International Telecommunication Union

Proceedings of the 2010
ITU-T Kaleidoscope
Academic Conference

Beyond the Internet?

Innovations for future networks and services

Pune, India, 13-15 December 2010

Supporters:

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The opinions expressed in these Proceedings are those of the paper authors and do not necessarily reflect the views of the International telecommunication Union or of its membership.

Foreword

Malcolm Johnson
Director
ITU Telecommunication Standardization Sector

Thomas Edison, one of the most prolific inventors in history famously said “the value of an idea lies in using it”. This is precisely why ITU-T strives to increase cooperation with academia, one of the most important sources of good ideas and innovations. Bringing ideas to fruition through standardization not only helps academics and their institution raise the profile of their work, but allows the wider community to benefit from their thinking. New and improved technologies can make life better in so many ways, making it a better, safer, cleaner, more efficient world for all.

I am very pleased therefore that the ITU Plenipotentiary Conference agreed a new academia category of membership at a much reduced membership fee. In part this is as a result of the lively discussions during the first and second Kaleidoscope conferences. I very much hope we will soon have plenty of new members from academia.

This is the third Kaleidoscope academic conferences and each has further evolved the concept. In Pune, India, we address the theme: Beyond the Internet? - Innovations for future networks and services. It promises to be an interesting discussion on what’s best for the future internet – evolution or clean slate? The many high-quality papers in these Proceedings address this issue, and venture on new services that could be provided by the future internet.

I am confident that discussions which start during this year’s Kaleidoscope on the social, economic and technical drivers will challenge the fundamental networking design principles of the Internet, and will assist ITU-T in the development of standards for future telecommunication networks and services to support the future internet.

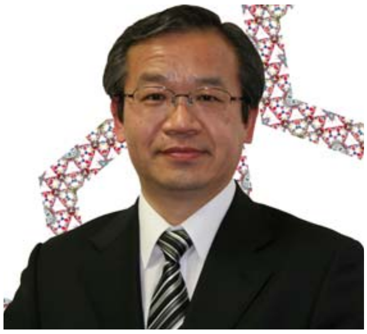
ITU greatly welcomes and encourages the contributions of academia in these debates. ITU standards enable interoperability between products on a global scale. All major ICT companies are ITU-T Sector Members, and there cannot be a more enlightening window on the world of ICT development than participation in ITU. The opportunity to be part of a team that creates a worldwide standard provides an exciting opportunity for any academic. We welcome you with open arms!

Malcolm Johnson

Director
ITU Telecommunication Standardization Sector

Chair's Message

Yoichi Maeda
General Chair



I am very pleased and proud that Kaleidoscope events look set to become an important event on both academic and ITU calendars. The success of the first two Kaleidoscope events - in 2008 and 2009 - demonstrates a real need for this type of outreach by ITU to the academic community. The outcomes of these events are having an impact on ITU meetings at all levels – from ITU Plenipotentiary Conferences to assisting the standardization work in ITU-T study groups.

A new category of membership in ITU for academia has just been approved – this is a practical outcome of earlier Kaleidoscope events and a real achievement of which we can be proud. We can now open the ITU doors much more widely to broaden academic participation in global standards development.

Another practical outcome from our earlier discussions is the ITU-T Focus Group on Future Networks. I am certain that the quality papers in Kaleidoscope 2010 will bring concrete results to enable further standardization towards future networks and services in the future internet.

This year's theme "Beyond the Internet?" has inspired fantastic contributions and we look forward to the presentation of some very interesting papers. As General Chair, I can assure you that selection was not an easy task and I would encourage those that have not been selected this year not to lose heart - next year's event - and the chance to try again - will be with us before you know it!

Over 115 paper proposals were submitted for double-blind peer-review by more than 150 ICT experts in our Programme Committee. I appreciate very much the hard work of this review panel in what is no doubt one the most difficult of tasks of the Conference: that of selecting for presentation the papers that stand out from the very high quality contributions across the board. Its findings are the 37 papers accepted for inclusion in these Kaleidoscope proceedings, in addition to four invited papers.

I am very glad that once again the Kaleidoscope Conference enjoyed the technical co-sponsorship of the IEEE Communications Society (ComSoc). This enabled us to make the papers for all three Kaleidoscope conferences available through the IEEE Xplore digital library, giving the global scientific community access to the wealth of ideas discussed. In addition, we published special features in the IEEE Communications Magazine, showcasing the top papers of the first and second Kaleidoscope events. We will do the same for Kaleidoscope 2010.

In addition to thanking the Programme management and Organizing Committee members, I must also express my sincere thanks to Cisco Systems, Nokia Systems Networks and MyFire that provided funding to support, among others, the 'best paper' cash awards. I also thank STES, GISFI, CMAI, and ITU-APT Foundation of India for promoting the event, and for the efficient local organization of the Conference. Last but not least, I wish to extend my sincere appreciation to the ITU Secretariat for the invaluable support. None of the achievements of this Conference would have been possible without them.

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Yoichi Maeda

General Chair

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SESSION 1

KEYNOTE SUMMARIES

MODERN ACADEMIA:
TEACHING, RESEARCH, DEVELOPMENT, PATENTS AND STANDARDS

Uday B. Desai
Director, Indian Institute of Technology, Hyderabad, India

Academia has traditionally focussed on teaching, research and development. These three aspects form the core of academic paradigm in most institutions. Moreover, they are in themselves quite demanding. In India, most academic institutions are working towards establishing themselves as leading research institutions – in fact, they are endeavouring to create an innovations culture. With the growing awareness on how important it is to create IPR, there is a new dimension that is added to academic pursuit, namely, patents and standards. Standards are very vital to today's technological development and are, perhaps, even more vital to take technology to the market. Moreover, standards are a source of revenue not only to institutions but to the nation.

Thus, today there is a need to rework the academic structure. It is not necessary that every faculty does all four; nevertheless, it is imperative that there are enough faculty that give emphasis on standards. In fact, once the realization sets in, that taking research to market is closely entwined with standards – there will be an automatic emphasis on standards.

It is also important to recognize that for research to get incorporated into standards there has to be active collaborations with industries. This collaboration is where Indian academia is weak.

In this talk, first a brief perspective on how standards activity can be incorporated into academia without compromising on the existing academic paradigms will be given. Then, some research activity in India, in the area of ICT that could have impact on standards will be mentioned. Also, there will be a brief description on some of the ongoing efforts on standards development, by academicians, in India. The talk will conclude on what are the possible avenues for academia to move forward and make major contributions to international standards.

VEHICLE COMMUNICATION: A FUTURE TELECOMMUNICATION MARKET

Tadao Saito
Professor Emeritus the University of Tokyo

Because of the rapid development of electronics, the market of information and communication technology changes rapidly. The cost reduction of advanced electronics made communication equipments inexpensive, and nowadays almost all people in the world have cell phone. This means that the market of telecommunication for human users is near to saturation. Although telecommunication technology changed rapidly and the performance of communication has improved, it is difficult to have higher price for higher performance, so and the market expansion will be incremental.

As a promising new market, vehicle could take advantage of telecommunications to develop a variety of applications for safety, comfort and operation support. The connected vehicle technology is a new competition edge in the car industry. In order to develop these applications, the telecommunication performance parameters need to be redesigned. Vehicle telecommunication could be subdivided in two different classes: “transport telematics using telecommunication” and “intelligent transport system (ITS) using dedicated short range communication”. The boarder of this classification is dynamically changing to expand the territory of transport telematics.

The presentation covers some examples of vehicle communication and explains new ranges of performance parameters. To promote the future “network of things” market, designing properly the performance parameter sets is needed; the current telecommunication development follows a different path. Finally, a new set of requirements for Next Generation Networks can be derived from the analysis of future applications.

FUTURE OF COMMUNICATIONS − THE INDIVIDUAL USER EXPERIENCE

Detlev Otto
CTO Nokia Siemens Networks, Germany

The advent of smart devices combined with HSPA network capabilities changed many things in the mobile communications food-chain. We could say the network changed from service provisioning to service enabling.

On the one side users are now masters of their services and the network is used as a transparent pipe. On the other side network resource utilization left predictability. The third "mega-trend" seen is connected objects or machine to machine (M2M) devices. Forecasts see as much as ten times more connected objects/devices in 2020 compared to 2010.

Obviously the users used their chance and with the help of smart devices (phones, tablets) and already started to individualize their communications needs and behaviour – and we are only at the beginning of that era.

The operators will have to respond to this and make sure their network resources are utilized in relation to the ARPU they can get. They will have to figure out how to participate in this changed "communications food-chain" and how to manage a service enabling network.

Here we can see three "next big things" for 2011 and beyond:

I. Capacity or resource management

II. Monetizing the service enabling network

III. Network transformation from network to service management

The first big thing focuses on the user experience. The users are individualizing their services and the operators need to individualize the user experience in the same way. The communication networks need to learn to differentiate between the services and which service and user should have which resource in which situation.

The second big thing is the need or wish to participate in innovative new services and revenue streams. These will be born mainly out of two areas: a) blended services combining user insight of the network operator with services from the cloud and b) enterprise services. Both require the network to do two things smarter than today: identity management and smart charging.

The third big thing is a network transformation from network management to service management. As part of this, today's BSS "food-chain" will need to manage millions of machines and objects connected and planning and operations processes will leave stove-pipes for radio, core, transport in favour of services to be managed end to end. IP will be a key enabler assisting this transformation.

SESSION 2

RETHINKING THE NETWORK

TOWARD A POLYMORPHIC FUTURE INTERNET: A NETWORKING SCIENCE APPROACH

Kavé Salamatian,

LISTIC Lab
Université de Savoie
Annecy-le-Vieux, France

ABSTRACT

In this paper, I will develop two major claims. First the, Future Internet should be polymorphic and conciliate different architectural paradigms networking. The second claim is that the Future Internet should be build on strong theoretical basis from a Networking science that is in course of development.

In this paper, I have used the concept of cooperation as an interpretation lens. Specifically, I will describe how virtualisation make possible a polymorphic future Internet and enables the easy deployment of new cooperation schemes. The next aspect that I describe in this paper is relative to security in the future Internet. Particularly the paper advocates the necessity of three major components: a secure execution platform, an authentication mechanism, and a monitoring component. Finally, I will show that it is possible to build scalable addressing and routing scheme but at the condition of following a clean slate approach.

***Index Terms*—** Future Internet, Internet Science, Network Architecture

1. INTRODUCTION

Based on one of the major stories about the origin of the Internet, Internet came to the age of 40 at 22:30 hours on October 29, 1969. During its 40 years lifetime, it grew from a small three nodes network build mainly for computer timesharing, to a network connecting an estimated 1,800 million users and a penetration rate of almost 25%. No human built system has ever reached such a growth rate in such a short time. What was once a tool known and used by only a small intelligentsia of high profile researchers, has become within one generation a universal commodity, like electricity, in the daily life of hundreds of millions of users. This shed light on the importance of the on going discussion about the "future Internet." Several initiatives envision the definition, the design, and the construction of this future Internet. On the research side, the US based GENI (Global Environment for Network Innovation) initiative [1], and the European Union FIRE (Future Internet Research and Experimentation) initiative [2] are noteworthy and many of other activities in different countries can be cited. More generally, the United Nations World Summits on Information Society held in 2007 and 2009 enlarged the scope of Internet as the main component of the Information Society of the future by introducing cultural aspects in a more formal way. That said, this flurry of interest on the future of Internet is also a source of confusion; different and competitive requirements as well as architectural concepts are fogging our vision of the future of Internet.

My aim in this talk is indeed not to add up to the existing fog by sprinkling my "*yet another*" new and clever architectural conception of how the community should shape the future of the Internet. I have in this paper two claims: first that future Internet should be polymorphic, *i.e.* that it will need to conciliate inside it different architectural paradigms. Particularly, the future Internet would have (at least in a long transitory period) to support the evolution of the current Internet in form of IPv6 or any other evolution of the current IP architecture along with other more revolutionary paradigms. Therefore, the main property of the future Internet should be flexibility to enable their smooth coexistence. My second position is that networking is not simply a technological artefact, but it is becoming a separate science that borrows some of its principles from other well-established sciences as computer science, physics, social science, *etc.* and has also its own particular fundamental laws and principles, similar to any other science. Describing the particular principles of this "*Networking Science*" and differentiating them from the principles of other sciences is one of the major scientific challenges that the Internet and more generally the networking research community has to overcome in the coming years. Indeed, by looking at networking in its broadest view, one can see that networking in form of roads, postal service and telephony have a very long history. However, it is only during the past years that with the development of social networks and wide availability of Internet that it becomes obvious that these historical networks and the new comer Internet should have common principles that have still to be fully investigated in the context of networking science.

Obviously, if such networking fundamental principles exist, the future Internet will also follow them naturally. Unfortunately, finding fundamental principles for networks is still a research effort. Being realistic, I will only be able in this paper, to shed some lights about major questions that the community will have to tackle in the path toward the future Internet. Being invited to give a keynote, I will be a little more radical than I am generally as I want to ignite thought-provoking discussions and open new perspectives.

2. COOPERATION: THE VERY ESSENCE OF NETWORKING

First, let's dig into the basics of networking. It is possible to define a network as a set of nodes that are cooperating with each other to distribute (exchange) information. However, one has to formalise the concept of cooperation further. The main role of a node in a network is to generate information in form of messages, packets, signals, *etc.* We can therefore state that a networking component receives from its neighbours (other components, nodes, or layers) and/or its environment a sequence of incoming messages or information and generates a sequence of outgoing messages. The relation between incoming and outgoing information streams defines the cooperation function (or forwarding function) the node implements.

We have been educated to consider networks through the layered approach of the Open Systems Interconnection (OSI) model. In this approach the inputs to the cooperation function come only from the predecessor layer and concern variables relative to components in the same layer; the resulting value is sent to next layer. Therefore, the layered architecture and the associated protocols constrains the space of possible cooperations between nodes to make it tractable and to formalise it. This layered view has been the major architectural paradigm in the past three decades. Layers opacity and independence enabled the programmers to concentrate on a single layer and to implement services without needing to tussle with other layers.

However, layering comes with a performance cost. Cross layering, *i.e.* enabling a layer to access information and to interact with any other layers, has been advocated for higher efficiency, performance, resource management, and security. These arguments have been important in the emergence of the "*autonomic network*" idea. In this approach, a network component is seen as an active element that is "*self-conscious*" and that interacts with its environment.

All the above considerations have resulted in a major shift of the networking paradigm that has moved away from a layered, to a puzzle view where autonomic components are "*co-operating*" with each other. This results in enabling the use of information coming from different layers for implementing the cooperation function. The classical view considered the operation of a network element as processing packets. Now the role of a network is considered as deciding (based on data received in the past and information's gathered from the environment) which type of cooperative functions the node (or the node owner) should be implemented to achieve the goals of the network (defined by the network operator, or the service provider), as well as its selfish goals (defined by the owner of the node). This opens the way for a node to behave differently from what the classical protocols predict (for example to go to a standby mode, or to open a tunnel to implements its specific cooperation function). Accepting that nodes can be selfish is a major change, motivated by applicative scenarios like ad-hoc wireless networks where one cannot assume that all nodes will belong to the same authority. Inter-domain routing at the Autonomous System (AS) level is another scenario where selfishness is essential. In this last situation, the different network operators have to cooperate even if they might be in fierce competition. Cooperation between selfish nodes is a central element of the future Internet architecture.

Moreover, cooperation is also the central concept of this developing science of networking. Networking science studies the production, distribution, and consumption of "information" and is cooperation for information exchange, precisely what a network does! Information has some specific properties that differentiate it from other goods and therefore necessitate a new theoretical treatment: information is universal and infinitely reusable, *i.e.*, a bit of information can be duplicated and shared infinitely at almost no cost. Another difference is that information is ambiguous, *i.e.* you might receive ambiguous messages when other goods and services are unambiguous. These two peculiar properties that are the basis of Information Theory as developed by Shannon differentiate Networking science from Classical economy. To illustrate this difference, one can look at the controversy existing in the economical literature on the Efficient-Market Hypothesis that asserts that financial markets are "informationally efficient". This hypothesis assumes that prices on traded assets (e.g., stocks, bonds, or property) reflect all past available information. However, the validity of this hypothesis has been questioned (critics even blame the belief in rational markets for much of the financial crisis of 2007–2010). This shows that unconsciously information is seen as an external concept helping in shaping the prices, and not as a normal asset and some mechanisms should be provided to make this information available to make market efficient.

While, in the layered view, network services are built over a predefined set of cooperation primitives (named Service Access Points in the OSI jargon), in autonomic networks the node has to permanently monitor its environment to optimise its cooperative behaviour based on information coming from a different layer of the classical layered architecture. While the optimisation can result in better performance, it adds a level of complexity and careless optimisation can even lead to lower performance as the information coming from different level of networks can be contradictory. A major challenge facing the networking research community consists of developing methods for online and autonomic optimisation of this cooperation.

3. TO CLEAN THE SLATE OR NOT? IS IT REALLY AN ISSUE?

From its inception 40 years ago, Internet was designed, developed and deployed simultaneously. More precisely whenever a problem arose, a solution was proposed and analysed following a technical consensus at the IETF resulting in a Request For Comment (RFC) solving the issue or implementing a new feature. The IETF consensus guaranted that the proposed evolution of Internet was backward compatible and was complying with the sacrosanct axiom of "no harm to what works". The previous guarantee is one of the explanation of the huge success and relative stability of current Internet. However, the drawback of this approach is that it contrains the future evolution and hinders the deployment of radical solutions that attack the problems at the source. This is one major reason for the clean slate approach advocated by a part of the community.

The clean slate approach comes from the belief that it is impossible to resolve the challenges facing today's Internet without rethinking the fundamental assumptions and design decisions underlying its current architecture. The incremental approach changes the Internet architecture by backward compatible patches; the clean slate approach advocates out of the box thinking with an architectural redesign with better concepts and abstractions to answer the current challenges.

However, as explained in the introduction we expect the future Internet to be polymorphic. Specifically the future Internet should be flexible enough to conciliate coexistence of the evolution of the actual Internet with the incremental patches with approaches coming out of the clean slate vision. Therefore, the future Internet should be designed so that the question of cleaning the slate or not should not be anymore relevant. Clean slate–based revolutionary research should go along with evolutionary approaches to ensure that the working Internet will still continue to work in parallel with new architecture with more features. We will describe later why we believe that such a flexibility is achievable and why therefore cleaning the slate or not is not an issue.

Nonetheless, imagining new architecture is a tough task. We need to choose among the large set of possible architectures, the few that could take the relay of the current Internet. For this purpose, we need to experiment different architectures in large scale experimental facilities. This explains why almost all initiative on the future Internet are backed by a large scale experimental facility like PlanetLab [3], GENI [1], *etc.*

4. A CRITICAL ANALYSIS OF THE FUTURE INTERNET MOTIVATIONS AND RATIONALES

It is natural to ask why people are questing for a future Internet. Internet as we know it today has certainly gone beyond the wildest expectations of it is first pioneers and even its current status addresses a large spectra of nowadays needs. However, there is a consensus in the research community and in larger audience that the current Internet has some shortages that make its evolution and/or revolution inevitable. I will give here some of the main reasons that are provided and shed some lights on the directions to go.

4.1. Flexibility or the future Internet contortionist

I explained previously that the future Internet should be polymorphic and its architecture be flexible enough to accommodate different cooperation paradigms in parallel. Another rationale for a flexible future Internet architecture is relative to new application deployment. The current Internet provides a very large freedom for developers to develop their own applicative protocols. However, it does not provide architectural hooks to deploy services beyond the socket interface; developers have no access to routing and addressing. But routing is an essential component for the cooperation provided by network. For this reason during the past decade, routing and addressing was frequently raised into the application level where the developers can have an impact on them.

Peer to Peer and overlay networks are examples of this approach and implement a complete cooperation scheme into the application level (more precisely above the socket interface). While this approach has been very successful, it is not really optimal as the packets have still to go through the underlying services narrow hip hourglass of IP that acts as a bottleneck. A network where one could implement, and deploy its new network protocols or cooperation schemes without disturbing other running protocols, would solve the application deployment issue, and will moreover provide a fantastic platform for innovative service deployment.

The quest for a flexible platform that will enable concurrent execution of different networking mechanisms along with easy deployment has been pursued in the networking research community with the objective of building a flexible experimentation platform for the future Internet research. This has resulted in the development of Planetlab [3], its European counterpart OneLab [4] and Global Environment for Network Innovations (GENI) [1]. The flexibility in these experimental platforms was attained thanks to the wide generalisation of virtualisation approaches [5] that enabled the parallel running of several virtual machines over a single hardware. As virtualisation ensures full isolation (fault, software, and performance isolation) between virtual machines, it enables the parallel execution of different networking systems (routing, addressing, *etc.*) and opens the way for the polymorphic future Internet I was advocating.

Virtualisation techniques also ensure the ability to encapsulate a full virtual machine into a single file that can be easily migrated to a virtualised hardware and being run on it. The encapsulation property opens the perspective of easy deployment of services by just distributing encapsulated virtual machine implementing the service over a large infrastructure of virtualised servers/routers. Last but not least, virtualisation approaches also ensure Interposition (to be discussed below) for monitoring and security. With the continuous increase in processing power available in commodity hardware, there has been a growing interest in developing new router architectures based on virtualisation running over clusters of multicore computers. This opens the perspective of building realistic routers implementing the polymorphic future Internet.

4.2. Security: the Achille's heel of the current Internet

One of the major rationales for the development of a future Internet is security. Indeed, the current Internet is plagued with spam, phishing, denial of service attacks, exploits and other security problems. However, one should be careful to not mix apples and oranges. Only a small proportion of problems referred as related to security, are resulting from the Internet architecture, *e.g.*, even if phishing is an important security issue, it cannot be related directly to Internet architecture. One has therefore, to separate what is relative to application security (ensuring that an application is doing what it is supposed to do), to communication security (ensuring that communication remain secret) and finally to network security (ensuring that cooperation on network is secure).

The approach of the current Internet architecture to security is minimalist. Security was not considered to be an essential component of the network architecture, even in IPv6 that integrates an IPSEC component. It was seen at best as an optional service. The absence of security-related elements inside the architecture can be seen as the root cause of the current security status, where we have an abundance of security service (VPNs, firewalls, proxies, SSL, *etc.*). As a reaction, some advocate for integrating all security primitives inside the architecture so that application can fully rely on network security services. This last view is also highly questionable as too much security has a heavy impact on network devices performance. The future Internet will have to find an in–between way between these two extremes. Indeed, the future Internet architecture should provide some support for application and communication security. However, we have still to determine the least common denominator of security support that should be integrated into the architecture and what should be seen rather as a service that will cooperate with other components through the architecture.

It is noteworthy that security is a negative concept: you do not know when you have it; you only know when you have lost it. This means that, rather than speak about providing security, one should talk about reducing the vulnerabilities. Almost 30 years of experience in Internet security has taught us that it is impossible (and too costly) to remove all risks, meaning that we have to accept that we will continue to live with a risky network. The consequence of the above statement in cooperation terms is that we have to increase the resilience of the future Internet architecture to ensure survivability and to reduce the impact of security risks. In other words, security risks should be assumed as plausible operational hypothesis in the design of networked system and architectural solution should be provided to detect and to contain them. This is a radically different position from the current approaches where the emphasis is rather put on authentication of users through passwords/biometrics and assuming that authenticated users are entitled to do whatever they do. In the collaborative approach, we have to assume that users (even authenticated) can misbehave and we should be able to detect and contain them.

Therefore, in the light of cooperation concerns, the future Internet architecture needs at least three basic security mechanisms: a mechanism shielding strictly and at the deepest level possible components running in the same execution environment (like a sandbox), a mechanism ensuring authentication (the type and level of authentication still pending) to ensure the identity of the running code owner, and a monitoring mechanism that will evaluate the cooperation behaviour of executing components and compare them with some normal or expected behaviours. None of these mechanisms exist nowadays in Internet, but the current proposal of building the future Internet with virtualised concepts goes in the direction of addressing the first and last needs, as system virtualisation should guarantee fault, performance, and execution isolation, and monitor (or hypervisor or virtual machine monitor) interposition. It is, however, noteworthy that even if we have now monitoring mechanisms in virtualisation kernels, very little is known on the methodology of monitoring to detect abnormalities of networking components. The authentication service is also still subject to discussion. It is not yet clear if a global authentication and/or identity mechanism is mandatory, or only a local, and trust-based scheme will be enough to cover the large spectrum of scenario the future Internet will have to deal with.

The necessity of monitoring results is a major tradeoff between performance and security; the more security we choose, the stricter and the more exhaustive would be the monitoring. This results in a higher share of processing power assigned to monitoring and therefore a loss of performance for the monitored activities. Moreover, monitoring means also to reduce the range of acceptable behaviour to be able to differentiate them from abnormal ones. We have also a tradeoff between flexibility (in term of the range of acceptable behaviour) and security. So, while security nowadays is an important issue in Internet, it seems that security for the future Internet should be considered with a paradigm shift rather than just trying to push existing approaches and mechanisms into the foundations of the new architecture.

4.3. Scalability or the delusion of grandeurs

Another issue that should be considered in the future Internet is scalability. The past decade has seen a mean growth of Internet traffic of almost 100% per year. The size of routing table that is the main indicator of the complexity of the routing operation has seen a yearly growth of 19.4% from 2002 to 2008 [6]. Even if this rate has decreased to 8% during the past two years because of exhaustion of IPv4 address space (that is expected to happen in mid 2011) the growth rate is still considerable. Moreover, mobile Internet revolution and the Internet of things will increase significantly the dimension of the devices connected trough Internet space.

The current Internet has dealt with scalability by using a hierarchical architecture separating the different issues of routing in three different levels. The lowest level deals with local connectivity and configuration of interfaces IP addresses link layer mechanisms (essentially through Ethernet). The second level introduces IP routing between subnets by assuming that the local connectivity is provided inside a network mask. The third level implements operators' policies through BGP filtering and announcement rules, assuming that an AS operator is wise enough to provide optimal connectivity inside itself.

Once upon a time, not so far in 1981, one could read in RFC 790, "*The assignment of numbers is also handled by Jon. If you are developing a protocol or application that will require the use of a link, socket, port, protocol, or network number please contact Jon to receive a number assignment*." Indeed, this situation was not tenable and Regional Internet Registries (RIR) took care of Internet addresses. However, address allocations had to remain compatible with previously allocated addresses. This backward compatibility constraint results in address fragmentation. CIDR (Classless Internet Domain Routing) was an attempt to reduce the burden of the past allocations. Indeed, this leveraged the pressure of address space exhaustion, but it did not solve radically the problem. The IPv6 standard solved radically the issue of address space exhaustion, and gave the impression that with an almost unlimited addressing space, its optimisation is not anymore needed. However, even with IPv6 the source of the scalability problem that was address space fragmentation remained. Moreover, IPv6 showed the difficulty of introducing radical changes into the network. Nearly a decade after most of the IPv6 standard was completed the vast majority of software and hardware still uses IPv4.

IPv6 never answered the cardinal question: "why do we need an address and how can we answer this need?" A trivial answer can be: "We need addresses to do routing." This lead to an even more radical question: "do we need routing?" The advent of Delay Tolerant Networks showed that routing might not be possible in some scenarios. It was even shown that network coding, which is not based on routing, is the forwarding scheme optimising the throughput [7]. More precise investigation shows that IPv6 or IPv4, rather than providing an answer for addressing needs, provides a roughly clever way of indexing the 32 or 128 bits address space. The past decade has seen first attempts at answering the general question of addressing. In these works, addressing was defined as a topological embedding adapted to a particular cooperation need, *i.e.* addressing is a function returning the position of the needed information into a topological space. It was shown that when the embedding is compact, *i.e.* when two close-by items are mapped by the addressing embedding into close addresses, addressing implies routing and *vice–versa*. In other words, if one knows the address of what he wants, he can derive directly from the address the path to reach it. This property means that scalable routing is possible and even trivial, when a compact embedding exists. Indeed, IP (either v4 or v6) addressing is not compact as close nodes are not necessarily close in the IP address space. Nonetheless, compact embeddings exist. For example, Content Addressable Network (with the assumption of no node withdrawal) [8] defines a compact embedding. The question of knowing whether we can embed the particular addressing need of a specific cooperation scheme into a compact embedding is one of the major questions of the Networking Science. For example, very general embeddings can be build that maps an IP like address space into a compact space with some performance costs [9]. Peer–to–Peer (P2P) and overlay networks have demonstrated that by lifting IP addressing backward compatibility constrainsts, scalability can be achieved and address fragmentation avoided. This validates the necessity of having a clean-slate approach rather than an evolutionary approach for developing the future Internet to enable deployment of new addressing/routing schemes. Indeed, one can note that IP addresses is still needed even on P2P or overlay networks. However, this is more a kind of link layer connectivity issue than a fundamental need of IP addressing.

This discussion sheds light on how to ensure the scalability of routing and addressing in future Internet. To summarize, contrary to the current Internet where routing tables are populated only with non-compact IP addresses, the future Internet should enable more flexible routing schemes with the choice of the suitable addressing embedding. Anyway, as explained before, the future Internet would be polymorphic and should simultaneously support execution of different addressing/routing schemes (embeddings), so that classical IPv4/v6 routing and addressing is expected to co-exist with more scalable schemes.

5. CONCLUSION

In this paper, we stated two main positions. First that the future Internet should be polymorphic, meaning that it should enable the coexistence of different networking paradigms in the same framework. I advocated that virtualisation techniques that are nowadays common provide the flexible technology needed for building such a polymorphic future Internet. The second position in this paper is that the future Internet needs a networking science to build strongly its foundations over it. I stated that the essential concept in the scientific approach to network is the concept of cooperation, and I used this concept to analyse some of the important issues I foresee for the future Internet design and deployment. This discussion resulted in some views on security, scalability, and flexibility in the current and the future Internet.

Specifically, I argued that the future Internet would be polymorphic. So the future Internet architecture should take advantage from the flexibility resulting from virtualisation to make possible a polymorphic architecture that adapts to the specific cooperation needs of the network applications. This flexibility will also be mandatory to ensure that new applications and services can be easily deployed in the future Internet, making this platform more attractive than the current Internet for innovation and businesses. The next aspect discussed in the paper was security. My position about security was that the future Internet will need to integrate some security mechanisms in its core architecture. I listed three basic and mandatory mechanisms: a secure execution platform (that could be provided by sandbox virtualisation), an authentication mechanism with an identification scope that has yet to be defined, and a monitoring component that could observe networking (cooperation) activity and eventually filter out all anomalous activities. The last topic developed in the paper was scalability. My position is that recent theoretical works showed that it is possible to construct infinitely scalable addressing and routing scheme by using suitable embeddings. This adds some arguments, to other strong rationales, in favor of a clean slate approach to future Internet design where out-of-the-box thinking with an architectural redesign is possible. Nonetheless, I argued that if future Internet is designed with polymorphism in mind, to clean the slate or not is not anymore a crucial question, as the future Internet should be able to accommodate a completely revolutionary networking architecture as well as a more evolutionary one.

At the end, I would like to thank Serge Fdida, Professor at Paris VI university about interesting discussions about the architecture of Future Internet, and the anonymous reviewers for their valuable advice.

6. REFERENCES

[1] Chip Elliott and Aaron Falk, "An update on the geni project," *SIGCOMM Comput. Commun. Rev.*, vol. 39, no. 3, pp. 28–34, 2009.

[2] Anastasius Gavras, Arto Karila, Serge Fdida, and Martin Potts, "M.potts. future internet research and experimentation," *The FIRE Initiative. ACM Computer Communication Review (CCR*, 2007.

[3] Larry Paterson and Timothy Roscoe, "The Design Principles of PlanetLab," *Operating Systems Review*, vol. 40, no. 1, pp. 11–16, January 2006.

[4] Ignacio Soto, Antonio de la Oliva, Bennoit Donnet, and Thierry Parmentelat, "A multi-homing architecture for onelab," *Paper presented at Real Overlays And Distributed Systems (ROADS) Workshop. (Warsaw, Poland)*, July 2007.

[5] Thomas Anderson, Larry Peterson, Scott Shenker, and Jonathan Turner, "Overcoming the internet impasse through virtualization," *Computer*, vol. 38, no. 4, pp. 34–41, 2005.

[6] G. Huston, "BGP table size," accessed 2010-10-11 online at http://bgp.potaroo.net/index-bgp.html.

[7] Junning Liu, Dennis Goeckel, and Don Towsley, "Bounds on the throughput gain of network coding in unicast and multicast wireless networks," *IEEE J.Sel. A. Commun.*, vol. 27, no. 5, pp. 582–592, 2009.

[8] Sylvia Ratnasamy, Paul Francis, Mark Handley, Richard Karp, and Scott Shenker, "A scalable content-addressable network," in *SIGCOMM '01: Proceedings of the 2001 conference on Applications, technologies, architectures, and protocols for computer communications*, New York, NY, USA, 2001, pp. 161–172, ACM.

[9] Julien Ridoux, Anne Fladenmuller, Yannis Viniotis, and Kavé Salamatian, "Trellis-based virtual regular addressing structures in self-organized networks," in *Proceedings of IFIP Networking*, 2005, pp. 511–522.

INTRODUCING ELASTICITY AND ADAPTATION INTO THE OPTICAL DOMAIN TOWARD MORE EFFICIENT AND SCALABLE OPTICAL TRANSPORT NETWORKS

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ABSTRACT

There is growing recognition that we are rapidly approaching the physical capacity limit of standard optical fiber. It is important to make better use of optical network resources to accommodate the ever-increasing traffic demand to support the future Internet and services. We first introduce an architecture, enabling technologies, and the benefits of recently proposed spectrum-efficient and scalable elastic optical path networks. In these networks, the required minimum spectral resources are adaptively allocated to an optical path based on traffic demand and network conditions. We then present possible adoption scenarios from current rigid optical networks to elastic optical path networks. We also discuss some possible study items that are relevant to the future activities of ITU-T. These items include optical transport network (OTN) architecture, structure and mapping of the optical transport unit, automatically switched optical network (ASON) control plane issues, and some physical aspects with possible extension of the current frequency grid.

Keywords— OTN, ASON, network utilization efficiency, adaptation, spectrum resources, elastic optical path network

1. INTRODUCTION

Optical networks have become widely spread and have assumed a role as mission critical infrastructures for our information society. This is due to worldwide intensive R&D activities and continuous initiative by ITU-T Study Group 15 toward optical transport networks (OTNs) and automatically switched optical networks (ASONs). A 10-Tb/s-class transport system that accommodates 100 Gigabit Ethernet (GbE) interfaces is under development [1]. Such advanced transport systems will probably employ dual-polarization, quadrature-phase-shift-keying (QPSK) modulation, and powerful digital signal processing, which is associated with sophisticated coherent detection, with a spectral efficiency reaching 2 b/s/Hz. Unfortunately, it is well known that bit loading higher than that for QPSK causes a rapid increase in the optical signal-to-noise ratio (OSNR) penalty, while further increase in the launched signal power results in serious impairment due to nonlinear effects in optical fibers. Therefore, it is becoming widely recognized that we are rapidly approaching the physical capacity limit of conventional optical fiber. Considering the forecasted doubling in the amount of Internet-protocol (IP) traffic every two years, the looming physical limit of optical fibers will lead to a capacity crunch in optical transport systems in the not-so-distant future [2].

Under these circumstances, there are two foreseeable issues related to the 100-GbE era and beyond. (1) Will the incremental improvement in transmission technology alone still meet the pace of traffic growth? (2) Will the electrical aggregation and grooming approach still be feasible in terms of cost, footprint, power consumption, and accommodation inefficiency due to stacked layers? Considering that the capacity of conventional optical fibers is not limitless as previously thought but rather a precious network resource, a practical strategy for accommodating ever-increasing traffic demands to support the future Internet and services in 2020 is to take advantage of synergetic effects, *i.e.*, continuous incremental innovation for higher capacity and spectral-efficiency-conscious networking evolution [3].

As a promising approach to address these challenges, we recently proposed to introduce elasticity and adaptation into the optical domain through more flexible spectrum allocation, where the required minimum spectral resources are allocated adaptively based on traffic demand and network conditions [4-5]. The effectiveness of the adaptation approach has been proven in other areas of technology. Link adaptation technology has been employed to increase the spectral efficiency of broadband wireless data networks and digital subscriber lines [6], and virtualization and elastic provisioning of computing resources has been applied to enhance flexibility, scalability, and robustness in emerging super data centers [7]. Facing an impending capacity crunch, the spectral-efficiency-conscious networking approach has attracted growing interest and a number of bandwidth-variable optical network models were investigated [8-13]. The introduction of elasticity and adaptation will be a big leap forward from conventional rigid and fixed optical networks. We, therefore, believe that early initiatives by the ITU-T in cooperation with other standardization bodies on studying possible extension of the OTN and ASON standards in terms of network resource utilization efficiency will greatly support the rapid advance and adoption of more efficient and scalable optical networks.

In this paper, we first present a spectrum-efficient and scalable optical network architecture based on the elastic optical path concept as a key for addressing these challenges. We then describe the enabling technologies for the elastic optical path network: bandwidth-agnostic wavelength crossconnects (WXCs) and rate-flexible transponders. Next, we present possible adoption scenarios from the current rigid optical networks to elastic optical path networks. Finally, we discuss potential study items and candidates from a standards viewpoint including network architecture, the structure and mapping of the optical transport unit, control plane issues, and some physical aspects.

2. ELASTIC OPTICAL PATH NETWORK ARCHITECTURE

2.1. Network architecture based on elastic optical path concept

Figure 1 shows the elastic optical path network concept in contrast to the conventional rigid optical network. The elastic optical path network consists of bandwidth-agnostic WXCs and bandwidth-agnostic reconfigurable optical add/drop multiplexers (ROADMs) in the network core and rate/modulation-format flexible transponders based on, for example, optical orthogonal frequency division multiplexing (OFDM) at the network edge. The aim of the elastic optical path network is to provide spectrally-efficient transport of various client data streams through the introduction of flexible granular grooming in the optical domain [4]. In an elastic optical path network, the required spectral resources on a given route are “sliced off” from the available pool and adaptively allocated to the end-to-end optical path.

It should be emphasized that in elastic optical path networks spectral resources allocated on a given route, through which an optical channel is transported, should be specified in an explicit manner. This can be understood as follows. Since channel spacing in conventional optical networks is fixed (See Fig. 1 (a)), network operators do not need to distinguish the optical channel itself and the spectral resources allocated on a given route. They only need to specify a center frequency for the optical channel when establishing the end-to-end optical connection. In contrast, the center frequency and the width of the spectral resource allocated to an optical path are variable parameters in elastic optical path networks, as shown in Fig. 1 (b). As well as the optical channel itself, network operators need to be aware of the end-to-end spectral resources in elastic optical networks. We hereafter refer to the end-to-end allocated spectral resources as an “optical corridor,” which is specified as a set comprising the center frequency and width, or low-end and high-end frequencies of the optical corridor.

2.2. Enabling technologies for elasticity and adaptation

2.2.1. Bandwidth-agnostic wavelength cross connects

The bandwidth agnostic WXCs can be established using a continuously bandwidth-variable Wavelength selective switch (WSS) as a building block. In a bandwidth-variable WSS, the incoming optical signals with differing optical

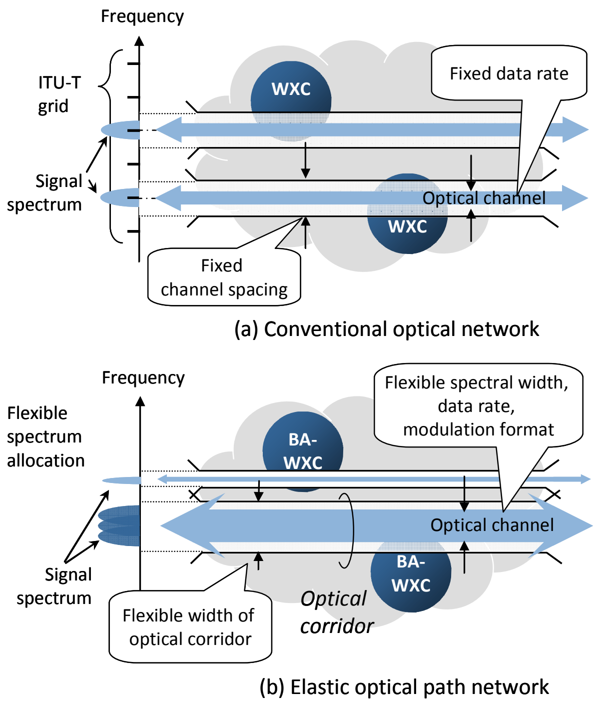

Fig. 1 Elastic optical path network concept and explicit designation of optical corridor

bandwidths and center frequencies can be routed to any of the output fibers. By using spatial phase modulation technology such as liquid crystal on silicon (LCoS), we can achieve variable optical bandwidth functionality. These technologies allow us to allocate the required bandwidth in nodes along the optical path.

2.2.2. Rat and format flexible transponder

Rate and modulation-format flexible transponders can be achieved, for example, by introducing optical OFDM technology. Optical OFDM is optical multiplexing of orthogonal optical sub-carriers that have a frequency spacing equal to the inverse of the symbol duration. The OFDM format allows us to achieve a high level of spectral efficiency and a flexible rate. We can tailor the bandwidth of an optical signal by adjusting the number of sub-carriers of the OFDM signal.

2.2. Benefits of introducing adaptation to the optical domain

2.3.1. Rate adaptive: Adaptation to actual user traffic volume

The network utilization efficiency in conventional optical networks is limited due to the rigid nature of the networks. One limitation originates from the mismatch of granularities between the client layer, which has a broad range of capacity demands with granularities from several to 100 Gb/s or more, and the physical wavelength layer, which has a rigid and large wavelength granularity. For

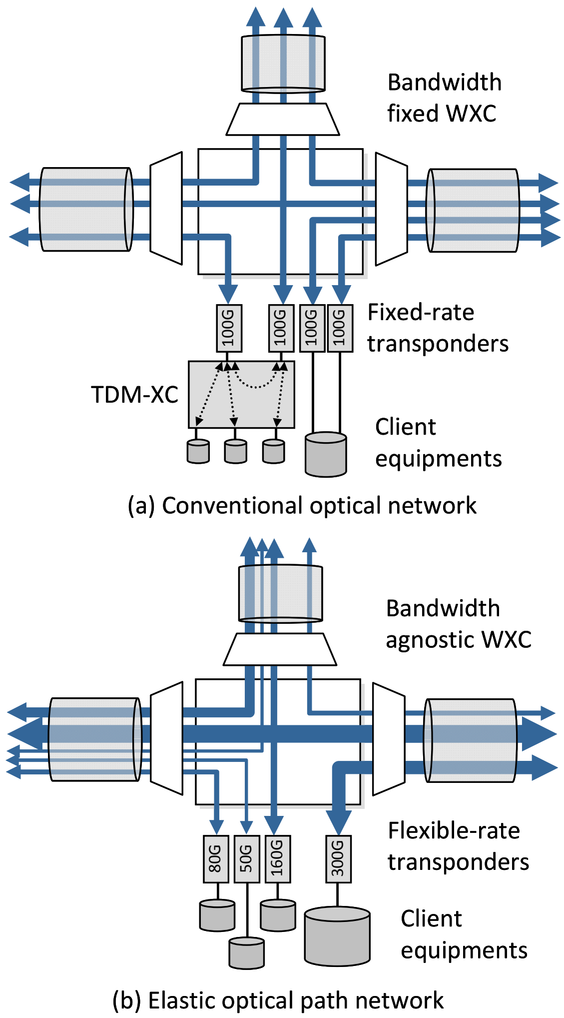

Fig. 2 Rate-adaptive spectrum allocation concept

example, bandwidth stranding occurs when the client traffic volume is not sufficient to fill the entire capacity of a wavelength. Current optical networks mitigate the stranded-bandwidth issue by aggregating and grooming low-bit rate data flow with electrical time division multiplexing (TDM) crossconnects (XCs) or packet transport switches as shown in Fig. 2 (a). However, such a multilayer approach has drawbacks in terms of extra cost, footprint, and power consumption especially in the several tens Gb/s regions and beyond, as well as accommodation inefficiency and complicated operation due to stacked layers. In contrast, if the requested end-to-end capacity is higher than that of the wavelength, several wavelengths are grouped and allocated according to the request as shown in Fig. 2 (a). The adjacent wavelengths in such groups must be separated by a buffer in the spectral domain for wavelength de-multiplexing, and this leads to low spectral efficiency.

Elastic optical path networks mitigate the granularity mismatch problem by dynamically allocating the required minimum spectrum resources in the optical domain as shown in Fig. 2 (b). As a result, they provide efficient, scalable, and future-proof accommodation of sub-wavelength and super-wavelength data traffic according to the actual user traffic volume.

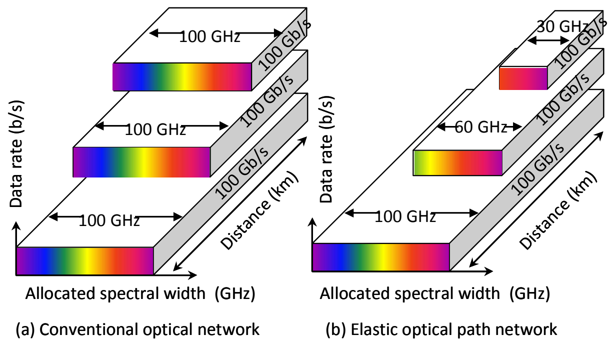

Fig. 3 Distance-adaptive spectrum allocation concept

We investigated the level to which the rate-adaptive elastic optical path network provides an increase in network utilization efficiency of an optical link between two nodes [4]. The case of allocation in the rate-adaptive elastic optical path network based on OFDM is compared to the cases of the fixed bandwidth optical path of 100 Gb/s and inverse-multiplexing where the path is broken into multiple lower bit rate WDM channels. We found that as the average traffic rate is increased, the elastic optical path network exhibits a large advantage over the case of the inverse multiplexed network. The efficiency when a network employing the fixed optical path approaches that of the rate-adaptive elastic optical path network only when the paths are fully utilized. Detailed evaluation results and experimental feasibility demonstrations are given in [4], [14], and [15].

2.3.2. Distance adaptive: Adaptation to physical conditions on the route

Another limitation of network efficiency in the current optical network originates from its worst-case design in terms of the transmission performance. Such design ensures that the worst-case optical path in the network, which usually is the longest path with multiple hops of linear optical repeaters, ROADMs, and WXCs, can be transmitted with sufficient quality. As a result, most optical paths with path lengths that are far shorter than that for the worst case have large unused margins in terms of OSNR, nonlinear impairment, and filter clipping at the receiving end.

By introducing distance-adaptive spectrum allocation, such unused margins for shorter connections can be used to conserve spectrum resources, while ensuring a constant data rate as shown in Fig. 3. A spectrally-efficient but shorter-reach modulation format, *e.g.*, 16-quadrature-amplitude modulation is utilized for shorter optical paths, while an OSNR-degradation tolerant but wider-spectrum modulation format, *e.g.*, QPSK, is employed for longer optical paths. The distance-adaptive spectrum allocation can conserve spectral resources for shorter paths, thus requiring far fewer spectral resources than the current worst-case spectrum allocation.

We evaluated the network utilization efficiency of a distance-adaptive elastic optical path network. A heuristic routing and spectrum assignment (RSA) algorithm with the spectrum-continuity constraint was developed and used in place of the conventional routing and wavelength-assignment (RWA) algorithm [5]. We found that the distance-adaptive elastic optical path network can conserve the required spectrum resources in excess of 45% for a 12 node ring network. A detailed experimental demonstration is given in [16].

2.3.3. Availability adaptive: Adaptation to available bandwidth on the route

Adaptive spectral allocation according to the available bandwidth on the route provides highly survivable restoration through the unique bandwidth variable feature of the elastic optical path [17]. When a link failure occurs and the detour route cannot provide sufficient capacity, we can squeeze the bandwidth of the failed working optical path in order to ensure the minimum connectivity at the expense of the channel bandwidth.

3. POSSIBLE ADOPTION SCENARIOS TOWARD ELASTIC OPTICAL PATH NETWORK

Introducing elasticity and adaptation into the optical domain is expected to yield significant cost savings and enhanced availability associated with the efficient and scalable use of spectral resources in the optical network. One possible adoption scenario for such technology is to be introduced on a step-by-step basis from the link level to network level and from the static level to dynamic level, most likely led by the development of future higher bit-rate OTN interfaces as described hereafter.

Step 1: Spectrally-efficient accommodation of future higher rate data stream
Future OTN line interfaces should transport client signals having a bit-rate higher than 100 Gb/s, for example, 400 Gb/s or even 1 Tb/s. The channel spacing optimized for future OTN line interfaces, which may not be aligned on any of the current ITU-T grids, can accommodate future ultra-high capacity client data-streams in a spectrally-efficient manner.

Step 2: Mixed-rate direct accommodation in the optical domain (Link level)
The next step is link-level direct accommodation of various data-rate client signals in the optical domain. Spectrally-flexible dense wavelength division multiplexing (DWDM) systems achieved by employing bandwidth variable wavelength Mux/Demux and/or flexible rate transponders eliminate electrical aggregation of TDM-XCs or packet transport switches at both ends of a DWDM system. We may reach this step by skipping Step 1.

Step 3: Mixed-rate direct accommodation in the optical domain (Network level)
Once bandwidth-agnostic ROADMs/WXCs are introduced, network-level direct accommodation of various data-rate client signals in the optical domain can be achieved. Electrical grooming at transit nodes employing TDM-XCs or packet transport switches is no longer required. This step leads to cost-effective and simple network operation achieved through collapsing of layers into a single optical layer.

Step 4: Distance-adaptive modulation and spectrum allocation
Evolutional change in designing and planning optical networks will be achieved by adopting distance-adaptive modulation and spectrum allocation. This improves spectral efficiency through significant savings of spectral resources at the network level. Thus, cost-effective accommodation of future Internet traffic and services can be achieved.

Step 5: Dynamic spectrum allocation
The final step is dynamic spectrum resource allocation. Optical bandwidth-on-demand services and cost-effective high-availability transport services will be achieved through sophisticated operation based on the optical version of link capacity adjustment scheme (LCAS) and bandwidth-squeezed highly-survivable restoration technologies.

An earlier adoption possibility may be the introduction of distance-adaptive spectrum allocation concept, which is described as Step 4 in the step-by-step scenario, to achieve cost-effective 100-Gb/s-class ROADM systems. This situation is a consequence of freeing ourselves of the worst-case physical design.

4. POTENTIAL STANDARDIZATION ITEMS

Since the transition from current rigid networks to elastic and adaptive optical networks will be a significant leap forward, we believe that early initiatives by the ITU-T will be absolutely indispensable. Clarifying what should be inherited, what should be extended, and what should be created is imperative as the stating point regarding studying the possible extension of OTN and ASON standards in terms of network efficiency. In the following, we present potential study items and some candidates from a standards viewpoint.

4.1. OTN Network architecture

ITU-T Recommendation G.872 “Architecture of optical transport networks” specifies the functional architecture of OTN from a network level viewpoint. G.872 defines an optical network layered structure that comprises an Optical Channel (OCh), Optical Multiplex Section (OMS), and Optical Transmission Section (OTS). Although the data rate, modulation format, and spectral width of an optical path in an elastic optical path network may change according to the user demand and network conditions, an elastic optical path is naturally mapped into the OCh of the current OTN layered structure. We, therefore, see no significant impact on the current G.872 when introducing the elastic optical path concept.

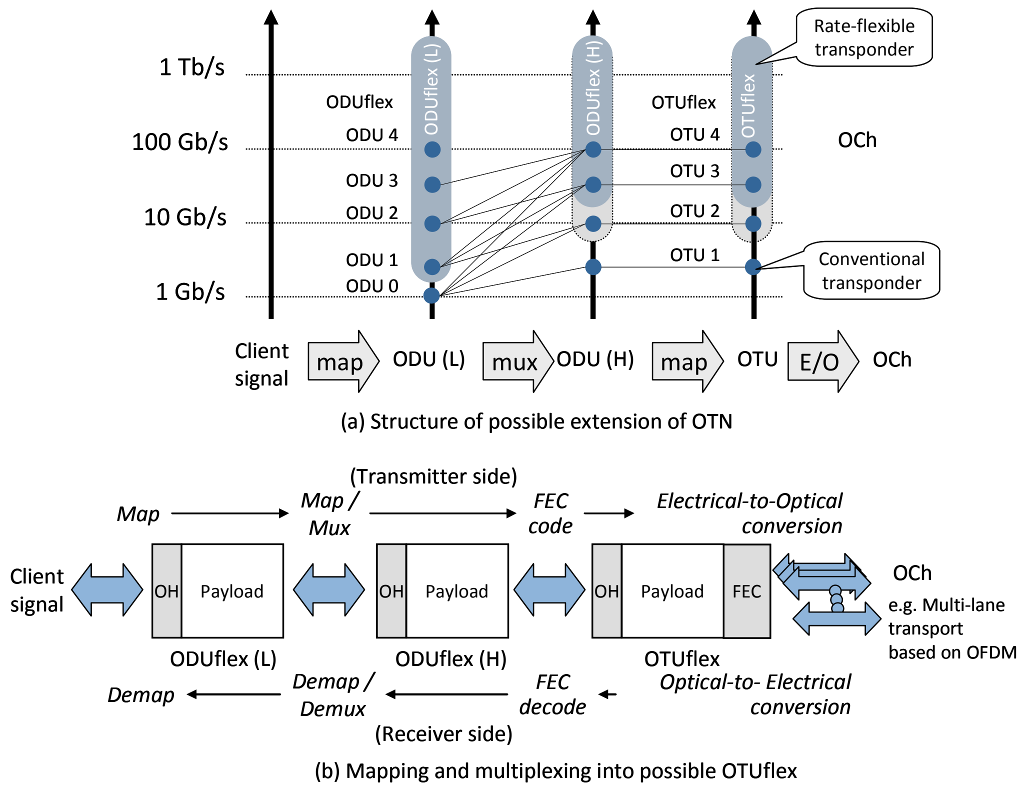

Fig. 4 Structure and mapping of elastic optical path network

4.2. OTN mapping and multiplexing

The interfaces and mappings of OTN are specified in ITU-T Recommendation G.709 "Interfaces for the optical transport network (OTN)." The OTN can accommodate various client signals and transport them over long distances. Originally the OTN specified client signal mapping into ODUk (k=1, 2, 3), which have bit rates of approximately 2.5 Gb/s, 10 Gb/s, and 40 Gb/s, and their multiplexing to ODUk with a higher bit rate if necessary. The multiplexed ODUk signal is then transported as an OTUk signal with a forward error correction (FEC) code.

A new ODU was recently specified in G.709 called ODUflex, which can have any bit rate, to accommodate any client signal efficiently. ODUflex must be multiplexed into ODUk (k=1, 2, 3, 4) with higher bit rates. In addition to the specification of ODUflex, the concept of a Lower Order (LO)/Higher Order (HO) ODU was introduced. The LO ODU accommodates client signals and LO ODUs are multiplexed into the HO ODU. Although network operators should transport a wide variety of client signals, they must keep the number of kinds of line-interfaces to as few as possible in order to reduce the capital expenditures, which are dominated by line-interface costs. The concept of the LO/HO ODUs can address these conflicting requirements. The LO ODU can have many kinds of bit rates in order to accommodate various client signals efficiently. Actually the LO ODU can have any bit rate because the ODUflex is one of the LO ODUs. On the other hand, the HO ODU has fewer kinds of bit rates. Now four kinds of ODUk (k=1, 2, 3, 4) are HO ODUs. As a result, the OTU also has four kinds of OTUk (k=1, 2, 3, 4). The LO ODU offers versatility to accommodate various client signals and the HO ODU offers simplicity in terms of the physical interfaces.

Once rate-flexible OCh based on optical OFDM transponders and bandwidth-agnostic ROADMs/WXCs is introduced, cost-effective transport of various client signals will be enabled in the fully optical domain without intermediate electrical multiplexing and grooming processes as described in previous sections. As a natural step toward a rate-flexible OCh, we may need to consider some extension of G.709. One possibility would be to introduce rate-flexible OTUs (OTUflex) as well as rate-flexible HO ODUs (HO ODUflex) as shown in Fig. 4 (a). The OTUflex and HO ODUflex will be specified in the region of over 10-100 Gb/s depending on the maturity of device technology at the time. Figure 4 (b) shows the client signal transport over OTN with a rate-flexible transponder. On the transmitter side, the client signal is mapped into the LO ODUflex, and then mapped/multiplexed to the HO ODUflex. An FEC code is added to the HO ODUflex to create OTUflex, which is envisioned to be mapped into the optical data stream by using a flexible rate transponder employing, for example, multi-lane transport based on optical OFDM. On the receiver side, reverse processing is performed to extract the client signal from the OTUflex frame. Another relevant standard that should be investigated for possible extensions is Recommendation G.798 "Functional characteristics of optical networking

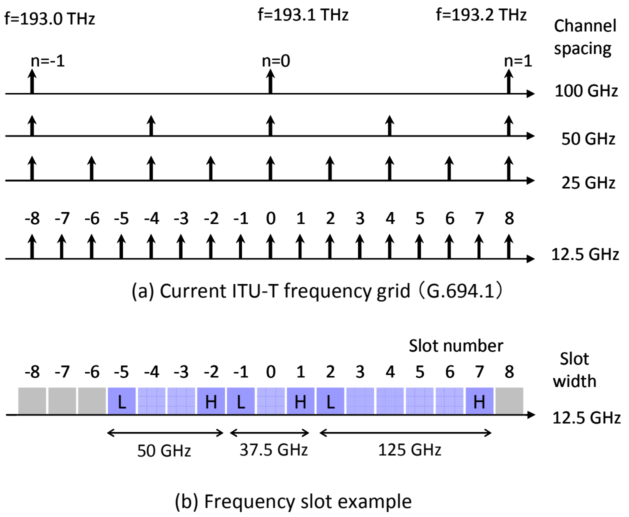

Fig. 5 Flexible spectrum resource designation example

equipment" which specifies the details regarding the OTN equipment.

4.3. ASON control plane

The ITU-T Recommendations on ASON provide requirements, architecture, and protocol neutral specifications for automatically switched optical networks with a distributed control plane [18]. The goal is not to define new protocols but to provide mappings between abstract protocol specifications and the existing candidate protocols. Development of new protocols such as generalized multiprotocol label switching (GMPLS) has been tasked by the Internet Engineering Task Force (IETF) and Optical Internetworking Forum (OIF). The ITU-T Recommendation G.8080 "Architecture for the automatically switched optical network (ASON)" defines fundamental functional components of the control plane. Three major processes in the ASON control plane are call and connection control, path control based on the dissemination of the network state information, and the discovery process for network self-configuration. Their protocol neutral specifications are provided in the ITU-T Recommendations G.7713, G.7714, and G.7715, respectively.

The ASON network resource model is based on a generic functional model for transport networks defined in G.805, functional models for Synchronous Digital Hierarchy (SDH) defined in G.803 and OTN defined in G.872. We have already examined G.872 in subsection 4.1, then again, as a result of preliminary investigation we consider that there will be no significant impact on the current ASON standards when introducing a distributed control plane into elastic optical path networks, although further studies are still necessary. As for the technology-specific aspects of routing and signaling in elastic optical path networks, we should discuss possible extension of GMPLS protocols in the IETF and the OIF in close cooperation with ITU-T Study Group 15, considering the explicit resource specification of the optical corridor width for example.

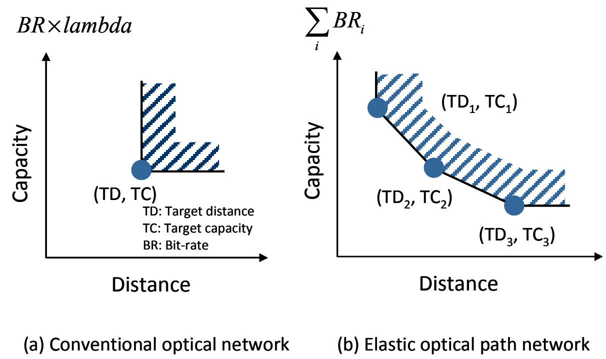

Fig. 6 Possible specification based on longitudinal compatibility approach

4.4. Physical aspects

The current ITU-T frequency grid specified in G.694.1 "Spectral grids for WDM applications: DWDM frequency grid" is anchored to 193.1 THz, and supports various channel spacings of 12.5 GHz, 25 GHz, 50 GHz, and 100 GHz as shown in Fig. 5 (a). In order to utilize fully the spectrally-efficient and scalable nature of elastic optical path networks, we may need to consider some extension of G.694.1 [3], [9]. From a practical viewpoint, one promising way would be to quantize the continuous spectrum into contiguous frequency slots with an appropriate slot width [3, 5]. The following is a possible candidate for the flexible spectrum resource designation scheme based on the frequency slot concept. In this scheme, the designated frequency f on the ITU-T frequency grid using channel spacing f_{cs} and frequency number n should be interpreted as a spectral slot having a frequency segment between $193.1+(n\pm1/2)f_{cs}$ (THz), as shown in Fig. 5 (b). ITU-T frequency number n corresponds to the frequency slot number. Spectral resources of an optical path can be allocated by assigning the necessary number of contiguous frequency slots.

In order to proceed in specifying other required physical layer specifications, we should pursue the following strategy. In general, physical layer standardization takes one of two approaches: transverse compatibility or longitudinal compatibility. The recommendation supporting multi-vendor transversal compatibility covers almost all the physical parameters and can usually be achieved when the relevant technologies are sufficiently mature. On the other hand, the recommendation supporting single vendor longitudinal compatibility requires the minimum number of parameters, *e.g.* target distance and repeater spacing, that usually have a strong relation to carrier requirements, and are relatively easily specified. Considering the advanced functionalities that we are trying to achieve, it is natural to start with the longitudinal compatibility approach for physical layer specifications of elastic optical path networks. Whereas, in conventional systems, the target distance and capacity are a fixed set of values as shown in Fig. 6 (a), in the elastic optical path network, there can be variable sets of parameters, as shown in Fig. 6 (b). The sets can be optimized, for example, according to the carrier requirements. This unique feature may bring additional degrees of freedom in defining the physical layer specifications and result in capital expenditure reduction.

5. CONCLUSIONS

We discussed an architecture, enabling technologies, and benefits of the elastic optical path network where the required minimum spectral resources are adaptively allocated to an optical path based on various network conditions including actual client traffic demand, physical network conditions, and the available bandwidth on the route. We presented possible adoption scenarios from current rigid optical networks to elastic optical path networks. One possible scenario is to introduce elasticity and adaptation on a step-by-step basis from the link level to network level and from the static level to dynamic level, most likely lead by the development of future higher bit-rate OTN interfaces. An earlier adoption possibility may be the introduction of distance-adaptive spectrum allocation to achieve cost-effective 100-Gb/s-class ROADM systems. We presented some possible study items that are relevant to the future activities of ITU-T Study Group 15. As the stating point for studying the possible extension of OTN and ASON standards in terms of network efficiency, we presented some clarification on what should be inherited, what should be extended, and what should be created. We also discussed some candidates in terms of structure and mapping of the OTU and some physical aspects with possible extension of the current frequency grid.

We hope that this paper contributes to ITU-T in future standardizations and revisions, particularly in the context of a more efficient and scalable optical layer as a mission critical infrastructure to support the future Internet and services.

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REFERENCES

[1] Y. Miyamoto and S. Suzuki, "Advanced Optical Modulation and Multiplexing Technologies for High-Capacity OTN Based on 100 Gb/s Channel and Beyond," *IEEE Comm. Mag*. Vol. 48, Issue. 3, pp. S65-S71, 2010.

[2] R.-J. Essiambre, G. Kramer, P. J. Winzer, G. J. Foschini, and B. Goebel, "Capacity Limits of Optical Fiber Networks," *J. Lightwave Technol.*, Vol. 28, No. 4, pp. 662-701, 2010.

[3] M. Jinno, H. Takara, and B. Kozicki, "Dynamic Mesh Networking: Drivers, Challenges, and Solutions for the Future," *ECOC 2009*, 7.7.4, 2009.

[4] M. Jinno, H. Takara, B. Kozicki, Y. Tsukishima, Y. Sone, and S. Matsuoka, "Spectrum-Efficient and Scalable Elastic Optical Path Network: Architecture, Benefits, and Enabling Technologies," *IEEE Comm. Mag*., Vol. 47, Issue 11, pp. 66-73, 2009.

[5] M. Jinno, B. Kozicki, H. Takara, A. Watanabe, Y. Sone, T. Tanaka, and A. Hirano, "Distance-Adaptive Spectrum Resource Allocation in Spectrum-Sliced Elastic Optical Path Network (SLICE)," *IEEE Comm. Mag*., Vol. 48, Issue. 8, pp. 138-145, 2010.

[6] S. Catreux, V. Erceg, D. Gesbert, and R. W. Heath, Jr., "Adaptive Modulation and MIMO Coding for Broadband Wireless Data Networks," *IEEE Comm. Mag*., pp. 108-115, June 2002.

[7] A. Weiss, "Computing in the Clouds," *netWorker*, 11(4) pp. 16-25, 2007.

[8] Q. Yang, W. Shieh, and Y. Ma, "Bit and Power Loading for Coherent Optical OFDM," *IEEE Photon. Technol. Lett*., Vol. 20, No. 15, pp. 1305-1307, 2008.

[9] A. Gumaste and N. Ghani, "Reach Optimized Architecture for Multi-Rate Transport System (ROAMTS): One Size Does Not Fit All," *OFC/NFOEC 2009*, OMQ3, 2009.

[10] A. Klekamp, O. Rival, A. Morea, R. Dischler, and F. Buchali, "Transparent WDM Network with Bitrate Tunable Optical OFDM Transponder," *OFC/NFOEC 2010*, NTuB5, 2010.

[11] R. Dischler, F. Buchali, and A. Klekamp, "Demonstration of Bit Rate Variable ROADM Functionality on an Optical OFDM Superchannel," *OFC/NFOEC 2010*, OTuM7, 2010.

[12] W. Zheng, Y. Jin, W. Sun, W. Guo, and W. Hu, "On the Spectrum-Efficient of Bandwidth-Variable Optical OFDM Transport Networks," *OFC/NFOEC 2010*, OWR5, 2010.

[13] O. Gerstel, "Flexible Use of Spectrum and Photonic Grooming," *IPR/PS 2010*, PMD3, 2010.

[14] M. Jinno, H. Takara, B. Kozicki, Y. Tsukishima, T. Yoshimatsu, T. Kobayashi, Y. Miyamoto, K. Yonenaga, A. Takada, O. Ishida, and S. Matsuoka, "Demonstration of Novel Spectrum-Efficient Elastic Optical Path Network with Per-Channel Variable Capacity of 40 Gb/s to over 400 Gb/s," in *Pro*c. ECOC 2008, Paper Th3F6, 2008.

[15] B. Kozicki, H. Takara, Y. Tsukishima, T. Yoshimatsu, K. Yonenaga, and M. Jinno, "1 Tb/s Optical Link Aggregation with Spectrum-Sliced Elastic Optical Path Network SLICE," *ECOC 2009*, 8.3.4, 2009.

[16] B. Kozicki, H. Takara, Y. Sone, A. Watanabe, and M. Jinno, "Distance-Adaptive Spectrum Allocation in Elastic Optical Path Network (SLICE) with Bit per Symbol Adjustment," *OFC/NFOEC 2010*, 2010.

[17] Y. Sone, A. Watanabe, W. Imajuku, Y. Tsukishima, B. Kozicki, H. Takara, and M. Jinno, "Highly Survivable Restoration Scheme Employing Optical Bandwidth Squeezing in Spectrum-Sliced Elastic Optical Path (SLICE) Network," *OFC/NFOEC 2009*, OThT2, 2009.

