- Memory allocation – how much memory (RAM) is allocated to the web application
- CPU time sharing segment – size of segment allocated for processing request for application data
- Transition / Transfer times – time for a request to travel from point to point (client to server and vice-versa)
- Router hops – number of routers on the route from end-point to end-point
- Hit rate of web host – number of requests a web host receives for a given period of time
- Number of web clients – number of sites hosted on a particular server
- Server time zone – where is the server located
- Client time zone – where is the request initiated from
- Time of request – when was the request initiated
- Distribution of hit rate w.r.t. time – when is the server served with the majority of hits
- Data size – size of page to be served
- CPU requirements – amount of CPU cycles needed to process request
- Others…

Some of these factors influence each other while others stand by themselves. For example transition and transfer times are influenced by the number of router hops en-route from client to server, which, in turn, is influenced by the server and client time zones. The CPU time sharing segment length is

[2] Available upon request

influenced by the number of hosts sharing the server, the distribution of request arrival rate and, the amount of CPU cycles required for the processing of each request.

As it turns out, the total request/response cycle time is influence by all these factors in one way or the other. In relation to our work, we are not interested in factors that are constant (such as the CPU cycles needed per request and the CPU speed). Obviously a faster server would give better results than a slower one, all other factors being equal. We wish to model only factors that are influenced by the location of the webhost (vis-à-vis time zone differences between client and server), and the fact that it's not a dedicated host.

There has been much research taking geo-location into account. Many algorithms were produced to accurately locate the geo-location of a web server given its IP address. See [1, 2, 3, 4, 5, 6], to name just a few. In addition, load balancing with utilization of distributed web farms was also looked into [12, 13, 14, 15]. Recetly, Malet and Pietzuch in [16] set out to migrate virtual web application components "in the cloud", so as to minimize total response times. They too, however, assume that response times "can be minimized by placing application components closest to the network location of the majority of anticipated users". To the best of our knowledge this is the first work done to investigate a way of utilizing geo-location of servers and clients for the exploitation of time zone differences so as to achieve quicker response times.

3. Model and simulation

3.1 Model

In the model presented, see Figure 1, the network consists of four nodes. Two nodes model the web server, and two nodes model the internet communication network as well as linear modeling internet traffic. The low-level details of the HTTP and TCP/IP protocols are purposely ignored. We use a simple file server over the internet, as every web page, image, etc. is actually transferred over the internet as a standalone file. We assume a single server multi-client architecture, adhering to the M/M/1 model [11].

Justification: The only difference between a server servicing requests and the standard M/M/1 model is the initialization time which is always present. It is common practice, however, to ignore initialization times if they are so minute with respect to the processing time. Here to, we assume the initialization time for processing each request is proportionally insignificant with respect to processing time; hence, it's ignored. In addition, in a web-hosted small company environment, it is typical to have only one server servicing the site; hence, M/M/1. It is possible that the web-hosting company is using co-location and load balancing techniques, but for the most part a single server is used for a specific web-hosted site.

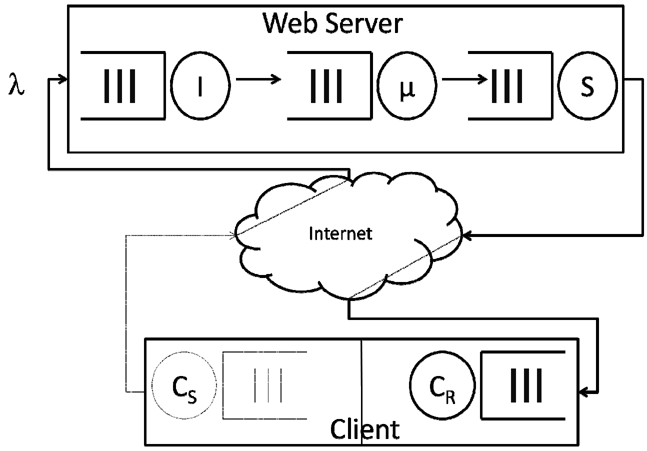

Figure 1: Jobs arrive at the Web server with rate λ. All one time initialization processing is performed at node I. The job proceeds to node μ, where it is processed and passed on to the network. At node S this block of data is travelled via the Internet at constant velocity and is received by the client browser, represented by the node C_R.

The total time for the request-response cycle is given by:

$$T = T_{response} + T_{transfer} \tag{1}$$

and each of the equation parts is defined as follows:

$$T_{response} = \overbrace{\frac{F_{request} + F_{reply}}{C}}^{client\ side} + \underbrace{\frac{F_{request} + F_{reply}}{S} + \frac{I}{1-\lambda I} + \frac{\rho}{\mu - \lambda}}_{server\ side} \tag{2}$$

where:

- $F_{request}/F_{reply}$ – the average file size being transferred for the request/reply
- C/S – the bandwidth of the client/server
- λ – request arrival rate at the server (assumed Poisson) as a function of max arrival rate and *Internet Traffic Percentage* ($ITP(t_{request})$) for a given time zone: $\lambda = \lambda_{max} * ITP(t_{request})$.
- μ – service rate of requests (assumed exponentially distributed)
- ρ – ratio of arrivals to service $\left(\rho = \frac{\lambda}{\mu}\right)$
- I – initialization time per request at the server

and,

$$T_{transfer} = \frac{x - x_0}{\langle v \rangle} = \frac{x - x_0}{\frac{4}{9}C} \tag{3}$$

as a function of distance between client (x) and server (x_0), the speed of transfer ($\langle v \rangle$) [9].

Although [8] stated that data travels through fiber optic cables at almost exactly 2/3 the speed of light in vacuum (C), experiments of [9] exhibited point-to-point (i.e. client to server and vis-versa) speeds of at most 4/9 the speed of light allowing for delays such as circuitous paths, packetization, and other similar delays. Thus, we used this latter measure for the equation 3.

We use the ITP to allow for the change of arrival rate and bandwidth usage throughout the day. It is a measure of traffic as a percent of peak usage measured. Values are extracted from [7] for North America – see Figure 2.

3.2 Simulation

In this section we provide the assumptions the simulation is built upon, the results of the simulation and, the conclusion we arrived at based on these results.

Assumption 1: In order to implement the theoretical model we assume that usage patterns are uniform around the world. That is, that at 6:00AM local time (for every time zone), internet usage is approx. 50% of the peak usage reached at 10:00PM.

Assumption 2: We assume that servers compared are on the same latitude. This assumption is placed so as to minimize the distance / time zone ratio. Having servers on different latitudes would allow for increase of transfer time without change in time zone, which is counter-intuitive to what we are trying to measure.

Assumption 3: Although possible that realistically a client will be closer to a server across time zone "boundaries", we currently assume that servers are strategically placed in the center of the time zone slices. We classify a server as local to its clientele if they are both in the same physical time zone. This, of course, can later be modified by re-creating "time zones" around the placement of servers.

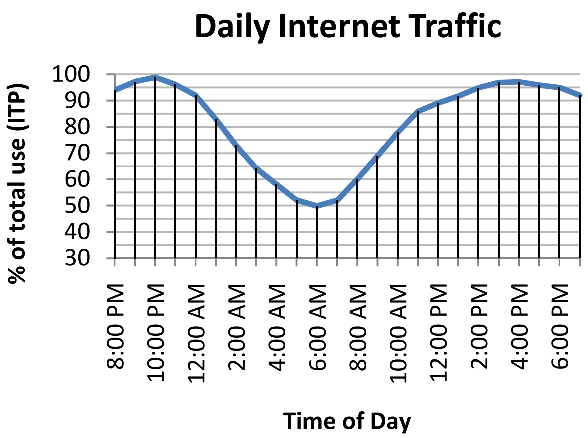

Figure 2: Daily internet traffic for the eastern cost of North America (EST – Eastern Standard Time) as the % of maximum usage. Thus, 50% at 6:00AM depicts half the requests within the system when compared to 10:00PM.

We used the above equations 1-3 to generate Figure 3, where: '+' is $T_{transfer}$, '*' is $T_{response}$, and 'Δ' is T, when placing the server in each of the 24 time zones given the client is in a specific location (EST) for a specific time of the day.

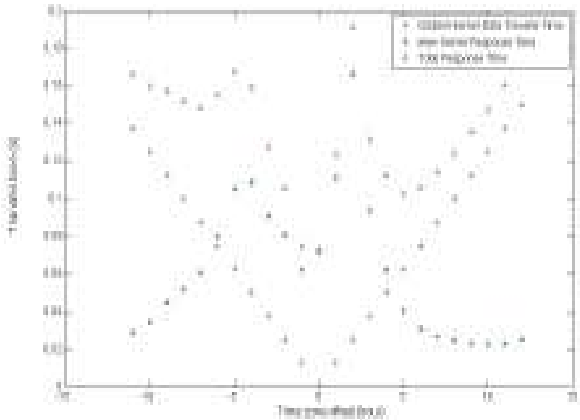

Figure 3: Total response time, T, for each time zone offset (±12 hours) from EST (set as 0 time zones differences between client and server). Each '+' represents the delay based on transition of distance between client (EST) and server location. '*' represents the total time for the server to produce the reply to the client's request. And 'Δ' is the summation of the two.

Generalizing Figure 3 for all 24 hours of the day, we generated Figure 4 in which we show via a 3-dimensional graph the time required for the client to receive a response (color coded) given the time of day at the client's location (assumed EST), and the time zone differences between client and server.

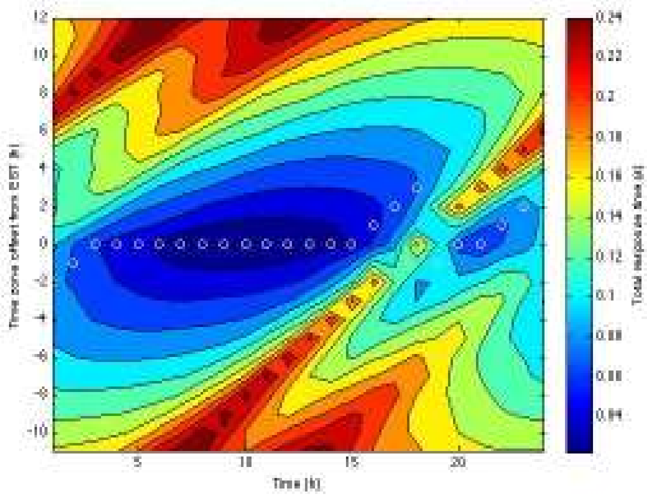

Figure 4: Total response time (s) for each of the 24 hours of the day when the distance between the client and the server is a number of time zones apart. The circles in the figure, depict the minimum time needed for the client to receive a response.

Figure 5 and Figure 6 further refine the information in Figure 4 to include only the minimum times from a side view and a bird's view, respectively.

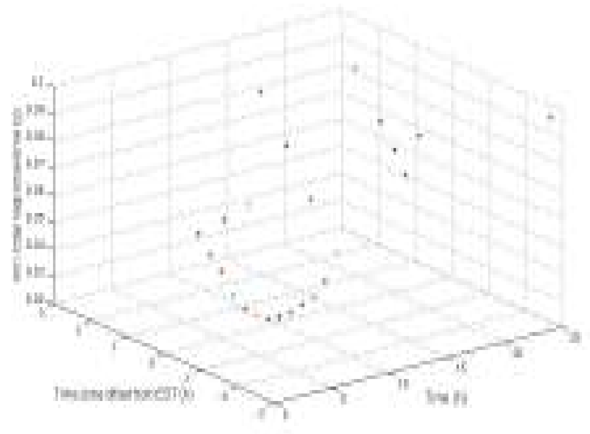

Figure 5: Minimum total response times for each of the 24 hours of the day. These values are the minimum values found in a collection of pre-generated values when executed for each of the 24 hours. This shows us the minimum possible response time when taking into account both time of day and distance (measured in time zone offsets) between client and server.

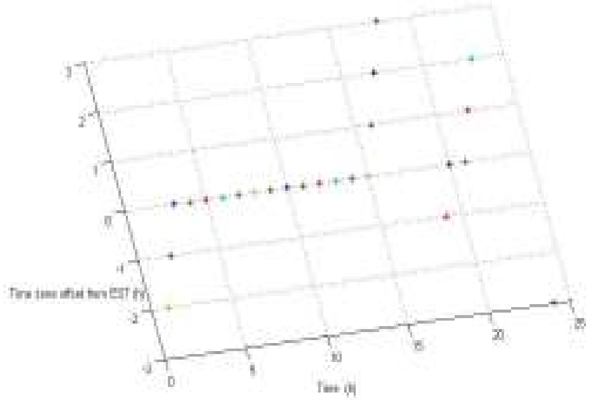

Figure 6: A bird's view of Figure 5, allowing for simpler understanding of time zone significance disregarding the actual response time values.

Examination of the graph (especially Figure 6) shows that indeed a local server (having the client and server in the same time zone) would perform better than off-shoring one for a better part of the day. If, however, one's clientele is primarily between 4:00PM and 2:00AM (such as a movie streaming sites, auction sites and other sites primarily geared to the evening crowds), it would be wise to off-shore a web-hosting plan to a location between 1 and 3 time zones ahead or behind. This would be true even if one doesn't expect peak usage of his server, as web-hosting allows for peak usage based on other sites hosted on the same server.

If, however, one's clientele primarily access the web-site during the work day (between 3:00AM and 4:00PM), the common notion of localizing servers still holds.

4. Future work and conclusion

4.1 Future work

Currently, we are engaged in further expanding this research venue towards testing our hypothesis that given today's common practice it is better to set up a hosting plan in a location where one's peak usage hours are the "off peak" for the majority of other sites hosted on the particular server. Our test plan includes building a benchmark website that will be either heavy on processing need, memory usage, or bandwidth requirements and any combination thereof; setting it up in different locations worldwide with comparable server capabilities; and, executing the request response cycle from different locations worldwide on different times of the day (through proxies or otherwise).

In addition, there are many other directions in which this work can be build upon, including: finding the real distribution of request arrivals and tie them to the above equations, developing a local/foreign host threshold ratio such that determining whether migrating hosted sites is advantageous, tying in the ratio of server to population savvy density (are servers in third world countries, where user bandwidth requirements are minimal, better than servers on backbones where requests are enormous. This might allow for off-shoring servers within the same time zone and while still gaining the "off peak" usage advantage. Further expansion of the model towards more complicated architectures such as multi-server (M/M/N), geographically distributed servers, web farm and/or web garden is also possible.

It is also possible to change the way co-location servers are spread out. Intelligent agents can predict (for a given time) where the best location for each of the servers might be, allowing for the slow migration servers (rather than redirecting requests via load balancers) so as to minimize total response times. This can be done as an augmentation to load balancing techniques, especially when load balancing reached equilibrium but is just not good enough.

Lastly, one can building a tool that will suggest to perspective hosting plan buyers where they should place their sites based on their perspective user location.

4.2 Conclusion

In this work we tried to give an initial model for calculating web response delays due to time zone differences. This allows us to come to a knowledgeable decision as to where to geo-place a hosted site. It was determined that buying a remote web-hosting plan can be beneficial and is desirable for sites that have clientele at evening hours. In addition, a max distance threshold of ±3 time zones from expected clientele is discovered.

Obviously, this work is assuming the fact that current practice doesn't encourage migration of sites. If all site owners of a specific geo-region were to migrate to a second geo-region, the conclusion would be quite different. Thus, the decision as to where to buy a hosting plan is a dynamic one, and the tool we hope to develop will give online answers.

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Quantitative Evaluation of App Inventor in Cloud-Based Chess Programming

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Abstract

App Inventor for Android is a web-based visual programming language that allows non-technical users to program applications for mobile devices with an Android operating system. We used App Inventor to create a chess application to show the capabilities of cloud computing. App Inventor was used to create the user interface (UI) of the chess application, while a chess engine was placed on Google's App Engine. Though App Inventor is a powerful tool, it still has some limitations. Our comprehensive analysis of App inventor includes a user experience as well as a quantitative analysis. From a user experience perspective, we found that the main limitations occurred in the Block Editor, which included programming lag time and the inability to generalize certain function blocks that can simplify the program. Quantitatively, we found that UI cost remained relatively constant with changing moves and difficulty. We also compared the App Inventor chess game with a native version of the game, in which we found that the App Inventor application was much larger than the native code application in terms of file size. We also found that the calculation time was constant between the two games versions, while the communication time varied based off network latencies.

Keywords: App Inventor, cloud computing, mobile computing, chess programming.

1. Introduction

App Inventor for Android (AIA) is a web-based graphical user interface (GUI) programming language which allows users to program applications for devices running the Android operating system. It was made available to the public by Google in 2010. App Inventor is now available to the general public. Anyone with a Gmail account can make an application using App Inventor by going to website in reference [2] and clicking on the "My Projects" link in the upper right hand corner.

App Inventor uses a drag-and-drop interface to piece together an application, just like putting together a puzzle. The two main goals of App Inventor are to allow all Android device users to be able to develop applications, even those without programming experience; and to be a learning tool used to teach students how to program applications for mobile devices [1].

A comparable software is Alice [3], a GUI built to teach students logical thinking and develop early programming skills that will enable students to transition easily into programming with a native language. Unlike Alice, which helps students understand object oriented languages such as Java C++, and C#, App Inventor helps non-technical users understand how to program applications for mobile devices.

We decided to use App Inventor to create an application to show the power of cloud computing. We wanted to choose an application that could partially in the cloud, while other aspects of the application that did not need the power of the cloud could run locally on hardware. For this reason, we decided to use chess, a computationally intensive game in which its user interface (UI) could run locally. Chess engines need to evaluate millions of nodes to calculate the next best move in the game. For this reason, running a chess application on a physically limited devices such as mobile phones can strain its resources. It makes chess an ideal application for cloud computing.

We used App Inventor to program a UI for a cloud computing chess application. Our chess application needed to connect with a Google App Engine server, which was done with the help of a web database component. Other design specifications, such as drag-and-drop versus text box input UI's, were considered and evaluated. The text box input design was chosen due to its relative simplicity, especially when using App Inventor.

The final application file was 2.70 MB compressed, with over two thousand code blocks used to program the game. This is compared with the 65 KB file of a native version of the game that we created. Only some of the rules of chess were implemented, due to time constraints and App Inventor limitations, though the rules that were implemented can easily be extended to fully develop the chess application. The major App Inventor constraints encountered were with the Block Editor, one of the three components that make up App Inventor. Some examples of these constraints include programming lag time (a noticeable time difference between the start of a code block being dragged and when it actually moves), inability to generalize some function blocks (such as the "MoveTo" block), and a fixed Block Editor height in which code length could easily exceed and cause arrangement issues.

We evaluated App Inventor quantitatively by gathering time measurements from the chess application for three different levels of difficulty. We also compared the App Inventor chess game to a native code version that we developed. We found that calculation time remains constant between the two versions, communication time varied due to network conditions, and UI time was much better on the App Inventor version, mainly due to the different implementation methods chosen for the different versions.

Not many related works exist currently toward evaluating App Inventor for Android. This is because App Inventor is a new application, still in its beta phase. An introduction to App Inventor has been written in media outlets such as TechCrunch [4] and the New York Times [5], but they do not exhaustively use App Inventor to create a large application such as chess. The applications they developed were very basic, merely done to give a general idea of what App Inventor can do. Our analysis of App Inventor is more in depth; we exhaustively put App Inventor to the test by using it to develop a chess application, which required a large number of components and code blocks to create.

The rest of the paper will be organized in the following manner: Section 2 will discuss the motivation for the evaluation of App Inventor; Section 3 will explain how we used App Inventor to program chess, including some examples and explanations; Section 4 will discuss App Inventor's comprehensive evaluation, including a user experience as well as a quantitative evaluation; and Section 5 will conclude our findings.

2. Motivation

Cloud computing is a popular concept that is being incorporated into our everyday computing world. Cloud computing offers on-demand resource allocation and a much larger amount of computational capability and storage space compared to local machines. Data in cloud computing systems is stored on a central server, whose services are provided by companies such as Amazon or Google. Some advantages of using a cloud system to run applications include saving energy lessening the burden on local resources, and being able to run applications that many not have been a feasible due to resource constraints.

Cloud computing is also being realized as a viable option for mobile devices. Mobile devices have significantly less physical resources than other computing devices due to their size. Consumers demand a smaller mobile device for ease of transportation, yet demand for more computing power is growing. Today's mobile devices have become a central tool that compact other electronic tools into one. Taking pictures, gaming, paying online bills, and finding directions are all examples of tools that are incorporated in today's mobile devices. However, mobile device resources are physically limited and may not be able to handle the computational demand required without draining a large amount of the battery's power. Mobile devices are then a good case for using cloud computing to alleviate the energy and resource problem.

We know that graphical or computational games are now being used as killer apps [8], or desirable software that proves the core value of larger technology. This is because games that require complex graphic capability or large computational capacity cannot be executed on hardware lacking the physical resources. Cloud computing can make the execution of such applications possible and less straining on local physical resources. We only outsource the computational part of chess to the cloud, while the rest of the application, mainly the UI, would run locally on the mobile device. A chess engine, computer software that can play a game of chess, was placed on Google App Engine servers, which computationally decides the next best move to make for the computer player, while the UI needed to be created that could handle user input and communication with the server.

Our application of choice is chess because of the complexity of the game. Chess is a heuristic game; in other words, the moves made in chess are based on experience. There is no one good way to play a game of chess; this is because a player's best move depends on their opponents moves. Chess engines use a computer's computational power to implement decision algorithms, such as the Minimax or Alpha-Beta Pruning algorithms [10]. In such algorithms, the chess engine can look multiple moves ahead, creating millions of nodes that must be searched to identify the next best move. The amount of computation needed to search such a large number of nodes make chess a desirable application for showing the power of cloud.

Chess proved to be the ideal application for a few reasons. One reason for using chess as our cloud computing application is because of the ease in modifying the search depth of the chess engine; the larger the search depth, the more difficult the chess game will be and the more search nodes the chess engine needs to evaluate. Being able to modify the difficulty of the game can give us comparable data without changing many variables. Another reason for using chess is because of its asynchronous capability. A chess move can be sent to the chess engine from the user interface (UI) and the engine can respond with its next best move without needing to synchronize with the UI. This simplified the communication process between the UI and the chess engine.

The Android operating system was chosen for this project because of the open source nature of its applications. We could create our own application and test it with an Android mobile device with ease.

App Inventor was the language of choice because of its promise of simplicity. Coding chess with native Java code proved some difficulty. Some open source applications were found, such as Honza's Chess [9] and DroidFish [7], but the UI and chess engine were programmed together, and could not be separated easily without rewriting the whole application.

We used a simple chess engine that we found called Pyotr [6], that we modified to communicate with the UI and so that it could be placed on Google's App Engine. It has three levels of difficulty, hard, medium, and easy, that change the amount of computation was done to calculate the next best move. Alpha Beta Pruning was used as the algorithm for computing the next best move. We chose this engine because of its simplicity and because of the fact that it did not incorporate a UI in its native code. Any other chess engine can be used with our App Inventor application with some modification.

The next section will describe some of the steps taken to program chess using App Inventor.

3. Programming Chess with App Inventor

To explain how App Inventor was used to develop a chess application, we will first look at the main parts that make up App Inventor. Then, details of how we developed the chess application will be given, including design considerations and programming examples. This paper will not be a comprehensive guide to using App Inventor; the examples here will only be applicable to our chess application. For more detailed information about how to use App Inventor can be found in reference [2].

3.1 App Inventor

App Inventor is made up of three components: the Designer window, where application components, such as buttons, text boxes, and pictures can be added to create the application view; a Block Editor, where each component is assigned a behavior; and a phone or emulator used to download an application for testing purposes.

App Inventor is designed to be used by anyone, even those with no programming background. App Inventor achieves this by using a using a GUI interface language to avoid restricting app developers to one syntax language. It prevents programming errors by allowing only certain blocks to connect together, to avoid programming something that does not make sense. For example, ImageSprites, an animation component in the Designer window that can be dragged, moved incrementally, or interact with other ImageSprites, can only be placed within a canvas, a two dimensional, touch sensitive panel. App Inventor will not allow you to use ImageSprites outside of a canvas, since the application cannot detect a dragged or touched event occurring with the ImageSprite otherwise. An example of App Inventor's error detection in the Block Editor is not allowing a number block to connect with a text variable block. In summary, App Inventor provides high-leveled components to simplify the programming.

3.2 Developing the Chess Application

3.2.1 Design Considerations.

Two main interface design options were considered. A drag-and-drop interface was one of the design options, in which a user can move a piece simply by dragging it to the user's desired location. The other option was providing an input box, in which the user could enter a valid chess move to make. The second design option, providing a text box and a button, was chosen to simplify the error-checking process. Checking a text box to make sure the user has entered a valid move is easier to implement on the App Inventor versus allowing the user to drag-and-drop a chess piece anywhere on the board. Once the game design was chosen, the components of the game where placed in the Designer window to create the visual image of the chess application.

3.2.2 How it Works: Programming Examples.

The chess application starts with a new game once the screen is initialized. All the ImageSprites (chess pieces) are made visible, enabled, and are placed in the correct starting positions. The user always plays with the white chess pieces and always starts the game.

The user must make an input in the way of "coordinate (letter and number) of piece that I want to move" combined with "coordinate (letter and number) of box I want to move the piece to." An example of this would be "e2e3," which means the user wants to move the fourth pawn from the right (located in e2) one space forward to e3. If a user inputs an incorrect move, such as selecting an empty box to move or selecting a letter or number that is outside the range specified on the board, the user will be notified that an error has occurred. Each letter and number correspond to specific x and y values that are used to move each piece to its correct position on the board. This correspondence is stored in the TinyDB, which is used as a reference table to match the user's text input with proper x and y values that can be used to move the chess pieces.

Once the user has entered the move that will be made and presses the "Send" button, the application scans the pieces for the correct piece that correspond with the user's selection. This is done by checking whether the current x and y values of each piece on the board match to the first x and y values provided by the user (the first two characters in the string). Once the piece is identified, it is moved to the x and y values that correspond to the user's specification

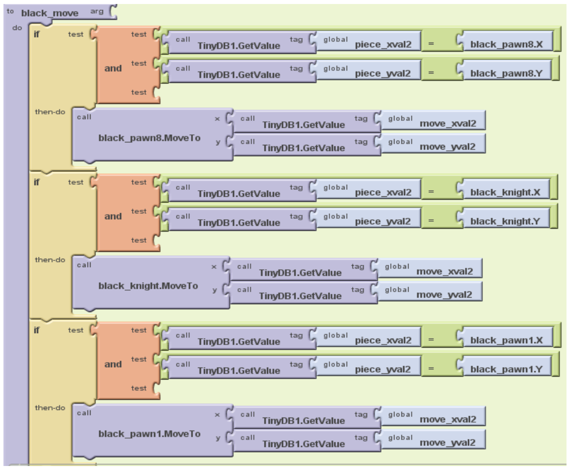

Figure 1: Sample code from App Inventor used to move the correct piece to its destination.

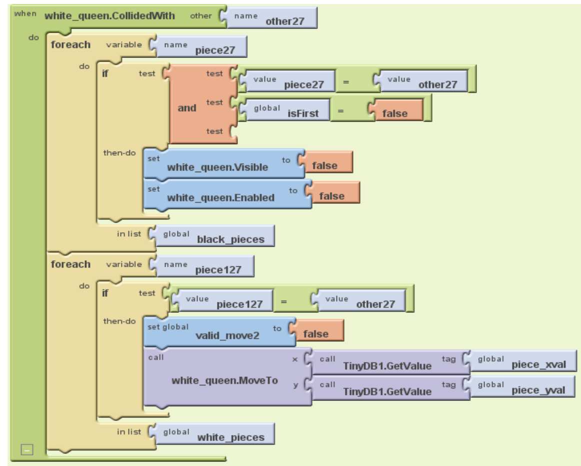

Figure 2: Example code for handling a collision event. A white queen collision event is being handled above.

(the second two characters in the string) using the "MoveTo" function block. Sample code of this search and move of a piece can be seen in figure 1. This search had to be made for all thirty-two pieces because the App Inventor cannot generalize function blocks for all pieces. For example, the app inventor cannot do the following:

```
for (int i = 0; i < white_pieces; i++){
    if (( i.x = xvalue_entered) & (i.y =yvalue_entered)){
      MoveTo(xvalue_entered, yvalue_entered)}
 };
```

The native code iterates through all the pieces until it finds the one that matches with the coordinates specified by the user, and then moves that piece to the user's desired location. It has been generically designed to work for any white piece. In the App Inventor, each "MoveTo" function (as well as all the other functions) is associated with a specific ImageSprite. Thus to move a block, it has to be checked as the block that needs to be moved first, only then can it be moved by calling the appropriate "MoveTo" function.

The user's move is sent to the server using the TinyWebDB component, which uses JSON requests, a protocol that allows the application to send a string to a URL (in our case, the server's URL) and waits until a response is sent back. Once the string is received by the chess engine, it is processed by the server and next best move to make is calculated. Once the move is chosen, the server sends another string back to the application with the same four-character string format described above. The application then moves the black piece for the computer just as it would a white piece for the user. Moving a white piece, however, is more complicated than moving a black piece because human error needs to be considered. Measures were taken to check for an invalid chess move made by the user. Since the computer's move is computed arithmetically, it is assumed that the computer will not make an invalid chess move.

In the case of one piece taking another, the "CollidesWith" function block is used. This function block checks to see if the ImageSprite has collided with another ImageSprite. Like the "MoveTo" function block, the "CollideWith" function block is associated with a specific ImageSprite and cannot be generalized. For this reason, each chess piece must handle colliding with another piece separately. If a white piece collides with a black piece and a black piece made the move that caused the collision, then the white piece has been taken. To take a piece, the ImageSprite is made invisible and disabled. If a white piece collides with another white piece, then the user has made an error in moving the piece. The white piece is moved back to its original position and the user is asked to make another move that is valid. This error checking is not performed for the black pieces because it is assumed that the chess engine will always make a valid move. Example code of handling a white piece's collision can be seen in Figure 2.

A new game can be started at any time by pressing the "New Game" button, which resets all the pieces to their original position, makes them all visible and enabled (in case there was a piece taken), and notifies the chess engine that a new game has been initiated so that the server can reset the game as well.

To test the effectiveness of our cloud-based chess application, a clock component was added to measure the time difference between sections of code. For example, the time was measured right after the "send" button was pressed by a user as well as right after the application receives a response from the server. The difference of this time in milliseconds is the total time spent by the server, which includes time spent calculating the next best move as well as communication time spent sending strings to and from the server. To find the communication time, we had the chess engine send its own calculation time that it computed. The calculation time was sent along with the move in the same string, which was then parsed in the application. The total server time calculated by the application level minus the calculation time received from the server equals the total communication time it took to send the information to and from the server from the mobile device.

Other time measurements were computed in the same manner. A Notifier component as well as text boxes were used to display the time measurements so that we may collect and analyze them. Section 4 will go into detail about the analysis of the results collected from the App Inventor Application.

4. Quantitative Evaluation of App Inventor

4.1 App Inventor User Experience

App Inventor was a helpful tool to use, but it did have its limitations. One limitation was the inability for function blocks to be generalized for different components. This made programming a chess application much more challenging since the same procedures had to be repeated for thirty-two chess pieces. The repetitiveness of the code created a very large program, which included over two thousand blocks of code (table 1 breaks down the code block count), a total of 2.70 MB file, 1.04 MB of which was code and 61.3 KB of which were pictures used for the interface. The application unpackaged on the phone was measured at 4.74 MB. The large App Inventor code was compared to a chess application (with the same functionalities as the App Inventor version) that we developed using native java code, which had a file size of only 65 KB. The large size of the code makes the Block Editor respond very slowly to the changes the programmer wants to make to the code. Quite a few seconds worth of lag time was noted when trying to move a block of code from one area in the Block Editor to another.

It should be noted that not all chess moves were implemented. The rules of chess were only applied to some pieces because of their simplistic movements. For example, the knight was relatively easy to program since it only moves in an "L" shape (e.g. two vertically up and one space over horizontally, or vice versa). The pawn, however, is much more difficult to program since it can move in different directions depending on its position on the board. To fully implement the rules of chess, the application would have been much larger, creating a larger lag while programming. The Block Editor is limited in other ways as well. For example, multiple blocks could not be selected and moved at the same without connecting them together first. In the case of a large program where the programmer starts to experience longer App Inventor lag time, moving chunks of large code could take quite a while. Moving each

Table 1: A breakdown of the number of code blocks used to program chess. A code block refers to a single puzzle piece, not a full block of code.

Breakdown of Code Block Count

	Initialize	Move	Collide With	Total
White Pieces	112	364	508	984
Black Pieces	112	272	284	668
Basic Blocks	169	224	-	393
				2045

block separately or connecting them first before moving them can mean hours of tedious work. This can seems very inefficient, especially when comparing it to a select, cut, and paste of chunks of native code, which can take seconds to accomplish.

Another example is that the Block Editor had a fixed window height. Thus, if a block of code was longer than the Block Editor's window size, it would run off the window and could not be seen by the programmer. If any changes needed to be made in the lower part of the code, it had to be proved time consuming, especially when the code had to be written in a certain order.

One minor drawback in the Block Editor is the inability for some code blocks to expand to allow multiple inputs. For example, two texts can be joined together to create a single text by using the "join" code block. The problem dislocated from its function block, sometimes in multiple fragments, and reinserted after the changes were made. This occurs if you have more than two texts that need to be joined together. Multiple "join" blocks then need to be used, one "join" within a larger "join" block, to achieve the task of combining multiple chunks of text. Similarly, in native code, a developer can write "if/else" statements comprised of multiple "else if" conditions. In App Inventor, an "if/else" code block only has one "if" condition and one "else" condition. Thus, to chain together multiple "else if" conditions, multiple "if/else" or "if" statements need to be used within a larger "if/else" code block. Intuitively, a developer might expect one "join" block or "if/else" block to expand, allowing multiple conditions or statements to be incorporated within one block rather than using multiple blocks to achieve the same task.

Even though a touch screen interface would have been more user friendly, it is much more difficult to implement because of the limitations previously discussed. A touchscreen interface would have needed a larger amount of error checking, especially to center the piece that was dragged to the middle of the square the user intended, would have made the code much larger, and in turn, much slower and harder to work with.

Persistent storage at the server side was also considered to reload the chess application with a game that was abandoned. This would not have been feasible on the client side when using the App Inventor since each piece needed

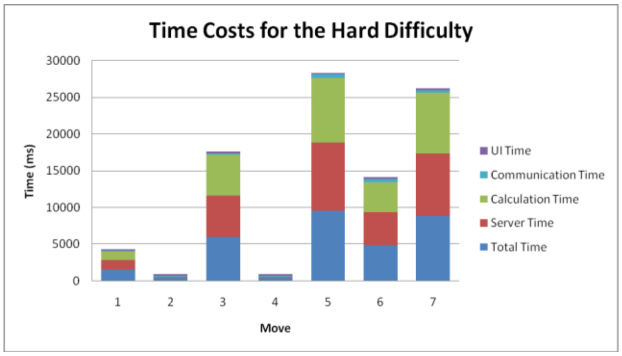

Figure 3: The time cost results for the hard difficulty level. It can be noted that the UI time remains relatively low throughout each move compared to the other time measurements taken. Communication time remains relatively low as well.

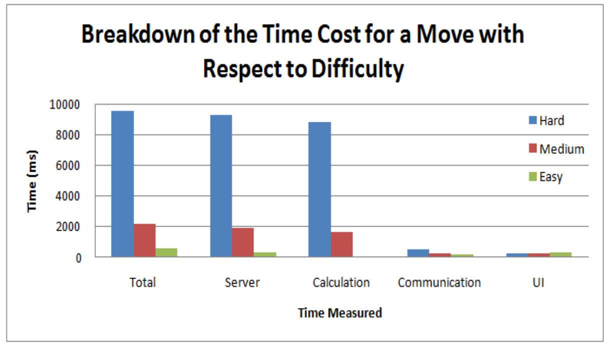

Figure 4: The breakdown of time cost for a single move based on the difficulty of the chess engine. Again, it can be noted that the UI and communication times were relatively low, especially with the higher difficulty.

to be checked for its last position on the board, as well as if it was previously "taken" off the board (when the piece was really made invisible and disabled to achieve the same visual idea). This again would have made the code much larger and more difficult to work with since it had to be implemented thirty-two times, once for each piece.

4.2 App Inventor's Quantitative Evaluation

The App Inventor chess application was tested using Google's Nexus One mobile device, which has the Android 2.2 operating system, a 512 MB memory, and a 1 GHz Qualcomm QSD 8250 Snapdragon processor. The results of the App Inventor chess application are recorded in terms of time cost. Total time includes the time the user interface (UI) took to make one white move and one black move, as well as the total time the server took to respond back to the application. The Server cost includes communication cost to and from the server and time the server took to calculate a move. Time cost was calculated for three difficulties hard, medium, and easy. A total of seven moves were made. The moves 1 to 7 are respectively as follows: e2e3, g2g3, b2b3, g1f3, f3e5, f1e2, b1c3. These moves were chosen because of their simplicity so that the order of moves can remain constant as the game difficulty changes. If more strategic moves were chosen, it could not be guaranteed that the same moves made in one difficulty could be made in a different difficulty.

Figure 3 and 4 summarize our findings. In figure 3, we can see that the UI time and communication time were relatively the same throughout each move. From figure 4, we can see that the higher the engine difficulty, the longer the server took to calculate the next move. This also effected the total time and total server time, which were also longer. Changing the difficulty did not have an effect on the communication cost and UI cost. Communication time ranged anywhere from 233 milliseconds (ms) to 829 ms, which is pretty quick considering the chess engine is located on a Google App Engine server in California. UI time ranged anywhere from 637ms to 1310 ms, depending on which chess piece was being moved since it executes in a different part of the code.

It should be noted that the calculation time does not relate to the move that is played. For example, the chess engine took 5 seconds to calculate move 3, yet it wasted no time in calculating move 4, replying back to the mobile device almost immediately. This is a characteristic of the chess engine itself and not the cloud nor the application itself.

We have also considered the location of deployment for our chess engine. We re-tested the App Inventor application with the chess engine running on a local machine to compare with the Google App Engine deployment. As shown in Figure 5, the total time cost per move is very close between the two locations of deployment, with the largest difference being around 200ms. The major difference comes down to the calculation time (which can vary slightly due to server resource availability) and communication cost (due to network latency). This difference can be neglected compared to the total time cost. We can conclude that it does not matter where the chess engine is located, whether it is in California (the location of Google App Engine servers), or right next to the mobile device; the total time cost will remain relatively same.

The App Inventor chess application was also compared to a chess application created using native java code. We first should mention that the native code was implemented differently than the App Inventor code. In the native code, the text box and send button because the native code can be generalized much easier than the App Inventor code. We were able to create a touch screen interface versus using determine where the user intended on placing the chess piece. For this reason, the UI time took the majority of the native code's time to execute as shown in Table 2. The App

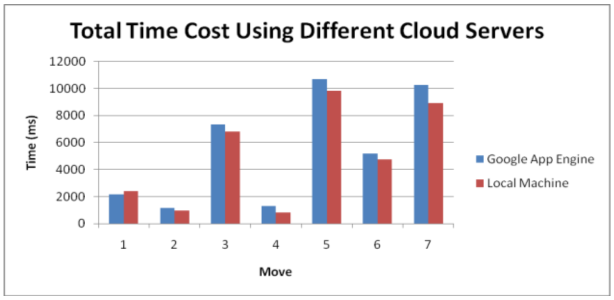

Figure 5: The total time cost for a hard level difficulty game using the chess engine on the Google App Engine servers as well as a local machine.

Table 2: A low level difficulty game's time average comparison for three time measurements collected from the App Inventor chess application and the native code chess application. The native code's UI time takes up a large portion of the total time spent per game. Communication time remains the same and communication time also remains relatively the same.

Comparison of App Inventor and Native Code Average Time Measurements (in ms)

	Calculation Time	Communication Time	UI Time
App Inventor	56.14	275.14	207.14
Native Code	57.71	130.28	1926.28

Inventor code did not have this problem, since it used the text box and a send button to receive the input from the user. The App Inventor code's UI time measurement is thus much less than that of the native code's UI time.

We also compared the actual values of the calculation time and communication time between App Inventor and the native code. From Table 2, we can see that calculation time is the same between the two applications as long as the same level of intelligence was used. We can also see that communication time between the two applications is relatively close.

5. Conclusion

App Inventor is a powerful tool when it comes to developing relatively small applications. It can simplify programming an application from a few hours of work to a few minutes. However, some drawbacks are experienced when developing a large application. The Block Editor's limitations make up the majority of the flaws with App Inventor, such as the inability to generalize some function blocks, a fixed Block Editor's window height, and lag time between dragging of code blocks. We found that even with such a large application, App Inventor's UI time is relatively low compared to the total time of move executions within the chess application. We also found that communication time depends solely on network conditions, and calculation time does not change between App Inventor and native versions. App Inventor file size is also much larger than its native code counterpart.

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CENTRALIZED BUFFERING AND LOOKAHEAD WAVELENGTH CONVERSION IN MULTISTAGE INTERCONNECTION NETWORKS

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ABSTRACT

In this paper, methods to alleviate the problem of internal blocking in interconnection networks based on WDM are studied. In an ordinary 8x8 Omega network, only 10% of all permutations are permissible in one pass, and it gets worse with larger switches. However, using WDM technology, the performance of these networks can be improved. In this paper, several architectures based on Omega network using the WDM technology are considered and in turn algorithms to resolve the problem of internal blocking in a centralized fashion are introduced. Performance of the Omega network is analyzed by simulation. It is shown that by using a few buffers and lookahead wavelength converters a considerable amount of improvement in the system performance is achieved.

Keywords: Omega Networks, Multistage Interconnection Networks, Internal Blocking, Buffering, Wavelength Conversion, Wavelength Division Multiplexing.

I. INTRODUCTION

Recently, significant developments have been made in photonic switching based on *Wavelength Division Multiplexing* (WDM). Therefore, researchers have used this technology to implement switches such as the *Crossbar* or *Multistage Interconnection Networks* (MIN) and to upgrade the performance of the available systems that use these networks. An $N \times N$ crossbar switch is a single-stage, strictly non-blocking network with N input ports and N output ports. It can realize $N!$ permutations (any one-to-one mapping between the set of inputs and the set of outputs). Therefore, it does not suffer from internal blocking. However, the hardware complexity of an $N \times N$ crossbar is $O(N^2)$ since it consists of N^2 cross-points. Therefore, it is expensive and only appropriate for small switches, say, with $N \leq 16$. The other class of interconnection networks is the Multistage Interconnection Networks (MINs) which consist of a few stages of a number of smaller switch elements. MINs with only a few stages suffer the problem of internal blocking. However, it is cheaper and faster. For example, by using 2×2 switch elements, only $\log_2 N$ stages are required to achieve full access capability in an $N \times N$ switch. Using, 2×2 switch elements, data might go through some unwanted changes when the two input channels try to access the output channel simultaneously. To resolve this problem, it is suggested to use two queues, one at each output port [1]. However, using the same switch based on WDM, two or more packets may share the output channel provided they use different wavelengths. For this reason and other advantages [2] [3], WDM is used in our network. Therefore, a 2×2 switch with additional configurations is introduced, namely the upper and lower mergers, and the upper and lower splitters. The internal blocking is redefined as two or more packets with the same wavelength trying to access a channel simultaneously. This problem can be eliminated by increasing the number of switch elements [4], the number of stages [4], or the size of the switch element [5]. However, all these techniques increase the cost and delay of such networks. Therefore, in this paper, utilizing the same few switch elements while maintaining full accessibility is attempted.

To alleviate the problem of internal blocking in MINs based on WDM, buffers [6] or wavelength converters [7] are used. The advantage of wavelength conversion over buffering is the ability to utilize the available channel bandwidth and to send a packet to its destination without waiting for the next switching cycle. Buffering and wavelength conversion techniques have been studied in detail in all-optical networks based on circuit switching and crossbar switches [8] [9]. Several algorithms are introduced defining the behavior of the central controller which acts as an interface in front of the MIN to resolve any internal blocking. Once the central controller resolves the internal blocking by buffering, wavelength conversion, or dropping of the packets, it directs the packets through the network without any collision.

Section II briefly describes our target multistage interconnection network, namely, the Omega network based on WDM. Section III presents an algorithm which also incorporates the concept of wavelength conversion. Section IV presents the concept of lookahead wavelength conversion. The performance of these algorithms is evaluated by simulation in section V. The final section concludes the paper.

II. ARCHITECTURE OF OMEGA NETWORKS BASED ON WDM

Based on the previously discussed characteristics of MINs, an $N \times N$ Omega network, first developed by Laurie (1975) [10], is best suited for WDM implementation. It consists of $\log_2 N$ identical stages, and each stage consists of a *perfect shuffle* connection followed by $N/2$ of 2×2 switch elements as illustrated in Fig. 1.

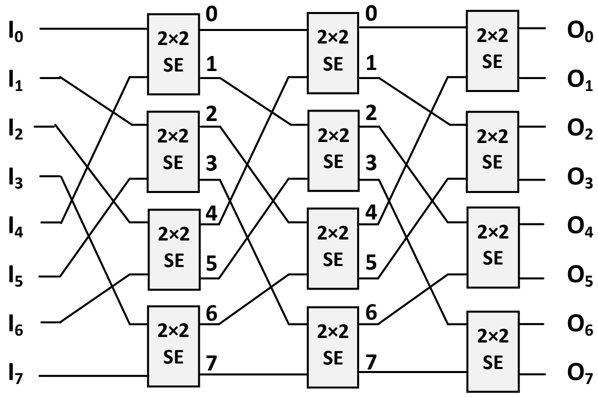


Fig. 1 An 8×8 Omega network

Since this network is based on WDM, and each input link can carry at most ω packets, each with a different wavelength from the set of available wavelengths, $\Lambda = (\lambda_1, \lambda_2, \ldots, \lambda_{w-1}, \lambda_w)$, one may merge both inputs of a 2×2 switch element and forward them to either the upper or lower output link when the sets of wavelengths on both input links are disjoint. Therefore, two configurations are to be considered, namely, upper merger and lower merger as illustrated in Fig. 2c. Note that these two configurations would have been considered as internal blocking in an ordinary Omega networks. Moreover, two additional configurations are required, namely, the upper splitter and lower splitter as illustrated in Fig. 2d to satisfy the requirements of one-to-one mapping.

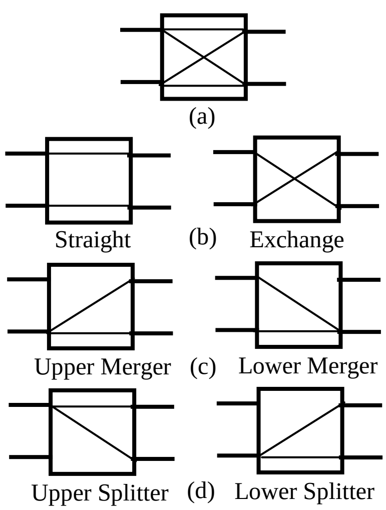


Fig. 2 Configuration of a 2 × 2 switch element

The set of permutations realizable by an Omega network is characterized by having $n-1$ windows, where $n=log_2N$, and each of them is a permutation. To understand this concept, concatenate the binary representation of all sources, $s_{n-1}s_{n-2}\ldots s_1s_0$, and the binary representation of the corresponding destinations, $d_{n-1}d_{n-2}\ldots d_1d_0$. This generates a table with N rows and $2 \times n$ columns, and which can be represented by $(d_{n-1}d_{n-2}\ldots d_1d_0\, s_{n-1}s_{n-2}\ldots s_1s_0)$. Now, $n-1$ windows can be defined as,

W_1 : $s_{n-2}\ldots s_2s_1s_0d_{n-1}$

W_2 : $s_{n-3}\ldots s_ns_{n-1}d_{n-2}$

...

W_i : $s_{n-i-1}s_{n-i-2}\ldots s_1s_0d_{n-1}d_{n-2}\ldots d_{n-i}$

...

W_{n-1}: $s_0d_{n-1}d_{n-2}\ldots d_2d_1$

A window W_i, illustrated in Fig. 3, has N rows, each with log_2N bits. If W_i is a permutation, i.e., no two rows in W_i are equal, then, it is guaranteed that there is no internal blocking in any switch element in stage i; otherwise, there is at least one switch element in stage i with both of its inputs competing on the same output link, which causes internal blocking in the ordinary Omega network. Since WDM is used, the switch element can be configured to either upper merger or lower merger. However, if the sets of wavelengths on both inputs of that switch element are disjoint, then there will be no internal blocking; otherwise, there will be internal blocking, and it can be resolved by either packet buffering or wavelength conversion.

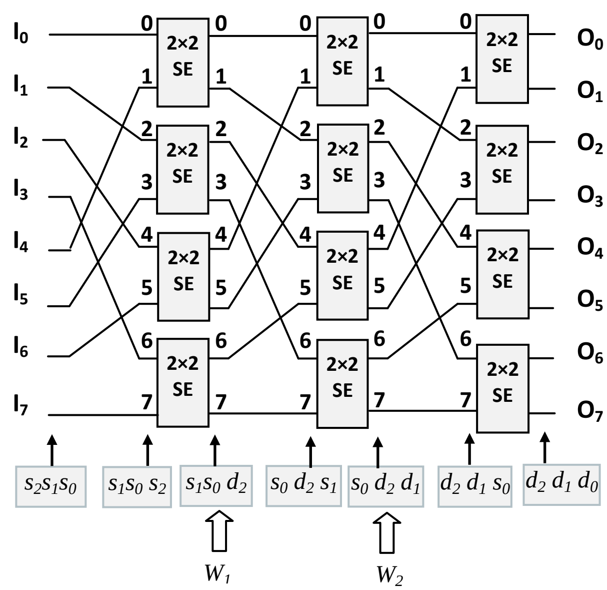


Fig. 3 No internal blocking if W_1 and W_2 are permutations

The main drawback of this architecture is that there is only one path between any source and any destination. Therefore, failure of any path causes a loss of full access capability. To increase the reliability of Omega networks, redundancy is added [11][12][13].

III. 8X8 OMEGA NETWORKS WITH WAVELENGTH CONVERSION

In this section, *Wavelength Converters* are included in the central controller of the Omega network as illustrated in Fig. 4. The purpose of a wavelength converter is to convert the wavelength λ_i of a packet to another wavelength λ_j, with λ_i not equal to λ_j. Therefore, if there is internal blocking, and wavelengths of packets that cause the internal blocking are converted, then the number of packets to be buffered by the central controller can be reduced, and the performance of the network will improve. Note that a packet can have at most one wavelength conversion and this happens inside the central controller. The number of packets which can have their wavelengths converted is limited by the number of available converters. Packets which will cause internal collision and cannot be wavelength converted are buffered. If no more buffers are available, packets are dropped.

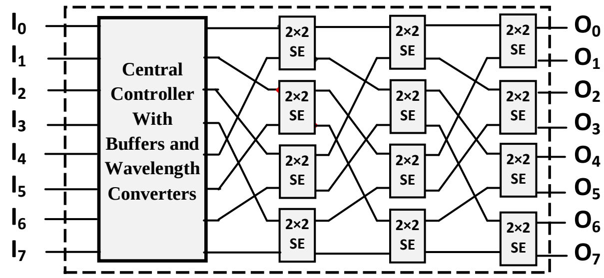
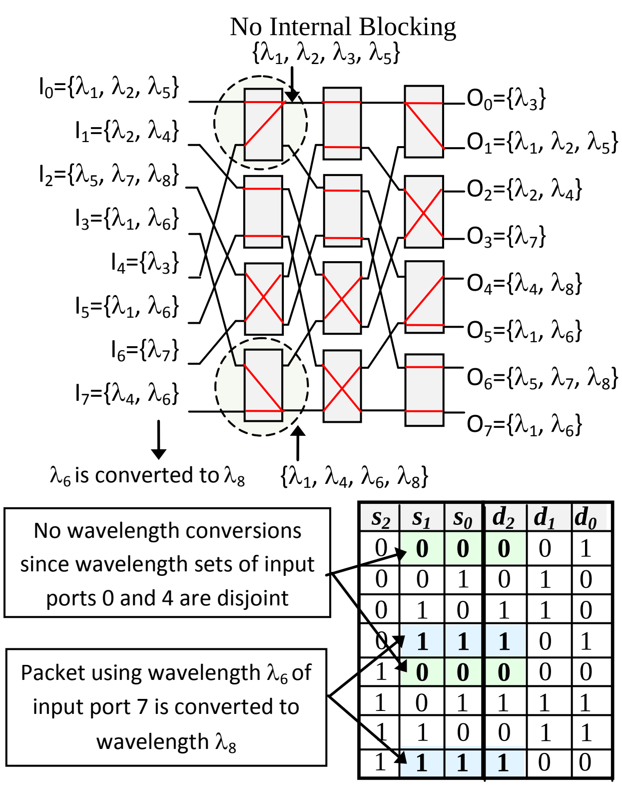


Fig. 4 Architecture of a collision free Omega network

Fig. 6 illustrates an example of resolution of internal blocking in Omega network by wavelength conversion. Assuming that the arriving set of packets to the input port (I_3) uses wavelengths $\{\lambda_1, \lambda_6\}$, and the arriving packet to the input port (I_7) uses wavelength $\{\lambda_4, \lambda_6\}$, and both are sent simultaneously to the same output port, then there will be internal blocking by the packets that use wavelengths λ_6. However, if the wavelengths λ_6 of one of the packets is converted to λ_8, then there will be no internal blocking at that switching element, and its output port can forward the set of packets with wavelengths $\{\lambda_1, \lambda_4, \lambda_6, \lambda_8\}$ to the next stage.



s_2	s_1	s_0	d_2	d_1	d_0
0	**0**	**0**	**0**	0	1
0	0	1	0	1	0
0	1	0	1	1	0
0	**1**	**1**	**1**	0	1
1	**0**	**0**	**0**	0	0
1	0	1	1	1	1
1	1	0	0	1	1
1	**1**	**1**	**1**	0	0

Fig. 5 Resolution of internal blocking in Omega network by wavelength conversion

To design the algorithm, a one-to-one mapping of the set of sources and the corresponding destinations is considered to be switched by an Omega network. By concatenating the binary representation of the sources, $s_{n-1}s_{n-2}\ldots s_1s_0$, and the binary representation of the corresponding destinations, $d_{n-1}d_{n-2}\ldots d_1d_0$, *n-1* windows are generated such as:

$$W_i = (s_{n-i-1}s_{n-i-2}\ldots s_1s_0d_{n-1}d_{n-2}\ldots d_{n-i}) \quad \text{for } i=1,2,\ldots,n-1$$

Algorithm Notations:

- $\boldsymbol{B}$ is the number of buffers in the central controller.
- $\boldsymbol{W}$ is the number of wavelength converters in the central controller.
- $\boldsymbol{\Lambda}$ is the set of all possible wavelengths.
- $|\boldsymbol{S}|$ is the number of elements in a set $\boldsymbol{S}$.
- $\boldsymbol{I}_u$ is the set of wavelengths arriving to the upper input port of a 2×2 switch element.
- $\boldsymbol{I}_l$ is the set of wavelengths arriving to the lower input port of a 2×2 switch element.
- $\boldsymbol{SW}$ is the shared wavelengths between I_u and I_l. It can be represented as: $\boldsymbol{SW = I_u \cap I_l}$
- $\boldsymbol{NU}$ is the set of wavelengths that are not used by either input port of a switch. It can be represented formally as: $\boldsymbol{NU} = \Lambda - (\boldsymbol{I_u \cup I_l})$
- $\boldsymbol{O}_k$ is the set of wavelengths that utilizes output port k without any internal blocking.

Wavelength Conversion Algorithm:

Step 1: Let $i=1$.

Step 2: Choose the corresponding W_i.

Step 3: Set switch elements at stage i. If W_i is a permutation, go to step 5.

Step 4: For every two rows in W_i which are equal to a value, say, k, perform the following on switch number $\lfloor k/2 \rfloor$ at stage i:

If ($SW = \phi$), then,

- Set $X = I_l$
- $O_k = I_u \cup X$

Otherwise,

- Compute NU
- If $|SW| <= |NU|$, then,
- Convert all wavelengths of the lower input link that are included in SW. This conversion is limited by the number of available converters (W). The resulting set of converted wavelengths is denoted by X
- $O_k = I_u \cup (I_l - SW) \cup X$

Otherwise,

- Convert only $|NU|$ wavelengths of the lower input links that are included in SW. This conversion is also limited by the number of converters (W). The resulting set of converted wavelengths is denoted by X
- Buffer the packets of the lower input link that are included in SW and not converted. If no buffers are available (B=0), discard the packets
- $O_k = I_u \cup (I_l - SW) \cup X$

Step 5: Tag the unused outputs of Stage *i*, and compute the new sets of inputs for the stage *i+1*.
Step 6: Let $i=i+1$. If $i < n=log_2N$, then go to step 2.
Step 7. Set the switches of stage *n*, and stop.

Given a permutation, a central controller initially can detect any internal blocking, and then it converts, if possible, some of the wavelengths, and buffers other wavelengths if required. This can all be done by the controller before the network performs the given permutation. Therefore, the network is collision-free.

IV. Lookahead Wavelength Conversion

In this section, a *16×16* Omega network with arbitrary connections is considered; i.e., the one-to-one mapping constraint has been relaxed. Therefore, the architecture of this network will be similar to Fig. 4 except it is 16×16 and the 2×2 switch element has the general configuration shown in Fig. 2a.
Considering the following problem, if a packet with wavelength λ_i is converted to λ_j to avoid collision at the first merger, and there is a packet with the same wavelength λ_j at the other input of the following merger, then this causes unnecessary internal blocking due to the poor choice of the first wavelength conversion as illustrated in Fig. 6. Therefore, an algorithm to avoid these unnecessary wavelength conversions is to be developed.

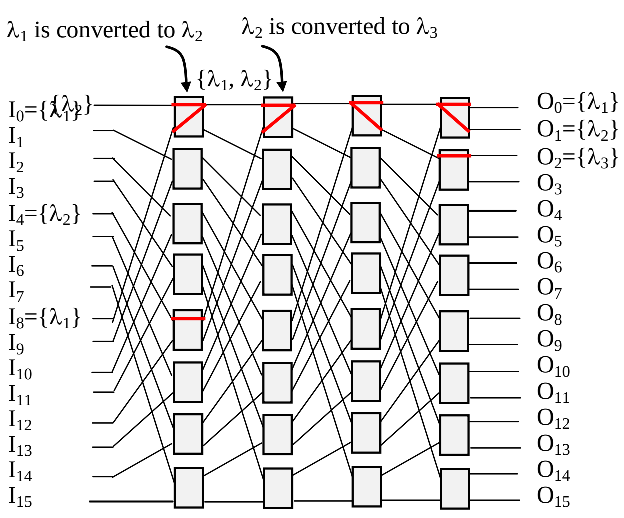


Fig. 6 Example shows unnecessary wavelength conversion

To reduce the complexity of this algorithm, the following approach is used to resolve internal blocking. Since an Omega network is well defined, it is easy to figure out the sources of all packets of any path from a source to a destination. In addition, all switch elements with lower or upper merger configuration are also known. Therefore, the central controller of this network can compute all wavelengths to be converted or buffered before forwarding these packets through the network.

The Lookahead Wavelength Conversion Algorithm:

By inspecting the destinations of the packets at the head of the *N* input links, the controller can calculate the *N* paths through the network.

Phase 1: Creating the list of sources that utilize a link.

```
For (j=0;  j<n; j++)        /* n is number of stages */
  For (i=0; i<N; i++)       /* N is the number of input channels */
     Compute the sources that use the output channel of SE(i,j).
    /* SE(i,j) is a 2×2 switch element indexed by i and j */
   END FOR i.
END FOR j.
```

Phase 2: Creating the conflict lists for every source.

```
For (i=0; i<N; i++)
    For (j=0;  j<n; j++)
        Compute the output channel of SE(i,j).
        If (SE(i,j) is MERGE)
            Add the sources that share the output channel to the
            conflict list of source i (excluding the source i itself
            and any redundant source).
        END IF.
    END FOR j.
END FOR i.
```

Phase 3: Resolving the internal blocking by wavelength Conversion and buffering.

```
For (i=0; i<N; i++)
        For (j=0;  j<k; j++)  /* k is the maximum number of
        wavelengths per channel */
          IF (Conflict List is not empty)
            IF (Other source has the same wavelength j)
           Add this wavelength j to the Collision List of source i.
           END IF.
           IF (all sources including source i not using
           wavelength j)
          Add this wavelength j to the Free List of source i.
          END IF.
        END IF.
        END FOR j.
/* Wavelength Conversion */
Convert as many wavelengths in the Collision List to the
wavelengths available in the Free List. This conversion is of
course limited by the number of converters (LWC>0).
/* LWC is the number of lookahead wavelength converters in
  the central controller */
Buffer the rest of the packets (wavelengths not converted) in the
Collision List. If no buffers are available (B=0), discard the
packets.
END FOR i.
```

The computational complexities of these phases are $Nlog_2N$, $Nlog_2N$, and Nk, respectively. Therefore, the central controller can perform the computations and identify the necessary wavelength conversion and buffering in real time.

V. Performance Results

This section presents the performance of a 16×16 Omega network with arbitrary connections. Three configurations are considered: First, a network with limited number of buffers. Secondly, a network with limited number of lookahead wavelength conversions and finally, a network with limited number of lookahead wavelength conversions in addition to limited buffering. The performances of these configurations have been analyzed by simulation.

