AFIN 2011

The Third International Conference on Advances in Future Internet

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AFIN 2011

Foreword

The Third International Conference on Advances in Future Internet (AFIN 2011), held between August 21-27, 2011 in Nice/Saint Laurent du Var, France, continued a series of events dealing with advances on future Internet mechanisms and services.

We are in the early stage of a revolution on what we call Internet now. Most of the design principles and deployments, as well as originally intended services, reached some technical limits and we can see a tremendous effort to correct this. Routing must be more intelligent, with quality of service consideration and 'on-demand' flavor, while the access control schemes should allow multiple technologies yet guarantying the privacy and integrity of the data. In a heavily distributed network resources, handling asset and resource for distributing computing (autonomic, cloud, on-demand) and addressing management in the next IPv6/IPv4 mixed networks require special effort for designers, equipment vendors, developers, and service providers.

The diversity of the Internet-based offered services requires a fair handling of transactions for financial applications, scalability for smart homes and ehealth/telemedicine, openness for web-based services, and protection of the private life. Different services have been developed and are going to grow based on future Internet mechanisms. Identifying the key issues and major challenges, as well as the potential solutions and the current results paves the way for future research.

We take here the opportunity to warmly thank all the members of the AFIN 2011 technical program committee as well as the numerous reviewers. The creation of such a broad and high quality conference program would not have been possible without their involvement. We also kindly thank all the authors that dedicated much of their time and efforts to contribute to the AFIN 2011. We truly believe that thanks to all these efforts, the final conference program consists of top quality contributions.

This event could also not have been a reality without the support of many individuals, organizations and sponsors. We also gratefully thank the members of the AFIN 2011 organizing committee for their help in handling the logistics and for their work that is making this professional meeting a success.

We hope the AFIN 2011 was a successful international forum for the exchange of ideas and results between academia and industry and to promote further progress in the area of future Internet.

We hope Côte d'Azur provided a pleasant environment during the conference and everyone saved some time for exploring the Mediterranean Coast.

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Extended Labeling Method for Achieving Maximum Data Message Flow in Wireless Multihop Networks

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***Abstract*—For multimedia data message transmissions in wireless multihop networks composed of wireless links with lower throughput, transmission capacity is reserved in wireless links included in one of wireless multihop transmission routes from a source node N^s to a destination node N^d. For reservation of transmission capacity required for applications, a method for achieving the maximum data message flow from N^s to N^d is required. For wired multihop networks, the labeling method has been proposed. However, in wireless multihop networks, since message transmission between neighbor nodes is realized by broadcast transmission of wireless signal, capacity of a wireless link is effected by data message transmissions by neighbor nodes due to the hidden-terminal and exposed-terminal problems. Thus, this paper proposes an extended labeling method based on a novel wireless network model for the maximum data message flow in wireless multihop networks and according to novel conditions for a wireless multihop transmission route to increase the amount of data message flow from N^s to N^d.**

***Keywords*- Ad Hoc Networks; Multimedia Communication; Throughput; Resource Reservation; Labeling Method.**

I. INTRODUCTION

In ad hoc networks, sensor networks and mesh networks, data messages are transmitted from a source wireless node to a destination one by wireless multihop transmission. In case that the destination node is not included in a wireless transmission range of the source one, intermediate wireless nodes in a wireless multihop transmission route forward the date messages. In order to realize multimedia communication which requires realtime transmission of vast number of data messages in such wireless multihop networks, reservation of transmission capacity in wireless links in the route in advance is required.

RSVP [1] is an internet-standard protocol for reservation of capacities in communication links along a transmission route from a source node to a destination one (Figure 1). In RSVP, it is assumed that available capacities in communication links are enough for requirements in network applications and capacity in each communication link can be reserved independently of the other communication links. That is, even when a certain amount of capacity in a communication link is reserved, available capacities in the other communication links are never reduced. However, in wireless multihop networks, available capacities in wireless links are not always enough for requirements in network applications. In addition, capacity reservation in a wireless link is not independent of available capacities of wireless links issued from the neighbor wireless nodes since wireless communication is intrinsically based on broadcast communication and there may be hidden- and/or exposed-terminals [4]. Therefore, it is difficult for multimedia network applications to be provided enough amount of data message flow by reservation of capacities in wireless links along a single wireless multihop transmission route.

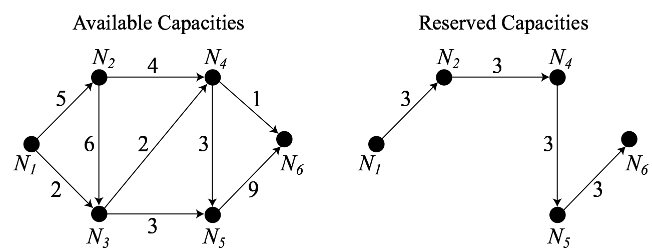

Fig. 1. Multihop Transmission along Single Route.

The authors have been proposed RSVMRD [7] link capacity reservation protocol in wired networks which satisfies required throughput of data messages for network applications by data message transmissions along multiple wireless multihop transmission routes from a source node to a destination one as shown in Figure 2. RSVMRD consists of the following 2-step algorithms. In the first step, the maximum amount of data message flows, i.e., the maximum available throughput of data messages, from the source node to the destination one along multiple transmission routes are calculated by using a well-known heuristic, the labeling method [2], modified for reduction of its computational complexity. Here, required capacities in communication links which are surely less than their available capacities to realize the maximum data message flows are also calculated. In the second step, based on the required capacities, capacities to be reserved to satisfy the requirements of network applications are finally induced and are reserved in nodes from which the communication links are issued by exchanging control messages. The authors propose the same approach to reservation of capacities in wireless links to satisfy requirements in multimedia network applications in wireless multihop networks. Since capacity reservation in a wireless link reduces available capacities in neighbor wireless links due to broadcast property in wireless networks, this paper discusses an extended labeling method to calculate the maximum data message flows in wireless multihop networks.

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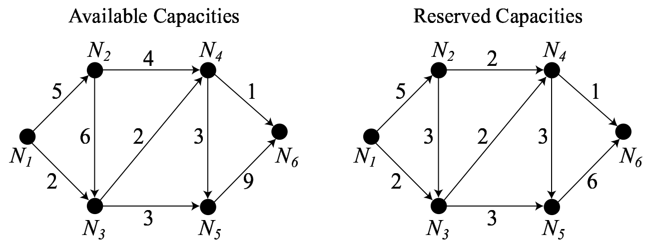

Fig. 2. Multihop Transmission along Multiple Routes.

II. Related Works

A. Wireless Multihop Communication

Let $\mathcal{NT} := \langle \mathcal{N}, \mathcal{L} \rangle$ be a wireless multihop network with a set $\mathcal{N} := \{N_i\}$ of wireless nodes N_i and a set $\mathcal{L} := \{\langle N_i N_j \rangle\}$ of wireless links $\langle N_i N_j \rangle$ between wireless nodes N_i and N_j. Wireless transmission ranges of all the wireless nodes are assumed to be equal and all wireless links are assumed to be bidirectional. Each data message is transmitted from its source wireless node N^s $(= N_0)$ to its destination one N^d $(= N_n)$ along a wireless multihop transmission route $R := ||N_0 \ldots N_n\rangle\rangle$ as in Figure 3. Each intermediate wireless node N_i $(i = 1, \ldots, n-1)$ in R forwards data messages. N_{i+1} is included in a wireless transmission range of N_i[1]. Thus, $N_{i+1} \in Nei(N_i)$ where $Nei(N)$ is a set of neighbor wireless nodes of $N \in \mathcal{N}$, i.e., a set of wireless nodes included in a wireless transmission range of N.

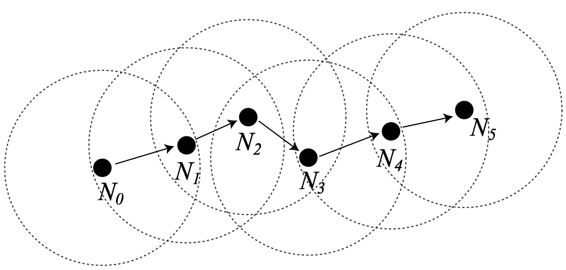

Fig. 3. Wireless Multihop Transmission of Data Messages.

Now consider forwarding of a data message from a transmitter wireless node N^t to a receiver one $N^r \in Nei(N^t)$ (Figure 4). Due to broadcast property of wireless communication, all data messages transmitted from N^t to N^r are also received by all its neighbor wireless nodes $N \in Nei(N^t)$. Thus, N is called an exposed-node of N^t. In most of wireless LAN protocols such as IEEE 802.11, a wireless node cannot transmit data messages simultaneously with its exposed-nodes for collision avoidance. The exposed-terminal problem is that neighbor wireless nodes which are exposed nodes each other but do not cause collisions are restricted to transmit data messages simultaneously. On the other hand, another type of collisions may occur when N^t and $N' \in Nei(N^r) - Nei(N^t)$ transmit data messages simultaneously. Since N^t and N' are not exposed nodes, they cannot detect their data message transmissions each other and collisions occur at N^r. Thus, N^t and N are hidden nodes each other and collisions between hidden nodes are called the hidden-terminal problem. As discussed, in wireless multihop communication, data message transmissions from an intermediate wireless node N_i to its next-hop wireless node N_{i+1} affects on data message transmissions from exposed and hidden nodes of N_i. That is, data message transmissions through wireless links are dependent one another, which is differrent from wired networks.

[1] N_{i-1} is also included in a wireless transmission range of N_i since all wireless links are bidirectional in this paper.

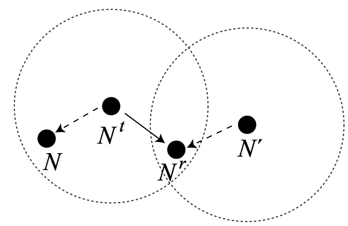

Fig. 4. Effects on Neighbor Nodes (Exposed and Hidden Nodes).

B. Labeling Method

Let $\mathcal{NT}' := \langle \mathcal{N}', \mathcal{L}' \rangle$ be a wired network where $\mathcal{N}' := \{N_i\}$ is a set of nodes and $\mathcal{L} := \{|N_i N_j\rangle\}$ is a set of wired links from a node N_i to its neighbor node N_j. In addition, for each wired link $|N_i N_j\rangle$, an available capacity $c(|N_i N_j\rangle) \geq 0$ is given and a potion of the available capacity is reserved for provision of required throughput to multimedia network applications independently of the other wired links. The labeling method [2] calculates the maximum amount of data message flows, i.e., the maximum data message throughput, from a source node N^s to a destination one N^d by using multiple multihop transmission routes. It also induces capacities to be reserved in wired links in one of the multihop transmission routes. The problem to achieve the maximum amount of data message flows is formalized as follows where a capacity to be reserved in a wired link $|N_i N_j\rangle$ is $r(|N_i N_j\rangle)$:

[Maximum Data Message Flows Problem]
Under the restrictions (1) $r(|N_i N_j\rangle) \leq c(|N_i N_j\rangle)$ in all wireless links $|N_i N_j\rangle \in \mathcal{L}'$ and (2) $\sum_{|N_k N_i\rangle \in \mathcal{L}'} r(|N_k N_i\rangle) = \sum_{|N_i N_j\rangle \in \mathcal{L}'} r(|N_i N_j\rangle)$ in all nodes $N_i \in \mathcal{N}' - \{N^s, N^d\}$, the maximum value of $\sum_{|N^s N_j\rangle \in \mathcal{L}'} r(|N^s N_j\rangle) = \sum_{|N_k N^d\rangle \in \mathcal{L}'} r(|N_k N^d\rangle)$ is calculated. □

In the labeling method, a multihop transmission route from a source node N^s to a destination one N^d which increases amount of data messages, i.e., data message throughput, one by one. Here, the maximum data message flow along a multihop transmission route is determined as the minimum available capacity of communication links along the route. If the maximum data message flow is greater than 0, the detected multihop transmission route increases total amount of data messages transmitted from N^s to N^d and it is called a flow increasing route. In the labeling method, flow increasing routes are detected one by one with a procedure to reduce the available capacities in communication links included in the

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route each time it is detected until no other flow increasing routes are detected. Many research results show that the labeling method is a better heuristic to achieve the pseudo maximum data message throughput and the required capacities to be reserved in the communication links in wired networks.

Figure 5 shows a naive example wired network $\mathcal{NT}' := \langle \mathcal{N}', \mathcal{L}' \rangle$ where $\mathcal{N}' := \{N^s, N_1, N_2, N^d\}$ and $\mathcal{L}' := \{|N^sN_1\rangle, |N^sN_2\rangle, |N_1N_2\rangle, |N_1N^d\rangle, |N_2N^d\rangle\}$. Suppose that available capacities are given as follows; $c(|N^sN_1\rangle) := 10$, $c(|N^sN_2\rangle) := 5$, $c(|N_1N_2\rangle) := 8$, $c(|N_1N^d\rangle) := 4$ and $c(|N_2N^d\rangle\}) := 8$.

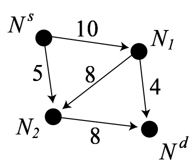

Fig. 5. Example Wired Network and Available Link Capacities.

As shown in Figure 6, at first, a flow increasing route $||N^sN_1N^d\rangle\rangle$ is detected and its maximum amount of data message flow is calculated as 4 since the available capacity of $c(|N_1N^d\rangle) = 4$ is the minimum in communication links along the route. Thus, capacity 4 is reserved in each link along the route and the available capacities are updated as follows; $c(|N^sN_1\rangle) := 6$ and $c(|N_1N^d\rangle) := 0$. Then, another flow increasing route $||N^sN_2N^d\rangle\rangle$ is detected. Since the maximum data message flow along the route is 5 ($= c(|N^sN_2\rangle)$), the total amount of reserved data message flows gets 9 and the available capacities are updated as follows; $c(|N^sN_2\rangle) := 0$ and $c(|N_2N^d\rangle) := 3$. Finally, a flow increasing route $||N^sN_1N_2N^d\rangle\rangle$ whose maximum data message flow is 3 ($= c(|N_2N^d\rangle)$) is detected and available capacities $c(|N^sN_1\rangle) := 3$, $c(|N_1N_2\rangle) := 5$ and $c(|N_2N^d\rangle) := 0$ are updated. As a result, there are no flow increasing routes and the total amount of data message flow 12 (=4+5+3) is achieved.

The order of multihop route detections from N^s to N^d depends on the route detection protocol. Hence, the multihop routes may be detected in different order. For example as shown in Figure 7, at first, a flow increasing route $||N^sN_1N_2N^d\rangle\rangle$ is detected. Its maximum amount of data message flow along the route is 8 ($= c(|N_1N_2\rangle) = c(|N_2N^d\rangle)$) and the available capacities in communication links along the route are updated differently from the previous example; $c(|N^sN_1\rangle) := 2$, $c(|N_1N_2\rangle) := 0$ and $c(|N_2N^d\rangle) := 0$. Then, another flow increasing route $||N^sN_1N^d\rangle\rangle$ whose maximum data message flow is 2 ($= c(|N^sN_1\rangle)$) is detected and the available capacities $c(|N^sN_1\rangle) := 0$ and $c(|N_1N^d\rangle) := 2$ are updated. Now, there are no multihop transmission route from N^s to N^d consisting of communication links whose available capacities are greater than 0. However, a multihop route $||N^sN_2N_1N^d\rangle\rangle$ is also a flow increasing route. This is because both $c(|N^sN_2\rangle) = 5$ and $c(|N_1N^d\rangle) = 2$ are greater than 0 and reduction of reserved capacity $r(|N_1N_2\rangle) = 8$ in a communication link $|N_1N_2\rangle$ is equivalent to reserve

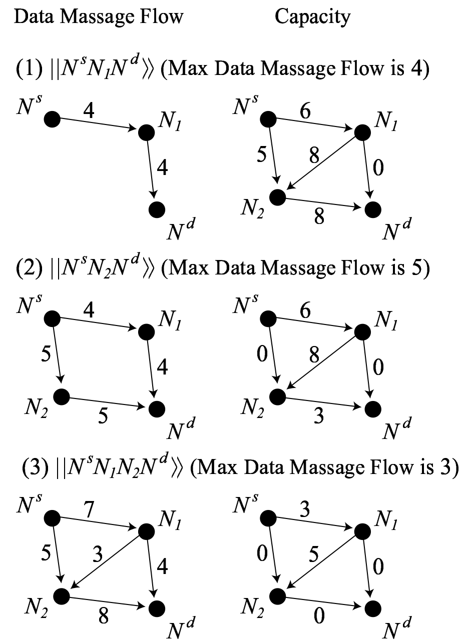

Fig. 6. The Original Labeling Method Example (1).

the same amount of flow in a reverse communication link $|N_2N_1\rangle$. Since the maximum amount of data message flow in the reverse communication link is $r(|N_1N_2\rangle) = 8$, the maximum amount of data message flow along $||N^sN_2N_1N^d\rangle\rangle$ is 2 ($= c(|N_1N^d\rangle)$) and the available capacities are updated as follows; $c(|N^sN_2\rangle) := 3$, $c(|N_1N_2\rangle) = 2$ and $c(|N_1N^d\rangle) := 0$. Now, there are no flow increasing route from N^s to N^d and the maximum amount of data message flow 12 (=8+2+2) is achieved.

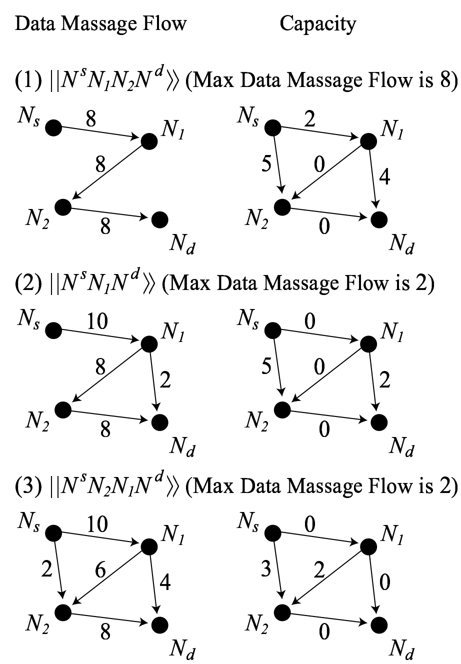

Fig. 7. The Original Labeling Method Example (2).

As a result of examination of the above 2 examples, there are 2 types of flow increasing routes and the finally achieved maximum amounts of data message flows are the same though reserved capacities in communication links are different.

[Flow Increasing Route in the Original Labeling Method]
A multihop transmission route $R = ||N_0 \ldots N_n\rangle\rangle$ satisfying one of the following conditions is a flow increasing route;

- $c(|N_i N_{i+1}\rangle) > 0$ in all communication links $|N_i N_{i+1}\rangle \in R$ (trivial flow increasing route)
- $c(|N_i N_{i+1}\rangle) > 0$ or $r(|N_{i+1} N_i\rangle) > 0$ in all communication links $|N_i N_{i+1}\rangle \in R$ (flow increasing route with reduction of already reserved capacities) □

C. Multiple Route Wireless Multihop Transmissions

As a result of the labeling method, multiple multihop transmission routes are detected which provide the maximum throughput of data messages from a source node to a destination one. Until now, venous ad hoc routing protocols for detection of multiple wireless multihop transmission routes have been proposed [3,5,6]. However, most of them are designed for continuous data message transmissions even with wireless link breakages and node failures and detect link- or node-disjoint routes. [8] and some papers propose that data messages are transmitted along detected multiple routes simultaneously for higher data message throughput. However, these protocols do not intentionally detect multiple routes to achieve higher throughput for multimedia data transmission. Of course, they do not provide the maximum throughput of data messages with consideration of the exposed and hidden terminal problems.

III. PROPOSAL

A. Wireless Network Model

In the original labeling method, reservation of capacity in a wired link $|N_i N_j\rangle$ does not affect the available capacity of wireless links other than $|N_i N_j\rangle$ itself. That is, available capacity and reserved capacity in a wired link is independent of those of the other wired links. Thus, the maximum amount of data message flows are calculated based on the available capacities in wired links in a wired network. On the other hand in a wireless multihop network, since wireless communication is intrinsically based on broadcast transmission, even if a wireless node N_i transmits data messages to its neighbor wireless node N_j, the data messages are also transmitted to all its neighbor wireless nodes $N \in Nei(N_i)$ in its wireless transmission range. During transmission of the data messages, N can neither transmit nor receive data messages. Hence, data message transmission through $|N_i N_j\rangle$ reduces available capacities of not only $\langle N_i N_j\rangle$ but also all wireless links $\langle N_i N\rangle$ (Figure 8). Therefore, in wireless multihop networks, available capacity should be assigned not to wireless links but to wireless nodes[2].

Suppose wireless nodes N_i and N_j are neighbor, i.e., they are included in their wireless transmission range each other.

[2]Capacities are reserved for wireless links to transmit data messages.

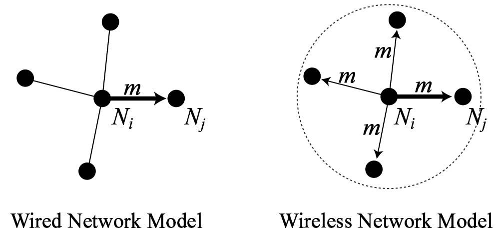

Fig. 8. Wired and Wireless Network Model.

Reservation of capacity in a wireless link $|N_i N_j\rangle$ reduces the capacity of a wireless node N_i. In addition, since N_i cannot transmit and receive data messages simultaneously, reservation of capacity in a wireless link $|N_j N_k\rangle$ where N_k is a neighbor wireless node of N_j also reduces the capacity of N_i. In addition, as shown in Figure 9, since N_i is an exposed node for data message transmission from its neighbor wireless node N to N^t, reservation of capacity in a wireless link $|NN^t\rangle$ reduces the available capacity of N_i. Due to the same reason, since N_i is a hidden node for data message transmission from N^f to N as in Figure 9, reservation of capacity in a wireless link $|N^f N\rangle$ also reduces the available capacity of N_i. In accordance with the above examination, the problem to achieve the maximum data message flows in wireless multihop networks is formalized as follows where available capacity $c(N_i)$ is defined for a wireless node N_i, reserved capacity $|N_i N_j\rangle$ is determined for a wireless link $|N_i N_j\rangle$ and a source and a destination wireless nodes are N^s and N^d, respectively;

[Maximum Data Message Flow Problem in Wireless Multihop Networks]
Under the following restrictions, the maximum value of $\sum_{|N^s N_j\rangle \in \mathcal{L}'} r(|N^s N_j\rangle) = \sum_{|N_k N^d\rangle \in \mathcal{L}'} r(|N_k N^d\rangle)$ is calculated;
(1) For all $|NN'\rangle$ satisfying $N \in Nei(N_i) \cup \{N_i\}$ or $N' \in Nei(N_i) \cup \{N_i\}$, $\sum_{|NN'\rangle} r(|NN'\rangle) \le c(N_i)$.
(2) For all $N_i \in \mathcal{N}' - \{N^s, N^d\}$, $\sum_{|N_k N_i\rangle \in \mathcal{L}'} r(|N_k N_i\rangle) = \sum_{|N_i N_j\rangle \in \mathcal{L}'} r(|N_i N_j\rangle)$. □

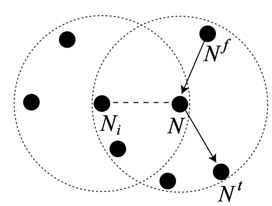

Fig. 9. Restrictions on Reservation of Capacity in Wireless Networks.

B. Capacity Increasing Flows

In the labeling method, multihop transmission routes from a source node to a destination one which increase amount of transmitted data messages, i.e., throughput, are detected one by one. Here, the conditions which the routes should satisfy are critical. In order to extend the original labeling method to be applied to wireless multihop networks, this

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subsection discusses the conditions where a wireless multihop transmission route R provides a capacity increasing flow from a source wireless node N^s to a destination one N^d.

At first, we examine how capacities $c(N)$ of wireless nodes N are updated when an amount r of data message flow is reserved along a wireless multihop transmission route R. The initial values of capacities $c(N_i)$ of wireless nodes N_i is determined by the specification of wireless network interfaces. The current available capacities $c(N_i)$ that have not yet reserved for any data message transmission flow restricts the amount of data message flows including N_i and its 1-hop and 2-hop neighbor wireless nodes due to exposed and hidden nodes relation. Thus, we examine the following 3 cases where N is included in R and a case where N is not included in R.

[$N = N_0 \in R$ or $N = N_n \in R$]

As shown in Figure 10(a), if N is a source wireless node $N_0 = N^s$, since $N_1 \in Nei(N)$, the amount of reduction of $c(N)$ is totally reserved capacities in wireless links $|NN_1\rangle$ and $|N_1N_2\rangle$. In the same way, as shown in Figure 10(b), if N is a destination wireless node $N_n = n^d$, since $N_{n-1} \in Nei(N)$, the amount of reduction of $c(N)$ is totally reserved capacities in wireless links $|N_{n-2}N_{n-1}\rangle$ and $|N_{n-1}N_n\rangle$. Therefore, $c(N) := c(N) - 2r$.

[$N = N_1 \in R$ or $N_{n-1} \in R$]

As shown in Figure 10(a), if N is a next-hop wireless node N_1 of N^s, since $N_0 \in Nei(N)$ and $N_2 \in Nei(N)$, $c(N)$ is reduced the total of reserved capacities in wireless links $|N_0N\rangle$, $|NN_2\rangle$ and $|N_2N_3\rangle$ along R. In the same way, as shown in Figure 10(b), if N is a previous-hop wireless node N_{n-1} of N^d, since $N_{n-2} \in Nei(N)$ and $N_n \in Nei(N)$, $c(N)$ is reduced the total of reserved capacities in wireless links $|N_{n-3}N_{n-2}\rangle$, $|N_{n-2}N\rangle$ and $|NN_n\rangle$ along R. Therefore, $c(N) := c(N) - 3r$.

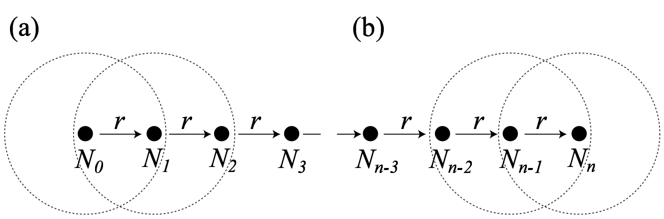

Fig. 10. Update of Capacities of Wireless Nodes (1).

[$N = N_i \in R$]

As shown in Figure 11, if N is an intermediate wireless node N_i in R where $i \neq 0, 1, n-1, n$, since $N_{i-1} \in Nei(N)$ and $N_{n+1} \in Nei(N)$, $c(N)$ is updated as reduction of the total amount of reserved capacities in wireless links $|N_{i-2}N_{i-1}\rangle$, $|N_{i-1}N\rangle$, $|NN_{i+1}\rangle$ and $|N_{i+1}N_{i+2}\rangle$ along R. Therefore, $c(N) := c(N) - 4r$.

[$N \notin R$ and $Nei(N) \cap R \neq \emptyset$]

For avoidance of collisions with exposed and hidden nodes in R, though N is not included in R, $c(N)$ is reduced. The amount of reductions is total reservation capacities in wireless links whose transmitter or receiver nodes are included in a wireless range of N as shown in Figure 12. That is, $c(N) := c(N) - lr$ where l represents a number of such wireless links.

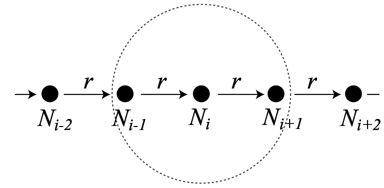

Fig. 11. Update of Capacities of Wireless Nodes (2).

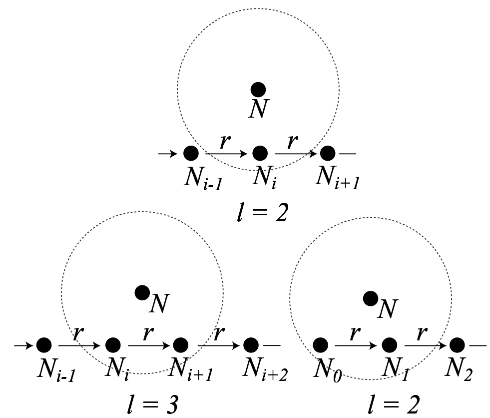

Fig. 12. Update of Capacities of Wireless Nodes (3).

According to the examination of update of $c(N)$, the following trivial condition for a wireless multihop transmission route R to be a flow increasing one is induced.

[Condition for Flow Increasing Route]

A wireless multihop transmission route $R = ||N_0 \ldots N_n\rangle\rangle$ increments data message flow if $c(N_i) > 0$ for all the wireless nodes in R and $c(N) > 0$ for all 1-hop neighbor wireless nodes $N \in Nei(N_i)$ of $N_i \in R$. $\square$

Next, same as in the original labeling method, we examine cases when a flow increasing route is configured by reduction of already reserved amount of data messages in wireless links. Due to effect on exposed and hidden nodes, the conditions induced for additional reservation of capacities along a wireless multihop transmission route $R = ||N_0 \ldots N_n\rangle\rangle$ by reduction of reserved capacity in a wireless link $|N_{i+1}N_i\rangle$ is different from the original labeling method.

Figure 13 shows an example where there has already been a wireless multihop transmission route R' from a source wireless node N^s to a destination one N^d along which capacity r' has been reserved in all the wireless links and an additional wireless multihop transmission route R provides an additional flow r of data messages from $N^s = N_0$ to $N^d = N_n$. Here, $|N_iN_{i+1}\rangle \in R$, $|N_{i+1}N_i\rangle \in R'$ and $r' > r$. By addition of R as a wireless multihop transmission route from N^s to N^d, reserved capacities $r(|N_jN_{j+1}\rangle)$ ($j \neq i$) increase r, i.e., available capacities $c(|N_jN_{j+1}\rangle)$ decrease r; however, $r(|N_{i+1}N_i\rangle)$ decreases r by update from r' to $r'-r$, i.e.,$c(|N_{i+1}N_i\rangle)$ increases r. Thus, though capacities $c(N_j)$ of intermediate wireless nodes N_j ($j \leq i-2$ or $j \geq i+3$) and

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those of its 1-hop neighbor wireless nodes are updated as usual by reduction of $4r$, i.e., $c(N_j) := c(N_j) - 4r$. On the other hand, capacities $c(N_j)$ of N_j $(j = n-1, n, n+1, n+2)$ and those of 1-hop neighbor wireless nodes of N_i and N_{i+1} reduces only $2r$, i.e, $c(N_j) := c(N_j) - 2r$. Hence, in order for R to be a flow increasing route, $c(N_j) > 0$ is required in all wireless nodes in R.

However, consider a wireless node N which is in a wireless range of N_i and is out of wireless ranges of N_j $(j \neq i)$. In this case, since reserved capacities $r(|N_{i-1}N_i\rangle)$ and $r(|N_iN_{i+1}\rangle)$ increases and decreases r, respectively, capacity of N is unchanged. Thus, even if $c(N) = 0$, R is a flow increasing route and r can be reserved along R. Same as this way, for a wireless node N' in a wireless range of N_{i+1} and out of wireless ranges of N_j $(j \neq i+1)$, even if $c(N') = 0$, it does not prohibit R to be a flow increasing route. This is also because $r(|N_iN_{i+1}\rangle)$ decreases r and $r(|N_{i+1}N_{i+2}\rangle)$ increases r.

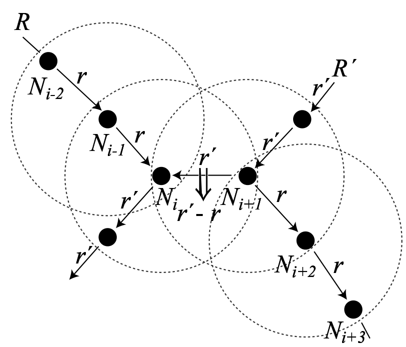

Fig. 13. Flow Increasing Route with Wireless Link in Another Route in Opposite Direction.

Figure 14 shows cases where a newly detected wireless multihop transmission route R contains some wireless links in which capacities have already reserved for another route in opposite direction. Here, reserved capacities in wireless links along R is r which is less than r' reserved in opposite direction in shared wireless links. As discussed above, if $r(|N_{i+1}N_i\rangle) \geq r$ has already reserved in $|N_{i+1}N_i\rangle$, $r(|N_{i+1}N_i\rangle)$ decreases r. Hence, distribution of reduction of capacities in wireless nodes are as Figure 14(a). In order for R to be a flow increasing route, all wireless nodes N in areas where the reduction of capacities are greater than 0 have positive capacities, i.e., $c(N) > 0$.

In cases that capacities have already been reserved in multiple successive wireless links in opposite direction along a newly detected route where these wireless links may be included in different routes, the restriction on the newly detected route to provide additional flow of data messages are relaxed. Figure 14 shows a case where $r(|N_iN_{i-1}\rangle) \geq r$ and $r(|N_{i+1}N_i\rangle) \geq r$ have been reserved and by reservation of r along R both $r(|N_iN_{i-1}\rangle)$ and $r(|N_{i+1}N_i\rangle)$ decreases r. Capacities in areas where amount of reduction of capacities is greater than 0 should be greater than 0 as N_j $(j \leq i-2$ or $j \geq i+2)$. However, in N_{i-1}, N_i and N_{i+1}, changes of capacities are 0. That is, even if $c(N_j) = 0$ $(j = i-1, i, i+1)$, R is a flow increasing route. In addition, capacities in wireless nodes in areas where the amount of reduced capacities by additional reservation is negative, their available capacities increases. Hence, in such wireless nodes N, $c(N) = 0$ is allowed for the reservation of r along R. Figure 14(c) shows a case that wireless links with reservation in opposite direction distributes separately along R.

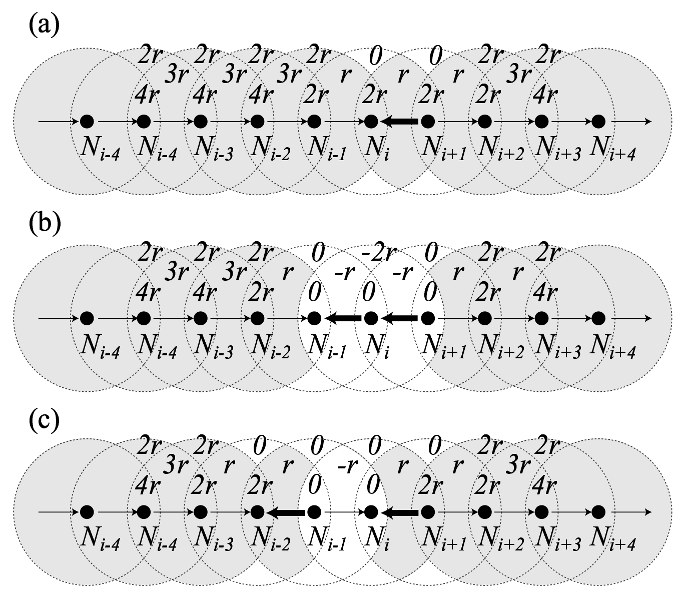

Fig. 14. Amount of Reduction of Available Capacities in Wireless Nodes with Capacity Reservation in Opposite Direction in Multiple Wireless Links.

Therefore, conditions for a flow increasing route containing wireless links with capacity reservation in opposite direction are as follows:

[Condition for Flow Increasing Route]
Suppose a pseudo capacity $r > 0$ is reserved in wireless links along a wireless multihop transmission route R from a sources wireless node N^s to a destination one N^d. For wireless links in R, reserved capacity $r(|N_iN_{i+1}\rangle)$ increases r if $r(|N_{i+1}N_i\rangle) = 0$ and $r(|N_{i+1}N_i\rangle)$ decreases r if $r(|N_{i+1}N_i\rangle) > 0$ by this reservation. According to this calculation, the amount of reduction of available capacities $c(N)$ of wireless nodes N are evaluated. If available capacities $c(N')$ of wireless nodes N' whose evaluated reduction of capacities are greater than 0 satisfy $c(N') > 0$, R is a flow increasing route from N^s to N^d. □

Figure 15 shows amount of reduction of available capacities of wireless nodes for a newly detected wireless transmission route R without reservation of capacities in wireless links $|N_{i+1}N_i\rangle$ in opposite direction. Since $r(|N_{i+1}N_i\rangle) = 0$ is satisfied in all wireless links in R, available capacities of all wireless nodes included in R and its 1-hop neighbor decrease for capacity reservation along R. Thus, $c(N) > 0$ is required in all wireless nodes N in R and its 1-hop neighbor ones. This is equivalent to the trivial condition mentions in this subsection and the latter condition contains the former one. In addition, since the pseudo reservation capacity may be any positive

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value for evaluation of changes of available capacities, only numbers of links which increase and decrease capacities of neighbor wireless nodes are required. Therefore, the condition for a flow increasing route is summarized as follows:

[Condition for Flow Increasing Route]

A wireless multihop transmission route R from a source wireless node N^s to a destination one N^d increases an amount of data message flow by capacity reservation if available capacities $c(N') > 0$ where a wireless node N' is included in R or 1-hop neighbor wireless nodes of them and the number of wireless links which connect to N' or its 1-hop neighbor wireless nodes and whose reserved capacity increases is larger than the number of wireless links which also connect to N' or its 1-hop neighbor wireless nodes and whose reserved capacity decreases. □

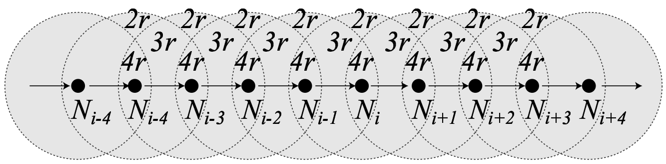

Fig. 15. Amount of Reduction of Available Capacities in Wireless Nodes without Capacity Reservation in Opposite Direction.

IV. Concluding Remarks

This paper proposes an extended labeling method for reservation of wireless link capacities in wireless multihop networks. For multimedia data transmission which requires high and stable throughput of data messages, capacity reservation is applied to multiple multihop routes. The original labeling method only applied to point-to-point based wired networks in which capacity in each wired link is reserved independently of the others. In wireless networks, due to broadcast based communication and existence of exposed and hidden nodes, reservation in neighbor wireless links is dependent each other. This paper proposes a modified condition of flow increasing wireless multihop transmission route which is critical to design a capacity reservation protocol based on the extended labeling method.

Based on the proposed condition, we are now designing a capacity reservation protocol. The extended labeling method with a depth-first search algorithm detects a set of wireless multihop transmission routes which realize the (pseudo) maximum throughput from a source wireless node to a destination one. Based on this examination, we are now considering a protocol to realize a set of wireless multihop transmission routes satisfying the application requirements with less communication overhead to exchange required control messages. In paper [9], we have already proposed a search cut-off algorithm to reduce the search area for lower overhead route search.

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Contention Aware Routing for Intermittently Connected Mobile Networks

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***Abstract*—This paper introduces a novel multi-copy routing protocol, called Self Adaptive Utility-based Routing Protocol (SAURP), for Delay Tolerant Networks (DTNs) that are possibly composed of a vast number of miniature devices such as smart phones, hand-held devices, and sensors mounted in fixed or mobile objects. SAURP aims to explore the possibility of taking mobile nodes as message carriers in order for end-to-end delivery of the messages. The best carrier for a message is determined by the prediction result using a novel contact model, where the network status, including wireless link condition and nodal buffer availability, are jointly considered. The paper argues and proves that the nodal movement and the predicted collocation with the message recipient can serve as meaningful information to achieve an intelligent message forwarding decision at each node. The proposed protocol has been implemented and compared with a number of existing encounter-based routing approaches in terms of delivery delay, and the number of transmissions required for message delivery. The simulation results show that the proposed SAURP outperforms all the counterpart multi-copy encounter-based routing protocols considered in the study.**

***Keywords*-Encounter based Routing, DTN.**

I. Introduction

Delay Tolerant Networks (DTNs) [1] are characterized by the lack of end-to-end paths for a given node pair for extended periods, which demonstrates a complete different design scenario from that for the conventional mobile ad-hoc networks (MANETs) [13]. Due to the intermittent connections in DTNs, a node is allowed to buffer a message and wait until it finds an available link to the next hop that will be able to store the message. Such a process is repeated until the message reaches its destination. This model of routing constitutes a significant difference from that employed in the MANETs, which is usually referred to as encounter-based, store-carry-forward, or mobility-assisted routing, due to the fact that nodal mobility serves as a significant factor for the forwarding decision of each message.

Depending on the number of copies of a message that may coexist in the network, two major categories of encounter-based routing schemes are defined: single-copy and multi-copy. With the single-copy schemes [5], no more than a single copy of a message can be carried by any node at any instance in the network. Although simple and resource efficient, the main challenge in the implementation of single-copy schemes lies in how to efficiently deal with interruptions of network connectivity and node failures. Thus, single-copy schemes have been reported to seriously suffer from long delivery delay and/or large message loss ratio.

On the other hand, multiple-copy (or multi-copy) routing schemes allow the networks to have multiple copies of a same message that can be routed independently and in parallel so as to increase robustness and performance. It is worth of noting that most multi-copy routing protocols are flooding-based [3], [4] that distribute unlimited numbers of copies throughout the network, or controlled tree-based flooding [20] that distribute just a subset of message copies, or utility-based approaches [2], [22] that determines whether a message should be copied to a contacted node simply based on a developed utility function.

Although improved in terms of performance, the previously reported multi-copy schemes are subject to respective problems and implementation difficulties. First of all, these schemes inevitably take a large amount of transmission bandwidth, and nodal memory space, which could easily dominate the network resource consumption [6]. In addition, they suffer from contention in case of high traffic loads, in which packet drops could result in a significant degradation of performance and scalability. Note that the future DTNs are expected to operate on a vast number of miniature and hand-held devices such as smart phones, tablet computers, personal digital assistants (PDAs), and fixed/mobile sensors, which are subject to a stringent limitation on power consumption and computation resources.

To cope with the deficiency of single-copy and multi-copy schemes, a family of multi-copy schemes called Utility-based controled flooding [15], [21], [14], [12] was proposed. The class of schemes generate only a small number of copies to ensure that the network is not overloaded with the launched messages. Although Utility-based controled flooding routing schemes have been reported to effectively reduce the message delivery delay and the number of transmissions, most of them assume that each node has sufficient resources for message buffering and forwarding. None of them have investigated how the protocol should take advantage of dynamic network status to improve the performance, such as

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packet collisions, wireless link conditions, and nodal buffer occupancy. There is obviously some room to improve for the Spray routing schemes in the DTN scenario considered in this study.

With this in mind, we introduce a novel DTN routing protocol, called Self Adaptive Utility-based Routing Protocol (SAURP) that overcomes the shortcomings of the previously reported multi-copy schemes. The main feature of the proposed protocol is the strong capability in adaptation to the fluctuation of network status, traffic patterns/characteristics, and user behaviors, so as to reduce the number of transmissions, message delivery time, and increase delivery ratio. This is achieved by jointly considering node mobility statistics, congestion, and buffer occupancy, which are subsequently fused in a novel quality-metric function. In specific, the link availability and buffer occupancy statistics are obtained by sampling the channels and buffer space during each contact with another node. In addition to this feature, we introduced new transitivity update rule and new adaptive time-window update strategy for updating the quality metric function. The developed quality-metric function targets to facilitating decision making for each active data message, resulting in optimized network performance. The accuracy of the utility fuction is verified a statistical mathematical model. We will show via extensive simulations that the proposed SAURP can achieve a significant performance gain over the previously reported counterparts.

The rest of the paper is organized as follows. Section II provides related work. Section III describes the proposed SAURP in detail. Then, Section IV provides the simulation results and the comparisons with the other counterparts. In Section V, we conclude the paper.

II. Related work

The previously reported encounter-based routing protocols have focused on the node mobility which is exploited and taken as the dominant factor in the message forwarding decision. Those schemes contributed by introducing novel interpretations of the observed node mobility in the per-node utility function. Spyropoulos et al. in [12], [6] developed routing strategies using different utility routing metrics based on nodal mobility statistics, namely Most Mobile First (MMF), Most Social First (MSF) and Last Seen First (LSF). S. C Nelson et al. [29] proposed an enhanced version of MSF, where the number of a message replica are transferred during contact is proportional to per-node utility function based on the evolution of the number of encounters a node has during a time-window. Lindgren et al. in [2] introduced a routing technique in DTNs which takes advantage of the predicted encounter probability between nodes. Jones et al. in [19] introduced a utility function for DTN routing which manipulates the minimum expected inter-encounter duration between nodes. Ling et al. in [23] designed a feedback adaptive routing scheme based on the factors solely determined by the node mobility, where a node with higher mobility is given a higher factor, and messages are transmitted through nodes with higher influence factors. A. Balasubramanian et al. in [22] considered statistics of available bandwidth and the number of message replicas currently in the network in the derivation of the routing metric to decide which message to replicate first among all messages in custodian buffer. The derivation of the routing metric, nonetheless, is not related to buffer status.

Another scheme is called delegation forwarding [14], [24], where a custodian node forwards a message copy to an encountered node if the encountered node has a better chance to “see” the destination. The key idea is that a custodian node (source or relay) forwards a message copy only if the utility function (represented by the rate of encounters between node pairs) of the encountered node is higher than all the nodes so far “seen” by a message, and then current custodian will update its utility value of that message to be equal to that of the encountered node. Mosli et al. in [17] introduced a DTN routing scheme using utility functions that are calculated from an evaluation of context information. The derived cost function is used as an assigned weight for each node that quantifies its suitability to deliver messages to an encountered node regarding to a given destination. A sophisticated scheme was introduced by Spyropoulos et al., called Spay and Focus [6], which is characterized by addressing an upper bound on the number of message copies (denoted as L). In specific, a message source starts with L copy tokens. When it encounters another node B currently without any copy of the message, it shares the message delivery responsibility with B by transferring $L/2$ of its current tokens to B while keeping the other half for itself. When it has only one copy left, it switches to a utility forwarding mechanism based on the LSF (time elapsed since the last contact). This scheme has proven to significantly reduce the required number of transmissions, while achieving a competitive delay with respect to network contentions such as buffers space and bandwidth. An approach very similar to the Spray and Focus protocol was proposed by Li et al. [7], which differs from that by [6] in the employed utility function and queuing policy mechanisms. In specific, the utility function is designed based on the probability of the duration of the contact time between pairs for a given time window interval.

Although some studies improved the previously reported designs by overcoming some of the shortages [6], [7], [2], [14], they are subject to various limitations in the utility function updating processes. These limitations are addressed in our previous work in [15]. More importantly, the channel capacity and buffer occupancy states have never jointly been considered as factor in the derivation of utility functions. These two factors could be overlooked/ignored if the encounter frequency is low, where the routing protocol performance is dominated by node mobility, while the network resource availability does not plays an important role. However, in the scenario that the nodal encounter frequency is large and each node has many choices for packet forwarding, the network resource availability could become a critical factor for improving routing protocol performance, and should be taken seriously in the derivation of utility functions.

Motivated by above observations, this work investigates encounter based routing technique that jointly considers node mobility and the network states, including wireless channel and buffer occupancy. This work proposes other strategies that can use fewer copies than the Spray and Focus scheme by spreading a number of copies that is less than or at most equal to the number of copies used in the Spray and Focus scheme, while obtaining better guaranteed results than those of other schemes described in the literature.

III. Self Adaptive Utility-based Routing Protocol (SAURP)

The most distinguished characteristic of SAURP is its ability of adapting itself to the observed network behaviors in order to reduce the number of transmissions, the message delivery time, and the delivery ratio. This is made possible by employing an efficient strategy for achieving a time-window based update mechanism for some network status parameters at each node. We use time-window based update strategy because it is simple in implementation and rather robust against parameter fluctuation. Note that the network conditions could change very fast and make a completely event-driven model unstable. Figure. 1 illustrates the functional modules of the SAURP architecture along with their relations.

The Contact Statistics (denoted as $CS^{(i)}$) is obtained between each node pair A, and B regarding the total nodal contacts durations, channel condition, and buffer occupancy state. These values are collected at the end of each time window and used as one of the two inputs to the Utility-function Calculation and Update Module (UCUM). Another input to the UCUM, as shown in Figure 1, is the updated utility denoted as $\triangle T^{(i)}_{new}$, which is obtained by feeding $\triangle T^{(i)}$ (the inter-contact time between any node pair, A and B) through the Transitivity Update Module (TUM). UCUM is applied such that an adaptive and smooth transfer between two consecutive time windows (from current time-window to next time-window) is maintained. Inter-contact time ($\triangle T^{(i+1)}$) is the output of UCUM, and is calculated at the end of current time window $W^{(i)}$. $\triangle T^{(i+1)}$ is thus used in time window $W^{(i+1)}$ for the completely the same tasks as in window $W^{(i)}$.

Forwarding Strategy Module (FSM) is applied at the custodian node as a decision making process when encountering any other node within the current time window based on the utility value (i.e., $\triangle T^{(i)}$).

It is important to note that CS, TUM, FSM, and message vector exchange are event-driven and performed during each contact, while UCUM is performed at the end of each time-window. The following subsections introduce each functional module in detail.

A. *Contact Statistics (CS)*

To compromise between the network state adaptability and computation complexity, each node continuously updates the network status within a fixed time window. The maintained network states are referred to as Contact Statistics (CS), which include nodal contact durations, channel conditions, and buffer occupancy state, and will be fed into UCUM at the end of each time window. The CS collection process is described as follows.

Let two nodes A and B are in the transmission range of each other, and each broadcasts a pilot signal per k time units in order to look for its neighbors within its transmission range. Let $T_{(A,B)}$, T_{free}, and T_{busy} represent the total contact time, the amount of time the channel is free and the buffer is not full, and the amount of time the channel is busy or the buffer is full, respectively, at node A or B during time window $W^{(i)}$. Thus, the total duration of time in which node A and B can exchange information is calculated as:

$$T_{free} = T_{(A,B)} - T_{busy} \tag{1}$$

Note that the total contact time could be accumulated over multiple contacts between A and B during $W^{(i)}$.

B. *Utility-function Calculation and Update Module (UCUM)*

UCUM is applied at the end of each time window and is used to calculate the currently observed utility that will be further used in the next time window. The two inputs to UCUM in time window $W^{(i)}$ are: (i) the predicted inter-contact time ($\triangle T^{(i)}$), which is calculated according to the previous time-window utility (i.e., $\triangle T^{(i)}$), as well as an update process via the transitivity property update (introduced in subsection 3.3), and (ii) the observed inter-encounter time obtained from the current $CS^{(i)}$ (denoted as $\triangle T^{(i)}_{cs}$).

1) Calculation of Inter-encounter Time ($\triangle T^{(i)}$): An eligible contact of two nodes occurs if the duration of the contact can support a complete transfer of at least a single message between the two nodes. Thus, in the event that node A encounters B for a total time duration T_{free} during time window $W^{(i)}$, the number of eligible contacts in the time window is determined by:

$$n_c = \left\lfloor \frac{T_{free}}{T_p} \right\rfloor \tag{2}$$

where T_p is the least time duration required to transmit a single message. Let $\triangle T^{(i)}_{cs(A,B)}$ denotes the average inter-encounter time duration of node A and B in time window

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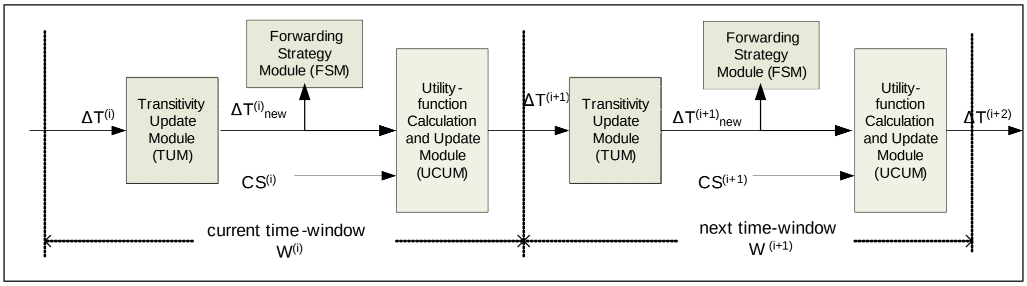

Figure 1. The SAURP Architecture

i. Obviously, $\triangle T^{(i)}_{(A,B)} = \triangle T^{(i)}_{(B,A)}$. We have the following expression for$\triangle T^{(i)}_{cs(A,B)}$:

$$\triangle T^{(i)}_{cs(A,B)} = \frac{W}{n_c} \tag{3}$$

$\triangle T^{(i)}{}_{cs(A,B)}$ describes how often the two nodes encounter each other per unit of time (or, the encounter frequency) during time window i considering the event the channel is busy or the buffer is full.

Thus, inter-encounter time of a node pair intrinsically relies rather on the duration and frequency of previous contacts of the two nodes than simply on the number of previous contacts or contact duration. Including the total duration of all the contacts (excluding the case when the channel is busy or the buffer is full) as the parameter is expected to better reflect the likelihood that nodes will meet with each other for effective message exchange. With this, the proposed routing protocol does not presume any knowledge of future events, such as node velocity, node movement direction, instants of time with power on or off, etc; instead, each node keeps network statistic histories with respect to the inter-encounter frequency of each node pair (or, how often the two nodes encounter each other and are able to perform an effective message exchange).

2) Time-window Transfer Update: Another important function provided in UCUM is for the smooth transfer of the parameters between consecutive time windows. As discussed earlier, the connectivity between any two nodes is measured according to the amount of inter-encounter time during $W^{(i)}$, which is mainly based on the number of contacts (i.e., n_c) and the contact time (i.e., T_{free}). These contacts and contact durations may change dramatically from one time window to the other and address significant impacts on the protocol message forwarding decision. Hence, our scheme determines the next time window parameter using two parts: one is the current time window observed statistics between node A and B (i.e., $\triangle T^{(i)}_{cs(A,B)}$), and the other is from the previous time window parameters (i.e., $\triangle T^{(i)}$), in order to achieve a smooth transfer of parameter evolution. The following equation shows the derivation of $\triangle T^{(i+1)}$ in our scheme.

$$\triangle T^{(i+1)}_{(A,B)} = \gamma.\triangle T^{(i)}_{cs(A,B)} + (1-\gamma)\triangle T^{(i)}_{(A,B)} \tag{4}$$

The parameter γ is given by

$$\gamma = \frac{\mid \triangle T^{(i)}_{(A,B)} - \triangle T^{(i)}_{cs(A,B)} \mid}{max(\triangle T^{(i)}_{(A,B)}, \triangle T^{(i)}_{cs(A,B)})},\ \triangle T^{(i)}_{(A,B)},\ \triangle T^{(i)}_{cs(A,B)} > 0 \tag{5}$$

The above relation is hold even if $\triangle T^{(i)}_{(A,B)} \geq W$ and $\triangle T^{(i)}_{cs(A,B)} \geq W$ which represents the worst case scenario, i.e. unstable node behavior, low quality of node mobility, or very congested area.

$\triangle T^{(i+1)}_{(A,B)}$ represents the routing metric value that is used as input to the next time window. This value is maintained as a vector of inter-encounter time that is specific to every other node, and the vector is called *routing metric table*. The routing metric table can be employed in the decision making process for message forwarding.

C. The Transitivity Update Module

When two nodes are within transmission range of each other, they exchange utility vectors regarding the message destination. With the update, the custodian node decides whether or not the message should be forwarded to the encountered node. This exchange of summary vectors is followed by another update, called transitivity update. We propose a new transitivity update rule that is adaptively modified according to ratio of the $\triangle T^{(i)}s$ between nodes. Although the idea of using transitivity updates are not new [2], the proposed SAURP has gone through a much different way. The transitivity property based on the observation that if node A frequently encounters node B and B frequently encounters node D, then A has good ability to forward messages to D through B. We formulated the updating rule as follows:

$$\triangle T^{(i)}_{(A,D)new} = \alpha\triangle T^{(i)}_{(A,B)} + (1-\alpha)(\triangle T^{(i)}_{(A,B)} + \triangle T^{(i)}_{(B,D)}) \tag{6}$$

where α is weighting factor that must be less than 1 to be valid.

$$\alpha = \frac{\triangle T^{(i)}_{(A,B)} + \triangle T^{(i)}_{(B,D)}}{\triangle T^{(i)}_{(A,D)}},\ \triangle T^{(i)}_{(A,D)} > \triangle T^{(i)}_{(A,B)} + \triangle T^{(i)}_{(B,D)} \tag{7}$$

α has a significant impact on the routing decision rule. From theoretical perspective, when a node is encountered that has more information for a destination, this transitivity effect should successfully capture the amount of uncertainty to be resolved regarding the position of the destination. Thus, a transitivity property is needed to update values only when $\triangle T^{(i)}_{(A,D)} > \triangle T^{(i)}_{(B,D)}$ in order to ensure that node A reaches D through B. Otherwise, if $\triangle T^{(i)}_{(A,D)} < \triangle T^{(i)}_{(B,D)}$, the transitivity property is not useful since node A is a better candidate for forwarding messages directly to node D rather than forwarding them through B. This rule is applied after nodes finish exchange messages.

D. The Forwarding Strategy Module (FSM)

The decision of message forwarding in SAURP is mainly based on the goodness of the encountered node regarding the destination, and the number of message copy tokens. If the message tokens greater than 1, weighted copy rule is applied, the forwarding rule is applied otherwise.

1) The Weighted Copy Rule : The source of a message initially starts with L copies; any node A that has $n > 1$ message copy tokens (source or relay) and that encounters another node B with no copies and $\triangle T^{(i)}_{(B,D)} < \triangle T^{(i)}_{(A,D)}$, node A hands over to node B a number of copies according to its goodness for the destination node D. Node A hands over some of the message copy tokens to node B and keeps the rest for itself according to the following formula:

$$N_B = \left\lfloor N_A \left(\frac{\triangle T^{(i)}_{(A,D)}}{\triangle T^{(i)}_{(B,D)} + \triangle T^{(i)}_{(A,D)})} \right) \right\rfloor \tag{8}$$

where N_A is the number of message tokens that node A has, $\triangle T^{(i)}_{(B,D)}$ is the inter-encounter time between node B and node D, and $\triangle T^{(i)}_{(A,D)}$ is the inter-encounter time between nodes A and D. This formula guarantees that the largest number of message copies is spread to relay nodes that have better information about destination node. After L messages have been copied to custodian nodes, each of the L nodes carrying a copy of the message performs according to the forwarding rule as descried next. This idea of weighted copy rule was examined in [15], [29] and has been proved with improved delivery delay.

2) The Forwarding Rule :

- If the destination node is one hop away from an encountered node, the custodian node hands over the message to the encountered node.
- If the inter-encounter time value of the encountered node relative to that of the destination node is less than that of the custodian node by a threshold value, $\triangle T_{th}$, a custodian node hands over the message to the encountered node.

The complete mechanism of the forwarding strategy in SAURP is summarized as shown in Algorithm 1.

Algorithm 1 The forwarding strategy of SAURP

```
On contact between node A and B
Exchange summary vectors
for every message M at buffer of custodian node A do
    if destination node D in transmission range of B then
        A forwards message copy to B
    end if
    if ΔT(i)(A,D) > ΔT(i)(B,D) do
        if message tokens >1 then
            apply weighted copy rule
        end if
        else if ΔT(i)(A,D) > ΔT(i)(B,D) + ΔT_th then
            A forwards message to B
        end if
    end if
end for
```

IV. Performance Evaluation

In this section a statistical analysis is conducted on the performance of the proposed SUARP. An adapted DTN simulator similar to the one used by [6] was created [30]. Community-Based Mobility model (CBM)[9] (the reader is referred to [9] for more details) is employed in the analysis. The problem setup consists of an ad hoc network with a number of nodes moving independently on a $\sqrt{N}X\sqrt{N}$ 2-dimensional torus (torus is used due to its symmetry; and similarity in the performance when we have run simulations in bounded networks) in a geographical region, and each node belongs to a predetermined community. Each node can transmit up to a distance $K \geq 0$ meters away, and each message transmission takes one time unit. Euclidean distance is used to measure the proximity between two nodes (or their positions) A and B. A slotted collision avoidance MAC protocol with Clear-to-Send (CTS) and Request-to-Send (RTS), is implemented for contention resolution. A message is acknowledged if it is received successfully at the encountered node by sending back a small acknowledgment packet to the sender. The performance of SAURP is examined under different network scenarios and is compared to some previously reported schemes listed below.

- Epidemic routing (epidemic) [3]
- Spray and Focus (S&F) [6]
- Most mobile first (MMF)[25]
- Delegation forwarding (DF) [14]
- Self-Adaptive utility-based routing protocol (SAURP)

The performance comparison was in terms of average delivery delay per message, and the total number of transmissions performed for all delivered messages.

A. Evaluation Scenarios

In the simulation, 120 nodes move according to the community-based mobility model [6] in a 600 x 600 meter

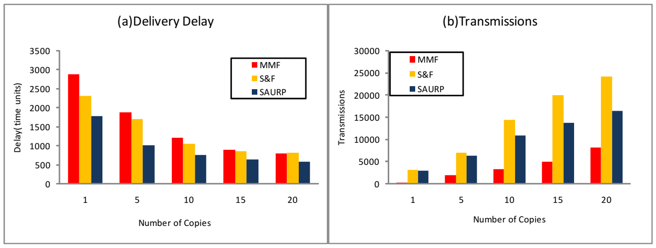

Figure 2. Impact of the number of message copies

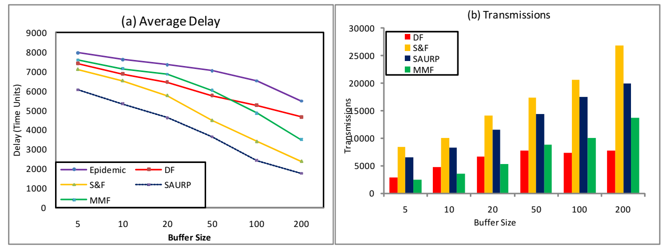

Figure 3. The effect of buffer size

network, and community size = 60x60. The message inter-arrival time is uniformly distributed in such a way that the traffic can be varied from low (10 messages per node in 40,000 time units) to high (70 messages per node in 40,000 time units). The message time to live (TTL) is set to 9,000 time units. Each source node selects a random destination node, begins generating messages to it during simulation time.

The performance of the protocols is evaluated with respect to the impact of the number of message copies. Second, with respect to the low transmission range and varying buffer capacity under high traffic load. Finally, with respect to the moderate-level of connectivity and varying traffic load.

1) Impact due to Number of Message Copies : We firstly look into impact of the number of message copies toward the performance of each protocol. The transmission range K of each node is set to 40 meters, leading to a relatively sparse network. In order to reduce the effect of contention on any shared channel, the traffic load and buffer capacity is set to medium (i.e., 40 generated messages per node in 40,000 time units) and high (i.e. 1000 messages), respectively. The number of message copies is then increased from 1 to 20 in order to examine their impact on the effectiveness of each protocol. The proposed SAURP is compared with the S&F and MMF schemes, since each scheme has a predefined L to achieve the best data delivery. Note that the value of L depends on the application requirements, the mobility model considered, and the design of the protocol.

Figure. 2 shows the results on message delivery delay, and number of transmissions under different numbers of copies of each generated message. As can be seen, the L value has a significant impact on the performance of each scheme. It is observed that best performance can be achieved under each scheme with a specific value L. These L values can serve as a useful rule of thumb for producing good performance.

2) The Effect of Buffer Size: In this scenario the performance of SAURP regarding different buffer sizes is examined under a low transmission range (i.e., K = 30) and a high traffic load (i.e., 70 messages generated per node in 40,000 time units). Due to the high traffic volumes, we expect to see a significant impact upon the message forwarding decisions due to the degradation of utility function values caused by buffer overflow. Note that when the buffer of the encountered node is full, some messages cannot be delivered even though the encountered node metric is better than the custodian node. This situation results in extra queuing delay, especially in the case that flooding-based schemes are in place. Figure 3 shows the experiment results where the buffer space was varied from 5 (very limited capacity) to 200 (relatively high capacity) messages to reflect the performance of the protocols under the considered traffic load. As shown in Figure 3, when the buffer size is small (50 messages or less) the performance of the protocols is very sensitive to the change of buffer capacity.

It is observed that Epidemic routing produced the worst delivery delay in all scenarios, since it has been critically affected by both the limited buffer size and mobility model. On the other hand, since SAURP takes the situation that a node may have a full buffer into consideration by degrading the corresponding utility metric, it produced the best performance. In specific, SAURP yielded a shorter delivery delay than DF by 40%. Although SUARP produced more transmissions than MMF, it yielded a smaller delivery delay than that of MMF by 70%. As the buffer size increased, the performance of all protocols was improved especially for MMF. When the buffer size is larger than the traffic demand, the MMF scheme has yielded a competitive performance due to the relaxation of buffer capacity limitation. SAURP still yielded the best performance with a smaller number of transmissions than S&F by 37%. At large buffer size, epidemic routing performs much more transmissions than other schemes at least an order of magnitude higher than the SAURP scheme does, and thus not included in the plot.

3) The Effect of Traffic Load : The main goal of this scenario is to observe the performance impact and how SAURP reacts under different degrees of wireless channel contention. The network connectivity is kept high (i.e., the transmission range is set to as high as 70 meters) under different traffic loads, while channel bandwidth is set relatively quite small (i.e., one message transfer per unit of time) in order to create an environment with non-trivial congestion. We have two scenarios for nodal buffer capacity: 1) unlimited capacity; and 2) limited capacity (15 messages). Figure.7 shows the performance of all the routing algorithms in terms of the average delivery delay, and total number of transmissions.

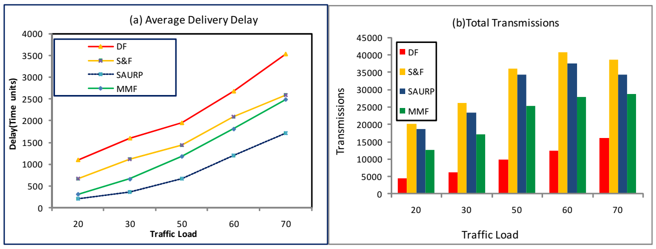

Figure 4. The effect of traffic load under high buffer capacity

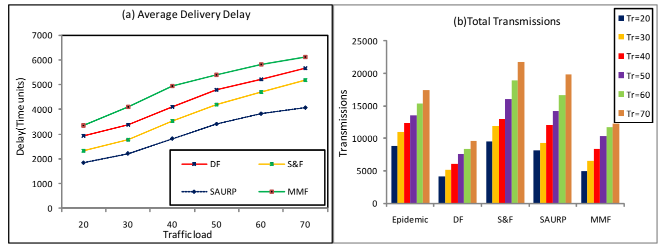

Figure 5. The effect of traffic load under low buffer capacity

It is observed that Epidemic routing produced the largest delivery delay and requires a higher number of transmissions compared to all the other schemes, thus it is not included in the figure. Note that the Epidemic routing is subject to at least 3 times of longer delivery delay than that by S&F and an order of magnitude more transmissions than that by SUARP.

As shown in Figure. 4(a), and 4(b), when the traffic load is increased, the available bandwidth is decreased accordingly, which causes performance reduction. When the traffic load is moderate (i.e., less that 50 messages in 40,000 time units), it is clear that the delivery delay is short in all the schemes, while SAURP outperforms all other protocols and MMF is the second best. This is because in MMF, the effect of buffer size is relaxed, which makes " "roaming nodes" buffer more number of messages while roaming among communities. SAURP can produce delay shorter than that of MMF, DF, S&F by 52%, 400%, and 250%, respectively. Although SAURP requires more transmissions compared to the MMF and DF, the number is still smaller than that produced by S&F.

As expected, the performance of all the schemes degrades as wireless channel contention is getting higher especially when the traffic load exceeds 50 messages per node during the simulation period 40,000 time units. We observed that SAURP can achieve significantly better performance compared to all the other schemes, due to the consideration of busy links in its message forwarding mechanism, where the corresponding routing-metric is reduced accordingly. This results in the ability of rerouting the contended messages through the areas of low congestion. However, such a rerouting mechanism makes messages take possibly long routes and results in more transmissions than that of MMF. In summary, the delivery delay obtained by the SAURP in this scenario is shorter than that of MMF by 70% , S&F by 90%, and DF by 247%, respectively.

As the buffer capacity is low (e.g. 15 messages) and the traffic load is high, the available bandwidth decreases and the buffer occupancy increases accordingly causing buffer overflow. Buffer overflow prevents messages that should be forwarded to wait longer time at the buffer of the current custodian node until it find es new custodian with available buffer space. This situation makes the performance of all protocols degraded, especially for Epidemic and MMF. Epidemic routing produced the largest delivery delay compared to all the other schemes. It is subject to at least 4 times of longer delivery delay than that by S&F, and thus it is not included in the figure. It is notable that SUARP outperforms all the multiple-copy routing protocols in terms of delivery delay under all possible traffic loads. When the traffic load is high, SAURP yielded shorter delivery delay than that of MMF by 52%, SF by 30%, and DF by 40%. Although SAURP requires more transmissions compared to the MMF and DF, the number is still smaller than that produced by S&F. Figure. 5(a), and 5(b) shows the performance of all techniques under this scenario. Note that the transmissions produced by Epidemic routing are affected by the buffer size, resulting lower transmissions and longer delivery delay.

V. Conclusion

The paper introduced a novel multi-copy routing scheme called SAUPR, for intermittently connected mobile networks. SAURP is characterized by the ability of identifying potential opportunities for forwarding messages to their destinations via a novel utility function based mechanism, in which a suite of environment parameters, such as wireless channel condition, nodal buffer occupancy, and encounter statistics is jointly considered.

Thus, SAURP can reroute messages around nodes experience either high buffer occupancy, wireless interference, or congestion, while taking considerably smaller number of transmissions. We verified the proposed SAURP via extensive simulation and compared it with a number of counterparts. SUARP has shown great stability and achieved shorter delivery delays than all the existing spraying and flooding based schemes when the network experiences considerable contention on wireless links and/or buffer space.

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Usability Heuristics for Virtual Worlds

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***Abstract*— Usability evaluation for applications based on emerging information technology brings new challenges. Virtual Worlds (VWs) are computer-simulated virtual environments accessed by multiple users, through their avatars. VWs constitute a growing space for collaborative play, learning and work. When evaluating VWs' usability, there is a need for new evaluation methods or at least for the use of traditional evaluations in novel ways. A set of heuristics is proposed, in order to help the usability evaluations of VWs applications.**

Keywords- usability; usability heuristics; virtual worlds

I. INTRODUCTION

Virtual worlds (VWs) are computer-based simulated persistent spatial environment that supports synchronous communication among users, who are represented by avatars [1]. Users have to choose or create theirs avatars, which will be able to interact with objects, the virtual environment and other avatars. Avatar's identity frequently differs from user's identity.

VWs are a growing space for collaborative play, learning and work. Usually researches focus on the use of VWs and the phenomenon they represent; it is hard to find studies on VWs' usability, especially on VWs' usability evaluation methodology.