[18] S. Tomic, B. Statovci-Halimi, A. Halimi, W. Muellner, and J. Fruehwirth, "ASON and GMPLS—Overview and Comparison," *Photonic Network Communications*, Vol. 7, No. 2, pp. 111-130, 2004.

INTRODUCING MULTI-ID AND MULTI-LOCATOR INTO NETWORK ARCHITECTURE

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ABSTRACT

The present day Internet has no separate namespace for host IDs. It uses IP addresses as host IDs, which are in fact locators. This dual role is problematic for mobility, multihoming, security, and routing on the Internet. To solve these problems, research has recently begun on ID/locator split architectures. Some standardization activities based on this concept are also progressing in ITU-T Study Group 13 and in the IETF. We expect that introduction of the ID/locator split concept into the new generation network or future Internet architecture can bring about additional functions, such as heterogeneous network protocol support, multicast, QoS, resource or service discovery, and flexible human-network interaction. Toward realization of these functions, this paper presents a study on an approach of introducing multi-ID and multi-locator support into the network architecture. The paper also lists items that have the potential to be standardized in ITU-T.

Keywords— new generation network, future Internet, network architecture, ID/locator split, multi-ID, multi-locator

1. INTRODUCTION

This paper aims at contributing to the activities of the International Telecommunication Union – Telecommunication Standardization Sector (ITU-T) Study Group 13 (SG 13) to develop standards for future telecommunication networks [1]. ITU-T SG 13 has established Question 21 (Q.21) to study the requirements of future networks and gradually standardize their functional architecture. These activities are stimulated by a great deal of research and experimentation on future networks progressing in Asia (AKARI Project in Japan [2] and Future Internet Forum in South Korea [3]), Europe (FIRE [4]), and the United States (FIND [5] and GENI [6]).

The current packet transport networks, both the Internet and Next-Generation Network (NGN), are based on two kinds of namespaces: domain names and IP addresses. Internet applications resolve the domain name into an IP address during a communication initialization phase via a domain name system (DNS) lookup. The IP address is then used by the networking protocols to identify communication sessions and locate the destination host in the network. However, there are problems in the use of IP addresses in the host protocol stack. Namely, an IP address is used in the network layer protocols as a *locator* to locate the destination host and forward packets toward it. The same IP address is also used in the transport and upper layer protocols as the *host identifier* (*ID*). This dual role of IP addresses as host IDs and locators makes it difficult to design efficient solutions for mobility, multihoming, security, and routing because such solutions require the ability to change locators used in the network layer without changing the host IDs used in the transport and application layers.

To eliminate the problems caused by the overloaded semantics of IP addresses as both host IDs and locators, different approaches of introducing the ID/locator split concept into network architectures have recently been discussed [7-14]. ID/locator split architecture uses distinct sets of values for host IDs and locators and allows the network layer to change locators without requiring the upper layers to change IDs.

In standards development organizations (SDOs) such as ITU-T SG 13 [1] and the Internet Engineering Task Force (IETF) [15], proposals for developing network architectural functions based on the ID/locator split concept have been progressing. To date, discussions in the IETF have been on enhancing mobility [13], multihoming [11], and routing [14] architectures. Similarly, in ITU-T SG 13, several recommendation drafts are progressing with the objective of enhancing mobility, multihoming, and routing in NGN, while ITU-T Recommendation Y.2010, which outlines the general requirements of ID/locator split in NGN [7], was approved last year.

Since network architectures incorporating the concept of ID/locator split are progressing in SDOs, additional functionalities need to be investigated that can easily be brought about by extending the concept to the new generation network or future Internet architectures. (Note: This paper uses the terms “new generation network,” “future Internet,” and “future network” interchangeably.) Therefore, with the objective of generating more work items for standardization, this paper presents a study on additional functions such as supporting heterogeneous network protocols, multicast, quality of service (QoS), resource or service discovery, and flexible human-network interaction that can be realized by the introduction of multi-ID and multi-locator into the network architecture. Multi-ID and multi-locator supporting architecture specifies service requirements through the host IDs. As the packets

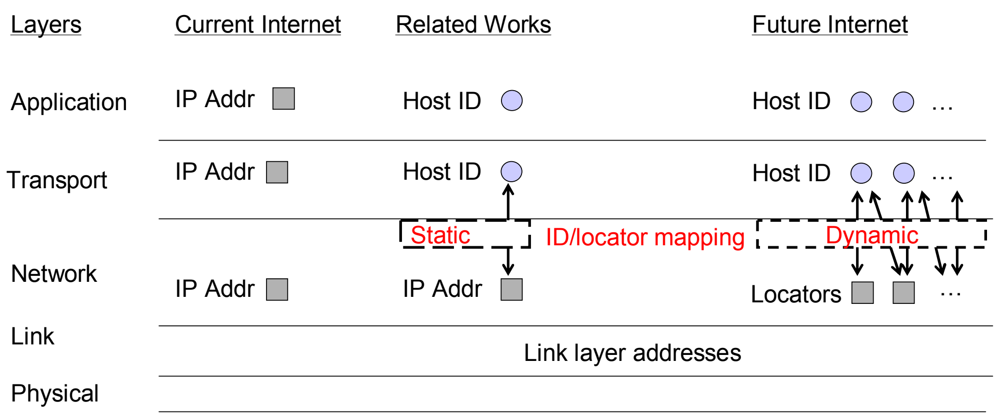

Fig. 1 Host IDs and locators in the current and future Internet protocol stack

containing the host IDs in their headers pass through the network nodes, such as gateways, the host IDs are mapped to appropriate network protocols and locators, which provide the communication modes (e.g., unicast, multicast, geo-cast) or paths (e.g., bandwidth, delay guaranteed) that optimally fulfill the service requirements. The latter part of the paper lists some research items that have the potential to be standardized in ITU-T.

This paper is organized as follows. Section 2 elaborates on the reasons for introducing multi-ID and multi-locators into the new generation network architecture. Section 3 presents a service scenario that would be feasible over the multi-ID and multi-locator architecture. Section 4 discusses the items for research and standardization and section 5 concludes the paper.

2. MULTI-ID AND MULTI-LOCATOR

This section discusses the reasons for introducing multiple host IDs and locators into network architectures. Fig. 1 compares the host IDs and locators used in the current Internet, related works, and future Internet.

As stated in the preceding section and shown in the figure, the current Internet has no separate namespace for host IDs. It uses IP addresses as host IDs in the transport and application layers.

In related research on ID/locator split architectures [13, 14], only a single type of host IDs is considered. The host ID to locator mapping is static. These host IDs cannot convey the service requirements from the application layer to the network layer.

2.1. Multiple Host IDs in the Future Network

The new generation network or future Internet architecture is required to support different types of host IDs for the following reasons:

(a) Network-protocol independent multicast, group-cast or geo-cast: In the current Internet, multicast applications use multicast IP addresses (i.e., locators) to utilize the multicast functions of the network. However, in ID/locator split architecture, the locators are not visible to the application layer; only host IDs are visible. Therefore, the host IDs should have the ability to represent not only individual hosts but also a collection of hosts. Such host IDs can represent a group of hosts that satisfy certain conditions; e.g., located in the same physical network, same logical network, same geographical area, or interested in the same service.

(b) Service differentiation: In the current Internet, only the transport layer protocol's port numbers are available to differentiate services, which are not sufficient for finely specifying service requirements in the network. For example, all services using the HTTP protocol are transported over port 80, which is not sufficient for distinguishing services in the network for QoS guarantee or for other purposes. The ID/locator split architecture can address this problem by designing different types of host IDs that can be used to discriminate between flows.

(c) Optimal network or path selection: Through the use of a particular type of host ID, applications can instruct the network layer to select specified protocols or networks that optimally satisfy the service requirements in a heterogeneous networking environment. The host IDs are mapped dynamically to different network layer protocols or locators for forwarding packets through preferred paths or networks; not only in hosts but also in network nodes, such as border routers or gateways.

(d) Private and public communication: Since host IDs are present in packet headers, private IDs that are unknown to unwelcomed parties are necessary for avoiding session tracing as well as protecting privacy in communication. However, the use of private IDs makes ID/locator mapping in the network (which is useful for routing and mobility) difficult. Thus we need to configure both public and private host IDs that can be used in different networking environments for different service requirements (e.g., public host IDs registered in gateways are used when gateway-assisted network-based mobility support is required, and private host IDs not registered anywhere are used when privacy or anonymity is important).

(e) Dynamic relationship between the user, service, and host: Support for multiple IDs is in demand in ubiquitous network society for identifying entities differently in different situations and connecting real society with the virtual network space. To create a flexible relationship between societal and network entities, it is important to have provisions of dynamically relating host IDs with other IDs such as service IDs, user IDs, and group IDs, and locators. Possession of multiple host IDs is helpful for this purpose.

In summary, multiple types of host IDs are required so each can be used to represent different service requirements. These host IDs play the role of media in carrying service requirements from the application layer to the network layer. The network layer handles the communication accordingly by selecting different locators and providing different communication modes.

2.2. Multiple Locators in the Future Network

The new generation network is required to support multiple locators as well as dynamic mapping between various host IDs and locators for the following reasons:

(a) Mobility: The new generation network must have a mobility-oriented architecture because most hosts connected to the network will be mobile. Mobility support requires hosts to possess multiple locators at the same instance or different instances of time. If the host possesses two or more locators simultaneously, a make-before-break type of seamless handover is possible. If the host possesses only one locator at a time, then a break-before-make type takes place. In both types of handovers, host IDs play the important role of relating the locators of before and after mobility together and continuing the ongoing communication sessions.

(b) Multihoming: This enables users to connect to different types of networks through different locators associated with different interfaces. Multihoming is currently in demand since it brings numerous advantages to users and service providers, such as cost-effective network selection, reliability, and ubiquitous connectivity, while generating fair business competition among the providers. For example, it is common for mobile phones to possess several interfaces such as 3G cellular, WiFi, Bluetooth, and WiMAX, and users can select an appropriate network for the given service (e.g., WiFi for web access and 3G cellular for voice conversation). Moreover, transferring a communication session from one network to another is essential for providing ubiquitous access (e.g., data downloading via WiFi at home is transferred to WiMAX when moving outside the home).

(c) Routing: To make routing scalable, it is important to keep the core network's routing functions independent of the edge network's addressing size and route update frequency. The use of different locator spaces in the core and edge networks with locator translation at the border between edge and core networks helps hide the core routing function from the changes in the edge networks' configurations. The use of host IDs allows referring of the host by different locators in the edge and core networks.

(d) Delay or disruption tolerant network (DTN): Like routing, DTN also requires referring of hosts by different locators in different parts of the network because of uncertainty in the end-to-end path availability.

(e) Heterogeneous network protocols support: The new generation network is required to support heterogeneous network layer protocols in different parts of the network for integrating existing and new networks and for leaving space for the development of new protocols suitable for emerging communication environments. These protocols may use different locators; e.g., IPv4 and IPv6 use different address (locator) spaces. Multiple locator support is therefore inevitable for the new generation network and its interoperability with other networks.

(f) Privacy: Referring of hosts by different locators is required so that a host's location or identity cannot be traced by an unwelcomed party.

Since we are heading toward designing disruptive technologies for the new generation network, we should take this opportunity to design different types of host IDs and the framework for dynamically mapping host IDs with other entities, such as service IDs, user IDs, group IDs, and locators. The framework should be able to keep the new generation network extendable so that it can easily accommodate future application requirements. We attempt to address these issues in this paper since there is no related research, to our knowledge, about the formation and mapping of such host IDs.

3. SERVICE SCENARIO FOR FUTURE INTERNET

This section presents a service scenario for the new generation network and elaborates on how the use of multi-IDs and multi-locators helps satisfy service requirements. While presenting the service scenario, it is assumed that the following functions are available in the new generation network:

(a) Simple user-interface: Hosts have a simple user interface where the user can provide descriptions of the intended service.

(b) Multiple host IDs: Hosts have multiple host IDs. Applications select appropriate host IDs on the basis of service requirements.

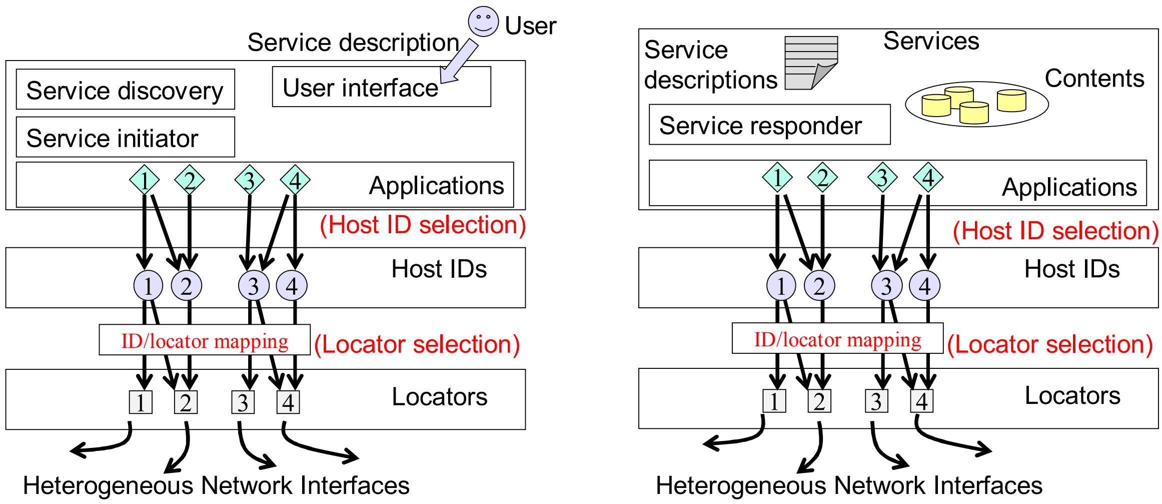

Fig. 2 (a) Client host and (b) server host functions in multi-ID and multi-locator architecture

(c) Multiple network interfaces: Hosts have multiple interfaces through which they can connect to different access networks. Each interface is assigned one or more locators. Host IDs are mapped to different locators depending on the service requirements.

(d) Various communication modes supported: The network can support various types of communication modes such as group communication, private communication, QoS-supported communication, and heterogeneity-aware communication. Host IDs, as well as locators present in packet headers, specify the intended communication mode for the service.

Fig. 2 (a) and (b) show the client and server host functions, respectively, that utilize multiple host IDs and locators to optimally satisfy service requirements. The service initiation process is as described below.

1. From the client host, the user issues a service request by providing service descriptions, in the form of service name and QoS requirements, via the user-interface. These descriptions specify the requested network resources in the form of content, connectivity or bandwidth, computing, storage, etc.
2. The service discovery function reads the user-interface and finds the corresponding service by querying a service registry in the network (not shown in the figure). The registry contains a list of service descriptions made available by different service providers. The service registry responds to the service discovery function with a list of the requested services that can be provided by different service providers. The service discovery function selects the best service on the basis of the user requirements and gives the selected service's logical locator to the service initiator function.
3. The service initiator function in the host selects its host ID that matches with the service description. It then sends a service request to the server. Based on the client host ID, the ID/locator mapping function selects the host's appropriate locator for utilizing the optimal access network for the service.
4. The service responder located in the server host (shown in Fig. 2(b)) selects the host ID suitable for the requested service description. The ID/locator mapping function then selects the host locator for using the appropriate physical network and communication mode for the service.
5. Once both the server and client hosts have each other's host IDs and locators they communicate by exchanging data packets containing host IDs and locators in the packet headers. In this case, the communication session is not affected even if the network configuration changes due to mobility, multihoming, route change, or other effects because the ID/locator mapping functions hide the impact of locator change from the application layer.

In order to realize the service scenario listed above, there are a number of issues to be resolved. In the next section we discuss these issues as research and standardization items of the new generation network.

4. ITEMS FOR RESEARCH AND STANDARDIZATION

This section mainly discusses research and standardization of service representation, host ID configuration, host ID to locator mapping mechanisms, and other items such as service discovery.

4.1. Service Representation

Unique representation of the combination of service type and quality is necessary for the service scenario described in the previous section. Such representation will help the host to select an appropriate host ID that will instruct the network to provide the intended QoS. Fig. 3(a) shows a proposal of the service representation, which we call a *service logical locator* (SLL). The SLL contains four fields: application name, host name, service content name, and a

Service_Logical_Locator format:

application_name | host_name | service_content _name| list{parameters=value}

e.g., video: | server1#providerA.com | titanic-1997-film | bw=10Mbps, delay=10ms

(a)

Host_ID format:

organization_prefix | scope | service_code | version | host_specific_suffix

e.g., 1FFF:0001 :FF 01 : 0001 :3EA3:82D2:B948:B35C

(b)

Fig. 3 (a) Service logical locator and (b) host ID formats

list of parameters of the QoS. The application name represents the application protocol used by the service, like HTTP or FTP, of the current Internet. We expect there will be several other protocols that more effectively support new applications such as video streaming, voice conversation, social networking, or games. The host name contains the name of the server holding the service. This name (e.g., server1#providerA.com) is formed by combining the host's local name (e.g., server1) and global domain name (e.g., providerA.com), as in [8]. The parameter list mentions the intended QoS requirements such as minimum bandwidth and maximum end-to-end delay, as well as mobility or heterogeneous network protocol support during the communication session.

4.2. Host ID Configuration

Since the host IDs have to convey service requirements from the application layer to the network layer for providing the appropriate communication support demanded by the application, comprehensive host IDs need to be designed that that can represent a wide range of service requirements. Fig. 3(b) shows a proposal of such host IDs (referenced from [9]). The host ID contains the following fields: organization prefix, scope, service code, version, and host-specific suffix. The globally unique organization prefix is assigned to each organization with which the hosts associate. Such prefixes make the host IDs hierarchical and help to simplify the host IDs' resolution (i.e., maintaining and retrieving locators or other information corresponding to the host IDs) system. The scope field indicates the scope of the host ID, such as global, local, private, public, individual, or group. Similarly, the service code represents the service quality intended by the application and version represents the version of the host ID. The host-specific suffix indicates a locally unique value for the host, which can be generated by hashing the hostname and other parameters. The use of hash value as the host-specific suffix is helpful for relating the host ID with the hostname as well as for introducing some randomness in the host ID value.

4.3. Host ID to Locator Mapping Functions

Both the host and gateway can possess host IDs to locator mapping functions, which are located in the identity sublayer [7, 9], as shown in Fig. 4. The mapping functions select appropriate locators based on the host IDs present in packet headers. The mapping function located in the host selects the locator (associated with an interface) that belongs to the network, which optimally satisfies the service requirements represented by the host IDs. For example, to have video-streaming communication, a mobile host may prefer a WiMAX or WiFi network to the 3G cellular network. Similarly, the mapping function located in the gateway can map host IDs to different network layer protocols and locators (also called late-binding) for supporting heterogeneous networks, network-based mobility, traffic engineering, site multihoming, dynamic route change, etc. The network-based mapping function is

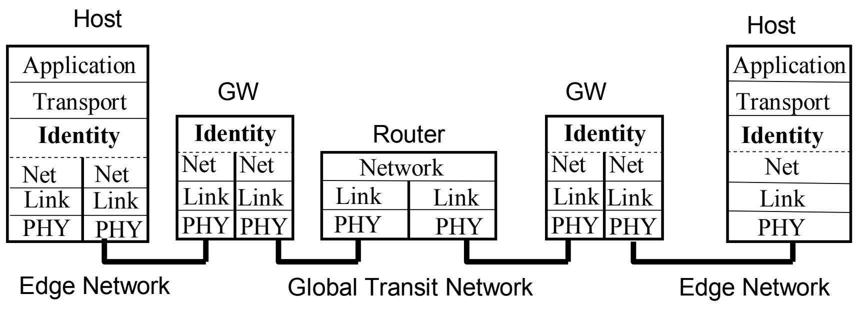

Fig. 4 Protocol stack of ID/locator split architecture.

also important for supporting different communication modes such as multicast, geo-cast, and group-cast.

4.4. Service Discovery and Other Issues

The service discovery function mentioned in the preceding section requires service registries, which hold the lists of service descriptions made available by different service providers. The means of providing service registry functions in the network is another research and standardization item. The service registry may be like the current DNS that provides various resources such as IP addresses and encryption keys related to domain names. Alternatively, the service registry functions may be provided through a search engine such as Google or Yahoo. The other issue is related with finding the server's physical network locator from the given SLL. For this purpose, as referenced from [9], the hostname field of SLL can be resolved from a logical control network that holds mapping information between the hostname and physical network locator of the server. Research needs to be conducted on different architectures of the logical control network to efficiently store, update, and retrieve host IDs to locator mapping information.

Having multi-ID and multi-locator support in the network architecture also promotes fair business competition among the network service providers and content service providers, while allowing users to enjoy cost-effective services from different service providers. For this purpose, the host IDs are required to be assigned independent of network or content service providers. A semi-government organization or national SDO such as TTC (http://www.ttc.or.jp) in Japan or TTA (http://www.tta.or.kr) in South Korea, or an international SDO such as ITU, can administer (or delegate) the host ID prefix assignment task so that none of the commercial service providers can monopolize the business. The locators, on the other hand, are associated with the network service provider with which the host has a service contract. Users can register their different host IDs with different network service providers, specifying the service requirements associated with each host ID.

The above are the issues that need broader research in order to develop a common understanding and make a plan for their standardization. We also need collaboration between different SDOs such as the IETF, ITU, 3GPP, ISO, and ETSI for developing common architectures and protocols.

5. CONCLUSION

This paper presented a study on introducing multi-ID and multi-locator into new generation network architecture. Different types of service requirements are conveyed to the network through the selection of appropriate host IDs. The host ID to locator mapping functions located in the identity sublayer of the protocol stack map host IDs into corresponding locators associated with the network that provides QoS demanded by the services. The paper listed some of the research and standardization items that are important for realizing effective services over multi-ID and multi-locator supporting network architecture.

In future work, the listed research items will be further explored and the research outcomes will be brought to ITU-T for standardization.

REFERENCES

[1] ITU-T SG13, Future networks including mobile and NGN, http://itu.int/ITU-T/go/sg13 (visited on 2010-09-09).

[2] AKARI Architecture Design Project for New Generation Network, http://akari-project.nict.go.jp (visited on 2010-09-09).

[3] Future Internet Forum, http://fif.kr (visited on 2010-09-09).

[4] FIRE - Future Internet Research and Experimentation, http://cordis.europa.eu/fp7/ict/fire/ (visited on 2010-09-09).

[5] NSF NeTS FIND (Future Internet Design) Initiative, http://www.nets-find.net (visited on 2010-09-09).

[6] Global Environment for Network Innovation (GENI), http://www.geni.net/ (visited on 2010-09-09).

[7] ITU-T Recommendation Y.2015: General requirements for ID/locator separation in NGN, Jan. 2009.

[8] V. P. Kafle, Hideki Otsuki, and M. Inoue, "An ID/locator split architecture for future networks," IEEE Communication Magazine, vol. 48, no. 2, pp. 138-144, February 2010 (Preliminary version of this paper was presented at the ITU-T Kaleidoscope event on Innovation for Digital Inclusion, Mar del Plata, Argentina, Sep. 2009.)

[9] V. P. Kafle and M. Inoue, "HIMALIS: Heterogeneity inclusion and mobility adaptation through locator ID separation in new generation network," IEICE Trans. Commun., Vol. E93-B, No. 3, pp. 478-489, March 2010.

[10] V. P. Kafle, K. Nakauchi, and M. Inoue, "Generic identifiers for ID/locator split internetworking," Proc. of First ITU-T Kaleidoscope Academic Conference on Innovation in NGN – Future Network and Services, Geneva, Switzerland, May 2008.

[11] E. Nordmark and M. Bagnulo, "Shim6: Level 3 multihoming shim protocol for IPv6," RFC 5533, June 2009.

[12] C. Vogt, "Six/one routers: a scalable and backward compatible solution for provider-independent addressing," Proc. of ACM MobiArch'08, Seattle, Washington, USA, Aug. 2008.

[13] R. Moskowitz and P. Nikander, "Host identity protocol (HIP) architecture," RFC 4423, May 2006.

[14] D. Farinacci, V. Fuller, D. Meyer, and D. Lewis, "Locator/ID separation protocol (LISP)," Internet-Draft, http://tools.ietf.org/html/draft-ietf-lisp-07, April 2010.

[15] The Interenet Engineering Task Force (IETF), http://www.ietf.org (visited on 2010-09-09).

HOW CAN AN ISP MERGE WITH A CDN?

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ABSTRACT

*As delivering contents has become the dominant usage of Internet, the efficient content distribution is being one of the hottest research areas in network community. In future network, it is anticipated that network entities such as routers will be equipped with in-network storage due to the trend of ever-decreasing storage cost. In this paper, we propose a novel content delivery architecture called Internet Service Provider (ISP) centric Content Delivery (i*CODE*) by which an ISP can provide content delivery services as well. i*CODE *can provide efficient content delivery services since an ISP can cache the contents in routers with storage modules considering traffic engineering and the locality of the content requests. Compared with CDN and P2P systems, i*CODE *can offer reduced delivery latency by placing the contents closer to end hosts, and incentives to ISPs by reducing inter-ISP traffic and allowing traffic engineering. We also discuss the technical and business issues to realize the i*CODE *architecture.*

Index Terms— Content delivery service, Content router, In-network storage, Future network architecture, Swarming

1. INTRODUCTION

International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) Study Group 13 has recently studied the requirements of future networks by investigating various technologies such as content-oriented network, network virtualization, etc. Indeed, more and more traffic on the Internet is attributed to content-oriented services such as file download and web access [1]. The original Internet architecture, however, is designed for host-oriented services like remote login. The discrepancy between the endpoint-based TCP/IP protocol suite and content-oriented user demands has brought some problems [2]. For instance, when multiple users close to each other download the same content file from a distant server, each download will take place separately and hence will take a long time to finish. Furthermore, if there is a surge of user access on a server (so-called flash crowd), it will overwhelm the server, which results in low throughput or even unavailability. The above inefficient delivery and lack of service availability happen since the Internet does not know the contents it carries.

To provide content delivery service efficiently with the current Internet architecture, there have been two representative solutions: peer-to-peer (P2P) systems (e.g., BitTorrent [3]) and content delivery networks (CDNs) (e.g., Akamai [4]). A P2P system is a distributed overlay network composed of cooperative end hosts (or peers). Since peers upload/download contents among themselves, there is no need of a centralized server responsible for contents distribution, which makes P2P systems scalable. In this way, P2P systems can handle a surge of user demands on a particular content file. One of the latest well-known P2P systems is BitTorrent that adopts *swarming*. The swarming technique allows a peer to receive multiple pieces of a content file from multiple peers in parallel, which even lowers the overhead of content transfer on each participating peer. However, since a P2P network is blind to the underlying network connectivity and a peer's topology information is limited, a peer in the P2P network may download the content file from a distant peer even if a peer in close proximity has the file. Also, if a peer downloads the file from a peer residing in another ISP, it will increase the incoming traffic from another ISP [5], which in turn will have a negative impact on the contract between ISPs. To reduce the inter-ISP traffic volume, ISP should be involved in the peer selection process [6]. Moreover, the crucial weakness of P2P systems is unavailability since peers are not stably connected to the P2P systems.

CDNs have been a successful business model, which provides stable and efficient content delivery services leveraging distributed data centers (often worldwide). A CDN provider distributes the copies of contents to its servers in data centers upon the requests of content providers. When a user requests a content file, the request is redirected to the CDN server in the closest proximity for the fast content delivery. In other words, a CDN provider pushes (or copies) the contents from the origin server to multiple (geographically distributed) servers towards end hosts. However, since the CDN provider coordinates its CDN servers to service end hosts independently of the ISP, the overall performance of the ISP network is not optimal in the perspective of traffic engineering [7]. Also, the CDN service

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is not affordable to small-scale content providers, which means that the CDN service can not be flexibly provided to a wide spectrum (in terms of traffic demand) of content providers.

Recently, there are emerging technical trends worth noting. The storage cost decreases steadily and exponentially, which enables many network devices (e.g., access points [8] and set-top boxes) to be equipped with ample storage modules [9]. Also, major router manufacturers (e.g., Cisco and Juniper) expose programming interfaces that allow packet manipulation by third-party applications to add new services at routers [10, 11]. Projecting this trend, it is expected that the network devices will be able to cache the contents in the foreseeable future [12, 13, 14]. That is, it is possible for routers or network entities to exploit the attached in-network storage modules for content caching and content delivery.

This paper develops this idea and proposes an efficient and flexible content delivery architecture, called ISP-centric content delivery (*i*CODE). The main advantages of the *i*CODE are summarized as follows.

1. User experiences: By placing contents at reliable network entities usually closer to end hosts, *i*CODE achieves stable and reduced latency of content transfer.

2. Traffic engineering: When *i*CODE places the cached contents into its routers, it can perform traffic engineering considering (a) the network topology and the bandwidth capacities of individual links, and (b) the popularity and the spatial/temporal locality of the contents.

3. Incentives to ISPs: Once contents from other ISPs are downloaded, they can be cached inside the ISP (that deploys *i*CODE) using in-network storage. In this way, *i*CODE reduces the inter-ISP traffic and hence gives the ISP financial advantages.

4. New business model: The *i*CODE architecture allows the ISP to provide the flexible CDN services for content providers with various traffic demands.

5. Incremental deployment: The facts that (a) the *i*CODE architecture retains the backward-compatibility with the current Internet and (b) a single ISP can provide the *i*CODE service independently of other ISPs facilitate an incremental deployment of *i*CODE.

The remainder of this paper is organized as follows. In Section 2, we describe the details of the *i*CODE architecture. Section 3 presents the simulation results, and Section 4 discusses the technical and business issues of *i*CODE. In Section 5, we compare the related proposals on content-oriented networking with *i*CODE. Section 6 concludes this paper with future work.

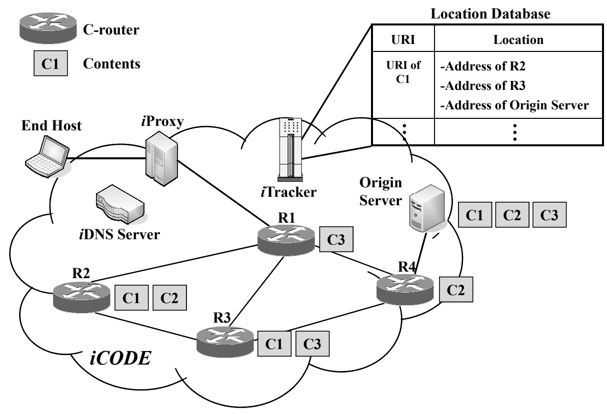


Fig. 1. An ISP operating the *i*CODE architecture is illustrated.

2. *I*CODE ARCHITECTURE

2.1. *i*CODE Overview

The *i*CODE architecture is illustrated in Fig. 1. To provide efficient and flexible content delivery services, the *i*CODE architecture employs the swarming technique adopted from BitTorrent and exploits in-network storage modules in network entities such as routers. The network entities cache the contents and service content requests from users. We anticipate that *i*CODE is a feasible business model to ISPs.

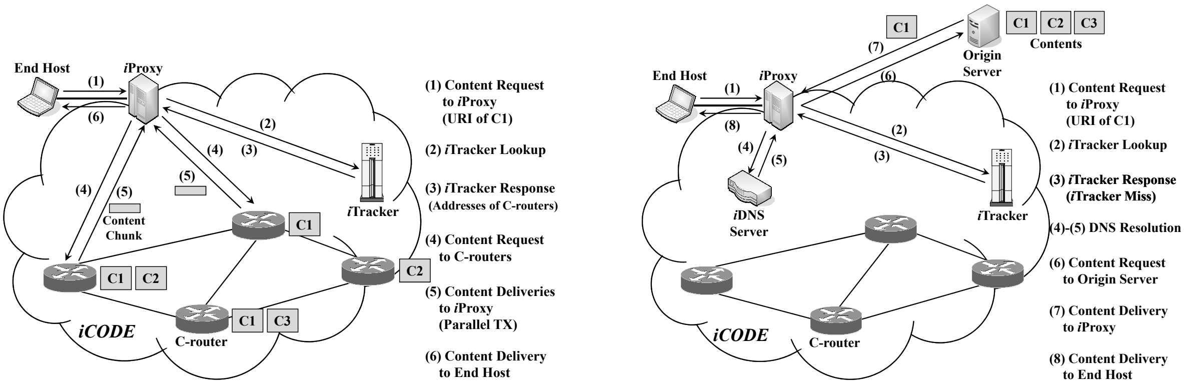
We assume that every content is identified by a uniform resource identifier (URI) to retain the backward-compatibility with the current Internet. When a user requests a particular content file, the *i*CODE architecture will intercept the request and check if the requested content file is cached inside the ISP network. If the content file exists, *i*CODE returns the addresses of multiple routers holding the file and the user will download the content file from the multiple routers through parallel transmissions. Otherwise, the content request will follow the current Internet practice (i.e. client-server model), but the downloaded contents may still be cached inside the ISP.

For the sake of exposition, we assume that one physical network (or autonomous system) is owned and operated by a single ISP. We assume that an ISP has many routers with storage modules for content caching [12, 13, 14], which are called content routers or C-routers for short. If *i*CODE concludes that a particular content file is popular and has to be cached, it will store the file at multiple C-routers. In this way, the following request for the same content file can be serviced from the C-routers within the ISP network.

2.2. *i*CODE Components

The main components of *i*CODE are as follows.

1. *i*CODE Proxy (*i*Proxy): An *i*Proxy receives a content request from an end host and sends the lookup request



(a) When the requested content is in the ISP.

(b) When the requested content is not in the ISP.

Fig. 2. Operations of *i*CODE when the request comes from the end host inside the ISP.

to an *i*Tracker to check whether the content is cached. If the *i*Tracker knows the location of the content, it will reply with the location (IP addresses of C-routers holding the content). Then the *i*Proxy will download the content, which in turn, is delivered to the end host. If the *i*Tracker does not know the content, the *i*Proxy performs domain name system (DNS) resolution for the specified URI and sends the content request to the corresponding server. The *i*Proxy is a functional entity that can be co-located with an end host, a middlebox like NAT or an access router. The main reason why the *i*Proxy is logically separated from the end host is to adopt the swarming technique transparently to the end host. That is, *i*Proxy may have to receive multiple chunks of the content from multiple C-routers in parallel, just like BitTorrent.

2. *i*CODE Tracker (*i*Tracker): An *i*Tracker is operated by an ISP and is responsible for managing contents inside the ISP. That is, it manages which contents to be cached and where to be cached. The *i*Tracker maintains the mapping between the content URI and its location (IP addresses of C-routers) in the location database as shown in Fig. 1. Also, the *i*Tracker maintains the mapping for the contents stored in the origin server which subscribes to the *i*CODE service. If there are too many contents in a single ISP, there should be multiple *i*Trackers in the ISP. *i*Tracker can be implemented in a distributed manner to avoid the single point of failure [15]. When the lookup request arrives from the *i*Proxy and if the *i*Tracker knows where the content is, it will choose multiple C-routers to deliver the requested content. Note that *i*Tracker can be extended to support ISP-friendly P2P services similar to P4P [6] by tracking the contents of end hosts within the ISP network.

3. Origin Server: An origin content server maintains the content published by the content provider. The origin server registers the metadata of the content to the *i*Tracker for the *i*CODE service.

4. Content Router (C-router): A C-router is a router that has a storage module, which can cache the copies of the contents. It performs content delivery services (in addition to packet routing) upon the request from the *i*Proxy. The *i*Tracker will manage which contents will be cached and replaced (if the storage is full) at individual C-routers.

5. *i*CODE DNS (*i*DNS): An ISP providing the *i*CODE service maintains *i*DNS servers for the DNS query redirection. When an end host outside the ISP requests a content file whose origin server belongs to the ISP, the DNS query will be forwarded to the authoritative DNS server that manages the domain name of the origin server. In the corresponding DNS record, there is a canonical NAME (CNAME) record to redirect the DNS query to the *i*DNS server. The *i*DNS server will contact the *i*Tracker to return the IP address of a C-router caching the requested content. Note that the host outside the ISP is not aware of *i*CODE and hence there is no *i*Proxy that performs swarming.

2.3. Operations for a content request from inside the ISP with *i*CODE

This section explains the operations of *i*CODE when a content request comes from an end host within the ISP that provides the *i*CODE service as illustrated in Fig. 2. The *i*CODE operations are different depending on whether the content copies are stored in the ISP or not.

1) The overall operation when there are cached content copies within the ISP is illustrated in Fig. 2(a). (1) If an end host wishes to download a content file, it will send the content request to the *i*Proxy in the form of a URI. (2) On receiving the content request, the *i*Proxy first consults the *i*Tracker which maintains the location database of contents. (which C-routers store the contents.) (3) After checking the location database, the *i*Tracker will reply with the IP

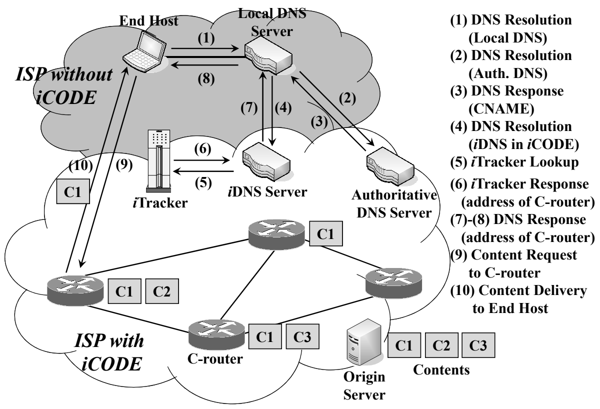


Fig. 3. Operation of *i*CODE when the request comes from the end host outside the ISP.

addresses of the C-routers. (4) After receiving the response, the *i*Proxy will request the content to the C-routers. (5) On receiving the content request, the C-routers will transmit the requested content to the *i*Proxy leveraging the swarming technique, which improves the download speed and mitigates the burden of individual C-routers. (6) The *i*Proxy will forward the received content to the end host. Note that even if the contents of the origin server are not cached in C-routers yet, the *i*Tracker will reply with the IP address of the origin server since the *i*Tracker maintains the metadata of the contents of the origin servers inside the ISP.

2) When there is no cached copy within the ISP, *i*CODE operates as illustrated in Fig. 2(b). (1) The end host sends a content request to the *i*Proxy. (2) Since there is no cached content within the ISP, the cache miss happens at the *i*Tracker. (3) In this case, the *i*Tracker informs the *i*Proxy of the cache miss event. (4) Then, the *i*Proxy will perform DNS resolution to find out the IP address of the origin server. (5) The DNS resolution will return the IP address of the origin server outside the ISP. (6)-(7) The *i*Proxy will establish a TCP/IP session with the origin server to retrieve the requested content. (8) After finishing content download, the *i*Proxy forwards the content to the end host. In case of successful downloading, the *i*Proxy informs the *i*Tracker of the result. Considering the download history of the contents, the *i*Tracker may decide to keep the copies of the content file at some C-routers.

2.4. Operations for a content request from outside the ISP with *i*CODE

Fig. 3 shows the operations of *i*CODE when a content request comes from an end host outside the ISP that provides the *i*CODE service. Suppose that *server1.com* is the URI of the origin server which subscribes to the *i*CODE service by the ISP and the CNAME of the origin server for the DNS redirection is *server1-com.iCODE1.com*. (1) When an end host requests a content file from the origin server that subscribes to the *i*CODE service, it will first perform the DNS resolution. The end host sends a DNS query to the local DNS server, which in turn consults the authoritative DNS server of the origin server. (2) The DNS query of the local DNS server arrives at the authoritative DNS server for *server1.com*. However, since the origin server subscribes to the *i*CODE service, the DNS query will be redirected to the *i*DNS server. For the redirection, there is a CNAME in the corresponding DNS record in the authoritative DNS server. (3) Accordingly, the authoritative DNS server will reply with CNAME, *server1-com.iCODE1.com*.

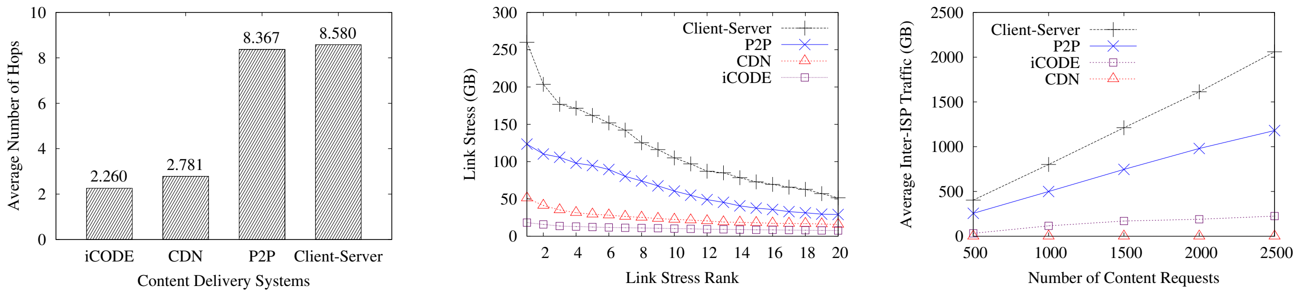
(4) On receiving the DNS response with the CNAME, the local DNS server of the host will proceed the DNS resolution process by sending a DNS query to the *i*DNS server maintained by the ISP. (5) The *i*DNS server will consult the *i*Tracker to locate the content. (which C-router caches the content.) (6) After checking the location database, the *i*Tracker will reply with the IP address of the C-router caching the content. As the end host does not use an *i*Proxy, it can download the file only from the single source. Note that the *i*Tracker can choose the C-router considering latency, current traffic load, and traffic engineering. (7)-(8) Now, the DNS response containing the IP address of the C-router is forwarded to the end host through the *i*DNS server and the local DNS server. (9) On obtaining the IP address of the C-router, the end host sends a content request. (10) The C-router will transfer the requested content to the end host.

3. SIMULATION RESULTS

We evaluate the performance of *i*CODE by using a discrete event-driven simulator. Our simulation environments are configured as follows. The Internet-like topology is generated using GT-ITM [16]. It consists of 1 transit domain with 5 routers, and 5 stub domains with 200 routers and 200 end hosts each. There are randomly distributed 1,000 contents with 1 GB size and the number of content requests follows Zipf distribution with parameter 1.0.

We compare the average hop counts, link stress, and inter-ISP traffic volume of *i*CODE with those of CDN, P2P, and client-server model. The cache size of all C-routers is set to 10 GB. The caching policy for the *i*CODE is a simple round-robin among C-routers. For the CDN server deployment, we assume the ISP-operated CDN model [17] which can deploy the server at the best position in the ISP network in terms of hop count. For the peer selection in P2P, each peer first selects peers within the same ISP network similar to P4P [6] (and then adds peers in other ISP networks only when it is not able to find 10 peers in the same ISP network).

(1) User Experiences: We evaluate the user-experienced performance of each scheme in terms of the average hop counts traversed for the content delivery. As shown in Fig. 4(a), users in an ISP with *i*CODE will experience the most reduced transfer delay due to the shortest hop counts. Although our simulation assumes that a CDN provider has deployed its server in all ISPs (which is not so likely), *i*CODE can perform better than the ideal CDN service. The reason is that the cached content will be delivered from the



(a) User experiences (Hop counts)
(b) Load balancing (Link stress)
(c) Inter-ISP traffic reduction (Volume)

Fig. 4. Performance comparison of *i*CODE against CDN, P2P, and client-server model.

nearby C-routers inside the network; that is, the C-router is typically closer to the requesting host than the CDN server. The client-server model and P2P incur much larger number of hops because the considerable amount of contents are originated from the servers or the peers outside the ISP.

(2) Load Balancing and Traffic Engineering: *i*CODE can achieve load balancing not only among C-routers but also among the links of the ISP network. We measure the link stress for all links, which is defined as the amount of traffic volume passed over a particular link, and plot the 20 links with the highest stress in descending order of the link stress in Fig. 4(b). *i*CODE and CDN use the links more evenly than P2P and client-server model. Compared with CDN, *i*CODE distributes contents to multiple C-routers; thus *i*CODE shows lower link stress performance than CDN (56.7 % reduction for the top 20 links). P2P can download from multiple peers resulting in better performance than the client-server model.

(3) Inter-ISP Traffic Reduction: *i*CODE offers economic incentives to ISPs by reducing the inter-ISP traffic. As shown in Fig. 4(c), *i*CODE incurs the smallest volume of inter-ISP traffic except CDN [1] since *i*CODE services the content from the C-routers inside the ISP once the content has been retrieved from other ISPs. Even with the large number of content requests, *i*CODE reduces the inter-ISP traffic consistently. On the other hand, both P2P and client-server model incur very large amount of inter-ISP traffic. P2P results in relatively less inter-ISP traffic than client-server model since P2P can download part of the content from peers within the same ISP network, whereas client-server model is fully blind to the underlying network connectivity among ISPs.

4. DISCUSSIONS

This section discusses remaining technical and business issues of *i*CODE.

(1) Customized Content Delivery Service: The *i*CODE service is affordable to small-scale content providers by supporting a wide spectrum (in terms of traffic demand) of content providers. Since the *i*CODE service is provided by the ISP that already provides the Internet connectivity, the content delivery service is possible with presumably smaller fee than the legacy CDN services. Also, it will be attractive to provide server-load-aware *i*CODE service to the small-scale content providers. Since the *i*Tracker participates in every content downloading process, it can control the source of the content transfer: the origin server or the specific C-routers. When the number of content requests is under a certain threshold, the requests are served by the origin server. If the number of requests exceeds the threshold (traffic overload), the requests can be served by the C-routers, which allows the small content providers to perform flexible server provisioning.

(2) Incremental Deployment: An ISP can deploy the *i*CODE independently of other ISPs. Even when end users, residing outside the ISP with *i*CODE, request content retrievals, the ISP can service the requests using C-routers caching the requested contents and the DNS redirection mechanism. Also, to provide *i*CODE services to end users transparently, the *i*Proxy performs the swarming and the in-network cache lookup. Note that *i*Proxy can be implemented as a network entity if the end user's device is not affordable.

(3) *i*CODE Substantiation over Multiple ISPs: The network virtualization is not only considered as one of the key technologies of future networks in the ITU-T Study Group 13, but also gaining momentum in the research community because of its potential to be a network infrastructure to test various proposals for the future network (e.g., Global Environment for Network Innovations (GENI) [18]). The network virtualization is an extension of the system virtualization in end hosts to the network entities such as routers. An instance of the virtualized fractions of the resources in the network entities is called a *slice*. So, it is possible for an ISP to lease a slice from other ISPs [19]. Assuming that ISPs in the future network will be virtualized, an ISP can substantiate *i*CODE across multiple ISPs by leasing virtualized slices of routers with storage modules from other ISPs.

(4) Swarming: Even though *i*CODE pushes the content files towards end hosts using in-network storage, there will be the transfer overhead on C-routers. To mitigate the

[1]Since our simulation assumes that all contents are stored in the CDN servers in advance before content retrievals, there is no inter-ISP traffic for CDN services. In reality, however, contents should be loaded at least one time across the ISP from the origin server to the CDN servers, which may occur comparable inter-ISP traffic with *i*CODE.

Table 1. Comparisons of innovative content delivery systems

	***i*CODE**	**DONA** [12]	**CCN** [13]	**PSIRP** [20]
Naming	URI	Flat name with a key to authenticate the publisher	Hierarchically encoded binary name based on URI	Flat id of the publication with the scope
Content resolution	*i*Tracker	Hierarchical Resolution Handler (RH) topology	Flooding	Rendezvous system
ISP incentive regarding traffic	Inbound and internal network traffic reduction, traffic-aware content delivery	Inbound traffic reduction	Inbound and internal network traffic reduction	Efficient network utilization for multicasting applications

transfer overhead and not to sacrifice the packet forwarding performance, we adopt the swarming technique. For this, the *i*Tracker will return the IP addresses of multiple C-routers that contain the requested content file. As the number of C-routers containing the same file increases, we can lower the transfer overhead of each C-router. Also, each C-router does not need to hold the entire file; the individual C-routers need to keep only the chunks of the file (like downloading different pieces of the same file from multiple peers in BitTorrent).

(5) Cache Management: Obviously, the total storage space is limited even though the number of C-router is large and the storage cost is not costly. We should note that enlarging the storage space of C-routers will incur more upgrade labor cost compared to CDNs due to the geographically distributed nature of routers. Thus, *i*CODE should efficiently manage the in-network storage with the *i*Tracker. As the popularity of a content changes, *i*Tracker can change the number of the content copies inside the ISP to manage the storage cost efficiently. Also, the request pattern for a particular content file varies spatially and temporally (e.g., time zones, diurnal pattern, spatial locality of contents). Thus, *i*Tracker will (be able to) flexibly migrate the contents among C-routers considering the above changes.

5. RELATED WORK

There have been innovative approaches to achieve the efficient content delivery in a non-Internet-compatible manner. We compare *i*CODE with the innovative proposals as summarized in Table 1.

Data-Oriented Network Architecture (DONA) [12] proposes to use flat, self-certifying names instead of URLs, and hence DNS name resolution is replaced by a name-based anycast. In DONA, the name resolution is done by a new class of network entities called Resolution Handlers (RHs). While a failed content request is forwarded to the origin server in *i*CODE, every content request in DONA is forwarded along the hierarchy of RHs, assuming that all the ISPs are DONA-compliant. Although DONA guarantees the perfect global availability of contents, the system works only when RHs are deployed over all the ISPs. On the contrary, users can benefit from *i*CODE even if only their ISP supports *i*CODE.

Content-centric networking (CCN) [13] extends the URI structure to name contents in a hierarchical manner for human readability, which in turn is mapped into the binary encoding. Content requests are binary-encoded as Interest packets and one Interest packet solicits one Data packet. The new network entity called a CCN node is somewhat similar to a router in the current Internet; it resolves and forwards Interest/Data packets; it also caches Data packets to reduce the network traffic and hence to enhance the availability. However, there may be many redundant Data packets in CCN if individual CCN nodes decide to cache their copies whereas the copies of data are managed by the *i*Tracker of the ISP in *i*CODE.

Publish-Subscribe Internet Routing Paradigm (PSIRP) [20] is the architecture with a goal of building a publish/subscribe-based network. Basically PSIRP tries to form a multicast tree for each content; the rendezvous system matches the publisher and subscriber and the topology system constructs multicast trees. Even though they propose to use Bloom filters, Merkle trees and special layer 2 hardwares, the multicast routing scalability will still be a concern. Also, the business model attractive to ISPs is not mentioned.

6. CONCLUSION

This paper proposes a new content delivery architecture called *i*CODE. With the in-network storage modules in routers, *i*CODE locates the contents closer to the end hosts, resulting in stable and reduced latency of content delivery. Also, *i*CODE provides incentives to the ISP by reducing the inter-ISP traffic with the locally cached contents and allowing traffic engineering considering the network status. Furthermore, *i*CODE opens a possibility of a new business model by which an ISP can support a wide spectrum of content providers. In future, we will investigate the details of content caching policy and delivery issues in *i*CODE over the large scale testbed.

7. REFERENCES

[1] ipoque, "Internet study 2008/2009," http://www.ipoque.com/resources/internet-studies/internet-study-2008_2009.

[2] K. Cho, J. Choi, D. D. Ko, T. Kwon, and Y. Choi, "Content-oriented networking as a future internet infrastructure: Concepts, strengths, and application scenarios," in *Proc. CFI*, 2008.

[3] "BitTorrent," http://www.bittorrent.com.

[4] "Akamai," http://www.akamai.com.

[5] T. Karagiannis, P. Rodriguez, and K. Papagiannaki, "Should internet service providers fear peer-assisted content distribution," in *Proc. ACM IMC*, 2005.

[6] H. Xie, Y. R. Yang, A. Krishnamurthy, Y. Liu, and A. Silberschatz, "P4P: Provider portal for applications," in *Proc. ACM SIGCOMM*, 2008.

[7] W. Jiang, R. Zhang-Shen, J. Rexford, and M. Chiang, "Cooperative content distribution and traffic engineering in an ISP network," in *Proc. ACM SIGMETRICS*, 2009.

[8] "Apple time capsule," http://www.apple.com/timecapsule.

[9] D. Ko, K. Cho, M. Lee, H. Kim, T. T. Kwon, and Y. Choi, "Decentralized and autonomous content overlay networking (DACON) with WiFi access points," in *Proc. CFI*, 2010.

[10] "Cisco Application eXtension Platform (AXP)," http://www.cisco.com/en/US/products/ps9701.

[11] J. Kelly, W. Araujo, and K. Banerjee, "Rapid service creation using the JUNOS SDK," in *Proc. ACM PRESTO*, 2009.

[12] T. Koponen, M. Chawla, B.-G. Chun, A. Ermolinskiy, K. H. Kim, S. Shenker, and I. Stoica, "A data-oriented (and beyond) network architecture," in *Proc. ACM SIGCOMM*, 2007.

[13] V. Jacobson, D. Smettersa, J. Thornton, M. Plass, N. Briggs, and R. Braynard, "Networking named content," in *Proc. ACM CoNEXT*, 2009.

[14] L. Dong, H. Liu, Y. Zhang, S. Paul, and D. Raychaudhuri, "On the cache-and-forward network architecture," in *Proc. IEEE ICC*, 2009.

[15] J. Choi, J. Han, E. Cho, H. Kim, T. Kwon, and Y. Choi, "Performance comparison of content-oriented networking alternatives: A tree versus a distributed hash table," in *Proc. IEEE LCN*, 2009.

[16] E. W. Zegura, K. Calvert, and S. Bhattacharjee, "How to model an internetwork," in *Proc. IEEE INFOCOM*, 1996.

[17] N. Kamiyama, T. Mori, S. Harada R. Kawahara, and H. Hasegawa, "ISP-Operated CDN," in *Proc. IEEE INFOCOM Workshop*, 2009.

[18] "GENI," http://www.geni.net.

[19] G. Schaffrath, C.Werle, P. Papadimitriou, A. Feldmann, R. Bless, A. Greenhalgh, M. Kind, O. Maennel, and L. Mathy, "Network virtualization architecture: proposal and initial prototype," in *Proc. ACM VISA*, 2009.

[20] A. Zahemszky, A. Csaszar, P. Nikander, and C. Esteve, "Exploring the pub/sub routing&forwarding space," in *Proc. FutureNet*, 2009.

SESSION 3

THE FUTURE INTERNET IS FOR ALL

CAN COMPUTATIONAL THINKING REDUCE MARGINALIZATION IN THE FUTURE INTERNET?

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ABSTRACT

Maths is presently regarded as the key driver that underpins Science, Education and Technology (SET) skills. In spite of significant studies, investment and efforts, math skills and widespread enthusiasm for SET remain elusive. In South Africa's disadvantaged communities, poor quality maths teaching and poor maths performance, both legacies of past political engineering, further fuel marginalization.

Computational thinking is a new characterization of some specific procedural thinking, abstraction, problem solving and organizational skills that are finding their way from computer science programs into other fields.

The paper describes our refocus of content in BingBee, a SET skill-building kiosk project targeting disadvantaged communities. As we shift to emphasize computational thinking more explicitly, we speculate that these skills could complement, and perhaps eventually displace, some elements of maths as the dominant driver of SET.

The confluence of better tools, open service interfaces, and the rapid spread of handsets and devices into marginalized communities is an opportunity to build more widespread computational thinking skills. This could in turn facilitate a future Internet which is more inclusive, and in which users are able to create their own services.

Keywords— Computational thinking, problem solving, abstraction, cognitive skills, marginalized communities.

1. INTRODUCTION[1]

Science, engineering and technology (SET) is widely accepted as a primary driver for innovation and economic growth [1]. But despite significant funding and attempts to promote SET in South Africa, success has been limited.

Maths is perceived as the key educational enabler of SET, and is the primary vehicle we presently use to develop problem solving, rigor, logical argument, formal notation, abstraction, and notions of proof and contradiction.

A vast body of knowledge and much effort has been expended in attempts to understand the intertwined strands of the learner's cognitive development, particularly in the area of teaching mathematics, and to integrate these theories into classroom practice and syllabi that will ultimately improve maths education. In Piaget's cognitive model, abstract thinking and deductive reasoning, the so-called *formal operations* stage, develops from ages 12+. The key discipline tasked with building these skills in the school system is maths. However, Moursund [2, p. 64] cites studies showing that even in industrialized societies, only about 35% of high school finishers achieve this stage, and "*college students' ability to reason with abstractions is strikingly limited"*.

Kilpatrick et. al. [3] identify five necessary and intertwined threads to mathematical proficiency: *procedural fluency*, *conceptual understanding*, *strategic competence*, *adaptive reasoning* and a *productive disposition*. Moursund [2, p. 65] further cites work suggesting that *the typical high school geometry courses was being taught at a developmental level considerably above that of the typical students taking such a course*. Such gaps undermine the attitudes – the productive disposition – that we require if we want to attract more students into SET.

Many have tried to close the gaps using *manipulatives* – artefacts, particularly software artefacts, that demonstrate ideas and attempt to build understanding independently of the formalisms. *Computing Technology for Maths Excellence* [4] has a rich discussion and links to numerous sites devoted to math manipulatives. Other sites with collections of quality software are the National Library of Virtual Manipulatives and Math Playgound.

However, the US National Mathematics Advisory Panel (2008) in its *Foundations for Success* [5] cautions as follows: *Despite the widespread use of mathematical manipulatives such as geoboards and dynamic software, evidence regarding their usefulness in helping children learn geometry is tenuous at best.*

In South Africa, maths teaching, mathematical performance, and the productive disposition towards mathematics is particularly poor in the marginalized groups. This is partly explained by a history in which high quality education in Science and Mathematics was deemed an "unnecessary luxury" for those who were destined to be labourers or semi-skilled workers in the political master-plan.

2. COMPUTATIONAL THINKING

Computational Thinking (CT) is an area that might offer a fresh approach to building abstract thinking, problem solving and deductive reasoning skills. The term was coined by Jeannette Wing [6] as a characterization of some specific thinking, abstraction, and organizational skills that

[1] This work was supported by the Telkom Centre of Excellence in the Department of Computer Science at Rhodes University.

Figure 1. Children at a BingBee kiosk

are commonly taught and used in Computer Science. Wing argues that the skills are fundamental across all disciplines and are widely applicable and useful generally. Just as the printing press gave rise to widespread literacy, the availability of computers creates an opportunity for shifting more emphasis onto computational processes and skills. Algorithmic thinking and capturing procedural knowledge is a core CT emphasis. Another is abstraction and organization. Working in software at multiple layers of abstraction simultaneously is also commonplace – we create on-the-fly factorings that are encapsulated in classes or services while we simultaneously develop the applications that consume them. Data representation, recursion, planning, and strategy are all elements that also fall under the CT banner. It also encompasses the ability to apply mathematical concepts such as induction, and to reason about pre-conditions, post-conditions, and correctness.

Denning [7] points out that CT is not new – it has been around in various guises in many disciplines for a long time. While CT may be a new lens through which we can regroup or view the notions, he cautions against drawing the narrow conclusion that *CT = Computer Science.* He is mindful of the previous costly mistake of characterizing *Computer Science = Programming*.

We are particularly interested in whether computational thinking is a viable way to build those "deep" problem-solving skills, especially for those who are not acquiring the skills from their maths education.

3. SETTING A NEW DIRECTION

Over the past five years our BingBee project [8] has endeavoured to indirectly build early childhood SET skills in computing, logic, language and mathematics. Located in an underprivileged area, it provides access to a range of edutainment activities running on unsupervised computer kiosks (Fig. 1). Some of the activities and games are pure entertainment, some, such as telling the time, Xhosa/English vocabulary, and arithmetic quizzes are for drill and practice, while others exercise memory, logic, strategy, spatial, and inductive skills.

Using the fresh lens of computational thinking, very few of the existing 50+ activities can be characterized as algorithmic or demanding of CT skills. Even where algorithmic skills are exercised or required (e.g. playing tic-tac-toe, solving sliding tiles, minesweeper, or solving a Sudoku puzzle), the CT components are implicit rather than explicit. One activity where we did attempt to explicitly teach an algorithmic "method of attack" was in the help documentation for Sudoku. The Sudoku engine can display "pencil marks" of remaining candidate possibilities for each empty cell. The algorithm is based on eliminating candidates from the possibility set, and the game can offer hints to the player. In practice, though, our documentation is ignored, and Sudoku has not caught on.

We also believe Kilpatrick's five threads for mathematical proficiency make an excellent framework for understanding and building proficiency in Computer Science. Our new goals are therefore to explicitly identify a few key elements of computational thinking and to develop virtual manipulatives to explicitly target these concepts. We do this in the style of an inverted curriculum in which we strive to get the higher-level understanding and a more positive productive disposition in place ahead of the low-level syntax and usual first-course programming concerns. We chose five CT themes:

- abstraction,
- programming,
- finite state machines,
- algorithms,
- graph structures.

Our first trial use was with a group of first-year university students, most non-majors in Computer Science, but we are in the process of retargeting the activities to our community kiosks.

3.1. RuBoT: a programmable robot

Our first new-breed activity is inspired by a delightful Internet game called light-Bot. The user builds a program by dragging simple robot instructions (turn, step, jump, flip the state of a light) into instruction slots in a program. If running the program successfully turns on all the lights in the current world, the player advances to the next challenge. The game provides two callable functions each with their own instruction slots. (Purists may prefer the term *procedure*.) These allow solving the advanced levels without running out of instruction space.

Our own implementation of the game added substantially to light-Bot – not by extending the command set, but by making more explicit the process of execution and abstraction. We've provided for more functions, and removed the fixed number of slots available (running out of space is a poor reason to create procedural and mental abstractions). Explicit comment blocks for each function can guide or encourage verbalization of the abstractions, e.g. *this function advances the robot over a complete row*, and we've provided a folding editor mechanism to collapse the body of code in each abstraction. Our CT message is that functions exist to facilitate mental "chunking", and folding the details of the chunks away mirrors how we can

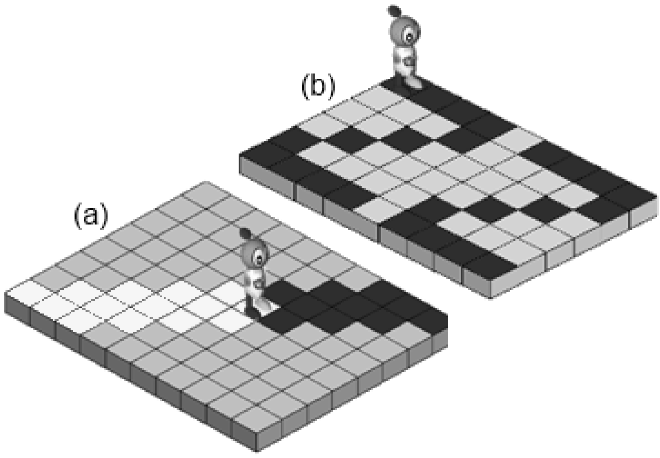

Figure 2. Two levels of RuBoT

use our mental abstractions. The documentation for the functions can also encapsulate pre- and post- conditions and invariants: *This function switches a diagonal of 8 lights to the right. Pre: robot is on the first light. Post: robot is on the last light, facing in the direction it started*. Functions can call other functions, providing for multiple layers of abstraction.

To reinforce early notions of a "programmable machine" and to build a concrete mental model of program execution, we highlight the current instruction at each step. The notion of function call and return is a vital early concept, so each function entry and return is an explicit step. (We considered showing the call/return stack too, but omitted it.)

There are no loops or conditional instructions in the instruction repertoire, but functions can recursively call themselves. (At the time the recursion is programmed, we warn the user that the program cannot terminate, but they might have fun anyway!) Recursion is a powerful idea, and we continue to seek good ways to introduce it early and gently.

But our primary goal with RuBoT is to promote abstraction and chunking. The various (extensible) levels of play introduce challenges that have embedded patterns of lights. It has been fascinating to observe how different students group and create their building blocks. In the challenge shown in Fig. 2(a), for example, some students will create a function to "light a diagonal". Others tackle the problem "row-by-row". In Fig. 2(b), one of the more creative solutions is to spot that the Z is symmetrical, so creating functions to "light half a Z", and then to "do one Z", and then to "do two Zs" works well.

3.2. Finite state machines

Understanding automation and computation in terms of states, events, transitions and responses is a powerful mental tool. The cell phone seems a great first introduction. Students have little difficulty (and great motivation) for identifying states (locked, unlocked, silent, general, ringing, busy, idle, awaiting PIN, etc.) and modelling the events and transitions between these states. We reinforce this with a

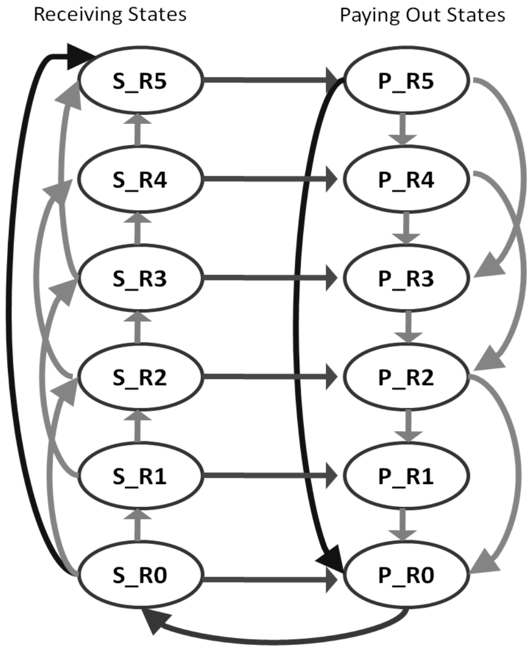

Figure 3. States of money changer machine

few software manipulators. The controller for a traffic-light simulator is programmed by building simple spreadsheet-like table of the current state, the event, the response, and the new state. We follow this with a more complex nondeterministic finite state machine for changing coins (suggested by Nievergelt [9]): the user puts coins into the machine, and presses a button for change. The machine makes nondeterministic choices at each step about which denomination coin to drop next. By repeated tries, the user can always eventually get the exact change she wants. A gentle introduction to probability and experimental method follows naturally: they can experiment many times with the machine, record the results, and estimate the probabilities of each transition at each state. Nievergelt suggests that if machines like this in parking garages had explicit animated state diagrams, novices would understand and appreciate the automation much better. We've taken this advice, so our money changer machine can display a view of its internal logic by showing an animated representation of its internal finite state machine (Fig. 3).

Discovering and mapping the states and transitions of a state machine is engaging, and reinforces notions of making trials against an unknown system, documenting evidence, building a model, and eventually ensuring that all possible transitions are mapped. We get students to feed inputs to black boxes, and to hypothesize about their internal logic. We also use a system of underground passages and caves that need to be discovered and mapped. This was inspired by Crowther's original Collosal Cave Adventure game in which players found themselves in mazes of twisty little passages. An activity from the Computer Science Unplugged [10] website is based on the same idea: children must discover and map a system of pirate islands in their search for the treasure.

3.3. Graph theory

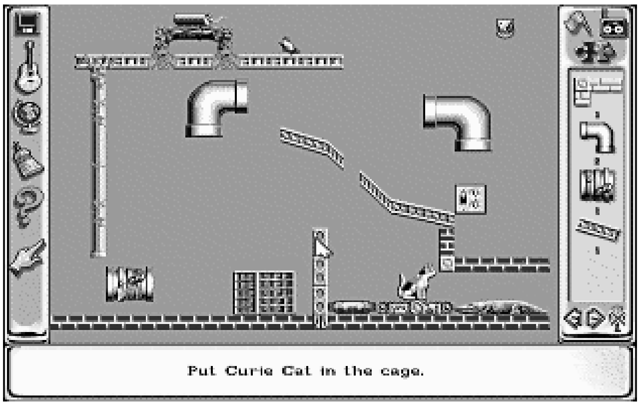

Figure 4. The Incredible Machine

Graph theory is a powerful formalism with many concrete applications and algorithms, especially in the field of telecommunications. In the unplugged *Muddy City* example [10], a town has a number of houses, but flooding makes the interconnecting pathways muddy. The goal is to help the mayor determine which pathways to pave, so that any house can be reached from any other only on paved pathways, using the least number of paving stones. Children colour in and count the paving stones they wish to lay.