The Simulator consists of two programs. The first one randomly creates 10,000 arbitrary connections and set of inputs. Each input channel has at most 16 random packets with different wavelengths. The first program will be executed 10 times for different arrival rates. The second program simulates the behavior of that network. Also, it reads the data generated by the first program and generates the performance parameters such as packet dropping probability and buffering probability.

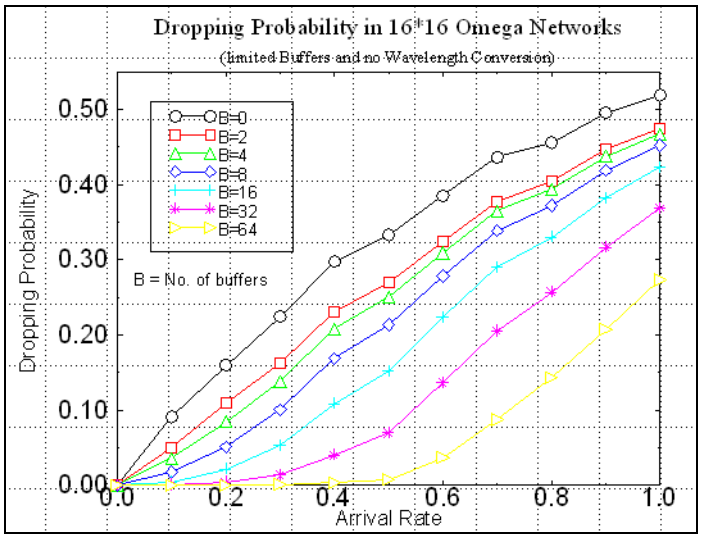


Fig. 7 Dropping probability versus arrival rate (limited buffers and no wavelength conversion)

Considering a network with a limited number of buffers, Fig. 7 illustrates the packet dropping probability versus arrival rate. As expected, the packet dropping probability decreases with increasing number of buffers. Fig. 8 illustrates the probability of buffering versus arrival rate. We define the probability of buffering as: when a random packet arrives to this network, the probability that this packet will be stored in a buffer and will be forwarded in Omega network in the next switching cycle. Initially, probability of buffering for different number of buffers increases linearly with the network load because the load of the network is satisfied with the available buffers. However, at some point these curves start to saturate and then go down. The reason is that the number of packets in the network becomes very large with respect to the available buffers and some of them are dropped. Also, Fig. 8 shows that the point of saturation moves to the right with the increase of buffers.

Using the concept of lookahead wavelength conversion, a considerable improvement can be achieved. Fig. 9 illustrates packet dropping probability versus arrival rate. It shows that a network with a few wavelength converters can improve considerably the network performance. Fig. 10 illustrates packet wavelength conversion probability versus arrival rate. Initially, the curves increase linearly, then start to saturate due to the increase in the number of packets.

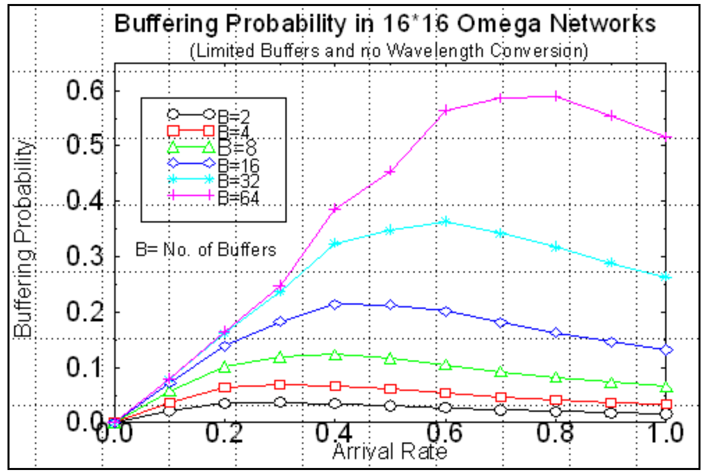


Fig. 8 Buffering probability versus arrival rate (limited buffers and no wavelength conversion)

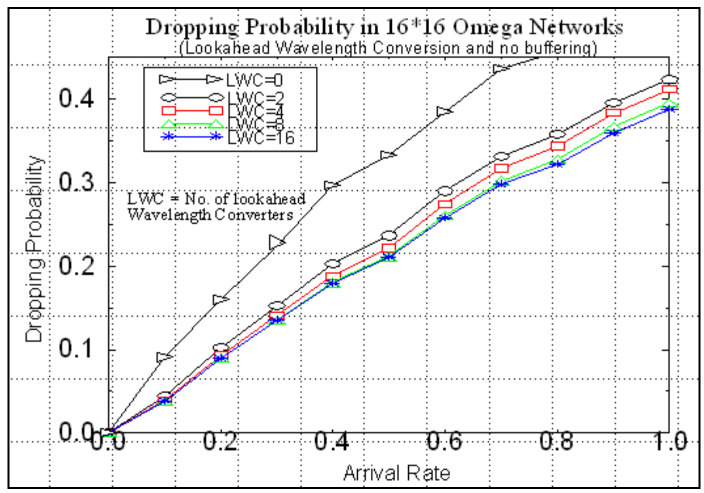


Fig. 9 Dropping probability versus arrival rate (lookahead wavelength conversion and no buffering)

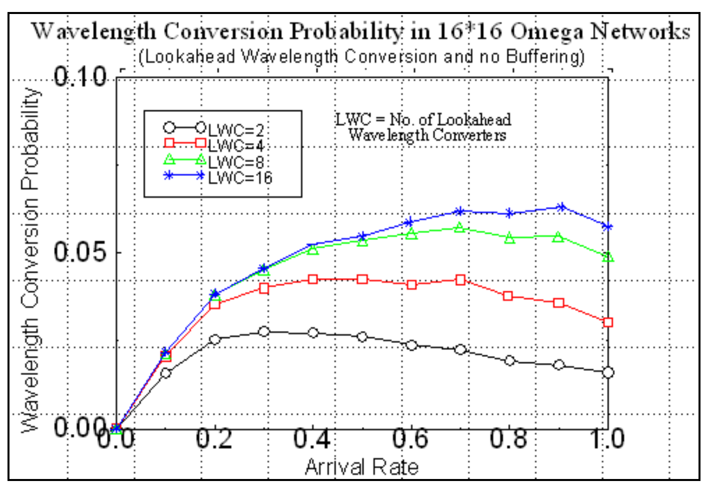


Fig. 10 Packet wavelength conversion probability versus arrival rate (lookahead wavelength conversion and no buffering)

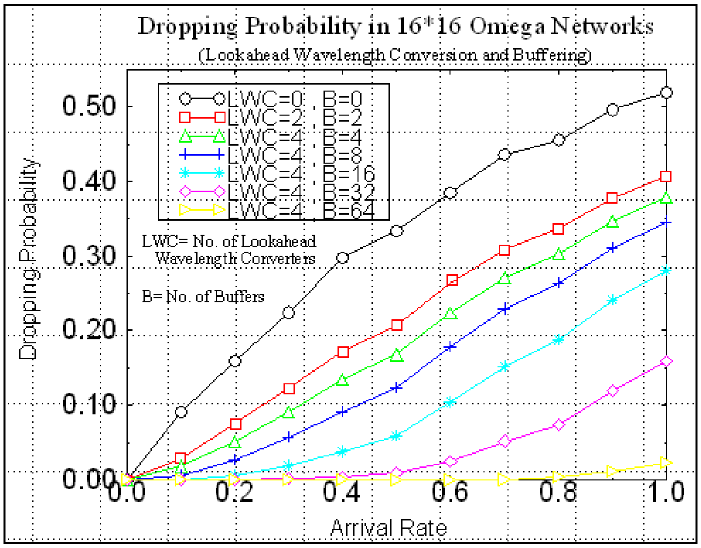


Fig. 11 Dropping probability versus arrival rate (lookahead wavelength conversion and buffering)

A network can achieve the most improvement when both lookahead wavelength conversion and buffering are used. Fig. 11 illustrates packet dropping probability versus arrival rate. As expected, the simulator shows the dropping probability will decrease when either the number of converters or the number of buffers is increased. Finally, Fig. 12 shows a comparison among different configurations. It shows that the performance is worst when using random wavelength conversion. This is due to the increase of unnecessary wavelength conversion. The network can improve by adding buffers. Also, the performance will be improved further by using lookahead wavelength conversion. Finally, best performance is achieved by using both buffering and lookahead wavelength conversion.

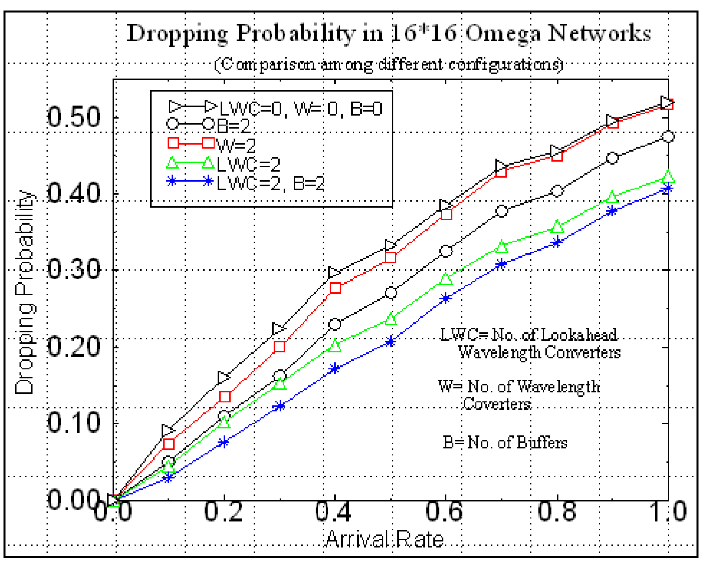


Fig. 12 Dropping probability versus arrival rate (comparison among different configurations)

CONCLUSION

Several architectures and algorithms have been introduced to alleviate the problem of internal blocking in WDM-based Omega networks. The first architecture is a collision-free Omega network based on buffering, the second one uses wavelength converters, in addition to buffering, and the last one considers lookahead wavelength conversion which eliminates unnecessary wavelength conversions. Also, the last algorithm is more appropriate for larger networks where arbitrary connections may be desired. It is shown that a few buffers and lookahead wavelength converters will considerably improve the system performance.

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Simulation Frameworks for Virtual Environments

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Abstract— *Scientific simulations have long been used to investigate the behaviour of systems governed by complex mathematical models. A wide range of simulation frameworks, visualisation methods and analysis tools have been developed to assist the development of these simulations. As computer hardware increases in power, computer games aim to create more immersive environments which now include physics engines which are similar in many ways to these scientific simulations. We identify different paradigms of simulation and how they can be incorporated into the development of more convincing and immersive virtual environments.*

Keywords: simulations; graphics; visualisation; artificial realities; model worlds.

1. Introduction

Simulation and modelling continue to play important roles underpinning the computational sciences [1], [2]. Simulations come in many different forms and degrees of complexity. These range from simple operational prediction models that might be made using nothing more sophisticated than a spreadsheet, through the use of general purpose modelling packages and environments to the development and use of custom hand-crafted and optimised simulation codes that run on supercomputers and other dedicated hardware systems.

Some simulations applications are very well known such as the problems of weather and climate prediction, both of which make use of many supercomputers around the world. Some are more mundane sounding but of high economic importance such as modelling air-flow and drag across new car and aircraft designs. Other applications are more esoteric and less well known such as the various simulation programs used to simulate the effects and associated phenomena of nuclear explosives.

An important idea that has been progressively up taken over the last fifteen years is Fox's concept of simulation on demand[3], whereby simulation programs are organised as services and can be accessed by client programs or indeed through a web interface. This approach can be used to make custom simulations more accessible to a wider user population. Complex simulation programs that are difficult to use can be wrapped up in a service-oriented software infrastructure such as web forms or even an immersive graphical interface. Another important related idea is Smarr's concept of steering computations[4] using advanced graphics or even totally immersive virtual reality systems. Support for this idea enables simulation users to home in on the parameter region of their problem that is of interest by facilitating a fast and close interaction between the user and the simulation running on supercomputing resources.

These ideas can be combined and it is possible to consider how the major classes or paradigms of simulation can make use of these notions and the tools and technologies that are already widely available. Many simulation categories have a strong need for good visualisation capabilities. A simulation is often best debugged (during development) and understood and interpreted (during production use) with the aid of a visual representation of the system configuration.

We discuss these simulation architectures and paradigms with respect to their relationship with virtual environments. Virtual environments aim to construct a convincing and immersive experience by simulating and visualising a virtual world. These environments require many different interacting systems to be modelled and simulated. These simulations are usually models of different paradigms, correctly understanding the paradigms of these different models is vital to determining how they can be incorporated together to model a single virtual environment.

In this paper we review some of the main simulation architectural ideas (Section 2) including: batch-driven simulations (Section 2.1); systems where the visualisation is driven by the simulation algorithm (Section 2.2) and systems where the simulation is driven by some intrinsic agent component (Section 2.3). We also discuss some cross-cutting issues for simulation developers such as: accuracy, repeatability, validation, complexity and performance.

Most simulations are built around some key idea or paradigm based on the way the system is represented and how they interact. Such paradigms include particle models (Section 4.1), field models (Section 4.2), event models (Section 4.3) and network models (Section 4.5).

2. Software Architectures

When implementing any simulation and visualisation environment, correct software design is vital to the re-usability and portability of the system. A simulation that is designed in an object-oriented and modular way can allow many components to be reused with little or no modification. Some simulations require a high-degree of integration between the simulation and the visualisation engine, which limits the modularity of the software. To create an virtual environment, many simulations of different natures may need to be combined to model separate parts of the environment. Correctly identifying the nature of each simulation can be invaluable when determining how to integrate them. Presented here are three common architectures used by scientific simulations: Batch-Driven, Simulation-Driven and Agent-Driven.

2.1 Batch-Driven Architecture

The **Batch-Driven** architecture is the simplest architecture to implement as it does not involve real-time observation or interaction. Multiple instances of the simulation are executed to provide a set of results about the simulation for certain parameters or parameter ranges. These calculated results can then be analysed to determine statistical properties of the model. Figure 1 shows the basic architecture of a Batch-Driven simulation, in which the analysis and visualisation are performed separately from the model.

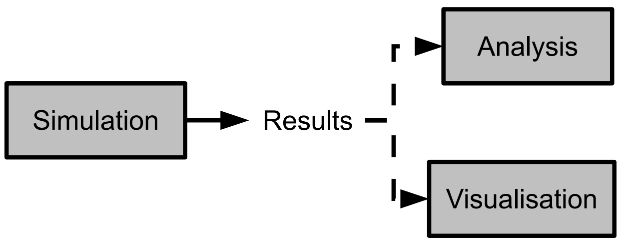

Fig. 1: Batch-Driven Architecture.

Simulations designed around the Batch-Driven Architecture are most commonly intended for gathering statistical data about a well developed simulation model. Each simulation instance is examined for some interesting phenomena or some property of each simulation is measured to provide this statistical data. Statistical gathering batches are often used to prove theories about simulations with certain configurations.

Batch-Driven simulations are also used to search a simulations parameter space. Batches of multiple simulation instances are executed with different parameters to search for a specific or many phenomena. These batches can be used to show or discover what conditions are necessary within the simulation for some event or phenomena to occur.

While Batch-Driven simulations would rarely be used within a virtual environment, they are applicable for generating random worlds, models etc. As virtual environments become more complex, generated rather than user-created worlds becomes an increasingly attractive feature. These worlds are usually created by randomly seeding a generator. This may be performed many times for different worlds or different areas. A process not dissimilar to a batch-driven simulation.

2.2 Simulation-Driven Architecture

A **Simulation-Driven** architecture simulation is driven by the model of the system. The entities within the simulation interact over time according to the governing model of the system. The simulation is decoupled from any visualisation engine or analysis process. The visualisation engine will simply display the entities of the system and not interact with them in any way.

The visualisation engine is responsible for displaying a list of entities and handling user input to change the method in which the entities are displayed. The visualisation engine will handle user input which controls only the visualisation. This input can be in the form of options displaying extra information about a simulation such as: grids, trails, and energy values.

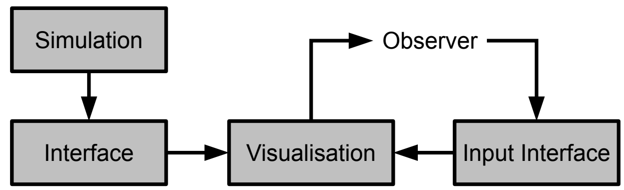

Fig. 2: Simulation-Driven Architecture.

In this architecture the operation of the simulation is the point of interest. The final result is defined by the interactions between the entities within it based on their starting configuration and other parameters of the simulation. This result may be in the form of a forecast where the simulation predicts the state of a system in the future given a starting configuration. Another possibility is that the visualisation of the simulation is examined by an observer to identify interesting events or phenomena inherent to the system. This architecture is useful for simulation systems such as Particle Dynamics (Section 4.1) and Field Equations (Section 4.2).

This architecture of simulation is very relevant for virtual environments. Immersive environments often contain many entities which are governed by models approximating the laws of physics. These entities should interact with each other according to the simulation.

2.3 Agent-Driven Architecture

In the **Agent-Driven** architecture the simulation itself serves a very different purpose, it provides a test-bed environment for intelligent agents. The simulation defines a set of rules and parameters about how the entities within it may act but allows the entities to change their state. What actions the entities perform to change their state is decided by an outside controlling agent. In this architecture the result of the simulation is dependent on the decisions made by the controlling agents.

These agents have a degree of control over one or more entities within the simulation and control them according to decisions they make based on the state of the simulation. The decisions made by the agents control how the entities act within the environment they exist in and how they interact with the other entities. These agents can be controlling human operators or artificial intelligence control programs. In Figure 3 the controlling agents are shown distinct from the simulation as they are most commonly separate programs that operate outside the simulation.

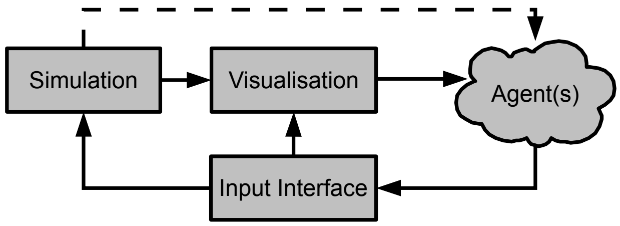

Fig. 3: Agent-Driven Architecture.

When a human operator is the controlling agent then the simulation is no longer merely visualised by the graphics engine, it is inextricably linked to it. The human operator processes the visualisation to determine the state of the simulation and from this information makes decisions about how to control its entities. This decision is then input into the simulation via an input interface (see Figure 3). Now the user input not only controls the method in which the simulation is visualised but also feeds back to control entities within the simulation.

When the controlling agents are artificial intelligence programs rather than human operators then the input into the simulation comes from an interface to the agent program. The information about the state of the simulation is sent to the agent directly without the need for the visualisation engine. The agent program can base its decision on the information it receives from the simulation and then makes its decision via the input interface.

The Agent-Driven architecture does not focus on the operation of the simulation but rather on the decisions made by the controlling agents. The simulation merely acts as a restricted testing environment to examine the performance of the agents. Each agent has a goal that it will try to achieve by controlling its entities within the limits of the simulation. Applications utilising this architecture include: most modern computer games, robot soccer simulators and robotic agent simulators (Section 4.6).

3. Simulation Considerations

For scientific simulations there are a number of important cross-cutting issues that affect all of the simulation paradigms that we have discussed. These include: accuracy and precision, repeatability, validation and verification, complexity and performance.

a) Simulation Accuracy and Precision: Numerical accuracy can vary from vitally important to relatively insignificant. The importance of accuracy depends on the workings and the purpose of the simulation. Accuracy is often most important for simulations built on the Batch-Driven and Simulation-Driven architectures. In these architectures the results produced by the simulation are vital, if the simulations are inaccurate then the results will be meaningless.

Simulations designed around the Agent-Driven architecture are often not required to be as accurate. Because these simulators are only providing an environment to test the controlling intelligent agents, the actual operation of the simulation is less important. However this is not always the case. In some simulators such as robot soccer systems (see Section 4.6, the models are designed to simulate a real-world environment where physical realism is important.

Virtual environments only require the simulation to be accurate enough to convince the user of their accuracy. Physics engines, graphics quality, artificial intelligence need only be accurate enough to create a convincing environment for the user. No statistical results or measurements are made other than providing the player with a enjoyable and immersive experience.

b) Simulation Repeatability: Repeatability is highly important for simulation and especially for those designed around the Batch- and Simulation-Driven architectures. Two executions of the same simulation that are initialised with the same configuration should both compute the same final state or result. This repeatability becomes increasingly hard to ensure when the simulation incorporates a degree of randomness within its computation. Simulations that incorporate randomness within experiments require a method of managing and repeating it. There is little point in discovering an interesting effect or phenomena if the experiment cannot be repeated.

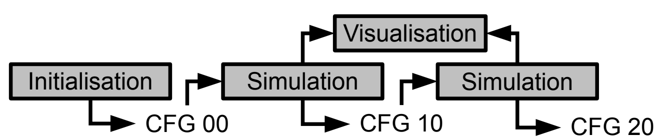

Fig. 4: Configuration-Chaining Architecture.

Figure: 4 shows a way of managing random-number generator (RNG) configurations. The simulator loads a configuration file which stores a starting configuration for the system as well as the RNG. The simulation then computes the interactions of the entities within the system over a period of time and saves a final configuration. The configuration of both the simulation and the RNG state is saved and can be reloaded at any time. It is important to ensure that executing the simulation for ten time-steps, twice should produce the same final configuration as one execution for twenty time-steps. This method of configuration managements allows the simulation configuration to be reloaded at the start or end of any execution.

c) Validation and Verification: Computer simulation is often used when testing the hypothesis that observed behaviour from a complex system truly emerges from a simple guessed microscopic model for the system's constituent parts. There may be strong microscopic physics arguments behind the choice of the component model or it may simply be a plausible guess. In these experiments it is important to conduct enough simulation runs over a properly varying set of choices of microscopic rules before it can be reasonably concluded that the model is consistent with the observations. One surprising feature of complex systems is the emergent universality of some sorts of complex behaviour independent of a wide variety of microscopic model parameters.

In some cases specific observations may be available for example of a physical system or a modelled crowd or a simulated network with which the simulation can be compared. This is not always the case however and a common driving force to use simulation is to help make a computational science link between the two conventional approaches of theoretical analysis and experimental observation. A well-posed simulation using well understood microscopic components can aid considerably in understanding the complex emergent macroscopic behaviour of a whole system.

d) Complexity: It is increasingly the case that we are targeting simulations of complex systems that may involve a hybrid of the techniques discussed in this article. This inevitably leads to a more complicated simulation architecture. An overly complicated simulation inevitably raises concerns about validation and verification and also reproducibility, and often performance. A clean simulation software architecture that can be validated as individual components helps allay these concerns.

Reuse-ability - which might deserve to be a criteria in its own right - is closely tied to complexity and performance. Generally we hope to have very modular software that can be unit tested and verified. This additional effort is amortized over greater use if the module is widely applicable. Unfortunately achieving performance often involves compromises and tradeoffs that reduce reuse and raise code complexity.

Virtual environments often require several simulations to be linked together to model all the interactions within the environment. This results in a highly complex simulation, but correctly identifying how to couple simulations within this environment can allow these complex systems to be simulated.

e) Performance: The performance of a computational simulation becomes increasingly important as the size and complexity of the simulation increases. Simulations must complete in a reasonable length of time for their results to be useful. In many cases this requires simulations to be structured differently in order to operate on supercomputers [**?**], computational grids or clouds [5], special hardware such as Graphical Processing Units (GPUs) [6] or even Field-Programmable Gate Arrays (FPGAs) [**?**] or other Application-Specific Integrated Circuits (ASICs).

Other simulations environments (especially game environments) require the simulations to run in real-time to be considered successful. Such performance constraints often require a trade-off against other considerations such as complexity and accuracy. Less accurate and complex simulations must often be used simply to allow them to be computed in real-time. Recently virtual environments have started making use of parallel accelerators to compute the simulations, notably the NVIDIA PhysX engine is used by many games to model the physical interactions of entities within the game.

4. Simulation Paradigms

There is considerable variety in the paradigms of simulations used for scientific research. The method of visualisation for each of these simulation paradigms is equally variable. This section presents a number of simulation paradigms and discusses the methods and libraries used to visualise them.

4.1 ParticleModels

Particle dynamics simulations model the motion and behaviour of particles or objects in space. A particle is considered to be a single point mass in space with a position and velocity [7], [8]. A particle's motion is normally constrained by Newton's Laws of Motion [9] and by some potential equation or some external force acting upon all particles. The potential equation of a simulation describes an attractive/repulsive force between a pair or particles. In each simulation, particles have a set of properties such as radius, charge or spin that also define their state in addition to their position, velocity and mass.

Within a particle simulation the two main issues are: their relative motion and attraction due to the potential between them; and their behaviour when the particles collide. These collisions can be approximated by potentials such as the Leonard-Jones or modelled as some event-driven collisions where there is no force model applied to compute accelerations, but simple point reflections are used to model hard spheres interacting with each other and with hard boundary walls [8].

This type of model is normally used within the Batch- and Simulation-Driven architectures which attempt to gain insight into emergent behaviour of many interacting particles. They also have applications in computer games as the physics engines in modern games are often performing calculations similar to these particle systems. Objects moving and interacting within some environment have to deal with similar collision events and move under the effect of some potential force (gravity).

4.2 Field Models

Field equations simulators model the behaviour of microscopic atoms over a discrete macroscopic cell. The modelled

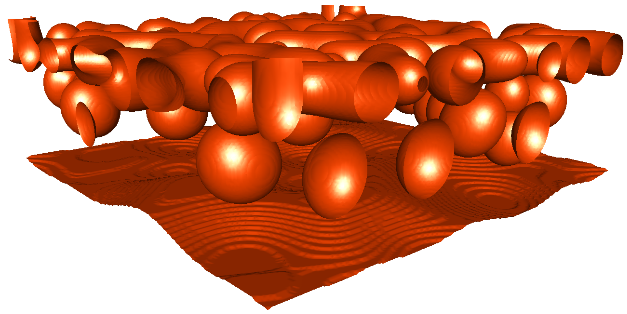

Fig. 5: A three-dimensional visualisation of droplets near a meniscus in a simulation of the Cahn-Hilliard-Cook field equation.

system is split into discrete cells and the state of each cell is defined by a set of properties. Every cell interacts with the cells in the area or volume surrounding it and changes its state according to the properties of the surrounding cells and field equations of the simulation. Visualising these simulations involves displaying some property or combination of properties that define each cell in a graphical way.

These field equations can be formulated for a wide-range of physical and mathematical systems such as binary alloys, superconductors, predator-prey systems and fluid-dynamics. These are usually simulated in either a Simulation- or Batch-Driven architecture to investigate the behaviour of the models. However, they can be used at a rough level to model the behaviour of systems such as fluids to produce a more convincing and immersive environment.

Figure 5 is in fact a visualisation of a three-dimensional Cahn-Hilliard-Cook field equation simulation. The Cahn-Hilliard Cook equation models phase separation in a binary fluid. Started from a random uniform configuration, surface tension effects drive the field cells to gradually coalesce and merge into separate domains. This domain separation forms the emergent camouflage pattern seen in the visualisation. This equation can be used to model the phase separation in cooling alloys to discover the optimum cooling process for forming real-world metals [10].

4.3 Event Models

The event driven paradigm is commonly used in cases where some aggregate or emergent behaviour arises from the collective interactions of many discrete participants. Events might be arrivals of data packets in a network, or transactions in a management situation, or encounters and conflicts in a defense scenario.

Many physical system can be modelled by the discrete movements of atoms or cells in a system. One famous discrete event model is Conway's game of life[11], which is a cellular automaton. Each spatial cell has very simple microscopic rules governing its temporal behaviour but some very complex and unexpected patterns emerge from the overall collective. Monte Carlo lattice models also fit into this paradigm. Models such as the Ising model[12], Potts model[13], Diffusion-limited aggregation models[14], Heisenberg model and clock models are essentially microscopic automata that interact with their local neighbouring cell and through a stochastic dynamical scheme or pseudo time imposed upon them they give rise to complex phenomena such as phase transitions[15].

Within virtual environments this idea of events-driven systems is important. However, rather than randomly occurring events they are normally tied to an agent or user performing some action which triggers the event. These events are also tied to the rest of the environment and the models that govern them.

Many systems that have a continuous time scale can also be modelled by specific events that occur at arbitrary times and which can be modelled through queues. These problems are notoriously difficult to fully distribute or parallelise although a distributed simulation can manage separate clocks or time queues that can be rolled back or time-warped to obtain synchronicity across all participating computers[16].

4.4 Cross-over Paradigms

A particular simulation to model a coupled set of phenomena may need to make use of a cross-over or hybrid approach. In some cases a hybrid model might link together multiple components of an overall model, each of which fit the same basic paradigm. An example of this would be simulations of the global weather or climate[17]. Typically models run by national bodies such as the UK Meteorological Office will comprise separate field-based models for the atmosphere and the ocean. These may be completely separate codes that exchange data[18] or they may be a single integrated simulation program. Simulation variables such as temperature and pressure fields are separately integrated in the ocean and atmosphere - likely using different model meshes and resolutions, but are coupled together to ensure the correct physical boundary conditions are available to the separate model components.

Other predictive simulations such as those used to predict extractive yields in oil reservoir systems are typically also a hybrid of a field model and discrete event information[19]. Particle methods have widespread uses for straightforward systems such as planetary dynamics or molecular dynamics [20], [21]. For some systems such as simulating very low density fluid flow around orbiter spacecraft re-entry for example, a hybrid model of particles and field equations has to be employed[22].

4.5 Network Models

Many interesting physical, technological and social systems can be characterised as a network or graph [23], [24], [25]. While some networks are small enough that they can be directly visualised and various direct counting methods can be used to study their properties, many systems are too big for such easy assimilation. It is however possible to develop a number of useful metrics or quantifiable characteristics that can be used to categorise network systems.

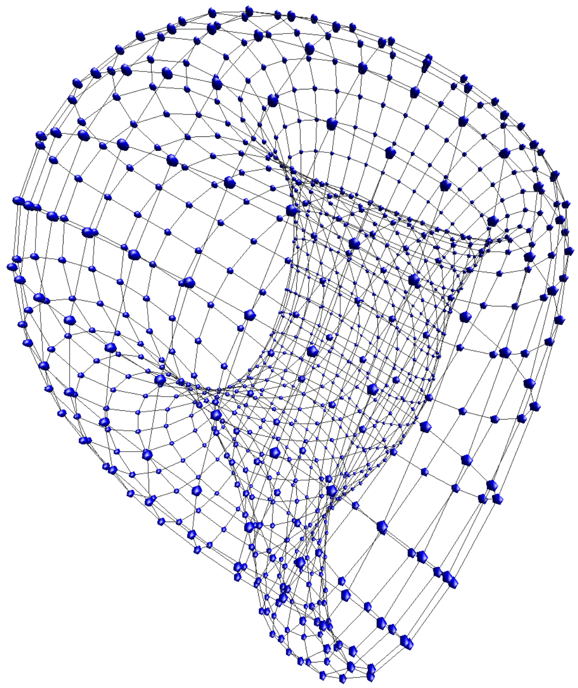

Fig. 6: A regular 30×30 lattice with periodic boundaries visualised with UbiGraph.

Networks have a wide range of applications within virtual environments. They can be used to represent connections between models, navigation pathways, communication between agents, interactions between cells in regular or irregular lattices, etc. Correct use of scientific network algorithms and technologies can be of great importance for the use of networks in virtual environments.

4.6 Simulations for Computer Animation

Visual immersion is an important aspect of real-world simulation as these simulations are typically run to either involve a human interactively, or represent the actions of intelligent agents to a human audience; the value here is then psycho-visual. An excellent example of this type of application is in modern film production - the MASSIVE software has been used to simulate huge numbers of interacting intelligent agents as actors in the Lord of the Rings trilogy [26]. Visual immersion can be aided by creating accurate models of real-world entities to-scale. This is best done using three dimensional modelling software, which can be aided by using real blueprints and photographs taken from different angles as a guide to scale and shape the model. We can then use the photos to accurately texture-map the model [27]. Our attempts at this process made use of the

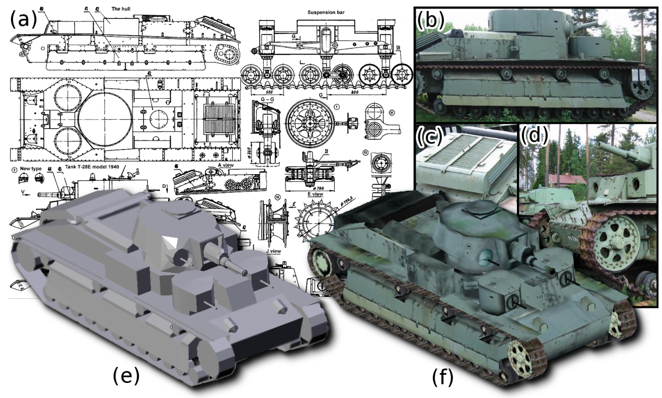

Fig. 7: A scale model of a vehicle is created for simulation from blueprints (a) and photographs (b-d), given a 3D mesh convex hull using a modelling tool (e), and rendered using a texture map made from photographs (f).

powerful, and freely available tools Blender3D and the GNU Image Manipulation Program (the GIMP) are illustrated in Figure 7. We have also made use of audio capture and manipulation software ffmpeg and Audacity to record and reproduce vehicle sound effects using a realistic 3D acoustic model.

Fig. 8: User-Assisted Dynamic Landscape Generation for games or simulations.

Most realistic terrain in 3D simulations is now generated by fractal modellers, which provide pseudo-realistic terrain but allow very little design control to scenario designers. The terrain illustrated in Figure 8, is however based on a novel method, in that it is visually created in 3D by hand, in real time, from within the simulation itself, with a variety of WYSIWYG ("What you see is what you get") tools that we have built into the simulator. This work is based on the ETM2 (Editable Terrain Manager) for the OGRE graphics library [28], and gives us the added advantage that we can allow simulated elements in our event driven models to deform the landscape during simulation according to the physics model. We have made use of algorithms such as ROAMing (Real-time Optimally Adapting Meshes) terrain [29] to optimally display the environment.

5. Summary and Conclusions

We have discussed a number of simulation paradigms and example applications. We have reviewed some of the complex systems issues facing simulation developers and users and have reviewed a number of the software tools and technologies that can be employed for a successful computational science experiment.

As we have shown, the overall architecture can be driven from the simulation or agent perspective. A better understanding of a simulation's architecture can help to identify and avoid design-breaking features which would otherwise reduce the software's portability and re-usability. Visualisation is important in any simulation but plays different roles depending on the software goals.

Correctly understanding these simulation architectures can allow games developers to reap the benefits of years of scientific simulations experience. Understanding and identifying simulation paradigms can allow different models to be integrated together to form more convincing virtual environments resulting in a more immersive experience for the user.

It seems likely that a service-oriented approach to running simulations may become more prevalent but there will likely also remain a strong need for advanced custom crafted simulations as a tool for investigating state of the art complex systems. Furthermore these ideas will remain closely linked to game engine architectures.

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A Case Study on Software as Service Approach to Model-Driven Development

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Abstract - *Business demands have been increasing for the past years due to the ever growing need to sustain competitive advantage. As a consequence, business processes require more agile and flexible information technology (IT) assets that can be easily and swiftly customized to meet their changing demands. Such challenges are continuously faced by IT professionals as they are required to develop innovative solutions to cope with business-related problems. In a dynamic business environment, innovation is a way to create value and, as this paper will show, it might result from the combination of cloud computing and model-driven development (MDD) foundations. As a consequence of bringing both theories together we present a software development environment that is hosted 'in the cloud'. By combining Web services and MDD tools we show that it is possible to create a software development service (SDS) which interprets application models, generating large parts of the application, ready to be deployed into any compatible application server.*

Keywords: MDA, cloud computing, domain models, UML, software as service, service-oriented architecture.

1 Introduction

As presented in [1], cloud computing is the future generation of computing. Although it is currently in its early development stages it is evolving rapidly.

Cloud computing is a new paradigm for deploying business services that is based on pervasive and on-demand access to Web-based applications. Such a paradigm can significantly impact the nature, complexity, and scale of business models, leveraging not only an organization's core business but also enabling new ways to do business itself. Some think it will radically change the way we think our internal processes and that it will change business and personal computing, moving all processing power to the cloud, going back to a few decades where one needed only a 'dumb' terminal to connect to the mainframe and gain access to processing power. It is now possible to see many applications available as services (such as with salesforce.com), proving that the technological evolution is at our doorstep and is evolving on a daily basis [13].

We have witnessed revolutionary results along this decade regarding the implications of bringing traditional applications into the 'cloud'. The increasing adoption of service-oriented solutions, supported by both technical and business communities are forcing a shift to the manner we should think software development.

From a technical perspective, service-oriented solutions can now be seen as an innovative approach to software development, where services provide reusable functionality with well-defined interfaces to create, from a pre-determined input, a software product as output [17]. The idea is to rethink the way software is built, by understanding that developers should no longer need to install different tools to create a domain-specific software application when one could only access an online software development service to have it done.

Such concept is called 'Dev 2.0' where the development environment is no longer needed to be in the developer's machine, and instead it can be used, executed, and consumed remotely as an iterative service [2]. Viewing such hosted development platforms from the perspective of traditional model-driven architecture (MDA), these Internet services can now be regarded as the future of the development industry [14].

Challenges faced and presented in this paper are related to proving that such approach is viable and how anyone should put several technologies to work together in order to reach this goal. Throughout the paper we demonstrate, step-by-step, our accomplishments from the early stages of modeling through the download of the prebuilt system. Essentially, this work targeted the creation of a development environment containing a MDD [6] service, running on top of a service-oriented architecture [8], capable of accepting business models as input and producing specific sets of software artifacts as output.

2 Motivation

Organizations have been reviewing their IT investments, seeking new integration models and innovative use of IT assets in order to come up with better strategies for their business domain problems. They seek to create services that must be performed in an effective and efficient way, in order to sustain their competitive advantages, creating value in the process.

The business world has reached a new competition age. Due to increasing domain-specific demands and ever growing knowledge consumption, investment in Information Technology (IT) is more and more necessary in order to keep up with time-to-market needs and competitive advantages.

What we now see is an increasing need for business integration, and such integration can only be achieved through strategic business alignment with IT services. At the same time, the organization must be aware that IT investments must be controlled and prioritized, as the word of order is to create value while relentlessly reducing costs. In fact, IT elements have become important business assets that not only contribute to achieving business goals but also revolutionize the organization as a whole [11].

Information technology is the single largest capital expense in many organizations that, when correctly managed, allows them to efficiently achieve business goals. Consequently, in seeking to achieve that efficiency, we present you the following questions:

1. How should we use SOA and MDA to improve the way we develop software?
2. How should we target efforts to use those theories to create a development environment capable of delivering what we are proposing?

3 The Software Development Service

The pursuit of an automatic and/or autonomic development approach, which efficiently gives the developer tools to speed up software development, allowing project managers to gain more insight on development schedule are, more and more, in the minds and hearts of any software organization [11]. Few techniques are available to help developers accomplish this [12]; and as we felt we had a chance of doing something to contribute, we present a case study which shows how software as services techniques can be used in conjunction with a model-driven development paradigm to create a deployable software development environment. The goal was to create a software development service, hosted in the cloud, which receives as input a specific analysis artifact and outputs full-fledged deployable Web-based software.

Let us clarify the proposal with a conceptual example: suppose we want to create a Web-based software with a set of requirements R = {r1, r2, r3, …, rn}. Each and every requirement is modeled in a W3C [15] compliant UML case tool. This model is exported and stored in a XMI file, and this file can now be used as input to a development service. The output is a package containing a Java/JEE Web application, a relational database script and configuration files ready to be deployed and executed. Any developer can register in to use the service and store all its business models in the portal. With only one click one can (re)generate the Web-based system and (re)deploy it at any time. At this point, we already have a MDA enabled development service, running in the cloud at www.mda4eclipse.com.br [3].

In order to prove that our approach would perform as previously detailed we decided to use it in a real-life project. This concept proof was based on a project conducted at the Brazilian Navy Bureau for Integrated Logistic Support (NALIM).

Our final goal was to understand whether it would be profitable to make this environment available to other project sites, hoping they would actually gain in development time and/or cost, by using business models already registered and by sharing business models themselves.

Step-by-step, the case was conducted on a set of requirements as presented in Figure 1. Attributes were omitted for the sake of confidentiality.

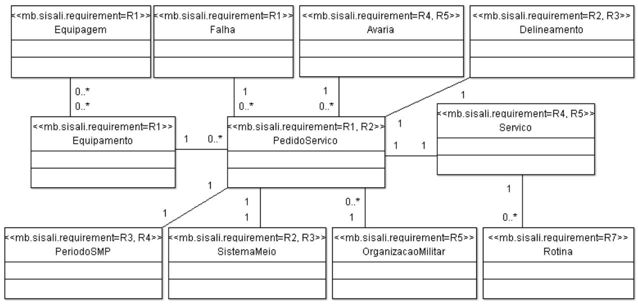

Fig. 1. The UML diagram used in the case study.

Using ArgoUML [9] as modelling tool, we generate the following XMI file. From it we extract all the information needed to create the system with the development service (Listing 1).

```
<?xml version="1.0" encoding="UTF-8" ?>
<XMI xmi.version="1.2">
<XMI.header>...</XMI.header>
<XMI.content>
  <UML:Model xmi.id="00B06" name="ModeloSemNome" >
    <UML:Namespace.ownedElement>
      <UML:Class xmi.id="0000000000000CAE"
       name="Falha" visibility="public" >
       ...</UML:Class>
      <UML:Class xmi.id="00CAF" name="PedidoServico"
       visibility="public" >...</UML:Class>
      <UML:Class xmi.id="00CB1" name="Delineamento"
       visibility="public" >...</UML:Class>
      <UML:Class xmi.id="00CB2" name="Rotina"
       visibility="public" >...</UML:Class>
      <UML:Class xmi.id="00CB3" name="Avaria"
       visibility="public" >...</UML:Class>
      <UML:Class xmi.id="00CB4" name="Servico"
     visibility="public" >...</UML:Class>
      <UML:Class xmi.id="00CB5"
```

```
    name="OrganizacaoMilitar"  visibility="public" >
     ...</UML:Class>
      <UML:Class xmi.id="00CB6" name="SistemaMeio"
       visibility="public" >...</UML:Class>
      <UML:Class xmi.id="00CB7" name="PeriodoSMP"
       visibility="public" >...</UML:Class>
      <UML:Class xmi.id="00CB8" name="Equipamento"
       visibility="public" >...</UML:Class>
      <UML:Class xmi.id="00CB9" name="Equipagem"
       visibility="public">...</UML:Class>
    </UML:Namespace.ownedElement>
     ...
  </UML:Model>
</XMI.content>
</XMI>
```

Listing. 1. Extracted from the system business model (XMI file). (Identifications and attributes were altered to fit this article).

After that, the developer accesses the online service at www.mda4eclipse.com.br, registers (Figure 2) as an user and uploads the XMI file (Figure 3). The MDA service reads the file and generates the system source code.

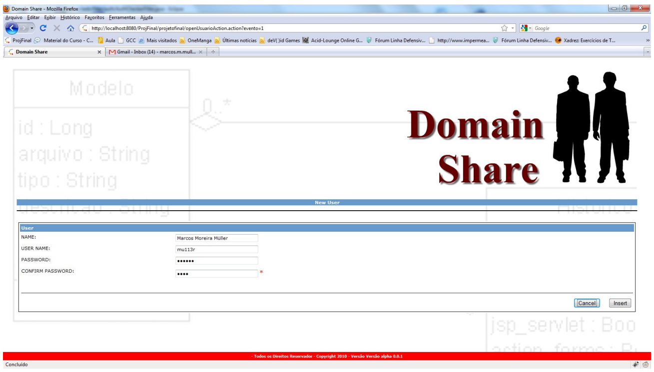

Fig. 2. Registering a user.

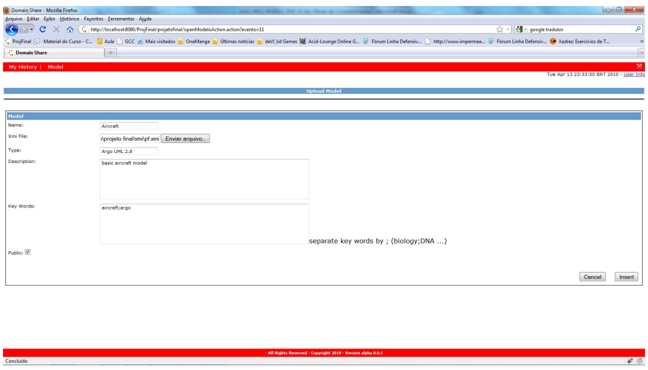

Fig. 3. Uploading a XMI file.

The package is now available to download and can be directly deployed in any application server (we used JBoss [16] as the application server). Figure 4 shows the download area.

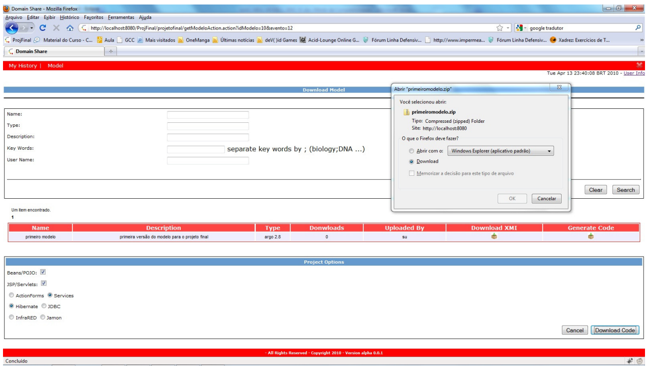

Fig. 4. Downloading the generated application.

Finally, the developer can benefit from a historical area were one can keep track of all versions and/or models ever uploaded into one's account. Figure 5 shows the user history.

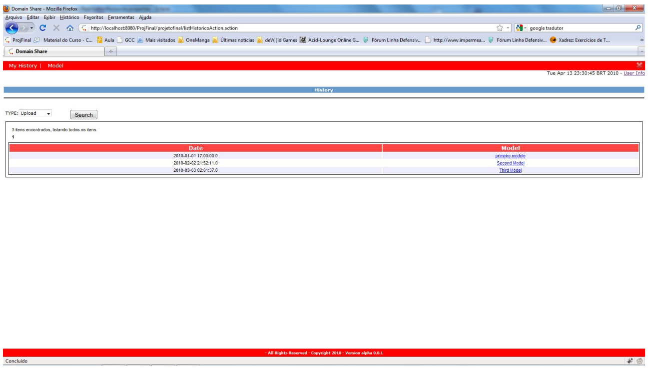

Fig. 5. Historical listing.

4 Preliminary Analysis and Results

In the light of the continuous evolution of a new development paradigm, we have presented in this paper a case study detailing the concept of a service-oriented development approach. The technique worked as planned and showed that the "Dev 2.0" paradigm can be explored and has a long road ahead for research and new solutions.

The SDS approach proved to be dynamic, effective, powerful, and fairly simple in concept, and highlighted many important issues as a consequence of using the SDS environment:

- Development services can optimize the way software is built.
- Reuse is one of the driving forces in deploying development services.
- Organizations do not need to worry about building complex development environments; once it is

available online they can use and benefit from ever coming updates.
- Project evolution is recorded through model versioning.
- Loose coupling between components is achieved, resulting into a flexible, scalable, and adjustable environment, enabling an easier replacement of services.
- It can help in cutting project budget and schedule [8, 18], as the development result is more reliable and is controlled by the SDS.

Finally, services can be built on top of an enterprise bus, where it can orchestrate a series of different services, each one producing software products with different attributes such as different programming style, architecture, languages, layouts, and patterns.

5 Opportunities and Future Work

Many opportunities can be drawn from this approach. One that is already in its early stages of research is the Model Sharing Service - MSS. It consists of a set of Web services, allocated to different project sites, aimed at storing and exposing different model artefacts to other project sites. The sharing process is straightforward: the project architect or the system analyst creates project analysis and design models, stores them in a local database and exposes them through the project's Web service. The Web service publishes the models via WSDL files, running in a Web application server [16], and any ongoing software project wishing to acquire related, pre-built business models, connects to the Web service and downloads it.

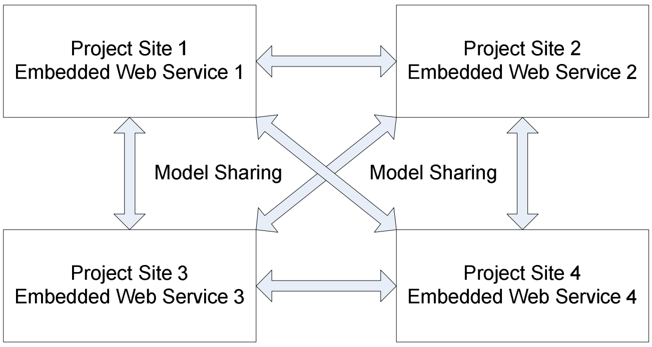

Fig. 6. Model Sharing concept.

From a "software as services" perspective, sharing is enabled in upload time. The user tags the models one wants to make public and it is visible in the global portal search.

A second related opportunity is the ability to identify change impact caused by software artefacts updates. For example, to analyse impact caused by changes in any functional requirement, we built a change impact analysis environment capable of acquiring source code information and creating a visual change impact graph. By using this approach we were able to significantly cut development time by means of severely reducing unnecessary (re)work and resource allocation. This is also an ongoing research and results have been submitted for evaluation to the 31st International Conference on Information Systems – ICIS'10 - http://icis2010.aisnet.org/.

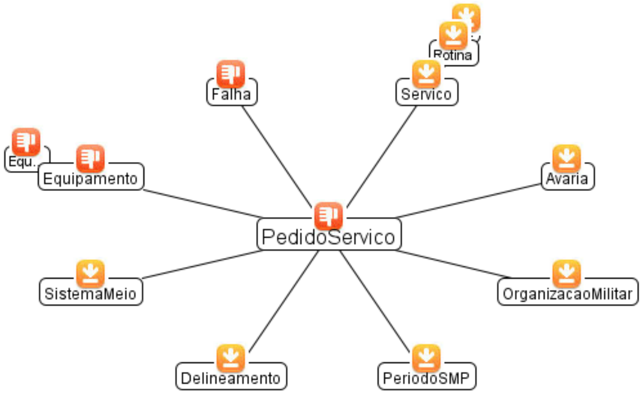

Fig. 7. Low level change impact analysis view. Classes associated with "PedidoServico" (ServiceRequest) are directly and indirectly affected by requirement changes.

6 Conclusions

This paper presented preliminary results of a software as service development approach. We showed is it possible to create a Web-based model-driven development service capable of accepting model artifacts (or more specifically XMI files) and to produce a fully-fledged Web-based application. To support such idea, we built and hosted the MDA service at the www.mda4eclipse.com.br portal. There, the user can register and manipulate many projects models one needs, allowing them to be shared with other users.

By using this approach we were able to contribute to the "Dev 2.0" efforts, proving that development services are possible and can produce effective results.

In spite of its being a preliminary analysis we can foresee more benefits to come, such as change impact analysis services and risk impact analysis services. We also intend, as soon as possible, to improve this approach, to keep contributing with the evolution of this new development paradigm.

Finally, we look forward to creating a fully-fledged environment where we can control the whole application development process, from the early stages of domain modeling, going through the code generation stage, and finally enabling a management environment where stakeholders can benefit from several analysis mechanisms such as the one presented in this paper.

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Applying Personalization Concept to an Online Newspaper

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Abstract - *With the rapid growth of the size and use of World Wide Web, users are provided with massive amount of information in which only some are of interest to them. Therefore, by applying the personalization concept, the information can be customized to the needs of different users. Personalization offers different users with a more tailored information content, structure and presentation. This paper presents a prototype of an online newspaper in which the content had been personalized using a profile-based personalization. The result of the personalized content will ease the reader in getting the section of news of their preference and therefore improve their user experience.*

1 Introduction

Nowadays, the growth of information on the web has becoming very rapid and causing information overload. In a typical online newspaper, there are several sub-sections of news such as nation, world, sports, education and entertainment. Under each sections, there are even more articles to be viewed. Once the users launch an online newspaper, users would be bombarded with a page full of snippet of news as well as options to navigate sub-sections of news. In order to look for an interesting article according to individual preferences, users may need to navigate through the whole website to look for it. In [1], it is stated that user's most normal action in finding an interesting article is to scroll through that particular article and look at it in detail after finding it interesting. This kind of action is time consuming and may cause unsatisfied user experience of the online newspapers. Furthermore, [2] also stated that newspapers are not only a mean to get personally interesting articles but also a way to get information that users are not explicitly looking for. There is also a problem in which visits to online newspapers are usually not related to previous visits. This is because the content of online newspapers changes on a daily basis.

In order to provide users with improved navigational experience, personalization technique can be applied to an online newspaper. This paper discusses the application of personalization concept to an online newspaper so that users can easily view section of news or columns of their interest only instead of navigating through the whole website. Using this technique, the information regarding users' preferences is obtained by explicitly asking the users to enter information about their preference of certain category of news. Based on the information given, the articles of news that are relevant or might appear interesting only will be showed as links that will then redirect the users to full articles.

This work is based on a local newspaper that does not have any personalization features. This work is done on an existing online newspaper in which the content has not been personalized. The implemented personalization features suggested are part of the developed prototype while the overall user interface design is still maintained.

2 Content Personalization

The concept of personalization has become a necessity in which only information that is needed or desired by the users will be given. Personalization is a concept which has been rigorously applied in many area of research such as web interaction, e-commerce and education. For web browsing, personalized content displayed to users can help to control aimless surfing activity [3] as well as to reduce the amount of information returned to users [4]. It is also stated in [5] that with the rapid increase of information available online, personalization has become a key component of Web applications to tailor information content, structure and presentation to the needs of particular user or a group of users. No matter what the domain of an application might be personalization can be generated based on some common techniques. Among popular techniques which have been used are the ones based on individual or group profile, behavior and collaborative filtering. The first technique often requires users to provide their demographic information as well as some preferences explicitly beforehand. Based on the information given by users, the application will match the information to the products or services provided. Weiß et al. [6] for example, used this technique to filter appropriate multimedia content for user viewing. This is the technique that has been chosen by the authors to filter the news articles outputted to the users. Further elaboration on the employment of the technique is further discussed in 'Personalize Online Newspaper' section.

According to a research done by the information system department of New Jersey Institute of Technology [7], most of the information service websites for example online newspapers only contain low degree of personalization compare to e-commerce or financial services websites. Example of information services includes http://www.cnn.com and http://www.nytimes.com. This is because the product of information service website is content. Compare to other types of websites, the visits to information service website is seldom related to its previous visits. This is mainly because the content of news website is changing daily. Therefore, it is more difficult for personalization concept to be applied. There is also an issue related to social point of view in which users not only want to read articles of their preferences but also want to look for information which cannot be found explicitly. Furthermore, interests will change eventually and sometimes, articles that users are interested in might be affected by its importance. For example, a user might be interested in entertainment news only. However, during the period when the tsunami occurred, the user may be very interested in following these news. On the other hand, there are also technical issues in newspaper personalization. According to Kamba and Bharat [2], the biggest technical problems are measuring the user's interest with reasonable accuracy without too much effort on the part of the user and to relate the presentation to the users' interest.

3 Related Work

Krakatoa Chronicle [2] is a personalized online newspaper. When this newspaper was being design, there are a few assumptions being made. Firstly, people are accustomed to the layout of printed newspapers. Secondly, keywords and weights can be used to describe the features in articles and user's interest. Another important assumption made is that multiple views were needed because not every times people want the newspaper to be personalized. The architecture of Krakatoa Chronicle was said to be different from other www based newspaper. The server side will be responsible in doing the user authentication, managing user's profiles, and collecting and processing the articles. To manage the daily articles, the articles will first be collected from several sites for the purpose of analyzing the content and re-formatting. Another important role at the server end is to manage the user profiles. In each profile, there will be keywords and each has its own weight. The weight will represent user's interest on the keyword, the higher it is, the more interested the user on that particular keyword. Another important role is to compute each article's weight. The server will compute each article's weight for a specific user based on how well the article's document vector and the user's profile match. Meanwhile, the client is responsible of managing user interaction with articles through operation like scrolling or peeking. All these operations provide the feedbacks of user's interests on a particular article. These operations will then be sent to server side and the user profile will be changed. Krakatoa Chronicle also allowed layout control. When the clients get the article's weight from the server side, the newspaper layout will be modified based on the information received.

Publico On-Line as discussed in [8] is another example of personalized online newspaper. The personalization of Publico On-Line is being done based on general personalization concept. The system will first perform the following tasks, noise filtering (removing irrelevant data), session's identification and storage in a repository. After that, data mining modeling operation is used for pattern analyzing purpose. One of the techniques of this operation is to discover frequent itemsets and its relationships. Frequent itemsets referred to groups of items which occurred frequently. For example, articles which had been reviewed several times. Another technique used is clusters identification to identify group of users with significant preferences.

In the experiment done by Sakagami and Kamba [1], a few methods had been tried in order for the system to learn user preferences. These experiments are based on an online personalized newspaper, ANATAGONOMY [1] which has similar architecture as Krakatoa Chronicle mentioned in [2]. It can be done explicitly or implicitly. Explicit method will ask the user to specify their interests using keywords or topics. However, this method is not effective enough as human interests will change as time passes and it required too many efforts on the user part. For implicit method, an experiment had been done based on the time spent in reading the news. It is based on the concept that user tend to spend more time on the articles of his/her preference. However, it is not accurate enough as the time when user leaves the terminal for other purposes were also being counted. Another experiment also being done by asking users to read the articles on ANATAGONOMY and rank the articles from A-E (explicit feedback). On the other hand, the system will automatically change article's score based on the bonus points which is set when user's perform operations such as scrolling and enlarging (implicit feedback). Later on, the relationship of the scores and the user operations is analyzed. The conclusion of this experiment is that personalization will be most effective by taking both explicit and implicit feedback into account. The method of learning user's preferences from their operations on an article is also appeared to be effective.

4 Methodology

To apply personalization concept for the online newspaper, a content-based profiling personalization technique is used. An explicit user profiling method is used to retrieve the user data. The user data is entered by the users when they first log in into the developed personalized online newspaper. Figure 1 shows the items needed to create a user profile. It includes section on the user preferences and keywords. These 2 sections formed the core of the personalize information presented to the users. Figure 2 shows the portion of the SQL query used to retrieve the relevant news article based on the profile.

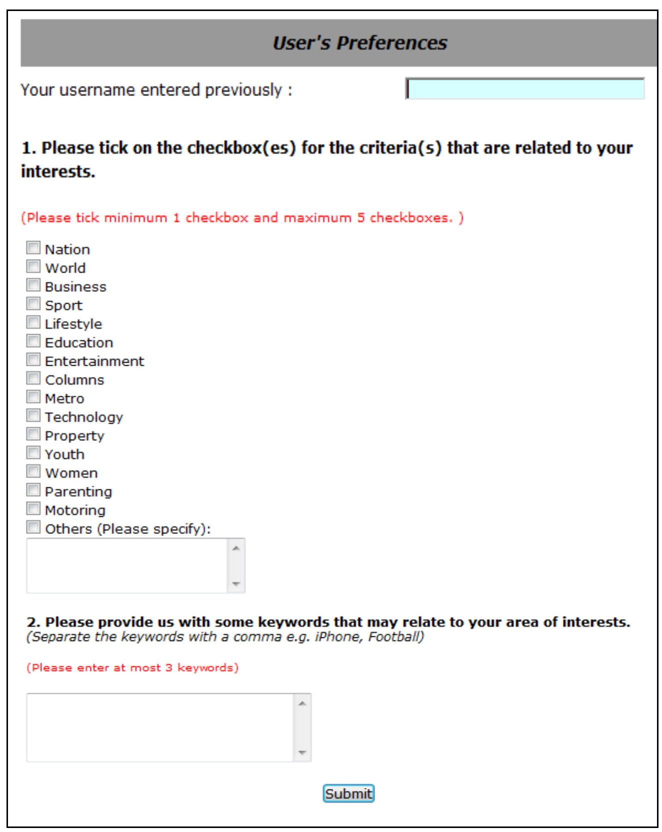
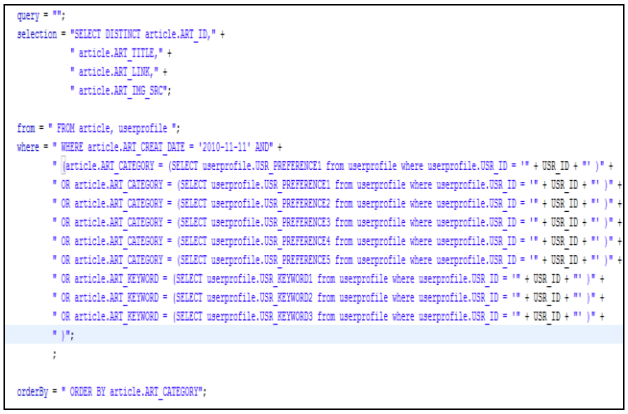


Figure 1: Capturing User Profile



```
query = "";
selection = "SELECT DISTINCT article.ART_ID," +
            " article.ART_TITLE," +
            " article.ART_LINK," +
            " article.ART_IMG_SRC";

from = " FROM article, userprofile ";
where = " WHERE article.ART_CREAT_DATE = '2010-11-11' AND" +
        " (article.ART_CATEGORY = (SELECT userprofile.USR_PREFERENCE1 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " OR article.ART_CATEGORY = (SELECT userprofile.USR_PREFERENCE1 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " OR article.ART_CATEGORY = (SELECT userprofile.USR_PREFERENCE2 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " OR article.ART_CATEGORY = (SELECT userprofile.USR_PREFERENCE3 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " OR article.ART_CATEGORY = (SELECT userprofile.USR_PREFERENCE4 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " OR article.ART_CATEGORY = (SELECT userprofile.USR_PREFERENCE5 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " OR article.ART_KEYWORD = (SELECT userprofile.USR_KEYWORD1 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " OR article.ART_KEYWORD = (SELECT userprofile.USR_KEYWORD2 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " OR article.ART_KEYWORD = (SELECT userprofile.USR_KEYWORD3 from userprofile where userprofile.USR_ID = '" + USR_ID + "' )" +
        " )";
        ;

orderBy = " ORDER BY article.ART_CATEGORY";
```

Figure 2: SQL query to retrieve news article based on the profile.