The ISO/IEC 9241 standard defines the usability as the extent to which a product can be used by specified users to achieve specified goals with effectiveness, efficiency and satisfaction in a specified context of use [2]. Usability evaluation methods are commonly divided into inspection and testing methods. Inspection methods find usability problems based on the expertise of usability professionals. Testing methods find usability problems through the observation of the users while they use (and comment on) a system interface [3].

Heuristic evaluation is a widely used inspection method [4] [5]. A group of evaluators (usually from three to five) inspect the interface design based on a set of usability heuristics. In order to ensure independent and unbiased evaluations, the inspection is performed individually. After all individual evaluations have been completed, the evaluators are allowed to communicate and have their findings aggregated in a single list of usability problems. Later on, each evaluator assigns scores to each problem's severity and frequency (on a 0 to 4 scale, from minor/less frequent to major/more recurrent). Severity and frequency are summed in order to get problem's criticality. Problems are ranked based on their average severity, frequency and criticality. The usability evaluation report includes usability problems, solution proposals, as well as positive findings.

Heuristic evaluation is easy to perform, cheap and able to find many usability problems (both major and minor problems). However, it may miss domain specific problems. That is why the use of appropriate heuristics is highly significant.

Usability evaluation for applications based on emerging information technology brings new challenges. Is it the classical concept of usability still valid? Which are the dimensions of the (new) usability? How can it be measured? How should we develop for (better) usability? There is a need for new evaluation methods or at least for the use of traditional evaluations in novel ways [6].

The paper focuses on usability evaluation of VWs applications, by heuristic evaluations. A set of 16 specific usability heuristics is proposed. Section 2 presents the methodology that has been used in heuristics' development. Section 3 highlights the main characteristics of VWs. The VWs usability heuristics proposal is presented in Section 4. Section 5 shows the proposal's preliminary validation. Section 6 presents preliminary conclusions and future works.

II. Defining Virtual Worlds Usability Heuristics

In order to develop usability heuristics for VWs, a specific methodology was applied [7]. The methodology to establish new usability heuristics includes 6 stages:

- STEP 1: An *exploratory* stage, to collect bibliography related with the main topics of the research: specific applications, their characteristics, general and/or related (if there are some) usability heuristics.
- STEP 2: A *descriptive* stage, to highlight the most important characteristics of the previously collected information, in order to formalize the main concepts associated with the research.
- STEP 3: A *correlational* stage, to identify the characteristics that the usability heuristics for specific applications should have, based on traditional heuristics and case studies analysis.
- STEP 4: An *explicative* stage, to formally specify the set of the proposed heuristics, using a standard template.
- STEP 5: A *validation* (experimental) stage, to check new heuristics against traditional heuristics by experiments, through heuristic evaluations performed on selected case studies, complemented by user tests.
- STEP 6: A *refinement* stage, based on the feedback from the validation stage.

Based on the well-known and widely used Nielsen's 10 heuristics, and extensively analyzing several VWs case studies (*Second Life, Club Penguin, Habbo Hotel, World of Warcraft, Ragnarok Online, ScienceSim*), a set of 16 new usability heuristics was developed for heuristic evaluations of VWs applications.

The methodology was applied iteratively; the set of new heuristics was refined in various steps. A specific usability checklist was also developed, detailing usability heuristics, in order to help the evaluation practice.

Section 3 synthetizes the findings of STEP 1 and STEP 2. Section 4 presents the results of STEP 3 and STEP 4. It specifies the refined heuristics proposal (based on STEP 5 and STEP 6). Section 5 presents the main results of STEP 5.

III. Virtual Worlds Characteristics

Nowadays VWs have a wide range of applications almost everywhere: organizations, educations, entertainment, training, virtual communities, e-commerce, scientific research, etc. There is no unique, widely accepted VWs' classification. Based on Porter's proposal (2004), Messinger, Stroulia and Lyons (2008) proposed a set of criteria, in order to establish the VWs typology [8] [9]:

- *Purpose* (content of interaction): The VW may be age focus, content focus, or open.
- *Place* (location of interaction): Players may be collocated or geographically dispersed.
- *Platform* (design of interaction): Communication may be synchronous, asynchronous, or both.
- *Population* (pattern of interaction): Is defined by the group's size, social ties, and characteristics of the target user market.
- *Profit model* (return on interaction): The VW may support single purchase price/registration fee, fee per use, subscription – based, advertising – based, pay – as you – go extras, and sale of ancillary products.

Some common features of VWs may be identified:

- *Avatar*: Each user is represented by its own (and only) avatar.
- *World's rules*: Each VW has its own unbreakable (physics) rules.
- *Shared environment*: A VW is shared by many users.
- *Interaction and communication*: User – user (through their avatars) and user – world interactions take place in real time.
- *Persistency*: The VW is (partially) persistent, regardless if individual users are logged in or out.
- *Customization*: VWs allow users to alter, develop, build, or submit customized content.
- *Graphic environment*: VWs offer computer-based graphic 2D, 2.5D or 3D environments.

Usability evaluations specifically focus on users, their needs and goals, and not on the inner part of the interactive software systems. Therefore, usability heuristics for VWs are meant to evaluate such products from the user perspective. As VWs are usually distributed systems, it is assumed that a set of basic (hardware, network, and platform related) requirements have to be accomplished. If not, the evaluation of applications' usability will be very difficult or even impossible.

IV. A Virtual World Usability Heuristics Proposal

VWs usability heuristics were specified using the following template:

- *ID, Name and Definition*: Heuristic's identifier, name and definition.
- *Explanation*: Heuristic's detailed explanation, including references to usability principles, typical usability problems, and related usability heuristics proposed by other authors.
- *Examples*: Examples of heuristic's violation and compliance.
- *Benefits*: Expected usability benefits, when the heuristic is accomplished.
- *Problems*: Anticipated problems of heuristic misunderstanding, when performing heuristic evaluations.

The 16 proposed usability heuristics were grouped in three categories: (1) *Design and Aesthetics*, (2) *Control and*

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Navigation and (3) *Errors and Help*. A summary of the proposed heuristics is presented below, including heuristic's ID, name and definition.

Design and Aesthetics Heuristics:

(H1) *Feedback*: A VW interface should keep user informed on both avatar's state, and the relevant facts and events that affect him.

(H2) *Clarity*: A VW should offer an easy to understand user control panel, using clear graphic elements, text and language, grouping elements by their purposes, and offering easy access to the main functionality.

(H3) *Simplicity*: A VW should provide easy and intuitive interaction with the environment's virtual objects. Only relevant information should be given, in order to avoid the control panel's overload.

(H4) *Consistency*: A VW should be consistent in using language and concepts. Avatar's actions and their effects on the VW's environment should be coherent and consistent. User – avatar, as well as avatar – VW's objects, should be consistent.

Control and Navigation Heuristics:

(H5) *Low memory load*: A VW should maintain main objects, options, elements and actions always available or easy to get to. It should provide ways to mark and remember places already visited and/or of user's interest.

(H6) *Flexibility and efficiency of use*: A VW should provide customizable shortcuts, abbreviations, accessibility keys or command lines. The user interface/control panel should be customizable.

(H7) *Camera control*: A VW should give user control over camera, allowing a customizable user's view.

(H8) *Visualization*: A VW should give user control over the objects and visual effects that he/she will get visible.

(H9) *Avatar's customization*: A VW should allow fully avatars' customization.

(H10) *Orientation and navigation*: A VW should provide full (customizable) information on avatar's position, paths to a desired destination, and passage ways from one position to another (according to VW's rules).

(H11) *World interaction*: A VW should clearly indicate the objects that user may interact with, as well as the actions that user may perform over the objects.

(H12) *World's rules*: A VW should clearly indicate its own rules and the rules that govern avatars, especially the actions that are impossible in the real (user's) world, but are possible in the VW (and vice versa).

(H13) *Communication between avatars*: A VW should allow easy communication among users, through their avatars.

Errors and Help Heuristics:

(H14) *Error prevention*: A VW should prevent users from performing actions that could lead to errors, and should avoid confusions that could lead to mistakes, during user – control panel interaction, as well as during (user's) avatar – VW interaction.

(H15) *Recovering from errors*: A VW should provide user appropriate mechanisms to recover from errors, and exit ways from unwanted situations. It should include clear messages, hopefully indicating causes and solutions for errors.

(H16) *Help and documentation*: A VW should provide an easy to find, easy to understand, and complete online documentation, accessible from both inside and outside of the world itself.

Table 1 presents the mapping between VWs 16 heuristics and Nielsen's 10 heuristics [5].

TABLE I. MAPPING BETWEEN VIRTUAL WORLDS HEURISTICS AND NIELSEN'S HEURISTICS

Virtual Worlds Heuristics		**Nielsen's Heuristics**	
ID	***Definition***	***ID***	***Definition***
H1	Feedback	N1	Visibility of system status
H2	Clarity	N2	Match between system and the real world
H3	Simplicity	N8	Aesthetic and minimalist design
H4	Consistency	N4	Consistency and standards
H5	Low memory load	N6	Recognition rather than recall
H6	Flexibility and efficiency of use	N7	Flexibility and efficiency of use
H7	Camera control	N3	User control and freedom
H8	Visualization		
H9	Avatar's customization		
H10	Orientation and navigation		
H11	World interaction		Various
H12	World's rules		
H13	Communication between avatars		
H14	Error prevention	N5	Error prevention
H15	Recovering from errors	N9	Help users recognize, diagnose, and recover from errors
H16	Help and documentation	N10	Help and documentation

VWs usability heuristics H1, H2, H3, H4, H5, and H6 particularize Nielsen's heuristics N1, N2, N8, N4, N6, and N7 (respectively), based on the VWs' characteristics.

Heuristics H7, H8, H9 and H10 are related to Nielsen's N3 heuristics. "*User control and freedom*" was detailed, considering relevant VWs aspects: visualization customization and navigation through the virtual environment.

Heuristics H11, H12 and H13 have no direct one – to – one relation to Nielsen's heuristics. They may be related to various Nielsen's heuristics (in different degrees of relevance).

Finally, heuristics H14, H15 and H16 put Nielsen's heuristics N5, N9 and N10 (respectively) into the context of VWs.

Based on the experiments made up to the date, the nature of the usability problems identified when applying VWs usability heuristics, and the problems that some evaluators had when applying such heuristics, a usability checklist was defined. It details the set of 16 heuristics and helps their use in heuristic evaluation practice. The checklist includes a total of 49 items (from 2 to 5 items per heuristic).

V. VALIDATING THE PROPOSAL: EARLY EXPERIMENTS

The 16 proposed VWs usability heuristic were checked against Nielsen's 10 heuristics, using *Club Penguin* as case study.

Club Penguin is a VW designed for 8-14 year olds children, a place where they can play games, have fun and interact with each other [10]. Users' avatars are penguins. Each player chooses a penguin, gives it an identity, and explores *Club Penguin*, interacting with other penguins by chatting, playing games, sending greeting cards, or using emoticons and actions (i.e. wave, dance, sit, walk or throw a snowball). By playing games, players earn virtual coins which they can eventually use to buy clothing and accessories for their penguin or furniture for their igloo.

Club Penguin was examined by two groups of 3 evaluators each. All 6 evaluators had similar (medium) experience in heuristic evaluations (with Nielsen's heuristics), but no experience in usability evaluation of VWs. The first group performed a heuristic evaluation of *Club Penguin* using only VWs usability heuristics, while the second group performed a similar heuristic evaluation, but using only Nielsen's heuristic.

A total of 52 problems were identified by the 6 evaluators. More usability problems were captured using VWs usability heuristics than using Nielsen's heuristics:

- 14 problems (26.9%) were identified by both groups of evaluators,
- 22 problems (46.2%) were identified only by the group which used VWs usability heuristics,
- 14 problems (26.9%) were identified only by the group which used Nielsen's heuristics.

The results seem to prove that VWs usability heuristics work better than Nielsen's heuristics. However, these are preliminary results, and more experiments are necessary. The experiments provided an important feedback for VWs usability heuristics (and the associated checklist) refinement.

VI. CONCLUSION AND FUTURE WORKS

VWs have nowadays a wide range of applications. Research usually focuses on VWs' use and the phenomenon they represent. There is a need for new usability evaluation methods or at least usability evaluations should be particularized for VWs applications.

A set of 16 specific usability heuristics and an associated (49 items) usability checklist were developed. Early validation proved their usefulness and potential. However, more experiments are necessary.

A right balance between specificity and generality should be follow. If heuristics are too specific, they will probably become hard to understand and hard to apply. General heuristics, complemented by specific usability checklists, will probably work better, most of the time.

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Advertising in Social Networks: Business-oriented Check-ins

A new business model and mobile service

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***Abstract* — This paper describes a new model for check-in process: customized (or business-oriented) check-ins. This approach introduces a new mobile service that lets any business publish customized records (statuses) in social networks (Facebook in the current release) in exchange for some benefits (discounts, gifts, coupons) provided for the customers. For the business, this service introduces a new way for advertising in the social networks. For the consumers, this service introduces a way for exchanging access to the own social graph for some benefits (e.g., gifts, discounts, coupons).**

Keywords — check-in; social network; mobile; HTML5; QR-code; coupon; Facebook

I. INTRODUCTION

Social networking has become very popular in nowadays' society. People will sign up for multiple social networking websites like Facebook, Foursquare, Twitter, etc.

At this moment, one of the popular models for the modern internet services (especially – for geo services) is "check-in". Check-in is a record with presence status, associated with some particular location. For example, in Foursquare users "check-in" at venues using a mobile website, text messaging or a device-specific application by running the application and selecting from a list of venues that the application locates nearby [1]. Each check-in awards the user points and sometimes "badges". This special status message is published in user's timeline (e.g., on the wall in Facebook).

Points are awarded for "checking in" at venues. Users can choose to have their check-ins posted on their accounts on Foursquare (communication service), Facebook (social network) or both. The common practice is to duplicate (translate) records in communication services to the social networks. Users can also earn badges by checking in at locations with certain tags, for check-in frequency or for other patterns such as time of check-in. Foursquare (communication service) was created with a core set of 16 badges, designed to reward and motivate all of users. The company has stated that users will be able to add their own custom badges to the site in the future.

So, earned points (badges) are finally the things users can exchange with some benefits at venue (e.g., discount, free offering, gift, etc.). Of course, it is true only if/when business that owns this venue participates in Foursquare's programs. In other words – venue owner participates in the program, developed by the communication service. It is very important for our future development. Let us just highlight again an important note for our future discussion – badges are being developed by the communication service, not by the business. And any customization here is actually an agreement between the business and communication services. The business (the source for rewards) is actually not free in the forms these rewards will be presented for business users. The format for the conversations is dictated by the communication service.

Figure1. Check-in record

The rest of this paper is structured as follows. We present the background to our QRpon mobile service in Section 2. In Sections 3 and 4 we present QRpon model and algorithm. We discuss the future work in Section 5 and conclude the paper in Section 6.

II. RELATED WORKS

The similar model is actually reproduced by the various implementations of "Places" services: Twitter places, Facebook places, etc. For example, as per official page, Facebook places let you easily share where you are, what you're doing and the friends you're with right from your mobile. You can check-in and your updates will appear on the Place page, your friends' News Feed and your Wall [2]. The business model is similar. You can check-in to get individual discounts, share savings with friends, earn rewards for repeated visits, secure donations for good causes, etc. Figure 1 shows a typical example for the check-in record:

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But here is at least one serious remark that remains true for all these services. All they are communication services at the first hand. And any check-in at the first hand solves communication tasks: how to let my friends/followers know where I am. The biggest question that remains is very simple and natural. Why do we ask business to deliver benefits via advertising some 3-rd party service? It looks very natural to let business define the format that should be used (exchanged) for benefits. It is actually the main idea our QRpon mobile service [3] was born from.

In this paper, we present QRpon– a new mobile service aimed to help connect sales with social networks. We are going to replace generic check-ins with the customized analogues. We are talking here about the check-ins used for the benefits delivery. That is why we have the words "business-oriented" check-ins in the title. We are not going to replace (dismiss, etc.) existing check-ins, this service also does not touch locations, etc. We are talking below about the special (user generated actually) forms for check-ins, that could be used in exchanges for benefits.

What are the advantages of suggested approach, what is missing the current models and why do we think it is a new model. Currently all the monetization efforts in LBS applications [4] like Foursquare, Facebook Places, etc. are based on the attaching (connecting, linking) some deals to the locations. Each deal (badge) is a position at the first hand. It is simply due to originality of the above mentioned systems. They are LBS (location based systems) applications originally. But it is very easy to conclude, that in the real case any business could offer several deals within the same location simultaneously. So, actually we should check-in in the deals (products) rather then in the places. On the other hand, the ability for the businesses to create any "deal" (discount, advertising) by themselves is obviously a big advantage too. They do not need to wait for any external development, deals could be created on the fly, etc. The open API for QRpon system lets create mobile sites (deals) automatically, getting data right from corporate ERP system.

The ability to collect stats directly from social network related to QRpons is the next big advantage. And we can note again as a big advantage the fact that statistical data is related to the products rather than to the places.

QRpon does not introduce any new social network, does not introduce any new authorization system, etc. Having a Facebook account is fully enough for using QRpon.

Lastly, QR-codes [5] simplify check-in process.

III. THE QRPON MODEL

Let us start from the business side. Here QRpon offers a specialized CMS (content management system) that lets any business create a special mobile web site. This web site lets users automatically, just after confirming the identity, post business-defined information on the Facebook's wall. In the exchange for this posting (action) mobile web site will show a confirmation for the benefits. For example, coupon, discount info, etc. In other words – anything that could be presented to the staff on the business side for claiming the benefits. All elements in this approach are user-defined: the offer, check-in info (what should be posted to the net), confirmation (badge) and even the rules for presenting badges.

How to present this mobile site for the potential users? It is where QR-codes help us. CMS lets businesses create mobile web site and an appropriate QR-code. Because it is mobile web (HTML5) application there is no need for downloading. Just scan QR-code and get URL opened.

Let us recall that HTML5 represents the cornerstone for modern Web [6]. There are several APIs for complex web applications. For example, we can easily add location info to our check-in process.

Automatically, this approach supports also physical check-ins. There is no way to mark you "at this location" being actually nearby (based on GPS location) or even far away (via API). QR-code should be scanned, and it is a physical action that could be performed on-site only.

So, for the business, this approach offered a mobile web site (sites – business can create more than one site, update them often, etc.), presented on-site with QR-code sticker, that lets visitors exchange posting in the social network (e.g., Facebook's wall) for some benefits. And all site's aspects (what is presented on the site, what should be posted to the social network, what should be presented as a confirmation) are defined by the businesses themselves. Another possible explanation from the business point of view – try to think about the current check-in system (e.g., Foursquare) and just replace the standard posting (notice) from Foursquare with your own text. Obviously your potential users do not need to download (install) mobile application and do not need to register in some new service (beyond their Facebook accounts). And another important difference from Foursquare (Facebook, Twitter, etc.) check-ins is the need for the physical presence.

The mobile CMS mentioned here is really simple. Practically, the business just needs to provide 3 pieces of text: the description (text for the first page of future mobile site – our offer and place for Facebook Connect button), the text that should be posted to social network and the text that should be displayed on the mobile site after the posting. So, the mobile site itself has got just two pages: the offer and the result (coupon, gift/discount info, etc.). And the transition from the first page to the second pages posts data to social network.

CMS creates a mobile site as well as the QR-code for the link to that site. This QR-code could be placed anywhere on the business side (Fig. 2). So, for access to the benefits (coupons, discounts) potential users need scan it with own mobile phone. QR code usage is very natural here. As per Google, the mobile phone acts as a cursor to connect the digital and physical worlds [7].

Figure 2. QR-code for mobile site

This QR-code points to the generated mobile web site. As soon as QR-code is scanned it is just one click deal to open an appropriate mobile URL. The user will see the first page, created by QRpon CMS (offer). After that user can confirm (accept) this offer, using his/her Facebook ID. Mobile site uses Facebook Connect [8] here.

So, our initial screen (offer) is user-defined text (probably images) plus automatically added Facebook Connect button. Actually it is some like Figure 3 shows (a screen shot from the service test page):

Figure 3. First screen

As soon user's identity is confirmed, the status text (also defined previously in our mobile CMS) is posted to the user's wall. And as a confirmation for that, mobile web site shows its second (final) page. It is a confirmation for getting benefits (coupon, discount info, etc). It could be presented to the staff at the business side, etc. Figure 4 illustrates this (the same text is already posted to user's wall in Facebook).

Figure 4. Confirmation screen

So, all the steps in this process are:

a) completely defined by the business
b) do not use any intermediate site / service beyond the social network itself

It is a fully customized check-in process (or completely customized, user defined badges in the terms of exiting communication services).

IV. QRpon Algorithm

QRpon service lets business completely customize check-in procedures: define an offer, check-in text (post) and badge (confirmation). It lets business ask clients check-in through the customized mobile web application and get reward for that. Lets consumers post links (data) in social network and get bonus for that.

Get coupons (bonuses, prizes, etc.) right on your mobile just for telling your friends about it. Deliver coupons (bonuses) right to mobile users shared info about your business.

Thus we have: bonus for link, on-site check-in for coupons, links building in the real world. Ask visitors to endorse your business in the social networks and deliver bonuses (coupons, discounts) for them. Just a simple sequence of steps: create mobile web site with your offer, prepare QR code for it, put QR code on your desks, walls, product wrap, etc. Ask users to scan QR code (check-in) in order to get confirmation for the reward. All the software you need is here.

Algorithm for business:

1. Create a mobile web site with your offer (coupon, discount info, prize, etc.). All the software you need is here. Just fill a simple form

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2. Basically you need the following things for creating your mobile site:

- your offer
- text for publishing in the social network
- response (confirmation) for the endorsement (coupon info, discount, prize, etc.)

3. Create a site (web page) and QR code for it
4. Test this QR code and the whole process right from your desktop and/or mobile
5. Print QR code and use it on-site. Place it on the table, wall, product's wrap, etc.
6. That is all. Ask your visitors/clients to scan this code for the reward. As soon as he/she scans your QR code and login to the network, your text is getting published in user's social circle (wall, timeline). And as a confirmation for that our service provides the above-mentioned response (badge). This response (data on the mobile screen, mobile web page) could be presented to staff at business location and used for claiming benefits.
7. Create a new site (a new check-in procedure) for new product (offer, etc.). See step 1 above.

Algorithm for consumer.

1. Make sure you have QR-code reading application on your phone (it is free).
2. Scan QR code from your iPhone, Android, etc on-site.
3. Open suggested mobile web page.
4. Login with your Facebook ID (check-in)
5. See bonus (discount, prize, gift, etc.) info right on your mobile
6. Show this info to the staff on-site and claim your benefits

Thus, we have win-win case: business get social networks marketing, consumers get rewards.

V. Discussions and Future Work

What could be added here in the future versions (or as some premium service)? This service at the moment of posting data to the social network has got access to the user's social graph. And it is very important moment. It means, particularly, that we can program output (our confirmation page) depending on the social graph size for example. For example, the more friends our customer has, the more potential readers will see our posting. So, the benefits could be increased for example, etc. Actually we can make our final screen (confirmation) depending on the social graph data.

In other words, the confirmation screen generation might be actually some production (rule based) system. It could be a set of rules (productions) like this:

IF (some condition) THEN (some conclusion)

where conditional part includes a set of logical operations against user's social graph data and conclusion is our output (coupon, gift confirmation, etc.). So, we are going to say here, that our system could be actually some sort of expert systems (production based) that generates conclusions (badges) by the social graph defined conditions. And our store for rules will present if the future versions some implementation of well known Rete algorithm [9-12].

The Rete algorithm is an efficient pattern matching algorithm for implementing production rule systems. The Rete algorithm provides a generalized logical description of an implementation of functionality responsible for matching data tuples ("facts") against productions ("rules") in a pattern-matching production system (a category of rule engine). A production consists of one or more conditions and a set of actions which may be undertaken for each complete set of facts that match the conditions. Conditions test fact attributes, including fact type specifiers/identifiers.

Also, because the output (badge) is actually programmatically controlled, it is very easy to add some randomness. So, our output (badge) could be randomly varied.

Another interesting stuff for the future development is open API for this QRpon service. It lets any business prepare custom check-in sites right from ERP system for example. Collect some stuff (items) from ERP database for rewards and create mobile web sites (check-ins) for them programmatically.

The next important thing here is the statistics. Obviously, that can collect the traditional web statistics for our mobile sites. For two pages site the most important value is the bounce rate – how many visitors do not accept the offer. But it is also obvious, that the proposed approach lets us accumulate an interesting statistics for the business from the social networks point of view. Facebook API has got a TOS (terms of services), we can not simply log raw data, but even the accumulated info could be very interesting. Just because our application gets access to the basic social graph data at the moment our user accepts an offer we can accumulate for example sex-age histogram for our buyers, etc.

In the same time, TOS for social networks API let us keep ID's for users. Just for keeping that info (without any data) we can easily discover new and returning user and easily implement such feature as "Majors" - users with the most visits, or users with the most visits within the given interval. In other words all the functionality (related to the business delivery) check-ins in the modern communication services could be provided directly. We simply do not need any intermediate communication service (e.g., Foursquare) for that.

And of course, the above-mentioned statistics could be processed programmatically too and could be used in the badges-generation expert systems. For example, the output (badges) could depend on the current (past) performance too.

VI. Conclusion

We presented a new service model for customized check-ins. This model lets businesses provide own forms for badges exchangeable for some benefits (e.g., discounts, gifts, coupons) without the external communication services as well as own rules for badges generation. It provides actually a new approach for advertising in the social networks. In this approach check-in records posted as exchange for some benefits (e.g., discounts, coupons, gifts) playing the role of ad messages. Finally, QRpon model presents mobile service that lets any business exchange some benefits for the clients with posting (advertising) in the social networks.

This service does not require downloadable mobile applications and based completely on the mobile web (HTML5). Via extensively used QR-codes this service builds a bridge between the virtual world of social networks and traditional retailing.

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QoS Support in UMTS Networks

Performance Evaluation and Perspectives towards an Autonomic Resource Management

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***Abstract*—In order to enhance the QoS support inside UMTS network architecture, this paper brings the perspective of a QoS support for self-managing resources based on knowing the service requests and the network capabilities. The originality of this approach lies in the perspective of certain integrated functionalities that will provide autonomy for the components of the managed system in terms of internal decisions and configurations. The analysis of the results will highlight the benefits of an autonomic resource management mechanism compared with native UMTS QoS support.**

Keywords-UMTS; QoS support; autonomic management.

I. INTRODUCTION

The success of Internet architecture, as evidenced by the variety of many types of applications and network technologies, has proven its centrality through the way it has influenced, and often determined the daily life. Internet's ubiquity has brought a number of shortcomings that the current architecture cannot solve, hence a multitude of solutions to approach problems as addressing, routing, congestion, resource management or traffic precedence have been developed.

We can distinguish two major trends in the way of the scientific community understood solving this set of problems, namely: the complete remodeling of the Internet architecture (i.e., 4WARD [1], AUTOI [2], 4D [3], GENI [4]), respectively the gradual improvement of functionalities in the existing architecture [5] (i.e., Self-NET [6]).

The complete remodeling of the Internet architecture offers a purist approach, a clean slate kind of modeling the new architectural elements. On the other hand, the gradual development of network architecture, ruled by a pluralistic approach, considers that the leap towards a new Internet architecture indifferent to the existing technologies is impossible.

Promoting the functionality promised by a clean slate approach (flexibility, reliability, fault-tolerance, autonomy and manageability), the authors of this paper believe however that passing to an architecture that will integrate all these features is progressive, at least for two reasons: the perspective of operators and Internet service providers on radical changes in the network and the difficulty in testing, evaluating and validating the proposed new architectural elements.

Beyond the need for new legislative and normative agreements between Internet service providers and network operators, agreements required by fundamental architectural changes, a major issue in the revolutionary innovation of the Internet architecture is the difficulty of assessing the new concepts in real experimental scenarios.

In [7], Peterson disputes the promotion of new architectural ideas, calling the scientific community to test the proposed solutions in the experimental testbed sites (i.e., PlanetLab platform with the Measurement Lab "M-Lab" back-end platform [8]) validation site completely different from what means the evaluation by simulation or emulation.

Although the reality of testing on an experimental platform is undeniable, just not to focus the proposed solution on an single extremely narrow issue, the authors of this paper believe that prior to live testing phase there are several steps that must be completed by simulation and emulation, namely: monitoring and highlighting critical situations to identify network problems, testing the effect of local parametric adjustments on the whole system, development and gradual integration of scalable features in a new architecture. Therefore, stepping towards a revolutionary architecture is a matter of time; the new capabilities added to the existing architectural elements represent the prerequisites for success in this matter. Because of this, we believe that it is impossible to jump towards an architecture which is independent of the existing technologies, the argument of this motivation being found in the evolutionary pluralist concept.

Starting from these premises, which combine the requirements of a clean slate paradigm with current technological reality, the paper aims at investigating and testing the benefits of integrating autonomic resource management capabilities into UMTS (Universal Mobile Telecommunications System) architecture to support QoS.

Besides analyzing the QoS parameters like average end-to-end delay, throughput or average jitter experienced by time-critical applications in the UMTS radio access network [14], the paper enhanced and extends the QoS support even

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to the UMTS IP sub-network segment of through virtualization.

As the UMTS QoS support acts only on the radio access network, by knowing the requirements of the source application, a best end-to-end performance could be offered through network virtualization in the core network. The analysis tool is QualNet 5.0.2 network simulator [13].

Thus, while in Section 2 the capabilities of the UMTS network to provide a QoS support to applications that have stringent requirements concerning the time component are tested, in Section 3 the premises of an autonomic resource management that ensures a higher quality support in the network are investigated. This fact is revealed by the ability to select an alternative route between entities SGSN (Serving GPRS Support Node) and GGSN (Gateway GPRS Support Node) based on knowing the source application requirements. Finally, Section 4 presents the conclusions of the realized study by showing the perspective of an autonomic management of network resources that is based on the conjunction of application requirements and network context.

II. QoS Support in UMTS Networks

In order to achieve a certain QoS support, UMTS network has defined a so-called "Bearer Service". A bearer service includes all aspects needed to enable the provision of a contracted QoS. These aspects are among others the control signaling, user plane transport and QoS management functionality. Various types of bearer service were established between different parts of UMTS network. More than that, each bearer service on a specific layer offers its individual services using services provided by the layers below. It is worth mentioning that bearers covering only a certain part of a system and being closer to the physical connection always have more stringent QoS requirements.

In order to solve the QoS problem, UMTS defines four types of traffic classes: Conversational (CO), Streaming (ST), Interactive (IN) and Background (BK).

The main difference between these QoS classes is the transfer delay value. Conversational QoS Class includes real-time applications that require stringent limits for delay value, while Background QoS Class is the most delay insensitive traffic class. Table I shows the main characteristics of the above mentioned QoS classes and examples of corresponding applications.

TABLE I. Main Characteristics of UMTS QoS Classes

Traffic classes	Characteristics	Application
Conversational (CO)	low delay, low jitter, symmetric traffic, no buffering	speech, VoIP, video
Streaming (ST)	moderate delay, moderate jitter, asymmetric traffic, buffering allowed	video streaming, audio streaming
Interactive (IN)	moderate jitter, asymmetric traffic, buffering allowed, request response pattern	web browsing
Background (BK)	destination doesn't expect data within a certain time, preserve payload content, asymmetric traffic, buffering allowed	email, file downloading

In order to define the traffic characteristics, UMTS architecture introduces a set of QoS attributes. It must be mentioned that a particular type of traffic class is itself a QoS attribute.

There are attributes specific to all classes (i.e., maximum bit rate, delivery order, maximum SDU (Service Data Unit) size) and some attributes that are applied only for a specific class (i.e., transfer delay is applied only for conversational and streaming classes; traffic handling priority is applied only for interactive class).

A. *Performance Evaluation of the UMTS QoS Support*

In order to evaluate the QoS support implicitly provided by an UMTS network, a scenario including a PLMN (Public Land Mobile Network) was simulated using QualNet 5.0.2 network simulator, a widely used platform in the defense and telecommunication network design and evaluation [13].

The network scenario includes eight UE (User Equipment) nodes: four source nodes (UE nodes 6, 8, 10, and 12) and four destination nodes (UE nodes 7, 9, 11, and 13), as presented in Fig. 1.

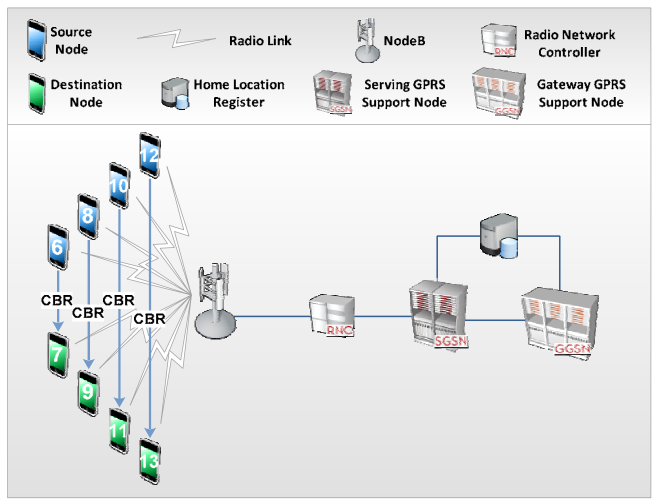

Figure 1. UMTS evaluation scenario

Global parameters configured at the physical (PHY) network layer of the simulation are given in Table II. It should be noticed that two different radio channels were used in order to access the network resources. The frequency values of these channels were chosen accordingly with European 3G bands for UMTS 2100 recommendations [11].

TABLE II. PHY Layer Configuration Parameters

Parameter	Value
Up-link channel frequency	1.95 [GHz]
Down-link channel frequency	2.15 [Ghz]
Path Loss Propagation Model	Two Ray
Shadowing Model	Constant
Shadowing Mean	4 [dB]
Propagation Limit (UE Sensitivity)	-150 [dB]

Between the source nodes (SN) and destination nodes (DN), four CBR (Constant Bit Rate) applications, each of them corresponding to one QoS class defined by the UMTS network, were considered. The characteristics of all applications are summarized in Table III.

TABLE III. CORRESPONDING QOS CLASS FOR EACH CBR APPLICATION USED IN THE EVALUATION SCENARIO

Application Type	SN	DN	Items to Send	Item Size [bytes]	Interval [s]	QoS Class
CBR	6	7	1000	32	0.1	BK
	8	9				IN
	10	11				ST
	12	13				CO

The results of the simulations concerning average end-to-end delay, average jitter and throughput for each QoS supported class are synthesized by Table IV.

TABLE IV. UMTS QOS PARAMETERS EVALUATION

UMTS QoS Class	Average end-to-end delay [s]	Average jitter [s]	Throughput [bits/s]
BK	3.493082507	0.068259929	2699
IN	0.180157479	0.053124915	1420
ST	0.130232583	0.022149065	1412
CO	0.103871283	0.016113024	2076

The average end-to-end delay results for each QoS support class are presented in Fig. 2 and the results regarding average jitter are illustrated in Fig. 3.

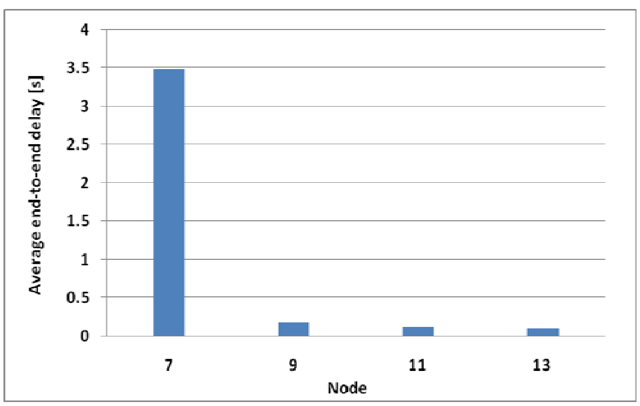

Figure 2. Average end-to-end delay experienced by the test applications in each QoS support class

Analyzing the obtained results, it can be noticed that application which corresponds to the Conversational and Streaming QoS class is characterized by the lowest value of average end-to-end delay and jitter delay, as expected for the type of application corresponding to this class (speech, VoIP, video or audio streaming).

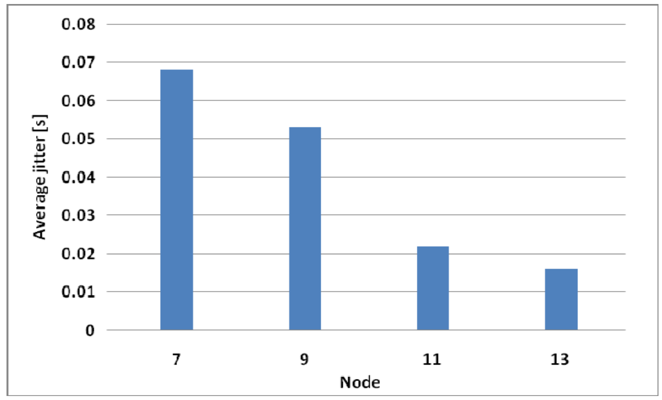

Figure 3. Average jitter experienced by the test applications in each QoS support class

If throughput values are compared, it can be seen that the application from Background QoS class has the highest value of this parameter, but it also has the highest value for the average end-to-end delay and the average jitter.

Summarizing, the network complies with the priority level imposed on the test applications that were modeled by CBR-type traffic sources, and this is reflected in the average values of the end-to-end transmission delays and in the jitter for each QoS priority class.

Correlating end-to-end transmission delays with the transmission flow rate and with the time interval between transmitted packets one can notice the increased flow rate in the case of Background class, which is due to transmitting a reduced total number of packets in a very short transmission time interval.

III. PERSPECTIVES TOWARDS AN AUTONOMIC RESOURCE MANAGEMENT IN A UMTS NETWORK

Developed within the 3GPP (3rd Generation Partnership Project), the 3G standard suggests an end-to-end QoS support based on a policy management system (Policy-Based Network Management) [9].

The network architecture presented by the first 3GPP public versions has evolved to an architectural model SAE (System Architecture Evolution) [10], which ensures the convergence of different access network categories such as UMTS, 3GPP, WLAN (Wireless Local Area Network) or any other non-3GPP radio access technology.

The latest public 3GPP version considers the IMS (IP Multimedia Subsystem) [11] network architecture to be completely separated from the access technology, having the specific access functions completely isolated from the core network.

In order to manage QoS resources, the SAE architecture integrates an informational QIF (QoS Information Function) function that interacts with all individual network models included within the IMS platform. This QoS resource management solution is an external part of the network, based on the use and interaction between a central entity and periferic elements.

Although in the clean slate approach the resource management and the QoS support are considered an

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integrated part of communications networks, this fact is not reflected in the characteristics of current systems or by the UMTS network architecture.

Therefore, the perspective of an autonomic resource management in an UMTS network proposed in this paper suggests the necessity of adding additional information at the level of the central UMTS network elements using virtualization technique.

Network virtualization represents a high level abstraction process that overlaps the implementation and physical network configuration details. Allowing co-existence of multiple virtual architectures overlaid on a common substrate physically shared, network virtualization promises flexibility and security, promoting diversity and increased management capacity [12].

In this way, UMTS core network nodes act autonomously, being able to sense the environment, to perceive the changes, to understand internal changes and to react in an intelligent manner by selecting the optimal path according to application requirements.

To demonstrate this, native UMTS QoS support is analyzed by comparison to the potential of the autonomic management offered through network virtualization, using QualNet network simulator [13].

Thus, in the case of an autonomic management system, the proposed analysis scenarios highlight the ability of selecting the best route according to the source application constraints in terms of maximum acceptable end-to-end delay.

A. *Scenario description*

According to [15], it is possible for an UMTS network to have multiple SGSNs and GGSNs entities which can be co-located or can be interconnected via an IP subnetwork in order to increase the geographical area served by an operator.

Considering the second approach, it was developed an evaluation scenario in which the SGSN and the GGSN are interconnected via a simple IP sub-network that consists of ten generic routers denoted R1 to R10, as depicted in Fig. 4.

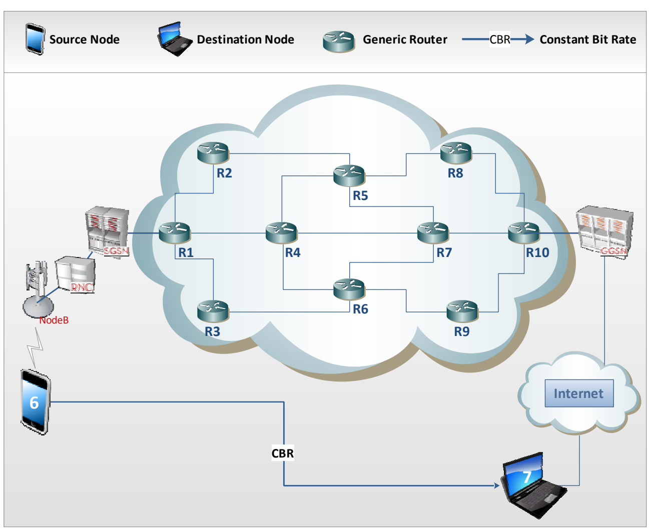

Figure 4. The architecture of the evaluation scenario

Obviously, in a real-life UMTS IP sub-network would be more complex, the motivation for this topology was to better illustrate the problems that may arise and the proposed solution.

In this evaluation scenario a CBR test application corresponding to Conversational QoS class (highest priority QoS class) was considered. Main parameters that describe the characteristics for the tested application are indicated in Table V.

TABLE V. CHARCTERISTICS OF THE MODELED APPLICATION

Application Type	SN	DN	Items to Send	Item Size [bytes]	Interval [s]	QoS Class
CBR	6	7	1000	32	0.1	CO

As we have already mentioned, the motivation behind this evaluation is to highlight the benefits of the autonomic management offered through network virtualization process.

In our case, the virtualization process could be illustrated by controlling, through the configuration files, the parametric values that characterize network architectural elements.

Knowing the values of these parameters, it is possible to indicate a dedicated virtual network that offers the best path form the source towards destination in terms of minimum average end-to-end delay, maximum throughput, packet loss rate on selected route or other stringent requirements specific to a certain type of application.

B. *Performance Evaluation*

In order to emphasize the path selection mechanism between SGSN and GGSN, core nodes of UMTS IP sub-network, two different cases were evaluated: a native UMTS QoS support selection compare to QoS network support provided by an autonomic network management selection, as a result of dedicated virtual network generation.

When it defines the links between two intermediate generic routers in the UMTS IP sub-network, the simulator allows the association of specific transmission throughput and delay on each link, as presented in Fig. 5.

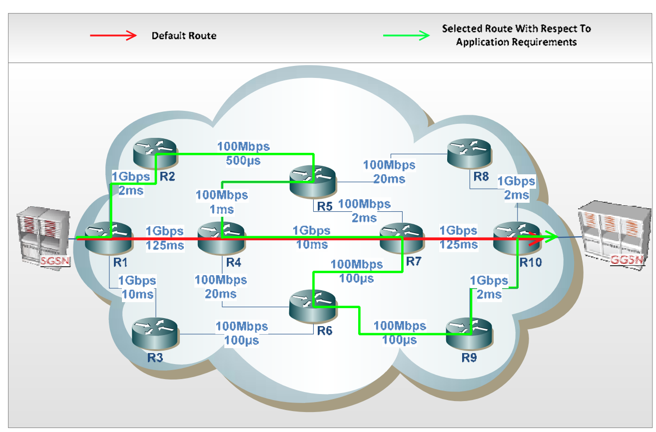

Figure 5. UMTS IP sub-network path selection

Broadcasting the test application scheduled as Conversational (CO) by the native UMTS QoS support in the access network, the default path from router R1 to router R10 in the IP sub-network was selected via routers R4 and R7, as illustrated in Fig. 5. This path selection was based only on the routing protocol mechanism (i.e., minimum hop count).

An autonomic network management QoS support should consider and accommodate both differentiated application requirements and dynamic network context. This mandatory integrated capability of the Future Internet (FI) network elements is performed based on network virtualization.

The objective of network virtualization process is to generate virtual networks and make each of these virtual networks appear to the user as a dedicated network infrastructure, with dedicated resources and services available for application requests. Therefore, the network virtualization process invokes a mode of selecting the virtual network that best integrates and satisfies the application requests at the physical network level.

It must be mentioned that for the generated virtual network, in our case, only the value of average end-to-end delay and jitter are considered as critical parameters for the source application.

Results of the simulations validate a virtual path from the router R1 to router R10 via routers R2, R5, R4, R7, R6, and R9, a corresponding physical network infrastructure that offers best performances in terms of requested average end-to end delay and jitter.

Parametric results of the average end-to-end delay, average jitter, and throughput, both for native UMTS QoS support and autonomic resource management QoS support, are summarized in TABLE V.

TABLE VI. PARAMETRIC EVALUATION OF QOS SUPPORT

Selected path in UMTS IP sub-network	Average end-to-end delay [ms]	Average jitter [ms]	Throughput [bits/s]
Native QoS support in UMTS access network			
R1→R4→R7→R10	307.19	21.26	2580
Autonomic resource management based on network virtualization			
R1→R2→R5→R4 → R7→R6→R9→R10	57.42	20.79	2575

As illustrated in Fig. 6, the UMTS IP sub-network decisively influences the applications performances in terms of average end-to-end delay and jitter.

Simulations results validates the perspective towards an autonomic resource management QoS support that makes the conjunction between the application's requests and the network context by choosing an alternate route in a virtualized environment.

The potential of the autonomic resource management is reflected by the capacity to identify an optimal path between the source node and the destination node, according to the imposed QoS constraints.

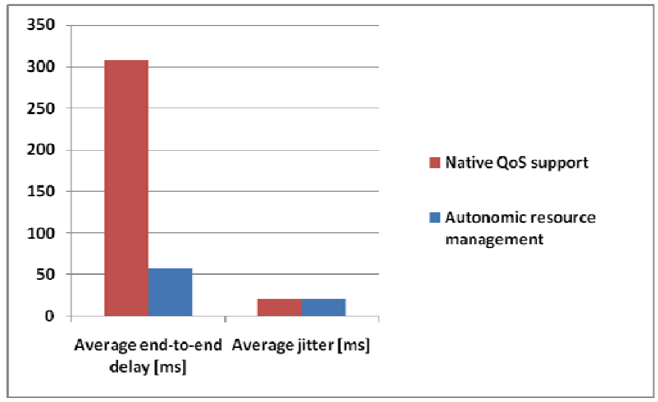

Figure 6. UMTS IP sub-network path selection

Some critical conditions or cost constrains could limit the resource optimization while using the QoS support only on the UMTS radio access network. These situations request for adaptation of the source application parameters to the network available resources in order to still broadcast the information on the channel.

An implemented and tested solution (i.e., virtualization algorithm, virtualized parameters, in-network message flow, and source adaptation) for the network virtualization concept is offered in [16]. The conjunction between the source application requests and the network context is achieved through the QoS profiles message exchange and it represents the process of network virtualization.

As long as this solution uses Multiprotocol Label Switching (MPLS) for route maintenance, it could be applied just in the UMTS IP-sub network. It could not offer an end-to-end QoS support because of the UMTS radio access which does not offer MPLS support. For the further work, the authors intend to extend this solution also for the radio access network in cellular networks.

As in [16], the paper assumes the network overflow by probing each UMTS IP sub-network link. In spite of that, the potential of the autonomic resource management is reflected by lower average end-to-end delay and lower jitter.

IV. CONCLUSIONS

This paper aimed to investigate the potential benefits that could reside from the integration of the virtualization process in current UMTS IP sub-network architecture.

In a first stage, the upper limit of native UMTS QoS support was analyzed in terms of average-end-to-end delay, average jitter, and throughput. Then, the efficiency of this QoS support was compared to the QoS support resulted from the usage of autonomic resource management implemented through network virtualization.

The obtained results suggest that such an approach could enhance the native UMTS QoS support by assuring an autonomic resource management that will allow overcoming the situations in which the existent QoS support would not permit even the service itself.

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M^2ANET Performance Under Variable Node Sleep Times

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***Abstract*—We investigate a new variant of ad hoc networking: a network of nodes whose sole function is to forward data in the network. This model differentiates between terminal nodes and transmission forwarding nodes in a network. We investigate the performance of the network under the condition when the forwarding nodes periodically enter the sleep state i.e., are turned off periodically. The results show the relation between the expected QoS (quality of service) and the duration of sleep time of a node (also a tradeoff between QoS and network lifetime). The higher the network node density, the lower the impact of introducing sleep state on QoS. ns2 simulation shows a M^2ANET network with sleeping nodes running the AODV routing protocol performing better than one running DSR for a range of node densities.**

Keywords- MANET; transmission medium; quality of service QoS; node sleep mode.

I. INTRODUCTION

An ad hoc network is a collection of wireless mobile nodes forming a temporary network without the aid of any stand-alone infrastructure or centralized administration. An ad hoc network with a changing topology due to node movements is called a Mobile Ad Hoc Network (MANET) [1]. In a conventional ad hoc network, the nodes not only act as hosts but also assist in establishing connection by acting as routers that route data packets to/from other nodes in the network. One can also propose a variation of an ad hoc network where mobile nodes (blue) would be used for forwarding only in order to support communication between designated terminal nodes (red in a large circle in Figure 1).

In [2], we defined the principles of operation of a mobile network made of forwarding nodes and called it M^2ANET: Mobile Medium Ad Hoc Network. The network nodes form a "cloud" of communicating nodes through which all data travels, not unlike sound waves traveling through air, thus forming the medium for data transmission (propagation). In this paper we test a new hypothesis that stipulates that a reliable transmission through a mobile medium (in a M^2ANET) does not depend on the continuity of operation of any one network node. If our new hypothesis is true, it would have significant implications on operations of ad hoc and sensor networks. In particular, one could create a M^2ANET using nodes with a limited battery capacity that would operate reliably for as long as new nodes are added to the network at the same rate as other nodes run out of battery power. In another scenario one could envision a network composed of nodes that switch on and off periodically (i.e., go to sleep) and, given sufficient number of nodes, such a network would still be able to reliably transmit data. Nodes that switch on and off periodically would not only conserve battery power but also would be more difficult to locate (hack or destroy).

In Section II, we introduce M^2ANET and then discuss using nodes with on/off sleep states in Section III. Experiments with different numbers of nodes, different probabilities of entering sleep state, and different routing protocols are described in Sections IV and V. Finally, we present the conclusion and future work, in Section VI and Section VII, respectively.

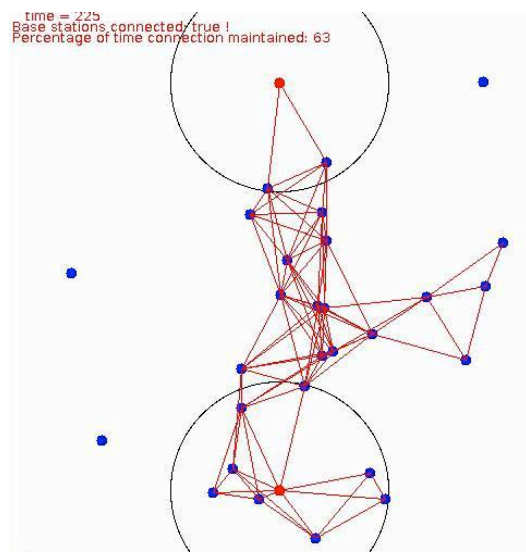

Figure 1. M^2ANET in operation: communication channel is established between two terminal nodes (red) using a network of mobile nodes (blue).

II. STATE OF THE ART

Performance of a MANET is impacted by many factors including node range, node mobility, number of nodes used in a network, and the protocol used [3-7]. A good example of protocol performance evaluation is in [5]; our own tests are reported in [8].

We find it interesting that the node density issue specifically and its impact on network performance was a subject of few studies. In [9][10] authors distinguish between

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a physical node density versus connectivity density. The performance of the MANET depends not only on the number of mobile nodes in a particular area, but also on how likely the nodes make connections. Bettstetter and Zangl derived the expected degree of a node for n nodes with range r_0 operating in an area A [11]. Their objective was to determine the required number of nodes to achieve an "almost surely connected network", i.e., an ad hoc network with a low probability of having isolated nodes. Based on discussion in [11], we provided our own definition of normalized node density [2][8]:

$$\delta = n\, c/a, \tag{1}$$

where n is the number of nodes, c is the transmission area covered by a node and a is the area over which the mobile network is deployed.

In our own studies we separated the network nodes into two categories: the terminal nodes and the communication nodes. In this scenario a "cloud" of communication medium nodes forms a medium through which the communication channel can be formed. With a sufficient number of nodes in the area a communication channel is formed across the mobile nodes of the M^{2}ANET allowing for designated terminal nodes to communicate. We call such a network a **M**obile **M**edium MANET, or M^{2}ANET (pronounced " *MANET square(d)*"). The main task of the M^{2}ANET is to establish communication between the terminal nodes (and not necessarily to link all other MANET nodes into a connected network). This is what makes M^{2}ANETs different from typical MANETs: as opposed to the MANET scenario described in [11], in M^{2}ANET some mobile nodes may become isolated without a detrimental effect on the network performance.

In [2], we showed how the different QoS criteria favour M^{2}ANET over a conventional MANET. In [8] we showed a slight advantage of AODV over DSR when used in M^{2}ANET.

III. NETWORK OF NODES WITH VARIABLE SLEEP TIMES

The main role of the network is to establish a route and to transfer the data. In a non trivial MANET only some of the mobile nodes would be needed to maintain the route (channel) at any given time, with other nodes not participating actively in the data transfer. MANET routing protocols take care of reestablishing the route during the initial set up and in case of a node or a link failure. In a MANET, given a sufficiently dense network, any routing node can be easily replaced with other nodes forming an alternative route. The non essential nature of any routing node suggests that they form a vast cloud of nodes serving as a transmission medium.

One can take advantage of the non essential nature of any one node by allowing for a node to be switched off (or removed) and still maintaining communication through the network. The nodes can be switched off in a coordinated fashion or at random [12]. Our work is concerned with a random sleeping mechanism and, apart from an obvious scenario of evaluating network resilience in the event of a node failure, we propose a scheme in which some routing nodes are switched off and on intentionally. In addition to showing that the M^{2}ANET network can maintain its reliable operation with some of its nodes switched off, we show the tradeoff between network lifetime and quality of service QoS. In our paper QoS is defined using a single metric: the delivery ratio. The network life time is defined as the period over which there is a sufficient number of functioning nodes in a M^{2}ANET, and we assume that extending the sleep time of individual mobile nodes would conserve power and extend the network life time by a corresponding amount. At the same time, nodes in the sleep state would be unable to forward data in the network, which in turn is likely to affect network connectivity and QoS.

To establish the relation between sleep times and QoS we investigate a M^{2}ANET with n mobile nodes and operating in a confined space (a rectangular region). We vary number of nodes used in the experiment and the duration of sleep time of a node. Sleep time of a node is randomly selected from a uniform distribution. The status of a node is modified periodically (e.g., every second), and p_{on} designates the probability that the next state of the node would be the ON state, similar to the Randomized Independent Sleeping (RIS) introduced in [13].

Application of a M^{2}ANET with nodes with variable sleep times can be considered for many scenarios, especially when a network needs to be established quickly and without much preparation. Nodes could be dropped into a zone of interest from an aeroplane, or can spread (move away) from each terminal station. QoS can be traded for the network lifetime by adjusting the probability p_{on}.

IV. JAVA SIMULATION USING NODE DISTANCE ONLY

A Java simulation of a mobile network is used with two nodes at stationary positions and the rest mobile. The mobile nodes are confined to an area bounded by a 500x500 rectangle (both the size of the bounded region and the transmitter range are defined in the same units; for the sake of the discussion below we can assume this unit distance as 1 meter). The two stationary terminal nodes shown in Figure 1 are at the opposite sides at locations (250,62) and (250, 437). The simulation is run over 10^5 time units, with mobile nodes moving in straight lines, and randomly changing speed and direction every 25 time units. Each node can be in the on or off state, and capable of forwarding messages p_{on}% of the time. A connection is formed between a pair of nodes if the distance between them is less than the pre specified threshold (called range and equal to 100). We test, and visualize, for the connectivity between the terminal nodes in the network under different node densities, and for different sleep times.

The graph in Figure 2 shows the percentage of time the connectivity was maintained when (i) the nodes were switched ON only 60% of the time (p_{on} = 60%), 80% of the time (p_{on} = 80%), and (ii) when nodes were ON all the time (p_{on} = 100%), all for different number of mobile nodes used in the simulation. The number of mobile nodes ranges from 10 to 100, for the 500x500 area and the mobile node transmission range of 100 units, corresponds to the range of normalized node density δ of 1.6 to 16 [2]. The percentage is calculated as a ratio of the total number of time units with connection criteria satisfied (path exists between two terminal nodes), to the total time. The results show that increasing the node sleep time (i.e., lowering p_{on}) impacts on the connectivity in the network and has a similar effect to changing the number of nodes in the network (note the translation between two functions in Fig 2). The higher the node density (or number of nodes in the network) the lower the impact on QoS of introducing the sleep state at network nodes. It should be noted that at higher node densities (e.g more than 80 nodes over the area 500x500, or δ>15) the connectivity stays at 100% whether or not the network nodes enter the sleep state, as long as it happens infrequently (Figure 3). This suggests that using nodes with variable sleep times in networks with high redundancy (δ>15) would not affect QoS while potentially extending the battery life of a node, and consequently, the life time of the entire mobile network.

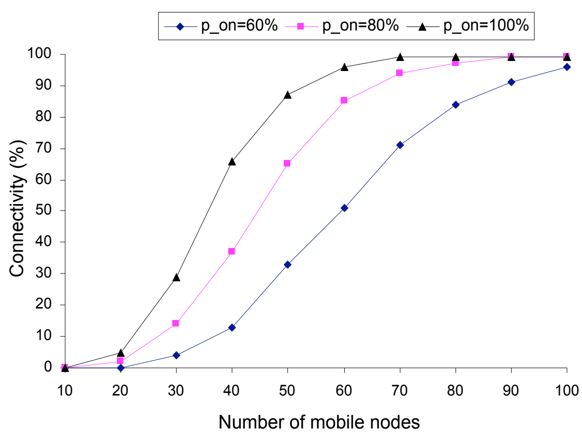

Figure 2. Comparison of M²ANET with variable sleep times, p_{on} at 100%, 80%, and 60% .

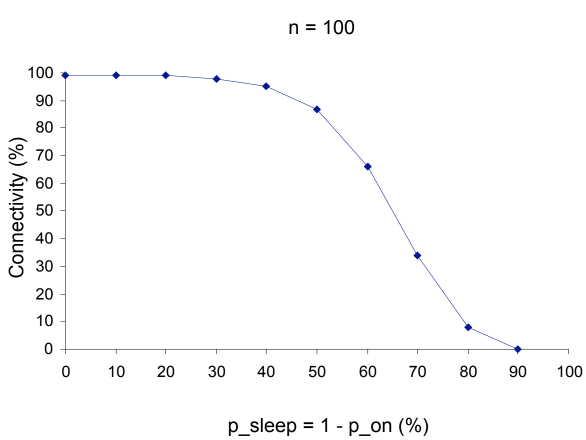

Figure 3. M²ANET performance reduction at increased sleeping times.

V. Comparison between AODV and DSR in NS2

While the Java simulation above takes into account the distance between nodes only, it does not account for the difficulties in reestablishing the routes when nodes switch on and off (i.e., enter sleep state) randomly.

We investigate the performance of two MANET protocols when used in a M²ANET scenario with variable node sleep times (p_{on} = 80%, ON/OFF status of a node reevaluated every 10 seconds). Each wireless mobile node uses the default 802.11 parameters and has a transmission coverage of approximately 125m x 125m. The random node mobility for the experiments is generated using a node movement generator "setdest" built into NS-2 [14]. The setdest application generates a node movement file using the random way-point algorithm. Data in the MANET are transmitted from the source to the destination node over a UDP connection at a constant bit rate (512 bytes sent every 0.1 seconds) and are generated using the NS-2 built-in CBR traffic generator. Two routing protocols are used: Ad-hoc On Demand Routing (AODV) [15] and Dynamic Source Routing (DSR) [16]. The number of nodes used in the ad-hoc network is varied from 30 to 70 (which corresponds to the normalized node density δ of 4.8 to 11.2 [2]). Each simulation is run for 50s and is repeated five times.

For performance comparison we define the delivery ratio as a ratio of the number of packets successfully received at the destination to the total number of packets sent from the source. The source is sending packets at a constant rate during the entire simulation experiment. As such the delivery ratio shows how successfully the connectivity between two nodes in a network is maintained (Figure 4). 100% packets received indicates that the connectivity was available all the time.

We attribute a slight advantage in the performance of the AODV protocol to the specifics of our experiment and to a different way the two protocols maintain and update the route information. Since we have only one source node and one destination in our network, it is only the source node that caches the route to the destination when the DSR protocol is used, while with AODV each node on the discovered route

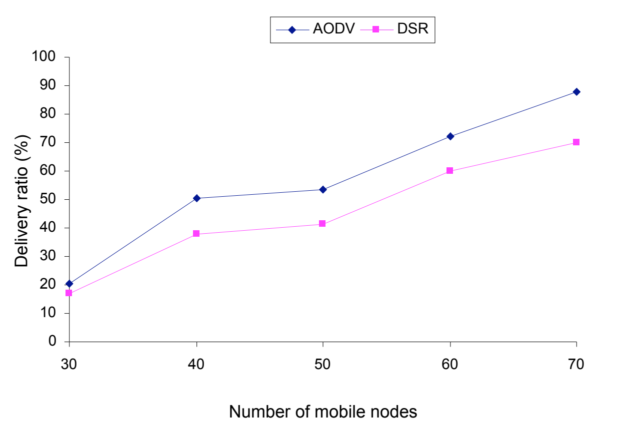

Figure 4. Comparison of AODV and DSR protocol performance for M²ANET with nodes with variable sleep times, p_{on} at 80%..

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maintains a routing entry. When a link break occurs due to a mobile node entering the sleep state (or simply getting out of the transmission range), AODV may be able to reestablish the route faster than DSR by reusing the existing routing tables at some of the nodes (down stream from the break, or by attempting a "local repair": sending RREQ from an intermediate node), resulting in a higher delivery ratio.

VI. CONCLUSION

We introduced and demonstrated a new view of the operation of a MANET, in which the mobile nodes form a "cloud" of transmission nodes, and act as a transmission medium. The performance metric for the M^2ANET is different form that of a typical network in that the transmission nodes are solely facilitating the transmission between the terminal nodes and are of no other value. The routing nodes that are not in the path of the signal between the terminal nodes do not influence the performance metric. The network can tolerate switching the routing nodes on and off periodically because the function supported by any individual node can be easily and automatically replaced by the routing protocol. The simulation shows that the performance of the network with nodes with variable sleep times is predicable: increasing the sleep time of a node has a similar effect as reducing the total number of nodes. We observed that in a network with many redundant nodes (high normalized node density δ) introduction of variable sleep times for each network node has negligible effect on QoS and over all network performance. The simulation showed that AODV is better than DSR in a network with nodes with variable sleep times (Figure 4).

VII. FUTURE WORK

More experimental work is required to investigate all aspects of performance of M^2ANETs with nodes with variable sleep times: in particular we plan to investigate the impact of the distribution of sleep times on M^2ANETs with different number of nodes. We also plan to investigate the performance of the M^2ANET theoretically, following the approach presented in [11]. We hope to introduce new routing protocols specific to M^2ANETs in the future. We would also like to investigate the role of the motion pattern of the mobile nodes (forming a "cloud" of transmission nodes) on M^2ANET performance.

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A Survey on Security in Future Internet and Cloud

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***Abstract*—The Internet was designed in the 1970s with limited scope and applications. It has been evolving during the last decades applying specific and ad-hoc solutions around the IP protocol to cover growing needs about security, mobility, interconnection, etc. Recent research has been increasingly focusing on the problem of Future Internet evolution; while one research line argues that a *clean-slate* approach is necessary to cover all future requirements, others maintain that Internet could continue to *evolve* adaptively, adopting new technologies as real requirements emerge. Increasing adoption of Cloud Computing paradigms could support the evolutionary approach. Beyond what would be the most valuable theory, security is one of the core issues future technologies must face. This paper gives an overview of Cloud Computing and Future Internet research issues and initiatives, oriented to security aspects. We analyze the common points regarding *Trust* and *Identity Management*, as well as identifying guidelines for future research.**

***Index Terms*—Future Internet, Cloud Computing, Security, Trust, Identity Management**

I. Introduction

Internet was designed in the 70's as a communication system between end-to-end machines targeting a community of users that could be considered experts. Thanks to the transparency of the design it has been quite easy to join new networks to the Internet's network-of-networks model. It permitted also the deployment of new services and applications leading to the well know hourglass model around the IP protocol.

Today's Internet scope is far from the original design. It has developed as a critical infrastructure for our society and economy and plays an active role in the daily life of millions of people. Internet has evolved from a limited academic scope to a mass phenomenon [1]. It has been taking more and more relevance in business and e-commerce since all processes have been significantly automated and improved through the usage of Internet technology. Concerning the user experience, Internet has influenced the evolution of computing paradigms from the mainframe to grid computing being the popularization of Personal Computers (PCs) an important milestone. Moreover, in the recent years we faced the popularization of service-oriented paradigms that have finally lead to the *Cloud Computing* approach that is envisaged to satisfy the constant demand of computing resource, data storage or software functionality [2]. Cloud computing brings to the Internet-of-services a high degree of flexibility and scalability.

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However, many aspects ave been traditionally deferred for later definition. Security is one of those aspects that were procrastinated. In fact, many early network protocols that are part of the Internet were designed without explicit security considerations [3], such as defining the security policies, managing and protecting identities, securing the interactions between heterogeneous systems, managing trust relationships between different administrative domains, monitoring and evolution of changing contexts, among others.

Thus it has become to agreement that a global re-design of the architecture may be needed pointing to the abstract concept of "Future Internet". The purpose of this paper is to summarize the most promising efforts concerning the definition of a reliable security framework for future Internet, since security is one of the tasks that are usually deferred during the definition of new architectures. Thus, we aim at bringing the reader a guide on security for the Future Internet that would help to bring robustness to the future network-of-networks.

The article is structured as follows. Some introducing notes on Future Internet evolution are given in Section III after the Section II, in which we define the basics of the cloud computing. In Section IV, we sketch out some proposals for Future Internet architecture. General security considerations and requirements are described in Section V. Finally, conclusions are given in Section VI.

II. Cloud Computing

Cloud Computing can be considered the prelude of Future Internet. Cloud computing is a new paradigm that offers scalability, reliability, availability when accessing resources across Internet. It is expected to abstract the details of the underlying infrastructure even when they are complex. Despite there is a lack of an accepted definition for this computing paradigm, Cloud Computing could be seen as the use of Internet-based technologies for the provision of services [2]. The term "Cloud" was originated from the way Internet is represented in diagrams. In general, the core concept behind Cloud Computing is Software as a Service (SaaS). Cloud Computing, and its complexity, born from squeezing or generalizing the SaaS concept to exhaustion. According to this, if in SaaS an application can be a service, also does the environment over which the application is executed, and even the hardware that executes the entire software. Following this reasoning, Cloud Computing is a resource aggregation of applications, components, frameworks that can be configured for serving several purposes.

When it comes to the user role, the interaction with Cloud Computing systems might be similar to already existing paradigms, in fact, Cloud computing can be seen as an evolution of the academia-oriented Grid Computing [4] or the next step in data center paradigm [2]. In a typical cloud infrastructure, a unique node, a huge "*black-box*" connection point, stays in the center of the configuration while a number of users connect with it to consume the services it offers. What really differentiates the cloud computing from traditional web service architecture is the type of services it provides. Three general types of services can be distinguished in the cloud:

a) Infrastructure as a Service (IaaS): At the basic level of abstraction there are providers who provide instantiation of virtual machines to their customers. These virtual machines are static configured and therefore have to be re-instantiated by the operator when the customer reach its limits.

b) Platform as a Service (PaaS): Moving up in the abstraction level, a provider can offer to its customers an entire environment, composed of virtual machines instantiated in the data center but invisible for the developer customers, where the programs are executed.

c) Software as a Service (SaaS): At the upper abstraction level, the providers offer entire applications to its customers. Those applications can be typically used as a desktop applications and offer storage and resources to the end user.

The combination of these services converge in three key factors for the success of the model: the illusion of infinite computing resources available on demand, eliminating the need for Cloud Computing users to plan far ahead for provisioning [5]; the possibility for companies to start small and increase hardware resources only when there is an increase in their needs; the ability to pay for use of computing resources when needed and release them when they are no longer useful.

III. What is Future Internet?

Besides Cloud Computing popularization could be considered the trigger for the definition of the Future Internet, the more Cloud Computing grows as operative technology, the more it disassociates from the definition of Future Internet becoming, as much, a component of it.

Nevertheless, at the time of writing this article, there is no "uniform" definition of what the Internet of the future will be. There exist several attempts all around the word that try to define it. The European Commission focused on research on Future Network in the Seventh Framework Programme (FP7) [6] while the US National Science Foundation launched the program Future InterNet Design (FIND) [7]. The National Institute of Information and Communications Technology of Japan launched in 2006 the AKARI project that aims to implement a new generation network by 2015.

Within a plethora of efforts and definitions, it is possible to recognize two different trends on Future Internet develop. On one hand we can find the EVOLUTIONARY approach. Followers of this trend found its view on the assumption that Internet is now a full commercial network and that the inertia introduced by operator investment and the lack of immediate gain for early adopters make the incremental enhance the only way of evolution. Some followers also point that most of the common problems, such as security, are not a problem of architecture.

On the other hand a more revolutionary idea can be found. In what is called the CLEAN-SLATE approach [8] the objective is to forget about the structural and commercial limitation imposed by the current Internet architecture in order to redefine network requirements and principles. Over them it would be designed a new architecture that would avoid known problems of IP in fields like QoS, security and mobility while provide best support for future applications.

IV. Prominent proposals

One of the key issues in the current Internet architecture has been identified in the use of IP addresses for both physically locate and identify hosts. Some proposals have been advanced to separate these concepts.

The Host Identity Protocol (HIP), defined in RFC4423, and the Accountable Internet Protocol (AIP) [9] use one public/private key pair to identify each host, where public key is used as public identifier or part of it while location is achieved through standard IP for HIP or, for AIP, through a hierarchical construction of Accountable Domains, which AIP addresses are concatenated with host address for routing purpose. Both rely on DNS services for address discovering. The Hierarchical Internet Mapping Architecture (HiiMap) [10] steps forward to redefine the DNS architecture for location. It divides Internet space in several regions, which was proposed to be real countries, and creates one mapping authority per region and one global authority to map region ones. Whit regard to the distribution of public keys HIP, AIP and HiiMap don't rely on a PKI infrastructure separated from the backbone Internet architecture but face the problem of the key distribution in different way. While AIP and HIP aim to integrate public keys in its address space through the use of self-certifying addresses, HiiMap integrates a PKI in its location system infrastructure [11], so that keys do not identify host but legal entities and solving the problem of key flexibility.