The abstraction into a graph with cost-labelled edges is simple, and in graph theoretic terms, the problem is to find a minimum spanning tree. (In network planning terms, one application is to determine which links require protection.) We used both Prim's and Kruskal's algorithms for this – both are greedy algorithms, and both are easy to teach.

3.4 The Incredible Machine

The Incredible Machine (Fig. 4) is a series of puzzle games originally published by Sierra. Users solve challenges (e.g. put Curie Cat in the cage) by dragging and dropping components from a toolbox on to a designer, and hooking them up. A falling ball can land on the mouse cage, causing the mouse to run on its treadmill. Adding a belt feeds power from the treadmill to a generator, which may light a laser which can be used to ignite a fuse on the rocket, and so on. Although we are not using the game presently, we regard it as an exemplar of a well-engineered interface that allows non-expert users to "program" sophisticated relationships and cascading sequences of events to achieve some objective. We will return to this in the next section.

4. SPECULATING BEYOND THE INTERNET

The Internet's historical strength was as a data repository. The transition to the so-called Web 2.0 saw the rise of social networking, sharing, and content provision "by the users, for the users". While the Internet provides the communication, sites like Facebook and YouTube provide the frameworks for sharing our data and for collaboration.

The same migration has not happened for our services, or for our programming tools that can express the computation necessary to build those services. Neither do we have an enabling framework analogue of Facebook that would enable sharing and creation of services by end users, for sharing with other end users.

If computational thinking and more explicit emphasis on the development of procedural and algorithmic skills does gain traction in our education systems, we will likely see more emphasis on the Internet as a programmable service creation platform. This trend will be supported by other factors: high-level graphical drag-and-drop development environments that enable users to "script" quite complex chains of logic and events to build games and services; more smart home networking; interfaces that permit interaction with our networked cars, personal video recorders, and buildings; open service interfaces from the vendors that can be wrapped as graphical components for easy use by our service-building tools. There is presently much protocol work in progress to enable this integration of the home, the Internet, and the triple-play services from service providers. The confluence of these would take us to a future Internet in which programming and expressing computation is not just the role of a select cadre of professional implementers: the tools and competencies for scripting their own services will be more accessible to large numbers of end users. As evidence that this trend is already underway, Microsoft Research recently released the Kodu Game Lab, a low-barrier tool that lets end users build multi-player networked games. Complex scenes, events and responses are tied together using a simple drag-and-drop user interface.

The ability to assemble one's own services and games would likely give rise to a large number of simple services that might not ordinarily warrant investment or attention from the telecommunications operators. It would also allow those in less privileged communities to set their own priorities about what services mattered to them. Consider these sample scenarios that could plausibly be programmed by do-it-yourself users using a TIM or Kodu style designer:

- Each day at 9am, poll my parents' online dispensary device at their retirement home. If daily medication has not been dispensed today, alert me via SMS.
- On Thursdays between 8am and 9am, the gardener will request access by sending an SMS from <number>. Respond by disarming burglar alarm circuits 6 and 7; unlock the tool shed; open the front gate. If the weather forecast service does not predict rain, and the grass has not been cut in the previous 14 days ago, send an request to the grass-mowing service to mow the grass between 11am and 3pm today. Revert burglar alarm circuits at 4pm.

5. CONCLUSION

Re-analyzing our existing SET-building activities and first computing courses through the perspective of computational thinking is influencing our approach significantly. We are actively seeking opportunities to identify, promote, and make more explicit the elements of procedural knowledge, problem solving, algorithms, abstractions, the notion of states and transitions, and the graph-theory formalisms early in the child's development. There appear to be opportunities, both in the existing school curriculum, (should one teach flowcharts or the binary lookup algorithm explicitly when teaching children to look up words in a dictionary?) and in board games, games with playing cards, and so-called "educational" games and activities. Activities with strong procedural, abstraction, and problem solving elements (as diverse as origami, Sudoku, Rubik's cube, chess, and card tricks) could be elevated from their current "recreational" status and included in the formal syllabi.

More appreciation and insight into the general value of algorithms and procedural knowledge might influence our school curriculum planners too: for example, we've recently seen the teaching of long division dropped from our schools. Even if the calculator has made this particular skill obsolete, we rue the loss of one of the very few explicit algorithms in the syllabus.

Wing [6] sees our widespread access to computing demanding a "new kind of literacy". We endorse that view, and have made a few first steps towards her vision in which computational thinking is explicitly taught as a fourth pillar of education, alongside reading, writing and arithmetic. Our earlier quote from [5] continues thus: *Students must eventually transition from concrete (hands-on) or visual representations to internalized abstract representations. The crucial steps in making such transitions are not clearly understood at present and need to be a focus of learning and curriculum research. (p. 29)*

Computational thinking is ripe for such focus. Not only is it emerging as a useful body of techniques and knowledge in its own right, but it potentially offers another approach for building or reinforcing those indirect skills of abstraction, problem solving, logic, strategy and formal reasoning.

The direct benefits of bringing communication technologies to communities marginalized in the digital divide are widely accepted. Can we further leverage the benefits by simultaneously addressing some cognitive skills gaps? Each new device is a computational engine – an ideal platform for shifting more emphasis onto procedural and algorithmic modes of thinking and problem solving. This will hopefully spark new interest in SET.

We next need to be able to pose long-term questions like *"Is teaching algorithms a more effective route to abstraction and problem solving skills than, say, the teaching of geometry?"* If we could measure the elements of computational thinking, and their correlation to internalized abstractions, we could make head-to-head comparisons with similar correlations between mathematical skills and ability to internalize the abstractions. A first suggestion is to urge assessments of scholarly competence, such as the SAT tests, to recognize, measure, and differentiate more explicitly this axis of procedural and algorithmic skills.

More widespread computational thinking skills would then become one of the drivers for a future Internet in which service definition and construction migrates into the hands of ordinary users.

I wish to thank the reviewers for their helpful comments.

REFERENCES

[1] http://www.britishcouncil.org/science-gost-introducing-uk-set-2.htm "Introducing UK Science, Engineering and Technology (SET), UK Science, and Overview", GOST version February 2008, British Council.

[2] D. Moursund, "Computational Thinking and Math Maturity: Improving Math Education in K-8 Schools", http://www.uoregon.edu/~moursund/Books/ElMath/K8-Math.pdf

[3] J. Kilpatrick, J. Swafford, B. Findell, (Editors) "Adding it up: helping children learn mathematics", National Academic Press (2001), ISBN: 978-0-309069-95-3.

[4] "Computing Technology for Math Excellence", http://www.ct4me.net

[5] National Mathematics Advisory Panel (2008). "Foundations for success: The final report of the National Mathematics Advisory Panel", Washington, DC: U.S. Department of Education. Also at http://www2.ed.gov/about/bdscomm/list/mathpanel

[6] Wing, J. "Computational Thinking", Viewpoint Article, *Comm. ACM.* 49(3), March 2006. Also at http://www.cs.cmu.edu/afs/cs/usr/wing/www/publications/Wing06.pdf

[7] Denning P. "Beyond Computational Thinking", Viewpoint Article, *Comm. ACM*, 52(6), June 2009. Also at http://cs.gmu.edu/cne/pjd/PUBS/CACMcols/cacmJun09.pdf

[8] Wentworth, P. "BingBee@RaglanRoad – a Field Trial with Unattended Educational Kiosks*", IST-Africa 2010 Conference Proceedings*, ISBN: 978-1-905824-15-1. Also at http://www.bingbee.com/?q=pub/ISTAfrica2010

[9] Nievergelt, J. "Lecture Notes on Theory of Computation", (Chapter 2, Finite State Machines), ETH, Zurich. http://www.jn.inf.ethz.ch/education/script/chapter2.pdf

[10] "Computer Science Unplugged", http://www.csunplugged.org

(All web references visited on 2010-08-21).

CHALLENGES THE INTERNET POSES TO THE POLICYMAKER

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ABSTRACT

This paper addresses policymakers at national and international levels -- regulators, standards bodies, politicians – arguing that there is no "beyond" the Internet. With the Internet so intimately intertwined with the lives of people, being used to build the backbone of large, important communities, an attempt to replace it with a new network would generate immense friction, and cost a lot. The transition would take long, because lots of complex software would need to be written, disrupting critical processes of the economy, indeed of governance. A plethora of regulators with very different manners and degrees of control would have to learn to work together at an international level, otherwise we might revert to the lawlessness of the Internet. The lost opportunity of Minitel, the botched attempt to look beyond the Internet in the 1990s via X.400 and the bankruptcy of large telecommunication companies in the wake of the dotcom boom are useful in appreciating the historical context and learning lessons from. Instead of looking beyond, the ITU should play a constructive role vis-à-vis the Internet. Suggestions presented are elimination of spam, and making the Internet accessible to all. These make commercial sense too

Keywords— ITU, X.400, Minitel, Internet, policy

1. A CLEAN SLATE?

The announcement for this Conference [1] "Beyond the Internet?", offers this para as justification for its title:

"Thus far, the Internet has proven to be robust and flexible and its continuous evolution has seen growth from a small experiment into a giant collaborative network capable of meeting the demands of more than one billion users. The rise of mobile access and its integration with optical transport networks present new challenges. Some experts question whether the current underlying architecture is sufficiently robust to address future demands or if a "clean slate" approach is needed to develop a really innovative Internet of the future."

The technologist can afford to dream. However, the policymaker at the national or international level has no such luxury. The mere thought of wiping a "slate" clean, one on which over a billion people work, play and learn, should have her breaking out in a cold sweat. Hard questions must be answered, before such an argument is treated with any seriousness.

This paper addresses policymakers at national and international levels - regulators, standards bodies, politicians – arguing that there is no "beyond" the Internet. The genie is out of the bottle, and no "really innovative" super genie is going to appear, to put it back in.

In looking at definitions of the Internet, the political aspect attracts more attention here, than the technical. As the charred fingers of seasoned policymakers will confirm, technology is usually the easier part. The political aspects are ignored at one's peril. With the Internet so intimately intertwined with the lives of people, building the backbone of large, important communities, an attempt to replace it with a new network would generate immense friction, to put it mildly, and cost a lot. The transition would take long, because lots of complex software would need to be written, disrupting critical processes of the economy, indeed of governance.

How would the new network be regulated? If the variety of activities found on the Internet were to all find homes in the new network, a plethora of regulators with very different manners and degrees of control would have to learn to work together at an international level, otherwise we might revert to the "anarchy" [27] of the Internet. There is plenty of scope for confusion, conflict, and mishap in any such transition.

Not knowing what one is exactly replacing, nor how, with severe doubts about who will do all this, the question of why should become loud in our minds. "The rise of mobile access and its integration with optical transport networks present new challenges" is not convincing. Surely our brilliant telecommunication engineers can find some way of fixing this within the framework of the Internet, without having to wipe the slate clean?

If not looking beyond the Internet, what role should the ITU play, vis-à-vis the Internet? While the author does find it easier to ask questions than to answer them, some humble suggestions are provided.

WHAT IS THE INTERNET?

You can step out onto the streets of Pune, and ask any passerby, where to get "Internet." Chances are, she will direct you to a nearby shop, which has the word written on its bill board. It might amuse our gracious roadside hosts, that if you search Google for a definition of the Internet [2], pointers to over a dozen pages show up. The one from Princeton University [3] is how a technologist sees the Internet:

"A computer network consisting of a worldwide network of computer networks that use the TCP/IP network protocols to facilitate data transmission and exchange." TCP/IP, one might paraphrase, is the "lingua franca" of a space in which any computer in the world can exchange information with any other.

The definition from a glossary at O'reilly [4], the leading publisher of Internet-related books, in talking of "a relatively loose federation", is already starting to get political:

"A relatively loose federation of computer networks that permits data to be widely transferred among computers."

A "Keywords and Definitions" page at MIT [5] talks about "Organizations using the Internet in innovative ways. (Technology)"

This is not just a computer network. There are organizations involved, communities as well. A concise yet comprehensive definition from the University of Chicago, [6] is perhaps best at outlining this mammoth called the Internet, which the policymaker will struggle to look past:

"Often confused with the World Wide Web, the term Internet actually refers to the combined collection of academic, commercial, and government networks connected over international telecommunication backbones and routed using IP addressing."

Indeed, our friends on the Pune street share in this confusion, as do surely hundreds of million others. The Internet cannot be imagined without the Web. An argument can therefore be made to include the html language and http protocol, which technically define the Web, in the definition of the Internet. But then, why stop there?

The first challenge the policymaker faces, therefore, is one of definition. The very inventors of TCP-IP, Robert Kahn and Vinton Cerf, wrote in December 1999, in a paper entitled, "What is the Internet (And what makes it work)" [7]:

"The authors feel strongly that efforts should be made at top policy levels to define the Internet. It is tempting to view it merely as a collection of networks and computers. However, as indicated earlier, the authors designed the Internet as an architecture that provided for both communications capabilities and information services. Governments are passing legislation pertaining to the Internet without ever specifying to what the law applies and to what it does not apply. In U.S. telecommunications law, distinctions are made between cable, satellite broadcast and common carrier services. These and many other distinctions all blur in the backdrop of the Internet. Should broadcast stations be viewed as Internet Service Providers when their programming is made available in the Internet environment? Is use of cellular telephones considered part of the Internet and if so under what conditions? This area is badly in need of clarification."

The same, of course, will apply to any new network one conceives. A mere definition of protocols will not suffice. If the definition is itself so hard, imagine how hard the implementation will be.

POLITICAL ANOMALIES OF THE INTERNET

This paper does not pretend to explain the political side of the Internet, merely point out how complex and unconventional it is. The Internet has allowed us to build communities across the globe, many of which would not exist without the Internet. A classic example is Linux, which was started by a student, Linus Torvalds, and announced on an Internet discussion forum in 1991 [8]. This operating system uses the Internet for distribution, support and collaborative ongoing development. A bunch of volunteers, including thousands of programmers, are holding their own against powerful companies such as Microsoft. The Linux "community" includes the Brazilian government, which has been more active than most in promoting it [9] and the Indian state of Kerala, which has mandated it for schools. [10]

What would wiping the slate clean and creating a new network mean for the Linux community, and myriad others, that rely crucially on the Internet for survival? If their future is not assured on the new network, they will fight it tooth and nail. To find a new home for such communities, not only would a lot of software need serious modification to run on a new platform but also, the developers and content providers of this community persuaded to migrate. These are daunting tasks indeed, surely not what the proponents of the "clean slate" envisioned.

Linux is only one example of an Internet-based community. If all such activities people carry out on the Internet are to be possible on the new network, and if the point of migrating is to have a more controllable network, then the policymaker must propose who will regulate these. The logical choice would be, of course, the entity that regulates this activity in the real world.

India has been struggling since 2001 [11] to bring just the communications and broadcasting sectors under one omnibus legislation, the Communications Convergence Bill, which only involve the telecommunication regulator, and a couple of ministries. Imagine trying to regulate several other Internet activities as well. Online money transactions would come under the Reserve Bank [12], trading in stocks and shares under SEBI [13], telephone services under TRAI [14], cinema content under the Censor Board, other kinds of content the relevant ministries. Then, there are issues relating to gender and minorities, and criminal activities, such as gambling and pornography, which have their own watchdogs. Merely bringing all these regulators around a table will be a challenge. Expecting any agreement in any realistic time frame, on how the new network should be regulated at an international level, is very unrealistic. Without compelling reasons, why would a policymaker want to embark on such a weary journey?

The Internet is not a political entity of the traditional kind. No person or organization is "in charge" with the power to speak on its behalf. Its resistance to censorship has been the subject of much discussion, best illustrated by an excellent quote from John Gilmore: "The Net interprets censorship as damage and routes around it." As he discusses on his home page [15],

"This was quoted in Time Magazine's December 6, 1993 article "First Nation in Cyberspace", by Philip Elmer-DeWitt. It's been reprinted hundreds or thousands of times since then, including the NY Times on January 15, 1996, Scientific American of October 2000, and CACM 39(7):13.

In its original form, it meant that the Usenet software (which moves messages around in discussion newsgroups) was resistant to censorship because, if a node drops certain messages because it doesn't like their subject, the messages find their way past that node anyway by some other route. This is also a reference to the packet-routing protocols that the Internet uses to direct packets around any broken wires or fiber connections or routers. (They don't redirect around selective censorship, but they do recover if an entire node is shut down to censor it.)

The meaning of the phrase has grown through the years. Internet users have proven it time after time, by personally and publicly replicating information that is threatened with destruction or censorship. If you now consider the Net to be not only the wires and machines, but the people and their social structures who use the machines, it is more true than ever. "

A multitude of companies own parts of the Internet [16]. They typically have little control over what content flows on their networks. These are the sorts of challenge policymakers would have little experience of. Why would they want anything to do with the total political mess a migration to a new network would imply?

LESSONS FROM HISTORY

This is not the first time that an attempt is being made to replace the Internet. The X.400 debacle is well researched from a standards point of view, but its failure was arguably less for technical than for political reasons.

In the 1980s the X.400 standard for e-mail was developed by the ITU in cooperation with the ISO. When X.400 based email services became available, (mid' 90s in India), Internet-based e-mail had already been in use for some years in research and educational environments. Although that may not have been the original intention, at some point in time SMTP (the Simple Mail Transfer Protocol; the Internet's e-mail protocol) and X.400 clearly were competitors. X.400 did not run on top of TCP-IP, but on a different transport service. Thus, in a way, they may be seen as proxies for the struggle between the Internet and the OSI world. X.400 received strong government backing, in that the policy allowing email services only allowed X.400 [17]. Several X.400 email serving companies did indeed start in India, but all of them folded in a couple of years. Some niche services such as EDI continue to use X.400, which now also runs on TCP-IP. Indeed this is now the most popular way to run X.400. [18].

The failure of X.400 against the Internet equivalent, SMTP, is explained thus, "X.400 was indeed 'installed-base hostile' in a way, largely due to the fact that it was considered an integral part of the OSI initiative, and accordingly initially required the use of underlying OSI protocols, full implementations of which were not readily available in 1984. This strict requirement regarding the underlying communications protocols implied that a prospective user company had to install a completely new OSIbased infrastructure if it wanted to employ X.400, a very costly exercise in terms of both time and money, not to mention training and other end-user related issues." [19].

X.400 lost also because the Internet had a far quicker manner of arriving at standards, such as those for the World Wide Web. Sadly from the point of view of those marketing X.400 based services, the Web began to take off simultaneously around the mid 1990s. Customers therefore were asked to choose between relatively complex X.400 email and an OSI based network, or opt for an Internet connection that gave them functioning email, with a far larger installed base of users, in addition to the Web – clearly no contest. Had X.400 then been able to run on TCP-IP, the choice might only have been between SMTP and X.400 email. It was a huge blunder to make this a choice between whatever package of services ran on the OSI networks, with the package of services that the Internet offered.

Even when the Internet was far smaller, and the communities reliant on it almost insignificant, an attempt to "wipe the slate clean" with X.400 and underlying protocols fizzled out very quickly. It may not have been intentional, but this attempt was, nonetheless, backed by all the members of the ITU. In the intervening 15 years the Internet has more or less doubled every year [20], and supported many large communities crucially since then, many of which would actively defend it against such attack. Surely current efforts to look beyond the Internet stand far less chance today?

BUYING CONTROL

Having failed to dislodge the Internet through the standardization process, the telecommunication establishment seemed to adopt a different tack. During the dot com boom that shortly followed the X.400 debacle, telecommunication companies invested a lot of money into the Internet. Part went into purchasing useless dotcoms, and part into vast quantities of optic fiber: hundred of thousands of km in the USA alone. Many large telecommunication companies went bankrupt as a result [21], and the 'dark' fiber became available at fire sale prices, thus subsidizing the growth of bandwidth-hungry multimedia services on the Internet.

If the original goal was to gain control over the Internet, this was the worst possible outcome. There does not seem to be much analysis of how these bizarre investment decisions were taken by companies that understood telecommunications well. However, one factor there may be some consensus on, is that the people taking these investment decisions did not know the Internet very well. There does seem to be a generation gap at work here. Most of the users of Internet services such as Twitter are relatively young [22], while decision makers in governments and telecommunications companies probably make far less use of such services, or even the SMS feature on their mobile phones.

NEGOTIATING CONTROL

Taking the Internet head on with the OSI networks and X.400 failed. Throwing money at it in the dotcom boom did not work either. The next effort to gain some control over the Internet was by talking. The main output of the World Summits on the Information Society (WSIS) in Geneva and Tunis, besides plenty of platitudes, was the Internet Governance Forum (IGF) [23]. But it was never clear what this forum would really do, other than talk a lot.

There is a famous quote "who do I call if I want to speak to Europe", incorrectly attributed to Henry Kissinger, which could also be applied to the Internet. Governments and the private sector already had a forum to talk to each other, the ITU. When discussing Internet governance, whom else do you invite, if you wish to have binding agreements?

While the IGF process is under review, this forum seems to be taking a new tack altogether: looking beyond the Internet. The problem with that is, that the Internet is not about to die. Instead, it is continuing to become more and more central to our lives. With the explosion in smart phone usage, even the phone is largely becoming an Internet appliance.

FULL CIRCLE : BACK TO MINITEL?

In concluding this brief tour of data network history, mention should be made of Minitel. [24]

"The Internet has reached mainstream adoption in the mid-1990s. But, few realize that the majority of the French population has enjoyed the conveniences of the Internet from the early 1980s through use of a videotext system called Minitel. Minitel consisted of a small monitor and keyboard, which used the phone connection to transmit information. Minitel was used for online banking, travel reservations, information services, online grocery shopping and messaging services."

In other words, there was flourishing e-commerce on the conventional phone network at least a decade before the Internet. Unlike the relative anarchy of the Internet, Minitel was controlled centrally, by government–owned France Telecom [25]. Unfortunately, Minitel was not extended as a global standard. How ironic it would be, if what this conference envisages, as a more secure, controllable network to replace the Internet, looks a lot like Minitel, which has been in the grasp of telecommunication companies for 25 years now! However, it is too late to think about Minitel now: the opportunity to make it an international network has been irretrievably lost. What would have been a worldwide smash hit in the 1980s, as it was in France, now has no hope of ever replacing the Internet.

There have been, without doubt, a series of bad decisions made by the telecommunication establishment in dealing with Internet related matters, and attempting now to look beyond the Internet will surely belong on that list. The Internet has no competition, while it enjoys the support of large communities for whom it serves as the spine.

LOOKING AT THE INTERNET, INSTEAD OF BEYOND

It is, of course clear, that not all is hunky-dory with the Internet. There are problems where it could do with a helping hand from the ITU. No doubt, many lessons learnt the hard way by standardization bodies have been thrown overboard, in an effort to keep up with the rate of change on the Internet. There is surely room for cooperation here.

How might mistakes as made in the past be avoided in the future? The process of decision making at the ITU deserves scrutiny. The ITU can ensure, first of all, that all stakeholders are present around the discussion table. A discussion of the future of the Internet should, for instance, involve the Linux, wikipedia and other large communities that have critical stake in it. Decisions in ITU on matters related to the Internet must be in the hands of people who know the Internet and modern technologies well. Particularly in the case of delegates from small developing countries, there may be need to improve technical training and assistance.

Most importantly, the ITU must seek to complement what other international bodies do, support and build on their work. In particular, it should look for problems that bedevil the Internet, which that public network is unable to fix by itself. One such is the vast amounts of junk mail filling our data pipes. The ITU, which can bring together all the governments of the world, would be perfect, to tackle this problem. The telecommunication companies, in any case, would be delighted to regain so much wasted capacity on their networks, and hence should also support this.

Providing access to the computer and the Internet to everybody, irrespective of disability, is not only a duty of the governments under Article 9 of the UN Convention on the Rights of Persons with Disabilities. It could also be a highly cost effective way of integrating the disabled into the education system, and a perfect medium for their gainful employment. Working with multinational companies to ensure that their products are accessible is far easier at the level of the ITU, than at the level of national governments. Access needs to be an integral part of standard setting at bodies such as the ITU, so that it is automatically part of the design of electronic products. This approach also makes good business sense. With large percentages of the world disabled or old, a large number of communication products will be sold this way.

While the first suggestion relating to the curtailing of junk mail would result in huge savings, the second one, relating to accessibility, would dramatically expand the market for telecommunication services and products. A significant segment of humanity finds keyboards and small screens inaccessible. Others suffer from communication handicaps that technology could reduce. It would take very little adaptation to produce hugely successful results, as was shown by the Raku-Raku phone which was originally meant for the elderly population. Fifteen million such phones were sold in a short time, because everybody loves easy-to-use technology. [26].

If the ITU works to help improve the Internet, it will quickly win the support of the many communities that depend on the Internet for their existence. Loose talk on replacing it with a new network, on the other hand, will attract huge opposition, as soon as the effort is perceived to be credible.

CONCLUSION

Had the telecommunications establishment made an effort to make Minitel into a global network in the 1980s, electronic commerce under its control would have become commonplace at least a decade earlier, and the Internet may never have become the "slate", the primary global network. This role the Internet won, even though it had far poorer capabilities than Minitel. The introduction into the market of Internet incompatible X.400 based email services, backed by the large telecommunications companies and governments, can be seen as an attempt to wipe the slate clean, with new OSI protocols instead of TCPIP. When that backfired, the Internet gained momentum, and the telecommunications establishment seemed to lose confidence. It put lots of money into Internet companies and bandwidth during the dotcom boom, which backfired too. The WSIS and the IGF have not gained the establishment any traction in Internet matters either. The Internet is very clearly different from any adversary that governments, multinational companies and organisations had encountered before, something with which they all must learn to live.

The ITU will need to learn how to work better with other groups that set telecommunications standards, in manners radically different from its own. Neither its members, nor it, enjoy the monopolies they once did. The Internet is here to stay, this "giant collaborative network capable of meeting the demands of more than one billion users". No doubt it has serious problems in its future. However, given the size and political clout of communities which cannot exist without the Internet, the ITU should forget about wiping the slate clean and starting afresh. It might help tidy up the slate a little.

An easy way for the ITU to become useful to the Internet would be to focus on serious problems relating to the Internet, which others do not seem able to solve, yet seem well within the abilities of the ITU. Solving these problems could also be in the business interests of ITU members. Helping the disabled get onto the Internet sells phones and bandwidth to a large, untapped segment of humanity. Reducing Internet spam reclaims bandwidth sewage, in a sense, besides showing the Internet community what countries and large companies working together can achieve.

The ITU should also help the telecommunications establishment avoid serious mistakes in dealing with the Internet, the kind that have cost money, time and momentum in the last decades. An argument can be made for a stronger leadership role in the telecommunications industry, but perhaps of a special kind.

The Internet has brought nothing less than a revolution to the world of telecommunications. Leadership in revolutionary times is different. Wise words presumably came from the politician Alexandre Auguste Ledru-Rollin, who was steeped in the lessons of the French Revolution: "There go my people! I better find out where they are going, so I can lead them there."

REFERENCES

NOTE - Except for [19], all references are freely available on the Internet. This has been done to make it convenient for researchers wishing to pursue the arguments made, without having to access paid databases.

[1] http://www.itu.int/ITU-T/uni/kaleidoscope/2010/

[2] http://www.google.com/search?q=define:internet&ei=RfVPTNeAOY3GrAeYutXXDQ&ved%20=0CB0QkAE

[3] http://wordnetweb.princeton.edu/perl/webwn?s=internet

[4] http://oreilly.com/catalog/debian/chapter/book/glossary.html

[5] http://ccs.mit.edu/21c/iokey.html

[6] http://itservices.uchicago.edu/docs/glossary/

[7] What is the Internet (And what makes it work), Robert E. Kahn, Vinton G. Cerf, December, 1999, http://www.cnri.reston.va.us/what_is_internet.html

[8] http://en.wikipedia.org/wiki/History_of_Linux

[9] http://news.cnet.com/Brazils-love-of-Linux/2009-1042_3-6245409.html

[10] http://news.softpedia.com/news/Kerala-039-s-Schools-To-Use-Only-Linux-80400.shtml

[11] http://www.lawyerment.com.my/library/publ/comm/review/ d_8.shtml

[12] http://www.rbi.org.in/home.aspx

[13] http://www.sebi.gov.in/

[14] http://www.trai.gov.in/

[15] http://www.toad.com/gnu/

[16] http://computer.howstuffworks.com/internet/basics/whoowns internet.htm

[17] Arun Mehta, A critical look at Indian telecom policy in the '90s, May 11 2001, Chowk, http://www.chowk.com/articles/a-critical-look-at-indiantelecom-policy-in-the-90s-Arun-Mehta.htm

[18] http://en.wikipedia.org/wiki/X.400

[19] Even Much Needed Standards Can Fail – the Case of email – K. Jakobs, The Journal of The Communications Network, Volume 5 Part 1 – January–March 2006

[20] Andrew Odlyzko, Internet growth: Myth and reality, use and abuse http://www.dtc.umn.edu/~odlyzko/doc/internet.growth. myth.pdf

[21] http://en.wikipedia.org/wiki/Dot-com_bubble

[22] http://www.llrx.com/features/generationgap.htm

[23] http://www.itu.int/wsis/outcome/booklet.pdf

[24] http://techtheory.blogspot.com/2009/02/two-technologicaltales-email-and.html

[25] http://www.dlib.org/dlib/december95/12kessler.html

[26] http://www.nttdocomo.co.jp/english/corporate/csr/report/_user/feature/

[27] http://www.abdn.ac.uk/philosophy/endsandmeans/vol1no2/graham.shtml

PARTICIPATORY APPROACH TO THE REDUCTION OF THE DIGITAL GAP IN AMAZON REGION OF ECUADOR IN THE FRAMEWORK OF THE "INNOVATION FOR DEVELOPMENT" PROGRAM

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ABSTRACT

This work illustrates the methodological approach followed in the Province of Orellana, Eastern Ecuador, for the realization of a telecommunication network infrastructure between the capital of the Province, the city of Puerto Francisco de Orellana (also known as El Coca), and some peripheral communities located in the surrounding of the tropical moist forest. The project has been implemented in one of the poorest countries of Latin America, in a remote and disadvantaged area where the lack of communication infrastructures and the absence of almost all public services generates a strong migration towards the capital. In this context, in 2008, it was conceived a project for the development of a communication system that allows the provisioning of basic intranet services for distance learning, telemedicine and internet connectivity. The main scope of the project was the development of an approach focused on the technological transfer to the local population, to start a reduction process of the digital gap in the area. The aim of the project has been achieved thanks to the direct enrolment of local municipalities, small entrepreneurs, communities and local NGO.

The technological transfer to local players and the choice of a suitable platform, designed for a simplified, low cost management, guarantee the sustainability and scalability of the project. The declaration of interest in the infrastructure by the Municipality enables the economic sustainability of the project.

Keywords— *Digital inclusion, Wireless networks, Wi-Fi, Anti-digital divide infrastructures, Broadband coverage, Technological transfer*

The project has been awarded a grant by the InterAmerican Development Bank under the "Innovation for Development" program.

1. INTRODUCTION

The Province of Orellana, in Amazon region of Ecuador, represents a typical case where IP based services and access to internet through broadband connectivity can support local social and economical development, and can be leveraged to support economic and social inclusion of the population, as well as avoid massive emigration of potentially productive human resources.

Among the several problems that affect the region of the municipality of Puerto Francisco de Orellana, the main one is population isolation caused by the inadequacy of transport and communication facilities and infrastructures. Telecommunication facilities are generally unsatisfactory: mobile coverage is not uniform and broadband internet connectivity is absent from most of the territory of the province. These problems affect not only the population, but also public facilities and services, such as scholar and health facilities. Orellana is the youngest province of Ecuador with a huge territory extending from the administrative capital, the city of Puerto Francisco de Orellana to Peru. **Fig. 1** indicates the location of the province. The city of Coca first appeared on maps at the end of the eighteenth century, and until even the 1970s remained a forgotten outpost in the moist tropical forest, cut off from the rest of the world except by boat or plane. Then, the discover of huge hydrocarbon reserves drew the settlement of oil companies in the area and the construction of several infrastructures boosted the capital's population, attracting new settlers from the Andean zone and the forest villages. The population has recently suffered a deep change of the environmental, social and living context, as a consequence of the exploitation of huge oil resources. The communities of the Amazonian Region are settled in a huge area with very low population density.

They are connected to the capital and among each other through unpaved roads and water ways. In this context, the realization of a wireless infrastructure between Coca and the village named Dayuma is proposed as a solution for the problem of the digital gap of the latter.

Fig. 1- Location of the area where the project is being carried out.

Dayuma is not far from Coca (about 40 km, 25 miles). Despite this, the poor life conditions of the inhabitants are tightly bound to the problems in the transportation system, lack of communication facilities, inadequacy of general services (school, health, etc.). These primary services, likewise the population, suffer from isolation. The realized infrastructure will facilitate access to information, and local use of ICT technologies and exploitation of its benefits, as well as enhance health assistance and schooling through on site digital applications.

Nowadays the access to communications means in developing countries is considered a major need, together with the supply of water, food and other primary goods [1]. IP-based intra and extra connectivity has proved to represent an effective way to connect poor and remote communities to enhanced services through, among others, applications of basic telemedicine tools and distance learning. The access to the internet itself, together with the possibility to deliver low cost services (such as VOIP) can generate self-fulfilling processes of inclusion and development [2].

The possibility to exploit low cost wireless technology in Developing Countries has been widely discussed in literature [3]. Several authors have demonstrated that wireless LAN (WLAN) based solutions are particularly convenient, as the possibility to make profitable investments becomes lower and lower [4], [5].

2. PROJECT DESCRIPTION

The project consists in the implementation and on-field validation of an ad-hoc solution which allows the construction of high performance digital connections with huge transmission capacity, by means of low cost scalable resources and easy to manufacture hardware components manageable locally by people with basic technical competence, to overcome the digital gap for rural developing communities [6]. The infrastructure that has build, based mainly on wireless connectivity, has provided:

1. direct connectivity between the school of the village and the school of the capital, for distance learning applications
2. direct connectivity between the infirmary of the village and the hospital in the capital, for telemedicine applications

The two connections use the same infrastructure and are configured as Intranet services, and will not require access to Internet, being independent from the presence of an Internet Service Provider and avoiding any access costs. The direct and dedicated communication between the local school in Dayuma and the primary schools in Coca will maximize the school system networking in order to make students of primary and secondary level able to have a distance learning support. The direct and dedicated communication between the local infirmary in Dayuma and the Hospital in Coca will be exploited for first level basic telemedicine diagnosis.

On the other hand, geographical and environmental conditions are not favourable for the realization of a wireless infrastructure. **Fig. 2** shows a zoomed representation of the area; as it can be observed, altimetric profile presents a lot of hills

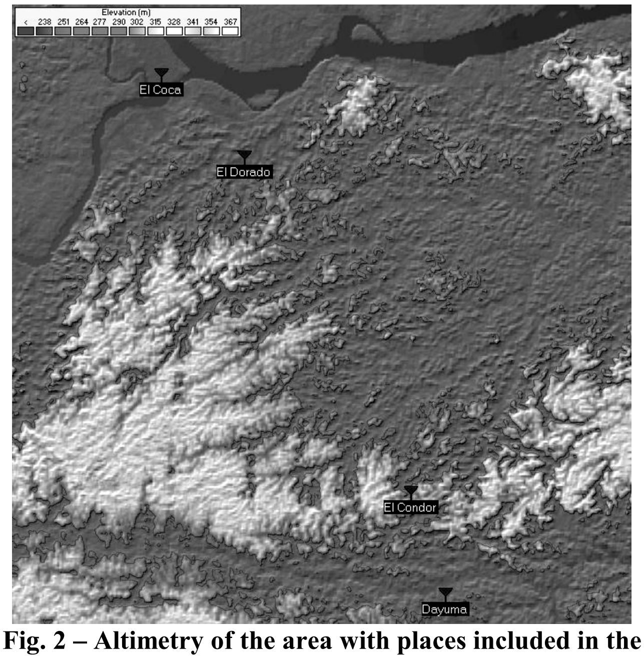

Fig. 2 – Altimetry of the area with places included in the network.

Moreover, due to the vegetation, the conditions for radiofrequency and microwave propagation became very critical.

The lack of streets and communication facilities does not facilitate the identification of places suitable for transmitters' installation. For all these reasons, the project can be considered representative of the construction of communication networks in harsh environments, like the one that can be typically found in several Developing Countries.

3. THE TECHNOLOGICAL SOLUTION

The platform chosen for the project is based on low cost, off-the-shelf x86 hardware components.

As wireless technology, radio equipment compliant to IEEE 802.11a/h standard has been chosen. The manufacturing process of transmitters constructed by assembling low cost 802.11x radios, suitable either for short range, point-to-multipoint wireless links and medium to long distance point-to-point IP backhaul links, ranging from 10 km up to 300 km, has been recently analyzed [6] [7].

The hardware platform is characterized by low power requirements, making feasible its use even in the absence of local electrical grid connection. Taking into account the local weather conditions, our tests showed that a network node hosting three radio devices can be supplied with a single 100W solar panel, with an average battery backup of two days. This characteristic will allow an easy installation in remote contexts, even without power grid connection. Moreover, the reduced dimensions and a very low weight of the equipment allow a great flexibility in term of ease of installation, either at ground level or on existing buildings and towers. It's worth noting that the network is a good testbed for analyzing the performance of low-cost 802.11x microwave links in the harsh weather conditions typical of the Ecuadorean Amazon Rainforest, as well as the durability of commercial hardware products in this context.

The use of a solid Linux-based operating system allow a great flexibility in terms of use of the devices, guaranteeing future upgrade of the system and the possibility to accommodated new devices when needed. Thanks to the presence of a complete graphical user interface and the use of commercial parts, the complexity of the platform is broken down in order to allow a very easy installation and maintenance of the network, easing the technological transfer process to the users with low technical skills.

4. ORGANIZATION OF THE COOPERATION

Our Labs identified as a partner a local Non Governative Organization (NGO). Thanks to the NGO, it has been possible to interact with local administrations, involving schools and the hospital as principal end users of the infrastructure. Moreover, during the activities, the direct relationship with the local administration was the key resource for the developing of the project. This approach is aimed to guarantee the continuity in the time of the project, related not only to the technical maintenance of the network but also to the usability of the services furnished by means of the intranet.

The goal of the project is to make the stakeholders able to implements the infrastructure as their needs require. From this point of view, the knowledge transfer has to be considered as the way to reach the this goal.

The implementation, guaranteed by the scalability of the pilot infrastructure, will be offer the possibility to extend the intranet making able the central administration of Coca to be connected with the rural communities in a bigger and bigger area.

5. THE REALIZATION PHASE

The realization phase consisted of three main steps:

- the construction of a data link between a starting point in the capital of Coca and an ending point in the Municipality of Dayuma; the realization of this link was carried out by means of the construction of multiple wireless hops, making use, where possible, of existing infrastructure and avoiding the use of tall towers in order to offer an exploitable solution for a natural extension of the network in the next future; Fig. 3shows the scheme adopted for the installation;
- the construction of a point-to-multipoint link (with the same wireless technology) to interconnect in Coca the primary schools, the hospital and the Municipality;
- the construction of a point-to-multipoint link (with the same wireless technology) in Dayuma to interconnect the school, the local infirmary and the municipality building that will become a meeting place for the access to the network services.

Fig. 3 - The network between Coca e Dayuma. .Images courtesy of Google Maps.

The realization is being carried on by engineers of the Politecnico di Torino supported by technicians and consultants of the municipality.
During the realization, local technicians have followed all the installations. The aim was to make a practical knowledge transfer to be followed, at the end of the realization of the network, by courses oriented to the used technology.
Thanks to the collaboration of the Municipalities, the access to the public infrastructures (schools, hospitals, infirmaries) was facilitated and installation procedures were accelerated.
Due to the hilly territory with very high tree (see in Fig. 4), it was necessary to individuate high place to install the antennas.

Fig. 4 – High trees on a hilly territory

In order to reduce the installation cost, the idea was to reuse the existing infrastructure to host the link terminals. The best choice was to use existing poles and the water tanks that are located in high place in almost all the area.
Moreover the water system infrastructures is well maintained and furnished of electricity. An example of this installation is shown in Fig. 5 and Fig. 6.
Thanks to the limited dimension of transmitters and antennas, a specific infrastructure is not required to install the equipments. Besides, a very simple wiring is necessary to connect and power the equipment as well. For these reasons, the used methodology represents a suitable solution for installation in atypical and harsh environments.

6. APPLICATION PHASE

One of the focal points of the project is the technological transfer to the local communities. For this reason, a great effort is being dedicated to the organization of courses, both for end users and technicians. In this way, fast and efficient scalability and replicability of the project will be guaranteed.

Fig. 5 – Water tank used in a intermediate hop.

Fig. 6 – Antennas installed on the top of the water tank by means of a little pole.

The training phase will consist in:

- the set-up of an intranet video-conference service for telemedicine applications in the infirmary;
- the set-up of an intranet video-conference service for distance learning applications in the schools;
- the training of local teachers and doctors to the use of video-conference services.

The technological transfer to local technical resources, for the management and maintenance of the technical realizations, was done during the installation phase and at the end of the installation work. In this way, the technicians indicated by the municipality to have in charge the maintenance of the network, have already received the training and can now help the training dedicated to the end users of schools, hospital and infirmary

Thanks to the assistance of the Municipalities, a basic course on the use of multimedia technologies was organized for teachers and doctors in Coca and Dayuma. The course represents the starting point for the use of distance learning and telemedicine facilities. Fig. 7 was taken in the framework of that project, during the training phase.

The courses are being offered under the coordination and technical assistance of the iXem Labs of Politecnico di Torino

Fig. 7 – Puerto Francisco de Orellana, training phase

CONCLUSIONS

The project described in the present paper refers to the possibility to adopt very low cost wireless technology as an efficient means to start the digital inclusion process in very remote and extremely isolated regions in Developing Countries. In this sense, a great effort is being devoted to the realization of a sufficient technological transfer. The realization of this transfer is in progress, but the first obtained results confirm the adequacy of the method and the fast scalability of the proposed solution.

REFERENCES

[1] "ICT and Development: The Kerala Model. Renee Kuriyan. Digital Kerala 2006", Green Chip Publications, Kerala State Information Technology Mission.

[2] S. M. Mishra, J. Hwang, D. Filippini, T. Du, R. Moazzami, and L. Subramanian, "Economic Analysis of Networking Technologies for Rural Developing Regions", *1st Workshop on Internet and Network Economics*, Dec 2005.

[3] L. Subramanian, S. Surana, R. Patra, S. Nedevschi, M. Ho, E. Brewer, A. Sheth, "Rethinking Wireless in the Developing World", *Hot Topics in Networks (HotNets-V)*, November 2006.

[4] S. Nedevschi, S. Surana, B. Du, R. Patra, E. Brewer, V. Stan "Potential of CDMA450 for Rural Network Connectivity", *IEEE Communications Magazine*, Special Issue on New Directions In Networking Technologies In Emerging Economies, January 2007.

[5] R. Patra, S. Nedevschi, S. Surana, A. Sheth, L. Subramanian, E. Brewer. "WiLDNet: Design and Implementation of High Performance WiFi Based Long Distance Networks". *USENIX NSDI*, April 2007.

[6] D. Trinchero, A. Galardini, R. Stefanelli, E. Guariso, F. Cambiotti, F. Troisi, L. Baldacci, D. Della Monica, E. Ragno, R. Moriondo, M. Ancilli, S. Schiavi, "An independent, low cost and open source solution for the realisation of wireless links over huge multikilometric distance", *IEEE Radio and Wireless Symposium*, Orlando, USA, 22-24 January 2008, pp. 495-498.

[7] D. Trinchero, R. Stefanelli, A. Galardini "Reliability and scalability analysis of low cost long distance ip-based wireless networks", *Innovations for Digital Inclusion*, ITU-T Kaleidoscope event, Mar del Plata, Argentina, 31 Aug. - 1 Sept. 2009

[8] D. Trinchero, A. Galardini, R. Stefanelli, P. Venuti, "Performance Of Low Cost Radios In The Implementation Of Long Distance Wireless Links", *2008 URSI General Assembly*, Chicago, Illinois, USA 7-16 August 2008.

SESSION 4

PROTOCOL EVOLUTION AND THE FUTURE INTERNET

A VISION ON THE INFORMATION AND COMMUNICATION TECHNOLOGIES (ICT) USING CLOUD COMPUTING ENVIRONMENT

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ABSTRACT

The government of Japan has announced the new ICT policy in June 2010. One of the points of the new policy is to start the 3D motion image content market in order to create new key industries in the near future as 3D motion image content will become most powerful media for CGM (Consumer Generated Media). In order to activate 3D motion image content industries, the development of an effective and simple tool for making 3D motion image content even by non-experienced people, is required. The Digital Movie Director (DMD) developed by the author, is being evolved as such an effective and simple tool. However, the big computational power requirement in making 3D motion image content has prevented DMD from being widely deployed. The cloud computing technology is supposed to solve this problem, thus, in this paper, the future prospects of the 3D motion image content industries with the cloud computing technology will be explained.

Keywords— Digital Content, Platform cloud computing, Information Communication Technologies (ICT), Consumer Generated Media (CGM**)**

1. INTRODUCTION

The world has strong interest in ICT (Information and Communication Technologies) and its usage. The WEF (World Economy Forum) ranking of countries on their usage of ICT every year is anxiously followed by policy makers worldwide. The WEF announced its ranking of 2007 at the end of March 2010. In this ranking, Japan was placed at the 14th position from the top. This made Japanese government and people feel unsafe, as they became aware of the gap between what they believed to be and the reality in the ranking of ICT deployment – they had the belief of their being top in the world. Because of this situation, Japanese government has proposed to make a new ICT policy which would completely review drawbacks of past policies like, e-Japan, U-Japan and the NGN/NWGN plans.

In this paper, in the first place, this new ICT policy of Japan is roughly explained. One of the points of this new ICT policy is that 3D motion image content is considered a a strategic area to create new industrial activities and markets in the near future. Thus, in the second part of this paper, we show how the activation of 3D motion image content market can be started. As the cloud computing technology would play a key role in this area, cloud computing technology and its influence on the future CGM (Consumer Generated Media) based on 3D motion image content will be explained.

2. NEW ICT POLICY OF JAPAN [1]

In Japan, the ICT Strategic Headquarters has just made the new ICT policy of Japan available on its website (http://www.kantei.go.jp/jp/singi/it2/), in the end of June 2010.

Three major targets of this policy are:

(1) Realization of e-Government for the People

Realize one-stop service for major application procedures and certification acquisitions 24 hour / 7 day by 2020. Over 50% of citizens should be able to use the services with terminals at convenience stores, etc. by 2013.

Realize government transparency and citizen's control of their personal information stored by the government by 2013, by local government by 2020.

Publicize government information in forms available through the Internet for secondary uses by 2013.

(2) Local Areas Vitalization

Realize high quality medical services, such as medical care at home for all citizens including the elderly, regardless of areas by 2020.

Build an environment for school education and lifelong education utilizing ICT by 2020.

Achieve broadband services for all households by around 2015, for improvement of everyday medical care and local vitalization.

(3) Globalization and new industries

Lower CO_2 emissions by the utilization of the smart grid, encouraging zero-energy house with ICT as, promoting reduction of CO_2 at home, and reduce traffic jam by half on major roads nationwide using ITS (Intelligent Transport Systems), etc.

Obtain intellectual property rights and promote international standards to activate content business.

Create a new market with about 70 trillion Yen by 2020, by introduction of new technologies including smart grid, cloud computing and 3D imaging.

In the new ICT policy, one of the strategic fields is the 3D motion image content market.

3. ACTIVATION OF 3D MOTION IMAGE CONTENT MARKET

Unexpectedly high growth of the amount of information in the network causes image information flooding, which is called as "Image Big Bang" now by some.

This "Image Big Bang" has been triggered by the surfing culture in the Internet depending on high usage of search engines by the people. Figure 1 shows this positive feedback spiral mechanism introduced by the progress of three elements, namely, broadband networks, search engines and easy presentation tools for motion image information.

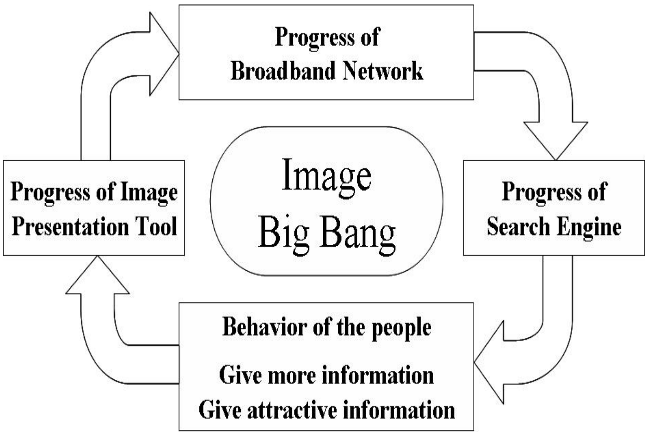

Figure 1: "Image Big Bang" Positive feedback spiral

In order to cope with the big amount of image information under this "Image Big Bang", the following items should be carefully considered.

1) To get a better scheme to gather/archive more image information from all over the world.
   The more, the better.
2) To get a better scheme to make a quicker search engine
3) To get a better scheme to reuse huge amount of archived information
4) To get a better scheme to bridge cross-cultural information
5) To get a better scheme to develop a personal search engine for the personal archive
6) To get a better tool to create 3D motion image content
7) To get a better scheme to accomplish trusted and safer presentation through the Internet. Not to be the shepherd boy of the Aesop's fables
8) To get a better Graphic User Interface (GUI) to manage the broadband Internet easier.

Concerning item (6), an easy tool for making 3D motion image content would be very helpful under the advanced ICT infrastructure for establishing a new CGM market.

However, there is no simple tool to create 3D motion image content by non-experienced people with no special skills in computer graphics or painting. Thus, such a tool should be developed very quickly to jump start the market.

4. CASE FOR VISUAL BY DMD [2, 3]

4.1 Unique Features of DMD

As stated in the previous section, a simple/ automated tool for making 3D motion image content would be helpful. We have developed a tool, namely DMD (Digital Movie Director).

There are several tools supporting the creation of motion image content, such as ALICE of CMU (Carnegie Mellon University), TV4U of NHK (the biggest broadcaster in Japan), and CPSL of NIT (Nippon Institute of Technology) Japan. Today, they are used to create 2D motion image content, however they have the potential to be leveled up to tools for creating 3D motion image content.

Compared with those potential forerunners, DMD is the only one which aims at the creation of 3D motion image content. Another major difference of DMD from those potential forerunners is that DMD has adopted the "iteration method" in the content creation process. Other technologies have adopted the "waterfall method" which is suitable for experienced people who already know how to make 3D motion image content.

However, the most important purpose of developing the new tool is to support people without any skills and experiences in making 3D motion image content. For non-experienced people, "iteration = trial and repeat" is the best method and DMD supports this.

DMD has many functions, described in the following section, to help non-experienced people to create 3D motion image content easily. After creation, he/she can post his/her 3D motion image content to Blogs, a Personal Diary, Travel Reports and etc. to make them more attractive.

4.2 GUI & Functions of DMD

To realize the "iteration method" described above, DMD consists of three important components and an iteration process shown in Figure 2.

The first component is the GUI developed to input information and create the scenario. The core of this GUI is the linguistic model used to input the sentences. The model

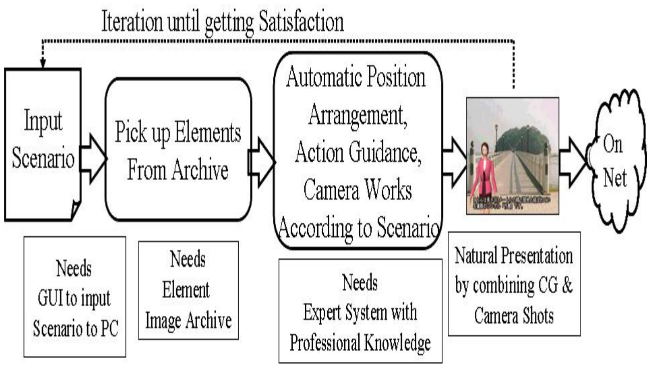

Figure 2: DMD Structure

Figure 3: DMD GUI

used is the subject-verb-object1-object2 (SVOO). Using this model, people can further input, “Facial Actions”, “Camera Works”, “Sound Effects” and pieces of “Background Music”. The GUI of DMD is shown in Figure 3. Every element to make 3D motion image content is stored in the archive of elements described below, thus what people should do is only to select the desired element by a click in the list showing element items.

The second component of DMD is an archive of elements. Today, DMD archive has 48 characters, 5 facial actions, 100 actions, 100 sound effects, 100 pieces of background music, 48 camera works and 47 stages. A creator who wants to make a 3D motion image content can select any of these elements.

People, particularly youngsters, tend to choose famous characters and non-human characters. The interface for adding elements to the archive is now open. Everyone can add his favorite element, like his/her portrait, and enjoy creating contents by using one’s unique elements.

The third component of DMD is the expert system, which provides a way to improve making 3D motion image content. By this expert system, 3D motion image content, made by non-experienced people, would look professionally made.

By those three components of DMD, non-experienced people can easily make 3D motion image content only by adding a scenario to the PC through above mentioned GUI.

4.3 Use Cases of DMD

DMD opens many application fields to the people as shown in Figure 4. The upper part of Figure 4 shows the basic use of DMD. Easy movie creation is the biggest major feature of DMD. As we can select archived element images, we need no skills in painting, computer graphics and camera works. To make a scenario is the only thing that you need to do to create a movie. Thus it is very simple and quick. Everyone from elementary school pupils to 70’s, 80’s even 90’s elderly can enjoy movie creation.

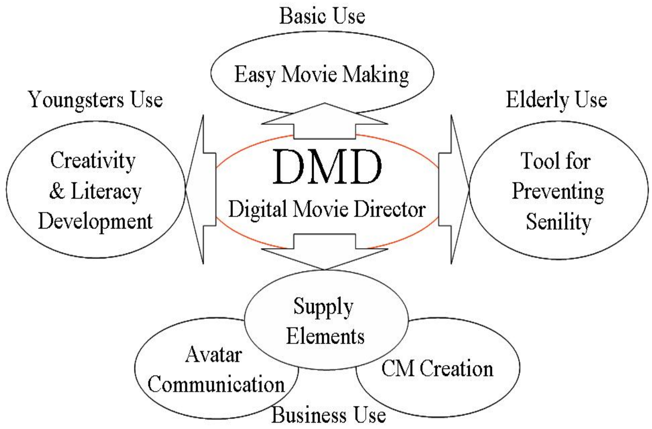

Figure 4: DMD use cases

For youngsters: creativity cultivation

As mentioned above, using DMD the only thing that needs to be done is creating scenarios. When we ask pupils to use DMD, they have shown strong interest in making 3D motion image content, compared with using other method, because DMD gives resulted movies in the shortest time period. Youngsters always ask to get their results very quickly and today only DMD fulfils this requirement. Because of this feature, even low grade pupils would willingly use DMD. This means that they become eager to make scenario. Scenario making is the best education for creativity, thus DMD is useful for youngsters to cultivate creativity.

For elderly: Preventing senility

As they can easily make their views in the form of powerful 3D motion image content, they will upload their creation in the Internet, thus they can enjoy active communication through the Internet. This makes elderly happier and healthier. DMD gives elderly the chance to recover from senility.

For business use: CGM activation

The lower part of Figure 4 shows a sample business use of DMD. There are two categories. One is the supply element business. If you create a popular character and put this character into the DMD archive, so many DMD users may use your character, and then you would be able to collect the usage charge of that character.

The other is the CGM type businesses. CM (commercial films) can be created by DMD, because the cost would be cheaper compared with CM made by human actors and actresses. The other CGM type usage is avatar communication. You would use your favorite avatar to appear in virtual chatting places and enjoy your avatar's life there. As described here, DMD may well have the potential to push CGM culture in the Internet.

As mentioned above and shown in Figure 4, , DMD can be used and enjoyed by everyone, form youngsters to elderly. As a large number of people may be expected to create 3D

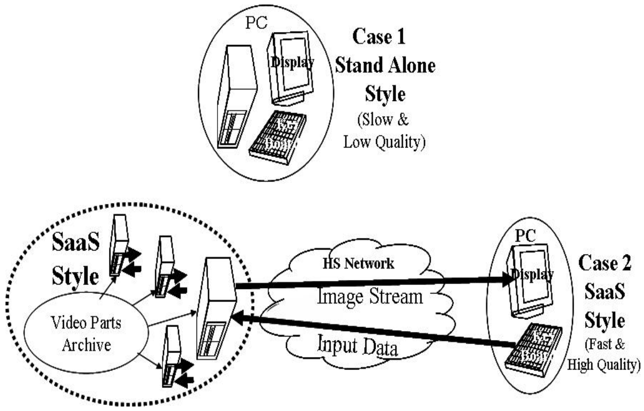

Figure 5: DMD in SaaS style

motion image content, this may activate new key industries and increase the market in the area of the 3D motion image content creation/exchange.

4.4 Defect of DMD

As described in the previous section, DMD has a great potential and can be used in many fields. However, one big limitation of DMD is that it requires high performance PCs for its operation.

DMD also requires a shift from stand alone style to ASP/SaaS (Application Service Provider / Software as a Service) style, as shown in Figure 5.

However, DMD needs more computing power for 3D motion image rendering. Cloud computing may provide the system with more rendering power, thus we would like to deploy cloud computing, as shown in Figure 6.

Using cloud computing for DMD, any low performance terminals, even mobile terminals, can be used as DMD terminals, thus people can enjoy 3D motion content creation by DMD anywhere/anytime.

5. ICT FINAL GOAL: CLOUD COMPUTING PLATFORM [4]

5.1 Evolution of Cloud Computing

A policy of "Buying and Having" ICT facilities is preferable when an organization has very special services/operations, which are so different from those of other organizations that they cannot share any of their facilities.

However, the software world is now changing. Customizable package software can accommodate almost all organizations' requirements. In this case, many organizations can use the same package software, thus the cost/performance of the package software goes down, as the package can be sold to many users.

On the other hand, within an organization, there will be people who want to use very special software for research/analysis/investigation. In this case, to buy this special software for small number of users would have a very poor cost/performance ratio. In this case, SaaS style is also better.

Because of above mentioned software situation today, SaaS style by the cloud computing shown in Figure 7 is being introduced.

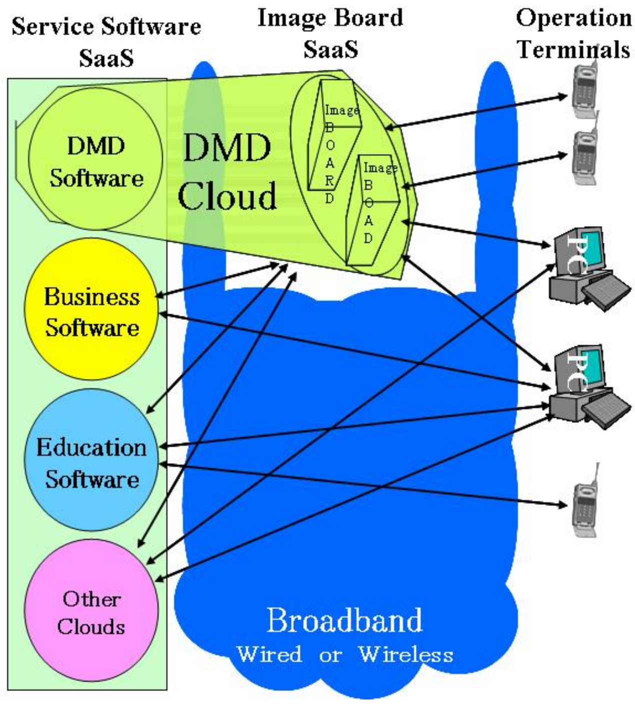

Figure 6: DMD with the cloud computing style

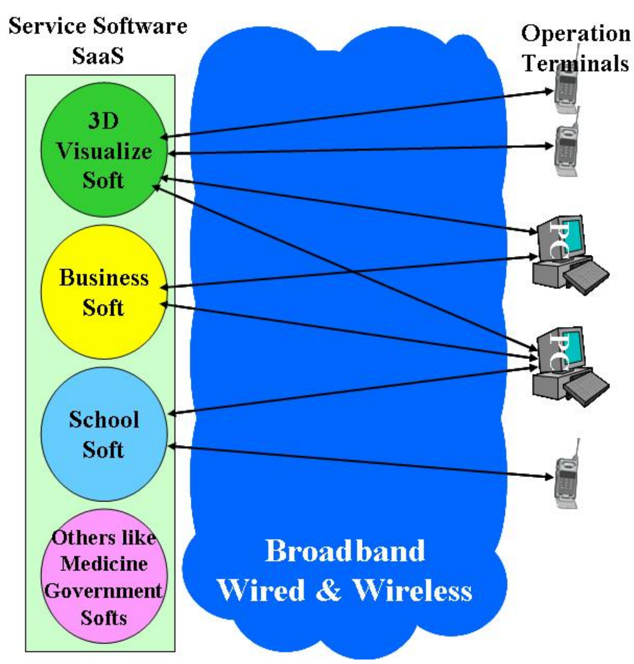

Figure 7: SaaS style cloud computing

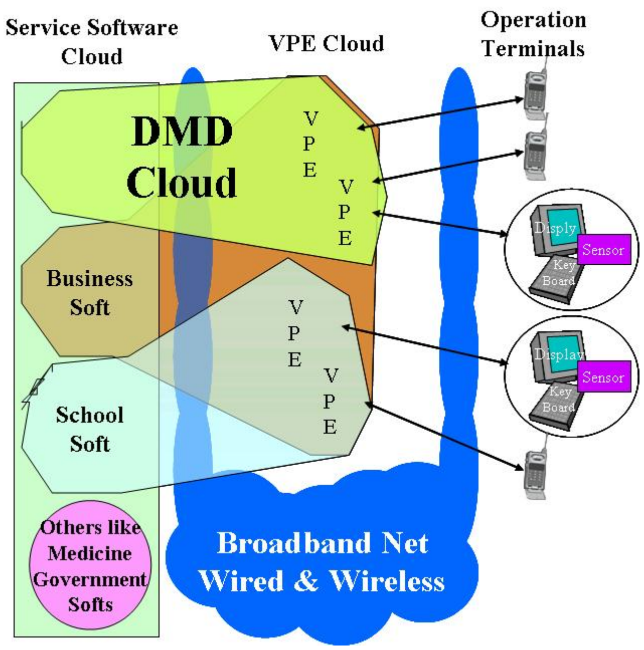

Figure 8: Platform cloud computing

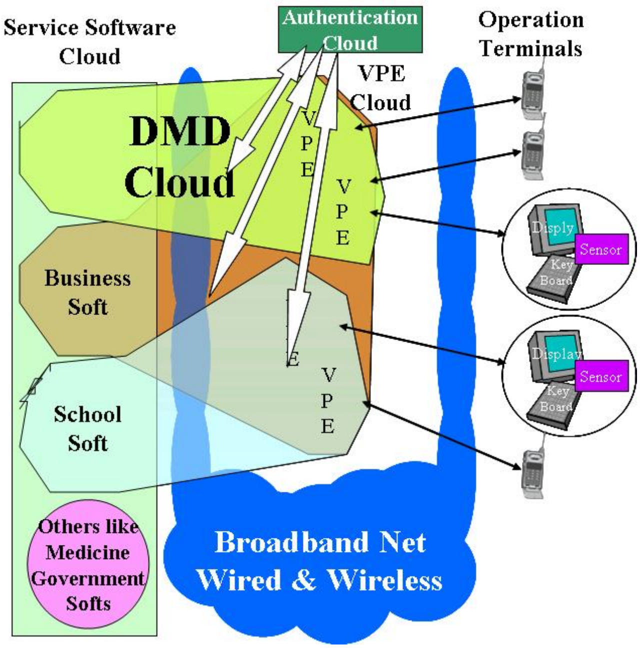

Figure 9: Secure platform cloud computing

Considering security, to supervise/maintain every PC in an organization is now becoming a very difficult task for in-house network managers. It is difficult for them to continuously catch up with the most advanced security technologies. In this case, a "Thin Client Solution" using cloud computing would again be a better solution. Thus, SaaS style will go up to the platform cloud computing shown in Figure 8.

5.2 Need from authentication

Platform cloud computing will give us better cost/performance and safer ICT, however, it needs another two key technologies, namely, independent secure authentication technology, and location dependent information storage technology.

The former would allow us to easily switch between various platform cloud computing systems. The latter would protect our personal information from being controlled by third parties.

These two important technologies are still under development, and we need accelerated research efforts to solve these problems. The final concept of the platform cloud computing, the secure platform cloud computing, is shown in Figure 9, and this will be used for the basis for DMD.

6. CONCLUSIONS

The new ICT policy of Japan has been defined. One of its major targets is to also cover the application level, and 3D motion image content is one of the identified strategic fields. DMD as a powerful tool for creating 3D motion image content is set to become an important element. The platform cloud computing, being the most powerful base for DMD, is considered a key technology.

The ICT market may be expected to change drastically by the introduction of secure platform cloud computing. Thus, sharp prediction and calm decisions over computing architectures will be the keys toward the future.

I appreciate everyone who has supported me in writing this paper.

REFERENCES

[1] ICT policy: http://www.kantei.go.jp/jp/singi/it2/

[2] H. Yasuda, "Digital Content Creation/Distribution in a Broadband-Ubiquitous Environment", IEICE Trans. Info. & Syst., Vol.E90-D, p76-p80, No.1 January 2007

[3] H. Yasuda, "Future Prospect of Digital Entertainment under Ubiquitous Environment", IIITE, Vol.57 No.11, p1399-p1406, 2003.11

[4] Comments on "The cloud computing" to ICT policy p16-p19, http://www.kantei.go.jp/jp/singi/it2/dai52/ sankou2.pdf

HYBRID CIRCUIT/PACKET NETWORKS WITH DYNAMIC CAPACITY PARTITIONING

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ABSTRACT

In this paper, we consider hybrid circuit/packet networks. A hybrid circuit/packet network consists of a circuit network co-existing with a packet network; generally the packet network is embedded on top of the circuit network. However, in certain cases such as the DOE energy sciences network (ESnet) [4], the circuit network and the packet network are deployed side-by-side (e.g. they have common end-node sites and equipment), but they are logically separate and they may have physically disjoint links. Currently, there is no capacity sharing between the packet and the circuit sections of the networks. In this paper, we propose and investigate the characteristics of schemes that enable efficient capacity partitioning between packet and circuit networks while ensuring survivability and robustness of the services. We conduct simulative experiments on ESnet topology with realistic traffic demands. We observe that capacity partitioning between packet and circuit networks enables to support services with enhanced quality of service and robustness along with improved resource utilization.

Keywords— Telecom networks, circuit network, packet network, wavelength services, packet services, network survivability.