User profiling is a method used for personalization as it can clearly identify the web user and provides an easier way to collect information about the web users. User profile stores information such as users' personal detail and preferences. This information will then be stored in database. Each time the web users log in to the site, the information will be retrieved and analyzed in order to come up with recommendation of articles which will match their interests.

As for the tools, HTML, Cascading Style Sheet (CSS), Javascript, JavaServer Pages (JSP) are used to develop the website and Java is used to do the analysis of users' profiles.

5 System Architecture And Prototype

Figure 3 shows the system architecture. There are basically three main components in the system architecture which are the web interface, database and system interface. The web interface is the interface which allows website users to sign up or login, edit user's profile and read articles. The system interface is the core of this personalized online newspaper. It is the place where predictions of user's favorite articles being made based on user profiling personalization strategy. Meanwhile, the database stores the information such as articles and user profiles.

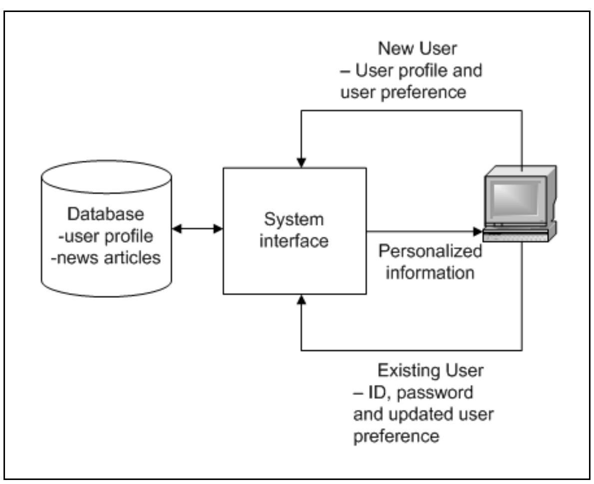


Figure 3: System Architecture

Figure 4 shows the system flowchart of the prototype. Basically a new user needs to register in order for the system to capture and store the user profile while current users are only required to login.

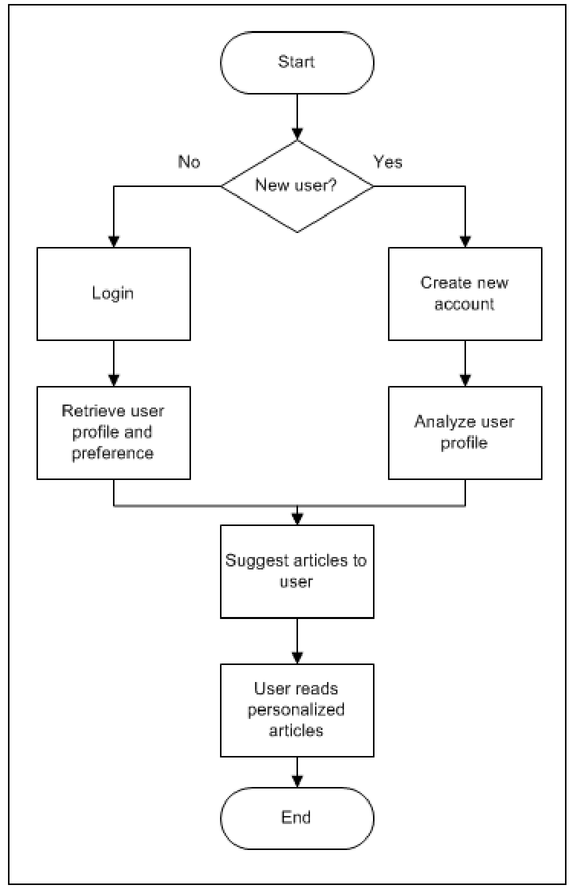


Figure 4: System Flowchart

6 Personalize Online Newspaper

In this system, the personalization strategy used is user profiling. Therefore, user is being identified by their login user ID. Each user profile includes user's preferences which user has to enter in a web form and the data are stored in the database. The articles suggested to the user will be based on the category of news user interested in and also the articles which have the keywords that match with user's entered keywords.

A content-based profiling is used to personalize the content of the developed online newspaper. An explicit user profiling method is used to retrieve users' data. User profiling is chosen because it can clearly identify the web users and provides an easier way to collect information regarding the users. The user profile stores information such as personal details and preferences. Personal details include data on username, password, name, birth date, gender address and occupation. Preferences include data on the user interest on category of news such as nation, business, entertainment, sport, lifestyles and so on. If the users choose news category such as entertainment, they will be presented with articles in that category as a tab option "Recommended to You". Users are also asked to provide keywords that might relate to their area of interest. For example if the user choose the word "iPhone", any articles related to the keyword will also be part of the articles recommended. Figure 5 shows the interfaces resulted from the captured user profile. All this information is stored in a database and each time the users log in, the information will be retrieved and analyzed in order to come up with recommendation of articles that match their interest.

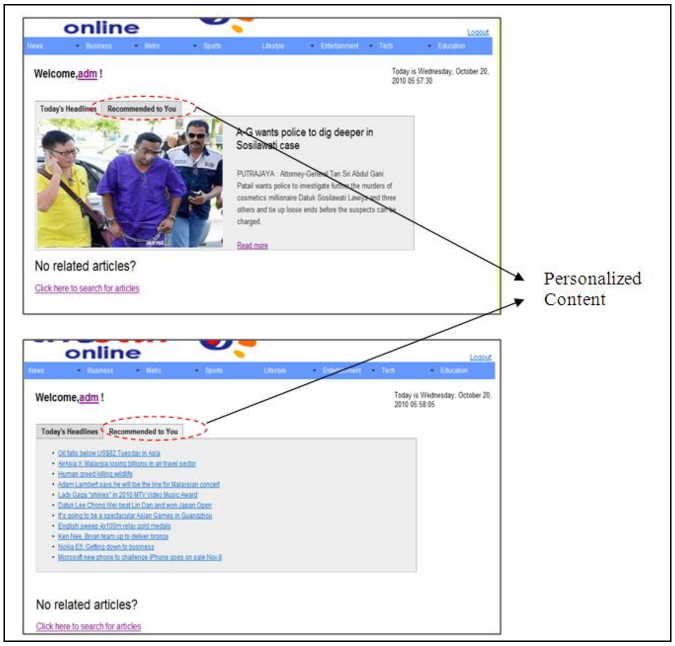


Figure 5: Interfaces Resulted from the Captured User Profile

Besides "Today's Headlines" and "Recommended to You", there is also a link to a search page where users can do an article search. The search page gives the users the capability to search certain articles that might not be included as part of their profile but of interest to them at that point of time. Search can be done by specifying the title (in part), author, category and/or dates.

7 Discussion

The developed personalized online newspaper is created based on an existing online newspaper that does not have any personalization features. It is developed by applying user profiling technique in which prediction was made based on users' preferences which is being acquired explicitly during user registration. There are pros and cons for using this personalization technique. The pro is that as the preferences were entered by users themselves, the prediction will somehow related to users' interests. Moreover, users can easily understand why certain articles are recommended to them. The disadvantage is that it requires too much effort from the users and sometimes users' interests are unconscious. To overcome this disadvantage, the authors suggest that a combination of user profiling personalization technique with a content-based filtering technique be used. This can be done by asking users to rate the articles that they have read. Content-based filtering technique will analyze articles that had been rated by the users and generate a set of related articles which associated with the contents of previously rated articles. As the related articles are generated based on users' reading and rating history, it can help to eliminate the problem that users' interests are unconscious and also more alert with the changing of users' interests.

8 Conclusion

A profile-based personalization had been chosen to be implemented on the online newspaper. This was done by asking the user to create an account on the online newspaper. The account will store user's basic personal information and also user's preferences. The system will then, based on user's preferences select articles which might appear to be interesting to the user. The personalized newspaper eases the users in looking for the articles of their liking. In the developed personalized online newspaper, recommended articles were being shown as links which will then lead the users to the full articles. With this, users can easily find articles that of interest to them instead of explicitly looking for a particular article by navigating through the whole website.

9 Future Recommendation

The authors had created a personalized online newspaper by applying user profiling personalization technique. With this personalization technique, prediction was being made based on users' preferences which had been explicitly acquired during user registration. Users are asked to enter category of news of their interests and also keyword(s) that might interests them. There will be situations that the keyword(s) that the users had entered does not match any article in the database for a particular day or archive articles because there are no such articles related to the keywords(s). However, in this project, this situation had been ignored. To further improve on this, a pop-up message can appear to inform users that the keyword(s) of their interests does not match any articles so that users can edit their profile and re-enter some other keyword(s).

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Energy-Efficient MAC Scheme for Sensor Oriented Future Internet Services

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***Abstract*— The design of an energy–efficient MAC protocol for provision of quality of service in sensor oriented personal area networks is challenging task. In this paper we propose energy-efficient MAC protocol with respect end-to end delays on IPv6-oriented-wireless sensor networks (IP-WSN). The proposed protocol can indentify multi-hop communication between source and destination for patient's monitoring applications. The hierarchical-cluster based schemes has been used for integrates IP-WSN (6Lowpan). This scheme (global communication) could be suitable for several Future Internet Services which one of the global healthcare monitoring applications (heart rate, three-lead electrocardiography, SpO2). The evaluation work of the protocol has implemented in NS-2.33 simulator and analysis the performance ratio of QoS with respect of throughput and packet delivery ratio.**

***Keywords*— IP-WSN, MAC, Energy-Efficient, uHealthcare, Future Internet Services.**

I. Introduction

The most important issue is the Internet Protocol connectivity over small low power embedded device. The IETF (Internet Engineering Task Force) working groups are continuously working to develop a standard 6Lowpan (IPv6 over Low Power Wireless Personal Area Networks) stack. In this stack, IPv6 is integrated to Lowpan device [RFC-4944]. The overall 6lowpan communication system into the PAN is offering global connectivity to the applications that have limited computational capacity, power and relaxed throughput. Some typical characteristics of 6Lowpan are: small packet size, support for 16-bit or IEEE 64-bit extended media access control addresses, low bandwidth, two kinds of topologies (mesh and star), low power, low cost and so on. Routing in different kinds of topologies should be implemented in such a way that computation and memory requirements are minimal. The IP-WSN node is using a web interface and the serial forwarder during the connectivity with sink node (gateway). The IEEE 802.15.4 standard defined reduced-function devices (RFDs) and full-function devices (FFDs) type of nodes. MAC layer beacons, RFDs can only communicate with FFDs in a resulting "master/slave" topology. FFDs can work in multi-hop mesh topologies [1-2].

The utilization of IP-based interconnection is the most common concern of industrial instrumentation makers. IP option is introduce to utilizing neither TCP/IP nor UDP/IP over Ethernet. However, this making some fear because of IP's ease of integration and broad interoperability. The (IP-WSN) nodes are use to transmit data which it receives from gateway and other sensor devices. The IP-WSN nodes are usually support mobility. Thus, the node can easily move and continue its communication in PAN but in association (joining or leaving node) would be a difficult task for wireless signal strength changes. Thus, the networks are working in a limited power capability and unable to carry large energy sources. These nodes usually operate in long hour of periods and making these devices to be build with little memory and modest processing capability.

The IP-WSN scheme is making more obvious and global connectivity with computers, laptops, and PDAs used by Wi-Fi (IEEE802.11) as their dominant wireless link. Wi-Fi is the most widely used in handheld client devices and embedded PCs, which are mains powered because of its high power consumption. The sensor oriented applications would be a part of Future Internet Services. For that, we have proposed a novel global healthcare monitoring applications. In technique, we need to design an intelligent wearable with fixed biomedical sensors over IP-WSN nodes. The biomedical sensors detect data such as ECG, glucose, or fitness related data from patient body and transmit it to the gateway operate by the consecutive forwarder over IPv6 networks. The intelligent wearable is considered "always connected" when there is a network connection between any two given nodes. The connected with the help of routing protocol and directly communicate with the IP-based networks. The intelligent wearable device fixed over patient body with integrated to IPv6 based gateway. Using this scheme the doctor can monitor his patient applications.

In this paper, we have worked on a novel energy-efficient MAC scheme to support several sensors oriented global monitoring applications. It is a very hot and challenging topic for researcher. To design energy consumption scheme while WSN network maintaining throughput and latency. Thus, we believe the novel energy efficient MAC scheme would be good choice to support Future Internet Services due to its global connectivity.

II. Energy Efficient MAC Scheme

The sensor oriented services for FI allows the ways to integrate the control of energy in everyday life easily with or without user awareness. With the load-balancing mechanism of shifting electricity usage and aggregating energy usage into cooperating pools, energy is used more efficiently. For example, a micro-controller controls the hardware modules inside the appliance and communicates with charging healthcare monitoring service over the Internet. The healthcare service monitors appliances current and predicted future power usage and reports to a pooling services that would construct an aggregated “profile” of the near-future power usage of connected units. This paper is discussing about the sensor oriented application monitoring with effective energy consumption with the respect of patient monitoring applications.

The transceiver characteristic distance dchar is the distance at which the transceiver characteristics are in equilibrium and it is the most energy-efficient communications distance.

$$d_{char} = \alpha\sqrt{}\,(e_{te} + e_{rx}) / e_{ta}\,(\alpha\text{-}1) \text{ -----------(1)}$$

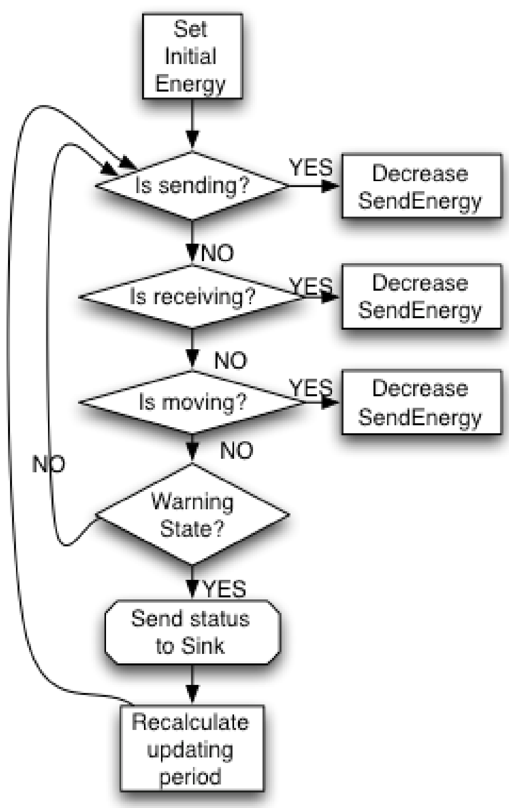

Figure1. Energy efficient MAC Scheme.

Where α is the path loss exponent, ete is the energy consumption of the transmitter electronics per bit, erx is the energy consumption of the receiver per bit, eta is the energy consumption of the transmit amplifier per bit over a distance of 1 meter. The energy model is a node attribute system in which energy will be decrease for each sending and receiving. When energy drops to a warning state, energy status will be periodically send to the sink. Also, the updating period is decreasing linearly as the energy is dropping. The sink collects energy data and sends it to the energy administration system so that necessary energy recovery can be carried out to the nodes. The overall algorithm is described in Fig.1.

Technically as we know the MAC layer used efficient communication of nodes network topology. It provides error free data transfer to the network layer. The design of protocol for MAC can be oriented towards the throughput, power consumption delay and quality of service. We use a Xemics 1209 transceiver designed for short range 2-5 m low frequency 36.86-45.05 data communication system. Sensor node worked discrete format for radio frequency and data communication between nodes. The data communication process master and each slave node in a wireless network to ensure, efficient and secure data field of each data packets. Cyclic redundancy checks standard polynomial in particulate CRC. The MAC frame format should fixed application information in PAN such as source to destination.

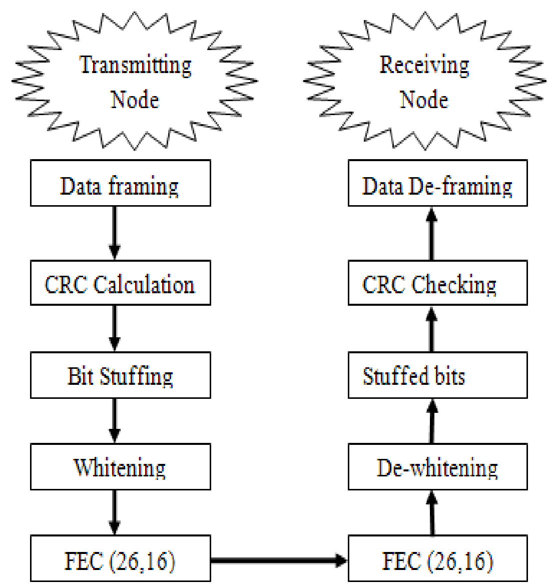

Figure 2. Simple Functional Diagrams for MAC Layer.

MAC layer has very important role during the communication between nodes in the network topology, which it provides error free during transmission to the network layer. In transmission node, it started with data framing. A data frame format is defined, then, it goes to cyclic redundancy check (CRC) calculation process. CRC calculation is used to detect any alteration of data during transmission. Thereafter, bit stuffing process that is the insertion of non-information bits into data. Whitening process is the next process and lastly gets to FEC (26, 12) encoding. For receiving node, it is directly opposite to transmitting node. It start with FEC (26, 12) decoding, and then de-whitening. Thereafter, stuffed bit is discarding. During this process, the non-information bits into data are taken out. Then, CRC checking, which is to checksum any alteration of data during transmission. Lastly, data de-framing is the last function flow.

III. Future Internet Service SCHEME: Global Healthcare Monitoring Applications

A. System Design

In our Prototype design, patient's body has fixed intelligent wearable to continuously monitoring application data. The wearable device should be in the range of IPv6-based gateway. The IEEE 802.15.4 standard defined reduced-function devices (RFDs) such as biological sensor nodes and full-function devices (FFDs) IP-WSN nodes. The MAC layer beacons RFD nodes commutate only FFD nodes via master/slave topology and the FFD nodes communicate both RFD and FFD to the gateway in Personal Area Network. Fig. 3 has shown several energy level of communication scheme.

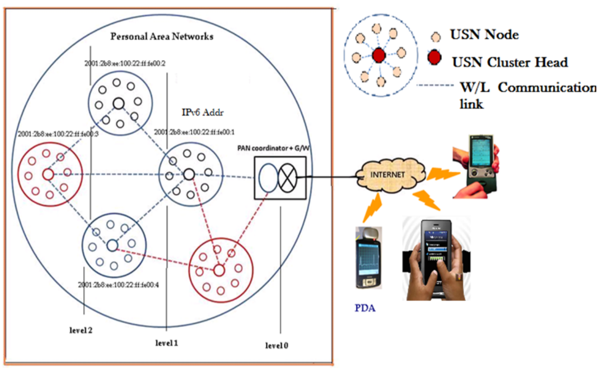

Figure 3. Global Patient's Monitoring System.

In each packet contains several bytes of preamble data with little energy to receiver. The energy efficiency measurement depends on the percentage of energy consumed by IP-WSN node during the delivery process. The analysis of energy consumption depends on the percentage of consumed energy at IP-WSN node and is calculated as initial energy. In networks system, the percentages of energy consumed by all IP-WSN nodes are measures of energy consumption of each node. The Intelligent wearable uses global unique IPv6 address for the identification and global connectivity between patient and doctor. Thus, the global connectivity technique has several challenges. We have given the solution of energy efficient MAC scheme of IP-WSN node. The doctor can send patient's query request to the gateway, and then gateway broadcasts query packet to all IP-WSN nodes. Thus, All intelligent wearable nodes transmit query response to the doctor with carry patient's IPaddr, query data, and signal strength of level 1 with gateway's level 0. After query request is received from gateway, the IP-WSN set their transmission power and reply to gateway its carry current position, energy consumption, and level. After data is received from IP-WSN, gateway can analyze energy consumption of all IP-WSN nodes.

The header contains destination EUID64 and source EUID64 that give information such as from where it came from and to whom it will receive. The relatively large identifier, EUID64 means at the time manufacturing, which is similar to Ethernet (IEEE 802.3) and Wi-Fi (IEEE 802.11). A 16-bit short address can be assigned to the device for the use of communication due to the lowpan packet size is small. The 6lowpan (IP-WSN) working group has presented IPv6 connectively over IEEE802.15.4. This working group also provides interconnection among lowpan devices and other IP links. Thus, it brings many advantages. One of the advantages is reduce a series of complex gateways. Gateway is a communication link for many adapters of the existing applications. Thus, it not only allows different company that using lowpan devices to be able to work together in a network, but able to work with many networked devices that already exists. Many industrial communication standards are able to support IP option.

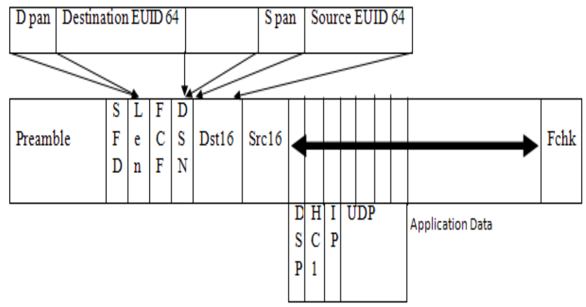

Figure 4. Header Compression Frame Format.

The working group makes these techniques because of the address, headers are large, and thus data transfers need to be larger so that it can fits in small 802.15.4 packets. Besides that, IP's utility is not free for usage. Fig. 4 show the example of extremely compact basic header expands where it also utilized the IP capability. There are only just 7 bytes, which are compressed from the entire 40-bytes IPv6 header plus the 8-byte UDP transport header. This is even smaller than a ZigBee header.

IV. Related Works

This paper has studies the basic technology available in today's healthcare monitoring applications in sensor networks and then designed energy-efficient scheme for future internet technology with respect patient's monitoring applications. In this part we have described the existing MAC frame work in Lowpan and then IPv6 based Lowpan for novel scheme.

A. Lowpan (Low Power Wireless Personal Area Networks) and Its Problems

Each communication link is corresponds to a specific low-level standards, which include a packet is coding scheme, and thus the physical device are able to communicate to each others. In 2004, IEEE standardized the latest wireless link, which is IEEE802.15.4. The IEEE802.15.4 is developing in compact, low power, low cost embedded devices, which can run on batteries for a certain periods. IEEE802.15.4 radio is use in home and industrial automation proprietary. It carries information at 2.5 GHz radio transceivers, at the power of 1mW, which is about 1% of the power Wi-Fi. Thus, it can say that it has low transmit power limits transmission range. Currently, in all wireless network protocols that used IEEE802.15.4 network use the same frame format and link-level header. Since all network protocols used the same frame format, additional information must exchange within the data payload section as a network-level header. This can be further explained that a network header will specifies where it starts, where it ends, and how it connecting each other. However, each of the current industrial protocols performs its network operation differently. Besides, there is no protocols address show how a packet is transferred out or into the IEEE802.15.4 network to the existing computers. The time synchronization with IEEE802.15.4 2006 MAC has defined beacon enabled and non-beacon enabled modes. The beacon enabled mode used only star topology networks, it cannot support mesh topology. During frame loss or collision period it uses beacons scheduling algorithm and periodic an overhead time has a lots of traffic overhead. However, non beacon mode required long idle listening in high power consumption. This frame format is not suitable for 6lowpan stack, it needs modification. For Global health care monitoring applications technique needs peer to peer & multi-hop mesh topology with beacon and without beacon enable network technique. It must be used time synchronization with guarantee time accuracy within error boundary and robust in change of topology.

The GTS (Guaranteed Time Slot) requests should compete with data frame in CAP (Contention Access Period) then it would increase collision probability, which reduces reliability. The acknowledgement in the next beacon period, so latency is the major problem. Maximum number of GTS is limited to 7 so inefficient overload networks. Quality of Services provision is difficult between Contention Access Period vs. Contention Free Period. For global healthcare monitoring applications, required reliable and fast channel access for communication because 6lowpan node broadcast req. packet by MAC layer to other neighbor 6lopwan nodes then it announces destination node id.

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The IEEE802.15.4 super frame is not flexible to accommodate dynamic traffic and channel hopping technique. The channel hoping technique provides high reliable link-level connectivity but current channel may not be sufficient to established requested channel connections. The quality of services provision may require more robust control of bandwidth requests instead of competing with data frame in contention access period. Flexible but robust out of band control frame may provide efficient channel access.

V. Simulation Work & Results

In this system, we assumed 11 6lowpan nodes their some of cluster head. Each node connected to cluster head and has its own IP address. The cluster head transmit its information to the gateway via multi-hop routing. The simulation parameters are presenting in Table 1.

TABLE 1

Parameter	**Value**
Transmission Range	15m
Simulation Time	120 s
Topology Size	100m * 100m
Number of Mobile Nodes	11
Number of Sources	2
Number of PAN Coordinator	1
Traffic Type	Constant bit rate
Packet Type	15 packets/s
Packet Size	32 bytes
Pause Time	5s
Maximum Speed	2 m/s

The initial energy of gateway is high due to its wired fixed power supply and it is not resource constraint. For simulation, we took 100j power supply for gateway. The initial energy of IP-WSN node is 1.5j and biomedical sensor is 0.5j are fixed. The IP-WSN node knows their location and distance with other IP-WSN node. But for simulation, we took a variable type 0, 1, 3 for IP-WSN, boi-medical sensor, and gateway respectively. Initially, we have used the signal strength of IP-WSN, biomedical sensor and gateway nodes are8, 3 and 10 respectively. Then, it will vary this signal strength afterwards and test its affect the network connectivity. Initially its value is enough to satisfy the coverage and connectivity. Initially gateway is – infinity (very low value) and IP-WSN node is –infinity (a very low value) due modified MAC protocol they choose their parents. By analyzing the data and running the simulation process is placing the left energy at initial energy.

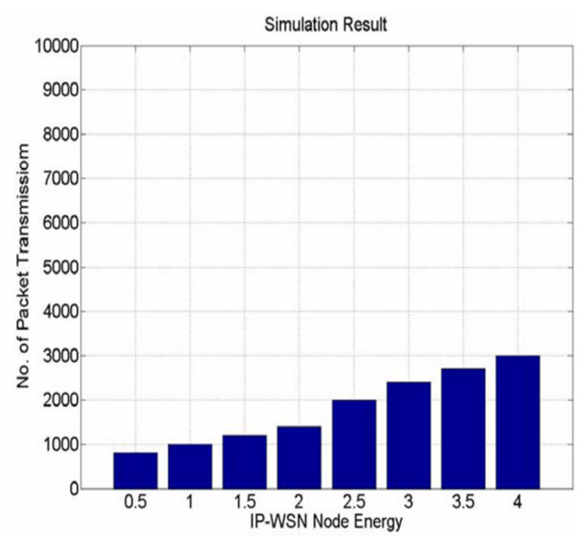

Figure 5. End-End Packet delivery Ratio.

In this paper, we have analysis about energy of IP-WSN. Thus, we get the total no. of transmission packet send to the gateway. We took IP-WSN Energy is 1.5, and then it varies up to 4 joule for total no. of packets.

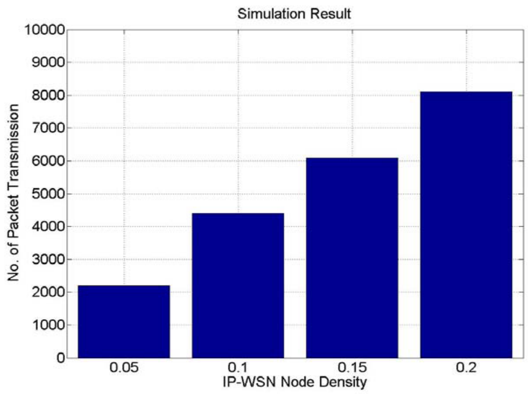

Figure 6. Throughput .

The simulation communication channels are also modeled and different type of fading and channel are used so due to fading simulation result varies with numerical results. Thus, the 3j energy of IP-WSN node's density is varies 0.05 to 0.2 Joule.

VI. CONCLUSION & FUTURE WORK

The paper has presented the energy efficient MAC scheme with respect delay and throughput. For that, the author has work on IP-WSN header and stack format. The stack has ability to fully realizable and highly pervasive with routing protocols, connectivity with external internet, service discovery, and coexistence with other peer technologies. The IP-WSN nodes are uniformly distributes in the PAN. The author has implemented novel MAC scheme on the NS-2.33. The designed scheme support IP-based networks to assist current status of the patient in hospital-based PAN. The next face of this paper is test-bed setup which is under progress work. We are developing an effecting protocol for fault tolerance in IP-enable USN as well as better performance of energy and routing especially biomedical data. We are trying to real time testing for whole global healthcare monitoring system.

ACKNOWLEDGMENT

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Terms Reduction with Web Page Clustering

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Abstract - *As the size of the World Wide Web has grown largely, it has become difficult to retrieve useful information quickly. User looking for information may have to browse lots of pages to get the desired information from the pool of World Wide Web (WWW). A technique is required which organizes documents content efficiently so that information can be easily obtained from largest data repository of WWW. Clustering is an unsupervised classification technique which puts related data in one set (Cluster). Clustering can help user to get interested information quickly from these abundance of information. However clustering methods are suffered from the huge size of documents with the high dimensionality of text features. We have proposed web page clustering scheme that works efficiently in higher dimension. We have presented the method to reduce the dimensionality of the feature vector by selecting the most informative words and still maintaining the quality of the clusters.*

Keywords: *Web page clustering, Feature selection, Cluster quality, Dimension reduction, Term reduction.*

1 Introduction

The size of World Wide Web (WWW) has grown largely and still rapidly increasing. From abundance of information on WWW, getting desired information quickly is important for the users. Web document clustering can help to achieving this objective. Document clustering gathers the semantically same meaning documents into one group so documents in the same clusters are highly related to each other compare to documents in the other groups. Generally, a user looking for information submits a query to the search engine and the search engine returns the linear list of URLs with short document summaries in the order of documents relevancy to the query. This query list approach may return thousands of pages if query is too general and problem increases when users are novice about the topic they are searching (i.e. having no domain knowledge of topic they are looking for) and the search engine retrieves documents on different topics. Clustering of search results can help the users to quickly get the interested information by selecting appropriate cluster. Apart from clustering search results other application of the web document clustering can be clustering news articles published on web site into different groups like sports, highlights, business and health etc [1]. Recently document clustering has becomes more important means for the summarization, organization and navigation of the information in the documents [2] [3] [4].

Many of the clustering algorithms suffer from the high dimensionality [5] [6]. This high dimensionality is due to bag of words vector representation of the each document. In bag of word representation, each element corresponding to a distinct word in the document and each document has thousands of words so resultant document corpus from all documents would have very large number of words. From bag of words, document to term matrix is constructed which has number of rows correspond to each document in the collection and number of column corresponds to each distinct words in the all the documents which would be very high. Value of the document to term matrix in ij cell indicates the frequency of term j in the document i. the column of the matrix depends on the number of the word in the document corpus which would be very high. The computation of the clustering process working on high dimension matrix results in very time consuming process and also it consume much memory to store and process this matrix in each round of the clustering process. So dimensionality of the matrix must be reduced. Here in this paper we have proposed term selection algorithm to reduce the dimensionality of the matrix and we have used in clustering process.

1.1 Clustering Problem Formulation

Clustering problem can be formulated mathematically in following way. Consider that X is the document set to be clustered, $X = \{x_1, x_2, \ldots, x_D\}$. Each document x_i is a vector of various dimensions, where each dimensions typically correspond to extracted features from the documents. D is the number of documents in the corpus. A clustering of documents collection X in “m” sets can be defined as $Q = \{q_1, q_2, ..., q_m\}$, so that the following conditions are satisfied:

$q_i \neq \emptyset$, $i=1, \ldots, m$,

$q_1 \cup q_2 \cup \ldots \cup q_m = X$,

$q_i \cap q_j \neq \emptyset$ for $i \neq j$ and $i, j = 1$ to m.

2 Related Work

Agglomerative Hierarchical Clustering (AHC) algorithms [5] are probably the most commonly used. These algorithms are typically slow when applied to large document collections. It is too slow to meet the speed requirement for one thousand documents. Because time complexity of agglomerative hierarchical clustering is $O(n^2)$. K-Means clustering algorithms are the best candidates to comply with the speed requirement of online clustering [7]. Time complexity of the K means clustering is the O(nkt) where n is the where n is the number of objects, k is the number of clusters and t is the worst number of iterations needed for reaching stable state. However with this lower time complexity it takes considerable time to generate the cluster due to high dimensionality of the document to term matrix. Therefore term reduction techniques along with clustering process are the prime requirement. The majorities of papers on feature selection involve complex calculation and work on the principle of the greedy algorithm that is taking first K terms based on some calculations.

Many of the researchers define some measures for term selection among which Dhillon [8] introduces the term variance quality and term profile quality based on some complex computation to reduced the dimension of the document to term matrix. In the approach proposed in [9], which uses the entropy to select the terms, the process is more sophisticated, since there is a concern that all documents are represented after term selection but this algorithm requires prior determination of the minimum number of terms. yuan-chao et. al. [10] proposed feature selection based on word co-occurrence frequency their algorithm used two threshold one for document frequency that is the number of documents in which term occurred and other for word co-occurrence frequency that is the number of document in which two terms W_i and W_j appear together. For every words in each document word co-occurrence must be found and then words are selected.

3 Clustering Process

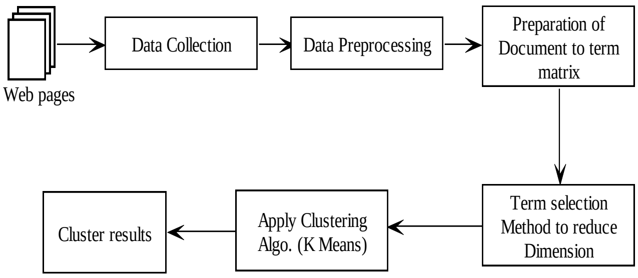

Figure 1: Clustering process

The main idea of our clustering process is to greatly reduce the feature dimension of the documents along with maintaining the accuracy. Figure 1 shows the complete process of the clustering. In first data collection step data is gathered in form of search results or it can be offline web pages on various topics. In data preprocessing step of the algorithm web pages are preprocessed by removing HTML tags, images and other advertising frames. That is converting web pages into text file. Special characters are also removed in this step. Articles, prepositions, and other common words that appear frequently in text documents but do not bring any meaning or help distinguish documents are called stopwords (for example 'are', 'a', 'an', 'the', 'on', 'in', 'and', 'at'). These words are usually removed. Stemming operation is also performed. The aim of stemming is converting words to their canonical form (base, root) be removing their grammatical suffix and prefix. This process allows matching different variants of words. These preprocessed documents are represented in the proper format before using it for the clustering. For documents representation we have used most commonly used vector space model (VSM) [11]. In VSM, each document is represented as an n-dimensional vector. The value of each element in the vector reflects the importance of the corresponding feature in the document. Each term in the VSM is assigned some weight using term weighting function to prepare document to term matrix. Most commonly used term weighting function is TF-IDF (Term frequency – inverse document frequency) function. TF-IDF is define as $TF_{ij} * IDF_i$. Where $IDF_i = \log (N/df_i)$. Here TF_{ij} is the number of occurrences of the term t_i in the Web page P_j, N is the total number of web documents and df_i is the number of Web pages in which term t_i occurs in the web document collection. Here in our clustering process we have used weighted inverse document frequency (WIDF) [12] instead of TF-IDF term weighting function to prepare document to term matrix. Consider following example to understand the drawback of the TF-IDF based approach for term evaluation.

Table 1**:** Document to term matrix.

Terms / Docs.	……. t_i ………... t_j …..
d1	……. 12 ………. 2 …..
d2	……. 25 ………. 3 …..
d3	……. 5 ………... 6 …..
d4	……. 9 ………... 7 …..
d5	……. 10 ………..4 …..

Consider the table 1 that contains number of documents in column and number of terms in rows. The intersection of document d_i and term t_i stands for the term frequency of the term t_i in document d_i. If we calculate TF-IDF values of document d_2 for terms t_i and t_j then it become 0 even if their frequency distributions in each document are quite different. IDF is not able to reflect frequency distribution because df_i in IDF equation reflect only absence or presence of the term in each documents. So WIDF for term "t" in the document "p" is defined as below.

$$WIDF(p,t) = \frac{TF(P,t)}{\sum_{i \in D} TF(i,t)} \quad (1)$$

Here D is the collection of web pages, TF (p,t) is the frequency of term "t" in the document "p". In above example WIDF values for term t_i and term t_j for document d2 are 25/ (12+25+5+9+10) and 3/ (2+3+6+7+4) respectively. So we will use WIDF term weighting function and evaluate the each terms. Next step in our clustering process is the term selection process. Dimensions of document to term matrix is very high as it contains all the terms for all web pages. Working with higher dimension is quite difficult so dimension of document to term matrix should be reduced. And term reduction should not result in the reduction of the cluster quality. Our term selection algorithm select most informative word from document to term matrix and that is very less compare to number of terms in original document to term matrix.

Algorithm: Terms selection
Input: D X T document to term matrix, where D is the number of documents and T is the number of terms.
Output: D X P matrix where P << T

1. Let M be the document-to-term matrix associated to a set of D documents and T terms.
2. Evaluate the coverage of each term of M in each document.
3. Sort the terms according to their coverage in the document i.e. term with highest coverage comes first.
4. Take the top P terms with the coverage of half or more than haft of the documents putting them in list NEWLIST. Neglect the terms that covers almost all documents.
5. Remove all documents (rows) covered by top P selected terms from matrix M and also remove from M all terms (columns) in NEWLIST.
6. If no document remains in matrix M, go to Step 7. Otherwise, go to Step 2.
7. If selected terms are too less compare to number of documents then add few other terms to NEWLIST based on their coverage so that number of terms should be 2 time total number of documents.
8. End.

Above terms selection algorithm is very simple to implement and does not involve any complex calculation to select the terms because it calculates only coverage of the term that is the numbers of documents covered by the term. Step seven checks that if algorithm results in very few term selections due to stop words or noise, then it takes other extra terms having high coverage to improve the accuracy of the clustering. Its time complexity is K*N log N where N is the number of terms and K is the number between 1<= K < D. Here D is the total number of documents in the collection.

Our clustering process has many advantages compared to other algorithms.

- Due to the high level of the document to term matrix's sparsity, it is possible that clustering process results in poor quality with other algorithms. This problem is not there in our proposed algorithm because term selection allows us to reduce the number of terms to a great extent.
- Because of high dimensionality of document to term matrix clustering process may become complex and takes more time to perform clustering.
- Our clustering process with term selection algorithm reduces the sparsity of document to term matrix by selecting most informative terms that results in lower dimension and makes the clustering process fast and accurate.

After term selection algorithm clustering algorithm is applied to the selected term. Here we have used K-means clustering algorithm due to its simplicity and speed. However any other clustering algorithm can also be used. Standard Algorithm K means is as below:

Input: selected terms with WIDF weight and number of clusters
Output: Partition of the documents into given clusters.

1. Initially select *k* objects randomly;
2. *Repeat*
3. Compute the centroids of the clusters;
4. Assign each object to the cluster with the nearest centroid;
5. *Until* no change in cluster assignments.

K means algorithm is relatively scalable and efficient in processing large document sets because its computational complexity is O (nkt). Where n is the number of data sets, k is the number of clusters and t is the number of iterations.

4 Experimental Simulations

4.1 Dataset & Experimental setup

For clustering process we have used collection of web pages located at the web site http://pami.uwaterloo.ca/~hammouda/webdata/ [13]. This dataset is the collection of 314 web pages. The pages belong to 10 different categories like blackbear-attack, campuse-network, Canada-transportation roads, career-services, co-op, health-services, river fishing, river-rafting, snowboarding-skiing & winter-canada. For clustering process web pages are preprocessed by removing HTML tags, images and other advertising frames. From preprocessed documents, CSV file is prepared containing words from each document. One single line in CSV file contains all the word from one web page separated by comma. Once CSV file is prepared, stops words (less informative words which are very common words such as "to", "am", "is" and "are") are removed and stemming operation (converting words to their original form by removing suffix or prefix) is performed. Then WIDF weight is assign to each term. Below figure 2 shows the intermediate step of the clustering process. In first step from collection of web pages one CSV file is prepared which contains one line for each web page in the collections. From this CSV file stop words are removed and stemming operation is applied. In step 2 file and generates the document to term matrix whose value gives the frequency of each word in each web page. After generation of document to term matrix we will apply WIDF weights function. Now document to term matrix contains WIDF values as shown in step 3. After step 3 term reduction algorithm is applied and term is reduced compare to terms in step 3. In step 4 we will have reduced terms and then K means clustering algorithm is applied to generate the clusters.

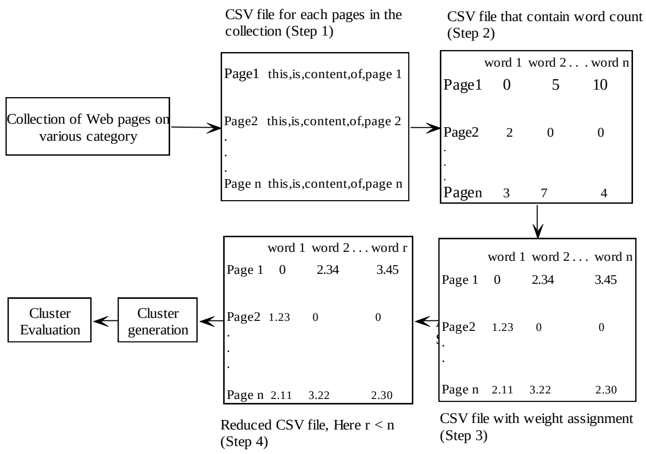

Figure 2: Processing steps for the clustering process

With K means algorithm to find the similarity between two documents many similarity measures are used among which Euclidean distance and cosine similarity are very famous. We have conducted the experiment to know which similarity measure perform better for our WIDF term weighting functions. For comparison of two similarities we have used two data sets one is mention above and for other dataset we have manually downloaded 150 pages on five different categories. Below figure 3 shows the F measure value for Cosine similarity and Euclidean distance for two data sets. Results show that Cosine similarity outperforms the Euclidean distance for both the data sets.

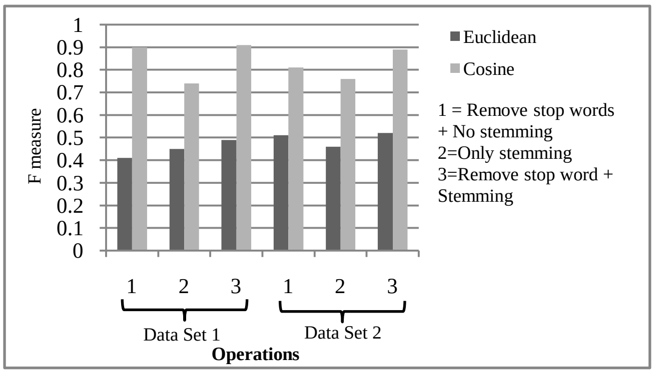

Figure 3: Comparison of Euclidean distance and Cosine similarity for two data sets

4.2 Evaluation measures

For evaluation of our cluster quality we will use F-measure which uses precision and recall [14]. The recall–precision framework is a useful method for evaluating Information retrieval system performance. It is also used to measure the quality of clustering [15]. The proportion of retrieved relevant documents to all retrieved documents is called precision. If all documents retrieved are relevant then the precision value is 1 but it may possible that some relevant documents may be missed out. Same way recall represents the proportion of relevant documents retrieved to all relevant documents. If recall value is 1 it means all relevant documents are retrieved but it may contain irrelevant documents. Same concept is applied to cluster evaluation if each cluster is treated as if it were the result of a query and each class as if it were the desired set of documents for a query. Then the recall and precision of cluster j for given class i is

$$\text{Recall}(i, j) = n_{ij}/n_i \qquad (2)$$

$$\text{Percision}(i, j) = n_{ij}/n_j \qquad (3)$$

Where n_{ij} is the numbers of members of class i in cluster j, n_j is the numbers of members of cluster j and n_i is the number of members of class i. Both precision and recall together are important measures for evaluation of clustering quality high value of any one is not encouraged for clustering so F measure is used which is harmonic mean of precision and recall of cluster j and class i is given by the following:

$$F(i, j) = (2 * \text{Recall}(i, j) * \text{Precision}(i, j)) / ((\text{Precision}(i, j) + \text{Recall}(i, j)) \qquad (4)$$

F measure considers the trade of between recall and precision. It discourages the clustering algorithms that sacrifice one measure for another too drastically. Higher value of F measure indicates good clustering quality. F-measure for the entire clustering is,

$$F = \sum_{i=1}^{m} \frac{n_i}{n} \max_{j=1,..,k} F(i, j) \qquad (5)$$

"n" is the total number of documents in the collection.

4.3 Results

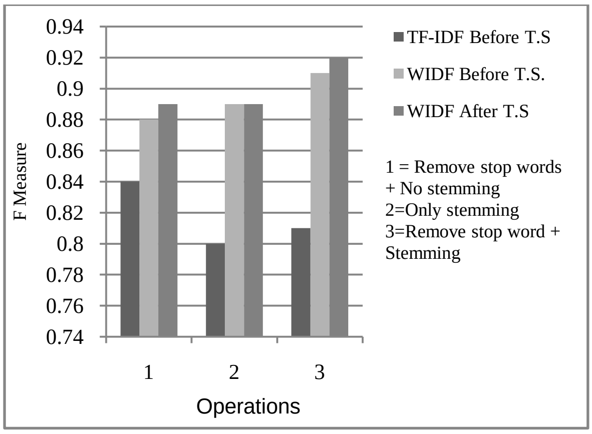

Figure 4: F measure for WIDF and TF-IDF for various operations.

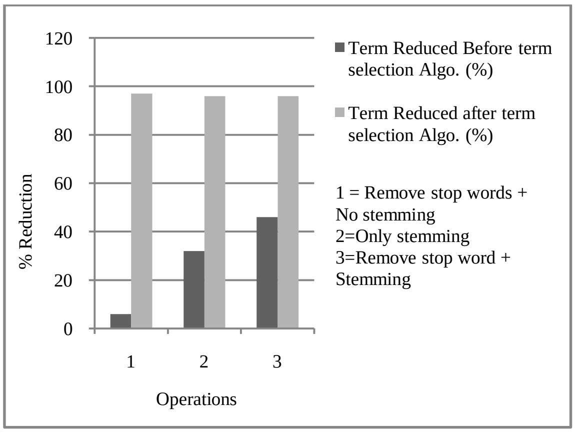

Figure 5: Reduction in terms with term selection algorithm for various operations.

Figure 4 shows the F measure value for TF-IDF and WIDF with various combinations of stop-words removal and stemming operations without applying term selection algorithm and WIDF after applying term selection algorithm. Here WIDF gives better cluster quality compare to TF-IDF with all combination of operations. After term selection, WIDF gives better or equal cluster quality compare to WIDF with full document to term matrix. F measure with WIDF for all combination of stop-words removal and stemming operation shows that even with the noise also our term selection algorithm gives better cluster quality. Figure 5 shows the percentage of terms reduced after term selection algorithm and before term selection algorithm with various combinations of stop-words removal and stemming operations. Results show that our term selection algorithm gives better quality for cluster generation with reduced terms. It also shows that term reduction process has reduced about 97% of terms from document to term matrix and with less terms also we have got even better result compare to F measure with full document to term matrix. With this reduced terms clustering process become fast and require less space to store the document to term matrix.

5 Conclusion and Future Scope

In this paper we have proposed term selection algorithm that largely reduced the number of term from clustering process and still maintains the proper cluster quality or even gives the better cluster quality. Our term selection algorithm also gives better results with the noise also that is with stop-words. With this reduction in the terms clustering process becomes fast and accurate. It requires less space to store complete document matrix and less computation to process the matrix. We have also shown the clustering process with this term selection algorithm that gives better results. For future work we can have term selection algorithm that works on combine text and link features of the web pages. Also we have used simple K-means algorithm for clustering but any other algorithm like genetic or particle swarm optimization or combination of two algorithms can also be applied to generate the cluster.

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The Impact of Video Conferencing on Campus

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Abstract- *Video Conferencing offers a whole new way to teach, learn, and interact. College campuses everywhere are turning to video conferencing (VC) technology as a new way to interact with students. Simultaneously, network administrators are struggling to keep up with ever-growing bandwidth requirements. Multimedia streaming applications, such as video conferencing, can consume large amounts of bandwidth, and negatively impact web browsing.*

In this paper, we will use IT Guru to simulate a simplified college campus dorm consisting of a "typical" wireless user generating moderate amounts of HTTP traffic. We will then measure the negative network effects introduced by a large amount of VC traffic.

Keywords: Video Conferencing

1 Introduction

Video Conferencing has traditionally been constrained by high costs, dedicated hardware and software requirements and limited flexibility [1]. However, the emergence of widespread, relatively low-cost broadband Internet access has caused many college campuses and corporations to utilize video conference as an effective way to interact over any distance [2]. Despite the benefits, video conferencing requires relatively large amounts of bandwidth.

In addition to VC, numerous bandwidth-intensive applications such as multimedia streaming and peer-to-peer file sharing can consume significant portions of bandwidth. These applications are becoming increasingly popular on campus, causing network administrators to turn to traffic control strategies such as throttling to mitigate any resultant negative network effects [3].

IT Guru network simulation software can be a powerful tool for network administrators seeking to study the impact of video conferencing, or similar applications, on a campus network. Accordingly, administrators will be better equipped to implement any necessary traffic control strategies while minimizing possible unintended consequences of such measures.

The purpose of this study is two-fold. First, we seek to establish the negative network effects of video conferencing on a representative web-browsing "typical" user in a simplified campus network. Secondly, we will establish IT Guru as a viable tool to model and study a campus-level network.

2 Model Description

Two separate campus models will be simulated within IT Guru to measure network performance. The first model will consist solely of HTTP traffic. The second model will introduce a large amount of VC traffic.

2.1 Baseline Model

We will use IT Guru to simulate a simple college dorm network. This dorm network will consist of a wireless access point and a wireless workstation. The workstation will simulate a "typical" college campus network user. This user will generate moderate amounts of HTTP traffic. An off-campus HTTP server will provide service to the user. Traffic will travel over the "ip cloud" model object to simulate the Internet. A campus firewall and additional network objects have been omitted for clarity.

For the purposes of this study, a typical user is simulated using the "light web browsing" application definition in IT Guru. However, the default settings need modification to more accurately reflect modern web browsing. Default settings for the web browsing definition include a page inter-arrival time of 60 seconds and a constant 1kB page size. Instead, we will use an inter-arrival time of one second. This will allow the workstation to generate sufficient amounts of traffic to model a typical campus environment, namely a fairly constant yet relatively low-bandwidth stream of steady HTTP traffic. Such an inter-arrival time precludes the need to use a large number of workstations to model the required traffic patterns. The page size has been increased to 320 kB to accurately reflect a representative modern webpage [4]. These settings can be easily adjusted to match constantly changing web characteristics.

This network will be used to establish a baseline for the simulation study. A variety of measurements will be taken during the simulation run in order to establish network quality without VC traffic. The network has been modeled to provide a high expected level of performance consisting of moderate link utilization, very low delays, and high reliability.

2.2 Video Conferencing Model

After establishing the network performance of the baseline model, we will then introduce the VC traffic. This traffic will be generated by two wireless workstations. The first workstation will be co-located with the "typical" user, utilizing the wireless dorm room access point. A second, identical workstation will be located outside of the campus network. This model has been chosen for simplicity and to clearly show the effects caused by video conferencing. IT Guru allows administrators to easily model a wide range of network configurations.

We will use the default IT Guru application definitions for the VC model. We will use the "VCR quality video" settings to reflect modern user expectations and typical resolutions. This setting uses a medium-resolution, 30 frame-per-second encoding scheme and "best effort" delivery service. These settings can easily be modified to fit particular real world network conditions. We expect these settings to have a significant negative effect on the web browsing user, such as increased delays and dropped packets, without completely crowding out HTTP traffic.

These two workstations will generate relatively large amounts of VC traffic. A variety of measurements will be taken during the simulation run to measure the negative network effects on typical web browsers caused by the VC users.

3 Simulation Results

The initial simulation run will consist of solely TCP traffic in order to generate a baseline measurement of network quality. We will measure the segment delay seen by the "typical user" as a key indicator of network quality. In addition, we will also measure both the point-to-point link utilization and the queuing delay between the campus gateway and the Internet. Recall that the typical user generates moderate amounts of HTTP traffic, equivalent to ordinary web browsing.

3.1 Baseline simulation results

The HTTP-only simulation shows reasonable network quality with minimal TCP segment delays of 50 milliseconds (Figure 1).

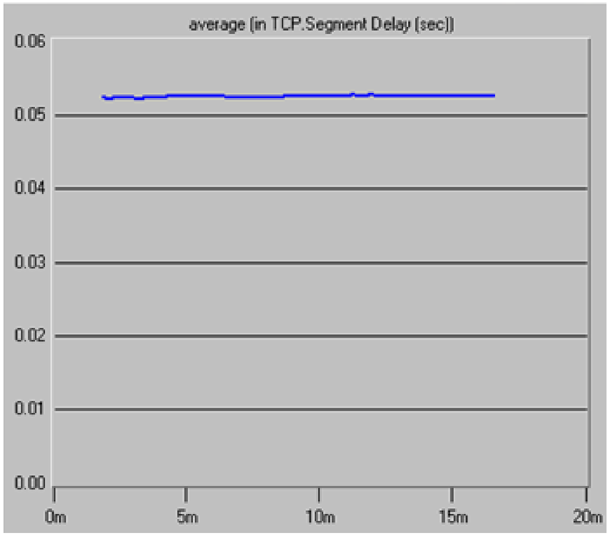

Figure 1. Average segment delay for typical user.

Similarly, the point-to-point queuing delays between the campus gateway and the Internet were minimal, as shown in Figure 2. The network required no TCP retransmits. In addition, the DS1 (T1) line between the Internet and the campus gateway shows approximately a 25% link utilization (Fig. 3), in line with our expectations given the moderate web browsing characteristics.

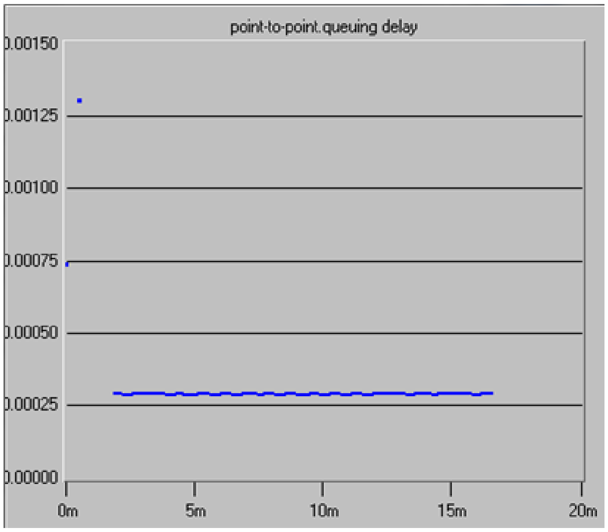

Figure 2. Queuing delay between gateway and Internet.

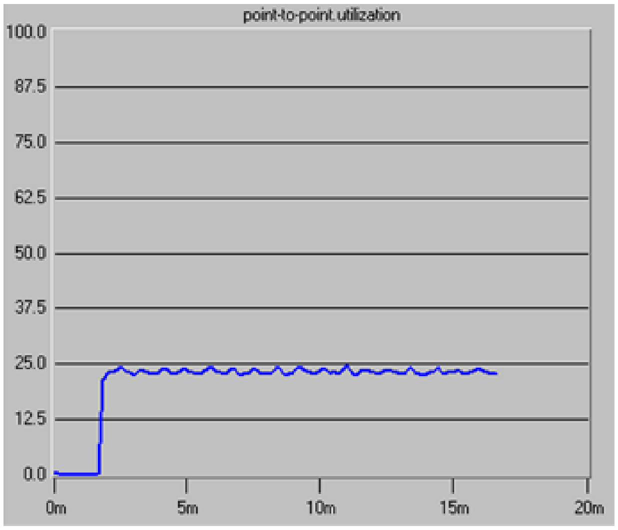

Figure 3. Link utilization between gateway and Internet.

3.2 HTTP and video conferencing results

The second simulation introduces relatively large amounts of VC traffic in addition to the HTTP traffic. Recall that our model includes both an on-campus and off-campus VC user. This additional traffic showed significant negative impact on the typical web browsing user. Despite the substantial impact, HTTP traffic was not completely overwhelmed by the video conferencing.

Of particular note, the segment delay was significantly impacted, showing an 18x increase compared to the baseline measurement. This is shown in Figure 4 below.

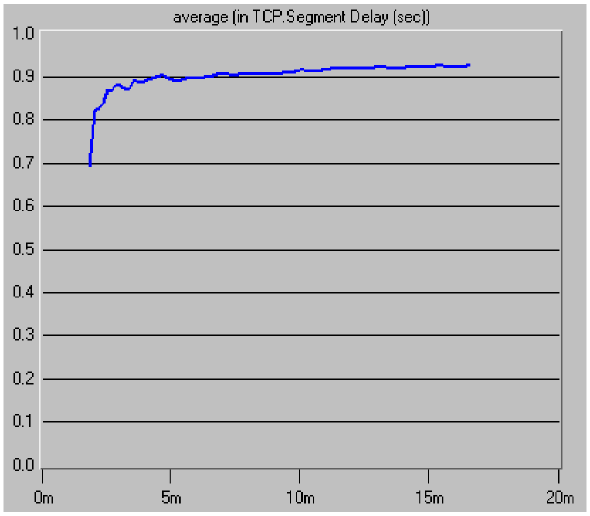

Figure 4. Segment delay with VC.

A similarly substantial increase in queuing delay was also measured on the Internet to campus link. Additionally, the user experienced frequent dropped packets; requiring approximately 45 TCP retransmits (Fig. 5).

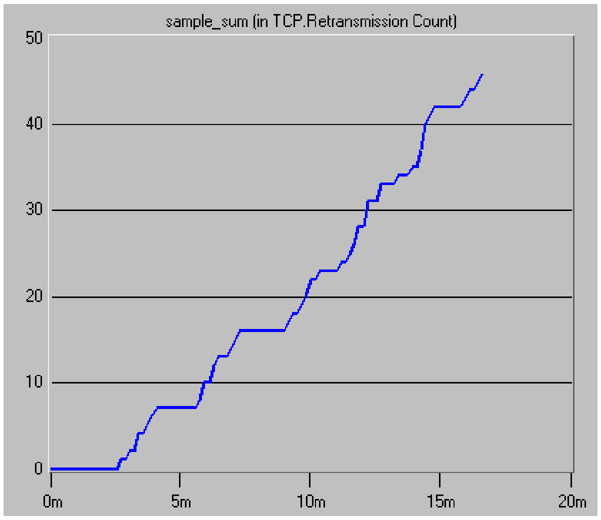

Figure 5. TCP retransmits with VC (sum).

The DS1 line between the campus gateway and the Internet showed nearly a 60% increase in link utilization with the introduction of VC traffic (Fig. 6).

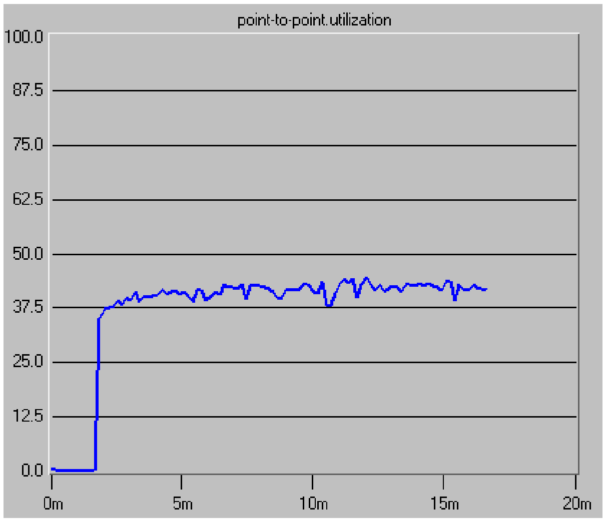

Figure 6. Link utilization between gateway and Internet.