RNA project [12] proposes a "single, flexible architecture based on the reuse of a metaprotocol over different regions", the stack of network protocols is thus dynamically composed by a particular instantiation of this protocol for each layer to avoid reimplementation. The metaprotocol composes capabilities currently dispersed in different layers providing services such as state management, congestion control and security association. The resulting service is thus configurable to mach the needs of the lower or the upper layer. The goal of RNA's metaprotocol is basically create a way to avoid the need of an ad-hoc service created adding a new ad-hoc layer between existing ones or virtualizing it over the current stack just to fit a particular context. RNA's metaprotocol provides security on the entire stack by reusing security features of existing protocols in different layers and coordinating them via a common metaprotocol module interface. Despite this is a very efficient solution, it does not specify any type of

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recommendation or requisites for security. A release of the RNA's metaprotocol is available in [13] as a patch to Click Modular Router software.

4WARD project [14] covers several areas of interest such as: Business Innovation and Regulation, New Architecture Principles and Concepts, Network Virtualization, In-Network Management and Forwarding and Multiplexing. It proposes an *architecture framework* that make possible to derive and deploy families of interoperable networks. It uses the notion of virtualization and particularly network nodes, the *netlets* [15], which can be seen as containers that hide protocol details but provide a number of properties via interfaces. A specific network is build assembling or multiplexing different netlets, which accomplish runtime requirements. It archives to run multiply different networks architectures in parallel and select the more appropriated one on runtime [16]. Based on the concept of network selection, the security provided by a network typology is just a parameter of the network selection process. Moreover in a virtual multi-network environment, a single node can be part of several virtual networks at the same time, each one with a different predominant paradigm, for example high bandwidth instead of security. In [16] the security is reduced to the selection of the appropriate security protocol by using a selection algorithm that evaluates the effects of adding such a protocol on a TCP/IP stack.

SELF-NET project [17] aims to design a self-managed network, where the concept is enhanced to cover distinct self-management methods defined as: self-optimization, self-configuration, self-healing, self-protection, self-awareness and self-organization. The object is achieved by defining a three-part closed cycle process (Monitoring - Decision Making - Execution) composing a *Cognitive cycle* that each element of the network would implement. In this context an element is intend to be either a network element (e.g., router, base station, and mobile device), a network manager, or any software element that lies at the service layer [18]. Such a system will be able to recognize its operational context (Monitoring phase), analyze it to extract a set of possible action and select the most appropriated one in each case (Decision Making phase). The selected solution will be applied to the system in the Execution phase through processes like self-reconfiguration or replacement of software components. An important aspect of these architecture elements is the ability to learn from past decisions discerning if they target the desired objective and use this experience in future decisions. Thus Decision Making, Monitoring processes and algorithms are strictly correlated.

SELF-NET proposes an implementation of the system it designed adding new autonomic network elements that should be aware of their internal and environment state and also have the ability of planning, deciding and adapting their operation in a way that best fits the operator's goals and objectives. Because of its autonomy property and distributed nature, multi-agent system (MAS) paradigm is used to support the Self-NET requirements; thanks to its decentralized approach MAS would be able to solve difficult problems in complex environments.

Slightly less centered on defining new specific network architecture, other projects focuses more on security aspects.

ECRYPTII project [19] centers on the cryptology fundamentals in networking. Divided in tree virtual labs, the project aims to address issues in symmetric and asymmetric encryption primitives and protocols as well as efficient implementation techniques in hardware and software. Those techniques might be useful for Future Internet. The WOMBAT project [20] focus on monitoring and identifying malicious code and attacks in order to generate new security practice and tools against emerging security threats.

INTERSECTION project [21] aims to build a security framework for interoperable networks by dividing the framework in two different layers, in-network and off-network layer. The project focuses on monitoring and identifying new security threats and vulnerabilities, as well as good countermeasures, at the in-network layer, and on providing a decision support system in the off-network one. The framework uses knowledge-based approach, in the off-network layer, to efficiently cross-relate monitoring results from different networks through the VIO (Vulnerability Intersection Ontology [22]).

Several more projects in FP7, center on the analysis of security in well-defined fields such as emergency (PACE [23]), financial frameworks (COMIFIN [24], PARSIFAL [25]) or industrial control systems (VIKING [26]).

The core concept that is perceived as fundamental for the success of any architecture is *trust*. Trust between network entities or end users, either to allow a distributed management of the network or to dynamically ensure end to end transactions. COMIFIN "wants to deliver a composable software system for large scale infrastructures that meets non-functional properties, such as responsiveness, predictability, *security and trust* by design". As authors in [25] state: "A lack of *trust* between entities where information is being exchanged goes to the heart of many of the challenges facing the domain of critical infrastructure protection".

Trust comes with the inseparable concept of *Identity*. For humans, both concepts can be reduced to one following the paradigm: "If I know you, I trust you". In the digital environment this model can not be applied. For example, even if the network can securely authenticate the user who wants to connect his equipment to it, this does not mean that his equipment is free of third-party malware and therefore safe to connect. Moreover either in the human or in the digital context we have to take into account the level of the trust that has to be assigned to a specific user identity and how to handle them.

Several FP7 projects take into account Identity Management. The SWIFT project [27] is designing an overlay infrastructure where the identity of the user is managed through an element called Identity broker. In the SWIFT context user owns several different *virtual identities* all of them related to a single real identity. Each virtual identity represents a "*face*" of the user, maintained to separate roles or for privacy reasons.

In parallel the PrimeLife project, whitch focuses on providing a life-long protection to the user privacy in emerging Internet applications, has defined interesting requirements for ensuring users' privacy in [28]. The documents focus

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on the analysis of social network sites and collaborative workspaces identifying several requirements for these kind of environments that can be easily dovetailed in a more general framework. The PrimeLife's work point out that: users should have control over contexts and be able to create different kinds of context relating to distinctions; A Management System (MS) should offer models for relationships, policies, etc., that mimic everyday's human social interactions; the MS should provide users with tools to inspect (and correct) the automated inferences made on the basis of their behavior in the network; the MS should offer users the option to terminate their identity which should result in deletion of all data pertaining to this user; the MS should provide a certain level of anonymity to its users and should provide features for creating, managing and deleting different partial identities in order to reduce linkability of all actions of the same user.

V. Security requirements

In this section, we analyze the efforts made in order to define high level indispensable requirements for research in Cloud and FI under the view of security, trust and identity.

A. Secure Cloud

Security aspects in cloud computing cover a large number of topics. To deal with that, the Cloud Security Alliance (CSA) was created as a non-profit organization to promote the use of best security practices. Within the *CSA domains* of work it is possible to find: application security, encryption and key management, identity and access management.

Identity and access management are among the most outstanding topics when defining a secure cloud. This is the domain of the Trusted Cloud Initiative (TCI). TCI published in 2009 the 2nd version of the research baseline for the CSA in order to define its certification criteria [29]. The same group is now working on version 3. The documents analyze what requirements have to be addressed in the fields of Identity Management (IdM), authentication and identity federation. In March 2010, CSA identified in [30] seven major security threats and proposed some directives for solving them. The threats described in the document can be summarized has follows:

a) Abuse and Nefarious Use of Cloud Computing: To overcome the problem, the CSA recommends to increase *accountability* degree, of services as PaaS or IaaS through stricter registration and monitoring processes.

b) Insecure Interfaces and Application Programming Interfaces (APIs): The inappropriate and unauthorized use of those programming primitives should be avoided, for instance, enforcing cryptography in authentication and using fine grained access control models.

c) Malicious Insiders: The lack for provider of transparency in managing security over the entire service chain let malicious insiders to manipulate users' data. To overcome this, the CSA recommends active participation of the user in the entire security process.

d) Shared Technology Issues: a strong compartmentalization must be used together with the enforcement of Service Level Agreements (SLA) in order to overcome problems derived from sharing infrastructure among users and Cloud provider.

e) Data Loss or Leakage: As long as the number of iterations increase so does the risk of information leakage in a cloud environment. For that reason, the usage of strong encryption mechanisms and access control should be mandatory.

f) Account or Service Hijacking: If attackers gain access to users credentials, current and future activities, transactions and exchanged data will be compromised. To overcome this problem, besides implement stronger authentication methods, monitoring credentials is key for noticing hijacking.

g) Unknown Risk Profile: The risk of losing track of the security ramifications is a drawback of cloud deployments. Partial or full disclosure of provider's infrastructure details as well as security logs could overcome or mitigate the problem.

After analyzing the threats, stronger authentication and monitoring practices seems to be essential to solve cloud security threats. Nowadays the adoption of Cloud Computing solutions by real world industry is clearly driven by the size of the organization. In fact, while a small organization could find in public cloud computing services a perfect solution for a cheap start-up, medium to large organizations need to define more complex environments, with restrictively security constrains, in order to accomplish their business project in a secure way. Hybrid solution, mixing public and private clouds, are used to merge high level protection, enabled by private clouds, with the greater flexibility offered by public cloud services. Those trends could be eventually changed in the near future with the outcome of several efforts from organization as DARPA [31] and IARPA [32]. These efforts rely on homomorphic cryptography techniques that allow operating directly over encrypted data sets, producing an encrypted output without knowing data itself. Due to the complexity of the problem the first system using homomorphic encryption is quite recent. It was developed in 2009 by Craig Gentry [33]. Other prominent effort is [34], where a conceptual simplification using Integer based scheme instead of Ideal Lattices scheme was presented. Recently an important improvement to Gentry's fully homomorphic scheme based on ideal lattices has been presented by Stehle at al. in [35]. In this work, they describe a system that reduces the complexity of binary operations of Gentry's system form 2^λ to $quasi - \lambda^{3.5}$ where λ is the security parameter. If research on homomorphic encryption continues reducing the complexity while maintains the security, in a near future, public clouds might be no longer considered as dangerous as today.

B. Secure FI

Despite several projects aim at designing and defining Future Internet architecture, not all of them tackle security aspects.

In the scope of design a global trusted architecture one project takes particular relevance. The THINK-TRUST project

[36] wants to provide guidance on policy and research challenges in the field of security and trust in the Information Society. The project aims to model and define new intelligent and user-friendly ICT security environments that fit the requirements of the Information Society. THINK-TRUST final report [37] identifies challenges in different areas that depict a set of requirements that should be followed in order to provide trustworthy hardware and software:

a) Trust engineering: Trust is not absolute and will be quantified by the preferences and intuitive policies of users. Thus the need of a *trust framework* appears where trust relationships between entities are established and managed to encompass trust *preferences*, trust *policy* and trust *weighting*. Alternative approaches such as reputation, recommendation and frequentation should also be explored.

b) Architecture: Architectural support must be provided first with regard to transparency - security monitoring, observability and measurability - for data logging and log access and secondly, with regard to the ability to function across multiple layers and domain. The requirements for accountability illustrate that the user can be fully accountable in the local context but his privacy has to be protected by that local domain.

c) Cyber-security: Techniques and mechanisms to provide protection, assurance and integrity are required. These have to be platform-independent to allow interoperability of trusted entities and have to consider the growing in complexity, size, capacity, speed of the digital environment. At the same time both scarce resource devices, such as sensor networks, and self-awareness ubiquitous systems have to be considerate.

d) Accountability: Faced to anonymity, represents a supposed dichotomy between security and privacy. Accountability is view as a priority if we consider that it allows traceability/identifability, making possible to establish responsibilities and liabilities. Two options are recommended: base the demand for traceability and accountability on global accountancy-type principles, which can encompass the whole network or reintroduce, on an intermediary network layer, a “territorialisation” of facts and participating parties. In both of them a certain degree of privacy have to be ensure and it is reflected in requirements for anonymous/pseudonymous charging and payment systems and requirements for anonymization or impersonation of heuristics to produce untraceable, but trustworthy, valid sources/channels for information; for example, for economic, social or health-related statistics.

e) Privacy: With nowadays data-recollection systems absolute anonymity may be neither possible nor applicable. [37] points out the need of a fine granularity access control to identity-related information, of tools and concepts for deleting data in the Internet (in order for it to “forget”) and the need for standardized techniques to assure privacy across the various Internet layers, throughout to network level and maintaining consistent privacy across different environments. Common points can be easily recognized in the recommendations found in PrimeLife documentation [28] depicted in sec V.

f) Protection: The protection of data processing, storage and transmission, as well as the shielding of resources and assets (information, services, devices, communications) require the following: domains, partitioning, compartmentalization, leading to trusted zones (and therefore, intermediate, semi-trusted zones), and to the localization of damage; fine granularity access control based on multiple bases for authentication and authorization; mutual authentication, with multiple devices; new cryptographic techniques which are low cost but high performing, in preparation for the quantum/post-quantum age.

g) Usability: FI and generally new generation network trends focus the attention on the end user but two opposite viewpoint emerge: one where the user is surrounded by a system that monitors him and automatically configure itself in order to suit user’s needs and requirements and one where the user actively influence and set his own environment. While in the second is the user who decides what and how much personal information provide to the system, in the first case the system “spy” the user in order to discover all kind of useful information. The trust of the user in both systems is a key factor in the analysis. The challenge consist in offer both possibilities to the user, solving trust issues in both scenario.

VI. CONCLUSION AND FUTURE WORK

The concepts of Cloud Computing and Future Internet have been presented in this paper. Cloud, as an almost fully deployed infrastructure, has been introduced as the last paradigm before the Internet of the future, Sec. II. Future Internet is not a well-defined concept, instead several proposals have been used in order to give an overview of the fields concerned by the change, Sec. IV. Regarding security, several aspects for both Cloud Computing and Future Internet are still an open issue and Sec. V has been dedicated to summarize results of major efforts in defining guidelines for research and development of secure infrastructures. Analyzing threats pointed out in this section, an high degree of overlap can be found, in fact beyond the requirement for stronger authentication and cryptographic techniques, useful in every kind of environment, a transverse need of new trust and identity management frameworks arise from almost every Cloud threat, keeping included in the more extended vision presented by FI requirements. Assuming stronger authentication and encryption will be available, most Cloud security issues could be solved through fine monitoring techniques and a complete users accountability, ignoring users’ privacy. Moreover FI requirements of *Architecture* and *Trust engineering* depict a scenario where multiple providers, of Cloud or new types of composable services, are required to interoperate ensuring trustworthy, dynamic and private transactions while Cloud security threats does not take into account interoperability.

Besides, some of the initiatives depicted in Sec. IV focus on aspects that can be directly mapped to requirements for a secure Future Internet: SWIFT and PrimeLife center on privacy and protection of users’ identity, INTERSECTION on cyber security, HIP and AIP on user accountability but architectural aspects such as monitoring and observability are a common issue present in most of them. The lack of a uniform view for architectural reform of Internet infrastructure could spread efforts in to many directions. At the same time, Cloud providers, whose infrastructures have already been deployed and are evolving to face security and privacy threats, could take advantage form research implementing solutions that fulfill the requirements for a secure Cloud. Being secure Cloud and secure FI somehow overlapped, this brings Cloud closer to the definition of FI strengthening the vision of the evolutionary approach at the expenses of the clean-slate one.

Cloud providers might thus be identified as those early adopters which could lead the Future Internet implantation.

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Ontology driven Augmented Exploitation of Pervasive Environments

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***Abstract*—Pervasive environments offer to software applications the possibility to interact with the reality in order to perceive the information surrounding the users and to adapt the environment and their own behavior. This problem is modeled as a set of context aware services which learn and augment the reality perceived by the user through her/his device. Application are enabled to perceive the information surrounding the users, to adapt the discovery and delivery of contents and software in order to optimize the user's satisfaction. We present a framework that implements the model to supports experts in the domain of the Cultural Heritage to augment the archaeological site with a set of multimedia contents which are delivered by innovative services to the visitors in order to guide their tour and to enhance their perception of the reality.**

***Keywords*-Augmented Reality; Ontology; Context Awareness; Mobile Devices**

I. Introduction

Pervasive environments offer to software applications the possibility to interact with the reality in order to perceive the information surrounding the users and to adapt the environment itself, or to improve the user's perception by exploiting available devices. Here we deal with the problem about how to program the user's personal device for enabling the interaction between remote services and environment by the devices themselves. Applications should be able to use pervasive sensors as their own extension in order to let the users benefit of context awareness in services exploitation. Many application contexts can benefit of a tight coupling between pervasive objects and services. Let's imagine some relevant scenarios:

- assisted exploitation of museums and archaeological parks;
- information guides in wide and crowded environments like big cities;
- discovery and recommendation of products in commercial centers;
- rescue of missing peoples in disaster situations by personal objects (smart-phones, RFIDs, Bluetooth devices);
- monitoring and safety check of activities in working areas.

For all the examples which have been cited before common issues must be addressed but it is not straightforward that the same technical and technological solution can be applied across the different scenarios and within the same class of applications. Some features to be considered in each new case study are:

1) the kind of environment: indoor, outdoor,with or without technological infrastructures;
2) dynamic changes and lack of landmarks;
3) heterogeneity of objects, entities and devices;
4) huge number of object/users and situations to be monitored and managed. In this case the complexity could increase and it could make impossible to perform the necessary checks at the required frequency, limiting the provisioning of real-time reactions.

Beyond the difficulties in dealing with the capability of getting the awareness about the environment and its perception, when this changes dynamically, services would implement adaptive and reconfigurable behaviors in order to optimize their delivery. Here we have to address many issues related to the heterogeneity of parameters and criteria to optimize the content delivery and to integrate the best suited techniques and technologies. For example, time constraints and user's positioning and interest have to be considered together but must be processed with different techniques. Furthermore additional properties to be considered in critical applications are security, traceability and certification of events, real-time reactions.

II. Related work

Many research contributions focus on the development of software/hardware architectures and frameworks for mobile context-aware tourist guides, also based on semantic techniques and technologies. In [2] authors present iJADE Free-Walker, a framework that integrates GPS, ontology and agent technologies to support location awareness for providing assisted navigation and classification of tourist information for the users. The system implements a context-aware tourist guide for the city of Hong Kong. The ontology is extracted using structural information from travel websites. CRUMPET, Creation of User Friendly Mobile Services Personalized for Tourism [3], is an agent based framework that provides a context-aware tourist guide adapting the information content and presentation according to user profile, device and connection characteristics. It is developed in microFIPA-OS, an agent platform based on the FIPA-OS that is an Open

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Source implementation of the FIPA (Foundations of Intelligent Physical Agents) standard. CRUMPET is modeled as a multi-agent system (MAS) whose agents are autonomous and share their knowledge using FIPA-ACL (Agent Communication Language). Agents use a common ontology in order to understand each other. The CHIP (Cultural Heritage Information Personalization) [4] project is a framework that uses Semantic Web technologies to provide personalized access to digital museum collections. The framework supports assisted navigation integrating heterogeneous technologies and provides semantic browsing, searching and semantic recommendations. In our approach semantic is used for knowledge representation and management, but the ontology is designed by experts of the application domain. It is used both as a common vocabulary to support interoperability aomng heterogeneous remot services, local applications and users, and for intelligent discovery of media contents. Beside pervasive technologies and augmented reality are used to enhance the visit experience and to enable visitors to explore easily pervasive areas according to their interests. Other relevant issues, like ubiquity [5], are not addressed here.

III. Problem modeling

In order to approach the described problem we have to model the environment where the user is moving and to reconstruct the perceptions of the user himself in order to get her/his particular vision about what is surrounding him. A real representation of the environment is necessary to identify landmarks and possibilities of intervention using pervasive actuators and sensors. On the other hand we need to get the special knowledge of the environment acquired by the user in order to support him by augmenting her/his perception of reality with something that can improve her/his satisfaction. This problem can be modeled as a set of context aware services which learn and augment the reality perceived by the user through her/his device. Reality can be augmented and adapted by interacting with neighbor objects and by delivering relevant media contents.

In Figure 1 an high level representation of this model is shown.

The environment will be modeled as a geo-referred map with itineraries, landmarks and point of interest. Of course localization of users and objects it is possible according the device technology, the available infrastructures and the kind of environment. Indoor or outdoor localization can be implemented using heterogeneous technologies, and often absolute localization could not be performed, but only nearby landmarks or object can be detected. In the following equation

$$E = \{\{POIs\}, Pos(U), \{path_i\}\} \quad (1)$$

the environment is modeled by a set of point of interest which are known a priori, or which have been detected dynamically by the user's device. $Pos(U)$ is the current

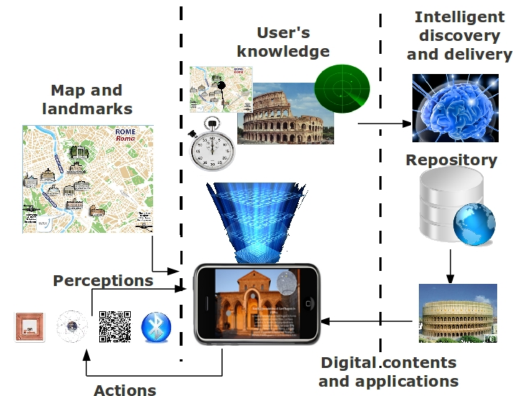

Figure 1. Problem model

position of the user. It could be an absolute position, or it could be a relative position according that has been estimated by detecting close POIs. $path_i$ represents an available itinerary for the user.

User's knowledge about the surrounding environment can be acquired by using peripherals of her/his personal mobile device, by recording and evaluating user actions or explicitly asking for user feedbacks. Some examples are user's position, interest, nearby objects, landscape, etc. Interaction with the environment and presentation of contents can be done again using user's device and her/his peripherals.

$$K(U(t)) = \{Per\{0,t\}, I\{0,T\}, D(t)\} \quad (2)$$

K(U(t)) represents the user knowledge, and eventually knowledge history, about the environment. It is reconstructed by using information coming from pervasive sensors $(Per\{0,t\})$; considering user's interests which come from what the user has done, what s/he declares, what are her/his feedbacks $(I\{0,T\})$; taking into account device capabilities $(D(t))$. Of course different components could have different weights according to their relevance or the time of perception. We could consider the knowledge at the current time, in a time widows, or the complete history. Services, in order to augment the user's knowledge and the their/user's capability to interact with the environment, have to choose, according their context awareness:

- what content and application they have to deliver;
- when it needs to execute the application or to present the content;
- how this should be done.

$$AR = \{C(K(U(t))), Mode\} \quad (3)$$

AR represents the set of contents, and the way they will be delivered, to be used for augmenting the reality perceived by the user. The optimal set, and its organization, is chosen to optimize the user's satisfaction.

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IV. Ontology based knowledge representation and reasoning

Semantic techniques can be used both to model the knowledge of applications and to reason about the decision to be taken about how to adapt the service delivery. An ontology implements the representation of the global knowledge. It is necessary to share a common dictionary and to describe the relationships among the entities/objects which are part of the model. In our model a common ontology include all the general concepts which are useful to describe a pervasive environment where mobile users are moving, using their devices and interacting with available facilities and other users. The general ontology is complemented with a domain ontology that is designed by an expert of the specific field. Concepts of the ontology are used on client side to describe a representation of the reality as it is perceived by the user. On the back-end the ontology is used to annotate digital resources like points of interest, contents, applications. It is also used to support reasoning. Users' behaviors, information from pervasive devices or from other users, device properties, external events are heterogeneous data that are perceived by the device and that are used to build a dynamic changing representation of the user knowledge about the reality within which s/he is moving. The applications are knowledge driven. The user's knowledge can be used by the application that is running on the device to adapt its logic locally, and it is updated remotely to improve the awareness of services at server side.

Application are events based. Events can notify an update of the user's knowledge or can be explicit service requests raised by the user. At each invocations a semantic queries, that depend on the user's knowledge, is built and processed to get the action to be performed and to content to be delivered. Results of the query are individuals of the ontology that are described by semantic annotations.

The user's knowledge is composed of many semantic concepts with *static*, *dynamic* or *locked* properties.

Components of the knowledge are:

- *Device technology and capability*. Among the static properties here we means hardware resources, on board peripherals, display size, total memory and storage. Dynamic ones can be power level, available memory and storage and bandwidth ;
- *User's position*. It is a dynamic property that can change over the time and can be evaluated using different techniques and technologies, depending on the devices, on the available infrastructures and the kind of environment (indoor or outdoor);
- *Pervasive objects*. They are dynamically discovered by the device. They can be sensors which provide information about the environment or can be used by the services themselves through the device or eventually directly if they are connected in some way to the network;
- *Time information*. We intend the current time at user side and the time that the user is spending, or has spent within the environment.
- *User's interests*. This part of the knowledge could be dynamically changed by the application according to the user's behaviors and to her/his feedbacks. The user could choose to start from an empty or a standard profile, to change it or to lock some properties interactively.

Semantic techniques are used for intelligent content and application discovery and delivery. Knowledge representation, ontology and annotations of digital resources are used to filter, organize and deliver contents and software to the device. Different techniques for reasoning can be experimented such as graph matching, description logics, neural networks or more simple ones like SPARQL (SPARQL Protocol and RDF Query Language) queries. Furthermore semantic can be integrated with other kinds of techniques to take into account constraints such as user's position and available time for exploitation.

V. Requirements and design

In Figure 2 the architectural solution of a framework that allows to apply the described approach is shown. Users,

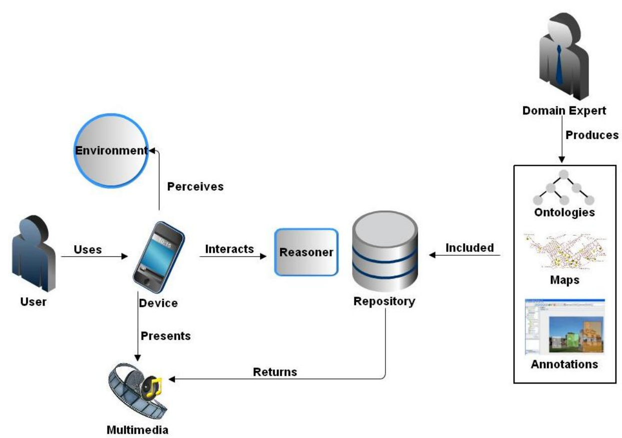

Figure 2. Architecture and roles

devices, services and producers are actors in this scenario.

On the left side the user is using her/his device that hosts a client application that is able to perceive information from the field by pervasive sensors.

The application executes autonomously and proactively in order to support the users' activity within the environment s/he is moving. It discovers surrounding objects, uses them to update the representation of the user's knowledge, reacts using the local knowledge to organize and propose the available contents and facilities by an interactive interface. It could also communicate with close devices. If the connection works the device can access remote services which can exploit a wider knowledge and complex reasoning capabilities to look for additional contents and applications.

Experts of the application domain define the ontology for the specific case study. They use or design a map to represent the environment. They add POIs to the map to geo-refer multimedia contents and can link them to a concept of the ontology. Furthermore they select relevant contents and annotate them using concept and individuals from the ontology.

Remote applications implement context aware services. They use personal devices to collect perceptions and for content delivery. They use reasoning to access content repositories or Internet, and are able to discover and adapt contents which are relevant to optimize the user's satisfaction.

In order to support these activities in a real scenario we need to provide:

- ***back-end tools*** for enabling content production, their semantic annotation publishing and retrieval;
- ***services*** for content discovery, adaptation and delivery;
- ***a client application*** to support user interaction with services, with the environment and with available contents.

VI. A REAL SCENARIO

Exploitation of archaeological sites can be very difficult because of a lack of supporting infrastructures and because of the complex recognition and comprehension of the relevant ruins, artworks and artifacts. The availability of personal devices can be used to plan and support the tourist by suggesting him the visit tours, the point of interest and by providing multimedia contents in the form of digital objects which can semantically augment the perceived reality.

In this context a relevant issue is the profiling of the user, the selection and the presentation of the contents which can improve the user's satisfaction, by providing new models of interactions with reality, trough her/his device. In this context the Second University of Naples is engaged on a multidisciplinary project with both cultural and a technological aims [1].

Following the approach defined above we are implementing a technological framework that supports the experts in the domain of the Cultural Heritage to augment the archaeological site with a set of multimedia contents which are delivered by innovative services to the visitors through their mobile devices in order to guide their tour and to enhance their perception of the reality and learning.

Three case studies have been chosen to test the approach and the framework. The S. Angelo in Formis Basilica, in Campania, near S. Maria Capua Vetere - i.e. the ancient Capua, is an interesting sample because of its uncountable layers which begin with the ancient temple of the IV century B.C., going on with the late- republican temple, unto the basilica that witnesses so many previous phases. The devices and the procedures allow to perceive some elements in the landscape, such as landmarks, which otherwise could not be noticed. It is necessary to understand the history and the details of ancient sanctuary and for the exploitation of the country nearby in ancient times. From the sanctuary one sees at the bottom the pianura campana, which still witnesses the centuriazione, and on the top the Monte Tifata, rich of archaeological evidence- such as temples, dwellings, thermal structures- to be studied with regard to the project. The project is carried out also on the ancient town of Norba and on the amphitheater of Capua: the former, because of its 44 ha in width, in a huge open park, has to be described with special regard to urban plan and architectural features because it is largely saved and still buried. It was founded in the V B.C. and destroyed in 81 B.C., later it was seldom inhabited just at the beginning of the Middle Ages. Usually visitors are surprisingly impressed by the walls in opera poligonale: that is why, in this case, the aim is to draw visitors attention on other landmarks (temples, houses, the street frame) and to represent by interpretation of aerial photographs, survey on the spot, specific digging sample, by means GPS, what is still underneath the surface. The last topic is the amphitheater of Capua, the second one as for width in the ancient world: in this case visitors shall be able to realize the whole building, with the service rooms and the devices underneath, with reference especially to the set of urban layers which the building stood in and which has been radically altered.

VII. IMPLEMENTATION

To satisfy the requirements of the presented case studies we need to provide a technological solution that does not need infrastructures for letting the software know the user location and her/his feeling about the environment. It means that Bluetooth, RFID, GPS, electronic compass, camera, network connection and others are the technologies which can be used together or independently to get information about the user perceptions and to augment her/his exploitation of the archaeological site. The user will be able to download at home, before to leave, or on site, if the network will be available, the map of the area to be visited. The map will include all the points of interest that identify the relevant objects of that area and different cultural itineraries which could be exploited on site. Also contents can be discovered and downloaded in advance. On board software and remote services will assist the cultural visit by augmenting the reality by the user's personal device. In the following we detail the technological choices which have been taken to implement each component of the framework.

A. Environement Map

To provide a description of the environment within which the user is moving we need a geo-referenced map that implements the model described in Equation 1 by describing

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buildings, roads, bans, itineraries and Points of Interest (POIs). We used the OpenStreetMap format to design open maps. In Figure 3 the map of the S. Angelo in Formis cathedral is shown. It has been built by exporting a model that was originally built by Autocad and has been exported into a GPX (GPS eXchange Format) format. The JSON (JavaScript Object Notation) tool allowed us to import the GPX (GPS eXchange Format) trace and to add manually details and POIs. Each point represents an artifact, a ruin or any other entities of cultural relevance and can be described using a list of key-value pairs. Some of them have been used to link the POIs to URLs of multimedia information, or to provide a semantic description of the POIs itself. The tool allow to export the map in an open format that can be read and used by the client application that is presented in this paper.

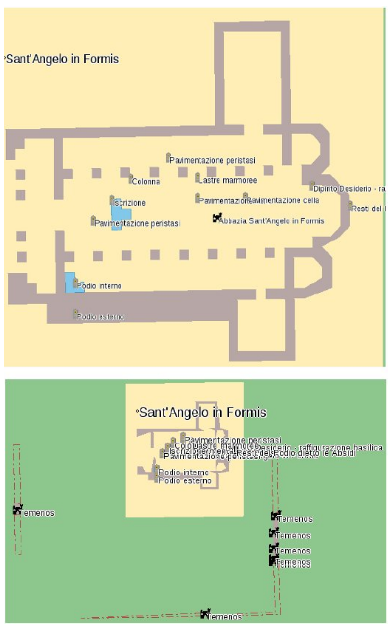

Figure 3. S. Angelo in Formis map

B. Ontology and annotation

An ontology has been designed to describe the sites of interest and to annotate the related media. It support the implementation of the model described in Equation 2. A general part includes the concepts which are common to all the class of applications that can be modeled according the proposed approach. Among the others the *Time* class and her/his properties (*CurrentTime*, *AvailableTime*, *ElapsedTime*, *ExploitationTime*) allow to organize and assist the visit taking into account time information and handling time constraints. *Position* class and its properties allow to localize the user and objects around him. An application specific part of the ontology include the concepts that belong to the domain of the cultural heritage and additional classes and individual which are proper of the case studies introduced in the previous section. In Figure 4 a snapshot of the ontology is shown. In particular we can see the *Building Element (Parte di Edifici)* class an its subclasses, *Amphitheater Element (Parte di Anfiteatro)* and *Domus Element (Parte di Domus)* which are proper of the sites of interests. The ontology is used: 1) for annotating the

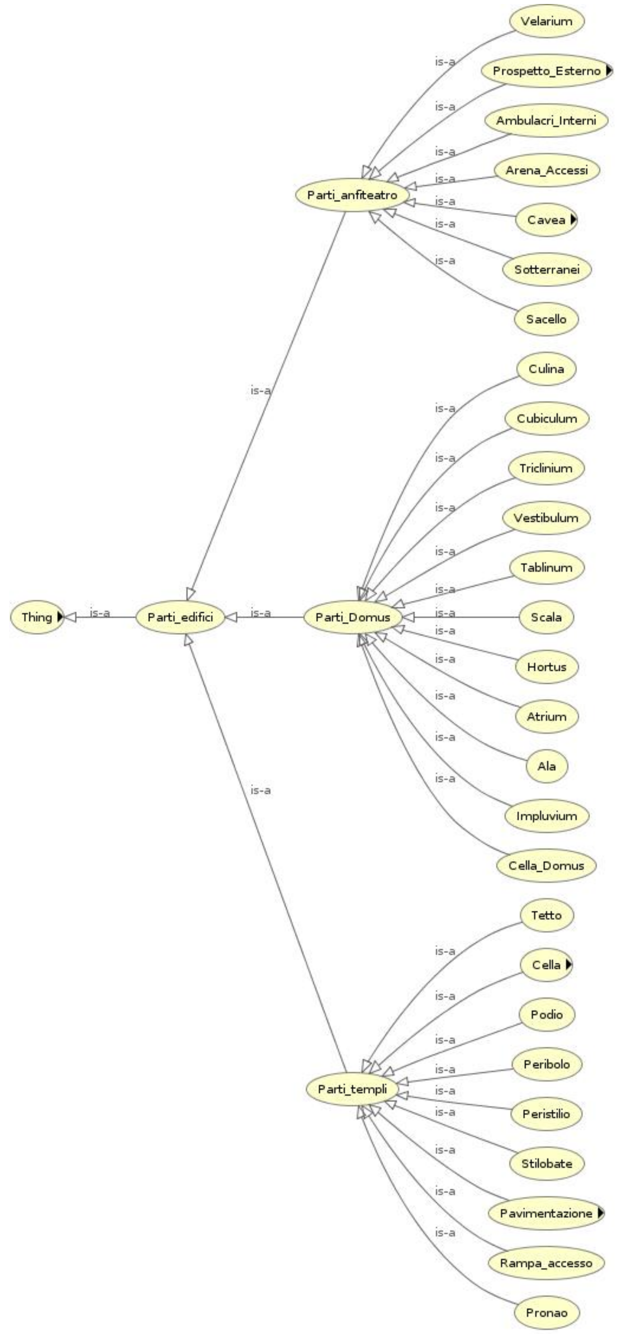

Figure 4. Ontology and annotations

multimedia contents; 2) to represent the user's knowledge;

3) to perform reasoning for intelligent discovery of those documents which are relevant to the optimize the quality of the cultural visit. To annotate texts, images and any kind of contents we chose the AktiveMedia tool (available at: http://sourceforge.net/projects/aktivemedia/). In Figure 5 a picture of the Amphitheater of S. Maria Capua Vetere is annotate with the *Column* and the *Arc* classes which are part of this kind of building.

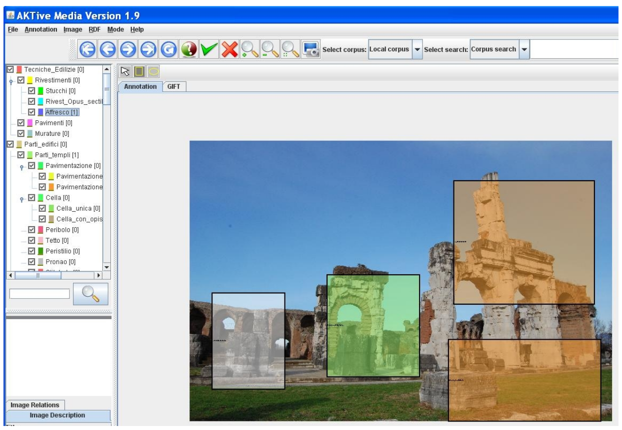

Figure 5. The annotator

The output produced by the annotator is an RDF file that use concepts and properties of the AktiveMedia ontology and of the domain ontology. An example of RDF annotation is described below. We filtered a set of properties. In particular *hasConcept* specify the ontology class or the individual by which the selected part of the image has been annotated.

```
<?xml version="1.0" encoding="windows-1252"?>
<rdf:RDF
   xmlns:j.0=""
   xmlns:rdf=""
   xmlns:dc=""
   xmlns:j.1="">
 <rdf:Description rdf:about="">
   <j.0:usesOntology></j.0:usesOntology>
   <j.0:hasAnnotation>
     <rdf:Description rdf:about="">
       <j.1:anID></j.1:anID>
       <j.1:hasContent></j.1:hasContent>
       <j.1:annotationText></j.1:annotationText>
       <j.1:botomX></j.1:botomX>
       <j.1:bottomY></j.1:bottomY>
       <j.1:hasShape></j.1:hasShape>
       <j.1:hasConcept></j.1:hasConcept>
       <j.1:topX></j.1:topX>
       <j.1:annotationHeight>
       </j.1:annotationHeight>
       <j.1:topY></j.1:topY>
       <j.1:hasComment></j.1:hasComment>
     </rdf:Description>
   </j.0:hasAnnotation>
   <j.0:width></j.0:width>
   <j.0:comment></j.0:comment>
   <j.0:hasHashCode></j.0:hasHashCode>
   <j.0:height></j.0:height>
   <dc:description></dc:description>
   <j.0:hasPath>
     image_example.jpg
    </j.0:hasPath>
   <j.0:content> </j.0:content>
   <dc:creator> </dc:creator>
 </rdf:Description>
</rdf:RDF>
```

C. Digital repository and semantic discovery

The Fedora repository (available at: http://fedora-commons.org/) is used to store digital objects, which are included in Equation 3, and supports their retrieval. Into the Fedora repository a digital object is composed of a set of files which are:

- *object metadata*: used by the client application to understand how to deliver the content;
- *binary streams*: which are images, video, text ... any kind of raw information to be delivered;
- *RDF annotation*: that describe the semantic of the object according to the ontology;
- *disseminations*: filters to be eventually used for adapting the object according to the target client.

We loaded the Aktive-Media ontology and the domain ontology into the Fedora repository in order to exploit its embedded SPARQL (SPARQL Protocol and RDF Query Language) engine that is used to select the optimal set of individuals that means contents. Multimedia contents are automatically stored into the repository after the annotation phase. The RDF output is automatically processed using an XSL (eXtensible Stylesheet Language) transformation to make it compliant with the format required by the Fedora repository.

D. Content types

Different types of content models have been defined and simple examples have been produced.

- Multiple images whose transparency can be graduate by the user to compare changes in different periods. In the same way real picture can be compare with paintings. Old picture can be compared with what is seen by the camera.
- Part of the image acquired by the camera are recognized and linked to related multimedia contents;
- Virtual reconstructions which are synchronized with the camera output or the detected RFIDs;
- Text, audio, video and composite media.

A content descriptor is attached to every digital object. It is used by the device when the content must be delivered. The descriptor defines the right player for that media, configuration parameter and necessary input.

In Figure 6 an example of delivered content is shown. The user focus the camera on a particular view of the S. Angelo in Formis Basilica and sees the original temple of

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Diana Tifatina. In particular the perspective viewed by the camera is the same shown by the video.

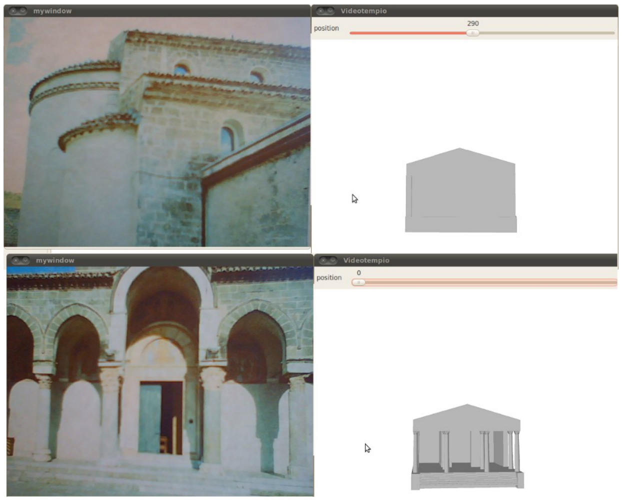

Figure 6. Synchronized video

E. Remote services

A remote service has been conceived to support content discovery and delivery.

It uses multiple criteria to discover and filter relevant multimedia contents. Some criteria are the available time for the visit, the user's position, the device technology, the user interests. The service is implemented by independent filters. Filters are executed on the occurrence of such event that update the user knowledge, or by direct asynchronous request from the user himself.

For example a change of the user position is used to filter POIs which are close to him or are relevant to her/his position. We could get information about a building located close to the user or about some other buildings located elsewhere but that have been designed by the same architect.

When the user shoots a new picture, a search by sample facility is used to find the images that are similar to the the current subject. This can be used to suggest a new set of contents, which are relevant to what the user is looking at.

Each time the interest of the users changes, or also when any other event occur, a reasoner generates dynamic SPARQL (SPARQL Protocol and RDF Query Language) queries, by which the repository is searched, and organizes the retrieved contents according to their relevance. The ontology is used to perform additional reflexive reasoning.

Additional filtering rules can be implemented using time information if they are available. For example it could be relevant to know the current time (to suggest a sunshine rather than to avoid a closed attraction or museum). The available time for the visit is important to limit the number of contents and to plan an itinerary. A delay is important to dynamically cut what is less relevant and cannot be exploited anymore.

We have implemented a search by sample filter that use the position of the user to select all the picture which represent the subjects s/he shoot, such as landscapes, monuments or buildings around him. Within this set a matching algorithm is able to identify an image that is the most similar to the picture. All the annotation for that image are also matched with on the display of the user's device.

Instead, some example of SPARQL query which are processed and whose results depends on the user's interest, are provided automatically by the software guide.

F. A Mobile Archaeological Guide

At client side, to implement a value added guide, we have extended an open source software navigator called Navit (available at: http://www.navit-project.org/). We used the android version. The navigator provides basic facilities for map visualization and to guide the user along some predefined itineraries by using the on-board GPS receiver. By new extensions the guide is able to sense the environment by the available peripherals, to understand the situations according to which the visit will be adapted, and to enhance the user experience. Even if many experiences on augmented reality are proposed nowadays, the exploitation of vision is used only to overlap real and virtual images. The extensions provide the following functionalities:

- access to device peripherals to sense the environment and to update the representation of the user's knowledge about the reality around him;
- access to remote services to update the user knowledge and to ask for available contents
- a local cache of objects for exploiting the visit without connection;
- a limited reasoner that is able to organize the content by itself when the connection does not work.

Perceptors are implemented by:

- GPS positioning to localize the user in open spaces and to guide him on cultural itineraries;
- RFID for positioning and detection of nearby POIs. This technology can be used to alert the user but also for her/his positioning in indoor environments;
- CODEBAR recognition to get information about artifact, monuments, ... when a RFID reader is not available;
- image recognition by search by sample techniques which are speed-up and improved using a position based filtering;
- monitoring of device resources and configuration;
- collection of user interest by feedback and by an analysis of her/his behavior;
- time monitoring.

On the other hand a list of functionalities are provided to deliver contents and to guide the visit. Content management (discovery client,organization and fruition) is supported both at client side and at server side. Knowledge visualization and management to allow the explicit specification of user's own interest is provided. The output of the camera is used as a component of the user's knowledge as well as a map on which semantic additions are anchored, not simply superposed.

The user will be able to ask for and exploit available multimedia contents, which are related to points of interest, or to personal interests expressed by semantic concepts. Multimedia content will be adapted at server side according to the device/user/session profile to provide to the user the best quality of service.

Figure 7 shows the output of the result of the content discovery service. The service invocation starts the camera that is used to take a photo of the landscape or of a subject of interest. In background, the client upload the image and wait for the content retrieval. The discovery of relevant contents and the download of retrieved information run in background meanwhile the user is interacting with other facilities provided by the client. The list of contents are presented to the user when they are available. Other events which open dialog for suggesting action, itineraries, POIs or media can be related to new perceptions.

Figure 7. Content retrieval

VIII. CONCLUSION

In this paper we presented an approach for delivering of context aware services in pervasive environment. Services adapt applications and contents to be delivered in order to optimize the user's satisfaction by augmenting and improving her/his perception of the reality. An ontology driven methodology has been used to describe the environment, the user's knowledge and application/contents to be delivered. Personal devices have been exploited to interact with the environment, to run interactive applications and to present contents to the user. Perceptions which come from on-board peripherals, from user's feedbacks and actions are communicated to context aware services to update the remote image of the user's knowledge, and are used at client side by the client application to adapt itself. We described a framework that implements the proposed approach. It represents preliminary research results about the aided exploitation and context awareness of complex archaeological sites by mobile devices. The framework that supports the experts in the domain of the Cultural Heritage to augment the archaeological site with a set of multimedia contents which are delivered by innovative services to the visitors in order to guide their tour and to enhance their knowledge of the reality. We are extending the framework using intelligent agents technology and modeling the interaction between the two agents according to learning by teaching approach: a student on the device and a teacher in remote. The teacher agent produces contents that the student agent consumes presenting them to the user. Future developments of this work could be the storing of user's information about the tour, so allowing for multiple visits, as well as the addiction of the possibility of exploiting the visit at home, or simply planning it before the trip.

ACKNOWLEDGMENT

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RDF2NμSMV: Mapping Semantic Graphs to NμSMV Model Checker

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***Abstract*— The most frequently used language to represent the semantic graphs is the RDF (W3C standard for meta-modeling). The construction of semantic graphs is a source of numerous errors of interpretation. The processing of large semantic graphs is a limit to the use of semantics in current information systems. The work presented in this paper is part of a new research at the border between two areas: the Semantic Web and the model checking. For this, we developed a tool, RDF2NμSMV, which converts RDF graphs into NμSMV language. This conversion aims checking the semantic graphs with the model checker NμSMV in order to verify the consistency of the data. To illustrate our proposal we used RDF graphs derived from IFC files (Building Information Modeling). These files represent digital 3D building model. Our final goal is to check the consistency of the IFC files that are made from a cooperation of heterogeneous information sources (plumbers, architects, electricians, etc.)**

Keywords*: *Semantic graph, RDF, Model-checking, temporal logic, NμSMV, IFC, BIM.

I. Introduction

The increasing development of networks and especially the internet has greatly developed the heterogeneous gap between information systems. In glancing over the studies about interoperability of heterogeneous information systems we discover that all works tend to the resolution of semantic heterogeneity problems. Now, the W3C (World Wide Web Consortium) suggests norms to represent the semantic by ontology. Ontology is becoming an inescapable support for information systems interoperability and particularly in the Semantic Web. Literature now generally agrees on the Gruber's terms to define an ontology: explicit specification of a shared conceptualization of a domain [1]. The physical structure of ontology is a combination of concepts, properties and relationships. This combination is also called a semantic graph.

Several languages have been developed in the context of Semantic Web and most of these languages use XML (eXtensible Markup Language) as syntax [2]. The OWL (Web Ontology Language) [3] and RDF (Resource Description Framework) [4] are the most important languages of the semantic Web, they are based on XML. OWL allows representing the ontology, and it offers large capacity machines performing web content. RDF enhances the ease of automatic processing of Web resources. The RDF (Resource Description Framework) is the first W3C standard for enriching resources on the web with detailed descriptions. The descriptions may be characteristics of resources, such as author or content of a website. These descriptions are metadata. Enriching the Web with metadata allows the development of so-called Semantic Web [5]. The RDF is also used to represent semantic graph corresponding to a specific knowledge modeling. For example in the AEC (Architecture Engineering Construction) projects, some papers used RDF to model knowledge from heterogeneous sources (electricians, plumbers, architects, etc.). In this domain, some models are developed providing a common syntax to represent building objects. The most recent is the IFC (Industrial Foundation Classes) [6] model developed by the International Alliance of Interoperability. The IFC model is a new type of BIM (Building Information Model) and requires tools to check the consistency of the heterogeneous data and the impact of the addition of new objects into the building.

As the IFC graphs have a large size, their checking, handling and inspections are a very delicate task. In [7] we have presented a conversion from IFC to RDF. In this paper, we propose a new way using formal verification, which consists in the transformation of semantic graphs into a model and verifying them with a model checker. We developed a tool called "RDF2NμSMV" that transforms semantic graphs into a model represented in NμSMV [8] language. After this transformation, NμSMV verifies the correctness of the model written in NμSMV language with temporal logic in order to verify the consistency of the data described in the model of the huge semantic graphs.

The rest of this paper is organized as follows. In Section 2 we present an overview of the semantic graphs, especially the structure of the RDF graphs and the model checking. Then, in Section 3, we describe the mapping of the semantic graphs into models and our approach is defined in Section 4. Finally, we end with the conclusion.

II. An Overview of Semantic Graphs and Model Checking

The RDF is also used to represent semantic graphs corresponding to a specific knowledge modeling. It is a language developed by the W3C to bring a semantic layer to the Web [9]. It allows the connection of the Web resources using directed labeled edges. The structure of the RDF documents is a complex directed labeled graph. An RDF document is a set of triples <subject, predicate, object> as shown in the Figure 1. In addition, the predicate (also called

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property) connects the subject (resource) to the object (value). Thus, the subject and the object are nodes of the graph connected by an edge directed from the subject towards the object. The nodes and the edges belong to the "resource" types. A resource is identified by an URI (Uniform Resource Identifier) [10, 11].

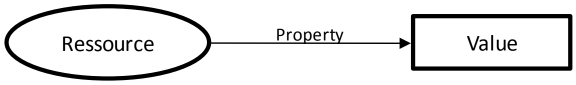
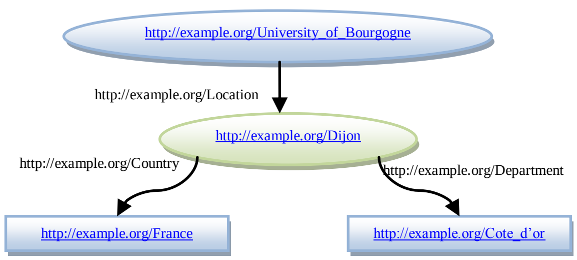


Figure 1. RDF triplet.

The declarations can also be represented as a graph, the nodes as resources and values, and the arcs as properties. The resources are represented in the graph by circles; the properties are represented by directed arcs and the values by a box (a rectangle). Values can be resources if they are described by additional properties. For example, when a value is a resource in another triplet, the value is represented by a circle.



Figure

2. Example of a partial RDF graph.

The RDF graph in the fig. 2 defines a node "University of Bourgogne" located at "Dijon", having as country "France" and as department "Cote d'Or". RDF documents can be written in various syntaxes, e.g., N3 [12], N-Triple [13], and RDF/XML. Below, we present the RDF\XML document corresponding to Figure 2.

```
<rdf:Description
rdf:about="http://example.org/university of
Bourgogne">
  <ex:Location>
       <rdf:Description
rdf:about="http://example.org/Dijon">
       <ex:Country> France</ex:Country>
       <ex:Department>Cote
d'or</ex:Department>
       </rdf:Description>
   </ex:Location>
</rdf:Description>
```

The model checking [14] described in fig. 3 is a verification technique that explores all possible system states in a brute-force manner. Similar to a computer chess program that checks all possible moves, a model checker, the software tool that performs the model checking, examines all possible system scenarios in a systematic manner. In this way, it can be shown that a given system model truly satisfies a certain property. Even the subtle errors that remain undiscovered using emulation, testing and simulation can potentially be revealed using model checking.

To make a rigorous verification possible, properties should be described in a precise unambiguous way. It is the temporal logic that is used in order to express these properties. The temporal logic is a form of modal logic that is appropriate to specify relevant properties of the systems. It is basically an extension of traditional propositional logic with operators that refer to the behavior of systems over time.

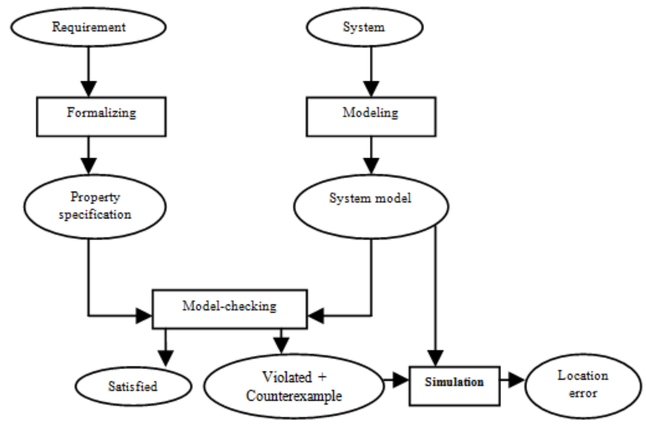


Figure 3. Model Checking approach

The following algorithm explains the way that the model checking works. First we put in the stack all the properties expressed in the temporal logic. All of them are verified one by one in the model and if a property does not satisfy the model, it is whether the model or the property that we must refine. In case of a memory overflow, the model must be reduced. Whereas formal verification techniques such as simulation and model checking are based on model description from which all possible system states can be generated, the test, that is a type of verification technique, is even applicable in cases where it is hard or even impossible to obtain a system model.

```
Algorithm: Model-checking
Begin
While stack ≠ nil do
   P := top (stack);
   while ¬ satisfied (p) then
       Refine the model, or property;
     Else if satisfied (p) then
     P := top (stack);
     Else // out of memory
     Try to reduce the model;
   End
End
```

III. THE MAPPING

The RDF graphs considered here are represented as XML verbose files, in which the information is not hierarchically stored (so-called graph point of view). These RDF graphs are not necessarily connected, meaning they may have no root vertex from which all the other vertices are reachable. The RDF graph transformation into a model is articulated in three steps: exploring the RDF graph, holding election of the root vertex, generating the model of the semantic graph.

A. *Exploring RDF graph*

In order to exploit the RDF graphs, we therefore have to determine whether they have a root vertex, and if this is not the case, we must create a new root vertex by taking care to keep the size of the resulting graph as small as possible.

We achieve this by appropriate explorations of the RDF graphs, as explained below. Let us consider that an RDF graph is represented as a couple (V, E), where V is the set of vertices and $E \subseteq VxV$ is the set of edges. For a vertex x, we note $E(x) = \{y \in V | (x, y) \in E\}$ the set of its successor vertices, and we assume that these vertices are ordered from $E(x)_0$ to $E(x)_{|E(x)|-1}$. This corresponds to the classical data structure for representing graphs in memory, consisting of an array indexed by the vertices and containing in each entry the list of successor vertices of the corresponding vertex. There are several algorithms to traverse a large graph, of these basic algorithms include the best known, depth-first search (DFS) and breadth-first search (BFS). We use depth-first search algorithm illustrated below to explore graph, knowing that the breadth-first algorithm also works in this context:

```
Algorithm: procedure Dfs (x):
begin
  visited(x) :=  true;
   // vertex x becomes visited
  p(x) := 0; // start exploring its successors
  stack := push(x, nil);
 while stack ≠ nil do
y := top(stack);
if p(y) < |E (y)| then
 // y has some unexplored successors
 z := E (y) p(y);
    p(y) := p(y)+1;
    // take the next successor of y
    if ¬visited (z) then
    visited(z) := true;  // visit it
    p(z) := 0;//start exploring its successors
    stack := push(z, stack)
    endif
   else //all successors of y were explored
   stack := pop(stack)
  endif
 end
end
```

We considered here an iterative variant of DFS, which makes use of an explicit stack, rather than the recursive variant given in [15]; this is required in practice to avoid overflows of the system call stack when the algorithm is invoked for exploring large graphs.

B. *Determining a root vertex*

If the RDF graph has no vertex root, we must create a root as to be the successors of all vertices of the graph but it will increase the number of edges. We look forward to doing this by adding a few edges as possible. A vertex x of a directed graph is a partial root if it cannot be reached from any other vertex of the graph. If the graph contains only one partial root, all other vertices of the graph can be reached from the root, otherwise there would be other roots in the partial graph. If the graph has multiple partial roots, the most economical way to provide a root is to create a new record with all the roots as a partial successor: this will add to the graph a minimum number of edges. We compute the set of partial roots in two phases, each one consisting in successive explorations of the graph. The first phase identifies a set of candidate partial roots, and the second one refines this set in order to determine the partial roots of the graph.

Remark: a property must always have a resource and a value; the resource should never be a value with the same predicate, i.e. a loop in the graph.

```
Algorithm: procedure RootElection():
// precondition: ∀ x ∈ V.visited(x) = false
Begin // first phase
 root_list := nil;
 forall  x ∈ V do
  if ¬visited(x) then
   Dfs(x);
   root_list := cons(x, root_list)
  endif
 endfor;
//second phase
 if |root_list|= 1 then
  root := head(root_list)
  // the single partial root is the global root
  else
  forall x ∈ V do  visited(x):= false;
  endfor;
  forall x ∈ root_list do
   // reexplore partial roots in reverse order
   if ¬visited(x) then Dfs(x)
   else
    root_list := root_list \ {x}
     // partial root is not a real one
   endif
  endfor;
  if |root_list| = 1 then
   root := head(root_list)
   // a single partial root is the global root
   else
   root := new_node();
   // new root predecessor of the partial roots
   E(root) := root_list
  endif
   endif
```

The first phase explores the graph until it is fully explored, and inserts in root_list all vertices that have no predecessor. If root_list contains a single vertex, so overall it is the global root of the graph since all the other vertex are accessible from it and it is useless to the second phase has passed. Otherwise, any vertex contained in root_list could also be a root of the graph: the role of the second phase is to determine which of the partial root the root of the global graph is.

The second phase performs a new wave of exploration of the roots contained in partial *root_list* in reverse order in

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which they were inserted in the list. If a root in *root_list* is to be visited by a partial root, it is removed from the list because it is not a partial root. At the end of this phase, all partial roots of the graph are present in *root_list*. Indeed, each vertex is unreachable from the partial roots which were explored during the second phase. A new root is created (see Fig. 4), having as successor all the partial roots of *root_list*, which ensures that all vertices of the graph are accessible from the new root. Therefore, such a summit is inaccessible from other nodes of the graph.

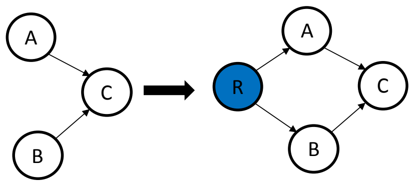

Figure 4. A root is a single node that has no predecessor. In this graph, we have node A and node B, two roots, and then we will create a new virtual root (blue circle "R") that points to the two roots.

The algorithm for determining a root has a complexity O(|V|+|E|), linear in the size of the graph (number of vertices and edges), since each phase visits every state and traverses every edge of the graph only once. Given that the graph must be traversed entirely in order to determine whether it has a root or not, this complexity is optimal.

C. Generating the model

The third step is divided into three sub-steps. The first one consists in creating the table of all triplets by exploring the entire graph; the second one consists in generating the correspondence table and the last one in producing the model representing the semantic graph in NµSMV language.

Table of triplets – Going through the RDF graph by graph traversal algorithms, we will create a table consisting of resources, properties and values. In our RDF graph, the resource is a vertex, the property represents the edge and the value is the successor vertex corresponding to the edge of the vertex. The table of triplets of the RDF graph is useful to the next sub-step.

Correspondence table – In this second sub-step, RDF2NµSMV generates a table of correspondence. This table contains an identifier for each resource, property and value.

The model – In this last sub-step, we will write the model in NµSMV language for RDF2NµSMV tool, corresponding to the RDF graph that we want to check.

IV. THE VERIFICATION WITH MODEL CHECKER

As we saw in Section 2, the model checker needs properties in order to check the model of semantic graphs. These properties are expressed in temporal logic. The concepts of temporal logic used for the first time by Pnueli [16] in the specification of formal properties are fairly easy to use. The operators are very close in terms of natural language. The formalization in temporal logic is simple enough although this apparent simplicity therefore requires significant expertise. Temporal logic allows representing and reasoning about certain properties of the system, so it is well-suited for the systems verification. There are two main temporal logics, that is linear time and branching time. In linear time temporal logic, each execution of the system is independently analysed. In this case, a system satisfies a formula f, if f holds along every execution. The branching time combines all possible executions of the system into a single tree. Each path in the tree is a possible representation of the system execution.

This Section details our approach which consists in transforming semantic graphs into models in order to be verified by the model-checker. For this, we have developed a tool called "RDF2NµSMV" that transforms semantic graphs into NµSMV language.

We use NµSMV as model checkers to verify the model of semantic graphs. NµSMV is the amelioration of SMV model checker, working on the same simple principles as SMV. NµSMV verifies the properties in both linear time logic and computation tree logic.

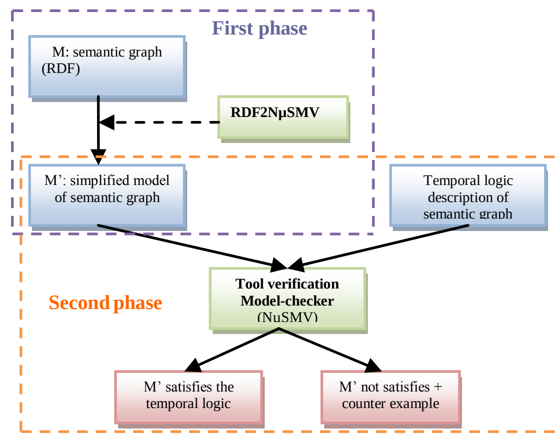

Figure 5. Our architecture.

The architecture of the fig. 5 is divided into two phases. The first phase concerns the transformation of the semantic graph into a model using our tool "RDF2NµSMV", as described in Section III. The second phase concerns the verification of the properties expressed in temporal logic on the model using the model-checker NµSMV.

To illustrate our approach, we take an RDF graph represented in the Figure 6 and a temporal logic expressed in the table 1 to verify if the BIM "b1" contains a floor.

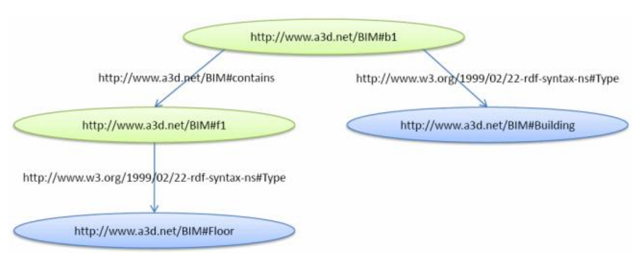

Figure 6. Example of partial RDF graph.

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TABLE 1. Temporal logic formula.

Temporal logic	Meaning	Result
Eventually (b1 → Next Next floor)	Is there a floor after two states starting from the state b1	True

We tested several RDF graphs on our tool “RDF2NµSMV”, graphs representing buildings as shown in Figure 7, using a machine that runs on a processor with a capacity of 2.4 GHz and 4 GB of RAM, calculating the time of conversion as shown in Fig 8. Note that the RDF2NµSMV tool is faster in converting semantic graphs. We have almost 22 seconds for a graph of 53 MB size. The transformation tool follows a polynomial curve. In Fig. 9, we see the size of the converted semantic graphs from RDF to NµSMV language.

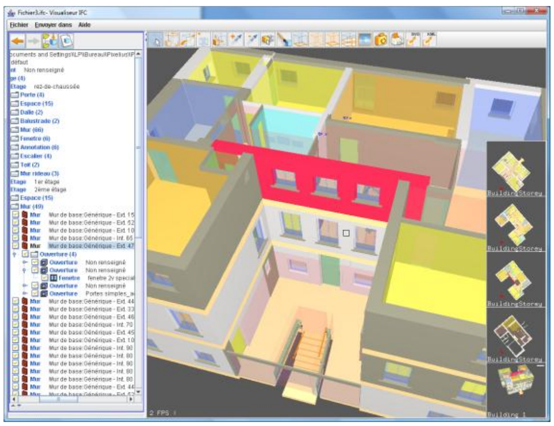

Figure 7. The 3D view of an IFC file.

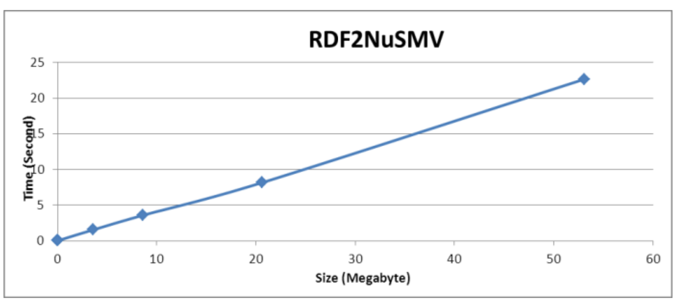

Figure 8. Time conversion of semantic graphs.

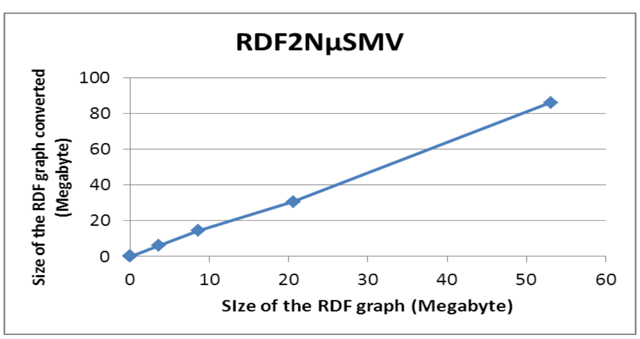

Figure 9. Size of the models

V. CONCLUSION

This paper presents how to transform a semantic graph into a model for verification by using a powerful formal method, that is the “model checking”. Knowing that the model-checker does not understand the semantic graphs, we developed a tool RDF2NµSMV to convert them into NµSMV language in order to be verified with the temporal logics. This transformation is made for the purpose of classifying large semantic graphs in order to verify the consistency of IFC files representing 3D building. We notice the advantage of NµSMV, whose verification can be made with both linear time logic and computation tree logic formulas.

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Tag Relevancy for Similar Artists

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***Abstract*—In music information retrieval, the rank position in similar artist lists have gained a lot of attention due to the surge in online music listening and semi-automated song recommendation approaches. Artist tags with respect to genre, style and mood are critical components in assisting these online communities. In this paper, we examine the relationship between an artist and associated similar artists by considering an artist's top tags. We are seeking to uncover patterns and correlations between pairwise rank positioning e.g., rank 1-2, rank 2-3, and so forth. The experiments show positive correlation between rank position pairs; however, the strength of the correlation is not as high as expected.**

***Keywords*-music information retrieval, knowledge dissemination, algorithms, experimentation**

I. Introduction

The music information retrieval field has grown significantly in recent years due to online music communities such as AllMusic, The Echo Nest, Idiomag, Last.fm and Pandora. In fact, the landscape for finding new music has been vastly transformed thanks to the Internet [3] as it has helped new artists such as Taylor Swift (country) and Sean Kingston (reggae) find new listeners as well. Several online music communities have user interface limitations and advantages. For instance, song 'replay' and/or previous song option is not allowed while permitting the use of 'skip' and 'pause' option with the advances of data streaming and network bandwidth capabilities.

Each online music listening website allows music listeners to create a user account in hopes of tracking music genre and artist preferences. In most cases, the user chooses a radio station with a programmed playlist. In contrast, Pandora also provides an option to construct a customized playlist with the input of a single music artist to begin the personalized user station. To assist their user in building a playlist, song recommendations are made through leveraging similar artists' ranking. Any form of music recommendation makes use of the artist profile, including primary genre, style and mood, and the user profile, including song ratings and song listening history.

Semi-automated song recommendation services are powered by the quality of similar artist rankings with the expectation that the most similar artists are ranked highly with respect to artist characteristics including audio features, genre, style, mood and music tags. As shown in Figure 1 [9], artist similarity is subjective. Daughtry (Rock), Bob Marley (Reggae) and Usher (R & B) are music artists from very distinctive genres; however, the number of similar artists vary and the proximity of their similar artists to the initial artist (center) differ widely. In this paper, we consider only music tag relevancy as related to a set of similar artists. Music tags can represent audio tones, genre, style and mood attributes unique to each artist. Hence, we expect that the music tag performance of similar artists would decrease as the rank positions increase.

Through experimentation, we record three types of performance e.g., precision, reciprocal rank and covariance, to show the relationship between an artist's music tags and corresponding similar artists' music tags. The precision performance measure records the number of tags overlapping between an artist and each of its similar artist. Reciprocal rank performance, on the other hand, computes the strength of the matching tags as higher rank positions receive greater weight. We then calculate the covariance between pairwise rank position e.g., rank 1-2, rank 2-3, and so forth, in order to investigate the quality of similar artist rankings.

The specific contributions of this paper are:

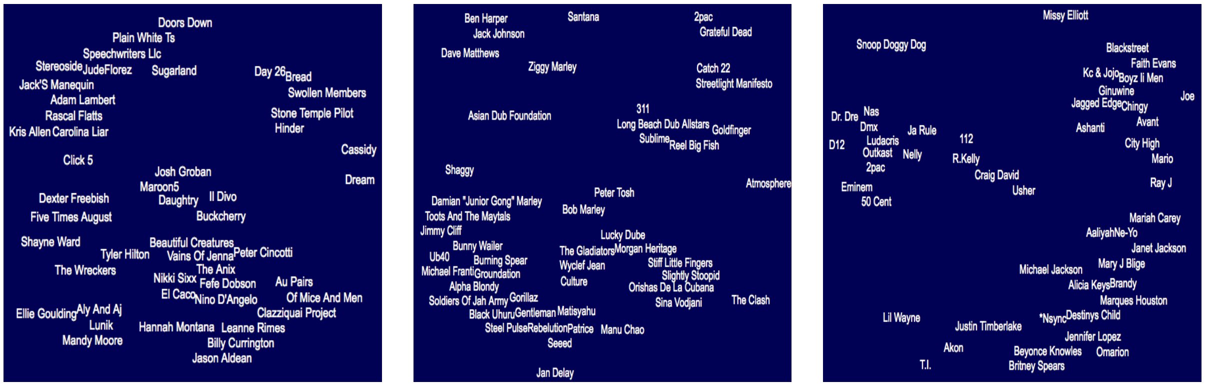
- study music tag relevancy in the context of similar artist rankings,
- perform a quantitative study showing the influence of music tags on similar artist rankings using Last.fm music data including 10 genres and nearly 500 artists
- conduct a performance analysis of tag relevancy considering precision, reciprocal rank and Spearman's ρ rank correlation coefficient.