1. INTRODUCTION

The use of optical networks for supporting bandwidth-hungry services is an attractive proposition to ensure wide-area reach and huge amount of inexpensive bandwidth [12]. Two services which are expected to be prevalent in future optical telecom backbone networks are packet/IP services and wavelength services (WL) [1]. The IP services include the traditional data services such as VPN, teleconference, data backups etc. These services are accommodated over IP packet technologies and are well established in carrier networks. Wavelength services include bandwidth-intensive applications such as terascale scientific experiments. These services, which are characterized by strict QoS requirements, require circuit-switched technologies to deliver guaranteed bandwidth and are managed directly at the optical layer. Wavelength services currently make up a small fraction of carrier traffic today, but are expected to grow as applications such as e-science emerge [2,3,4]. So, important and timely research is needed to enable telecom carriers to design cost-effective hybrid circuit/packet networks by jointly supporting packet and wavelength services.

Hybrid circuit/packet networks consist of a circuit network co-existing with a packet network; generally the packet network is embedded on top of the circuit network. However, in certain cases such as the DOE energy sciences network (ESnet) [4], the circuit network and the packet network are deployed side-by-side (e.g. they have common end-node sites and equipment), but they are logically separate and they may have physically disjoint links. The circuit network in ESnet known as science data network (SDN) provides dynamic and scheduled circuit services. ESnet's SDN is based on traffic-engineered circuits that can be likened to a comparable dynamic wavelength circuit network in terms of flow handling end to end. This paper will make that assumption when dealing with SDN circuits. Unfortunately, the bandwidth partitioning between these two networks is relatively fixed. However, it would be very desirable to have dynamic partitioning of the bandwidth between circuit and packet services, so that the bandwidth can be easily migrated from the packet network to the circuit network and vice versa. For example, traffic variation over the time of the day can be exploited by accommodating large data transfer for terascale science applications during the night and leaving exploitable capacity for packet traffic during peak hours of the day. Note that several approaches for hybrid networks supporting jointly packets and services were proposed more than a decade ago (e.g. in [13] for the specific case of a STM/ATM network architecture). But, in our work, there is no mixing of packet and circuit traffic on the channels, while we only consider borrowing of *capacity* between circuit section and packet section of the network as it will be exemplified in the following over the ESnet.

In this paper, we consider the opportunity to dynamically partition the network capacity across circuit and packet services. The capacity can be migrated from the circuit network to the packet network, e.g., by loaning idle backup circuits of wavelength services to support IP services. In practice, wavelength circuits are often protected by dedicated backup circuits (1+1 protection) [5], which are generally idle and are not utilized unless there is a network failure. The backup capacity is normally carrying duplicate traffic, which can lead to huge capacity wastage with future 100G transmission systems. A solution to address this capacity wastage and enable dynamic capacity partitioning is to use 1:1 protection instead, and to loan the backup circuits to packet services, so that all network resources are properly utilized. Similarly, the idle capacity in the IP network can be borrowed to provision wavelength services. When the capacity in the circuit network is exhausted, the wavelength services and their backups can be supported over the idle capacity of the packet network. The re-routing of wavelength traffic over the packet network can be operated quite easily by provisioning new traffic engineered tunnels. Note that traffic engineering in such a network becomes very challenging, since wavelength services must be provisioned in such a manner that the services do not get disrupted during IP link failures.

Efficient traffic engineering techniques also need to be developed for successful operation of hybrid circuit/packet networks. Researchers have already conducted some relevant studies on provisioning survivable IP services in hybrid circuit/packet networks. In [6], the authors have formulated an Integer Linear Program (ILP) for mapping the IP links (lightpath) onto the physical topology so that the IP topology remains connected even if a physical fiber fails. In [7], a heuristic mapping to ensure survivability of IP services in the network was proposed. In [8], survivable routing and wavelength assignment in Layer-1 VPNs has been discussed. Even though these studies are very essential for designing efficient capacity sharing mechanisms between packet and circuit networks, they consider a classical approach where packets are transported over circuit.

In this paper, we explore dynamic capacity partitioning mechanisms to improve quality of service and robustness in ESnet, when the circuit network and the packet network are deployed side-by-side (as shown in Fig. 1). We propose schemes that enable us to borrow capacity from the packet network to protect the wavelength services in the circuit network. In Section 2, we discuss techniques for dynamic circuit/packet partitioning in hybrid circuit/packet networks. In Section 3, we present mathematical formulations for capacity migration from the packet network to the circuit/SDN network for protecting the wavelength services in the ESnet. In Section 4, we present results from our experiments in the ESnet. Section 5 concludes the paper.

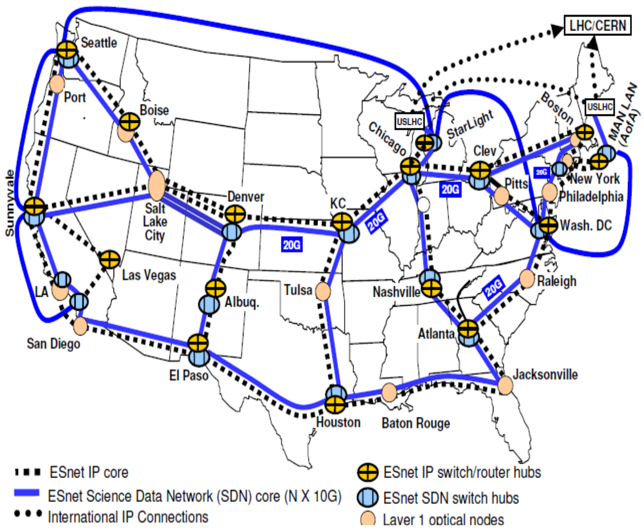

Fig. 1: DOE Energy Science Network (ESnet).

2. DYNAMIC CIRCUIT/PACKET CAPACITY PARTITIONING

As already mentioned in the previous section, bandwidth can be dynamically migrated from one portion of the network to another. We briefly discuss principles for bandwidth migration from packet network to the circuit network and from circuit network to packet network.

Bandwidth can be migrated from the circuit network to the packet network, e.g., by reusing the already deployed backup circuits capacity. The idle backup capacity in the circuit network can be loaned to support traffic in the packet network. However, the IP/packet traffic over a backup circuit gets pre-empted in the event of failure of the backup circuit or failure of its primary circuit. In case of failure of the primary circuit, the backup circuit needs to be restored for serving the wavelength traffic. To reroute the pre-empted IP traffic over alternate paths, the IP topology must remain connected at all times [6]. Thus, only those backup circuits can be loaned which do not disconnect the IP topology in case of a link failure. We note that using backup paths to route IP traffic implies that a large set of logical failures can be caused by a single physical failure (e.g., when a link failure blocks both directly a path routed on that link and indirectly the backup path of a primary path routed over that link). Algorithms and mathematical formulations for this approach are proposed in [9], [10].

Similarly, bandwidth can be migrated from the packet network to the circuit network. In ESnet, the packet and circuit (SDN) networks are physically disjoint. So, when the capacity in the circuit network (SDN) is exhausted, the excess capacity from the packet network can be borrowed to support services in the SDN network. While provisioning the wavelength services over the packet network, we need to ensure that the service does not get disrupted due to IP link failures. Robust and survivable algorithms for provisioning of wavelength services over IP links need to be developed.

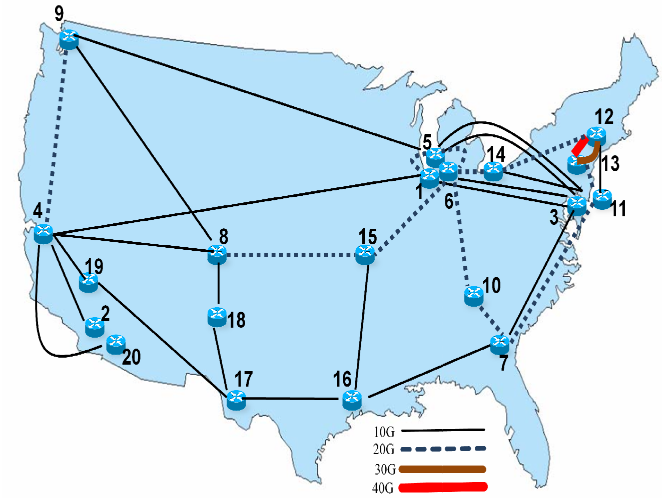

Fig. 2: ESnet Science Data Network (SDN) Topology.

In this paper, we consider a specific case of bandwidth migration from the packet network to circuit network (SDN), which arose when we tried to provide dedicated protection to all SDN reservations in ESnet. In September 2009, we took a snapshot of the ESnet topology and traffic, focusing mainly on the circuit network (SDN). The topology of the SDN network is shown in Fig. 2. There were 15 active wavelength service reservation requests in the SDN network as shown in Fig. 3. The SDN topology comprises of SDN nodes and the logical links between those SDN nodes. The SDN nodes comprise of optical cross-connects (OXC's) and routers. Note that these logical SDN links are mapped over a physical network of optical fibers by leasing capacity from Level 3 network [11]. There are 20 nodes in the SDN network. Every link in the SDN network is bidirectional and is of 10 Gbps capacity. When multiple links exist between a pair of nodes, we aggregate their capacity into a single link. After aggregation, there are 58 links in the SDN network. Out of these 58 links, there are thirty five 10 Gbps links, twenty one 20 Gbps links, one 30 Gbps link, and one 40 Gbps links. We note from Fig. 3 that not all the nodes in the SDN network issue reservation requests. Some of the nodes have multiple reservation requests to them. The reservation requests are of 1 Gbps, 2.5 Gbps, and 3 Gbps capacity. The total capacity lit-up in the SDN network is 840 Gbps. The ESnet team at LBNL has established primary circuits for supporting these 15 reservations. The capacity used is 85 Gbps. The rest of the capacity is idle. In our experiments, we want to protect these 15 SDN reservations by providing dedicated protection (since the network seems to have a substantial amount of excess capacity).

We note from Fig. 3 that there are two reservation requests to node 20. However, there is only one logical SDN link connecting node 20 to the rest of the network (Fig. 2). Thus, it is not possible to provision backup circuits for reservation requests to node 20 with the existing lit-up capacity. *In order to protect these reservations to node 20, we need to either light up new wavelengths on some of the SDN links, or borrow capacity over few IP links from the packet network. The new wavelengths over the SDN links must be lit to ensure two-connectivity at node 20. Similarly, capacity borrowed over the IP links from the packet network must ensure two-connectivity at node 20.* In this paper, we propose two different approaches. In the first approach, we protect as many SDN reservations as possible by using only the existing SDN capacity. We do not light up any new wavelengths. In the second approach, we borrow capacity from the packet network so that we can protect all the SDN reservations including those to node 20. These approaches are discussed in detail in Section 3.

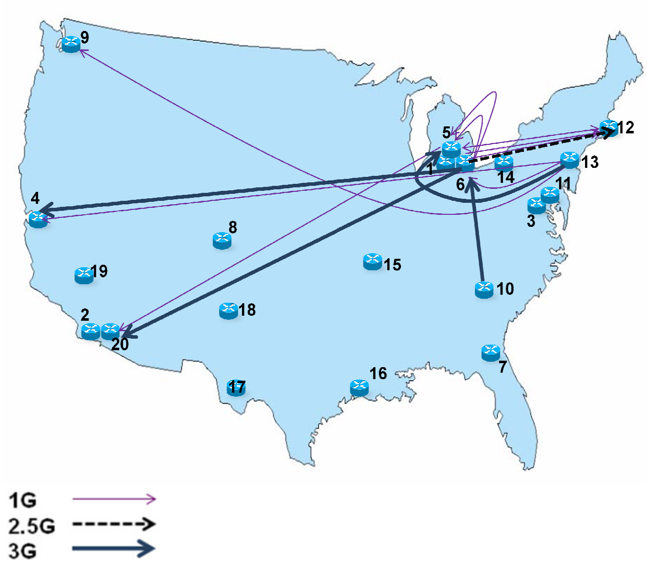

Fig. 3: ESnet Science Data Network (SDN) Reservation Requests.

3. MATHEMATICAL FORMULATIONS

Our objective is to protect each of the SDN reservations with a dedicated backup. We consider two approaches, namely "separated" capacity and "shared" capacity. In both the approaches, we consider the primary circuits that have been provisioned by the ESnet team (we refer to these as fixed primaries), and we aim at finding a set of backup paths for these circuits. The objective is to minimize the total number of wavelengths consumed while providing backups for all reservations. We establish lightpaths over the SDN and IP logical topologies. The exact physical routing of these lightpaths is handled by the level3 network [11]. The two approaches are modeled by means of integer linear program (ILP) formulations described below.

3.1. Separated capacity

In this approach, we protect as many SDN reservations as possible by only using the existing SDN links. Obviously, we cannot protect reservations to node 20. The total lit-up capacity in the network is 840 Gbps, out of which only 85 Gbps is currently used to support the fixed primaries. We can use the rest of the capacity to establish the backups for the SDN reservations. We are not allowed to light new wavelengths or borrow capacity from the IP network. The following are the input parameters and variables used in the ILP formulations for this approach.

Input Parameters:

$P_{l,k,c,t}$: Number of wavelengths carried by link *(l, k)* assigned to the primary circuit of t^{th} connection between source-destination pair c.

s_c: Source node of the source-destination pair c.

d_c: Destination node of the source-destination pair c.

$T_{c,t}$: Traffic in Gbps for t^{th} connection between source-destination pair c.

$W_{l,k}$: Number of wavelengths of 10 Gbps capacity lighted on link *(l, k).*

$C_{l,k}$: Capacity of wavelength channels on link *(l, k).*

A_i : Set of nodes adjacent to node *i*.

$\in$: is a constant.

Variables:

$B_{l,k,c,t}$: Number of wavelengths carried by link *(l, k)* assigned to the backup circuit of t^{th} connection between source-destination pair c.

$\alpha_{c,t}$: is 1, if the t^{th} connection between source-destination pair c is protected and is 0, otherwise.

The integer linear program (ILP) formulation is shown below:

Maximize: $$\sum_{c,t} T_{c,t}\alpha_{c,t} \qquad (1)$$

Subject to

$$\sum_{k\in A_i} B_{k,l,c,t} - \sum_{k\in A_i} B_{l,k,c,t} = \begin{cases} \alpha_{c,t} & if\ l = d_c \\ -\alpha_{c,t} & if\ l = s_c \quad \forall l,c,t \\ 0 & otherwise \end{cases} \qquad (2)$$

$$P_{k,l,c,t} + P_{l,k,c,t} + B_{k,l,c,t} + B_{l,k,c,t} \le 1 \quad \forall (l,k),c,t \qquad (3)$$

$$\sum_{c,t} (P_{l,k,c,t} + B_{l,k,c,t})T_{c,t} \le C_{l,k}W_{l,k} \quad \forall (l,k) \qquad (4)$$

Binary: $B_{l,k,c,t}, \alpha_{c,t} \quad \forall (l,k),c,t$ (5)

The objective function in equation (1) maximizes the total amount of traffic in Gbps that is protected. This approach tries to establish backups for as many SDN reservations as possible. The backup circuits are established over the logical topology. Equation (2) ensures that backup circuits of desired capacity are established. Equation (3) ensures that the primary and backup circuits are link-disjoint. Equation (4) is the capacity constraint which ensures that the total traffic on any link is less than that of the lighted up capacity. Equation (5) ensures that the variables used in this approach are binary. We slightly modify the above formulation to incorporate two possible policies for the routing of the backup paths: load balancing vs. shortest-path routing. We conduct experiments in the separated capacity case with load balancing and with shortest-path routing approaches. The objective function for separated capacity with load balancing is given in equation (6) below. M_L is the maximum flow on any link in the SDN topology. In this approach, we distribute the traffic on every link in the network by pursuing the minimization of the maximum flow on any link. In other words, we try to avoid congestion on some links by ensuring that the flow on every link is below a certain threshold. Another intuitive approach to obtain load balancing would be to minimize the fractional utilization on every link. For ensuring load balancing, we add one more constraint to the set of constraints as shown in equation (7). The drawback of this approach is that the routes may be longer and use more capacity.

Separated capacity with load balancing:

Maximize: $$\sum_{c,t} T_{c,t}\alpha_{c,t} - \in M_L \qquad (6)$$

$$M_L \ge \sum_{c,t} (P_{l,k,c,t} + B_{l,k,c,t})T_{c,t} \quad \forall (l,k) \qquad (7)$$

The objective function for separated capacity with shortest-path routing is given in equation (8). In this approach, we route the backups so that they consume minimum number of wavelength channels. In the objective function (8), we simultaneously maximize the amount of traffic being protected and minimize the number of wavelength channels being consumed. The drawback of this approach is that some of the SDN links may be over crowded.

Separated capacity with shortest-path routing:

Maximize: $$\sum_{c,t} T_{c,t}\alpha_{c,t} - \in \sum_{(l,k)} W_{l,k} \qquad (8)$$

3.2. Shared capacity

In this approach, we protect all the SDN reservations. We can protect all SDN reservations either by lighting up new SDN links or by borrowing capacity from packet network, or both. In our approach, we borrow capacity from the packet network. Idle capacity on some of the IP links is used to support the backups. We can assure protection to all SDN reservations without any additional cost. The IP lightpaths borrowed or the SDN lightpaths established are over the respective IP and SDN logical topologies. The following variable is used in addition to those of separated capacity approach.

Additional Variables:

$\beta_{l,k}$: Number of wavelengths that need to be borrowed from the IP link *(l, k).*

The ILP formulation is shown below:

Minimize: $$\sum_{(l,k)} \beta_{l,k} \qquad (9)$$

Subject to

$$\sum_{k \in A_i} B_{k,l,c,t} - \sum_{k \in A_i} B_{l,k,c,t} = \begin{cases} 1 \; if \; l = d_c \\ -1 \, if \; l = s_c \\ 0 \quad otherwise \end{cases} \quad \forall l, c, t \qquad (10)$$

$$P_{k,l,c,t} + P_{l,k,c,t} + B_{k,l,c,t} + B_{l,k,c,t} \le 1 \quad \forall (l,k), c, t \qquad (11)$$

$$\sum_{c,t} (P_{l,k,c,t} + B_{l,k,c,t}) T_{c,t} \le C_{l,k} W_{l,k} + \mathrm{C}_{l,k} \beta_{l,k} \quad \forall (l,k) \qquad (12)$$

Binary: $B_{l,k,c,t} \quad \forall (l,k), c, t$ (13)

Integer: $\beta_{l,k} \quad \forall (l,k)$ (14)

The objective function in equation (9) minimizes the number of wavelengths that we need to borrow from the packet network for protecting all the SDN reservations. The constraint in equation (10) ensures that a backup is established for every SDN reservation request. Equation (11) ensures that primaries and backups are link-disjoint. Equation (12) indicates that the total traffic on a link is less than sum of the wavelengths that are already lit-up on it ($W_{l,k}$) and the wavelengths borrowed from the packet network ($\beta_{l,k}$). $\beta_{l,k}$ in equation (12) can also be used to indicate the number of additional wavelengths lit-up on the SDN link *(l, k)*. We do not consider that possibility in our formulation. Similar to the separated capacity approach, we consider load balancing and shortest-path routing on top of this formulation. The objective function and the additional constraint for load balancing are given below:

Shared capacity with load balancing:

Minimize: $\sum_{(l,k)} \beta_{l,k} + \in M_L$ (15)

$$M_L \ge \sum_{c,t} (P_{l,k,c,t} + B_{l,k,c,t}) T_{c,t} \quad \forall (l,k) \qquad (16)$$

The objective function for the shortest-path routing is

Shared capacity with shortest-path routing:

Maximize: $\sum_{(l,k)} \beta_{l,k} + \in \sum_{(l,k)} W_{l,k}$ (17)

Finally, we also modify the above ILP formulations for jointly provisioning both primaries as well as backup circuits simultaneously. The primaries provisioned by these approaches are known as "variable primaries".

Shared capacity approach enables us to dynamically partition the capacity between the packet and circuit networks. The network resources can be more efficiently utilized. However, management of both packet and circuit networks will require sophisticated centralized control mechanisms.

4. ILLUSTRATIVE NUMERICAL EXAMPLES

We have conducted experiments using separated capacity and shared capacity approaches on the 15 active reservations in the SDN network. The 15 active reservations amount to 24.5 Gbps of required capacity in total. Using the separated capacity approach, we are able to protect all the reservations except those to the node 20. The two reservations that cannot be protected are from node 5 to node 20 (1 Gbps) and from node 6 to node 20 (3 Gbps). As there is only one bidirectional link connecting node 20 and all other nodes, we were unable to find two link-disjoint routes to node 20. As a result, we could not protect the connections to node 20 using the separated capacity approach.

We recall that fixed primaries are the primary circuits established by the DOE ESnet team and the variable primaries are those primary circuits established by us. We can notice from Fig. 4 that the capacity used for separated capacity approach with load balancing and fixed primaries is 171.5 Gbps whereas the capacity used for separated capacity approach with load balancing and variable primaries is 171 Gbps. In the separated capacity approach with load balancing and fixed primaries, 85.5 Gbps is the additional capacity used for supporting the backup routes. We note from Fig. 4, that there is not much improvement in using variable primaries for load balancing approach. This implies that the primaries established by the ESnet team (fixed primaries) are nearly optimal. Similarly, for separated capacity with shortest-path routing, the capacity used with fixed primaries and variable primaries is 151.5 Gbps and 116.5 Gbps, respectively. Even after establishing backups for most of the reservations, there is still a lot of idle capacity in the network as shown in Fig. 4. Thus, there is sufficient capacity in the SDN network for protecting the existing reservations and supporting newer reservations as well.

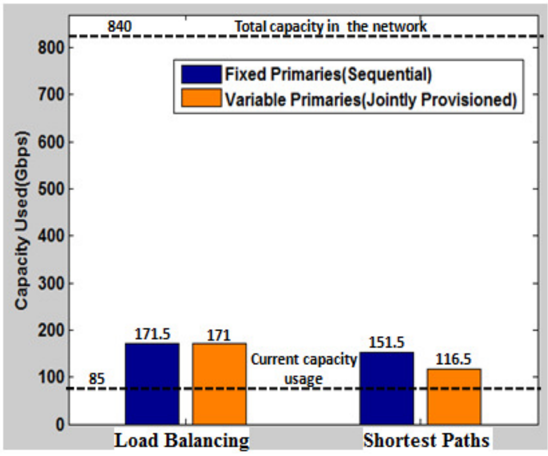

Fig. 4: Capacity used for dedicated protection using separated capacity with load balancing and shortest path routing.

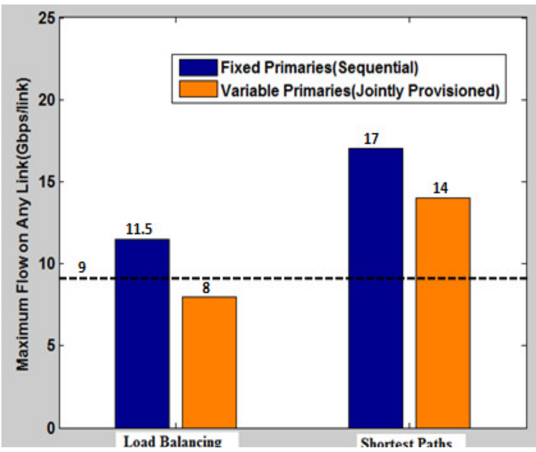

Fig. 5: Maximum flow on any link for dedicated protection using separated capacity with load balancing and shortest-path routing.

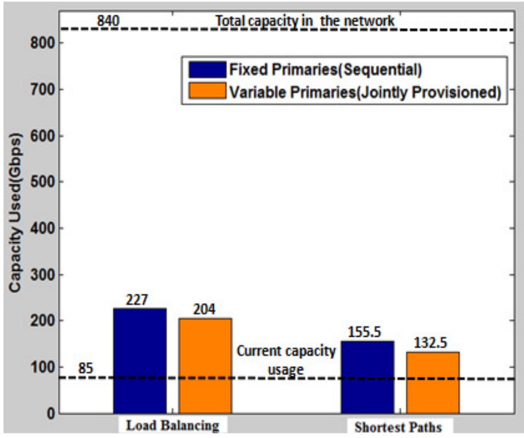

Fig. 6: Capacity used for dedicated protection using shared capacity with load balancing and shortest paths.

Fig. 5 shows the maximum flow on any link using separated capacity with load balancing and shortest paths for fixed primaries and variable primaries. As expected, we can notice that separated capacity with shortest paths has larger maximum link flow than separated capacity with load balancing. For both fixed and variable primaries in separated capacity with shortest paths and fixed primaries in separated capacity with load balancing, the maximum flow is greater than 10 Gbps. In these cases, the maximum flow must be on one of the aggregated links of larger capacity.

Figure 6 shows the results for the shared capacity approach. In this approach, we borrow capacity over some of the IP links that already exist, to protect all the reservations in the SDN network. Even though there is sufficient idle capacity in the SDN network, we are borrowing capacity over some of the IP links from the packet network, so that node 20 is connected to at least two other nodes in the SDN network. When node 20 is two-connected, we can find two link-joint routes and thus, protect the reservations requests to it. For shared capacity approach with shortest paths, we borrow 10 Gbps capacity over the logical link (6-20) for both fixed and variable primaries. For shared capacity approach with load balancing, we borrow 10 Gbps capacity over the logical link (13-20) for fixed primaries and 10 Gbps capacity over the logical link (14-20) for variable primaries. In this manner, by borrowing capacity over a single IP link, we are able to ensure two-connectivity at node 20 and protect all the reservations in the SDN network. However, the capacity used is slightly greater than the separated capacity approach since we are protecting more connections. Thus, we can borrow capacity from the packet network, when

a. The capacity in the circuit network is exhausted. The borrowed capacity from the packet network can be used to provision primaries and backups of the reservation requests.

b. Some of the nodes in the circuit network are not two-connected. Then, the borrowed capacity from the packet network can be used to ensure two-connectivity at all nodes and thus, ensure protection.

Fig. 7 shows the maximum flow on any link using shared capacity with load balancing and shortest paths for fixed primaries and variable primaries. In future, approaches that wisely choose which IP links to be borrowed based on geographical distances have to be developed.

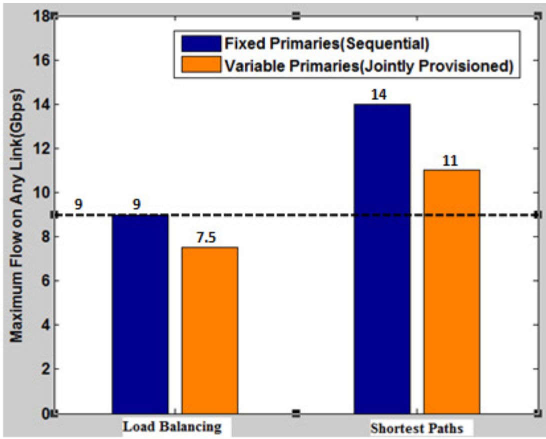

Fig. 7: Maximum flow on any link for dedicated protection using shared capacity with load balancing and shortest paths.

5. CONCLUSION

In this paper, we have discussed techniques for dynamic capacity sharing between packet and circuit network in hybrid circuit/packet networks. We have proposed a novel technique that enables us to borrow capacity from the packet network to the circuit network. We have simulated this scheme, and studied its performance on the DOE Energy Sciences Network (ESnet). From our experiments on the ESnet Science Data Network, we have noticed that, by borrowing capacity on just one IP link from the packet network, we were able to ensure two-connectivity at all nodes and provide dedicated protection for all the active 15 wavelength service requests. When the capacity in the circuit network is exhausted or some of the nodes in the circuit network are not two-connected, we can still support the services in the circuit network and provision backups, by borrowing idle capacity over few links in the packet network. However, when the number of requests increases, it will be efficient to develop a heuristic that is computationally efficient and scalable to large volumes of traffic. We conclude that capacity partitioning between packet and circuit networks enables to support more services and enhance the quality of service and robustness of the existing services along with much higher resource utilization.

REFERENCES

[1] J. Simmons, *Optical Network Design and Planning*, Springer, 2008.

[2] A. Chiu et al, "Network design and architectures for highly dynamic next-generation IP-over-Optical long distance networks," *IEEE/OSA Journal of Lightwave Technology*, vol. 27, pp. 1878-1890, 2009.

[3] http://www.darpa.mil/STO/Solicitations/CORONET/index.htm.

[4] http://www.es.net/pub/maps.

[5] L. Sahasrabuddhe, S. Ramamurthy, and B. Mukherjee, "Fault management in IP-Over-WDM networks: WDM protection versus IP restoration," *IEEE Journal on Selected Areas in Communication*, vol. 20, pp. 21-33, 2002.

[6] E. Modiano and A. Narula-Tam, "Survivable lightpath routing: A new approach to the design of WDM-based networks," *IEEE Journal on Selected Areas in Communication*, vol. 20, pp. 800-809, 2002.

[7] M. Kurant and P. Thiran, "Survivable Routing of Mesh Topologies in IP-over-WDM Networks by Recursive Graph Contraction," *IEEE Journal on Selected Areas in Communications*, vol. 25, pp. 922-933, 2007.

[8] A. G. Yayimli, "Selective survivability with disjoint nodes and disjoint lightpaths for layer 1 VPN," Proc. *IEEE International Symposium on Advanced Networks and Telecommunication Systems (ANTS)*, 2007, Mumbai, India.

[9] C. S. K. Vadrevu and M. Tornatore, "Survivable IP topology design with re-use of backup wavelength capacity," Proc. *IEEE International Symposium on Advanced Networks and Telecommunication Systems (ANTS)*, 2009, Delhi, India.

[10] C. S. K. Vadrevu, M. Tornatore, and B. Mukherjee, " Dynamic Protection-Capacity Sharing for Survivable IP and Wavelength Services in Optical Backbone networks," Proc. *IEEE/OSA Photonics in Switching (PS,)* 2010, Monterey, USA

[11] http://www.level3.com/downloads/Level_3_Network_map.pdf.

[12] J. Berthold, A. A. M. Saleh, L. Blair, and J. M. Simmons, "Optical Networking: Past, Present, and Future," *IEEE/OSA Journal of Lightwave Technology*, vol. 26, pp. 1104-1118, 2008.

[13] Y. Yoshida, A. Arutaki, J. Shimizu, and K. J. Miyahara, "IP/ATM/STM converged next generation network," *NEC Technical Journal*, vol. 52, pp. 28-33, 1999.

A NEW PROTOCOL LAYER FOR USER SPACE FUNCTIONALITY

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ABSTRACT

Evolution of the Internet user has brought attention to the lack of standards for ideal levels of user interaction. The core Internet architecture has not evolved much since its inception, and its user-driven limitations typically constrain one's personal computing infrastructure so that the goals of pervasive and ubiquitous computing are only incipiently achieved. We propose to consider the user's image, or user space, as a significant entity in the Internet model by introducing a new layer of protocols into the Internet protocol stack to support future usage in the Internet. We also present the Identifier/Interlocutor/Locator split architecture for flexible addressing. Standards for such architectures would provide generic user support across heterogeneous networks.

Keywords— Future Internet, user space, ubiquitous computing, Identifier/Locator split, Interlocutor

1. INTRODUCTION

The core Internet architecture has not evolved as fast as the user's ideal levels of interaction. Multiple extensions have provided temporary relief for users, but the need for modification into the core architecture is becoming increasingly evident. Additionally, the user's space is ignored in the design of networking protocols which inadvertently ties down the user space to one single device, thus impairing multi-device management and user mobility.

We use the term *user space* to refer to a user's personal computing environment provided by the operating system (OS), consisting of all configurations relating to the user such as personal settings for applications, network, Internet, desktop, all open connections, sessions, user processes and their related data and open files, including the related data of system processes which are relevant to the state of the user space. We use the term *user space live migration* to refer to a user space being transferred from one host to another while still executing. Lastly, the term *distributed user space* refers to a user space encompassing multiple devices.

In order to separate mobility from portability, and facilitate live migration between heterogeneous devices, we aim to standardize procedures such as user space live migration. Similarly, we introduce the notion of distributed user space, and propose that multiple devices under a single user should all be included under a single user space. This allows grouping of devices for collective communication and management of information towards a single activity. A user space could be set up for a group of users, each with their own set of devices at different sites on the Internet, for the purpose of collaboration.

Our central theme is to consider the user space as a significant entity in the Internet model. We introduce a new layer of protocols, namely the Entity Layer, into the TCP/IP stack to support the user space and provide services to all its processes as a whole. Although explicit significance and support to the user space is overlooked in the Internet model, its addition would address issues in ubiquitous computing such as mobility, live migration, multi-device management, supportive addressing schemes, supportive business infrastructure, sharing, and collaboration over the Internet. We emphasize that user space functionality should be standardized by protocols positioned between the Transport and IP Layer of the Internet model.

An IP address plays three roles in the Internet including Identifier (ID), Interlocutor (IL) – the liaison for the communication session, and Locator (Loc). We propose the Identifier/Interlocutor/Locator split architecture to decouple ownership and network connections from any single device. It allows a clean separation between the user, the host system, and the network of devices, thus reflecting the user's independent identity/ownership and the flexibility of communications in the Internet.

Our concept provides a foundation for previous and future works on mobility and multi-device management [1-5]. ID/Locator split architectures [6-11] separate the different roles of the IP address by associating a permanent ID with the device. Existing works on migration include process migration [12-14], Thin Clients [15, 16] and Thick Client [17] approaches which use virtualization [18]. Our approach differs from existing works by the addition of a third role/attribute, namely Interlocutor, as the communication endpoint and liaison of a communication session. We expand the ownership feature of the Identifier to include multiple devices within the same user space and avoid using virtualization technology for migration.

This paper presents our architecture for the Entity Layer. Section 2 describes issues related to the Internet user. Section 3 discusses related work. We present our architecture of the Entity Layer in Section 4 and discuss its impact in Section 5. Finally, we give our conclusion and describe our future work.

2. BACKGROUND

User-based generic models for ubiquitous computing have gained importance in recent years. However, the Internet lacks a common framework for the acceptance of these models. Efforts have been aimed towards meeting challenges which include seamless mobility, multi-device management, user-driven connectivity solutions and QoS criteria [19], security associations between users and service providers, and support and evolution of business infrastructure for Internet services [20]. In this paper, we address the challenges of mobility and multi-device management.

2.1. Mobility and Migration

Mobility has become synonymous with Portability, compelling the user to carry a device and risk its breakage, loss or theft, while being limited to the hardware of the portable device. The mobility of the user's computing session is only as fast as the personal mobility of the user and portability of the device. Thus, mobility is only incipiently achieved. Seamless mobility, though possible via several solutions, can benefit from improvements in addressing mechanisms, routing, and connection updates [21].

The Internet Protocol (IP) address undertakes multiple roles in communication systems, including Identifier (ID), Interlocutor (IL), and Locator (Loc). To support mobility, the device should be able to change its point of attachment to the Internet, but the liaison for the communication should persist for the entire session. The Interlocutor/Locator split decouples network connections from the device making the connections flexible.

Live migration and mobility of the user space over the Internet provides many advantages such as the independence of mobility from portability constraints, mobility at high speeds over large geographical distances, migrating a computing session off a faulty device or to a device with more resources, the movement of computing processes closer to the data store they are querying to improve computation throughput, or quick transition between devices.

2.2. Multi-Device Management

Earlier, there were multiple users per computer, but today a user owns multiple heterogeneous computing devices and combines them for executing activities due to the following factors:

- Portability in relation to hardware resources
- Specialized roles for each device
- Transitioning between devices
- Specific activities are better accomplished by devices with a particular range of features

Activities are getting progressively complex by requiring multiple devices, access to multiple data stores at different locations on the Internet [1], sharing of data, and collaboration. Complex activities are inherently collaborative in nature, though they are executed on computers made for individual tasks. Hence, managing information across multiple devices and achieving a user experience over the Internet is difficult. We present the notion of distributed user space which enables the user's ownership to span multiple devices, while the interlocutor and liaison of each communication session resides on their respective host and takes a format supported in their respective networks. The Identifier/Interlocutor split decouples ownership from the device, making a user's ownership more inclusive.

3. RELATED WORK

Multiple contributions have addressed issues relating to mobility, migration, and multi-device management. *Beyond device mobility* [22] recommends a communication substrate that delivers a common set of functions to applications in order to provide generic support to preserve a communication across suspend/resume operations. ID/Locator split architectures [6, 7] like Shim6 [8], HIP [9], and LISP [10] improve mobility through flexible addressing mechanisms. They bind a permanent Identifier to the host device, while the Locator remains variable. They may experience latency during mapping between Locator and Identifier. ITU-T Study Group 13's Recommendation [11] provides the general requirements for Identifier/Locator separations in Next Generation Networks.

Several works focus on separation of mobility from portability [23]. Developers could design applications which would save and restore their computing state, but this would neglect the large share of legacy applications available to the Internet. Thin Client systems [15, 16] allow remote connectivity to the user space, but require continuous network access with high bandwidth and low latency. Process migration can be achieved through several approaches [14], and it enables load balancing and fault resilience in distributed computing systems such as Sprite [12] and Chorus [13]. However, it is difficult to perform on heterogeneous architectures, and the lack of applications for process migration has resulted in its lack of research. ZAP [18] uses OS virtualization, which prevents system level dependencies, to provide a consistent view to closed groups of processes for migration. ISR [17] saves the computing state of a user space into the network, along with the entire guest OS of the virtual machine (VM) for restoration on a different machine. ISR requires reestablishment of network connections during mobility, and are problematic to multi-device management since they are based on serial usage of devices [1].

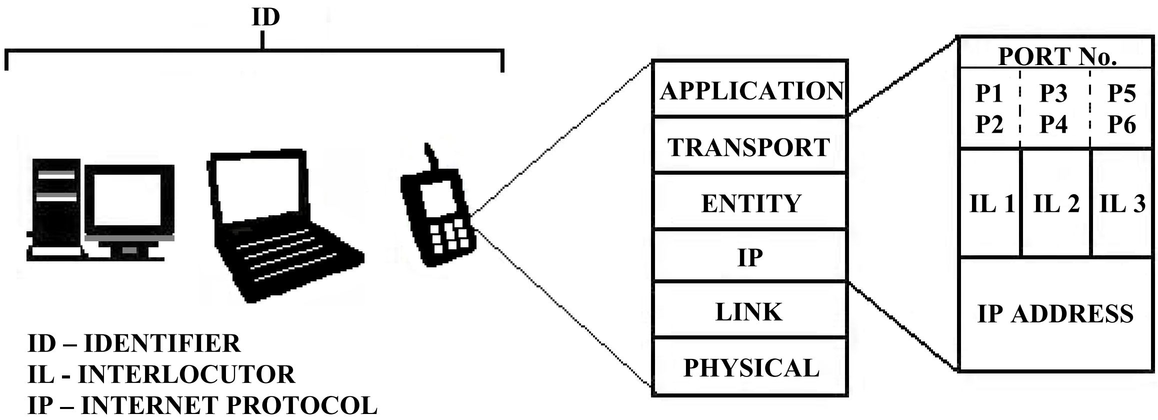

Figure 1: Internet Model with Entity Layer

To enable easy communication between a user's personal devices, [3] elevates device ownership to a first class property and uses an Instant Messaging (IM) architecture [4]. Connecting an ensemble of devices [5] and transferring information between them [2] are some of the approaches to solving the issue of multi-device management. Multi-device interoperability [24] can be achieved via a set of protocols and standards that provide a generic interface for communication between devices having very little knowledge of each other.

For future wireless communications, [20] proposes a consumer-based model to achieve ubiquitous "consumer user's" functionalities in wireless access services. It emphasizes the importance of full number portability achieved via a person-centric identity in the form of an IPv6 address class. It also describes the need for a network-independent trusted means of payment by which the user can procure access services.

4. ARCHITECTURE

In this section, we discuss the design of the Entity Layer and its inclusion into the Internet architecture. The Entity Layer will provide protocols for the decoupling or distribution of the user space from the underlying OS and host device, while the kernel will provide the functionality for mobility, migration, and distributed user space. These functions will be accessible to any second or third party applications, through an interface exported by the kernel. Applications will be created, specifically, to facilitate user space migration and user space distribution.

We make three assumptions in the design: First, users leave their devices on for extended lengths of time. ICT studies on computer usage [25] reveal that up to 60% of household and individual users leave their computer on for extended lengths of time, even when no tasks are being performed. Second, each device hosts a single user space at a time. Third, we explain the details for homogeneous devices using IP communications which is prevalent in the Internet, but the concept can be extended for heterogeneous devices and networks and will be part of future work.

We envisage both live migration and distributed user space being achieved by the user on any device which is party to the operation. This means, the user can be on either the source or destination device for migration, and any joining device for distributed user space.

4.1. Services

We briefly outline the services the Entity Layer would intend to provide: First, it provides a variable to undertake the current role of interlocutor by the IP address. Second, it groups processes under their respective Interlocutor. Third, it provides protocols to support and enable the decoupling of a user space from its host. Fourth, it separates mobility from portability by enabling live migration across the Internet. Fifth, it supports the collection, or a part of, a user's activities by managing its collection of information and tasks on a computer or multiple devices. Sixth, it supports sharing and collaboration of a user's activities.

4.2. Layer Placement

Establishing the appropriate boundaries for the user space in a communications system is ignored in the research of networking protocols [26]. This neglect ties down each user space to one single device, thus impairing user mobility and multi-device management. Communication between devices in the Internet, or heterogeneous networks, with very little prior knowledge of each other, requires a standard set of protocols and generic interfaces which are acceptable to all entities [24]. A new layer of protocols for the user space would fit naturally into the layered structure of the Internet model, and provide the added services, along with a generic interface, to support the future Internet user.

Figure 1 shows the modified TCP/IP stack including the Entity Layer. The Transport Layer provides protocols for the functions on individual processes, and provides services for the applications. The Entity Layer will provide protocols for the user space. The Network Layer provides protocols for functions related to the host system, device, and network that the user space is on.

For mobile and stationary communications, users who are directly involved in a communication are at the end hosts only, and not on any of the intermediate nodes. For interoperability, sharing, and collaboration, information is replicated or exchanged only with end-devices which constitute end-points. By the end-to-end argument [26], the functions relating to the user space can completely and correctly be implemented only with the knowledge and help of the user profile at the end hosts of the communication system. To avoid redundancy and incompleteness of functions, the functionality of the user space should be standardized by protocols which are end-to-end in nature. Hence, in the TCP/IP stack, the Entity Layer and its protocols should be placed above the IP Layer of the Internet model. Again, if the functionality is supported at or above the Transport Layer, it will have to be implemented individually and separately for each process or application. To improve effectiveness and efficiency of the functions, we place them below the Transport Layer so that it can be implemented for all the processes in the user space as a whole. In conclusion, we emphasize that the Entity Layer must be positioned between the Transport and the IP Layer of the Internet Model.

4.3. Addressing

Figure 1 shows the Identifier/Interlocutor/Locator split architecture, where the Identifier is elevated to cover multiple devices under a single ownership, while the Interlocutors reside on their respective host systems and act as liaisons for their communication sessions. Interlocutors are used as control information in upper layer (TCP, UDP) connections and data packets, so it is imperative that they be in a format which is supported in their respective network.

Communication sessions are built on persistent Interlocutors routed through the current Locator, which in turn is used for network functions of locating, routing, and forwarding. Network connections, denoted by port numbers in Figure 1, are grouped under their respective Interlocutors. The Interlocutor is generated identical to the current Locator, taking the format of the IP address, which is prevalent in the Internet, and playing the role of a virtual IP [27].

Mapping of Interlocutors to a new Locator is instantaneous, and requires no complex setup. Mapping for persistent connections is achieved through Mobile IP and Co-located Care-of Addresses (Coa). The ability of the user space to own a single permanent Identifier, multiple persistent Interlocutors, and a variable IP address, within the Identifier/Interlocutor/Locator architecture, enables user flexibility and progress on issues of ubiquitous computing.

4.4. Mobility and Migration

The Entity Layer uses the services of Mobile IP provided by the IP Layer. For persistent connections, mapping of Interlocutors to a new Locator is achieved through Mobile IP and Co-located Care-of Addresses (Coa). Using DHCP, a mobile host can acquire an IP address on the new subnet and use it as a Co-located Coa, negating the need for a separate Foreign Agent (FA). As a node moves from one subnetwork to another, its current Locator changes and generates a new Interlocutor for the particular subnetwork. All Interlocutors remain persistent for the lifetime of their respective connections, but the Identifier remains constant and permanent. Hence, the Locator is location-dependent, the Interlocutor is of a particular location-origin but location-free, and the Identifier is location-independent.

Network connections use Interlocutors of different subnets or domains, and the communication for each separate interlocutor will be directed via the Home Agent (HA) of its respective domain. Hence, we achieve duality [28] and remove the Home Agent as the bottleneck or single point of failure in the Mobile IP architecture. By avoiding a single method of delivery, the mobile host only pays for the extra cost of mobility support, triangular routing, or security perimeter traversal when it is truly needed.

To avoid duplicate addresses, all IP addresses corresponding to live Interlocutors must be reserved on their respective DHCP servers until the related Interlocutor is released and can be recycled via DHCP. The DHCP server could demand that the user space send it regular messages to maintain its Interlocutors and to indicate that the address is still in use. If the message for a particular Interlocutor times out, DHCP recycles the IP address.

User space live migration across the Internet enables complete mobility for the user, who can enjoy freedom from devices while on the move. When the user is ready, the user space can be migrated to a destination device from any point of attachment to the Internet, while its processes and connections are still computing. We assume that a device hosts a single user space at a time, enabling all the device's resources to be available for its current user space. Unlike ZAP [18], resource and process naming conflicts are absent and a private namespace is not required. Hence, resource and system level dependencies can be maintained, allowing processes to use all IPC mechanisms such as shared memory, queues, messages, etc.

For user space migration, a checkpoint–restart [17] mechanism could suspend the user space and resume it on another machine. Like ZAP, this would avoid leaving any residual components on the source machine. The user space is suspended by pausing all the user processes and saving their process state, including memory, CPU registers, file pointers, all open files, and data related to requisite system processes, daemons, and network connections. User configuration, desktop, and Internet settings are also stored before effective migration. The suspended user space can now be transferred via the Internet. On reaching the destination host, the user space is resumed by first, triggering the required daemons and system processes, and feeding them the requisite data. This makes the system environment familiar and conducive to resumption. Requisite files are then opened and linked with file handles. Subsequently, the suspended processes are restored and resumed.

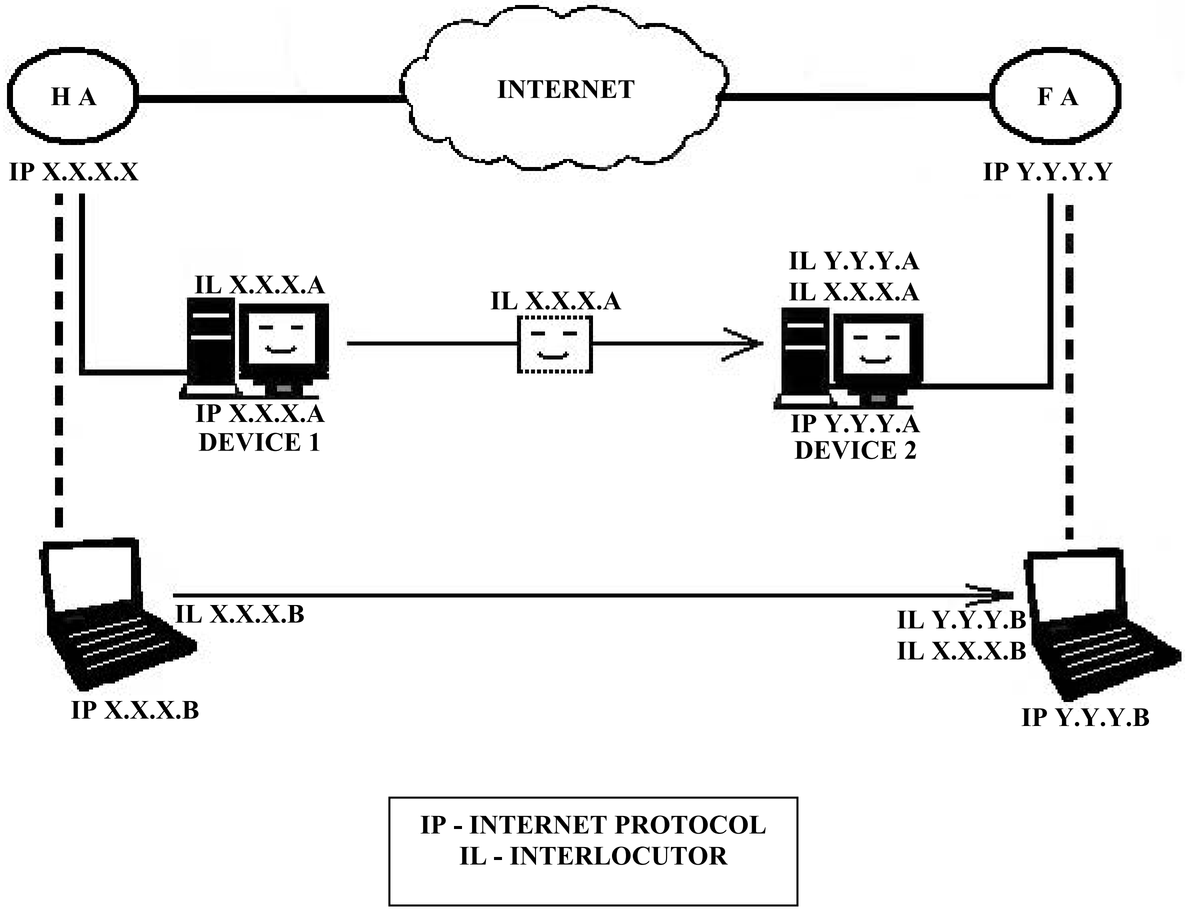

Figure 2: Mobility and User Space Live Migration

To ensure the completeness of its execution, *user space migration* must be an atomic function. After migration, the OS components and processes left behind are insignificant. The device must not retain its previous IP address, but acquire a new IP address from the DHCP server. If a resource is not available on the destination device, the related process is held in suspension as a dormant process until such resource becomes available.

Figure 2 demonstrates the concept of mobility and user space live migration. The Foreign Agent (FA) is labeled to depict the destination domain or network. We explain the procedure for live migration of the user space, from source Device 1 to destination Device 2, using the concepts of Interlocutors and Locators. We assume that the user space has already acquired multiple persistent connections from several domains, and their packets are being tunneled via a separate Home Agent for each respective domain:

- Before user space live migration, the Entity Layer on Device 1 informs each Interlocutor's HA of its intention to move.
- On getting the information, the HAs start buffering the packets for all connections of the respective Interlocutor, and sends back an acknowledgement.
- Upon getting all the acknowledgements, the user space starts to effectively migrate. This is achieved through a separate session and the exchange of control messages between the Entity Layers on both devices.
- After the migration is complete, Device 2 now hosts the user space. Its Entity Layer uses the current Locator to generate an Interlocutor for future connections, and sends the current Locator as its new co-located Coa to the HAs of all previous Interlocutors.
- On receiving the new co-located Coa, the HAs tunnel the buffered packets to the Coa and resume Mobile IP.
- After migration is complete, Device 1 acquires a new Locator from its DHCP server, and is ready to host a new user space.

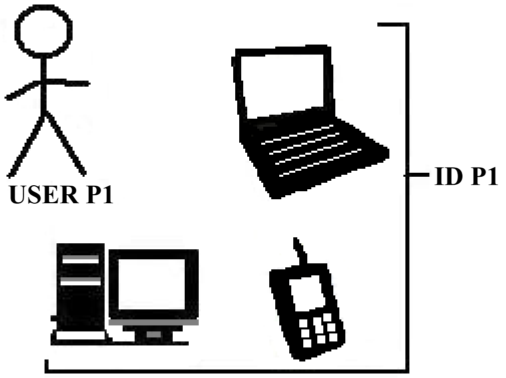

Figure 3: Distributed User Space

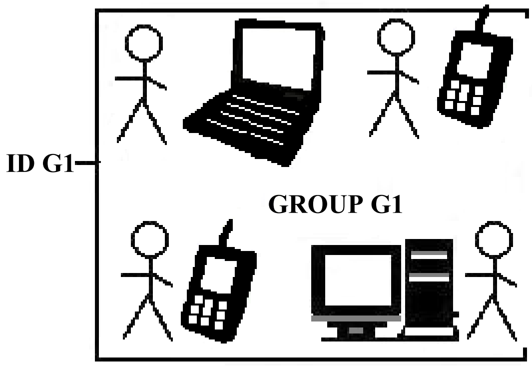

Figure 4: Distributed Group User Space

4.5. Distributed User Space

In this subsection, we present the *distributed user space* based on the following idea: Multiple devices under a single user should all be included in a single user space. We elevate the role of the Identifier, as shown in Figure 1 and Figure 3, to encompass all devices under the ownership of the user. By distributing the user space, the user can perform several tasks on multiple devices and efficiently manage information relating to activities spanning multiple devices. Multiple services, information sources, and tools, which are located at different geographic locations, may be aggregated and used simultaneously for a single user task or collaboration.

By making a distributed user space for a group, the user space can be shared either asynchronously, by each group member alternating tasks on the user space, or synchronously. This notion is illustrated in Figure 4, where all devices come under the group user Identifier. If two or more group members login to the group user space on different devices during the same interval, they can synchronously collaborate by explicitly inputting the other members' interlocutors or locator, or by searching over the Internet for all sessions and computing based on the common identifier. Although technology mentioned in [2-4] can be used, based on the end-to-end argument as described in section 4.2, we emphasize that any technology used for synchronous collaboration will be better placed on the Entity Layer, that is, between the Transport and Network Layers.

We do not assume the availability of a centralized server so as to allow applicability when utilizing public computers as well as personal devices. Hence, all current data of a user space must be present on the user's set of devices. A Distributed User space would use a distributed file system (DFS) that aggressively caches data on all the user's devices across the Internet. Each device acquires its image of the user space by regularly refreshing its state from the DFS. Unlike ISR [17], the distributed user space would require a DFS where devices directly communicate with each other in a peer-to-peer fashion [29] compared to the classic client-server model.

The Entity Layer will support the distributed state of a user space spanning multiple devices, and its protocols will provide standard methods for management of information, and automatic file and data synchronization between devices. The devices need to be connected to each other either directly or indirectly.

5. IMPACT

The benefits of our proposal are:

- separation of mobility from portability
- local communications do not use Mobile IP
- Home Agent is not the bottleneck or single point of failure for all mobile communications
- easier design of applications to provide a user experience spanning multiple devices
- increase in limits of mobility and device ownership/interoperability over the Internet
- Internet can provide localized services for individual connections based on its Interlocutor, since each Interlocutor belongs to a particular domain and subnet.

User space functionality will drive new user-centric technologies, creating opportunities for new utilities based on the evolution of the Internet user. As the concept gradually sinks in, fresh demands will arise from users in the future. Consequently, we would develop similarity standards for host systems and compatibility standards for applications on heterogeneous systems. For example, to provide a rich user experience spanning multiple devices, we will need to develop standards for identifying other devices and communicating with them.

6. CONCLUSION AND FUTURE WORK

We discussed the exigencies for considering the user space as a significant entity in the Internet model, and introduced the Entity Layer of protocols between the Transport and IP Layers of the Internet protocol stack. The Identifier/Interlocutor/Locator split architecture was then presented for flexible addressing. Implementation of our concept is part of future work, and can be extended to develop standards for providing generic support across heterogeneous networks. We envisage the evolution of the Entity Layer side-by-side with the user's role, enabling the user space to become a contemporary image of the user on the future Internet.

REFERENCES

[1] D. Dearman et al., "It's On My Other Computer!: Computing with Multiple Devices," *Proc. 26th Annual SIGCHI conference on Human factors in computing systems*, Florence, Italy, April 2008

[2] R.C. Miller et al., "Synchronizing Clipboards of Multiple Computers," *Proc. UIST 1999*, ACM, pp. 65-66, 1999.

[3] J. S. Pierce et al., "An infrastructure for extending applications' user experiences across multiple personal devices," *Proc. 21st Annual ACM symposium on User interface software and technology*, Monterey, CA, USA, October 2008.

[4] J. Ahn et al., "SEREFE: Serendipitous File Exchange Between Users and Devices," *Proc. Mobile HCI,* pp. 39-46, *2005*.

[5] B. N. Schilit et al., "Device Ensembles," *IEEE Computer*, vol. 37 (12), pp. 56-64, December 2004.

[6] V. P. Kafle et al., "'An ID/locator split architecture of future networks," *Proc.* Second ITU-T Kaleidoscope Academic Conference on Innovations for Digital Inclusion, Mar del Plata, Argentina, August-September 2009

[7] V. P. Kafle et al., "Generic identifiers for ID/locator split internetworking," *Proc. First ITU-T Kaleidoscope Academic Conference on Innovation in NGN – Future Network and Services*, Geneva, Switzerland, May 2008.

[8] E. Nordmark and M. Bugnulo, "Shim6: Level 3 multihoming shim protocol for IPv6," *Internet-draft*, February 2009.

[9] R. Moskowitz and P. Nikander, "Host identity protocol (HIP) architecture," *RFC 4423*, May 2006.

[10] D. Farinacci et al., ``Locator/ID separation protocol (LISP)," *Internet-draft (work in progress)*, March 2009.

[11] "General requirements for ID/locator separation in NGN," *ITU-T Recommendation Y.2015*, January 2009.

[12] F. Douglis et al., "Transparent Process Migration: Design Alternatives and the Sprite Implementation," *Software - Practice and Experience*, vol. 21 (8), pp. 757-785, August 1991.

[13] M. Rozier et al., "Chorus (Overview of the Chorus Distributed Operating System)," *Proc. USENIX Workshop on Micro-Kernels and other Kernel Architectures*, Seattle, WA, April 1992.

[14] D. Milojicic et al., "Mobility: Processes, Computers, and Agents," *Addison Wesley Longman*, February 1999.

[15] R. Baratto et al., "MobiDesk: Mobile Virtual Desktop Computing," *Proc. ACM MobiCom,* Philadelphia, PA, September 2004.

[16] B. K. Schmidt et al., "The Interactive Performance of SLIM: a Stateless, Thin-Client Architecture," *Proc.17th ACM Symposium on Operating Systems and Principles*, Kiawah Island, SC, December 1999.

[17] M. Kozuch et al., "Internet Suspend /Resume," *Proc. IEEE WMCSA*, Calicoon, NY, June 2002.

[18] S. Osman et al., "The Design and Implementation of Zap: A System for Migrating Computing Environments," *Proc. 5th Symposium. Operating Systems Design and Implementation,* ACM Press, pp. 361-376, December 2002.

[19] M. O'Droma et al., "Always Best Connected Enabled 4G Wireless World," *Proc. 12th European Union IST Summit on Mobile and Wireless Communications*, pp. 710-716 , Aveiro, Portugal, June. 2003.

[20] M. O'Droma and I. Ganchev, "Toward a Ubiquitous Consumer Wireless World," *IEEE Wireless Communications*, vol. 14 (1), pp. 52–63, February 2007.

[21] N. Golmie, "Seamless Mobility: Are We There Yet?," *IEEE Wireless Communications*, vol. 16 (4), pp. 12-13, August 2009.

[22] Y. Ismailov, "Beyond device mobility," *Proc. 4th Asian Conference on Internet Engineering*, Bangkok, Thailand, 2008.

[23] D. Johansen et al., "Environment mobility: moving the desktop around," *Proc. 2nd workshop on Middleware for pervasive and ad-hoc computing*, pp.150-154, Toronto, Ontario, Canada, October 2004.

[24] W. K. Edwards et al., "Challenge: Recombinant Computing and the Speakeasy Approach," *Mobicom 2002*, Atlanta, GA USA, 2002.

[25] T. Beauvisage, "Computer usage in daily life," *Proc. 27th International conference on Human factors in computing systems*, pp. 575-584, Boston, MA, USA, April 2009.

[26] J. Saltzer, D. Reed and D. Clark, "End-to-end arguments in system design," *ACM Transactions on Computer Systems*, vol. 2 (4), pp. 277-288, November. 1984.

[27] F. Teraoka et al., "A Network Architecture Providing Host Migration Transparency," *Proc. ACM Sigcomm*, September 1991

[28] X. Zhao et al., "Flexible network support for mobility," *Proc. 4th Annual ACM/IEEE Mobicom*, pp.145-156, Dallas, Texas, United States, October 1998.

[29] R. Hasan et al., "A Survey of Peer-to-Peer Storage Techniques for Distributed File Systems," *Proc. International Conference on Information Technology: Coding and Computing*, vol. 2, pp. 205-213, April 2005.

QUALITY OF SERVICE IN THE FUTURE INTERNET

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ABSTRACT

Whatever the Network of the Future turns out to be, there is little doubt that QoS will constitute a fundamental requirement. However, QoS issues and the respective solutions will not remain unchanged. New challenges will be raised; new ways of dealing with QoS will be enabled by novel networking concepts and techniques. Thus, a fresh approach at the QoS problem will be required. This paper addresses QoS in a Future Internet scenario and is focused on three emerging concepts: Network Virtualization, enabling the coexistence of multiple network architectures over a common infrastructure; In-Network Management, improving scalability of management operations by distributing management logic across all nodes; the Generic Path based on the semantic resource management concept, enabling the design of new data transport mechanisms and supporting different types of communications in highly mobile and dynamic network scenarios.

Keywords— QoS; Network Virtualization; Generic Path; In-Network Management

1. INTRODUCTION

Quality of Service (QoS) has played a relatively minor role in the innovations proposed in the framework of Future Internet research initiatives. Paradoxically, the importance of QoS, in a wide range of application scenarios, is likely to become more crucial than ever before – pervasiveness of network-based applications and emerging trends, such as cloud computing, will contribute to exacerbate the need for QoS and to make network performance more crucial than ever before for an increasing number of application scenarios. In fact, the capability to guarantee deterministic QoS is usually included in the “wish list” for the Future Internet.

In spite of considerable research effort devoted to QoS technologies in last few years, QoS mechanisms have been deployed in large scale commercial environments to a limited extent. Part of the problem lies in the complexity of implementing QoS models, which often encourages network operators to use “brute-force” solutions based on resource overprovisioning.

While best effort can be considered good enough in many cases, it is clear that for a lot of highly important applications, either for their potential business value (e.g. voice, interactive video), or for their role in crucial aspects of social welfare and quality of life (e.g. telemedicine, security), strict fulfillment of QoS parameters is a crucial requirement.

Several recent trends have had a significant impact on QoS. Overwhelming growth of P2P traffic has had a disruptive impact on ISP network backbones. In addition, the increase of access network capacity (which traditionally represented the main bottleneck), combined with the explosion of video-based applications, tends to provoke a huge traffic growth in core networks which will likely be exacerbated in the near future [5].

On the other hand, the adequacy of the original Internet design principles to cope with future Internet requirements has been put into question in the last few years. Despite the huge success of the Internet, largely enabled by the simplicity and scalability properties of the IP protocol, it is becoming increasingly clear that novel ideas and fresh technical approaches are needed to fulfill the requirements of future applications and services. Inevitably, this will have a major impact on how QoS can be provisioned, managed and controlled.

Traditionally, QoS is handled by the proper combination of network resource provisioning with techniques such as admission control, packet scheduling and active queue management. In relatively static networking environments, this kind of approach is adequate to control QoS in most cases. However, in the future this is likely to be challenged by higher elasticity, dynamicity and scalability requirements.

There is no doubt that QoS will remain a fundamental requirement in the Future Internet. But it is also clear that QoS is a moving target and new challenges will be raised by emerging trends, for which the traditional QoS tools may be no longer adequate.

The FP7 4WARD Project [1] has addressed key challenges of dynamic and scalable internetworking posed by the Future Internet and aims at creating a new architectural approach, more flexible and better adapted to present and future requirements [7]. Innovative networking concepts have been developed, referring to different views or behavioral aspects of the network, and paving the way to new approaches for provisioning, managing and controlling traffic and network resources.

The following sections are focused on three major concepts developed by 4WARD and analyze their potential impact on QoS, both in terms of new challenges and new capabilities:

- Network Virtualization to enable the deployment of multiple networks and architectures over a common infrastructure and foster the emergence of novel Internet paradigms;
- The Generic Path based on the semantic resource management concept to enable the design of new data transport mechanisms in order to flexibly establish and manage connectivity, supporting different types of communications in highly mobile and dynamic network scenarios;
- In-Network Management to simplify and improve scalability of management operations by pushing management intelligence into the network and distributing management logic across all nodes.

A detailed discussion of these concepts is beyond the scope of this paper. Readers are advised to look for relevant information in [2], [3], [4], respectively.

2. NETWORK VIRTUALIZATION

Network Virtualization is the concept of sharing physical resources (the substrate), i.e. nodes and links, in order to create virtual networks (VNets) on top of this shared infrastructure. 4WARD considers Network Virtualization in a competitive environment [13] and especially considers shared infrastructure in an inter-provider setting. Within VNets, arbitrary network architectures, which are not necessarily based on IP, may be deployed. Figure 1 shows an example VNet instantiated on top of the physical resources of three Infrastructure Providers with some end users attached to the virtual network.

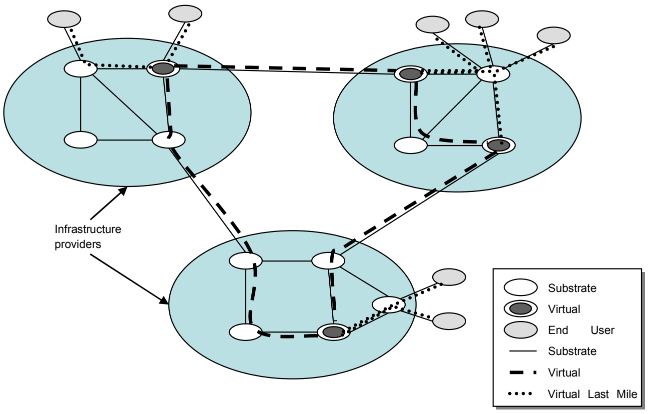

Figure 1. Basic Network Virtualization Scenario

This example already implies that there are at least two levels of QoS involved in network virtualization:

1) QoS at the substrate level

At the substrate level, QoS mechanisms are required for multiple reasons. From a security point of view, a well-defined degree of isolation has to be provided between virtual networks in order to prevent VNets from mutually affecting each other adversely — be it on purpose or inadvertently. Therefore, a differentiation between traffic of different virtual networks is necessary and the associated resources need to be isolated from each other to a certain degree. A stringent resource isolation requires resource reservation before use as well as policing during use. Thus, before the creation of a new virtual network, admission control is performed and the required resources are reserved for use by the virtual network afterwards. While the virtual network exists, usage of these resources needs to be policed accordingly in order to guarantee the SLA negotiated for this virtual network. These SLAs can comprise, e.g., the three classic levels of QoS assurance depending on the actual requirements on the VNet:

(a) <u>Best effort</u>: A virtual network is asserted only best effort service as usual in today's Internet. Virtual networks in the best effort category would then share with each other the resources remaining from higher QoS classes. This behaviour may be acceptable for some virtual networks that are not too sensitive to QoS parameters and that need to be operated very cost-efficiently. Real isolation is not provided and thus a DDoS flooding attack on one best effort VNet may adversely affect other best effort VNets on the same physical link.

(b) <u>Statistical Multiplexing</u>: In order to allow the Infrastructure Providers, which operate the substrate, to make efficient use of their resources, they may apply statistical multiplexing between virtual networks if the associated SLAs permit them to do so. For instance while providing a minimum throughput for a virtual link, additional bandwidth may be shared between several other VNets and may be utilized if available.

(c) <u>Hard QoS Guarantees</u>: For virtual networks that are very sensitive to QoS parameters or for which strict isolation is of uttermost importance, the substrate must enforce those hard QoS guarantees. That usually implies the use of admission control procedures on the respective resources.

2) QoS at the virtual network level:

The second level of QoS supporting mechanisms is located inside of virtual networks and inherently depends on the QoS guarantees given to the VNet by the substrate. Based on those guarantees, the Operators of virtual networks can then apply QoS models and mechanisms they prefer inside the virtual network. This includes especially all QoS mechanisms spanning multiple virtual links, including those providing end-to-end QoS guarantees. We note that QoS guarantees at this level are not possible if the substrate level QoS guarantee for virtual links is only best-effort.