Table 1 summarizes the network quality measurements of the two simulations.

Table 1. Summary of network quality measurements

	Segment Delay	Link Utilization	Queuing Delay
HTTP only	50 ms	24.5%	0.25 ms
HTTP & VC	900 ms	40%	9.25 ms

4 Conclusions

The introduction of video conferencing showed a significant detrimental impact on network quality, particularly to the HTTP traffic of the typical user. As expected, large delays were created throughout the network. In particular, the nearly 1 second segment delay seen by the typical browser effectively ruins the web browsing capability of network users.

The VC traffic also caused numerous dropped packets for the web browser, requiring a TCP retransmission, on average, approximately once every 25 seconds. However, in many instances, a dropped packet causes a domino effect, with the resultant retransmission being dropped as well. Consequently, the network exhibited a large degree of packet delay variance, also known as “jitter.” Minimizing network jitter is a key requirement for quality media streaming and video conferencing [5]. Such behavior must be minimized in order to retain network reliability and ensure user satisfaction.

Video Conferencing, while offering many advantages to both college campuses and corporate environments, can negatively impact ordinary web browsing. However, using network simulations such as IT Guru, network administrators can easily model, and therefore prepare for, the network effects of video conferencing.

5 Future Work

This study has modeled a simplified campus network for clarity and to demonstrate the viability of IT Guru as a network modeling tool. Future studies will expand upon the model in order to more accurately simulate a real-world campus network. Kennesaw State University utilizes video conferencing technology as an effective teaching method for both local and long-distance students. IT Guru can be used to model the KSU campus network as a means to measure the impact of increased VC usage for lecture dissemination and student-teacher interaction.

Furthermore, future models can incorporate a wide variety of traffic, including such popular applications as peer-to-peer sharing, Voice over IP, and multimedia streaming. The popularity of such applications continues to grow and requires effective simulation tools such as IT Guru to model and predict the impact of such usage on a campus network.

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Research of Custom Management System for C2C E-Commerce

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Abstract - *Through researching customer management of shops online in C2C e-commerce mode now, find problems on business development and analyze the advantages and practical significance of Taobao's customer management system. Mainly analyze Taobao shop business problems in the customer management and help them to manage the customer effectively. By analyzing the existing pages extraction technology, and understand the basic processes and page extraction method, then design and develop Java program to extract shop's goods information, trading records, account information in Taobao, design the database to save classify and manage data, and then use the Google chart API generates charts, and use the website based B/S structure to display information on charts, which provide a reference for product marketing in Taobao.*

Keywords: CMS, information extraction, e-commerce, Data management; generate reports; statistical tables and charts

1 Introduction

Customer resources decide the core competitiveness. On taobao net shopkeepers have less knowledge for their business and lack of sales of similar shop the collection and analysis, not the product of reliable reference data for marketing information, to defend their clients. Taobao face increasing competition, businesses are required to protect their customers, concern for their ideas, needs, purchase intent, and good with customers to establish good, long-term customer relationships, strengthen communication with customers, retain old customers and expand New customers, enhance customer value, and thus to enhance the competitiveness and profitability, which came into being in this topic.

2 Related research

Please use the styles contained in this document for: Title, Abstract, Keywords, Heading 1, Heading 2, Body Text, Equations, References, Figures, and Captions.
Do not add any page numbers and do not use footers and headers (it is ok to have footnotes).

2.1 *A Li Wang Wang*

Currently, C2C online shop for customer management, provide Ali Taobao Want software. By using this software, businesses can add as a friend buyers account, you can send them information, and this software can also display for business, link lead to Taobao shop business background to handle orders, shipping, returns, has been sold Out of the commodity information. But there is no statistic of information for buyers, the analysis of consumption, can't provide a reference for the marketing business.

2.2 *Jin Suan Pan Platform*

This is a powerful enterprise website management system, integrated network marketing ideas, using the generated static html page as the template isolation procedure, a powerful content tagging, and acquired in the establishment of site management functions very well, but not for the management of customer Function, not the customer information analysis and statistics.

2.3 *Maimaile, pat, eBay and other C2C websites*

Although the front of these beautiful interface C2C site, the background is also more perfect management functions, but not-to-business customer management, statistical analysis of consumption and other functions.

2.4 *The advantages of the system*

The subject of research and development for Taobao business customer management system is designed for Taobao business tailored customer management system, capable of business customers personal information, consumer information, product information for statistical and management, and generate charts and Detailed reports product marketing for businesses to conduct a reference..

3 Program Design Information Extract-ion

This procedure is mainly the use of Java technology to implement the interface using JSP and some AJAX techniques. Some use of the database is SQL Server 2005 Enterprise Edition, corresponds to the operating system to Windows Server 2003 Enterprise Edition

3.1 *The basic flow of information*

extraction

WEB information extraction in four steps: (1) WEB crawl pages; (2) page clean; (3) data extraction; (4) data loaded. First crawled WEB page for the data source, and then use the web page pre-cleaning procedure to remove the tag and information related to produce structured HTML documents, HTML documents and then transformed into the final follow-up procedures can be identified so that the standard structure. Then use the program's interface to select the information to be collected for the program to provide samples. At the same time generate the corresponding data table with the sample. Automatically set by the sample extraction procedure rules, the use of the data extraction rules to crawl out from the HTML document, and ultimately saved into the database, SQL Server 2005.

Extraction flow chart:

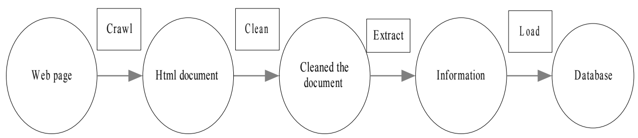

Fig 1 Extraction flow chart

3.2 *Web crawler*

Web crawler feature is the URL of a web page through the HMTL code extracted to a text document. The program will automatically access by entering the URL address of the URL domain name, you can traverse the web pages to ensure that the scope of information collected in the site. This can more effectively target a web site for informational extraction, extraction can also increase the efficiency and accuracy.

3.3 *Web preprocessor*

Pretreatment of the page in the whole process of information extraction is a very important part. A large number of pages of useful data often are disturbed by a lot of noise data, including advertising, navigation bar, copyright descriptions. Although these noise data for browsing on the Internet users have a certain function. But they also prevent the automatic web data collection and mining, including Web page automatic categorization, clustering, information extraction and information retrieval accuracy, efficiency and performance. Web pages include HTML code preprocessing correction and noise data filtering. Here I have divided into three pre-processing: HTML code fixes, URL processing and information processing.

3.4 *Wrapper*

Wrapper is a software process that has been defined using information extraction rules. The Web crawler to collect the information to the WEB page data extracted. Converted to the format， a specific description of the information. A wrapper points at a class of pages from a particular data source. Wrapper using the rules of the actual implementation of procedures to extract the data source extraction. Wrapper generally: extraction rules and the extraction device of two parts.

Figure Wrapper:

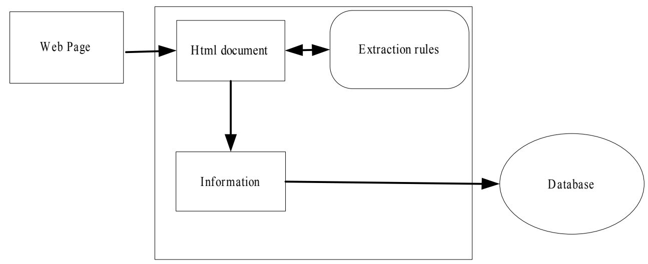

Fig 2 Wrapper

3.4.1 The development of extraction rules

Extraction rule describes rule-making, extraction step, the output methods. Extraction rules of the procedure are mainly composed of two aspects, one is the URL extraction rules, and the other is the information extraction rules.

a) URL extraction rules are very important and taken directly affect the efficiency of a part. Since this is a site for information extraction, although the website is limited to crawl a site, but the amount of pages are still very huge. A Web site URL must be on the extraction have a certain relevance to ensure the accuracy and efficiency of extraction.

b) informational extraction rule is given by the user to automatically generate a sample, so you can extract all the pages without having to make changes to the extraction rule.

3.4.2 The workflow of wrapper

When the user specifies a good URL that contains the information extracted, the wrapper will extract the URL for the rules. Then the user to specify sampling, wrapper and will rule extraction samples shall also record the number of samples used to automatically generate data tables, and waits for the user specified sampling again. When the extraction rules finalized, the program will be based on the number of selected articles extracted information to automatically generate the corresponding data table, then began to information extraction.

3.5 *Information Storage*

Extraction is based on information extraction rules to extract one by one, but the information you need to store the data table row by row insert. This resulted in that the extraction and storage are not synchronized. Therefore, data table generated by the number of fields of information are stored in the configuration table. Stored information, according to the number of field splicing SQL statement cycle, each time the rules of comparative information is a cycle. In order to ensure that the SQL statement is correct, so making the SQL statement complete at the beginning and end of the loop.

4 Data Maintenance & Chart Generation

Has been saved to the database of the data for further finishing, maintenance, and then use the Google chart API generates charts.

4.1 *Data Maintenance*

Taobao buyers mainly on the information, consumption records and product information, to view, modify and update.

B.Chart Generation

Through selecting and analysising goods data of web in a shop in Taobao, we obtain Taobao's transaction log information.

Taobao shop about "Doman flash cards ", which part of the transaction records are as follows:

Taobao account	Product name	price	amount	time
yxd001yw	al circle back Doman flash cards 【Dianka dig	89	1	2010/12/19 13:0
blueword20008	al circle back Doman flash cards 【Dianka dig	89	1	2010/12/18 9:3
tehchengsim	l flash card reader series: the English alpha	38	1	2010/11/24 20:5
kaka_happylife	al circle back Doman flash cards 【Dianka dig	89	1	2010/12/15 14:5
blueword20008	Encyclopedia of World Architecture Series Car	16	1	2010/12/18 9:3
me010909	Encyclopedia of World Architecture Series Car	16	1	2010/12/13 21:2
jessica811028	Encyclopedia of World Architecture Series Car	16	1	2010/12/3 15:2
zzg6066	Encyclopedia of World Architecture Series Car	16	1	2010/11/30 9:4

Fig 3 transactions

Taobao shop buyers' information about "Doman flash cards" are as follows:

Taobao account	Residence	buyer
yxd001yw	Baoshan District, Shanghai	Yu niuniu
blueword20008	Futian District, Shenzhen, Guangdong Province	MAYXU
kaka_happylife	Qingpu District, Shanghai	kaka
badujqu	Futian District, Shenzhen, Guangdong Province	silvia
laiyl	Dongguan City, Guangdong Province	Dun jianfeng
aaabbb8881	Ruian City, Wenzhou City, Zhejiang Province	Liao shanlian

Fig 4 Buyer's Information

Statistical data analysis generated by the chart:

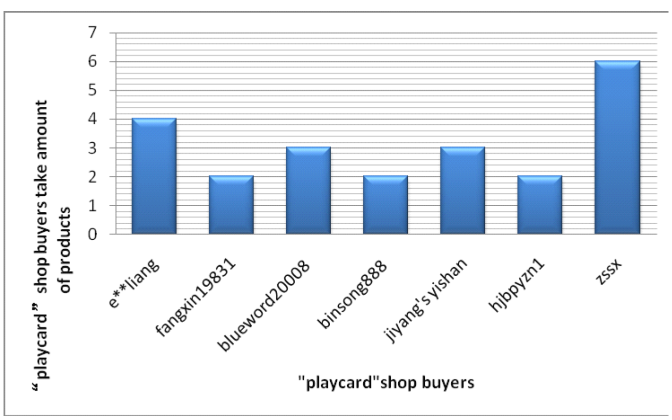

Fig 5 Data Statistics

5 Structure of the site design Based on B/S

Taobao customer management system framework shown in Figure:

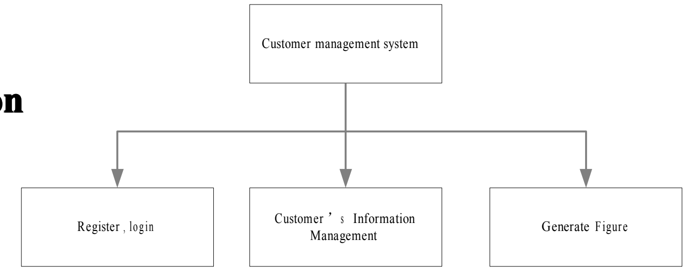

Fig 6 Framework of management system for customer

Charts shows on the page:

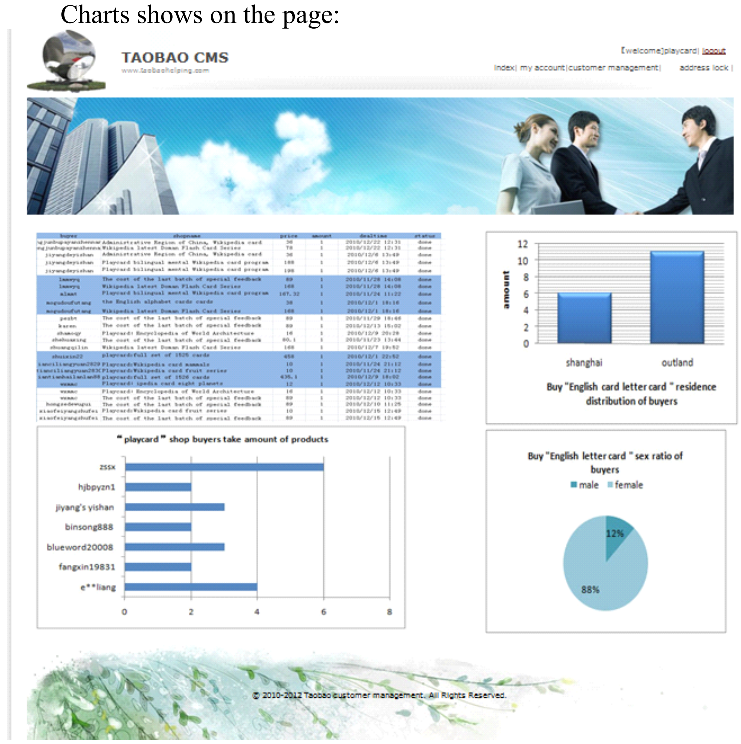

Fig 7 Data Analysis Chart

6 Conclusion

6.1 *Summary*

According to trading records, statistics generation business customers to purchase the type of goods, quantity, we can see the best customers, customer loyalty and purchase the product features that can be recommended to those customers interested in special offers and promotions products; Taobao buyers under information and statistics from their place of residence, then be able to live in dense local customers launch postage promotional activities; statistics of the gender characteristics of a commodity purchase, the merchant can adjust with the commodity related products supporting the recommended promotion.

6.2 *Outlook*

Statistical analysis using third-party tools are Taobao, buy to buy music and other C2C online shop on the platform to analyze the management of business customers, low cost, high efficiency, ease of use and strong. However, the web information extraction based on customer information management system, the main problem is still the information extraction accuracy and robustness to be improved, and information encryption and access issues. More limited functions of the system, but also towards diversification, comprehensive, and intelligent direction, there is great development space.

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KSU News Portal: A Case Study

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Abstract--**Information, now days, is a need instead of an option. The News Portal of King Saud University is an effort to provide information about the current happenings in the university nationally and internationally. The news portal has emerged as a reliable way of sharing and publicizing information across the world. The trust level for the news portal has increased with passage of time and the number of visitors visiting this site has also increased overwhelmingly. This research is focused to discuss the tools, technologies and practices used to implement this success story.**

Keywords: KSU News, News portal, KSU portal department, WordPress news portal.

I. INTRODUCTION

King Saud University, being the oldest and largest in the kingdom of Saudi Arabia, was established 1953 to provide quality education to the youth of this holy nation. The university has been blessed with the dynamic leadership since its origin and consequently that dynamic leadership has selected the prominent personalities to work on the key positions. As a recent initiative of the university a "Deanship of E-Transactions and Communication" has been formed to envision and implement the strategic decisions to enhance and extend the electronic environment by providing the services electronically. The deanship is working under the able supervision of Dr. Yousef Al- Ohali [5]. The deanship is further categorized in five departments to undertake specialized tasks. The portal department (PD) is one of the five departments of the deanship and works under the supervision of Abdul Aziz Al-Oraij. The PD [6]has envisioned and implemented state of the art electronic services to ensure that the university can quickly move towards the paperless environment and the ease of use and reliability of data is also increased at the same time to facilitate the information seekers.

In order to make the university visible globally an initiative was taken to launch a news portal of the university. The PD under the able guidance of its supervisors and with the support of the team of individuals that it has attracted in the past few years has implemented another very important service called KSU News Portal, which is responsible to publish and advertise the events and other happenings in the university as and when they happen. The portal was launched in 2008 and since that time the trust, security and archival has increased to make it a reliable source of information sharing.

The trust can be observed by the fact that almost all announcements, achievements and events are primarily advertised on this portal, where each college and administration unit can upload the information they want to make public.

II. SELECTION OF TOOL AND DEVELOPMENT

The choice of technology to implement a portal like this was a hectic process and a team of individuals dedicatedly identified three different tools, Drupal, Joomla and Wordpress. The tools were thoroughly studied and their characteristics were identified. Since beginning of this process, the team focused to identify an open source product so not it could not only be customized according to the requirements of this project but the open source community could also be benefited with this. An online comparison tool [1] helped the team to select the best out of available tools that could be used for the development of the news portal in King Saud University. The appropriateness of the tool was accessed based on 8 primary parameters including: System requirement, security, Support, Ease of use, Performance, management, Interpretability, Flexibility, Built-in Application and commerce. Based on these parameters, it was observed that

Wordpress is not only an open source platform but also can be more effectively used in establishing the huge solutions like this portal. Wordpress 3.0 [2] contains huge functionality as compared to the previous version (WP 2.0, WP 2.2, WP 2.3, WP 2.5, WP 2.7, WP 2.8 and WP 2.9) the new factures in functionality are discussed in this section. The improvement in the functionality can be categorized in four sub areas, namely general, dashboard, posts and media.

A. General

- Notify if the server environment does not meet the requirements to run php and mysql.
- wp_login_form() provides a simple login form
- New WordPress logo for admin header is provided
- It Place "Search Engines Blocked" mini-alert in the wp-admin header that keep monitoring that the user does not delist itself accidently.
- Quite a few menu items have been renamed, that include but not limited to Posts->Edit becomes Posts->Posts, and Links->Edit becomes Links->Links, etc.
- Administration menu icon has been re-considered before updation
- Gray Admin Color Scheme has been revised by updating that.
- The 'readme' file that provides the basics for any software/tool has been extensively revised and updated.
- Login form is displayed as a response to recover/reset email to help the user login.

B. Dashboard

- A new link has been added that becomes accessible after some core update have been made.
- The **Tools->Upgrade** menu option has now been moved to **Dashboard->Updates** and the user interface and themes have been updated and plug-in and core updates have been brought under one panel.
- New way of managing the menu by " menu management (navigation) "via Menus option under Appearances Menu has been introduced.
- Password nag has been provided for the newly registered users of the system.
- The "Right Now" widget has now been further categorized into Content | Discussion to describe possess more effectiveness.

C. Posts

- Has a quick edit mode which allows the removal of all tags.
- Allow themes to style the visual editor with editor-style.css file, using add_ editor_ style() function.
- Enable "custom taxonomy" User Interface for pages as well as for posts
- An interface has been provided that is used for the purpose of comparing the revisions of user interfaces.
- Shortlinks have been enabled, but still requires a plug-in or theme support to fully realize this features
- Added capital_P_dangit() filter to change 'Wordpress' to 'WordPress'.

D. Media

- Add FunnyOrDie.com oEmbed support
- Allow Deletion of Media Alt text
- Change Media User Interface labels from "Post Thumbnails" to "Featured Image"
- Cleanup of the edit media screen
- Don't let "Crunching" overlap image name while uploading
- Optimize scan facility is provided for the attachments that are lost during the process of uploading or later.

- Support for additional file type extensions is provided.

III. CHANGES FOR CUSTOMIZATION AT KSU

Along with the features provided in the wordpress 3.0, quite a few localized customization were expected to be provided I order to make it a more effective way of information dispention and communication. Each college and administrative unit is facilitated to upload the news about the current happenings in their domain with their authorized access on the portal, only in the prescribed area of relevance. The news on the portal can be commented by the individuals who require the authentication by the manager of the portal. Following features have been added in the wordpress 3.0, to customize it for the needs of KSU news portal.

A. Arabization

Arabization of the system As the university is located in the capital city of Saudi Arabia., where Arabic is widely used, it was deemed necessary that an Arabic version of the portal is also provided in order to enable people understand appropriately the contents of this portal in their own language.

B. Layout

The portal layout was built on the world usability standards where before implementing the standard layout a comparative study among the available standards was made and the best was chosen out of that. It was also considered that quite a few people with disabilities around the globe may be interested to view the contents of the portal. This portal is capable of supporting the people with disabilities to gain meaningful information from this portal. The layout of the website supports the Blind (or with restricted vision) people reading application where we make the system table free and make every element of the website associated with text alias, even the images are included with this feature. The feature has been incorporated with a great zeal and effort just to ensure that the academic community, including the students can be equally benefited with the information on the news portal. The layout is built to support people with low vision, color blindness, and genetic defects. A button gives the visitor the ability to increase/decrease the font size without losing the distorting the portal layout.

C. One click services

- The ability for the user to send the article as an email to anyone with one click.
- Print the article with one click and gives it a proper printing layout.
- Extracting a PDF image from the website for the article.
- Get a Word Document version with one click.
- Sharing: on facebook and twitter

D. Image Zoom

we implemented a jQuery code for getting the original sizes of the images in the website where you click on an image and it will give you the original size in a very smooth way and you can navigate between images using the keyboard arrows.

E. Integration

Integration between WP and the KSU Active Directory has been made and also have Implemented a SSL login where the login URL will automatically take you to a **secure** page to login to the system.

IV PRESENT STANDING OF THE KSU NEWS PORTAL

The KSU new portal has been among the news despite the fact that it actually is a source of originating and publishing the KSU news to the world. A comparative study was conducted in 2010 [3,4], to see the features that other news portals contain as compared to KSU News Portal, which yielded the results shown in Table 1

Main Strength Points	KSU News Portal	Harvard University	University of Saskatchewan
Layout:			
Low Visibility	☑	☒	☒
Text size Changer	☑	☑	☒
Support Blind	☑	☑	☑
RSS support	☑	☑	☑
RSS easy to find	☑	☑	☒
Front Page slide show	☑	☑	☒
Clear information spreading on the page.	☑	☑	☒
Images with every news article	☑	☑	☒
Image captions	☑	☑	☒
Every image has an alter	☑	☒	☒
Ability to share articles with other website with one click	☑	☑	☒
Clear distributed Categories	☑	☑	☒
Visitors Can comment on news	☑	☒	☒
Multi lingual Portal	☑	☒	☒
Diversity in the news	☑	☑	☑
Using Social Networks (Face book, & Twitter)	☑	☑	☒
Using video on the news portal	☑	☑	☑
News search support	☑	☑	☑

Table 1: The results of comparative study of three news portals

It can be observed that the KSU News Portal contains the highest functionality as compared to the news portals of other leading universities in the world. Thus KSU news portal may very promptly claim of being the portal with most functionality among the world.

As the results of study above suggest that the KSU News Portal has more functionality as compared to most of the news portals of leading universities, the amount of confidence and trust to this portal has also increased. This can also be depicted by the last six month statistics (Oct' 10 – Mar'11) about the web portal, in Figure 1 and Table 2.

Visitors Expand	256,436	+108%
Actions Expand	726,597	+156%
Average actions per visit	2.8	+22%
Total time spent	795d 3h	+112%
Average time per visit	4m 28s	+2%
Bounce rate	21%	+11%

Fig 1: The KSU News portal Statistics

	6 Month	1 Month	Daily	Hourly
Visitors	256,436	42740	1424	60
Action	726597	121,100	4036	168

Table 2: Time wise Portal Statistics

Conclusion

It can be concluded that the KSU News portal follows a state of the art technology for its development and includes a huge list of features out of which some are only present in the KSU News Portal. It has a documented way of allowing access to publish information and also a check that which comments can be made public. The Statistics show that the trust on this portal is increasing day by day. This can be seen by the increases in visitors and the number of actions per visit. It can further be concluded that this portal has been a huge source of making the KSU visible in national and international discussions.

Acknowledgement

The authors are thankful to the authorities in Portal department and Deanship of E-Transaction of the King Saud university, for sharing their valuable data and providing help in translation, explanation and understanding of contents.

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Implementation of Mobile based Hospital System using Web Services

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Abstract **-***By combining mobile computing and WebServices technologies, pervasive computing expects more portability and location transparency for accessing information in anytime from anywhere. Though, a direct integration of two technologies imposes performance limitations because of XML's verbose nature and physical limitations of mobile computing. We present hospital management system with mobile based solution which is helpful for making the appointment with doctors. The benchmark results show the performance advantage of using the architecture when a session composes a sequence of messages.*

Keywords: Mobile Computing, Web Service, Web Application Domain, Hospital Management

1 Introduction

Mobile computing gives a pervasive computing the way to access information any-time and anywhere by its portability and remote connectivity. And Web Service technology gives a pervasive computing the way to interoperate remote resources and diverse services. Because of their importance in the pervasive computing, it is a no big surprise that there are so many recent researches of adopting mobile computing as one
of flexible platform of Web Service technology. Even a half a decade ago, there is just handful of people who access remote information from their mobile device. The information access from mobile devices, however, has became easier than ever recently with the help from advanced mobile devices and widespread availability of packet switched, always-on cellular phone networks. There are many projects that try to adopt smart phones and cellular phone with data connections as major elements in Web Services, since huge synergy effect of interoperability and removing physical-location constraints are expected. However, the verbose nature of current XML based SOAP [1] approach imposes performance limitations in integrating mobile computing applications and conventional Web Services directly. SOAP achieves ubiquity by using highly universal XML as a form of data exchanging between disparate and distributed computing resources. Though, XML-based SOAP possesses three major characteristics that may affect SOAP performance. First, the in-memory data model must be converted to textual format to build a SOAP message object and to extract information from it. Secondly, because of inevitable mobile computing characteristics – high latency,

narrow bandwidth, limited computation, and small memory space, SOAP message processing consumes valuable resources [2]. Finally, mobile communications suffer from a larger data size by XML's descriptive tags and structure. It is usually not a problem on the powerful wired networks, although the bandwidth is pricey in mobile networks.

High performance SOAP encoding is an open research area [3], [4], and [5]. Web Services in mobile environment is a benefited of the researches, since it also need to overcome the performance limitations because of its characteristics above. Even the small size, regular frequency message exchanges could cause performance overheads in such an environment. In this paper, we present our new architecture design to achieve an optimized communication using binary message stream and a SOAP negotiation as well as the prototype implementation of the architecture and the performance benchmarks.

2 Background

We see several notable projects from industry and academia that try to overcome performance limitations of current Web Services approach. Extreme! Lab researched the limits of SOAP performance for scientific computing where large data sets including arrays are common and the design of a SOAP implementation suitable for systems with small memory and bandwidth [3], [4]. Throughout the experiments, the result of research shows the major improvements from using schema-specific parser mechanism for arrays, persistent connection, and streaming of messages to prevent full serializing objects to determine the length. It also shows that the most serious overhead is conversion to textual form from in-memory float numbers. To resolve the limitations, they recommend using multiple communication protocol incorporating with a binary representation and fast protocols other than SOAP. The condition they are facing with the conventional Web Services is similar to the constraint of mobile computing because of its limited computing environment characteristics.

Both need to overcome performance limitations of SOAP.

The report of the W3C Workshop [6] on Binary Interchange of XML Information Item Sets (Infoset) [7] is the result of the increasing demand of binary form of XML-based communication. The report includes conclusion of workshop meeting on September 2003 as well as several dozens of position papers from various institutes [5], [8], and [9]. The purpose of the workshop is to study methods to compress XML documents and transmit pre-parsed and schema specific object. It identified the requirement of binary XML Info set, for example 1) maintaining universal interoperability, 2) a generalized solution that is not limited to a specific application domain, 3) reducing process time including a data binding time, and 4) negotiation - fall back to XML/SOAP text format if receiver can't understand binary. The discussion leads W3C form XML Binary Characterization Working Group for further researches. Sun's Fast Web Services [5] and Fast Info set project [10] specifies a representation of an instance of SOAP Info set using binary encoding. They use Abstract Syntax Notation (ASN). 1. [11] to abstract encoded messages that may be encoded using it. The higher level protocols (WSDL [12] for contract definition of service etc.) remain unchanged, thus you could use standard SOAPXML for development, and have a switch that turns on the binary protocol for production deployment.

W3C XML Protocol Working Group released the draft of Message Transmission Optimization Mechanism (MTOM) [13] and XML-binary Optimized Packaging (XOP) [14]. Combined together, the specifications are targeted to two data type - multimedia data that already have standardized formats, such as JPEG, GIF, and MP3 and data that includes digital signature. The XML encoding would damage the data integrity. XOP is an alternate serialization that looks like a MIME package. It avoids data binding overhead, though still preserves XML structure – tags. Thus, XOP and MTOM, which describes how XOP is layered into SOAP HTTP transport, still possess a parsing issue inherited from SOAP/XML.

Cross Format Schema Protocol (XFSP) [9] is another project that serializes XML document based on schema. Initially it is motivated by the flexible definition of network protocols. It is written in Java and uses DOM4J model to parse the schema. With XML Schema-based Compression (XSBC) [15], XFSP provides binary serialization and parsing framework. Naval Postgraduate School provides lots of research on Streaming X3D documents in the XFSP framework.

Data Format Description Language (DFDL) [16] is a descriptive language that is proposed to describe a file or a stream in a binary format for Grid computing. Like Extensible Scientific Interchange Language (XSIL) [17], it is XML based and comes with an extensible Java Data model. DFDL define the structure of data. For example, it defines a number format of data whether it is a big-endian or little-endian and a complex data format such as an array. Also DFDL is designed to be process able through DFDL parser and data model. We designed the message format description of our Flexible Representation based on DFDL. In our Handheld Flexible Representation architecture, we define simple XML-schema based descriptive language and develop a language parser using XML Pull Parser (XPP) [18]. Our prototype implementation will not be in-depth like DFDL, though it will be enough to show advantages of our approach.

3 Service Oriented Architecture and Web Services

An SOA is essentially a collection of services to execute business processes. These services can communicate with each other. The communication can involve either simple data passing or it could involve two or more services coordinating some activity. Some means of connecting services to each other is needed. Those connections are web services.

Web service provides a standard means of interoperating between different software applications, running on a variety of platforms and/or frameworks. As defined in [3], a web service is a software system designed to support interoperable machine-to-machine interaction over a network. It has an interface described in a machine process-able format (especially WSDL). Other system interact with the Web service in a manner prescribed by its description using SOAP messages, typically conveyed using HTTP with an XML serialization in conjunction with other web-related standards. The web service architecture is interoperability architecture; it identifies those global elements of the global web service network that are required in order to ensure interoperability between web services.

Web services technologies provide a language-neutral, environment-neutral programming model that accelerates application integration inside and outside the enterprise. Application integration through Web services yields flexible loosely coupled business systems.

Web services technologies provide a language-neutral, environment-neutral programming model that accelerates application integration inside and outside the enterprise. Application integration through Web services yields flexible loosely coupled business systems.

A Web service is a collection of functions that operate as a single entity and are published to the network for use by other programs. Web services are building blocks for creating open distributed systems Web services are basically designed to allow loose coupling between client and server, and they do not require clients to use a specific platform or language. In other words Web services are language neutral. Mainly for these reasons among others, these services are becoming very popular.[19]

Web services are well defined, reusable, software components that perform specific, encapsulated tasks via standardized Web-oriented mechanisms. They can be discovered, invoked, and the composition of several services can be choreographed, using well defined workflow modeling frameworks.

A Web Service is a software program identified by an URI, which can be accessed via the internet through its exposed interface. The interface description declares the operations which can be performed by the service, the types of messages being exchanged during the interaction with the service, and the physical location of ports, where information should be exchanged.
Following figure shows the overall architecture for the web service and how it is used for communication into web domains.

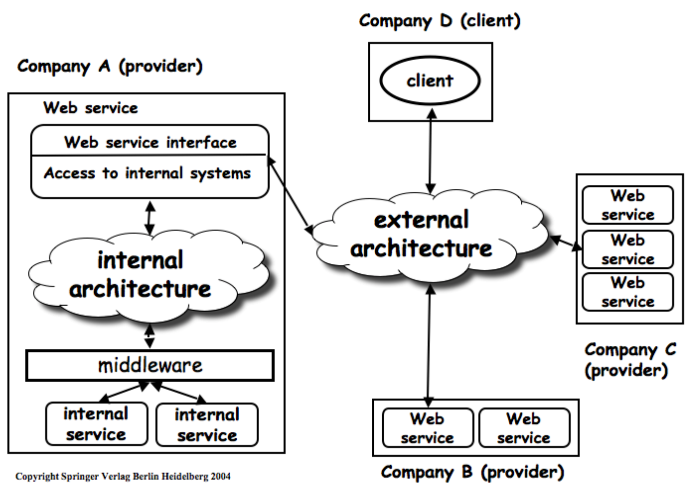


Figure-1: Web Service architecture

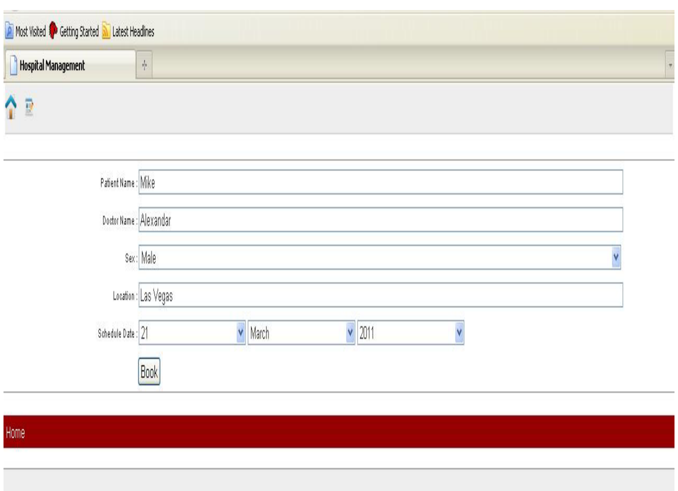
Here it shows the how the different web services can be used into different applications and how they are communicating with each other for passing information.[19]

The same thing we are demonstrating with the developed prototype for the airline reservation application which can be used by the client and which ultimately called the different web services from different domains and check the appropriate information and it found then returns the desired output.

4 Design Overview

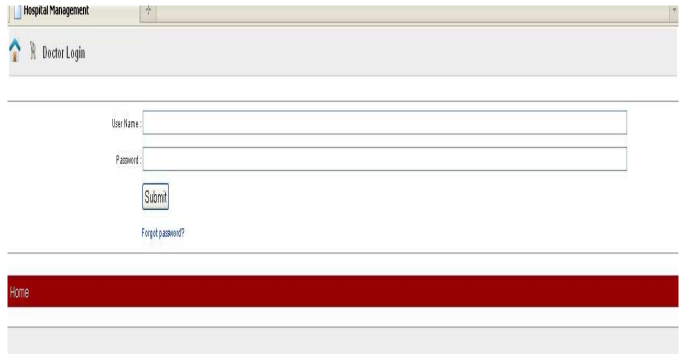
System is design to give the idea regarding the mobile based hospital management system developed for making the schedule with doctors and get the idea about whether appointment is fixed or not. The system is developed using the web services. Web services are such a handy thing that

could even make the system more stable and more efficient to work.
Here different classes have been created for making the schedule with doctors. Here the patient can open the applications and search for the available free slots for doctors and if the slot is available then the appointment is fixed. Here we used web service to get the information for available doctors and once we get that information then it is displayed to the user and user have to make the necessary selection to further process. As we all know about web service as it follows the XML and basic communication language so whether the application is developed into another language it does not matter. That's another advantage of using web services and its provides far efficient way compare to another languages.



The above screen shows the schedule form to book the appointment with the doctor. User need to enter certain information like User/Patient Name, Doctor Name, Sex, Location, Appointment Date etc. Once all the information has been filled up then user can submit/book the appointment.

Following screen shows the login screen for the doctors. Doctors can check how many appointments have been made today. Doctor can also get the future appointments list so that they can prepare in advance.



Following screen shows the schedule appointments with Doctor Alexander on 21st March 21, 2011. It shows the list of available patients who booked for Dr. Alexander on 21st March, 2011. Doctors can also search the booked schedule. The advantage of using this system is that doctor can check the scheduled appointments and also check how many patients are booked for the appointments, so that doctors can prepare in advance.

Using web services into mobile provides the advantages which can resolve several issues which requires time consuming and sometimes it also requires to go physical place.

5 Conclusion

We have developed prototypes to analyze the importance of web services in mobile applications and observed that it may be quite useful tool to provide the communication. It reduces the work from user side as well as from the doctor side. It also reduces the efforts and hence it improves the efficiency.

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20. Impact of Web Services in Web Domains by Dr. Samrat Khanna and Mijal Mistry

SESSION

TOOLS + CLOUD COMPUTING + AGENT TECHNOLOGY + ONTOLOGY + DATA MINING + APPLICATIONS AND ALGORITHMS

Chair(s)

Prof. Hamid R. Arabnia

Legal Feasibility of Automated Data Collection for Statistical Purposes in the EU

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Abstract. Internet-based measurement is using Internet as source of data gathering and it is a method of automated data collection. The three most common Internet-based measurement approaches are user-centric, network-centric and site-centric measurements. User-centric relies on in-depth analysis of behaviour of users by installing software and application; network-centric measures traffic flows between users and content throughout the network; and site-centric collects data from one to more websites. Internet as a source of data gathering could lower the costs and increase the speed of data collection for statistical purposes compared with traditional manual methods. Statistics analysis is important as it may contribute to value added service. On the other hand, data privacy rights may be at risk under such measurement approaches if technical measures for data security are not appropriate or users did not give prior consent to the use of such data. This paper discusses the current EU data privacy protection legislation and analyses the overall legal feasibility of the development of Internet-based measurements with regard to automated data collection for statistical purposes by looking into the detail of the reform of the new EC Directive on e-Privacy and the current review of the EC Directive on Data Protection.

Keywords: Data Privacy Protection, Automated Data Collection, Statistical Research

1 Current EU Legal Framework on Online Privacy

"Internet-based measurement is a set of methods that have been applied to quantitatively describe the structure, workload and use of the Internet. They provide a practical means of doing a kind of virtual 'fieldwork' on the Internet using online tools and network monitoring techniques to gather fine scale primary data, as opposed to relying on aggregate secondary data sources (such as government statistics)."[1]

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[1] International Encyclopaedia of Human Geography, MS number: 457.

In other words, Internet-based measurement is using Internet as source of data gathering and it is a method of automated data collection. The three most common Internet-based measurement approaches are user-centric, network-centric and site-centric measurements. User-centric relies on in-depth analysis of behaviour of users by installing software and application; network-centric measures traffic flows between users and content throughout the network; and site-centric collects data from one to more websites.[2] Internet as a source of data gathering could lower the costs and increase the speed of data collection for statistical purposes compared with traditional manual methods. Statistics analysis is important as it may contribute to value added service, which may, for example, "consist of advice on least expensive tariff packages, route guidance, traffic information, weather forecasts and tourist information".[3] On the other hand, data privacy rights may be infringed under such measurement approaches if technical measures for data security are not appropriate or users did not give prior consent to the use of such data.

Due to the ever fast-growing technology, legislation is always one step behind the latest invention of computing network services. This leads to a situation where computer scientists and entrepreneurs try to adjust or improve the application of products in order to comply with the existing law, or legislators try to amend the existing law in response to the new technology in order to protect the users' rights and enhance the public safety without jeopardising technological innovation and market development. Currently, there are three main pieces of legislation concerning data and privacy protection in the European Union (EU): 1) EC Directive on Data Protection in 1995;[4] 2) EC Directive on e-Privacy in 2002;[5] and 3) EC Regulation on Personal Data Protection in 2001.[6]

The EC Directive on Data Protection has been under the review of European Commission since 2009. In 2011, the Commission will propose a new general legal framework for the protection of personal data in the EU, covering data processing operations in all sectors and policies of the EU. This framework will then be negotiated and adopted by the European Parliament and the Council. The EC

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[2] Go with the Dataflow, available at http://www.umic.pt/images/stories/publicacoes1/annexes.pdf (last visited on 18 April 2011).

[3] Recital 18 of the EC Directive on e-Privacy.

[4] Directive 95/46/EC of the European Parliament and of the Council of 24 October 1995 on the protection of individuals with regard to the processing of personal data and on the free movement of such data, OJ L 008 , 12/01/2001 P. 0001 – 0022.

[5] Directive 2002/58/EC of the European Parliament and of the Council of 12 July 2002 concerning the processing of personal data and the protection of privacy in the electronic communications sector (Directive on privacy and electronic communications), OJ L 201 , 31/07/2002 P. 0037 – 0047.

[6] Regulation (EC) No 45/2001 of the European Parliament and of the Council of 18 December 2000 on the protection of individuals with regard to the processing of personal data by the Community institutions and bodies and on the free movement of such data, OJ L 008 , 12/01/2001 P. 0001 – 0022.

Directive on e-Privacy has been amended by the Directive 2009/136/EC. Member states are required to implement the new EC Directive on e-Privacy by May 25, 2011.[7]

To understand the legal feasibility of the adoption of the three common measurement approaches for statistical research, basic legal concepts that are arisen from the approaches shall be firstly identified and interpreted, for example, the definitions of personal data and sensitive personal data. Personal data is defined as "any information relating to an identified or identifiable natural person ('data subject'); and identifiable person is one who can be identified, directly or indirectly, in particular by reference to an identification number or to one or more factors specific to his physical, physiological, mental, economic, culture or social identity",[8] whilst sensitive personal data could be understood as "personal data revealing racial or ethnic origin, political opinions, religious or philosophical beliefs, trade-union membership, and the processing of data concerning health or sex life".[9]

The possibility of the implementation of automated data collection for statistical research in business organisations and statistical institutions depends on the feasibility of legal compliance. It is debatable whether automated data collection for statistical research is allowed under the current EU data privacy legislative framework. This paper intends to provide an overview of the EU data privacy protection legislation and discuss the overall legal feasibility of the development of Internet-based measurements with regard to automated data collection for statistical purposes by looking into the detail of the reform of the new EC Directive on e-Privacy and the review of the EC Directive on Data Protection.

2 Necessary Legal Compliance for Automated Data Collection for Statistical Purposes

According the current EU data privacy protection legal framework, there are four underlying steps in the EC directives that intend to ensure that privacy rights are put into action:

1) Member states should take appropriate technological and legislative measures to safeguard security and ensure the protection of personal data and privacy.

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[7] Directive 2009/136/EC of the European Parliament and of the Council of 25 November 2009 amending Directive 2002/22/EC on universal service and users' rights relating to electronic communications networks and services, Directive 2002/58/EC concerning the processing of personal data and the protection of privacy in the electronic communications sector and Regulation (EC) No 2006/2004 on cooperation between national authorities responsible for the enforcement of consumer protection laws (Text with EEA relevance), OJ L 337, 18.12.2009, p. 11–36.

[8] Article 2(a) of the EC Directive on Data Protection.

[9] Article 8(1) of the EC Directive on Data Protection.

2) Service providers have a legal duty to inform users prior to obtaining their consent.

3) Service providers shall allow users to give and withdraw their consent freely as users have “the right to be forgotten”. It is debatable what constitutes a meaningful consent and whether “privacy by default” is sufficient.

4) Member States shall enhance enforcement of data privacy protection because any legislative and technological measures to protect users’ privacy can only be effective if they are properly implemented and enforced. EU citizens’ data privacy rights should be protected equally no matter where the service provider and data are situated. The service provider shall duly notify data breach and take appropriate measures to avoid escalation of the problem.

Among the above steps, the rightful implementation of consent determines the lawful processing of data. Member states may understand the valid form of consent differently, for example, the UK law interprets consent as ‘reasonable grounds for believing’ that consent to do, which do not comply with EU rules defining consent as "freely given specific and informed indication of a person’s wishes".[10] Under the new EC Directive on e-Privacy, the use of cookies requires users’ prior consent. Article 29 Working Group on Data Protection addressed that “currently three out of the four most widely used browsers have a default setting to accept all cookies. Not changing a default setting cannot be considered as a meaningful consent.”[11] It is expected that there are various interpretations by member states when the new EC Directive on e-Privacy comes into force in May 2011.

With regard to data collection for statistical purposes, the exemption clause of prior consent remains unchanged in the new legislation. In general, data can be processed solely for the purpose of **scientific research** or kept in personal form for a period which does not exceed the period necessary **for the sole purpose of creating statistics**[12] subject to the implementation of conditions:

1) adequate legal safeguards – the data are not used for taking measures or decisions regarding any particular individual;

2) clearly no risk of breaching the privacy of the data subject;

3) data kept only for necessary period and employ other appropriate safeguards provided by member states.

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[10] Recital 17 of the EC Directive on e-Privacy.

[11] “Opt-out is not sufficient”, European Commission Press Release, 24 June 2010, available at http://ec.europa.eu/justice/policies/privacy/news/docs/pr_26_06_10_en.pdf (last visited on 15 April 2011).

[12] Recital 29, 39 & 40 and Article 11(2) & Article 13 of the EC Directive on Data Protection.

That is, automated data collection from the Internet for statistical purposes could be legitimately processed, provided that they fulfil the above three basic conditions except for the category of processing sensitive personal data that needs to meet the additional condition of public interest.[13] There are four different layers of data collection that could possibly be used for statistical purposes: first, general personal data; second, further processing of personal data previously collected; third, data that are not obtained from the end users directly; and fourth sensitive personal data.

As to the last but not least important issue – enforcement of data privacy protection, service providers shall duly notify data breach to the competent national authorities and take appropriate measures to protect data privacy. In the author's opinion, the interpretation of "without undue delay" for data breach notification under Article 4 of the EC Directive on e-Privacy is vital as the timing affects the certainty of data-privacy protection. The determination of the appropriation of time limit on notification and remedial action shall be taken into account of the speed, scope and capabilities of spreading personal data under the current and future development of technologies in particular automated information systems. In addition, the consideration of the time-limit issue for notification and remedial action can be learned from the interpretation of the time-limit requirement on the exercise of the right to access in Article 12(a) of the EC Directive on Data Protection regarding information storage and disclosure in the case of *College van burgemeester en wethouders van Rotterdam v M.E.E. Rijkeboer Netherlands* (judgement of 7 May 2009). The judgement provides that:

> "Article 12(a) of Directive 95/46/EC of the European Parliament and of the Council of 24 October 1995 on the protection of individuals with regard to the processing of personal data and on the free movement of such data requires Member States to ensure a right of access to information on the recipients or categories of recipient of personal data and on the content of the data disclosed not only in respect of the present but also in respect of the past. It is for Member States to fix a time-limit for storage of that information and to provide for access to that information which constitutes a fair balance between, on the one hand, the interest of the data subject in protecting his privacy, in particular by way of his rights to object and to bring legal proceedings and, on the other, the burden which the obligation to store that information represents for the controller.
>
> Rules limiting the storage of information on the recipients or categories of recipient of personal data and on the content of the data disclosed to a period of one year and correspondingly limiting access to that information, while basic data is stored for a much longer period, do not constitute a fair balance of the interest and obligation at issue, unless it can be shown that longer storage of that information would constitute an

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[13] Recital 34 and Article 8 of the EC Directive on Data Protection.

> excessive burden on the controller. It is, however, for national courts to make the determinations necessary."

Accordingly, it shall be for Member States to fix a time-limit for notification of the personal data breach and remedial action. Where the length of time for which a personal data breach is to be informed to the competent national authority or remedial action is to be taken is very long, the adverse effects of the breach of the personal data or privacy of a subscriber or individual may be higher as the implementation of appropriate technological protection measures may be delayed. The issue of a fixed time limit for notification and remedial action shall be further assessed when the Commission examines the modalities for the introduction in the general legal framework of a general personal data breach notification, including the addressees of such notifications and the criteria for triggering the obligation to notify according to the EU Comprehensive Approach 2010. The obligation of a time-limit for notification of data breach shall also be contained in the future EU standard forms of "privacy information notices".

3 Conclusion and Recommendation

It is important to strike the balance between data privacy rights protection and the free movement of data within member states in order to build users' trust on the Internet without jeopardizing technological innovation and market development. The recent European Commission review on the EC Directive on Data Protection has paid attention to that.[14] Statistical methods on Internet as a source of data gathering could provide statistical outcomes faster than the traditional paper-based questionnaire methods. Using Internet as a source of data gathering could also collect data that is difficult or even impossible to be gathered in the offline world. The implementation of Internet-based measurements could bring us great added value to improve products and services and allow us to promptly respond to the market development. From a legal perspective, the successful implementation of the statistical methods on Internet as a source of data gathering depends on the appropriate use of the exemption clause, the correct operation of informing users and requesting consent where necessary and the strict compliance of lawful data storage and data breach notification system. The building of automated data collection systems for statistical purposes has to comply with appropriate legislative and technological measures.

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[14] "A comprehensive approach on personal data protection in the European Union" (known as "the EU Comprehensive Approach 2010") – Communication from the Commission to the European Parliament, the Council, the Economic and Social Committee and the Committee of the Regions, European Commission, Brussels, 04.11.2010 COM(2010) 609/3.

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Directive 95/46/EC of the European Parliament and of the Council of 24 October 1995 on the protection of individuals with regard to the processing of personal data and on the free movement of such data, Official Journal of the European Union, OJ L 281, 23 November 1995, P. 0031–0050 (known as "EC Directive on Data Protection").

Directive 2002/58/EC of the European Parliament and of the Council of 12 July 2002 concerning the processing of personal data and the protection of privacy in the electronic communications sector (Directive on privacy and electronic communications), Official Journal of the European Union, OJ L 201, 31 July 2002, P. 0037–0047 (known as "EC Directive on e-Privacy");

Directive 2009/136/EC of the European Parliament and of the Council of 25 November 2009 amending Directive 2002/22/EC on universal service and users' rights relating to electronic communications networks and services, Directive 2002/58/EC concerning the processing of personal data and the protection of privacy in the electronic communications sector and Regulation (EC) No 2006/2004 on cooperation between national authorities responsible for the enforcement of consumer protection laws, Official Journal of the European Union, OJ L 337/11, 18 December 2009, P.0011 – 0036;

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Network Topology Analysis in the Cloud

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***Abstract*—This paper demonstrates how Cloud resources can be used for scientific calculations in a cost effective manner. The target audience consists of researchers who have to calculate a large amount of data. The paper does not deliver new scientific results in the traditional way in the area of Cloud computing, but explains the economic and technical advantages and disadvantages of Cloud computing for use in science. It is meant to foster the use of cloud resources as a cost effective way to run scientific analyses. For this, a cost estimation is given for an example implementation. A running example originating in the research area of computer network optimization is explained from the technical background to the final implementation.**

***Index Terms*—Cloud computing, network analysis, scientific computing.**

I. Introduction

During empirical studies researchers typically have to analyse large amounts of data. In this paper we use an actual calculation needed in an ongoing scientific project to investigate if commercial cloud providers could be an alternative to local high performance computers or computing clusters. The calculations used as case study take time series of network parameters originating from a large wireless mesh network as input and deliver several quality indicators.

The remainder of this paper is organized as follows. Section II explains the technical and functional background of the example application used in this paper, section III compares two possible cloud service levels and gives a short overview about available commercial and academic cloud providers. In section IV the example implementation is discussed as well as costs and performance are assessed. Section V concludes this paper and suggests further activities.

II. Functional background of the example application

This section explains the functional background and gives and overview about needed calculations and data sets collected for examination.

A mesh network basically consists of nodes that share a wireless communication channel. Every node in the network forwards data packets for other nodes according to a routing scheme to its neighbours. This decentralized structure allows cost-effective networks but requires a sophisticated routing mechanism.

The mesh network [7] under investigation uses Optimized Link State Routing (OLSR) [12]. OLSR is a pro-active link-state routing protocol. The mode of operation of pro-active link-state routing protocols requires that current information describing the complete topology is available at every node before data packets are being sent. Hence, a large amount of topology information have to be permanently distributed through the entire network. Using this topology information all nodes are able to calculate paths for outgoing data packets themselves.

In OLSR every router broadcasts topology control (TC) messages with a pre-configured update rate. These TC messages consume a non-neglectable share of the available throughput and generate interference in a larger mesh network. The final goal of the parent research project is to reduce the update rate and pre-compute topology changes, which might be caused by recurring events, such as human behaviour (daily routine) or natural phenomenons.

A. Data set

We use historic data describing the topology at certain times to analyse those phenomenons. For this purpose the routing situation at every time stamp in a long term time series needs to be reconstructed. TC messages are available at every node participating in the network. Delays caused by the propagation of TC messages are inevitable. Hence, all data represent the current situation at the point within the network where all data is being collected.

According to OLSR principles every link between two nodes is attributed by two parameters, link quality (LQ, from current node to neighbour node) and neighbour link quality (NLQ, reverse direction). Note that links are not considered to be symmetric. Both values describe the probability that a single data packet reaches the node at the opposite side of the link. Probe packets (“Hello” messages) are used to determine the link quality.

Data containing LQ and NLQ values for every available link have been snapshotted every minute for several months. The routing algorithm uses this topology information and chooses the best paths from source to destination according to LQ and NLQ values. More detailed, the expected number of transmission attempts needed before a packet reaches its destination (expected transmission count - ETX) is used as metric.

Currently the data collection consists of about 195 million tuples each representing a single link at a certain point in time. [1] The data base containing these 195 million tuples has a size of about 22 gigabyte. About 240.000 snapshots (distinct time stamps) are available by now. On average there are 180 nodes

[1] All data and software used in this paper is available for download from open science repository co-maintained by the authors at http://opsci.informatik.uni-rostock.de/.

and 810 available links forming the mesh network at every point in its history. Each tuple represents one link at a time and has the following structure:

Attribute	Meaning
timestamp	Timestamp of snapshot, same for all links being active at the time of taking the snapshot.
thisNode	Name (IP address) of local node
otherNode	Name (IP address) of remote node
lq	Link quality from thisNode to otherNode (ratio of received OLSR "Hello" messages to sent probe messages)
nlq	Link quality from otherNode to thisNode

TABLE I
MEANING OF DATA ATTRIBTUES.

A short example for several links looks like this (II:

timestamp	thisNode	otherNode	lq	nlq
2010-09-10 00:00:02	B 192.168.1.129	A 192.168.0.254	0.780	0.800
2010-09-10 00:00:02	B 192.168.1.129	E 192.168.1.14	1	1
2010-09-10 00:00:02	B 192.168.1.129	D 192.168.1.15	0.450	0.450
2010-09-10 00:00:02	E 192.168.1.14	D 192.168.1.15	1	1
2010-09-10 00:00:02	E 192.168.1.14	C 192.168.1.2	0.800	0.800
...	...	...	...	...
2010-09-10 00:01:02	B 192.168.1.129	A 192.168.0.254	0.820	0.800
2010-09-10 00:01:02	B 192.168.1.129	E 192.168.1.14	1	1
2010-09-10 00:01:02	B 192.168.1.129	D 192.168.1.15	0.550	0.500
2010-09-10 00:01:02	E 192.168.1.14	D 192.168.1.15	1	0.980
2010-09-10 00:01:02	E 192.168.1.14	C 192.168.1.2	0.990	1
...	...	...	...	...

TABLE II
EXAMPLE DATA SETS DESCRIBING SEVERAL LINKS AT TWO DIFFERENT TIMESTAMPS.

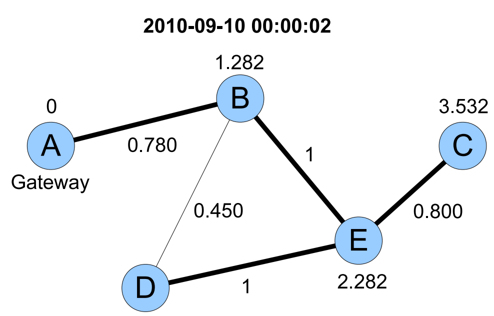

Fig. 1. Network condition at a point in time. For simplicity only LQ values are attached to links. Bold lines show used links.

B. Calculations

All tuples in a snapshot (those tuples carrying the same time stamp) together describe the network topology as seen by the node where all data are being collected. The network has several gateways to the Internet. Internet usage is by far the main usage of the mesh network. TC and "Hello" messages are broadcast by every node without routing. So, basically all other traffic is routed towards Internet gateways and in reverse direction towards the nodes directly serving users. The quality of the path between user node and gateway determines

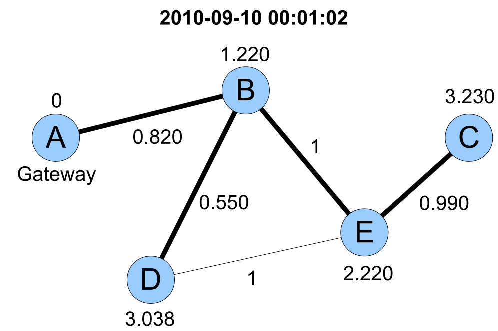

Fig. 2. Network condition at another point in time. For simplicity only LQ values are attached to links. Bold lines show used links. Routing has changed due to a change of link quality.

the quality of the Internet connection. There have been four gateways during the data collection campaign. Every node is assumed to connect to the nearest gateway in terms of ETX metric.

With computing the path quality to and from all available gateways for all active nodes and for every snapshot we hope to find recurring patterns usable for later predicting the network status. For this purpose the path quality to all gateways is being computed. At first, the shortest paths according to the ETX metric is calculated towards all gateways with the traditional Dijkstra algorithm. This delivers the same paths as for real data packets during at the time of data collection. As result every node is labeled with four path ETX values. Additional quality describing values such as hop count to the gateways, and path link quality are also computed.

The following estimation illustrates the complexity of calculations. The database contains 240.000 snapshots (growing). Every snapshot contains 810 available links. Most of them are never becoming active links, but all of them have to be considered when the shortest paths are calculated. Those 810 available links connect about 180 active nodes. There are four gateways to be independently considered. For every node the shortest paths (in terms of the ETX metric) to all four gateways have to be computed. The path ETX has to be stored with every node for each of the four gateways. The minimum ETX among those four ETX values represents the node's connection quality towards the Internet.

Calculating path ETX values for one snapshot and one gateway takes about 900ms on a local computer (Intel i7 CPU, sufficient RAM to hold all data, algorithm implemented in Java and python). For all 240.000 snapshots and four gateways it would take an estimated 240 hours computation time ($0.9s \cdot 240,000 \cdot 4$). The process as a whole has to be repeated when routing parameters or other aspects of the analysis are changed, as if they are used in a simulation. This is was not acceptable for a detailed analysis. The calculation process is parallelizable (distributable) in several dimensions. Therefor the calculation time can be decreased by using more hardware. As a feasibility study a cloud based implementation was preferred over a local cluster.

III. Cloud computing used for research

This section introduces relevant concepts of cloud computing. It is not meant to give a full overview about cloud providers, but shows significant advantages and possible pitfalls.

The term “Cloud computing” as used in this paper refers to computing resources available on the Internet. In most cases resources are leased to customers. Several levels of services are available for leasing. Applications are deployed to unknown locations. “Cloud computing” is a new form of distributing software, using resources, and delivering services on the Internet. A major advantage of using clouds is that applications are highly scalable since most cloud computing infrastructures deliver services through a network of computing centers and built on a large number of servers. These servers can be used in a distributed way for scientific calculations when algorithms are parallelizable.

From an economic point of view cloud computing could decrease investment costs for computational complex analyses and simulations, especially when computing power is needed for short periods only. In these cases building own infrastructure that idles most of the time is uneconomic.

Cloud providers can deliver services on different levels, which are usually referred to as Platform as a Service (PaaS), Infrastructure as a Service (IaaS), and Software as a Service (SaaS). PaaS and IaaS are most suitable for scientific problems as they facilitate deployment of applications without the cost and complexity of buying and managing the underlying hardware and in case of PaaS also the underlying software. A more general overview about cloud technologies and different service levels is given in [21]. Concrete examples can be found in the following sections.

A. Platform as a service and Infrastructure as a service

As mentioned before, mainly two service levels are available, Infrastructure as a service and Platform as a service. Basically both are suitable for the example network calculation described in this paper.

The main difference is that using Infrastructure as a service requires to set-up a computer as it would happen to a local machine, including operating system, libraries, and software, while using a Platform as service requires deploying of software to the platform only. But, the latter approach limits resources to those provided by the platform.

Amazon EC2, as an example for Infrastructure as a service, provides several pre-configured images as a starting point for configuration. After installing the software the prepared image can be started several times. This is controlled by the user through web services. Each instance works with its own separated resources. Amazon EC2 provides different instance types (machines) with different hardware configurations, for example high memory, high CPU, or cluster compute instances. The granularity of adding and removing resources is limited, since an entire instance has to created or destroyed at once. Additionally, different quality of service levels can be purchased, for instance the so-called spot instance which only runs when the spot price for computing capacities is below a threshold. This is useful since scientific calculations as described here could be computed during off-peak times in order to save money.