The rest of the paper is organized as follows. Section II reviews the relevant literature in music recommendation research. In Section III, we discuss the popular online music communities and describe our approach to music tag relevancy for similar artists. Section iV contains our experimental evaluation. We summarize our findings and discuss future work in Section V.

II. Related Work

Music recommendation research has three main branches: (1) content-based through audio processing, (2) pre-defined or user-generated tagging of artists, albums and/or songs and (3) mixed music content-based and tagging methods.

a) ***Content-based approaches****:* A large section of music research focuses on content-based methods by processing the audio file in order to correctly determine a song's genre. However, content-based methods are computationally expensive,



(a) Daughtry (b) Bob Marley (c) Usher

Fig. 1. Sample Artist Similarity Maps from Music-Map

but has led to improvements in the music genre labeling, artist style classification and identifying song moods. These improvements have been applied to construct music recommendation systems [2], [11], [18] by using collaborative filtering methods to provide user-specific results using information from many users. Other prior work [2], [18] concentrates on the users' playlist through song properties including pitch, duration and loudness. While users' playlist are customized, artist similarity can assist in generating a playlist but has a wider appeal with greater song and artist diversity. The primary disadvantages of content-based approaches are the higher computational cost of pre- and post-processing of sound recordings and the identification of relevant audio features to assist in distinguishing songs and vocalists.

*b) **Tagging approaches**:* In recommending music using text, tags associated with artists and/or songs are typically classified into either pre-defined expert opinion or community-based labeling categories. Assessing the quality of artist tagging and similarity for determining appropriate ground truth remains a challenging problem due to the current state of ill-formed music tag semantics [4].

Regardless, tagging offers the average user and experts an opportunity to catalog music in a detail unachievable with audio features. Bischoff et al. [1] emphasize that music listeners tend to label music and artist according to genre style and enjoy providing personalized opinions of the music. In Shao et al. [17], style and mood tags produced nearly identical results using a content-based approach comparing 12 artists. Magno and Sable [13] show the similarity of human recommendation with automate music recommendation services provided by *Last.fm* and *Pandora*. Nevertheless, the results also note the limitations of human recommendation as some dependencies are not captured for an individuals musical taste.

*c) **Mixed method approaches**:* Combining sound recording and tag content has become a popular method through the explosion of digital songs/albums e.g., Apple's iTunes and Google's YouTube. One goal of 21st century music research is building personal music information retrieval systems. Mixed method approaches tend to focus on either establishing customized artist similarity ([12], [19]) or song recommendation ([15], [16]). Through clustering, Li et al. [12] propose bimodal learning technique to semi-automate the process of grouping similar artists based on songs, albums and artists using the AllMusic repository as ground truth. Based on the MapReduce framework, Zhao et al. [19] consider tag semantic similarity in hopes of minimizing semantic loss and tag noise, while ensuring attribute diversity. The research reveals the scarcity of style and mood tags, but the need to give this content more importance when available. Both MusicWiz [15] and MusicBox [16] proposes specialized music management systems. MusicWiz is individual user-focused as to personalize song playlists according to an individual's feelings and memories while MusicBox is community-focused with the goal of exploiting correlation between users, tags and music content.

III. Music Tagging

Mainstream music listening is no longer primarily on the radio and playing CDs, but has become to mainly digital activity. On the Web, online music communities have been designed as a digital repository of artists, albums, songs, artist similarity and music influences & followers. In III-A, we discuss benefits and limitations of five popular music communities. Then in III-B, we describe specific challenges of music and artist tagging. We also discuss an approach to evaluate relevancy of tags for Last.fm.

A. Online Music Communities

1) **AllMusic. [6]** Originally All Music Guide or AMG, *AllGuide* offers consumers access to artist/group information containing biography, discography, songs, credits and charts & awards. The AllMusic database also includes descriptive content (genres, styles, tones, moods, themes and nationalities) and relational content (similar artists, influences and followers). The genres, styles and moods content are assigned to each artist from a pre-defined expert- approved list.
2) **The Echo Nest. [5]** Due to intellectual property rights, *Echo Nest* has not unveiled their process of relating artists. Nevertheless, the company has revealed that artist information is generated in a number of ways such as the analyzation of the raw music, blogs, song lyrics

and message board postings. While the exact method in rating artist similarity is unknown, test queries have shown that their artist database is multi-faceted with many musicians and genres.

3) **Idiomag. [7]** *Idiomag* uniquely labels, or tags, a given artist by weighted genre names from a preset list of 144 acceptable genre names maintained by the company's staff. Then a manual weight is applied to each of the tags by Idiomag's expert musicians/music lovers. Lastly, the artists are ordered according to their labels' values.
4) **Last.fm. [8]** *Last.fm* is highly user-centric by allowing any user to create self-defined tags. In contrast to *Idiomag*, *Last.fm* supports user-tagging which has led to a number of issues, including *duplicate tags* due to grammatical errors and *maliciously false tags* applied to artists. *Last.fm* combats this challenge by counting multiple occurrences of a single tag for a single artist as a vote for that artists tag. A tag's votes are reflected in how each tag is weighted. To determine musical similarity for an artist, *Last.fm* compares the tags of all artists in their database to the target artist.
5) **Pandora Radio. [10]** Originally the Music Genome Project (2000-2008), *Pandora Radio* is an automated customizable music recommendation service available only in the United States. The music recommendations are made based on nearly 400 attributes to describe songs using a U.S.-patented mathematical algorithm. At the core, the 5 music genomes are Pop/Rock, Hip-Hop/Electronica, Jazz, World Music and Classical. Artist and song labels are embedded within *Pandora Radio* and only partially accessible through the "Why this song?" choice in the Menu tab.

The Echo Nest, Idiomag and Last.fm offer well-developed Web APIs which allows anyone to develop specialized programs using their music data. Marshall [14] aggregates the artist tags from these three Web APIs in order to extract better and more consistent similar artists. However, Echo Nest, Idiomag and Last.fm return relatively diverse similar artists lists with a majority of agreement occurring within the top-3 similar artist list. Hence, artist similarity aggregation broadens the identification of similar artists; however, these online music communities do not have a consensus on artist similarity given a particular artist. We now examine the quality of these artist similarity rankings.

B. Tag Relevancy of Similar Artists

Much of the previous work [1], [4], [13], [16], [17], [19] used Last.fm music data since Last.fm remains an open-source environment since its creation. As a result, we use Last.fm music data as our experimental research platform.

Based on the artists presented in Figure 1, we display the top-10 tags and score values in Table I. We notice that all tags in rank position 1 have a score value of 100 indicating a universally recognizable tag associated with the artist. In addition, the tags at rank position 1 represent the primary artist genre. From rank position 2 downward, the score values decrease significantly with ties allowed where the tags are mainly genre subcategories and includes the artist's name.

Daughtry	Bob Marley	Usher
rock (100)	reggae (100)	rnb (100)
alternative rock (80)	roots reggae (25)	Hip-Hop (52)
alternative (47)	Bob Marley (18)	soul (38)
hard rock(34)	ska (15)	pop (38)
American Idol (22)	rock (9)	rap (26)
post-grunge (13)	jamaican (4)	Usher (20)
daughtry (9)	classic rock (3)	hip hop (11)
male vocalists (7)	chill (3)	male vocalists (8)
pop rock (7)	singer-songwriter (3)	r & b (7)
american(5)	roots(2)	r and b (6)

TABLE I
LAST.FM TOP TAGS WITH FREQUENCY COUNTS

With Last.fm music data, we observe some inherent tag semantic overlap, such as rock, alternative rock, alternative, that makes assessing the quality of the similar artists' tags more difficult. For instance, depending on the matched tags between two artist, a distinctive similarity relationship may exist. In the *TRAS* function below, we follow a template in evaluating the quality of tags with respect to the similar artist list returned from the Last.fm GETTOPTAGS method. A string IA denotes the initial artist, a string array $similars$ holds n similar artists of IA and retrieving the top-k tags serve as input and returns an array of performance analysis calculations for each (IA, SA_i) pair.

function TRAS(initial:IA,similars:$\{SA_1, \ldots, SA_n\}$, count:k)
resultArray=empty //holds the result of performance measure
IAtags = LASTFM.GETTOPTAGS(initial,k)
for $i = SA_1$ to SA_n **do**
  SAtags = LASTFM.GETTOPTAGS(i, k)
  //MEASURE$(\cdot,\cdot)$ is a place holder for a performance measure e.g., precision, reciprocal rank, covariance
  result = MEASURE(IAtags, SAtags)
  resultArray.append(result)
return resultArray

IV. EXPERIMENTAL STUDY

To test the degree of similarity amongst music artists, we ran nearly 500 artist queries over 10 popular music genres. We manually selected the query artists as to guarantee the return of at least 10 similar artists from Last.FM. We present a sample of the artists queried in Table II.

For each artist query, we first gather the top 10 similar artists. Then, we extract the top 10 popular tags associated with each artist query, we denote as initial artist (IA) and its similar artists, we denote as SA. We chose the first 10 tags because the score values associated with these rank positions are consistently above 0. We examine the tag match performance between IA and each of its SA using two measures. The first error measure, precision P, is calculated by taking the two tag lists m_{IA}, m_{SA} and finding the number of common tags in relation to the number of returned elements k. We chose $k = 10$. Formally, precision is defined as follows

$$P_k(m_{IA}, m_{SA}) = \frac{m_{IA} \cap m_{SA}}{k}$$

Alternative	50	Disturbed, Korn, Muse, Nickelback, Papa Roach, The Fray
Blues/Jazz	48	B.B. King, Nina Simone, Otis Redding, Ray Charles, Stevie Wonder, Doris Day
Country	50	Garth Brooks, Faith Hill, Toby Keith, Vince Gill, Willie Nelson
Electronic	48	Air, Daft Punk, Depeche Mode, Massive Attack, Zero 7
Funk	49	Culture Club, Funkadelic, Musiq, Parliament, Sade
Hip-Hop	48	Janet Jackson, John Legend, TLC, Monica, Black Eyed Peas
Pop	50	Blackstreet Boys, Britney Spears, Coldplay, Justin Timberlake, Lady Gaga
Rap	50	Dr. Dre, Eminem, Lil' Wayne, Run DMC, Snoop Dogg, Young Jeezy
Reggae	45	Bob Marley, Black Uhuru, Matisyahu, Shaggy, Toots & The Maytals
Rock	49	Creed, Finger Eleven, Hinder, Rob Thomas, Train

TABLE II
SAMPLE ARTISTS

Precision is a commonly used measure to distinguish between relevance and non-relevance. However, precision does not indicate the degree of relevance, e.g., the rank position of relevant data. To assess relevancy based on rank position, we use the reciprocal rank measure. Formally, reciprocal rank is defined as follows

$$RR_k(m_{IA}, m_{SA}) = \sum_{l=1,\ldots,k} \frac{1}{l} \;\; \text{if } m_{IA}(l) = m_{SA}(q)$$

where l, q ($l = q$ or $l \neq q$) refer to a position in a ranking. In the case when precision is 100% and thus, the reciprocal rank value is 2.93 (= $\sum_{l=1}^{k=10} 1/l$).

Genre	$P_{10}(min, max)$	$RR_{10}(min, max)$
Alternative	(33.20%, 73.8%)	(1.35, 2.51)
Blues/Jazz	(23.92%, 75.29%)	(1.06, 2.54)
Country	**(8.00%, 54.00%)**	**(0.41, 1.93)**
Electronic	(25.91%, 75.51%)	(1.08, 2.45)
Funk	(17.21%, 67.60%)	(0.73, 2.37)
Hip-Hop	(2.29%, 66.04%)	(0.08, 2.35)
Pop	(33.40%, 72.00%)	(1.39, 2.48)
Rap	(25.40%, 70.80%)	(1.15, 2.43)
Reggae	(20.21%, 62.82%)	(0.93, 2.26)
Rock	**(38.20%, 79.20%)**	**(1.51, 2.67)**

TABLE III
PRECISION AND RECIPROCAL RANK AVERAGE INTERVALS

In our first set of experiments, we observe the interval range of both precision and reciprocal rank for each music genre as shown in Table III. We record the minimum and maximum value observed for each IA and its 10 SA e.g., for alternative music, we record 50 minimum and 50 maximum precision values (as well as 50 minimum and 50 maximum reciprocal rank values). The observed precision values ranges from the lowest maximum of 54% for Country music to the highest maximum of 79% for Rock music. The reciprocal rank values mirrored those observed in the precision calculations. The reciprocal rank, however, assist in determining the rank positions of the matching tags. Hence, for Country music, the minimum 8% precision is seen on average 1 out of the 10 rank positions at position 2 ($RR(\cdot,\cdot) = \frac{1}{2}$) or position 3 ($RR(\cdot,\cdot) = \frac{1}{3}$) in order to compute a minimum reciprocal rank of 0.41. To achieve the maximum precision and reciprocal rank in Country music, 5 or 6 rank position has matching tags giving the average maximum precision of 54% and the matching rank position are either {1, 3, 4, 5, 6} ($RR(\cdot,\cdot) = \frac{1}{1} + \frac{1}{3} + \frac{1}{4} + \frac{1}{5} + \frac{1}{6} = 1.95$) or {1, 4, 5, 6, 7, 8} ($RR(\cdot,\cdot) = \frac{1}{1} + \frac{1}{4} + \frac{1}{5} + \frac{1}{6} + \frac{1}{7} + \frac{1}{8} = 1.88$).

We further investigate the relationship between rank positions with respect to the precision and reciprocal rank values in our second and third experiments. We consider the change in performance values between consecutive rank positions e.g., Rank 1 to Rank 2, Rank 2 to Rank 3, and so forth. We expect that the performance between consecutive rank positions would be increasingly negative. In other words, when compared to the initial artist IA, the tag similarity of a similar artist SA at rank 1 (SA_1) is greater than the tag similarity of a similar artist at rank 2 (SA_2). In addition, we assume that as SA_w in which $w \to 10$, the difference between rank w and $w + 1$ increases. Last.fm returns a "count" value for each tag indicating the popularity of the tag with the corresponding artist. These counts are consistently and quickly decreasing in value.

Rank Position Sum Difference. For each consecutive pair of rank positions, we compute the average precision sum difference for each genre. The results are displayed in Table IV. We anticipate that the precision at rank position 1 would be higher than precision at rank position 2, precision at rank position 2 would be higher than precision at rank position 3 and so forth. However, our observations did not lead to this conclusion. Instead, we notice a *low and mainly positive* precision difference between consecutive pairs of rank positions. The highest precision value difference is -0.34 (or increase of 34% in the Hip-Hop genre from Rank 1 to Rank 2). The majority of the precision value difference is $\leq$ +/- 0.050 (or increase/drop of precision by less than 5%). In fact, no genre tested is consistently negative or positive in terms of their consecutive rank positions. This oscillating precision performance implies that the similarity ordering of artists is not primarily based on tag label.

Through precision performance, we can only assess *how many* tags consistently matched. We have no information about *which rank position* the artists appeared in the ranking. The reciprocal rank performance provides this evaluation. In Table V, we compute the average sum difference for each genre and rank position pair. Each cell value records the magnitude change between rank positions. A majority of the change is in a [-0.100,0.100] interval or within one position

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Genre	Rank 1-2	Rank 2-3	Rank 3-4	Rank 4-5	Rank 5-6	Rank 6-7	Rank 7-8	Rank 8-9	Rank 9-10
Alternative	-0.006	0.044	-0.010	0.050	-0.056	-0.002	0.034	-0.064	0.054
Blues/Jazz	0.082	0.002	0.037	-0.012	0.006	-0.012	0.047	-0.027	0.031
Country	-0.034	0.028	-0.036	0.028	-0.064	0.076	0.012	-0.050	0.012
Electronic	0.016	0.027	-0.012	0.041	-0.006	0.031	-0.018	-0.039	0.076
Funk	0.078	-0.066	0.048	-0.030	0.028	0.016	-0.018	-0.014	0.018
Hip-Hop	**-0.340**	-0.112	-0.021	0.060	-0.019	-0.010	0.033	-0.029	-0.025
Pop	0.044	-0.036	0.032	-0.034	0.030	0.010	0.016	0.058	-0.042
Rap	-0.082	0.044	0.016	0.018	0.012	-0.036	0.006	0.032	-0.028
Reggae	0.004	0.054	-0.002	0.009	0.004	0.020	0.043	-0.061	0.011
Rock	0.020	0.004	0.002	0.022	-0.004	0.034	0.006	-0.004	-0.016

TABLE IV
PRECISION SUM DIFFERENCE: RAW VALUES (MULTIPLY BY 100 FOR PERCENTILE)

Genre	Rank 1-2	Rank 2-3	Rank 3-4	Rank 4-5	Rank 5-6	Rank 6-7	Rank 7-8	Rank 8-9	Rank 9-10
Alternative	-0.076	0.154	-0.078	0.151	-0.142	0.009	0.146	-0.210	0.116
Blues/Jazz	0.186	-0.016	0.060	0.034	-0.042	0.029	0.101	-0.100	0.103
Country	-0.130	0.053	-0.102	0.111	-0.230	0.208	0.024	-0.108	0.166
Electronic	-0.017	0.130	-0.078	0.090	-0.052	0.112	-0.004	-0.155	0.265
Funk	0.224	-0.189	0.150	-0.085	0.097	0.032	-0.107	0.099	-0.030
Hip-Hop	-1.419	-0.371	-0.037	0.165	-0.115	-0.043	0.097	-0.088	-0.077
Pop	0.053	-0.148	0.139	-0.088	0.074	0.001	0.011	0.189	-0.059
Rap	-0.317	0.101	0.048	0.100	-0.038	-0.121	-0.005	0.085	-0.007
Reggae	0.052	0.169	-0.033	0.042	-0.103	0.117	0.209	-0.265	0.036
Rock	0.071	0.023	0.003	0.041	0.028	0.074	0.030	-0.015	-0.114

TABLE V
RECIPROCAL RANK SUM DIFFERENCE: RAW VALUES (DIVIDE BY 2.93 FOR NORMALIZATION)

Genre	Rank 1-2	Rank 2-3	Rank 3-4	Rank 4-5	Rank 5-6	Rank 6-7	Rank 7-8	Rank 8-9	Rank 9-10
Alternative	0.544	0.651	0.668	0.640	0.627	0.617	0.588	0.578	0.574
Blues/Jazz	0.333	0.376	0.404	0.448	0.452	0.475	0.459	0.465	0.438
Country	0.083	0.165	0.276	0.310	0.309	0.330	0.329	0.325	0.352
Electronic	**0.699**	**0.627**	**0.633**	**0.638**	**0.608**	**0.598**	**0.569**	**0.559**	**0.539**
Funk	0.358	0.397	0.380	0.347	0.358	0.377	0.382	0.387	0.377
Hip-Hop	**-0.010**	**0.017**	**0.127**	**0.216**	**0.181**	**0.211**	**0.235**	**0.236**	**0.257**
Pop	0.551	0.470	0.515	0.530	0.555	0.553	0.554	0.557	0.578
Rap	0.260	0.426	0.461	0.518	0.521	0.512	0.530	0.537	0.542
Reggae	0.565	0.648	0.581	0.530	0.528	0.543	0.545	0.544	0.547
Rock	0.512	0.495	0.375	0.361	0.359	0.364	0.383	0.401	0.411

TABLE VI
SPEARMAN'S ρ EVALUATION OF PRECISION VALUES

Genre	Rank 1-2	Rank 2-3	Rank 3-4	Rank 4-5	Rank 5-6	Rank 6-7	Rank 7-8	Rank 8-9	Rank 9-10
Alternative	0.592	0.669	0.688	0.638	0.611	0.602	0.594	0.579	0.577
Blues/Jazz	0.470	0.490	0.483	0.530	0.520	0.524	0.495	0.499	0.483
Country	0.254	0.371	0.436	0.470	0.486	0.493	0.500	0.488	0.484
Electronic	**0.831**	**0.714**	**0.686**	**0.666**	**0.636**	**0.623**	**0.594**	**0.593**	**0.570**
Funk	0.326	0.381	0.390	0.354	0.375	0.386	0.402	0.397	0.393
Hip-Hop	**0.016**	**0.041**	**0.145**	**0.218**	**0.210**	**0.240**	**0.259**	**0.270**	**0.290**
Pop	0.544	0.506	0.553	0.544	0.564	0.576	0.569	0.587	0.607
Rap	0.345	0.509	0.545	0.603	0.595	0.593	0.620	0.629	0.625
Reggae	0.571	0.663	0.621	0.551	0.546	0.568	0.564	0.553	0.555
Rock	0.468	0.364	0.246	0.228	0.259	0.291	0.313	0.330	0.344

TABLE VII
SPEARMAN'S ρ EVALUATION OF RECIPROCAL RANK VALUES

at rank 10. Hence, we conclude that, on average, the matched tags differ by one or two rank positions in the higher ranks e.g., rank 8, rank 9 or rank 10. The anomaly is the Hip-Hop genre at rank 1-2 with a value of -1.419, which implies rank 1 has low matching tags to a much higher matching tags at rank 2.

Spearman's ρ Evaluation. However, precision nor reciprocal rank measure the covariance of two variables or two rank positions in our case e.g., rank 1-2, rank 2-3, etc. To assess the covariance between rank positions, we use the Spearman's rank correlation coefficient (Spearman's ρ). Formally, Spearman's ρ is defined as follows

$$\begin{aligned} rho_k(rp_X, rp_Y) &= \frac{cov(X,Y)}{\sigma_X \sigma_Y} \qquad (1) \\ &= \frac{\sum(X-\bar{X})(Y-\bar{Y})}{\sqrt{\sum(X-\bar{X})^2}\sqrt{\sum(Y-\bar{Y})^2}} \end{aligned}$$

Spearman's ρ values have a [-1,1] interval in which -1 denotes that the two variables are negatively correlated, 1 denotes that the two variables are positively correlated and 0 denotes no correlation between these two variables. We calculate the covariance between rank positions for both the precision and reciprocal rank performance, which are displayed in Tables VI and VII. The Electronic genre is the most positively correlated while Hip-Hop genre is the least correlated. The Spearman's ρ evaluation of the reciprocal rank values are only slightly different from the equivalent precision values in Table VI. We conclude that there is moderate correlation between consecutive rank positions for both performance measures. The oscillating nature between rank positions for each genre observed in Tables IV and V is observed once again as the positive correlation slightly fluctuates.

V. Conclusion

Music listening has become a mainly digital activity with online music communities such as Pandora and Last.fm. Music listening has become a science with relevant song recommendation methods at its core. Song recommendations are guided by appropriate artist similarity rankings. Using the frequently-used Last.fm music data across 10 genres, we investigate the relevancy of music tags with respect to the similar artists' ranking by looking at pairwise rank positions. We expected to discover a strong relationship between an artist and its corresponding similar artists. With precision, we explore how many tags consistently matched while with reciprocal rank, we evaluate the rank position of matching tags. In both performance analysis measures, we did not observe any logarithmic, polynomial, power or exponential trendlines. We also perform a Spearman's ρ evaluation across genre and rank positions. We found a moderately positive correlation between pairwise rank positions.

In the future, we plan to examine how to better randomize music song recommendation by mixing highly and moderately similar artists. To this end, we will consider a two-prong approach: 1) artist tag classifications and 2) artist song collaborations. We will investigate the influence of artist song collaborations on artist similarity. These song collaborations can increase the interestingness of the song by highlighting the unique sound of each artist. The tends in popular song collaborations based on determining genre, style and mood correlation of each artist.

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Usability Heuristics for Interactive Digital Television

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***Abstract*— Usability evaluation for applications based on emerging information technology brings new challenges. Interactive Digital Television (iDT) is considered as the convergence of television and computer technologies. The main iDT feature is that the user may interact with the application; therefore usability should be a main concern when designing iDT applications. Current research usually focuses on iDT applications from a technical point of view, rather than a user–centered approach. There is a need for new usability evaluation methods or at least for the use of traditional evaluations in novel ways. A set of heuristics is proposed, in order to help the usability evaluations of iDT applications.**

Keywords- usability, usability heuristics, interactive digital television

I. INTRODUCTION

Interactive Digital Television (iDT) is considered as the convergence of television and computer technologies, which gather three typical features: interactivity, customization and digitization [1]. The iDT exceeds the analog TV in several aspects: capacity, better use of the spectrum, greater immunity to noise and interference, better sound and picture quality, potential for transmission of data simultaneously, saving power transmission. However, the main iDT advantage is that the user may interact with the application [2].

Usability evaluation for applications based on emerging information technology brings new challenges. Is it the classical concept of usability still valid? Which are the dimensions of the (new) usability? How can it be measured? How should we develop for (better) usability? There is a need for new evaluation methods or at least for the use of traditional evaluations in novel ways [3].

The ISO/IEC 9241 standard defines the usability as the extent to which a product can be used by specified users to achieve specified goals with effectiveness, efficiency and satisfaction in a specified context of use [4]. Usability evaluation methods are commonly divided into inspection and testing methods. Inspection methods find usability problems based on the expertise of usability professionals. Testing methods find usability problems through the observation of the users while they use (and comment on) a system interface.

Usability evaluation is needed particularly if the design concept is new. Users look for more than just a usable product; they look for a pleasing and engaging experience [5]. Therefore, usability should be a main concern when designing interactive iDT applications. Current research usually focuses on iDT applications from a technical point of view, rather than a user–centered approach. There is a necessity to establish methodologies that could lead to applications with a high level of usability. Such methodologies have to include accurate usability evaluations.

Heuristic evaluation is a widely used inspection method [6] [7]. A group of evaluators (usually from three to five) inspect the interface design based on a set of usability heuristics. In order to ensure independent and unbiased evaluations, the inspection is performed individually. After all individual evaluations have been completed, the evaluators are allowed to communicate and have their findings aggregated in a single list of usability problems. Later on, each evaluator assigns scores to each problem's severity and frequency (on a 0 to 4 scale, from minor/less frequent to major/more recurrent). Severity and frequency are summed in order to get problem's criticality. Problems are ranked based on their average severity, frequency and criticality. The usability evaluation report includes usability problems, solution proposals, as well as positive findings.

Heuristic evaluation is easy to perform, cheap and able to find many usability problems (both major and minor problems). However, it may miss domain specific problems. That is why the use of appropriate heuristics is highly significant.

The paper focuses on usability heuristic evaluation of iDT applications. A set of 14 specific usability heuristics is

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proposed. Section 2 presents the methodology that has been used in heuristics' development. Section 3 highlights the main characteristics of iDT applications. The iDT usability heuristics proposal is presented in section 4. Section 5 presents preliminary conclusions and future works.

II. DEFINING USABILITY HEURISTICS FOR INTERACTIVE TELEVISION

In order to develop usability heuristics for iDT, a specific methodology was applied [8]. The methodology to establish new usability heuristics includes 6 stages:

- STEP 1: An *exploratory* stage, to collect bibliography related to the main topics of the research: specific applications, their characteristics, general and/or related (if there are some) usability heuristics.
- STEP 2: A *descriptive* stage, to highlight the most important characteristics of the previously collected information, in order to formalize the main concepts associated with the research.
- STEP 3: A *correlational* stage, to identify the characteristics that the usability heuristics for specific applications should have, based on traditional heuristics and case studies analysis.
- STEP 4: An *explicative* stage, to formally specify the set of the proposed heuristics, using a standard template.
- STEP 5: A *validation* (experimental) stage, to check new heuristics against traditional heuristics by experiments, through heuristic evaluations performed on selected case studies, complemented by user tests.
- STEP 6: A *refinement* stage, based on the feedback from the validation stage.

An early version of the iDT usability heuristics was proposed in 2009 [9]. Later on, STEP 1 to STEP 4 of the methodology were performed, and a refined usability heuristic proposal was defined. As no specific iDT usability heuristics were found, the proposal is based on the well-known and widely used Nielsen's 10 heuristics [7]. However, heuristic proposals for other fields, such as Social TV, Virtual Worlds and Grid Computing were also used. STEP 5 and STEP 6 are yet to be performed.

III. INTERACTIVE DIGITAL TELEVISION CHARACTERISTICS

Nowadays the concept of television does not refer to a specific device, but rather a specific kind of content available almost everywhere, freeing television from the TV set and bringing it out of home. Additionally, current trends combine iDT and the web; users may navigate on internet from their TV sets, download and use applications, download content or customize the TV schedule.

iDT applications have a set of basic features that should be considered when evaluating their usability:

- *Interactivity:* iDT applications should offer bidirectional communication, a fundamental requirement of any interactive system. An iDT application should invite user to participate, in order to have a more active experience while watching content. Interactivity is the ability to offer additional content to the television programs, allowing the user to view associated information with audiovisual content, to view the TV channels' schedule, to participate in contests, polls, to buy products or services, and even to participate in the television programs creation/customization.
- *Customization:* iDT applications should allow customization in terms of content, appearance and others, taking into account users' needs, skills, personal preferences, etc.
- *Physical features of interaction:* Human vision is optimal at a particular distance from the screen; therefore, iDT applications should take into account screen resolution and contrast. Traditionally, users watch TV in an environment that is oriented to relaxation and comfort. However, nowadays users can access this medium in various environments, from multiple devices (TV sets, phones, etc.) and using different technologies (high-definition, 3D, etc.).
- *Consistency of applications and content:* iDT applications should be related to the content itself, and relevant for specific users.
- *Adaptability*: iDT applications should be adaptable to different target public and environments. They should even suggest content/programs based on users' preferences and history of selection (among others).

IV. A USABILITY HEURISTICS PROPOSAL FOR INTERACTIVE TELEVISION

iDT usability heuristics were specified using the following template:

- *ID, Name and Definition*: Heuristic's identifier, name and definition.
- *Explanation*: Heuristic's detailed explanation, including references to usability principles, typical usability problems, and related usability heuristics proposed by other authors.
- *Examples*: Examples of heuristic's violation and compliance.
- *Benefits*: Expected usability benefits, when the heuristic is accomplished.
- *Problems*: Anticipated problems of heuristic misunderstanding, when performing heuristic evaluations.

The 14 proposed usability heuristics were grouped in three categories: (1) *Design and Aesthetics*, (2) *Flexibility and Navigation* and (3) *Errors and Help*. A summary of the proposed heuristics is presented below, including heuristic's ID, name, definition and explanation.

Design and Aesthetics Heuristics:

(H1) **Match between the system and the real world**: *An iDT application should speak the user's language, with* *words, phrases and concepts familiar to the user.* iDT applications should use specific conventions of the real world and should show the information in a natural order. The sequence of activities should follow user's mental processes. Metaphors should be easy to understand; there should be an intuitive mapping between controls and their functions.

(H2) **Simplicity**: *An iDT application should not overload users with irrelevant and/or unnecessary information.* Every extra unit of information competes with the relevant units of information and diminishes their relative visibility. iDT applications should show concise (but all necessary) information.

(H3) **Consistency and standards**: *Design should be coherent and consistent throughout the iDT application; it should follow the norms or conventions for TV design in general, as well as for new specific elements of iDT.* iDT applications should present similar elements in similar ways. Terminology, controls, graphics and menus should be consistent throughout the system; there should be a consistent look and feel for the system interface. As there are not yet widely recognized standards for iDT applications, highlights the importance of the consistency over standards.

(H4) **Feedback**: *An iDT application should provide feedback to the user, at least when he/she is performing key actions.* iDT applications should provide feedback on user's key actions, in a clear manner and within a reasonable time. User should be able to clearly identify their location into the application, and the available options.

(H5) **Physical constraints**: *An iDT application's elements should be visible at the visual range of watching TV, and in various types of lighting.* iDT applications design should consider issues related to the size, distances between elements displayed on screen, lighting, and others environmental factors. The concept of television is being redefined, television becomes ubiquitous; therefore specific factors should be considered.

(H6) **Extraordinary users**: *An iDT application should be inclusive, attending (all) special users' needs.* iDT applications should at least use appropriately color restricted and provide alternative mechanisms for users with hearing problems.

Flexibility and Navigation Heuristics:

(H7) **Structure of information**: *An iDT application should organize information hierarchically, from general to specific.* Related pieces of information should be clustered together; the amount of information should be minimized; option, titles and headlines should be straightforward, short and descriptive.

(H8) **Navigation**: *An iDT application should allow simple navigation; user should easily move through the application and locate information of interest.* iDT applications should provide navigational feedback (e.g. showing a user's current and initial states, where they have been, and what options they have for where to go) and navigational aids (e.g. find facilities).

(H9) **Recognition rather than recall**: *The iDT application's main elements and options should be always kept available; user should not have to remember information from one screen to another.* Help and instructions should be visible or easily accessible when needed; relationship between controls and their actions should be obvious; input formats and units of values should be indicated.

(H10) **Flexibility and efficiency of use**: *An iDT application should allow a wide range of user expertise; it should allow users to personalize the application according to their skills; it should adapt to different environments.* iDT applications should offer appropriate guide to novice users. Experienced users should get appropriate mechanism to customize applications according to their needs, skills, and personal preferences.

(H11) **User control and freedom***: An iDT application should offer users control over their actions and should allow free exploration.* iDT applications provide "undo" (or "cancel") and "redo" options; exits should be clearly marked (when users find themselves somewhere unexpectedly); facilities to return to the top level should be provided, at all stages. Facilities to return to previous points and to the main screen should be provided, from anywhere in the application. Users should be able to freely explore the application, without castigation.

Errors and Help Heuristics:

(H12) **Error prevention***:* An *iDT application should provide appropriate mechanisms to prevent errors*. iDT applications should provide appropriate messages in order to prevent users' errors. User confirmation should be required before carrying out a potentially "dangerous" action (e.g. deleting important information).

(H13) **Recovering from errors**: *An iDT application should provide clear messages, hopefully indicating causes and solutions for errors.* Error messages should adequately describe problems; they should assist in diagnosis and suggest ways of recovery in a constructive way; error messages should be written in a non-derisory tone and refrain from attributing blame to the user.

(H14) **Help and documentation**: *An iDT application should provide users a clear and simple help, in their own language.* iDT applications should offer clear, direct and simply help, expressed in user's idiom, free from jargon and buzzwords; help should be easy to search, understand and apply.

When refining the usability heuristics proposal, a usability checklist was also defined. It details the set of 14 heuristics in order to help their use in heuristic evaluation practice.

As Table 1 shows, a mapping can be made between iDT 14 heuristics and Nielsen's 10 heuristics. However, as the heuristics' specification shows, the proposal is not just a particularization of Nielsen's heuristics.

TABLE I. MAPPING BETWEEN IDT HEURISTICS AND NIELSEN'S HEURISTICS

iDT Heuristics		**Nielsen's Heuristics**	
ID	***Definition***	***ID***	***Definition***
H1	Match between system and the real world	N2	Match between system and the real world
H2	Simplicity	N8	Aesthetic and minimalist design
H3	Consistency and standards	N4	Consistency and standards
H4	Feedback	N1	Visibility of system status
H5	Physical constraints	N8	Aesthetic and minimalist design
H6	Extraordinary users		
H7	Structure of information	N7	Flexibility and efficiency of use
H8	Navigation	N3	User control and freedom
H9	Recognition rather than recall	N6	Recognition rather than recall
H10	Flexibility and efficiency of use	N7	Flexibility and efficiency of use
H11	User control and freedom	N3	User control and freedom
H12	Error prevention	N5	Error prevention
H13	Recovering from errors	N9	Help users recognize, diagnose, and recover from errors
H14	Help and documentation	N10	Help and documentation

Heuristics H1, H3, H4, and H9 particularize Nielsen's heuristics N2, N4, N1, and N6 (respectively), based on iDT applications' characteristics. Heuristics H12, H13 and H14 put Nielsen's heuristics N5, N9 and N10 (respectively) into the context of iDT applications. Heuristics H2, H5 and H6 particularize Nielsen's N8 heuristics. Heuristics H7 and H10 denote Nielsen's N7 heuristic. Heuristics H8 and H11 detail Nielsen's N3 heuristic.

V. CONCLUSION AND FUTURE WORKS

As iDT is nowadays a reality and the number and type of users is growing fast, the usability of iDT applications became a main issue. There is a need for new usability evaluation methods or at least usability evaluations should be particularized for iDT applications.

A set of 14 specific usability heuristics and an associated usability checklist were developed. As no specific iDT usability heuristics were found, the proposal is based on the well-known and widely used Nielsen's 10 heuristics as well as on heuristic proposals for other fields (such as Social TV, Virtual Worlds and Grid Computing). However, as the heuristics' specification shows, the proposal is not just a particularization of Nielsen's heuristics; the set of 14 usability heuristics was specifically designed for iDT applications.

As future work, the proposal has to be validated trough experiments. Heuristic evaluations will be performed, in order to check the iDT usability heuristics' potential in practice. Heuristic evaluation experiments will also provide an important feedback for heuristics' refinement.

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Making an Android Tablet Work as a Set-Top Box

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***Abstract*—The widespread diffusion of connected devices capable of receiving and decoding multimedia streams is inducing a change in the market of set-top boxes from dedicated proprietary appliances to software modules running on top of off-the-shelf devices. In spite of the large number of devices we use every day, smart phones are the favourite answer to our communication needs because of their availability, of their user friendliness, and of the great opportunities of personalization offered by user-generated mobile applications. The last generation of tablet PCs, capable of handling HD multimedia streams while also retaining the distinguishing features of mobile devices, enable the convergence between personal communication devices and home entertainment appliances. In this paper we discuss how to use an Android tablet PC as a set-top box, in order to allow end-users to take advantage of the tailored run-time environment of their personal mobile devices while whatching television in the comfort of their living rooms.**

***Keywords*-Set-top box; Tablet PC; openBOXware; Android; Streaming**

I. Introduction

IP traffic trends and forecasts [1], [2] indicate that multimedia contents delivered over residential and mobile IP networks are among the main driving forces of next generation networks.

The analog switch-off and the advent of *digital video broadcasting* (DVB) have enabled the technological convergence of client-side equipment required to take advantage of broadcast TV channels, IPTV services, and Internet multimedia streams. Nowadays, all new television sets come with embedded decoders, and most of them are Internet enabled. In this scenario, software components running on top of off-the-shelf connected devices are replacing proprietary *set-top boxes* (STBs), while traditional IPTV models are undergoing deep changes in order to face the pressure of *over-the-top* (OTT) multimedia contents streamed across global *content delivery networks* (CDNs).

At the same time, the widespread diffusion of smart phones and Internet enabled mobile devices, together with the growing coverage of broadband wireless networks, have induced operators to move from *triple-play* offers (i.e., Internet access, VoIP, and IPTV) to *quadruple-play* offers (which includes wireless connectivity) [3], accelerating the convergence between mobile and residential broadband markets and creating the conditions for delivering mobile TV services [4].

In spite of the wide diversity of connected devices which might work as multimedia boxes (including connected TV sets, media centers, DVB decoders, video game consoles, and personal computers), end-users spend most of their connected time using personal smart phones (or similar handheld devices) which have several competitive advantages: they are available everywhere and at any time, they offer intuitive user interfaces, they provide suitable answers to any communication need, and they provide unprecedented opportunities of personalization thanks to the thriving market of user-generated contents and applications [5].

If, on one hand, exploiting addins and configuration options to create a perfectly tailored run time environment on a smart phone is an intriguing pastime, both the actual quality of experience offered by the device and the effort devoted to personalize it keep end-users from using other devices.

Although a new generation of STBs has recently sprouted which allow end-users to create their own applications and to easily install third-party addins [6], they are far away from gaining the popularity of their mobile counterparts and the gap is hard to be closed in the near future. In fact, mobile devices are always at users' disposal and they will maintain their dominant role of personal communication equipment. Moreover, STBs are typically installed in a living room where they are mainly expected to provide a *lean-back* usage experience, which is in contrast with the *lean-forward* attitude typical of smart phone users, which has sustained the market of mobile applications [7], [8].

On the other hand, personal handheld devices have never threaten the market of media centers and STBs because of their tight design contraints, imposed by portability requirements, which made them unsuitable to sustain the workload of high definition multimedia streams. The gap between personal mobile devices and multimedia boxes is about to be closed, however, by the last generation of tablet PCs, which support HD video streams and are equipped with HDMI interfaces.

This paper investigates the possibility of making an *An-*

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droid tablet PC work as a STB, thus allowing end-users to take advantage of their personal runtime environment in the comfort of their living room, possibly switching from a lean-forward to a lean-back usage experience. The starting point is *openBOXware* (http://www.openboxware.net/trac/), an open-source framework for the development of bandwidth-aware multimedia applications originally implemented on top of *Mono*, using *GStreamer* for the multimedia subsystem and *Qt* for the graphic user interface. The concept, the main features, and the architecture of openBOXware are outlined in Section II, while Section III shows how to port the key features of openBOXware on top of Android, also discussing the switching between the lean-forward interface typical of an Android tablet PC and the lean-back interface of a STB.

II. OPENBOXWARE

OpenBOXware is on open-source framework which works as a general handler of multimedia flows streamed from heterogeneous sources to both local and remote targets. The framework automatically creates the streaming pipelines between media sources and media targets, possibly including the required transcoding stages. The capability of handling multiple simultaneous pipelines while streaming them to remote targets makes it suitable to be installed not only at the receiving end point, but also at the intermediate nodes of a content delivery network.

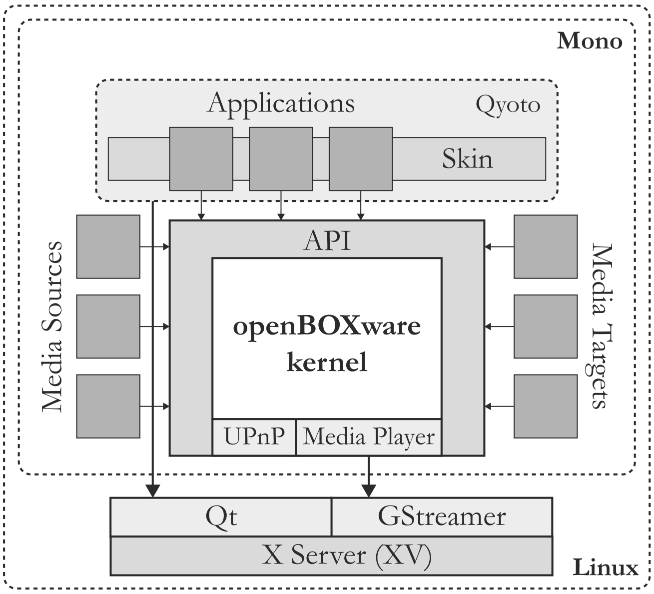

Figure 1. The software architecture of openBOXware.

In particular, openBOXware provides support for incoming and outgoing multicast streams, thus enabling the implementation of bandwidth-aware content distribution mechanisms within managed IP networks [9], [10].

OpenBOXware has a layered architecture, as shown in figure 1, which grants portability by abstracting the underlying HW platform and software components. The first level, which is the *kernel*, includes the implementation of all actual components and makes them interact with each other. It also handles bootstrapping and loading.

The second level, which is composed of the application programming interfaces (API), provides the abstraction to the functionalities exposed by the kernel. This interface level also provides means for external components, built by third-party developers and deployed as *add-ins*, to create custom applications that run on top of openBOXware and to describe media sources and media targets in an abstract fashion.

Those add-in components represent the highest architectural level, which includes all other software directly relying on the API. In particular, a special component, called *skin*, acts as application manager and determines the main user interface of the platform. All other applications, including *media sources* and *targets*, can expose new functionalities to the system. By changing the skin it is possible to change the usage experience while maintaining compatibility with the applications. Media sources and media targets, implemented as high-level add-ins, expose external resources to the system and, thus, to any other component willing to use them.

A media source represents an abstract browsable tree of media elements, each one described in such a way to enable a media target to open it and start playing it back. Given the abstract descriptions of the source media element and of the media target of choice, the framework handles multimedia loading, streaming, transcoding (if needed) and delivery to the target. For instance, a media source could describe an online video service, whose videos can be streamed to a specific media target (for instance, a remote TV set). This is done either by using a general application, called *media library*, which allows the end-user to browse media sources and easily bind them to a media target, or by using specific applications which create ad-hoc pipelines between predefined sources and targets, exhibiting a brand-specific user interface.

Other possible add-ins include client-side web applications, ranging from social network clients and feed readers to fully fledged web browsers, and server-side applications, such as *UPnP* services for home entertainment/automation or any kind of web services. Moreover, any application can expose a remote interface, possibly built by taking advantage of the embedded web server. Each application can run in one or more *execution modes*, including *fullscreen* (which takes over the whole screen area, covering up other applications), *sidebar* (which share the foreground with the top-level fullscreen application), and *background* (for services that do not need any graphic user interaction). The execution modes implemented by a given application are declared in its *manifest*, which is an XML file providing to the framework the information required to present it to the end-user and to load its components when required.

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OpenBOXware is currently implemented on top of *Mono*, an open source implementation of the Microsoft .NET framework, providing the required abstraction from the underlying hardware and software components. The multimedia subsystem relies on *GStreamer*, while the graphic user interface is based on *Qyoto*, a binding library of the *Qt* framework for .NET. Video output is enabled through the XV overlay mechanism of *X server*.

The current 1.2 software release is freely available from the official website of the openBOXware project, including its source code (http://www.openboxware.net/trac/). This first release has been developed and tested on x86 PCs running the GNU/Linux operating system, while work on a port to an ARM based embedded platform (namely the IGEPv2 board equipped with a TI OMAP processor) is currently under way, specifically targeting the MeeGo operating system [11], whose core distribution includes all required components and exists in both x86 and ARM flavors.

III. PORTING OPENBOXWARE ON ANDROID

The Android architecture is built on top of Linux kernel and consists of three main layers: the *Android runtime*, based on the *Dalvik* virtual machine (VM) with additional *support libraries*, the *application framework*, and the *applications* which run on it [12]. An Android application can be made of several components. For our purposes, the most important types of components are *activities*, which represent screens with specific user interfaces, and *services*, which run in background. Each application runs in a separate VM instance for security and protection. Communication among applications is guaranteed by an asynchronous message passing mechanism which allows a component to issue an *intent* message which is handled by another component possibly belonging to a different application. Each intent contains action and data specifications which are used by the application framework to dispatch the intent and trigger one of the components registered for performing the requested action on the specific type of data. The main graphic user interface is provided by a *launcher*, which is a special activity registered to react to a particular intent issued by the operating system at start up. The launcher allows the end-user to browse and launch activities which publish the `MAIN` intent filter. In addition, the launcher can also act as a *widget host* to allow end-users to customize the main page by embedding their preferred miniature applications.

The porting of openBOXware on top of Android, schematically represented in Figure 2, can take advantage of the features of the Android application framework in order to avoid re-implementing the bootstrapping and component handling functionalities of the kernel. The task of handling, installing, and loading component packages can be easily left to the default package manager provided by Android, ensuring that all required openBOXware components are correctly mapped to standard Android activities and services.

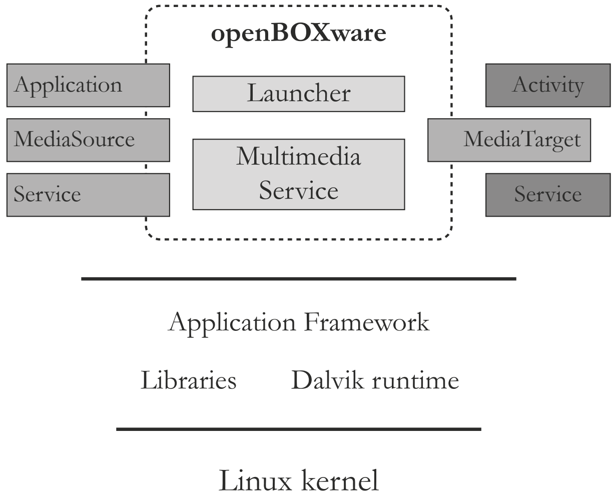

Figure 2. The openBOXware ecosystem running on top of Android.

In order to provide the usage experience typical of a STB system, openBOXware on Android sports a custom launcher conceived to offer an easy to use lean-back experience and to discriminate between normal Android applications and special openBOXware applications (identified by the intents they are registered to handle, as detailed below). It is worth noticing that multiple launchers can be installed on the same device, but only one at the time can be running. Hence, a home switching mechanism is required to allow the user to switch from a lean-forward to a lean-back use of his/her own Android device. There are three main ways to achieve this switching functionality: by changing the default launcher in the Android settings, by avoiding to specify a default launcher (in this case the choice is made by the end-user on a dialog box which appears whenever he/she presses the home button on the device), or by means of a specific *home-switching* application. Once the openBOXware launcher is selected, it becomes the main graphical user interface of the system, which shows a status bar, allows the user to launch applications, and displays the installed media sources and targets. In practice, the launcher essentially takes over the role of the skin in openBOXware, providing three different home screens: *i)* the media library, *ii)* the openBOXware application grid, and *iii)* the Android application grid. By default, the launcher presents the media library home screen with the previews of the available media sources that can be easily selected and played back on the default media target (which is the built in media player) as conventional TV channels. Advanced functionalities provided by add-ins can always be accessed from the other home screens.

Applications which provide a graphical interface are to be implemented as activities. By default, activities that desire to be listed and started by the launcher must handle the `android.intent.action.MAIN` intent, which tells them to start up and present their user interface. This mechanism is extended in openBOXware by adding one custom intent, `openboxware.app.FULLSCREEN`, which allows the launcher to recognize openBOXware applications, providing a dedicated STB-like viewing experience, and to list them into the specific home screen.

On openBOXware, applications could be run both in fullscreen and sidebar mode. On the Android port however, only the fullscreen mode has been retained. The decision to drop the sidebar mode for applications is due to some limitations of the Android window manager, which does not correctly support interactions with windows other than the one in foreground. Displaying a sidebar application is technically feasible, but it could provide an inconsistent interaction with the background fullscreen application, if any. Thus, instead of allowing applications to run in sidebar, a separated system sidebar has been implemented, which is displayed in overlay and acts as a widget host, containing any number of widget applications according to users' choices. This solution allows openBOXware to take advantage of the existing widgets in a manner which fits into the Android architecture.

Component mapping

The main components of the openBOXware API are mapped on Android as follows.

Notifications: openBOXware offers a simple API to enqueue text notifications and display them on the system status bar, allowing the user to react to the corresponding events. This API may be used by applications running in background to ask for user interaction. The notification hub is directly implemented on top of the Android notification manager and provides an additional user interface which makes it compatible with the STB-like experience.

Persistence: The original framework includes a simple associative key/value map that is persisted to disk, in isolated storage for each installed application. On Android, applications can make use of the system *SQLite* database storage to this purpose.

Networking: Web server functionalities and UPnP browsing and service consumption rely on external libraries which are readily available for Java on the Android operating system.

Multimedia playback: OpenBOXware 1.0 exposes an API that wraps the GStreamer multimedia framework. Similarly, a thin wrapper around the built-in Android media player can be provided to developers as a public API. However, playback capabilities of the system multimedia backend are heavily constrained due to the nature of the platform. Hence, many functionalities which are easy to provide using GStreamer must be emulated, reimplemented, or dropped altogether (e.g. UDP/RTP output streaming) on top of Android.

Media sources: Add-ins that export a media source to the system are implemented as Android services, which react to the `openboxware.mediasource.ACCESS_MEDIA` intents. This intent allows other applications to bind to the service and start a bidirectional RPC communication session, which allows the application to browse the media tree and the service to return structured data bundles representing media elements. These data bundles encapsulate all data required to play back the media resource and they can be consumed directly by the media player API.

Media targets: Since additional media targets cannot rely on GStreamer, their whole streaming backend must be implemented from the ground up. Thus, media targets are implemented as full-fledged media player instances that can be used instead of the default one provided by the system.

Dæmons: Applications that do not provide a graphical interface are naturally implemented as Android services.

IV. CONCLUSIONS

In summary, openBOXware can be ported on Android as an ecosystem which encloses a multimedia subsystem, a custom launcher, a set of specific intents and a development framework for applications that want to be recognized as openBOXware add-ins.

As a last remark, the default multitouch input device of Android has to be complemented by a remote control in order to provide full support to a lean-back living room experience. The bluetooth interface makes it possible to use off-the-shelf input devices to this purpose. More advanced control functionalities can be achieved by using an Android smart phone as remote.

The proposed architecture is currently under development on Android 2.2 running on two different devices: a *Samsung Galaxy Tab* with dock station, representative of state-of-the-art tablet devices, and an *IGEPv2 board*, representative of open-source embedded hardware platform. In both cases, a *Logitech diNovo Mini* is used as bluetooth remote and keyboard.

ACKNOWLEDGMENT

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In-network Hop-aware Query Induction Scheme for Implicit Coordinated Content Caching

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***Abstract*—The breadcrumb scheme was proposed to realize content distribution in next-generation networks. In the breadcrumb (BC) scheme, each node maintains a small piece, called a "breadcrumb," of query induction information. The BC scheme can, however, increase the amount of traffic as well as file acquisition delay since a query for a file is may diverted away from the nearest server with the file even if the query passes close to the server. This paper proposes a hop-aware breadcrumb scheme, in which the query induction information includes latest file acquisition node ID, which enables a query to be diverted toward a closer node, the latest file acquisition node, or the server. Simulations show that the hop-aware breadcrumb scheme yields shorter file acquisition time than the breadcrumb scheme.**

***Keywords*-breadcrumbs; hop-aware; query induction; in-network cache.**

I. Introduction

The Internet is now being used to transfer large contents (or files) such as video and music more often. This has stimulated research on content-centric networking such as Contents Delivery Network (CDN)[1] and Peer-to-Peer (P2P)[2]. Most studies consider overlay systems on the current Internet architecture. Users are, however, interested in contents themselves rather than where they are. Therefore, more recently, content-centric architectures have been targeted such as the "clean slate" approach for the next-generation Internet[4], [7], [8], [9]. In particular, in-network processing for content naming, search, routing and storage is one of major issues for content-centric architectures.

[4] proposed a simple content caching, location, and routing system that adapts an implicit, transparent, and best-effort approach towards in-network caching. The system is based on Transparent En-Route Caches (TERC) [6], [7]. Each node maintains a small piece, called a "breadcrumb," of state for queries and the direction in which a file was transferred. For convenience, we call the system proposed in [4] the "Breadcrumb (BC)" scheme. A breadcrumb helps a query to locate content, so that cache hit-ratio increases, i.e. the access load for content server is reduced.

In the BC scheme, however, even if a query for a server passes close to the server of the file, the query may be diverted away from the server. This may increase the amount of traffic as well as the content acquisition delay. Furthermore, the BC scheme is weak against dynamic changes to the topology. This paper tackles these problems by proposing the Hop-aware BC (HBC) scheme, which utilizes hop-count information.

The rest of the paper is organized as follows: Section II briefly explains the BC scheme. Section III introduces the HBC scheme as a significant improvement over the BC scheme in terms of file acquisition delay and the amount of traffic. Section IV conducts simulations to confirm the effectiveness of the HBC scheme. Finally, Section V concludes this paper.

II. Breadcrumb Scheme

The Simple Best-Effort Content Search (S-BEACONS) scheme, described below, is a traditional implementation of BC [4].

In the BC scheme like the IP scheme, a query initiated for a file by a request node is transferred to the (original) server of the file. Each file (or content) is assumed to be indexed by a global file ID. In the BC scheme, routers are assumed to have the function to cache not only files but also their corresponding BCs. The cache policy is assumed to be Least Recently Used (LRU), which discards the least recently used item first.

A BC contains the following information:

- Global file ID.
- ID of node from which the file arrived (ID of upstream node).
- ID of node to which the file was forwarded (ID of downstream node).
- Previous file transfer time: most recent time the file passed through the node.
- Previous query transfer time: most recent time the file was requested at the node.

A BC is generated in a router when a file is transferred through the router for the first time, and it is updated every time the file or the corresponding query traverses the node.

A. Query Induction

In the BC scheme, if a query for a file traverses a router with a BC for the file, the query is diverted toward the downstream node of the file and it traces the corresponding BC trail. Suppose that a query for a file arrived at time t at a router and found that the file was not cached at the router. Then, with timeout thresholds T_{f} and T_{q}, the router forwards the query downstream if-and-only-if

- The file was cached or refreshed (via successful query) at the router within $[t - T_{\mathrm{f}}, t]$; or
- The previous query passed through the router within $[t - T_{\mathrm{q}}, t]$ and sent downstream.

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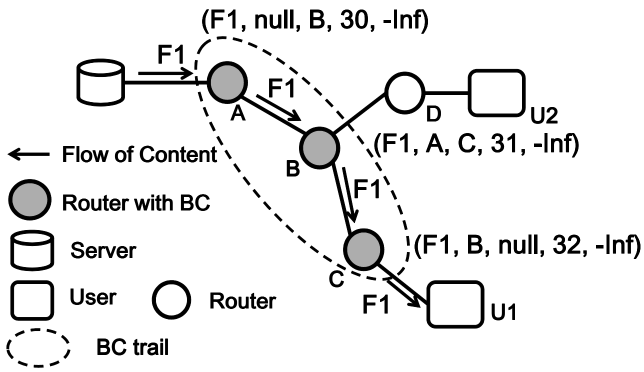

Fig. 1. BC trail and query transfer.

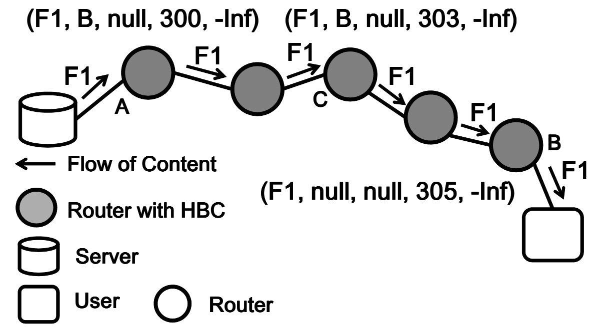

Fig. 2. Example of HBC trail.

1) Invalidation of BC: If the query can not find the file over the BC trail and reaches a dead end, the BC trail is regarded as being stale, so the invalidation procedure is invoked for the trail. More precisely, when the query encounters a node with its downstream entry null (i.e. dead end) and the file is not cached there, the query turns back along the trail and deletes the corresponding BCs along the trail.

2) Update of BC: If a file being transferred finds the corresponding BC in a router, then the BC entries, i.e. the upstream node ID, downstream node ID, and the most recent time the file passed through the node, are updated. If a query for a file finds the corresponding BC in a route, the BC entry of the previous query transfer time is updated.

B. Example of Query Induction

Fig. 1 shows an example of the query induction in the BC scheme. Given this figure, suppose that file F1 is transferred via routers A, B and C and reaches user U1, so that BCs are newly generated on the path. Note that the entry of the previous query transfer time is set to $-\mathrm{Inf}$, and the entry of the upstream node ID in the router A, to which the server is attached, and that of the downstream node in the router C, to which the user U1 is attached, are set to "null." The BC of each router is updated every time the file or the query traverses the router. Next, suppose that user U2 requests the same file and sends its query to the server. It is then transferred via routers D and B and finds an available BC for the file at router B. In this case, the query is diverted downstream toward router C instead of upstream router A. If the query finds the file on the way, the file is transferred from there instead of the server to user U2. This reduces the access load of the server. On the other hand, if the query reaches router C where the downstream BC entry is "null," BCs are invalidated in the upstream direction from router C to the server since the file is not cached on the path.

C. Problems with BC Scheme

In the BC scheme, even if a query for a file encounters a BC for the file near the server of the file, it is once diverted in the downstream direction away from the server. This may cause not only the requesting node to suffer long acquisition delay but also an increase in the amount of traffic in the network. Furthermore, since a BC trail is created as a sequence of upstream nodes (or downstream nodes), it is weak against dynamic topology changes.

III. Hop-aware Breadcrumb Scheme

To tackle the above problems, we propose the HBC scheme in which a piece, called HBC, of query induction information contains the ID of the node that acquired the file most recently instead of the downstream node ID. This modification enables a query for a file to be transferred to a closer node, the source of the file or the node that acquired the file most recently. We assume that the hop information can be obtained from the routing table.

An HBC contains the following information:

- Global file ID.
- File acquisition node ID: ID of node that acquired the file most recently.
- ID of the node with HBC for the same global file ID from which the query arrived (ID of upstream node)
- Previous file transfer time: most recent time the file passed through the node.
- Previous query transfer time: most recent time the file was requested at the node.

Note that we assume that each query includes a server ID.

It is highly possible that the node that acquired the file most recently still has the file in its cache. Since a HBC for a file directly points to the node that acquired the file most recently rather than the downstream node, the HBC scheme is robust against dynamic changes in network topology.

Like the BC scheme, when a file is transferred through a router, an HBC is newly generated or updated like the BC scheme. LRU is assumed as the cache policy of HBC scheme, too. Fig. 2 shows an example of query induction in the HBC scheme. In this figure, file F1 has already been transferred from the server to the user. In this case, routers A through B generate HBCs whose entry indicating the acquisition node is the ID of the requesting user. Router B is the access router of the user and its HBC entry of file acquisition node is "null" to denote the access router itself.

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Unlike the BC scheme, for the invalidation process explained later, a query has a field to convey an upstream node ID, which is initially set to "null." Note that the upstream node does not have to be a physically neighboring node in the HBC scheme. The query also has a field to convey an invalidated node ID, which is also set to "null," initially.

1) Query Induction by HBC: Suppose that a query for a file is traveling to the source node of the file and the query finds an available HBC for the file on the way for the first time. (The definition of availability will be explained later.) The router diverts the query toward the closer (hop-count basis) node, server, or previous acquisition node whose ID is stored in the HBC.

When the acquisition node is selected, the router enters its own node ID into the upstream node ID field of the query before sending it.

Also the router sets the entry of the upstream node ID in the HBC to "null." The router then sends the query to the acquisition node. Every time the query finds an HBC that points to the same acquisition node for the same file at another router on the way, the upstream node ID field in the query is copied into the entry of the upstream node ID in the HBC of the router, and the router writes its own node ID into the upstream node ID field of the query before sending it to the next hop. This process makes the HBCs point to the same acquisition node for the same file to be linked in a serial manner in the reverse (or upstream) direction.

If the file is found on the path toward or at the access router of the acquisition node, the file is transferred from there to the new request user. Otherwise, the invalidation procedure explained in Section III-2 is invoked. In this case, the file will eventually be transferred from the server to the requesting node.

The availability of the HBC is determined as follows. Suppose that a query for a file arrives at time t at a router. If the router has the HBC for the file, the HBC is available if-and-only-if

- The file was cached or refreshed (via successful query) at the router within $[t - T_{\mathrm{f}}, t]$; or
- The previous query passed through the router within $[t - T_{\mathrm{q}}, t]$.

Otherwise the HBC is deleted.

2) Invalidation of HBC: If the query could not find the file on the path toward or at the access router for the acquisition node, the HBCs pointing the acquisition node over the path should be invalidated (Note that the access router has the HBC whose the acquisition node field is set to null.) This can occur due to the cache policy, LRU in this paper. The access router returns the query toward the upstream node whose ID is stored in the HBC after the invalidated node ID field of the query is set to the acquisition node ID. The query is successively sent back following the upstream node IDs of the HBCs over the reverse path until it arrives at the router whose HBC entry of the upstream node ID is null. (Note that the router is the first one where the query found the HBC for the file.) After that, the query is diverted to the source, and eventually the file is transferred to the request node. On the way, if the query finds an HBC whose entry of the acquisition node for the file is the same as the invalidated node, the HBC is also invalidated.

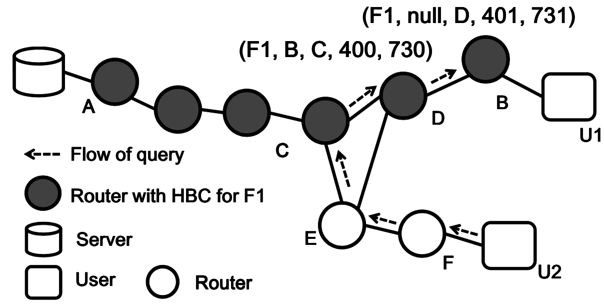

Fig. 3. Example of HBC query induction.

3) Update of HBC: When a file traverses a router, an HBC for the file is newly generated or updated in the router. At this time, the entries of the upstream node ID and the previous query transfer time are set to null and the current time, respectively. The entry of the file acquisition node is set to the ID of the closer (hop-count basis) node, current requesting node, or previous file acquisition node. As a result, the next query will be sent to the closer node. Recall that we assume that the hop-count is obtained from the routing table. Similar to the BC scheme, when a query for a file traverses a router with an existing HBC for the file, the entry of the previous query transfer time is updated to the current time.

A. Example of Query Induction

Fig. 3 shows an example of query induction in the HBC scheme. In this figure, suppose that file F1 is transferred via routers A through C and reaches user U1, so that HBCs are newly generated. Next, suppose that user U2 requests the same file and sends a query to the server. It is transferred via routers F and E and finds an available HBC for the file at router C. Router C refers to its routing table, and then diverts the query to router B which is closer (hop-count basis) than the server from router C. For the invalidation process, the query conveys the ID of router C which will be written in the entry of the upstream node of the HBC of router D. On the path from router C to router B, if the query finds the corresponding file, it will be transferred from there to user U2, so that HBCs are newly generated or updated on the file transfer path. Otherwise, the query deletes the HBCs from router B by following the upstream entries of the HBCs, and then the query arrives at router C which has an HBC whose entry of the upstream node is null. Therefore, the query is transferred to the server, and the file is downloaded from the server to the request node.

IV. Performance Evaluation

In order to compare the HBC scheme with the BC scheme and the IP scheme for the case of several thousand routers, we developed an event-driven simulator in C++ instead of utilizing ns-2 or ns-3. Router topologies used in the simulations were generated based on the BA model by BRITE [3].

TABLE I
BASIC PARAMETER.

Item	Value
Link capacity	1 packet/unit time
Number of files	10,000
Number of routers	6,000
Number of servers	300
Number of users	600
Query size	1 packet
File size	100 packets
Router cache size	100 files

We suppose that servers and users are located in the core and the edges of networks, respectively. Therefore, in the scenarios of the simulations, the servers are attached to the routers one by one in decreasing order of link degree. The users are attached to the routers one by one in increasing order of link degree. Furthermore, since routes in the core network cannot be expected to have enough high-speed memory even in the future, we assume that each router has only a cache to store BC or HBC, and only edge routers to which users are attached have enough cache capacity to store files. The query interval of each user follows an exponential distribution. A requested file is selected according to a Zipf-like distribution with $\alpha = 0.75$[5]. We also assume that query and file transfers are without packet loss.

Table I shows the parameters used in the simulations.

As shown in Table I, each link is assumed to have the capacity that one packet can be sent in one unit time, that is, the transmission time $T_{\rm s}$ is one. Queueing delay $T_{\rm w}$ of an output link of a router is estimated from the amount of traffic by using the M/M/1 queueing model[10]. Let $D_{\rm q}$ denote the average file discovery delay which is the time from the epoch that a query is issued to the epoch that the query finds the file. Using the average number $h_{\rm q}$ of hops taken by a query, which is obtained from the simulation results, the average file discovery delay $D_{\rm q}$ is roughly estimated as

$$D_{\rm q} = L_{\rm q} + h_{\rm q}(T_{\rm w} + T_{\rm s}), \tag{1}$$

where $L_{\rm q}$ denotes query length in packets, and $L_{\rm q} = 1$ in this case. Next, let $D_{\rm f}$ denote the average file transfer delay which is the time from the epoch that file is found to the epoch that the file is received by the requesting node. Using the average number $h_{\rm f}$ of hops taken by a file, which is also obtained from the simulation results, the average file transfer delay $D_{\rm f}$ is roughly estimated as

$$D_{\rm f} = L_{\rm f} + h_{\rm f}(T_{\rm w} + T_{\rm s}), \tag{2}$$

where $L_{\rm f}$ denotes file length in packets, and $L_{\rm f} = 100$ in the simulations. Finally, total file acquisition delay D is given as

$$D = D_{\rm q} + D_{\rm f}. \tag{3}$$

A. Simulation Results

1) Influence of Timeout Threshold: Figs. 4 and 5 show the relative delay ratio in the IP scheme and the download ratio

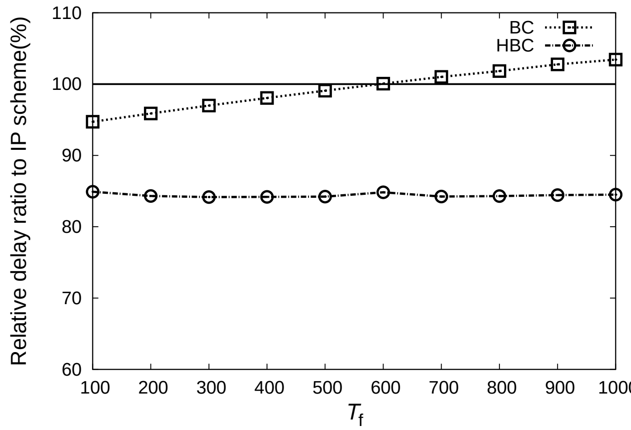

Fig. 4. Characteristics of relative delay ratio in IP scheme as a function of $T_{\rm f}$ ($T_{\rm q} = 10$, $T_{\rm i} = 70$).

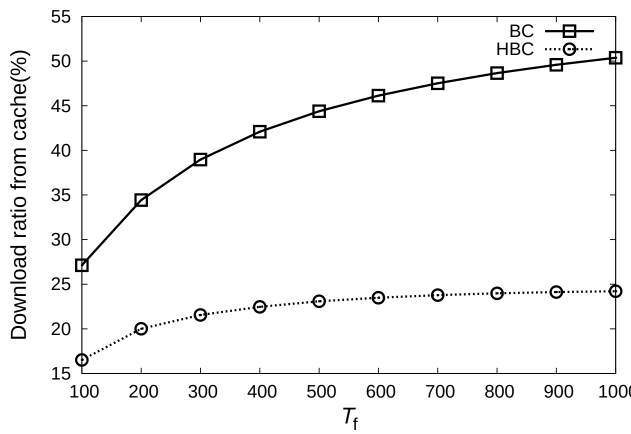

Fig. 5. Characteristics of download ratio from cache as a function of $T_{\rm f}$ ($T_{\rm q} = 10$, $T_{\rm i} = 70$).

from cache (i.e. cache hit ratio), respectively. Here, the average query interval per user is set to $T_{\rm i} = 70$. The timeout threshold $T_{\rm f}$ is varied from 100 to $1,000$ while the timeout threshold $T_{\rm q}$ is fixed to $T_{\rm q} = 10$.

From the figures, we can see that the relative delay ratio of the BC scheme increases while the download ratio from cache increases as the timeout threshold $T_{\rm f}$ increases. This is because the life time of BC increases and the possibility that files are transferred from further away increases, which means that, in one sense, cache is utilized efficiently. On the other hand, it is shown that the HBC scheme can suppress the relative delay ratio by about 10 to 18 points compared to the BC scheme. The download ratio from cache, however, does not increase as much as that of the BC scheme.