This two-layer QoS model leads to the following implications with respect to a globally deployed virtualization framework:

(a) In order to provide virtual networks with network resources spanning multiple infrastructure provider domains with QoS guarantees, an inter-domain QoS solution needs to be standardized and deployed. This at least implies a standardized notion of interoperable QoS descriptions as for example provided by the QoS NSLP QSPEC template. Whereas the QoS models and mechanisms by which the QoS constraints are enforced inside a domain may be left to the responsible parties, at the inter-domain boundaries, a common language for QoS specifications and interoperable mechanisms are required.

(b) Operators of virtual networks may deploy their preferred QoS models inside their seemingly homogenous VNets at potentially global scale without requiring global agreement. This, however, comes at the cost of an agreed-upon QoS solution at infrastructure provider level, which has to be highly flexible, extensible, and scalable, e.g., by considering aggregates of virtual links wherever possible, in order to support future needs and to deal with a vast amount of virtual networks running in parallel.

Summing up the above, many of the QoS-related problems emerging with the advent of network virtualization can be solved by applying existing QoS approaches, such as basic mechanisms of Differentiated Services. Some of them are already deployed whereas some others have been rather investigated from a scientific point of view only and are not widely deployed in practice or used within a limited scope only. For instance, from an inter-domain perspective, standardised interfaces, deployable business models, and inter-provider agreements are necessary in order to get network virtualization deployed in a global manner. With this prerequisite met, however, global deployment of QoS provisioned virtual networks could foster the development of service-tailored networks for innovative future applications.

3. GENERIC PATH

Aiming to overcome limitations present in the current Internet through a set of radical architectural approaches, we design a new end-to-end communication abstraction, the Generic Path (GP) [3]. One main aim of the GP clean slate concept is to support various types of communications in highly mobile and dynamic network scenarios between two or more end systems. It aspires to adapt transport and QoS procedures to the capabilities of the underlying network when multiple routes as well as advanced techniques such as network coding and different types of diversity are considered for improving efficiency of information transport.

3.1. The Generic Path Architecture

The current Internet architecture is founded on the layered model. At the time, this was one of the powerful software engineering foundations for abstracting functionality and separating engineering concerns. This model, however, seems to be less than adequate today. The main reasons are its rigid opacity and the lack of recursiveness that is often needed to explain and capture the repetition of functionality in different contexts and scales. This means that in the current layered model, functionality, semantics, APIs and algorithms are not re-usable or cooperative across different levels of abstraction; thus leading to an explosion of APIs, repetition of functionality in overlays, and uncontrolled feature interactions due to competitive objectives across layers. Last but not least new functionality is impossible to introduce outside the scope of a layer without violating the architecture.

Moreover, in today’s Internet architecture the socket level API is supposed to be the main generic abstraction that hides the complexity of an underlying network sub-system and offers the network service developer a generic access mechanism to different layer functionalities. This, however, is frequently not the case. As a result, today, most network application developers resort to the use of higher level network APIs that are offered by middleware platforms. Those, although they partially solve the problems, are non-standard and system centric as opposed to network centric.

In our work on the Generic Path, we focus on the aspect of resource management by applying shared semantic concepts and formalizing the heterogeneous communication technology with ontologies. At the same time, apart from the foundational work, we back the proposed architectural framework with mechanisms and actually apply them in a number of key application domains such as routing and mobility.

3.2. Semantic Resource Management Based On Ontologies

Generally, an ontology defines a “network” of information associated with logical relations used for structuring and as a means for data/information exchange. In this way, information and mechanisms can be shared between different systems.

The use of ontologies in future networking is particularly suitable to support and facilitate network interoperability. In the current situation, there subsists a huge semantic gap between the service and the transport layer. So the general objective should be to improve the vertical interplay by introducing semantic methods (ontologies) into the network. Ontologies additionally open up opportunities to amend the horizontal resource handling of heterogeneous network technologies. Semantic methods are already established in applications (Web services) and in the QoS area. Web services are supported by ontologies describing the concepts of an application domain. QoS and traffic characteristics are represented by QoS ontologies. However, network resource ontologies still need to be developed to map and match the capabilities of different technologies in order to support the requesting applications.

3.2.1 Ontologies for Traffic Profiles and QoS

The consideration of QoS requirements in the GP context leads to an approach similar to the semantic Web services. The aim is to support automated QoS-aware network resource selection and composition, which addresses the service QoS requirements and objectives.

There is a substantial need to represent GP QoS features with a QoS ontology in order to adopt it for the GP traffic profile construction. More precisely, the QoS profile parameters (bandwidth, delay, error tolerance) are deduced from the QoS ontology descriptions. Figure 2 shows an example for a QoS metric ontology, where a metric is separated into static and dynamic metrics [6].

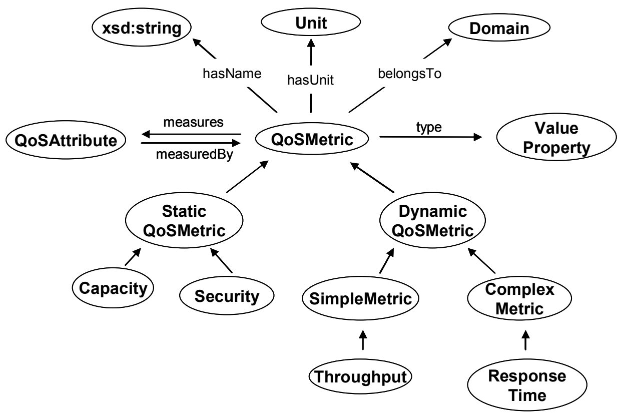

Figure 2. QoS Metric Ontology Example

3.2.2 Network Resource Ontology

In the first step, a generic representation of the network resource parameters has been developed. It provides a standard model defining the characteristics of associations between attributes and the way they are measured based on the following main classes:

- Network Resource Parameter representing a property of the resource within a specific compartment;
- Metric defining how each parameter is assigned with a Value;
- ConversionFormula class enabling the transformation from one metric to another;
- Impact representing the way the parameter value contributes to the perceived quality;
- Type specifying the category of the ontology vocabulary, where the parameter belongs to;
- Aggregated describing the property of aggregation.
- Relationship characterizing the way a Parameter is correlated with others.

Figure 3 shows an example of a network resource ontology model for arbitrary network resource parameters.

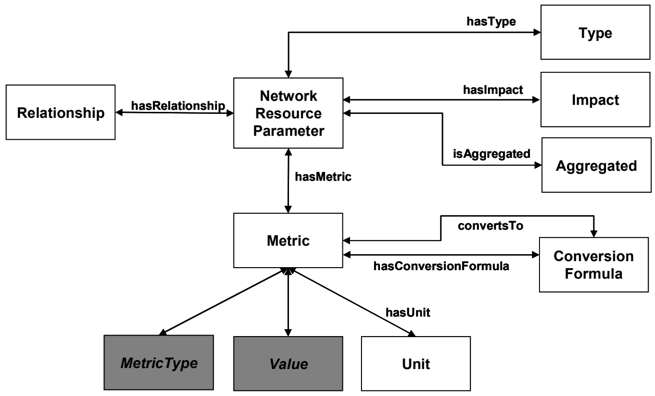

Figure 3. Network Resource Parameter Ontology

If the discovered network resources are accompanied with descriptions of their non-functional properties, automated network resource selection and aggregation is possible. Figure 4 sketches an exemplary network resource ontology representing the relationship between diverse resource types on different levels.

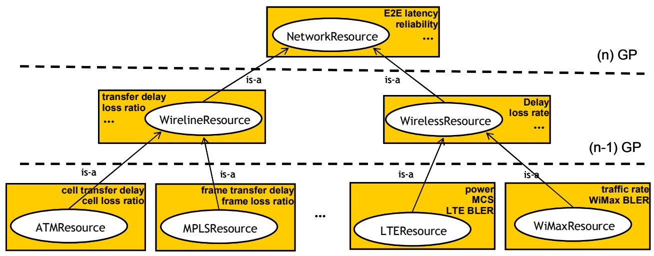

Figure 4. Exemplary Network Resource Ontology

It also shows examples for resource type associated network resource parameters, instantiated as subclasses of the network resource parameter class. The WirelineResource and the WirelessResource classes are examples for an abstract resource type providing an encapsulation for resource aggregation. Generally, the ResourceObject represents physical and abstract resources at a specific GP level.

3.2.3 Semantic Resource Management

The semantic resource management supports fair resource strategies by combining best effort and strict allocation policies to a hybrid approach.

Arriving packet flows are classified according to the QoS profile, resulting in a traffic resource request. This request arrives at the top-level (GP) ResourceObject (Figure 5), representing the current status of the abstracted E2E Resource. Based on the current route path the resource assignment starts and checks if the status of the ResourceObject is UpToDate. In this case, the process verifies whether the available resources fulfill the requirements. If the required resources are available, resource assignment can be performed. Otherwise, a re-routing has to be triggered. In case re-routing does not succeed, the request has to be delayed or rejected. In this way, the GP-using application gets immediate feedback to a dedicated resource request by accessing the top-level (GP) ResourceObject.

At the bottom-most GP level ResourceObjects represent real physical resources, which are located and distributed directly at the physical resource level. There technology specific resource functionalities are performed and information about the current link state, e.g. link degradation etc, are available. As the physical resource is a highly shared object, it is very beneficial to provide a pro-active behavior at each layer by iteratively advertising its resource status to all ResourceObjects in the compartment (abstraction) layers above. In Figure 5 the resource status is periodically advertised starting from the bottom-most ResourceObject level to vertical and horizontal adjacencies. The advertising process can be triggered from a resource request in the case when the ResourceObject is not UpToDate.

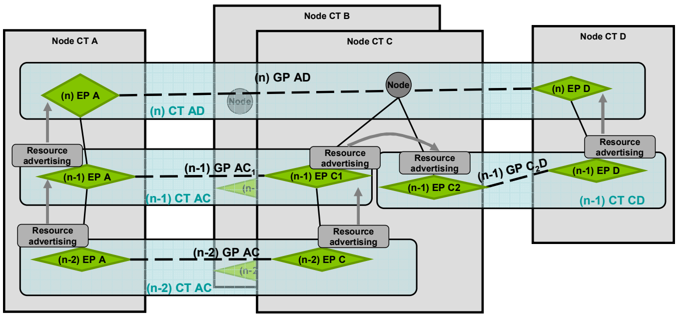

Figure 5. Resource status advertising within the GP architecture

In vertical direction, the advertising process leads to an aggregation (concatenation) of different and possibly heterogeneous ResourceObjects (level (n-1)) resulting in a more abstract representation of the ResourceObject (level (n)). Aggregation and abstraction are supported from the network resource ontology. In particular specific resource parameters are converted to a new metric with the help of the conversion formula depicted in Figure 3.

3.3. Defining QoS for challenged networks

One of the innovations incorporated in the Generic Path architecture is support for intermittent connectivity in Delay-Tolerant Networks. The feasibility of Delay-Tolerant Networking (DTN) and the efficiency of its implementation are closely related to features provided by the GP architecture. In the case of DTN message switching, the GP represents and manages the delivery of the message (or, more generally, of the Information Object) from a source application to a destination application.

The main question that a DTN routing algorithm has to answer is: "How many copies of a message should be distributed into the network?". The answer may range from "one" to "as many as the present nodes are". We argue that this question has to be answered taking into account the desired service outcome, which is not the same in all cases.

In conventional Internet communications end-to-end connectivity exists at all times and therefore, QoS techniques are investigated accordingly. For example, different service priorities, over-provisioning and queuing management, just to name a few, have been of great interest the past few years. In the Internet, however, QoS techniques can be triggered reactively, i.e., reside above the network layer of the TCP/IP protocol stack, since this is indeed possible given the small end-to-end delays. The rules change, however, when discussion comes to challenged, partitioned, disconnected, delay-tolerant networks. Thus, we attempt to re-define the term "QoS" for such networks.

In particular, there exist two ultimate goals a DTN forwarding algorithm attempts to achieve: (i) (high) delivery ratio, (ii) (low) delivery delay. We call these two goals the service targets of a DTN system; we contend that these service targets form the main QoS guarantees that a routing/forwarding algorithm should be able to provide. Furthermore, given that DTNs consist mainly of mobile, battery-powered devices, they will be constrained with regard to: (i) energy consumption, (ii) storage space. We refer to these constraints as system constraints. Although delivery delay may sound contradicting as a QoS service target for delay-tolerant networks, we note that delay-tolerant may be an application that can tolerate one minute of delay, but delay-tolerant may also be an application that has to tolerate one week of delay. In that sense, some sort of service differentiation seems to be essential.

Since the DTN "killer app" is yet to be found [8], we pick some well-known, but still DTN-applicable applications in order to illustrate the difference. Email requires 100% delivery ratio, but is not strict in terms of delivery delay. In contrast, "web-on-the-move", or non-critical telemetry data becomes useless if not delivered within (relatively) strict deadlines. In that sense, QoS targets appear to be diverse among DTN applications.

Applying similar forwarding policies to the above applications may drain the system's energy in case of e-mail flooding, or saturate the system's storage in case of reliable telemetry delivery, for instance. In other words, QoS guarantees may be difficult (if possible at all). We argue that holistic, "one-fits-all" approaches to DTN routing, which appear to be the norm until now, will lead to QoS deadlocks, similar to the ones that the research community faces presently in the conventional Internet (e.g., mice vs. elephants). In contrast, proactive, service-target driven QoS designs have the potential to provide, inherently, supply according to demand, instead of reactive patches that would regulate demand according to supply. A first step on that direction has been made in [9].

4. IN-NETWORK MANAGEMENT

The network of the future is believed to be a network with converged services. The same network will provide access to data, voice or high quality video content to the end users. Because not all the data flows require the same traffic parameters, we have to classify them according to the transported information type and treat them differently. All these operations are made in order to maintain a specific quality of the service, provided by the operator to the end user and specified in the SLA between those two entities. Because a usual traffic flow crosses different communication domains, each one with its own rules, it is still a challenge to guarantee end-to-end QoS services on a communication channel, because each domain can implement different QoS mechanisms that are not always compatible, or worse, are not offering QoS at all. The reason that many network administrators choose not to enable these functions in their network is that they are difficult to configure properly, requiring a thorough understanding of the mechanisms behind.

We consider that QoS is an important management functionality that should be supported by future networks. The architecture of each network node should contain several quality of service related management capabilities. They are included into an SE (Self-Management Entities) and have two types of interfaces. The organizational interface (ORG) is used by a manager or another entity to send high level commands to a specific INM (In-Network Management) entity. The collaboration interface (COLL) is dedicated to facilitate the communication between two management entities residing either in the same or in different nodes

The INM CLQ (Cross-Layer QoS) accesses the hardware directly, through the collaboration interface and it has actually two approaches:

- Bottom-up approach: will enable collecting traffic parameters like: ATR (Available Transfer Rate), OWD (One-Way Delay), BER (Bit-Error Rate),, and other information that is able to characterize a specific physical link. This is an objective way of evaluating a communication channel. The results could be obtained directly from the hardware driver where the technology will permit, or using different dedicated tools that will perform passive or active measurements between nodes.

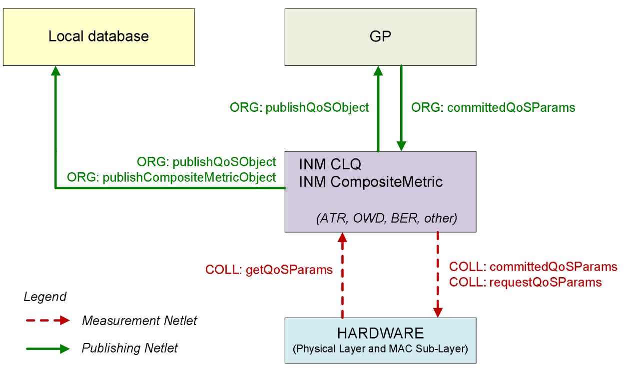

Figure 6. Interaction between INM Cross-Layer QoS, INM Composite Metric, hardware and other managed entities

- Top-down approach: will impose a specific transfer rate to the hardware through INM platform and hardware driver. The requested traffic parameters will be received from different applications (e.g. GP) through the organizational interface and sent directly to the hardware, using the collaboration interface.

An immediate example of INM CLQ's beneficiary would be the real-time composite metric (CM) calculation. The preliminary formula used for an overall perspective of the links with the neighbors (for hop-by-hop data transport) was:

$$CM = \frac{k_0}{ATR[bps]} + \frac{OWD[s]}{k_1} + k_2 \times BER \qquad (1)$$

where $k_0 = 10^9$ *[bps]*, $k_1 = 10^{-5}$ *[s] and* $k_2 = 10^{12}$. This CM could help the management as criteria for triggering network-coding based GP activation, QoS-aware routing, etc. The formula should be interpreted in a similar way Cisco's EIGRP composite metric is used in Network Layer. This means that the maximum ATR envisaged was 1 Gbps, the minimum OWD was 10 microseconds and the BER was not involved in fixed networks-based testbed. Obviously, additional work is needed to demonstrate that this formula seizes the dynamicity of the physical links. However, as a first step we may consider the composite metric provided as useful.

This paper proposes and demonstrates that a combination between INM and GP is feasible. Based on our evaluation of existing solutions in [10], we designed Cross-Layer QoS as a particular example of in-network management, according to the new paradigm. To resume, INM supposes inherent management (or at least integrated) into the monitored network, and not external as in the legacy solutions. The generic path is an abstraction facilitating the development of applications that use the data transport and enhancing the communication's reliability and quality.

By combining Cross-Layer QoS with a simplified generic path, i.e. a multi-point-to-multi-point communication based on network-coding (NC), congestion control is one of its immediate applications. Note that our approach addresses the case of preserving the performances of the running services, despite the congestion which cannot be eliminated. Thus we offer an enhanced distributed routing on a real-time implementation on the existing hardware (no dedicated platforms are needed). This is valid for all types of networks and it involves dedicated software that could be automatically activated/ deactivated by the Cross-Layer QoS. An example of employing the in-network management capabilities for building a GP based on network coding (NC) is discussed in [11], [12]. Network coding was intended for improvement of the multicast transfer rates. It was mathematically demonstrated that NC could improve significantly the multicast transfer rate when congestion is present in the network. Even if the principle of the NC is a relatively simple one, the implementation of such a technique is difficult due to the supplementary operations required: selection of the encoding nodes and of the interconnecting links (shortly the coding network topology), appropriate choosing of the links transfer rates and of the local coding coefficients in the encoding nodes, network wide synchronization of the flows transmitted between neighbor encoding nodes.

The QoS management element collects information about available resources in each strategic node, i.e. in a node that includes GP, CLQ and major management capabilities (neighbor discovery, registry, resource control, event handling, security etc.). The testbed, presented in Figure 7, includes six routers with cross-layer & network coding capabilities (R1 - R6), each one running in a Linux-based virtual machine. The data flow generators (S1, S2) and the destinations (D1, D2) are PCs performing cross-layering only. Specialized software is running on each node to monitor the substrate resources, i.e. the transfer rates and the one-way delays between the neighboring nodes, in order to assist congestion control mechanisms to get a global perspective and to have statistics on link status.

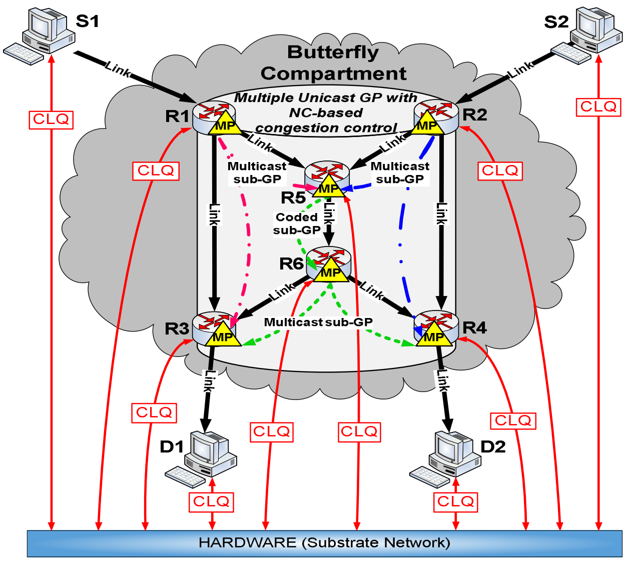

Figure 7. Testbed for Cross-Layer QoS and NC-based GP

Briefly, the experiments included three cases: a) Case 1 (no congestion, no NC-based GP): due to enough available transfer rates on link R5-R6 in Figure 7, the quality of experience at the destinations was very good. b) Case 2 (congestion on link R5-R6, no NC-based GP): both receiving nodes R3 and R4 experienced a bad quality of the movies because of the packets lost in the congested link. c) Case 3 (congestion on link R5-R6, with NC-based GP awareness): the link between R5 and R6 remained congested as in Case 2, but a NC-based GP was instantiated whenever CLQ triggered the situation. Note that a pre-congestion was experimentally detected using the INM algorithm. Thus the mechanism was activated in advance, before severe congestion might occur. The measurement results presented in Figure 8, according to [12], show that the number of packets lost is very low (0.75%, compared to about 18% in the previous case).

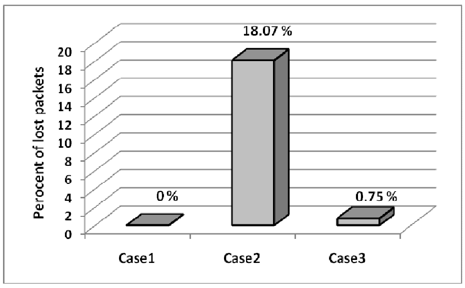

Figure 8. Summarized results proving the efficiency of INM CLQ combined with GP in case of congestion [12]

5. CONCLUSION

Considerable research effort will be necessary to address the challenges raised by the design of the Networks of the Future.

In this paper, three concepts developed in the framework of the FP7 4WARD project have been analyzed from the point of view of the potential impact on QoS – Network Virtualization, Generic Path Semantic Resource Management and In-Network Management.

While these novel networking concepts have not been specifically targeted at handling QoS issues, they undoubtedly enable new QoS approaches and solutions for the networks of the future, especially in view of requirements such as dynamicity, flexibility, adaptability and scalability.

Network virtualization decouples networks from infrastructure and allows infrastructure resources to be shared among multiple isolated networks. By enabling the capability to build service-tailored networks and overcoming the limitations of the traditional "one-size-fits-all" approaches, network virtualization inherently brings potential advantages in terms of QoS. However, a two-layer QoS model will be required, which raises new challenges, particularly in multi-domain scenarios.

The Generic Path is a new end-to-end communication abstraction that aims at overcoming the inadequacies of the traditional layered network model. Resource management is accomplished by applying shared semantic concepts and formalizing the heterogeneous communication technology with ontologies. The QoS features of the Generic Path can be represented by an ontology from which QoS profile parameters, such as bandwidth, delay and error tolerance can be derived. Semantic resource management enables fair resource strategies by combining best effort and strict allocation policies.

Finally, in-network management enables the incorporation of QoS management capabilities in the network elements, thus facilitating QoS configuration, even without a thorough understanding of the usually complex mechanisms behind it. Cross-Layer QoS has been presented as a particular application of in-network management in a scenario combining cross-Layer QoS with a multi-point-to-multi-point communication based on network-coding. In INM network management is inherent, or integrated, into the network, and not external as in legacy solutions.

Integration of these new concepts with traditional QoS mechanisms will require further study. The need for interoperability will surely make standardization play a fundamental role in this scenario.

ACKNOWLEDGMENT

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REFERENCES

[1] The FP7 4WARD Project, http://www.4ward-project.eu/

[2] S. Baucke et al, "D-3.1.1 Virtualisation Approach: Concept", http://www.4ward-project.eu/index.php?s=file_download&id=68, September 2009

[3] S. Randriamasy et al, "D-5.2 Mechanisms for Generic Paths", http://www.4ward-project.eu/index.php?s=file_download&id=78, May 2010

[4] G. Nunzi et al, "D-4.2 In-Network Management Concept", http://www.4ward-project.eu/index.php?s=file_download&id=37, March 2009

[5] "Approaching the Zettabyte Era", Cisco White Paper, http://www.cisco.com/en/US/solutions/collateral/ns341/ns525/ns537/ns705/ns827/white_paper_c11-481374.pdf, 2008

[6] L. Lin, S. Kai, S. Sen, "Ontology-Based QoS-Aware Support for Semantic Web Services", Technical Report at Beijing University of Posts and Telecommunications, 2008

[7] M. Johnsson et al, "Introduction to a 4WARD Architectural Approach", 4WARD Deliverable 0.4, February 2009

[8] J. Crowcroft, E. Yoneki, P. Hui, T. Henderson, "Promoting Tolerance for Delay Tolerant Network Research", SIGCOMM Comput. Commun. Rev., Vol. 38, No. 5, 2008, pp. 63-68

[9] I. Psaras, N. Wang, R. Tafazolli, "Six years since first DTN papers. Is there a clear target?", ExtremeCom, 2009

[10] A. B. Rus, V. Dobrota, "Overview of the Cross-Layer Paradigm Evolving towards Future Internet", ACTA TECHNICA NAPOCENSIS, Electronics and Telecommunications, ISSN 1221-6542, Vol.50, No.2, 2009, pp.9-14

[11] Z. Polgar, Z. Kiss, A. B. Rus, G. Boanea, M. Barabas, V. Dobrota, "Preliminary Implementation of Point-to-Multi-Point Multicast Transmission Based on Cross-Layer QoS and Network Coding", 17th International Conference on Software, Telecommunications & Computer Networks IEEE SOFTCOM 2009, Split-Hvar-Korkula, Croatia, September 24–26, 2009, pp.131-135

[12] A. B. Rus, M. Barabas, G. Boanea, Z. Kiss, Z. Polgar, V. Dobrota, "Cross-Layer QoS and Its Application in Congestion Control", 17th IEEE Workshop on Local and Metropolitan Area Networks LANMAN 2010, Long Branch, NJ, USA, May 5-7, 2010, http://www.ieee-lanman.org/

[13] G. Schaffrath, C. Werle, P. Papadimitriou, A. Feldmann, R. Bless, A. Greenhalgh, M. Kind, O. Maennel, L. Mathy, "Network Virtualization Architecture: Proposal and Initial Prototype", p. 63-71, ACM, Barcelona, Spain, August 2009.

SESSION 5

SERVICE INNOVATIONS IN THE FUTURE INTERNET

CROSS-LANGUAGE IDENTIFICATION USING THE WAVELET TRANSFORM AND ARTIFICIAL NEURAL NETWORK

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ABSTRACT

With the advent of the Internet, search engines were developed for English language because English language was a lingua franca. Currently, most of popular search engines such as Google and Yahoo! are available in more than 50 languages. However, these search engines have received less attention in South Asian languages especially, Urdu language.

In this paper, we propose a novel approach for feature extraction and classification of queries in cross-language search engines. This novel approach presents an automatic method for classification of English and Urdu languages identification. The classifier used is a three-layered feed-forward artificial neural network and the feature vector is formed by calculating the wavelet coefficients. Three wavelet decomposition functions (filters), namely Haar, Bior 2.2 and Bior 3.1 have been used to extract the feature vector set and their performance results have been compared. The performance results of the Haar filter have given superior results than other filters.

***Keywords*—**Wavelet transforms, artificial neural network, language identification, cross-language, Unicode.

1. INTRODUCTION

During the first decade of the search engines, most of the search engines were developed for English language because the English language was a lingua franca. Currently, a report published by Internet World Stats (www.internetworldstats.com/stats7.htm) at the end of 2009 showed that more than 65% of all Internet users are from non-English speaking areas. Therefore, the population of non-English speaking Internet users is growing much faster than that of English speaking users. Asia, Africa, the Middle East and Latin America are the areas with the fastest growing online population.

Nevertheless, the unequal use of the language and the users' lack of access to the relevant materials in a language that they speak and understand seem to reinforce the growing knowledge gap between information and cognition processing on the Internet. This problem has motivated United Nations Educational, Scientific and Cultural Organization (UNESCO) to consider one of its important goals, "to achieve worldwide access to e-contents in all languages, improve the linguistic capabilities of users and create and develop tools for multilingual access to the Internet." In [1, 2], Chowdhury, Peters and Picchi confirm that the multilingual information retrieval has become a major challenge to access to the prolific information on the Web and digital libraries. In addition, they divided the problem of Web multilingual information retrieval into two sets:

(1) Identification, manipulation, and display of multiple languages.

(2) Multilingual or cross language information search and retrieval.

The first problem is based on finding technology solution to enable users to use and access information in whatever language it is stored. The second problem implies allowing the users to get relevant information in another language based on the information need for one language by user.

Due to the aforementioned fact, we should have an adequate solution to the problem of multilingualism on the Internet. The foremost reason being, we are in a multilingual and multicultural world. Everyone in the world has the right to utilize multilingualism on the Internet and universal access to cyberspace.

The search engines (or search services) and Web directories are the most general approaches to facilitate searching for huge number of information on the Web. Most of popular search engines such as Google (www.google.com) and Yahoo! (www.yahoo.com) are available in more than 50 languages. However, these search engines are not supporting searching and translation from English into Urdu and vice versa at the moment. Urdu language is a Central Indo-Aryan language of the Indo-Iranian branch which belongs to the Indo-European family of languages. There are about 104 million Urdu language speakers, including those who speak it as a second language (http://www.omniglot.com/writing/urdu.htm). Urdu scripts and numbers are written from right to left and left to right respectively. Its scripts have shapes similar to Arabic, Persian language characters.

Cross-language information retrieval (CLIR) is a subfield of information retrieval which is an important field of language identification as well. CLIR allows a search engine user to obtain relevant information in another language based on the queries entered for one language by user (English into Urdu or Urdu into English). Therefore, query language identification plays an important role for information retrieval in increasing the performance of search engines. The study of language identification is a critical problem, especially identifying the language of user queries in the Web search engines.

The major difficulties of language identification may be summarized into two questions: (a) How can the search engine identify the language for the required information and get the relationships among these topics? (b) How can the search engines go in depth of content and make classification of these huge numbers of information in the Web? One of the answers to these questions is the automatic language identification.

The researchers have been studying the challenges of language identification over several decades. The survey of previous work for this problem is written by Hughes et al, [3]. The most related works for language identification are based on statistical algorithms, short-words, frequent words, and n-gram methods. Ceylan and Kim proposed a language identification method using two decision tree classifiers. Their data set is created automatically from the Yahoo! search engine. The experimental results have shown that the proposed methods give the satisfactory performance of language identification [4]. Gottron and Lipka compared some of the performance of some public methods for language identification on very short and query–short texts. The accuracy of typical methods was more than 80% and close to 100% for single words and slightly longer texts, respectively [5]. W. Anwar et al. showed Urdu language and its difficulty of processing and summarized several linguistic methods such as Part of Speech tagging, parsing and named entity recognition [6]. There are some innovative and new ideas related to information search and retrieval based on representation of the word position or the narrative of order of the words within the document as signal.

In [7], Brewster and Miller convert the document to terms frequency matrix of signals as a method for providing document characterization and so as to utilize one feature or benefit of wavelet transform to allow the user to produce a visual representation of the semantic structure of the document. Some of researchers have begun representing the word relatives, and the word position or the narrative of order of the words within the document as signal by identifying content-bearing words using Bookstein's [8] proposed method.

Most of these research works based on removing stop words, stemming words, parsing words, filter the documents according to word frequency or topicality, or achieving some combination of these functions which are called preprocessing operations. Therefore, they have several disadvantages of which are the following:

(1) Every language needs a special algorithm to be applied preprocessing operations such as parsing or stemming words etc; (2) Preprocessing operations increase the computational complexity; (3) They may provide a crude picture of the document's content; (4) The word position in the document can not be understood and tracked in the content of document, thus information about order are lost especially, when the document implements as a bag of words; (5) Parallel processing of query or document may not be applied because the concept of frames is not supported.

So, we proposed a novel method to use the wavelet transform as information retrieval and indexing as shown in our previous works [9,10] whereas our novel method converted the query entered by the Internet user into a signal using its Unicode standard. Unicode is the international standard for representing the characters used into a plurality of languages. Also, it provides a unique numeric character, regardless of language, platform, and program in the world. Furthermore, it is popularly used in the Internet and any operating system of computers, as a device for the text visual representation in the whole world. In addition, many software companies often use it.

Some of the advantages of our proposed model are: (1) Our proposed method is based on just one algorithm for most of the languages using the Unicode standard; (2) It facilitates applying mathematical transformations, especially the wavelet transform where the signal has been transformed by the availability of mathematical transforms. Therefore, we can obtain further information from the signal that is not readily available in the raw signal or normal language; (3) the query of proposed method can be a word(s), small and large sentences, snippets, or document. These signal queries do not exclude anything from the query (such as the spaces, commas, question marks, and prepositions or applying parsing or stemming on the query). One of the reasons of that, we need to keep the position of every word in the query in order to get full meaning of the query and track the content of document; (4) By using the wavelet transform, only a few transform coefficients would suffice as the index, thus reducing the index size; (5) Some of the properties of the wavelet transform enable indexing of large size data with a small number of terms, which in turn facilitate faster and more accurate searches where the computational complexity of wavelet transform is O(n); (6) We can search easily about chemical formals and mathematical/scientific symbols; (7) It may enable to be used parallel processing of query because our method supports framing query or document

We consider our previous works as preprocessing phase to select suitable filter and speed up the classification phase. In [9,10], we applied a new direction for the wavelet transform for multilingual Web information retrieval of 14 languages belonging to 5 language families (English, Spanish, Danish, Dutch, German, Greek, Portuguese, French, Italian, Russian, Arabic, Chinese (Simplified and Traditional), Japanese, and Korean (CJK)). Forty-two wavelet functions (mother wavelets) of six families, namely, Haar (haar), Daubechies (db2-10), Biorthogonal (bior1.1- 6.8 and rbior1.1 – 6.8), Cofilets (coif 1-5), Symlets (sym 1 - 10), and Dmey (dmey) are investigated in order to evaluate a suitable wavelet function for multilingual Web information retrieval. Three wavelet decomposition functions (filters), namely Haar, Bior 2.2 and Bior 3.1 have been approved for the aptitude to be multilingual Web information retrieval. In [11], the aptitudes of multi-wavelet transform to represent one language (domain) of the multilingual and multicultural world regardless of type, script, word order, and direction of writing and difficult font problems of the language were further investigated with more languages and families.

As a result of our work with 31 sentence queries for 29 languages belonging to 8 language families, we expect that the wavelet transform or multi-wavelet transform becomes a multilingualism tool on the Internet. We recommended it as a universal approach for creating multilingual software.

One can gain a lot by adopting a combination of wavelets and artificial neural networks (ANN) for feature extraction and classification of a signal query and this is exactly what we intend to discuss in this paper. One of the benefits of that is to mimic the brain processes of human being in bringing and searching the information required. The brain of human being deals with the query (reading or thinking) as a signal and gets the information required. As a result, the study of cross-language Web information retrieval has become an interesting and challenging research problem in the multilingual world.

The rest of this paper is organized as follows. In section 2, we discuss preliminaries that are related to wavelet transform and ANN. In section 3, we describe methodologies pertaining to our work. In section 4, we discuss simulation environment. Section 5 contains results and discussions. Finally, section 6 is dedicated to conclusions and future directions.

2. AN OVERVIEW THE WAVELET TRANSFORM AND ARTIFICIAL NURAL NETWORK

2.1 The Wavelet Transform

Wavelets are basis functions that allow transformation of signals from their original domain to another in which some operations can be performed in an easier way. In mathematic, a wavelet is a function $\Psi(t) \in L^2(R)$ where $L^2(R)$ space is the space of all square-integrable functions defined on the real line **R** with a basic property [14, 15].

$$\int_{-\infty}^{\infty} \Psi(t)\,dt = 0 \qquad (1)$$

This means that the average value of the wavelet in time domain must be zero, and therefore it must be oscillatory.

In other words, $\Psi(t)$ must be a wave, and the wavelets are generated from a single basic function $\Psi(t)$, called Mother Wavelet, by using both scale a and translate b factors and the constant $|a|^{-1/2}$ is used for energy normalization [13, 16].

$$\Psi_{a,b}(t) = |a|^{-1/2}\, \Psi\left(\frac{t-b}{a}\right) \qquad (2)$$

$\Psi_{a,b}(t)$ stands for the wavelet basis.

Wavelet transforms can be divided into the Continuous Wavelet Transform (CWT), the Series Wavelet Transform (SWT) and Discrete Wavelet Transform (DWT) based on the variables **a** and **b** which are continuous values or discrete numbers and type of input signal [13, 14].

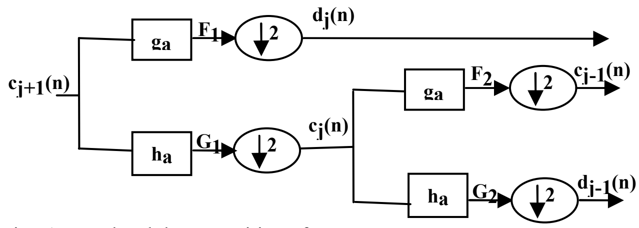


Fig. 1. Two level decomposition of FWT.

As shown in figure 1, the low pass h_a and high pass g_a filters achieve convolution of the $c_{j+1}(n)$ input signal and subsequently down sampling them by factor 2. An output of this level is $c_j(n)$ and $d_j(n)$, where $c_j(n)$ is approximation coefficients and $d_j(n)$ is detail coefficients.

2.11 Fast Wavelet Transform (FWT)

In 1989, Mallat proposed that Multi Resolution Analysis (MRA) can be used to obtain the Discrete Wavelet Transform (DWT) of a discrete signal by replacing the scaling function Φ(n) and mother wavelet Ψ(n) with low pass and high pass filters respectively. Thus, an increase in speed for the wavelet transform is giving. Figure1 shows some of wavelet filters. For increasing of speed the algorithm, Mallat proposed replacement equations (3) and (4) with (8) and (9), respectively [13, 14].

$$c_{j,k} = \sum_{n=-\infty}^{\infty} x(n)\Phi_{j,k}(n) \qquad (3)$$

$$d_{j,k} = \sum_{n=-\infty}^{\infty} x(n)\Psi_{j,k}(n) \qquad (4)$$

$c_{j,k}$ and $d_{j,k}$ illustrate both approximation and detail coefficients.

Here Φ(n) is Scaling function

$$\Phi_{j,k}(n) = \sqrt{2^j}\,\Phi(2^j n - k) \qquad (5)$$

Ψ(n) is Mother wavelet function:

$$\Psi_{j,k}(n) = \sqrt{2^j}\,\Psi(2^j n - k) \qquad (6)$$

$$c_j(n) = \sum_{m=-\infty}^{\infty} h_a(m-2n)c_{j+1}(m) \qquad (8)$$

$$d_j(n) = \sum_{m=-\infty}^{\infty} g_a(m-2n)c_{j+1}(m) \qquad (9)$$

Where h_a is low pass filter and ga is high pass filter

$$g_a(n) = (-1)^n h_a(n-1) \quad (10)$$

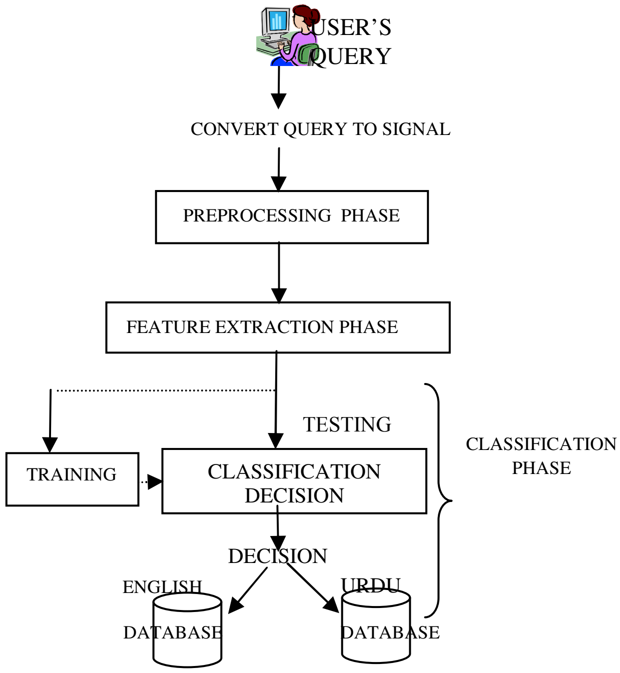


Fig. 2. Methodology for classification of the processed signal.

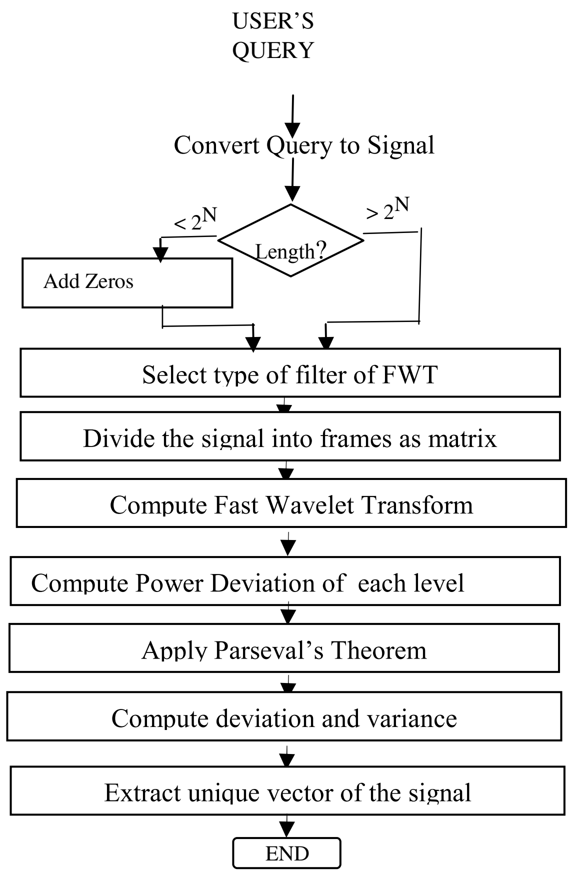


Fig. 3. Flowchart for Power Deviation method.

2.2 ARTIFICIAL NEURAL NETWORKS

Artificial Neural Networks (ANN) have been developed as a generalization of mathematical models of biological nervous systems which is designed to mimic the decision–making ability of the brain by providing a mathematical model of combination of numerous neurons connected in a network [17, 18]. It possesses a good learning capability that learns from given input/output data pairs and adjusts the design parameters through minimization of error function using a suitable learning algorithm. In the area of pattern recognition, signal and image processing, feature extraction is important for dimensionality reduction procedure [18]. Back Propagation (BP) is one of the most widely used training algorithms for multi-layer neural networks [19]. The multi-layer neural networks have one or more layers of nodes (neurons) between the input and the output nodes, which is called the hidden layer [20]. BP is a gradient descend method and was first described in detail in 1986 by Hinto, Rumelhart and Willians, and is sometimes called Generalized Delta Rule (GDR).

3. METHODOLOGIES

In traditional language identification methods, it is not so easy for search engines to find relevant language database of a given query. Therefore, there is a need to identify the relevant user's natural language query of unknown document database in a better way by automatic language identification. In several CLIR problems, the number of variables is very large. As a result, the processing of the data can be slow, and may require a lot of memory. Also, classification algorithms may cause over-fitting or overtraining of the training algorithm. Feature extraction is a general term for methods of constructing combinations of the variables which overcome the above problem but still describe the data sufficiently and accurately. In other words, the most useful information is chosen from the complete feature space to form a feature vector in a lower dimensional space. Therefore, the feature extraction removes any redundant and irrelevant information that may have detrimental effects on the performance of recognition.

The main objective of the proposed model is to make the Internet user easily access it using one's language or any language for the required information as shown in figure 6. In addition, the objective of the proposed model is to use a smart way for automatic language identification and reducing human efforts in translation and reducing the search engine depending on translation machine.

One of the disadvantages of translation machine systems is the difficulty of creating the translation and making mistakes as well. Therefore, we try to increase the performance of a search engine. This work extends our previous works as presented in [9, 10, 11].

As shown in figure 2 and figure 6, we divide the methodology into three phases: preprocessing, feature extraction, and classification. In the preprocessing phase, a series of steps is applied to the signal query. This is done to select a good filter from wavelet filters and speed up the next step i.e. feature extraction. So, the data is now ready for the feature extraction phase.

In the feature extraction phase, it maps the signal into a unique vector for every language that is used by the classification phase. Hence the main tool used in the feature extraction phase is the fast wavelet transform (FWT) decomposition with three filters, namely, Haar, Bior 2.2, Bior 3.1 and Power Deviation (PD) method [12].

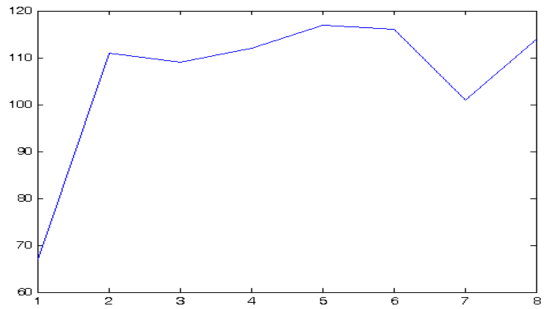
In the classification phase, the signal query is recognized by ANN from the unique vector obtained as an output of the feature extraction phase. We use an approach that is a combination of wavelet transform and artificial neural network. Also, we use the PD method for signal classification which had proved its previous ability for identifying any change on the signal [12]. Figure 3 shows the flow of control for PD method. In that, after the native language users pose his/her query into a search engine, the algorithm converts the query to Unicode and deals with the query as signal query without removing anything from it [9, 10, 11]. One example of converting the query to a signal is shown in table 1 and figure 4. After that, we first check whether the length of the signal query is greater or less than 2^n. Depending upon the outcome, either zeros are inserted (zero padding) or/and the signal query is divided into frames. Then, the FWT is computed to select a good filter which can be applied depending on its length and quality of construction (decomposition) and reconstruction. Thereafter, we first compute power deviation of each level and apply Parseval's theorem to the power levels, and then deviation and the normalization of power are carried out. Finally, the signal query is extracted from the unique vector so obtained.

Table 1. Some of the queries for English and Urdu languages.

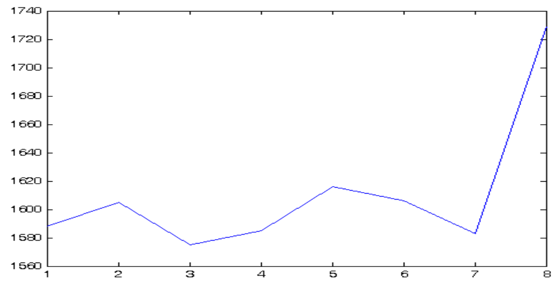
No	English Queries	Urdu Queries
1	Computer	شمارندہ
2	Computers cannot "think" for themselves in the sense that they only solve problems in exactly the way they are programmed to, and arrive at the correct answer (500,500) with little work. In other words, a computer programmed to add up the numbers one by one as in the example above would do exactly that without regard to efficiency or alternative solutions.	اور اس متبادل راہ کے استعمال سے انسان وہی درست جواب (500500) نکال لیتا ہے جو شمارندہ اوپر دی گئی ہدایات سے نکالے گا۔ بس یہ فرق (سوچنے کا) شمارندے اور انسان میں ایسا ہے کہ جس کی بنا پر شمارندے مکمل خود مختار نہیں ہوتے۔
3	Punch Card	کا ایک سوراخی بطاقہ
4	What's your name?	آپ کانام کیاہے؟
5	Where do you live?	آپکا تعلق کہاں سے ہے؟

4. SIMULATION ENVIRONMENT

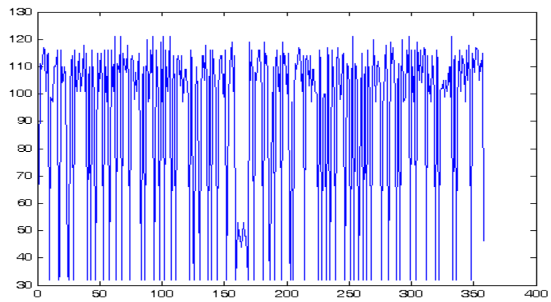
We use Wavelet Toolbox and Neural Network Toolbox of Matlab™ 7.6 for computing the fast wavelet transform, and for training and testing of ANN, respectively. Three wavelet decomposition filters, namely Haar, Bior 2.2 and Bior 3.1 from Forty-two wavelet filters of six families, namely, Haar (haar), Daubechies (db 1-10), Biorthogonals (bior 1.1-6.8 and rbior 1.1-6.8), Cofilets (coif 1-5), Symlets (sym 1-10), and Dmey (dmey) have been used in order to identify a suitable wavelet filter for language identification [21]. The performance results of the three filters have given superior results than other filters. The processing of signal queries achieved in the feature extraction phase produce 5 coefficient inputs with 4 levels of FWT decomposition which feed the fifth input nodes of ANN to achieve training and test processes.



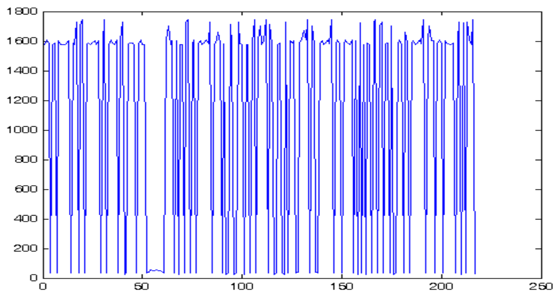
a) English query No.1.



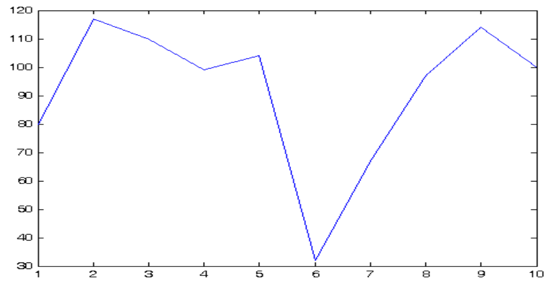
b) Urdu query No.1.



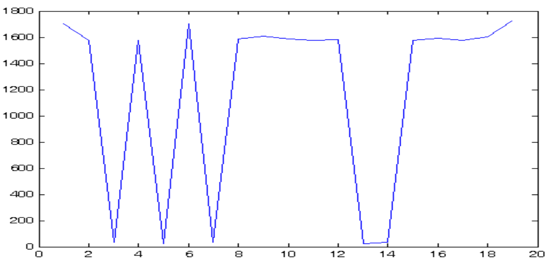
c) English query No.2.



d) Urdu query No.2.



e) English query No.3.



f) Urdu query No.3.

Fig.4. Some of signal queries (a to f) for English and Urdu languages.

Hence, we propose after many trials and errors that a good architecture is with three layers of back-propagation algorithm and structures 5-15-2 (five input layers, fifteen hidden layers and two output layers) for ANN of PD method as shown in figure 5. This number of neurons is suitable to perform best for most of tests.

For evaluation of quality and effective ANN, we need to create database for training of ANN and part of it to make a test process. There is a lack of finding any standard or common available multilingual data set [4, 5]. Therefore, our experiment is applied on data collected to evaluate the practical performance of our method from different websites. The data collected is from English and Urdu headlines and some contents (snippets) of Wikipedia's websites (http:/en.wikipedia.org/wiki/Computer) and (http://ur.wikipedia.org/wiki/%D8%B4%D9%85%D8%A7%D8%B1%D9%90%D9%86%D8%AF%DB%81) and we used a native Urdu user to get some data and confirm our database. Our database contains 52 types of every language with different types of queries such as a word(s), small and large sentences, snippets, or document as shown in table 1 and figure 4. The total of our dataset is 104 headlines, a word(s), small and long sentences, snippets, and questions whereas we use the first 26 of every language (English/Urdu) in this database to train the artificial neural network and the remainder of data of every language to test its performance.

Table 2. Training time and classification ratio of ANN.

Filters	Training Time (SEC)	Accuracy %		Average %
		English	Urdu	
Haar	4.3	100%	100%	100%
Bior 2.2	18.1	92.3%	76.9%	84.6%
Bior 3.1	11.6	57.7%	84.6%	71.2%

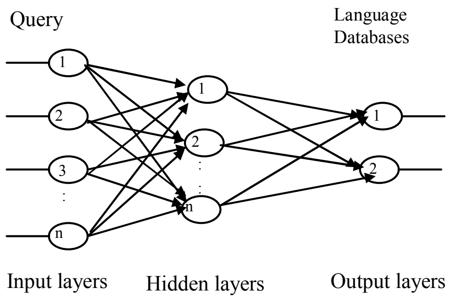


Fig.5. The proposal architecture using ANN of PD method for English and Urdu languages.

5. RESULTS AND DISCUSSIONS

The primary result for selecting a good filter in preprocessing phase has shown that the filters Haar, Bior 2.2, and Bior 3.1 are better among Forty-two wavelet filters of six families, namely, Haar (haar), Daubechies (db 1-10), Biorthogonals (bior 1.1-6.8 and rbior 1.1-6.8), Cofilets (coif 1-5), Symlets (sym 1-10), and Dmey (dmey) [9, 10, 11].

The performance results of Haar filter are 100% and 100% as classification ratio with training time 4.3 seconds for English and Urdu languages, respectively. The performance results of Bior 2.2 filter are 92.3% and 76.9% as classification ratio with training time 18.1 seconds for English and Urdu languages, respectively. Also, the performance results of Bior 3.1 filter are 57.7% and 84.6% as classification ratio with training time 11.6 seconds for English and Urdu languages, respectively. So, the total average of performance of the Haar filter has given superior results than other filters. In addition, the Haar filter is a good filter because it has low order coefficients and good results in FWT construction and reconstruction. Further, the Haar is the simplest wavelet.

One of the reasons of getting different results is that the network needs more training and more amount of training sets. Also, we can say that there is no standard method to select a wavelet function in wavelet transform with signals and languages. Some criteria have been proposed to select a wavelet. One of them was that the wavelet and signal should have good similarities. This also appears on our results in preprocessing phase where the type of the sentence query entered of language and wavelet filters should have good similarities as shown also in table 2 [9, 10, 11].

As a result, the best performance of language identification of English and Urdu languages is the Haar filter with 100% as classification ratio and training time 4.3 seconds.

6. CONCLUSIONS AND FUTURE DIRECTIONS

In this paper, we propose a novel approach for feature extraction and classification of signal queries. This novel approach presents an automatic method for classification of English and Urdu languages identification. The classifier used is a three-layered feed-forward artificial neural network which is called Back Propagation and the feature vector is formed by calculating the wavelet coefficients. Our algorithm deals with the signal query to speed up preprocessing. The query of proposed method can be a word, small and large sentences, snippets, or document. These signal queries do not exclude anything from the query (such as the spaces, commas, question marks, and prepositions or applying parsing or stemming on the query). Also, one of the reasons of that we need to keep the position of every word in the query in order to get full meaning of the query and snippet of document. The performance results of Haar filter are 100% and 100% as classification ratio with training time 4.3 seconds for English and Urdu languages, respectively. The performance results of Bior 2.2 filter are 92.3% and 76.9% as classification ratio with training time 18.1 seconds for English and Urdu languages, respectively. Also, the performance results of Bior 3.1 filter are 57.7% and 84.6% as classification ratio with training time 11.6 seconds for English and Urdu languages, respectively.

Based on the results, we may refer that the best performance of language identification of English and Urdu languages is the Haar filter with 100% as classification ratio and good training time 4.3 seconds.

For future work, we suggest comparing our results with combining multi-wavelet and ANN and use multi-wavelet to solve the problem of selecting a suitable wavelet function of these languages.

REFERENCES

[1] G., Chowdhury, "Natural Language Processing". Annual Review of Information Science and Technology (ARIST), vol. 37, pp. 51-89, 2003. ISSN 0066-4200.

[2] C. Peters, and E. Picchi, "Across Languages, Across Cultures" Issues in Multilinguality and Digital Libraries, D-Lib Magazine, 1997. http://www.dlib.org/dlib/may97/peters/05peters.html

[3] B. Hughes, T.Baldwin, S. Bird, J. Nicholson, and A. Mackinlay, "Reconsidering Language Identification for Written Language Resources", In 5th Internatonal coference on language resources and evaluation(LREC2006), pp. 485-488, Genoa, Italy, 2006.

[4] H. Ceylan, and Y. Kim, " Language Identification of Search Engine Queries", In the 4th International Joint confernce on Natural Language Processing of the AFNLP, vol. 2, pp. 1066-1074, Singapore, 2009.

[5] T. Gottron, and N. Lipka, "A Comparison of Language Identification Approaches on Short, Query-Style Texts", In the proceedings of the 32nd European Conference on Information Retrieval (ECIR-2010), 2010.

[6] W. Anwar, S. Wang, and X. Wang, "A Survey of Automatic Urdu Language Processing",The International Conference on Machine Learning and Cybernetics, IEEE Computer Society Press ,vol. 4, pp.4489-4494,2006.

[7] E. Brewster, N. E. Miller, "Information Retrieval System Utilizing Wavelet Transform" ", United States Patent No. 6070133, 2000.

[8] A. Bookstein, et al., "Clumping Properties of Content-bearing Words", Journal of the American Society For Information Science, Vol. 49, No. 2. pp. 102-114, 1998.

[9] S. A. Al-Dubaee, and N. Ahmad, "New Direction of Applied Wavelet Transform in multilingual web information retrieval", The 5th International Conference on Fuzzy Systems (FSKD'08), IEEE Computer Society Press ,vol. 4, pp.198-202,2008.

[10] S. A. AL-Dubaee, and N. Ahmad, "The Bior 3.1 wavelet transform in multilingual web information retrieval", The 2008 International Conference on Data Mining (DMIN'08), a track at (WORLDCOMP'08), Las Vegas, Nevada, USA, pp.707-713, 2008.

[11] S. A. AL-Dubaee, Vaclav Snasel, and N. Ahmad, "Wavelet, Multiwavelet, and Multilingualism on Internet", The 2009 International Conference on Data Mining (IKE'09), a track at (WORLDCOMP'08), Las Vegas, Nevada, USA, pp.716-722, 2009.

[12] J. W. Resende, M.I.R. Chaves, C. Pnna, "Identification of Power Quality Disturbances Using the MATLAB Wavelet Transform Toolbox ", IPST 2001 Conference, 2001.

[13] C. S. Burrus, R. A. Gopinath and H. Guo , " Introduction to Wavelets and Wavelet transforms " ©1998 by Prentice Hall Inc.

[14] S. Mallat, "A theory for multiresolution signal decomposition: the wavelet representation" IEEE Trans. On Pattern Analysis and Machine Intell, vol.11,no.7, pp.674-693 ,1989.

[15] A.Graps, "An Introduction to Wavelets", IEEE Computational Science and Engineering, Vol.2, No.2, 1995.

[16] C. Valens , " A Really Friendly Guide to Wavelets " , 1999, website: www.robots.ox.ac.uk/~parg/mlrg/papers/arfgtw.pdf

[17] G.qian-jin,Y.Hai-bin , X.Ai-dong, "Wavelet Fuzzy neural network for fault diagnosis", IEEE Communications, Circuit and Systems, Vol.2, No.993-998, May 2005.

[18] E. Oropesa , H.Cycon, M.jobert, " Sleep Stage Classification using Wavelet Transform and Neural Network" 1999, website: http:// www.icsi.berkeley.edu/techreports/1999.html.

[19] Y. Wang and C. Lin, "A Second-Order Learning Algorithm for Multiplier Networks Based on Block Hessien Matrix", Neural Network, vol. 11, no. 9, pp. 1607-1922, December, 1998.

[20] R. Lippmann, "An Introduction to Computing with Neural Nets", IEEE ASSP Magazine, pp. 4-22, April, 1987.

[21] M. Misiti, et al., "Wavelet Toolbox MATLAB user's guide Version 4", Mathworks Inc, 2008.

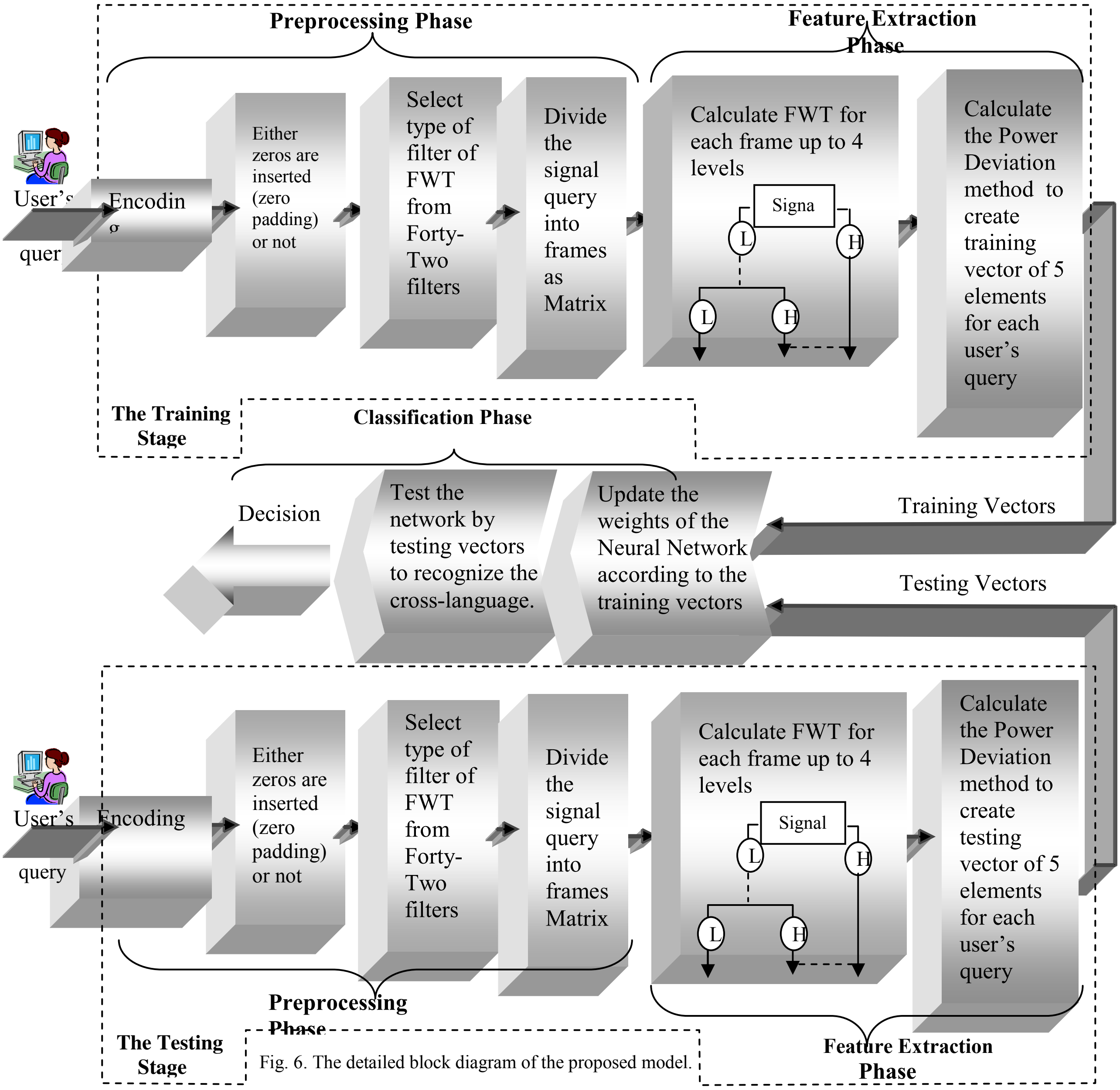


Fig. 6. The detailed block diagram of the proposed model.

GEOHYBRID: A HIERARCHICAL APPROACH FOR ACCURATE AND SCALABLE GEOGRAPHIC LOCALIZATION

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ABSTRACT

Geographic location and Grid computing are two areas that have taken off in recent years, both receiving a lot of attention from research community. The Grid Resource Brokers, which tries to find the best match between the job requirements and the resources available on the Grid, can take benefits by knowing the geographic location of clients, for a considerable improvement of their decision-taking functions. A measurement-based geolocation service estimates host locations from delay measurements taken from landmarks, which are hosts with a known geographic location, toward the host to be located. Nevertheless, active measurement can burden the network. Relying on database-driven geolocation and active measurements, we propose GeoHybrid. GeoHybrid estimates the geographic location of Internet hosts with low overhead as well better accuracy with respect to geolocation databases. Afterwards, we propose a geolocation middleware for grid computing. By defining the architecture and the methods of this service, we show that a promising symbiosis may be envisaged by the use of the proposed middleware service for grid computing.

***Index Terms*—** Geolocation, Measurement, Grid performance optimization, Resource Broker

1. INTRODUCTION

Geographically locating an Internet host from its IP address enables a diversified and interesting new class of location-aware application. Nowadays a lot of services depend on the geographic location of Internet hosts. Examples of such applications comprise targeted advertising on web pages, displaying local events and regional weather, automatic selection of a language to first display the content of web pages, restricted content delivery following regional policies, and authorization of transactions only when performed from pre-established locations.

Multimedia delivery systems, such as Content Distribution Networks (CDNs) that offers a world wide service but has limited servers, can also benefit from knowing the location of their clients [1]. For example, benefits include the indication of nearby servers to clients or the location-based adaptation of the multimedia content. In other words, the nearest geographically located server, which in most cases is likely to have the lowest latency and/or highest bandwidth path.

Active measurement-based IP geolocation techniques have been proposed [2–6], and these may achieve desirable properties, such as accuracy, and robustness. These techniques use a set of reference hosts, called landmarks, to estimate the location of other hosts. However, these properties come at the expense of scalability, high overhead and very high response time ranging from tens of seconds to several minutes to localize a unique IP address. This is several order of magnitude slower that what is achievable with database-driven geolocation, representing the passive approach.

Database-driven geolocation usually consists of a database-engine (e.g., SQL/MySQL) containing records for a range of IP addresses, which are called *blocks* or *prefixes*. Geolocation prefixes may span non-CIDR subsets of the address space, and may even span only a couple of IP addresses. Examples of geolocation databases are *GeoURL* [7], the *Net World Map* project [8], and free [9–11] or commercial tools [12–16]. When coupled with a script embedded in a website and upon a client access to the website being detected, a request can be sent instantly to the database. This request can be to check if the IP address has an exact or longest prefix match (LPM) with a corresponding geographic location and coordinate. Since there is no actual measurement involved but merely a simple lookup, the request can be served in a matter of milliseconds. The expected time for which a website should be fully loaded, without causing any nuisance, is in general within one second.

Nevertheless, exhaustive tabulation is difficult to manage and to keep updated, and the accuracy of the locations is unclear. In practice however, most location-aware applications seem to get a sufficiently good geographic resolution for their purposes. Siwpersad et al. in [17] have shown that the geographic resolution of databases is far coarser than the resolution provided by active measurements, typically several times coarser than the confidence given by active measurements. As most geolocation databases do not give confidence in the accuracy of their location records, they are likely not to be trustworthy sources of geolocation information if precise IP address-level locations are required. Also, the geographic dispersion between results from several databases can span an entire region.