Google App Engine, as an example for Platform as a service, hides the infrastructure from users. Instances of user applications are created on-demand. Separate virtual machines are indistinguishable for users and applications, and, hence, neither have to be set-up, created, or destroyed. Resources are provided to each application instance. The application developer has to regard the APIs provided by the platform. Additional resources beyond those API are not available. In the concrete case of Google App Engine the service is provided through HTTP, which imposes limits on calculation time.

A general recommendation for the application of either PaaS or IaaS for scientific computing cannot be given. IaaS offers more variability while PaaS reduces the effort to maintain the system. General criteria are price, performance, and ease of development and use. The task of calculating network parameters used as example in this paper could be fulfilled with both designs.

B. Scientific and commercial cloud providers

A brief overview about commercial cloud providers can also be found in [8]. It is obvious that major Internet companies also provide cloud services. This is reasoned by the history of cloud computing. Most commercial providers of the first generation have developed cloud services to sell their excess capacity in computing centers to others.

Larger commercial cloud providers are Amazon Elastic Compute Cloud (EC2), Google App Engine (GAE), and Windows Azure (WA). All of them offer services on a per-use basis, which is common for cloud computing.

Although universities and other research bodies maintain large computer farms and although several initiatives have been started to provide cloud services for scientific calculations currently no cloud-like computing environments or only prototypes are available for external researchers.

Scientific cloud computing is discussed since around 2008 [25]. For this purpose both non-commercial and commercial cloud providers are considered [14] [17] [16]. Examples for non-commercial initiatives in North America are ScienceClouds [24] and a NSF project [15], [27]. ScienceClouds, for instance, is able to provide cloud services at the same interface as Amazon Elastic Compute Cloud.

C. Available cloud computing resources

For the purpose of analysing large amounts of data the Cloud provider must at least offer facilities to compute and store data. Input-output-functionality is needed for transferring data into the cloud and downloading results.

Amazon EC2 and Windows Azure mainly provide a basic infrastructure. Typically this includes a set of virtual hosts. On those hosts users can install software as needed. Typically file system images are used for this. Scaling the number of virtual hosts up and down is a basic function of the cloud to adapt resources to changing needs. Developers can use memory, CPU, and network connections on a virtual system. They are free to install frameworks and use programming languages at their discretion. In case of Amazon EC2 developers can choose among several operating systems and different instance types. Instance types are available from so called micro instances up to cluster instances [1].

Persistent storage has to be provided by the developer through conventional relational database management systems. Alternatively, Amazon Simple Storage Service (S3), Amazon SimpleDB, or Amazon Elastic Block Store (EBS) can be used, all three are very well integrated with Amazon EC2.

With Eucalyptus [20] an open source platform is available for the implementation of private cloud computing on computer clusters. This means, Eucalyptus allows owners of hardware to provide Infrastructure as a service.

Contrary to that, Google App Engine provides a closed platform which hosts the final application. The platform consists of a set of APIs which enable the application to use a variety of services such as authentication, fetching web pages via HTTP, sending e-mails, and persistence of data. Applications have to be written in Python or Java and will be executed in a sandbox. Data storage is provided by Google through its Datastore service [9]. Applications are deployed and run as web service or servlet, both of them providing an HTTP interface. This limits compute time to one request-response-cycle.

The example used in this paper (calculating routes and network quality values) can be easily split into small sub-tasks. For this reason the time limit for processing HTTP requests is not an issue. An alternative way to deal with that limitation are task queues [22]. A task queue collects background work that is organized into smaller units. Tasks are executed when cloud provider system resources permit.

An open source platform (PaaS) that is compatible with Google App Engine and which can be deployed into an IaaS cloud such as Amazon EC2 or Eucalyptus (see above) is available with AppScale [10].

D. Data storage and data retrieval

Traditional database management systems can be used within Amazon EC2 and Windows Azure beside file based concepts whereas Google App Engine only provides a non-relational (NoSQL) [19] [18] storage concept called Google datastore [11]. The advantage of those storage is that it can easily be distributed among instances. A disadvantage is that queries are limited to selections (choosing data sets). Projections (choosing a set of attributes) and aggregations (for instance calculating sums or grouping values) are not permitted. Application developers should be aware of those disadvantages when designing software. In some cases operations such as grouping values have to be performed within the application itself instead of leaving this to the database.

In the concrete example the datastore itself could not deliver a list of timestamps since this would require to group several data sets by distinct attributes. Building this list require either a separate table (actually it is not a table in the sense of SQL because not all attributes have to be present for all data sets) could be used or the entire datastore would have to be read.

Scientiest should also consider to upload their data into public data sets, such as discussed in [4]. This collection of data is sponsored by Amazon and made available to cloud applications via the Amazon S3 API.

E. Application control, security, and costs

Cloud users have to control the resource usage of their applications. For scientific computing this includes to set the number of instances, access rights, scheduled tasks, and financial limits. The framework reports for instance the current load, throughput, amount of used and remaining storage space, used CPU time, and all other billable resources. Google App Engines for instance provides a web based control center, see Figure 3.

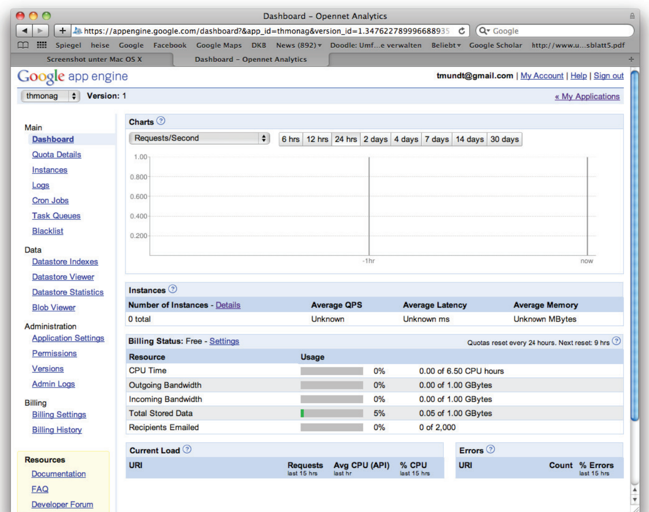

Fig. 3. Google App Engine “Dashboard” reporting current load and billing information.

All cloud providers bill for CPU usage and data storage. Further billable resources include transactions, data transfer, API calls such as for sending and receiving e-mails, building datastore indexes and for image manipulation.

Tables III and IV show examples for pricing models. Please note that CPU hours not necessarily comparable between different systems, but these figures give a hint about real costs.

Cloud provider	Costs per CPU and hour
Amazon EC2	8.5 US-Cent
Google App Engine	10 US-Cent
Windows Azure	12 US-Cent

TABLE III
COSTS FOR CPU USAGE. [2] [5] [6]

Several payment methods are available. Using resources subject to charge requires a credit card or bank account. Cost

Cloud provider	Costs per GB storage space and month
Amazon EC2	14.03 US-Cent
Google App Engine	15 US-Cent
Windows Azure	15 US-Cent

TABLE IV
COSTS FOR STORAGE USAGE. [2] [5] [6]

limits can be set for each application. Applications can easily be deployed via HTTP. Setting up a new application can typically be realized within a few minutes.

Cloud computing environments bear several risks in terms of availability and data security. Generally data security is an issue when applications and data are distributed to the cloud. Although cloud providers promise confidentiality a certain risk of technical problems remains. Availability of resources raises further concerns. Although the short history of cloud computing showed that outages are very rare critical applications should not be deployed to a cloud provider. The sheer amount of redundant resources decreases those risks dramatically.

F. MapReduce

MapReduce is a technology invented by Google to handle large amounts of data [13]. It works by performing a two stage computation. In the first stage (Map), the input is divided into smaller subsets and distributed to so-called worker nodes. A worker can either process the smaller subset or divide it into further subsets. The latter case would lead to a tree structure. In the second stage (Reduce) the answers from all workers are combined to get the output. This is exactly what is needed to perform the calculations in the example.

Hadoop [26] is an open source implementation supported by the Apache Foundation. Hadoop is, for instance available in pre-configured Amazon EC2 instances under the name Amazon Elastic MapReduce [3].

IV. EXAMPLE IMPLEMENTATION OF NETWORK ANALYSIS SOFTWARE

In order to understand the following cost and performance estimations an overview about relevant components of the application are given in this section. The focus is on PaaS(concretely Google App Engine). An IaaS based approach would have the same basic architecture, except that the platform on top of the infrastructure has to be provided by the application developer. Aspects such as consistent storage are not relevant for the concrete example application.

A. Software architecture

In this section all concepts are named by their respective name within Google App Engine. Other cloud provider use different names for similar concepts.

The application is very simple, hence, the architecture remains very straightforward. It needs to store data and performs many calculations where the same data is used multiple times. For this reason all data is stored in the cloud in order to increase access speed. A major reason is that data sets are replicated transparently through the scaling mechanism maintained by the cloud provider. Alternatively, data could be stored in the local data center and being transferred on demand, but this would decrease access speed significantly.

Figure 4 shows a general overview about the layers used to access the datastore and to compute the results for the example used in this paper. Of course, other APIs can be called as well, but are not used in the example.

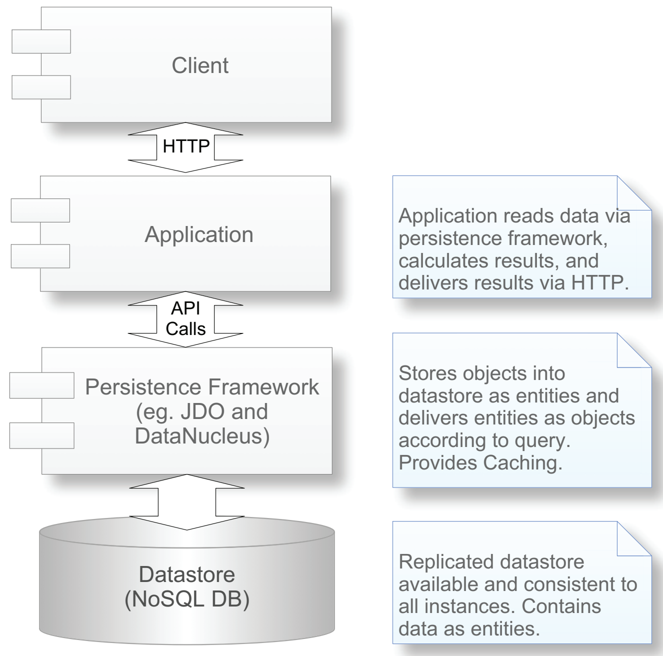

Fig. 4. Components forming the sample application on top of Google App Engine.

The datastore keeps data objects with their persistent attributes as entities. These entities are comparable to tuples in relational databases, but the main difference is that each entity can have arbitrary named values. The result is not a table but a collection of entities of a kind. Two entities of the same kind can have different properties. Hence, the Google Query Language (GQL) does not support projections and aggregations. Several data types are supported, for instance string, integer, or references to other entities. Distributed transactions, which occur when multiple users access the same data at the same time, are supported by the datastore. The datastore uses a distributed architecture to scale. Furthermore, customers can choose between a master/slave replication where one master copy of data is kept in one of Google's data centers, or high replication (at a higher cost).

It is common that application classes do not access the database or datastore directly through a low level interface. Such a low level interface is provided by Google, but more sophisticated interface is also availabe. In case Java is used as platform, persistence is provided through the DataNucleus project [23] which finally provides a JDO implementation to the application. This eases datastore access dramatically.

Basically all persistent attributes of the entity class have to be labeled as "Persistent" such as in this example:

```
@PersistenceCapable
public class Link implements Comparable<Link> {

  @PrimaryKey
  @Persistent(valueStrategy =
    IdGeneratorStrategy.IDENTITY)
  private Key key;

  @Persistent
  private Date timestamp;

  @Persistent
  private String thisIP;

  @Persistent
  private String otherIP;

  @Persistent
  private double lq;
  ...
```

Particular entities or sets of entities can be queried from the datastore through a persistence manager which accepts GQL or JDO syntax.

Calculations can be triggered by HTTP requests. Results of calculations can be stored in the datastore or can be delivered to the client via HTTP, as HTML or XML for instance. An XML web service would be another suitable bidirectional interface.

For the example application results are put into an XML document like this:

```
<?xml version="1.0"?>
<?xml-stylesheet type="text/xsl" href="/calc.xsl"?>
<timestamp time="Fri Sep 10 00:00:02 UTC 2010">
 <root name="192.168.0.254">
  <calctime>1707</calctime>
  <node>
   <sequencenumber>1</sequencenumber>
   <name>10.2.0.247</name>
   <quality>8.056671</quality>
   <parent>192.168.1.27</parent>
  </node>
  ...
  <node>
   <sequencenumber>2</sequencenumber>
   <name>10.2.0.251</name>
   <quality>10.0566718</quality>
   <parent>192.168.1.130</parent>
  </node>
 </root>
</timestamp>
```

B. Software development process

The software development process for PaaS clouds is very straightforward, especially when the frameworks provided within the PaaS cloud are well established in the off-cloud world. This is the case with the Java environment provided by Google App Engine. In an IaaS cloud the software development process is comparable to regular server based system.

A local testing environment is preferable, as it limits deployment efforts and shortens the development and test cycle. The completed application can than be released to the cloud environment. Deployment of applications (Google App Engine) or instance images (Amazon EC2) is supported through helper applications or plug-ins for IDEs, such as Eclipse. Although very straightforward, deployment consumes a certain amount of time.

Debugging an application on a PaaS cloud is a hassle, since step-by-step execution or console output is generally not available and logging remains the only way to inform the developer about the current state of the software. Therefore, local testing of components is inevitable.

V. Test results, costs, and conclusion

Each computation step (calculating all paths to one gateway for one timestamp) consisting of fetching the data for this timestamp, building the internal model, performing the Dijkstra algorithm, and calculating. The entire calculation explained in section II takes about 13 hours, which is approx. 18 times faster than computation on high end local resources (approx. 240 hours). To protect their infrastructure, cloud providers limit the number of possible instances. Google App Engine for example creates up to 20 instances when many requests are queued and consequently workload is high. The factor of 18 fits very good with the number of available instances considering some overhead.

For this calculation 750 CPU hours have been charged, which results in costs of 75 US-Dollar.

The entire dataset needs around 48GB when stored in the datastore (data plus meta-data). This costs additional 7.20 USD per month. The amount of uploaded data is about 22GB (note, that all data are uploaded via HTTP which imposes an overhead). Uploading of these data costs 2.20 USD one time for data transport and approx. 2 USD for CPU time (eg. creation of indexes). Uploading can and should be parallelized as well through multiple instances, otherwise uploading data takes several days time.

Some figures shall be given for a very rough comparison. A local machine would need 10 days. The total cost of ownership for a server system in the University's computing center the authors are using is estimated at around 3000 EUR per year (750 EUR amortization of hardware over 4 years, 700 EUR electricity, 100 EUR external networking, 25 EUR rent for rack space incl. climate control etc., 1400 EUR administration). At an idealized full capacity this would occasion costs of around 80 EUR per 10 days or 110 USD (at an exchange rate of 1.40 valid in February 2011).

The problem described in this paper as a running example is very well parallelizable. For this reason it benefits much from additional computation resources. Additional resources can be provided through cloud computing or through local hardware. Local hardware needs to be working to maximum capacity to decrease costs. In comparison, cloud computing is more cost effective when resources are needed at peak times or the workload is unevenly distributed over life time.

Generally, the following conclusions can be drawn from the example project:

- Cloud computing is a suitable option for scientific computing. For calculations that require shorter peaks of computation time cloud computing is a cost-saving option.
- PaaS supports instant software development without the need of installing operating systems and runtime environment. This is subject to the availability of all needed functions within the platform.
- Availability of cloud resources (both long term and short term) should be considered when larger software development efforts are necessary.
- Cloud computing concepts are suitable for resource sharing withing the academic community. This would increase the load factor and decrease costs.

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NET-COMPUTER: Internet Computer Architecture Based on Intelligent Agents

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Abstract - *Research in Intelligent Agents has been ongoing for sometimes now yielding interesting results, some of which have been translated into commercial ventures. Intelligent Agents are executable software components that represent the user, performs tasks on behalf of the user and when the task terminate, the Agents send the result to the user. Agents are best suited for the Internet: a collection of computers connected together in a world-wide computer network. A number of Agent systems have been implemented, the simulation test results do show the feasibility of Intelligent Agents and Agent systems. Agents' technology gives rise to a new computer architecture: the NET-COMPUTER based on HYDRA computer architecture in which the computing power resides in the Internet. In the NETCOMPUTER, the Internet computers form the hardware and software resources, and the user is provided with a simple input/output interface through which he/she can access the Internet to run user tasks. The simulation test result do show the feasibility of the NET-COMPUTER based on HYDRA computer architecture and Intelligent Agents.*

Keywords: Intelligent Agents, Internet, HYDRA computer architecture, NET-COMPUTER

1. Introduction

Agents' technology developed from related works on distributed Artificial Intelligence (AI), where the researchers aimed at spreading intelligence behavior through distributed computer systems to tackle inherently distributed problems. They developed intelligent software programs (or Agents), where the most advanced can communicate, collaborate and even learn from each other as they use their knowledge for problem solving.

The Internet has made it possible for computers at both ends of the world connected to one worldwide network: the "Internet" to talk to each other. From the comfort of their office or home, the user can log onto the Internet, and browse through documents on a remote machine several thousands of miles away. Combining information superhighway with the Internet opens the door for all sorts of communication and information processing possibilities. Automatic banking or Automatic Teller Machines (ATM), Imaging, Remote Teaching and Telemarketing are among the many services that can be provided [1].

A growing segment of the Internet is Electronic Commerce. Consumers are looking for suppliers selling products and services on the Internet. Meanwhile, the suppliers are looking for buyers to increase their market share. The vast amount of information on the Internet causes a great deal of problems for both the consumer and the seller. Searching the vast information on the Internet, a task executed online is not only time consuming but boring as well. Intelligent Agents are best suited for this type of task. Intelligent Agents will surf through the clutter on the Internet, resulting in the selection of specific information of interest to the user. The Agents speed up the process of locating items on the Internet and leave the users more free time to do more productive or enjoyable tasks.

2. Intelligent Agents

The definitions of an Agent fall everywhere along a continuum, from simple macros in which the user enters a few parameters to truly intelligent Agents which demonstrate learning abilities and artificial intelligence. An Agent [2],[3] is a software entity with some degree of autonomy, carries out operations on behalf of the user or another program, and thus, represents or has knowledge of the user's goals or wishes.

Agents are the user's personal representative or assistant on the Internet carrying the user's identity, access rights (permissions) and responsibilities. The Agents acts on behalf of the user at the user's request or using some agreed user protocol. Agents communicate with their peers by exchanging messages on behalf of the user [4]. While Agents can be as simple as subroutines, typically they are larger entities with some sort of persistent control (e.g. distinct control threads within a single address space, distinct processes on a single machine or separate processes on different machines).

2.1 Agents on the Internet (Distributed network)

Agents on behalf of the user freely roam the Internet or information superhighway searching for relevant information or services which are of interest to the user. These Agents do not take actions on behalf of the user, they provide the user with the information, and the user is free to act otherwise. However, a higher level of Agent sophistication involves *service performing* Agents which execute the specific tasks on behalf of the user (e.g. find me the cheapest flight to Paris or arrange a meeting with the managing director someday next week). Finally there are *predictive* Agents that volunteer information or services to the user without explicitly being asked [3]. For example, the Agent may monitor news groups on the Internet and return discussions that it believes to be of interest to the user.

Agents' execution on the Internet raises a host of issues which needs to be addressed if Agents processing is to be accepted by the computing community. For example, how ambitious should the Agent representing a user be? And, how much should the user trust the Agent especially when one considers that the Agents are delegated important decisions such as making financial commitments on behalf of their user. How about the security of such a transaction? In traversing the network searching for information, the Agents will need to have access permission at every site they visit in addition to the site authenticating the identity of the user that the Agent purports to represent [4]. This is to prevent Agents masquerading as some other user's representative. The need to guard against wayward programs that insta viruses, compromise host, or pilfer the database. Th resources used by the Agent at each site needs to b charged to the user. These and a host of many others ar the issues which must be addressed.

2.2 Agents Motivation

Four main motivations exist for the Agent paradigm. Th first, performance improvement is an importar motivation. In Figure l (a), we have the common situatio where a client and server are located on two differer nodes. The sever will typically manage some data. T carry out computation, several messages will often have t be transferred back and forth between the two partie There are performance advantages in moving the client t the server side of this connection. The amount of networ interactions can be reduced as a result of co-locatin service requesters and service providers heavily engage in communication. This scenario is illustrated in Figure (b).

A second motivation for the Agent model is that it is intuitive. Agents acting more or less independently on behalf of somebody are a well known concept. Figure l (c), represents a similar situation as in Figure l (b), but now the client itself does not move. It sends a representative to the server side to do the actual computation. The Agent will typically reduce the amount of data that has to be transferred over the network. In this example, the result is finally sent back the client.

A third motivation for the Agent paradigm is that communication is reduced to a site local issue. All that is needed is to move, or co-locate, Agents that wish to communicate. Locality of Agents is not hidden, but communication channel are.

Finally, this model gives cleaner and simpler failure semantics. In a regular distributed system, the state between the client and the server is normally distributed. A failure half-way makes it tricky to ensure that both parties roll back their state to maintain consistency.

Alternative Agent architecture model such as server-server models shown in Figure 2 are also in existence. In the server model, the Agent server co-ordinate activities of all local Agents arriving from other hosts.

A number of Agent-based systems have been in implemented, tested and the simulation results obtained point to the feasibility of Agents processing. The Agent systems are implemented either as interpreted program code WAVE [7], [8], TACOMA [9], Telescript [10], [11], [12] and Agent Tcl [13] or compiled program code [14], [15]. In the interpreted program code version, the execution engine runs on each host, receiving, interpreting user program (string) and executing the user command. In the compiled program code version, the compiled program code (compiled code) is loaded on to each host. The user commands (compiled program code) is loaded and executed by the execution engine.

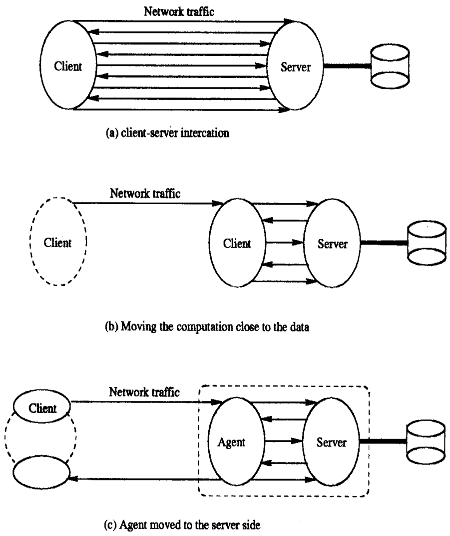

Figure 1: Client-server agent model

3. HYDRA Computer Architecture

HYDRA: Parallel and Distributed Swarm computer architecture for Agent execution [15] is based on the Swarm computer architecture [14]. The HYDRA computer depicted in Figure 3, consists of four modules: An Access Point (AP) through which the user can access the HYDRA computer to issue tasks and collects the results after task termination, Routing Module (RM) responsible for routing Agents to local destinations (PM), Processing Module (PM) is the execution engine and also the destination of an executing Agent, and Network Router (NR) that connects hosts in the network, routes Agents between the hosts and ensures message delivery to correct destinations. Agents navigate a network of hosts using hosts' addresses (PM and node), executing a task, and when the task terminates, the Agents collects the result and send to the user. The HYDRA computer is a software implementation coded in the C programming language running in the UNIX/Linux environment using the TCP/IP transport protocol.

The HYDRA computing environment consists of several hosts connected together by an interconnection scheme to provide a distributed computer system in which Agents execution can be investigated.

PM2 → host2…….. PM_n → $host_n$. The NR then starts the RM running at each host, creates a connection point for Access Point (user interface) at each host and then, enters a continuous loop monitoring network hosts, RM and AP connections for user connection.

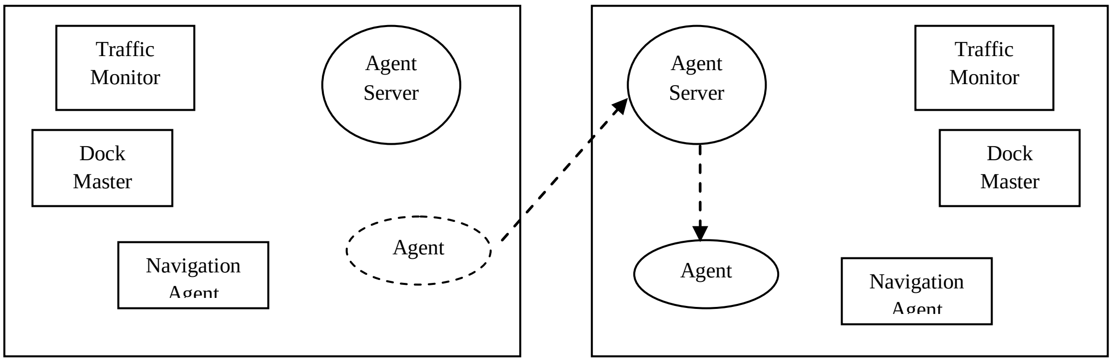

Figure 2: Server – Server Agent model

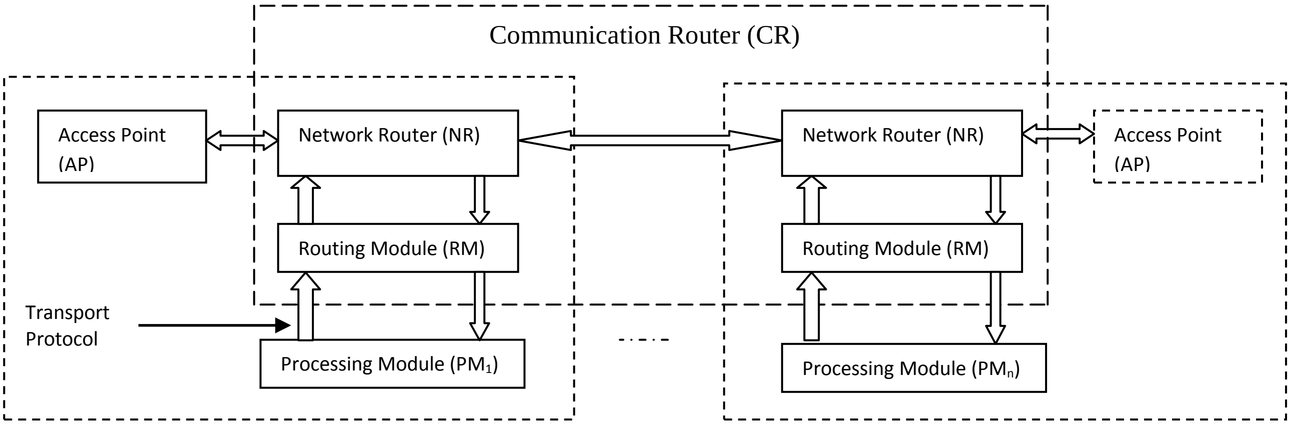

Figure 3: The HYDRA Computer Architecture

The Communication Router (CR) in the HYDRA computer connects hosts in the network and reliably delivers messages to the hosts. During the start up, the CR connects the hosts, starts the routing module at each host, establishes a listening end-point at each host and monitors message arrival at the end-points. The CR identifies both local and remote destinations to reliably deliver messages to these destinations. The CR uses an interconnection scheme [15] to connect hosts in the network, assign host numbers to host names and establish communication channels among the hosts. The hosts in the network are grouped into domains, and the domains interconnected together to form a distributed network. Hosts in a domain are directly connected to every other host in that domain and within each domain, a single host acts as gateway handling out-of domain Agents i.e. Agents crossing domain boundaries. After the hosts in the domain have been connected, the CR at each hosts opens the network file containing names of all hosts in the network, reads the hostnames and maps hostnames to PM i.e. PM1 → host1,

4. The NET-COMPUTER Computer System

The NET-COMPUTER implementation is based on HYRDRA computer architecture developed specifically for executing Intelligent Agents. The motivation for developing the NET-COMPUTER is to shift the computing resources to the Internet and provide users with a simple input/output device through which users can access the Internet computing resources.

Current estimates indicate that:

- Nearly 90% of the users run document processing applications.
- In an 8-hour office working day, effective computer usage is about 40% (the time when the computer is actively in use). In the home environment, this is less than 20%.
- The ordinary user uses less than 10% of the computing

resources e.g. processor speed etc (except for scientific applications and data processing applications e.g. payroll processing).

From the above statistics, one can rightly argue that there is no point in providing the user with a powerful machine which spends most of its time idle and thus, forms part of office/home decoration. Thus, why not concentrate the computing resources on the Internet and provide the user with a simple input/output terminal through which to access the computing resources. The user input/output terminal can be as simple as a pager, ipad, tablet, personal organizer, etc.

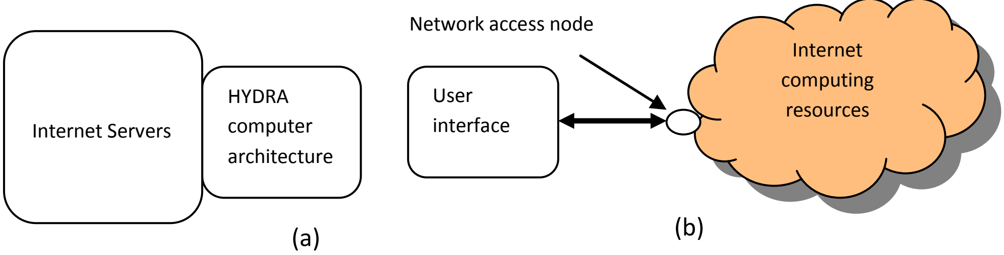

Figure 3: Configuration of the NET-COMPUTER

The NET-COMPUTER is a collection of Internet computers (servers) running user programs on demand. Both the hardware and software resources reside on the Internet. The user need not be aware of the exact location of the resources. The HYDRA computer architecture attached to the Internet servers shown in Figure 3 (a) receives and executes the Agents. The Internet servers are connected together to form the Internet computing resources shown in Figure 3 (b). The Internet or cloud provides the computing resources. The user interface to the NET-COMPUTER is a simple device with less complexity in hardware and software since the computing resources reside on the Internet.

5. Simulation Test

The NET-COMPUTER was simulated in a network of three hosts: ainur, kira and yavanna connected as shown in Figure 4. The hosts: are connected to networks A, B and C shown in Figure 5 and, run UNIX operating system.

The objective of the simulation test to test the following Agents characteristics:

1. Mobility: the ability of Agents to navigate the network and reach destination address (host).
2. Demonstrate the ability of the Agents to carry out useful work.
3. Test the ability of the gateway hosts to route Agents to host in the other domains.

In addition, the user is at liberty to configure and select the hosts on which to run HYDRA computer system as explained in section 3.0. This is done through selection of hosts in each domain. At the beginning of execution, a node file containing node distribution (node reference number and links to other nodes in the network) is prepared for each PM. At each node, is stored a unique number which is accessed by Agents visiting the node. The aim here is demonstrate the ability of the Agents to access a stored data item. The unique number is stored in the persistent workspace 3 (PW3) which is accessible and can be modified by Agents belonging to the same task. The unique number stored at each visited destination is retrieved by the Agents and used in a simple addition routine. The node file is then uploaded to the respective PMs.

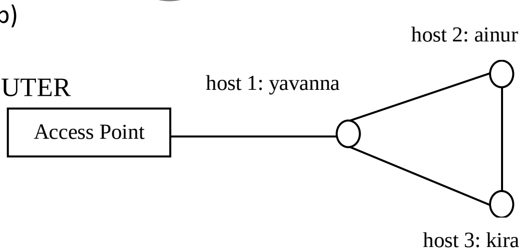

Figure 4: Simulation test network

In the test routine, the Agents load a value of 0000001Ahex to the Agent's workspace 2 (AW2), the Agents then navigate the network to node 1 in all PMs. On reaching its destination, the Agents' retrieve the number stored at PW3 and add it five times to the number carried by the Agents in AW2 and store the result in PW3 as shown in table 1. The Agents then navigate the network to node 1 in PM3 where each Agent add contents of its AW2 to the contents of PW3 in PM3 and store the result in AW3. The final result is stored in PW4 in PM3 shown in table 2.

The test program is described as follows:

AW2, AW3, AW4 - Agent variables in the Agent workspace.
PW3, PW4 - persistent workspace (nodal variables)
01 Assign a value to AW2.
02 Insert destinations PM (PM1-3) address in the destination list (DL).
03 Insert destination node address (node 1) in the destination list (DL).
04 Spawn to destination address in the destination list.
05 Load a value from persistent workspace PW3 to AW3.
06 Add contents of AW3 to contents AW2 and store the result in AW2.
07 Add one (1) to contents of AW4.
08 Test if contents of AW4 equal 5.
09 If contents of AW4 are less than 5, go back to 06.

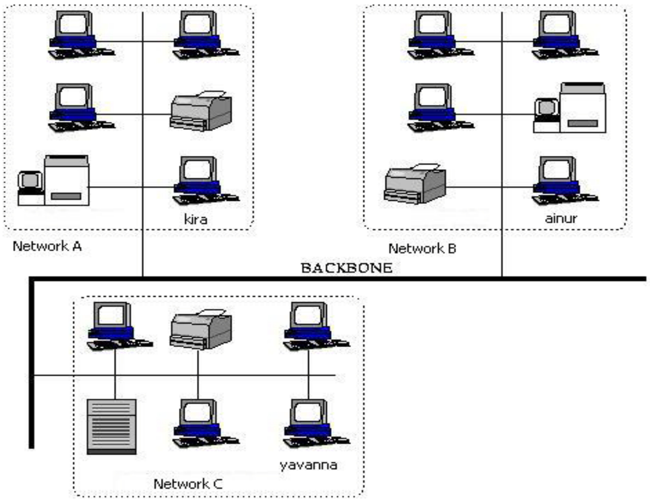

Figure 5: The NET-COMPUTER execution network

10 Store the contents of AW2 in persistent workspace PW3.
11 Store the value 0 in persistent workspace PW4.
12 Insert destination PM (PM3) address in the destination list.
13 Insert destination node (node 1) address in the destination list.
14 Spawn to the address in the destination list.
15 Load a value from persistent workspace PW3 to AW3.
16 Add contents of AW3 to contents AW2 and store the result in AW3.
17 Load a value from persistent workspace PW4 to AW4.
18 Add contents of AW3 to contents AW4 and store the result in AW3.
19 Store the contents of AW3 to persistent workspace PW4
20 Terminate.

6. Simulation Test Results

During the simulation test, Agents traversed the network using destination addresses carrying a data item. At each visited PM, the Agents retrieved data item stored at the site and added the data to the data item carried by the Agent, executing the addition loop five times and stored the final result at the visited site. At the end of the addition, the Agent retrieved final result stored and stored it in Agents workspace. The Agent then, traversed the network to PM3. At the destination PM3, the Agent retrieved the stored data, added it to the data carried in the Agents workspace and stored the result at PM3. The results are presented below in tables 1 and 2.

7. Conclusion

The simulation tests were carried out to test Agents ability to traverse the network using destination (PM) addresses, access a data item stored at a site, manipulate the data item and store the final result at the site or communicate the data item (result) to a different location. In the NET-COMPUTER, the computing resources are distributed over a wide geographical area. Thus, Agents navigate the network to reach their execution environment and the user at the end of the task execution to deliver the results. The hosts used in the simulation test are connected to physically separate networks. We can therefore conclude that the Agents traversed the network to reach their destinations. Thus, confirming Agents navigation ability in the HYDRA computer architecture. The simple addition exercise demonstrated that Agents can be used to carry out some useful work such as manipulating user program. The ability of the Agents to access stored information (data), manipulate the data and carry the data to different locations within the network, demonstrates the Agents' ability to collect/access, modify data and communicate the same. The simulation test results have demonstrated the feasibility of the NET-COMPUTER based on HYDRA computer architecture and Intelligent Agents. However, more research work is needed to adapt the HYDRA architecture for deployment on the Internet.

Table 1: Agents add contents of AW2 to the contents of PW3

PM Number	PM1	PM2	PM3	Loop count
Value in PW3	22	23	24	
Value in AW2	0000001A	0000001A	0000001A	
Sub total in AW2	0000003C	0000003D	0000003E	1
Sub total in AW2	0000005E	00000060	00000062	2
Sub total in AW2	00000080	00000083	00000086	3
Sub total in AW2	000000A2	000000A6	000000AA	4
Sub total in AW2	000000C4	000000C9	000000CE	5
Total in PW3	000000C4	000000C9	000000CE	

Moreover, the delays experienced when sending Agents and during execution of user program and, the fact that the participating hosts run other programs in addition to HYDRA computer architecture has not been taken into account. These are matters and others will be the subject of further investigation in the next phase of the research.

Table 2: Agents add contents of AW2 to the contents of PW3 and store the results in PW4 in PM3

Loop count	PM1	PM2	PM3	Sub Total	Total in PM3
	000000C 4 (AW2)	-	000000CE (PW3)	0000019 2 (AW3)	0000019 2 (PW4)
		000000C 9 (AW2)	000000CE (PW3	0000019 7 (AW3	0000032 9 (PW4)
	-	-	000000CE (PW3) 000000CE (AW2)	0000019 C(AW3)	000004C 5 (PW4)

The NET-COMPUTER aims to provide the user with the ability to log to the system, issue tasks to be executed, log off and at a later time log to the system to collect the task results. Thus, providing an execution environment similar to the mail server described in section 4.0.

The simulation was run with the user was connected online, the next phase of the research will investigate the feasibility of the user to log to the NET-COMPUTER, issue a task and log off. At a later time, the user can log to the NET-COMPUTER to collect the results of the task execution.

The combination of Intelligent Agents and the NET-COMPUTER might just provide the mechanism for furthering Electronic Commerce on the Internet. However, a lot of work still needs to be done to make Agents processing on the Internet feasible. Issues such as security, accessibility, fault-tolerance, authentication, charges for consumed resources, etc still need to be adequately addressed.

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Semantic based synchronization of Profiles in a pervasive environment

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Abstract - *Personalization provides a convenient means to ease the use of pervasive computing systems and related services. Profiles are one of the major building blocks for making real personalization happen. In the existing heterogeneous and upcoming pervasive computing environment typically numerous different profiles are used and maintained by the user or different services, devices or operators. In order to avoid multiple copies of the same data profile synchronization and -linking should be used and applied between the different locations and administrative domains, in which the different types of profiles reside. But keeping in mind the different contexts', in which these profiles have been constructed, it is mostly unknown how to match the meaning of one profile to another or how to harmonize the different kind of values in the different profiles. In this paper, mechanisms for efficient synchronization of profiles based on semantic distinctions are introduced. By using ontology's to distinguish between semantic entities profiles can be kept synchronous even using low-bandwidth radio links and mobile devices like smart-phones and PDAs. The concepts presented in this paper have been developed by the Wireless World Research Forum (WWRF) Working Group 2 (WG2)[3].*

Keywords: Personalization, Semantic Profiles, Ontology, pervasive computing

1 Introduction

Manifold services and vast amount of resources makes it difficult for the user to find what he needs in the Internet. Personalization filters information and resources and adapts services to the needs, requirements, expectations and behaviour of the user. Thus, personalization improves user experience and increases the value of using the Internet to the user. Today, in the Internet different services exist separate from each other. Services depend on other services rarely. Each service is able to provide his own personalization architecture, interoperability is rarely required. In pervasive computing services interact with each other to perform tasks of the user. This interaction requires seamless personalization architecture to allow interoperability and to provide a personalized pervasive computing environment to the user. Personalization covers three aspects:

- gathering the context of the user,
- use configured and learned behaviour and capabilities of the user, device, network and service environment from profiles
- and adapting service presentation to the available capabilities of the user and input/output devices.

The context of the user enables services to predict the user his intention. Context awareness is meant to capture the situation of the moment. Adaptation meets the capabilities of the user and his devices by converting content and service interfaces to the format preferred by the user. Profiles provide configured and learned behaviour, preferences and capabilities of users, devices, networks and services to a personalization system.

Profiles follow profile definition standards. These standards, open or proprietary, focus on specific profile information like device capabilities, privacy preferences, contact information, service description etc. The heterogeneous nature of pervasive computing we assume will feature several profile definitions. In pervasive computing where different services of various domains and vendors interact with each other the profile of the user, his device and the configuration of previously used services must be accessible by newly encountered services. While a user is indisposed to enter his name, email address, preferred colour scheme etc. several times, every time he enters a new environment with new, previously unknown services, a pervasive computing system should provide a mechanism to access and process this information. Thus, configurations and learned behaviours must be exchanged between different profile definition standards.

To allow convenient exchange of profile information, translation rules and interoperability standards must be defined. On the one hand these rules permit for exchanging profile information. On the other hand these rules free service developers from developing and using custom filters for profile information exchange like as it is done by today PC applications, e.g. Word or Photoshop.

"The Resource Description Framework (RDF) provides a lightweight ontology system to support exchange of knowledge on the Web". RDF maps information from one document to another document by describing the elements in the documents. This description is then used to define an exact mapping between the elements of both documents. While RDF does not provide translation rules between different documents it helps to understand the content of documents.

In this paper we separate the profile structure form the profile information. Views, a concept derived from relational database systems, hold the structural information and are linked to profiles which hold the profile information. By separating the profile structure from profile information two different profile definitions can share their common data, avoiding redundancy and keeping integrity of data, which is actually a transformation for compatible profile definitions. Furthermore, attributes augment profiles describing implicit information and processing instructions for services using the information stored in profiles.

In the next chapter we scope the need and problem of profile semantics in pervasive computing. In section 3 we give a definition for profiles and introduce profile augmentation and views. Two examples on how to use views and profile semantics are given in section 4 and 5. Before the paper closes with the conclusion a general discussion on profile placement and security concerns in pervasive computing environments is given in section 6.

2 Motivation

In pervasive computing, services perform tasks for the user. Services not owned by a particular user are not specialised towards serving this particular user. Personalisation adapts services to the user for a convenient use. Profiles, beside context awareness and adaptation, are one cornerstone of personalisation. Profiles contain configuration and behavioural inputs about the user, his preferences and capabilities for the service to learn about and adapt to the particular user. Profiles provide the stored information to interested services, thereby releasing the user from inputting contained information and increasing convenience for the user to use a pervasive computing environment.

There are several types of profiles. The following is just an example on what types of profiles are possible and what type of data they may contain. User profiles keep the configuration, behaviour, preferences and capabilities of a user. The configuration covers the name and phone number of the user, the preferences knows about the favourite movies and music the user consumes, behaviour specifies how the user consumes such music and movie content and how the user usually interacts with other users, devices and services in a pervasive computing environment and the capabilities hold information about the physical capabilities of the user. Device profiles keep configuration and capabilities of the devices used. Configuration parameters are setup by the user. Examples of configuration parameters are the volume of speakers, colour schemes and setup of network interfaces of the device. Capabilities are defined by the manufacturer of the device. Examples of capabilities are the maximum screen resolution of a display, physical capabilities of network interfaces, available input and output mechanisms and available memory and processing power. Service profiles contain configuration of the service, e.g. customisation parameters like maximum allowed CPU and memory usage or locations of data sources and data drains. Characteristics of networks like bandwidth, number of users, cost models and charging information etc. is part of the network profile.

We assume that a pervasive computing environment is a heterogeneous environment with many different services, users, devices and networks. In addition, services are not only offered by operators and professional service providers but by normal users of pervasive computing environment as well. Possibly, the user developed his own service and then provides his service in a pervasive computing environment to other users.

Profiles of services, users, devices and networks follow different proprietary and standardised profile definitions. Services, depending on the information stored in the different profiles, must be able to understand the profile definition. The information stored in the profiles must be available to the services taking this information as their input. But to be able to synchronize the profiles of different service providers, operators, or even user devices, the semantics of these profiles must be defined somewhere. Therefore an ontology or an appropriate RDF description should be given, defining the vocabulary and the meaning within such a profile.
In order to be able to keep different profile synchronous and to understand the different meanings of several profiles, these kinds of basic information need to be provided somehow.

3 The profile architecture

In order to understand the definitions made for profiles, we start with defining the scope of a profile itself. A profile consists of profile entries. Each of the profile entries is identified by a key and contains values. Since multiple values are aggregated under one key, we say that this collection of values is a single value to the key. The key of a profile entry can be unique to allow only one key/value mapping for this particular key or occur multiple times, allowing for multiple similar key/value mappings. Users of the profile, i.e. services, use the key to retrieve the value of interest from the profile. If the key is not known search strategies may be defined to retrieve either the key of the profile entry of interest or the value of interest.

Services interpret values of profile entries and perform their operations. Besides the native meaning of values there may be implicit meanings and operations to perform on values of profile entries. Such operations are security, i.e. access control, extended type information, e.g. for postal address, country codes, phone numbers, colour schemes, and semantic information to provide hints on how to utilise the profile information. This implicit meaning, the attributes, is attached to the actual profile information as attributes to the profile entries. These attributes then augment profile entries and their stored profile information. Attributes are single values without key while multiple attributes are assignable to one profile entry. This attribute augmentation has close similarity to the XML and makes established XML technologies such as XPath and XPointer applicable to the profile architecture.

To address the need for profile semantics we define views on profiles. A view is a representation of the profile with its own meaning and structure. Views are derived from database views in relational database systems. A database view is defined on top of selected database tables and table data. This view is then be used like any normal database table. The definition of a view on a profile provides a new meaning on the underlying profile. Further, views may aggregate and reorganise profile information.

In a view the key of a profile entry is replaced by an identifier. Instead, the key moves to the view and is mapped to the now used identifier of the profile entry. Consumers of the profile do not access the profile directly but instead access the view defined on the profile.

```
<?xml version="1.0"?>
<rdf:RDF xmlns:rdf="http://www.w3.org/1999/02/22-rdf-
syntax-ns#" xmlns:vCard="http://www.w3.org/2001/vcard-
rdf/3.0#">
 <rdf:Description
rdf:about="http://qqqfoo.com/staff/corky">
   <vCard:FN>Corky Crystal</vCard:FN>
   <vCard:TEL rdf:parseType="Resource">
    <rdf:value>+61 7 555 5555</rdf:value>
    <rdf:type rdf:resource="http://www.w3.org/2001/vcard-
rdf/3.0#work"/>
    <rdf:type rdf:resource="http://www.w3.org/2001/vcard-
rdf/3.0#voice"/>
   </vCard:TEL>
   <vCard:EMAIL rdf:parseType="Resource">
    <rdf:value>corky@qqqfoo.com</rdf:value>
    <rdf:type rdf:resource="http://www.w3.org/2001/vcard-
rdf/3.0#internet"/>
   </vCard:EMAIL>
 </rdf:Description>
</rdf:RDF>
```

vCard in RDF/XML vCard profile definition.

```
BEGIN:vCard
VERSION:3.0
FN:Corky Crystal
TEL;TYPE=work;TYPE=voice:+61 7 555 5555
EMAIL;TYPE=internet:corky@qqqfoo.com
END:vCard
```

vCard in vCard 3.0 profile definition.

Above is the classic approach of profile translation. Both vCard profiles contain the same data. Using e.g. XSLT duplicates the profile, making both profiles independent of each other. A modification to any value in any of the both vCard profiles is not reflected in the other profile, resulting in inconsistencies.

RDF/XML vCard element	**Profile entry identifier**
rdf:RDF/rdf:Description/vCard:FN	11a3
rdf:RDF/rdf:Description/vCard:EMAIL	33cf
rdf:RDF/rdf:Description/vCard:TEL	e92b

Profile entry identifier	**Profile entry value**	**Profile entry attributes**
11a3	Corky Crystal	
33cf	corky@qqqfoo.com	TYPE=internet
e92b	+61 7 555 5555	TYPE=work TYPE=voice

vCard element	**Profile entry identifier**
FN	11a3
TEL	e92b
EMAIL	33cf

Profile information and profile definition information are separated. vCard representation in both RDF/XML vCard profile definition and vCard 3.0 profile definition have their own views accessing the same profile containing the actual data. A modification to the vCard in any one view automatically updates the vCard information accessible via the other view.

Figure 1: Comparison of profile translation by duplication with definition of views on profiles.

Views avoid duplication of profiles by performing translation of profile definitions in the views. Profiles only contain the data of the profile but no structural information. Views define the structure of the data in the profile the view is defined on. An example of defining two views on one profile is compared with the standard approach of profile duplication in Figure 1. The first view on the profile defines a hierarchical profile structure following the RDF/XML vCard profile definition while the second view on the profile defines a flat keyword based profile structure following the vCard profile definition.

As seen in the example in Figure 1, integrity of profile information is an integral feature of the separation of profile information and their representation in profiles and views defined on profiles by avoiding profile duplication. Additionally the use of views makes the need for reconciling between duplicates of the same profile obsolete.

Figure 2: Screenshot of the CUCA, setting up a mapping between the RDF/XML vCard profile definition and the vCard 3.0 profile definition.

Accessing profiles through views avoid duplication of data when access in different profile definition formats is required. Thus, integrity of profiles is guaranteed and reconciliation between identical profiles following different profile definitions is not necessary. The translation of a profile between different profile definitions is done by translating the views representing the profile definition.

The next two chapters describe example applications to point out the practical benefit of views on profiles. First, an application for semi-automatic profile translation, the ComTec User Centered Application AddOn for Profiling (CUCA), is presented where users are able to map elements of profile definitions to each other to perform automatic translation between different views on a profile. The second application is a general profile editor to edit profiles regardless of a profile definition.

4 CUCA – Translation of profiles

As learned in the previous section views defined on profiles avoid profile duplication and keep profile information integer. However, there must be then a mechanism to translate between the views representing a profile definition.

The ComTec User Centered Application AddOn for Profiling, CUCA, is a graphical user interface for setup of translation rules for interpreting views on a single profile. For this purpose CUCA uses attribute dictionaries to augment profile entries. An attribute dictionary consists of related attributes. Such relation could be the belonging to a profile definition. CUCA then creates one view on the profile for every attribute dictionary.

Attribute dictionaries show similarities of different attributes. The similarities aid services and applications to process attributes and to design appropriate search strategies for profile entries where the key to the profile entry is unknown.

Attributes represent the keys in profile definitions and augment profile entries. Attribute dictionaries representing a profile definition summarise these attributes representing the keys in the profile definition. Thus, there would be a plain vCard attribute dictionary consisting of all the keys defined in the plain vCard profile definition standard where each key is represented by a single attribute in the plain vCard attribute dictionary. The RDF/XML vCard profile definition would form another attribute dictionary with all their keys are attributes in this RDF/XML vCard attribute dictionary.

To provide automatic translation of a profile between different profile definitions CUCA maps attributes of two attribute dictionaries. Figure 2 shows a screenshot of CUCA to setup an attribute mapping between a plain vCard attribute dictionary and a RDF/XML vCard attribute mapping. How the attribute mapping is then applied on a profile and his corresponding views is shown in Figure 3. CUCA defines one view per attribute dictionary.

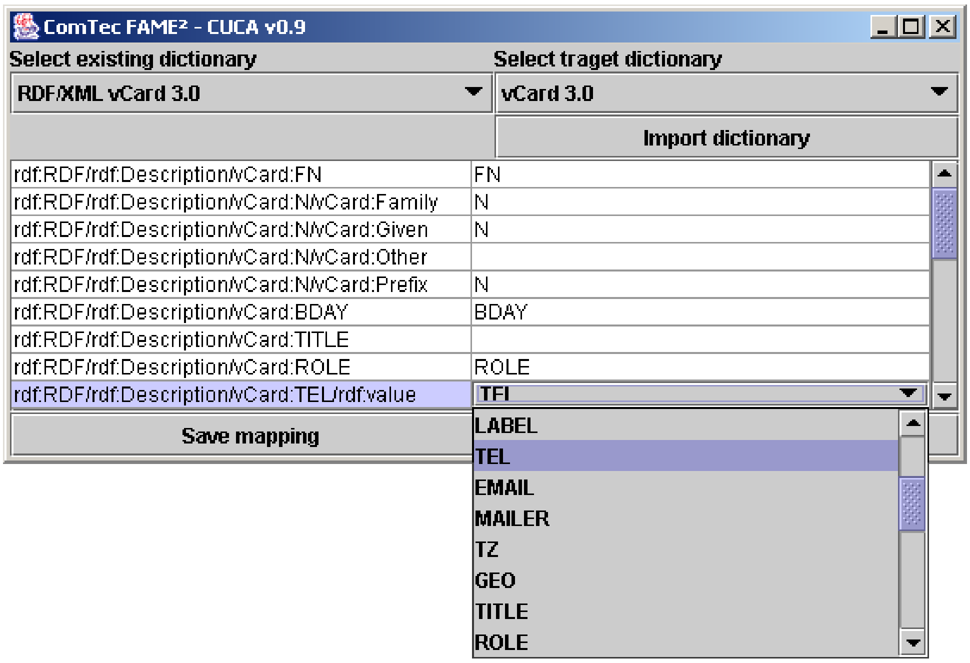

Figure 1: Screenshot of the CUCA, setting up a mapping between the RDF/XML vCard profile definition and the vCard 3.0 profile definition.

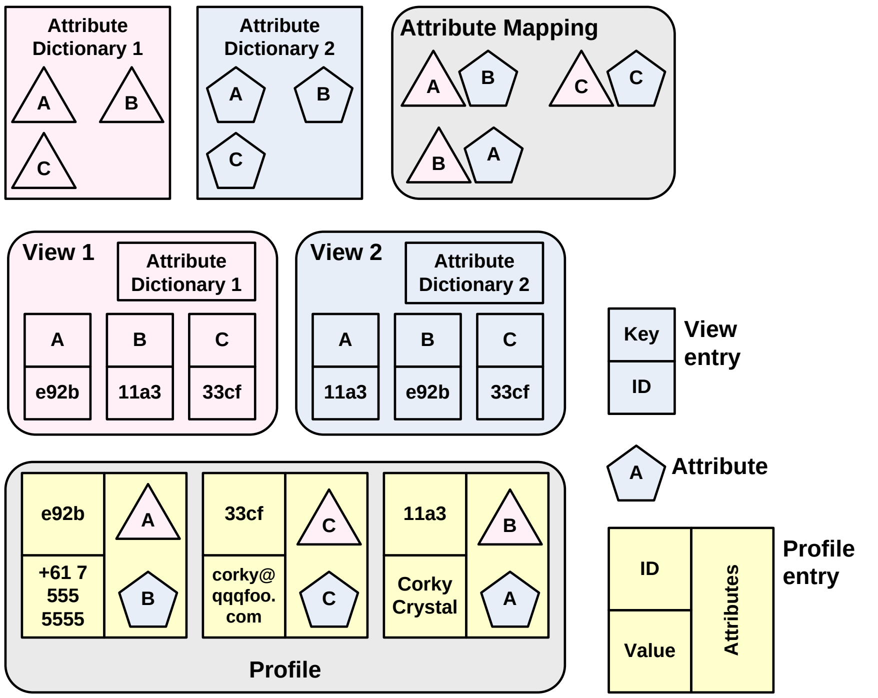

Figure 2: Definition of views on profiles basing on attribute dictionaries to translate profiles between different profile definitions. Each profile definition defines its own attribute dictionary.

CUCA is a graphical user interface to define translation rules for translation of views defined on a profile. The translation bases on attribute dictionaries where for every profile definition an attribute dictionary is defined. Translation of views on a profile is then automatically performed by a mapping of attributes of different attribute dictionaries.
In the next section attribute dictionaries and views on profiles are used to create a general profile editor.

5 General Profile Editor

Profiles are one cornerstone in personalization is to ease the user using services in pervasive computing. Several services in pervasive computing environments use and define profiles. Editing profile information contained in these profiles by the user is done by profile editors.

Today there is no standard format for profiles used by services in pervasive computing. Thus, users have to learn to deal with several different profile editors, each specific to a profile definition or to a single profile in the worst case. The specialized profile editors are optimized towards the profile or the profile definition they are designed for. This specialization allows for enhanced user interface design but requires the user to learn different user interfaces. Additionally optimized profile editors may choose to hide information from the user, making it inaccessible to the normal user. While hiding profile information may be useful for the normal and inexperienced user it is an inconsistence to the conception of putting the user in control. Furthermore, where services share a profile the specialized profile editors provided by the services may disagree in how to edit the shared profile. This is mainly because the specialized profile editors are self-contained. Relationships between services in their profile use are not visible by specialized profile editors.

A general profile editor allows for standardized access to profile information for the user in just one user interface. This general profile editor disburdens the user to learn several user interfaces. Further, the general profile editor is a single place for editing shared profiles, avoiding inconsistencies in editing the shared profile. Additionally, unless there is an overlapping attribute that hides a profile entry from editing, service specific profile editors cannot hide a profile entry by their user interface. Moreover an attribute may indicate the use of a profile entry in different services.

The design of a general profile editor requires the translation of profiles of different profile definitions. The introduction of views in section 3, "The profile architecture", enables a seamless, duplication free translation of a profile into different profile definitions.

The general profile editor defines his own attribute dictionary. Attribute dictionaries are introduced in section 4, "CUCA – Translation of profiles". All profile entries of a profile edited with the general profile editor are augmented by one or more attributes from that attribute dictionary. The attributes contain rendering information for the augmented profile entry in the general profile editor. This rendering information is pure graphical information, e.g. to show a Yes/No option, a selection list with appropriate selection model to allow selection of one or more values, a free from text field and so on. Further, the profile entry to edit is augmented by profile entry type information. This type information influences the values that are assignable to the profile entry. Such type information could be the limitation of certain numbers (e.g. only positive numbers), certain formats (e.g. social security number, date and time) or a set of predefined data the user can select from (e.g. colors, fonts). A specialized view defined by the general profile editor access the profile to edit, provides the rendering and profile entry type information and gives access of the profile entries contained in the profile to the general profile editor. This specialized view translates the profile into a profile definition understandable by the general profile editor.

Figure 4: Screenshot of prototypic general profile editor. The user "bjoern" edits his profile via a RDF/XML profile definition view.

Similar to a web browser, the general profile editor then renders the profile to edit accordingly to the rendering information and profile entry type information the profile entries in the profile are augmented with. A screenshot of a prototypic general profile editor is shown in Figure 4.

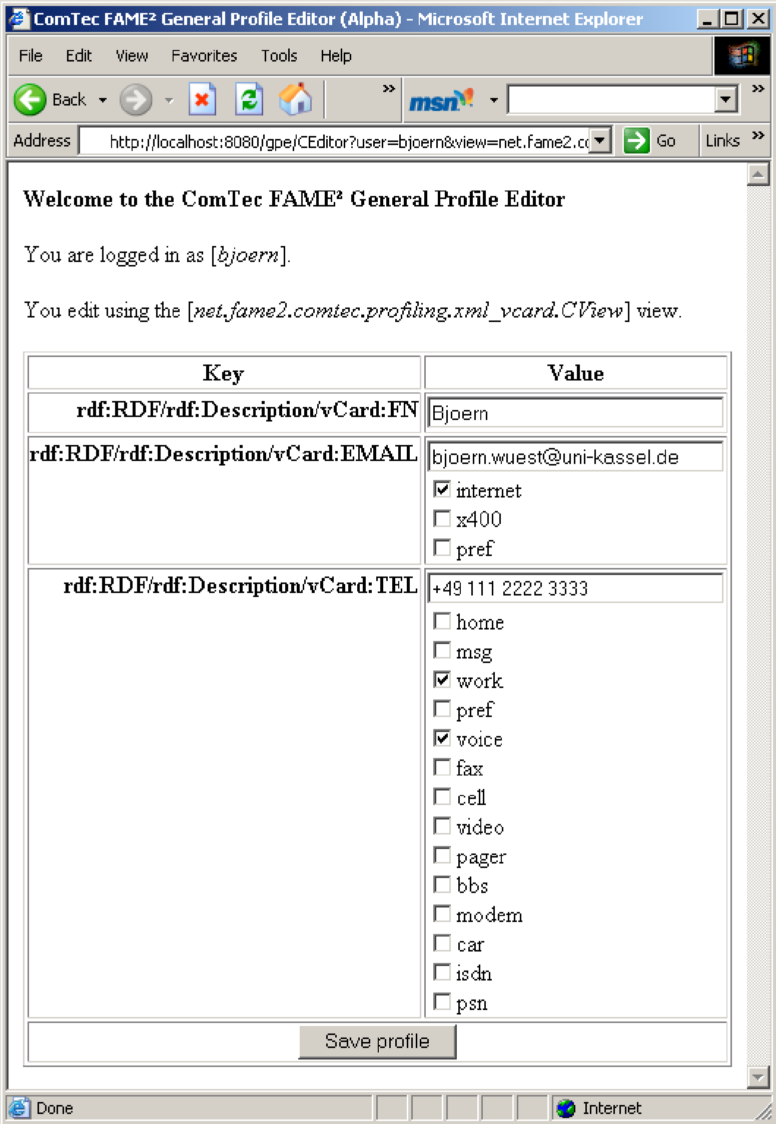

Figure 3: Screenshot of prototypic general profile editor. The user "bjoern" edits his profile via a RDF/XML profile definition view.

Specialized profile editors may provide optimized profile information but lack the transparency for profile editing expected by the user in a pervasive computing environment where the user remains in control. The general purpose editor augments a profile by rendering and profile entry type information. A view specific for the profile editor translates the profile to edit accordingly.