2) Influence of Query Interval: Next, the average query interval $T_{\rm i}$ per user was varied from 65 to 200. The timeout threshold $T_{\rm q}$ and $T_{\rm f}$ were fixed to 10 and $1,000$, respectively. Figs. 6 to 7 show the relative amount of traffic in the IP scheme, the relative delay ratio in the IP scheme, and the

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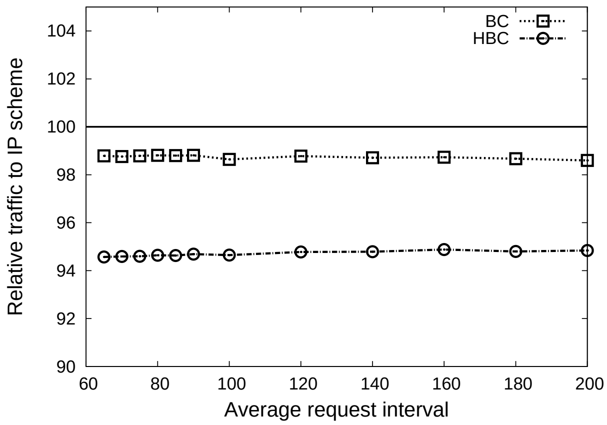

Fig. 6. Characteristics of relative traffic ratio in IP scheme as a function of request interval.

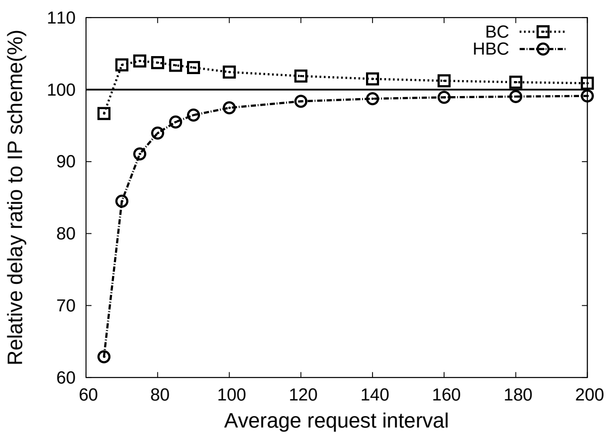

Fig. 7. Characteristics of relative delay ratio in IP scheme as a function of request interval.

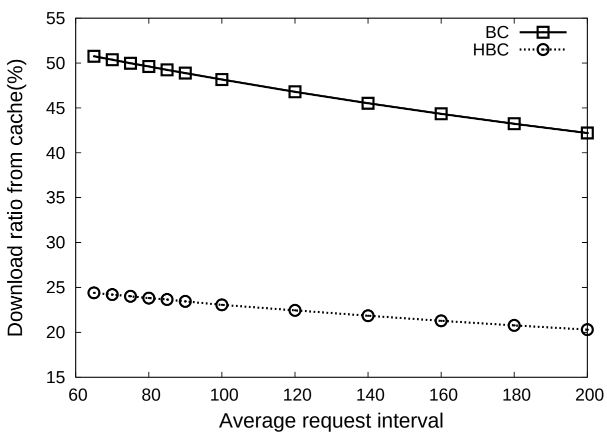

Fig. 8. Characteristics of download ratio from cache as a function of request interval.

download ratio from cache, respectively.

Fig. 6 shows that the HBC scheme reduces the amount of traffic compared to the BC scheme. This is because the HBC scheme shortens the length of the path on which the query and the file are transferred.

Fig. 7 shows that the relative delay ratio of HBC becomes smaller than that of BC as the query interval shortens, i.e. the offered load increases. For instance, the HBC scheme reduces the relative delay ratio in the IP scheme by about 34 points compared to the BC scheme. This is because even a slightly difference in traffic amount leads to a quite different queueing delay at high offered loads.

Fig. 8 shows that the download ratio from cache increases in both the BC scheme and the HBC scheme as the query interval shortens, i.e. the offered load increases. This is because more frequent requests for a file lead to more frequent updates of BCs or HBCs, so they are available for longer times. It is also shown that the BC scheme has larger download ratio from cache than the HBC scheme regardless of the offered load.

V. Conclusions

In this paper, we proposed the HBC scheme which takes account of hop-count information to reduce file acquisition delay and decrease the amount of traffic.

Simulations were conducted to compare the HBC scheme to the BC scheme and the IP scheme. Consequently, it was shown that the HBC scheme is effective since it suppresses not only the file acquisition delay but also the amount of traffic, even though the download ratio from cache increases.

Acknowledgement

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A Core/Edge Separation Based Bridging Virtualization Approach Oriented to Convergent Network

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***Abstract*—Internet has been developing for more than three decades ago. Considering its popularity, ubiquity and scale, it causes immense difficulty for other networks with different architectures to be deployed or even get adequately evaluated. This paper introduces a core/edge separation based architecture model, which allows core networks with various architectures to coexist, enables end users to choose whichever core network at will, and simplifies the design, deployment, operation and management of edge networks. In this model, Carrier Grade Ethernet, which is still on-going evolving, is selected as transport technology in the edge networks. A radically new approach called Bridging Virtualization is used to form a convergent access platform to these different core networks for end users. Besides the increasing flexibility for the edge transport carriers, this model also improves end users' independence from the access provider. Moreover, it makes it possible the execution of network convergence step-by-step and can be used to realize it in the edge first. Finally, we apply this architecture model to design the campus networks of Tsinghua University, analyze its pros and cons and make some conclusion.**

Keywords-Virtualization; network convergence; Carrier Grade Ethernet; transport; Future Internet

I. INTRODUCTION

Due to the research and development effort revolved around the concept of New Generation Networks (NGN), Carriers' networks have been improved significantly. The access and aggregation networks especially, even the core networks, are undergoing architectural evolution. Although there are different approaches to NGN, convergence, ubiquity, mobility and security are the common goals for all of them.

The concept of network convergence has twofold meanings, i.e., first and foremost it means that there is a uniform packet transport platform; second it means various kinds of applications can be built on a common network architecture such as CCN (Content Centric Network) or some other virtualization based future network. Here we distinguish them as two functions and locate them in different layers of the network system which is divided into edge part and core part. For the second meaning, it is mainly implemented in the core part; for the first meaning, it is implemented in both the core and the edge, and the architecture and technology of the implementations in both of them may be different depending on their individual characteristics and requirements. For us, network convergence not only facilitates the deployment and application of new network services, but also encourages the emergence, setup and evaluation of innovative network architectures.

Currently, there are two main alternatives to achieve network convergence: IP/MPLS and Carrier Ethernet. In this paper we focus on the Carrier Ethernet proposal for the edge networks, especially the Carrier Grade Ethernet (CGE) solution aiming at a transport technology for carriers' networks, since the former one is closely related to IP technology, whereas the latter does not care about whatever the upper layer architecture and protocol will be and fits in with future Internet initiatives much more likely.

About Carrier Grade Ethernet, there are several technological components such as Provider Bridge (PB) [12], Provider Backbone Bridge (PBB) [13], Provider Backbone Bridge – Traffic Engineering (PBB-TE) [14] and Shortest Path Bridging (SPB) [15]. To date, all of them are IEEE standards except the SPB, of which PB and PBB address scalability and management issues, PBB-TE adds traffic engineering and SPB introduces a link-state protocol into Ethernet. In addition, there are other standards pertaining to Ethernet OAM. All of the above existing standards together with the future ones will finally make Ethernet an adoptable packet transport technology for carriers.

In order to join different transport carriers' edge network together and increase the ubiquity of NGN, virtualization technique is utilized to achieve a virtual connectivity platform, where different Carrier Grade Ethernet technologies coexist and cooperate closely to provide access to various core networks for end users. Each edge transport Carrier has its own geographical footprint, and it provides network connectivity for all the potential users without the discrimination of which core network clients they are (It can charge the core network providers for their customers' usage of its transportation infrastructure). This way ensures the customer independence from the individual edge transport carriers, which not only brings flexibility for the edge transport networks but also increases the mobility for the core networks. Different edge transport carriers' networks are interconnected through virtual bridging approach to form a

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complete platform to provide access to each kind of core networks, such as the current Internet, the next generation Internet based on IPv6 and some other totally different networks as the Future Internet (FI) candidates.

By this way, we separate the whole network system into two parts, the edge part and the core part. The infrastructure in the edge just provides the convergent access platform for end users to different core networks, while at the core various networks with different architectures compete with each other to meet end users' increasing service requirements. Users can switch to different protocol stack or utilize different virtual machines in their terminals to choose among services provided by these core networks, the final judgment and decision about what architecture the future network will be depend on the most people who approve its services, then other core networks will diminish gradually and the transition to the Future Internet can be realized smoothly.

In this paper, we first present Carrier Grade Ethernet as a solid alternative to achieve convergent edge transport infrastructure in carrier's domain. Second bridging virtualization approach is proposed to provide isolate transport platform for each core network, and then Click tool is used to apply this design to Tsinghua University campus network (THUNET). Finally, we analyze, evaluate this model and complete this article with some conclusions.

II. Net Convergence and Related Work

What we mean network convergence is that people can use various network services through just one type of network. Although it is universally agreed the future network will be a convergent network, to what degree that the convergence will reach still remains uncertain. In this paper, we adopt the strictest definition of it: convergence of various services and convergence of connectivity. Meanwhile we need to point out that this definition of network convergence will not affect its realization step by step.

In the scenario of network convergence, the whole network will consist of three different parts; those are the users' networks, networks of connectivity providers and networks of service providers. Just as we say in the introduction, this paper concentrates on the convergence of connectivity providers' networks in the edge, and carrier grade Ethernet is chosen as the basis for this kind of convergence.

There are three main options for transporting Ethernet frame: Ethernet over SONET/SDH, Ethernet over MPLS and Carrier Grade Ethernet. For the first solution, its drawback lies in that the SONET/SDH equipment is costly and it has severe problems for multipoint solutions. The second solution does allow both point-to-point and multipoint connections, but again it is complex and rather expensive because it needs a routable network and IP/MPLS solutions at the core. For Carrier Grade Ethernet, it is the most suitable technology to provide Ethernet services since almost all the Internet traffic originates and ends in Ethernet. What's more, organizations for standardization including IEEE, IETF, ITU and MEF are still striving toward extending the scale and scope of Ethernet, especially aiming it at a suitable transport technology for carrier's networks. In the following we briefly describe and analyze the evolving process of Carrier Grade Ethernet.

Some requirements must be met for Ethernet if it can be considered as a transport alternative for carriers. First of all, it must provide standardized services without the need to change customer equipments. Besides this, it must support different types of quality of service in terms of bandwidth, packet loss, delay or jitter. Also, it must be scalable to be able to provide services for millions of customers accessing simultaneously to voice, video and data services. To guarantee reliability, resilience is an essential and a recovery time inferior of 50 ms (of SDH technology) is mandatory. Finally, standardized mechanisms for network monitoring, diagnosis and management are indispensable.

To fulfill the above requirements, first traffic from different customers must be segregated. IEEE802.1q is available for this purpose, but because the VLAN tag in 802.1q frame has only twelve bit length, at most 4094 VLANs can be used, which is not enough for carriers. Moreover, many customers need to assign VLAN ID for their internal networks by themselves and want to maintain this VLAN designation across the carrier's network, so there need much cooperation between carrier and customer which imposes great difficulty for carrier's operation if it is not impossible.

IEEE 802.1ad is developed to solve this problem, which uses a separate VLAN tag called S-tag for carrier's network in the 802.1ad frame. Similarly, the length of S-tag is twelve bits too, which has the same scalability problem as 802.1q. Besides, because the carrier's network and the customer's network share the same MAC address space in 802.1ad, which make them seem to be in a same bigger network, any changes in customer's network will have an impact on carrier's one, which is not desired by the carrier. From the customer's point of view, if their internal MAC addresses are exposed to the carrier's network, it may cause security concern. The most important of all, the BPDUs from carrier's network and customer's network should not interact with each other, which otherwise may cause unpredicted serious consequence. But currently in 802.1ad, there is no efficient way to distinguish them.

For all of the above reasons, IEEE802.1ah is proposed, whose frame format is as Figure 1 shows. 802.1ah uses a new carrier's MAC header to encapsulate (decapsulate) incoming (outgoing) frame. In this header, instead of a 12-bit S tag, a 24-bit field called I-SID (I-tag) is used to differentiate the service instances, which drastically resolve scalability problem. The forwarding is based on the new header's fields (B-DA, B-SA and B-VID), totally isolated from customer's addressing scheme. But 802.1ah still maintains flooding and STP mechanisms.

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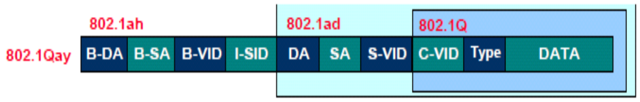

Figure 1. Carrier Grade Ethernet frame format

For carriers, traffic engineering capability of transport network is critical, but STP does not have such one and flooding wastes so much bandwidth that the carrier cannot bear. IEEE 802.1Qay aims to solve this problem. It is based on 802.1ah encapsulation, by disabling some well known mechanisms of Ethernet like STP, flooding, broadcasting or MAC learning, it eliminates the problems faced by the 802.1ah but needs additional management plane/control plane to populate its forwarding table. The forwarding decision is based on the combination of destination MAC address and VLAN ID (60 bits), providing enough capacity to traffic engineering.

IEEE 802.1aq [15] (Shortest Path Bridging) is another development that proposes an alternative to STP dependence. 802.1aq is a draft standard that uses 802.1ah data plane combined with the well-known link state protocol IS-IS [6]. This enhancement adds carrier-grade any-to-any infrastructure capabilities to the 802.1Qay point-to-point model. This is done by changing from a management system to IS-IS associated states and protocols to rule forwarding behavior. 802.1aq technology enables Ethernet to use the shortest path from any source to any destination, thereby allowing full use of the entire mesh connectivity and eliminating the need for complex Multiple Spanning Tree Protocols.

OAM (operation, administration and maintenance) is critical for Carrier-Class Ethernet. There are several standards regarding these fields: IEEE 802.1ag [7], it provides a mechanism for service fail proactive signaling; IEEE 802.3ah, it defines OAM capabilities for the first mile; IEEE 802.1AB, it allows topology discovery; ITU-T G.8031, it adds Ethernet protection mechanisms; and ITU-T Y.1713 [8], it gives additional management capacities to 802.1ag.

With all of the above developments in Carrier Grade Ethernet, carrier's network will converge to common transport platform based on native Ethernet and customers will be able to access the specified core networks via it.

III. System Architecture and Prototype Implementation

A. System Architecture

Today telephone network, Internet and cable TV network are still separate networks through which people access voice service, data service and video service accordingly, and these networks each have their own architecture and underlying technology. Even if there are some telecommunication operators or cable TV operators who claim that they can provide all these three services by themselves, users still hope that the communication media, terminal equipment and even the connectivity technique for these services could be unified too. This object seems simple and clear at a sudden, but the technical factor supporting this uniformity are extremely complex and difficult, lots of work is still in process of revision, especially for deployment in large scale.

As we mentioned before, the network convergence we think should include the service convergence and the connectivity convergence, and we separate the whole network into core and edge. In this scenario, there will be three different business roles that do not match those of today exactly. They are users, transport carriers and service providers.

Because there are already Ethernets deployed in users' premises, and Ethernet has the advantage of low cost, ease to use, maintain, upgrade, etc., we continue to use Ethernet as a convergent communication platform in user's domain.

In the domain of edge transport carrier lying between the user's network and the core network, based on the analysis in the second section, we choose Carrier Grade Ethernet (CGE) to construct a convergent packet transport platform, in which various technical components of CGE are utilized depending on actual needs to aggregate users' networks into the POPs of core networks. There users' networks are connected to various types of core networks through different gateways. These gateways may be IPv4 router, IPv6 router, CCN router, and so on.

At the interface between user's network and the edge transport network, a layer two device is used instead of a layer three router. The system architecture will look like as Figure 2 shows.

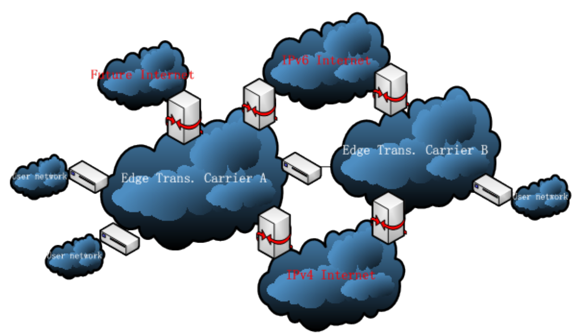

Figure 2. System architecture

Due to the large number of users' networks, heavy routing and processing burden will be imposed on the core network's gateway, e.g., they are demanding for huge port number, link bandwidth and processing power, etc. About the problem of port number, port virtualization can be used to subdivide one physical port to lots of sub interfaces or virtual interfaces, and then the technique of "router on a stick" may be used to provide routing among huge number of networks in one device. The problems of link bandwidth and processing power of a single router will be resolved or improved by the distributed router technology. Furthermore, along with the advancement of router virtualization, some device which merges the IPv4 router, IPv6 router and CCN router will finally appear and get ready for the utmost realization of core network convergence.

In the edge transport network between user's network and core network, in order for end users to connect to different types of core network, network virtualization is utilized to partition the end users' traffic for different core networks into different network instances. Because the transport network embeds the users' Ethernet frame into a new Ethernet frame when it reaches and de-encapsulates it when it leaves, which separates the transport carrier's network from the user's network hierarchically and processes the user's traffic recursively, and all these actions happen in the layer two, we call it connectivity virtual bridging or bridging virtualization. Any Layer 3 protocol from the core networks can use this connectivity transparently. In addition, due to the limited footprint of every edge transport carrier's network, similar virtual bridging technique among carriers can be used for expansion. Hence there are two layer bridging virtualization in our approach. The first layer virtualization is within each carrier's domain, where there is an individual virtual transport platform per core network. The second layer virtualization is across carriers, which further virtualizes the virtual networks in the first layer in each carrier which are for the same core network.

The second layer virtualization has two purposes. First it can be used to extend the geographical footprint of the Future Internet candidates with innovative architectures. For these networks, its initial deployment is in small scale and provided by few carriers, by this way, more carriers can provide access to it and lots of end users are able to try out it at its beginning. Second for extensively deployed IPv4 and IPv6 Internet, this type of virtualization could provide reliability and performance guarantees for end users' networks by multi-homing.

The schematic diagram of bridging virtualization is demonstrated in Figure 3.

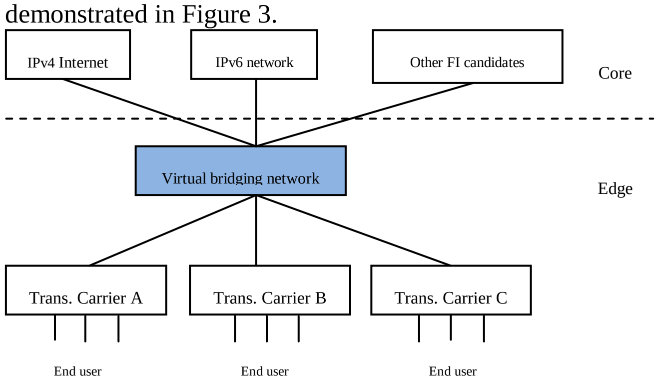

Figure 3. Schematic diagram of bridging virtualization

The advantage of this architecture model is that new network instance can be added easily and any solutions in control plane other than IEEE proposal can be used. This means that any new network architecture can be incorporated into a common transport platform by adding a new instance whose forwarding behavior can be customized for it.

B. *Prototype Implementation*

We use Click tool [9] to prove this network prototype for Tsinghua University Campus Network (THUNET). This tool was developed at MIT initially and it allows simulation networks to interact with real network nodes. It needs to be pointed out that because there is only one operator for THUNET, so we don't consider virtual bridging across multiple carriers in our initial prototype implementation. If Tsinghua University or CERNET wants to expand the footprint of IPv6 Internet or the future FI, it can permit other telecommunication operators (such as China Mobile or China Unicom) to provide access to these networks for ambient social users through this virtual bridging platform of Tsinghua University, meanwhile keep them separate from THUNET.

The Click design for the component at the interface between user network and edge transport network is shown in Figure 4.

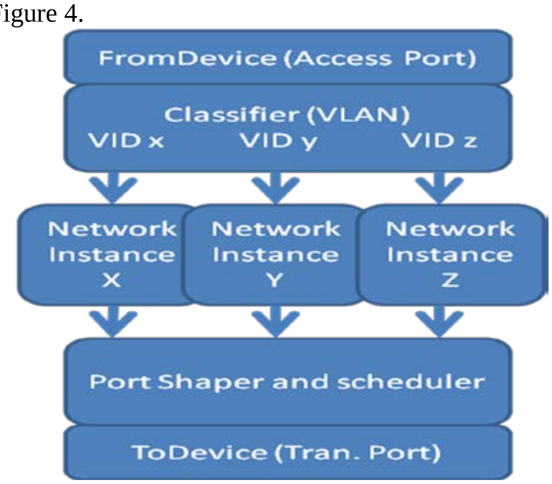

Figure 4. Click design for virtual bridging

When a frame enters the bridge node, first it must go through a classification process to determine which network instance it belongs to, then it will be processed and encapsulated with new header, after that some specific component of CCE will determine and forward this frame to proper output port. Since all the network instances share each access port of the bridge node, there must be a scheduler for the frame forwarding among these network instances.

In addition, in the Click design, the edge bridge and the core bridge are different. The edge bridge need encapsulate, process and forward frames, while the core bridge need only forward frame according to control protocol or management system. The edge bridge sets value of fields in the new header according to the type of packet, the destination gateway, the operator's policies, and so on. Once the frame finishes encapsulation process and enters some specific network instance, many protocols such as STP, IS-IS, proprietary protocols and (or) management system can be applied in this network instance to direct its forwarding action.

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IV. Discussion

This type of network architecture model has the following advantages:

First, it eliminates the use of low-end branch routers completely and decreases the campus IP network's burden of distributed network configuring, managing and troubleshooting.

Second, by using core routers to complete the function of routing, security enforcement, user authentication, protocol and address translation, etc., it will improve the IP network's reliability, stability, security, manageability and controllability, and increase the utilization rate of core router's backplane, which makes campus network construction much more cost efficient.

Third, since low-end branch routers or layer 3 switches do not support IPv6 and FI related protocols generally, elimination of them will make easy or possible for the deployment or expansion of IPv6 Internet and Future Internet, and the application based on them will spread, which allow these types of internets to get adequate usage and efficient evaluation.

Fourth, in the edge transport network, it is easy to implement traffic engineering to maximize its usage and improve the average Internet bandwidth per user.

Fifth, there is no need to make any change of the current IP routers and end hosts for this network architecture to work.

Sixth, this type of network architecture is more likely to support multiple architecturally different networks including Future Internet, facilitating their deployment and coexistence.

However, this architecture model has its drawbacks too:

First, for big organizations with multiple VLANs, because the traffic across inner VLANs need to be transported to the POP for routing, their delay will increase, though this will not affect the routine applications with no strict requirement noticeably. Besides, this model will cause security concern for those who has some security requirement on the traffic across their inner VLANs. But there are solutions for all of the above problems, for example, they can be solved by the scalable enterprise Ethernet architecture, for more details, please refer to SEATTLE [18].

Second, to aggregate users' networks and transport them to the operator's PoP for routing, the number of VLANs accumulated there will be very high, so the VLAN ID should be carefully designed and the map of routers' ports to VLAN IDs must be planned in detail to avoid confliction, which will be more demanding for the capability of network planning and management than the current IP solution.

In summary, this network architecture does not intend to replace IP protocol of present Internet; instead, it still needs protocols including IP to interconnect with other networks. But it can free end users from tyranny of local ISP, simultaneously support internetworks with different IP versions, and much likely support Future Internet and its alternatives deployed in the core.

The purport of this architecture is trying to expand the scope of edge networks, minimize the impact of IP protocol, which is mainly located or applied in the core in our model, and pave the way for the appearance and transition to post-IP Internet paradigm conceptually and practically.

Our contribution lies in the following three aspects:

First, our model changes the architecture of edge networks. It corresponds to Cloud Computing trend in network field, uses virtualization-ready core routers with rich function, high performance and high reliability to route, enforce security, translate between protocols and addresses, etc.

Second, this model design is oriented to the convergent new generation networks, it separates whole network into core and edge, and makes possible for the realization of network convergence step by step. Moreover, it realizes convergence in the edge firstly and preliminarily, which is helpful and promotive to convergence in the core.

Third, the two-layer virtualization approach makes possible for coexistence of multiple core networks, lowering the barrier to entry for network innovation, and fulfills the function of multi-homing for user's network in layer 2.

V. Conclusion

Until now, there is still confusion about which way to take for the future Internet design: incremental approach or clean-slate approach [17]. Unsettlement of this problem impedes the rapid development for Future Internet. The approach introduced in this paper can bring an end to this argument and allow two kinds of design to coexist and develop in parallel, while the final decision about what the future Internet will look like depends on the choice of most users.

In this paper, based on the concept of network convergence, we propose to partition the whole network into two parts: the core and the edge. The different networks with various architectures, such as IPv4 Internet, the next generation Internet based on IPv6 and other future Internet candidates, lie in the core. While at the edge, a two-layer bridging virtualization approach is designed to form a uniform access platform for all the core networks, enabling end users choose among multiple core networks freely. This model lowers construction cost of campus networks with better security and manageability, makes them scalable for new network architectures, and improves user networks' reliability and performance.

With its advantages and advancement of Carrier Grade Ethernet, we are optimal the deployment and execution of this architecture model will become a reality.

VI. Future Work

In our architecture model, we define business roles as transport carriers and service providers, who cooperate to provide the network connectivity and network services. Considering how to setup the proper business relation between them to spur the development of this network architecture is valuable. Moreover, when network faults or service interruption happen, how to settle responsibility among them or improve the accountability of this type of network is meaningful.

The technique for scalable Ethernet architecture is very helpful to our architecture model, which not only facilitates its execution but also can be explored for its application in the edge transport networks. So the next step for us is to adapt this technique and apply it to CGE, proposing new Carrier Grade Ethernet draft and promoting it to be standardized.

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EIE: An Evolvable Internet Environment

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***Abstract*—The end-to-end characteristic of Internet enables easy deployment and modification to the application layer protocols running on host. And the competition among these applications promotes the development of the Internet. However, new protocols related to core network layer or network forwarding equipment are hard to deploy, which hinders the evolution of the core network technology. To solve this problem, this paper proposes an Evolvable Internet Environment (EIE), which runs as a programmable platform and provides a mechanism for normal network researchers (not only the equipment vendors like Cisco and Juniper) to implement and deploy new network architectures or try their new ideas on the forwarding device and operate the hardware resources on it. EIE supports incremental deployment and provides a collaborative experimental environment for academia, industry achievements. With EIE, various Internet architectures or protocols can plug themselves into EIE network forwarding equipment and run simultaneously for experiment and actual development. Architectures upon EIE coexist and compete with each other, and some of them maybe succeed or are eliminated under natural selection. EIE uses such competition to promote the evolution of the core network.**

Keywords-Internet Architecture; Evolution; Competition; Platform.

I. INTRODUCTION

With 30 years of development, Internet itself got an unprecedented development and prosperity. However, new protocols involving core network layer or core network forwarding equipment (mainly refers to switches or routers) such as Differentiated services, IP Multicast, RBA [1], RNA [2], SILO [3], secure routing are suffering from the large-scale deployment embarrassment:

1) New protocols need to be standardized.

2) Need large-scale test.

3) Need realization into forwarding equipment.

4) Deployments are always costly, and lack of enough deployment incentives.

5) Furthermore, there is no large real user traffic, economic factors, routing strategy or deployment incentives in testbed. Experiments running successfully in testbed cannot prove that they will run successfully in real network. This further hinders the realization maturity of equipment and the deployment decision of operators.

The evolution of Internet core technology is at a standstill.

Besides, only one Internet architecture cannot meet all requirements of the future Internet; if we try to meet all requirements by improving TCP/IP, it would be a problem of seeking for an optimal solution in multiple constrained conditions. If these constraints have conflicts, there would be no optimal solution. It is necessary to solve various problems by a coexistence of varieties Internet architectures.

Furthermore, in the past years, it was impossible for commercial switches and routers to provide an open software platform due to competition. And the internal detail of network equipment is hidden for new experiments; this might lead to the network crash down. However, in recent years, with the development of the technology and thousands of demands from the vendor’s customers, the equipment vendors start to open up more and more programming interfaces for customers to develop their needs by themselves.

Based on the analysis above, this paper proposes an evolvable platform EIE to support the coexistence of various Internet architectures. It allows normal researchers to participate to the development of the core network. Internet architectures can be easily plugged into EIE platform, and the TCP/IP stack is one of them. The EIE upper architectures can form a competitive relationship to solve a same problem, similar to the relationship among Skype, MSN, Google Talk, or complement each other to solve different problems. Users can become clients of one or several network architectures.

The following sections are organized as follows: Section II describes the related work. Section III gives a full explanation of the EIE mechanism. Then, the deployment issues are discussed in Section IV. Finally, this paper presents evaluation in Section V and conclusion in Section VI.

II. RELATED WORK

During the past few years, several experiment platforms have been proposed for the network Innovation. OpenFlow [4] is a solution providing real data flows for researchers to carry out their Internet innovations. Currently, it works in the local area network, and mainly deals with IPv4 protocol. It faces lots of challenges to achieve the worldwide deployment, such as the scalability problem, management problem and so on.

PlanetLab [5] is an open, shared testbed for developing, deploying, and accessing planetary scale applications. Researchers can use a slice made up of several dedicated hosts or servers to carry out their experiments. In contrast, EIE is to achieve a multi-architecture coexisting Internet, and focuses on programming in network forwarding devices rather than hosts.

The Global Environment for Network Innovations (GENI) [6] is organized around several focus areas, facility architecture, the backbone network, and distributed services. GENI is a clean-slate solution and possesses its own experiment devices, this is the main difference between GENI and EIE, as EIE is dirty-slate and running on production network.

Active network [7] allows customized computations on packets in programmable routers to accelerate infrastructure innovation. However, its design is based on TCP/IP by adding an Active Layer upon TCP/UDP layer. Besides, it does not support the deployment of revolutionary architectures like RBA and RNA, which are equal to TCP/IP stack and need the hardware change on routers. Thus, it is hard to deploy them.

Virtual router [8] achieved the router underlying resource isolation. But the function in virtual router is a subset of the master router. Currently, it is used in several special occasions and does not support the proposed novel architectures or protocols by researchers like a new OSPF or BGP.

So far, the easy and efficient deployment issue of new architectures is still unresolved. Novel ideas related to core network layer or core network device are still need years to be deployed to the production network.

III. THE EIE ARCHITECTURE AND MECHANISM

A. EIE targets

EIE mainly has two goals: 1) Researchers can program and easily try their new architectures or protocols in production network device. 2) EIE is able to accommodate varieties of other architectures, as a platform to enable the competition among the upper architectures. Through natural selection, the fittest survive. Besides, different architectures share the same underlying hardware resources.

B. EIE model

symbol	meaning
NE	network entity
NC	network connector
CS	constrains, the relationship between NE and NC
h	host
le	logical forwarding equipment
sas	safety assurance system

EIE = Host (*NEh*)
+ Logical forwarding device (*NEle*)
+ Logical link (*NC*)
+ Safety assurance system (*NEsas*)
+ EIE mechanism (*CS*)

Definition: *EIE = (NE, NC, CS)*

EIE = (network entity, network connector, constrains)

$$NE = \sum_{k\in(h,le,sas)} NE_h + NE_{le} + NE_{sas}$$

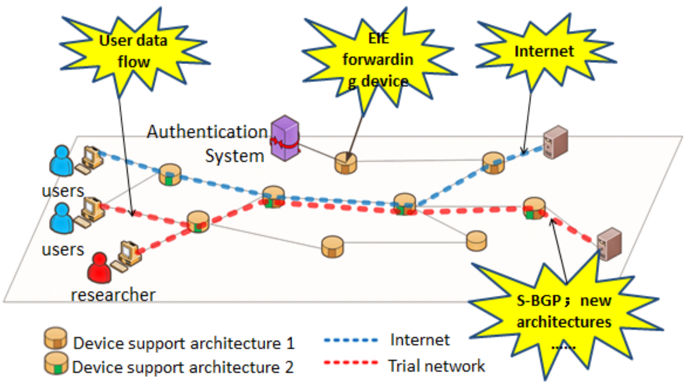

Figure 1. Future Internet with EIE

In Figure 1, the trial architecture network in blue line and current Internet in red line coexist in the future. And the trial architecture network runs in production environment. Internet users can contribute traffic to new architecture network. Each architecture network may only cover a part of the entire Internet; if one architecture has not been used for a long time by users, finally, it will completely be out of stage by natural selection.

The authentication system or safety assurance system controls which researchers can deploy resource code to the EIE forwarding equipment.

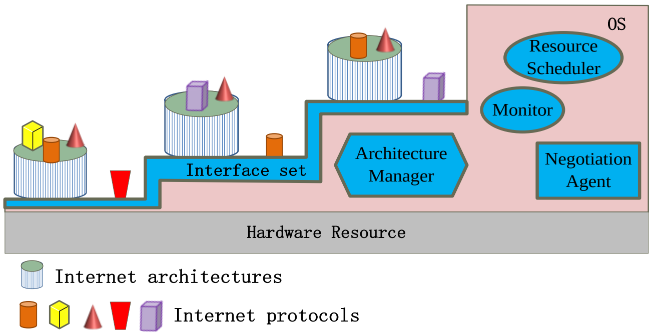

Figure 2. Components in EIE forwarding device

In Figure 2, EIE forwarding device contains five components:

① Interface Set: open up to researchers for programming. Ladder type interfaces means supporting various granularity functions. New protocols proposed by researchers can run directly on the EIE interfaces or on an architecture running in EIE.

② Negotiation Agent: as the representative to negotiate new protocol deployment requests from Internet.

③ Architecture Manager: responsible for architecture access control, management, withdraw, and maintaining records {resource size, lease length, developer ID, process ID …}

④ Monitor: monitoring the health status of architectures and feedback to Architecture Manager.

⑤ Resource Scheduler: responsible for resource allocation and isolation.

Outside EIE forwarding equipment, there are two other components in the whole EIE platform:

⑥ Safety assurance system: protecting the safety of programmable forwarding device.

⑦ Virtual build environment: for developers.

Figure 3 below describes how the seven components work in EIE mechanism.

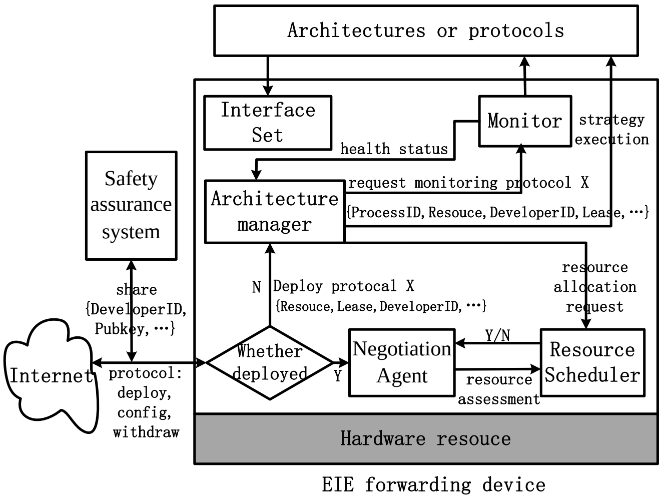

Figure 3. The whole EIE components interaction mechanism

C. *EIE compositions*

EIE includes two parts: 1) Normal hosts with EIE software to support multi-architecture; 2) EIE supported network devices (switches or routers). Besides, there should be a safety assurance system to control which researchers can deploy programs to EIE device in Figure 1.

D. *EIE host*

1) *Basic model*

As the figure below, the existing hosts need an additional EIE module in the link layer to support multiple architectures. Users just need to download the EIE software and install it.

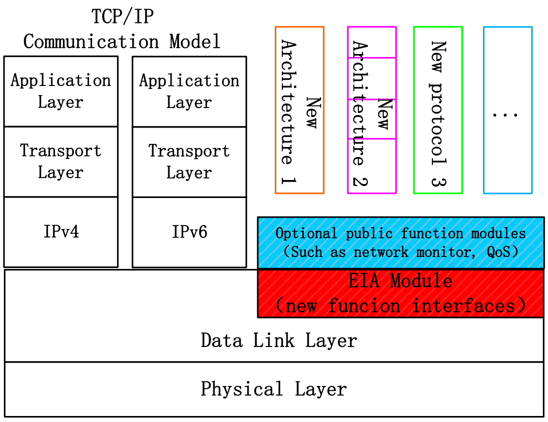

Figure 4. Basic model of EIE host to support multiple architectures

The existing TCP/IP communication protocol stacks (IPv4, IPv6) run as two special architectures and coexist with other new architectures.

After installing the EIE module, the tasks of the link layer are as follows: it receives a packet from the Internet, then passes the packet to the appropriate upper protocol stack according to the architecture or protocol ID; when receives a packet from the upper architectures, it will forward this packet to a specific port or make a call to the underlying hardware resources according to the instruction of upper architectures.

2) *Implementation in host*

EIE on host is achieved by WinPcap [9], which allows application to capture and transmit packets bypassing the TCP/IP stack. In this way, EIE can get an entire data frame, and directly put it onto the network adapter and then to the Internet.

E. *EIE network equipment*

1) *How to support multiple architectures for EIE network equipment*

EIE supporting multiple architectures to coexist requires those architectures to share the underlying hardware resources.

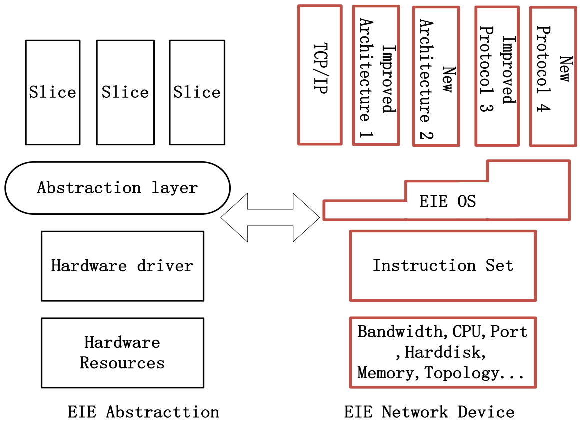

Figure 5. EIE network equipment

As shown above, the instruction set in EIE network device will be provided as primitive operation to operate the hardware resources, similar to the x 86 instructions in computer. EIE operating system layer encapsulates instructions into interfaces and provides them to the upper architectures or protocols, such as packet forwarding, discarding, and rewriting functions.

2) *Resource management*

Each architecture can enjoy a certain number of hardware resources like hard disk space, memory, bandwidth, and so on. Once the amount of usage exceeds the pre-negotiated size, EIE will carry out strategies like suspending this process for a certain time. In this way, EIE guarantees that new architectures have negligible effect to the existed data traffic.

3) *How to support EIE for commercial network equipment*

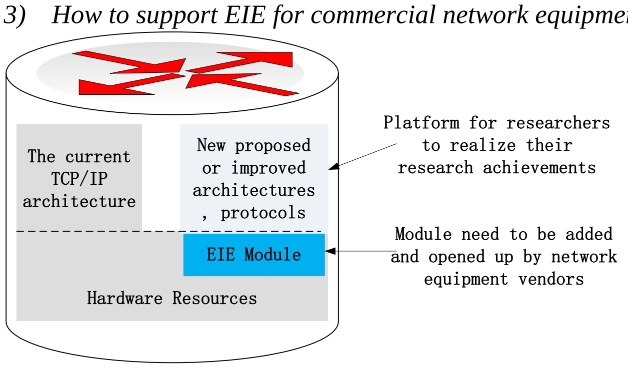

Figure 6. Commercial forwarding equipment to support EIE

The EIE forwarding equipment does not change the current TCP/IP architecture. Network equipment vendors just need to add EIE module and open up some interfaces to developers as the figure above. Once researchers gain permission to deploy a new architecture to EIE equipment, they also can send commands dynamically to change their code, configurations, parameters, and others.

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II. DEPLOYMENT

A. *The compatibility of EIE with the current Internet*

EIE achieves compatibility with the existing network, by taking the current TCP/IP protocol stack architecture as a special case among the multiple competing architectures. EIE does not add any new protocol layer to the current TCP/IP architecture, and just adds a software mechanism to the traditional network equipment. So the EIE solution can achieve faster implementation and easier transition.

B. *EIE supports incremental deployment and arbitrary virtual topologies*

In the initial stage, EIE uses the Overlay technology [10] for a packet of new architecture to transport from one EIE network device to another to achieve the incremental deployment.

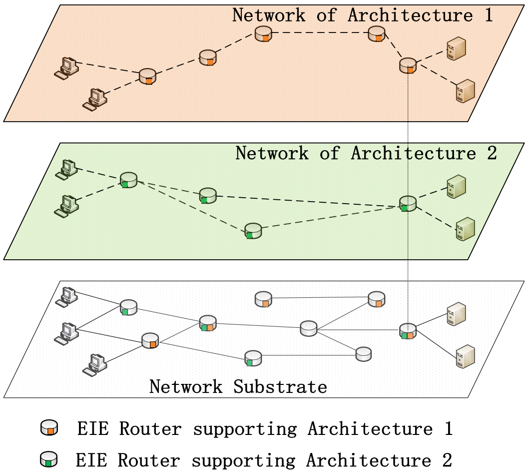

Figure 7. Current network infrastructure with EIE routers

As in the figure above, different architecture networks have different views of topology, such as the topology of network 1 and the topology of network 2.

III. EVALUATION

A. *The advantages of EIE*

1) *The benefit to researchers*

Researchers can do experiments in the real world; experiments will carry real traffic of end users at a large scale. Compare with the traditional way to deploy a new protocol to the real network in section I, in EIE the cycle time for a new protocol from the conception design to the commercial use is greatly shortened.

2) *Delplyment Incentives for Internet Service Provider*

There are mainly two ways to motivate the EIE deployment for an ISP (Internet Service Provider). : 1) Through government support. Currently academic achievements are always difficult to be deployed to the real network. EIE can increase the conversion rate of the scientific research achievements. 2) ISP can charge equipment rental fees from researchers. Researchers maintaining their protocols reduce the management cost of ISP.

B. *Performance of EIE equipment*

Each architecture can enjoy a certain number of hardware resources, once the usage amount exceeds the pre-assigned number, EIE will carry out certain strategies. New proposed architectures or protocols also can run as pluggable hardware modules. In this way, EIE guarantees that new architectures or protocols do not affect the existed traffic, and that the new architectures on EIE can get a high forwarding speed.

IV. CONCLUSION AND FUTURE WORKS

At the current stage, new architectures or protocols involving network core layer or core forwarding equipment (switches, routers and others) are hard to deploy. This impeded the evolution of the Internet core network. Although researchers have already proposed various Internet architectures, almost no architecture has been large-scale deployed and widely used on the Internet so far. The main contribution of this paper is the proposed EIE theory and its model, which runs as an open and programmable platform achieving easy deployment of innovative architectures or ideas from researchers.

Researchers participate in the core network development to realize their ideal future Internet, rather than relying on vendors to implement. Architectures upon EIE can coexist and compete to achieve the continual evolution of the core network.

With EIE, we believe that the Internet will become more and more powerful, and meet the growing demands of users. The future work of EIE will focus on defining the detail of the interface set for network equipment vendors, improving the EIE theory, and implementing the EIE prototype.

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Analysis of the Collaboration Structure in Router-level Topologies

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***Abstract*—As the Internet is one of infrastructures, the reliability of the Internet is becoming crucial to survive against failures of network equipment. Physical connectivity of the network is essential to characterize the reliability. There is collaboration structure, which is one of topological structures where two or more nodes are connected with a node, and collaboration is observed in transcriptional regulatory networks and ISP's router-level topologies. The collaboration structure relates to reliability of the network. Here, the main objective of this paper is to indicate whether the increase of collaboration improve the reliability or not. For this purpose, we first categorize the topology into three-level hierarchy; top-level layer, middle-level layer and bottom-level layer. It is apparent that the collaboration between top-level layer and middlelevel layer is much lower in the router-level topologies. We then calculate reliability of each network. The result indicates that reliability of most transcriptional regulatory networks is higher than one of router-level topologies. Finally we confirm that reliability of router-level topologies can be improved by rewiring to increase collaboration between top-level and middle-level layer.**

Keywords-power-law network; network reliability; transcriptional regulatory network; router-level topology; collaboration.

I. INTRODUCTION

As the Internet becomes the social and economic infrastructure, the reliability is becoming crucial to survive against failures of network equipment. Many approaches to improve the reliability have been investigated either at the network layer [1] or more higher layer. The reliability is also investigated in the optical communication systems through the protection/restoration techniques [2].

These approaches greatly improve the reliability of the network, however, physical connectivity of the network is more essential to characterize the reliability of the network. That is, if the physical connectivity of the network is easily disrupted by network failures, the approaches to improve the reliability at network layer are no longer effective. In fact, the physical topologies used in the previous studies inherently assume that the physical connectivity is kept after the network failures. In order to design a reliable network, it is important to make the physical topology to be reliable against the network failures. For this purpose, the topological characteristic and topological structure that makes the physical topology more reliable is necessary to investigate.

As for the topological characteristic, Faloutsos et al. [3] demonstrates that the degree distribution of ISP's router-level topologies in the Internet exhibits power-law attribute, meaning that the existing probability $P(k)$ of degree k node that has k links is proportional to $k^{-\gamma}$. The modeling methods of router-level topology are investigated in Refs. [4], [5]. Barabási et al. presents the well-known BA model that generates topologies having power-law degree distribution [4]. Albert et al. also investigated the failure tolerant characteristics of BA topologies generated by the BA model [6]. The results show that BA topology has relatively few high degree nodes. Thus, a random failure of nodes will mostly remove low-degree nodes with little effect on the physical connectivity of the network. However, only the degree distribution does not determine the performance of router-level topologies. In Ref. [5], the author enumerates several topologies that have the same degree distribution but have the different topological structure, and then evaluates the amount of traffic that the network accommodates. Because of the constraints of router's processing capacity and the product lineup of commercial routers, the amount of traffic differs dependent on the topological structure of router-level topology. The results show that router-level topologies have the structure that high-degree nodes connect with low-degree nodes while the topology by the BA model has the structure that high-degree nodes are connected each other.

Li et al. [5] demonstrates that the topological structure greatly affects the network-level performance of router-level topologies. However, Abilene network examined in [5], which is one of scientific networks, is different to the other ISP networks [7]. The main difference may come from the fact that scientific networks like the Abilene network provide fewer opportunities to enhance their network because of budgetary constraints, while ISPs make their efforts on enhancement of network and/or reduce the traffic load on network. The difference on the redundancy of the topology can clearly be seen from the graphs of the Abilene network (Fig. 6 (e) of Ref. [5]) and the Sprint network (Figs. 7 and 8 of Ref. [8]). One of our motivations in this paper is to reveal a topological structure that makes the router-level topologies more reliable.

In this paper, we investigate a collaboration structure in router-level topologies. Here, the collaboration is one of topological structures where two or more nodes are connected with a node. The collaboration contributes robustness or reliability of topologies because it introduces multiple paths between nodes.

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The collaboration contributes robustness or reliability of topologies because it introduces multiple paths between nodes. The network where the nodes are more collaborated has a chance to have larger number of paths between nodes. Note that unlike the clustering coefficient defined in [9], we consider the hierarchy of ISP topologies. That is, we first categorize the topology into three layers; top-level layer, middle-level layer, and bottom-level layer. We then investigate the collaboration between the three layers, which clarifies the topological structure of routerlevel topology. The definition of the collaboration structure is defined in Section II.

The collaboration structure is also investigated for transcriptional regulatory network [10]. The transcriptional regulatory network is one of the biological networks where transcription factors are regulated by the gene in the cell. In Ref. [10], the collaboration structure is investigated for several species. The authors show that the complex organism like human is more collaborated than the other organism such as *E. coli* or yeast. One of the main reasons to focus on the collaboration structure in router-level topologies is to answer the question that the current router-level topologies are designed well in terms of biological contexts. Our results show that there is a clear difference on the collaboration between the transcriptional regulatory network and the router-level topologies: The collaboration between top-level layer and middle-level is much lower in the router-level topologies. We therefore investigate the effect to increase the collaboration in the router-level topologies through the rewiring operation to discuss the future direction to design of router-level topologies.

This paper is organized as follows. In Section II, we define the collaboration in the networks. Section III presents the degree of collaboration and reliability against random node failures of router-level topologies, and compares with the collaboration in biological networks. Then, we investigate the effects of collaboration structure on the reliability by changing the physical topology through the rewiring process. Finally, we conclude this paper in Section V.

II. Collaboration in networks

A. Collaboration in biological networks

The collaboration structure in transcriptional regulatory network is investigated in Ref. [10]. In the cell of organisms, there is a transcriptional regulatory network consisted of transcription factors that are a kind of protein. The network transmits information to regulate genes depending on environmental insult. Collaboration in the transcriptional regulatory networks is a co-regulation relationship where two transcription factors regulate a transcription factor. According to the results of [10], more complex organism has more collaboration structure.

There are some analogies between transcriptional regulatory networks and router-level topologies. For example, the degree distributions of both networks exhibit power-low attribute. Another analogy is the hierarchical structure. In the transcriptional regulatory network, there are three level of hierarchy; top-level, middle-level, and bottom-level. Router-level topologies also have the hierarchy in the network; for example, a core network connects with several regions and/or state, a regional network, and an access network. We therefore investigate the collaboration structure in router-level topologies and show the difference, and then examine for changing the collaboration structure to discuss the future direction to design the router-level topologies.

B. Definition of hierarchy in router-level topologies

A key to identify the collaboration structure is to find a hierarchy, i.e., top-level, middle-level, and bottom-level in the router-level topologies. We define top-level, middle-level, and bottom-level nodes in the router-level topologies as follows. We first calculate H_i as the average hop-counts from a node i to the other nodes. Then, we set a directed link from node i to node j when H_i is lower than H_j. That is, when the node is located at the "center" of the network, the node tends to be the top-level node that belongs to the top-level. When the node is located at the "edge" of the network, the node tends to be the bottom-level node that belongs to the bottom.

More precisely, the top-level node is determined through the modularity analysis [11]. We divide the topologies into modules, and when the node has one or more links that connect with other module is classified into the top-level node. Note that, when there is a directional link from middle-node to top-level node, we reverse the direction of the link so as not to have links from lower level layer to the top-level layer. When there is a directional link between top-level nodes, we also change the directional link to be undirected. For the remaining nodes, nodes that have both incoming and outgoing links are classified into the middle-level nodes, and nodes that have only the incoming links are classified into the bottom-level nodes.

C. Definition of collaboration

Degree of collaboration is defined in [10]. It is the fraction of genes that are regulated by multiple transcription factors. In this paper, in order to investigate collaboration inside of topology, we adjusted the definition, that is, degree of collaboration is the fraction of nodes that are regulated by multiple nodes. Degree of collaboration does not depend on number of nodes and links. Bhardwaj et al. [10] introduces two types of degree collaboration. One is degree of collaboration in each layer D^L_{collab} and the other one is degree of collaboration between layers $D^{L_1,L_2}_{collab-betw}$.

1) Degree of collaboration in each layer: Degree of collaboration in each layer D^L_{collab} represents the average of D^i_{collab} for all nodes i in L-level, where D^i_{collab} is the number of nodes that are co-regulated by node i and the another node (A, for instance) divided by the nodes that are regulated by node i. The formal definition of D^i_{collab} and D^L_{collab} is as follows;

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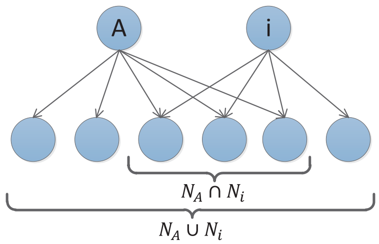

Figure 1. Collaboration between node i and node A: $|N_A \cap N_i|$ is the number of nodes regulated by node A and node i. $|N_A \cup N_i|$ is the number of nodes regulated by node A or node i.

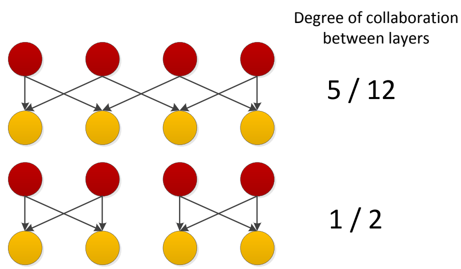

Figure 2. Illustrative example of differences on degree of collaboration between layers even when he number of nodes, links are the same. In the upper topology, the collaboration is $\frac{5}{12}$. In the bottom topology, the collaboration is $\frac{1}{2}$.

$$D^i_{collab} = \frac{\sum_{A \in N} |N_i \cap N_A|}{|N_i|}, \quad (1)$$

$$D^L_{collab} = \langle D^i_{collab} \rangle_i \quad \forall i \in L, \quad (2)$$

where N is a set of nodes in the network, and N_i is a set of nodes that are regulated by node i. Then, $|N_i \cap N_A|$ represents the number of nodes that are regulated by both node i and node A as shown in Fig. 1. $\langle\ \rangle$ represents arithmetic average.

2) Degree of collaboration between layers: Degree of collaboration between layers $D^{L_1,L_2}_{collab-betw}$ indicates fractions of nodes that are co-regulated by the node at L_1-level and the node at L_2-level, and is defined by the following equations.

$$D^{L_1,L_2}_{betw-level-collab} = \frac{\sum_{A \in L_1} \sum_{B \in L_2} \frac{|N_A \cap N_B|}{|N_A \cup N_B|}}{|L_1| \cdot |L_2|}, \quad (3)$$

where $|N_A \cup N_B|$ is the number of nodes regulated either by node A or by node B (see Fig. 1 for illustrative example). $|L|$ is the number of nodes including in L-level. Note that the collaboration defined by Eq. (3) depends on the number of nodes in L-level. To compare with several ISP topologies that have different numbers of nodes/links, we modify the definition of collaboration between layers to represents the number of collaboration:

$$D^{L_1,L_2}_{betw-collab} = \frac{|S_{L_1} \cap S_{L_2}|}{|S_{L_1} \cup S_{L_2}|}. \quad (4)$$

Figure 3 illustrates $S_{L_1} \cap S_{L_2}$ and $S_{L_1} \cup S_{L_2}$. $|S_{L_1} \cap S_{L_2}|$ is the number of nodes regulated by both a node including in L_1-level and another node including in L_2. $|S_{L_1} \cup S_{L_2}|$ is the number of nodes regulated by nodes including in L_1-level or nodes including in L_2-level.

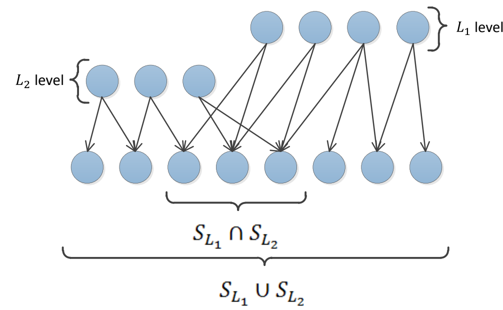

Figure 3. Modification of definition of degree of collaboration between layers. In this case, degree of collaboration between layers is $\frac{3}{8}$

III. Collaboration Structure and Reliability of Router-Level Topologies

A. Degree of collaboration

We first evaluate the collaboration structure in eight router-level topologies; AT&T, Sprint, Ebone, Exodus, Level3, Telstra, Tiscal and Verrio [8]. These topologies are obtained from traceroute-based network measurements, which may require alias resolution. The rocketfuel in Ref. [8] extended Mercator project's method [12] and relaxed the possibility of IP aliasing of routers to some extent. For comparison purpose, we compare the results of router-level topologies and five transcriptional regulatory networks, *E. Coli*, human, mouse, rat and yeast, and model-based two topologies (BA topology and ER topology [13]). For each topology, we calculate the hierarchy and then obtain the degree of collaboration in each layer and degree of collaboration between layers. Note that we do not show the degree of collaboration related to the bottom-level since nodes in bottom layer do not regulate other nodes by our definition of hierarchy.

We show the degree of collaboration in Figs. 4 and 5. From the results of router-level topologies in Fig. 4, we observe that difference between the degree of collaboration of top-level and the degree of collaboration of middle-level is less than 0.1. In contrast, the difference on transcriptional regulatory network is large in general. More distinctive characteristic of router-level topologies can be seen from Fig. 5. In the router-level topologies, the collaboration between top-level and middle-level is marginal, whereas it is not in the transcriptional regulatory network. One possible reason to have such the marginal collaboration is the functionality of middle-level nodes in the router-level topologies. That is, the traffic is first aggregated at the middle-level nodes and then forwarded to the top-level nodes. Thus, there are no consideration on load-balancing between top-level nodes and middle-level nodes.

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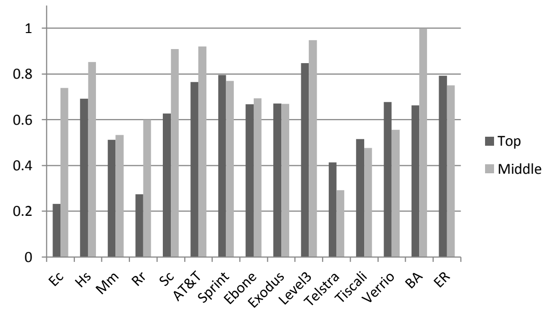

Figure 4. Degree of collaboration in each layer.

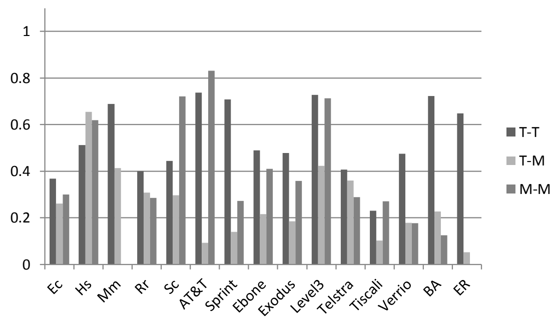

Figure 5. Degree of collaboration between layers.

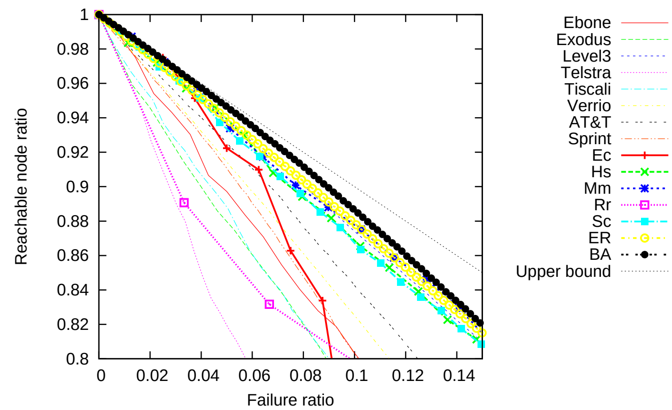

Figure 6. Reliability of networks for the random node failures

B. Reliability

We next compare the reliability of router-level topologies with that of the transcriptional regulatory networks. The purpose of the comparison is to investigate how the degree of collaboration discussed in previous section relates to the reliability of networks. For this purpose, we consider the random node failures in each network, and we evaluate the ratio of nodes that are reachable from top-level nodes to the number of nodes in the network. Hereafter, we call the ratio as the *reachable node ratio*.

Figure 6 shows the reachable node ratio dependent on the failure ratio. The failure ratio is defined as the number of failed nodes normalized by the number of nodes in the original network. In obtaining the figure, nodes to fail are selected randomly from a set of nodes in top-level or middle-level nodes since bottom-level nodes are located at the edge of the network and removing them does not give the impact on the reachable node ratio. The results of the router-level topologies are depicted as solid-lines without symbols. We also depict the upper bound of the reachable node ratio in the figure. From this figure, we observe that the results of human (Hs), mouse (Mm), and yeast (Sc) is most reliable among the organisms that we investigated, and is close to the results of model-based topologies (BA topology and ER topology). Looking again at Fig. 5, we notice that these organisms exhibit a high collaboration between top-level nodes and middle-level nodes, which increases the number of alternative paths between top-level nodes and bottom-level nodes. That is, it is expected to construct more reliable network by incorporating such the collaboration structure. In the next section, we will discuss the effect of the collaboration structure on the reliability in detail.

IV. Effects of Collaboration Structure on the Reliability

In the previous section, we show that the transcriptional regulatory networks of human (Hs), mouse (Mm), and yeast (Sc) is most reliable among the organisms that we investigated, and find that these organisms exhibit higher collaboration between top-level nodes and middle-level nodes, while the router-level topologies exhibit lower collaboration between them.

In this section, we investigate effects of collaboration structure on the reliability. More specifically, we increase the collaboration between top-level nodes and middle-level nodes through the link-rewiring, and investigate differences before and after the link-rewiring. Note that the actual ISP network may increment links or capacity of links rather than rewiring the links. However, we still consider to rewire the link because our primal concern here is whether the increase of collaboration improve the reliability or not.

A. Rewiring to increase the collaboration

We explain the link-rewiring operation to increase the collaboration between top-level and middle-level. The operation consists from four steps as described below. The illustrative example of each step is shown in Fig. 7.

Step. 1 Find a node X regulated by three or more nodes in the same level. If several nodes are found, a node is selected randomly.

Step. 2 Select a node Y randomly from the several nodes that regulate the node X and that are in the same level.

Step. 3 When the node Y is the middle-level node, find a node Z that is co-regulated only by the top-level nodes. Otherwise, i.e., when the node Y is the top-level node, find a node Z that is regulated only by

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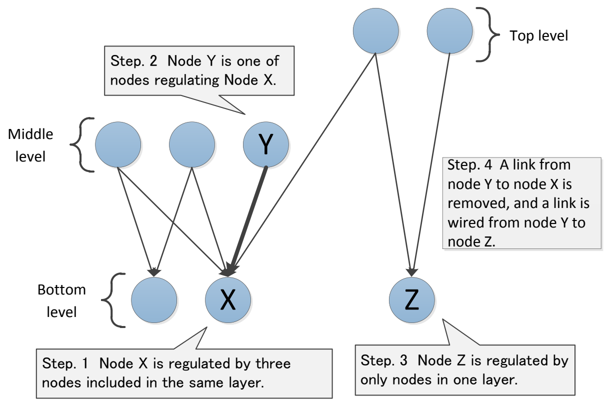

Figure 7. Illustrative example of the link-rewiring operation

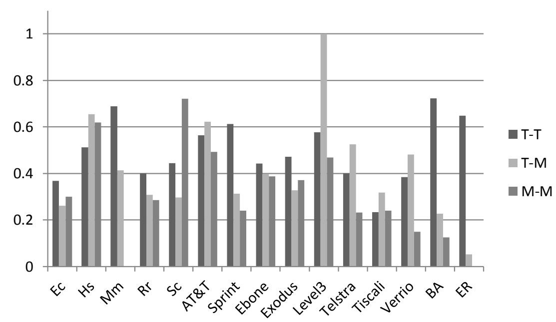

Figure 8. Degree of collaboration between layers after the rewiring operations

the middle-level node. If there are several candidate for the node Z, select the node Z randomly.

Step. 4 Rewire a link between node Y and node X; remove the link from node Y to node X, and wire a link from node Y and node Z.

Note that, in Step. 1, when the node X is regulated by only two nodes, rewiring the link leads to decrease the collaboration in the layer (middle-layer in Fig. 7) that the node Y belongs to.

This rewiring operation is continued until either of following termination conditions is satisfied.

1) There is no candidate for node X or node Z.
2) All nodes are regulated by both of top-level nodes and middle-level nodes. With this case, the collaboration is maximized, thus we do not need the link-rewiring.

The collaboration between layers after the rewiring operation is summarized in Fig. 8, and shows that the operation certainly increase the collaboration between top-level and middle-level.

B. Reliability of topologies after the link-rewiring

Lastly, we investigate the reliability of topologies after the link-rewiring that increases the collaboration between top-level and middle-level. Unlike Fig. 6 that evaluates the connectivity of directed network after the node failures, we present connectivity after the random node failures by using the un-directed links instead of directed links, and evaluate the difference between the original router-level topologies and the topologies after the link-rewiring. In particular, we use the *cover ratio*, which is defined as $\frac{S_i}{N}$ where S_i is the number of nodes in the largest strong component after i-th node failure, as the measure of the reliability.

Figure 9 shows the cover ratio of each topology after the link-rewiring. We examined 300 trials of random node failures and the average of them is plotted in the figure. We can see that the cover ratio improves for the most of router-level topologies except Sprint, Exodus, and Level3 which show a little improvement. The reasons are as follows. The original Level3 topology has a lot of links and is already collaborated well. That is, there is a little space to improvement. The situation for the Sprint and Exodus is different from Level3; the number of rewiring operations in the Sprint and Exodus is 59 and 14 respectively, while it is 221 in the AT&T topology. That is, the Sprint and Exodus have a little opportunity to increase the collaboration between the top-level and middle-level. The results of this section indicate that the collaboration structure of topologies characterizes the reliability, and the reliability improves to some extent by increasing the collaboration.

V. Concluding Remarks

In this paper, we investigated the collaboration structure in router-level topologies, and showed that the collaboration between top-level nodes and middle-level nodes in router-level topologies is less than that in the transcriptional regulated networks. Because of this, the connectivity of router-level topology is easily collapsed when node failures occur. In order to reveal the possible evolution path to improve the reliability of router-level topologies, we demonstrated that the reliability of several topologies improves when the collaboration between the top-level and middle-level increases. However, the improvement of reliability is limited in Sprint and Exodus topologies due to the limited opportunity of link-rewiring operations. That is, the evolution path to construct a reliable network is limited. Our future work is to identify the topological structure to prevent the link-rewiring opportunity in router-level topologies.

Acknowledgement

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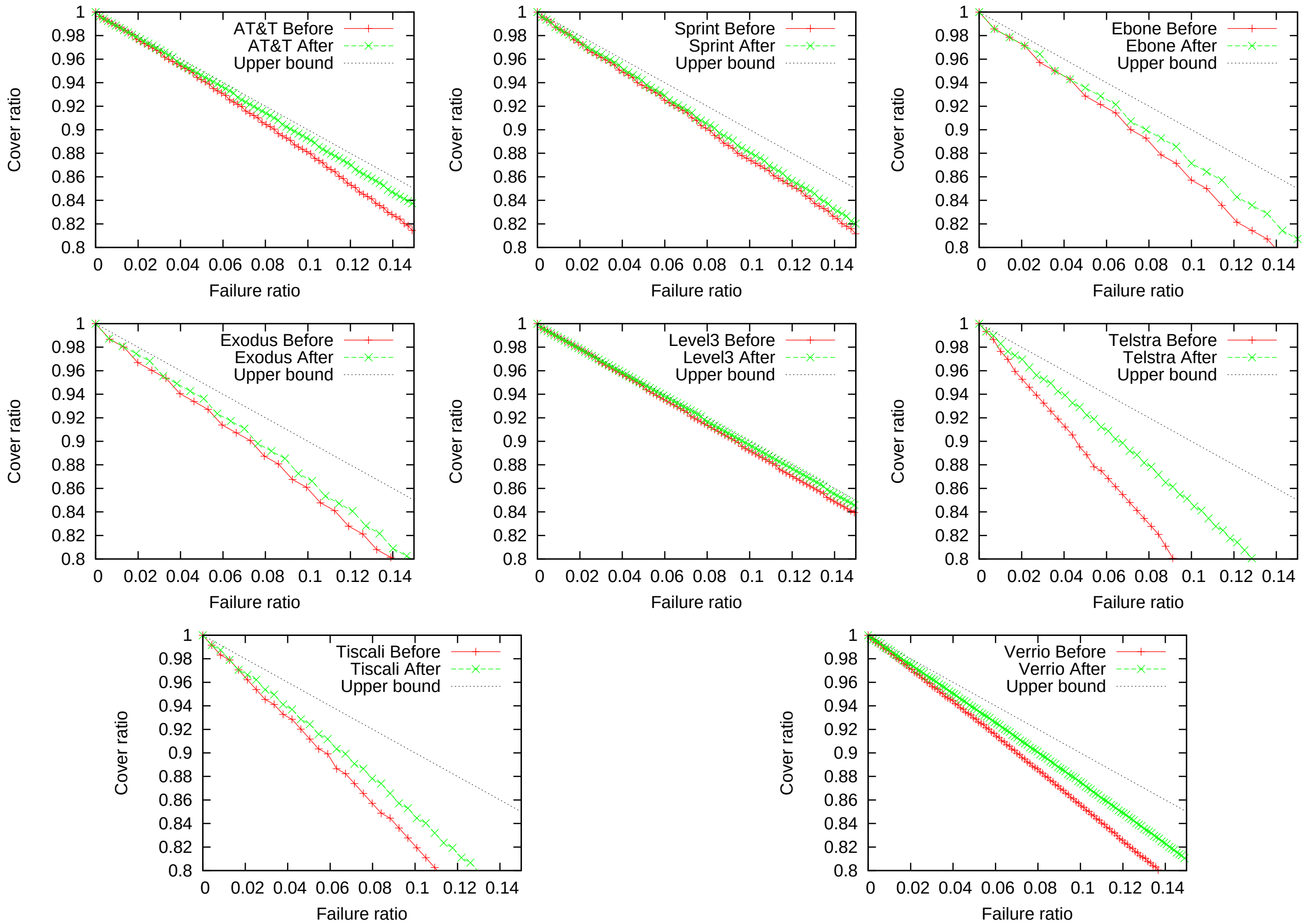

Figure 9. Difference of reliability between topologies before and after rewiring.

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Hybrid Multicast Management in a Content Aware Multidomain Network

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Abstract—**This paper proposes an architectural management solution for a multicast hybrid infrastructure optimised to transport real time multimedia flows. The multicast capabilities are offered to the upper layers as connectivity services by an overlay multi-domain Content-Aware Network. The overall system is based on new concepts like Content-Aware Network and respectively Network Aware Application. A hybrid multicast framework, Quality of Services capable is proposed, where IP intra-domain multicast is combined with inter-domain overlay multicast. The management aspects of this infrastructure are discussed. The design, validation and implementation of a system based on this architecture are currently under development in the European FP7 ICT research project, ALICANTE.**

Keywords - multicast; overlay multicast; content-aware nertworking; network aware applications; multimedia distribution; quality of services; Future Internet.

I. INTRODUCTION

The multicast transport service has received more attention in the last years, in the context of increasing of group communication needs and also of real time flows and content/media distribution to large groups of users like IPTV, Video on demand (VoD), peer-to-peer (P2P). The traditional IP multicast (IPM) despite its two decade age[1], is not however globally deployed [2][3] , due to problems related to group management, needed router capabilities, inter-domain issues and QoS problems. On the other hand, Overlay Multicast (OM) has received increased attention in the last decade, including tree based and P2P solutions. OM [4][5][6], has lower efficiency and speed, but it is quickly implementable, due to not relying on network layer multicast capabilities. A hybrid approach (IPM + OM) can be an attractive trade-off, both in terms of scalability, efficiency and flexibility. One can benefit from intra-domain IPM where it exists, and use OM outside the IP multicast area. Therefore, in this paper, a hybrid solution is considered.