It became clear that solely relying on databases leads to incorrect results or results that have a high geographic dispersion. Furthermore, measurement-based geolocation can burden the network with extra traffic and can therefore trigger intrusion detection systems. We aim at mitigating the number of measurements generated in the network. To overcome theses limitations, we propose an hybrid geolocation service called *GeoHybrid*. Firstly, the technique GeoHybrid uses a database to find the geographic location of the IP block which hosts the IP of the target. Secondly, in order to improve the provided localization, GeoHybrid selects either few landmarks located at the vicinity of the geographic location of the IP block (heuristic choice) or randomly selects few landmarks. Afterwards, we localize target hosts with lower number landmarks compared to [2], and thus, mitigate the impact of measurements. Note that, the measurement tasks are done with the Constraint-Based Geolocation (CBG) technique. Furthermore, we improve the accuracy of geolocation databases. Afterwards, we compare both approaches (*i.e.* heuristic choice and random choice). The obtained results show that the heuristic choice outperforms the random choice.

The geographical distributed computing architectures - the so-called Grid - appear as new trend in supercomputing and distributed computing [18]. The users that perform operations such as submit jobs, control their execution and retrieve their output, demand resources allocation simultaneously. The quality of this service depends directly on the network condition, and the computation capacity of each cluster. Therefore, geolocation tools may contribute in supporting a highly dynamic environment where operational conditions are constantly changing. In fact, job execution may require one or more files and produces output data, thus, given the distribute nature of the databases, the input/output process can produce considerable data traffic across the Grid. Furthermore, if the same amount of resources are available everywhere, GeoHybrid can permit to the *Workload Management System* (WMS) to delegate jobs to the closest cluster. Note that, the WMS has the responsibility of managing the Grid resources. Furthermore, we can do a geographic mapping of different resources available on the Grid, and thus allow users the possibility to send their jobs following geographic constraints.

This paper is organized as follows. Section 2 reviews the related work on this field. Section 3 describes the CBG approach to estimate the geographic location of a given target host. In Section 4, we introduce our hybrid geolocation service and points out the contributions of GeoHybrid in contrast to previous approaches. Following that, we present results for datasets in Section 6. We illustrate the use of geolocation in case of Grid computing in Section 7. Finally, we conclude and present some research perspectives in Section 8.

2. RELATED WORK

A DNS-based approach to provide a geographic location service of Internet hosts is proposed in RFC 1876 [19]. This proposition, however, is not widely adopted since it requires changes in DNS structure and administrators have no motivation to register new location records. Tools such as [20,21] query Whois databases in order to obtain the location information recorded therein to estimate the geographic location of a host. This information, however, may be inaccurate or stale. Moreover, if a large and geographically dispersed block of IP addresses is allocated to a single entity, the Whois databases may contain just a single entry for the entire block.

There are also some geolocation services based on an exhaustive tabulation between IP addresses ranges and their locations. This is the case of some projects [7, 8] or commercial services [12, 15, 16]. Exhaustive tabulation is difficult to manage and to keep updated and unreliable, since the accuracy is hard to determine and it also relies on how truthfully a user has submitted his personal information. Furthermore, the results are usually coarse grained and not suited for applications where accuracy is required. The authors of [22] quantify the extent to which locating all IP addresses within a block leads to an inaccurate geolocation of Internet hosts. With active measurements, they show that the geographic span of block of IP addresses make their location difficult to choose. Therefore, using a unique location for a block of IP addresses as an estimate of the location of its IP addresses leads to significant localization errors, whatever the choice made for the location of the block.

Different techniques [3] estimate the geographic location of an Internet host from DNS names, from clustering the IP address space with BGP prefix information, or from delay measurements. An example of a discrete measurement is the GeoPing [3] approach where the location is based on the nearest landmark, thus having a discrete space of answers. In contrast, the Constraint-Based Geolocation (CBG) [2], where landmarks are used as well, the estimation is based on multilateration providing a continuous space of locations. The authors of [23] present a topology-based geolocation method. They extend multilateration techniques with topology information. In fact, they use traceroute from landmarks to map topology.

Nevertheless, measurement-based approaches burden the network with extra traffic and can therefore trigger intrusion detection systems (IDS). If an IDS is alarmed, it might block future access at some points in the route, which evidently will lead to incorrect measurements as well.

3. BACKGROUND ON CBG APPROACH

In this section, we present a brief background on how CBG provides geolocation estimation for target hosts based on delay measurements.

3.1. Multilateration with geographic distance constraints

The physical position of a given point can be estimated using a sufficient number of distances or angle measurements to some fixed points whose positions are known. When dealing with distances, this process is called multilateration.

Consider a set $\mathcal{L} = \{L_1, L_2, \ldots, L_K\}$ of K landmarks. Landmarks are reference hosts with a well-known geographic location. For the location of Internet hosts using multilateration, CBG [2] tackles the problem of estimating the geographic distance from these landmarks towards the target host to be located, given the delay measurements from the landmarks. From a measurement viewpoint, the end-to-end delay over a fixed path can be split into two components: a deterministic (or fixed) delay and a stochastic delay [24]. The deterministic delay is composed by the minimum processing time at each router, the transmission delay, and the propagation delay. This deterministic delay is fixed for any given path. The stochastic delay comprises the queuing delay at the intermediate routers and the variable processing time at each router that exceeds the minimum processing time. Besides the stochastic delay, the conversion from delay measurements to geographic distance is also distorted by other sources as well, such as circuitous routing and the presence of redundant data. Anyway, it should be noted that no matter the source of distortion, this delay distortion is always additive with respect to the minimum delay of an idealized direct great-circle path.

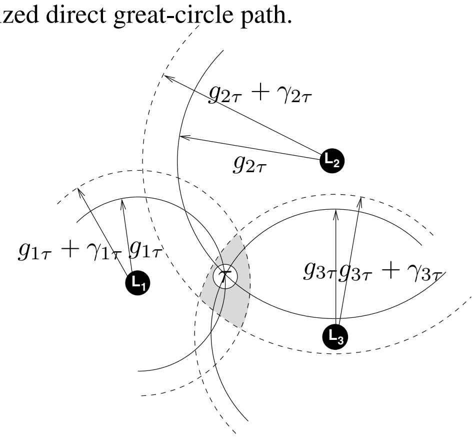


Fig. 1. Multilateration with geographic distance constraints.

Figure 1 illustrates the multilateration in CBG using the set of landmarks $\mathcal{L} = \{L_1, L_2, L_3\}$ in the presence of some additive distance distortion due to imperfect measurements. Each landmark L_i intends to evaluate its geographic distance constraint to a target host τ with unknown geographic location. Nevertheless, the inferred geographic distance constraint is actually given by $\hat{g}_{i\tau} = g_{i\tau} + \gamma_{i\tau}$, *i.e.* the real geographic distance $g_{i\tau}$ plus an additive geographic distance distortion represented by $\gamma_{i\tau}$. This purely additive distance distortion $\gamma_{i\tau}$ results from the possible presence of some additive delay distortion. As a consequence of having additive distance distortion, the location estimation of the target host τ should lie somewhere within the gray area (*cf.* Figure 1) that corresponds to the intersection of the overestimated geographic distance constraints from the landmarks to the target host.

3.2. From delay measurements to distance constraints

Previous work [3, 25] has investigated the correlation between geographic distance and network delay. Figure 2 provides an example of the relation between the distance and the delay for one of the landmarks we used in our measurements towards the remaining landmarks of our dataset (further details on the experimental data used are found in Section 6). The *bestline* shown in Figure 2 for a given landmark L_i is defined as the line that is closest to, but below all data points (x, y), where x expresses the actual great-circle geographic distance between this given landmark and all the other landmarks in the set, while y represents the measured RTT between the same pairs. The equation of the bestline is defined as

$$y = m_i x + b_i. \quad (1)$$

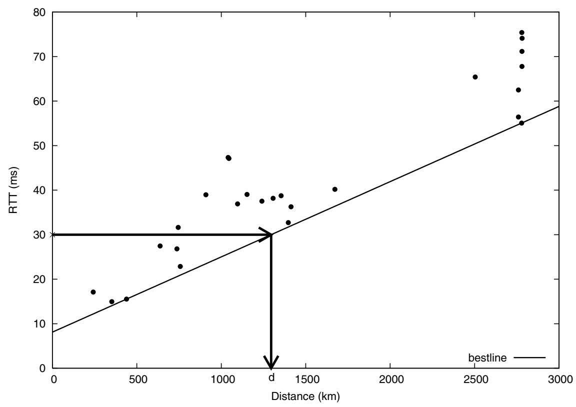


Fig. 2. Sample scatter plot of geographic distance and network delay.

It should be noted that each landmark finds its slope m_i and its positive intercept b_i based only on delay measurements between the available landmarks. For further details about the computation of b_i and m_i, we refer the reader to [2]. The presence of a positive intercept b_i in the bestline reflects the presence of some localized delay. Each landmark uses its own bestline to convert the delay measurement towards the target host into a geographic distance constraint. A delay measurement from the considered landmark of Figure 2 towards a particular target host τ is transformed into a distance constraint by projecting the measured delay on the distance axis using the computed bestline of this landmark. For example, if the measured delay is 30 ms, the distance constraint is d, as illustrated by the thick arrow in Figure 2. This estimated geographic distance constraint $\hat{g}_{i\tau}$ between a landmark L_i and a target host τ is derived from the delay $d_{i\tau}$ using the bestline of the landmark as follows:

$$\hat{g}_{i\tau} = \frac{d_{i\tau} - b_i}{m_i}. \quad (2)$$

Each landmark L_i localizes a given destination τ inside a circle whose radius is the obtained distance constraint $\hat{g}_{i\tau}$. The region formed by the intersection of all these circles from the set of landmarks is called in CBG the *confidence region*. CBG provides the centroid of this confidence region as the location estimation for the target host.

4. GEOHYBRID LOCALIZATION FRAMEWORK

The goal of GeoHybrid are twofold: (*i*) mitigate the number of measurements by reducing the number of landmarks used for geolocating target hosts, and thus enhance scalability; (*ii*) improve the accuracy of geolocation databases. In fact, using the single location for a block of IP addresses as an estimation of the location of its IP addresses leads to significant localization errors, whatever the choice made for the location of the block [22].

4.1. Hybrid geolocation framework

Fig. 3 illustrates the different components of our hybrid geolocation service. The geolocalization framework can be decomposed as follows :

- A database which contains block of IP addresses (entries). In fact, a database entry is composed of a pair of values, corresponding to the integer representation of the minimum and maximum address of a block. Each block is then associated with several informations helpful for localization: country code, city, latitude and longitude, and Zip code.
- A given server where is implemented the heuristic which allows to trigger, if necessary, measurements from landmarks towards a fixed host.
- Afterwards, if measurement task is needed, the server delegates the measurements to few landmarks which are chosen following a fixed rules. It is worth noticing that the selected landmarks use CBG technique 3 to localize target hots.

The process of locating a given target with GeoHybrid host is more explained in Section 5.

4.2. Structure of database used for IP geolocation

According to the GeoHybrid framework, as illustrated in Fig. 3, when a request arrives for geolocation purposes, the server should use a database to geolocate the target host. In the sequel of this paper we restrict our attention to one commercial database called GeoIP. This database, GeoIP by Maxmind [16] is used because of its popularity (see [16] for a listing of some of their customers) and its expected reliability.

In fact, the Maxmind database is split into two parts as depicted in the Table 1: *table 1* and *table 2*. One part contains the IP prefixes and a location identifier (loc id). The other part consisted of the representation of the location identifiers such as country, region, city, zip code and geographic coordinate. Maxmind contains more than 3 millions of block of IP prefixes. It should be noted that “lon.” and “lat.” means longitude and latitude respectively in Table 1.

Using an exhaustive tabulation as in [7,8,16], we find the IP block which owns the IP address of the target host. By knowing the location of this IP block, one can determine from table 2 the geographic location of the target host. It should be noted that the goal of the exhaustive tabulation is to check if the IP address has an exact or longest prefix match with a corresponding geographic location and coordinate. As we know the coarse grained location of the target host, we can select the set of landmarks $\mathcal{L}$, following a given criteria, that should perform measurement task. Otherwise, if the IP target belongs to any database’s block, we should use all landmarks available in our measurement infrastructure to estimate the position of the target.

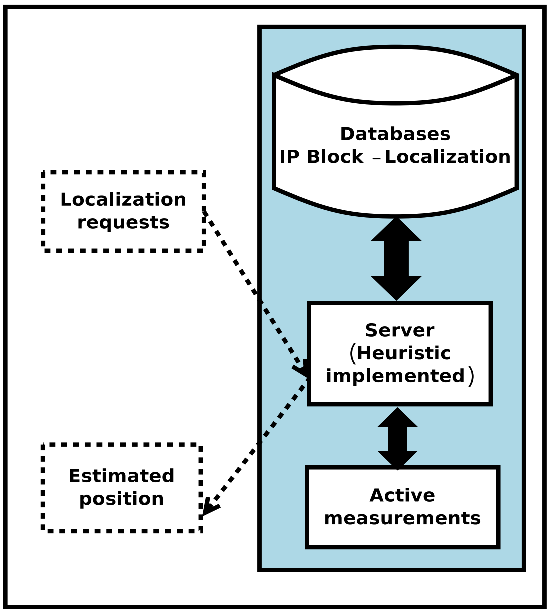


Fig. 3. Hybrid geolocation framework.

Table 1. Database fiels.

table 1	
IP prefix	**loc id**

table 2						
loc id	**country**	**region**	**city**	**zip code**	**lat.**	**lon.**

5. HEURISTIC CHOICE OF LANDMARKS

As shown in the GeoHybrid framework (Fig. 3), the server implements several heuristics simultaneously for the selection of probes (landmarks). The core feature of GeoHybrid is its capability to use only the set of landmarks located at the vicinity of the IP prefix that owns the target host. It is worth noticing that the geographic location of the set of landmarks $\mathcal{L}$ is known. After having the geographic location of the IP prefix that hosts the target, from Maxmind database, we can estimate the distance between the set of landmarks and the target host. Based on [26], the geographic distance between each landmark L_i and the target host τ can be estimated as follows:

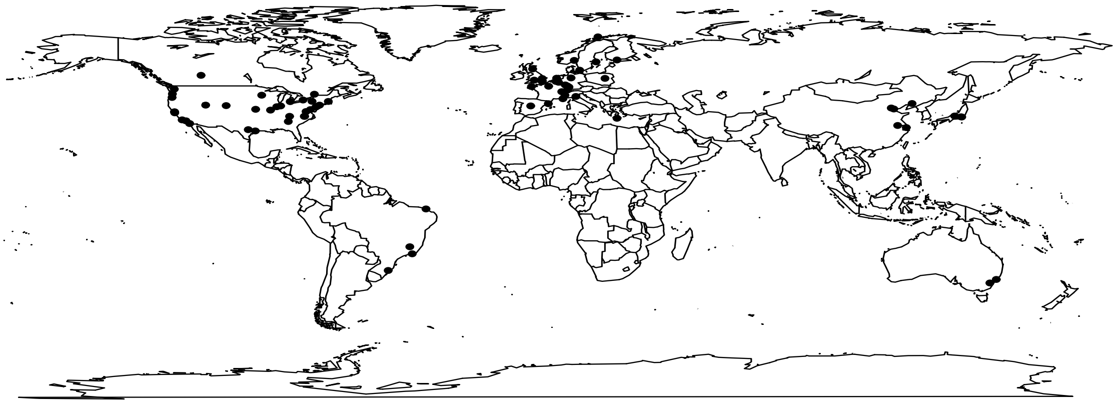


Fig. 4. Geographic location of landmarks.

$$\beta = \sqrt{\left(\sin\left(\frac{lat_i - lat_\tau}{2}\right)\right)^2 + \cos(lat_i) \times \cos(lat_\tau) \times \alpha} \tag{3}$$

$$\alpha = \left(\sin\left(\frac{lon_\tau - lon_i}{2}\right)\right)^2 \tag{4}$$

$$\hat{dist}_{i\tau} = 6371 \times 2 \times \arcsin(\beta) \tag{5}$$

It should be noted that lat_i and lon_i represent the latitude and longitude, expressed in radian, of landmark L_i; lat_τ and lon_τ, also in radian, represent the latitude and longitude of the target host τ. Afterwards, the geographic distance (in km), between landmark L_i and the target τ is obtained from equation 5. The value 6371 used in equation 5 represents the radius of the earth and the product $2 \times \arcsin(\beta)$ gives the geographic distance expressed in radian. It is worth noticing that in section 6, the distance are expressed in km. For the target host τ, we obtained the following distance vector:

$$D_\tau = [\hat{dist}_{1\tau}, \hat{dist}_{2\tau}, \ldots, \hat{dist}_{K\tau}], \tag{6}$$

where K represents the total number of landmarks of $|\mathcal{L}|$, and $\hat{dist}_{i\tau}$ represents the geographic distance (in km), computed between the landmark L_i and the target τ for $1 \leq i \leq K$.

Assume that we would like to choose only n among the K landmarks which form the set of landmarks $\mathcal{L}$ for measurement purposes. The goal of our heuristic is to find the n nearest landmarks towards the target hosts. In other words, we should find the smallest distances $\hat{dist}_{i\tau}$, $1 \leq i \leq n$ with respect to equation 6.

6. EVALUATION

6.1. Datasets

To validate our heuristic, we use two datasets formed by RIPE hosts [27] and AMP hosts [28]. The experimental datasets comprise 127 hosts located in United States and Europe. The main reason for this restriction is that the datasets we have had correspond to hosts located in these regions. Unfortunately, datasets that provide the geolocation of the involved hosts are uncommon. Nevertheless, we indeed believe that the results we report in this paper are interesting and promising in spite of being limited to the U.S. and Europe.

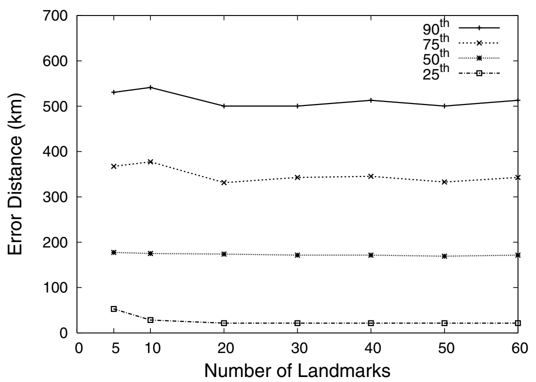
In this paper, we consider *MaxMind* [16] which is a commercial geolocation database. Maxmind database is formed by more than 3 millions IP blocks and each block is associated with several informations helpful for localization: country code, city, latitude and longitude, and Zip code (Table 1). Note that block prefixes are between /8 and /32. Nevertheless, most of IP block from Maxmind correspond to subnet smaller than /25.

In our experiments, for geolocating target hosts, we consider 74 PlanetLab [29] nodes as landmarks. Their geographic distribution is illustrated in Fig. 4. Landmarks perform *ping* measurements towards a given target host to locate it. Each ping is composed by 10 packets sent by interval of 1 second. The inter-packet spacing is due to the fact that we do not want to trigger IDS alarm. Each packet has a size of 1024 Ko. Only the minimum RTT (Round Trip Time) is considered. In order to locate target host we use the CBG methodology described in section 3.

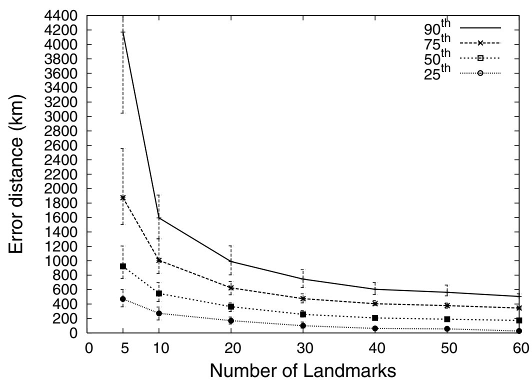
6.2. Results

In this section, we evaluate the impact of the number of adopted landmarks in the performance of GeoHybrid. After inferring the point estimate for each considered target host, we compute the error distance, which is the difference between the estimated position and the real location of the target host τ.

Fig. 5 shows different percentile levels of the error distance of GeoHybrid location estimates as a function of the number of adopted landmarks. For example, the 90^{th} percentile



(a) Landmarks chosen by an heuristic



(b) Landmark chosen at random

Fig. 5. Error distance as a function of the number of landmarks.

curve represents the error distance at which the CDF plot of mean error distance meets the 0.90 probability mark. The x-axis is the number of chosen landmarks among all landmarks, and the y-axis is the difference between the estimated position and the real location of the target host. The number of landmarks varies between 5 and 60.

Fig. 5(a) illustrates the case where landmarks have been chosen according to their vicinity to the location of the IP block which hosts the target (*i.e.* heuristic choice). We remark that a certain number of landmarks, typically about 20, is needed to level off the error distance (Fig. 5(a)). Nevertheless, for curves illustrated the 90^{th} and 75^{th} percentile, we have a slight rise of the estimation error. Probably, the reason is due to the presence of some distortion in our delay measurements caused by the added landmarks, which are far with respect to the target hosts. Nevertheless, the general trend observed in Fig. 5(a) is, more chosen landmarks are the closest towards the target hosts and more the estimation is better. Indeed, by considering only the closest 20 landmarks, 50% of target hosts are located with an error distance lower than 175 km.

Fig. 5(b) illustrates the impact of the number of adopted landmarks in the performance of GeoHybrid. Note that, the choice of landmarks is done randomly. We compute the mean error distance as the average of all error distances corresponding to several random sets of k landmarks chosen out of the total number of available landmarks (74 landmarks). Because the number of possible placement combinations become very large as we increase k, we do not consider all the possible choices of k landmarks. Error bars indicate the 99% confidence interval. These results suggest that a certain number of landmarks, typically about 30, is needed to level off the mean error distance. Nevertheless, the obtained error with random approach is upper than the heuristic choice. Indeed, with 30 landmarks chosen randomly, 50% of target hosts are localized with an error lower than 400 km. In contrast, the heuristic choice has an estimate error lower than 175 km for 50% of target hosts.

By considering a few number of landmarks we reduce the amount of time needed to localize a target host, and thus the response time is widely shortened. Furthermore, we mitigate the number of traffic generated in the network.

7. GEOLOCATION SERVICE FOR GRID COMPUTING MIDDLEWARE

The integration of geolocation information can be extremely useful for the optimization of the decision taking process of a Grid Resource Broker. For instance, the GeoHybrid service can be used for the improvement of data management among different *Storage Elements*: for the selection of the nearest replica of a given file if multiples copies of it are present in different storage elements.

7.1. DataGrid overview

The Workload Management System (WMS) is the component of the Grid that has the responsibility of managing the Grid resources, (*i.e.* in each Site i (Fig.6)), in such a way that applications are conveniently, efficiently and effectively executed. It is formed by the :

- *User Interface (UI)*: it allows a user to interact with the Grid in order to perform operations such as submit jobs, control their execution, and retrieve their output.
- *Resource Broker (RB)*: it is the core component of the WMS. The RB tries to find the best match between the job requirements and the resources available on the Grid whose characteristics are retrieved from *Information System* (Fig. 6). The output of the search is a *Computing Element* where the job, while running, has access to all resources specified in the job description, such as data or storage space.
- *Logging and Bookkeeping (LB) Service*: it is the Grid service responsible to store and manage logging information which concerns the WMS itself. Furthermore, the bookkeeping collects information about active jobs, i.e, jobs that are within the WMS. It consists of the job definition, its status, resource consumption. In particular from the events stored in the logging and

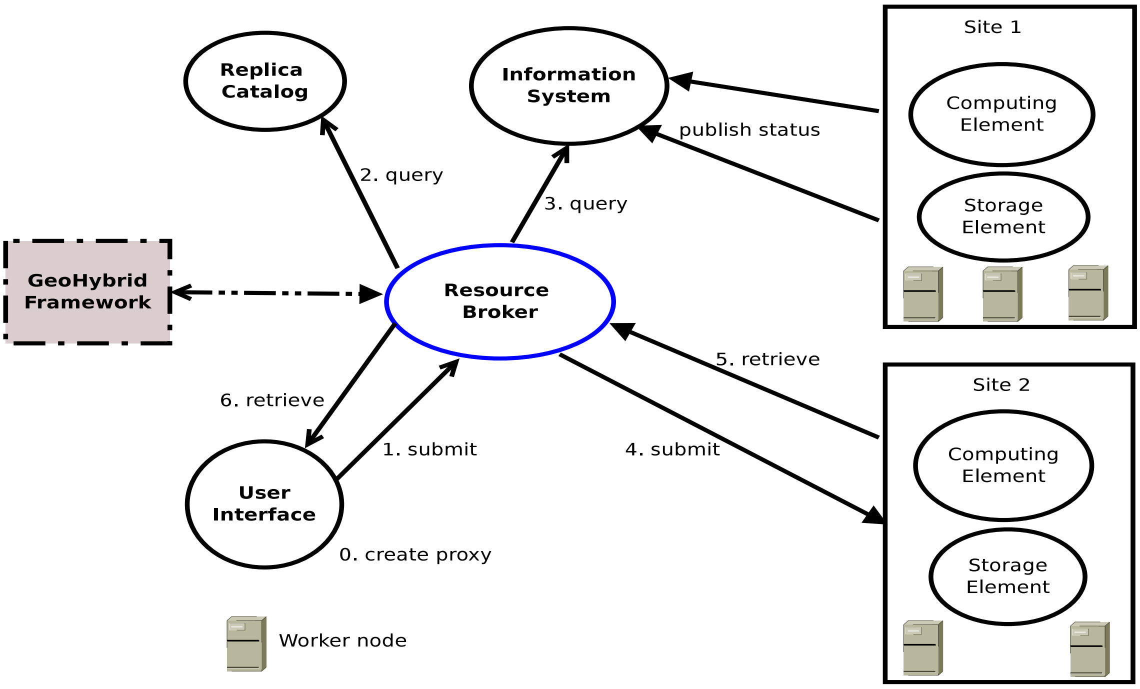


Fig. 6. A logical view of the Geolocation-based Grid Optimizer.

bookkeeping databases it is possible to reconstruct the status of a job that was previously submitted to a Resource Broker for execution on the Grid. The LB is located inside the Information System shown in Fig. 6.

- *Top-BDII (Information Index)*: it is a LDAP (Lightweight Directory Access Protocol) server which collects the different resources available in the Grid. It is used by the Resource Broker in order to select resources. It should be noted that each site can have its own BDII called Site-BDII. In such case, it collects the available resources in the site, from the Computing Element, and shares this information with the Top-BDII. The Top-BDII is located inside the Information System.

7.2. Optimization scenarios

Fig. 6 illustrates a grid optimization service. Let us assume that a user wants to send a job to a RB. Firstly, it needs to access to User Interface; it obtains a timeout of 24 hours for doing its job ("*create proxy*") (see Fig. 6). Note that, this timeout can be renewed. Afterwards, the User Interface submits the job to the RB, and then the RB sends a request to a *Replica Catalog* (Fig.6) in order to verify if it is possible to realize this task. In such case, the RB queries the *Information System*, and thus receives a list of candidate worker nodes often geographically distributed. Note that, this list contains the best computing element for a given job execution. Afterwards, the RB can send a request to the GeoHybrid server, as illustrated in Fig. 3, in order the find the geographic location of the user and the worker nodes. Following the obtained responses from GeoHybrid, the RB selects the closest worker node towards the user among the list of candidate worker nodes. Therefore, according to this heuristic, we mitigate the amount of traffic exchanged across the Grid.

8. CONCLUSION

In this paper, we proposed the GeoHybrid framework, a scalable measurement-based method to estimate the geographic location of Internet hosts. Relying on geolocation database and active measurement, GeoHybrid estimates the geographic location of Internet hosts with lower overhead by reducing the number of used landmarks. Using active measurement, GeoHybrid provides also better accuracy with respect to geolocation databases by improving their geographic estimation which is coarse grained.

Our experimental results show that the heuristic choice, where we select only the closest landmarks towards a given IP block, outperforms the approach where landmarks are chosen randomly. Indeed, with 30 landmarks chosen randomly, 50% of target hosts are localized with an error lower than 400 km. In contrast, the heuristic choice has an estimate error lower than 175 km for 50% of target hosts and typically about 20 landmarks, is needed to level off the error distance.

The synergy between the areas of grid computing and geographic location points out the importance of a specific measurement middleware service. Based on GeoHybrid, we improve the selection mechanism of worker nodes from the Resource Broker. Indeed, the candidate worker nodes can be sorted following their vicinity to the user which sent the job. Therefore, the amount of traffic generated across the Grid is minimized.

Our future work consists to implement this middleware in the Research Education Network which interconnects different Universities and High schools in Senegal.

9. REFERENCES

[1] Kunwadee Sripanidkulchai, Bruce Maggs, and Hui Zhang, "Efficient content location using interest-based locality in peer-to-peer systems," in *Proc. IEEE INFOCOM*, April 2003.

[2] B. Gueye, A. Ziviani, M. Crovella, and S. Fdida, "Constraint-based geolocation of Internet hosts," *IEEE/ACM Transactions on Networking*, vol. 14, no. 6, pp. 1219–1232, December 2006.

[3] V. N. Padmanabhan and L. Subramanian, "An investigation of geographic mapping techniques for Internet hosts," in *Proc. ACM SIGCOMM*, August 2001.

[4] E. Katz-Bassett, J. John, A. Krishnamurthy, D. Wetherall, T. Anderson, and Y. Chawathe, "Towards IP geolocation using delay and topology measurements," in *Proc. ACM SIGCOMM Internet Measurement Conference (IMC)*, October 2006.

[5] B. Wong, I. Stoyanov, and E. G. Sirer, "Geolocalization on the Internet through constraint satisfaction," in *Proc. USENIX Workshop on Real, Large Distributed Systems (WORLDS)*, Seattle, WA, November 2006.

[6] B. Gueye, S. Uhlig, A. Ziviani, and S. Fdida, "Leveraging buffering delay estimation for geolocation of Internet host," in *Proc. IFIP Networking Conference*, Coimbra, Portugal, May 2006.

[7] "GeoURL," http://www.geourl.org.

[8] "Net World Map," http://www.networldmap.com.

[9] "Host IP," http://www.hostip.info.

[10] IPInfoDB, "Free IP address geolocation tools," http://ipinfodb.com/.

[11] Software 77, "Free IP to country database," http://software77.net/geo-ip/.

[12] GeoBytes Inc., "GeoNetMap - geobytes' IP address to geographic location database," http://www.geobytes.com/GeoNetMap.htm.

[13] Qwerks Inc., "WhereIsIP - IP whois tool," http://www.jufsoft.com/whereisip.

[14] Hexasoft Development Sdn. Bhd, "IP address geolocation to identify website visitor's geographical location," http://www.ip2location.com.

[15] Quova Inc., "GeoPoint - IP geolocation experts," http://www.quova.com.

[16] MaxMind, "Geolocation and online fraud prevention from MaxMind," http://www.maxmind.com/.

[17] S. Siwpersad, B. Gueye, and S. Uhlig, "Assessing the geographic resolution of exhaustive tabulation for geolocating Internet hosts," in *Proc. of PAM*, Cleveland, Ohio, USA, April 2008.

[18] Ian Foster, "The grid: A new infrastructure for the 21st century science," in *Grid Computing: Making the Global Infrastructure a Reality*. 2003, pp. 51–63, John Wiley & Sons.

[19] C. Davis, P. Vixie, T. Goodwin, and I. Dickinson, "A means for expressing location information in the domain name system," *Internet RFC 1876*, Jan. 1996.

[20] University of Illinois at Urbana-Champaign, *IP Address to Latitude/Longitude*, http://cello.cs.uiuc.edu/cgi-bin/slamm/ip2ll/.

[21] D. Moore, R. Periakaruppan, J. Donohoe, and k.c. Claffy, "Where in the world is netgeo.caida.org?," in *Proc. of INET*, Yokohama, Japan, July 2000.

[22] B. Gueye, S. Uhlig, and S. Fdida, "Investigating the imprecision of IP block-based geolocation," in *Proc. Passive and Active Measurement Conference (PAM)*, Louvain-la-neuve, Belgium, April 2007.

[23] Ethan Katz-Bassett, John P. John, Arvind Krishnamurthy, David Wetherall, Thomas Anderson, and Yatin Chawathe, "Towards ip geolocation using delay and topology measurements," in *Proc. of IMC*, New York, NY, USA, 2006, pp. 71–84, ACM.

[24] C. J. Bovy, H. T. Mertodimedjo, G. Hooghiemstra, H. Uijterwaal, and P. van Mieghem, "Analysis of end-to-end delay measurements in Internet," in *Proc. of PAM workshop*, Fort Collins, CO, USA, Mar. 2002.

[25] Artur Ziviani, Serge Fdida, José Ferreira de Rezende, and Otto Carlos Muniz Bandeira Duarte, "Toward a measurement-based geographic location service," in *Proc. of PAM*, Antibes Juan-les-Pins, France, Apr. 2004, Lecture Notes in Computer Science (LNCS) 3015, pp. 43–52.

[26] *Ed Williams*
, http://http://williams.best.vwh.net/avform.html.

[27] *RIPE Test Traffic Measurements, 2000*, http://www.ripe.net/ttm/.

[28] *NLANR Active Measurement Project, 1998*, http://watt.nlanr.net/.

[29] PlanetLab, "An open platform for developing, deploying, and accessing planetary-scale services," 2002, http://www.planet-lab.org.

CONTEXT-AWARE SMART ENVIRONMENTS ENABLING NEW BUSINESS MODELS AND SERVICES

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ABSTRACT

This work describes innovative smart environments with embedded context-awareness technologies, enabling new business models and consequently the creation of new services. The context-awareness framework presented in this paper is taken from the results of an EU Framework Programme (FP) 7 Information and Communications Technologies (ICT) project. Major novelties include a business shift from traditional and conventional telecommunication or ICT services towards highly personalized, customized and user targeted services, empowered by a myriad of pervasive and ubiquitous interconnected environments employing various kinds of context information. In this work, we show how these context data can be technically made available as a service and business enabler and be used by any entity or application built within these environments, using context for adapting service logic or for targeted service customization. Moreover, it considers customer's needs and privacy aspects, providing users with a more immersive and less intrusive experience at the same time.

Keywords**—**business model, service, smart environment, context

1. INTRODUCTION

The number of mobile user terminals permanently connected to the web over wireless technologies as potential context sources (e.g., embedded sensors, cameras) together with large interworking Wireless Sensor Networks (WSN) interconnected with Building Sensor Network and Automotive Sensor Networks are increasing significantly, allowing real ubiquitous systems. All these information sources create heterogeneous and very pervasive environments collectively identifiable as Smart Spaces. Users immerged into this technology expect "smarter" and more useful services with richer multimedia content.

In parallel, the advent of Web 2.0 technologies and their consequent application deployments revealed that user generated content, when tied to a certain geographical location and part of an augmented reality with virtual social communities, became a trend in what concerns user most used services. In this sense, leveraging smart space technologies with a context-aware framework would allow objects, devices, people and entities to effectively interact in real time.

Among all players involved in the business value chain of the innovative services, users themselves are the ones demanding for appealing services. To satisfy this demand, a mentality shift needs to occur, as the technological one occurred with the emergence of context-awareness and WSN technologies, enabling new business models and revenue schemes.

The context management technology used in this work has been designed in the C-CAST European research project, which aims at combining mobile content distribution techniques (both unicast and multicast) and context-awareness in order to optimize multimedia content distribution. This work shows how it can be used for new business models in smart environments. The remainder of the paper is organized as follows. Section 2 presents the state-of-the-art in context-awareness for smart spaces. The technical and business aspects of context management are presented in sections 3 and 4, respectively. Then, section 5 describes some context-aware commercial services and trials designed by Telecom Italia Labs. Finally, section 6 concludes the paper and introduces future work.

2. STATE OF THE ART

This section introduces the terminology and provides an overview of related research activities in the areas of context management.

2.1. Definitions and Terminology

Context – Ryan et al. [1] referred to context as the user's location, environment, identity and time. Dey [2] defines context as the user's emotional state, focus of attention, location and orientation, date and time, as well as objects and people in the user's environment. Hull et al. [3] describe context as the aspects of the current situation. Brown [4] defines context to be the elements of the user's environment, which the computer knows about. The notion of situation is closely related to the concept of context. Zimmermann [5] defines it as "the state of a context at a certain point (or region) in space at a certain point (or interval) in time, identified by a name".

Smart Space – a smart space is any real or virtual location equipped with passive and active artifacts. These artifacts can be any kind of sensors and actuators, mobile devices, software agents, autonomous vehicles, management systems, and also human beings. In smart spaces, these artifacts have the processing and communication capabilities to interact with each other in a (mutually) beneficial way. Examples include smart homes, smart factories, smart transportation, smart grid or ambient assisted living environments [6], [7].

2.2. Context-Aware Systems for Smart Spaces

Ailisto et al. [9] and Dey and Abowd [10] proposed a layered conceptual architecture that augments layers for detecting and using context by adding interpreting and reasoning functionalities. In middleware-based systems, the Service-Oriented Context-Aware Middleware (SOCAM) project [11] introduced an architecture for building and rapid prototyping context-aware mobile services. One further extensible centralized middleware approach designed for context-aware mobile applications is a project called Context-Awareness Sub-Structure (CASS) presented in [12]. A context-broker-based approach has been presented by Chen [13] (Context Broker Architecture, CoBrA), and further investigated in the SOUPA, and GAIA [14] frameworks. Many EU projects have explored context-awareness aspects, for example, SPICE [15] and MUSIC [16]. One of the reasons of little context-awareness usage, despite the mentioned efforts, was the lack of end user devices that could host the context-aware applications for users. With the wide scale availability of devices embedded with sensors, including the WSN sensors, it is now possible to leverage context-aware services. Most known types of services where context-aware smart spaces could play a significant role are advertisement, e-recommendations, e-tourism and gaming, opening the path for personalized content creation, customized service adaptation and intelligent content distribution, enabling new business models in these smart spaces applications.

3. TECHNICAL ASPECTS OF CONTEXT-AWARE SMART SPACES

Before focusing on the business aspects, it is important to understand the technical requirements of smart environments and how context is managed and represented.

3.1. Context Management Requirements for Smart Spaces

Intelligent environments and context-aware services are useful in both business and private application areas. Telecom services, as well as energy saving, gaming, chatting, entertainment, information and payment services are exemplary constituents of smart environments. Further smart space deployment areas are indicated in Figure 1.

The design of a context management system shall consider objectives and requirements from various domains, such as data management, communication, user identity management, system reliability, privacy, security, etc. The novelty here lies in the application of context information in smart spaces for supporting conventional, as well as context-aware applications. This section gives an overview of the most common high-level objectives and requirements [17], [18] which are the following:

1) Enhance and leverage context data through
   - filtering mechanisms as well as evaluating and improving its quality;
   - aggregation, fusion, inference, and prediction.
2) Efficiently exchange data and provide context services within and between different smart spaces by
   - establishing rules for context discovery, storage, and distribution;
   - designing common and open interfaces for communication within and across smart spaces;
   - specifying communication mechanisms between smart space artifacts
   - establishing management procedures for smart space entities owning context data.
3) Ensure end-to-end data security by
   - monitoring and assuring the non-violation of privacy, security and trust within and across smart spaces;
   - guaranteeing the enforcement of rules and policies.

These overall objectives yield a set of design requirements for the desired context-aware smart environment functionalities.

Figure 1. Context-Aware Smart Spaces: Potential Deployment Areas

Furthermore, the system needs to be capable of obtaining and representing context data from a wide range of sources, and at the same time, processing and storing them according to rules and policies defined by the artifacts (users, objects, etc.) present in a smart space. Moreover, mechanisms for computation of semantically more abstract context from simple types of contexts (aggregation, fusion, reasoning, prediction, and inference) for situation recognition and adaptive behavior in smart spaces have to be made available. This includes the usage of widely standardized communication protocols and interfaces. A smart space has to dispose of a (preferably distributed) context service directory that stores information about where and which context sources are available within a smart space. In addition, a joining mechanism for new context sources shall be available. Finally, mechanisms for identification, authentication and authorization guaranteeing end-to-end security and privacy of context within and across smart spaces, have to be included.

3.2. Context Management Architecture

The context-awareness architecture as described in this section is shown in Figure 2 and is based on the producer consumer role paradigm.

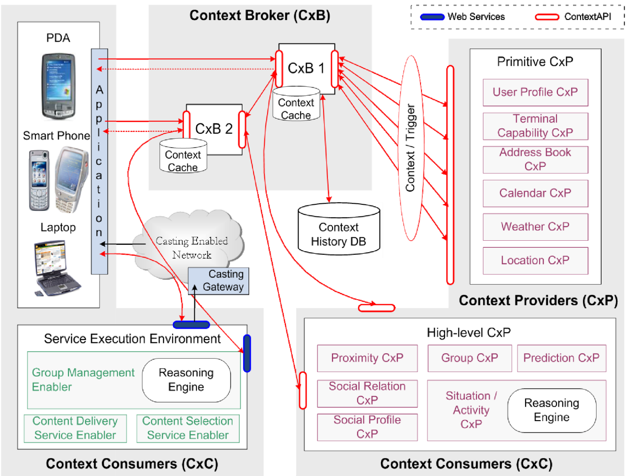

Figure 2. Context-Awareness Architecture [19]

The system is built upon three basic functional entities: Context Consumer (CxC), Context Broker (CxB) and Context Provider (CxP), where a CxP is a source of context information, a CxB is an arbitrary node handling context data as a reference point for all active entities in a smart space and a CxC is a consumer of context information. As shown in Figure 2, simple CxP usually only provide data, whereas more complex CxP functionalities, such as prediction or situation recognition, both consume and provide context information. The decoupling of provisioning and consumption is important as it impacts on the overall scalability of the system. Finally, the architecture disposes of so-called Group Management as well as Content Selection and Delivery modules to enable group-based multicasting of selected contents to different terminals.

3.3. Context Modeling and Representation

Context information needs to be represented and modeled for being machine interpretable and exchangeable using well-defined interfaces. The goals are to support easy manipulation on many devices, including resource constrained mobile devices, achieve low overhead in keeping the model up-to-date, easy extension, efficient search and query access, having cheap and simple mechanisms for adding new types of information and scalability. Within our work, we produced a representation and exchange format for context information in smart environments based on XML.

This format can handle several layers of abstraction as illustrated in Figure 3. Therefore, the context-aware architecture must be capable of building representations and models of these layered abstractions.

The main context handling features are context acquisition, aggregation & fusion, dissemination, discovery and lookup. Adapting such a structure to existing infrastructures could allow a context handling interface to be embedded into a service exposure layer as a SOA-like web service, allowing secure and controlled access to context information.

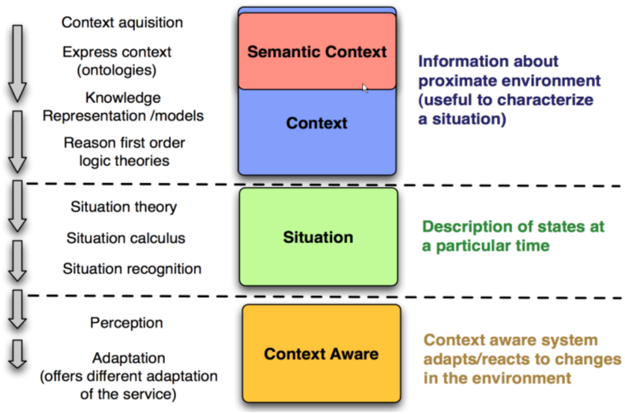

Figure 3. Layered Context Model

3.4 Context Data Classification

Context information can be derived from the physical environment or the physical state of an entity (e.g., temperature, humidity, light, noise levels, etc.) as well as its logical attributes (e.g., activity, friend, parent, preferences, etc.). Moreover, context may be of primary or derived origin, where derived context is effectively reasoned from primary context data.

For operational purposes, we therefore separated context data into three different categories: real-time, historical and reasoned context. Table 1 provides some examples of these context data types.

Real-time context represents the data available at a certain time from the Context Management System. This includes data from the context cache containing still valid (i.e. non-expired) context for fast access to increase system performance. Historical data refers to data whose validity has expired.

3.5 Context Prediction for Generating Higher-Level Knowledge

Reasoning refers to the information that can be inferred from analyzing data and combining different context information. This is usually more complex to obtain and should have a confidence level associated with it. One example for reasoning includes context prediction [20].

Context prediction deals with the issue of generating awareness for future context. This implicitly includes the proactive control and adaptation of systems, such as smart spaces, according to (expected) future user behavior and context. Generally, an entity of a context management system (e.g., CxC) can query or subscribe future values of a specified context scope. An exhaustive prediction contains the following parameters:

- Name of context scope to be predicted
- Name of entity the scope belongs to
- Future point in time the context should be computed for
- Confidence interval of the prediction

For the actual prediction process, a set of algorithms for analysis of historical data are available. Depending on the prediction request, the most appropriate algorithm is selected from the following ones: Markov models, hidden Markov models, or Bayesian networks. Relevant criteria include the time horizon of prediction, type of context (i.e. expected data characteristics), performance of previous predictions for specific applications (e.g. service execution in smart space environments), as well as, real time requirements. Another important aspect is the degree to which the algorithm needs supervision, i.e. human interaction. Usually, for smart environments, prediction should be performed completely unsupervised. As such, the combination or parallel execution of different algorithms shall provide predictions with higher confidence and reliability.

Table 1. Types of Context Data Available within a Smart Environment

Real-Time	**Historical**	**Reasoned**
Location	Previous locations	Movement
Orientation	Previous orientation	Movement, destination
Temperature	Previous temperatures	Weather
Humidity	Previous humidity	Weather
Brightness	Previous brightness	Weather, Time
Noise level	Previous noise levels	User activity, user location
Proximity	Previous proximity	Friends, colleagues, customers
Telecom Service usage	Call/VAS Logs	Relationship
Phone profile settings	Previous profile settings	Proximity
HW version, FW release and SW packages installed	Previous devices settings	Service type, capability, content format
Battery level	…	Service usage

4. BUSINESS ASPECTS OF CONTEXT-AWARE SMART SPACES

This section shows how context-awareness enables new business models in smart environments.

4.1. Relevant Applications and Business Models

Context-aware smart spaces may be used by many applications and services. The most interesting application classes that can benefit from context-awareness in smart spaces are the following:

- sustainability – including assisted living, smart cities, traffic management, intelligent energy management, tele-working, smart remote (business) process control
- machine to machine communications – including smart factories, logistics, and vehicular communication
- e-health – including smart hospitals, personalized fitness programs, elderly support from remote, advanced national health and wellness program
- emergency management, recovery and rescue teams support – fast recovery from catastrophes, remote monitoring, support for rescue forces and efficient material allocation
- gaming – including distributed and remote entertainment, education and tourism systems.

Any type of company from both the Telecom and IT industries may leverage on the potential of the presented context-aware applications in smart spaces. Some of the most promising business models applicable in this area will be described in the following sections.

4.1.1. The In-House Model

One of the best ways to avoid churn is to offer users what they want and need. Using context intelligence to improve services customization and personalization became a trend and operators are starting to use context to improve or create, existing or new services, respectively. A great example of such deployment is provided by some Telcos where a service merges a simple address book into an immersive social experience. Similar ideas can be employed for enhancing user experience in smart spaces. Although very often third party applications (e.g., Facebook, Twitter) are used, there is not any sort of revenue sharing. With this approach, they improve not only customer satisfaction but also revenue generation resulting from the associated data consumption from the operators' network. Furthermore, as these services are usually associated with fidelity plans, this helps operators to rethink long-term strategies.

4.1.2. The Revenue Model

This particular method can exist in two variants. The first occurs when the Telco improves its offer by using context information from external sources (e.g., social networks, traffic or news provider). This is similar to the outside-in methodology or to a white labeling product/service where the operator leverages its customer base to a new offering and earning a percentage out of it. The second approach resembles the inside-out approach, where third party applications are built not only based on operator infrastructure (using APIs) but also its context information. In this situation, although the Telecoms do not own the end user of these applications; they still earn a part of the revenue generated from this service.

4.1.3. The Advertising Model

Advertising can be leveraged by a smart space infrastructure in a myriad of ways. It can be done in a non-intrusive way when the customer requests a specific service (e.g. in a shopping mall), which is then sponsored by advertising. In any case, advertisement can be targeted according to a user's preferences and context. Therefore, advertisers are no longer limited to little customer segmentation options. In a way, this model is a mix of revenue sharing and context exposure selling as they earn money by using both their infrastructure (e.g., SMS or MMS) and their context information (e.g., location).

The main challenge of this model is to provide advertisements in an unobtrusive way, simultaneously considering users' interests for a specific moment in time. These targeted events will consequently improve the probability of service sales since the offer will match his or her needs, interests and context.

4.1.4. The Context Exposure Selling Model

Operators can extend the usage of their customers' context to outside their domain in order to improve their long-tail experience and creating revenues from businesses built by third parties. In this model there is no strict correlation between the service offered by third party providers and the operator. In this sense, the Telco exposes context through well-defined APIs. Then, depending on what is necessary the context can be accessed by:

- Request – can be done per user, per group of users, or per context. In this case, the billing is performed on a per request basis or per response basis (both models could work).
- Subscription – for some services, in case an application needs to keep an updated status of the context (e.g., to track movement you need to update location regularly). For this case, a fee can be applied for the subscription of a certain context, to certain users, group of users or a combination of all, as the updates can occur without any specific pattern and therefore is difficult to estimate a price per notification.
- Bundling – bundling context decreases the offer price when compared to single purchases, while significantly improving or completing the service offer.

4.1.5. The Self-Service Model

In today's interactive networks, users are no longer just consumers but also producers. Although for now this concept is mainly associated with content, it is predictable that in the future the same tendency is reflected on service composition and execution, where people can build their own applications based on intuitive graphical interfaces that represent logical interfaces to a myriad of context enablers. In this scenario the operator would also need a service orchestrator based on context. This would pave the way to a whole new way of service creation and consequently revenue generation. In a first instance, it would allow customers to engage with the service and network providers and would accelerate new service development, paying back the network operator's investments in customer tailored services. In a way, this can be seen as the "Application Stores" business model, where customers could then publish their services, using both the operators' infrastructure and context information. For this to happen, a revenue sharing model together with context exposure billing model should be defined to charge an intermediate or an end user.

4.2. Limitations

Despite the simplicity and flexibility in which these business models can be implemented, the processing and management of personal data obviously raises security and privacy issues that need to be addressed. In this sense, users need to stay in control of the whole process, enabling them to specify when, what, why, by whom, where and how the data is or can be accessed. In other words, user context should be disclosed according to contextual privacy policies and settings. Even the most primitive model should allow users to opt-in and opt-out from such profiling.

In addition, it is necessary to achieve a consensus and agreement among different stakeholders so that information can be provided in a standardized way, as envisioned in [21]. This will allow for compatibility within and across different smart spaces and it will facilitate the reusability of applications in different smart spaces. Examples include developer-independent software APIs as well as manufacturer-independent "hardware APIs". Moreover, context shall be efficiently acquired, encoded and transformed into a determined high-level context, distributed and provided to requesting entities for consumption, while simultaneously maintaining its real-time properties. This is a very complex and resource-consuming task, requiring appropriate context sources, and efficient supporting context management systems with embedded intelligence, employment of interoperable and lightweight standards, fast communication channels and secured interfaces for context retrieval and distribution.

The main difficulty lies in the creation of intelligent and comprehensive smart spaces as they need to be equipped with myriads of sensors, which requires huge investments. Therefore the users' mobile terminals are considered as sensor devices in smart environments, while the WSNs penetrate limited spaces such as buildings, delimited areas and cars. Altogether, the wide level adoption will allow prices to fall considerably, increasing the feasibility of such concepts.

5. TRIALS OF NEW APPLICATIONS AND SERVICES QUASI-SMART ENVIRONMENTS

Telecom Italia is already providing some context-aware services tied to a certain location or a territory, to its customers, especially for mobile users in the eTourism field, where the B2B2C model is implemented with the following elements:

- the development of a model of integrated and aggregated offer on a local basis, or "under the patronage" of public administrations responsibility;
- the involvement of the PAs (Public Administrations) for the sponsorship of the model and the involvement of the proper actors active on the business value chain;
- the collection and delivery of content (e.g., information for the enhancement of heritage artists);
- a strong leverage on mobile devices for the end customer and on the operator's infrastructure based on assets (connectivity, data centers, Telco Capabilities) solutions;
- the hypothesis of conditional business to the establishment of a collaborative model based on a local public-private integration.

The different roles and their interactions in the eTourism scenario are shown in Figure 4 where the CA&SA Provider is responsible for integrating context information and self-adapt it into its services. To some actors in the network, this context information has an economic value. Therefore, it is assumed that actors do not give away data or services without an adequate compensation. Consequently, the analysis here has been done in terms of value exchanges between the different actors. By playing the role of context enabler, the operator limited its involvement to providing network data that could enable context reasoning. In any case, it could be an important distributor of context data in the market of CA&SA services. By playing the role of context provider, it would increase the involvement in the organization of these services and would trade context data from different sources.

There are no sensor networks involved in the described services as abovementioned, however, mobile terminals and location-aware services can be considered as sensors. Therefore, the described environment is a first step towards a field trial smart space facilitated by a telecom operator. In the presented scenario, it plays the role of network operator, service provider and seller. This combination of roles puts it in a privileged position to collect user and network context data.

The business model of the aforementioned services is based on the bundling of a customized platform/service solution to third parties interested in a promotion of certain initiatives and events at large national, regional and local scales. These entities, (e.g., City Hall, Government, Event sponsors) interested to involve and to attract, for any reason (e.g., advertisement, social visibility, tourism), as much people as possible, have now a way to target this offer within a specific and contextualized scope. A prepared ad-hoc system usually involves context management platforms, preparation and customization of the website and its content, where users can register and upload their content. Then, different interactions are offered: voting, rating, the best content, the most popular place, as well as interactive game scenarios where every participant is invited and potentially involved. These web portals leverage User Generated Content (UGC) upload and sharing capabilities, mainly based on the user location, including the proximity to objects and other people, presence status, willingness preposition, preferences, and social contexts.

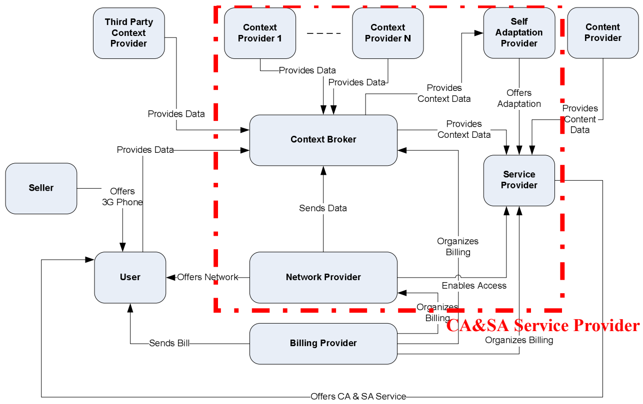

Figure 4. Different Roles and Their Interactions in a General eTourism Service Scenario.

6. CONCLUSIONS AND FUTURE WORK

In this work we have presented the business potential of context-aware smart environments, demonstrated by a technical solution implemented in the C-CAST FP 7 ICT project and employed in a major telecom company's premises for trial concepts and scenarios. We have highlighted the context management requirements of smart spaces. As seen, most use cases involve more than two entities (a smart space consists of many active entities), therefore disrupting the traditional business models. Despite the presented limitations, we have shown that small variations to well-known models can generate positive financial flows. Moreover, it became clear that the application of context-awareness as a wide spread technology for adapting services and managing personal data shall be further analyzed for potential issues regarding privacy, business models and law regulations before it can be widely adopted and deployed in smart spaces. Nevertheless, when context-awareness and context data are exposed as business enablers to third parties in a guaranteed and protected way, all entities within the value chain will be able to deliver attractive solutions. Still lacking is the definition of the right pricing structure with an appropriate revenue sharing scheme. Altogether, this work will leverage context exposure over Web2.0 interfaces and other new business propositions.

REFERENCES

[1] N. Ryan, J. Pascoe and D. Morse, “Enhanced reality fieldwork: the context-aware archaeological assistant”, Proc. of 25th Anniversary Computer Applications in Archaeology, 1997.

[2] A.K. Dey and G.D. Abowd, “Towards a better understanding of context and context-awareness”, Proc. of Work on the What, Who, Where, When and How of Context-Awareness, ACM Press, New York, 2000.

[3] R. Hull, P. Neaves and J. Bedford-Roberts, “Towards situated computing”, Proc. of the First International Symposium on Wearable Computers (ISWC ‘97), 1997, p.146.

[4] P.J. Brown, “The stick-e document: a framework for creating context-aware applications”, Proc. of Electronic Publishing, Palo Alto, 1996, pp.259–272.

[5] A. Zimmermann, “Context Management and Personalisation”, 2007, p. 260.

[6] Y. Oh et al., “A Context Management Architecture for Large-Scale Smart Environments,” IEEE Communications Magazine, March 2010.

[7] R. Casdell-Oliver, W. Liu, “Representation and recognition of Situations in Sensor Networks,” IEEE Communications Magazine, March 2010.

[8] R. Want, A. Hopper, V. Falcao and J. Gibbons, “The active badge location system”, ACM Transactions on Information Systems, 1992, vol. 10, No. 1, pp.91–102.

[9] H. Ailisto, P. Alahuhta, V. Haataja, V. Kylloenen and M. Lindholm, “Structuring context aware applications: Five-layer model & example case”, Proc. of Workshop on Concepts & Models for Ubiquitous Computing, Sweden, 2002.

[10] A.K. Dey and G.D. Abowd, “A conceptual framework and a toolkit for supporting rapid prototyping of context-aware applications”, HCI Journal, 2001, vol. 16, Nos. 2–4, pp.7–166.

[11] T. Gu, H.K. Pung and D.Q. Zhang, “A middleware for building context-aware mobile services”, Proc. of IEEE Vehicular Technology Conference (VTC), Milan, Italy, 2004.

[12] P. Fahy and S. Clarke, “CASS – a middleware for mobile context-aware applications”, Workshop on Context Awareness, MobiSys 2004, 2004.

[13] H. Chen, T. Finin and A. Joshi, “An Intelligent Broker for Context-Aware Systems”, Article, Adjunct Proc. of Ubicomp 2003, 2003.

[14] M. Roman et al., “GAIA: A Middleware Infrastructure to Enable Active Spaces,” IEEE Pervasive Computing, vol. 1, no. 4, 2002, pp. 74–83.

[15] Service Platform for Innovative Communication Environment (SPICE), FP6 EU Project, http://www.ist-spice.org.

[16] N. Salis, C. Licciardi, S. Hallsteinsen, K. Geihs, “Telecom Italia’s MUSIC Exploitation: Mobile eTourism”, D13.12, FP6 EU Project MUSIC, December 2009.

[17] Y. Oh, W. Woo, “How to build a Context-aware Architecture for Ubiquitous Virtual Reality”, Proc. of the International Symposium of Ubiquitous Virtual Reality, 2007.

[18] M. Baldauf, S. Dustdar and F. Rosenberg, “A survey on context-aware systems”, International Journal of Ad Hoc and Ubiquitous Computing, 2007, vol. 2(4), pp. 263-277.

[19] C-CAST – Provide an End-to-End Context-Aware Communication Framework, FP7 EU Project, Deliverable 6, http://www.ict-ccast.eu/files/C-Cast_D6.pdf/.

[20] R.M. Mayrhofer, “An Architecture for Context Prediction,” Dissertation, Johannes Kepler University Linz, 2004.

[21] SOFIA – Smart Objects For Intelligent Applications, FP7 EU Project, http://www.sofia-project.eu/.

INNOVATIVE TANGIBLE USER INTERFACE AS A MEAN FOR INTERACTING TELECOMMUNICATIONS SERVICES

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ABSTRACT

While modern telecommunications are becoming ever more useful and even necessary in everyday life, not all groups of people are equally capable of using them. Due to inevitable demographic changes the elderly are growing in number, yet they are not very well served by user interfaces for the various telecommunications tools. The prime target group for our proposed technology is people with cognitive and motor disabilities, whether due to age, illness or traumatic events. They require a user interface which enables them to make or redirect calls, create conferences, set forwarding and/or access different voice XML services - without the complexity of keyboards or menus with tree structures. The motivators behind this are: simplicity, accessibility, usability and efficiency – all within the scope of potential user groups and usage scenarios. The key enablers are Next Generation Network (NGN) open interfaces and Near Field Communication (NFC) technology as a part of Radio Frequency Identification (RFID) family.

Keywords— Tangible User Interface, Next Generation Network, Near Field Communication, People with Special Needs, Interactive Services, Internet of Things.

1. INTRODUCTION

NFC technology connects physical tagged objects with the existing digital systems and networks (e.g. Internet of Things). This feature brings countless new possibilities to the world of telecommunications. On the other hand NGN network offers open interfaces to developers via communities [1]. Those interfaces could be used to access different functionalities supported by the network. Normally we talk about high level interfaces based on technologies such as Simple Object Access Protocol (SOAP) or Representational State Transfer (REST). The first one represents a standardized way of encoding and decoding remote procedure calls, while the second is not strictly defined and is often used by web developers.
The before mentioned interfaces are provided by an additional network element, which performs the role of a gateway and maps complex telecommunications protocols to user-friendly technologies. In the context of our research and development work we have developed our own gateway that uses Parlay X Web specification of telecommunications services at its front side (Figure 1).

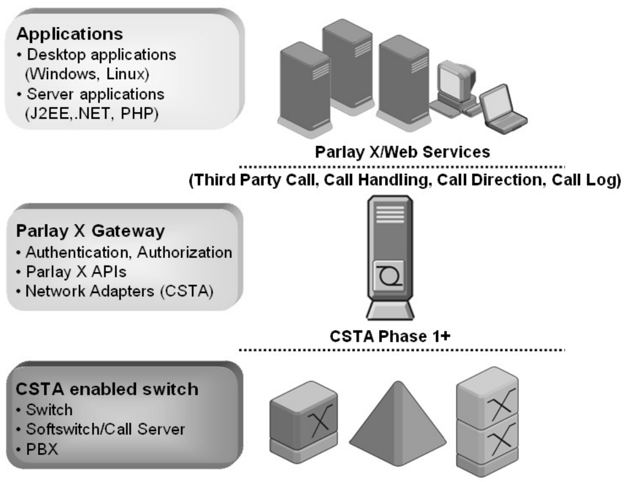

Figure 1: CSTA / Parlay X Gateway [2]

Amongst the sustainable advantages of the presented solution is the unique fusion of modern packet-based telecommunications services with state-of-the art tangible interaction modality. This enables our solution to bridge the gap between pure information & communication services and the physical object manipulations in the real world. This presents a basis for enriching and enhancing user experience, proving crucial in numerous environments where people with disabilities interact with information and communication technologies.

2. PARLAY X OPEN INTERFACES

Parlay X is a set of abstract APIs/interfaces that expose communication capabilities to the applications via SOAP Web Services [3]. The Parlay X Gateway provides application developers with Parlay X APIs for managing call related features on switches and soft switches. Using these APIs, developers can enrich the applications with call related features [4].
SOAP Web Services provide a simple but powerful means for one software component to invoke action on another via the use of message interactions. The specification for SOAP is a world-wide standard, administered by the World Wide Web Consortium (W3C). All of the major software

platform vendors have agreed to implement the specification, which means that a software component installed on one type of platform can communicate with another component on any other type of platform.
The interface for communication is described in Web Service Description Language (WSDL). The communication itself is performed by exchanging SOAP messages that are based on the Extensible Markup Language (XML) format. The messages are usually transferred over Hypertext Transfer Protocol (HTTP) although other underlying protocols can be used as well. The Parlay X APIs that we have implemented are shortly described in the table below (Table 1) [5]:

Table 1: Implemented Parlay X APIs

Parlay X API	**Functionality**
Third Party Call [6]	Creating and managing a call initiated by an application.
Call Notification [7]	Handling calls initiated by a subscriber in the network. Notifying applications about calls and enabling them to further impact calls (routing of a call).
Call Handling [8]	Specifying how calls are to be handled for a specific number; providing a variety of features e.g. call forwarding, white/black list handling etc.
Call Log	Getting Call Log (dialed / received / missed).

By using the APIs in the table above one can develop innovative applications that trigger and manage calls, build conferences, set forwarding access call log etc. [9]. NFC technology plays the role of the service trigger since many listed functionalities could be accessed only by a touch between a mobile NFC device and a tagged object.

3. NEAR FIELD COMMUNICATION

Near field communication brings radio frequency identification technology to mobile phones and some other devices that people use almost every day (e.g. communication, payment and ticketing devices) [10]. Technology consists of two hardware parts – reader and tags. Reader communicates with tag over the air (electromagnetic induction with central frequency 13.56 MHz) and reads the information written on the tag's chip or writes some data on it. Communication distance is usually very low (it depends on the size of reader's and tag's antenna) – approx. 2-5 cm. Technology is robust and as such very convenient to use, also in harsh environments. NFC tags are passive tags, which means they are not battery assisted and could be very thin and small. As they are usually produced in the form of labels, they could be easily installed on the object's surface.

This technology can be ideal for any integration of the physical world (objects that are marked using the NFC tag) and the existing digital systems. Such principles are known under the term of "Internet of Things". The idea behind this is very simple - if all objects of daily life, from bread to an airplane, are equipped with RFID tags, they can be identified and managed by computers, like they can be by humans.
Today NFC technology is adopted already by many mobile device vendors (limited to some selected device models). NFC is also being standardized by the NFC Forum - a non-profit industry association.

4. APPLICATION DESCRIPTION

The prime target user group for our solution is people with cognitive and motor disabilities (people with special needs), due to age, illness or traumatic events. They require a user interface which enables them to make or redirect calls, create conferences and access different voice XML services (e.g. e-Books, e-News) without the complexity of keyboards or menus with tree structures. The main idea of our solution is to simplify the usage of certain telecommunications services with the usage of Tangible User Interface (TUI) [11]. In the first step we have moved some of the contacts from the digital phone book (mobile phone) to the personal contacts pin board. Each person is represented by his/her photo which is accompanied by the NFC tag that holds the telephone number of the person (Figure 2).

Figure 2: Personal contacts pin board – phone book

Within the NFC mobile device we have installed our own application which is launched by a touch of the NFC tag that is attached to the photo. The application sends the SOAP request (using Wi-Fi or one of the 3G access technologies) to the application server that sends the appropriate HTTP Parlay X request to the Parlay X gateway. Possible operations are the following:

- Initiating a call between application's user and the person selected;
- Adding selected person to the conference;
- Redirecting incoming calls to selected person;

- Accessing the list of missed calls;

Before the first operation, a user must enter some settings inside the application in order to enable it to communicate with the application server – Internet address (IP) of the application server, username and secret for authentication purposes and user's calling phone number (mobile number or local phone number). The solution is comprised of two software parts:

- NFC application on the mobile device;
- Application on the application server (Java HTTP Servlet);

The main task of NFC device application is sending HTTP/SOAP requests to the server-side application. Server application is a web service with two methods:

- *String NFCevent(String calling_party, String tag_value)*,
- *String CNchanged(String calling_party).*

The *NFCevent* method contains two parameters – *calling_party* and *tag_value*. *calling_party* parameter indicates the calling device's phone number. *tag_value* is the value written inside the NFC tag. There are two possibilities: it could be the value of the telephone number of the called party's device, or the character string "MissedCalls", which tells the server side application that the user wishes to receive a list of missed calls. Another method, *CNchanged*, contains a single parameter - *calling_party*. When the user enters information about the calling number within the application (using settings window), this method is called. Consequently the gateway starts monitoring selected number. Next figure presents the overall architecture of the system (Figure 3).