Some general discussion on further profile issues like security, synchronization etc. and how the definition of views on profiles and augmentation of profile entries by attributes aids follows in the next section.

6 General discussion on profiles

The above two sections demonstrate how definition of views on profiles and augmentation of profile entries with attributes from attribute dictionaries as introduced in section 3, "The profile architecture", provide translation of profiles without duplicating profiles and profile editing by a general profile editor is realised. Following is a discussion on profile issues like limiting access on profile information, profile placement and profile synchronisation.

Information stored in profiles, especially in user profiles, may contain sensitive information that should not be publicly accessible. Access control may provide a level of security and privacy on this sensitive profile information. Access control restricts access by evaluating access policies. With the use of views on profiles these access policies may be given implicitly by limiting the profile information accessible through a view. Thus, a view may provide only a subset of the information stored in the profile the view is defined on. Using this approach no explicit access control check is necessary when accessing any particular profile information but only one single access control check to determine whether the user is allowed to use this view. In an environment with limited device capabilities and processing power like the pervasive computing environment, this single access control check on granting access on a view as a whole is more efficient than performing the same access control every time any profile information is accessed. Further, read-only views - allow for reading but disallow for writing profile information - simplify the definition of access control rules and their evaluating logic.

The pervasive computing environment, where users own multiple and different devices and a service is not located in a single fixed place, raises the question on where to place profiles. Centralised servers for profile storage allow for simplicity of the overall profile system and allow formulation of a guarantee on profile availability. This centralisation has three major disadvantages. First disadvantage is how profiles are accessible by services located in the current environment in situations where no global communication infrastructure and thus no access to the centralised profile servers is available. Reasons for the unavailability of a global infrastructure are places where disasters or war destroyed the infrastructure or where the setup of a global infrastructure is inefficient from an economically point of view. Second disadvantage is the problem of scalability when millions of users and services access the centralised profile servers to ask for information from hundreds of thousands of profiles. Finally, users like to be in control over their property, whereas profile information is a property of theirs. Thus, users may not trust in a centralised system.

On the other hand distributed profile systems raise the issue of profile synchronisation. If profiles are copied, replicated and distributed any change to any profile must reconcile with existing copies and replicas. While views make duplication of profiles unnecessary for translating profiles between different profile definitions views do not solve the physical distribution of profiles. Views, however, give support for synchronisation of profiles by first using read-only views on replicas and second integrate reconciliation mechanisms in the view rather than in the profile. Additionally augmentation of profiles with attributes provides additional information on the origin of profile information, identification of replicas and topicality of profile information.

The above discussion gives a hint on how the application of views on profiles and attribute augmentation aids in solving these profile issues. On some issues like synchronisation there is work ongoing, other issues like profile access control and restriction are not under active research within our department.

7 Conclusion

In pervasive computing heterogeneous profiles are used. Their access by services requires profile translation. Existing solutions to profile translation duplicate profiles, introducing the problems of how to maintain data integrity and profile synchronisation. In this paper we introduce views on profiles. Views are, similar to relational database systems, structural representation of underlying data. All structural information as defined by profile definition standards is removed from the profile and shifted to views on the profile instead. Views avoid duplication of profiles, making reconciliation of profiles and profile integrity protection mechanisms obsolete. Attributes augment profiles and provide support for various applications related to profiles. Attribute dictionaries and their mapping allow for semi-automatic profile translation as demonstrated with the CUCA application. Attributes representing information on how to render a profile entry allow the design of a general profile editor. Profile synchronisation and access control is simplified and enhanced by views and augmenting attributes on profiles. Further work aims for synchronisation of physically distributed profiles and future exploitation of views and attributes on profiles.

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Internet of Things: Services and Applications Categorization

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Abstract—**In this paper we attempt to categorize the services provided by the Internet of Things (IoT) in order to help application developers build upon a base service. First we introduce the four main categories of services, and then follow by providing a number of examples of each of the service categories so as to provide an example of how each type of service might be implemented, and how it can be used to build an IoT application.**

I. INTRODUCTION

The Internet can be described as a ubiquitous infrastructure that has evolved from being a technology for connecting people and places to a technology connecting things. The future is the Internet of Things (IoT), which aims to unify everything in our world under a common infrastructure, giving us not only control of the things around us, but also keeping us informed of the state of the things around us.

One of the main problems with IoT is that it is so vast and such a broad concept that there is no proposed, uniform architecture. In order for the idea of IoT to work, it must consist of an assortment of sensor, network, communications and computing technologies, amongst others. But when you start putting together different types of technologies, the problem of interoperability arises. One proposed solution is to adopt the standards of the services-oriented architecture (SOA) deployed in business software systems [1]. Another takes a similar approach, suggesting the integration of Web Services into sensor network with the use of IoT optimized gateways, which would bridge the gap between the network and the terminal [2]. In general, it may be beneficial to incorporate a number of the technologies of IoT with the use of services that can act as the bridge between each of these technologies and the applications that developers wish to implement in IoT. This paper breaks down four main categories of services according to technical features, as proposed and described by [3]. In categorizing IoT services, we aim to provide application developers a starting point, giving them something to build upon so that they know the types of services that are available. This will allow them to focus more on the application instead of designing the services and architectures required to support their IoT application.

II. TYPES OF SERVICES

There are an exceptional number of applications that can make use of the Internet of Things, from home and office automation to production line and retail product tracking. The number of applications is endless. For each application, a particular IoT service can be applied in order to optimize application development and speed up application implementation. Note that the categorizations that follow come from [3].

A. *Identity-Related Services*

Identity-related services can be divided into two categories, active and passive, and can serve either individuals or enterprise, which can lead to a number of different kinds of applications.

The general identity-related service consists of two major components: (1) the things, all of which are equipped with some kind of identification identifier, such as an RFID tag; and (2) the read device(s), which read the identity of the thing based on its label, in this case reading the information encoded into the RFID tag. The read device would then make a request to the name resolution server to access more detailed information about that particular device.

Active identity-related services are services that broadcast information, and are usually associated with having constant power, or at least under battery power. Passive identity-related services are services that have no power source and require some external device or mechanism in order to pass on its identity. For example, an active RFID tag is battery powered and can transmit signals once an external source has been identified. A passive RFID tag, on the other hand, has no batteries, and requires an external electromagnetic field in order to initiate a signal transmission. In general, active identity services can transmit or actively send their information to another device, whereas passive services must be read from.

B. *Information Aggregation Services*

Information aggregation services refer to the process of acquiring data from various sensors, processing the data, and transmitting and reporting that data via IoT to the application. These types of services can be thought of, more or less, as one way: information is collected and sent via the network to the application for processing.

Information aggregation services do not have to implement a single type of communication channel in order to work together. With the use of access gateways, an information aggregation service could make use of different types of sensors and network devices and share their data via a common service to the application. For example, an application could make use of RFID tags to be aware of the identity of some devices, while also using a ZigBee network to collect data from sensors, then use a gateway device to relay this information to the application under the same service, say a Web Service such as JSON or XML. Not only would this allow a developer of an application to incorporate a number of different technologies into the application, but it could also allow the application to access various IT and enterprise services that may already be in place.

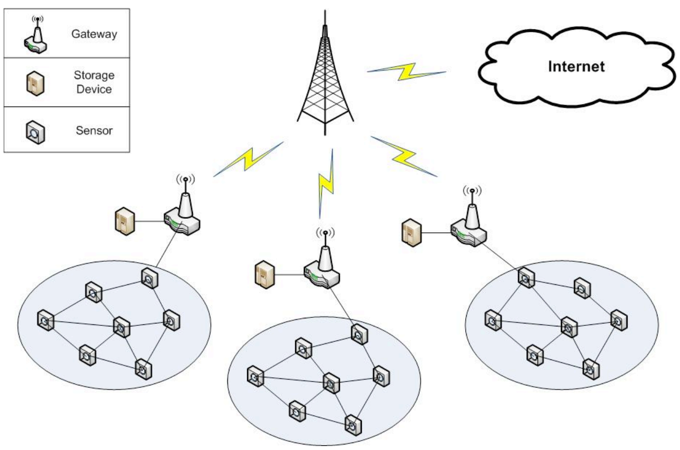

Figure 1: Aggregate Network diagram with sensor network and access gateways

C. Collaborative-Aware Services

Collaborative aware services are services that use aggregated data to make decisions, and based on those decisions perform an action. As IoT takes shape, it should bring about the development of complicated services that make use of all of the data that can be retrieved from the extensive network of sensors. This will require not only being able to retrieve information, but to relay back responses to the collected information to perform actions. These services will thus require "terminal-to-terminal" as well as "terminal-to-person" communication. By providing collaborative aware services, the IoT infrastructure naturally requires greater reliability and speed, and will require the terminals to either have more processing power or be linked with some other device that does.

D. Ubiquitous Services

Ubiquitous services are the epitome of the Internet of Things. A ubiquitous service would not only be a collaborative aware service, but it would be a collaborative aware service for everyone, everything, at all times. In order for IoT to reach the level of providing ubiquitous services, it would have to overcome the barrier of protocol distinctions amongst technologies and unify every aspect of the network. There is no particular system architecture for the Internet of Things, but there have been numerous papers written about the use of Web Services or REST (representational state transfer) APIs (application programming interfaces) to unite loosely coupled things on the Internet under a single application so that they can be reused and shared. IPv6 is also a protocol that could greatly benefit the increase in ubiquitous services. Reference [4] proposes such an architecture that, if implemented, would be considered a ubiquitous service.

III. Applications of IoT Services

Moving past what each of the categories means, the following subsections provide examples of each type of service in an attempt to offer developers a starting point when developing their own application. The idea is to provide a series of examples for each service type that use a common technologies so as to provide a basic framework to build an application upon a specific type of service.

A. Identity-Related Services

Identity-related services are the most simple, yet maybe one of the most important, services to be provided to an application of the Internet of Things. Applying an identity-related service to an application provides the developer with vital information about every device, or every *thing*, in their application.

The most prominent technology used in identity-related services is RFID. RFID is a technology that enables data to be transmitted by a tiny portable device, called a tag, which is read by an RFID reader and is processed according to the needs of that particular application. RFID provides an upgrade from the traditional form of device identification: barcode scanning. RFID is more versatile because it does not require line of sight transmission, and, in the case of active RFID tags, can transmit its data as opposed to simply just being read by a reader device.

Most IoT applications that are aimed at providing an identity-related service make use of RFID technology. As described in [5], the RFID tag stores an identification code unique to that device. The RFID reader reads that code, and looks up the device in the RFID server, which then returns the detail information require by the application.

Production and shipping are two common applications that would benefit greatly from the use of an identity service. Another application that uses an identity-related service describes a model that can solve the information asymmetry problem in supply chain management and supply chain information transmission [6].

Every IoT application will either be based on, or at least incorporate some instance of, an identity-related service. This is because for the IoT to incorporate everything in the physical world to the digital world, the application will need to be able to identify all of the devices that are connected.

B. Information Aggregation Services

Information aggregation services incorporate identity related services, along with other components such as Wireless Sensor Networks (WSNs), and access gateways to collect information and forward it to the application for processing. The information aggregation service is just responsible for providing the application with all of the information that is collected, and potentially processed along the way, from the terminals of the system (sensors, RFID tags, etc). In this regard, the WSN can be a powerful tool for collecting and communicating data between terminals and the platform (host of application), as long as the platform is within range of the WSN. But this would not be an IoT application on its own; an IoT application would consist of multiple WSNs all configured to work together to provide information about the world around them. The link between these networks is an access gateway. The general structure of this network is shown in Figure 2 below. Each access gateway in the IoT network will have access to the database server, thus every device would be connected and information from the entire network aggregated at the database server.

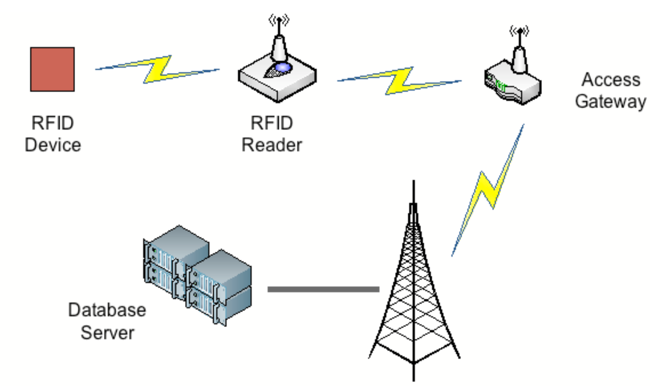

Figure 2: RFID network example

There are a number of applications out there that make use of information aggregation services and access gateways. In [7], the importance of extending the information aggregation service to beyond the WSN is proposed by using a cellular network (CN) to extend the range of the WSN. The idea is that if a terminal is outside of the WSN of interest, it uses CN resources to access that information through the use of an "IoT gateway," which essentially implements both WSN and CN resources.

Information aggregation services are useful in monitoring situations, such as energy monitoring in the house and in the enterprise, or, if the Internet of Things has been realized, monitoring of anything, anywhere. For example, [7] introduces a monitoring and control system for use in an agriculture greenhouse production environment. The system measures and records critical temperature, humidity and soil signals which is then transmitted through the network to the platform for processing. Another application [9] uses a ZigBee WSN to monitor physiological data of patients that automatically generates electronic medical records.

C. Collaborative-Aware Services

The key difference between information aggregation services and collaborative-aware services are the use of the data collected to make decisions and perform actions. As mentioned before, the keys to creating a collaborative-aware service are network security, speed, and terminal processing power. Terminals can no longer be just simple sensors that collect information, or if there are simple sensors in the network, there must be separate embedded devices within the network that can make use of the data.

There are fewer applications published in terms of IoT and collaborative-aware services. We can, however, attempt to apply new technologies to a collaborative-aware service. An example of a new technology that will help shape the way the Internet of Things grows is IPv6. IPv6 is a new version of the Internet Protocol (IP) that allows for a significantly greater number of addressable devices to be connected to the Internet. Although the use of IPv6 has had a slow start, it is definitely the Internet protocol of the future due to the lack of available IP addresses. Moving forward, one of the most important factors in IoT becoming reality is being able to address each of the embedded devices in the world, which converting to IPv6 would allow. Reference [10] offers a number of applications for IoT, many of which could be considered collaborative-aware services, or which could at least provide a baseline for such a service. They propose integrating every object into the IP infrastructure using both IPv6 and 6LoWPAN, which is the use of IPv6 over low power wireless personal area networks. They propose a network with three types of nodes, all of which can be reprogrammed to function as any of the three types. The three types essentially are a base station node (IPv6 router), a mobile node (wireless dongle that allows WSN connectivity to a standard laptop) and specialized nodes, which are used for specialized tasks. This becomes a collaborative-aware service because it incorporates terminal-terminal and terminal-person communication, which is accomplished due to the use of the IPv6 protocol.

D. Ubiquitous Services

Ubiquitous services are the ultimate goal of the Internet of Things, taking collaborative-aware services to the next level by providing complete access and control of everything around us, whether it be through a computer or a mobile phone or something else.

Ubiquitous services have yet to be realized in the world today, but most research in IoT is ultimately aimed at providing some piece to the puzzle that will ultimately be ubiquitous services. In [4], the authors first talk about why the Internet of Things is so difficult to realize. One of the biggest hurdles for IoT is having a single architecture that allows the many different application layer standards to communicate and interoperate. The authors in [4] proposed an architecture, based on RESTful services, in which a universal API would be created so that everyone who creates devices to be used in the Internet of Things has an architecture to adopt in order to be interoperable with the rest of the world's devices.

IV. CONCLUSION

This paper outlined the four main categories of services of the Internet of Things and attempted to provide some examples of each in order to give the developer of an IoT application a starting point for his application. Many people are using Web Services and access gateways in order to interface with the terminals, while others are moving towards the use of IPv6, which will allow for more devices to be connected directly to the Internet and be IP addressable, as opposed to being a part of some subnetwork that is connected to the Internet through a gateway. Overall, there is still much work to be done in IoT, specifically in finding a way to incorporate all of the services into an omnipresent, omnipotent service aimed at delivering communication anytime, anywhere, for anybody, and for everything.

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Reducing Travel Time by Incident Reporting via CrowdSourcing

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Abstract— *The contribution of this work is the creation of a novel system that enables motorists who witness incidents to submit reports to our system via the web. These reports are aggregated, validated and verified automatically. Then, they are used to update the road network graph. In this work, we designed and implemented an incident reporting system whereby users can report an incident such as an accident or construction on a road network. We extended the FreeSim simulator to accommodate our incident reporting system. Experimental results showed that our system is capable of reducing the travel time of users. We also presented our verification algorithms that are used to verify that reports are fact. Current approaches to congestion detection such as loop detectors probe vehicles, and video image detection may not be available on arterial streets. These technologies may not be available in countries whose transportation budget is low. Our model is less expensive, easy to implement and can work in any environment (e.g. extreme weather). This work is dependent on people's incident response input and not from sensor signals converted to traffic measurements. To the best of our knowledge, this approach is the first to consider web-based incident Crowd Sourcing with automatic incident verification. Other driver based models are telephoned based.*

Keywords: Crowd Sourcing, Shortest path, Vehicular networks, Incident reporting,Travel time reduction

1. Introduction

The aim of this research is to provide end users with the most optimal route for a particular trip. Most routing engines are based on static information and do not take congestion into route planning on arterial streets. We clearly differentiate between *recurring congestion* and *non-recurring congestion*. Recurring congestion is caused by peak travel time when most motorists are expected to be travelling. Figure 1 depicts recurring congestion during a single day in a city. Non recurring congestion is spontaneous and may result from accidents, road maintenance or extreme weather. This project is based on the latter, which is non-recurring congestion, since 2/3 of traffic delays are caused by non recurring incident based congestion [1]. This system proposed and presented in this paper enables the end users to save time and energy on a trip. Congestion information updated by motorist will not include multimedia data. Instead the information sent will be text based because of the limitations of mobile devices (e.g. battery life) and the demands of multimedia (e.g. processing power and storage). According to the 2007 Urban Mobility Report, delays due to heavy traffic are now costing Americans $78 billion in the form of 4.2 billion lost hours and 2.9 billion gallons of wasted fuel [1]. As seen in Figure 1, vehicular traffic congestion is a growing problem with more drivers experiencing severe and extreme delay where the travel time is in excess of 1.5 times of the free-flow trip time. In addition, 2/3 of traffic delays are not caused by recurring congestion. However, due to the traffic incidences, they are caused by point-based spontaneous congestion [1]. We intend to enable the latter, which is spontaneous congestion, to be taken into consideration when returning a trip to the end user.

Effective routing engines must consider both recurring traffic patterns and non-recurring spontaneous traffic events. Also, We intend to take this work further by including spontaneous causes of congestion such as accidents and construction work in our model. We refer to accidents and construction work as incidents. For a transportation graph G = (V,E)where V is a set of vertices and E is a set of edges. V corresponds to a point where two roads intersect and E is the road connecting two vertices. Congestion on the road graph results in an $edge\ e \in E$, that becomes larger for travel time or less for current travel speed and thus the characteristics of G has changed and resulted a new shortest path graph. For the congestion, it should be observed that the edge e

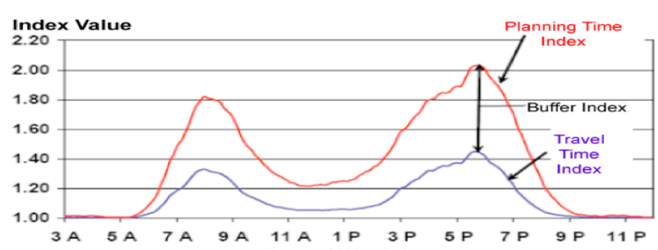

Fig. 1: Diagram Illustrates the Recurring Vehicular Traffic Congestion

will return to its original value after a period of time.

The goal is to be able to tackle incident based congestion (e.g. peak time, accidents, events such as a football game and constructions). Our application accepts congestion or incidents reports from third parties (e.g. police, ambulance drivers, highway patrol and motorist) into consideration when returning possible trip routes to end users of the system. Our goal is not to create a trip planner, instead, it is to update the routing server of trip planners with congestion information to produce more accurate trips to end users.

Accident and Construction reports will be able to be submitted by persons (e.g. police, ambulance, fire personnel and regular motorists) at the scene of the incident with regular motorist given lower priority due to trust factors. Current incident detection frameworks do not have users to submit incidents using the web and automatically verify the reports [12]. Normally, the users may make a telephone call to the transportation center and report the incident information. Using the GPS sensor on the person's mobile device maybe fruitful in identifying an incident's location. Our approach is a web-based incident Crowd Sourcing system with automatic incident verification. Some other strategies to detect incident in prior work are loop detectors, CCTV imaging, probe vehicles and microwave radar [12], [13]. These may not be available on arterial roads or in countries who's transportation budget is low. The remainder of the paper is organized as follows: Section 2 explains the prior work. Sections 3, 4, 5 and 6 introduces new edge weight computation, validation and trust, updating shortest path graph and system design respectively. Section 7 contains the simulation, algorithms, experiments and system evaluation. Section 8 and 9 contains the future work and conclusion respectively.

2. Related Works

A wide variety of sensors can be used to determine where and when an incident has manifested. Inductance loops, microwave radar, probe vehicles and video detection are some of the common sensors based approach [12], [13].

Inductance loops are inexpensive, robust and common in incident detection [12], [13]. The pitfall of inductance loop is that they require road closure for installation and maintenance. Also, these devices under perform whenever the weather condition is extreme, such as several inches of snow on the ground that cover the inductance loop. Video based approaches also have limitations [12], [13]. Since they can be installed above ground, road closure may not be necessary for installation and repair. Video detection may under perform during extreme weather such as fog or any other low visibility weather conditions. Also, we may have the scenario where vehicles hide each other from the cameras. For example, a large truck may hide several cars from the camera and the correct speed of the cars become undetected. Probe based vehicles [13] can be broken down into two categories which are *Beacon based* and *GPS or positioning based* [13]. Beacon based relies on a device called a beacon that uses Dedicated Short Range Communication (DSRC) to communicate with vehicle tags as vehicles passes the beacon. By observing the temporal and spatial components as vehicles pass from one beacon to another, the travel time can be determined. The pitfall of this approach is that all vehicles in this system are expected to have a tag that can be used to communicate with the beacons. GPS or positioning based eliminates the cost of constructing a beacon based network. This facilitates more coverage of the road networks. By taking the GPS coordinates of a probe vehicle at different time intervals, we may be able to infer the current status of the road network.

Observe that the models described above are not common on arterial roadways. Furthermore,it is unrealistically expensive to deploy any of the above systems over the entire road network of a country. The models above operate by first determining a threshold in terms of speed or travel time for a road network. Then, algorithms continue to take sensor readings and if the speed or travel time falls below the threshold an alarm is triggered[12]. Our approach is less expensive and can be used under extreme weather conditions. Based on the fact that we consider people as sensors, the entire road network of a country can be covered. Observe, our approach is not the first driver based model [12]. Other driver based models are not done via the web. Instead, they rely on phone calls for reporting. One of the challenges we faced in driver based models is the inconsistency of the reports submitted by different users.

3. New Edge Weight Computation

One of the major issues for this research is the re-computation of the weight of the road edge that has been affected by the incident. In transportation graphs, edges have weights which may be related to average time to travel an edge or the length of an edge. In our system, the weight of an edge is the current travel speed. We assume a free flow travel speed of 65 mph in this work. If there is an incident on an edge, then the edge attracts a new weight. The research issue is how to determine the new value that should be assigned to the weight of the affected edge. The approach that we use to determine the new edge weight, is dependent on the verification rule that is selected in our system.

Our two verification rules are discussed thoroughly in the algorithm section. The first rule is called "majority rule" and the second called "GPS rule". For the "majority rule", if the reporter is a "regular motorist", then the incident is treated as a fact if the total number of distinct reporters reporting the same incident exceeds some threshold. In our system, for regular motorist the threshold used is three. We then take the mean travel speed of the three reports as the new edge weight. Additionally, if the incident is reported by a uniformed personnel (e.g. police, highway patrol and ambulance driver), then we treat the incident report as a fact immediately since we assume uniformed personnel are more trusted than regular motorists. In this case, we use the current travel speed reported by the uniform personnel as the new weight of the road edges affected by the incident

For the GPS rule, we treat an incident that is reported as a fact if and only if the reporter is close to where the incident has occurred. Closeness is determined by extracting the reporter's location coordinates (latitude and longitude) and then computing the Euclidean distance between the reporter and the incident's location point. This rule enforces the fact that persons can only reports incidents that they are a close distance from. Privacy issues concerning the location of the mobile user is addressed in our prior work [15], [16]

4. Validation and Trust

While reputation-based trust management protocols have been proposed for mobile wireless networks, the scale and potential dangers of misinformation may lead the system into chaos. The large number of motorist pose a problem, which may be reduced if we have persons willing to participate in congestion control conduct a registration and then we issue to the users them a key or unique id. When these persons witness an incident and see traffic piles, they can report online to the validation system.

We define an end user of the system as persons who submit request for trips. And a trip is defined as a route between two points, a start point and a destination point. We also define an untrusted user as a motorist who is at the scene of a congestion scenario and need to update our routing component so that other motorist can be presented with trips around the congested area. A trusted user in our system is defined as an authority with more permission or trust such as police, ambulance personnel and fire department personnel. For the untrusted users, we will require them to sign up to participate in congestion management afterwards they are issued with a key to use our system. The virtual ratings of this key may increase or decrease depending on the information they provide. We could also offer virtual incentives if the incident information that they provide is correct. The validation for this type of user (untrusted) will be more rigorous than the trusted user. The robustness of the proposed model is determined by the number of similar reports that must be received before the incident is verified as a fact. Also, the constraint that ensure that users must be close to an incident to submit a related incident report (GPS rule).For trusted parties, the validation system may be less vigorous since we assume that these persons have a higher truthfulness level.

5. Updating Shortest Path (SP) Graph

After we compute the new weight for the incident edge, recomputing the entire shortest graph may be time consuming. To compute the shortest path between two vertices in a directed graph, there are three general classes of algorithms which are Naive, Dynamic class and Pre-Computed class [14].

The *Naive* class of algorithms includes Floyd-Warshall's Algorithm [3] and Johnson's Algorithm [4]. Both of which compute the shortest path for all pairs of vertices in a graph. The complexity may be up to V^3. These algorithms, with respect to the specific problem discussed, are required to be executed every time a link update occurs, which is when a trusted/untrusted third party update has been validated. Although the other algorithms to be discussed next may be faster for updates, re-running these algorithms every time a edge update is received allows the fastest path to be determined in constant time, since the fastest paths will already have been computed.

The *Dynamic* All-Pairs Shortest Path algorithm, developed by *Demetrescu and Italiano*, attempts to improve over the naive algorithms by determining the path with minimum cost between two vertices in a graph in constant time where there is an edge update cost [5], [6].

The *Pre-Computed* class of algorithms [14] takes advantage of the fact that the graph is static. With no edge insertions, all of the paths between all pairs of nodes can be precomputed. Then, they do not require the application to take the extra step of determining all of the paths between two vertices when a request for a fastest path is being answered [14].

The *Constant Update Algorithm* [14] sacrifices the speed of retrieving fastest paths for the amount of time it takes to update the weight of an edge. The algorithm does not maintain the fastest paths at all times, but does compute the fastest path when requested by an end user. When the speed on an edge has changed by a specified threshold, the time to traverse that edge will be updated, which can be accomplished in constant time [14].

The *Constant Query Algorithm* [14] sacrifices the speed of updating an edge for the time to retrieve the fastest path. This algorithm always maintains the fastest path between all pairs of vertices by recalculating the fastest paths for all pairs of vertices that have a path containing the updated edge whenever an edge update occurs. When the speed on an edge has changed by a specified threshold, the time to traverse that edge will be updated. Also, the fastest paths for all pairs of vertices, that have a path containing that edge, will be recalculated. In our system, an edge is updated with its new weight each time an incident is verified.

6. System Design

The operation of the incident reporting system is shown in Figure 2. We clearly differentiate between the two types of users that submit incident reports in our system. First, we define a trusted user according to our system as a "uniformed personnel" such as police. The reports, that these personnel generated are more trusted than "regular

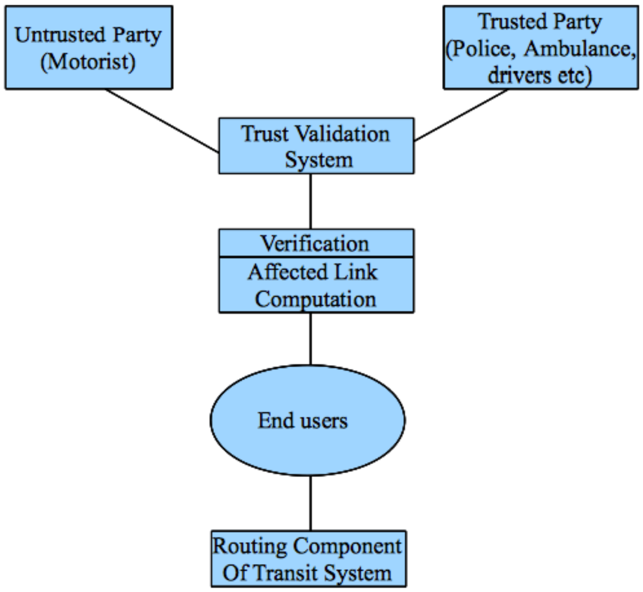

Fig. 2: Diagram Illustrates the System Design

motorist" and incident verification for these reports, are less rigorous. A single report by a "uniformed personnel" may be considered as a fact. Secondly, we classify the other motorist as untrusted parties. These reports, that are generated by untrusted parties, may be more rigorously verified. In our system, if the same incident is reported at least three times by three distinct untrusted parties and the users are close to the incident, then the report is considered a fact.

A user, either trusted or untrusted, submits a report reporting an incident that they witness on a road network. We first validate the user to ensure the user is who they claim to be and is authorized to use the system. After validation we then consider verification which is done by two rules conjunctively (1) Majority rule and (2) GPS rule. These rules are used to verify that an incident, that was reported, is a fact. When verification is completed, the next step is to compute a new weight for the incident edge. This may be done by taking an average of all the current travel speed reported for that edge by the untrusted users. The weight of the edge may also be taken directly from the incident report by the "uniformed personnel". The current travel speed of the incident edge is a parameter for report input.

Finally, the routing component, in our case FreeSim traffic simulator [2], [8] is updated with the new weight for the incident edge. In publicly available transit itinerary planning systems, the updated incident edge update would be sent to the routing engine of the transit itinerary planning system e.g Graphserver. End users, that request shortest path from an origin to a destination, may now get a path where the incidents are taken into consideration when returning the shortest or fastest path.

7. Simulation and Experiment

We extended the FreeSim freeways simulator [2], [8] to accommodate our congestion reporting system. FreeSim is a transportation simulator that enables users to insert transportation graphs and query the shortest or fastest point from an origin to a destination. The simulator implemented a variety of shortest path algorithms such as Dijkstra [9], Bellman Ford [3] and Johnson's [4]. For our experimental evaluation, we used two real world road network. The first is a road network representing the University of Illinois area of Chicago (UIC)consisting of 14 nodes and 15 edges. The second road network is represented a subsection of the Los Angeles area consisting of 50 nodes and 64 edges. A map of the UIC evaluation area in Chicago is shown in Figure 3. The simulation environment was a HP Notebook PC running Windows Vista containing a P8400 Intel DUO 2.27 GHz processor with 4GB RAM. The system is developed using JAVA and the development environment is Eclipse version 3.4.1.

7.1 Architecture and Methodology

In Figure 4, we discuss the architecture of our system. We assume that each user has a mobile device with an Internet connection. Each mobile device have access to positioning information such as GPS, cell tower triangulation or WiFi positioning.

A user *u* submits a request to our intelligent transportation simulator simulator FreeSim for the shortest or fastest path from an origin node to a destination node on the road network graph. FreeSim engine then runs a selected shortest path algorithm on the graph and returns the shortest or fastest path as requested by the user. All users in our system, that are traversing the road network, have access to the incident

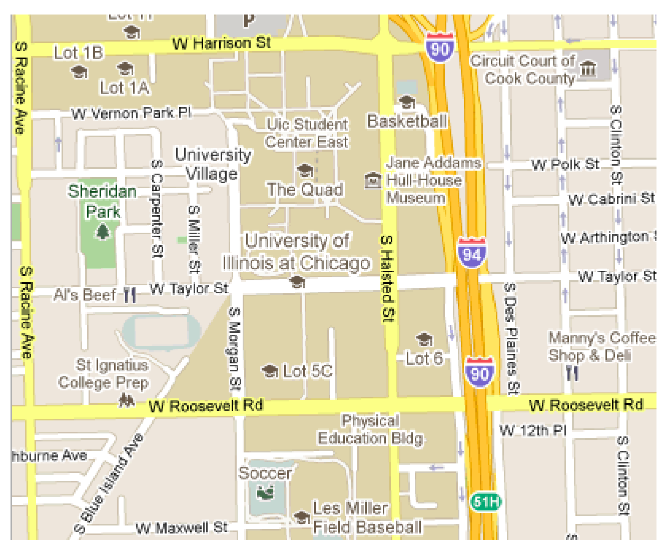

Fig. 3: Diagram Illustrates the Evaluation Area

reporting system and report an incident that they witnessed. A typical report has the following properties *reporter name, reporter id, incident, report location*. The *reporter name* and *reporter id* is the name and identification of the person that submit the incident. The report location is the latitude and longitude of the reporter which may be accessible if the users have positioning features on the mobile device. If the mobile device is limited and cannot infer a latitude or longitude, our system still works. An incident in the report has the following properties *incident location, current travel speed, incident time, incident type , source node, destination node*. Below we explain the properties of an incident as defined in our work.

- *Incident location* represents the latitude and longitude of the incident
- *Current travel speed* represents the current travel speed of vehicles after the incident. Before the incident, we assume that vehicles travel at 65mph.
- *Incident time* represents the time of the incident
- *Incident type* represents the type of incident example accident or road construction
- *Source node and Destination node* are nodes defined by our system. The source node is a node before the incident point and the destination node is a node after the incident point.This information assist us in detecting the incident edge.

7.2 Data Structures

Three hash maps are used to track the reports and the incidents. There is an hash map called *allIncidents* that keep track of all incidents reported in the system. Another hash map called *allReports* that maintain a list of all reports in the system. The reports are stored with reporter identification as index in the hash map. If an incident is confirmed by the GPS rule or majority rule, that report is added to the*confirmedReports* hash map and removed from the *allReports* hash Map. Before we add a report to the confirmed hash map of reports, we first verify that the hash map does not already contain the possible new report.

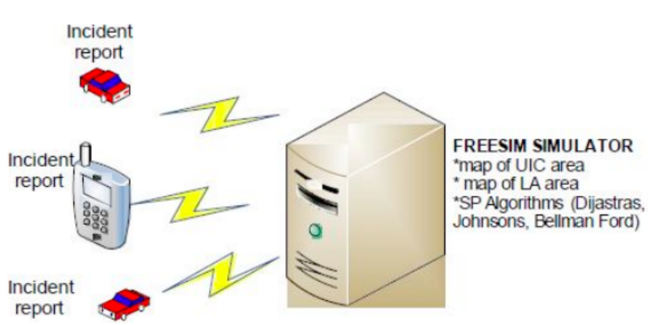

Fig. 4: Diagram Illustrates the System Architecture

7.3 Algorithms

We discuss two algorithms that we used in our system to verify an incident that was reported is a fact. The first algorithm is called *majority rule*. It considers a report or set of reports as a fact if the same incident was reported by at least three different users in the case of untrusted users. The second algorithm is called *GPS rule*. It considers a report or set of reports to be a fact if the location that a user submitted a report from is in close proximity to the report's incident location. Both rules have to be satisfied before a report can be treated as a fact. Algorithm 1 is our majority rule verification algorithm. These algorithms are used each time a new report is entered into the system. In the first step, the algorithm checks to see if the user that submitted the incident report is a "Uniformed Personnel" or not. If the user is uniformed personnel, then the data that they provide is more trusted and a single report from these personnel are considered as a fact. We then update the traffic simulator and add to the confirmed incident hash map the new report and update the road network graph in constant time with the new edge weight. Secondly, if the reporter is a "regular motorist", the algorithm compares against all other reports in the system searching for reports with a common incident source and destination. If the algorithm discovers at least three other reports from different users in the system, then that incident is confirmed. For all confirmed incidents, we update FreeSim traffic simulator with the new weight of the edge containing the incidents. The weight of the new edge is determined by taking an average of the current travel speed as reported by the three users. In line 17 of the algorithm, before updating the simulator, we ensure that this report is not duplicated the list of confirmed reports.

Algorithm 2 is our GPS rule for incident verification. For

Algorithm 1: Majority Rule

```
     precondition: allReports !=null
     input: Hash Map allreports,confirmedReports
                  /* Hash map of all reports and confirmed reports in the system*/
          Report newReport
     method:
1.        if(newReport.reporterId == "Uniformed Person")
2.             allConfirmedReports.put(newReport);
3.             updateTrafficSimulator()/** update FREESIM **/
4.             return
5.        endif
6.        int count = 0;
7.        for(j=1 to allReports.size())
8.         if(allReports.get(i)!=null && newReport!=null)
9.          if(allReports.get(i).incident.sourceNode==newReport .incident.sourceNode)
10.      if(allReports.get(i).incident.DestNode==newReport .incident.DestNode)
11.                 if(allReports.get(i).reporterId!=newReport .reporterId)
12.        count++
13.      endif
14.     endif
15.    endif
16.        endif
17.       if (count==3&& newReport!ϵ allConfirmedReports)
18.            allConfirmedReports.put(newReport)
19.            updateTrafficSimulator()/** update FREESIM **/
20.            allReports.remove(j)
21.       endif
22.       endfor
23.
24   end
```

each report, the rule operates as follows. For each report, it checks to see if the incident location is close to the reporters location. This clearly means that reporters can only report incidents that you are close to. To determine if a reporter is close to an incident we compute the Euclidean distance between the reporter's location and the incident's location. Even though we refer to the ruleas GPS rule, positioning techniques used by this rule is not limited to GPS. Other positioning techniques may be considered such as cell tower triangulation or WiFi positioning.

7.4 System Evaluation

In Figure 5, we present the system during operation. The graph is a section of the Downtown area of Chicago in particularly of the UIC area along Halsted Street. In the diagram above to the left is the FreeSim traffic simulator [2], [8] and to the right is the incident reporting system that we extended the simulator to accommodate. The nodes in the graph are as follows starting from the upper left: Halsted and 20th, Halsted and 19th, Halsted and 18th, Halsted and 17th, Halsted and 16th, Halsted and 14th, Halsted and Roosevelt, Roosevelt and Morgan (South West of Halsted and Roosevelt), Roosevelt and Jefferson (North East of Halsted and Roosevelt), Halsted and Taylor (South of Halsted and Roosevelt), Halsted and Polk (South of Halsted and Taylor), Halsted and Harrison(South of Halsted and Polk) Morgan and Taylor (West of Halsted and Taylor), Harrison and Jefferson (East of Halsted and Harrison).

In the demonstration in Figure 5, the user submits a query for the fastest path from Halsted and 20th as origin and Halsted and Harrison as destination. The path recommended is highlighted in red as shown above. The total distance to be traveled is 10.30 miles and duration of travel is 9 mins and 30 seconds. The shortest path algorithm used is Dijkstra [9]. We assume that users travel at 65mph and that each block is 1/10 of a mile. Also, we discuss the situation where an incident is reported and verified and the simulator is updated with the new edge weight after a set of incidents have been reported and verified. In Figure 6, an incident is reported by a user (Alice). The incident is reported to have occurred on the edge between Halsted and Roosevelt and Roosevelt and Jefferson and the current travel speed is 1mph. Two other users before

Algorithm 2: GPS Rule

```
precondition: allReports !=null
input: Hash Map all reports,confirmedReports
        /* Hash map of all reports and confirmed reports in the system*/
method:
    for(j=1 to allReports.size())
      if(allReports.get(i)!=null)
        if(allReports(i).incidentLatitude closeTo allReports(i).reportLatitude)
            if(allReports(i).incidentLongitude closeTo allReports(i).reportLongitude)
                allConfirmedReports.put(allReports.get(i))
                updateTrafficSimulator()/** update FREESIM **/
                allreports.remove(i)
            endif
        endif
      endif
    endfor
end
```

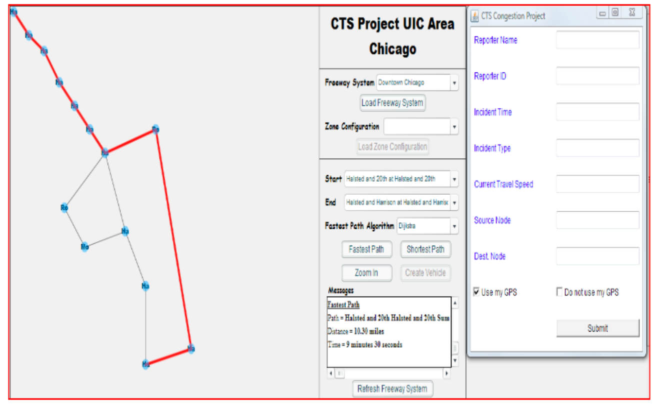

Fig. 5: Diagram illustrates the System GUI-A

Alice had submitted a similar report on the incident and was close to the incident. Based on the majority rule, the incident is verified and the updated edge speed sent to the simulator as shown in blue in the message section of the simulator. After the edge is updated, another user requested the same query as described in the first screen shot. The new fastest path is highlighted in red above and takes 10 minutes and 26 seconds and the total distance is 11.30 miles. Observe that, if the previous shortest path as in the first screen shot was taken by the user, in the event of the incident, the total travel time would have been 27 minutes and 14 seconds. This reduces the travel time of Alice by over 16 minutes. Also, there is also a reduction in fuel consumption, however we did not compute the amount of fuel that was saved.

8. Future Work

We intend to extend this work by using a mobile object generator [10], [11] to generate synthetic users moving in the streets and submitting requests to the system. These mobile objects generated by the moving object generator will report

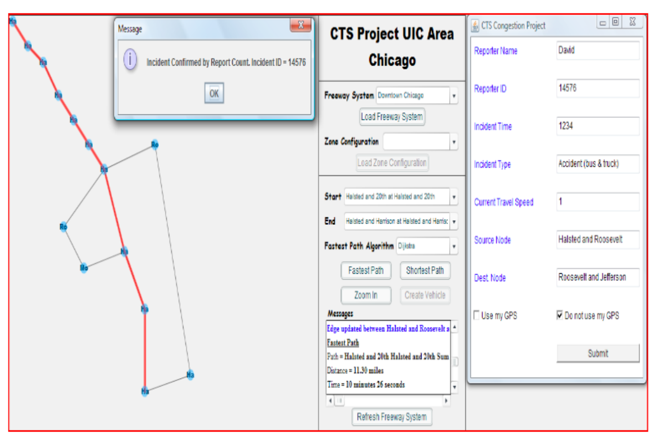

Fig. 6: Diagram illustrates the System GUI-B

incidents and the current travel speed of an edge. Incidents will be generated randomly on the road network, statistical results could then be used to show how much time can be saved by the usage of our system.

9. Conclusion

We present a system that reduces the travel time and fuel consumption of motorist whenever there is an incident on the road network. In our system, the road network graph is very dynamic and new edge weights are added to the graph when incidents are detected. The system we created is an extension of the FreeSim traffic simulator [2], [8] to accommodate our incident reporting system. Algorithms that we used for incident verification in our system are presented. Also, a demonstration example of our system during operation is provided. The demonstration shows a motorist saving over 16 minutes of travel time when our system is used.

Current sensor based models for incident detection may not be available on arterial streets. It is impractical to cover the entire road network with these sensors. Our approach uses people as witnesses and sensors. Clearly, in our model, users can perform collaborative cheating to lead the system into chaos. In subsequent work on this paper, the plan is to develop a collaborative cheating prevention algorithm for this system.

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Routing with QoS using bioinspired Models: An overview

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***Abstract*—During the past forty years, routing services have tried to solve problems of the best route without considering real Quality of Service parameters and just in the last fifteen years researchers have been analyzed natural behaviors for developing routing algorithms using analogies with bio-inspired models. The use of bioinspiration for routing protocols has many advantages and their application over next generation networks could improve the users networking experience. This paper tries to show some routing bioinspired models in order to find research opportunities to finally propose an analysis with hybrids implementations.**

I. Introduction

The natural mechanism has been the inspiration to solve routing problems with multiple QoS constrains, this problem has been classified to be NP-complete and complex to solve [1]. Different kind of heuristics and nature mechanism such as Genetic algorithms, Ant Colony Optimization, neuronal networks, Particle Swarm Optimization, has helped to solve this kind of complex problems demonstrating good solutions with rapid convergence time. In the same way the application of pure heuristic to solve multiple QoS routing problems have showed some lacks, then hybrid algorithm approaches has been proposed by many researchers in order to solve the heuristics problems, the paper presents some hybrid algorithm and the features extraction in order to apply them to other heuristics, the heuristics lacks and simulation results comparing the performance of pure heuristics and hybrid heuristics algorithms.

The rest of the paper is organized as follows: Section II shows a brief about some bioinspired models used in routing protocols with QoS and their bases on natural behavior. Section III presents a in-deep review of two Routing protocols with QoS constrains based on Bioinspired model and some introduction to different kind of networks which routing protocols are designed for, finally shows a general review of some heuristics proposed in other Routing protocols. Section IV presents some approaches using hybrid models, some heuristics lacks and features to overcome them. Section V shows conclusion which abstracts approaches with hybrid heuristics.

II. Bioinspired Models

Different kind of bioinspired models have been used based on natural behavior or biological features to solve routing problems with multiple constrains, this section makes a review to some of them.

A. GA. Genetic Algorithms

GA (Genetic Algorithms) were proposed by Holand in 1970 [2] with an Algorithm called Simple Genetic Algorithm [3]. This kind of algorithms are based on evolutionary mechanism, one example is the competence between individuals in order to survive and to adapt to the environment. This adaptation is controlled by a unit called gen, a set of gens makes a chromosome, these chromosomes have the information which will be exchanged between individuals, in order to exchange the information there exits a number of genetic operator which are: Crossover, mutation, selection. Crossover let to exchange gens between chromosomes to get information about order individuals. There is a value associated to each chromosome "the Fitness", this fitness value let to know how good is one chromosome compared with others chromosomes. The highest fitness chromosomes have more probability to continue in the evolution process, in the same way the lower fitness chromosome will be eliminated. The selection operator is used to select the chromosomes based on their fitness value, different methods for selection are used, one of them uses the average fitness value within the complete population divided into the fitness per each chromosome, this approach allow to control that only the chromosomes with a fitness value higher than the average fitness value could be selected, there is another method called roulette wheel selection [2]. The mutation operator can change some information randomly and sporadically based on a mutation probability, it is used to scape from local optimums. PGA (Parallel Genetic Algorithms) are variants of GA, this algorithms was developed due to the extremely computer complexity in the implementation of GA, it is intended for adapting the heuristics to parallel computer systems. Three groups could be found: Master-Slave Algorithms, finite-grained algorithms, coarse-grained algorithms [4] [5] [6].

B. Neuronal Networks

A neuron is the basic unit of the brain, they can process signals received from the dendrites and pass them to other neurons through the axon, the action of processing a signal is called “fire”, there is a threshold level which makes a neuron to fire. Each neuron is connected to other neurons through the synapses and not all the connections are equal, they have some weights assigned depend on the connection. The neuronal networks has to be trained in order to recognize some patterns, the training is the action to assign some weight values to the connections. Different ways to train the neuronal networks have been developed and can be grouped in supervised and unsupervised. Supervised method is based on the action to give some data to the input of neuronal networks and some sample of the output from this given input, the training process is repeated until the output error is reasonably small with the comparison to the given output and the actual output of the neurons. Unsupervised training is equal to supervised method unlike there is not anticipated output. Another hybrid methods have been worked which uses both supervised and unsupervised training. Neuronal networks are divided in layers. Layers are the group of neurons that perform similar task and can be divided in three types: Input Layer, Output Layer and one or more Hidden Layers [7].

C. ACO. Ant Colony Optimization

Ant Colony Optimization heuristic was proposed in early 1990s by M. Doringo as a possible solution of combinatorial optimization problems based on ants behavior [8]. The ants have some features such as self-organizing and foraging behavior which are the bases for many studies and have helped to develop different kind of heuristics to solve problems, one of the most successful algorithms based on ants is ACO [9]. This algorithm uses the foraging behavior in which the ants travel from the nest to the food, in order to find the best path, the ants use a chemical substance called pheromone. The pheromone allow the ants to follow the path with higher pheromone's level, in this way the shortest path will be selected (the ants will have more probability to go to the food and return to the nest fastest than in other cases). Deneubourg [10] made an experiment called “Double Bridge Experiment”, the nest of Argentine ants was situated near to a food source and separated by two bridges with equal distances, the first ants randomly explored the path to the food choosing different bridges, after some time a bridges was elected because of random conditions and pheromone concentration. Another experiment was done with different distance bridges, it showed the shortest bridge was chose to get food. This kind of experiment was the base to develop ACO. It uses artificial ants with some special features such as memory and pheromone evaporation [11], in real ants colony, the level of pheromone intensity decreases over the time and this method is applied to artificial ants in order to allow to explore other paths to the food source, in the same way memory is applied to avoid loops.

D. PSO. Particle Swarm Optimization

Particle Swarm Optimization was firstly introduced by James Kennedy (Social psychologist) and Rusell Eberhart (electrical engineer) in 1995 [12], based on the social behavior of some species such as bird flocking, fish schooling, and swarming theory. In PSO there are some entities which are called particles, this entities are placed in the space of the problem and constantly are looking for an optimum value. PSO are similar to GA, in which a population is generated randomly (possible solutions to a problem) and through the generations and optimal solution is updated. Unlike Genetic Algorithms, Particle Swarm Optimization does not use genetic operator such as mutation and crossover, then in order to get the optimal solution the particles follow the optimal particle through the solution space. In the original version of PSO proposed by Kennedy and Eberhart each of the particles have three dimensional vectors associated, current position $\vec{x_i}$, previous position $\vec{p_i}$, and velocity $\vec{v_i}$, this values are described similar to fitness value, which allow to know how good is the solution in reference with others. The best solution found so far is stored in a variable called $pbest_i$ (previous best), this variable is updating constantly in order to get the best solution [13].

III. Routing Protocols

Different kind of Routing algorithms have been proposed for multiple networks with multiple bioinspired models, this section presents an in-deep review of two algorithms, an introduction to some kind of networks used for the algorithms and a general review of the application with bioinspired models to QoS routing problem:

A. WMSN. Wireless Multimedia Sensor Networks

Wireless Multimedia Sensor Networks has been deployed in the recent years due to the facility in the implementation and acquisition of hardware such as CMOS cameras and Microphones, this kind of devices are able to ubiquitous capture video, audio, images, temperature, humidity and pressure information. WMSNs has been study from many researches because of its different abilities to collect information from the physical environment, process and transmit it to remote devices in different locations. This features shows an extensive range of new applications intended to combat the crime and terrorist attacks monitoring or controlling public events, private properties and borders. The vehicular traffic can be controlled in order avoid violations or car accidents, thefts, and develop systems to offer route information depend on the state of the traffic at different time hours. To support the multimedia traffic it is necessary to develop algorithms that deals with the different kind of QoS for specific applications requirements [14]. The architecture in this kind of networks normally works with many sensor nodes and one principal node called sink node or Gateway sensor node.

1) Routing based on Genetic Algorithm of Game Selection: The algorithm proposed by CHEN Niansheng ,LI Zhi, KE Zongwu ,GUO Xiaoshan in 2010 from the HuBei Normal University , China [15], is an hybrid Genetic Algorithm and Game selection routing protocol for WMSNs with QoS constraints, in last paragraph QoS is a main topic for Multimedia Services (real-time images, audio, video). In the proposed algorithm the selection genetic operator is used with the Nash equilibrium solution of select game. The QoS parameters for path selection in the route algorithm are bandwidth, delay, loss and cost. The principal goal of the algorithm is to find an optimal multicast tree which deals with the multiple QoS constrains in WMSN. In order to evaluate a multicast tree solution found by the algorithm there is a Fitness function associated to each one, the fitness is based on QoS values and only gives a fitness value to solutions (multicast trees) that satisfy the QoS constrains. The architecture proposed runs the algorithm in the sink, ones an optimum path is found the sink node sends it to the corresponding nodes.

$$f(x)=\begin{cases} \frac{1}{cost(t)}, & b(t) \geq B_{min} \wedge d(t) \leq D_{max} \wedge l(t) \leq L_{max} \\ 0, & \text{others} \end{cases}$$

Where:

- $b(t)$ is the minimum limit bandwidth value of the multicast tree.
- $d(t)$ is the delay value for the multicast tree.
- $l(t)$ is the lost packet rate.
- $cost(t)$ is the average value from the source node to the sink node.

Instead of use roulette wheel in the selection algorithm, the authors propose a selection game $G = \{N, S, U\}$ where:

- N is the number of players if $N \geq 2$.
- S is the strategy per player. $S_i = \{0, 1\}$. If $S_i = 1$ it is supposed that player i is selected. If $S_i = 0$ it is supposed that player i is failed.
- U is the payoff function.

For the payoff function, node i Energy is used, in this way the lifetime of the node and in general of the network is used to evaluate the performance and to pay to each of the nodes based on it. Finally a Nash equilibrium for selection game is calculated.

The general description of the algorithm is as follows:

1) Generating the initial population with the random walk method.
2) Calculating the fitness value of initial population.
3) Copying the initial population as a child population.
4) set gen=1.
5) while(termination condition is not satisfied) do:

{

- Retaining the best individuals to new populations.
- Using game selection algorithm to select other individual of a new population.
- Crossover operating on the new population.
- Implement mutating on the new population.
- Set gen=gen+1.

}

The gen variable is a counter for the genetic algorithm iterations.

Genetic Algorithm - Game Selection show a convergence of the algorithm (the fitness value keeps stable) around 30 generations and the average fitness value is closed to the best fitness value after the convergence time. In the same way the algorithm was compare with basic GA algorithms, the results show the game selection GA algorithms can improve the lifetime of the network.

B. WMN. Wireless Mesh Networks

Wireless Mesh Networks has been widely deployed in different environment : home networking, high speed metropolitan networks, enterprise networking and more. Because of their capabilities, more than MANETs, WMN can improve some aspects: simple network maintenance, robustness, power supply for routers. The WMNs are composed by two kind of nodes [16]: mesh router and mesh clients. The infrastructure could be classified in three types: Infrastructure-Backbone (Many routers giving connection to wireless or wired clients to Internet), Client WMNs (A kind of Ad-Hoc network) and Hybrid WMNs. Different groups are working on some new features or specifications for WMNs such as IEEE 802.11, IEEE 802.15, IEEE 802.16. At the same time technologies like MIMO has been developed to improve the capacity and flexibility of the networks. WMN special capabilities in comparison with traditional Wireless networks are self-configuration and self-organizing [17] which means they can automatically establish and maintain connection.

1) Multi-Constrained QoS Routing Optimization of Wireless Mesh Network Based on Hybrid Genetic Algorithm: The algorithm for WMS (Wireless Mesh Networks) developed by Hua JIANG, Liping ZHENG, Yanxiu LIU, and Min ZHANG in 2010 at the College of Computer Science,Liaocheng University [18] , is based on GA and ACO. The authors took some features from this heuristic in order to get fast convergence and to avoid local optimal. Genetic Algorithms were implement for solving QoS routing problems but they lack to maintain feedback information, hence they can fall in local optimal and converge slowly. In the other hand, ACO could maintain feedback information based on pheromones then get optimal solutions and converge faster, but this heuristic lack of the initial pheromones at the start time, therefore the time at the begin is longer. In order to overcome this problem the initial pheromones information is generated based on GA solutions, the time between the GA heuristic pheromones generation and ACO heuristic is determined by different number of GA iterations.

The QoS parameters for the proposed algorithms are:

- Delay
- Bandwidth

- Packet loss ratio
- Cost (hop count)

The main object of the algorithm is to minimize the cost of a path that satisfy the QoS constrains. Then P (Path) needs the following constrains:

- Bandwidth must be greater than the minimum value allowed.
- Delay must be less than the maximum value allowed.
- Packet loss ratio must be less than the maximum value allowed.
- Cost must be the minimum.

The genetic operator for the GA phase works as follows:

- Initial Population: The population is generated based on random selection of multiple path from the network.
- Selection Operator: Selection is performed using the optimum individual and uses the roulette wheel selection method.
- Crossover operator: The crossover is performed selecting two paths, then it finds common nodes between them, next it randomly selects 2 nodes a,b from the common nodes, finally exchange the part between nodes a,b and delete duplicate parts in the path. This process is realized in such manner to avoid illegal paths.
- Mutation operator: The algorithms select one randomly path and take a node from this path. The nodes before the selected node (k) has to be added to the offspring, then the k node is to be the start node the the destination node the end point, next randomly search a path can not contain nodes before k node. It is added to the offspring. The proposed method avoid illegal paths.
- Termination Condition: The termination judgment is performed based on an evolutionary rate R which is calculated every genetic iteration, if R is satisfied for 3 times successfully where $0 < R < R_{min}$ (R is a threshold value), GA heuristic is terminated and ACO heuristic starts.

  In order to get the optimum individual a fitness function is proposed based on QoS parameters and the cost.

The ACO phase takes information from the output of GA phase to add it to initial pheromones, this GA process takes the 10% of the paths with the fittest values which are defined as genetic optimize set. Once the initial pheromones are generated the path selection process is started, this is accomplish by the ant visit to different nodes, there is a probability to select a node j from an ant in node i which let to avoid local optimization. Final steps are partial update which performs pheromone update when an ant complete a hop from node i to node j and global update which is performed when all the ants complete one cycle. There is an evaporation function to explore new paths and avoid local optimal.

The simulation shows a comparison between pure GA Algorithm, ACO Algorithm and Hybrid Algorithm (HGA), the result was a lower delay value for HGA and a higher value for GA, the convergence time comparison is presented, the result was a lower time for HGA and higher value for GA.

C. Mobile Ah Hoc Networks

Researches in Ad Hoc networks started actively in 1995 in a conference of the IETF (Internet Engineering Task Force), in 1996 this evolved to Mobile Ad Hoc Networks (MANETs), finally in the charter of MANET WG (Working Group) of the IETF in 1997. MANETs base on mobiles node interconnected each other trough wireless links without infrastructure and centralized administration, the nodes acts as router to pass the packets. Mobile Ad Hoc networks allow implementation in different applications environments where wired networks could not be possible: catastrophes (floods, earthquakes...) or battlefield communication [19].

D. Bioinspired Models and Routing with QoS as a General use

In previous two algorithms examples, bioinspired models were used to find optimal solutions for Routing Problem with multi-constrain QoS parameters. General QoS parameters presented in bioinspired-routing problems are:

- Bandwidth.
- Delay.
- Lost packet rate.
- Cost.

One of the most used Bioinspired model to solve QoS routing problems is GA, due to its ability to solve optimization problems which deals with cost optimization problem for routing, it is a perfect heuristic for applying in this field, but it has some lacks which many researches have tried to overcome using some hybrid heuristics. In explained algorithms, GA has been used in conjunction with Game theory and ACO to improve Genetic Operators and to avoid local optimal, different approaches are presented with other models used for routing problem as a base for an analysis of hybrid heuristic implementations:

The algorithm proposed by Chen Xi-hong, Liu Shao-wei, Guan Jiao, Liu Qiang in [20], performs an hybrid between ACO an PSO, this algorithms uses ant sub-colonies for every destinations nodes, each of the ant sub-colonies can generate a multicast tree, in the same way each sub-colonies represent particles to perform PSO, it regulates the solutions from ACO depend on the current position of the particle, the updating task is based on position updating and velocity updating. With PSO the solutions from ACO could be compared with current solutions, best solution of particles and all particles respectively, it avoids prematurity of the algorithm. LI Taoshen, XIONG Qin GE Zhihui [21] refers another approach using GA and PSO, the algorithms initially perform PSO to solve the anycast routing problem, if the algorithm fall in a local optimum value, genetic operators are applied (crossover and mutation), it helps to scape from local value, next many iterations are performed until the conditions are satisfied, when the algorithms converge (the optimal value does not change during many iterations) the optimal value is reached. P.Deepalakshmi and Dr.S.Radhakrishnan [22] use an ACO algorithm applied to Manets networks with QoS constrains, the ants have additional features than pheromones, such as number of visited nodes, delay from destination to source and bandwidth per link traveled. Once the ants come back to the source node with the gathered information, it is used to decide if the path is suitable for the QoS constrains. The algorithms performs two phases: Route discovering and Route Maintenance. The last phase is applied because of the constantly change of the topology in Manets networks. Algorithm proposed by Mahsa Armaghan, Abolfazl T. Haghighat, Mohammadreza Armaghan [23] bases on Tabu search to solve Routing with QoS, the algorithm starts with a pre-processing phases intended to eliminate nodes an links do not satisfy constrains, a small network is obtained as the result of this phase. It combines the Dijkstra algorithm to find an initial solution. The performance applied to Tabu search is an Elite Candidate List, it allows to keep some predefined moves to get better solutions and to avoid function deterioration and cycling.