For the Future Internet (FI) it is generally accepted that it will be [8][9] strongly service-content oriented and media oriented. A new concept proposes to increase the coupling between the transport/network and application layer, resulting in Content-Aware Networks (CAN) and Network Aware Application (NAA). Such CANs can be constructed as overlays, on top of traditional IP networks using partial or full network virtualization (VCAN). Note that virtualization is seen as a main way to make the Internet more flexible [10],[11]. Compared to the traditional ones, the CAN routers have additional tasks such as content/context-based classification and filtering, QoS processing, routing/forwarding, adapting and transforming the packet flows.

This paper considers the multicast service realization in a CAN/NAA context. The work is performed inside the European FP7 ICT research project, "Media Ecosystem Deployment Through Ubiquitous Content-Aware Network Environments", ALICANTE, [15]-[19]. An architectural management solution for a multicast hybrid infrastructure, QoS capable is proposed here, optimized to transport real time multimedia flows. The multicast services are offered to the upper layers by the VCANs. The solution is based on hybrid multicast (H-Mcast) framework, QoS capable, where the IP intra-domain multicast (if available) is combined with inter-domain overlay multicast. The management aspects of this infrastructure are explored.

The paper is organized as follows. Section II presents samples of related work. Section III summarizes the overall ALICANTE architecture. Section IV is focused on the multicast hybrid framework. The multicast management solution is proposed in Section V. Section VI contains some conclusions and outline of future work.

II. RELATED WORK

The new approach CAN/NAA is currently investigated in many studies, targeting better performance (for multimedia) but without losing modularity of the architecture. The CAN and NAA are studied in the framework of Future Internet architecture discussions.

The CAN/NAA approach can naturally lead to a user-centric FI and telecommunication services, as described in [8],[9],[12],[13]. The works [12][13] consider that CAN/NAA can offer a way for evolution of networks beyond IP. The capability of content-adaptive network awareness is exploited in [13]. The CAN/NAA approach can also offer

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QoE/QoS capabilities of the future networks, [12]. The architecture can be still richer if, to content awareness we add context awareness [13]. To provisioning methods for QoS assurance one can add content adaptation as another set of methods [14][18]. The CAN approach, on the other hand, requires a higher amount of packet header processing in the CAN elements, similar to deep packet inspection techniques; therefore, new methods are needed to minimize this processing task. Organizing the Internet in VCANs is a solution that fits naturally into the concepts of Parallel Internets [20]: each plane can be associated to a VCAN and to different QoS classes having different granularities, as described in [20], [22].

The approach of this paper is to enhance the VCAN capabilities with a powerful multicast service, QoS-enabled CAN, and multi-domain capability.

III. SUMMARY OF ALICANTE SYSTEM ARCHITECTURE

The ALICANTE main concepts and general architecture has been already described in [15]-[19]. Here, a summary only is inserted to allow the next presentation of the multicast framework as shown in Figure 1.

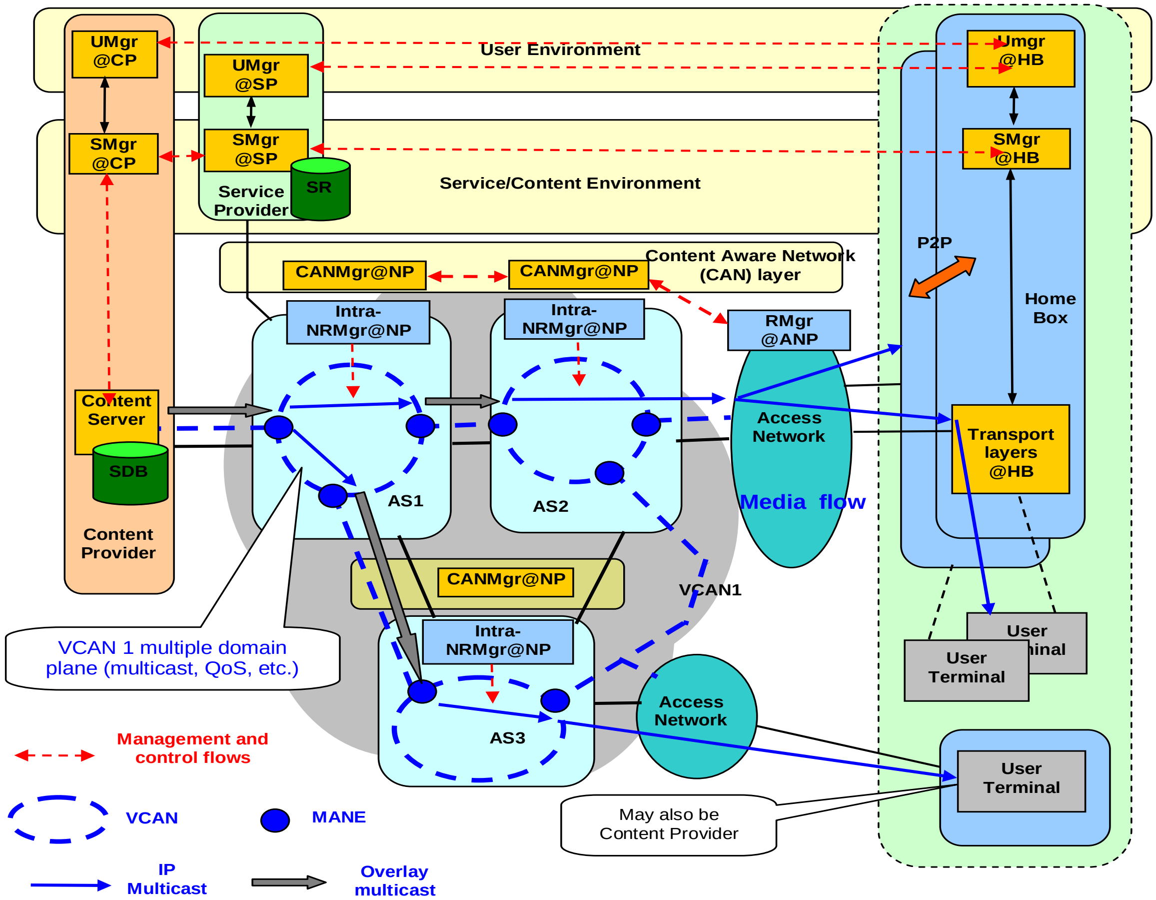

Figure 1 The ALICANTE Overall Architecture
Notations: UMgr, SMgr, CANMgr, NRMgr, RMgr – are respective managers at user, service, CAN and network levels. MANE – Media Aware Network Element; SR Service Registry; SDB- Server Data Base

ALICANTE proposed a new *Media Ecosystem*, comprising business entities, having roles of consumers and/or providers. It defines inter-working environments, [15], [16]: *User Environment (UE)*, to which End-Users (EU) belong; *Service Environment (SE)*, to which Service Providers (SP) and Content Providers (CP) belong; *Network Environment (NE)*, to which the Network Providers (NP) belong. The business model contains a new actor - *CAN Provider (CANP)* which is the virtual layer connectivity services provider seen as an enhanced NP and offering VCAN services. A new entity is also defined: *Home-Box (HB)*, partially managed by the SP, the NP, and the EU. The HB is located at end-user's premises and manages *content/context-aware and network-aware* information. The HB can also act as a CP/SP for other HBs, on behalf of the End User (EU). The HBs cooperate with SPs in order to distribute multimedia services (e.g., IPTV) in different modes (e.g., native multicast or Peer to Peer -P2P).

Two novel virtual layers [15],[16] exist: CAN layer for network level packet processing, working on top of IP and HB layer for the actual content delivery, in the user proximity. The HB layer hosts the service adaptation,

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service mobility, security, and overall management of services and content. The CAN routers are called *Media-Aware Network Elements (MANE*) having additional capabilities: content/context - awareness, controlled QoS, security, monitoring, etc. The CAN/MANE approach offers advantages over conventional routing/forwarding, but raises several challenges given more tasks to be performed by the MANE.

The CAN layer offers to the SP, *Parallel Internets* specialized for different types of applications/content, including multicast services. One or several CANs with different capabilities can be defined, installed and offered by each domain. They also can be chained in order to obtain multi-domain spanned CANs. The CAN data operations are performed by MANEs which are installed at the domains' edges (scalability reasons). The MANEs process the flows according to the content properties derived from the data flow and depending on VCAN network properties and its current status. The flows are classified in MANE based on *Content Aware Transport Information (CATI)* inserted by the content servers in the data packets, or *packet headers analysis* or information derived by *on-the-fly content-type statistical analysis.* The Management and Control Plane supports contracts/interactions [19] of SLA/SLS type (appropriate protocols are developed in ALICANTE): *SP-CANP*- through which the SP requests to CANP to provision/ modify/ terminate VCANs; *CANP-NP* - through which the NP offers or commits to offer resources to CANP; *CANP-CANP* – for multi-domain VCANS; *Network Interconnection Agreements (NIA)* between the NPs or between NPs and ANPs (these are not new ALICANTE functionalities but are necessary for NP cooperation).

In Figure 1, a VCAN1 is configured, spanning AS1, 2, 3 and offering multicast services. IP multicast is used inside ASes (if available) and overlay multicast between domains. The MANE is both IP multicast and OM capable.

IV. HYBRID MULTICAST OVERLAY

This section describes the multicast functionalities provided by the CAN, first at intra-domain level, and then expanding the scope to cover multi-domain multicast via VCANs.

A. *MANE Multicast Functionalities*

For intra-domain multicast, ALICANTE will use Protocol Independent Multicast – Sparse Mode or Source Specific Multicast (PIM-SM/SSM), [5],[7]complemented with QoS assurance mechanisms. For the first phase of system development, PIM-SM has been selected as an existing mature solution. In case that a given domain has no IPM capabilities, still the H-Mcast system can extend the OM to transit the respective domains.

The inter-domain links between MANEs will transport UDP packets containing a header and the multicast data. Thus, in the case of native multicast traffic generated by the CP server, an Overlay Module (OMd) inside the MANE, encapsulates the native multicast packet or payload into a unicast one and send it to the next domain. The latter will strip the outside header and, based on OMd header, CATI, or SVC[24] layer information and then forwards the multicast data to the HBs that are subscribed to it, using transport method selected for that session in that domain.

MANEs can also contribute to P2P intra-domain multicast [17][18]. In such scenarios, the MANE is first instructed to forward the content as unicast flows, one flow per HB. Later, if a few more HBs subscribe to the content, in order to save bandwidth, switch to P2P distribution mode is triggered by the CAN Manager. Therefore, from the MANE point of view, only unicast sending is necessary. The P2P management and control actions will be done by the CAN Manager and HBs. The MANEs are supposed to be both IPM and OM capable. MANE has a *multicast data plane module (MDPM)*. It receives an input packet and retransmits it in a number of replicas as required. The MANE receives instructions of how to multiply and forward multicast traffic (forwarding table), from CAN Manager via Intra-NRMgr.

The details of the MANE multicast functionality will be presented in a future paper. A short description is inserted here. The MDPM has two input drivers, one for multicast and another for unicast input, a high-level Multicast Bridge/Switch (MB) sub-module, and two output drivers, one for multicast and another for unicast. The MB contains a *Forwarding Table* storing the tree information of that MANE. As an example, for the case of an SVC application flow there will be associated a multicast tree for each SVC layer [18]. This is also done by the LOLCAST protocol [23], but it only supports OM, and not H-Mcast as is proposed here. The input driver receives packets from a network interface and analyzes them to extract the parameters {VCAN_ID, Layer#}. The input module delivers to the MB the original full packet, plus VCAN_ID, Layer# information. The MB generates the replicas to the output ports, in native multicast mode or unicast mode depending on the next device/network which follows that MANE.

B. *Multi-domain VCAN Multicast Trees*

An H-Mcast tree is composed of several MANE nodes (H-MCast nodes) distributed in several NP domains. Each one may assume one or more of the following roles: *root node*, *intermediate node*, and *leaf node*. The latter are usually located at the ingress of ANs. Intermediate MANE nodes are capable of making an interface between the IPM and OM parts of the tree.

A domain peering problem appears in multi-domain VCAN cases: how to determine the domains to compose the VCAN. The hub model has been proposed in ALICANTE, [19]. The SP contacts an initiating CAN Manager, which in turn determines and negotiates with other managers in order to establish the multi-domain CANs. The initiating *CAN Manager is supposed to have inter-domain topology information* (it should know all AS domains participating to this VCAN). The advantage of this approach with respect to others (e.g., cascade model) is that one has complete control of the VCAN, but knowledge on inter-domain topology is needed. However, the number of ASes involved in a CAN communication is low (< 10), and they can be localized in

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an Internet region, therefore the scalability problem is not so stringent.

Figure 2 shows an example of a multicast VCAN spanning three autonomous systems AS1, 2, 3, 4. The AS1 and AS4 network routers are PIM-SM capable, while AS2 and AS3 are not (except MANE). A multicast capable VCAN is constructed having a tree topology, with root in AS1 and leaves in other domains. The left side MANE routers in AS1 and AS4 plays also the role of *Rendezvous* Points *(RP)*[7] for the PIM-SM protocol. The thin lines of the tree in the picture represent the unicast links composing the overlay part, while the thick ones the IPM part. The leaf MANEs play also the role of Designated Routers for PIM-SM protocol. The HBs can subscribe to the multicast tree by using the *Internet Group Management Protocol IGMP*. The construction of such tree will be described in the following sections.

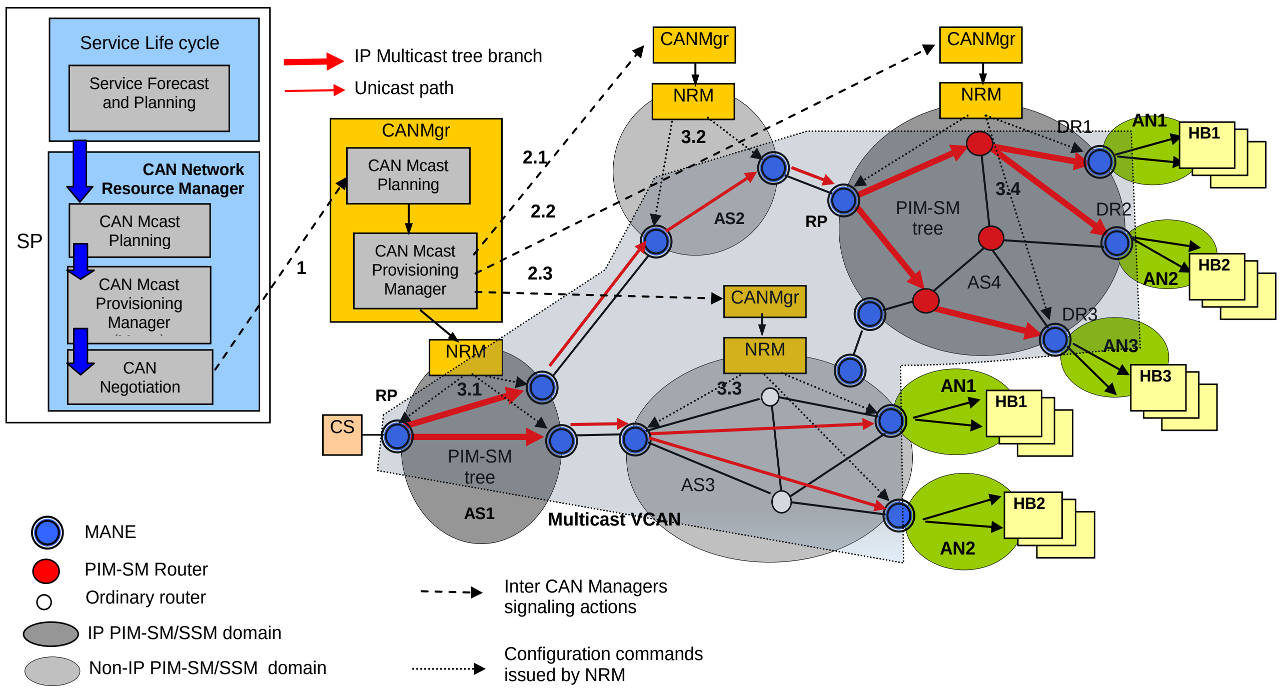

Figure 2 Hybrid multicast tree and management actions
AS – Autonomous System; CANMgr- CAN Manager; NRM – Intra-domain Network Resource Manager; DR – Designated Router; RP – Rendez-Vous Point (PIM-SM); HB- Home Box

V. HYBRID MULTICAST MANAGEMENT

In this section, the hybrid multicast management framework is described. We focus first on defining the management entities involved, then the process of multicast VCAN construction, and finally QoS aspects of multicast trees.

A. *Management Entities*

This section outlines the management framework for H-Mcast in ALICANTE. The decision to construct multicast enabled VCANs belongs to the Service Manager (SM@SP) of the Service Provider (Figure 2). More complete description of the SM@SP is given in [19]. Here we only emphasize those ones involved in multicast management.

The *CAN Network Resources Manager* (*CAN_RMgr*) is a SM@SP functional block performing all actions to assure the VCAN (unicast, multicast) support to the SP. It negotiates actions like *VCAN planning, provisioning and operation*. An SLA between SP and CANP establishes the provisioning and operation clauses for the future VCAN.

The CAN_RMgr@SM of the SP interacts with other modules of the SM@SP: *Service Forecast and Planning* - an *offline process* performing service predictions and their associated plans of deployment, considering the business needs as input; *Service Deployment Policy* (not shown in Figure 2) - containing predefined rules for service planning and deployment. This information is derived from the high-level SP business interests. The detailed functionality of these are out of scope for this paper.

The CAN_RMgr@SM contains: *CAN Mcast Planning, CAN Mcast Provisioning and CAN Negotiation* modules as a main tuple to provision VCANS. Not shown in Figure 2, are: *CAN Operation and Maintenance* intervening during VCAN exploitation; *VCAN Repository* data base to keep all data related to VCAN provisioning, installation and current status; *CAN Deployment and Operation Policies* to guide the other blocks of the CAN_RMgr@SM. The interface implementation for communications between external

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modules will be based on SOAP/Web Services interfaces, used for SOAP requests and responses.

At CAN layer, the *CNMgr@CANP* has a Multicast Manager (*McastMgr@CANP*) to perform actions to execute the SP multicast service requests, i.e., related to multicast VCAN provisioning and operation. In Figure 2, two modules called *CAN Multicast Planning and CAN Multicast Provisioning Manager* are suppose to perform the actual planning of the multicast VCAN in terms of all tree elements and domains to be spanned.

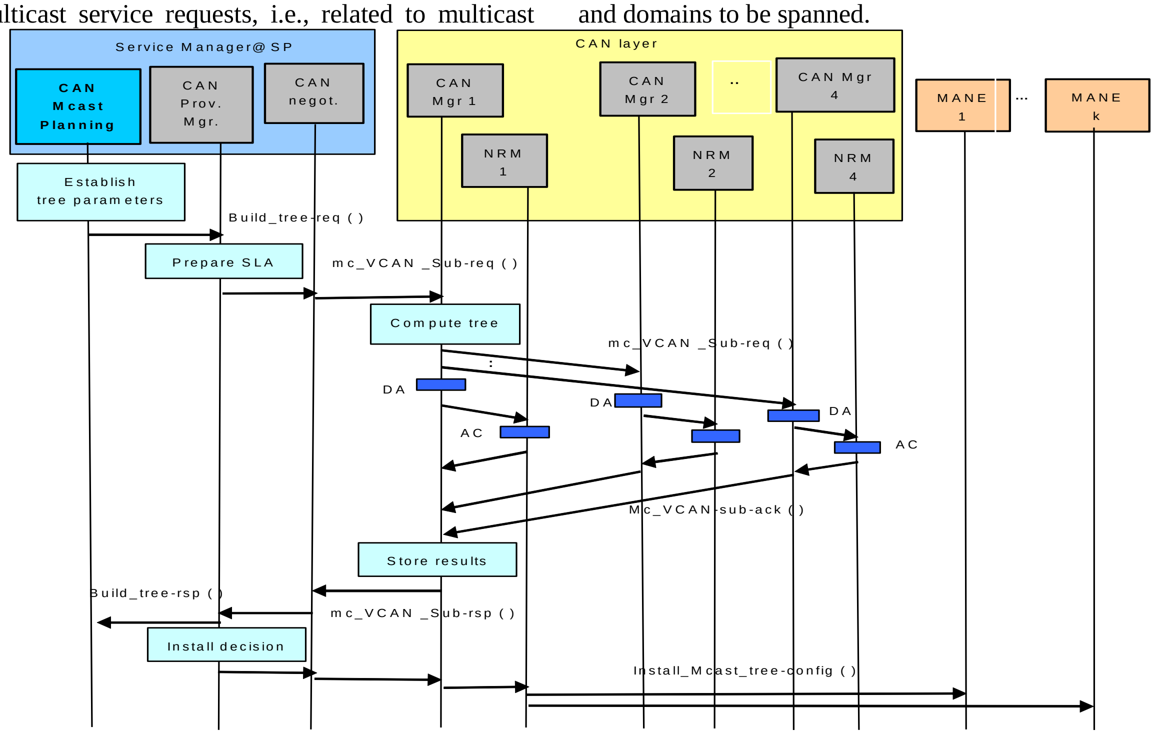

Figure 3. Message Sequence Chart example of communication between management entities to build a H-Mcast tree

B. *Multicast VCAN Construction*

To following SALs negotiation protocols/interfaces are currently specified and implemented in ALICANTE to support the multicast VCAN framework:

- *CAN Mcast Provisioning@SP – CAN Mcast Planning@CANP* (Figure 2, action 1)
- *CAN Mcast Provisioning@CANPk - CAN Mcast Provisioning@CANPm* (Figure 2, action 2.1,2.2, 2.3)
- *CAN Mcast Provisioning@CANPk- Intra-domain NRMk* (Figure 2, action 3.1, ..3.4))

Figure 2 shows the sequence of actions to construct the multicast VCAN. Figure 3 shows the Message Sequence Chart presenting the required signaling between management entities. The sequence of action is:

1.CAN Multicast Planning at SP, establishes (after cooperation with Service Forecast and Planning) the tree characteristics, from the service point of view, i.e., the root and leaves IDs (where servers and current or future users are located) QoS classes of services characteristics, bandwidth necessary, static/ dynamic characteristics, etc. We recall that the SP does not have to know the inter-domain topology but only the root and the network nodes that are leaves of the tree. How the VCAN Planning at SP is performed is out of scope of this paper.

2. After this planning the *CAN Mcast Provisioning Manager* at SP, via its *CAN Negotiation* module, asks to CANP and negotiate with it the VCAN construction (Figure 2, action 1). Let it be *CANP1* at AS1.

3. The *CAN Mcast Planning@CANP1* computes details of the multicast tree. Initially, the other NP (let them be *n, m, p,..*) domains involved are determined. Then a mesh of possible ingress/egress MANE nodes is determined. Using an appropriate metric, a tree (containing not only number of "hops" but bandwidth constraints, etc.) is computed by using a modified constrained routing Dijkstra shortest path algorithm (SPF) algorithm. The details of this algorithm will be presented in a future work.

4. The *CAN Mcast Provisioning@CANP1 will contact the CNAP2, 3, 4* to negotiate with them the multicast VCAN (Figure 2, action 2.1, 2.2, 2.3). Details of such negotiations and cases of success failures wil be a future work.

5. Each CAN Manager will negotiate with its Intra-domain NRM the possibilities to realize the VCAN tree part in the corresponding domain. (Figure 2, action 3.1, ..3.4).

6. Supposing success scenario, acknowledgments are returned to the *CAN Mcast Provisioning@CANP1*, and from this to SP concerning this multicast VCAN subscription.

7. Immediately or later the SP can ask the tree installation in the network. To this set of actions the Provisioning managers will contribute at SP and CANP and also the involved IntraNRMs and associated MANEs.

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C. QoS assurance and Resource Management

The approach adopted is that one CAN is associated to a given *QoS class*. The QoS classes in ALICANTE have reused the framework defined in [20],[21], but adapted to VCAN context. One may have several levels of granularity when defining CANs, while the main common idea is preserved: that CAN layer offers to the SP, *Parallel Internets* specialized for different types of applications content. In ALICANTE the VCANs are constructed after successfully accomplishing Admission Control (AC) in each NP domain. This is the basis of capability to guarantee the QoS. In multicast case also AC is applied. This check can be applied statically, at VCAN subscription time, or dynamically at VCAN invocation time.

VI. CONCLUSIONS

The paper proposed an architectural solution for management of hybrid multicast (H-MCast) capable to construct virtual Content Aware Networks, QoS capable in a multi-domain network context. The overall system architecture is introduced and then the multicast-related management entities roles are defined. The necessary interfaces/protocols between the management entities are defined. A multi-domain peering solution is proposed and multicast scenarios are presented in order to emphasize the signalling phases required. Further work is going on to develop the above mentioned protocols and also the resource management framework in order to add QoS capabilities of the VCANs.

ACKNOWLEDGMENT

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Designing Improved Traffic Control in Network-based Seamless Mobility Management for Wireless LAN

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Abstract—**Seamless mobility management which prevents packet loss when mobile terminals (MTs) move is an indispensable feature for future mobile networks. Proxy Fast Mobile IPv6 (PFMIPv6) has been standardized to reduce packet loss of user data in network-based mobility management. By predicting the movement of MTs, it can minimize packet loss by forwarding user data from the previous mobile access gateway (MAG) to the new MAG where MT makes a handoff, and by buffering the forwarded data at the new MAG until the MT is attached to it. When shared wireless access technology (e.g., wireless LAN) is employed as a wireless access network, the released packets from the buffer in the new MAG degrade the communication quality of the other (resident) MTs already attached to the wireless access network. In this paper, we propose a MAG design treating the buffered packets to prevent degradation of the communication quality of all MTs (resident MTs and MTs making handoffs). Using a packet-based simulation, we investigate the communication quality and show the proposed method's validity.**

***Keywords*− Network-based seamless mobility management; Wireless LAN; QoS; VoIP**

I. INTRODUCTION

Wireless access technologies have greatly advanced in the past few years. Among these advances IEEE 802.11-based wireless LAN plays an important role in offering convenient network connectivity and high-speed access at affordable costs. With these wireless access technologies, wireless networks are evolving toward all-IP systems. IP (network-layer) mobility needs to support transparency with applications and independence within mobile networks.

Mobile IPv6 (MIPv6) [1], a mobility protocol within the IP layer, provides mobility management for mobile terminals (MTs), but only MTs with the host implementation of MIPv6 can acquire the function of the mobility management. Proxy Mobile IPv6 (PMIPv6) [2], network-based mobility management without MTs participating in related signaling, has now been standardized. When PMIPv6 is adopted in mobile networks, all MTs with IPv6 functions can acquire mobility management service. MTs adopting PMIPv6 cannot communicate while the MT makes a handoff, that is, the MT changes mobile access gateways (MAGs), and in many cases some packet loss occurs in this period.

To minimize this packet loss, Proxy Fast Mobile IPv6 (PFMIPv6) [3] has been standardized as network-based seamless mobility management. When the MT makes a handoff, PFMIPv6 can prevent packet loss by buffering the MT's user data forwarded from the previous MAG (PMAG) to the new MAG (NMAG), to which the MT is attached before and after the MT's handoff. PFMIPv6 will be utilized in future mobile networks to provide seamless mobility management for numerous MTs

When multiple MTs in PFMIPv6 simultaneously make handoffs into the same NMAG, a multiple set of buffered user data is concurrently released in a bursty manner, as described in Section II-A. This leads to degraded communication quality for the other MTs ("resident MTs" hereinafter) already attached to the NMAG. Even if IEEE 802.11e [4] is applied to guarantee QoS, the buffered packets in the MAG lead to degraded communication quality of the resident MTs when they are categorized as high-priority traffic that the resident MTs are also using.

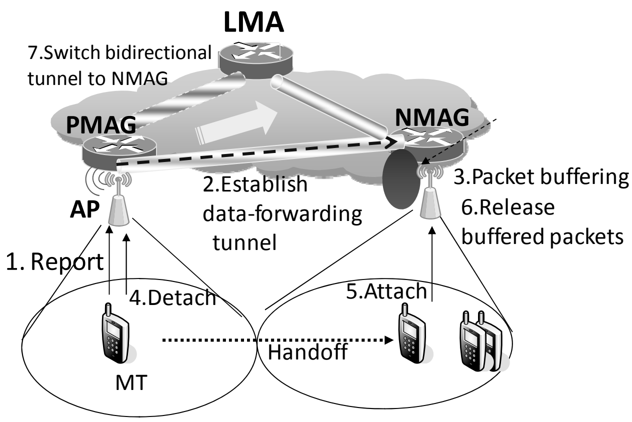
For seamless mobility management, the one-way delay of the traffic into the MTs making handoffs ("handoff MTs" hereafter) should also be a concern. Real-time applications have acceptable values for a one-way delay. Some have the standardization [5] defining requirements of one-way delays as a QoS requirement. Packets with a large one-way delay exceeding the acceptable value are treated as actual losses by these applications, even if they are ultimately delivered to the applications. If the traffic of the resident MTs is prioritized and the delay of the traffic into the handoff MTs is prolonged, seamless mobility management sometimes becomes meaningless. A traffic control method that does not degrade the communication quality of resident MTs and handoff MTs should also be considered for cooperating with seamless mobility management.

Considering these issues, we propose a MAG design that treats the traffic of the resident MTs and the handoff MTs separately along with a buffered-packet releasing traffic control method to prevent degraded communication quality for all the MTs. This paper evaluates the effects of the proposed traffic control method. Section 2 addresses issues of wireless LAN in seamless mobility management. Section 3 introduces the design of MAG and the proposed traffic control method. Section 4 discusses an experiment with packet-based simulation. Section 5 concludes the paper.

II. Wireless LAN Issue in Network-based Seamless Mobility Management

This section explains the handoff procedure used in PFMIPv6 and shows that PFMIPv6 potentially causes bursty traffic in the wireless LAN.

A. Handoff Procedures of Network-based Seamless Mobility Management Protocol



Figureure. 1. PFMIPv6 handoff procedure

Figure 1 shows the PFMIPv6 handoff procedure. It begins after an MT connects to a PMAG that has a bi-directional tunnel with a localized mobility anchor (LMA) for the MT's traffic. In this procedure, the MT makes a handoff to the NMAG. The following procedures take place:

1. Before the MT makes a handoff, the PMAG is notified which MAG the MT will make a handoff into.
2. The NMAG begins preparation for the MT's handoff, and the PMAG and NAMG establish a data-forwarding tunnel to transfer the traffic.
3. The PMAG begins to transfer the downlink user data to the NMAG through the data-forwarding tunnel, and the NMAG begins to buffer the user data arriving through the tunnel.
4. The MT is notified that the preparation for the seamless mobility procedures is finished, and starts to be detached from the PMAG.
5. The MT is attached to the NMAG. Layer 2 authentication optionally takes some time for the MT to connect the new wireless access network.
6. The NMAG begins to release the buffered user data.
7. The LMA switches the bidirectional tunnel from the PMAG to the NMAG.

A method to adjust the timing in releasing the buffered packets has been proposed [6]. For mobile networks that accommodate many MTs, it is not feasible for the MAGs to adjust the timing adaptively for each MT in consideration of the characteristics of the application each MT is using. When the MAG adjusts the timing for each packet in releasing it, it needs a great deal of buffer space (e.g., the resources to buffer user data and compute until buffered packets are released). When multiple MTs execute step 6, we assume that the buffered data is released one after the other from the MAG and that some amount of buffered packets is released simultaneously. There has been no research into how the MAG should manage the buffer space for each MT and executes the traffic control method for the buffered packets when MT makes a handoff.

In many cases, buffered-packet releasing must cooperate with the packet delivery scheduler realized (or implemented) in the wireless access network when the wireless access technology has QoS control capabilities. Studies on the control of buffered packets of TCP traffic [7, 8, 9] have improved the throughput of TCP traffic by discarding some packets beforehand, before the wireless LAN becomes congested. Because the real-time application of UDP traffic is sensitive to packet loss, the method of these studies treating TCP traffic cannot be applicable. The proposed design can cooperate with the design of the existing research concerning TCP traffic.

B. Wireless LAN Issue in Releasing Buffered Packets

The IEEE 802.11e EDCF (enhanced distributed coordination function) classifies the traffic into four access categories (AC*i*, *i*=0, 1, 2, 3) according to QoS requirements for access points (APs). EDCF is a typical example of decentralized controlled mechanisms that do not require a centralized controlled coordinator. Because the decentralized controlled mechanism is used in most cases for wireless access network, we assume that APs adopt the IEEE 802.11e EDCF in this paper.

Each access category in the IEEE802.11e EDCF treats the traffic class that the network operators defined and follows carrier sense multiple access with collision avoidance (CSMA/CA). The IEEE 802.11e assigns different parameter values to different access categories in order to differentiate the flows based on the defined traffic class. Each traffic class is shared by the same multiple users. As a result, a bundle of flow transferred from an access category in an AP into the resident MTs and the handoff MTs is treated in the same way as the same flow transferred from the same access category.

APs that support IEEE 802.11e EDCF do not mitigate the influence of bursty traffic released from the MAG. Bursty traffic degrades the communication quality of the traffic categorized in the same access category. In this case, delay fluctuation is still very large, owing to the burst feature of the back off mechanism in the 802.11e EDCF. The traffic transferred to the resident MTs and the handoff MTs is delayed longer because of the burst traffic. It degrades the communication quality of the resident MTs and handoff MTs because of the congestion in the wireless LAN.

A great deal of research has focused on developing QoS capabilities of the MAC protocol for real-time applications [10] [11]. These researches accommodate real-time traffic by differentiating real-time traffic from non-real-time in order that real-time traffic achieves relatively small delay.

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However, these researches do not address the case where the MAG treats bursty traffic in releasing the buffered packets when multiple MTs make handoffs. In this case, even if the wireless LAN supports IEEE 802.11e, the bursty traffic released from the buffer space of the MAG is stacked altogether in an access category used for the traffic class. We adopt IEEE 802.11e as the wireless access technology and propose a novel traffic control method for the case in which multiple MTs make handoffs.

Mobile service providers are now considering data offloading in the wireless LAN in order to achieve cost reduction of data service and availability of higher bandwidth compared to cellular networks. If the data offloading technology is adopted, most traffic that flowed into the cellular networks will be transferred into the wireless LAN, and the chance of MTs making handoffs into the wireless LAN will increase. QoS capability of seamless mobility management into wireless LAN is important for the future mobile networks.

III. PROPOSED DESIGN FOR SEAMLESS MOBILITY MANAGEMENT

This section explains the proposed MAG structure and two-phase traffic control in consideration of handoffs of numerous MTs in the wireless LAN.

A. Requirements of MT, AP, and MAG

To have novel traffic control in consideration of the handoff of multiple MTs, we assume the following properties are included in the MT, AP, and MAG.

1. MAG establishes point-to-point links when MTs are attached.
2. MAG has output queue management for the traffic and AP has the QoS capability specified in IEEE 802.11e EDCF.
3. MT, AP, and MAG support Layer 2 Handoff (L2 HO) signaling and prediction of which MAG the MT will be attached to, in order to inform the MT's decision beforehand.
4. MAG can identify whether resident MTs or handoff MTs the traffic will be transferred into.

The PMIPv6 standard defines that the logical connections between the MAG and MT are point-to-point links and unique network prefix is assigned for each MT. To assign the network prefix for each MT, the MAG needs to establish the logical connections with each MT by using the IP tunnel in the network segment of the wireless LAN.

To reduce packet loss during the handoff, it is important to exploit the timing of the MT's handoff, and which AP the MT will be attached to, as early as possible. L2 HO signaling has been used to detect the MT's handoff decision in advance [12]. This signaling contains the information about an MT identifier and new MAG identifier. For IEEE 802.16e, MOB_HO_IND messages play the L2 HO signaling role for the handoff. The current IEEE802.11 product does not usually support such signaling, but some research [13] [14] is addressing this. In this paper we assume that MAGs, APs, and MTs support such signaling and functions.

In order to report to the NMAG that the traffic will be transferred the handoff MTs, the PMAG executes packet marking for the traffic of the data-forwarding tunnel. As the NMAG knows the rule of the packet marking beforehand, NMAG can identify whether the resident MTs or the handoff MTs the traffic will be transferred into based on the rule of the packet marking.

We propose a MAG structure that controls the buffered packets into the handoff MTs and the traffic which are transferred directly into the resident MTs separately. The current mobility protocol and MAC protocol in wireless LAN do not consider how the MAG should perform the traffic control for the traffic which will be transferred into the wireless LAN when many MTs make handoffs. The detail of the proposed traffic control is described in III-C. Note that the proposed method requires no modification to the MAC protocol in the wireless LAN and extends the function of the MAG.

B. Proposed MAG Structure for Seamless Mobility Management

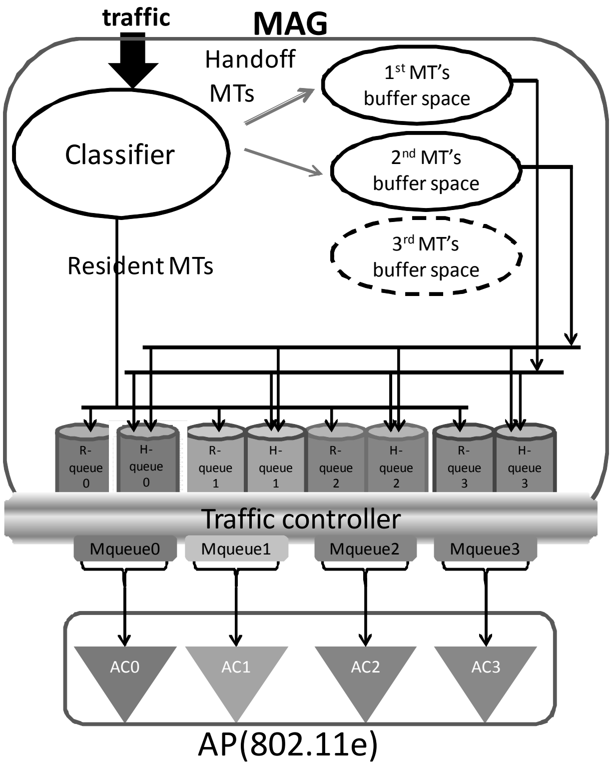

Figure. 2. Proposed MAG structure in the consideration of handoffs of multiple MTs

Figure 2 shows the proposed MAG structure, which adopts IEEE 802.11e as the wireless access technology. To control the traffic which is transferred into the handoff MTs and resident MTs separately, we propose that the MAG has queues dedicated for both types of MT. The queues labeled H-queue are dedicated for the handoff MTs, and those labeled R-queue are for the resident MTs in Figure 2. In Figure 2, the traffic of H-queue[0-3] and R-queue[0-3] are transferred into Mqueue [0-3] and AC [0-3] in AP. Mqueue corresponds to the IP queue of the interface in MAG. We assume the number of the queues that the MAG needs to prepare is not large because IEEE 802.11e defines a maximum of just four types of access category and the AP cannot support so many queues.

Classifier in Figure 2 distinguishes whether the traffic is transferred into the handoff MTs or the resident MTs based on the rule of the packet marking which the PMAG executed. When Classifier recognizes the traffic of the handoff MT, the traffic is transferred into the dedicated buffer space for the MT. When it recognizes the traffic of the resident MT, the traffic is transferred into the R-queue based on the traffic class when the packets is released from the buffer space. Traffic Controller transfers the traffic from each R-queue and H-queue into each Mqueue in the first phase of the proposed traffic control method based on the defined traffic class, as described in the next subsection.

C. Proposed Traffic Control Method in MAG

We propose that the MAG executes two-phase traffic control for the traffic into all the MTs in the wireless LAN. In the proposed method, first the MAG transfers the traffic from each R-queue and H-queue, which treats the same traffic class in a round-robin manner. Second the MAG executes the priority queueing (PQ) for the traffic from each Mqueue following the policy of the traffic control in AP.

We assume it is not suitable for the communication quality of either type of MT to be significantly degraded. Whether the traffic of the resident MTs or the MTs making handoffs is prioritized actually depends on the network operator's policy. However, if the delay characteristics of the traffic which is transferred into the handoff MTs are too large, the procedures of the seamless mobility management themselves become meaningless. We aim here to prevent degraded communication quality of all the MTs when multiple MTs make handoffs in the wireless LAN. Thus, we propose the traffic of resident MT and handoff MT in a round-robin manner in the first part of the two-phase traffic control method. And then we aim not to generate the bursty traffic into the wireless LAN in a round-robin manner. In the second part, the MAG prioritizes the traffic following the priority that IEEE 802.11e defined for each access category. In the second-phase, the MAG executes the PQ discipline in a normal way.

IV. EVALUATION

This section shows the experiment environment of the simulation for the proposed traffic control and the result of the simulation.

A. Experiment Environment

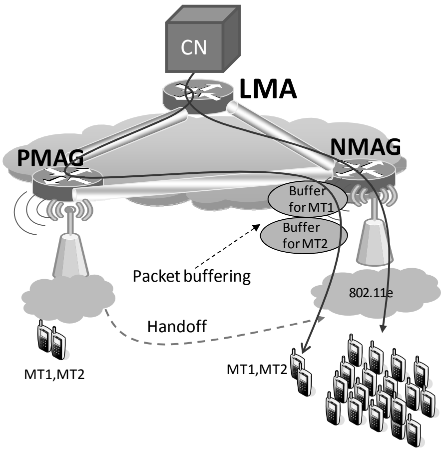

Figure. 3 Overview of experiment environment

Table 1 IEEE 802.11a default parameter values

Parameter	Value
SIFS	10 (*usec*)
SlotTime	20 (*usec*)
BasicRate	54 (*Mbps*)
DataRate	54 (*Mbps*)
LongRetry	4
ShortRetryLimit	7

Table 2 IEEE802.11e EDCF default parameter values for the 802.11a physical layer

Parameter	Value
CWmin	7
CWmax	15
AIFSN	2
TXOP	3.264

We evaluated the degree to which our proposed design would affect the one-way delay values of the high priority traffic after handoff in a simulated environment where certain MTs make handoffs by adopting PFMPv6 as the seamless mobility protocol. This simulation was performed with the QualNet software package [15]. We implemented the two-phase traffic control by extending the queueing discipline of the Qualnet. We employ an IEEE 802.11a-based wireless LAN in AP with an RTS/CTS mechanism and adopt the default IEEE 802.11a configuration, as shown in Table 1. For VoIP traffic, the AP followed the IEEE 802.11e EDCF, which the parameters of the configuration are shown in Table 2.

Twenty MTs communicated with the corresponding node (CN) through the LMA, as shown in Figure 3. The bandwidth of the physical link was 1Gbps. We focused on the downlink traffic buffered from the NMAG when certain MTs make handoffs. We adopt VoIP traffic as the high priority traffic during the seamless mobility. The CN and MTs sent CBR traffic (UDP packet, 200-byte packet, and 20-millisecond inter-packet gap). We considered a case in which only VoIP traffic exists in the wireless LAN.

We evaluated the delay characteristics of VoIP traffic from CN to MTs by changing the number of MTs making handoffs (1, 2, 4, and 6). In the simulation, we assumed multiple MTs concurrently making handoffs. We compared the delay characteristics in two cases of traffic control. In the first case, the MAG does not execute traffic control for the buffered packets in the first of the two-phase traffic control. In the second case, the MAG executes traffic control for the buffered packets in the first of the two-phase traffic control in a round-robin manner (which is the proposed traffic control).

B. *Experiment Result*

We simulated four cases, in which one, two, four, and six MTs make handoffs concurrently. One-way delay values are shown spanning the time when the MAG buffered the packets to 50 ms after when the last buffered packets were released from the MAG (we term this period "handoff affection period"). To investigate the affect of the buffered packets into the communication quality of nineteen, eighteen, sixteen and fourteen resident MTs, we get the one-way delay values in the cases where MTs do not make handoffs. Seeing those delay characteristic values in the cases which no MTs make handoffs, we show how much the traffic of the resident MTs is delayed.

The delay values of the handoff MTs in the case that the MAG executes traffic control in the first of the two-phase traffic control in a round-robin manner are termed "RR Handoff" and those of the resident MTs are "RR Resident". In the same way, we define "No Control Handoff" and "No Control Resident" respectively as the delay characteristics of handoff MTs and resident MTs when the MAG does not execute traffic control in the first of the two-phase traffic control. The delay values of the MTs in the case that no MTs make handoffs are termed "No Handoff".

The parameters of CSMA/CA were randomized by the seed in QualNet. We executed simulations 10 times for each case by changing the value of the seed in order to get the affection of the proposed traffic control. Figure 4 shows the average delay values of all the handoff MTs and all the resident MTs over the ten times. Figure 5 shows the maximum delay values during the handoff affection periods over the ten times. The X-axis shows the number of the handoff MTs in Figures 4 and 5. Y-axis in Figure 4 shows the average delay values (milliseconds). Y-axis in Figure 5 shows the maximum delay values (milliseconds).

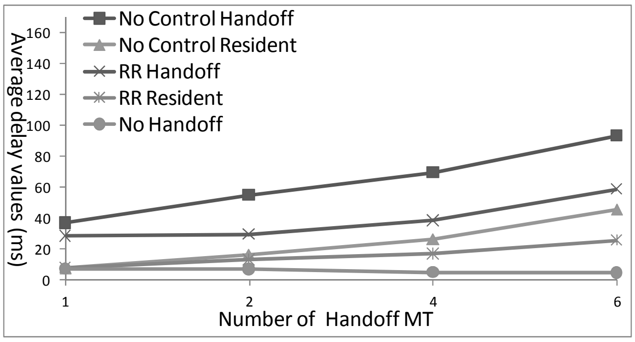

Figure 4. Average delay values of the traffic during handoff affection period

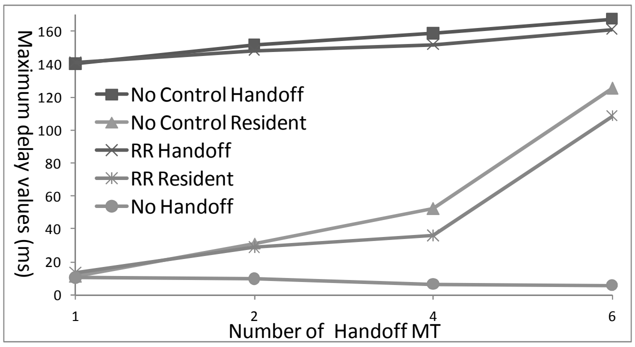

Figure. 5. Maximum delay values of the traffic during handoff affection period

The average and maximum delay values of "No Handoff " in Figures 4 and 5 decrease as the number of handoff MT increases. It is because the number of MTs which receive the traffic through the wireless LAN decreases. Compared with the delay values of the handoff MTs and those of the resident MTs, the values of handoff MTs become much larger because some periods are required for the seamless handoff procedures of PFMIPv6. Figure 4 shows that the released packets from the MAG give large impact on the communication quality of the resident MT. The value of "No Control Resident" becomes much larger than one of "No handoff". When only one MT makes a handoff, the differences between "No Control Handoff" and "RR handoff", and between "No Control Resident" and "RR Resident" is little.

However, applying the proposed traffic control method ("RR Control") when two, four or six MTs make handoffs, the average delay values of both resident MTs and handoff MTs become 0.5-0.6 times smaller than those of "No Control". The bursty traffic in the case of "No Control" delays the traffic which is transferred into both resident MTs and handoff MTs in AP because the backoff mechanism of EDCF makes transferring the bursty traffic into the wireless access networks wait for a few times. When the traffic is transferred in a round-robin manner by preparing the dedicated queue for resident MTs and handoff MTs, the ratio of the bursty traffic is reduced and the backoff

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mechanism does not make the traffic of each MT wait for being transferred from AP. The proposed traffic control method improves one-way delay for both resident MTs and handoff MTs when multiple MTs make handoffs.

Figure 5 shows that the maximum delay values in all the cases are almost same. This shows that the maximum time during that the traffic is made to wait for a few times is not changed even if the traffic is transferred into the wireless LAN in a round-robin manner.

I. Conclusion

Much existing research has tackled the prevention of packet loss and effective signaling for the mobility management protocol. Assuming that the MAG can predict the movement of the MT, packet buffering in the MAG is a key technique for the seamless mobility management because it can prevent packet loss during the handoff. However, such buffered packets give large impact on the one-way delay of the resident MTs in the wireless LAN. The current researches do not treat the communication quality of the resident MTs.

This paper focuses on the management of the buffered packets in the MAG in consideration of the communication quality of resident MTs and handoff MTs. We proposed a MAG structure which prepares the dedicate queue for resident MTs and handoff MTs and two-phase traffic control method using a round-robin manner in the first phase. We used packed-based simulation to evaluate the proposed traffic control method. When only one MT makes a handoff, the effect of the proposed traffic control method was not large. However, when multiple MTs make handoffs, the proposed traffic control method always improved on the delay values of the resident MTs and the handoff MTs because it prevent the generation of the bursty traffic in the wireless LAN which adopts the IEEE802.11e EDCF. As the future work, we need to execute the evaluation in the environment where MTs use the various applications with multiple priorities beside the VoIP.

The proposed traffic control method does not need to modify the AP's functions and is easy to deploy in the commercial mobile networks. The proposed traffic control method is applicable for the cases e.g., passengers in a car are always located in the same network and move in the same direction. In addition, family members and friends tend to spend a significant amount of time together. These cases will more often appear in the future mobile networks.

In this paper, we adopt PFMIPv6 as the seamless mobility protocol. However, the proposed traffic control method and design of the MAG is applicable for the other mobility management protocol (e.g., [16] [17]) besides PFMIPv6. It is because our proposal does not require the change of the protocol and focuses on the function of access gateway. Our proposal can devote the traffic management of the future mobile networks.

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Multi-view Rendering Approach for Cloud-based Gaming Services

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***Abstract*—In order to render hundreds or thousands of views for multi-user games on a cloud-based gaming at interactive rates, we need a solution which is both scalable and efficient. We present a new cloud-based gaming service system which supports multiple viewpoint rendering for visualizing a 3D game scene dataset at the same time for the multi-user games. Our multi-view rendering algorithm maps well to the current graphics processing units (GPUs) and we have evaluated its performance on two different GPUs with different rendering resolution. The experimental results demonstrate the multi-view rendering method can be successfully applied to the multi-user games.**

***Keywords*-gaming on demand, multi-view rendering, video encoding, video streaming, cloud computing**

I. Introduction

Cloud computing is a general term for complete services delivered over the networks using self-service end user portals and flexible business model that incorporates hardware, software, and services into a single revenue stream. This computing model allows a performance focus at a single location, the cloud server, and enables user mobility and pervasive access for the users. One of the latest advancements in gaming technology that enables such a ubiquitous gaming are *cloud-based gaming service systems*, also called *Gaming on Demand* (GoD) [1]. They are networked media platforms that offer web-based services allowing game play on smaller end-devices like low-cost PCs or set top boxes without requiring the games to be installed locally.

The ultimate goal of the cloud-based gaming services is to provide pervasive game access on devices such as set top boxes and mobile devices that typically do not have a full set of technical requirements to run high-quality video games. This goal can be achieved by processing the game execution logic such as rendering, audio, artificial intelligence (AI) and physics, on at remote high-end servers being streamed as an interactive video over a network to be played on lightweight clients. Recent work in this area has been focused on video encoding and streaming techniques to reduce the latency in games. Most of the earlier systems were serial in nature and designed for a single core or processor in terms of 3D rendering.

The recent trend in computer architecture has been toward developing parallel commodity processors, including multi-core CPUs and many-core GPUs. It is expected that the number of cores would increase at the rate corresponding to Moore's Law. Based on these trends, many parallel game engines [2] and parallel rendering algorithms [3][4] have been proposed for commodity parallel processors.

In this paper, we propose a new cloud-based gaming service system to support the multi-view rendering based on multi-threaded game engine [2] for the multi-user games. To provide convincing streaming-based GoD services, we describe the key requirements of cloud-based service systems as follows:

- *User responsiveness*: Latency is defined as the time between a player's action and the time the actual resulting game output on the player's screen. Since computer games are highly interactive, extremely low latency has to be achieved. The interaction delay in the games should be kept below 80ms in order to guarantee suitable user responsiveness [5].
- *High-quality video*: In order to provide a high-quality video (above 720p) interactively, data shall be reduced as much as possible but keeping quality. However, video encoding is computationally quite demanding.
- *Quality of services*: In the case of network congestion, the network problems like increased latency, jitter, and packet losses distribute evenly on all competing traffic. However, the quality can be enhanced using quality of service (QoS) technologies to giver higher priority to game traffic in the network bottlenecks [6].
- *Operating costs*: Since the servers of cloud-based gaming service system have high-performance CPUs and GPUs, the operating costs for the servers are quite expensive. So, it is necessary to develop the optimization technologies to minimize power consumption and network bandwidth [7].

Our contributions: We present a novel system architecture for the cloud-based gaming services, which utilizes parallel commodity processors, multi-core CPUs. We also present a novel *multi-view rendering* algorithm to efficiently support multi-user game on the server, which has a single GPU with multi-core CPUs. Our algorithm can easily handle insertion and removal of viewpoints and can also take advantage of scalable and parallel processing using multi-core CPUs. In addition, our approach give the benefits in terms of *arbitrary focal positions* for viewpoints and better rendering quality over prior parallel multi-view rendering methods [8].

The rest of the paper is organized as follows. We briefly

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survey previous work on Gaming on Demand (GoD), parallel rendering and video encoding for the cloud-based gaming services in Section II. Section III describes the proposed the system architecture and the core systems in our system. We explain implementation details of our multi-view rendering algorithm and describe the performance result in Section IV. In Section V, we compare our system and algorithm with prior GPU-based algorithms and highlight some of the benefits. Finally, we discuss future work and conclude in Section VI.

II. Related work

In this section, we give a brief overview of related work on cloud-based gaming technology and parallel rendering algorithms. We also highlight many technical characteristics of cloud-based gaming services, parallel rendering and video encoding.

Gaming on Demand (GoD): There are a number of commercial Gaming on Demand systems that have been presented to the market [6]. OnLive is a gaming-on-demand entertainment platform, announced at the Game Developers Conference (GDC) in 2009 [7]. Gaikai launched GoD beta service; *Gaikai beta*, based on a cloud-based gaming technology that allows users to play major PC and console games [9]. The clients of their service can display audio/video (AV) game streams, which were streamed from the cloud, by using previously installed plug-ins such as Java or Adobe Flash on client devices. Even though both are cloud base gaming, OnLive and Gaikai have different goals in mind. OnLive sells full games, provides demos, brag clips, and being able to watch other players play games (Arena) while Gaikai advertises games via a webpage as demos [7].

Visual effects rendering based on *global illumination* that commonly requires extensive hardware and processing time. However, the OTOY can provide visual effects rendering in real-time using the cloud; *Fusion Render Cloud* (FRC), through the power of *server side rendering* [10]. However, there is very little detailed technical information publicly available about these commercial systems.

The *Games@Large* (G@L) framework enables commercial video game streaming from a local server to remote end devices in local area networks (LANs) [11]. This system and streaming protocols are developed and adapted for highly interactive video games [1] [12].

There are two major approaches for the game streaming. One is *3D graphics streaming* approach which exploited for streaming the game's output is to directly transmit the graphics commands to the client device and render the image on the client device [3]. The other approach is *video streaming* that the server renders the game graphics scene, the framebuffer is captured, eventually downsampled to match the target resolution, and the current image is encoded using standard video codes such as MPEG-2, MPEG-4 and H.264 [13][14]. The video streaming is intended for thin-client devices lacking hardware accelerated rendering capabilities [15]. In our research, we exploit a video streaming method since our system should support the thin-client devices.

Parallel rendering: Much of the recent work in the area of parallel rendering has focused on using networked clusters of commodity PCs. Such systems can generally drive a tiled display using a commodity local network as well. There are three major approaches according to a sorting classification of parallel rendering such as *sort-first*, *sort-middle* and *sort-last rendering* [16].

Also, there have been various research efforts to multi-view rendering and scalable rendering [4]. However, those methods cannot be directly employed for multi-view rendering for multi-user games since those methods usually focus on multipipe display systems, workstations with multiple monitors, walls build out of multiple screens or projectors as well as immersive environments.

A parallel multi-view rendering architecture in a cluster of GPUs has been proposed in [8]. This system have shown a theoretical analysis of speedup and scalability of the proposed multi-view rendering. However, the critical limitation of this method is that all the cameras are always looking to the center of arbitrary tile. This is not suitable for common mutli-user game applications. Moreover, it is difficult to apply this method to a *high visual quality* (photo-realistic) games since they used a simple phong shader for lighting and shading.

In this paper, we exploit a parallel game engine for improving the multi-view rendering performance as well as the visual realism in the games as shown in Fig. 1.

Video encoding: Many techniques have been proposed to accelerate the performance of video encoding algorithms. In H.264/AVC encoders, *macroblock partitioning* and *motion vector calculation* are computationally very demanding. An acceleration based on render context information has been developed, which allows the direct calculation of motion vectors, similar to [13]. The parallel model of the encoder using multiprocessor platforms has been introduced in order to improve the encoding performance in [17].

OnLive introduced *interactive video compression* method designed for video games. In order to achieve high performance encoding, they developed two dedicated compression hardware for video encoding; *optimized compressor* based on *human perception* and *live compressor* similar to conventional compressor [7].

Current GPUs are regards as high-throughput processors, which have a theoretical peak performance of a few Tera-Flops. In order to accelerate the performance of the motion estimation, fast motion estimation implementation using CUDA on GPU has been proposed in [18]. Recently, OTOY introduced new video encoding method, so-called *ORBX*. ORBX has been designed from the ground up to take advantage of OpenCL based GPU servers (FRC). ORBX encodes video entirely on the GPU, with more than 30-100x the scaling of H.264 encoding solutions requiring either a CPU or specialized encoding ASIC [10]. Unfortunately, the technical information of the ORBX encoding method is not publicly available.

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Fig. 1. The result of our multi-view rendering algorithm (eight views with 640x480 resolution for each view.)

III. System Architecture

In this section, we describe the proposed system architecture for the cloud-based gaming services. Our architecture consists of three major systems such as distributed service platform (DSP), distributed rendering system (DRS) and encoding/QoS/streaming system (EQS) as shown in Fig. 2.

A. System Overview

The Distributed Service Platform (DSP) is responsible for launching the game processes on the game execution nodes or rendering job on the Distributed Rendering System (DRS) after client-side invocation, monitoring its performance, allocating computing resources and managing user information. And, the DSP is responsible for processing user's game input via UDP from the client-side devices. In client-side, the user's game input is captured and transmitted via UDP by the user input capturing and transmission software on the client devices. Also, the DSP performs execution management of multiple games. In order to perform streaming the game A/V streams to the clients, the DSP requests capturing rendered frame buffer for video encoding and streaming to the Encoding/QoS/Streaming System (EQS).

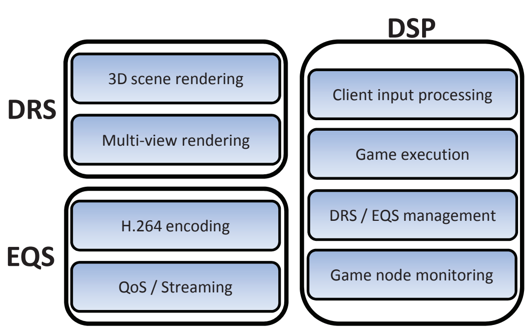

Fig. 2. Our system architecture: *DSP-Distributed Service Platform, DRS-Distributed Rendering System, EQS-Encoding, QoS and Streaming System*

The DRS is responsible for rendering a 3D scene and *multi-view rendering* for multi-user games. To improve 3D rendering performance in games, we utilize the multi-threaded game engine [2] that is designed to scale to as many processors as are available within a platform.

The EQS is responsible for audio/video encoding and streaming the interactive game content to the clients. In order to implement the visual capturing for the games, we utilize the DirectShow SDK. And we utilize the H.264 video coding standard for low-delay video encoding of the captured game content. Before the EQS performs the H.264 encoding, we perform a color space conversion from RGB to YUV on the captured frames. Finally, we exploit the Real Time Protocol (RTP) packetization to transmit the encoded video stream in real-time [19][20].

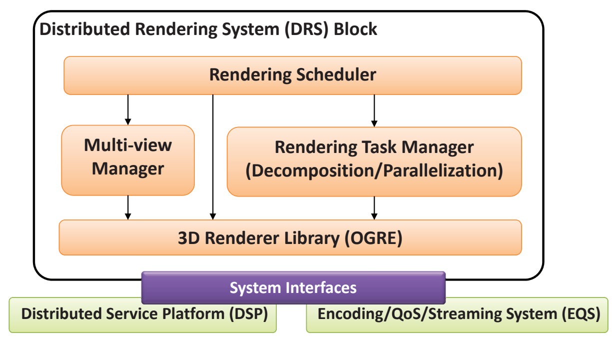

Fig. 3. The DRS block architecture

B. Distributed Rendering System

The Distributed Rendering System (DRS) consists of four major block components such as *rendering scheduler*, *multi-view manager*, *rendering task manager* and *renderer library* as shown in Fig. 3. The rendering scheduler is responsible for rendering process monitoring, performance timer control, rendering statistics management and communicating other modules for external rendering requests in the DRS blocks. The key performance improvements for the game applications is the use of per-thread task queues. This eliminates the synchronization checkpoint when one shared task queue is used.

Advanced task schedulers may use heuristics to determine which thread to steal from and which task to steal and this may help cache performance. In order to implement the rendering scheduler, we use the Intel Threading Building Blocks (TBB) [21], which is highly optimized scheduler for Windows, Mac OS X, and Linux. The multi-view manager is responsible for performing the management of user's viewpoints (such as insertion, deletion, update and search operations) for the shared spaces in the multi-user games. The rendering task manager module performs the rendering task decomposition and parallelization in order to improve the rendering performance. In our work, we use the Object-oriented Graphics Rendering Engine (OGRE) [22], which performs a 3D scene graph management and rendering. In order to provide the cloud-based gaming service, the DRS should have common system interfaces to the DSP and the EQS.

C. Multi-view Rendering

In the case of multi-user games, multiple viewpoints are needed if we want to support several users visualizing a given 3D scene at the same time. However, rendering multiple views using the standard graphics pipeline is a challenging problem.

In order to provide the interactive multi-view rendering results for the cloud-based gaming service, we utilized the shared resources for the rendering such as scene graph, textures and shaders in a GPU as much as possible and keeping the quality of the rendering results.

If R_i denotes a i-th rendered image in framebuffer of the DRS, then $\mathcal{S}_k$, which has i image sequences is defined as:

$$\mathcal{S}_k = \{R_1, R_2, ..., R_i\}$$

The $\mathcal{CP}_i$ denotes the i-th viewpoint parameters, which contains internal parameters such as focal length $f_l(f_x, f_y)$, center $c(c_x, c_y)$, aspect ratio a and external parameters such as position $p(c_x, c_y, c_z)$ and orientation $r(r_x, r_y, r_z)$. The DSP generates this $\mathcal{CP}_i$ according to the requests of the clients.

Algorithm 1 Viewpoint addition algorithm.

procedure ADDVIEW($U_i, \mathcal{CP}_i$)
    RenderWindow W;
    Camera C_i;
    Viewport V_i;
    RenderedFrameBuffer R_i;
    $C_i \leftarrow$ createCamera(U_i, $\mathcal{CP}_i$);
    $V_i \leftarrow$ addViewport(C_i);
    $R_i \leftarrow$ renderOneFrame(W, V_i, C_i);
    return R_i
end procedure

The DRS provides the function for adding the multiple viewpoints to support the multi-view rendering. First, the DSP receive the service requests from the clients. These requests include several user information, U_i, such as user identification, selected game, which they want to play and initial or previous viewpoints in the 3D game space. Then, the DSP sends these information to the DRS to request for multi-view rendering. According to this request, the DRS provides the function for adding viewpoints, $\mathcal{CP}_i$. To perform this function on the DRS, we create the cameara C_i and viewport V_i objects to attach the viewport to the render window W_i. After the viewport was successfully added to the render window, the DRS performs the rendering procedure to generate an image on the framebuffer in a GPU. The pipeline of our algorithm for multi-view rendering is shown in **Algorithm 1**. Another function for multi-view rendering is deletion function for viewpoints in the multi-user games. This function can be easily implemented similar to the viewpoint addition algorithm.

If EA_i and EV_i denote a i-th encoded audio and video in interactive game content respectively, then $\mathcal{ES}_k$, which has i encoded audio/visual gaming sequences is defined as:

$$\mathcal{ES}_k = \{(EA_1, EV_1), (EA_2, EV_2), ..., (EA_i, EV_i)\}$$

Therefore, the EQS performs the streaming $\mathcal{ES}_k$ to the clients for the cloud-based gaming services. In order to address the game's audio/visual output capturing, we develop the capturing module on the EQS in C++ and DirectShow SDK. We also develop the H.264 encoder for achieving low-delay video coding. Before the EQS performs the H.264 encoding, a color space conversion from RGB to YUV (convertRGB2YUV function in **Algorithm 2**) takes place on the captured frames F_i. We utilized the 4:2:0 method for the YUV sampling to achieve the reduction of storage data. We also capture and encode the audio data for the games to transmit the interactive game content to the clients. In our work, HE-AACv2 is utilized for audio streaming. And then, the EQS transmits the encoded AV content for the game to the client via RTP/RTCP [20]. The details of our video encoding and streaming algorithm for interactive gaming content is shown in **Algorithm 2**.

Algorithm 2 Video encoding and streaming algorithm.

procedure ENCODINGANDSTREAMING(U_i)
    YUVImage Y_i;
    FrameCapture F_i;
    FrameNumber f;
    EncodedAudio EA_i;
    EncodedVideo EV_i;
    $F_i \leftarrow$ captureRenderedFrameBuffer(U_i, f);
    while $F_i \neq$ NULL **do**
        $Y_i \leftarrow$ convertRGB2YUV(F_i);
        $EV_i \leftarrow$ encodeFrame(Y_i);
        $EA_i \leftarrow$ captureAndencodeAudio(U_i, f);
        $\mathcal{ES}_i \leftarrow$ transmitAVstream(EA_i, EV_i, f);
        $F_i \leftarrow$ captureRenderedFrameBuffer(U_i,f);
    end while
end procedure

On the other hand, the client side devices for our system support the H.264 decoding functionality. Also, the client is responsible for capturing the commands of the input controller such as keyboard and mouse, and sending them to the DSP via UDP.

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IV. Implementation and Performance

In this section, we describe the implementation of our system and highlight the performance of our multi-view algorithm.

Implementation: We have implemented our multi-view algorithm on two different commodity GPUs: a NVIDIA GTX 480, a NVIDIA Quadro 4000. In order to test the performance of our system, we used five workstations (Intel Core i7, 8G RAM) which were connected to a 100 Mbps switch via a wired Ethernet connection. Also, the eight laptops as thin-clients (Intel 2GHz, 1G RAM) were connected to the 100Mbit LAN. These laptops are capable of H.264 decoding and displaying the game videostream as shown in Fig. 4.

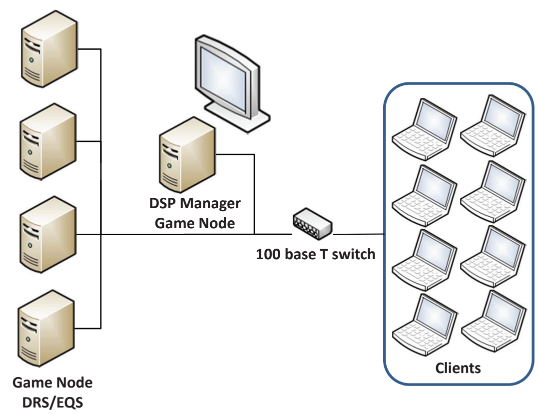

Fig. 4. Performance test setup: *This figure shows the system configuration; five workstations for servers, eight laptops as thin-clients, for the performance testing.*

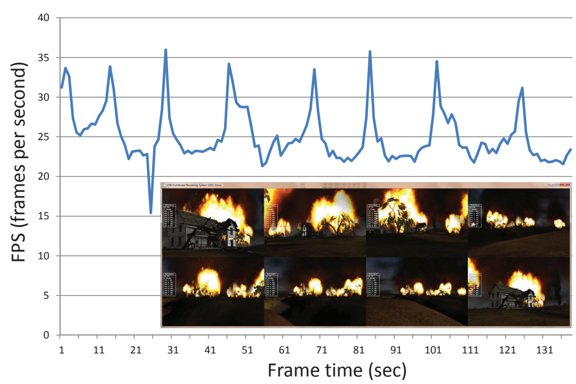

Fig. 5. Performance of multi-view rendering: *This figure shows the frames per second (fps); 25.4 on average, for multi-view rendering (640x480 resolution for each view) on a NVIDIA Quadro 4000.*

Performance: First we evaluate the performance of multi-view rendering on a PC running Windows 7 operating system with Intel Core i7 2.93GHz CPU, 8GB memory and a NVIDIA Quadro 4000. We used OGRE library based on DirectX as a graphics API and Microsoft HLSL for a shading language.

The frames for second (FPS) is the number of frames per second that have been rendered by the DRS. High FPS results with smooth movements in the 3D scene. Our system rendered 8 views at 25.4 fps on average with one GPU. We measured the FPS every second at the DRS for rendering at 640x480 resolution for each view. Fig. 5 shows the performance result of multi-view rendering.

We utilized the parallel game engine; *Intel Smoke* [2], and we adopted our multi-view rendering algorithm. Then we ran it on a 8 core system with a NVIDIA GTX 480 to measure the scalability of our rendering system as shown in Fig. 6.

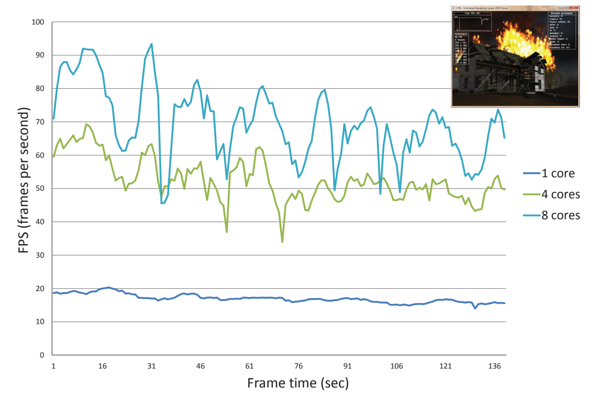

Fig. 6. Scaling performance of the DRS according to the number of CPU cores: *This figure shows the scalability of our parallel (multi-threaded) rendering (1600x1200 resolution) on a Intel i7 8-core (quad-core with hyperthreading) with a NVIDIA GTX 480. Average FPS - 1 core: 16.9, 4 cores: 52.6, 8 cores: 69.7*

In terms of the performance of our encoding system, our system can encode in 25.6ms on average for eight views (interactive gaming videos) with 640x480 resolution in parallel and 24.9ms for a 1600x1200 video. **Table I** shows the supported technical features of AV encoding in the proposed system.

TABLE I
THE SUPPORTED FEATURES OF AUDIO/VIDEO ENCODING IN OUR SYSTEM.