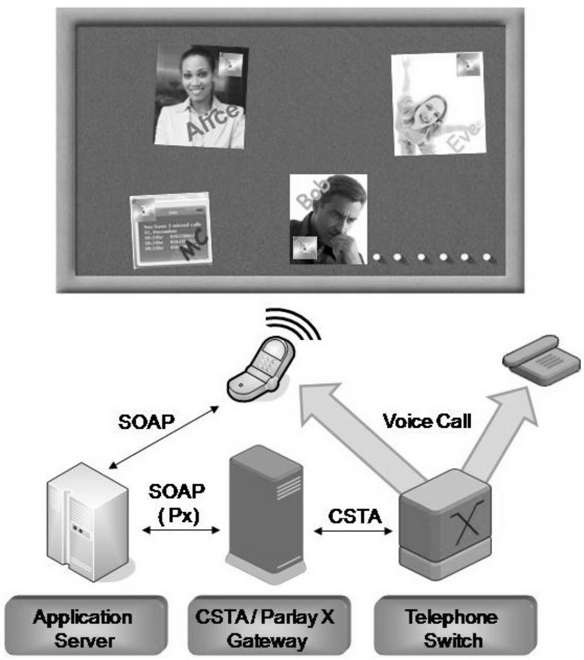

Figure 3: Complete solution architecture

The Application Server is the main executor of the entire logic. It receives requests by the NFC enabled devices and talks to the Parlay X gateway through the SOAP interface. The application server also monitors the status of calls (for the calling party number) on a telephone switch. This is how the application server knows about call events and consequently how to properly react on any request made by the NFC device. The following section describes the operation process of the whole system.

5. PROCESS DESCRIPTION

In the first step the user of the client side NFC application enters his or her user's data (IP address of the application server, credentials and phone number). The application provides the information about the user's phone number to the application server using the remote method *CNchanged*. The application server sends a subscription to the Parlay X gateway (*startCallDirectionNotification*) to receive call notifications about a selected phone number. This is how the server knows when the user is being called (*handleCalledNumber*). If the application server receives a request (*NFCevent*) from the NFC application when the user is being called, it recognizes that it must instruct the gateway to perform call redirection. This is how the user can divert incoming calls to a selected person in real time (*handleCalledNumberResponse*) by simply touching the person's photo using his or her NFC mobile device. The next figure shows the entire message flow (Figure 4).

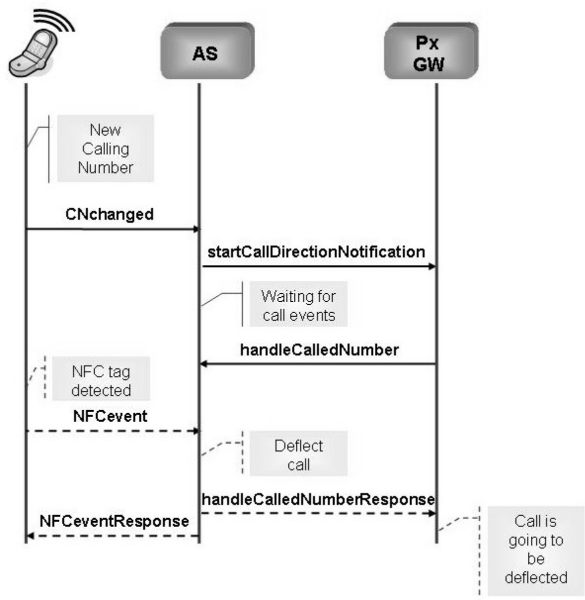

Figure 4: Call deflection

When a user is not in a call and the application server receives a new *NFCevent* request by the NFC application, the Parlay X gateway is instructed to make a third party call by sending a makeCall request. The application server receives a *makeCallResponse* by the gateway with the identifier of the call. This identifier could then be used to obtain additonal information about a call (call state) by using the *getCallInformation* method (Figure 5).

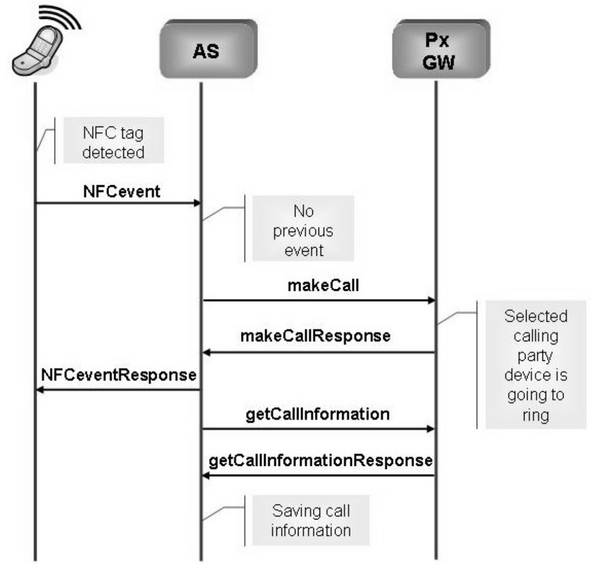

Figure 5: Make third party call

If the server receives a new request *NFCevent* by the user who previously created a third party call, then it is aware that the user wants to upgrade an existing call to the conference. Therefore, a server sends a new request (*addParticipant*) to the gateway and another user is added to the existing conversation (Figure 6).

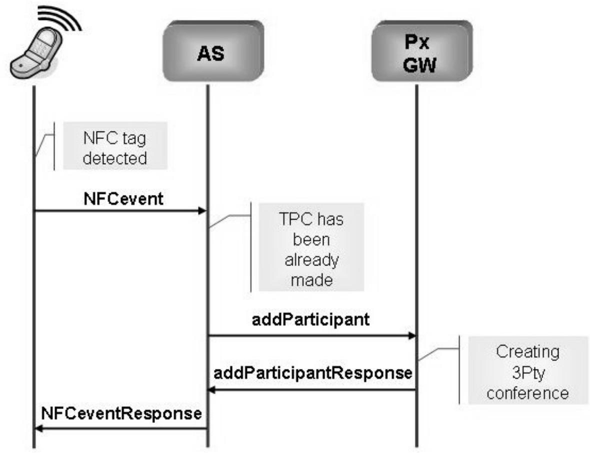

Figure 6: Upgrading third party call to the conference

In all previously described cases the value of the parameter *tag_value* inside the method *NFCevent* was the telephone number. On the other hand, the value could also be a reserved string "MissedCalls". In this case, the application server knows that the user is requesting a list of missed calls for a selected telephone number. For this reason the server sends the request *getCallLog* towards the Parlay X gateway, which then returns a list. Information is then sent to the NFC application and appears on the screen of the mobile phone. The application could also check for those numbers in the device's local phone book and connect them with names. This functionality is very interesting when the user controls an old stationary telephone device, which does not have the option to print out missed calls (Figure 7).

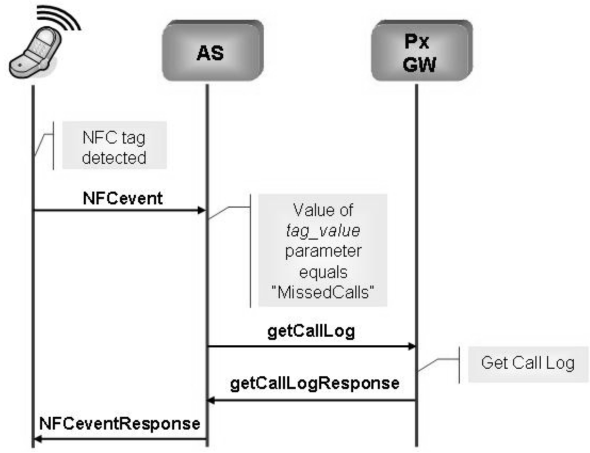

Figure 7: Get missed calls information

6. USAGE SCENARIO

The elderly and people with special needs may have great difficulties using communication devices. The problem is usually the fact that many (especially mobile) devices support various functionalities. For some people they are very welcome, while others may have huge problems accessing even basic features - like browsing through a phone book.

On the other hand, some people still use their old stationary phone, which could also be controlled by using the proposed solution. In both cases NFC technology simplifies the use of mobile/fixed telephone devices for people who need such help. It is possible to access some popular telecommunications services by a simple touch of the phone device with NFC tags.

In our scenario, we can have a grandmother (Mary), who has put photos of her relatives and friends on her kitchen wall. An NFC tag that contains the person's phone number is attached on the back of each photo. Mary can call her son (Bob) by simply touching his image with her NFC enabled device. The application is triggered and her telephone rings (mobile or stationary). Mary accepts a call initiated by the local switch and waits for Bob to answer. During the conversation with her son, Mary also wants to talk with her daughter (Alice) to invite them both to lunch. That is why she decides to create a conference. She can do so by a simple touch of Alice's photo with her NFC device.

Sometimes Mary is called by an agent, who offers her new insurance packages for her house. Mary doesn't want to talk with him and therefore redirects such calls directly to Bob. She does it by touching Bob's photo using her NFC device.

When Mary comes home from the market she wants to check her missed calls. The screen on her old telephone device is too small to properly display them. Therefore, she uses her NFC enabled mobile device by touching a special

tag, which carries a "MissedCalls" string. All her missed calls are then listed on her mobile phone screen.
Mary can access various e-contents by selecting the appropriate image on her board. For example, when she touches an image of the New York Times, she can listen to the latest news and weather report directly from the voice XML machine.
Last, but not the least, the proposed system could also implement the functionality of composing and sending short text messages using Short Message Service (SMS) within Tangible User Interface.

7. FUTURE WORK

A hot topic for future work is semantics – especially solving questions as: how to semantically connect various physical objects (e.g. photos, images, pictograms) inside a physical frame. We will also make an empirical evaluation of user experiences on a selected number of individuals inside target user groups. The results will help us improve our solution regarding semantics and service selection.

8. CONCLUSION

In our research work we present the innovative use of NFC technology together with modern telecommunications networks. NFC technology plays an important role when we try to combine the physical (human) world with the complex digital world of telecommunications and build user-friendly tangible user interface.
Despite the fact that the elderly are growing in numbers, there is still a lack of adjusted telecommunications devices solutions. The paper aims to present an innovative solution to how users can interact or control some modern voice services by means of exploiting RFID/NFC technology available on a NFC mobile device.
The visual part of the user interface is comprised of selected objects (e.g. photos, images, pictograms), which are semantically connected inside a physical frame (e.g. pin board). Simply by interacting those visual objects, end users are able to initiate and control different services (communications, voice XML services etc.) which are otherwise difficult to use for them. The main goal of the presented solution is to achieve a great impact on people with special needs using communications services.

REFERENCES

[1] ITU-T, Recommendation Y.2011: General principles and general reference model for next generation networks, 2004.

[2] B. Imperl, Functional Specification of Product AS6112AX, Iskratel, 2007.

[3] U. Sedlar, L. Zebec, J. Bešter, A. Kos, "Bringing Click-to-Dial Functionality to IPTV Users", *IEEE Comm. Mag.,* vol.46 no.3, pp. 118-125, 2008.

[4] N. Blum, T. Magedanz: "The Importance of a Service Oriented Approach: Open Interfaces, Network Abstraction and Service Brokers", *Informa 5th Annual Global SDP Summit*, London, 2009.

[5] L. Zebec. I. Humar, D. Bodnaruk, A. Kos, J. Bešter, "NGN service development - overview and Parlay X implementation", *Elektroteh. vestn.*, vol.72, no.1, pp. 45-51, 2005.

[6] ETSI Standard. ETSI ES 202 391-2 V1.2.1 Open Service Access (OSA), Parlay X Web Services, Part 2: Third Party Call (Parlay X 2), 2008.

[7] ETSI Standard. ETSI ES 202 391-3 V1.2.1 Open Service Access (OSA), Parlay X Web Services, Part 2: Call Notification (Parlay X 2), 2008.

[8] ETSI Standard. ETSI ES 202 391-10 V1.2.1 Open Service Access (OSA), Parlay X Web Services, Part 2: Call Handling (Parlay X 2), 2008.

[9] E. Mikóczy, P. Podhradsky, B. Zovko-Cihlar, "Evolution of Services and Applications in Environments of the Converged Networks and NGN", *13th International Conference on Systems, Signals and Image Processing*, 2006.

[10] K. Finkenzeller, "RFID Handbook: Fundamentals and Applications in Contactless Smart Cards and Identification 2nd Edition", Wiley, 2004.

[11] H. Ishii, "The tangible user interface and its evolution", *Communications of the ACM*, vol.51, no.6, 2008.

SESSION 6

REGULATION, STANDARDIZATION AND STAKEHOLDER PARTICIPATION

HOW MANY STANDARDS IN A LAPTOP? (AND OTHER EMPIRICAL QUESTIONS)

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ABSTRACT

An empirical study which identifies 251 technical interoperability standards implemented in a modern laptop computer, and estimates that the total number of standards relevant to such a device is much higher. Of the identified standards, the authors find that 44% were developed by consortia, 36% by formal standards development organizations, and 20% by single companies. The intellectual property rights policies associated with 197 of the standards are assessed: 75% were developed under "RAND" terms, 22% under "royalty free" terms, and 3% utilize a patent pool. The authors make certain observations based on their findings, and identify promising areas for future research.

Keywords— standards, SDOs, consortia, intellectual property rights (IPR), RAND, royalty free

1. OVERVIEW

Our effort began with some simple questions: how many standards are embodied in a modern laptop computer? How many of these standards are developed by formal standards development organizations and how many by consortia? What type of intellectual property rights policies – e.g. "RAND" or "royalty-free" – apply to each of these standards?

Answering these seemingly-simple questions proved dauntingly complex. Nonetheless, subject to the limitations and qualifications described in this paper, we were able to reach the following conclusions:

- We identified 251 interoperability standards that are embodied or directly utilized in a modern laptop computer. We focused only on standards that facilitate technical interoperability, and did not count quality, safety, performance, measurement, environmental, accessibility, design process, manufacturing process or electromagnetic compatibility standards. Further, our count of interoperability standards is not comprehensive: we have become aware of significant omissions. Accordingly, we believe our count sets only a floor: a modern laptop embodies or utilizes *at least* 251 interoperability standards, but the actual number is certainly much higher (the authors would be unsurprised by a total number of 500 or more). Including other types of relevant standards, such as environmental or safety standards, in addition to interoperability standards would further raise the count dramatically.
- Of the 251 standards we identified, 112 (44%) were developed by consortia, 90 (36%) by formal standards development organizations, and 49 (20%) by individual companies (see Figure 1).
- We were able to allocate 197 of the 251 standards into one of three broad intellectual property model categories: RAND, RF or patent pool (we lacked sufficient information to categorize the remaining 54 standards). Of the 197 we categorized, 148 (75%) were RAND, 43 (22%) were RF, and 6 (3%) utilized a patent pool (see Figure 2).

In order to meaningfully assess our data it is imperative that readers understand our terminology and our methodology. These are described in Section 2, below. Section 3 highlights some limitations of our approach, and identifies some gaps in our research. Section 4 explores some preliminary observations and conclusions based on our data. Section 5 identifies opportunities for further research. Finally, the appendix contains a table listing the particular standards we identified and the values we assigned to each.

2. TERMINOLOGY AND METHODOLOGY

We began by examining the specifications of various current-generation laptop computers produced by different manufacturers, and developing a vision of a composite, hypothetical laptop that drew from the features of each. We also gave ourselves some flexibility to include a few features that are widely expected to be included in laptops in the imminent future (e.g. hi-definition wireless display capabilities).

Next, we created a set of broad categories – display, graphics, sound, storage, BIOS, input device, processor, power, file system, networking, wireless, I/O ports, memory, software, codecs, content protection, security and "other" – and sought relevant standards. Using a variety of methods, including interviews with experts and extensive primary research, we identified standards in each category

Brad Biddle is an Adjunct Professor at ASU and Standards Counsel for Intel Corporation; Andrew White and Sean Woods are law students. This article reflects the authors' personal views. The authors thank Steve Balogh, Carl Cargill, Wayne Carr, Kevin Cornelius, Bob Grow, Earl Nied, Ken Salzberg, Kim Turner, Steve Whalley and Andrew Wilson for their invaluable input. Errors are solely the responsibility of the authors.

that would be embodied in or directly utilized by our hypothetical laptop computer.

For our purposes, "standards" included not just standards developed by formal standards development organizations like ISO, but also industry specifications developed by consortia like PCI-SIG. We also encountered a number of specifications intentionally promulgated by a single company for broader industry adoption, and we counted these as "standards" as well. We limited our count of company-promulgated standards to those that a company intentionally and specifically made available for adoption as an industry specification; we did not include proprietary technologies that have significant market share but that are not otherwise intentionally made available for industry adoption.

As noted in the introduction, we focused only on standards that facilitate technical interoperability, and did not count quality, safety, performance, measurement, environmental, accessibility, design process, manufacturing process or electromagnetic compatibility standards.

We identified the developer/promoter of each standard as either (a) a formal standards development organization or "SDO," (b) a consortium, or (c) an individual company. For this step we utilized the taxonomy suggested by the IPO Standards Setting Committee in their 2009 "Standards Primer" document. [1] We counted as SDOs: (a) the "Big I" international standards organizations (ITU, ISO, IEC), (b) the "Little I" international organizations (IEEE, ASTM), (c) government-sanctioned regional bodies such as ETSI, (d) government-sanctioned national bodies, such as BSI, and (e) organizations sanctioned or accredited by a national body, such as all of the ANSI-accredited organizations (e.g., JEDEC, TIA). All other group-focused specification-development efforts were classified as "Consortia." The consortia category contains a wide variety of different groups, ranging from formal organizations like the W3C to very informal open source development efforts. We called specifications created by single commercial entity "Company" standards.

Assessing the intellectual property rights (IPR) policies associated with each standard proved difficult. Many IPR policies were extraordinarily complex. Further, IPR policies for some organizations were not publicly available, leaving us to rely on second-hand accounts or draw inferences. Noting some risk of oversimplification or error, we allocated each standard to one of four broad categories:

- *RAND.* This category included standards that were developed under RAND or F/RAND terms – (fair,) reasonable and non-discriminatory patent license commitments, without precluding the option of patent owners collecting patent royalties for essential patent claims. If a SDO or Consortia permitted a RAND option, even if it contemplated other options as well, we included it in the RAND category. We note that the fact that an IPR policy *permits* collection of royalties does not mean that parties *in fact* collect royalties. (IETF provides an example: the IETF IPR policy permits RAND, and thus we categorized all IETF standards as RAND, but in practice parties attempt to collect royalties on few, if any, IETF standards.)
- *RF.* This category included standards that were created under terms that prohibit the participating companies or individuals from collecting patent royalties for essential patent claims (usually subject to important limitations). For our purposes, IPR models such as "RF-RAND" (royalty-free RAND) and "RAND-Zero" (RAND with zero royalties) fall into this category. We also included standards with IPR policies that rely on promises not to assert essential patent claims here. Note that our designation of a standard as RF does not mean that the standard is in fact royalty free to implement, as entities not bound by the IPR policy could assert patents, for example.
- *Patent pool.* The term "patent pool" is sometimes defined in a way that would sweep in virtually any RAND or RF IPR policy, but for our purposes we adopted a narrow definition. We focused on the scenario where a specification is made available subject to execution of a license agreement, and that license agreement conveys a license to patents pooled by multiple parties. The DVD specifications provide an example.
- *NA ("not available").* In 54 of our 251 cases we simply could not determine the intellectual property policy associated with a particular standard. Figure 2 below includes only the 197 bodies that we were able to categorize.

We should emphasize that our taxonomy glosses over a great deal of complexity, including the key issue of whether the RAND or RF promise extends from participants in the standards development process to all implementers or only to those implementers that join the relevant consortia or SDO. For our categorization purposes, either approach sufficed: e.g., if a group required that participants promise to license on RAND terms only to members of that group, with no other license obligation, we counted that group as RAND.

3. LIMITATIONS AND GAPS

Our hypothetical/composite laptop approach potentially allows some ambiguity or duplication. For example, we include file systems standards for both Linux and Windows computers, even though in many cases they would not co-exist in a single machine. Likewise, we include wireless display standards that might be competitive rather than co-existing. Focusing on a single, specific "real world" machine would have mitigated this risk. However, our composite approach enabled us to avoid singling out a specific vendor, and enabled us to anticipate soon-to-be implemented standards.

A related point: while our primary focus was on standards that would be fully implemented in our hypothetical laptop, we also included some standards that would be directly *used by* our hypothetical machine, but that are not necessarily fully implemented on the client side (e.g., basic Internet standards like IPv4, DNS or TCP). This involved some judgment calls and line drawing. Similarly, we included standards related to some basic software applications (e.g., OpenXML), but tried to avoid expanding too far "up the stack" into the software application world.

Another issue: our data are imperfect. The authors bring legal expertise to the table rather than deep technical expertise. Understanding each of our various technical focus areas – display, graphics, sound, storage, BIOS, input device, processor, power, file system, networking, wireless, I/O ports, memory, software, codecs, content protection, security – sufficiently to assess the relevant standards in each area proved challenging. We suspect there are errors of both under-inclusion and over-inclusion in some of our focus areas. Further, we have realized that our focus areas may have been too narrow. For example, battery technologies, biometrics, camera hardware, solid state drives and docking systems standards are currently underrepresented in our list. We will continue to refine and improve the quality of our data set. However, we do not believe that this refinement will dramatically change our observations or conclusions.

4. OBSERVATIONS

The focus of this stage of our effort has been primarily on collecting empirical data rather than interpreting it. However, a set of fairly obvious conclusions are immediately apparent:

- *The critical role of standards in ICT.* The fact that a modern laptop computer implements or relies on over 250 (and probably closer to 500, we estimate) interoperability standards is remarkable. While certainly no one doubted the importance of standards to the information and communications technology (ICT) industry in the absence of this data, quantifying the volume of standards embodied in a common ICT device is striking. We believe that as technological convergence continues, and ICT devices increasing include elements from the computing, telephony and consumer electronics sectors, the number of relevant standards will only increase.
- *The importance of consortia for ICT standards development.* Of the 251 standards we indentified, only about one-third were developed by formal SDOs. Consortia developed 44%, and single companies developed 20% (see Figure 1). We suspect the dominant role played by the private sector in at least this aspect of ICT standardization will come as a surprise to some policymakers and other standardization stakeholders.

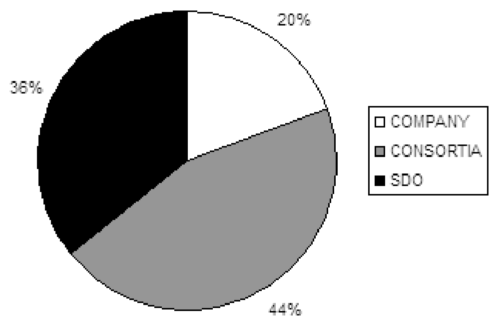

Figure 1: TYPE OF STANDARD/SPEC DEVELOPER

- *The preponderance of RAND as IPR model.* The merits of RAND and RF IPR models are fiercely debated by their respective proponents. Our data suggests that historically RAND has been effective in the computing sector, if measured by implementation of associated standards: we see that 75% of the standards we examined were developed under RAND terms (see Figure 2). Conceivably the financial industry axiom that "past performance is not indicative of future results" may be applicable, given the emergence of open source, increasing patent litigiousness, or other factors. Further, the practical impact of RAND policies seems to be different in different contexts (e.g., IETF standards, while nominally RAND, appear to be largely RF in practice; other RAND standards, such as the IEEE's 802.11 standards, are the subject of licensing and patent litigation). Nonetheless, the strong dominance of RAND in our set of successful (i.e., implemented) standards is notable. Our data also suggest that patent pools, to date, have not played a significant role for at least the computing sector of the ICT industry.

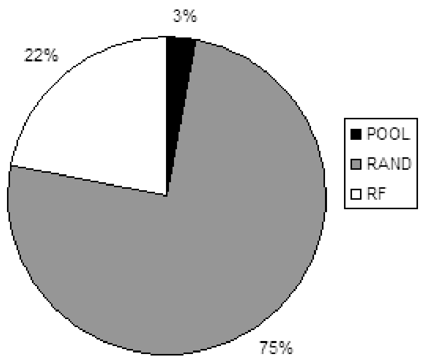

Figure 2: IP MODELS

5. NEXT STEPS AND CONCLUSION

As noted in Section 3 above, a key next step for us is to expand our data set and refine our data. We welcome constructive input and will happily make our spreadsheets available to interested parties.

While we utilized a single RAND category, we have noted that in the ICT environment there seems to be two broad subcategories of RAND standards: those for which the de facto reality seems to be a RF environment, and those for which there are active royalty-collection efforts. We believe that it would be interesting to attempt to count the number of standards in each subcategory.

Another promising focus area for additional empirical analysis is an assessment of consortia. For our purposes consortia occupied a single category, but in fact we saw a bewildering variety of approaches among consortia in the course of our research. Identifying different types of consortia, and analyzing the implementation of standards produced by different types, strikes us as a fascinating research question.

Additionally, assessing each of our identified standards against various criteria of "openness," along the lines of Per Anderson's recent study [2], could prove quite interesting. Our working theory is that the development and distribution processes associated with a significant percentage of the successfully-implemented standards we identified would *not* meet typical definitions of openness, transparency or consensus decision-making. If true, this would be an interesting data point to bring into, e.g., current policy debates over "good practices" for consortia, such as the current BSI PAS 98 effort. Further, it would be interesting to consider whether the empirical data could demonstrate either a positive or negative correlation between the "openness" of a standards development effort and its effectiveness as measured by widespread implementation of that standard in the commercial marketplace.

* * *

The academic literature on standardization often bemoans the dearth of empirical analysis of standards. Our hope is that the analysis documented in this paper helps to fill this gap, and enables policymakers, academics, commercial stakeholders and others to better understand ICT standards and industry specifications.

REFERENCES

[1] IPO Standards Setting Committee, *Standards Primer: An Overview of Standards Setting Bodies and Patent-Related Issues that Arise in the Context of Standards Setting Activities*, 2009 IPO Articles & Reps., Pats. Sec. No. 16 (10/09/2009). Available to IPO members via http://ipo.org; excerpts publicly available at <http://standardslaw.org/seminar/class-2/excerpts-from-ipo-standards-primer/>.

[2] Per Anderson, *Evaluation of Ten Standard Setting Organizations With Regard to Open Standards*, IDC Study commissioned by the Danish National IT and Telecom Agency, January 2008. Available at <http://www.itst.dk/it-arkitektur-og-standarder/standardisering/abne-standarder/baggrundsrapporter/Evaluation%20of%20Ten%20Standard%20Setting%20Organizations.pdf>.

APPENDIX:
LIST OF STANDARDS/SPECIFICATIONS

Name of standard/specification	Developer	Developer type	IP type
.NET	Microsoft	COMPANY	NA
16x9 Notebook Panel ver. 1a	VESA	CONSORTIA	RAND
3GP	3GPP	SDO	RAND
8P8C/"RJ-45" IEC 60603	IEC	SDO	RAND
AC'97 v2.3	Intel	COMPANY	NA
ACS-2 [ATA/ATAPI Command Set 2]	T13 INCITS	SDO	RAND
Advanced Configuration and Power Interface Spec 3.0	ACPI	CONSORTIA	RAND
Advanced eXpress I/O Module [AXIOM]	ATI	COMPANY	NA
AES (U.S. FIPS PUB 197)	NIST	SDO	NA
AGP	Intel	COMPANY	NA
AIFF	Apple	COMPANY	NA
ALC889	RealTek	COMPANY	NA
Allegro 4.9.19	open source project	CONSORTIA	RF
ANSI INCITS 207-1991[R2007]	ANSI	SDO	RAND
ANSI INCITS 346-2001[r2006]	ANSI	SDO	RAND
ANSI INCITS 407-2005	ANSI	SDO	RAND
ANSI INCITS 417-2006	ANSI	SDO	RAND
APM	Microsoft	COMPANY	NA
ASF	Microsoft	COMPANY	NA
Atom	IETF	CONSORTIA	RAND
AVI	Microsoft	COMPANY	NA
Bluetooth spec.	Bluetooth Sig	CONSORTIA	RF
Blu-ray Disc Read-Only Format ver. 1	Blu-ray Disc Association	CONSORTIA	POOL
Blu-ray Disc Recordable Format ver. 1	Blu-ray Disc Association	CONSORTIA	POOL
Blu-ray Disc Rewrittable Fromat ver. 2	Blu-ray Disc Association	CONSORTIA	POOL
C	ANSI/ISO	SDO	RAND
C++ (ISO/IEC 14882:2003)	ISO/IEC	SDO	RAND
CD audio ("Red book") - IEC 60908	IEC	SDO	RAND
CDROM	ISO/IEC	SDO	RAND
CIM [Common Information Model] 2.250	DMTF	CONSORTIA	RAND
Cinepak	SuperMac Technologies	COMPANY	NA
COLLADA 1.5	Khronos	CONSORTIA	RF
Compact Flash	Compact Flash Ass.	CONSORTIA	RAND
CSS (Cascading Style Sheet)	W3C	CONSORTIA	RF
CSS (Content Scramble System)	DVD Forum	CONSORTIA	NA
DDR3	JEDEC	SDO	RAND
Dirac	BBC Research	COMPANY	RF
Direct Drive Monitor [DDM] v1	VESA	CONSORTIA	RAND
Direct3D 11	Microsoft	COMPANY	NA
DirectCompute API	Microsoft	COMPANY	NA
DirectX	Microsoft	COMPANY	NA
Display Identification Data [DisplayID] Structure v1.1	VESA	CONSORTIA	RAND
Display Port Panel Connector	VESA	CONSORTIA	RAND
Display Subsystem Power Management	VESA	CONSORTIA	RAND
DisplayPort Interoperability Guidline v1.1	VESA	CONSORTIA	RAND
DLNA	Digital Living Network Alliance	CONSORTIA	RAND
DMI2 [Direct Media Interface]	Intel	COMPANY	NA
DNS	IETF	CONSORTIA	RAND
DOM	W3C	CONSORTIA	RF
DVB-H/EN 302 304	DVB/ETSI	SDO	RAND
DVD Multi	DVD Forum	CONSORTIA	POOL
DVI	DDWG	CONSORTIA	RAND
DVI 1.0 Spec	Digital Display Working Group	CONSORTIA	NA
ECMA 262 3rd edition	ECMA	CONSORTIA	RAND
ECMA C#	ECMA	CONSORTIA	RAND
ECMA CLR	ECMA	CONSORTIA	RAND
ECMA-378	ECMA	SDO	RAND
ECMA-384	ECMA	SDO	RAND
EDD-4 [Enhanced Disk Drive - 4	T13 INCITS	SDO	RAND
EHCI	Intel	COMPANY	NA
Embedded DisplayPort Standard (eDP)	VESA	CONSORTIA	RAND
Ethernet [802.3]	IEEE	SDO	RAND
EXT4	open source	CONSORTIA	RF
Fat16	ECMA	SDO	RAND
Fat32	Microsoft [Open Spec]	COMPANY	NA
Firewire/1394	IEEE	SDO	RAND
Flash (FLV, F4V)	Adobe	COMPANY	NA
FMOD	Firelight Technologies	COMPANY	NA
FTP	IETF	CONSORTIA	RAND
Guidline for transmission and control for DVD-video/audio through IEEE1394 Bus	DVD Forum	CONSORTIA	POOL
Guidline for Transmission and Control for DVD-video/audio through Most Bus	DVD Forum	CONSORTIA	POOL
H.263	ITU-T	SDO	RAND
H.264	ITU-T/ISO/IEC JVT	SDO	RAND
HDCP	DCP	COMPANY	NA
HDMI	HDMI	CONSORTIA	RAND
HFS	Apple	COMPANY	NA

HFS+	Apple	COMPANY	NA
HTML5	W3C	CONSORTIA	RF
HTTP	W3C	CONSORTIA	RF
HTTPS	W3C	CONSORTIA	RF
HuffYUV	Rudiak-Gould	COMPANY	NA
IEC 60320	IEC	SDO	RAND
IEC 60958 type II (S/PIF)	IEC	SDO	RAND
IEEE std. 1212.1-1993	IEEE	SDO	RAND
IEEE std. 1680.1-2009	IEEE	SDO	RAND
IETF RFC 5545 iCalendar	IETF	CONSORTIA	RAND
IMAP	IETF	CONSORTIA	RAND
INCITS 370-2004(1510D): ATA Host Adapter Standards	T13 INCITS	SDO	RAND
INCITS 437-2008	ISO	SDO	RAND
INCITS 452-2008(D1699): AT Attachment 8 ATA/ATAPI Command Set	T13 INCITS	SDO	RAND
Intel 64 architecture x2APIC Spec	Intel	COMPANY	NA
Intel AHCI	Intel	COMPANY	NA
Intel High Definition Audio	Intel	COMPANY	NA
Intel Platform Innovation Framework for UEFI	Intel	COMPANY	RAND
IPSEC	IETF	CONSORTIA	RAND
IPv4	IETF	CONSORTIA	RAND
ISO 8601 is dates and time	ISO/IEC	SDO	RAND
ISO 9241-300:2008	ISO	SDO	RAND
ISO 9241-302:2008	ISO	SDO	RAND
ISO 9241-303:2008	ISO	SDO	RAND
ISO 9241-304:2008	ISO	SDO	RAND
ISO 9241-305:2008	ISO	SDO	RAND
ISO 9241-306:2008	ISO	SDO	RAND
ISO 9241-307:2008	ISO	SDO	RAND
ISO 9241-400:2007	ISO	SDO	RAND
ISO 9241-400:2007	ISO/IEC	SDO	RAND
ISO 9241-410:2008	ISO	SDO	RAND
ISO 9241-410:2008	ISO/IEC	SDO	RAND
ISO/IEC 1064 is Unicode (and uft-8, utf-16)	ISO/IEC	SDO	RAND
ISO/IEC 11002:2008	ISO	SDO	RAND
ISO/IEC 11989:2010	ISO	SDO	RAND
ISO/IEC 13170:2009	ISO	SDO	RAND
ISO/IEC 14772-2:2004	ISO	SDO	RAND
ISO/IEC 14776-150:2004	ISO	SDO	RAND
ISO/IEC 15412:1999	ISO	SDO	RAND
ISO/IEC 15948:2004	ISO	SDO	RAND
ISO/IEC 19774:2006	ISO	SDO	RAND
ISO/IEC 19775-1:2008	ISO	SDO	RAND
ISO/IEC 19775-2:2004	ISO	SDO	RAND
ISO/IEC 19776-1:2008	ISO	SDO	RAND
ISO/IEC 19776-2:2008	ISO	SDO	RAND
ISO/IEC 19776-3:2007	ISO	SDO	RAND
ISO/IEC 19777-1:2006	ISO	SDO	RAND
ISO/IEC 19777-2:2006	ISO	SDO	RAND
ISO/IEC 24739-1:2009	ISO	SDO	RAND
ISO/IEC 24739-2:2009	ISO	SDO	RAND
ISO/IEC 24739-3:2009	ISO	SDO	RAND
ISO/IEC 24757:2008	ISO	SDO	RAND
ISO/IEC 26300:2006 Open Document Format	ISO/IEC	SDO	RAND
ISO/IEC 29121:2009	ISO	SDO	RAND
ISO/IEC 29171:2009	ISO	SDO	RAND
ISO/IEC 29171:2009 [iVDR spec]	ISO	SDO	RAND
ISO/IEC 29500 Office Open XML	ISO/IEC	SDO	RAND
ISO/IEC 9995-1:2009	ISO	SDO	RAND
ISO/IEC 9995-2:2009	ISO	SDO	RAND
ISO/IEC 9995-3:202	ISO	SDO	RAND
ISO/IEC 9995-4:2009	ISO	SDO	RAND
ISO/IEC 9995-5:2009	ISO	SDO	RAND
ISO/IEC 9995-7:2009	ISO	SDO	RAND
ISO/IEC 9995-8:2009	ISO	SDO	RAND
ISO/IEC TR 24784:2009	ISO	SDO	RAND
ISO/IEC TR29106:2007	ISO/IEC	SDO	RAND
ISO 32000-1:2008	ISO	SDO	RAND
JCP JSR 270 Java SE 6	Java Community Process	CONSORTIA	RF
Magsafe	Apple	COMPANY	NA
MATHML	W3C	CONSORTIA	RF
Matroska	open source project	CONSORTIA	RF
MD5 (RFC 1321)	IETF	CONSORTIA	RAND
Micro SD	SD Association	CONSORTIA	RAND
MIDI	MIDI Manufacturer s Ass'n	CONSORTIA	NA
MIME	IETF	CONSORTIA	RAND
Mini Displayport	VESA	CONSORTIA	RF
MINI-DVI	Apple	COMPANY	NA
MiniSD	SD Association	CONSORTIA	RAND
MJPEG (RFC 2435)	IETF	CONSORTIA	RAND
MMS	Open Mobile Alliance	CONSORTIA	RAND
Monitor Control Command Set [MCCS] Standard v2.2	VESA	CONSORTIA	RAND
MP3 (MPEG-1 Layer 3)	ISO/IEC	SDO	RAND
MP4 (ISO/IEC 14496-14:2003)	ISO/IEC	SDO	RAND
MPEG-2	ISO/IEC	SDO	RAND
MPEG-2 (ISO/IEC 13818)	ISO/IEC	SDO	RAND
MPEG-4 Part 2 (ISO/IEC 14496-2)	ISO/IEC	SDO	RAND
MSFT Silverlight	Microsoft	COMPANY	NA
MXF	SMPTE	CONSORTIA	NA
MXM Graphic Module Software Spec 3.0 revision 1.1	MXM Group/SIG	CONSORTIA	RF
MXM Graphics Module Mobile PCI Express Module Electromechanical Spec version 3.0 rev 1.1	MXM Group/SIG	CONSORTIA	RF

Net2Display Remoting Standard (N2D)	VESA	CONSORTIA	RAND
NTFS	Microsoft [Closed Spec]	COMPANY	NA
NTP (time synchronization)	IETF	CONSORTIA	RAND
OGG	Xiph.Org Foundation	CONSORTIA	RF
OpenAL	Creative Technology	COMPANY	RF
OpenCL	Khronos	CONSORTIA	RF
OpenGL 4.0 Compaitbility Profile Specification	Khronos	CONSORTIA	RF
OpenGL 4.0 Core Profile Specification	Khronos	CONSORTIA	RF
OpenGL ES	Khronos	CONSORTIA	RF
OpenGL SC 1.0	Khronos	CONSORTIA	RF
OpenGL Shading Language 4.00.7 Specification	Khronos	CONSORTIA	RF
OpenKode	Khronos	CONSORTIA	RF
OpenMAX	Khronos	CONSORTIA	RF
OpenML	Khronos	CONSORTIA	RF
OpenSL/ES	Khronos Group	CONSORTIA	RF
OpenVG	Khronos	CONSORTIA	RF
OpenWF	Khronos	CONSORTIA	RF
PCI Express Base Specification 2.0 [x8,x16]	PCI-SIG	CONSORTIA	RAND
PCI Local bus Spec 3.0	PCI-SIG	CONSORTIA	RAND
PCI Local Bus Specification 3.0	PCI-SIG	CONSORTIA	RAND
PCMCIA/PC Card	USB-IF	CONSORTIA	RF
PGA-989 socket	Intel	COMPANY	NA
PGP (RFC 4880)	IETF	CONSORTIA	RAND
PNG	W3C	CONSORTIA	RF
POP	IETF	CONSORTIA	RAND
Quicktime	Apple	COMPANY	NA
RealVideo 3&4	RealNetworks	COMPANY	NA
RJ-11 (TIA-968-A)	TIA	SDO	RAND
RSS	Various	CONSORTIA	NA
RSVP	IETF	CONSORTIA	RAND
RTMP	Adobe	COMPANY	RF
RTP	IETF	CONSORTIA	RAND
RTSP	IETF	CONSORTIA	RAND
S/MIME	IETF	CONSORTIA	RAND
SATA	Serial ATA Int'l Org.	CONSORTIA	RAND
SD	SD Association	CONSORTIA	RAND
SDL 1.3	open source project	CONSORTIA	RF
SDP	IETF	CONSORTIA	RAND
SDRAM	JEDEC	SDO	RAND
SHA-1 (FIPS PUB 180)	NIST	SDO	NA
SIP	IETF	CONSORTIA	RAND
SmartMedia	Toshiba	COMPANY	NA
SMTP	IETF	CONSORTIA	RAND
SOAP	W3C	CONSORTIA	RF
SODIMM	JEDEC	SDO	RAND
Sorenson	Sorenson	COMPANY	NA
SQL - ISO/IEC 9075	ISO/IEC	SDO	RAND
SVCD (IEC 62107)	IEC	SDO	RAND
SVG	W3C	CONSORTIA	RF
TCG EFI Platform Spec 1.2	UEFI	CONSORTIA	RAND
TCG EFI Protocol Spec. 1.2	UEFI	CONSORTIA	RAND
TCG Physical Presence Interface Spec	Trusted computing Group	CONSORTIA	RAND
TCP	IETF	CONSORTIA	RAND
Theora	Xiph.Org Foundation	CONSORTIA	RF
TKIP	IEEE	SDO	RAND
TPM 1.2 Protection Profile	Trusted computing Group	CONSORTIA	RAND
TSR jack 3.5mm (PCXX version)	Intel	COMPANY	NA
UDP	IETF	CONSORTIA	RAND
UEFI Platform Initilization Distribution Packaging Spec 1.0	UEFI	CONSORTIA	RAND
UEFI Platform Initilization Specification 1.2	UEFI	CONSORTIA	RAND
UEFI Shell Spec 2.0	UEFI	CONSORTIA	RAND
UEFI Specification Version 2.3	UEFI	CONSORTIA	RAND
Universal Audio Architecture	Microsoft	COMPANY	NA
UPnP	UPnP Forum	CONSORTIA	RF
USB	USB-IF	CONSORTIA	RF
VC-1 (SMPTE 421M)	SMPTE	CONSORTIA	NA
VCD ("White Book")	Various companies	CONSORTIA	NA
VESA DDC2/E-DDC	VESA	CONSORTIA	RAND
VGA	IBM	COMPANY	NA
VOB	DVD Forum	CONSORTIA	NA
VP5	On2 Technologies	COMPANY	NA
VP6	On2 Technologies	COMPANY	NA
VP8	Google	COMPANY	NA
WAV	MSFT and IBM	COMPANY	NA
WebGL - OpenGL ES 2.0	Khronos	CONSORTIA	RF
WIGIG 1.0	Wireless Gigabit Alliance	CONSORTIA	RF
WiMax (IEEE 802.16)	IEEE	SDO	RAND
Wireless 802.11 [a/b/g/n]	IEEE	SDO	RAND
Wireless HD 1.0	Wireless HD Consortium	CONSORTIA	NA
WMV	Microsoft	COMPANY	NA
WSDL	W3C	CONSORTIA	RF
x.509	ITU-T	SDO	RAND
x86-64 Instruction Set	Intel/AMD	COMPANY	NA
XHCI	Intel	COMPANY	NA
XML	W3C	CONSORTIA	RF

A USER-CENTRIC APPROACH TO QOS REGULATION IN FUTURE NETWORKS

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ABSTRACT

The evolution of current networks to Next Generation Networks (NGNs) constitutes arguably the most significant transformation in the Telecommunication sector in recent decades. Quality of Service (QoS) is one of the key aspects in this evolution. In the NGN environment, networks are designed to be multi-service, supporting a wide range of premium services. Each of these services may have different QoS requirements which should be established based on the overall end users' perception. In this emerging context, novel QoS policies are required to adapt the traditional QoS regulatory model to the new scenario. This paper presents a user-centric approach to identify key factors that contributes to the development of quality of service regulation in future networks. A case study on the application of our user-centric QoS model to the Internet QoS regulation in Spain is described. The results of the study demonstrate the need for adapting current regulatory frameworks in order to ensure competition, pluralism and diversity in the new network environment.

Keywords— QoS, Internet, regulation, NGN

1. INTRODUCTION

The successful development of Next Generation Networks (NGN) should be accompanied by sound regulations. Such regulatory program if implemented early on would help to reduce the investment risks and accelerate the deployments of networks through competitive pressure. A late or ineffective regulatory program could otherwise result in a non-competitive environment.

In contrast to the legacy networks whose regulation came after full-fledged network deployment, in ITU-T's vision "*the move to NGNs represents an opportunity to establish in advance ground rules for ensuring the continued passage to effective competition and minimize damage during transition*" [1].

The move to NGNs is the next logical and evolutionary step in technological progress but the R&D community is already stepping forward and the next big challenge, the NGN evolution towards the Future Internet, is already under study. In accordance, the new regulatory framework should strive for technology neutrality in order to better suit the evolution and ever-expanding capabilities of the underlying networks and services. In future networks we are expected to observe the convergence of different services with different QoS requirements and a drastic increase in the users' demand for new QoS guaranteed services.

In this context, QoS regulation becomes one of the key issues in the new Internet regulatory framework. The largest part of the current Internet QoS regulation has been developed based on traditional legacy network regulation model which is technology-centric. In the future networks context, the user's behavior will play a major role in quality of service management as it is important to satisfy user's requirements and expectations to ensure the transition to the new network environment.

This paper presents a new approach aimed at identifying the key factors that will contribute to the development of the future networks QoS regulation by considering users requirements and ITU-T QoS framework. In order to demonstrate the effectiveness of the approach a case study of applying the regulation proposal to the current Spanish situation is also described.

The remainder of this paper is organized as follows. Section 2 summarizes current QoS related regulations and the challenges in adjusting them to the future networks environment. Section 3 describes our user-centric regulation approach. In section 4 we present the Spanish case study and obtained results. Finally, section 5 contains conclusions and final remarks.

2. BACKGROUND AND CHALLENGES

Since Internet QoS regulation is a relatively recent notion [2] most of the current regulations are only specified to be applied to the Internet Access Service and, basically, only concerning the broadband networks [3].

In addition, although many of the National Regulatory Authorities (NRAs) assert to have a QoS regulation based on the ITU-T established QoS framework, i.e. Ecuador [4] and Colombia [5], none of them fully comply with it. Even in Europe, where the European Commission has developed initiatives to regulate the sector [6], each of the NRAs has defined a different approach. As an example, Ofcom [7], the U.K. regulator and one of the most active proponents of the Internet regulation published in July 2008 a voluntary code to ensure consumers have QoS information about broadband speeds. Ofcom asked Internet Service Providers (ISPs) to sign up the Code to present QoS related statistics to the end users.

In other countries, like Spain [8] or Portugal [9] the regulator obligates ISPs to publish the QoS information offered and achieved. Finally, other NRAs, like HAKOM (Croatia), RTR (Austria) or FICORA (Finland) have not taken any measure to ensure the transparency of Internet QoS.

In light of these data National Regulatory Agencies should seize the opportunity provided through the evolution to NGN to unify criteria regarding the new Internet QoS regulation that, clearly, will condition the development and establishment of the new networks.

At the moment some countries [10] are already working on developing initiatives for the establishment of regulations in the area of the NGN [11, 12]. Therefore, an effort is required from the International Standardization Organizations and the NRAs to join forces in defining the global QoS policies for the new scenario.

ITU-T has already pointed out some key points and recommendations [13]:

"*The real differences in terms of regulatory implications in NGN are related to the multi-service and multi-operator environment and to the fundamentally global character of the Internet.*

Regulation should contribute to opening the sector for innovations and be seen as part of a broader national strategy for innovation of the communication area.

QoS for end-users can be secured by way of regulatory provisions. Users will often experience that the QoS delivered does not correspond to the promises made."

Regarding the latter point, users should be provided with QoS information useful for them in a "relevant, comparable and reliable" fashion. As stated in [3], only if those three requirements are assembled, could the QoS published information be used to assess the QoS of the providers. Furthermore, the three requirements are also applicable when comparisons between the QoS provided in different countries are to be made so they should be adopted by all the National Regulatory Agencies.

On the other hand, it is necessary for regulators and providers to know the level of user's satisfaction of a delivered service. This will help in better guiding the market and steering the networks evolution. For that reason, as suggested in [14] the exact way to express QoS in the future may vary but users' global level of satisfaction are likely to be specified.

New QoS methodologies and algorithms linking network parameters to QoS perceived by users (QoP) should be defined. User's requirements for different application and services should also be analyzed in order to obtain baseline QoS levels that must be upheld. The publication of all this information should be carried out by independent and neutral bodies and must be supervised by regulators to guarantee fidelity of information. A problem to be addressed is who should pay the costs of ensuring this neutrality [15].

Finally, it must be remarked that future QoS regulation should be technology neutral since ex-ante regulation for interoperability or QoS issues in a high technical detail is much more likely to impede market development than to support it.

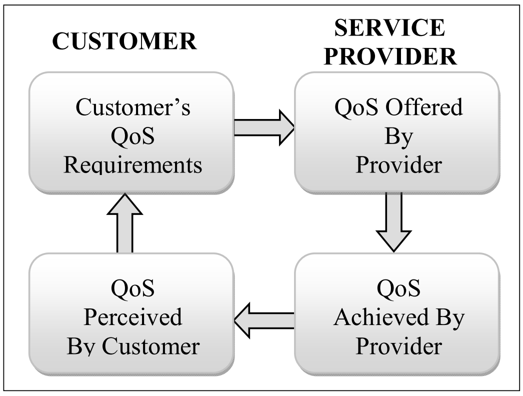

Fig. 1 – G.1000: Four viewpoints of QoS

3. A NEW APPROACH BASED ON THE ITU-T QOS FRAMEWORK

The proposed approach for enhancing Internet QoS regulation comprises using the QoS model and implementation methodology presented in [16]. The QoS model is based on the ITU-T QoS framework [17] and, therefore, complies with its standards and requirements and could be adopted by global NRAs.

3.1. Principles of the new regulation proposal

The proposed QoS model and the deployment methodology supply a thorough analysis of the four view points of QoS of ITU-T G.1000 (Fig. 1) on the analysis for user's satisfaction. In addition, with the new proposal the different points of view are closely linked and users' satisfaction is evaluated in terms of these relationships.

In this sense, the new regulation approach should force ISPs to make available, at least, the following information about their services:

- *QoS offered by provider (QoO)*
- *QoS delivered by provider (QoD)*
- *QoS perceived by user (QoP)*
- *QoS required by user (QoR)*
- *Global user's satisfaction with the service*

Nearly all of the current Internet QoS regulations in force consider only the service provider's point of view. Furthermore, in most cases, the QoS offered and the QoS delivered by provider are defined only in terms of technical networks parameters, like the ones detailed in ETSI EG 202 057-4 [18].

In the new regulation proposal, according with the ITU-T Rec. E.802 [19], even the objective QoS (QoO/QoD) will be expressed in terms meaningful to users. For that purpose, the *Key Quality Indicators (KQI)* relevant to the users must be identified and their relation with the associated *Key Performance Indicator (KPI)* established [20].

Therefore, not only network performance parameters must be taken into account but also other related parameters such as provision, operation, customer services, etc. For that reason, as described in [16], the proposed QoS model gathers QoS indicators under four categories: *Network QoS, Availability, Customer Care and Content*. The global QoS perceived by user will be computed in terms of users' preferences for each of the QoS categories.

3.2. Analytical methodology for QoS evaluation

Many literature [21] and standards [22] on defining and evaluating the QoS offered and delivered by provider can be found. In fact most of the current Internet QoS regulation is already forcing or suggesting to Internet Service Providers the publication of these QoS aspects. Unfortunately, in most cases these regulations often results in an endless list of technical parameter measurements that, although may be useful for the providers or even the regulators, do not mean much to the users. For that reason, we consider that it will be more effective to provide, in addition to the results of the Key Performance Indicators measurements, a *Global Criteria Key Quality Indicator (Q)* value for each of the QoS categories. The purpose of this indicator is to supply a global idea of the quality of service that the customer may perceive by means of the influence of possible misalignments between the QoS offered and the QoS delivered by provider. In addition, the Global Criteria Key Quality Indicator values will be normalized in the ranges [0,1], for unified and easy comparison among QoS of different providers.

For the computation of this global indicator, the user's preferences for each of the Key Quality Indicator that may contribute to the QoS category should also be considered, since they will also influence the user's perception of the service.

3.2.1. Assessment of the Global Key Quality Indicator (Q)

For the evaluation of the *Global Key Quality Indicator (Q)* of one QoS criteria category, the following formulation is suggested:

$$Q_{criteria} = \sum_{i=1}^{n} w_i \, Scal_i(KQI)_i \qquad (1)$$

where $w_1+w_2+..+w_i=1$.

The weight factors w_i denotes the user's preferences for each of the KQI that contributes to the QoS criteria category. User surveys must be designed to allocate the suitable weighing values for each of the KQI based on the user´s opinion because some QoS indicators may be critical for most users while others may be completely irrelevant. In this way the user's requirements are taken into account. $Scal_i\ (KQI_i)$ is a scaled value for each of the KQI that contributes to the QoS criteria category. This value is computed, based on related QoS work [23], in terms of the misalignment between the QoS offered and delivered.

The proposed scaling function for $Scal_i\ (KQI_i)$ is:

$$Scal_i\,(KQI_i) \begin{cases} \dfrac{(KQI_i) - (KQI_i)_{min}}{(KQI_i)_{max} - (KQI_i)_{min}} & \text{if } (KQI_i) \in KQI^{+} \\[2ex] \dfrac{(KQI_i)_{max} - (KQI_i)}{(KQI_i)_{max} - (KQI_i)_{min}} & \text{if } (KQI_i) \in KQI^{-} \end{cases} \qquad (2)$$

where KQI^{+} is the subset of more-is-better-like key quality indicators (e.g. speed transmission), and KQI^{-} the subset of less-is-better-like indicators (e.g. time to repair faults).

KQI_{max} and KQI_{min} denote the maximum and minimum values established within the QoS offered by provider in the provision of the service and the KQI_i value corresponds to the QoS actually achieved by provider.

In this way, the four *Global Key Quality Indicators*, $Q_{Network}$, $Q_{Availability}$, $Q_{CustomerCare}$, $Q_{Content}$, will provide, in the new regulation approach, very simple, precise and helpful QoS information to the user of the service.

3.2.2. QoS perceived (QoP) by users

One of the major innovations in the new regulation approach is the consideration of the *"subjective QoS point of view"*. The evaluation of the QoS as perceived by the users is critical in determining whether the users' requirements have been met. As mention before, this is also one of the main challenges in defining the NGN regulatory framework where the user preferences and capabilities should be analyzed in order to ensure the success and development of new network service scenarios.

ITU-T defines QoS perceived by users (QoP) [24] as: "*A statement expressing the level of quality that customers/users believe they have experienced.*"

Based on this definition, we compute the QoP utilizing the four Global Key Quality Indicators. Each of the global indicators may have different impact in the final QoS perceived so QoP is also weighed by a factor α according to user's preferences for each of the Global Key Quality Indicators. Again, appropriate user surveys to obtain the suitable values for the weighing factors should be developed.

As a result, the QoS perceived is estimated in terms of:

$$QoP = \alpha_1 Q_{Network} + \alpha_2 Q_{Availability} + \alpha_3 Q_{CustomerCare} + \alpha_4 Q_{Content} \quad (3)$$

where $\alpha_1 + \alpha_2 + .. + \alpha_i = 1$.

In this way, the QoP will be provided as an aggregate value ranged in [0,1] such that the users, regulators and providers have a unified global view over the QoS perceived by the users.

Table 1- Key Quality Indicators for Internet Access

CRITERIA	FUNCTIONAL ASPECT	KQI
Network QoS	Data transmisión speed achieved (from networt to user)	Downstream speed achieved
Availability	Generic DNS service availability	Successful log-in ratio
	Authentication service availability	
	Access network (and equipments) availability	Unsuccessful data transmissions ratio
Customer Care	Fidelity/accuracy for service provisioning	Provisioning time for the Internet Access
	Reliability for fault repair service	Fault report rate per fixed access lines
		Fault repair time on access network
	Accuracy of on-line support	Response time for admin/billing enquiries
	Accessibility to the documentation and information about the service	
	Fidelity/accuracy of the charging/billings system	Rate of bill correctness complaints
	Capability/Efficiency of the complaint management	Customer complaints resolution time
		Frequency of customer complaints

3.2.3. User's satisfaction

Despite the fact the term "Quality of Service" has been traditionally linked to network performance and technical parameters, ITU-T defines quality of service as "*Totality of characteristics of a telecommunications service that bear on its ability to satisfy stated and implied needs of the user of the service*" [24].

Therefore, it is very important for any QoS regulation to have an effective measure of a user's satisfaction over his/her services. User satisfaction assessment is thus an important part of the quality assurance. In addition, nowadays, in the "Internet era", the users' behavior can vary to a great extend and customer satisfaction is one of the key factors influencing the user repurchase intention. Therefore, user's satisfaction may also be critical for the development and expansion of the new markets and services like the ones defined in the NGN.

The new approach will consider CSAT model [25] that states that customer satisfaction can be modeled as a function of the user's perception and disconfirmation, the difference between user's perception and user's expectations. Then user's satisfaction in terms of the ITU-T QoS framework will be estimated as:

$$S = f_1(QoP) + f_2(QoP - QoR) \qquad (4)$$

where functions f_1 (the perception function) and f_2 (disconfirmation function) must be defined to control the effects of user's tolerance and other contextual factors. On the other side, *QoR* denotes the user's requirements for QoS which should be inferred through market studies and surveys. Users update their future requirements based on present expectation and perception. So, regular auditing of the QoR is necessary in order to adapt the quality objectives to the changes in user perception and expectation.

Table 2- KQI, KPI and parameters for Internet Access

KQI	KPI	PARAMETER
Downstream speed achieved	Mean value of transmission rate	Mean value of throughput (Kbps)
	Maximum transmission rate	Maximum value of throughput (percentil 95 -Kbps)
	Minimum transmission rate	Minimum value of throughput (percentil 5- Kbps)
Successful log-in ratio	DNS resolution time	Mean time for DNS resolution (sg)
	Authentication time	Mean time for authentication (sg)
Unsuccessful data transmissions ratio	Correct file transmisión time	Delay
		Packets lost

4. CASE STUDY

In the previous section we have introduced the principles and methodology of a new regulation approach. In this section, we present the implementation of this regulation in a real world scenario: the Internet QoS regulation in Spain.

4.1. Spanish Internet QoS regulation

The Spanish Ministerial Order ITC/912/2006 of 29 March 2006 [8] was issued to regulate the conditions of the quality of service for electronic communications service provision. At the same time the order was published, the Ministry for Industry, Tourism and Trade (MICyT) established a Working Group (GT3) [26], comprised of Internet Service Providers (ISPs) and Internet users associations with the main objective of identifying a specific set of QoS parameters and measurement methodology that, based on the International Standards, will help to define an accurate and precise QoS regulation for the Internet Access Service. The result of the work was a methodology that has been adopted by the main Spanish ISPs to comply with the Internet QoS regulation. Even though the methodology meets the three criteria of relevance, comparability and reliability, it only covers two of the four points of view in the ITU-T QoS framework [17].

The lack of the user's QoS points of view, as recommended in ITU-T framework, leads to results that do not comply with the objective of the regulation: guaranteeing best practices in the provision of the service according to user's QoS requirements.

4.2. Results of Spanish Internet QoS regulation

Since the end of 2007, the main Spanish ISPs quarterly publishes the results of the QoS offered (QoO) and the QoS delivered (QoD) for their Internet access services in the regulator's website [27]. One of the major limitations of this regulation is that it only covers the provider's QoS point of view (objective QoS). In some way, the Spanish regulator seems to be aware of this shortcoming because, at the end of 2009, a survey report about the user's QoS point of view was also published on their website [28].

Table 3- KQI, KPI and parameters for Internet Access

KQI	***Scal (KQI)***							
	2008				***2009***			
	Q 1	*Q 2*	*Q3*	*Q 4*	*Q 1*	*Q 2*	*Q 3*	*Q 4*
Downstream speed achieved	0,27	0,60	0,73	0,74	0,40**	0,30	0,65	0,60
Successful log-in ratio	0,75	0,93	0,90	0,98	0,98**	0,92	0,95	0,95
Unsuccessful data transmissions ratio	0,53	0,87	0,93	0,80	0,96**	0,96	0,96	0,88
Provisioning time for the Internet Access	0,38	0,38	0,50	0,88	0,88	0,88	0,88	0,75
Fault report rate per fixed access lines	0,57	0,71	0,63	0,63	0,39**	0,63	0,49	0,07
Fault repair time on access network	0,36	0,97	0,60	0,56	0,41	0,92	0,55	**-0,11**
Response time for admin/billing enquiries	0,20	0,31	0,37	0,65	0,97**	0,79	0,52	0,67
Rate of bill correctness complaints	**-0,61**	**-0,53**	**-0,11**	**-0,11**	**-0,02***	0,57	0,55	0,25
Customer complaints resolution time	**-1,00**	**-1,00**	0,17	0,69	0,45	**-0,15**	0,42	0,68
Frequency of customer complaints	**-0,93**	**-0,83**	**-0,36**	**-0,44**	0,04*	0,55	0,54	0,24

* QoS offered values enhanced.

The report included information about the user's perception of QoS and the degree of user satisfaction. However even with this new information it results difficult for the users to draw conclusions about the global QoS of ISPs. In addition, it is not easy to find the relation between the subjective QoS data on user's satisfaction and the objective QoS data offered by providers. In fact, a first glance at the global results suggests inconsistencies between both statistics. Another problem is that the ISP's information is provided in terms of a list of technical parameters instead of users' Key Quality Indicators (KQI), as recommended by ITU-T [19]. The Spanish regulator seems to be also aware of the complexity of understanding the data because it has made available a set of information that provides a more "user-friendly" summary of the published data.

In the end, even though Spanish Internet service regulation has been a great advance and relevant QoS information is published, it is still far from meeting the crucial objective of providing useful information to the users. This situation motivates the application of the user-centric approach to the Spanish case.

4.3. Implementation of the new regulation approach

One of the principles of the new regulation approach is that, as recommended in the ITU-T QoS framework, QoS should be expressed in terms of the *Key Quality Indicators (KQIs)*. This is also a key issue to be considered in regulation of NGN where QoS requirements tend to be technology independent due to the heterogeneous nature of the networks. According to this principle, the first step to accommodate the current Spanish regulation to the new approach is the definition of an adequate set of Key Quality Indicators for each of the QoS criteria categories defined in the QoS model: *Network QoS, Service Availability, Customer Care and Content*. Table 1 shows the list of KQIs that have been identified, based on international standards [18, 29, 30] in the case study and that fits closely the list of parameters defined by GT3.

Table 4- KQI, KPI and parameters for Internet Access

Global Indicators	***2008***				***2009***			
	Q1	*Q 2*	*Q 3*	*Q 4*	*Q 1*	*Q 2*	*Q 3*	*Q 4*
$Q_{Network}$	0,271	0,602	0,730	0,736	0,398	0,304	0,647	0,603
$Q_{Availability}$	0,642	0,898	0,917	0,890	0,967	0,942	0,955	0,915
$Q_{CustomerCare}$	-0,108	0,048	0,276	0,422	0,455	0,613	0,563	0.356
QoP	**0,285**	**0,536**	**0,657**	**0,694**	**0,616**	**0,624**	**0,730**	**0.637**

It can be observed that no functions have been defined for the content QoS category. We note that indicators of major importance, such as cyber-security, could have been taken into account in this category. Nevertheless, some recent user surveys [31] show that this QoS indicator is not critical to residential users.

After KQIs have been identified, the *QoS offered* for each KQI must be established by determining the maximum and minimum thresholds. These values should be in accordance with the service user's requirements. Then the Key Performance Indicators (KPI) and parameters to measure the QoS achieved for each of the KQI must be defined (Table 2).

The analysis up to this point fits well with the current regulation except for the use of technical parameters rather than KPIs. We now can implement our methodology using real data obtained from the Spanish regulator's website about the QoS of one of the main ISPs in Spain. Based on the data results of the QoS offered and delivered by this provider, the evaluation of the $Scal_i(KQI_i)$ for each of the Key Quality Indicators is carried out using the analytical formulation proposed in section 3 (eq. 1 and eq. 3). The results for the four quarters of 2008/2009 are presented in Table 3.

According to the analytical formulation described in section 3, values close to 1 denote that the QoS delivered is very similar to the QoS offered by provider. On the contrary values close to 0 indicates a large misalignment between them. Last, negative values (bolded) of the $Scal_i(KQI_i)$ indicate that the QoS achieved do not even comply with the user's requirements. As can be observed, this provider does not meet the user's requirements in most of the results related with complaints for the 2008 year. For that reason, this provider has fixed a new offer lowering the level of QoS for these indicators in 2009 year (*) as can be observed in [27].

On the other hand, for those KQI signed with (**) the provider has increased the level of QoS offered to the users. In this way the ISP has improved the values of the $Scal_i(KQI_i)$ for the 2009 year by avoiding the misalignment between the QoS offered and achieved. Nevertheless, this may not improve the user's satisfaction with the service because, in fact, in many cases the QoS achieved has not been improved. This is one of the reasons of extending regulation to include subjective aspects such as user satisfaction.

Table 5- User's preferences for Internet Access

Criteria	α_i	KQI	W_i
Network QoS	0,34080	Downstream speed achieved	1
Availability	0,35239	Successful log-in ratio	0,50133
		Unsuccessful data tr. ratio	0,49866
Customer Care	0,30679	Provisioning time	0,14031
		Fault report rate	0,15460
		Fault repair time	0,15460
		Response time for enquiries	0,15169
		Rate of bill correctness complaints	0,13914
		User'scomplaints resolution time	0,12981
		Frequency of customer complaints	0,12981

Once, the scaled values $Scal_i(KQI_i)$ for each of the Key Quality Indicators have been evaluated the *Global Key Quality Indicators (Q)* for each of the QoS criteria and the *Global QoS perceived* for each quarter are computed (Table 4). Based on [29] the weighing values w_i and α_i (Table 5) to consider user's preferences regarding KQIs and QoS Criteria categories have been established. With this new approach, additional global QoS information is provided.

Finally, the user's satisfaction with the service will be modeled. As described in section 3, the perception function f_1 and the disconfirmation function f_2 that fits with the user's profile must be obtained [25]. For this purpose a data fitting process, based on the results of the surveys carried out by the Spanish regulator at the end of 2009 [28], has been made.

As a result, the perception function f_1 (Fig. 2) and the disconfirmation function f_2 (Fig. 3) have been obtained. Based on these functions, the user's satisfaction with the service can be modeled by means of the following expression:

$$S = \begin{cases} 0.7083(QoP)^3 - 2.125(QoP)^2 + 2.4166(QoP) - 0.2025(QoR) & \text{if } (QoP\text{-}QoR) \geq 0 \\ 0.7083(QoP)^3 - 2.125(QoP)^2 + 2.21416(QoP) + (QoP - QoR + 1)^{3.235} - 1 & \text{if } (QoP\text{-}QoR) \leq 0 \end{cases} \quad (5)$$

Based on the survey results, the user's QoS requirements for the case study are close to QoR=0.65. The resulting satisfaction curve for the defined QoS requirements is presented in Figure 4. As it can be observed, the domain of satisfaction is bounded between -1 and 1 (the same as the disconfirmation function). Values below 0 denote a prolonged user's dissatisfaction with the service due to a chronic condition of QoP under user's requirements (e.g. complaints QoS indicators in this ISP).

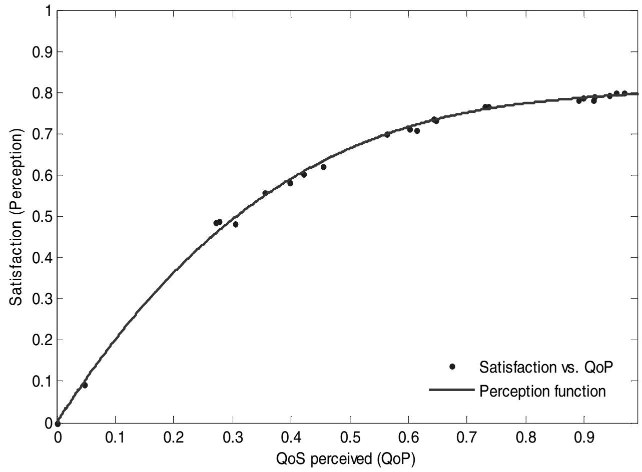

Fig. 2 – Perception function for residential users

On the other hand, customer satisfaction is slightly increasing when QoS perceived exceeds user's requirements and it is drastically reduced when QoS perceived falls below QoR.