IV. Analysis of approaches in Hybrid Bioinspired Models

In previous sections bioinspired models were used to solve routing with multiple QoS parameters, different heuristics present robustness and weakness in their application, in order to overcome the weakness some approaches such as hybrid bioinspired models has appeared, it means combination of different heuristics which let to solve weakness problems, the next section presents an analysis about hybrid algorithms that overcomes heuristics lacks:

A. Genetic Algorithm - Ant Colony Optimization. GA-ACO

Pure GA routing protocols implementation showed to be slow in convergence and fall in local optimum values as described in [18], to speed up the convergence time and to avoid the local optimum values GA-ACO is proposed, due to the ACO's ability to maintain a track of the visited nodes, ACO is applied in GA to improve its abilities and allow the new algorithm to keep a feedback which will be reflected in local optimum avoidance, consequently better fitness values always could be found and less iterations (generations) are necessary to the algorithm's convergence. in the other hand, pure ACO algorithm is a good approach for routing protocols due to its feedback ability, but it is hard to discover new paths if high pheromone level is spread in one path, as a result pheromone evaporation is implemented. Mutation operator from GA improves ACO, it allows to jump in the problem space to scape from local fitness values. the graphs presented in [18] indicate a performance comparison between pure GA, pure ACO and HGA (GA-ACO), as expected better performance is achieved by HGA, getting lower average delays values, the next better is ACO and finally GA, the simulation displays the time wasted by the algorithm with different number of nodes, the time execution difference is higher when network has greater number of nodes, in simulation with 60 nodes the time is considerable different between each of the three compared algorithms, as discussed before GA takes more time than ACO and finally HGA get best time than others.

B. Ant Colony Optimization - Particle Swarm Optimization. ACO-PSO

The ACO-PSO hybrid based on the interaction of Ant Colony Optimization and Particle Swarm Optimization uses sub-colonies, every sub-colonies is intended to search a multi-cast tree, in order to know the quality of the multi-cast tree found by the ACO algorithm, PSO is used, in this way every sub-colony represent a particle, once ACO phase is finished multiple multi-cast tree are consider (particles for PSO), based on optimization heuristics with PSO (velocity update and position update) the best multi-cast tree is elected. The simulation presented for this hybrid algorithm indicate a convergence in relative few iterations (about seven) and find low Jitter, delay and cost values. Comparing ACO-PSO with GA and ACO, the run time is performed, GA is the slowest algorithm, later ACO and finally ACO-PSO. The hybrid algorithm has good performance because it can result in multiple multi-cast trees based on ACO to evaluate them with PSO in order to find better solutions [20].

C. Particle Swarm Optimization - Genetic Algorithm. PSO-GA

The hybrid algorithm uses Genetic Algorithm to perform PSO, the approach aims to integrate genetic operator (based on GA) to PSO. PSO does not use genetic operators such as GA does, consequently PSO could fall in local optimum however GA is slow to converge, as a result PSO-GA is presented. The operators implemented in PSO are: crossover and mutation, when PSO fall in local optimum GA operators are used, the operators help to scape from local optimum and overcome the shortcoming. Simulation presented in [21] indicate a comparison between GA, PSO and GA-PSO based on the fitness value, the result shows better fitness values for hybrid algorithm and convergence about 25 iterations, in the PSO case it get lower fitness values and a rapid convergence, it could be the result from local optimal, in the other hand GA get better fitness values than PSO but the convergence time is longer, it is due to the genetic operators but as showed GA has lower convergence time.

V. Conclusion

Many heuristics has been applied to routing with multiple QoS constrains, the paper showed multiple hybrid models in order to improve the performance and to overcome heuristics lacks, the hybrid models were applied to solve the QoS problems and in comparison, the simulations display better performance using hybrid algorithm than with pure algorithm, because hybrid is performed to take features from heuristics and apply them to other heuristics which lacks these, it allows to speed up the convergence time, the quality of the results, the number of iterations. Based on simulation results some pure heuristics shows better result when applying to multiple QoS routing problems, an example is ACO, where the features of path selection and ant's pheromone model is designed exactly for these type of routing problems, therefore in many simulations ACO showed better performance that GA. Another approach is based on GA and PSO comparison, where the simulations showed fastest convergence and simple implementation of PSO because of its simple design, in the same way due to PSO does not have genetic operator the probability to fall in local optimal values is higher, hence GA present better fitness values but longer convergence time, the same approach is applied to GA-ACO, as referred GA compared with ACO manifested low performance due to the lack of GA to maintenance a feedback of the results which is an ability of ACO. The solution to overcome the earlier discussed shortcomings is the implementation of Hybrid heuristics which emphasis the last part of the paper.

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On Mining DOM Trees to build Information Extractors

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***Abstract*—The Web is the largest information repository. The information it contains is usually available in human-friendly formats. Companies are interested in using this information. The problem is that they need it in structured formats so that they can use it in automated business processes. In the literature, there are many proposals to infer information extractors. They build on machine learning techniques that attempt to infer a pattern in the HTML or XPath sources. To the best of our knowledge, no-one has ever explored using datamining techniques on DOM trees. In this paper, we report on a methodology that builds on datamining CSS features and a few other DOM features. Our results prove that this methodology is promising.**

***Index Terms*—Datamining Techniques, Information Extractor, Machine Learning**

I. Introduction

The web is an enormous and increasing source of information. It has been defined by some authors as the biggest store of knowledge of the humanity and the reason is that the information is usually available in a friendly format for the users that allows it to interact easily with the Web.

The problem is when the information is necessary to use an automated business process.

Currently, the two most accepted solutions are the use of Semantic Web technologies or the use of Wrappers. The Semantic Web is an extension of the current Web where all the resources have associated annotations that represent the information in structured way [2]. The Semantic Web is not a new Web but is a complement of the current web that will help the programmers make use of the information contained in the Web more easily. The Semantic Web is advancing, but, unfortunately, it is not widely deployed nowadays. Wrappers can be an effective short-term solution. A web wrapper aims at offering an API to abstract programmers from simulating human behaviour interacting with the Web. One of the main components of a web wrapper is the information extractor. Information extractors usually are general purpose algorithms that rely on a number of extraction rules. These rules can be hand-crafted but there are a number of proposals in the literature that report on learners to infer them.

To the best of our knowledge, all of the proposals in the literature build on ad-hoc machine learning techniques; for that, the use of standard datamining techniques to extract information from web pages is novel.

In [5] we reported on the results of a preliminary experiment in this field. We labeled DOM trees with labels to indicate which nodes contained relevant information. The pages must be homogeneous, i.e., they all must be about books, films, conferences or DVDs. The reason is that our technique builds on features of their DOM tree, so they all must be generated by the same server-side template; then we used the standard Naive Bayes, C4.5, IBk, ZeroR, SVM, and VFI datamining techniques to infer a classifier that performed quite well in practice. This motivated us to keep working to develop a complete methodology to create information extractors. In our experiment, we used the DOM trees features, chiefly CSS features. Initially we use the features as are, i.e., they all had to be nominal features, but we have learned that the results can be improved by normalising some of them. For instance, feature color may have values red, #12ab, or rgb(0, 255, 180), which makes it difficult to define the domain of this feature; in our methodology, we normalise this feature by transforming it into three independent numeric variables called R, G, and B that correspond to the red, green, and blue component of every colour. In this paper, we also compare our methodology with other state-of-the-art proposals in the literature. Our results prove that our methodology achieves a precision and recall that is comparable to other proposals, but the time required is significantly better.

The rest of the paper is organised as follows: In Section §II we review briefly related work, in Section §III we explain our methodology and then in Section §IV we evaluated our methodology and shows our results. Finally, Section §V presents the conclusions and our future work regarding this methodology.

II. Related Work

Literature contains many techniques to infer extraction rules from semi structured and free text web pages. Proposals such as S-CREAM [6], KIM [10], Armadillo [3] and MnM [18] works over free text but in our work we focus on information extraction from semi structured web pages. These techniques can be classified into two main groups: a heuristic based group and a rule based group.

Heuristic based group contains those proposals to extract information that are based on predefined heuristics which can not be modified by users. Although these heuristics can be seen as rules, the difference between this group and the rule based group is that they are totally independent of the web page over which they are applied. These heuristics are not inferred by any automatic technique neither learn extraction rules. Below we describe some of these techniques briefly:

- Stavies [15] uses signal theory and clustering techniques to localise data region and extract contained records from this region.
- Alvarez et. al [1] localises data region by comparing DOM trees and then works on separating this region into records and extracted records into attributes using string alignment and other techniques.
- ViPER [16] divides input web page into visual blocks to identify data regions and then breaks them up into data records. String alignment is then used to separate attributes inside extracted data records.

The rule based group contains those information extractors that are configurable by means of rules. Beyond handcrafted information extraction rules, there are many proposals in the literature to learn them in a supervised and in an unsupervised manner. Supervised techniques require user to label a set of web pages by selecting and assigning a type for the relevant information in a set of web pages used then in the learning process. Below we describe few supervised information extraction learning proposals:

- Softmealy [8] constructs a transducer where states identify the data to extract and transitions contain the learned regular expressions to pass from one attribute to another. Transition condition are learned using a tokenisation hierarchy and an alignment technique.
- WIEN [11] is a tool that provides an implementation to six algorithms that build on the general idea of learning common prefixes and suffixes. In the rest of the paper, we focus on the NLR algorithm, which allows to extract simple fields.
- Stalker [14] uses a tree like structure called embedded catalog where each node identify and extract one type of attributes. Rules are learned by using a coverage algorithm which tries to use previous and post tokens for each annotation type, covering the maximum number of annotations and minimum number of un annotated text.
- Thresher [7] searches for sub trees inside the DOM tree similar to that one annotated by user. The annotated one is then generalised to cover all data records and the result is a sub tree which contains fixed and variant nodes.
- DEByE [12] rule learning algorithm uses these annotations to learn both: rules to extract a complete object and rules to extract attributes inside this object. Rules are inferred by searching for the largest common prefix and suffix tokens for the attributes to extract.

This group also contains unsupervised techniques which do not require the user to provide labels. Some unsupervised learning algorithms are described here:

- Fivatech [9] uses clustering technique to label HTML nodes and the alignment and pattern detection techniques to learn a grammar that defines the web page's structure.
- RoadRunner [4] compares input web pages by keeping similar string fragmentes and replacing unfixed ones by wildcards.
- DEPTA [22] uses MDR [13] to detect and separate data records distributed horizontally (in rows) or vertically (in columns) and then applies a partial tree alignment algorithm to separate attributes inside detected records.
- DeLa [20] localises data region by using [19] and then detects repeating patterns by tokenising content and using suffix tree.

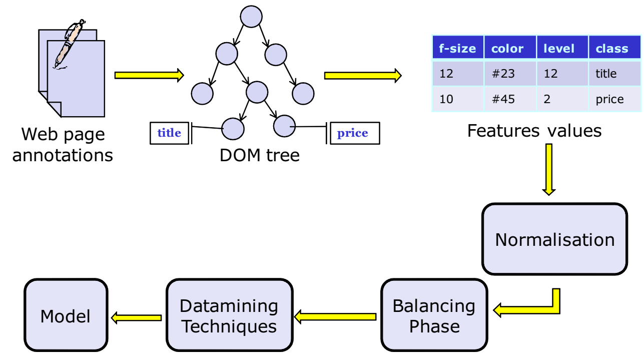

Figure 1. Datamining CSS features to learn a classification model

III. METHODOLOGY

Our methodology is as follows, Figure §1:

1) Gather a collection of web pages from the site we wish to extract information.
2) Labeled the DOM trees using user-defined classes to mark the nodes that contain the relevant information, or the pre-defined class NA to mark the remaining nodes.
3) Extract the CSS features of every node, plus level in the DOM tree and length of the corresponding text. We have observed that these additional features may help in some cases.
4) Normalise the CSS features according to the categories defined in table §I.
5) Use standard datamining techniques in the literature to infer classifiers, cross-validate them, and compare the results to decide which one is the most appropriate.

The normalisation process depends on the features being analysed. We have grouped them into five categories, cf. Table §I:

Category 1 This category includes all of the features whose domain is nominal and bounded. For instance, feature font-weight ranges over domain {normal, bold, 100, 200, ..., 900}; note that no intermediate values area allowed, e.g., 110 or 850. The features in this category are not normalised.

Category 2 Here we include features whose domain is numeric, but has a few pre-defined nominal values. For instance, the range of feature vertical-align includes real numbers, and the following nominal values: {baseline, sub,

Category	CSS features
Category 1	font-style, font-variant, font-weight, background-repeat, background-attachment, text-decoration, text-transform, text-align, border-top-style, border-right-style, border-bottom-style, border-left-style, float, clear, display, white-space, background-image, list-style-image, list-style-type, list-style-position
Category 2	vertical-align, background-position, word-spacing, letter-spacing, margin-top, margin-right, margin-bottom, margin-left, padding-top, padding-right, padding-bottom, padding-left, border-top-width, border-right-width, border-bottom-width, border-left-width, width, height, text-indent, line-height
Category 3	font-size
Category 4	color, background-color, border-top-color, border-right-color, border-bottom-color, border-left-color
Category 5	font-family

Table I
CSS NORMALISATION

super, top, text-top, middle, bottom, text-bottom, auto}. These features are normalised so that nominal values are transformed into extreme real values that are not likely at all. For instance, baseline is normalised as -35000, sub is normalised as -40000, and so on.

Category 3 The features in this category may have values that are actually macro-definitions. For instance, feature font-size ranges over {xx-large, x-large, large, middle, small, x-small, xx-small} which are defined as 1.6 S, 1.4 S, 1.2 S, 1.0 S, 0.8 S, 0.4 S, and 0.2 S, where S denotes the normal font size, which is browser dependent. The normalisation consists of unrolling the macro definition.

Category 4 This category includes colour-related features. In CSS a colour can be expressed as a symbolic name, as a hexadecimal number, or in RGB notation, e.g., blue, #12ab00, rgb(12, 0, 128). These features are transformed into three independent numeric variables, each of which accounts for an independent RGB component.

Category 5 The only feature in this category is font-family. The domain of this feature is an unbounded set of sets of nominal values that includes Arial, Times New Roman, {Verdana, Helvética}, {Courier New, Tahoma} and other well-known fonts or combinations of fonts. We have identified a collection of 394 popular font names, and we have transformed this feature into a collection of 394 boolean variables: these variables are assigned true if the corresponding font family is in the value of the feature, and false otherwise. To account for other uncommon fonts, we have added a variable called other-font-family.

IV. EVALUATION

To evaluate our methodology, we have conducted several experiments. We use some datasets previously annotated with the relevant information and each dataset contains thirty web pages. The following techniques were used: Naive Bayes, C4.5, IBk, ZeroR, DecisionTable, SVM [21]. We compared the results to SoftMealy, and NLR. For each technique, we measured precision (P), recall (R) and time to build model (TBM). Then, we compared the results obtained from each dataset and we made a ranking.

The analysis of our experiments rely on statistical inference. The idea is to define two hypothesis, the null hypothesis, which always considers that there are not significant differences between values and the alternative hypothesis that considers that there are differences. The goal is to reject the null hypothesis and show that there are differences. Accepting or rejecting the null depends on the comparison of the p-value calculated by the tests and significance level (α). In our study we used the standard value ($\alpha = 0.05$). If p-value is less than α then the null hypothesis is rejected.

First, we used Kolmorogov-Smirnov's test with the Lilliefors correction and the Shapiro-Wilk's test to check normality. The results show that none of the measures follow a normal distribution, cf. Table §II. The implication is that we need to resort to non-parametric tests. We used the Kruskal-Wallis test to determine if P, R and TMB means can be considered equal or not. These results proved that all techniques are statistically different and therefore it is possible make a ranking, since all of the p-values were 0.00. The next step is to use Friedman's algorithm to calculate the scores and Bergmann's algorithm to rank them. The results are shown in Table §III. We can conclude that datamining techniques we have analysed are better than the SoftMealy and the NLR techniques, which are well-known in the literature. With regard to precision, the datamining techniques are as good as the other techniques, but regarding recall and time to build models, the datamining techniques are better.

V. CONCLUSIONS

In this paper, we have presented a methodology to extract information from web pages that relies on datamining CSS features of DOM trees. To the best of our knowledge, this is the first methodology that explores this field, and the results are promising since precision and recall are comparable but the time to build models improves significantly.

In future, we plan on a) extending our methodology to extract data records, instead of just data items; exploring other techniques in the literature to balance the NA class [17]; c) exploring if using CSS features of parent or sibling nodes can help improve precision and recall, without a significant impact on the time to build models.

Technique	Kolmogorov-Smirnov (P-Value)			Shapiro-Wilk (P-Value)		
	Precision	Recall	Time	Precision	Recall	Time
Naive Bayes	0.20	0.90	0.20	0.52	0.24	0.09
C4.5	0.20	0.20	0.20	0.37	0.31	0.97
IBk	0.20	0.20	0.25	0.11	0.10	0.11
ZeroR	0.00	0.00	0.02	0.00	0.00	0.23
DecisionTable	0.20	0.20	0.20	0.16	0.61	0.71
SVM	0.03	0.62	0.00	0.34	0.21	0.03
NLR	0.14	0.20	0.50	0.64	0.49	0.04
SoftMealy	0.20	0.20	0.00	0.30	0.09	0.00

Table II
KOLMOGOROV-SMIRNOV'S AND SHAPIRO-WILK'S TESTS

Ranking	**Precision**	**Recall**	**Time to build model**
1	SoftMealy, C4.5, IBk, DecisionTable, SVM, NLR	C4.5, IBk, DecisionTable, SVM, Naive Bayes	IBk, Naive Bayes, C4.5, ZeroR
2	Naive Bayes	SoftMealy, ZeroR	SVM
3	ZeroR	NLR	DecisionTable, NLR, SoftMealy

Table III
BERGMANN'S RANKING

ACKNOWLEDGMENT

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E-reading strategy model to read E-school book in Libya

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Abstract:- defining the stages for the reader to follow when reading e-resources is only one of several factors which can provide a significant understanding of the actual reading behaviour and cognitive process of reading. This article aims to compare reading processes that students follow when reading school books in two different media (paper and electronic). A sample of 80 students, studying in Libyan primary schools, and aged 9 to 12, were selected to investigate how students use and interact with both print and digital school–books, how they identify the e-reading process, and to define what students like and dislike in both versions. The results showed differences in the reading process between paper and electronic books read.

Keywords- reading process, reading online, school book.

I. INTRODUCTION

Reading e-texts brings several challenges to readers, such as the difficulty of reading on the screen in the case of long lines, difficulty of browsing and moving from page to page, and inability of the search tools to satisfy the needs and requirements of the reader. These aspects draw attention from many research areas, such as information science, computing science and human science. There are three categories to digital reading research:

I. Research has focused on the usability of e-text, e.g. comparing reading on-screen to paper reading [1, 2]; measuring the legibility and comprehension of text, and examining user behaviours in digital environments.
II. Research presents the 'new approach of technologies-supported reading on screen' which concentrates on new software and hardware, hypertext, and interface design[3-5] [4].
III. Finally, research focused on the phenomenology of reading, like studying human interaction with e-resources and reading process in both linear text and hypertext[6, 7].

Generally, reading is a complex activity. It requires different skills according to the purpose of reading. Furthermore, the reading scenario changes on the basis of the type of e-materials, starting from short text reading (i.e. e-mail) to long text reading (i.e. e-books). For instance, scholarly articles are not similar to books. In the first case, readers usually skip the abstract of an article, skim the introduction, read the research problem, and subsequently read various paragraphs on the theoretical perspective.

Notably, using hypertext raises several issues relating to their impacts and ability to improve the accessibility to information on e-environments and reading processes. Moreover, the impact of hypertext on academic reading remains unclear, although, there are several researchers who argue that reading in an e-environment becomes more of an interactive process [8], which can engage a reader in having a further authorial role and ultimately induces decision-making processes [9]. However, the design of hypertext in e-learning materials requires a good understanding of human cognition features, such as previous knowledge, memory structure, and the ability to impress in terms of the organisation of information.

The aim of this study is to observe how students read school books in e-format, to accordingly investigate reading behaviours adopted by students, and to examine how students' reading behaviours are changing in both versions. However, understanding these changes in the reading process would support us in designing an effective computer-based learning environment.
In this work, we will attempt to build an e-reading strategy based on users' cognitive and behavioural processes. The remainder of this paper is organised as follows. First, we illustrate related works in reading e-resources behaviours. Second, we define the methodology of research and dataset for analysis. Third, we provide two reading strategy models. Finally, we summarise findings and set forth our generalisations and recommendations for the reading process.

II. RELATED WORK

With the increase in the number of students who read e-text for pleasure or learning, there are still deficiencies in studies that seek to understand how the digital text is read as previous studies have shown, i.e. most research on e-reading have just focused on comparing issues without having a clear idea of how readers deal with the e-text, or, whether this technology has affected the way that the reader reads the e-text.

Thus, research in the field of usability are concentrated on addressing questions such as why, what and how do people read a document to use it with any research interest in reading generally and e-reading specifically. Also, research is focusing on cognitive and behavioural aspects of the reader by asking these questions and bringing up issues related to context which appears in certain factors that affect the presentation of the medium.

For instance, Dillon [1] built two models of reading process, one for reading academic journal and the other for manual as be seen in *Fig. 1* and 2. This researcher noted that there is a distinction between text types in the characterization of usage they suggest.

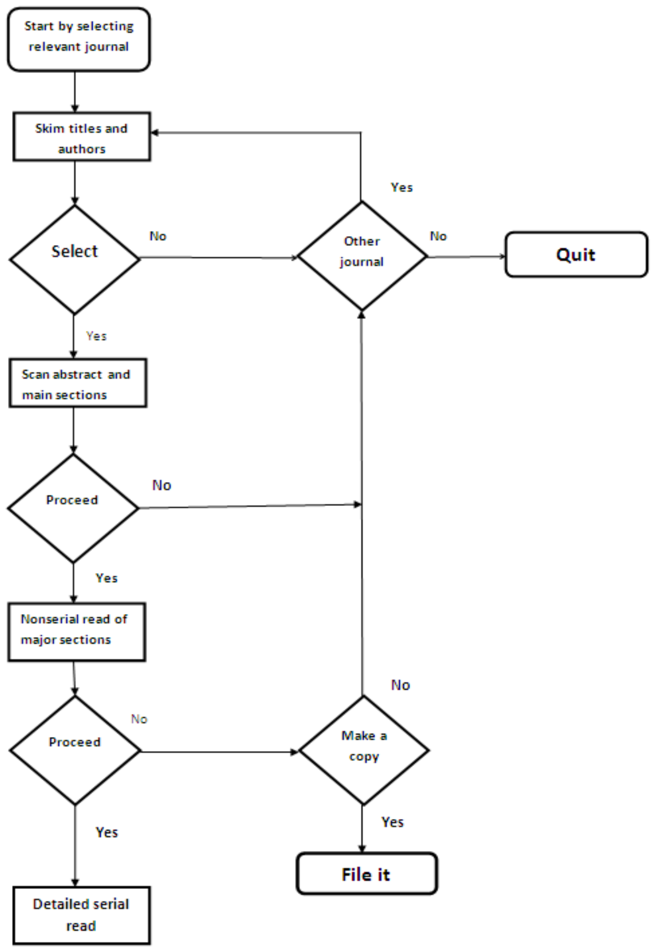

Figure 1. Generic model of journal usage[1].

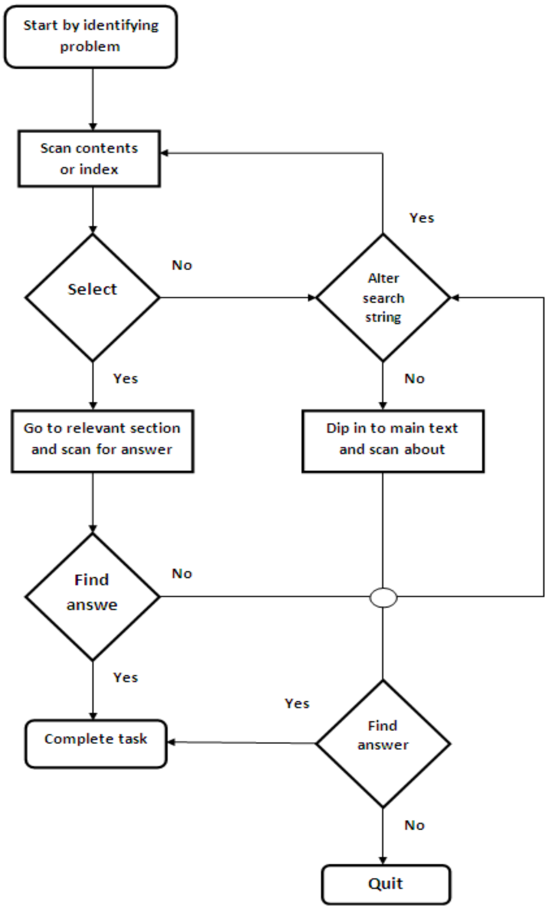

Figure 2. Generic model of manual usage[1].

Moreover, Terras [2] has suggested a model in terms of how experts read an ancient text by understanding a complex process in the humanities. Based on content analysis, focused interviews and thinking aloud protocols, the model was built. In addition, the study reported that the three experts used different methods to examine the document. They also spent a long time checking the text and the words in different orders. In addition, they dealt with visual features before building up knowledge about the document. Moreover, the reading process is un-liner and is based on the interaction of different facets of expert knowledge as can be seen in Figure (3). These findings help to implement a computer system that can work in several approaches

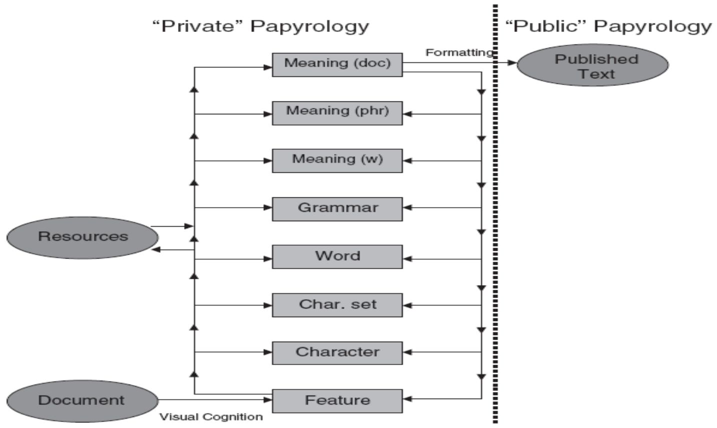

Figure 3: Model explains how experts read ancient text[2].

Alternatively, the reading process differs according to the purpose of reading and the type of material. For instance, Kol and Schcolink [10] argued that understanding reading strategy and teaching students how to deal with e-text could help students to read effectively from screen.

At the early stage, the reader tends to read documents from beginning to end. But with the growing in the amount of information acquired, the reader tends to skim, scan, and browse the document to search for relevant information. In the same perspective, Harrison [11] defined several reading behaviours, which can be classified into two types: comprehensive reading such as reading to learn, critique or edit; the other is skim reading such as reading to support listing, cross-reference or to answer questions.

III. METHODOLOGY

In order to follow-up and investigate in-depth users' cognitive behaviours with e-books, a sample of 80 participants—all of whom considered themselves confident with e-books—were selected. A follow-up method was selected to collect data in order to:

- ➢ Investigate how students use and interact with both print and digital school–books;
- ➢ Measure participants' achievements in both versions;
- ➢ Identify the e-reading process; and
- ➢ Define what students like and dislike in both versions.

The target population comprised students aged between 9 and 13 who were attending Libyan public primary schools. The total sample comprised 80 respondents, distributed as follows: 20 participants from Level 4; 30 participates from Level 5; and 30 participants from Level 6. Participants carried out three different observations, which required dividing the participants into eight groups, each of which had 10 participants. Four groups used e-books (available at: http://skooollibya.com/), whilst the other four groups used paper versions. During the first observation, participants were asked to prepare a lesson for discussion in class. They were given an open time to complete the task. The next day, teachers discussed the lesson with students and provided explanations, which took 45 minutes. Finally, the students were asked to search the text, answer the questions, and take a small quiz. Figure (4) shows the steps comprising the follow-up study.

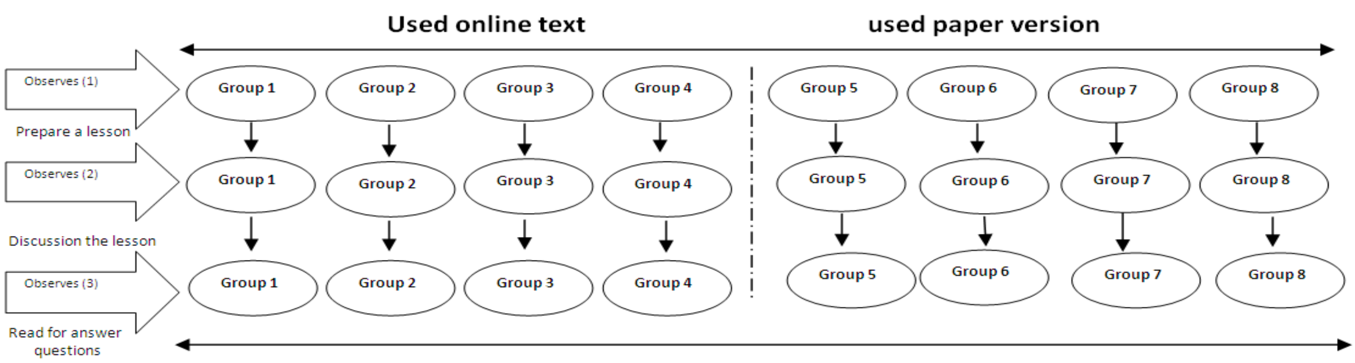

Figure 4. The follow-up survey stages.

The talk-aloud technique was used in order to identify users' cognitive and behavioural processes, and to collect quantitative data, which could not be obtained via any other method. It was also used to obtain more in-depth details from participants, such as describing their actions and reactions with the book interface.

A. Participants' Selection

The target population comprised students aged between 9 and 13, attending Libyan public primary schools. The total size of the participant sample was 80, distributed as follows:

- 36 males, thereby representing 45%; and 44 females, thereby representing 55%;
- The age of the participants was between 9 and 13;
- The sample was distributed across three studying stages: level 4 comprised 16 students; level 5 comprised 19 students; and level 6 comprised 35 students. These distributions are shown in Table 1.

Table 1.The sample by gender and education levels.

Education levels	Male	Female	Total
Level 4	13	7	20
Level 5	15	10	25
Level 6	15	17	35

B. Material Selection

The survey focused on primary school books within Libya, examined in order to recognize the main characteristics of such books and the overall structure of the text. This was done in order to obtain a clear picture from the analysis of the content of the books and to accordingly highlight the aim of the education system. Table 2 shows the distribution of school books at each level.

Table 2. Distribution of subjects studied at each level.

School levels	Models										
	math	Science	reading	grammar	History	English	Geographic	Religion	Computer		
Level 1	/	/	/					/			4
Level 2	/	/	/					/			4
Level 3	/	/	/					/			4
Level 4	/	/	/	/	/	/	/	/	/	/	10
Level 5	/	/	/	/	/	/	/	/	/	/	10
Level 6	/	/	/	/	/	/	/	/	/	/	10
Total											42

School books differ from other types of resources in the way of presentation of information, the amount of information, and the structure of the content.

The author has examined a random sample from books in order to identify the main combination of school books, from which a document model is provided, as shown in Figure 5, which illustrates the document in a top-down method, complete with independent descriptions for the document elements. The document content is expressed as a set of parts. The document's contents are divided into many parts such as the text part, which is divided into many regions, such as the Table of Contents and Lessons. Each lesson component is made up of several sections, and each section breaks down into paragraphs, with each paragraph combining sentences, which are further broken down into words. In addition, each document then comprises a hierarchical structure of abstraction levels, with each level representing elements in the document. However, the analysis of the school books shows the following:

- The majority of the contents of books at the three first levels comprise images and limited exercises, which means we need to focus on colour dynamics and flexibility.
- The books of levels 4, 5 and 6 are completely different to those of other levels in many respects, such as the technique in which information is offered and organised, the structure, and the amount and type of information—all of which require the use of different formats, such as HTML and PDF

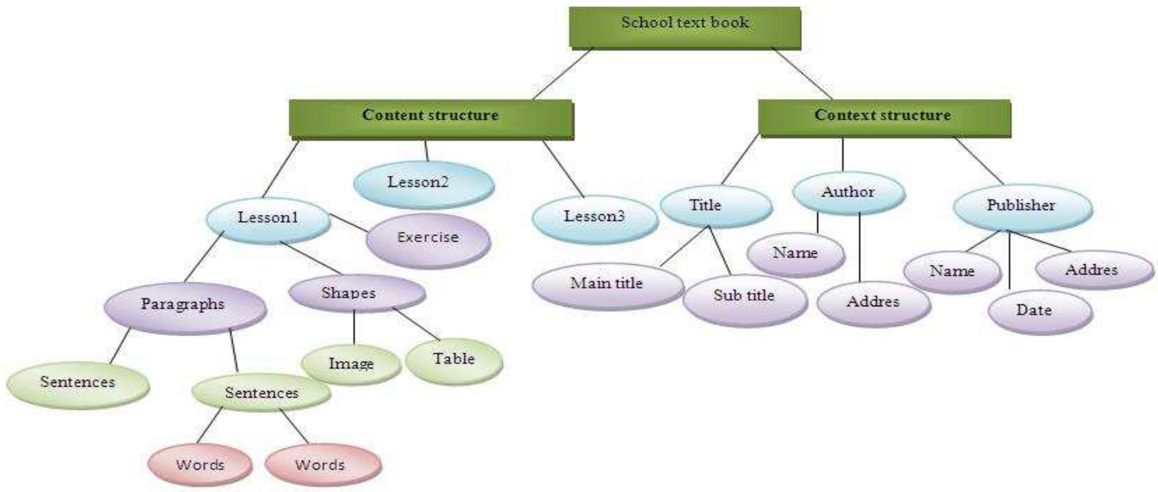

Figure 5.Structure of school text book.

IV. Dataset Analysis

A. *Using E-book and Paper Books in Primary Education*

The use of paper version of books is still the first option to most readers. For instance, Perry reports that 84% of participants preferred using a paper version. Alternatively, students in primary schools in Libya are also more confident when using paper school books, as can be seen in Table (3). Furthermore, 75% of participants have found using the e-version to be very difficult, whilst 88% of participants could not deal with the paper version easily. Navigation is another challenge which participants face in e-versions. In addition, 91% of respondents found that transition from one page to another is difficult, which in turn influences the communication between students and teachers. On the other hand, 67% of participants found browsing the lessons easier in the paper version. Moreover, identifying the location of information in both versions was not easy, but was more difficult in the case of the e-book (74%) compared to the paper book (45%).

Table 3. Participants' opinions about e-version of school book compared to paper version.

	e- version		Paper version	
	yes	No	yes	no
➢ Easy to used.	25%	75%	88%	12%
➢ Can use without any help.	44%	66%	93%	7%
➢ Easy to search.	9%	91%	67%	43%
➢ Easy to find answer.	12%	88%	67%	33%
➢ Sound help to learn	19%	81%	-	-
➢ Easy to identify the location of information.	26%	74%	55%	45%

Moreover, there are several factors contributing to students' performance when reading electronic versions of school books, some of which relate to technical aspects, such as cohesion of content, linking, navigation, screen layout, segmentation of data, interface design and location of data. On the other hand, there are personal or human aspects which affect reading on-screen, such as previous knowledge on the topic, age, and memory structure and stages.

However, paper books usually have a good structure and the same structure is used to design e-materials, which are not the most suitable method for several reasons. The printed method has a long history, and has improved over time to become more efficient for readers by ensuring understanding of many different aspects, such as reader behaviour, reading process, and types of reading.

In order to improve reading on-screen, student behaviours need to be studied at first in order to define the reading process. This requires clarification of the reading stages which students follow, and accordingly defining the reading strategy to be used.

In order to investigate the reading strategies used by students, two questions were asked: 'Do you read the lesson first, or, do you read the question first?' As seen in Table 4 which presents the reading strategy used by students, 65.8% of participants always read the lesson first and then answer the question, whereas 34.1% read the question first. Consequently, both strategies are used by students based on the purpose of reading. For instance, if the student is reading for an exam, s/he will read just the questions, whereas if reading for understanding, s/he would then read the lesson first. This confirms the correlation between the purpose of reading and the strategy that the reader will use to achieve his/her goal

Table 4. Strategy of reading used by students when reading school book.

Strategies	Read the lesson first						Read the question first					
levels	Always		usually		Sometime		Always		usually		Sometime	
	N	P	N	P	N	P	N	P	N	P	N	P
Level 4	92	18.2	22	4.3	20	3.9	16	3.1	32	6.3	44	8.7
Level 5	104	20.6	50	9.9	26	5.1	88	17.4	60	11.9	32	6.3
Level 6	136	26.9	36	7.1	18	3.5	68	13.4	48	9.5	64	12.6
total	332	65.8	108	21.4	64	12.6	172	34.1	140	27.7	140	27.7

B. Models of Reading Strategies

The survey reports that there are several possible scenarios when using a school book. These scenarios are built based upon the purpose of using the school book which is always used either at school or home. In each case, the purpose of use is different.

- **Use at school:** at school, the teacher directs students by telling them the number of the page, the lesson title, the number of questions, and so forth. Thus, all the stages in this case are controlled by the teacher.
- **Use at home:** at home, some students get support from their parents, while others do not. In both cases, students use the school book for two purposes: firstly, to memorise the lessons taught at school; or preparing for the next lesson. In the case of the latter, the teacher prefers comprehensive reading.

Notably, there are differences in the steps followed by readers when reading an e-text. Such differences are highlighted in Figure 6 which summarizes the school book reading stages used by students. Firstly, students start by opening the book and skimming the Table of Contents (TOC). Worthy of note is the fact that 54.7% of participants always use the TOC to access the lesson, whereas 11% access the lesson using the page numbering. Accordingly, students usually check the lessons by identifying the subtitle, how long it is, and the number of questions before starting to read. This technique is also used when students read the lesson for the first time. In addition, when students decide to read the lesson, there are two ways to view the text, as the survey reports: either by viewing the text or viewing the questions. In each case, students use different reading methods. Finally, during the reading, students always take notes and save it for later use.

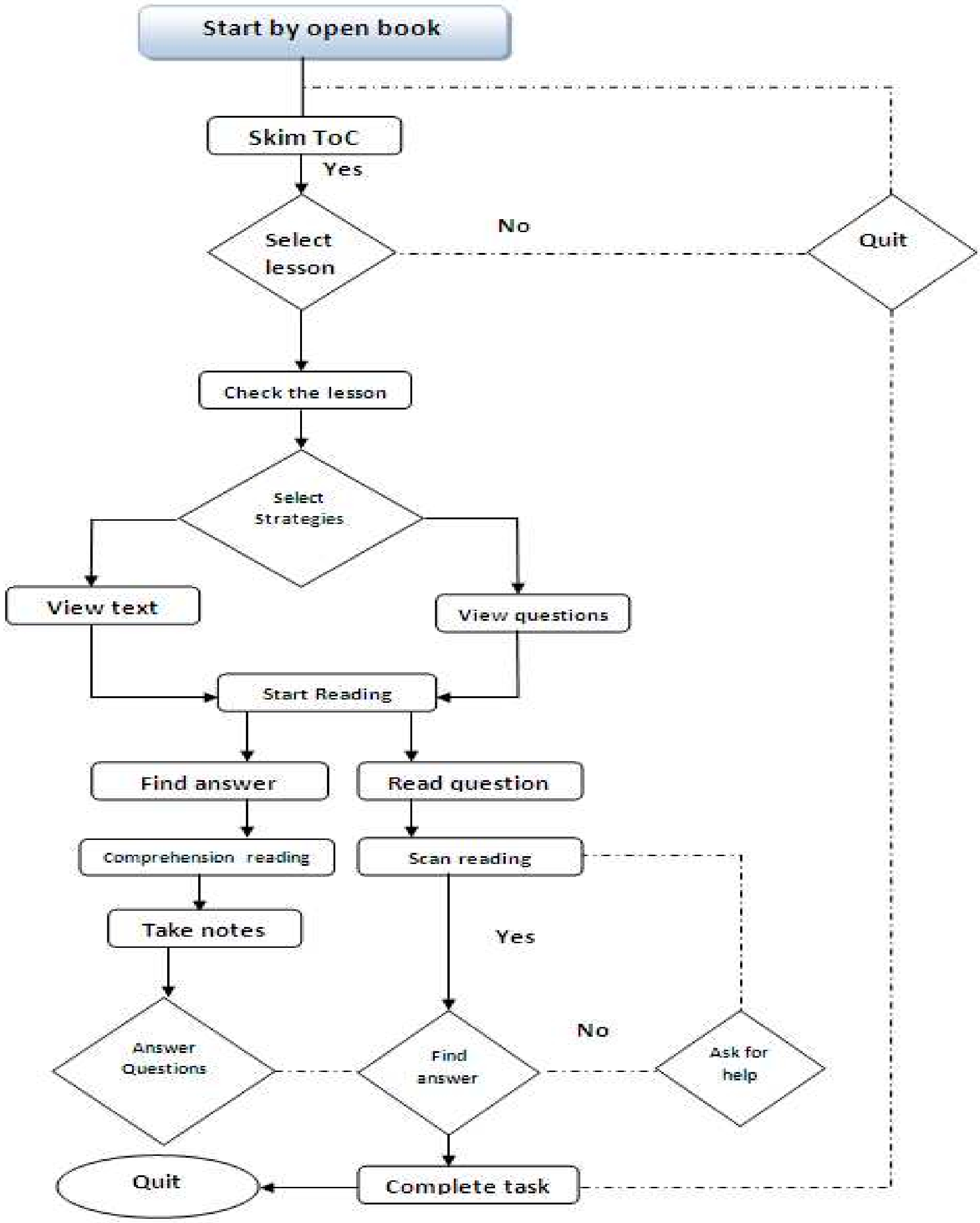

Figure 6. Reading strategies with paper school book.

On the other hand, based on qualitative feedback, the e-reading strategy for school books was built. Figure 7 shows a generic description of the e-school book reading strategy, which starts by viewing the home page of the system. Subsequently, the student can access the book by viewing the models and then selecting the level, or otherwise, by viewing the education levels and then selecting the model. The first action will be the opening of the book. At this point, the student has two options: to view the text or to view the questions. In the case of selecting the questions, the student will access the questions which link to the parts that include the answer rather than answering the questions, simply because the aim is to encourage students to read. Notably, if students select and view the text, s/he will then start with the introduction to the lesson before going through the lesson and learning the main bulk. Students can take notes and save them for later use.

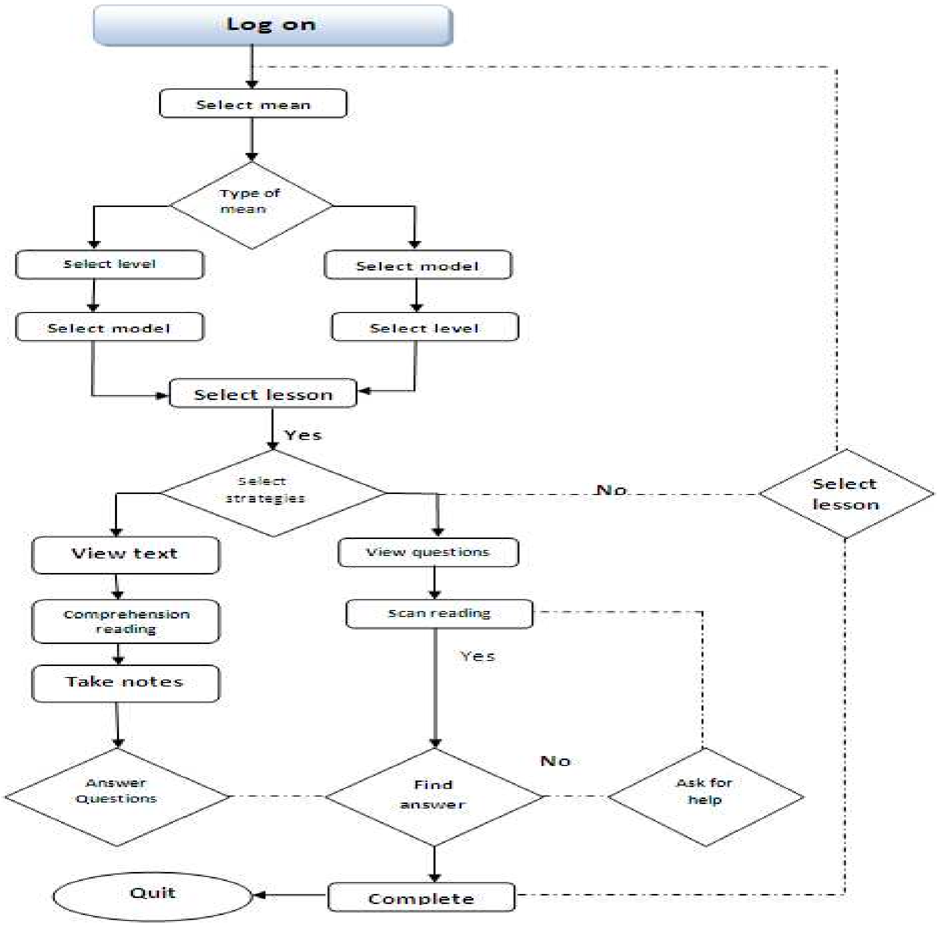

Figure 7. Reading strategy model of e-school book.

Finally, there are no changes required for user needs when students read the paper version or e-version of school books. However, the student's reading action changes in each version which makes a difference in the overall reading process. For example, the page number is the main tool used by students to access the book content in the paper version, while just 54% of participants use TOC. On the other hand, in the case of the e-version, students access the lesson by means of the lessons. Thus, defining the reading process can help define searching and reading stages. In the searching stage, students frequently search for a title or subtitle, or question and answer; this requires searching the content for concepts and titles. In the reading stage, students exploit two reading strategies: scanning and comprehension.

V. CONCLUSION

The general conclusion is that students preferred reading the paper version of the school book. This is due to a number of reasons. For example, dividing the text into a set of parts according to the paragraph for display on screen has led to a range of difficulties in browsing the text, making it difficult for the student to link and move between these parts. Moreover, the reading process changes according to the reading purpose. For instance, students usually use two strategies when reading school book: scanning and comprehension reading strategies. Each strategy requires specific tools and techniques, such as highlighting the sentence, taking notes, or using a finger when reading the text. In addition, the reading strategy is selected according to the aim of the reading. Therefore, determining the purpose of reading is necessary for designing readable e-learning material. However, these models will not only help define the interaction between users and e-books, but will also help designers understand user behaviour regarding e-books to establish the most appropriate functions when building the e-book interface.

VI. FURTHER WORK

In the next stage of the research, the following questions will be asked:

1. What are the most significant elements to consider when designing e-school books interface?
2. How should e-school books be designed to improve satisfaction, information recall, levels of confusion, and mental overload?
3. Measuring the factors that affect reading e- text such as font size, line length and color.

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Performance Enhancement of SIP- based VoIP Server

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Abstract - *The performance improvement on SIP-based VoIP server is given in this paper. The signaling server "SIP Proxy" and media streaming server "RTP-Relay" are employed. Firstly, the performance of VoIP server is analyzed and evaluated in detail. Then the sophistic method is employed to improve the performance of VoIP server.*

In the analysis results of "SIP Proxy", the CPU time for the registration and call-setup sessions need 3.241 ms and 7.985 ms respectively. Moreover, in the "RTP-Relay" system, when the packet size from 1 to 32 bytes, the maximum throughput of packet is about 82,000 and 16,000 per second for UDP and TCP respectively. The large the packet, the less throughput is founded.

In the improved method, three methods are employed to improve the performance. The capacity by 31 times is achieved. The capacity with 3,524 for each "RTP-Relay" is achieved. The improved method proposed in this paper is reasonable good.

Keywords: Voice over IP, SIP Proxy, RTP-Relay

1. INTRODUCTION

In the past, the VoIP application is growing like crazy. The VoIP protocols such as H.323 [1], SIP [2], IMS[3], and Skype [4] were designed for real applications. However, due to the simplicity and flexibility, many efforts [5,6,7] were paid on the SIP protocol and most of VoIP product followed the SIP standard. In the future trend, the SIP-based VoIP solution will become more popular in the broadband and wireless networks.

The capacity of SIP-based VoIP server is critical to the planning and operation of IP-telecom system. However, the capacity of SIP-based VoIP server is not well-studied or understood. Moreover, the improvement of SIP-based VoIP server is also not well-studied and understood. The clear analysis results on capacity will help to plan a suitable IP-telecom system. Moreover, the improved method on the VoIP server will reduce the cost of operation.

In this paper, the performance analyzed and evaluated on the SIP-based VoIP server is given in the next section. The improved method and results on the VoIP server will be proposed and described in the section III. The conclusions will be given at the last section.

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2. EVALUATION ON VOIP SERVER

The SIP-based VoIP application is employed to evaluate the performance of VoIP server. The VoIP server includes "SIP-Proxy" and "RTP-Relay". The "SIP Proxy" handles the signaling of call-setup. However, the "RTP-Relay" exchanges the media streaming.

In the SIP client, the regular registration process is needed to keep client always on-line. Otherwise, the call-in process will fail. The flow-chat of SIP-based registration process is given in Figure-1. The flow-chat of call-setup process is also given in the Figure-2. The CPU time for the registration and call-setup processes is evaluated in the Table-1. There are two types of registration process. The first, "Registration-I", includes the authentication process. The MD5 algorithm and password are employed to authenticate the user. If the "SIP UAC" doesn't change the IP address and repeat the registration process, the "Registration-II" is employed to bypass the authentication process. The call-setup process is also with two types. In the Table-1, the Pentium CPU with 2.4GHz is employed to evaluate the CPU time and capacity. The capacity of registration and call-setup processes is 309 and 125 times per second. It means that "SIP Proxy" can process 309 REGISTER requests or 125 INVITE requests within one second.

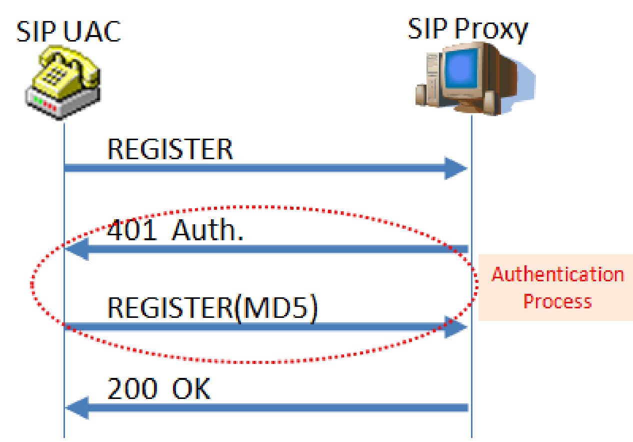

Figure-1. The flow-chart of SIP-based registration process.

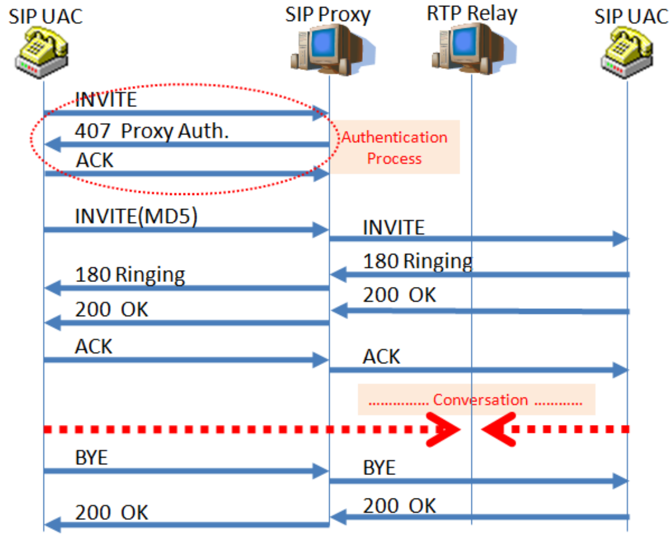

Figure-2. The flow-chart of SIP-based call-setup process.

Table-1. The CPU time and capacity for each session.
(not include Internet Tx/Rx time).

Session	Command Flow	CPU Time	Capacity: (session/ sec.)
Registration-I (with MD5)	REGISTER+401+ REGISTER+200	3.241 ms	309
Registration-II (without MD5)	REGISTER+200 OK	1.838 ms	544
Call-Setup-I (with MD5)	INVITE+407+ACK+ INVITE+180+200+ ACK+BYE+200	7.985 ms	125
Call-Setup-II (without MD5)	INVITE+180+200+ ACK+BYE+200 OK	5.254 ms	190

In the "RTP-Relay" system, the audio and video packet is received, exchanged and re-sent to each other. The maximum throughput (MT) of packet with different size is listed in the Table-2. The MT of packet is same when the packet size is small than 32. The MT will decrease when the packet size increase. Moreover, the MT of UDP packet is more TCP packet.

Table-2 The maxima throughput and CPU times with different packet size.

Packet Size (Bytes)	**Maxima Throughput (Packet No/Second)**		**Percentage of CPU Usage (%)**	
	UDP	**TCP**	**UDP**	**TCP**
1	82410	15104	61	66
2	82374	19532	59	48
4	82998	12358	60	52
8	83556	16656	59	50
16	82264	16440	57	56
32	66734	16918	11	38
64	44208	15918	10	36
128	35246	16758	10	32
256	23900	11218	15	30
512	17372	9760	7	28
1024	11318	8292	8	26

3. IMPROVEMENT ON VOIP SERVER

In this session, the improved method on the "SIP Proxy" and "RTP-Relay" is proposed and evaluated. In the "SIP Proxy" system, three methods are employed to improve the performance. These methods are described and listed as below:

1. In the registration process, the MD5-based authentication process can be bypass when the IP address and port number of SIP UAC is unchanged. The capacity of "SIP Proxy" will be improve from 309 to 544. The capacity of "SIP Proxy" in the registration process increases 76%
2. The "SIP Proxy" detects the keep-alive time interval (KATI) of NAT automatically and asks "SIP UAC" to register every KATI seconds. The capacity of "SIP Proxy" can be optimized and maximized. The KATI of different trademark™ NAT is listed in the Table-3.
3. In the NAT device, the KATI of TCP is large than UDP. The TCP-based registration process will improve the capacity of "SIP Proxy" dramatically. The capacity will be improved more than 31 times.

Table-3. The keep-alive time interval (KATI) of NAT for each trademark.

Trademark of NAT	Keep-Alive Time Interval (unit: Seconds)	
	UDP	TCP
PCI CQW-MR500	75	1800
AboCom FSM410	90	>3600
AboCom MH1000	65	>3600
Corega CG-WLR300GNH	75	1800
Buffalo WZR-HP-G300NH	100	>3600
Buffalo BBR-4HG	65	2700
TP-Link TL-R402M	90	1800
D-Link DI-604	95	>3600
D-Link DIR-655	80	>3600
D-Link DIR-320	80	>3600
D-Link DIR-100	75	>3600
Lemel LM-IS6400B	70	900
DrayTek Vigor2104p	100	>3600
I.O DATA NP-88RL	80	>3600
Zonet ZSR0104B	90	1800
ZyXEL P-330w v2	175	>3600
ZyXEL Prestige 304	70	900
3Com WL-537	175	1800
Asus RX-3041	90	>3600
SMC WBR14-G2	65	>3600
Average	90.25	2835

In the "RTP-Relay" system, the bottleneck of capacity is the maximum throughput of packet. Two methods can be employed to enhance the capacity of "RTP-Relay". These methods are described and listed as below:

1. The UDP-based RTP packet can improve the capacity of "RTP-Relay" 2~4 times. For example, in the Table-2, with 16 bytes packet size, the maximum throughput is above 82264 and 16440 respectively. The performance of UDP is much better than TCP.
2. The large size of RTP packet will also enhance the capacity of "RTP-Relay". For example, in the G.729 mode, the capacity of "RTP-Relay" for each time-stamp interval is listed in Table-4. The time-stamp interval increases, the capacity of "RTP-Relay" increases also.

Table-4. The time-stamp interval vs. capacity of RTP-Relay.

Time-Stamp Interval (Sample Points)	RTP packet size (RTP Header=12 bytes)	Packet No./ Second	Capacity of RTP-Relay
10ms(80)	22 bytes	100	66734/100= 667
20ms(160)	32 bytes	50	66734/50=1332
30ms(240)	42 bytes	33.3	44208/33.3=1473
40ms(320)	52 bytes	25	44208/25=1768
50ms(400)	62 bytes	20	44208/20=2210
60ms(480)	72 bytes	16.7	35246/16.7=2110
70ms(560)	82 bytes	14.3	35246/14.3=2464
80ms(640)	92 bytes	12.5	35246/12.5=2819
90ms(720)	102 bytes	11.1	35246/11.1=3175
100ms(800)	112 bytes	10	35246/10=3524

4. CONCLUSION

The performance of SIP-based VoIP server is evaluated and improved in this paper. In the "SIP Proxy", three methods are employed to improve the performance of "SIP Proxy". More than 31 times on capacity is achieved. Moreover, two methods are employed to improve the performance of "RTP-Relay". The capacity with 3524 for each "RTP-Relay" can be achieved. The analyzed results and improvement method can help to plan the VoIP system and architecture.

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A Survey of Online Collaboration Tools

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Abstract - *Nowadays using online tools to support group collaboration has become a necessity. Online collaboration tools allow users to work together in easy and efficient way. A great number of such tools have been developed that provide a variety of different features. In this paper, we investigate the available features of online collaboration tools, the meanings of these features and identify their common and key features. Based to our review, we propose to divide features provided by these tools into three main categories: collaboration, coordination, and communication. Each tool supports different sets of features each of which belongs to a specific category. The proposed categorization aims to identify how well a collaborative tool supports a group work by offering different ways of collaboration, coordination, and/or communication functionalities.*

Keywords: Collaboration, Communication, Groupware, Group work, Online collaboration tools**.**

1 Introduction

Nowadays the use of collaboration tools in order to complete or facilitate group work is becoming increasingly widespread[1]. Web 2.0 providing advanced Internet technology and applications that enhance group communication and collaboration[2]. Many online collaboration tools have emerged which make group collaboration more powerful and convenient and allow people to work together in an efficient and simple manner. Most of these online collaboration tools facilitate information sharing, document management, tasks coordination and communication among groups. And also they support different-place and/or different-time collaboration. There are a great number of such tools that provide a variety of different features. Some of these features are emails, chats, web conferencing, white boards, Wikis, and resource sharing. Each tool provides a different set of features in different ways. In this paper, we will investigate the different features provided by online collaboration tools, the meanings of these features and identifies their common and key features.

2 Background

A collaboration tool is defined as “an implementation of one or more collaboration technologies offered as an integrated package” [3]. A collaborative tool has to fulfil four basic requirements in order to facilitate the collaborative work.

The four requirements are Connectedness, Awareness, Sharing, and Communication[4]. Subsequently, to enable collaboration between participants, participants have to be linked and aware of each other which allows them to share and exchange information and communicate about the common task they want to accomplish. A collaboration tool is also known as groupware. Groupware refers to “any type of software that supports collaboration and/or communication through computer systems” [5]. Also, groupware can be defined as “computer-based systems that support groups of people engaged in a common task (or goal) and that provide an interface to a shared environment” [4]. The main purpose of groupware tools is to help a group of people to accomplish the objectives of a common task[1]. The advantages of using groupware systems are numerous, including: faster communication, saving time and cost in coordinating group work, increased productivity, and the automation of routine processes[5-7]. There are many forms of Groupware, supporting features such as e-mail, instant messaging, shared document storage, group meeting spaces, chats, wikis and shared calendars[5].

Groupware tools can be categorized into synchronous or asynchronous depending on the type of interaction they support[8]. Synchronous tools support real-time interactions where multiple users synchronously manipulate the shared data and all users are immediately notified about the updates produced by other users [8]. For example online chat, video and audio conference, instant messaging and white board all of which is a synchronous collaboration tools. Asynchronous tools support non-real time interactions where users can collaborate by accessing and modifying shared data without having immediate knowledge of the updates produced by other users[8]. Examples of asynchronous collaboration are emails, documents, newsgroups, forums, wikis and discussion boards. Both synchronous and asynchronous collaboration are important. Synchronous collaboration allows people to work together at the same time, providing them with instant feedback[9]. On the other hand, asynchronous collaboration is more flexible where people can work in different time and contribute to the collaborative work providing them with non-immediate feedback.

Due to advancements in Web technology, groupware tools have become more accessible. In the past, a groupware tool had to be physically installed onto the computers to support communication and collaboration, which is called desktop-based groupware tool. But today, many of desktop based systems are replicated by browser-based groupware systems that do not require the installation of additional software and are thus viewed as a more convenient solution[5].

3 Related Work

Xu et al. [10] studied ten current asynchronous collaboration tools and their features in order to identify common and key features in asynchronous collaboration tools. They organized the common features by four major functional categories: communication, information sharing, group calendar and project management. Rama & Bishop [6] compared a number of groupware systems against a set of multidimensional criteria which they proposed. The comparison includes three commercial systems and four academic systems. The criteria proposed include functional, architectural, focus, time, user involvement and platform dependency of each system. Mogan & Weigang [5] conducted a survey to compare three different real-time groupware systems and the result showed that browser-based real-time groupware that takes advantage of Web 2.0 technologies have scored better. Hazemi et al. [11]surveyed a range of popular groupware tools that are useful in the academic sector and support functionalities such as information sharing, contact management, group document management, workflow, and group communication.