Supported features	Audio	Video
Codec	HE-AACv2	MPEG-4, H.264
Resolution	-	320x240 - 1600x1200
Bitrate	16Kbps - 64Kbps	384Kbps - 5Mbps
Frame Rate	-	5 - 30fps
Sampling Rate	22.05KHz - 48KHz	-
Channel	Mono, Stereo	-

V. Analysis

In this section, we evaluate the performance of our system in terms of rendering and encoding functionalities and highlight some of the benefits.

Analysis: Our rendering system provides good performance scaling of multi-core CPUs for multi-view rendering. And the multi-view rendering algorithm maps well to the current GPUs and we have evaluated its performance on two different GPUs with different rendering resolution. Furthermore, it is relatively simple to combine the video encoding methods and optimizations in the streaming-based gaming service framework. This makes it possible to develop a more flexible

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GPU-based framework for the video encoding methods like H.264/AVC or ORBX which is GPU-based encoding schemes.

Limitations: Our approach has some limitations. First, we support the multi-view rendering for one multi-user game, since it is difficult to share the rendering resources in a GPU among different games. We believe that this can be resolved by using multi-GPUs. Secondly, our system performs directly rendering to the framebuffers on the server-side machines. However, in terms of efficient services in the cloud-based gaming, we should exploit the *off-screen rendering* approaches and *GPU virtualization* techniques.

VI. Conclusion and Future Work

In this paper, we have presented a system architecture for the cloud-based gaming service and multi-view rendering. Our rendering system greatly improves utilization of hardware resources present in the system, allowing to utilize both multi-core CPUs and a GPU simultaneously.

We found that the proposed system provide the multi-view rendering for different focal positions for each viewpoint with high visual quality. Moreover, our approach is flexible and maps well to current GPUs in terms of shared resources such as textures and shaders for rendering. In addition, we demonstrate that the proposed rendering system could prove to be scalable in terms of parallel rendering. So, we believe that our rendering system will provide high-quality with good performance for the cloud-based gaming services.

There are many avenues for future work. It is possible to use new capabilities and optimizations to improve the performance of the video encoding especially H.264/AVC through the GPU-based implementation. Furthermore, we would like to develop algorithms for integrating the multi-view rendering with the video encoding in a GPU.

Acknowledgments

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Location-based Service with Spatial Data Analysis within IP Multimedia Subsystem

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***Abstract*—Nowadays applications and services, which utilize information about client's geographical position, are becoming more and more popular. Such kind of applications are usually called Location-based services (LBS). Location-Based Service technology has great potential to add value to existing or new wireless mobile data services and for example for LBS developers, mobile advertising and spatial data providers. Virtually all of the big mobile device and services-oriented companies are acquiring start-up companies dedicated to mobile advertising based on location data. The IP Multimedia Subsystem (IMS), on the other hand, is meant to be global and access-independent and to have all Internet protocol (IP) based connectivity and service control architecture, as defined originally by 3GPP in their Release 5. Since that time, it also has evolved to cover the Fixed Mobile Convergence (FMC) and provides reliable charging, security and quality of service. This research investigates the methods of analyzing spatial data (movement pattern extraction) and shows how "smart" location services can be build on top of the IP Multimedia Subsystem. It shows that open Internet standards and open source solutions can be utilized to build an LBS service on top of IMS. As the standards are open, it is easy for anyone to see what's behind the service and satisfy themselves that no proprietary solution has been used. The proposed software architecture opens possibility to add other models for spatial data analysis.**

***Keywords*—Location-based services; IP Multimedia Subsystem; spatial data analysis.**

I. Introduction

The IP Multimedia Subsystem was a significant step forward to integration of different multimedia services into a single infrastructure which allows the user to access those services via a common interface and one (Session Initiation Protocol - SIP [3]) account. Unfortunately, location services were not included into the IMS core, and the current solutions do not allow the use of geographical location within IMS services. LBS differ quite a lot in their functionality and in their approach for location data processing. Some of them are comparatively "simple" and inform mainly on the current position of object. Other services, which are more "intelligent" deal with a set of location data that usually represents the history of movement for a certain period and object. By analyzing such kind of data, it is possible to find some rules which describe how an object behaves in space-time dimensions. The aim of present research is twofold. First, to develop location-based service that collects movement history data and analyzes it in order to predict the future location of an object. Second, to connect that service to IMS and show how both of these technologies can work together. By combining these, the common software architecture is created that can be easily expanded by adding additional modules to the location-based service. The prediction of the future location is then linked with timestamps. As a result of this, the location-based service can deliver services beforehand. This brings great value in case of advertising as they would not have to wait until the user is in the vicinity of the service. Instead the ads could be sent based on the history prediction. Sections II and III are giving overview of the location-based services and the IMS, while Section IV presents the proposed architecture and implementation.

II. Related Work on Location Based Service

A Location-based service infrastucture can be accessed with a mobile device utilizing geographical position data. The key to LBS is to know the location of the user, so that an appropriate service can be delivered. However, that location might not necessarily have to be related to the current position but could also be some future location of the user. The most common examples of a location-based services are: turn-by-turn navigation, locating people, requesting the nearest service point. Nowadays location-based services can be used in combination with messaging, for example in mobile advertising.

A. Positioning technologies

Today, there are many ways to get position information for the mobile device. There are many terrestrial systems based on radio signals including Angle of Arrival, Time of Arrival, Time Difference of Arrival and satellite-based systems such as Global Positioning System which is the best-known and the only fully operational satellite positioning system [25]. Assisted-GPS [6] is used in the implementation part of this research.

B. Building blocks for LBS

A Location-based service infrastructure consists of several components: mobile device (User-side device that is able to use location-based services), mobile network, service and

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content provider (Owner of a geographical data storage; offers actual location-based services), positioning system. Detailed investigation of mobile devices and mobile network components is not within the scope of the current research. Mapping data that content providers use is discussed further in this chapter. LBS applications typically use information from several content databases: the road network (digital maps); business and landmark information, often referred to as Yellow Pages, or Point-of-Interest (POI) information; and dynamic data such as traffic and weather reports. One additional data type used in this research is history data on the user's movement over time.

C. Maps data

Building LBS applications starts with the collection of road data. Map database vendors collect and convert raw geographic content into digital formats. The map data are captured in many ways, ranging from satellite imagery to scanned maps to manually digitized paper maps. Some vendors physically drive along each road segment in a GPS-equipped car, recording every change of direction and photographing road signs be able to provide information on specific road conditions such as turns and height/weight restrictions. Each vendor's data are different, which accounts for some of the discrepancy in the maps and routes generated. One of the easiest ways of getting maps data and using it to utilize the application programming interface (API) of some known available maps applications such as Google Maps [9] or Microsoft Bing Maps [12]. This research employs the Google Maps API.

D. Point-of-Interest Information

One of the most popular LBS applications is Yellow Pages, or concierge services. Mobile concierge-type services help users locate businesses near a specified location. These services help answer questions such as: "Where is the airport?" or "Where is the nearest gas station?" Concierge applications use business and landmark information that has been compiled into POI databases. Integrating a map database with a POI database creates a detailed, digital representation of a road network and business services available along it. These POI databases contain the kind of detailed information typically found in a phone directory and add value to the map database's geographic content [23].

E. Movement Patterns

By definition, moving objects are entities whose positions or geometric attributes change over time. However, in many cases the dimension of an object is not as important as its position. Hence, moving objects are considered as moving points, whose trajectories (paths through space and time) can be visualized and analyzed. In most cases, moving object data sets are quite large. Therefore, it is necessary to develop efficient data mining algorithms in order to extract useful information. This information contains knowledge about object behavior, which is known as movement pattern. Generally, movement patterns include any recognizable spatial and temporal regularity or any interesting relationship in a set of movement data [24]. Every movement, by nature, has several spatial, temporal and spatio-temporal parameters. In case of spatial movement, the parameters are position, distance, and direction. In case of temporary movement, the additional parameters are duration and travel time. And in case of spatio-temporal movement, the additional parameters are speed and acceleration. The importance of each characteristic depends on domain and the problem we need to solve. In other words, in most cases there is no need to record and analyze all information about object movement. In addition to movement parameters mentioned above, we should consider a set of other environment variables, which influence the choice of pattern extraction algorithm. One of the most significant factors is the number of objects involved with movement (individual or group movement). However, it is important to understand the difference between groups which consist of independent objects and groups in which objects are linked to each other. In the first case, pattern extraction task can be divided into subtasks, and extraction an algorithm can be applied to each object separately. In the second case, we deal with the behavior of the entire group. This type of movement is usually much more difficult to analyze, and advanced data mining techniques should be applied [10]. Another important factor to consider before collecting information about object movement is path type. Paths of movement objects may take different forms. Most objects travel more or less continuously, generating a continuous path (a pedestrian, a car moving on a road). Such a continuous path is typically discretized into regular steps prior to computing the movement parameters. This research employs the individual and continuous movement.

III. IP Multimedia Subsystem

In order to support universal IP connectivity the IMS should be able to use multiple transport technologies to guaranteed connectivity. So regardless of the underlying access network or the user's terminal features, IMS-related services should be usable. The IMS can be seen as a glue between different services [7]. The OpenIMS core [14] has offered an open source based possibility to try out the IMS deployment. Although the OpenIMS is not a commercial solution its performance can be considered to be suitable for smaller players in the telecommunications business [5], [4]. Other additional reasons for the IMS deployment are:

- IMS offers, for 3GPP/ETSI-TISPAN, a standardized and developed IP system environment containing standard interfaces to external systems and networks
- IMS contains, as a default Authentication, the Authorization and Accounting (AAA) system for user identification and service authorization.
- IMS enables SIP/Voice over IP (VoIP) capability by default when using public or private WLAN networks anywhere in the world, and usually comes with a single flat rate fee.

The core elements of the IMS architecture are called: Call Session Control Functions (CSCF). There are three main CSCFs:

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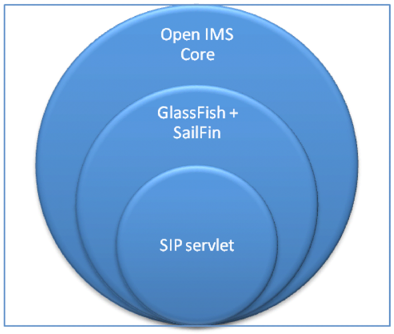

Fig. 1. The LBS integration

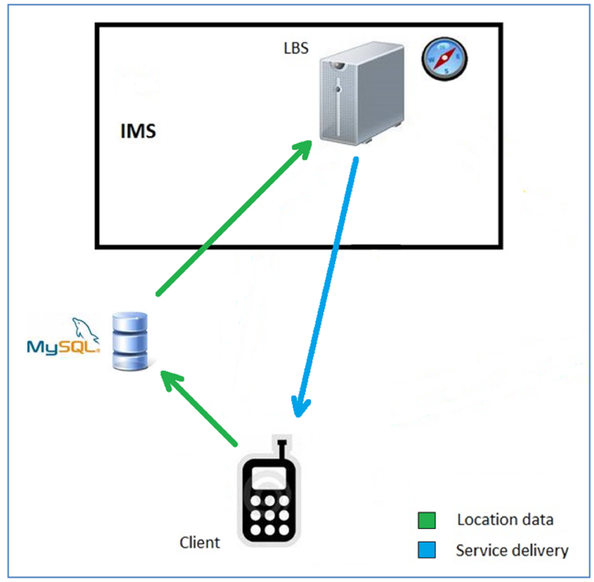

Fig. 2. The data flows

Proxy CSCF (P-CSCF), Serving CSCF (S-CSCF), Interrogating CSCF (I-CSCF). Each of these has its own special tasks [22]. Proxy-CSCF is the first contact point for users within IMS. This means that all SIP signaling traffic from the users terminal will be sent to the P-CSCF. S-CSCF is responsible for handling registration processes, making routing decisions, maintaining session states, and storing service profiles. I-CSCF is a contact point within the service operator's network for all connections destined to a subscriber of that network operator. The Home Subscriber Server (HSS) can be considered as the main data storage for all subscriber and service-related data of the IMS. The most important data that the HSS holds is user identities, registration information, access parameters and service triggering information [1]. For the research the Open IMS Core implementation was deployed. This implementation used only the basic IMS Core elements, consisting Proxy-CSCF, Serving-CSCF Interrogating-CSCF and HSS, for the test platform. These elements are made by the Open IMS Core project by Fraunhofer Fokus [14]. The CSCFs used were from Open IMS core project SVN version 732 and HSS was Java based FHoSS (Fokus Home Subscriber Server) running with MySQL server version 5.0.51a. Java used was that of Sun Microsystems Java Runtime Environment 1.6.0 update 12. The hardware configuration had the following characteristics; XCSCF servers and HSS located in the same virtualization based Linux PC framework; Server Dell PowerEdge 2950 with 2 Intel Xeon E5420 processors and 32 GB ECC-DDR2 memory; 4 750 GB SAS disks using RAID 10 configuration. Virtualization was done with Citrix's XenServer 5.0.0. A virtual machine was configured with 2 virtual processors, each using a 4096 MB main memory, 15GB of disk memory and one 1Gb Ethernet port.

IV. Proposed LBS Architecture for the LBS Application Server

IMS architecture supports the extension of core functionality with the help of application servers (AS). Thus, integrating LBS into IMS means implementing and configuring the application server, which will function as part of IMS. In the current research, the GlassFish application server and the SailFin SIP servlet container were chosen as the deployment platform for a location-based service (application server) [18], [11], [21]. To test the LBS functionality in a real IMS environment, the Open IMS Core implementation was deployed as described in chapter III. The LBS itself is implemented as a SIP servlet [2] and written in Java. The implementation environment had following characteristics Sailfin 1.0 which is based on GlassFish v2.1 (Java EE 5) and SIP Sevlet API 1.1 (JSR 289). This developed application server was based on Intel Core 2 Duo E8400 CPU with a 4 GB DDR2 memory and a 1Gb Ethernet port. The operating system was Debian GNU/Linux 5.0. The integration approach described above is shown in Fig. 1. Another important part of the architecture is the client. In the context of a developed system, the client is a mobile device with an installed application, which "understands" the SIP protocol and periodically sends the user's location data. However, for testing purposes these requirements could be simplified. During the development phase of the SIP Test Agent plug-in for NetBeans IDE, was used to send SIP messages to a location-based service [20]. As shown in Fig. 2, the interconnection between the Client and IMS consists of 3 components (Client, LBS and MySQL database [19] and of 2 data flows (collecting location data and providing the service). The MySQL database includes 2 major tables: the table of users ("users") and the table of services ("services"). The purpose of the "services" table is to provide information about POIs during the service delivery phase. The "Users" table is used for saving the user's data, such as: location information (latitude, longitude), the name of the user, presence status ("online" or "offline") and some other fields. The important feature of the database implementation is that the the "users" table "knows" only the current position of the client at a particular moment. The application Server (Location-based

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service) scans the table once every minute to obtain location data and merges all the coordinates into one route. LBS then uses this information to create a user profile and to choose services with the best match with the user's everyday routes. When enough statistical data have been collected, the application server switches to the service mode, in which mode it sends advertising messages to the client. Position data saved on the application server is in the Extensible Markup Language (XML) [8] format and tagged for latitude, longitude, date and time.

A. Implementation of the LBS Application Server History Analysis Module

The main idea of the application server functionality is to collect location / time information about client, and based on this data, try to predict the user's movements in the nearest future. In anticipation of these movements, advertising message will be sent to the user with a POI description. The client's routes are saved in the XML format and include a set of "coordinates" tags such as latitude, longitude, date and time. In addition, the user should also provide information about home location and the location of work office, so that the user's movement pattern can be extracted correctly. Fig. 3 gives a general view of the application server functionality. In the figure "History" means the user's routes that were described above (a set of XML files). The "Spatial data analysis" part will be covered later in this chapter. The POIs database is implemented with the help of the MySQL server, thus application server will connect to it to obtain needed information. Finally, "Potential services" is an intersection between the POIs database and the user's movement pattern. Preparation for the extraction of a movement pattern consists of several steps. First, a spatial area between the home and work locations should be divided into regions (Fig. 4). When choosing the size of that region, we should try to find a compromise [17]. The region should be small enough to be accurate, but should not be too small to avoid generation of unnecessary data and to provide better pattern extraction. In the current research, region width equals a 0.003 longitude coordinate difference (about 200 meters) and region height equals a 0.002 latitude coordinate difference (about 220 meters). Second, we should save the regions in a format suitable for data processing. Thus we save them in a matrix in which each region is represented by 4 borders so that there are left, right, top and bottom borders. Third, we parse the history data of the user's route and represent this information in a similar (tabular) way, including latitude, longitude, date and time. Fourth, we should find the correspondence between the latitude/longitude coordinates in the movement history table and region number (region number equals row index in the regions table) so that there would be region, date and time. This can be achieved by checking a simple condition: "latitude (top border AND latitude) bottom border AND longitude (right border AND longitude) left border". We then divide all the history data by days, so a sequence of regions can be composed for each day (Fig. 5). Finally, we create a table of time stamps. It

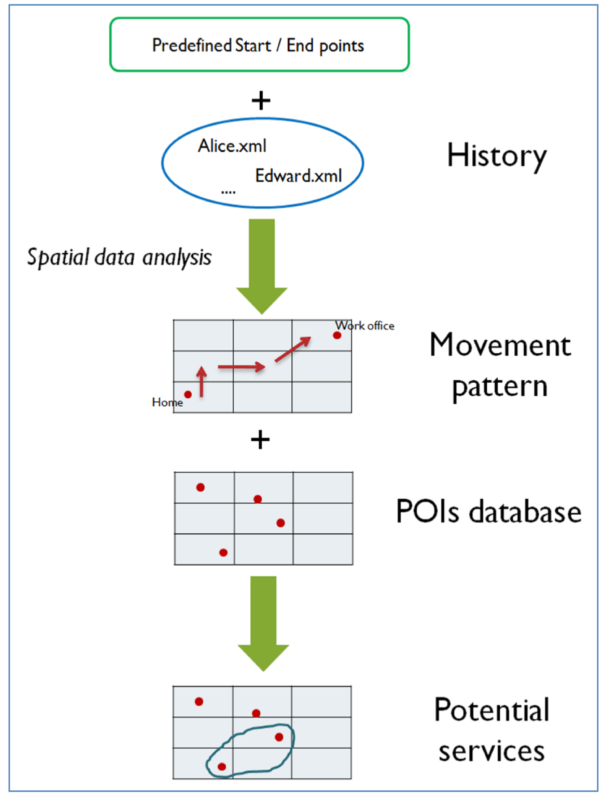
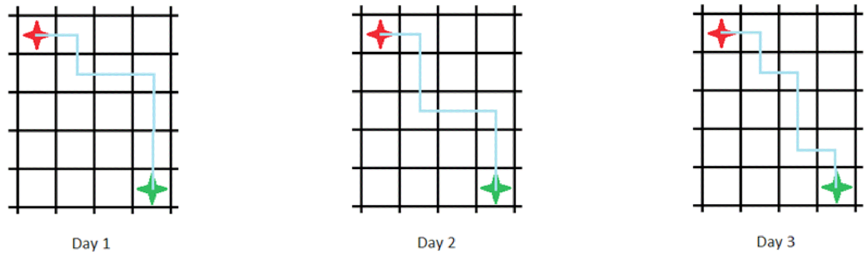


Fig. 3. Application server functionality in general level

has the same dimensions as the table of region sequence, but consists of corresponding time values. Now everything is ready for the extraction of movement pattern. Movement pattern means relations between the present location of a moving object, its past location(s) and time. There are several techniques and algorithms to find out the rules according to which the object moves. These algorithms are dependent on the application domain and information needed in a particular situation [13]. In our current research we were not so interested about relations between location/time values inside a particular sequence. What we needed to know is "What are the common (the most frequent) locations (and corresponding time stamps) of an object during a trip from home to work (from work to home)?". The movement pattern extraction algorithm goes as follows: input is the region sequence table, output is the movement pattern. And for:

1) Movement pattern = day 1 sequence
2) For each region in sequence if (region in movement pattern NOT EQUAL region in next day sequence) region in movement pattern = UNKNOWN
3) Repeat step 2 for next day sequence

After applying this algorithm to the example shown in Fig. 5, we will get the next pattern: 1, 2, 6, UNKNOWN, UNKNOWN, UNKNOWN, 16, 20. To complement this information with time values, we calculate the average time using



Day	Region	Region	Region	Region	Region	Region	Region	Region
1	1	2	6	7	8	12	16	20
2	1	2	6	10	11	12	16	20
3	1	2	6	7	11	15	16	20
...	...	...	...	...	...	...	...	...

Fig. 5. Example of region sequence per day

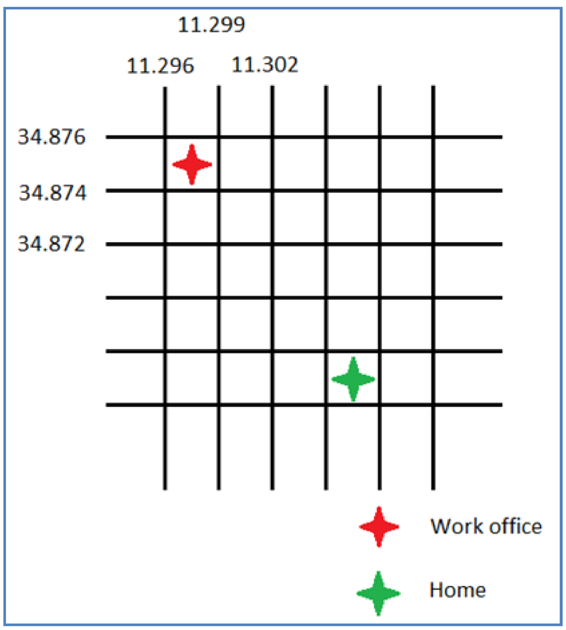


Fig. 4. Spatial regions

the time stamps table. For example, the average time for region 1 could be: (16.45 + 16.50 + 16.51) / 3 = 16.48. After calculating all the average values we will get the next time sequence:16.48, 16.49, 16.50, NOT NEEDED, NOT NEEDED, NOT NEEDED, 16.55, 16.57.

B. Overall architecture of the LBS system with Client, Server and Application Server

In order to get real history data to the application server, a supporting client and servers were implemented. This way real movement data could be collected and the analysis would be based on real-life data. There were also other reasons for having a proper client and server for LBS. The server implementation resembles the application server described previously. It runs on top of Open IMS and was deployed into the Glassfish and Sailfin combination. The relevant features of the server implementation are:

- Provides administration tasks for the whole LBS implementation. It has a web interface for administrative tasks such as manage users and POIs
- Provides a map view presentation of the existing users and POIs
- Can add tasks for a specific user group and can monitor the progress of the task, which is location-dependent and requires actions from the user

The client is mobile device based, and in this case it is implemented with Qt [16] and runs on top of the Maemo 5 [15] platform in Nokia N900 device. The benefits of the N900 device are: built-in IMS support through the SIP protocol and a capable web browser environment. Thus it is easy to connect to IMS services and Google maps data. The main features of the client implementation are:

- User is able to see him/herself, friends and POIs on the map
- User is able to interact with the tasks that are given: one can view, accept and set them as done
- User can communicate via IMS calls and messages with other users

The overall architecture is shown in Fig. 6.

C. Architecture and Implementation Considerations

To get better understanding of the proposed architecture and implementation done, the following aspects were discussed; performance, history data analysis module and need for any further modules. The performance of the Open IMS Core is, as previously mentioned, studied in [5], [4]. Based on these studies Open IMS Core performance was adequate very well for small enterprise usage. This research utilized the same

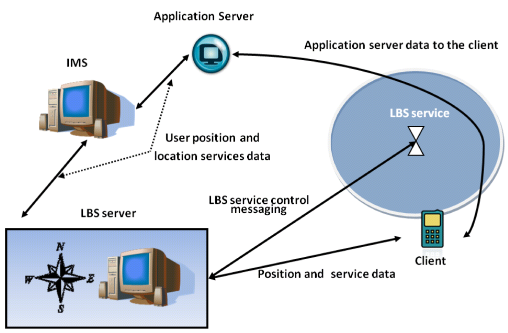


Fig. 6. Overall LBS architecture

hardware configuration as in the previous performance measurements. Therefore, overall hardware and software configuration was considered to be sufficient for a robust environment. In the history analysis module the regional split (see Fig. 4) was chosen to be similar then the average POI radius in LBS service. The service POI area was defined as a radius but for the history pattern analysis the region was divided to rectangle areas. For pedestrian movement analysis this arrangement worked fine. However, it may need to be reconsidered when applying algorithm for other forms of traffic such as car traffic. Performance of the algorithm was not any issue with tens of real users. However the scalability of the current implementation will be part of the future work.The presented history analysis module provided needed information that the extraction algorithm itself works. In addition, other models could be added to further enrich the feature set of the LBS application server. For example algorithm that detects the end points of user's routes could be useful in the same context as the presented model. The same applies for the detection of the user's movement type and the detection or if the user is moving somewhere else then usually (traveling). All these further details could be build as a separate modules of the LBS application server.

V. Conclusion and Future Work

We have presented a location-based service architecture and implementation. The architecture was build on top of an Open IMS implementation and other open protocols and standards like SIP and XML. The spatial data analysis based on the user's movement history was possible and it provided, for example, a way to find services that are presumably located on the route that the user moves along between work and home. Timestamps are added so that it is possible to estimate the time when the user is taking this route next time. All this information is potentially very valuable for advertisement purposes, for example. Our future research continues with fine tuning of the movement pattern algorithm and adding other algorithms to the developed location-based service module. The aim is that the application would automatically find the correct end locations and detect the corresponding transport method user is using so that they would not need to be marked manually.

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Automatic Generation of Efficient Solver for Query-Answering Problems

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***Abstract*—The Query-Answering (QA) problem is a class of the logical problem that is more general than the proof problem and the database searching problem, and can be applied in the semantic web. In this paper, we develop a new technology about how to generate an efficient solver (C program) corresponding to a given QA problem. We expect to generate the specific solver, not the general solver for all kinds of QA problem. The solver is generated based on the bottom-up solution used to update models in the QA problem. We have also developed the technology to suppress the size of the solver to deal with the large-scale QA problem.**

Keywords-Query-Answering problem; bottom-up solution; specific solver; unfold transformation; support set.

I. INTRODUCTION

We know that Description Logic (DL) is a cornerstone of the Semantic Web [1] for its use in the design of ontology, and adding a rule layer on top of the DL-based Web Ontology Language (OWL) is currently the central task in the development of the Semantic Web. In the final analysis, all of these efforts are going to get exactly the right answers of problems, not only judging problems by right or wrong like a proof problem.

The QA problem is a problem whose all answers should be obtained as ground facts. However, adding rules to DL to solve the QA problem in an efficient way is an incomplete research. Moreover, to deal with the logic problem in the real Semantic Web, we should extend the logical expression of the QA problem such as being composed of FOL, DL, Horn clauses, etc. Therefore, the domain of QA problem in our research is larger than the problem defined in the current Semantic Web. An efficient solution corresponding to the QA problem is very important in the development of the Semantic Web. However, a sound solution for QA problems has not yet been satisfactorily established. The process speed of the proposed solution [2] decreases when the size of the problem grows. And it might be impossible to be processed as the size goes beyond a certain size.

In the bottom-up solution proposed by the previous research [2], the data structure of the model is enumeration type, in the pre-model' updating process before generating the representative model, it has to scan the constituted atom from the first atom of the pre-model in the operation like the retrieval, the addition, and the deletion. The calculate speed is not so fast. Moreover, towards a large-scale QA problem the data might explode, and it is possible that the problem cannot be compiled because of the limited memory [3, 4, 5].

In our research, to solve these problems (speed and memory), we generate the specific solver corresponding to the given QA problem by using the specific properties of the problem instead of the general solver for all kinds of QA problem. We think that the entire efficiency improvement could be realized if the processing speed of the specific solver is even fast though it needs to cost the process of generating the solver. Concretely, the specific properties of the problem in this research are clauses of the problem.

Moreover, towards a large-scale QA problem the data size might explode and the computing time takes a lot. Sometimes the problem cannot be solved because of the limited memory. Therefore, we have also developed the technology to suppress the size of the solver to deal with the large-scale QA problem. To generate the efficient solver, simplifying initial clauses obtained from the given QA problem based on the idea of the top down solution named unfold transformation is applied.

The solution of the QA problem proposed in this research is a combination of the top-down solution and the bottom-up solution. And there are four steps in generating the specific solver for the given QA problem.

A. *Clauses of the QA problem are simplified by unfold transformation based on Equivalent Transformation theory*

B. *The set (support set) including all possible ground atoms obtained from clauses are requested in a minimized size.*

C. *A bit array that corresponds to this support set is made.*

D. *Clauses, which are used to update the model, are transferred into "if statement or for loops" in C program as few as possible.*

E. *The final generated C program is the specific solver corresponding to the given QA problem.*

II. QA PROBLEM AND RESEARCH PURPOSE

A. *QA Problem*

In this research, the QA problem is a more general class of the logical problem than the proof problem and the database searching problem, and can be applied in the semantic web.

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The QA problem contains knowledge (Δ) and the query atom (q), in which Δ not only includes the definite clause (atom0 ← atom1, atom2…), which means the clause has only one atom in the left side of the arrow, but also the negative clause (← atom1…), which means there is no atom in the left side, and non-definite clause (atom0, atom1, … ← atom2, atom3…), which means there are more than one atom in the left side. In this research, the QA problem is described by the logical expression and we have to obtain all answers to it. It is possible to describe such problem's answer generally as follows.

$$A = \{g|\Delta|=g \in G,\ g = q\theta,\ \theta \in S\}$$

Here, we take the *Oedipus* [6] problem as an example.

"OE is the child of IO. PO is the child of IO. PO is the child of OE. TH is the child of PO. OE is a patricide. TH is not a patricide. A person's child is a patricide, but his/her grandchild is not a patricide. Who is the person?"

This problem is composed of knowledge (Δ, "OE is the child of IO. …, but his/her grandchild is not a patricide.") and the question (q, "Who is the person?"). The result (A) is "IO". This QA Problem can be rewritten as the following clauses (There are two atoms existing in the left side of the arrow including the question atom, which indicates to wider meaning QA problem).

Knowledge (Δ):

isChild(oe, io) ←. isChild(po, io) ←. pat(oe) ←.
isChild(th, po) ←. isChild(po, oe) ←. ← pat(th).
prob(*x),pat(*b)←isChild(*a, *x),pat(*a),isChild(*b, *a).

Query Atom (q): prob(*x)

Answer (A): {(Δ, q)} ==> {io}.

B. Research Purpose

In this research, we want to develop a new technology about how to generate an efficient solver (C program) corresponding to a given QA problem. Before generating the specific solver (C program) for a given QA problem, not the general solver for all kinds of QA problems, we need to do the memory saving work. Therefore, the present research purpose is shown as follows.

1) *As the size of the QA problem grows, suppressing the memory consumption is important. To generate the efficient solver, the reduction of the size of the QA problem by simplifying initial clauses based on the idea of the top down solution named unfold transformation is applied.*

2) *We propose how to generate the solver corresponding to the given problem by using the unfolded clauses based on the bottom-up solution. Because the final solver is generated by C program, the data structure of the model is the bit-array type, not the usual enumeration type.*

III. APPROACH OF THE RESEARCH

To achieve the research purpose, we do the following four steps (Figure 1) [7, 8].

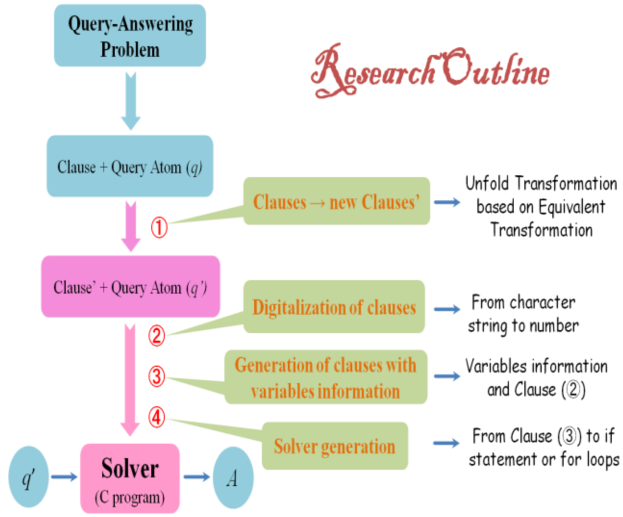

Figure 1. Approach of the research

A. Simplification Processing of Clauses based on the Unfold Transformation

In this research, as the size of the QA problem grows, the memory consumption for updating the model will become very big. Before applying the clauses obtained from the QA problem, we firstly do the unfold transformation based on the equivalent transformation theory, the simplification of clauses is pursued by substituting the definite clause, which will be introduced in Section 4.

B. Digitalization of Clauses

As introduced in the research purpose, we want to generate the specific solver (C program) for each QA problem, it is necessary to make the mechanism about how to convert clauses, which are the result of the unfold transformation in step A, into the corresponding C program. In order to transfer clauses to C program, the algorithm about how to convert the atom, the basic element consisting of the clause, into the index number of the bit array used in C program is very important.

C. Generation of Clauses with Variables Information

When we convert each clause into C program (if statement or for loops) by using the result of process B, in order to consolidate the size of the generated C program, we want to generate for loops as far as we can. For this reason, we need to firstly obtain the variables information of each clause (the information of clauses without variables is empty, and this kind of clause will be transferred into if statement), then based on this variables information, the clause with variables will be converted to for loops.

D. Solver Generation

Based on atoms-index's corresponding algorithm obtained in step B and clauses with variables information generated by step C, all clauses will be transferred into the corresponding if statement or for loops in C program. As a result, the solver corresponding to the given QA problem is generated. Finally, the answer of the given QA problem will be got by executing the generated solver.

In Section 4, we will introduce the simplification processing of clauses based on the unfold transformation. In Section 5, we will explain the processing of digitalization of clauses. In Section 6, generation of clauses with condition and automatic generation of solver will be introduced.

IV. SIMPLIFICATION PROCESSING OF CLAUSES BASED ON THE UNFOLD TRANSFORMATION

To apply a large-scale QA problem, the size and the complexity of clauses requested from the QA problem can be reduced by the unfold transformation based on the Equivalent Transformation theory [9, 10, 11].

In this research, the unfold transformation is started by deciding the target predicate of the atom, and other atoms (with different predicates) exist in the same clause would be substituted by definite clauses. The definite clause will be finally removed after being applied. In this way, the simplification of clauses obtained from the given QA problem could be accomplished.

Here, there is an example showing the unfold transformation for a non-definite clause (e.g.:← (Wolf *A) (Fox *B) (eat *A *B).), which is done by using two ground clauses (e.g.: (Wolf wolf) ←. (Fox fox) ←.) (Figure 2).

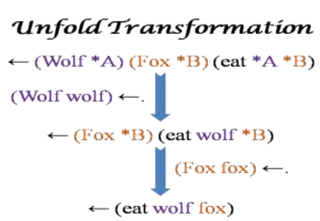

Figure 2. Process of the unfold transformation

V. DIGITALIZATION AND LIMITATION OF CLAUSES

A. Generation of the Support Set

In the process of generating the solver from clauses, we need to first decide the set of all possible atoms which constitute clauses. In this research, the set of all atoms are called the support set. The digitalization of clauses can be made by deciding the support set. The support set requesting process has been divided into three steps (A, B, C) shown in Figure 3. Clauses requested by the unfold transformation is input, and the support set corresponding to the problem is output. In the approximate processing of A, the atom corresponding to the problem is roughly requested. The smaller and more accurate the support set requested, the higher calculation cost for generating the support set is. Based on the approximate idea, we do not look for the most accurate support set, but within an approximate range, search a little wide-ranging support set efficiently, and finally generate the program used to request the support set. The support set is requested by executing the program.

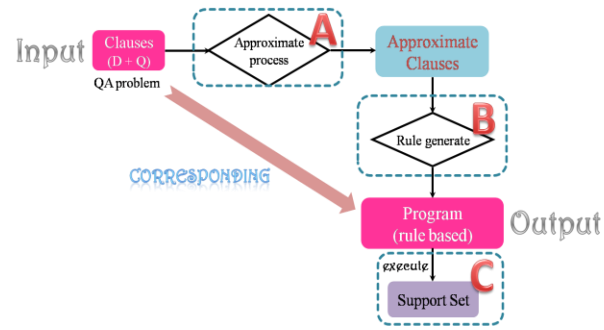

Figure 3. Process of generating the support set

*1) Approximate Process (**A**)*

Clauses in the Oedipus problem:

(isChild oe io) ←.	(isChild po io) ←.	***Input***
(isChild po oe) ←.	(isChild th po) ←.	
(pat oe) ←.	← (pat th).	
(prob *x), (pat *b) ← (isChild *a *x), (pat *a), (isChild *b *a).		

New generated clauses:

(isChild1 oe) ←.	(isChild2 io) ←.	***Output***
(isChild1 po) ←.	(isChild2 oe) ←.	
(isChild1 th) ←.	(isChild2 po) ←.	
(pat oe) ←.	← (pat th).	
(prob *x) ← (isChild1 *a), (isChild2 *x), (pat *a), (isChild1 *b), (isChild2 *a).		
(pat *b) ← (isChild1 *a), (isChild2 *x), (pat *a), (isChild1 *b), (isChild2 *a).		

*2) Generation of the Approximate Clauses (**B**)*

1: (isChild1 oe) ←.	2: (isChild2 io) ←.	***Input***
3: (isChild1 po) ←.	4: (isChild2 oe) ←.	
5: (isChild1 th) ←.	6: (isChild2 po) ←.	
7: (pat oe) ←.	8: ← (pat th).	
9: (prob *x) ← (isChild1 *a), (isChild2 *x), (pat *a), (isChild1 *b), (isChild2 *a).		
10: (pat *b) ← (isChild1 *a), (isChild2 *x), (pat *a), (isChild1 *b), (isChild2 *a).		

Output:

(whole *isChild1 *isChild2 *pat *prob *x1 *x2 *x3 *x4), {(addelem *isChild1 (oe) on *isChild1new)}→(whole *isChild1new *isChild2 *pat *prob *x1 *x2 *x3 *x4). ➔1

…

(whole *isChild1 *isChild2 *pat *prob *x1 *x2 *x3 *x4), {(inter (*pat *isChild2 *isChild1) *mid1), (inter (*isChild2) *mid2), (inter (*isChild1) *mid3), (cons *mid1), (cons *mid2),

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(cons *mid3), (addelem *pat *mid3 on *patnew)}→(whole *isChild1 *isChild2 *patnew *prob *x1 *x2 *x3 *x4). ➔10

*3) Generation of the Support Set by Executing the Rule Based Program (**C**): By applying rules, each predicate set (isChild1, isChild2, pat, prob) that constituted the support set has been expanded (Figure 4).*

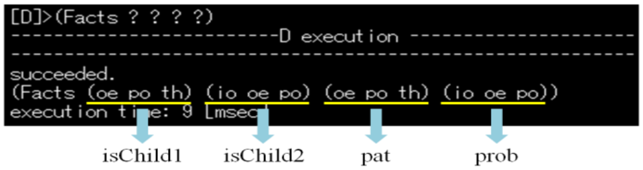

Figure 4. Generating of each predicate set

The support set is composed by combining these predicate sets, and the support set of the Oedipus problem is shown as follows.

{(isChild oe io), (isChild oe oe), (isChild oe po), (isChild po io), (isChild po oe), (isChild po po), (isChild th io), (isChild th oe), (isChild th po), (pat oe), (pat po), (pat th), (prob io), (prob oe), (prob po)}

Expression in clauses (Figure 5) :

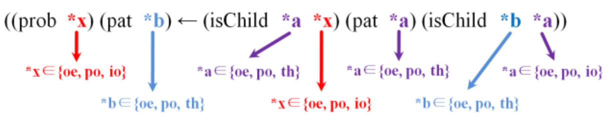

Figure 5. Support set expressed in the clause

B. Digitalization of the Support Set

First of all, for all atoms in the support set, the symbol set that includes all the symbol values which can substitute the argument are requested. The order of symbols in the symbol set is decided, and each symbol is converted into the natural number. Second, all atoms in the support set are sorted in alphabetical order of the argument by the order of this symbol set (Figure 6)。

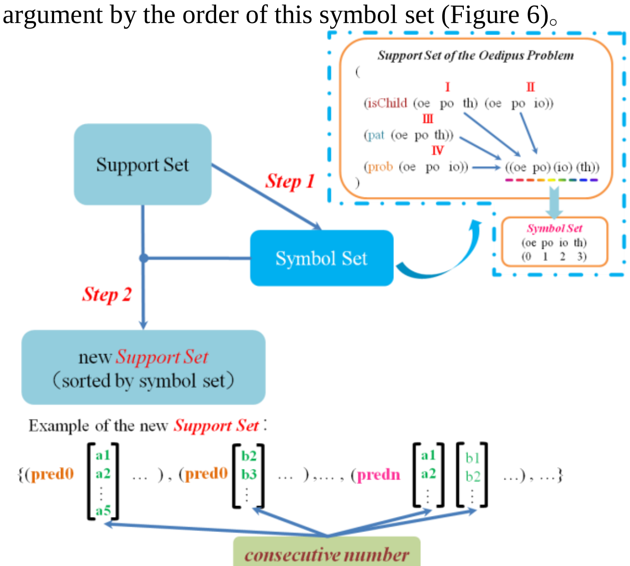

Figure 6. Digitalization of the support set

Symbol Set:

(oe po io th) → (0 1 2 3)

Input:

{(isChild oe io), (isChild oe oe), (isChild oe po), (isChild po io), (isChild po oe), (isChild po po), (isChild th io), (isChild th oe), (isChild th po), (pat oe), (pat po), (pat th), (prob io), (prob oe), (prob po)}

Output:

{(isChild 0 0), (isChild 0 1), (isChild 0 2), (isChild 1 0), (isChild 1 1), (isChild 1 2), (isChild 3 0), (isChild 3 1), (isChild 3 2), (pat 0), (pat 1), (pat 3), (prob 0), (prob 1), (prob 2)}

Expression in clauses (Figure 7) :

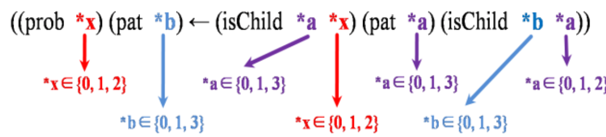

Figure 7. Digitalization of support set expressed in the clause

C. Digitalization of Clauses

In this research, we will finally convert each clause into the corresponding the C program (if/for loops), it is necessary to make the mechanism about how to convert the atom into the address of the bit array. Here, the atom-address calculating function, used to make all basic atoms in the support set correspond to the address of the bit array, is made. Based on the function, it is possible to access the address which corresponds to the atom quickly in the updating process.

In the sorted support set (from the result of B), the argument of atoms with a consecutive value is brought together. Then, the address function corresponding to this kind of atom is generated. It generates completely different address function for discontinuous argument value in spite of having the same predicate (Figure 8).

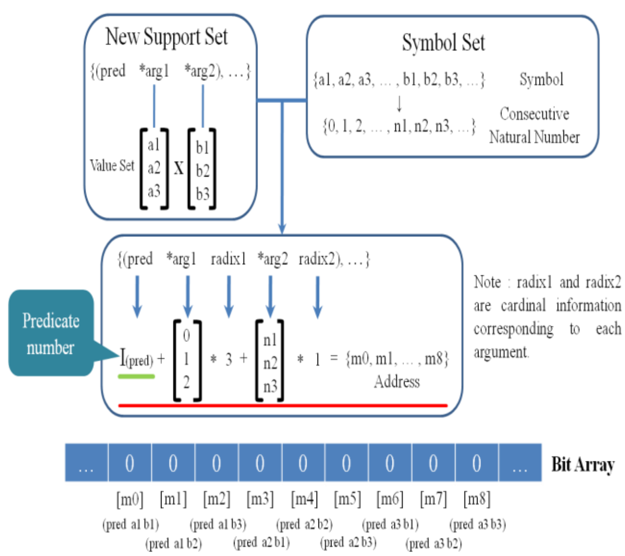

Figure 8. Digitalization of clauses

The address "PAdr(s1,...,sn)" of all basic atoms "Pred(s1,...,sn)" (There are n arguments) can be decided by the introduced algorithm. It is requested from the predicate number "I(pred)", and the relative address "Rel(s1,...,sn)" of the predicate "Pred", based on the following formula.

$$\mathrm{PAdr}(s1,...,sn) = \mathrm{I}(pred) + \mathrm{Rel}(s1,...,sn) \quad (1)$$

Relative address "Rel(s1,...,sn)" is requested from the 1st argument value "Sym(s)" and its cardinal "R(pred,s)" of the last argument by the following formula.

$$\mathrm{Rel}(s1,\ldots,sn)=\mathrm{Sym}(s1)*R(pred,s1)+\ldots+\mathrm{Sym}(sn)*R(pred,sn) \quad (2)$$

Cardinal "R(p,i)" is requested by using symbol number "Ssym(p, k)". For instance, based on the support set how many symbols can substitute the back arguments.

$$R(p,i) = \prod_{k=i+1}^{n} S_{sym}(p,k)$$

By applying the above-mentioned function, atoms in the clause can be conver into address of the bit-array. Expression in clauses (Figure 9) :

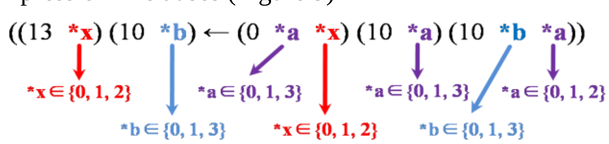

Figure 9. Example of digitalization of clauses

D. Limitation of Clauses

Based on the result of clauses requested in process C, the atom with the same variable is extracted, and the intersection calculation of the value set of the argument is done. The result is shown in Figure 10.

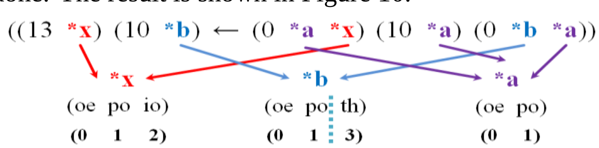

Figure 10. Example of Limitation of clauses

As shown in Figure 10, because the value set corresponding to the variable (*b) is not a consecutive value, the clause with condition is generated based on the value set corresponding to the argument (limitation of clauses). The condition part of ground clauses is empty.

eg.: {(*x 0 2) (*b 0 1) (*a 0 1)}

((13 *x) (10 *b) ← (0 *a *x) (10 *a) (0 *b *a))

{(*x 0 2) (*b 3) (*a 0 1)}

((13 *x) (10 *b) ← (0 *a *x) (10 *a) (0 *b *a))

VI. SOLVER GENERATION

A. Solver Generation

In this research, the solver is composed by three parts, which are main function definition, bit-array declaration and if/for loops. We input the query (*q*), the solver will output the corresponding answer (*A*). The generation of if/for loops is requested by using the conditional clause generated in the process D of Section 5. Here, we will introduce the method about how to convert various conditional clauses into if/for loops of C program.

1) From Ground Clauses to if loops

The transmission from a ground clause to an if loops in C program is shown as follows (Figure 11).

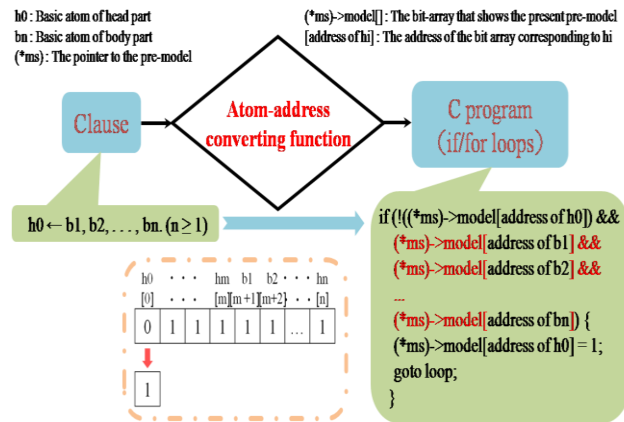

Figure 11. From ground clauses to if loops

2) From Not Ground Clause to for loops

A clause that contains variables will first substitute all variables for possible symbols, and then generate new ground clauses. The patterns of the symbol those can substitute the variable increase while the size of the QA problem grows. Therefore, the size of the generated C program will grow, sometimes it will become impossible to compile.

In this research, because the value set corresponding to the variable of each atom in the clause is obtained based on the support set, and been changed into natural numbers (starting from 0) based on the symbol set, the consecutive value will be expressed by one "for loops". The solver result of this "for loops" is already shown in Figure 12.

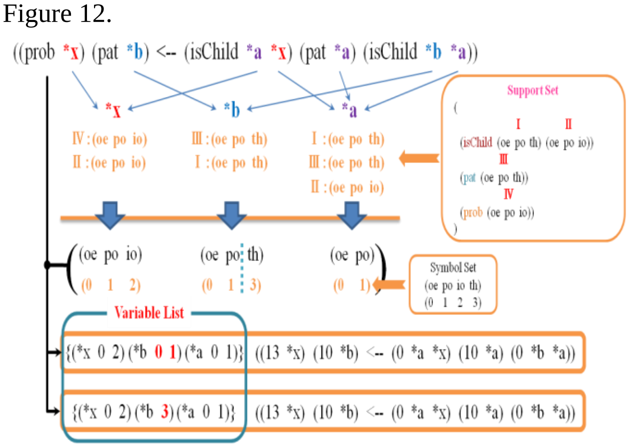

Figure 12. Consolidate processing by "for loops"

3) Example of Solver Generation

Here is an example of generating the solver corresponding to the Oedipus problem. Chiefly three parts is shown (Figure 13)。

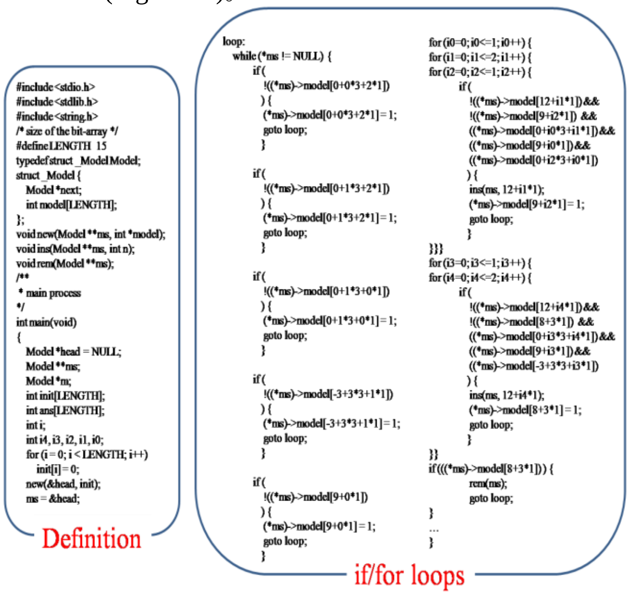

Figure 13. Example of solver generation

VII. SUMMARY

In this paper, we talk about generating a special solver (C program) for a given QA problem. The solver is composed by main function definition, bit-array declaration, and if/for loops. We input the query (*q*), the solver will output the corresponding answer. Because models generated by the solver are based on the bit calculation, the speed is absolutely fast. We also develop the technology for suppressing initial clauses by the equivalent transformation process.

As future work, we think the answer of the QA problem can be more quickly obtained in a smaller search space by simplifying process of clauses obtained from the QA problem before clauses being used to update the pre-model.

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Context-Oriented Knowledge Management for Intelligent Museum Visitors Support

Ontological approach

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***Abstract*—The paper presents an approach to context-oriented knowledge management for supporting visitors in museum smart environment. Museum smart environment is a private network consisting of museum rooms equipped with devices that allow identifying visitor location, his/her movement and providing information about exhibits. Visitor has a mobile device, which stores visitor's preferences and communicates with the museum smart environment. The mobile device calculates museum visiting plan for the visitor based on the visitor's preferences, amount of visitors in each museum room at the current time and other context information (closed exhibits, reconstructions, seasonal exhibitions and other) acquired from Internet and intranet services.**

Keywords-knowledge management; ontologies; Internet services; user profiles; smart environment.

I. INTRODUCTION

Recently, the tourist business has become more and more popular. People travel around the world and visit museums and other places of interests. This fact increases popularity of the museums. But overwhelming majority of museums has limited space for visitors causing accumulation of visitors and increasing waiting time for them.

In this regard an approach is needed, which allows assisting visitors (using their mobile device), planning their excursion plans depending of the context information about the current situation in the museum (amount of visitors around exhibits, closed exhibits, reconstructions and other) and visitor's preferences.

Smart environment is an aggregation of mobile and stationary devices, which can interact with each other and use pertinent services regardless of their physical location. Such technology allows seamless integration with other system, services, and program modules.

Research efforts in the area of the smart environment have become very popular recently. Such topics of research as smart home, smart car, etc. are widely discussed in the modern computer science. In such systems all elements have to interact and coordinate their behavior without any user intervention.

Modern tendencies of information and telecommunication technologies require development of stable and reliable infrastructures to extract and keep different kinds of information and knowledge from various members of the smart environment. The smart environment assumes more than one device that uses common resources and services. One of the most appropriate approaches to implement such infrastructure is applying knowledge management systems.

A museum can be considered as a smart environment (Figure 1) where each exhibit or a group of exhibits is represented by a stationary device of the smart environment. Each device can interact with other stationary devices representing exhibits and with visitors' mobile devices. Visitor's mobile device interacts with other smart environment devices and provides for the visitor an acceptable excursion plans inside the museum based on the museum context (amount of visitors around exhibits, closed exhibits, reconstructions and other) and visitor preferences.

Visitor's mobile device also can provide textual, graphical, video and audio information for the user in his/her language.

The following scenario can be considered. A tourist arrives in Paris. He/she is going to attend Louvre Museum and he/she has the intelligent museum visitors support system on his/her mobile device. The tourist adds his/her interests to his/her user profile within the intelligent museum visitors support system, for example it can be Leonardo da Vinci paintings. When the visitor enters the Louvre the mobile device guides him/her based on interests defined him/her earlier and museum context. For example, if at the moment there are no too much people in the room where the Mona Lisa painting is shown the mobile device proposes to start from this exposition because usually this room is overcrowded. After that he/she can see Madonna of the Rocks, Lady with an Ermine, Benois Madonna and other. When the tourist approaches the painting or another exhibition he/she gets audio, textual and video information about it from appropriate service through the Internet or intranet.

The rest of the paper is structured as follows. Section II presents an overview of mobile museum guides systems. Section III introduces developed approach to knowledge management in museum smart environment. The museum ontology is shown in Section IV. Information model of museum visitor profile is given in Section V. The case study can be found in Section VI. Main results and drawings are summarized in Conclusion.

II. Mobile Museum Guides Systems Overview

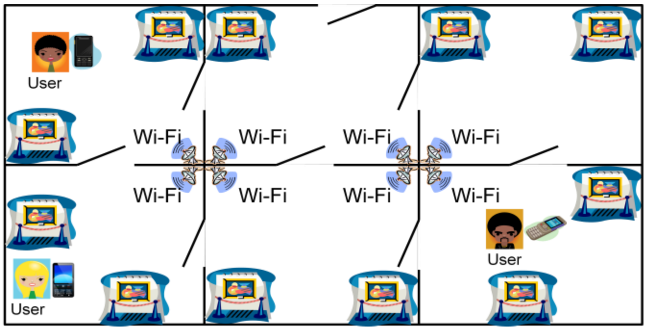

Figure 1. Museum smart environment infrastructure

spaces (i.e., museums, art exhibitions). The system is meant to provide the visitor with personalized information about the relevant artworks nearby. The information is mainly audio in order to let the user enjoy the artworks rather than interacting with the tool.

Bohnert et al. [2] describes a system for providing a visitor with the challenge of selecting the interesting exhibits to view within the available time. It includes the recommendation and personalization process, i.e., the prediction of a visitor's interests and locations in a museum on the basis of observed behavior.

Kuflik et al. [3] describes an approach for supporting users in their ongoing museum experience, by modeling the visitors, "remembering" their history and recommending a plan for future visits. This approach identifies some of the technical challenges for such personalization, in terms of the user modeling, ontologies, infrastructure and generation of personalized content.

Project CRUMPET [4] has realized a personalized, location-aware tourism service, implemented as a multi-agent system with a concept of service mediation and interaction facilitation. It has had two main objectives: to implement and trial tourism-related value-added services for

The main research activities of HIPS project [1] are development of approach for navigating artistic physical nomadic users across mobile and fixed networks, and to evaluate agent technology in terms of user-acceptability, performance and best-practice as a suitable approach for fast creation of robust, scalable, seamlessly accessible nomadic services.

These systems don't take into account information about current situation. Proposed approach allows monitoring the current situation in the museum and uses it for visitor assistance.

III. Knowledge Management in Museum Smart Environment

The approach presented in the paper relies on the ontological knowledge representation. The conceptual model of the proposed ontology-based knowledge management is based on the earlier developed ideas of knowledge logistics [5]. In this work the ontology is used to describe knowledge in the smart environment.

The architecture of the approach is presented in Figure 3. Mobile and stationary devices interact through the smart environment. When the visitor registers in the system, his/her mobile device creates the visitor profile that allows specifying and complement visitor requirements in the smart environment and personifying the information and knowledge flow from the system to the user. Every time the visitor appears in the smart environment the mobile device shares information from the visitor profile with other devices.

The proposed ontological approach to context-oriented knowledge management in the museum smart environment is presented in Figure 2.

The following scenario for mobile and stationary device interaction support in the smart environment is considered.

A visitor enters a museum. His/her mobile device finds the museum smart environment using Wi-Fi connection.

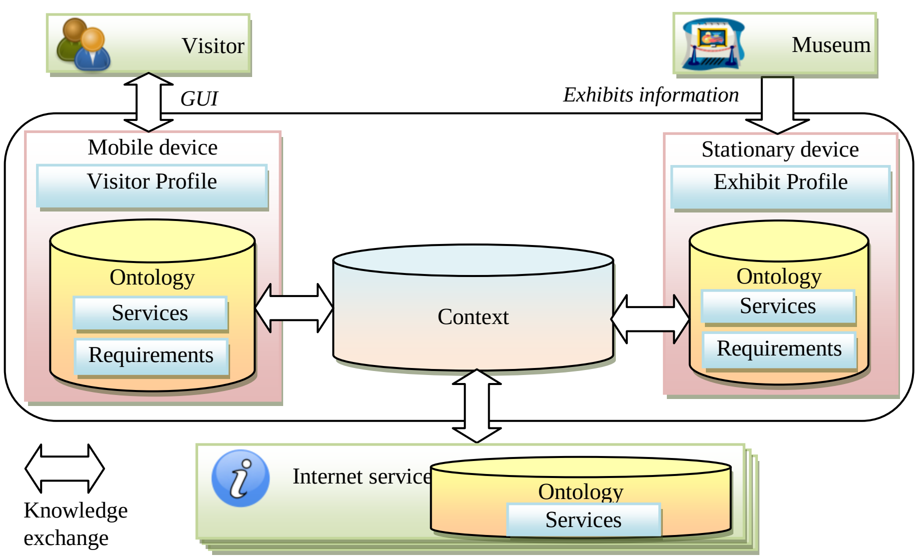

Figure 2. Ontological approach to context-oriented knowledge management in the museum smart environment

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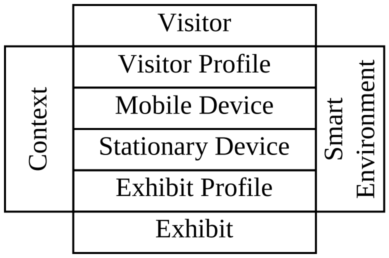

Figure 3. Architecture of the proposed approach

The mobile device informs the visitor that the smart environment has been found and based on the visitor's preferences and the current situation in the museum (the context) an acceptable path for visiting museum rooms is built. The context is formed based on the interaction process between the visitor's mobile device and other mobile and stationary devices through the smart environment. The context is the description of the visitor's task in terms of the ontology taking into consideration the current situation in the museum. The ontology in the knowledge management system describes the main terms used for the museum smart environment description and relationships between them.

The mobile device of the visitor provides services for sharing information with other devices of the smart environment. For example, it shares the visitor's location, visitor's personal information, preferences and preferred exhibits to see.

For these purposes, the visitor profile has to contain personal information about the visitor, domain specific information (e.g., preferred exhibits to see), information that describes user preferences, feedback information and history that contains previous user activities in the system.

IV. MUSEUM ONTOLOGY

There are several different ontologies of intelligent museum visitors support systems has been analyzed [3][7][8]. The upper-level museum ontology based on these ontologies is presented in Figure 4.

It consists of Device entity divided into Infrastructure Device, Mobile Device, and Stationary Device entities. Infrastructure Devices provide infrastructure for the Smart Environment (e.g., Wi-Fi, hot spots).

Figure 4. Upper level of the museum ontology

Figure 5. Device entity of the museum ontology

Mobile Devices are used by museum visitors and allow visitors to connect to Smart Environment, to get assistance, and information about exhibits. Stationary Devices represent exhibits in Smart Environment (provide to Smart Environment all necessary information about exhibits).

A more detailed description of Device entity is presented in Figure 5. Every Device has location information (associative relationship to Location entity). Infrastructure devices have attribute Type (e.g., Wi-Fi or Bluetooth).

Mobile devices have attributes which characterize hardware and software information and rights in the museum smart environment. Every Mobile Device entity has associative relationship to Visitor entity, which means that this visitor uses that mobile device. Stationary device has associative relationship with Exhibition, which means that this exhibition is represented by this stationary device in smart environment.

A more detailed description of Excursion entity is presented in Figure 6. Every excursion is described by the following attributes: description, duration, name, and start time. Excursion entity has associative relationships to Exhibition, Visitor, and Museum entities.

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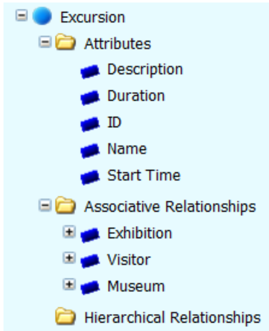

Figure 6. Excursion entity of the museum ontology

Description of Location entity is presented in Figure 7. It includes the following attributes: Accessibility for Disabled Visitors which means that this location is equipped to assist disabled visitors, and Description which describes the location. Class compatibility relation with Disabled Visitor entity determines if this Location is compatible or incompatible with the Disabled Visitor, based on Type of Disability (see Figure 12).

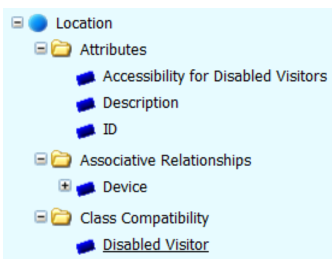

Figure 7. Location entity of the museum ontology

Figure 8 represents different roles for a museum visitor. Role entity has associative relation to Visitor entity. The following roles can be mentioned: researcher, school teacher, regular visitor, tourist, and other. Additional information for the every role can be used by the system. E.g., researcher will stay in museum for a long time, school teacher attend the museum with children, etc.

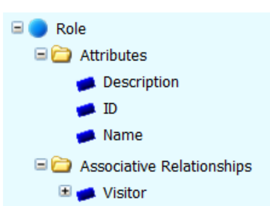

Figure 8. Role entity of the museum ontology

More detailed description of Museum entity is presented in Figure 9. This entity has attribute Name (the name of the museum), associative relations to Excursion and Smart Environment entities, and hierarchical (part-of) relations to Exhibition entity.

Description of Exhibition entity is presented in Figure 10. It is characterized by the following attributes: capacity, description, name, style, type.

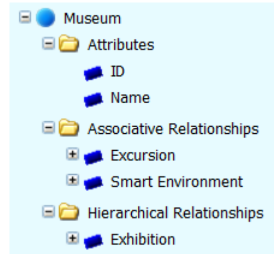

Figure 9. Museum entity of the museum ontology

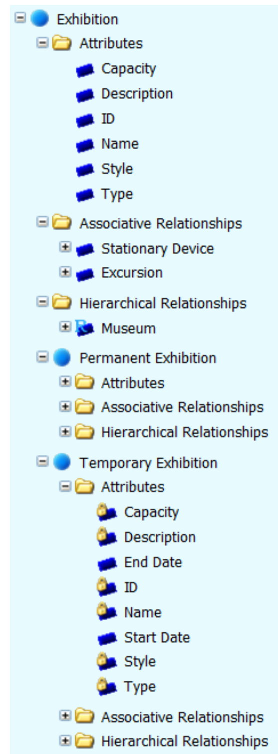

Figure 10. Exhibition entity of the museum ontology

Associative relation to Stationary Device entity means that this exhibition is represented by this stationary device in the smart space. Relation to the Excursion entity means that this exhibition is included into the excursion. Hierarchical relation to Museum entity means that this exhibition is part-of the Museum. Temporary exhibitions have additional attributes Start Date and End Date, which determine start and end date of exhibition.

Figure 11 represents Smart Environment entity. It has associative relation to Museum entity and hierarchical relation to Device entity.

Description of the Visitor entity is presented in Figure 12. The following attributes describe the visitor: name, gender, date of birth, e-mail, list of languages for communication, phone number, and position. Associative relation to Role entity means that the visitor has a role. Relation to Excursion entity determines that the visitor is connected to the excursion. Relation to Mobile Device entity determines that the visitor uses the mobile device to communicate with museum smart environment.

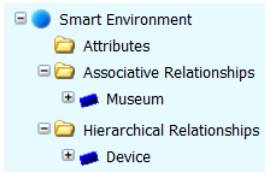

Figure 11. Smart Environment entity of the museum ontology

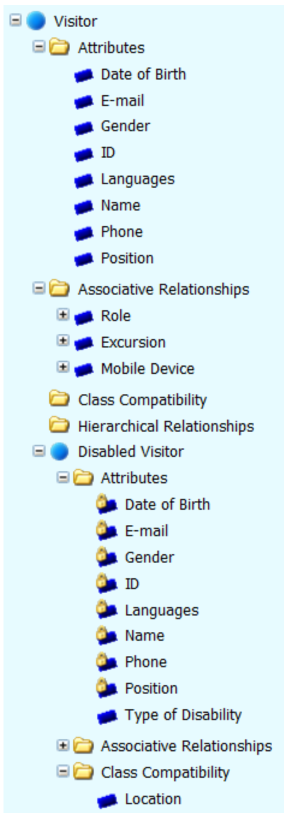

Figure 12. Visitor entity of the museum ontology

V. VISITOR PROFILE

Most of user profile models include such information as: first name, last name, gender, date of birth, languages, and contact information and user position. This information is also important for intelligent museum visitor's support. It is stored in the "Personal Information" module (Figure 13).

Museums visitors can have different roles (e.g., individual visitor, family, group of schoolchildren and other). Intelligent museum visitor's support system can take into account this information for building the plan of the excursion. Some parts of the visitor profile can be hidden from other visitors (for example if the user would like to attend museum anonymously). For this purpose the visitor has to choose which information can be accessible to other devices. It is needed to provide the knowledge management system with information about visitor's hardware and software capabilities, because based on this information the system suggests the visitor which types of exhibit descriptions (audio, video, textual) he/she can use. This information is stored in the "System Information" module (Figure 13).

Since the knowledge management system is context-oriented, it is necessary to determine the location of the visitor in time. For this purpose the module "Context Information" is proposed.

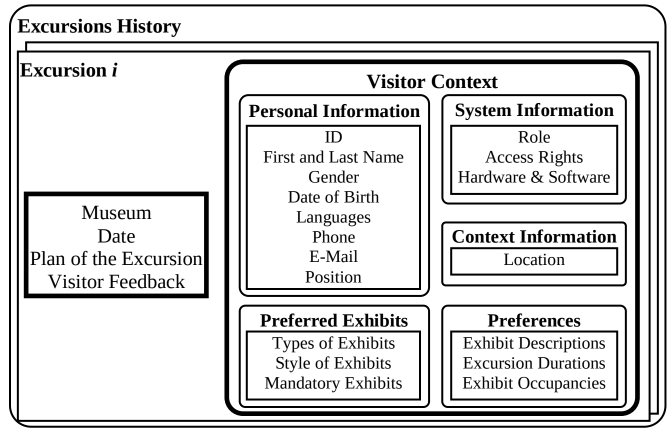

Figure 13. Model of museum visitor profile in intelligent museum visitor's support system

For building an acceptable plan of excursion the system needs information about exhibits preferred by the visitor: types of exhibits (paintings, ancient items and other), styles of exhibits (modern, impressionism and other), and mandatory exhibits which visitor have to see (e.g., Venus de Milo, Mona Lisa in the Louvre Museum). Also visitor profile has to keep the information about preferable types of exhibit description (audio, video or/and textual), excursion duration (how much time the visitor can spend at this museum), exhibit occupancies (in case of high occupancy of an exhibit the visitor might prefer to skip this exhibit or to try to see it later).

For the purpose of keeping the history of interaction between the visitor's device and the museum smart environment, and its further analysis all excursions of this visitor, including the museum name, date, plan of the excursion, visitor's feedback about the excursion, and user context at the moment of excursion are stored in visitor profile. Based on this information, the visitor's preferences and preferred exhibits can be semi-automatically identified using ontology-based clustering mechanisms described in [6].

VI. CASE STUDY

The intelligent museum visitor's support system has been implemented based on proposed approach and ontology. Maemo 5 OS – based device (Nokia N900) and Python language are used for implementation.

An open source software platform [9] that aims to provide a "Semantic Web" information sharing infrastructure between software entities and devices is used for system implementation. In this platform the ontology is represented via RDF triples. Communication between software entities is developed via Smart Space Access Protocol (SSAP) [9].

Different entities of the system are interacting with each other through the smart environment using proposed in Section IV ontology. Every device has a part of this ontology and after connecting to smart environment it shares own ontology part to the smart environment.

The system has been partly implemented in Gymnasium of Karl May History Museum [10] which located in the same building with St. Petersburg Institute for Informatics and Automation RAS.

When the visitor enters the museum he/she can connect to the museum intranet network and download appropriate software for getting intelligent museum visitors support. Installation of this software takes few minutes depending on operating system of mobile device (at the moment only Maemo 5 OS is supported). When the visitor runs the system first time the profile has to be completed. This procedure takes not more than 10 minutes. The visitor can fill the profile or can use a default profile. In case of default profile the system can not propose preferred exhibitions to the visitor.

Response time of the Internet services depends on the Internet connection speed in the museum, number of people connected to the network, and workload of the services. Average response time should not exceed a one second.

On the three top screenshots (Figure 14) the visitor profile is presented. According to the model of museum visitor profile in intelligent museum visitor's support system it consists of personal information, system information, visitor preferences (preferred exhibits and other preferences). The fourth screenshot shows an exhibit description acquired from an external Internet service (e.g., Wikipedia).

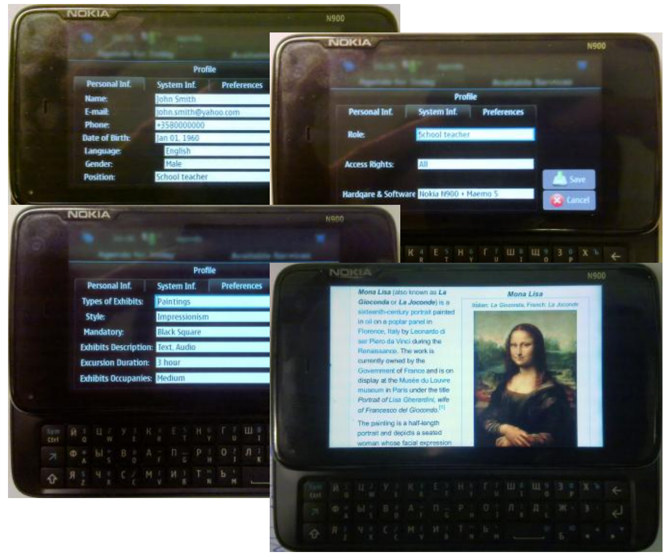

Figure 14. demonstrates some screenshots of the prototype.

VII. CONCLUSION

The paper presents an innovative approach to context-oriented knowledge management for supporting visitors in museum smart environment. This approach allows different devices in the museum smart environment to interact with each other for the purpose of guide visitor in the area of museum. User profiles allow keeping important information about the user and using it in the smart environment.

Since there is no the centralized server in the proposed system, the performance is affected by the number of visitors indirectly. If there are many visitors using the system, the bottleneck of the system performance will be network capacity and Internet services.

ACKNOWLEDGMENT

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Web Service Enhanced Home Energy Management System

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***Abstract*—In the last years energy consumption in the home environments has increased considerably. There is an interest to provide users with means to reduce their energy consumption. Introducing a Home Energy Management System (HEMS) into user residences might provide the necessary tools to reduce and optimize the energy consumption in home environments. The main element of HEMS is the home gateway. In this paper, a home gateway suitable for HEMS is presented. The home gateway proposed uses rules to implement the HEMS. The rules can be downloaded through web services from a rule server. Furthermore, web services are used to provide modularity to the home gateway by enabling the deployment of the different logical components into different devices.**

***Keywords*-Home gateway, Home Energy Management Systems (HEMS), ontology, OSGi, web-services**

I. Introduction

Despite the fact that home appliances, such as washing machines and fridges, have become more energy efficient, electricity consumption in users' residences has increased 30% over the last 30 years [1]. This growth in consumption is due to the fact that the number of appliances that can be found in households is also increasing. According to the International Energy Agency (IEA), European electricity consumption is going to increase 1.4% per year up to 2030, unless countermeasures are taken [2]. In order to avoid this, users should become more conscious of their energy consumption and try to reduce it and avoid demand peaks.

The appliances found in users' premises are usually manufactured by different producers and may use different communication technologies which can lead to interoperability issues between devices. Therefore, the main challenge in home networks is the variety of technologies, providing different communication methods, as well as the diversity of producers, providing different types of devices and services.

A Home Energy Management System (HEMS) [3] is a system from which the user can control the devices in the home network through an Graphical User Interface (GUI) and apply energy management strategies to reduce and optimize their consumption. The herein proposed home gateway is technology and device type independent, in order to offer a common "pluggable" platform to different devices in the home network, which makes them interoperable at the service level.

The home gateway is developed using OSGi and offers a GUI from where devices can be controlled and queried. A detailed description of the home gateway developed can be found in [4]. Furthermore, the home gateway accesses a knowledge base data repository from where the capabilities of the devices can be obtained. This knowledge base data repository is implemented by an ontology, where the home devices are classified according to their functionalities and capabilities. Moreover, the ontology also contains a classification of the different commands and states a home device can support. Energy management strategies can be performed by applying a set of rules, which are based on the energy consumption of the home devices, information from the electrical grid and users' preferences. More information about the knowledge base data repository and the energy management method used can be found in [5].

The home gateway presented in [4] and [5] has been extended to include a web services interface allowing modularization of the system. In this paper, the web services incorporated into the HEMS and the required changes to the previous developed home gateway will be described. The following sections will describe in detail the developed home gateway explaining each logical component and the interaction between them. The main motivation was to create a simple home gateway which would be easily scalable and that had the necessary capabilities to create a home energy management system: offer control of devices through a GUI, run rules to apply energy management strategies and interoperability between devices even if they belong to different subnetworks. Furthermore, as our home gateway is developed using OSGi, new bundles offering other OSGi services or web services, such as communication with the electricity providers, can be easily incorporated.

The rest of this article is organized as follows: Section II introduces relevant related works. Section III briefly describes the OSGi framework. Next section, Section IV, provides an overview of the web services, while Section V describes the home gateway architecture. Section VI describes the interaction between bundles and web services, within the chosen OSGi architecture. Finally, Section VIII provides final remarks and conclusions.

II. Related Work

The implemented home gateway is based on Open Service Gateway Initiative (OSGi) [6] using Equinox which is an Eclipse project that provides a certified implementation of the OSGi. To create the knowledge base data repository for the home gateway implemented, OWL ontology [7] is used.

Protégé-OWL API 3.4.4 [8] provides the necessary mechanisms to manipulated ontologies from a JAVA application.

An example of a home gateway using OSGi and ontologies is Domotic OSGi Gateway (DOG) [9] by Politecnico di Torino. DogOnt [10] ontology is used in the home gateway developed in [9] and is reused as well in the implementation herein described. This ontology provides a good classification of the devices that can be found in a home environment. However, this article will not describe this ontology as an extensive description can be found in [10]. The main difference between DOG and the home gateway herein presented is the fact that DOG is focused on domotics while the home gateway herein is mainly concerned with energy management. The user can use it to define their own energy management system by creating, modifying and deleting rules, which may reduce the total electricity consumption.

In [11], another example of a home gateway developed using OSGi can be found. As with DOG, no energy management strategies are applied, since the necessary tools, for example a rule engine, are not provided.

Furthermore, the authors in [12] describe the IntelliDomo's learning model which is an ontology-based system, able to control a home automation system and to learn users periodic patterns, when using home appliances.

In [13] a HEMS has been implemented to reduce stand-by consumption by setting a power line network. Similarly, in [14], a HEMS implementation using ZigBee and infrared communication to reduce stand-by consumption of power outlets and lights is presented. THE Hems proposed in this paper has two main advantages over [13] and [14]: 1- the energy management strategy can help reduce the consumption of all users' appliances and not only stand-by consumption; 2- the HEMS may communicate using different technologies and not only power line communication or ZigBee.

A HEMS is implemented in [15] to monitor the consumption of appliances. The implemented system does not offer any tools o apply energy management strategies unlike the HEMS presented in this paper.

In this article the home gateway presented in [4] and [5] has been extended to include web services. This web services are used to enable the deployment of the different logical components of the home gateway into separate devices. Furthermore, web services are also used to download rules for the energy management system included in the home gateway. This supposes an advantages to the users, the rules can be downloaded from a rule server instead of being introduced to the system by writing them.

III. OSGi

The OSGi Framework [6] is an open service platform for the delivery and control of different JAVA-based applications, called bundles. Bundles are JAR files containing extra metadata, making them fully self-describing. This metadata is contained in MANIFEST.MF files. Besides the MANIFEST.MF, the bundles consist of packages containing JAVA classes. The packages of a bundle are private by default and they are not visible to other bundles. However, packages can be made public and may be imported by other bundles if the package is exported. MANIFEST.MF is used to declare which packages are imported and/or exported. Furthermore, bundles can export and import services by using the OSGi Service Registry. A service is defined using a public JAVA interface which must reside in an exported package, usually contained in the bundle exporting the service. This bundle implements the service, instantiates it and then registers the instance using the OSGi Service Registry under the service interface name and using the exported JAVA interface. Other bundles can import this service by using the JAVA interface and use its methods.

IV. Web Services

The W3C defines a web service as "a method of communication between two electronic devices over a network. 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-processable format (specifically WSDL)." [16]

Web services are incorporated into the home gateway developed to offer modularity. The first implementation of the home gateway considered that all the bundles developed would be deployed in the same device, the home gateway. However, this may not be the case and some of the bundles may be deployed in a separated device from the home gateway.

As mentioned, the home gateway developed offers energy management strategies by using rules. Web services are used to download these rules from a rule server. In this case only one Rule Server bundle has been implemented. However, the user could use more than one rule server to obtain the rules for the energy management system from different sources.

To incorporate web services into the home gateway developed, Apache CXF Distributed OSGi [17] is used. This distribution enables an easy integration of web services into OSGi platform. Furthermore, CXF-DOSGi will autogenerate the Web Services Description Language (WSDL) from the java interface, at the deployment time. The web services are implemented in a so called server. The consumer of the service is called the client. In order for the client to use the web service, a java interface of services has to be deployed in the client side. The server bundle will contain an implementatin of this interface. The interfaces of the web services are deployed in Web Services Interface bundle which will be enclosed in the server and client sides.

V. Home Gateway Architecture

The home gateway designed is aimed for a HEMS. It provides a knowledge base data repository where information about the devices is stored and also a rule engine. The rule

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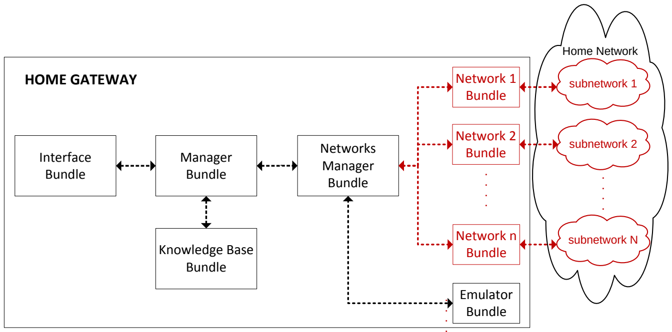

Figure 1. Home Gateway Architecture

engine is used to run rules that will help the user reduce the energy consumption at home. Furthermore, the home gateway provides a user interface from where the devices and rules can be managed.

An ontology is used as a knowledge base data repository. DogOnt ontology is used to create the initial knowledge base data repository of the home gateway. However, this article will not describe this ontology as an extensive description can be found in [10].

The first design of the home gateway implemented consists of the bundles shown in Fig. 1. This first design assumes that all the bundles will be found in the same device. However, some of the bundles could be found in different devices. The interaction between bundles found in different devices can be handled by using web services. In the second implementation, which is presented in this paper, it is assumed that there are two GUI: the Administrator GUI and the User GUI, as shown in Fig. 2. The Administrator GUI is contained within the home gateway. On the other hand, the User GUI could be contained within a separate device, for instance the users' computer. Furthermore, the Network n Bundles, which are equivalent to Network Emulator bundle, could be deployed separately from the home gateway implementation. This is a likely situation as home appliances use different communication technologies and bridges could be used, together with the home gateway, to enable communication to all the devices in the home network. The presented HEMS offers interoperability at the service level between its components, which interact through web services. In order to integrate web services into the new design, three new bundles, called Home Gateway Web Services Interface bundle, Rule Server Implementation bundle and Rule Server Interface bundle, are added into the implementation. A short overview of each bundle is provided in the following subsections.

A. Knowledge Base Bundle

This bundle handles the interactions with the knowledge base data repository and rule engine. To implement the home gateway knowledge base data repository DogOnt ontology is used. Instances of OWLIndividuals are used to represent these home devices, their functionalities and their status based on DOGOnt OWLClases. The information about the devices, including their status, features and functionalities,

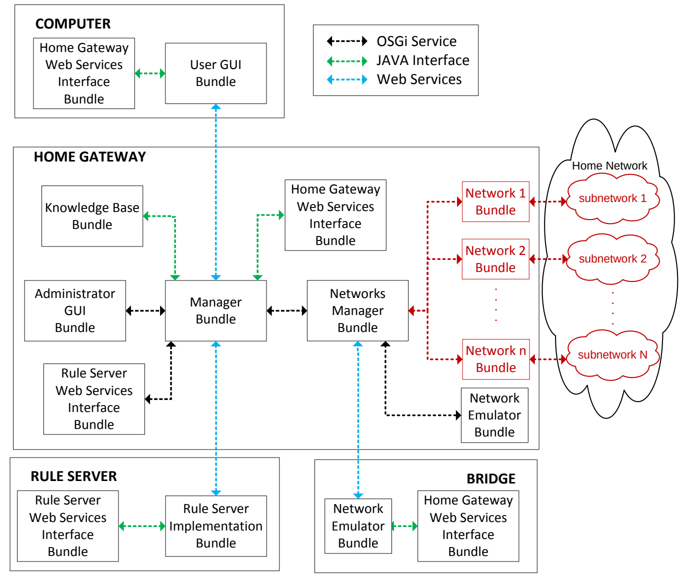

Figure 2. HEMS Architecture

can be obtained from the knowledge base data repository by using queries. SQWRL (Semantic Query-Enhanced Web Rule Language) [18] is used to write the queries. SQWRL is based on SWRL language for querying OWL ontologies and provides SQL-like operations to retrieve knowledge from them. Furthermore, SWRL (Semantic Web Rule Language) [19] is based on a combination of the OWL-DL and OWL-Lite, which are sublanguages of the OWL.

In addition this bundle contains the means to apply energy management strategies by using rules. Rules are written in SWRL, which are used to reason about OWLIndividuals by using OWLClasses and properties terms. Jess Rule Engine [20] is used to run the rules and queries. Protégé-OWL API 3.4.4 [8] is used to manipulate the ontology through the implemented home gateway. This API also provides methods to handle the interaction between Jess Rule Engine, SQWRL and SWRL. Further information details about the Knowledge Base Bundle can be found in [5].

B. Administrator GUI Bundle

The home gateway developed provides a Graphical User Interface (GUI) which is contained in this bundle. This GUI is used to communicate to the devices, obtain information about them and manage the energy management rules. This GUI is aimed at users with knowledge about the ontology and the capabilities of devices. The user can use this GUI to send commands to the devices in the home network, get their status, get the valid commands for a device, incorporate a new device to the knowledge base data repository, manage rules to apply energy savings and save the knowledge base data repository containing the information about the devices in the network. The Administrator GUI provides the list of devices in the home network and also the list of possible commands. Not all commands can be sent to all devices, so when the user tries to send an invalid command to a device an error will be prompted.

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C. User GUI Bundle

This bundle is contained within a different device of home gateway, for example the users' computer. This GUI is less advanced than Administrator GUI. The User GUI is a simple GUI with limited capabilities which is aimed at users with limited knowledge about the devices and energy management strategies. This GUI is used to send commands to the devices and get their status. The energy management rules cannot be managed from this GUI. However, this functionality could be incorporated in this GUI if desired.

This bundle obtains the necessary information about the devices by using web services to communicate to the Manager bundle. For instance, the list of devices in the home network and also the list of possible commands can be queried. Web services are also used to enable this separated user GUI to send commands and receive notifications, for instance status updates.

D. Network n Bundle

The home network found in the users' premises can contain devices using different communication technologies, for example power line or wireless. Each of these Network n bundles will handle the communication with the devices in the subnetwork n in the home network. These bundles will send messages and forward notification messages to/from the devices contained in the n subnetwork.

The home gateway could provide some of these communication technologies. However, the home gateway may not provide all the communication technologies found in the home network and the possibility of using bridges to communicate to some of the home appliances is a likely scenario. For this reason these bundles could be found in the home gateway or in a different device such a bridge as shown in Fig. 2.

E. Network Emulator Bundle

The focus of this paper is not on the enabling technologies in the physical layer and their interoperation, but on the software mechanisms that allow use of the different elements, regardless of the connectivity mechanisms towards the home gateway. The home network is therefore emulated and an GUI is provided to emulate changes in the devices status. This bundle functionalities and interaction with the rest of the architecture are equivalent to Network n bundle. Two Network Emulator bundles are used, one contained in the home gateway and another contained in a bridge separated from the home gateway device. The Network Emulator bundle contained in the bridge interacts with the Manager bundle by using web services. On the other hand, the Network Emulator bundle contained within the home gateway uses the OSGi services directly provided by the Manager bundle. Both of this Network Emulators provide a user interface from where the devices status can be changed, for instance, the status of a switch or sensor.

F. Networks Manager Bundle

Due to the fact that various Network n Bundles may exist, inside the home gateway and also in bridges, this bundle is created to handle the communication with these Network n bundles. This bundle will receive all the messages that have to be send to the home network devices and forward it to the corresponding Network n bundle of the subnetwork hosting the target. In order for this bundle to function properly there has to be a mapping between the the Network n bundle and the information contained in the knowledge base data repository so the message can be forwarded to the proper Network n bundle. At the time of writing this bundle only interacts with Network Emulator bundles. However, a Network bundle to support communication with KNX networks is under construction.

G. Manager Bundle

This is the central bundle and handles the interaction between the different bundles and contains the web services implementation and therefore acts as the server. To handle this interaction, this bundle offers different services, one for each functionality. The services used in the home gateway will be explained in more detail in section VI. Some of this services are exported as web services and others are simple OSGi services.

H. Home Gateway Web Services Interface Bundle

In order to implement the web services for the home gateway this bundle is needed. It provides the JAVA interfaces needed by the client and the server of the web service to make the exchange of information possible. This bundle is used to define the methods that can be used through the provided web services. This bundle must be included in both components, the client and the server, as the interfaces are need in both sides to successfully implement the web services.

This bundle contains different packages, each containing the necessary interfaces for each web service provided by the home gateway which will be explained in section VI.

I. Rules Server Implementation Bundle

This bundle is used as a rules provider for the energy management system of the home gateway implemented. This bundle emulates a rules server that can be found in the internet. This server mainly contains a data base where SWRL rules are stored and offers access to them through web services. The user is able to connect to it using the GUI and downloading the rules (s)he selects.

J. Rules Server Web Services Interface Bundle

Similar to the Home Gateway Web Services Interface Bundle bundle, this bundle provides the JAVA interface needed by the client and the server of the web service to make the exchange of information possible. However in this case the exchange of information is between the Manager bundle and the Rules bundle An overview of how the rules are imported to the home gateway is given in section VII.

Table I
HOME GATEWAY SERVICES

Service	Provided by bundle	Used by bundles
Message Listener	Manager	Networks Manager
Message Sender i	Network i	Networks Manager
GUI Printer Listener	Manager	GUI
Rules Provider	Rules Server	Manager
Knowledge Base Operator	Knowledge Base	Manager
Manager Functions	Manager	GUI
		Network Manager
		Network i

VI. BUNDLES INTERACTION

In this section the services offered by the different bundles will be explained briefly and are summarized in Table I. A more detailed description of the services can be found in [4]. Not all the services are made available through web services as some are used only inside the home gateway. The details of the web services is given in the next section.

The following services can be found in the developed home gateway:

- **Message Listener Service:** This service is provided by the Manager bundle. It is used by the Networks Manager to receive all the messages that have to be sent to the devices in the home network. After receiving a message the Networks Manager will find which subnetwork is hosting the target and forward the message only to the corresponding Network bundle. The messages send to the devices can be generated by the user through the provided GUIs or by Knowledge Base bundle when a rule has been fired.
- **Message Sender i Service:** This service is used by the Network Manager bundle to forward the message to the corresponding Network bundle hosting the targeted device. Each Network bundle provides its own Message Sender service which the Network Manager has to import. One may argue that the Networks Manager bundle could be removed from the implementation and all Network bundles could subscribe to Message Listener and Message Sender service could be omitted. However, this would mean that all Network bundles would receive the message even though the targeted device is not found in its subnetwork. Furthermore, considering that some of the Network bundles may be found outside the home gateway device, Networks Manager bundle, Message Listener Service and Message Sender services are necessary. Due to this fact this service is offered as a web service so bridge devices can use it.
- **GUI Printer Listener Service:** This service is imported by all the user GUIs, in this case the Administrator GUI bundle and the User GUI bundle, which imports it through web services. This service is used to receive notification messages, for instance when a device changes status a message will be received by the subscribed bundles and printed in the GUI to notify the user.
- **Rules Server Service:** This service provides access to the rules stored in the Rule Server bundle. It is implemented as a web service where the Rule Server bundle is the server of the web service and the Manager bundle is the client. This service will provide methods to download the list of rules, the rules' names and brief description. It is also used to download the SWRL rule to be deployed in the home gateway energy management system. An example of how the user can download rules from a rule server is provided in section VII.
- **Knowledge Base Operator Service:** This service is used to access the knowledge base data repository, containing the information about the home network devices and rules. It is used to query knowledge base data repository and obtain information about the devices and energy management rules. The only bundle importing this service is Manager bundle. The Manager bundle will offer the functions provided by this service to the rest of the bundles. In this way there is a separation between the knowledge data base repository and the rest of the architecture in case it is modified.
- **Manager Functions Service:** This service offers different functionalities. Firstly, this service can be seen as a mid-step to query the knowledge base data repository. This is done to facilitate the modularity of the system. The Knowledge Base Operator service is used to directly access and query the knowledge base data repository. The Manager Functions service is used by bundles that need to obtain information contained in the knowledge base data repository. The bundles inside and outside the home gateway used this service to obtain information about the home network and its devices. For example, the GUIs bundles use this service to obtain the list of the home network devices and commands for those devices. The Network Manager bundle uses this service network information of the devices, for instance the subnetwork hosting the device. The Manager Functions service will use the Knowledge Base Operator service to obtain the information from the knowledge base data repository.

  In addition, the Manager Functions service offers two other functionalities related to the Message Listener and the GUI Printer Listener. This service offers a method to forward the messages for a home device to Message Listener subscribers, in this case only the Network Manager bundle. In a similar way, when a message has to be delivered to the GUIs, the Manager Functions service is used to send the notification to all the bundles subscribed to GUI Printer Listener.

  The last functionality of this service is to provide access to the rules servers by using Rules Provider service. Manager Functions service includes the necessary methods to bridge the GUI and the Rules Provider service.

As the Manager Functions service is used by the GUI bundles and Network bundles it has been offered through web services.

VII. Accessing Rules Server

The Administrator GUI bundle offers a GUI from where the user can manage the energy management system rules. These rules are designed to decrease the energy consumption by, for example, turning off appliances when they are not need. For instance, the home gateway can have a rule which turns off the light in a room when there is enough natural light coming through the window or when there is no one in the room. These rules can be introduce in the HEMS through the Administrator GUI, by introducing them manually or downloading them from a Rules Server using web services. The rules in a Rule Server are presented to the user through a GUI, which displays the names and a brief description of each rule in the Rule Server. This information is obtained by the Manager Functions using the Rules Provider Service accessed thought web services. If the user is interested in incorporating any rules into his/her energy management system, the rule is downloaded and loaded into the rule engine in the Knowledge Base Bundle. The rule is loaded by using Knowledge Base Operator service through Manager Functions service.

Using rules servers and web services the user is not forced to write his/her own rules, (s)he can obtain them from a rule server or even from other users.

VIII. Conclusion

In this paper, a home gateway using OSGi framework, ontologies for the knowledge base data repository and web services has been presented. Protégé-OWL API 3.4.4 has been used to handle the interaction between the OSGi home gateway, SQWRL queries, SWRL rules and Jess Rule Engine.

The home gateway presented can support different technologies as long as the corresponding Network bundle is created. These Network bundles can be found outside the home gateway device. For this reason web services have been incorporated into the home gateway. Through web services the remote Network bundle can easily consume the home gateway services. In a similar way, the GUI could be also found outside the home gateway, for instance a computer or mobile phone. By offering the home gateway services though web services freedom to chose between different display devices is provided. Furthermore, web services are also used to download rules for the energy management system provided in the home gateway from a rules server.

The main motivation was to create a simple home gateway which would be easily scalable, new bundles can be incorporated to the HEMS to offer new functionalities and the bridges to communicate with all home devices can be deployed when necessary. The system developed has the necessary capabilities to create a home energy management system: offer control of devices through a GUI, run rules to apply energy management strategies and interoperability between devices even if they belong to different subnetworks. The home gateway presented in this paper is therefore suitable for Home Energy Management System (HEMS).

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Proposal of Functional Exchange Networking for Distributing Data Services across Multiple Network Generations

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***Abstract*—Continuous evolutions of network management and control technologies are producing a variety of different network functions such as network-oriented authentication, cross-layer operation, administration and management, and session-based quality of service control. The continuous growth is also producing a byproduct that blocks the global distribution of data services because the evolved network functions are effective only within a single network domain. In short, the global network will have the network generations of conventional IP-based network, the next generation network, and the emerging future network. In order to achieve advanced network services utilizing evolved network functions across multiple network domains and generations, this paper proposes a network function exchange architecture. The proposed network function exchange intermediates the various differences related to control protocols, management information, and data format. A service scenario using the network function exchange and detailed architecture is described. Functional requirements of the exchange, a design of the universal interface protocol, and an operational procedure based on the design are described.**

Keywords-NGN; Future Network; Network Function Exchange; Multiple Network Generations

I. INTRODUCTION

Penetration of the cloud computing services and the Internet accessibility are driving the global distributions of the information and communication technology (ICT) services. The recent growth of emerging countries is the economical background to ICT globalization, and accordingly, activities of enterprises tend to go across borders. Indeed, various Internet applications have already been provided globally across multiple network operators, but they are usually provided with the best-effort quality. On the other hand, mission critical or bandwidth-sensitive ICT services cannot be globally achieved by best-effort quality. Today, the standardization of the next generation network (NGN) is enabling session-based quality of service (QoS) control even over all Internet protocol (IP) networks [1]. However, the capabilities achieved by the NGN are effective within the NGN operator, and achieving the capabilities across multiple network operators is unlikely since the NGN has not been widely rolled out yet. To accelerate the global distribution of the enterprise cloud services, various network functions, such as QoS or authentication interworking not only between conventional IP network and NGN but also between NGNs are required. So far, an attempt for global distribution of session initiation protocol (SIP) based services (typically the voice over IP (VoIP) service) has been made [2], however the service exchange technique cannot apply to non-SIP services including the cloud. For the limited scale of inter-operator network interworking businesses, the open access networking has been also discussed, where a common access operator is the hub for network service distribution [3, 4]. In addition, there was past activity for interworking among different types of networks [5], but the activity handled only single generation (i.e., IP network) and handled only single network function (i.e., QoS control).

Recently, a standardization of the future network (FN) as the next of NGN has been initiated [6], and an advance evaluation of the FN testbeds are also underway [7, 8]. FN will have additional functional capabilities, such as the network virtualization that enables secure isolation of user networks [9]. In the future, the global service distribution must transcend architectural barriers at the network borders of conventional IP networks, extended conventional IP network having the bandwidth broker [10] mechanism, NGN, and FN [11]. Not only a simple connectivity but also additional functions (e.g., QoS, authentication, charging) are required to be interworked between those network generations. The attendant issue of the interwork will be filling gaps regarding available functions between the network generations. In addition, depending on countries and operators, exchanging functions may face policy differences regarding, for example, the regulation of the data allocation and the business process of authentication and charging.

To achieve the network interworking, two typical models called the exchange model [12] and the private peering model [13] have been discussed. The exchange model has been employed for Internet exchange (IX) [12], and many networks are interconnected at the cost-effective concentrated exchange point. While the exchange model enables consolidation of interconnection points, the policy management (e.g., defining QoS class, authorization, and applied function itself) cannot be unified since each interconnecting provider has a different policy. On the other hand, in the private peering model, it is relatively easy to negotiate a universal policy and interworking functions at the interconnection point. But the private peering model is unlikely when the number of interconnected network increases, and thus the private peering model tends to require more interconnecting interfaces than the exchange model. Therefore, the exchange model is expected to be suitable for the service distribution in the large-scale global environments. However, there have not been discussions regarding the suitable functional architecture for interworking various network functions among multiple network generations.

This paper proposes a network interworking architecture for the global-scale service environment across multiple network generations based on the network function exchange (NFE). This paper also proposes the functional design of the NFE and the design of the interworking interfaces. The structure of this paper is as follows. First, an example of the service scenario by utilizing proposed NFE is described in section 2. Section 3 describes the proposed functional architecture to achieve the exchange. Based on the functional design, technical issues that need to be resolved and the requirements to resolve the issues are identified in section 4. Next, detailed functions and interface design to fill the requirements are described in section 5. Finally, the proposed procedural design of the service distribution operations is described in section 6.

II. SERVICE SCENARIO

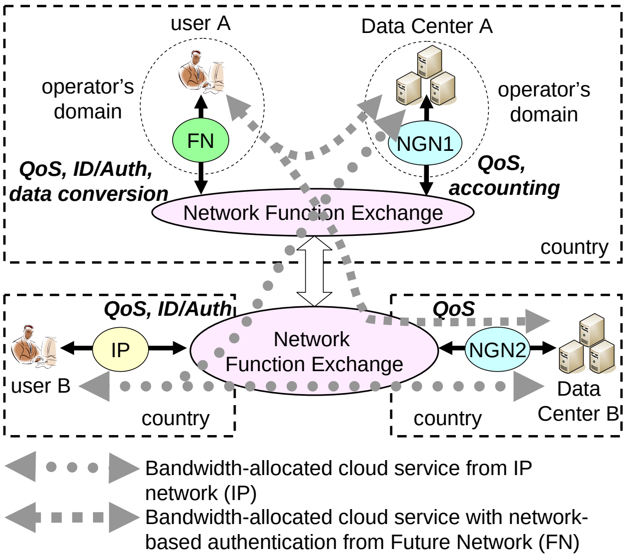

Figure 1. Example of advanced cloud service scenario by mediating functions among multiple network generations as well as multiple countries and operator's domains.

Fig. 1 shows an example scenario of a cloud computing service where the NFE intermediates the functions among multiple network generations owned by different countries and operators. The NFE is operated by an independent company dedicating NFE services, or operated by one of the network service operators providing FN, NGN or IP. The ***terms*** in Fig. 1 show examples of the major functional categories that are intermediated by the NFE. In this scenario, the intermediated functional categories are QoS control (e.g., bandwidth allocation, priority management, guarantee of latency and jitter), network-based identification and authentication (ID/Auth) (e.g., fixed-line or user-terminal based authorizations with the network operator-driven strict confirmation and proof of no-spoofing) and accounting. In addition, if the conversion of the data format or the control signal is necessary, the NFE makes the conversion. In this scenario, a data center A is connected to NGN1 which provides the QoS control and accounting functions. User A is connected to the FN that supports QoS control and ID/Auth functions. By intermediating the available functions of QoS control, ID/Auth and accounting between NGN1 and FN, the NFE provides a QoS-guaranteed cloud service with strict user authentication. The NFE knows the differences of protocols, data formats and functions between network generations, and has the translating/conversion functions. In this scenario for the cloud service between the data center A and the user A, when the NFE receives a request for network services across multiple network generations and domains from user A to data center A through the FN and NGN, the NFE identifies the detailed requested information such as the destination to be connected. The NFE intermediates the QoS control function with the translation of the control protocol to an understandable one in NGN1. The most important thing in this scenario is that the QoS control function is achieved across multiple network generations without any modifications to the current implementation of the control and management scheme on each network generation and domain.

In addition, in order to prevent spoofing, the NGN1 may ask the FN to provide the network-based authentication for identifying user A before providing the service. It is important that network-based authentication (i.e., ID/Auth) function is asked not directly to the user A but to the FN. The FN checks the subscriber information of user A and informs the result to the NGN1 through the NFE. If necessary, the NFE translates the ID/Auth information in order for the NGN1 to be able to understand. The NGN1 also requires the accounting to the user A, and the NFE asks the FN by proxy for the NGN1's request. This function is also translated in the NFE and then requested to the FN with the translation of the management protocol. In the same manner, the data center B connected by NGN2 which supports the QoS control function can also provide a QoS-guaranteed advanced cloud service with the user authentication for user B. Next, the user B connecting to conventional IP network that supports only the QoS control function can use the QoS-guaranteed cloud service without strict network-based user authentication using the data centers A and B. In such a scenario, the NFE can provide advanced services over multiple network generations, network operators and the limitation of country by intermediation of the various network functions.

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III. PROPOSED ARCHITECTURE

A. Architectural Fundamentals

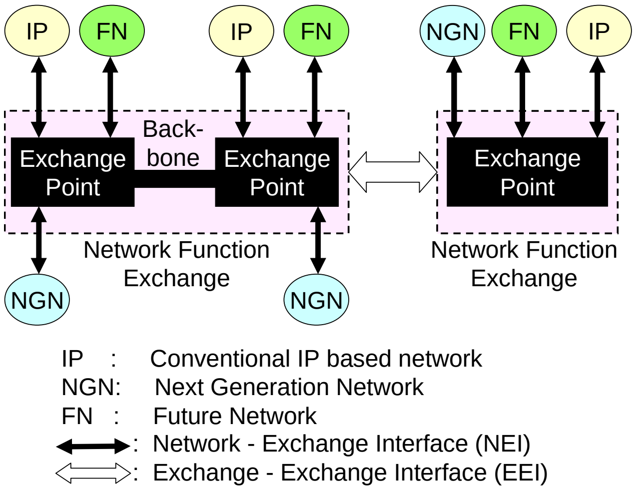

Figure 2. Basic topology of the proposed NFE

Fig. 2 shows the basic topology of the proposed NFE with two interconnection models bridging multiple generation networks. A NFE consists of more than one Exchange Point and a backbone link as optional. The backbone is required if one NFE needs to scale out by distributing the exchange point. If more than two exchange points are interconnected by the backbone link, the aggregated entity acts as an NFE for the distributed architecture. Various kinds of network generation (e.g., conventional IP network, NGN, and FN) are assumed to be interconnected to the exchange point through an interface called the network-exchange interface (NEI). All the data transition and the mediation of control information are transacted within the exchange point through the NEI. The most important characteristic of the exchange architecture is that the NEI is a unified single interface regardless of the network generation. The proposed design of the NEI is described in section 5. As indicated in section 2, the conversion of the data format and the control protocols between the different networks generations are carried out within the exchange point. Therefore, the exchange point provides not only simple data bridging, but also negotiating and brokering of network functions. If it is necessary to interwork between multiple NFEs, the exchange points are interconnected by the Exchange-Exchange Interface (EEI). The difference between the backbone link and the EEI is that the backbone link is a simple transport of control signals and data traffic, and EEI has a functional negotiation role but also a simple transport role.

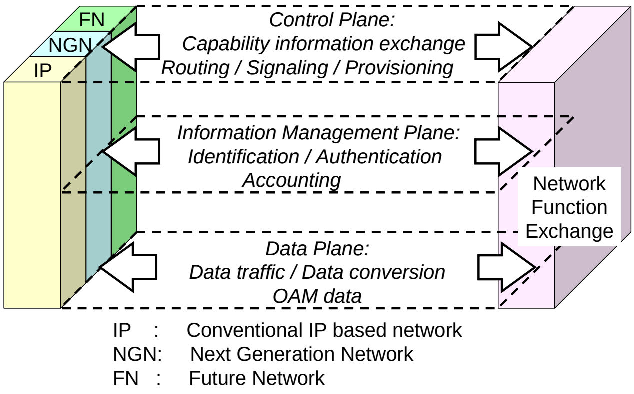

Figure 3. Proposed stratum model of the NFE architecture

The proposed stratum model of the NFE is shown in Fig. 3. The top of layer is the control plane where the supported functions of the network, routing, and reachability information are exchanged. In addition, the signaling information for establishing the QoS-managed path or for admission control is also exchanged in the control plane layer. The middle layer is the information management plane where information on authentication, accounting, network statistics, and operation and management (OAM) status is exchanged. The bottom layer is the data plane where all of the users' data traffic and OAM signals are exchanged. If the data structure or address information of one network generation's traffic must be converted into another generation's format, it is done via the data plane. The detailed data link layer, network layer or transport layer of FN has not been defined yet. Therefore, if the definition of FN will be completely different from the conventional IP or NGN, the Exchange Point must absorb the differences utilizing the data plane. The backbone includes these triple layers within it as a simple link interconnecting multiple exchange points, and delivers the information and data to each exchange point.

B. Architectural Comparison

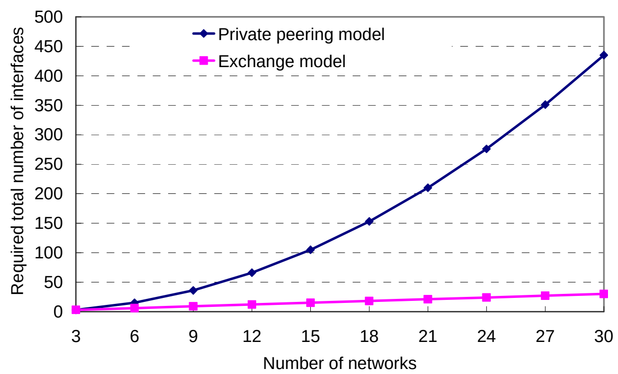

Figure 4. Comparison chart of required number of interfaces for interconnection

Fig. 4 shows a comparison of the required number of interconnection interfaces between the private peering model and the proposed exchange model in the case of fully meshed interconnection among networks that can be conventional IP, NGN or FN. In the private peering model, when the number of interconnecting networks is *n*, the total number of interfaces can be calculated by an equation of the number of *2*-combinations from *n* elements, ${}_nC_2$. As the result, the total number of required interfaces increases at $O(n^2)$. Hence, when the number of interconnecting networks is 15, the total number of required interfaces is more than 100. Considering service provision on a global scale, it is no longer scalable. On the other hand, in the proposed architecture, a network only has to have an interconnection interface connected to the NFE, regardless of the total number of other interconnecting networks. As a result, the total number of interfaces increases on a linear scale of $O(n)$. In addition, considering the redundancy, many more interfaces are required for a full mesh model. However, the definition of the interface may be much more complicated than the full mesh model, and the NFE must play many advanced roles in order to achieve the proposed architecture. The detailed design is described in section 4.

IV. Technical Issues and Requirements

With the intermediation of network functions, a newly deployed function in one network must also be available to the other interconnecting networks. This means that any network of any generations must be indirectly but continuously upgraded based on functional demand or the plan of the network. In addition, interoperability issues must be partially resolved by emulating some of the functions implemented in the advanced network side. In order to achieve the above requirements in the control plane, each network must report the list of available functions exposed to the NFE. This concept is similar to the capability information exchange or discovery in the link-layer discovery protocol (LLDP) [14] and the link management protocol (LMP) [15], but those protocols can handle only link-layer capabilities. As for routing or signaling protocols, it is important for existing networks, such as the conventional IP and NGN, to reduce the impact on implementing additional interface protocols in order to ensure interoperability. Therefore, interface protocols used in the current networks should be continuously applicable for the NEI as much as possible. Absorbing protocol differences or converting the data format must be performed by the NFE. Even if the same interface protocols are used in interconnecting networks, the meanings or definitions of used parameters may be different and that results in failure of interworking routing or signaling operations. In order to prevent such failures, the introduction of common meanings and definitions within the same context parameters are required. Additionally, if conversion of the data format or addresses is necessary on the exchange point, additional latency and jitter caused by the exchanging operations are desirably required for consideration in the control plane.

As for the protocols and handled information regarding ID/Auth, accounting, and OAM control exchanged using the information management plane, there are few discussions and standardization activities for the functional exchange. Especially for the ID/Auth function, contexts and granularities of the information, such as the identifier (e.g., user account) and the locator (e.g., IP address), should be coupled beyond the network generations. In order to bridge the different ID/Auth protocols, universal hi-level schemes to exchange information should be defined as applicable to multiple network generations. Standardizing the high-level scheme aims to reduce the implementation impacts on all of the network generations, while standardizing new ID/Auth protocols or choosing one existing protocol has huge impacts on existing networks to be interconnected. The proposed high-level scheme is described in section 5. The approach of the ID/Auth functional exchange is expected to be applicable for the accounting function, since the accounting operation is tightly coupled with the ID/Auth information. Regarding the regulation policy of data allocation, configuring the data cache has legal constraints depending on the situations of each country. To address this constraint, the capability of disclosing location information for each exchange point or caching area is effective within the functional exchange. If multiple NFEs are involved, the capability of determining NFE for the conversion is required on EEI in order to avoid duplicated capability allocation and to balance the functional load.

V. Detailed Functions and Interfaces Design

A. *Functions Provided by Each Network Generation and Applicable Protocols*

Table 1 shows the proposed list of typical network functions and shows the eligible protocols for each function. The category of Table.1 shows the major types of functions and the function name represents the specific function belonging to each category. These categories of network functions listed in Table I are already-available ones in current NGN or IP (i.e., QoS path control, ID/Authentication, Accounting, OAM, and Conversion) and a new function available in the FN (i.e., Virtualization/Separation). Eligible protocols for each function are shown in the applicable protocols field in Table 1 per the network generation. Fields represented as “New” in “Applicable Protocols” column means that the protocol needs to be newly defined. As for the “Conversion” row, there are needs for converting between different addressing such as IPv4 and IPv6, and for converting between different transports such as Ethernet and SONET/SDH.

TABLE I. List of Proposed Functions and Applicable Protocols at Each Network Generation

Function		Applicable Protocols		
Category (plane)	**Function Name**	**FN**	**NGN**	**IP**
QoS path (Control)	Bandwidth allocation	New	SIP, Ri, RSVP	RSVP, SIP
	Priority	New	RSVP	RSVP
	Delay	New	New	New
	Jitter	New	New	New

Function		Applicable Protocols		
Category (plane)	**Function Name**	**FN**	**NGN**	**IP**
ID/Authentication (Information management)	Identification	New	New	New
	Authentication	New	Radius	Radius
Virtualization/Separation (Control)	Establish of virtualized/separated slice, participant management	New	New, L2TP, MPLS, GMPLS, VLAN	New,L2TP MPLS, GMPLS, VLAN
	Address conversion	New	New	New
Accounting (Information management)	Accounting	New	Radius	Radius
OAM (Control, Information management)	End to end quality Reachability	New	New	New, LMP
	End to end delay	New	New	New, LMP
	End to end jitter	New	New	New, LMP
Conversion (Data)	Data/format conversion (address, data, codec, IPv4-IPv6, etc)	-	6rd, DS-lite	6rd, DS-lite

B. *Design of the Interface Protocols*

As a basis of the protocol, a design of the control commands required for each function is proposed. Commands for each interface protocol are assumed to be performed in the control plane or the information management plane, and the commands are designed so that the typical operations of each categorized function in Table 1 can be covered as much as possible as well as the exchange of functional capability information described in section 4. In terms of the data conversion function in Table 1, it is a capability within the data plane, and thus specific commands are not introduced.

First, four commands are proposed to cover the exchange of functional capability information described in section 4.

- Functional capability data base (DB) creation
- Functional capability DB update
- Functional capability DB deletion
- Functional capability DB ack

The functional capability DB creation command is used when the new network is attached to the NFE so that a new functional capability database has to be created. The functional capability DB contains both the availability information of each network function and the routing/reachability information. The functional capability DB update command is used when current functional capability information is changed in supporting of new function. The functional capability DB deletion is used when a network is removed from the NFE. Functional capability DB ack is sent by NFE in order to notify that the NFE surely receives the command (the functional capability DB creation or update or deletion) to the FN, NGN and IP. The functional capability DB contains the area information where the required network function is available.

Second, five commands are proposed to cover the typical QoS functions.

- QoS path creation
- QoS path deletion.
- QoS path modification
- QoS path confirm/provisioning
- QoS path ack/nack

The QoS path creation command is used when new QoS guaranteed (bandwidth allocation, priority control, delay control and jitter control) path wants to be setup, and it contains the parameters of the QoS (e.g., bandwidth in Mbits per second) and path information (e.g., ingress, egress and transit points) to be created. The QoS path deletion command is used when the already setup QoS guaranteed path wants to be deleted. The QoS path modification command is used when the already setup the QoS guaranteed path wants to be modified. The most important factor of this command is that the availability of target path must be kept without any disruption of data traffic. If the modification cannot be achieved without the data disruption by a technology such as the "make before break", the network providers have to notify that they have no functional availability of the QoS path modification by using the functional capability DB messages. The QoS path confirm/provisioning command is used when the required commands (creation, deletion, modification) are really operated. In the QoS path ack/nack command, Ack is used to report that the sent command (QoS path creation or deletion or modification or confirm/provisioning) is received by the opposite network or NFE. Nack is used to show the refusal of the received command and the reason to the opposite network or NFE.

Next, three commands are proposed to cover the typical ID/Auth functions.

- ID/Auth request
- ID/Auth reply
- ID/Auth ack/nack

The ID/Auth request command is used when ID/Auth information is required from one network or NFE to another NFE or network. The ID/Auth reply command is used when one network or the NFE reply with the ID/Auth information to another NFE or network. In the ID/Auth ack/nack command, Ack is used to report that the sent command (request or reply) is received by the opposite network or NFE. Nack is used to show the refusal of the received command and the reason to the opposite network or NFE.

As for the accounting function, four commands are proposed to cover the typical operations.

- Accounting request
- Accounting reply
- Accounting confirm/provisioning
- Accounting ack/nack

The accounting request command is used when charging is required from one network or NFE to another NFE or network. As the parameter, detailed information such as the cost is included in this command. The accounting reply command is used when replying with the received accounting request command, such as "accept" or "not accept." The accounting confirm/provisioning command is used when the real charging transaction is done. In the accounting ack/nack command, Ack is used to report that the sent command (request or reply or confirm/provisioning) is received by the opposite network or NFE. Nack is used to show the refusal of received command the reason to the opposite network or NFE.

To cover the OAM function, five commands are proposed.

- OAM path creation
- OAM path deletion
- OAM path modification
- OAM path confirm/provisioning
- OAM path ack/nack

The OAM path creation command is used when the OAM function is newly required. The OAM path deletion command is used when the coexisting OAM path is deleted. The OAM path modification command is used when the coexisting OAM path is modified. The OAM path confirm/provisioning command is used when the required commands (creation, deletion, modification) are really operated. In the OAM path ack/nack command, Ack is used to report that the sent command (creation or deletion or modification or confirm/provisioning) is received by the opposite network or NFE. Nack is used to show the refusal of received command and the reason to the opposite network or NFE.

Finally, five commands are proposed to cover the typical virtualization functions.

- Virtualized/separated slice creation
- Virtualized/separated slice deletion
- Virtualized/separated slice modification
- Virtualized/separated slice confirm/provisioning
- Virtualized/separated slice ack/nack

The virtualized/separated slice creation command is used when the virtualized/separated slice or layer is newly required. The virtualized/separated slice deletion command is used when the coexisting slice or layer is deleted. The virtualized/separated slice modification command is used when the coexisting slice or layer is modified. The virtualized/separated slice confirm/provisioning command is used when the required commands (creation, deletion, modification) are really operated. In the virtualized/separated slice ack/nack command, Ack is used to report that the sent command (the virtualized/separated slice creation or deletion or modification or confirm/provisioning) is received by the opposite network or NFE. Nack is used to show the refusal of received command and the reason to the opposite network or NFE.

VI. Procedure for the Service Distribution

All of the services are achieved using the combination of the functions defined in Table 1 and the commands defined in section 5. For instance, the procedure for the global cloud service scenario of exchanging the secure bandwidth allocation functions is represented in Fig. 5. In this service example, following three functions are utilized.

- QoS: bandwidth allocation (QoS path)
- Identification/Authentication (ID/Auth)
- Accounting

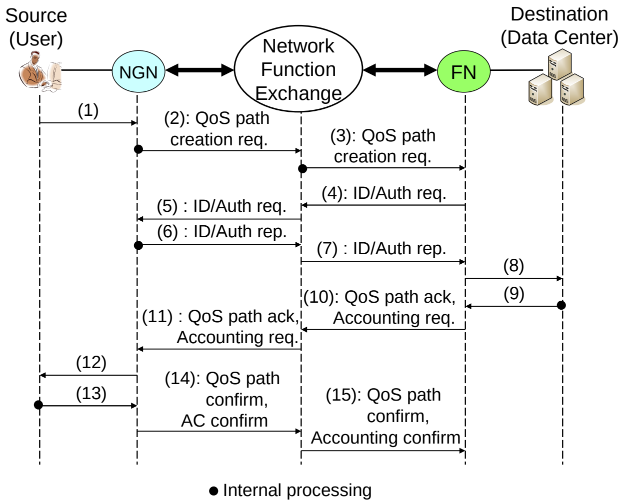

Figure 5. Example procedure of global cloud service scenario with exchanging the secure bandwidth allocation functions

The step-by-step procedure in Fig. 5 is explained below. It is on the premise that user knows the destination information as IP address, domain name or a kind of aliases representing the service name.

(1) The user sends a service request command for bandwidth allocation between the user and the data center to the NGN.

(2) The NGN confirms the availability of the bandwidth resources between the user and the NFE, and then sends the QoS path creation command to the NFE.

(3) The NFE confirms the functional capability of the FN belonging to the data center, and then sends the QoS path creation command to the FN. If the FN does not support the requested function, the NFE sends the QoS path nack command to the NGN and operation ends.

(4) In order to authenticate the requesting user, the FN sends the ID/Auth request command to the NFE.

(5) The NFE forwards the ID/Auth request command to the NGN. As the basic service policy using the network-based ID/Auth functions, the data center and the content provider must clearly specify the utilization to the user. Thus the confirmation process or privacy protection scheme must be prepared to the user.

(6) The NGN checks the identity of the requested user and then sends the ID/Auth reply command to the NFE.

(7) The NFE forwards the ID/Auth reply command with the requested user's identity information to the FN.

(8) The FN asks the data center's approval to create a bandwidth-allocated path to the requested user.

(9) The data center confirms and approves the user by the ID/Auth information.

(10) The FN sends both the QoS ack command and the accounting request command to the NFE.

(11) The NFE forwards both QoS ack command and the accounting request command to the NGN.

(12) The NGN asks for approval from the user.

(13) If the user agrees the details of service contents with accounting/charging information, he sends the final confirmation to the NGN. The final confirmation message is the trigger of actual service provisioning.
(14) The NGN sends both the QoS path confirmation command and an accounting confirmation command to the NFE. In addition, the NGN provisions the bandwidth-allocated path between the user and the NFE.
(15) The NFE sends both the QoS path confirmation command and the accounting confirmation command to the FN. In addition, the FN and NFE provision the bandwidth-allocated path between the NFE and the data center. Finally, the NFE bridges the bandwidth-allocated paths.

VII. Discussion

The proposed architecture has advantages in reducing the number of interconnecting interfaces and also simplifying implementation of functional interworking scheme. However, those advantages are only effective for the network service providers which operate a single generation network and want to interconnect to different network generations for wider service distribution. However, there are some remaining issues regarding complicated implementation of functional interworking scheme especially for NFE provider. First, NFE has to know the detailed functional capability information of each network as well as the routing information, and to determine the availability of services from that information. For the further study, the scalability about the number of connecting networks should be done from this viewpoint. Second, the scalability of protocol and data conversion functions [16] in an NFE has to be considered.

VIII. Conclusion

In order to achieve global distribution of data services utilizing various network functions across multiple network generations and domains, a functional interworking architecture of NFE is proposed. The architectural fundamentals and design of the NFE with a triple layer structure of the control plane, the information management plane and the data plane are proposed. With the detailed functional description, the universal control commands between the NFE and each network generation is proposed to intermediate the various functions. Finally, the utilization of proposed commands is identified with a secure bandwidth-reserved cloud service scenario.

Acknowledgment

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A Middleware Framework for the Internet of Things

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***Abstract*—After the traditional Internet (with program-to-human communication), and after the Internet of Services (with program-to-program communication), the Internet of Things is a new paradigm of communication aiming at integrating the state of everyday things into the digital world. But things are everywhere, have different colours, come in different flavours, so, building reliable applications that depend on such things imposes great challenges and demand for new approaches to integrate heterogeneous devices smoothly. These new methods should use public and resilient networks, like the Internet, to secure and facilitate the access to things. This paper addresses such constraints and proposes a middleware framework to interact with the devices and their data, all this supported by the use of Web services. It is our goal to design and implement a generic framework where services represent functionalities of sensor networks and provide a dynamic way for high-level applications to interact and program such devices with heterogeneous hardware. By using Web services we benefit from the interoperable technology, cross platform, independency of programming languages, and available on the Web. Those attributes are ideal to combine heterogeneous systems like sensor networks. It is also our goal to create an event-driven system, where the user can subscribe the available services and receive notifications as network data is being processed.**

***Keywords*-middleware, Web services, sensor networks, internet of things, service-oriented architecture.**

I. Introduction

The Internet of Things (IoT) is a novel paradigm that aims at bridging the gap between the physical world and its representation within the digital world. The idea is to integrate the state of the "things" that form the world into software applications, making them benefit from world's context information.

There are various forms of capturing the state of things, ranging from simple bar codes identifying objects, to more sophisticated technologies involving radio-frequency identification (RFID), near field communication (NFC), or even more complex devices, such as sensor nodes, that come equipped with an internal memory, context awareness devices (e.g., GPS), and, specially, with computation capabilities [1]. These last mentioned devices are of special interest, since they allow things, named *smart objects*, to perform local computations and interact and collaborate among them.

Sensor networks [2] are a hot research topic in academia and an expansion business area in industry, with applications in many fields. These networks are composed of a set of nodes with the capability of sensing physical phenomena (e.g., luminosity, temperature, or humidity). Wireless Sensor Networks (WSNs) are a specialization of sensors networks, where communication among devices occurs via radio-frequency and nodes (usually) rely on a battery power supply. Those characteristics allow devices to operate remotely, but complicate the access and the process of sensed data.

Building high-level applications that exploit information from smart objects are of valuable interest, but they do not come without a cost. In fact, interacting with devices that run on top of different operating systems (e.g., TinyOS [3], Contiki [4]) or virtual machines (e.g., Squawk [5]), that use distinct programming languages (e.g., nesC [6]), and that use different communication protocols, makes it very difficult, and undesirable, to handle the complexity from the high-level application.

Our work focus on abstracting the interaction among applications and smart objects (e.g., nodes connected via a WSN), by both hiding the communication (and other hardware idiosyncrasies) and the programming capabilities of such objects. It is our intent to accommodate these differences in a middleware layer such that, from an application perspective, all smart objects are reprogrammable and present a common interface.

The programmability of smart objects depends, more often than not, on manufactures and whether they provide hardware specifications. As far as we are aware, there are not so many platforms that allows devices to be reprogrammed remotely, while running. In Callas [7] we presented a framework that supports reprogramming of nodes in a WSN. Here, we lift this idea to the middleware level by equipping it with reprogramming capabilities irrespectively of these capabilities being supported by the underlying infrastructure.

From the high-level application point of view deploying code into (a network of) smart objects is performed regardless of the ability of these devices to be programmed. Based on the configuration of smart objects, the middleware either installs the code in the devices or in itself and behaves as if the code was effectively deployed into the devices. For this approach we put together a Data-Flow engine that runs client deployed modules; these work upon data received

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from smart objects and are organised in dataflow chains acting as filters to process incoming data according to the client's requirements.

Another problematic area is the interoperability among observation centres and their clients. We follow the Sensor Web Enablement standard (SWE) [8] from the Open Geospatial Consortium. This standard defines a set of interoperability interfaces and metadata encodings that enable real-time integration of heterogeneous sensor webs. Our implementation respects two SWE standards: *Observations and Measurements* (O&M) [9][10], and *Sensor Observations Service* (SOS) [11].

To summarise, our main contributions are:

- the abstraction of code deployment, communications, and hardware of (networks of) smart objects, allowing for client applications to program heterogeneous devices either physically, by installing code into the devices, or logically, by creating a Data-Flow network of filters to process the data received from devices.
- the management of observations received from smart objects and its broadcast to client applications via Web services respecting the SWE specification;

The remainder of this paper is structured as follows: Section II presents state-of-the-art related work. Section III presents our approach to fulfil the aforementioned goals. In Section IV we detail the internals of out middleware, and, finally, the last section draws our conclusions and provides an overview of the intended future work.

II. Related work

Middleware frameworks are widely used to aggregate and manipulate sensor networks. Rakhi Motwani *et al.* [12] present a Service-Oriented Architecture (SOA) to coordinate observation centres and to share their information via Web services. The authors' motivation is that many observation centres are geographically dispersed and isolated, and it is difficult to share information among them. We adhere to this view and our middleware proposal, MufFIN (Middleware For the Internet of thiNgs), provides features to manage the data received from network devices and allow information to be registered and accessed via SWE standards.

Another open source implementation of SWE is the 52North Sensor Web community application [13]. Since the project seems to have achieved a mature state, we tried to build Muffin on top of it. However, it has revealed to be quite difficult to integrate a stand-alone application with our framework features. Instead, we used part of their specifications (e.g., database ER model) as the base to our SWE implementation.

The communication between heterogeneous services is possibly by using a known protocol on top of a common format, like XML. However, this kind of technologies needs more computational resources to marshal and unmarshal the contained information. Choon-Sung Nam *et al.* [14] propose

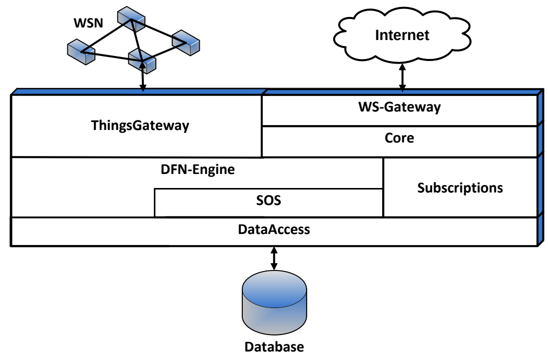

Figure 1. The bundles layer diagram illustrating the communication between bundles from the gateways to the database layer.

an Event-Driven Architecture middleware equipped with a publish-subscribe paradigm, allowing client applications to subscribe topics of interest and to receive notifications when the server publishes information on such topics. This type of communication reduces the amount of messages exchanged among clients and servers, as opposed to pulling techniques used within synchronous scenarios.

João Santos also implemented a middleware framework [15] that combines SOA and Event-Driven Architectures to achieve interoperability among WSNs and client applications via Web services. The framework supports the creation of pipelines (fixed dataflow chains), although it does not provide for dynamic composition of filters in dataflow chains, and its current implementation does not support the SWE specification.

Catello Di Martino *et al.* [16] present an adaptive and configurable architecture for accessing sensor networks based on their specifications. These specifications result in filters and focus of information, where clients connect to, in a fast way, and gather or receive notifications with the needed data.

Our middleware solution, like most of the aforementioned works, makes use of Web services, supports synchronous and asynchronous requests, complies with the SWE standards, and abstracts the sensor's hardware. New to MufFIN is the ability to transparently reprogram smart objects, being this our main scientific contribution.

III. Our Approach

This section presents our middleware design and the decisions made to address the requirements identified previously.

A. Architecture

We choose to design our middleware framework based on a Service-Oriented Architecture composed of a group of seven loosely coupled bundles. Figure 1 shows the framework layer diagram depicting the communication dependencies between components.

On top, we provide two communication gateways: the *thingsGateway* for communicating with smart objects (e.g., network sensors) and the *WS-Gateway* for communicating with high-level applications via Web services. The middleware supports both synchronous and asynchronous communications with clients. The *Core* bundle provides the Web interface implementation to invoke framework operations. *DFN-Engine* bundle manages client deployed modules as well as the dependency chains between them. It instantiates the client modules and creates the publish-subscribe connections for their communication. The details of the deploying operation is presented in Section III-B. As for the *Subscriptions* bundle it receives clients subscriptions, processes them, and saves the subscriber's information. This bundle publishes notifications for clients to the Web. Section III-C details the bundle's internals. The processing of XML documents for the Sensor Observation Service standard is delegated to the *SOS* bundle. All operations of this bundle must follow the OGC specifications. At the basis of our middleware lies the *DataAccess* bundle that provides a set of façades for other bundles to access the database layer. Database tables related to smart object observations are based on the O&M specification.

The following section describes how our framework handles devices' programming and data management.

B. Network Programming and Data Management

For network programming we allow clients to upload modules (filters) and to instantiate them. If the target smart object (network) allows for code installation, the middleware deploys the received module to it. Otherwise, the code is instantiated locally and runs directly at the middleware layer. For that, the client must also specify the module's dependencies. This specification is given in an XML document, like the example shown in Listing 1. Notice that the document also specifies whether the module will be made available as a Web service; the client may decide not to expose the module to the Web because it may be used to compute intermediate results. To discover module's information the client may invoke the Web service *getModulesInfo()*.

Listing 1. An example of XML file to instantiate a new modules

```
<deploy>
 <instance serviceId="TemperatureDiff" service="true">
  <dependency serviceId="LisbonTemperature"/>
  <dependency serviceId="PortoTemperature"/>
 </instance>
</deploy>
```

The created modules and the triggered events build a dependency chain as a publish-subscribe network, like it is shown in Figure 2. This chain results in a Data-Flow Network (DFN) where the raw data, coming from the smart objects (e.g., WSN nodes), is further processed to produce the information subscribed by the high-level applications.

Figure 2 illustrates the interaction with two networks of things, accessed through gateways *LisbonSink* and *PortoSink*. This modules receive raw observations directly from the networks. Whenever new data is available this modules

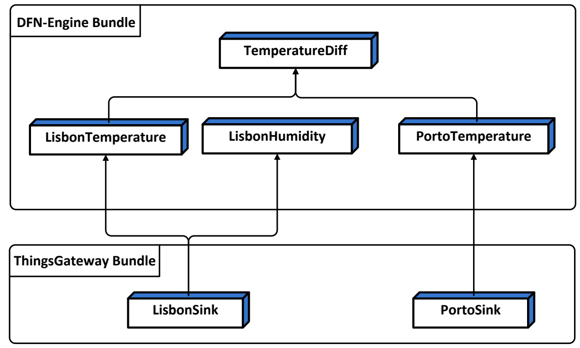

Figure 2. The modules dependencies chain.

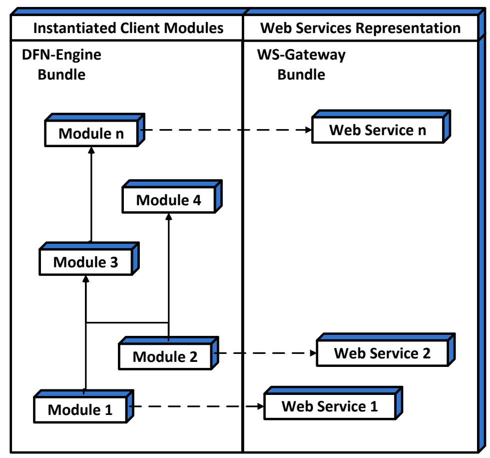

Figure 3. Conceptual view of the relation between module instances and their representation on the Web.

will store it and notify their subscribers, which will process the supplied data, store it, and further notify theirs subscribers (and so on). In case a module is available to the Web, its remote subscribers will be notified as well. For instance, the *LisbonSink* notifies modules *LisbonTemperature* and *LisbonHumidity*. In its turn, *LisbonTemperature* will inform *TemperatureDiff* that, dependent on a value received from the *PortoTemperature*, will compute its difference and make it available for its subscribers. If no value is available from the *PortoTemperature* module, *TemperatureDiff* will suspend until a value is obtained.

Each module stores its processed information, allowing for backdated queries.

C. Subscriptions

After instantiation, a module becomes available for subscription via its corresponding Web service. The *Subscriptions* bundle maintains information about subscribers and which services they subscribe. When the DFN processes information, it notifies the respective module and an event is sent to the subscriber.

Figure 3 represents the mapping between modules and

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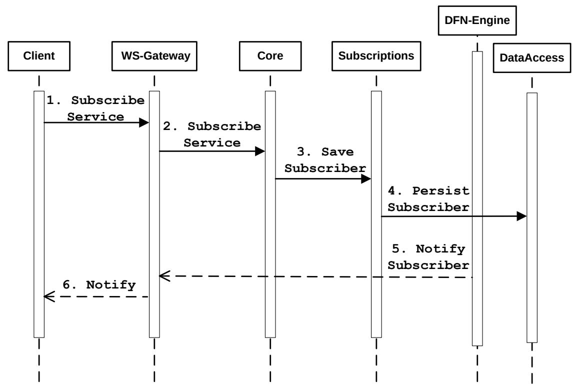

Figure 4. Sequence diagram with the invocations between modules from service subscription to client's notification.

Web services. In the present case, only modules 1, 2, and n are available for subscription. Modules 3 and 4 just compute intermediate information that is further refined and made available. *Module 4* exemplifies a kind of extension point that can be refined later.

The framework provides two methods of subscribing services:

- *Direct subscription:* by specifying the service to be subscribed, given its identification.
- *Discovery subscription:* by querying specific service characteristics. This allows for a client to discover (and subscribe) services based uniquely on the partial matching of the client's topics of interest with the module's characteristics. Our approach is to classify modules in ontologies (upon module's definition) and allows for querying these ontologies for finding relevant services.

To enable notifications it is necessary a client Web-service endpoint where the events will be delivered. The endpoint is correlated with the client at subscription time. Figure 4 shows the sequence diagram corresponding to the actions from a client subscription until its notification.

D. Framework Instances Integration

This framework was designed with interoperability in mind. Not only to integrate with different smart objects, but also to be integrated with other frameworks. This is particularly useful when a client wants to access to various observation centres. He may do it in two ways: (1) access each observation centre per se; or (2) if possible, some observation centres acts as a subscriber for the others and the client accesses only one centre that aggregates all the data. Notice that (ideally) all communications between middleware instances must be invisible to the client. The integration with other frameworks that are not compliant with SOS standard have to be tackled, not surprisingly, case by case, but it amounts to develop specific things gateway adapters for each case.

IV. Implementation

This section presents our decisions about MufFIN implementation and gives an overview of the two SWE standards we implement.

MufFIN is implemented in the Java programming language and runs on top of Fuse Enterprise Service Bus [17], based on Apache ServiceMix [18]. Fuse ESB implements the OSGi functionalities, allowing the complete and dynamic decoupling of the system components. The ESB is an architecture that facilitates the integration between services [12], [19], [20] and allows the creation of dynamic flows between system components.

The middleware introduced in this paper implements two types of communication with clients:

- *Synchronous:* allows clients to filter and receive observations already stored in the middleware. This communication respects the OGC standard, and returns the observations that follow the received constraints.
- *Asynchronous:* allows the middleware services subscription to receive observations that will occur in the future. When some action related with the service happens, it triggers an event that sends to the client the observations in OGC standard format.

Both communication types respects the following SWE standards that we present a brief overview.

Observations and Measurements (O&M)

Standard models and XML schemas for encoding observations and measurements from a sensor. An observation is defined as an act of measuring a property or phenomenon, with the goal of producing an estimate of its value.

The observation has the following mandatory fields:

- *Sampling Time:* time when the measurement was made.
- *Procedure:* process used to generate a result. Could be a sensor observation, an algorithm, a computation, or a complex processing chain.
- *Phenomenon:* defines the environmental characteristic to be read, and its unit.
- *Feature of Interest:* the observation target; the real-world object on which the observation is made.

O&M aggregates observations that have common characteristics; in particular, observations that have a similar *Feature of Interest* and observe the same *Phenomenons* should be related by an *offering*. *Offerings* define what is provided by the system and it is the base property for all observation requests.

Sensor Observations Service (SOS)

Web services interface for requesting, filtering, and retrieving observations and sensor system information. The

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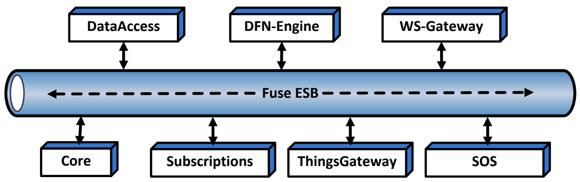

Figure 5. Main bundles that compose MufFIN integrated on Fuse ESB.

goal of SOS is to provide access to observations from sensors and sensor systems in a standard way that is consistent for all sensor systems including remote, in situ, fixed, and mobile sensors. The filtering arguments allow the client to specify the time, the location, the observed phenomenon, and the feature of interest of each sensor observation. All these arguments are dependent on an offering that is going to be used as the primary search criteria.

A. The Middleware Bundles

Figure 5 shows the middleware overview with the main bundles deployed on the Fuse ESB. The middleware is structured in decoupled bundles to facilitate code maintenance. Below, we present the bundles implementation details:

- *ThingsGateway:* allows the dynamic loading of adapters that bridge the communication between smart objects and our framework. These adapters are the root nodes of the Data Flow Network tree and are responsible for deploying new code into devices. Our framework is ready to send code to devices that run the Callas virtual machine, and can be customized to other languages with the deploying of new network adapters. This is possible because the framework receives byte-code, which facilitates the process since it is difficult to create a framework built-in with a large range of source language adapters.
- *WS-Gateway:* responsible for presenting the framework features as Web services. We follow the Web Service Notification protocol (WS-N) [21] to notify remote subscribers. When a high-level application subscribes a service, it must send an endpoint reference to where the notifications can be published. When a notification is triggered, our middleware dynamically connects to the client endpoint and sends the subscribed data.
- *DFN-Engine:* we use the *ActiveMQ*—a Fuse ESB built-in broker that implements the Java Message Service and provides reliable communication—to implement the following communications: *Queues*, to implement one-to-one communications among bundles; and *Topics*, to implement one-to-many communication among dynamically deployed DFN filters.
  The deployed services follow the command pattern [22], extending a class *Service* and overriding the method *doAction()*. This method is invoked whenever a notification arrives from a module, on which the self depends. The class Service also offers methods to get and set the observation properties specified in the O&M standard. After processing an incoming event a module publishes data via the *send()* method.
- *SOS:* responsible for the processing of the SOS standard XML documents. We implemented the XML parser on top of Apache XML Beans [23].
- *DataAccess:* provides the interfaces to access the database layer. All system data (e.g., observations, subscribers data) are stored in a MySQL database where the access is performed using the Java Persistence API (JPA) to decouple the database implementation.

V. Conclusion and Future Work

In this paper we proposed MufFIN—a generic middleware framework—that allows for managing and programming Internet of Things smart objects.

The IoT aims at bridging the gap between physical and digital worlds, by integrating world's context information, described by the state of the "things", into software applications, making them context aware. An important topic in this area is how to manage the heterogeneity among smart objects that equip these networks.

To improve the interoperability among observation centres, our framework conforms with Web service schemas that follow two Sensor Web Enablement standards from the Open Geospatial Consortium, namely the Observations and Measurements and the Sensor Observations Service. This allow for high-level applications and observation centres to get smart object data via the Internet using Web services.

The main goals of our framework are to manage the data received from WSNs and to create a generic framework to program their devices in two transparent methods: by installing code directly into devices, and by creating a Data Flow Network in the framework. The DFN is created with modules received from the clients and installed in the framework. This provides, to high-level applications, the perspective that the code was deployed into the sensors, even though the code runs at the middleware side. The DFN filters the data received from the WSN and returns the data that clients are expecting.

In the near future, we plan to pursue four major areas: (1) improve the discovery subscription of services, with the use of ontologies to bind directly the services with their *Feature of Interest* and the network device locations; (2) software testing and validation, in particular, load testing; (3) support for SensorML [24], a SWE standard to specify sensor devices and their characteristics; (4) field-testing, MufFIN is planned for being deployed in a real use case scenario, where all the presented features will be tested. The scenario involves collecting environmental values (temperature and moisture) from farms in the Azores archipelago, in order to establish the conditions wherewith a fungus that causes light skin hypersensitive in cattle may appear. The project has both financial and public health impact, since an accurate assessment of the fungus appearance symptoms will reduce causalities and prevent overmedication among animals.

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