4 The Proposed Categorization

Based on our review, we propose to divide the features provided by online collaboration tools into three main categories: Collaboration, Coordination, and Communication. Each tool supports different sets of features each of which belongs to a specific category. The proposed categorization aims to identify how well a collaborative tool supports a group work by offering different ways of collaboration, coordination, and/or communication functionalities (see Fig. 1). We briefly describe each category with its features.

Collaboration: Collaboration features facilitate collaborative work by providing the means to share different types of information in different ways. Features include file sharing, discussion board (forum), links sharing, polls, online office and Wiki. In general, file sharing enables users to upload and download files of different format such as document, spreadsheet, presentation, audio, video, PDF ... etc. Uploaded files are stored in a centralized place where users can access them anytime from anywhere. Sometimes, file sharing is enriched with a version control (or versioning), so that users have an access to all versions of stored files. Discussion board is a way of sharing information where users can discuss and exchange ideas about a specific topic by posting messages and comments in an asynchronous matter. Links sharing feature allows user to share bookmarks. Polls feature allows collecting users' votes on some topics/issues. Online office enables collaborative creation and editing of different types of office documents such as word, presentation, and spreadsheet. Finally, wiki allows users to collaborate by directly editing the contents of a web page.

Coordination: Coordination features facilitate managing tasks and group members. Features include shared calendars, to-dos, milestones, and alert/notification. Shared Calendar usually shows milestones, scheduled meetings, appointments, upcoming events and the participants of the events. To-dos feature represents tasks that have to be done. Task is usually assigned to person/s and has a start date, and end date. Milestone is used to mark the due date of important events or deliverables of a project. Alert/notification feature is used to notify group members about new events such as, new file is uploaded, new message is posted, etc.

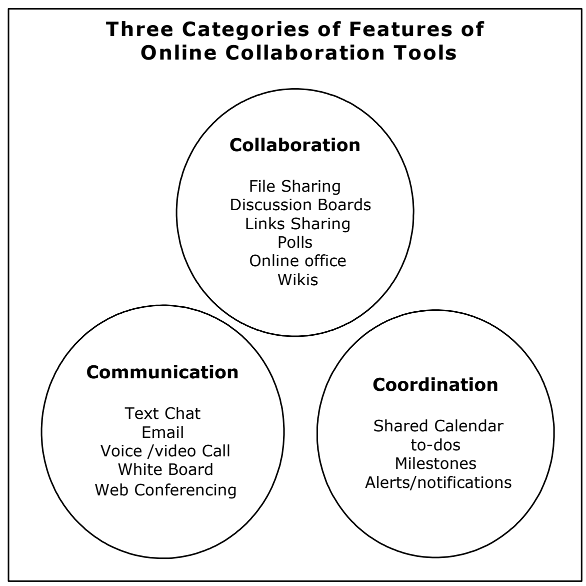

Fig. 1 The proposed categorization

Communication: Communication features are an interactive way of collaboration. The common communication features include Email, text chat, voice/video calls, white board, and web conferencing. Email feature is integrated with the tool such that user can directly send a message to email address of other users that are registered in the system. Text chat means exchanging of instant messages between group members synchronously. Usually, text conversations are saved and archived for future reference. Voice/video calls enables users to communicate using Voice over IP and/or video stream. Whiteboard is a shared space that allows users to communicate visually with others in real-time using drawing, writing and conducting presentations. Web conferencing is a real-time communication that enables group meetings. Web conferencing usually combines many functionalities such as Voice Over IP, live videos, text chat, slide presentations, desktop sharing, polls, and meeting recording.

5 Conclusion

When people need to accomplish group work, they need a reliable collaboration tool that can help them to reach their goals in an effective and efficient manner. They have to select the collaboration tool that meets their requirements and needs. This paper investigates online collaboration tools and their features. We proposed to divide features supported by these tools into three main categories: collaboration, coordination, and communication. Each category has its own features and functionalities. The proposed categorization can help us to assess online collaboration tools. As a future work, we plan to apply this categorization to a number of online collaboration tools, and compare them to show how well they support collaboration, coordination, and/or communication functionalities.

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Mobile User Interfaces and their Utilization in a Smart City

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Abstract - *Up-to-date web (2.0) is collaborative, mobile and contextual. It takes into account human actors as well as different things, i.e. LBS (location based services) and internet of things are now an integral part of internet. We are working on this approach with the acronym MOCOCO (Mobility, Contextualization, Collaboration). Our research work is generic with multiple applications in working, learning and societal situations. Professional and home working situations, professional and teenager contextual mobile learning situations as well as Smart City applications are taken into account: transportation, goods distribution and local sport and cultural activities. In the paper we propose to present our approach for contextual mobile user interfaces and their application to the Smart City.*

Keywords: Mobile web, Location-based services, Internet of things, mobility and nomadism, in-environment user interface, environment dependent and independent user interface, mobility, contextualization, collaboration

1 Introduction

Ambient intelligence (AmI) refers to electronic environments that are sensitive and responsive to the presence of people [1]. In an ambient intelligence world, devices work together to support people in carrying out their everyday life activities, tasks and rituals in an easy and natural way using information and intelligence that is hidden in the network connecting these devices. AmI is a distributed system [2]. It represents a new vision of daily life, consisting of several kinds of sensors and computing devices, which lead to pervasive intelligence in the surrounding environment supporting user activities and interaction. This means that computing technology will exist in everything that surrounds us (devices, appliances, objects, clothes, materials) and that everything will be interconnected by a ubiquitous network. The system formed by all these intelligent things (also called the Internet of Things) will interface with humans by means of more advanced interfaces, which are natural and flexible and which adapt to the needs and preferences of each user. The final goal is to obtain an adaptive and "intelligent" system that assists humans in their day to day activities.

To attain Ambient Intelligence (AmI), first some kind of mechanism is required that enables sensors and computing devices to achieve it. Furthermore, it must also detect changes in context, which could affect the applications executed in user devices. In computing, there is a discipline that deals with this kind of problem, namely "context-aware computing" [3]. Context-aware computing, which was first discussed by Schilit and Theimer [4] in 1994, refers to a general class of mobile systems that can sense their physical environment, and adapt their behaviors accordingly. Inside this area, a special kind of application has emerged, the context-aware middleware solutions. In general, they provide the same functionality associated with traditional middleware in terms of communication. However, they also provide context management and adaptation to deal with the different resources present in the environments [5].

From a device point of view, desktops and laptops are no longer unique in their kind: more mobile devices such as TabletPC, PDA, and Smartphone, known as wearable or handheld computers, are being increasingly used.

In this paper we explain our proposal for a context-aware middleware for ambient intelligence, a taxonomy of user interfaces allowing mobile and nomadic actors to participate in this real augmented environment. We also describe a particular situation, which is a subset of the Smart City, i.e. City 2.0 as a real augmented environment for digital born actors [6].

2 Context-aware middleware for ambient intelligence

The goal of our research is to develop a context-aware middleware for ambient intelligence. This middleware can collect contextual information from various interaction devices (data gloves, Wii Mote, etc.), techniques (gesture recognition, markers or object recognition, etc.) and sensors (RFID, QR codes, etc.), and use these contexts to provide relevant information and/or services to the user, where relevancy depends on the user's task [7].

We propose a context-aware middleware for building and rapid prototyping of the context-aware services in the ambient intelligence, which consists of various computational entities (Smartphone, computer and tablet) with appropriate UI devices and various sensor-based devices (RFID, camera, and marker). The architecture is made up of the following components: sensor data fusion, reasoning engine, Context Knowledge Base (KB), context database, context query engine. OWL is used to express a context model taking into account mainly, but not only, our laboratory sensors. Its main component services are (figure1):

- Sensor data fusion: collect and transform the information from the sensors to OWL (we use ontology OWL for context description);
- Reasoning Engine and Context KB: check context consistency and deduce the high-level context from low-level context;
- Context database: store the context data;
- Context query engine: handle the query from the application.

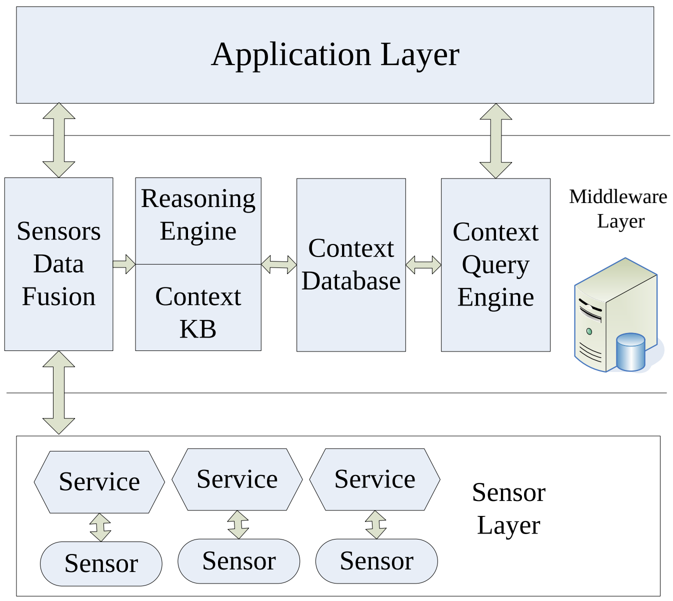
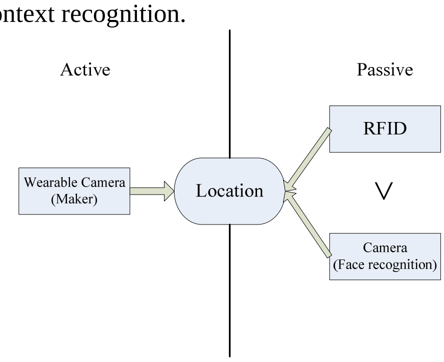


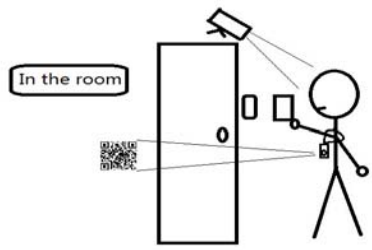
Figure 1. The architecture of our context-aware system

In our case, the in-door location system is one of the most important context providers for our context-aware middleware. It locates people by three sensor-based devices: camera, RFID, and marker. We set the camera on every door in the building. When someone enters, his/ her face could be captured by the camera on the door, after which the face recognition system compares his/ her face in the database to try to find who he/ she is. We put the RFID reader in the room. When the person is already in the room, the RFID ID system confirms presence via the card worn. These two sensors can obtain location information without the user is participating to this collection. This is the passive method. We also propose an active method. The marks (QRcode or ARtoolkit) [12, 14] with location information are affixed all around the room. The person can use the wearable camera to capture the location information and transfer it to his / her Smart phone, tablet and the location system database. According to precise data and robustness, we define the priority for the location data of this location system to avoid conflicts. This active method has top priority. Data from the RFID take priority over data from camera (Face recognition).The structure of the in-door system is shown in Fig.2 (a).

We try to use a simple scenario to interpret how this system works (Fig.2 (b)). In the morning, Tao enters the laboratory, his friend behind him says hello and Tao turns round and smiles at him. The camera has missed Tao's face, and yet the RFID reader obtains Tao's information precisely according to Tao's card, and then transmits Tao's location to the server. Tao comes in and hangs his coat up in the hall, before going to his room. But he leaves his card in his pocket. It doesn't matter, as the marker in his room can help him locate his position and update data by his wearable camera. In this way, by multiple sensors, we are able to obtain more robust context recognition.



(a) Structure of the system



(b) A particular scenario
Figure 2. The in-door location system

Our context-aware middleware provides a rapid programming interface to handle context information for multiple sensors.

3 In-Environment User Interfaces: A taxonomy

As we all know, in the late 80s and early 90s, Mark Weiser [8] published a vision for the next generation of computers, which he termed as ubiquitous computing. Ubiquitous computing is a promising research area, which focuses on the integration of technology in daily life with the end goal of making technology invisible to the user. This technology represents a move away from the current second generation desktop mode, in which the user’s interaction with technology is intentional and deliberate, to a third generation computing model, in which the user may engage several technologies at once, virtually without being aware of the interaction. However, as the world moves ever closer to integration of technology in every aspect of life, a greater need for innovative research and development into various aspects of ubiquitous computing has emerged. Issues relating to ubiquitous computing and pervasive computing vary from the practical problems of user interfaces and miniaturization of devices, to the more ethical issues of privacy and data protection. In our opinion, we found that the traditional user interface, used on the desktop computer, is no longer appropriate for the new mobile and connected devices. Thus, our aim is to study the innovative wearable user interfaces which can help the user to interact freely in an environment with mobility and in context. Our theoretical background is based on mobile interaction in real augmented environment [9]. In this case, mobility, as we understand it, is not that the user is physically mobile, nor equipped with conventional mobile devices such as the laptop computer, but rather that the user wears several computing devices like the pico projector, webcam, goggle attached small screen and so on. Many researchers in this field investigated these new interfaces and interaction techniques. MIT students have designed and developed the WUW prototype system [10], a wearable gesture interface, which attempts to bring information out into the tangible world. Moreover, the Skinput technology [11] enables the skin to be used as an input surface, and provides an always available, natural portable finger-input system. Besides the background of mobile interaction, our approach is inspired by contextual markers in order to acquire a conscious augmentation. This can be achieved by active markers such as RFID stickers and passive markers such as QR codes and AR-Toolkit markers [12,13,14]. In our current work, we add AR-Toolkit markers to the interface, which link the digital world and the real world in a simplified and economical way.

We propose three types of interface on the basis of the relationship with the environment: in-environment interface, environment dependent interface and environment independent interface.

1/ We define the in-environment interface (IEI) as a interface which is fixed in the environment and the nomadic user, i.e. the user without his/ her UI devices, can interact with the application.

2/ We define the environment dependent interface (EDI) as the interface which is closely connected with in-environment information and markers (on the wall of the corridor, on the door, on the surface of an appliance or any other surface). This interface has the ability to provide intuitive interaction techniques, which can recognize and understand the situation of the user and the real environment around him/ her. The information media, which are the tangible markers, are static and protected against vandalism. In this way, public or professional guiding information can be used for contextualization based on webcam recognition. Mobile users have the UI devices, which allow them to dominate the interaction.

3/ We define the environment independent interface (EII) as the interface without any contact with the environment. The user is the sole source of contextualization, i.e. he/ she can acquire information at any time and any place. What this means is that contextualization is carried out by the actual user by showing appropriate contextualizing markers to the webcam. These markers can be grouped on a grid and selected by finger, mask or by turning over the pages of a notebook. In addition, on his/ her own initiative, the user can project the menus, schedules, websites, videos and other information on a flat surface (the wall) in the environment or on a personal surface (a blank sheet of paper, or part of the human body). The user can move freely in his/ her working environment and obtain contextual information independently from this environment.

We have designed and implemented a series of innovative interface prototypes, which allow the user to interact while moving with at least one hand free. These prototypes are mainly based on: a webcam for perceiving context and user interaction; a goggle-attached small screen for visualizing text, image and video, or a pico projector for acquiring a larger view and interface; a computing device for calculating, located in the pocket or carried on the back. The software for these prototypes was developed on the Microsoft Windows platform using C, C++, OpenCV, GTK and AR-Toolkit.

We first studied the in-environment interaction techniques by fixing the webcam on a plastic bracket on the desk (Figure 3). We segmented the whole visual webcam field into the rectangular grids and linked each grid rectangle to a single event. The user wears a color sticker on his index finger and can trigger events chosen by pointing the grid. We then added the markers to the grid rectangles, which can be recognized by AR-Toolkit. Two main points have been focused at this stage: the study of grid size and organization, and the study of marker recognition and their arrangement.

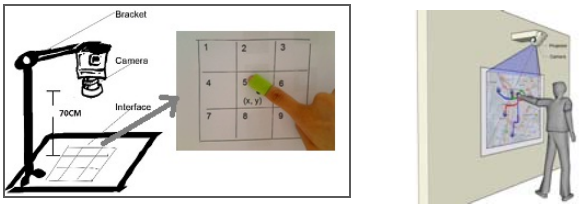


Figure 3. In-Environment Interface (IEI)

We then fixed the webcam to the user's body as an input device, and goggles attached small screen as an output device. Three interaction techniques are designed: the finger selection technique, the mask selection technique and the page selection technique. We created several scenarios and implemented the relevant applications in order to show how these prototypes can promote interaction in the context and help the mobile user access information freely in a specific environment or a general setting. The applications based on the dependent interface (Figure 4 (a,b,c)) include an indoor way-finding application, a "Research Team interaction System" application, and the bus stop instruction application.

Figure.4 Environment Dependent Interface (EDI) **a)** Camera and goggle based screen configuration **b) c)** in-environment markers

Furthermore, the independent interface applications (Figure 5 (a,b,c)) based on a the projector instead of goggle attached small screen, cover the industrial maintenance assistance application, which can offer help in a specific working situation. The webcam and the projector are combined as input / output configuration, fixed on the head of the user.

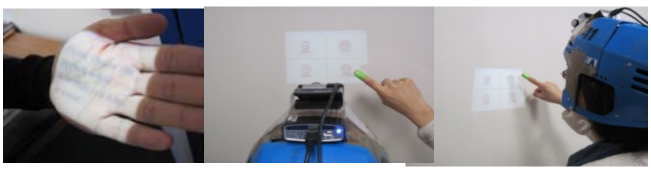


Figure 5 Environment Independent Interface (EII) **a)** information projected on the palm **b)** Camera and picoprojector configuration **c)** information projected on the wall

We conducted an experiment to compare the finger selection technique with the mask selection technique on a small scale. The goal was to contextually study the differences in the efficiencies between two selection techniques. We found that though the finger selection technique takes less time to select one item, it generates an uncomfortable feeling for the user. The user feels cumbersome in that he/ she aims at a target carefully in a spatial interaction situation. Though limits exist, our method and prototypes still have advantages. Our system allows mobile interaction and leaves the user with at least one hand free. With the marker, the user can obtain information immediately quicker and more easily than by using several text inputs and menu selections via his mobile phone.

To better understand the issues for interacting without the limits of paying attention, our goal is to design and implement the novel WIM-HG style interfaces, which support the user with one single hand to interact. Compared with the WIMP (window, icon, menu, pointing device), we defined the **WIM-HG** style as **window, icon, menu and hand-gesture** interaction techniques [15]. It is worth studying such alternative interfaces, which should be easy to learn, have appropriate feedback and reduce mental load. While continuing to investigate our Environment Dependent Interface and Environment Independent Interface, we chose the same exploratory approach and almost the same configurations that we performed previously. We added the WPF, QR decoder as well as the encoder to the prototypes. At the present stage, two prototypes are studied and more will be investigated in the future. We created a pointing technique consisting of two gestures in the first prototype. In a scenario of picture browsing, a traveler wants to browse the photos in a photo gallery. He/ she finds a flat surface and then projects the photo gallery onto the surface. He/ she then uses one of their hands to navigate the interface. He/ she uses the gesture of full hand to move the indicator of a cursor. When he/ she wants to select one photo to operate, he/ she closes his/ her hand where the cursor locates, then opens their hand to select the photo. Then, when he/ she wants to delete this photo, he/ she only needs to select the delete button in the same way (Figure 6 (a,b)).

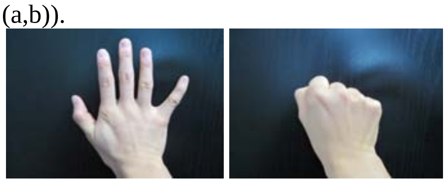


Figure 6 Hand gestures **a)** Full hand **b)** Closed hand

The disadvantage of this method is that the user still needs to do a gesture before navigating the buttons. Thus, in the second prototype, we divided the function of the photo gallery into four parts: browse, rotate, zoom and delete. Then, we discard the pointing technique, and instead, select five common hand gestures which correlate with the commands. In this way, we avoid the constraint of careful pointing. However when choosing an appropriate gesture, it is necessary to consider whether this gesture is easy to learn, remember and recognize.

4 Case Study: Smart City context

In our approach, a bus stop (Figure 7) is an important example of a significant component of a Smart City. It is located at a strategic place where buses stop for passengers to board or leave a bus. In a previous paper [16] we studied and proposed a huge number of services located around or in relation with the bus shelter. We can implement our AmI environment and different sorts of user interface (IEI, EDI and EII).

In our approach as use of a bus shelter as a hotspot, we compare classical www distributed and virtual Communication / Cooperation and situated hotspot based local Communication / Collaboration. We are mainly concerned with contextual mobile situated collaboration in two cases: transport cooperation management and social neighborhood management. In this context we are able to implement IEI (In-Environment Interface) as a part of the Bus Shelter for monadic users (without interaction devices). An EDI (Environment Dependent Interface) can also be supported by the bus shelter, the way in which static information, such as bus shelter name, time table referential, line numbers, etc. are written in the environment and can be used by a mobile user, user equipped by his/ her interaction devices (camera and Smartphone or goggles attached small screen) in order to propagate the interaction on his/ her backpack computer and receive the answer on the screen. This interaction is naturally more personal respecting user privacy and autonomy. Finally, UI, EII (Environment Independent Interface) is currently a little more expensive (using picoprojector), but allows a higher level of interaction and enhanced visual comfort. This kind of interface is for the time being reserved for professionals, i.e., transportation and bus shelter maintenance personal, requiring a large amount of information accessible on site.

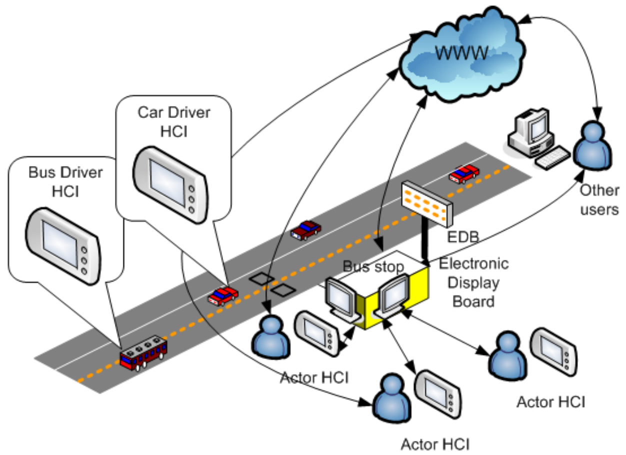


Figure 7. Bus shelter physical and informational environment

On figure 8 we show an in-environment support for EDI (Environment Dependent Interaction). Visual menus affixed on the bus shelter wall are the support for this interaction.

Naturally, several electrical sensors as RFID readers can be located in the bus shelter in order to capture personal information on the in-coming user. The shelter can thus display on the main screen of the shelter, appropriate information for this newcomer. On the screen, a QR-code can be displayed to allow the user to collect information of interest to him/ her and that he / she can save to read later on his/ her private display unit, as a goggles screen.

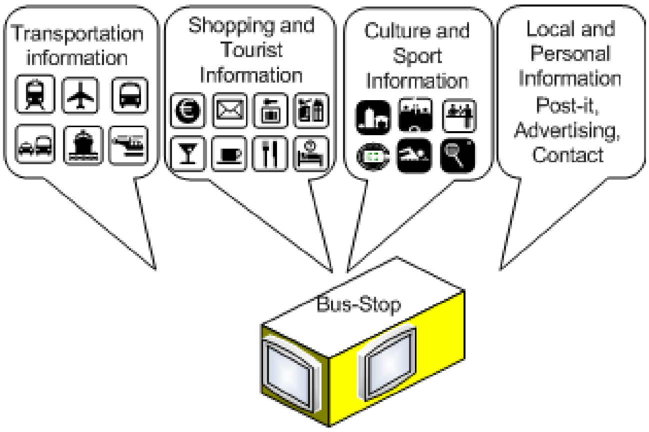


Figure 8. In-environment support for contextual interaction

The picoprojector based EII (Environment Independent Interface) can be used in professional activities as a contextual mobile learning medium [17] in order to master, on site, specific functionalities not yet mastered by the technician.

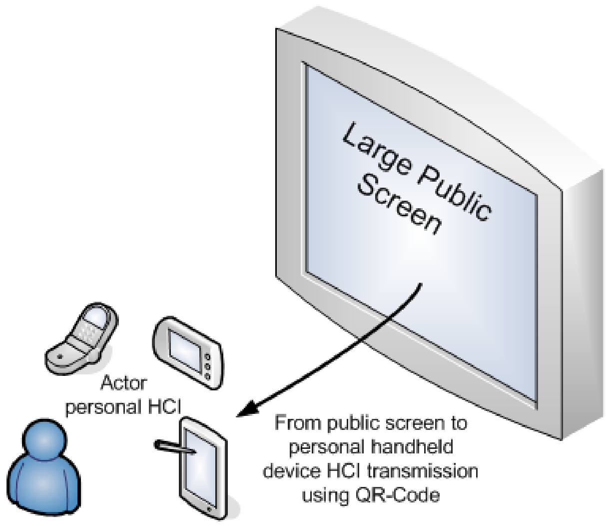


Figure 9. Information exchange between a large public screen and personal handheld devices by QR-Code to allow personal and confidential exploration and treatment based on ID and PIN.

Following initial use of these 3 kinds of interface, we consider that they are satisfactory and acceptable in the context in which they have been used. On course, a more complete evaluation, with a larger population, must be conducted.

5 Final discussions

New User Interactions are required in the context referred to as Real Augmented Environment, the goal of which is to support ambient intelligence and, ubiquitous computing. We proposed 3 kinds of interface (IEI, EDI and EII), which we then concretized with a first set of interaction devices (webcam, goggles based screen and, if applicable, picoprojector) in relation with complementary context-sensitive stickers (visual or electromagnetic) used for active or passive contextualization.

Naturally, this configuration is not the final one, but can be considered as an intermediate stage leading to new interaction paradigms required for nomadic and mobile situations.

Our case study, based on Smart City situation and on an actual example formulated by communication Bus Shelter, seems to be at least for two of the three proposed UI (IEI and EDI) an appropriate test bench. For the last one (EII), a more professional context is necessary, relating to the cost of a picoprojector.

Whatever the circumstances, we will continue to propose and evaluate these different kinds of interface in order to find appropriate configurations and use cases.

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Causes of Mistrust and the Most Effective Authentication Tools for Improving Saudi Arabian Consumers' Willingness to Participate in E-commerce

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Keywords: Mistrust, consumer identity theft, user authentication

Abstract:

E-commerce is facing different challenges globally, and in particular, in developing countries. In some countries e-commerce adoption is slow because of some barriers; such as lack of trust [3][5]. Customer trust is a significant issue in e-commerce since online services and products are typically not immediately verifiable [6]. Both vendors and consumers are forced to assume that the information that is being provided is accurate and non-fraudulent [7]. Therefore, customer trust significantly affects new customer acquisition, customer retention, and customer purchase intentions [8].

In addition, PC sales and Internet growth in the Kingdome of Saudi Arabia (KSA) are considered to be among the highest in the region (KSA Ministry of Commerce, Council of Saudi Chambers of Commerce and Industry). Also, the growth of internet usage in Saudi Arabia is considered to be tremendous and estimated to be 3,750% from 2000 to 2009. The number of Internet users is still growing and has recently reached 8.5 Million [8].

Although the number of internet users is increasing; still, e-commerce acceptance in KSA is modest. The lack of trust is considered as a significant barrier to e-commerce adoption, and it is one of the main reasons for customers avoidance of Internet purchasing [12]. From this perspective, how to encourage trust in buyers is an important research issue.

Consumers typically have no problem navigating a site and acquiring information that is needed, however the issue of trust arises at the time when a decision needs to be made regarding making an electronic payment [9]. Wakefield mentioned, "Trust in a Web vendor is an important variable for the completion of an online transaction, and consumers who trust are more likely to exhibit positive online purchase intention." In fact, when online customers have high confidence in the web-site's security measures they will be more willing to shop and purchase online.

One of the main problems that causes fear to online shoppers is the issue of consumer identity theft. In fact, studies showed that “the growth of identity theft is damaging the consumer trust and confidence in e-commerce” [10]. Moreover, one survey reported that “57% of consumers believe responsibility for protecting their online identities and personal details lies with web-based companies” [10].

Identity theft is obtaining another’s personal information and using it without his/her knowledge or consent to commit fraud for financial gain or for another criminal purpose. The most common types of identity theft problems are related to payments card fraud, which hinders buyers from conducting online transactions. In fact, approximately 50% of all identity theft victims reported that a Credit Card was opened in their name or unauthorized charges were placed on their existing cards.

There are different ways to build customer trust online. One of the most successful examples is eBay for its penalized policy. That is “trustworthy behavior is rewarded and untrustworthy behavior is penalized by the traditional process of communicating experiences from person-to-person” [11]. Other companies are using different devices to build trust with customers and are exploring the role of

design, style, presentation, and electronic devices that ensure trust in e-commerce transactions.

Another important method for improving consumer trust is by adopting strong user authentication techniques [1]. There are several different technologies that can be used for authentication, and these may operate with one another. For instance, in a study by the *Federal Deposit Insurance Corporation* (*FDIC*) in 2004[2], they listed some authentication methods such as passwords, the use of one time password, tokens, digital signature, and the use of some biometric authentication methods: fingerprints, iris scan, and face recognition. However, regardless that more effective solutions of authentication are generally more expensive, trust of customers may be more important.

This research-in-progress intends to investigate the main causes of mistrust and hesitance of people in Saudi Arabia in purchasing products online. Moreover, it also looks into the Saudi consumers' opinion about different authentication methods and how favorable they are of each, and determining if such authentication methods would improve the consumers' willingness to increase their level of e-commerce participation.

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Using mouse movements in the authentication process as behavioral authentication technique

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Abstract - *The majority of current information systems applications are based on retrieving the explicit data and information stored in databases while some other applications like data mining take further steps through extracting the implicit knowledge that can be deduced from the explicit ones using some techniques and algorithms. These Techniques and algorithms need to be mature enough to extract such knowledge. Lack of certainty in these tools lead to poor results and wrong findings sometimes.*

In fact, in the last decade a lot of research was conducted in behavioral studies as unique identifiers for humans. In this paper, we will focus on the area of analyzing users mouse movements to identify and re-authenticate the users behaviorally since many physical studies proved that every single person has his own unique mouse movement signature.

Most of the current authentication techniques are based on something the user knows, something the user has, or something the user is. In this paper, we will add another dimension which is something the user does. Our proposed technique might be used as a supporter tool since it still suffers from high rates of false acceptance and false rejection

Keywords: Authentication, Mouse movement, False Acceptance Rate, False Rejection Rate.

1 Introduction

Getting the advantages of behavioral based techniques will not come without suffering some serious pains as we will see in this paper. This section will highlight the problem of certainty, and the confidence level of these techniques, and how accurate they are.

To answer the question of why behavioral techniques are not widely used, a variety of conditions may ultimately led to that:

•A lack of skills, knowledge, or awareness in the data mining and other intelligent solutions.
•Technical challenges and limitations.
•Limited competent resources.
•Lack of communication between IT and other fields experts like mathematicians, physicians, statisticians, etc.

In the e-business environment, both consumers and service providers suffer high levels of insecurity. Concerns about trust, authentication, security, and fraud are often in people minds which creates some barriers for ecommerce growth [1].
Most traditional authentication mechanisms became not robust enough for e-commerce environment because of different nature, processes, and interfaces.

Here we propose a new authentication framework for online authentication which has several advantages if combined with the traditional techniques. Also, we will explain how the proposed framework can be used to increase the security level in the e-commerce environment. The new technique is not a replacement for any current mechanism, it will work with the current techniques to maximize the security and confidence levels.

In a few words, the solution is about authenticating, and continuously re-authenticating users behaviorally in an automatic fashion simultaneously in addition to other traditional techniques.

One of the proposed techniques was studying the mouse movements (called mouse dynamics as well) through developing some statistical models. Although this technique is effective, it still suffers from some accuracy issues called False Acceptance Rates (FARs) & False Rejection Rates (FRRs) that affect its trustworthiness since people will not invest more in shaky systems and they will prefer to work with traditional techniques like *Username/Password* or any other stable identification and authentication techniques like *fingerprint*.

2 Current Authentication Techniques

Authentication by definition is not a new concept in information security. On the other hand, authentication techniques keep changing to keep pace with new applications and environments. In Table 1, we list some authentication techniques and evaluate their performances.

	Performance Criteria					
Authentication Mechanisms	Robustness	Acceptance	Cost	Ease of use	Portability	Security
1. Buyer Registration at Seller's Web site	Low	Low	Low	High	High	High
2. Buyer's Machine/Access Device	Low	Low	Low	High	Low	High
3. Credit Cards	High	High	Low	High	High	High
4. Smart Cards and Tokens	High	Low	High	High	Low	High
5. Digital Certificates	High	High	Low	High	Low	High
6. Digital Certificates and Financial Institutions	High	Low	Low	High	Low	High
7. Biometrics	High	Low	High	High	Low	High
8. Cookies	High	Low	Low	High	Low	High
9. (URL) Advertising	High	High	High	High	High	High
10. Web site Content originating from Seller	Low	Low	Low	High	High	Low
11. Web site Content originating from Public Sources	High	Low	Low	High	High	High
12. Product Try-outs	High	High	High	High	High	High
13. Virtual Communities hosted by Seller	Low	High	Low	High	High	High
14. Independent Virtual Communities	High	Low	Low	High	High	High
15. Online Trust Services	High	High	High	High	High	High
16. Active Digital Market Makers	High	Low	High	High	High	High
17. Passive Digital Market Makers/Reputation Systems	Low	Low	Low	High	High	High

Table 1: Current Authentication mechanisms and their performance levels [1]

3 The proposed Solution

The solution provided here depends mainly on mouse metrics values like number of clicks, double clicks, curves, silence, etc. in specific durations. It collects such information and inserts in the model and build a profile for each user based on his behavior. The accuracy and maturity of each profile will improve overtime since it should go through a learning curve.

The following components are the main components of the proposed solution as per [2]:

1) Data Interception Unit
2) Behavior analysis Unit
3) Behavior Standardization stage
4) Behavior comparison Unit

4 Conclusion

As mentioned earlier, mouse dynamics technique is a new behavioral biometric, recently introduced. The concept in this technique is to observe all the actions of the mouse which are generated as a result of the user interaction with the application, and then all these actions should be processed in order to analyze the user behavior. In the extended version of this research we shall provide details on how this new technique should work, in addition to describing its advantages, and drawbacks.

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Adaptive Histogram Algorithms for Approximating Frequency Queries in Dynamic Data Streams

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Abstract – *Histograms can be used as summaries of frequency data. However, staying within the error tolerance becomes problematic when dealing with dynamic data streams. For dynamic data streams, the histograms can be reconstructed every time data is either discarded or collected - which is very inefficient. If a histogram is to be employed as a quick estimate of stream data, updating the histogram non-destructively can be done using the following approach: decrement one from each bucket where data is to leave the histogram, and increment one to each bucket where data is to enter the histogram. In this paper, we empirically prove this method to be a generally strong way to control loss of accuracy. The costs of executing this error-minimizing layer are trivial to processing, memory, and should consequentially maximize uptime. This method was tested on two histogram algorithms including Equivalent Width and Variance Optimal in four specified histogram data-density scenarios including sparse, balanced, dense, and very dense, while using two different random value distribution sources including the Uniform distribution and Gaussian distribution.*

Keywords: Histogram, Frequency queries, Data Stream, Approximation, Algorithm

1 Introduction

Histograms, utilized as a summary of frequency data, have been proven to be accurate-enough measures to approximate count (or frequency) queries. Staying within the error tolerance becomes problematic when dealing with dynamic data, such as streams, due to the potential for shifts in source data. This shifting can happen even if a stream is modeled by using a single random distribution, especially when observing a relatively smaller number of values as related to a large or infinite data set. Elements can also expire and become irrelevant, as well as new elements can come into existence. Attempting to run a histogram on dynamic data without a method of controlling error will become disastrous, especially on specialized histograms such as the Variance-Optimal and Maximum Difference due to the way they interpret data inherent to their original construction. In lieu of reconstructing a histogram every time data is either discarded or collected - which is prohibitive in processing power, memory space, and real life uptime - a method for reducing cumulative error is necessary if not imperative. Therefore, if a histogram is to be employed as a quick estimate of stream data, updating the histogram non-destructively can be done using the following approach: decrement one from each bucket where data is to leave the histogram, and increment one to each bucket where data is to enter the histogram. The costs of executing this error-minimizing layer are trivial to processing, memory, and should maximize uptime. In this paper, we have tested this method on two histogram algorithms including Equi-Width and Variance Optimal (also known as V-Opt or V-Optimal).

2 Background

Here we will discuss some of the basics of queries and histograms. Various types of frequency based queries are discussed that can benefit from histograms.

2.1 Queries

There are three types of queries that can benefit from histograms.

2.1.1 Selection queries with equality constraints

Selection queries with equality constraints return the tuples in a table that satisfy a certain equality criteria. You can also associate a count function to these selection queries. For example, in order to determine the number of employees with age 65, the following SQL-like query could be executed:

select count(*) from employee where age=65

This query can be further extended to include other values in the following manners:

select count(*) from Employee where age=62 or age=65

select count(*) from Employee where age in (62,65,67)

If a certain amount of error is acceptable, a histogram can be used to estimate the frequency of a certain value in a data source. The individual frequencies can then be added to get the combined counts of extended queries above.

2.1.2 Range queries

The range query can have either a lower or an upper limit on its input, or both limits explicitly declared. For example, suppose an executive in a company wants to know how many of their employees are making salaries between the range of 50,000 and 100,000 inclusively. The SQL-query would be structured as:

select count (employee_id) from Salary where salary >= 50000 and salary <= 100000

Here also histograms can be used to compute the count for the whole range by adding frequencies for each individual value in the range or by manipulating frequencies of whole buckets.

2.1.3 Join Queries

Join queries are more targeted towards pattern matching. Just as in selection queries, a count function can be attached to join queries as well. Consider the scenario where personal info for employees is stored in table Employee and their salary information is stored in table Salary.

Select emp.first_name, emp.last_name, emp.age, s.salary from Employee emp join Salary s on (emp.empId =s.empId) where s.salary>=100000 and emp.age <=35

If accuracy can be sacrificed for efficiency, histograms can be used to provide estimates on each data source, which in turn can be used to generate join query result-counts. Histogram information on the individual tables can also be used to create an optimized query plan that would run faster.

2.1.4 Histogram Algorithms

Histograms are compressed versions of an entire data set that are used as statistical tools of approximation. There are many ways to create a histogram that mainly differ in the method of setting boundaries commonly known as buckets (Ioannidis Y. 2003). Each individual bucket can represent a range of values where the range can be as small as a distinct value, and as large as the whole data range. How to break up a data set into subsets to store in a single bucket is up to the histogram algorithm. The easiest way to visualize a histogram is to picture a bar graph with the bucket ranges on the horizontal axis and the frequency counts on the vertical axis. By design, histograms are excellent tools for single count queries and range queries. In the following subsections, we will briefly describe two of the most popular algorithms that will be used in this research. Both the examples will be using the same sample array: {0, 2, 2, 3, 3, 4, 5, 8, 8}

2.1.5 Equivalent Width

One of the more basic histogram algorithms, the equal-width or equiWidth method separates data into horizontally equal-sized buckets. This means that each bucket will represent, as closely as possible, an equal number of values. The benefits of this algorithm lie in the simplicity – and consequentially the inexpensive computing cost to construct it. The time complexity is O(n) due to the single pass required to process through the pre-sorted source array to build the histogram.

2.1.6 Variance-Optimal (V-Opt)

The variance-optimal (also commonly referred to as V-Opt and V-Optimal) histogram is one of the more complicated histogram algorithms. It is also broadly regarded as one of the more accurate histogram algorithms. The premise behind this algorithm is to minimize the sum of all intra-bucket variances. There exists a dynamic programming algorithm with O(n^2B) time complexity that follows the following recurrence relation

$$SSE^*(i,k) = \min_{1\le j<i} \{SSE^*(j,k-1) + SSE([j+1,i])\} \quad (1)$$

In the above equation, n is the number of elements in the data array, B is the number of buckets to construct the histogram and *SSE*(i, k)* is the minimum sum of squared error *(SSE)* for the prefix vector *F[1, i]*, i.e. the first i values of the frequency array corresponding to the data array using at most k buckets. Notice, for this algorithm we first need to convert the data array to a frequency array F, where *F[i]* is the frequency of the value mapped to slot i. As you can see that this histogram construction algorithm can become cost-prohibitive as n grows bigger.

3 Approximating count queries using Histograms

Suppose we would like to know the frequency of value v. In order to do this, we would first find out which bucket i value v belongs to and then find the frequency f_i and number of values n_i for that bucket. Then the approximate frequency for v can estimated to be f_i / n_i. For our sample array, according to the equiWidth histogram the frequency of 3 is 4/3 = 1.33 and according to v-optimal histogram the answer is 6/6 = 1. As you can see, for v = 3, v-opt has a higher error (2 – 1 = 1) than equiWidth (2 – 1.33 = 0.67). However, for v = 2, error in v-opt estimation is 2 – 2 = 0, whereas equiWidth has an error of 2 – 1 = 1. On average, v-opt produces lower error if all values in the data range are equally likely to be queried as it is designed to minimize average intra-bucket variance (SSE) which ultimately minimizes average error.

3.1 Histograms for Dynamic Data Streams

The goal of histograms is to be as accurate in estimating data distribution as possible while improve speed of answering queries and also the speed of histogram construction. The speed of query execution results from the faster access of a histogram array versus searching and counting an entire data array. Since error is produced by the averaging of the frequency among all the values located in the same bucket, it can be reduced by increasing the number of buckets in the histogram. At one end of the spectrum is the one-bucket histogram that will have the highest error and fastest construction. On the other end is the *n*-bucket histogram where *n* is the number of distinct values, which will have the slowest construction time and no error. As one can see, the two goals reducing error and increasing construction speed are at odds with each other. One may think one should just use *n* buckets since accuracy is more important than offline histogram construction time. However, using an *n*-bucket histogram not only increases construction time but it also requires more space to store the histogram. As a result choosing the right number of buckets and the right algorithm is crucial for system performance.

With that established, the problem becomes even more difficult when running frequency queries on continuous data streams comes into picture. Since a histogram could become increasingly inaccurate with each new data point entering the stream, we investigate the effectiveness of a simple but fast method in mitigating the potentially distortional effects a stream could have on histograms.

3.2 Dynamic Data Streams

In order to emulate a dynamic data source, a sliding window based method is used. This sliding window approach used in experiments is very similar in nature to a stock or financial ticker on a news channel. The television can only show as much data that can fit on the screen, just like a fixed-size sliding window stores data. As the ticker scrolls, stock data exits the screen and new and more relevant information comes into the picture. The point of this parallel is to illustrate the core workings of the sliding window mechanism, as applied to processing data stream simulations. In this paper we consider only fixed-size sliding windows as opposed to variable-sized sliding windows.

3.3 Reducing Inaccuracy

In the scope of this paper, inaccuracy, I, is measured using a normalized error technique given by equation 2.

$$I = \frac{|V_A - V_H|}{V_A} \qquad (2)$$

Here V_H is the answer returned by the histogram and V_A is the the actual frequency in the sliding window.

By establishing a metric for representing inaccuracy, a measured comparative difference can be calculated from various testing simulations. Static histograms have error built in to begin with. Therefore, if a distribution changes skew even slightly, the histogram would become increasingly less accurate, depending on how significant a change. Even in the same distribution, depending on how many values the histogram stores and covers in the overall range, shifts could be expected – still leading to an increase in inaccuracy. As a result, to be able to answer queries with higher accuracy, histograms need to be updated in accordance with change in streaming data. This is not a big problem for equiWidth since the bucket boundaries stay the same and only frequencies change. However, this is very problematic for v-optimal histograms since the new data could render the existing histogram boundaries incorrect as far as minimizing expected intra-bucket variance is concerned.

In an attempt to control for such a force, this paper explores a bucket tweaking method. To combat the complications that dynamic streaming data presents to histogram algorithms, some sort of mitigation method must be installed to keep the histogram up to date so to speak. The histogram algorithms are capable of handling static data, but cannot compensate themselves for data growing stale and driving inaccuracy up, as occurs with dynamic data. Since this shifting effect appears to affect the specialized partition scheme algorithms the most, it would be prudent to attack the problem at its core: the buckets. Instead of moving the buckets around in a desperate attempt to account for an entire distribution shift, consider smaller, incremental changes designed to be an agile response to dynamic activity in the simulated data stream. By using the sliding window approach previously discussed and already proven to be an effective way to manage data streams, it is possible to determine what data is becoming stale, and what data is fresh. In combination of the agile concept proposed and the sliding window approach, the following adaptive method is used to keep the histogram as up to date as reasonably possible. For every data point that leaves the sliding window, decrement the corresponding bucket in the histogram by one. Conversely, for every data point entering the sliding window, increment the corresponding bucket by one. In these experiments, a single unit of data expires as a single unit of data becomes relevant. That is an inherent property of a fixed-size sliding window, however, it would be possible to change the size of the stream sample if the situation deems necessary.

4 Experimental setup

Two different pseudo-random distributions are used in this paper to avoid single distribution bias – Uniform and Gaussian. Similar to the problem of distribution bias is the issue of histogram density. To be precise, density in the scope of this research refers to the ratio between the number of values in the histogram as it relates to the range the histogram attempts to model. For example, a histogram summarizing 100 distinct values with 150 total occurrences has density 1.5. To address the density bias, four different histogram densities given below were tested.

Name	Density
Sparse	1
Balanced	1.5
Dense	2
Very Dense	2.5

4.1 Performance metric

For each combination of input characteristics, we use 0 – 1000 as the value range. Once the data arrays are filled, queries are posed after each new data addition. Altogether 10000 queries are executed and their error measured. At the end of each trial, the normalized errors obtained from equation 2 are summed, and divided by the number of queries processed to determine an average normalized error per query.

5 Results

This section discusses all the simulation results and presents several charts. The x-axis of each chart shows the number of buckets as a percentage of the size of the zero-based range, and the y-axis shows the measure of inaccuracy determined by the normalized difference per query. Each chart title and legend clearly dictates what simulation data is presented. These charts are separated based on pseudorandom distribution and histogram algorithm. Each chart shows the performance of the histogram algorithm as it applies to all four density-ratios and three types of source data and histogram pairings – static data with static histograms (*static* scenario), dynamic data with static histograms (*dynamic* scenario) and dynamic data with adaptive histograms (*adaptive* scenario). The density-ratios are color-coordinated. The source data and histogram pairings are coordinated on the graph by line-type, such that: static scenario simulations are represented by a solid line, dynamic scenario simulations are shown with a dashed line, and the *adaptive* scenario simulations are shown as a dotted line. The static scenario represents the baseline as the results cannot get any better than this. The purpose of testing the dynamic scenario is two-fold. One, it determines the feasibility of using a static histogram in a data stream management system. If the skew of a distribution in data stream suddenly morphs, how does that affect the histogram? Two, as alluded to with the prior question, if and how does the accuracy of the histogram change as the data stream progresses with new information processed by the sliding window? Finally, the adaptive scenario is used to test the effectiveness of our adaptive histogram method in estimating answers to frequency based queries.

5.1 EquiWidth on Uniform Data

This distribution is usually better suited for analysis by histograms because the skew is very low and consistently even, regardless of the point in question within the distribution chart. The results shown in Figure 3 are as expected, from the static showing less inaccuracy with more buckets, to the dynamic data losing accuracy with more buckets, through to the adaptive method following the same pattern as the static data. The static data here is very consistent in accuracy from the sparse density-ratio up through the very dense density-ratio. As previously mentioned, the dynamic data does increase in inaccuracy, which again proves the point that histograms alone are ill suited to model streaming data. Combined with the fact that they become less accurate with more buckets, this means that there must be a different solution. In this case, the solution again is the adaptive method. With the EquiWidth histogram, queries were nearly just as accurate as the static data in the lower number of buckets scenarios, and proceeded to follow the static data very closely in accuracy. This strongly suggests that the adaptive method works well in these categories.

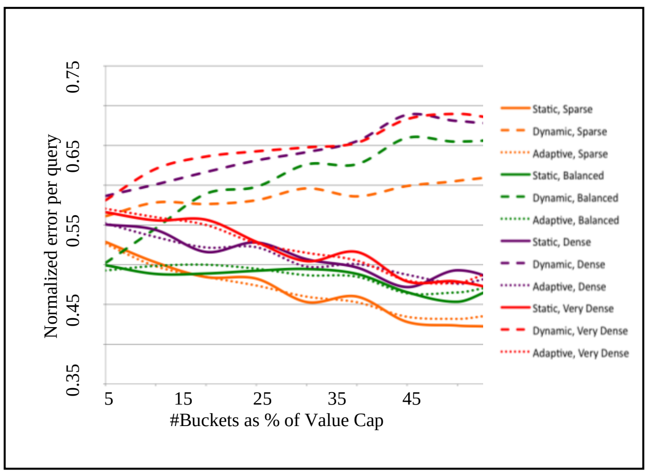

Figure 1: Equiwidth error on Uniform data

5.2 V-Opt on Uniform Data

Proceeding onward from the equivalent width analysis, we now study the results of the variance optimal algorithm processing uniform pseudorandom data. The chart is shown in Figure 4. Again, this is excellent result. The graph shows the decreasing slope for static data, showing that the more buckets the variance optimal histogram has at its disposal, the more accurate the results it can return.

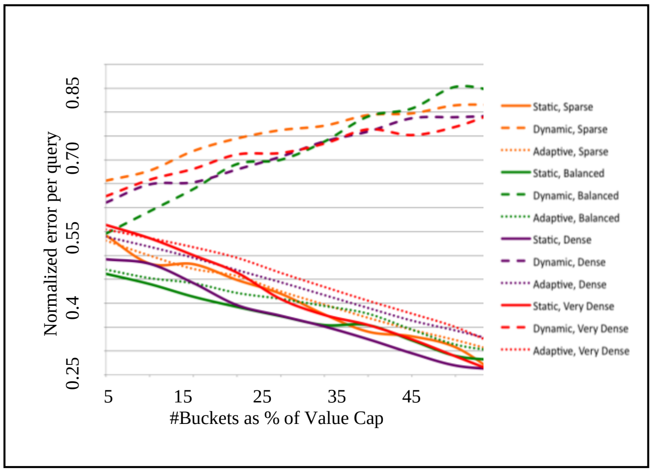

Figure 2: V-Optimal error on Uniform data

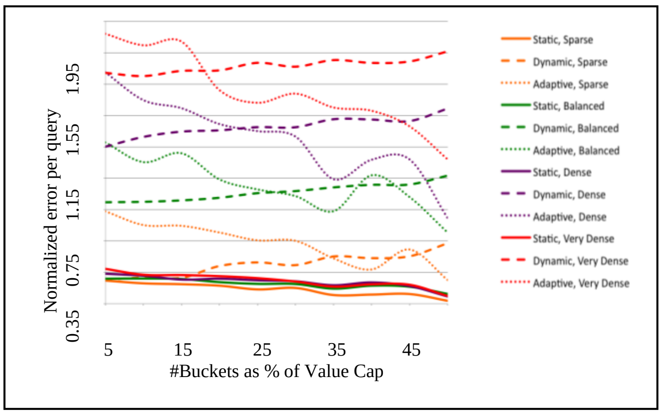

Figure 5: EquiWidth error on Gaussian data

Variance optimal histogram test data shows consistency in accuracy when the density-ratio is at least 1:1. The dynamic data lines tell a story. As more buckets are introduced in a dynamic setting, the histogram gets less accurate rapidly and consistently. This shows that the variance optimal algorithm is sensitive and susceptible to changes in its underlying data. Due to the nature of the algorithm basing its partitioning scheme on minimizing variance, it is of no surprise that it loses accuracy as rapidly as it does in these stressful scenarios. Conversely, the adaptive method again shows excellent mitigation of error creep introduced by dynamic data. The slope nearly matches the static data slope. Also, the level of accuracy provided by the adaptive method nearly matches the level of accuracy that the static data typically returns. At peak effectiveness, at the 50% number of buckets mark, the difference in accuracy is around 0.05 units. By contrast, the dynamic data at the same point was ten times less accurate.

5.3 EquiWidth on Gaussian Data

Gaussian pseudorandom data provides a different challenge, as it is a more erratic distribution with regard to frequency. This poses a complication to histograms, because the skew is both greater than a relatively flat line and changing depending on the point in question within the distribution chart.

The chart displaying the results of this simulation is at the end of this subsection, shown in Figure 5. Proceeding to the analysis, the vast majority of the results are as expected. The static lines are consistent and at a downward slope – confirming that the more buckets, the more accurate. They are also closely clustered, suggesting that density-ratio does not have a great effect on inaccuracy. Regardless of how much data is in the histogram, the appearance is that the query will be approximately a half point away in accuracy, or fifty percent of the actual frequency. The dynamic data lines are ascending in slope, which is also expected. Bear in mind that over the same sized range, the larger the number of buckets, the smaller range size they individually cover normally. This smaller range size on one hand does make them better representations of the distribution curve, but also makes them more sensitive to changes in underlying data. A larger bucket may not be more accurate, but it will not lose accuracy as quickly either. Additionally, the dynamic data scenario is affected by the density ratio. The more data packed into the histogram, the less accurate. A balanced histogram is already off by one full point in estimation in the five percent bucket situation, and nearly reaches two full points away from the true value in the very dense scenario with fifty percent. Bearing in mind that each point represents a multiplicative factor applied to the actual frequency value, inaccuracy rising to an increased error of 200% is problematic. The adaptive lines show a lot of promise here. As they have more buckets to work with, they return greater accuracy. This is exactly the opposite of what occurs with dynamic data, and precisely what the goal of this research sought out to determine. Data density does appear to have an effect as well, such that the denser the histogram, the sooner the adaptive benefits take place, with respect to an ascending number of buckets. All of these effects combine to solve many of the problems dynamic data present, and that is with the more difficult pseudorandom distribution to model in histograms. At peak operating efficiency, the data charts suggest that it is possible to re-gain over a half point of accuracy in the dense and very dense scenarios. This does not adversely affect the attempt to devise a generally acceptable method of mitigating dynamic error creep, due to the fact that generalizations are not universally perfect. Although this does make the adaptive method appear to be more of a situational fix at first glance, given enough situations where the adaptive method presents benefits, this can be utilized as a generalized solution to counteract the negative effects of dynamic data on histograms.

5.4 V-Opt on Gaussian Data

Moving forward, the next algorithm for analysis is the variance optimal algorithm using Gaussian data. The chart displaying the results of this simulation is shown in Figure 6. As previously exposed with the equivalent-width algorithm, the expected results of the static data increasing in accuracy as more buckets are provided is expected. Also expected, is the overall decrease of accuracy when running on dynamic data, and the increase in inaccuracy with more buckets on dynamic data. The dynamic data can result in as much as two points of divergence from the true frequency count, as shown in the fifty percent case with a density-ratio of very dense.

The adaptive technology, on the other hand, is really starting to shine. As seen with the equivalent width algorithm, the adaptive method works better with denser histograms. In two categories, balanced and dense data density ratios, the adaptive method was completely more accurate than the dynamic scenario. The very dense scenario was more accurate starting with the ten percent bucket point, forward to all counts of buckets higher. The overall downward sloping dotted line in all scenarios is very encouraging for the adaptive method to mitigate the inaccuracy complications that dynamic data can present.

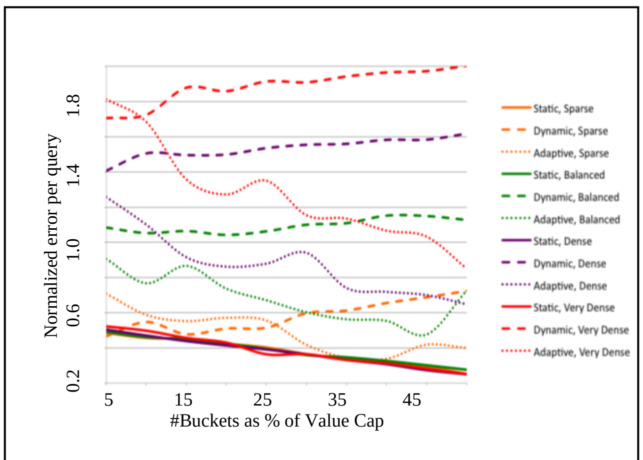

Figure 6: V-Optimal error on Gaussian data

6 Future Applications

The future applications for research exist as an extension of what was done in this paper. Iteratively speaking, more algorithms could be tested with more random distributions – which could yield either more evidence confirming these results, or information pointing to the contrary. Similarly, another way to extend the research presented in this paper could be to develop other methods of keeping histograms fresh. Perhaps it is possible to dynamically alter the bucket boundaries to reflect a best-guess real-time estimation for the summary of a dynamic data stream. Following the same logic, it could also be possible to trigger a full histogram reconstruction should the level of inaccuracy exceed a specified error threshold. These would certainly benefit the field.

Further ahead, it is possible that one could start testing these concepts against higher-dimensional histograms, determining if the results found here extend to more complicated algorithms. Common sense is not factual until empirically proven in the eyes of the scientific community, and as such, results could be proven contrary to common sense. This research was also limited to the scope of single selection count queries based on frequency. It would not take much to extend this research to range queries and join queries.

7 Related Work

Amongst many researchers in the field studying histograms, one in particular took the time to compose a comprehensive study of the history of histograms: Yannis Ioannidis of the University of Athens. In his paper entitled, *"The History of Histograms (abridged),"* Again jumping in time, now to the mid-1990s, Yannis Ioannidis writes another paper, now with Viswanath Poosala at the University of Wisconsin, *"Balancing Histogram Optimality and Practicality for Query Result Size Estimation"* (Ioannidis & Poosala, 1995). Jagadish et. al. developed a dynamic programming method of calculating the bucket boundaries in a variance optimal histogram in the paper *"Optimal Histograms With Quality Guarantees."* By using an inherent property of the variance optimal histograms, the sum-squared error (SSE) is determined to be possible to calculate in an iterative fashion. This ability, in turn, saves repeated and redundantly wasteful calculations when intermediate results are stored in an internal matrix. Later Yannis Ioannidis and Viswanath Poosala delve into answering queries with an approximation while improving database response time (Ioannidis Y. E., 1999), in *"Histogram-Based Approximations of Set-Valued Query Answers"*. At that time, database responses were slow to return information – even when precision was not necessary. By using histograms, they were able to generate approximations for early reports in on-line database queries when the precise answer was not absolutely critical, but response time was. With regard to utilizing sliding window technology, Arvind Arasu and Gurmeet Singh Manku pioneered the efforts of maintaining approximations and quantiles over both fixed and variable-sized sliding windows processing streaming data (Arasu & Manku, 2004). Gurmeet Singh Manku and Rajeev Motwani devised two methods of approximating frequently occurring singleton frequency counts over data streams using sliding windows with small memory footprints and output error limited to user-specification (Manku & Motwani, 2002). In other related work, sliding window technology has been used in conjunction with other algorithms designed to count and maintain aggregate data in dynamic data streams with provable low-memory (Datar, Gionis, Indyk, & Motwani, 2002).

8 Conclusions

While histograms inherently are not designed, nor are capable of by default, to handle dynamic data sources, they can be adjusted with certain methods of error-minimization to compensate for what otherwise could be considered cumulative error. This paper has empirically shown that when using a sliding window approach for using the histogram over a dynamic data stream, the histograms grow stale and less accurate. The data analyzing partition scheme based histogram algorithms suffer the most, which is problematic because they are also the most accurate under static data. In order to make them work on dynamic data, their buckets must be modified in such a manner to keep them fresh.

Overall the adaptive method is a strong success. In the Equivalent Width histogram tests using Gaussian pseudorandom data, the adaptive histogram process was an okay performer. It did increase accuracy over the dynamic data on static histograms scenarios with more buckets or more dense data. In overlapping cases, performance was even better. Specifically, there was a 20% reduction of error at the 25% number of buckets ratio when the histogram was very densely packed. The Variance Optimal histogram on Gaussian data benefitted strongly from the adaptive method. In the balanced, dense, and very dense ratios, inaccuracy decreased between 40% and 70% from the dynamic data estimates across the spectrum of number of buckets, and above 15% number of buckets ratio for very dense data. For EquiWidth histograms operating on uniformly distributed data, the adaptive method proved to be an excellent performer. The accuracy rivaled static data results, within 2% throughout all density ratios and number of bucket ratios. Additionally, VOpt histograms with uniformly distributed data maintained a level of accuracy within 10% of all of the static data tests, throughout all number of bucket ratios and data density ratios. In summary, the adaptive method showed okay to excellent results, depending on pseudorandom distribution. Although not as strong on Gaussian data, the adaptation drastically improved accuracy when histograms were processing uniformly distributed data. There was an overall trend of a greater decrease in inaccuracy with more buckets.

This paper has shown that an agile method using minimal incremental changes can aid in maintaining the accuracy of the histograms. In conclusion, this method is a strong candidate for error mitigation in histograms applied to dynamic data sources for a few reasons. It is inexpensive to maintain, as two high level seek operations and two basic operations at most are required per update: seeking the corresponding bucket in a histogram for a given value (twice if inserting and removing), and incrementing or decrementing that value as necessary. Those operations are not processor or memory intensive tasks. Finally, the decrease in inaccuracy over a broad range of tests suggests the potential for true generalization.

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