It must be remarked that the obtained results of user's satisfaction are very close to the survey results published by the regulator which verifies the suitability of the regulation approach. Furthermore, the "objective" and "subjective" QoS are linked by the proposed QoS methodology through the evaluation of the QoS perceived by means of the QoS offered and the QoS delivered misalignments.

In this way the ITU-T QoS framework is completely fulfilled: "*For any framework of QoS to be truly useful and practical enough to be used, it must be meaningful from these four viewpoints: Customer's QoS requirements, Service provider's offerings of QoS (or planned/targeted QoS), QoS achieved or delivered, customer perception of QoS.*"[17]

5. CONCLUSIONS

In this paper we have presented a new regulation approach that aims to provide simple and useful QoS information for the users, regulators and providers.

In contrast to the "traditional objective" QoS information, a new wider perspective is presented, offering new QoS subjective information that are technology and network neutral as required in future networks.

Based on international standards, the user-centric regulation approach makes use of the Key Quality Indicators (KQIs) and the Key Performance Indicators (KPIs) in order to consider the QoS aspects that are relevant for users.

The new approach links, through the Global Key Quality Indicators, the four QoS points of view in the ITU-T QoS framework. In addition, the QoS results are provided in normalized ranges [0, 1], for unified and easy comparison among QoS of different providers.

To conclude, the proposed methodology helps in the adaptation of regulation to the new networks environment.

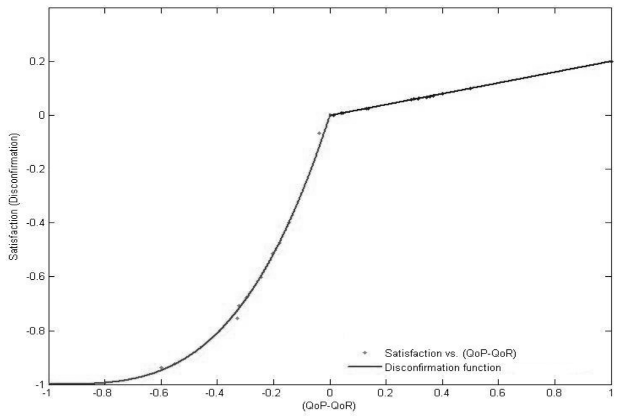

Fig. 3 – Disconformation function

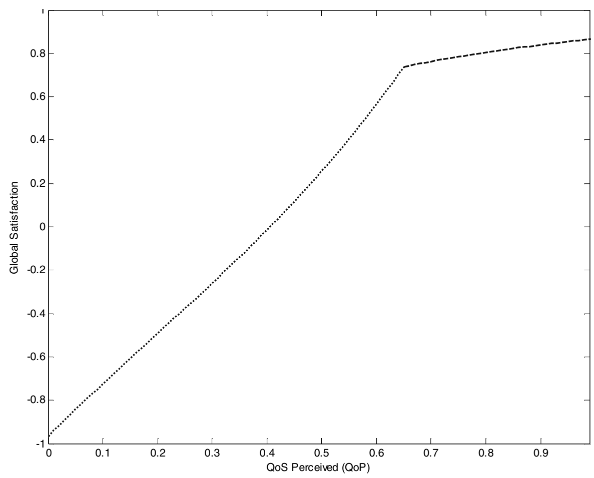

Fig. 4 – Satisfaction vs QoP perceived

REFERENCES

[1] ITU-T, "ITU-T and Next Generation Network (NGN) Global Standards Initiative (GSI), Focus group on NGN," 2007.
[2] EIS, "Overview of the EU regulatory framework for the electronic communications sector," 2008.
[3] ERG, "Report on Next Generation Access - Economic Analysis and Regulatory Principles," in http://erg.ec.europa.eu/doc/publications/erg_09_17_nga_economic_analysis_regulatory_principles_report_090603_v1.pdf, 2009.
[4] CONATEL, "Norma de Calidad de los Servicios de Telecomunicaciones," in *Consejo Nacional de Telecomunicaciones de Ecuador*, 2006.
[5] CRT, "QoS conditions in telecommunication services," in *Comisión de Regulación de Telecomunicaciones*, República de Colombia, 2006.
[6] EC, "Directive 2002/22/EC of the European Parliament and the Council of the European Union -on universal service and users' rights relating to electronic communications networks and services (Universal Service Directive)," *Official Journal of the European Union*, 2002, pp. 27.
[7] OFCOM, http://www.ofcom.org.uk.
[8] BOE, "Orden Ministerial del 29 de Marzo de 2006 ICT/912/2006 ", 2006.
[9] ANACOM, "Evaluation of the Internet Access Service - Fixed Broadband (Autoridade Nacional de Comunicações de Portugal)," 2007.
[10] ITU-T, "Developments of Next Generation Networks (NGN): country case studies," 2009.
[11] Ofcom, "Next Generation Networks: Developing the Regulatory Framework," 2006.
[12] Ovum, "Policy Principles for the Irish NGN Regulatory Framework," 2007.
[13] ITU-T and InfoDev, "ICT regulation toolkit: Key points an recommendations on QoS regulations," in http://www.ictregulationtoolkit.org, 2008.
[14] N. Encarnação, "Future Networks & Services, Standards & Regulation, private and public domains, general aspects," presented at ITU-T Workshop on "End-to-End QoE/QoS", 2006.
[15] LIRNEasia, "Broadband QoSE Benchmarking," 2008.
[16] E. Ibarrola, F. Liberal, A. Ferro, and J. Xiao, "Quality of service management for ISPs: a model and implementation methodology based on the ITU-T recommendation E.802 framework," *Communications Magazine, IEEE*, vol. 48, pp. 146-153, 2010.
[17] ITU-T, "G.1000: Communications quality of service: A framework and definitions," 2001.
[18] ETSI, "EG 202 057-4: Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 4: Internet access," 2008.
[19] ITU-T, "E.802: Framework and methodologies for the determination and application of QoS parameters," 2007.
[20] TMF, *SLA Management Book* - Volume 2- Concepts and Principles, vol. 2, 2005.
[21] A. Botta, D. Emma, A. Pescape, and G. Ventre, "Systematic performance modeling and characterization of heterogeneous IP networks," *Journal of Computer and System Sciences*, vol. 72, pp. 1134-1143, 2006.
[22] ETSI, "EG 202 765-3:Speech and multimedia Transmission Quality (STQ); QoS and network performance metrics and measurement methods; Part 3: Network performance metrics and measurement methods in IP networks ", 2009.
[23] N. Limam and R. Boutaba, "QoS and reputation-aware service selection," presented at Network Operations and Management Symposium, 2008. NOMS 2008. IEEE, 2008.
[24] ITU-T, "E.800: Definitions of terms related to Quality of Service," 2008.
[25] J. Xiao and R. Boutaba, "Assessing Network Service Profitability: Modeling From Market Science Perspective," *Networking, IEEE/ACM Transactions on*, vol. 15, pp. 1307-1320, 2007.
[26] GT3, "Criterios adicionales para la medición de los parámetros de calidad de servicio específicos para el servicio de acceso a Internet," 2008.
[27] MICyT, "Publicación niveles calidad de servicio" http://www.mityc.es/telecomunicaciones/es-ES/Servicios/CalidadServicio/Paginas/Calidad.aspx, 2007.
[28] MICyT, "Estudio sobre la percepción de los usuarios acerca de la calidad de los principales servicios de telecomunicaciones," 2009.
[29] ETSI, "TR 102 276: User's Quality of Service Criteria for Internet Access in Europe," 2003.
[30] ETSI, "EG 202 009-2: Quality of telecom services; Part 2: User related parameters on a service specific basis," 2007.
[31] SSRC, "Survey on residential broadband internet access services," 2007.

COMPETITION AND COOPERATION IN THE FORMATION OF INFORMATION TECHNOLOGY INTEROPERABILITY STANDARDS: A PROCESS MODEL OF WEB SERVICES CORE STANDARDS

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ABSTRACT

Standards formation is a key dimension in the competitive strategy of ICT firms, as a successful strategy would result in the emergence of favorable IT interoperability standards. This paper examines the standardization efforts of core Web services standards and the results indicate that resource dependencies and strategies adopted by dominant firms to extend their platforms influence the standards formation process. Communities of practice and standard-setting bodies are leveraged by dominant firms in the formation and adoption of standards. We propose a process model of standard setting consisting of five intertwined states: resource pooling, linkages, signaling and implementation, institutionalization, and extension.

Keywords— Web services, ICT Standards, Standardisation, Process model

1. INTRODUCTION

Standards are emerging as an important dimension of competition in information and communication technologies (ICT) sector, where significant modularization and network externalities exist. Standards emerge through two broad processes: *de jure* and *de facto* standardization. *De jure* standardization concerns standards promulgated by legislative bodies or voluntary standards published by independent organizations [1], [2], [3]. *De facto* standards emerge when a standard arises from a standardization struggle between different technologies, each of them sponsored by a firm or a coalition of firms [1], [4]. Standardization is a complex longitudinal process involving activities that include decision-making, technical construction, and coordination across myriad firms, technologies (artifacts) over time and space [5]. Firms get involved in the standardization process through signalling their willingness to participate/develop a standard; agree on the content and form of the standard; develop a contractual agreement in relation to the standardization process and its outcomes; and finally, agree upon what conforming to the standard specification actually means [6]. Often, firms may have to deal with informal groups such as Communities of Practice (COP) interested in non-commercial applications of a technology. Communities of Practice are a group of people informally and contextually bound and working through a codified exchange of information [7], [8].
Specifically with regard to the firms in the ICT sector research has proposed, that open standards are the key to volume and protected standards are only viable in small high-priced niche markets [9]. This is important given the open standards nature of core Web services standards, which is the context of this research. According to ITU-T, "Open Standards are standards made available to the general public and are developed (or approved) and maintained via a collaborative and consensus driven process. Open Standards facilitate interoperability and data exchange among different products or services and are intended for widespread adoption" [10]. With the exception of [11], [12], [13] there is a lack of empirical research which addresses the development, adoption, and outcomes of IT-related standards. Past research has also looked at IT standards issues such as the emergence of industry-wide collaborative standards in the case of vertical IT standards [14], vendor strategies towards maintaining switching costs in scenarios where product features are defined by open standards [15], economics of standard adoption from the perspective of diffusion of a communication standard [16], importance of relationship between timing of events in reaching an agreement and managing critical relationships [17], standards life cycle models [18] and standards development and diffusion strategies [19].
The term Interoperability could be used to define the ability of disparate systems to work together. In the context of ICT sector, interoperability could mean seamless, cost effective integration and extension of disparate systems at data, format, schema and business process levels. Previous research [20] has shown higher degree of ICT interoperability fosters innovation by reducing lock-in effects and lowering entry barriers, enhances user choice and autonomy and growth of diverse applications. Given the importance of the outcomes of IT interoperability standards for organizations in ICT industries, our objective is to understand the competitive as well as cooperative behavior of dominant firms involved in the process of standards setting. Towards this, we examine the strategies espoused by key players, process of standard setting, how institutionalization happens and how communities of practice are involved. In this paper we propose an empirically grounded process model of standardization that

helps analyze the formation and evolution of standards as a dynamic process.

Process theories offer a useful methodology to understand the cooperative and competitive behavior of firms in the formation of IT interoperability standards. Process theories focus on sequences of activities to explain how and why particular outcomes evolve over time [21], [22]. In this paper, we derive an empirically grounded process model by examining the standardization processes of three inter-related core Web services standards: Universal Description, Discovery and Integration (UDDI), Simple Object Access Protocol (SOAP) and Web services Description Language (WSDL). The core Web services standards are chosen as they involve a rich history of a) competitive and cooperative standards formation strategies exhibited by dominant firms in the ICT industry, b) the standards are inter-related and were formed almost in parallel, c) private and public participation, including informal groups such as COP participated and influenced several stages of evolution, and d) involvement of multiple standard setting bodies reflecting the dynamics of institutionalization. Moreover, the sole reason for the large scale interest in Web services is due to the industry adoption of the three core interoperability standards. Therefore, the analysis of the standard setting process would offer useful insights into the competitive and cooperative strategies of dominant firms in influencing the formation and adoption of the standards.

The remainder of the paper is organized as follows. In the next section, we review the standards literature followed by the research methodology. Next we cover the three IT interoperability standards, followed by the process model of standardization. The final section details the discussion of the implications of the proposed process model and contributions for future research.

2. STANDARDS AND STANDARDISATION PROCESS

Standard setting is a multistage process, which comprises of profiling, standard development, testing and deployment [23]. Standard setting can be dictated by a government institution or by voluntary agreement between independent firms with their efforts coordinated by trade or professional organizations or set unilaterally by a firm's own management [24]. Standard setting organizations based on degree of strategic purposes can be classified as research consortia, specification consortia and strategic consortia [25]. A research consortium connotes a cooperative research effort among companies, universities, industries and/or government, typically aimed at helping the participants maintain their leadership position. Specification groups such XML/EDI or OMG are essentially concerned with development of a usable, robust standard for the benefit of the industry. Strategic consortia are formed and funded by a small number of companies for their individual benefit in order to promote the adoption of certain technology as open technology.

Multiple theories that have been suggested within the areas of economics, sociology, and management can be used to analyze standardization processes [26], [27], [28], [29], [30], [31], [32], [33]. While such models are analytically compelling, their weakness is that they lack the concepts needed to identify and analyze the standardization process in inter-related ICT standards and standardization outcomes. Moreover, they fail to describe the emergent features of standardization that are part of the standard setter's experience [34] and lack the concepts required to understand the internal logic through which the process becomes enacted and subsequently results in specific outcomes. Most theoretical research on standardization research has not sufficiently explored this issue and ignore the actual processes of standard setting [35], [1]. While social-technical network literature [28], [30] provides a processual account of the building of the socio-technical network related to standardization actors, it does not account for rationalities involved in actors' decision making. Hence, there is a need to theorize standardization studies by drawing upon multiple theoretical disciplines.

3. RESEARCH METHODOLOGY

Since this research aims to understand the process of standards formation, a longitudinal qualitative analysis was conducted. We adopted a process theory approach as they also offer the scope for generalisability and prediction. The unit of analysis was a particular standard i.e. SOAP, UDDI and WSDL. Process theories focus on sequences of activities to explain how and why particular outcomes evolve over time [21], [22]. Practitioners value process theories as they are easier to understand and are high in relevance [22]. In considering our research questions, process research is found to be more suitable to investigate the phenomena of standards formation. Similar to previous research [36], we attempt a process model involving antecedent conditions, encounters, episodes, and outcomes during standards formation. This methodology has the challenge that the events need to be presented objectively.

The data sources for our methodology were 1. *technical notes of standard bodies* (OASIS, W3C, IETF etc.), 2. *research forums* (IBM Developerworks etc.), 3. *analyst reports* (Zapthink, Forrester and Gartner), 4. *magazine articles* (O'Reilly Web services, XML.com, Infoworld, XML Journal, Webtechniques, Computer Weekly, JavaWorld, Intelligent Enterprise, TechWeb, XML Magazine, MSDN Magazine, CNET News, Web Services Journal, Network World, Web services.org, Computer Reseller News, The Register, Internetweek, Software Development Magazine, XMLmodeling.com, ZDNet, CIO.com, DevXpert, Itworld.com and B2B Magazine), 5. books (Professional XML Web services) and 6. *practitioner journals* (Dr. Dobb's Journal). All these data sources are publicly available and accessible. In all we analysed 200 plus articles, analyst reports, technical reports etc. relating to SOAP (published during the time period 1999–2003), 150 plus articles, analyst reports, technical reports, etc. relating to UDDI (published during the time period 2000–2005) and 200 plus articles, analyst reports, technical reports, etc. relating to WSDL (published during the time period 2000–2007). We also accessed archives of developer discussions regarding SOAP, WSDL and UDDI. Other sources of data included company releases, and promotional material thereby achieving triangulation of sources and methods triangulation [37]. Each standard was analyzed by first preparing a visual representation of sequence of events by preparing a process map. The various events, activities and decisions were categorized and the time dimension of progression was also captured minutely. See Table 1 for the timeline of progression of the three standards.

Table 1: Timeline of progression of the three standards

Year	SOAP	UDDI	WSDL
1999	• Microsoft develops SOAP along with DevelopMentor and Userland	-	-
2000	• Microsoft submits SOAP to the Internet Engineering Task Force (IETF) for review • IBM offers support for SOAP • SOAP receives support from over 20 companies including Intel, Ariba etc. • Microsoft submits SOAP v1.1 to W3C along with Ariba, CommerceOne, DevelopMentor, HP, IBM, SAP, Userland Software etc. • IBM reference implementation of SOAP v1.1 • Release of SOAP v 1.2. by IBM • W3C forms working group for standardizing SOAP	• IBM, Microsoft, Ariba and 33 other companies team up to develop UDDI specs • UDDI Business Registry goes live • The initiative gets widespread industry support including Oracle, HP, Dell, Intel, Nortel, Sun Microsystems, Ford Motor, Webmethods etc.	• WSDL 1.0 is developed and released by IBM, Microsoft and Ariba to describe Web Services for their SOAP toolkit • IBM releases WSDL Toolkit
2001	• SOAP extension by Microsoft, HP, Webmethods • SOAP Security extensions by IBM and Microsoft • Updated IBM Web Services Toolkit v 2.2 which supports UDDI, SOAP, and WSDL • ebXML Integrates SOAP into Messaging Services Specification • Microsoft announces SOAP toolkit v2.0 • Microsoft Publishes XML Web Services specifications for review • W3C draft of SOAP 1.2 standard • Sun supports Web services standards • Microsoft Releases new XML Web Services Specifications • Microsoft submits a Web services related standard (DIME) to IETF	• IBM releases UDDI4J, an open-source Java implementation of UDDI • RosettaNet registers 83 business process standards within UDDI • UDDI registry becomes live • HP becomes registry operator • UDDI.org releases UDDI v2 • IBM offers its UDDI registry • SAP offers its UDDI registry • IBM, HP, and SAP announces support for UDDI4J	• WSDL 1.1 is published • IBM, Microsoft along with leading players, submits WSDL to W3C • IBM releases Web Services Invocation Framework (WSIF), complementary to WSDL
2002	• Amazon.com Web Services Facility Supports XML/HTTP and SOAP	• UDDI is adopted by OASIS • IBM releases UDDI registry extensions • NTT launches UDDI registry	• IBM releases WSDL Explorer Web Application • Cape Clear releases free WSDL Editor
2003	• SOAP Version 1.2 Published as a W3C recommendation	• OASIS ratifies UDDI v2.0 as an open standard	• W3C releases WSDL 1.2
2005	-	• OASIS ratifies UDDI v3.0.2 as an open standard	-
2006	-	• IBM, Microsoft, and SAP close their public UDDI nodes	-
2007	-	-	• WSDL 2.0 becomes a W3C recommendation

The collection of multiple types of data from different sources provided triangulation and increased the reliability of the study. The above process was repeated for each of the three standards viz. SOAP, UDDI and WSDL. The standards were analyzed by first preparing a visual representation of sequence of events from the founding of the firm and subsequent changes by preparing a process map. The data was classified according to 5 headings as given in Table 2. The data collected was analyzed to understand their impact on the entire process, and key themes were identified using analytic induction methods [38].

Table 2: Data Classification Criteria

1. Resource pooling formulated by the involved firms
• Standards idea mooted by a dominant firm • Standards idea mooted by a group of firms • New firms supporting and joining the standards formulation exercise • Extent of co-operation amongst organizations • Extent of competition amongst organizations • No. of dominant players • No. of dominant groups • Participation by end user industry groups • Participation by Software Integrators (SI) • Presence of meta-standard bodies
2. Creation of linkages with communities of practice and standard institutions
• Formation of a working group within the standard body • Sponsoring of Communities of Practice (CoP) • Sponsoring of Development forums • Membership in Standard setting bodies (number of memberships) • Control over standard setting bodies • Primary membership in Special Interest Groups (SIG)
3. Signaling implementation, experiments by willing adopters
• Reference implementation of the standard • Submission of a new standard to the standard body • Submission of a new standard to the working group
4. Institutionalization
• Acceptance of a standard by the working group • Acceptance of a standard by the standard body • Release of complementary standards by the dominant firms • Patent announcements by the firms who are part of the standards formulation exercise
5. Extensions • Announce extensions to standards, which are complementary to the firm's proprietary IP

The process maps along with the detailed write-ups helped analyze each standard. The collection of multiple types of data from different sources provided triangulation and increased the reliability of the study.

4. WEB SERVICES CORE STANDARDS

There are two main standard bodies dealing with Web Services standards: The World Wide Web Consortium (W3C) and The Organization for the Advancement of Structured Information Standards (OASIS). W3C played a critical role in the formation of standards such as SOAP, XML and WSDL whereas OASIS played an import role in the standardisation of UDDI. W3C mainly focuses on basic specifications right from HTML and HTTP whereas OASIS focuses on developing higher level standards. W3C has received positive comments about the robustness of its standards. Criticisms against W3C include its slow and deliberate approach towards standards setting, which may run to two to three years for the entire process from start to acceptance of the standard. The slow pace may not find takers in fast moving technology businesses. Standards formation timelines for OASIS are much shorter and may be suited for certain stakeholders. OASIS has been criticized for the lower degree of usefulness and quality of its standards.

4.1 WSDL

WSDL 1.0 was developed by IBM, Microsoft and Ariba to describe Web Services for their SOAP toolkit by combining two service description languages: NASSL (Network Application Service Specification Language) from IBM and SDL (Service Description Language) from Microsoft. WSDL defines a standard description mechanism for Web services. A WSDL document describes what functionality a Web service offers, how it communicates and where it is accessible. WSDL 1.1 was published in March 2001. WSDL 1.2, published in June 2003 is still a working draft at W3C and is not supported by most of the vendors. WSDL 2.0 became a World Wide Web Consortium (W3C) recommendation on June 2007 and WSDL 1.2 was renamed to WSDL 2.0.

4.2 SOAP

Simple Object Access Protocol (SOAP) is a XML based lightweight protocol for exchange of information in a decentralized, distributed environment. SOAP defines a mechanism for expressing application semantics by providing a modular packaging model and encoding mechanisms for encoding data within modules. SOAP provides a built-in extension mechanism that allows additional functionality, such as security and transactions, to be added to the basic transport. SOAP Version 1.2 became a W3C recommendation on June 24, 2003. SOAP was originally developed by Microsoft and the SOAP specification is currently maintained by the W3C.

4.3 UDDI

Universal Description, Discovery and Integration is a platform-independent registry for businesses worldwide to list their web services on the Internet. UDDI provides a standard discovery mechanism for Web services. UDDI uses WSDL to describe interfaces to Web services. UDDI provides an open, platform-independent service architecture framework that enables businesses for publishing services, discovering businesses offering services and integrating business services over Internet along with a registry with publicly listed information. Public UDDIs did not find industry support and in 2006, IBM, Microsoft, and SAP closed their public UDDI nodes

5. PROCESS MODEL

The process of standard creation involves the following five intertwined states: 1) resource pooling formulated by the involved firms, 2) creation of linkages with communities of practice and standard institutions, 3) signaling and implementation experiments undertaken by a set of firms that are willing to follow and embrace such solutions, 4) institutionalization and preservation of proprietary control and 5) extension. Our analysis shows that standardization processes unfold as a dynamic interplay of these five activities. Figure 1 presents the process of standard setting in Web services standards.

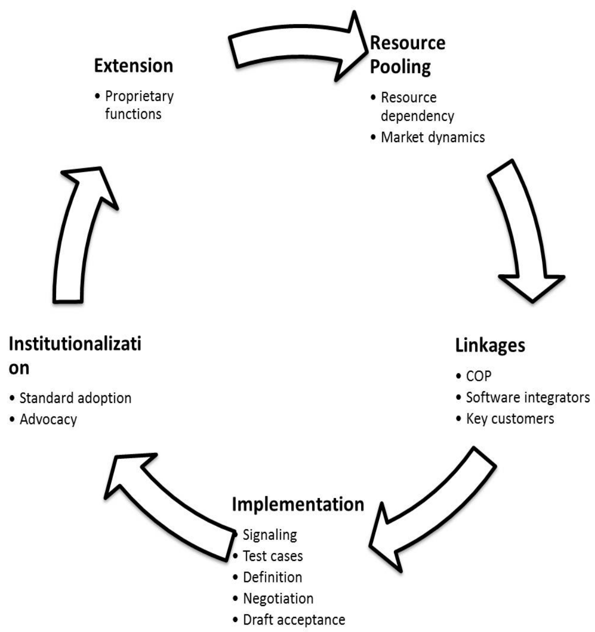

Figure 1: Process of Standards Setting

5.1 Resource Pooling

The first activity that happens is resource pooling. Resource pooling functions underlie creation of resources, platforms for member and membership development, and information exchange. Understanding of the opposite point of view at the resource pooling stage helps in establishing workable definitions of standards and good working relationships.
Resource endowment functions include a) promotion of firm specific technological efforts, and b) secure and manage funds for the promotion of technological choices. All the key firms in the Web services landscape had committed considerable resources towards Web services standards. They had pooled their resources with those of their competitors. For example, IBM and Microsoft had committed considerable resources towards SOAP, UDDI and WSDL. Microsoft went to the extent of formulating its entire Internet strategy around SOAP. The initiative saw resource pooling from not only the dominant players but also smaller firms such as Ariba, DevelopMentor, Userland Software, etc. The firms were pooling their resources and developing Web services architecture stacks. IBM co-developed the SOAP/UDDI stack with Microsoft, Ariba, etc. IBM has been aggressive in committing its resources towards the Web services initiative and was leveraging the horizontal capabilities ingrained in its software divisions to ensure a unified approach.

5.2 Linkages

In the ICT sector, system and component compatibility are the levers around which standardization strategies are pursued [39]. Since these activities need participation and acceptance of several parties including competing firms and institutions, standardization process is a social and community process. Linkage functions include a) promotion and dissemination of artifact logic, b) providing a forum for dissemination of specification, c) organize meetings, symposiums, conferences which provide a forum for presentation and discussion of the specification and draft standard, and d) establishing linkages, collaboration and coordination among other standard setting bodies, scientific academies, government scientific departments, and communities of practice.
It is critical for the dominant players such as IBM and Microsoft to create strong linkages with the ecosystem partners. The critical role of IBM in the Web services standardization process is visible in the dominant role of the firm in various standard bodies. Microsoft has been building its .Net strategy into its development tools, operating systems and applications and entered into agreements with companies like eBay so that everyone can put/list items for auction via a simple XML interface. Moreover Microsoft.Net supports open standards such as XML, SOAP, WSDL and UDDI. This is part of the Microsoft strategy to create and sustain linkages with players in the IT industry. Microsoft's strategy towards Web services standard has been one of continued openness. Microsoft adopted the approach of accommodating diverse legacy technologies and the firm's strategy was tuned to exploit the richness offered by XML.
IBM and Microsoft collaborated with Ariba, which was a leading player in the e-Marketplace landscape to come out with joint specifications for UDDI. Microsoft also collaborated with DevelopMentor and Userland Software for its SOAP specifications. Similarly, for WSDL, IBM and Microsoft worked closely with added support from Ariba. The strong COP linkage strategy followed by IBM included the firm's contribution of Java-based UDDI code to the open source community, which led to the widespread interest towards UDDI in the open source community. This helped IBM to garner support of the open-source community and use the same as a competitive advantage. In a related effort, IBM also made available to the open source community, its Eclipse code which was valued at about $40 million.

5.3 Signaling and implementation

Signaling is a mechanism available for participants to convey their degree of seriousness and commitment towards the standardization process they are part of.

Signaling can be in the form of announcements about potential new products/platforms, extensions of existing product/platforms, product/platform support for the standard etc. Implementations are in the form of reference implementations which are representative of actual usage scenarios.
Microsoft was aggressive in incorporating SOAP into its offerings such as adding the Web services toolkit to its Visual Studio 6.0. The MSDN Web Services Toolkit (WSTK) complied fully with SOAP v1.0. Similarly, HP came out with its Web Services Platform, which supported UDDI, WSDL, SOAP and ebXML. Oracle was a late entrant to the Web services standards initiative and now supports UDDI, SOAP and WSDL standards. A number of other key players such as Sybase, TIBCO, Vitria, Borland, Mercury Interactive, etc. and several smaller players such as Cape Clear Software, IONA Technologies, Flamenco Networks, etc. started supporting the basic Web services standards.

5.4 Institutionalization

International policy regimes, standard setting bodies and governments enhance the viability of a standard by signaling and supporting legislation, regulation and standardization efforts. One of the ways in which the firms collectively create and maintain these institutional legitimating devices is through industry councils, technical committees, and trade associations. Industry associations, in turn, approach, educate, and negotiate with other institutions and governmental units to obtain endorsements and develop regulatory procedures. Whatever means are used, the typical process for setting standards is influenced more by social and political dynamics than by technical considerations [40], [41]. These socio-political dynamics vary with the relative benefits to public and private parties of promoting a standard, the extent to which interested parties (producers, consumers, regulators) have different views about the standards that are chosen, as well as the evaluation complexity of a new technology [41]. According to Network effect theory, the end user value of a software platform is directly proportional to the number of users of the platform as well as the number of complementary software applications [42]. Past research using EDI and Web standard adoptions in e-commerce shows that network effects play a significant role in new standard adoptions, while adoption and switching costs can vary depending on earlier standard choices [43]. Institutionalisation can be seen in the case of UDDI, wherein four companies (IBM, Microsoft, NTT and SAP) were operating the UDDI Business registries.

5.5 Extension

Presence of network effects allow software platform firms to pursue strategies that will secure a dedicated user base, supported by complementers, who in turn attract more users [44], [45]. Complementers are firms who provide software applications which are compatible to the software platform. Direct network effects help to attract new users. Complementers trigger indirect network effects by making available useful, innovative and compatible software applications [46]. Firms involved in IT standardisation can have two pronged strategy, with the primary strategy of promoting network effects by large scale adoption by new users and the secondary strategy of enhancing value to the end users by leveraging indirect network effects by promoting adoption by complementers.
Firms have two ways to extract revenue from standard setting: primary licensing or extending proprietary control of higher-level services (layers). The leading players in the web services standardization process; IBM and Microsoft own significant intellectual property within specific Web services protocols. This gives them motivation enough to work towards extensions to standards while maintaining their proprietary rights. Microsoft has been a keen follower of the extension strategy wherein Microsoft first publicly announces its support for a standard and assigns employees to work with the standard's bodies. This is followed by partial/full support for the standard while at the same time adding proprietary extensions which work only with Microsoft interfaces. As Microsoft enjoys a dominant position in server products and developer tools, the increased use of proprietary extensions results in the Microsoft version to be the dominant one.

6. DISCUSSION

The process framework espoused above draws upon and integrates several separate lines of inquiry into a dynamic analysis of standardization processes. A process theory of standardization, outlined above enables us to theorize over standardization processes and their outcomes. Our process analysis suggests that the process of standard creation involves five intertwined states and the standardization processes unfold as a dynamic interplay of these five activities, albeit not in a linear-fashion.
The paper is one of the first efforts to analyse the process of IT interoperability standards formation involving inter-related standardization efforts progressing in parallel. The stakeholders in the process not only involved the dominant firms of the industry, but also involved numerous end users of the standards as well as technology implementation partners. The standardization efforts of SOAP, UDDI and WSDL were progressing in parallel with the two dominant firms; IBM (bridge firm with interest in both open source Java and proprietary groups), and Microsoft (star firm) playing very important roles in all three standardization efforts. Across all three standards we find that IBM and Microsoft are playing a dominant role in not only proposing the standards, but also in deciding their paths of evolution and final adoption.
Our study makes several important contributions to the literature on standard setting in general and IT interoperability standards in particular. First, our results allow for an understanding of the nature of standard definition, selection, adoption and institutionalization, indicating that a particular specification may be selected due to not only transaction efficiencies but also because of

resource and existing technology path dependencies. The results indicate standard setting tactics are influenced by prior relationships, more closely firms work on different technical committees, more likely they will come together for a collaborative standard setting, thus supporting network theory. Maintaining proprietary control was important for firms in extensions and later stages of standardization, thus influencing firms' decisions related to licensing agreements. External factors such as COP play a significant role in defining the standards and the standard setting process. Finally, our study enhances our understanding of how inter-related technology standard strategies are pursued by dominant firms. Star and Bridge firms seem to agree for public ownership of basic layers, while they could enforce proprietary control over extension or emerging top layer. This should be a cause of concern for open source evangelists.

The proposed model opens new research avenues on several fronts. First, we need to extend the analysis of activities to other standardization processes. Studies could examine the generalisability of the proposed model by examining the standardization efforts in other Web services standards such as WS-orchestration, WS-security, etc. Further, the research context could be extended to include both the reactions of standard-setting bodies, and firms to identify the factors, the terms of negotiations and common trade-off for a particular technology selection. Another extension of the study worth pursuing is to use alternate forms of research design and data collection. Survey based research could reveal variations in modes of standard setting process. This would allow for greater breadth of data and predictions of evolution of product technology standards. Further research should also focus on network relationship effects, especially at the level of dominant firm and bridge firm, standard-setting bodies and major sponsors, and COP. This we believe would be particularly relevant in the context of emerging IT standards.

REFERENCES

[1] David, P.A., Greenstein, S., "The Economics of Compatibility Standards: An Introduction to Recent Research," *Economics of Innovation and New Technology*, (1), pp. 3-41, 1990.

[2] David, P.A., Steinmueller, W.E., "Economics of compatibility standards and competition in telecommunication networks," *Information Economics and Policy*, (6), pp. 217-241, 1994.

[3] Lehr, W., "Compatibility Standards and Industry Competition: Two Case Studies," *Economics of Innovation and New Technology*, (4), pp. 97-112, 1996.

[4] Axelrod, R., Mitchell, W., Thomas, R.E., Bennet, D.S., Bruderer, E., "Coalition Formation in Standard-setting Alliances," *Management Science*, (41:9), pp. 1493-1508, 1995.

[5] Jorgensen, U. and Sorensen, O. "Arenas of Development: A Space Populated by Actorworlds, Artefacts, and Surprises," *Technology Analysis and Strategic Management*, (11:3), pp. 409-429, 1999.

[6] Fomin, V., Keil, T. and Lyytinen, K., "Theorizing about Standardization: Integrating Fragments of Process Theory in Light of Telecommunication Standardization Wars," *Sprouts: Working Papers on Information Environments, Systems and Organizations*, (3:1), pp. 29-60, http://sprouts.case.edu/2003/030102.pdf, 2003.

[7] Lave, J. and Wenger, E, *Situated learning. Legitimate peripheral participation*. Cambridge: Cambridge University Press, 1991.

[8] Brown, J. S., and Duguid, P. "Organizational learning and communities of practice: Toward a unified view of working, learning, and innovation," *Organization Science* (2:1), pp. 40-57, 1991.

[9] McGee, J., and Sammut Bonnici, T. A. "Network industries in the new economy", *European Business Journal*, (14:3), pp. 116-132, 2002.

[10] ITU's Telecommunication Standardization Sector (ITU-T), http://www.itu.int/en/ITU-T/ipr/Pages/open.aspx (visited on 2010-08-03).

[11] Damsgaard, J., and Lyytinen, K. "The Role of Intermediating Institutions in the Diffusion of Electronic Data Interchange (EDI): How Industry Associations Intervened in Denmark, Finland, and Hong Kong," *The Information Society* (17:3), 2001.

[12] van Baalen, P., van Oosterhout, M., Tan, Y.H., and van Heck, E. *Dynamics in Setting Up an EDI Community*, Eburon Publishers, Delft, The Netherlands, 2000.

[13] Nickerson, J. V., and zur Muehlen, M. "The Ecology of Standards Processes: Insights from Internet Standard Making," *MIS Quarterly* (30): Special Issue on Standard Making), pp. 467-488, 2006.

[14] Markus, ML, Steinfield, C., Wigand, R. and Minton, G., "Industry-Wide Information Systems Standardization as Collective Action: The Case of the U.S. Residential Mortgage Industry," *MIS Quarterly*, 30, pp. 439-465, 2006.

[15] Chen, P. Y., Forman, C. "Can Vendors Influence Switching Costs and Compatibility in an Environment with Open Standards?," *MIS Quarterly*, pp. 541-562, 2006.

[16] Weitzel, T., D. Beimborn, and W. Koenig, "A unified economic model of standard diffusion: the impact of standardization cost, network effects, and network topology," *MIS Quarterly*, 30: p. 489-514, 2006.

[17] Lyytinen, K., Fomin, V. V. "Achieving high momentum in the evolution of wireless infrastructures: the battle over the 1G solutions," *Telecommunications Policy*, 26 (3-4), pp. 149-170, 2002.

[18] Söderström, E. "Formulating a general standards life cycle," *Lecture Notes in Computer Science*, 3084, pp. 263-275, 2004.

[19] Boh, W. F., Soh, C., Yeo, S. "Standards development and diffusion: A case study of RosettaNet," *Communications of the ACM,* 50(12), pp. 57-62, 2007.

[20] Gasser, U., Palfrey, J. "Breaking down digital barriers: When and how ICT interoperability drives innovation", Cambridge MA: Berkman Center, http://papers.ssrn.com/sol3/papers.cfm?abstract_id=1033226, 2007.

[21] Mohr, L. B. *Explaining Organizational Behavior*. San Fransisco: Jossey-Bass, 1982.

[22] Shaw, T., Jarvenpaa, S. "Process models in information systems," *Proceedings of the IFIP TC8 WG 8.2 international conference on Information systems and qualitative research*, pp.70-100, Philadelphia, Pennsylvania, United States, 1997.

[23] Cargill, C. F. *A five-segment model of standardization, in Brian Kahin and Janet Abbate, eds, Standards Policy for Information Infrastructure*, Cambridge: MIT Press, 1995.

[24] Farrel, J and G. Saloner, "Standardization, Compatibility and Innovation," *Rand Journal of Economics*, 16, pp. 70-83, 1985.

[25] Updegrove, A. "Forming, Funding and Operating Standard-Setting Consortia," *IEEE Micro*, (13:6), pp. 52-61, 1993.

[26] Miller, W. L. & Morris, L. *4th Generation R&D: Managing Knowledge, Technology, and Innovation*. New York: John Wiley, 1999.

[27] Besen, S.M., and Farrell, J., "Choosing How to Compete: Strategies and Tactics in Standardization," *Journal of Economic Perspectives*, (8:2), pp. 117-131, 1994.

[28] Farrell, J. and Saloner, G. "Coordination through markets and committees," *Rand Journal of Economics*, (19), pp. 235 – 252, 1988.

[29] Williams, R. and Edge, D. The Social Shaping of Technology. *Research Policy*, (25), pp. 865- 899, 1996.

[30] Hanseth, O. Monteiro, E and Hatling, M. "Developing information infrastructure: The tension between standardization and flexibility," *Science Technology and Human Values*. (21:4), pp. 407-426, 1996.

[31] Sirbu, M., and Zwimfer, "Standards setting for computer communication: The case of X. 25," *IEEE Communications*. (23:3), pp. 35-40, 1985.

[32] Sirbu, M., and Hughes, "Standardization of local area networks," Paper presented at the *14th Annual Telecommunications Policy Research Conference*, 1986.

[33] Besen S.M., and Johnson, L. *Compatibility standards, Competition, and Innovation in the Broadcasting Industry*, RAND Corporation, Santa Monica, CA., 1986.

[34] Autio, E. "New technology-based firms in innovation networks: Symplectic and generative impacts," *Research Policy*, (26:3), pp. 263-281, 1997.

[35] Farrell, Joseph and Garth Saloner, "Economic issues in standardization," *Telecommunications and Equity: Policy Research Issues,* (eds.), Miller, J, North-Holland, Amsterdam, 1986.

[36] Newman, M. & Robey, D. "A social process model of user-analyst relationships," *MIS Quarterly*, (16:2), pp. 249-266, 1992.

[37] Patton MQ. "Enhancing the quality and credibility of qualitative analysis," *Health Services Research,* (34:5), pp. 1189-1208, 1999.

[38] Miles MB, Huberman AM. *Qualitative Data Analysis: A Sourcebook of New Methods*. Sage Publications: Newbury Park, CA, 1984.

[39] Adams, W. and J. Brock, "Integrated Monopoly and Market Power: System selling, compatibility standards, and Market Control," Quarterly *Review of Economics and Business*, 22, pp. 29-42, 1982.

[40] David, P.A., "Some new standards for the economics of standardization in the information age," in *Economic policy and technological performance*, Partha Dasgupta and Paul Stoneman (eds.), Cambridge University Press, Cambridge, 1987.

[41] Tushman, M. L. and L. Rosenkopf. "Organizational Determinants of Technological Change: Towards a Sociology of Technological Evolution," *Research in Organizational Behavior*, (14), pp. 311-347, 1992.

[42] Katz, M. and Shapiro, C. "Technology adoption in the presence of network externalities," *Journal of Political Economy* (94:4), pp. 822-841, 1986.

[43] Zhu, K., Kraemer, K.L., Gurbaxani, V. and Xu, S.X. "Migration to open-standard interorganizational systems: network effects, switching costs, and path dependency," *MIS Quarterly* (30), pp. 515-539, 2006.

[44] Shapiro, C. and Varian, H.R. Information Rules: A Strategic Guide to the Network Economy, *Harvard Business School Press*, Boston, MA, 1998.

[45] Suarez, F.F. "Network effects revisited: the role of strong ties in technology selection," *Academy of Management Journal* (48:4), pp. 710-720, 2005.

[46] Lin, L. and Kulatilaka, N. "Network effects and technology licensing with fixed fee, royalty, and hybrid contracts," *Journal of Management Information Systems* (23:2), pp. 91-118, 2006.

SESSION 7

RADIO TECHNOLOGIES AND THE FUTURE INTERNET

PERFORMANCE COMPARISON OF INTELLIGENT JAMMING IN RF (PHYSICAL) LAYER WITH WLAN ETHERNET ROUTER AND WLAN ETHERNET BRIDGE

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ABSTRACT

The very nature of Radio Frequency (RF) technology makes Wireless LANs (WLANs) open to a variety of unique attacks. Most of these RF-related attacks begin as exploits of Layer 1 (Physical – PHY) & Layer 2 (Media Access Control – MAC) of the 802.11 specification, and then build into a wide array of more advanced assaults, including Denial of Service (DOS) attacks. In Intelligent Jamming the jammer jammed physical layer of WLAN by generating continuous high power noise in the vicinity of wireless receiver nodes. In this paper, we study the threats in an Intelligent jamming Comparison with WLAN Ethernet Router and WLAN Ethernet Bridge and the security goals to be achieved. We present and examine analytical simulation results for the throughput for different scenario performance, using the well-known network simulator OPNET 10.0 and OPNET Modeler 14.5 for WiMAX Performance. IEEE 802.11b has two different DCF modes: basic CSMA/CA and RTS/CTS. Intelligent jamming, which jams with the knowledge of the protocol, the jamming describe in our paper is based on the basis of Fake AP Jamming. When we have applied same concept in WiMAX system under the influence of jamming we have received same effect of router performance.

***Keywords*—**Intelligent Jamming, Radio frequency, WLAN Ethernet Router and WLAN Ethernet Bridge OPNET, WiMAX

1. INTRODUCTION

RF interference is a common problem in WLAN system Because WLANs broadcast over the ISM (Industrial, Scientific and Medical) and U-NII (Unlicensed National Information Infrastructure bands) public bands with a limited number of available channels. As interference increases, signal quality and network availability decreases. In Frequency Jamming Attackers with high gain directional antenna attached to RF noise generators can easily jam a WLAN resulting in a Denial of Service attack. The result is dramatically reduced performance or no successful data transmissions at all on the jammed channels as valid WLAN traffic struggles with excessive noise for the limited 802.11 bandwidth. RF jamming attacks can be detected by a WLAN system by monitoring the received signal strength indication (RSSI) threshold and noise floor levels at the PHY layer as well as the CRC error count levels at the MAC layer. RF jamming can be contained through dynamic channel assignment, whereby affected APs change channels to avoid excessive noise and interference. We compare the security issue in RF (Physical) by using two different Routers this different Routers basically acts as a Aps.

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[OPNET] allows comparing both at a same time by using different scenarios. The objective is Create the four Scenarios, two for WLAN Ethernet Router and two for WLAN Ethernet Bridge. We also compare the result with and without Jamming. The paper is divided as follows: **Section 2** back Ground Information about APs. **Section 3** and **4** present the design, implementation and simulation results. **Section 5** concludes this Paper.

2. BACKGROUND INFORMATION

WLAN systems that use X.509 certificates to authenticate APs to the wired infrastructure prevent unauthorized devices from masquerading as valid APs via MAC spoofing. As with user based authentication, WLAN systems that employ X.509 certificates for AP authentication cannot be compromised by APs MAC spoofing attacks. In an 802.11 WLAN, MAC addresses are openly broadcasted over the air. The security implication is that potential attackers can sniff the air looking for valid MAC addresses associated with authorized WLAN users, APs, and even wired infrastructure components, such as switches and routers. Once detected, programs exist to spoof these addresses, whereby intruders can masquerade as a valid WLAN client or AP.

2.1 Denial of service attacks

This can be described in following ways:

Airjack – AirJack is a Linux device driver for 802.11 cards that enables raw frame injection into WLANs. The AirJack

DoS attack can be mounted against a single client or all wireless clients within a BSS. To target a single client, the attacker first spoofs the MAC address of the AP to which AirJack "victim" is connected. Next the attacker transmits de-authentication management packets to that client causing it to disconnect from the AP. To target an entire BSSID group, the attacker broadcasts de-authentication packets to all listening clients. The result of this de-authentication flood is a DoS attack to the entire group of clients. The AirJack attack exploits a loophole in the 802.11 specification which states that "stations shall not refuse a de-authentication notification".

Fake AP – Fake AP is another Linux based packet injection attack whereby 802.11 beacon frames are rapidly generated with random ESSIDs, BSSIDs, and channel assignments. The result is that clients in the vicinity of a Fake AP attack will see thousands of bogus APs. Fake AP is a program that could also be used as part of a "honey pot" to divert potential intruders from accessing restricted WLANs. As with the Airjack and void 11 attacks, WLAN systems with real-time IDS signature analysis can detect bogus beacon packets transmitted by the Fake AP program. Once detected, the WLAN system can mitigate this attack with rogue containment by isolating the offending device from the rest of the WLAN. In This paper the main attack is Fake Jamming This is introduced in two different scenario where jamming phenomena accurse.

void11 – This attack is similar to the AirJack exploit in that an attacker forges a MAC address of an AP and floods the WLAN with de-authentication packets resulting in a denial of service attack. In addition to the network BSSID flood attack, void11 can also mount a DoS attack on an AP by transmitting a flood of bogus authentication requests from random MAC addresses. The result of this attack may cause the AP to crash due to the overload of authentication requests. Intelligent WLAN systems are capable of detecting the Airjack or void11 attacks through IDS signature analysis and location tracking. Once detected, the WLAN system can mitigate these attacks by isolating the offending devices with rogue containment from the rest of the WLAN.

In addition, the WLAN system can identify the precise location of the offending device so that it can be removed from the environment.

AusCERT DoS Attack – This attack exploits the Clear Channel Assessment (CCA) function at the PHY layer and causes all WLAN nodes within range, both clients and APs, to defer transmissions of data for the duration of the attack. When under attack, any 802.11 device behaves as if the channel is always busy, preventing the transmissions of any data over the wireless network. Since the AusCert attack targets the PHY layer, WLAN systems that combine real-time RF monitoring and location tracking are capable of mitigating this attack. The RF monitoring system will detect the presence and location of a device transmitting the CCA interference so that remedial action can be taken to remove the device from the environment.

EAP Flood Attack – This attack causes DoS by generating a flood of EAPOL messages requesting 802.1X authentication. As a result, the authentication server cannot respond to the flood of authentication requests and consequently fails at returning successful connections to valid clients. WLAN systems that are capable of enforcing an authentication "exclusion list" for devices that repeatedly fail 802.1X authentication is protected against EAP flood attacks.

3. SIMULATION MODEL DESIGN AND IMPLEMENTATION

We have designed this architecture because our goal was to analyze the performance and Jamming comparison with WLAN Ethernet Router and WLAN Ethernet Bridge. The performance was based on the measurement of the following parameters: Throughput (Bit/sec), media access delay (In sec), Load in all scenarios (Bit/sec).Data dropped at particular node to check the status of jamming in (Bit/sec).

The simulation results between both the cases have been given by using the Optimized Network Engineering Tool simulation environment [OPNET]. Our Model is consists with 10 Node (Station) name rakustn and APs. Since Jammer Parameter is so important, The Jammer band base frequency 2.402(MHz), Jammer Bandwidth 1(Mbps), Jammer transmitted power powe.001 (Watt), pulse width (.001) and silence width is (1).Wireless parameter of station is also play an important role to measure these parameters. Like AP Beacon interval 0.5(sec), traffic type of service Best Effort (0) we also defined the pulse off time and on time and its description given in process model. Details parameter has been given in **Figure 5.** and **Table 1.** Jammer attributes was taken from pulsed_advanced_jammer.

Table 1. Jammer Attributes

Attribute	Value
Model	Jammer_pulsed_advanced
Altitude	10
Jammer band base frequency	2.402
Jammer bandwidth	100
Jammer transmitted power	0.001
Pulse width	0.001
Silence width	1

3.1 Description of process model

The process model is consists with Scheduling packets, they send a packet up to last with proper Duration and then scheduling the next packet to be sent. The Jammer node transmits for 0.001 Seconds continuously and then waits for a period of 1 second. The packets being sent by the jammer are not received by either the clients or the AP.These packets serve as RF noise on the channel. Details have given in **Figure 6.**

3.2 Implementation

In this research, we have divided our work into four different scenarios with the help of OPNET Modeler.

Scenario1: Here our network is encloses with jammer, WLAN Ethernet Router (APs) and ten numbers of stations. This scenario has been given in **Figure 1.**

Scenario 2: This scenario is covers with without jammer, WLAN Ethernet Router (APs) and ten numbers of stations. This scenario has been given in **Figure 2.**

Scenario 3: This scenario is comprises with jammer, WLAN Ethernet Bridge (APs) and there are ten number of station. This scenario has been given in **Figure 3.**

Scenario 4: This scenario is covers with without jammer, WLAN Ethernet Bridge (APs) and there are ten numbers of stations. This scenario has been given in **Figure 4.**

4. RESULT ANALYSIS

4.1 Throughput (Bit/sec)

We have seen from **Figure 7.** The throughput is highly affected in the case of WLAN Ethernet Router (APs) in this case the value is 1465307(Bit/sec), but in case of WLAN Ethernet Bridge (APs) the value is 1671790(Bit/sec) and In other two cases when jammer is not there throughput is same i.e 1505200(Bit/sec).So we have judged that jamming is active in this environment.

4.2 Media access delay (sec)

Now we have referred **Figure 8.** Jammer with the Media Access Delay (sec) is near about 10.15(sec) and in other case it is near about zero. That is jammer with WLAN Ethernet Router (APs highly affected.

4.3 Load (Bit/sec)

We have seen from **Figure 9.** When we are comparing Load in the case of WLAN Ethernet Router and WLAN Ethernet Bridge. In case of with jammer load is higher than without jammer. So we have judged that jammer disrupt the Communication. From simulation result the load with jammer is 3181181(Bit/sec) and without jammer is 2856016(Bit/sec).

4.4 Data Dropped At Station (Bit/sec)

Lastly we have referred **Figure 10**. The data dropped status on particular station (rakustn). We have observed that in case of WLAN Ethernet Router (APs) and WLAN Ethernet Bridge. (APs) with Jammer have dropped data packet are very high but in the case of without jammer dropped is zero.

All the above results have been implemented in OPNET Modeler 10.0 O and PNET Modeler 14.5.OPNET tools have hierarchical structures they are follows.

Network Model => Node Model => Process Model =>Source Code Model

OPNET Modeler is provides simulation up to packet level and we can change the parameter through parameter involved or can be designed through Source code.

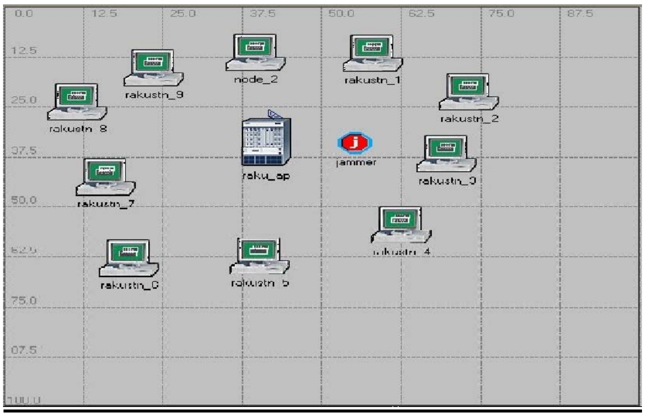

Figure 1. Jamming model with WLAN Ethernet Router

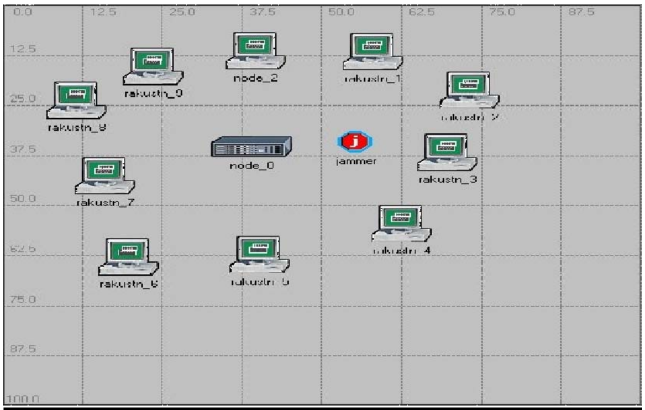

Figure 2. Jamming model with WLAN Ethernet Bridge

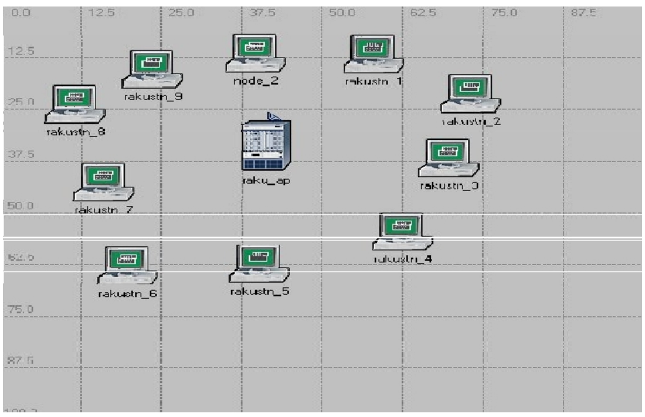

Figure 3. Without jamming model with WLAN Ethernet Router

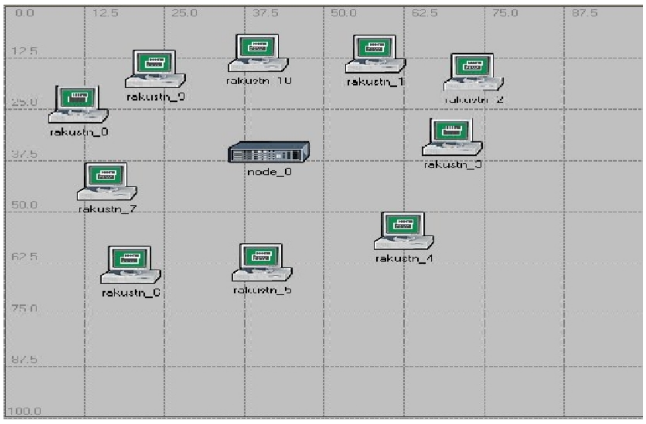

Figure 4. Without jamming model with WLAN Ethernet Bridge

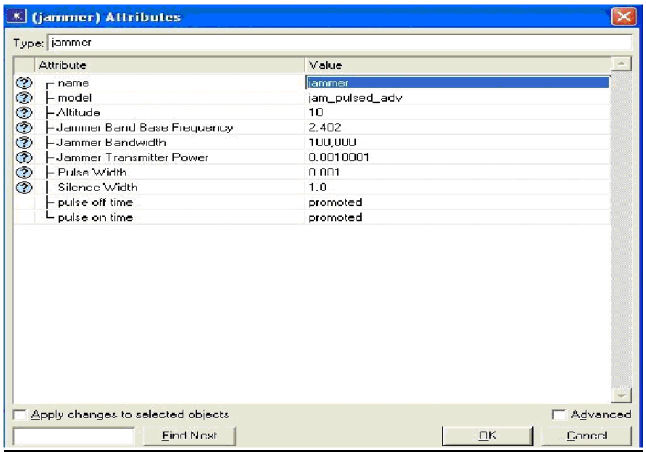

Figure 5. Jammer parameters

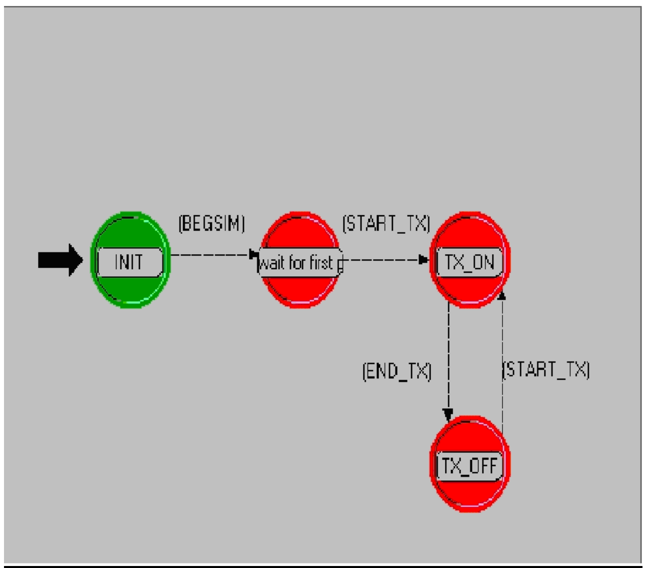

Figure 6. Process Model for jammer

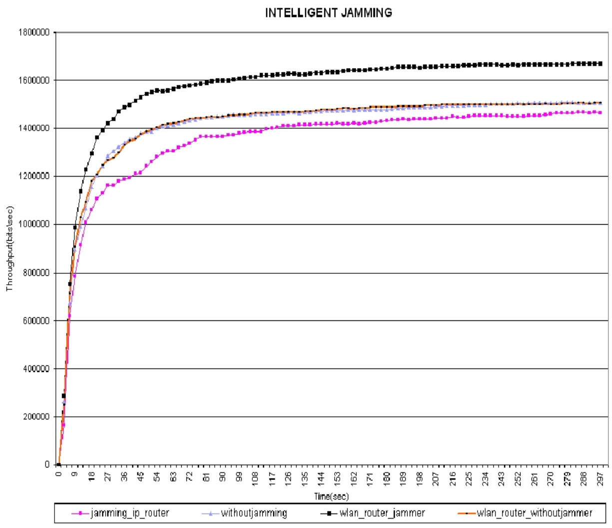

Figure 7. Comparison of average throughput of different scenarios in case of Jammer and without jammer

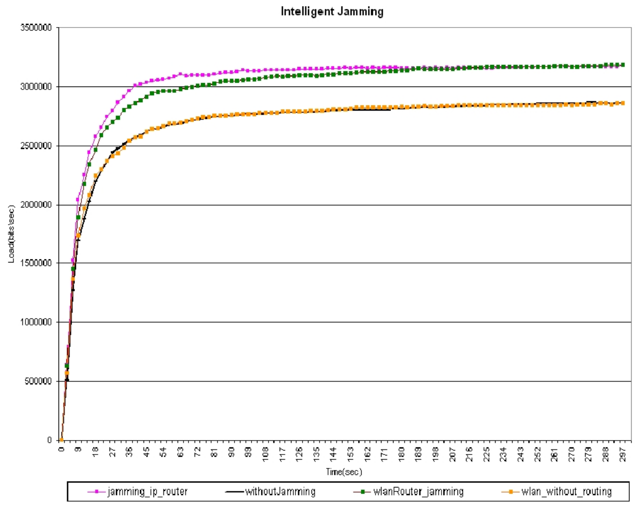

Figure 9. Comparison of average load of different scenario In case of Jammer and without jammer

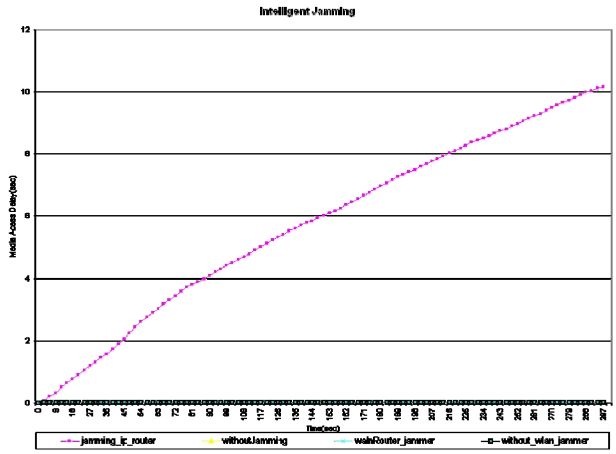

Figure 8. Comparison of average Media access delay of different scenario in case of Jammer and without jammer

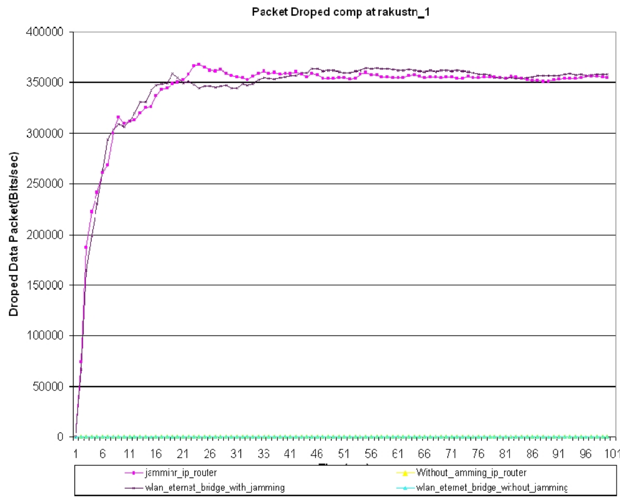

Figure 10. Comparison of average Dropped Data Packet of different scenarios in case of Jammer and without jammer

5. CONCLUSIONS

In this paper we have seen the performance of router under the influence of jamming, that can be launched in an access point based 802.11b.Our network is consist with WLAN Ethernet Router (APs) and jammer which is highly affected RF (Physical Layer).We had two observation in this research and I am giving both observation in case wise observation.

Case 1. Jammer is highly affected in the network. When we have referred our results we have observed that dropped data packet, load, and Media access delay all result was showing that jammer is highly affected in the Network.

Case 2. When we were compared all the result scenario wise we have concluded that if we will use WLAN Ethernet Bridge at the place of WLAN Ethernet router the influence of jamming attack can minimize and throughput can increase. So I am finally saying that in case of jammer we should use WLAN Ethernet Bridge.

6. RECOMMENDATIONS

We have observed from Simulation result WLAN Ethernet router is unable to perform good result for security issue so that we are recommending that the WLAN Ethernet Bridge should be preferred for security issue. We are also recommending that this concept will apply in WiMAX system under the influence of security attacks. In our view this research is very useful in defense sector that's why we are endorsing our research to our country defense sectors.

7. FUTURE SCOPE OF THE WORK

The area of security issues in ad hoc networking has many issues which need to be resolved. In this paper we have focused on router for intelligent network for security issues but still a lot of scope in future consideration for further research. In case of jamming network you can protect the network by using some algorithm and .c code inside source code of router to provide high secure wireless and cyber security network. We have observed the effect of Router in WiMAX System this was given a really good comparable result so we can follow this path in future for security environment. Since In wireless communication we are moving towards 4G so this concept may be apply for LTE, WiMAX upper layer to enhance the performance of throughput under the influence of security attacks.

8. REFERENCES

[1] IEEE Std. 802.11, “IEEE Standard for Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specification,” Edition 1999, ISO/IEC 8802-11: 1999.

[2] Acharya, M., T. Sharma, D. Thuente, D. Sizemore, “Intelligent Jamming in 802.11b Wireless Networks”, OPNETWORK 2004, August (2004) 1-10, Paper number 1689.

[3] Rakesh jha,Hardik Patel and Upena D Dalal “WiMAX System Simulation and Performance Analysis under the Influence of Jamming”,vol 1, no 1,pp 20-26, July 2010 by Scientific research journal under Wireless Engineering Technology(WET).

[4] R. Negi and Arjunan Rajeswaran, “DoS analysis of reservation based MAC protocols,” Tech. Memo Carnegie Mellon Univ., Feb, (2003) 3632-3636.

[5] W. Xu, T. Wood, W. Trappe, and Y. Zhang, “Channel Surfing and Spatial Retreats: Defenses against Wireless Denial of Service,” in 2004 ACM Workshop on Wireless Security, October, (2004) 403-404.

[6] IEEE Std 802.11b-1999/Cor 1-2001 Standard for wireless LAN medium access control (MAC) and physical layer (PHY) specifications, 2001.

[7] http://www.opnet.com/products/opnet-products.html.

[8] http://www.opnet.com/products/modeler/home-1 html.

[9] Ahmadi, M.R.; Satti, M.M.; “A security solution for Wireless Local Area Network (WLAN)” 18-20 Nov. 2007 Page(s):1 -6.

[10] IEEE Std 802.11b-1999/Cor 1-2001 Standard for wireless LAN medium access control (MAC) and physical layer (PHY) specifications, 2001.

[11] Jamie Van Randwyk, Dimitry Averin, Ryan P. Custer, Jason Franklin, Franklin Hemingway, Dominique Kilman, Erik J. Lee, Mark Lodato, Damon McCoy, Kristen Pelon, Amanda Stephano, Parisa Tabriz, Eric D.Thomas “Intrusion Detection And monitoring For Wireless Networks”, November,(2005) 21-29.

[12] Jha R. K, Kosta, Y.P., Dalal, U.D., “Security Comparison of Wired and Wireless Network with Firewall and Virtual Private Network (VPN)”2010 International Conference on Recent Trends in Information, Telecommunication and Computing, pp 281-283, 12-13 March 2010 .

[13] Rakesh Kumar Jha, Suresh Limkar and Dr. Upena Dalal “A Performance Comparison of Routing Protocols(DSR and TORA) for Security Issue In MANET(Mobile Ad Hoc Networks) ” IJCA Special Issue on MANETs (2):78–83, 2010. Published by Foundation of Computer Science.

Self-organized Spectrum Chunk Selection Algorithm for Local Area LTE-Advanced

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ABSTRACT*: -*

This paper presents a self organized spectrum chunk selection algorithm in order to minimize the mutual inter-cell interference among Home Node Bs (HeNBs), aiming to improve the system throughput performance compared to the existing frequency reuse one scheme. The proposed algorithm is useful in Local Area (LA) deployment of the Long Term Evolution-Advanced (LTE-A) systems, where the HeNBs are expected to be deployed randomly and without coordination in distributed manner. The result shows that the proposed algorithm effectively improves the system throughput performance with very limited signaling exchange among the HeNBs.

Keywords: - Frequency Reuse, Local Area, LTE-Advanced, self-organized.

I. INTRODUCTION

The LTE-A which is generally recognized as the evolved Long Term Evolution (LTE) system, aims to provide high capacity for improving user experience [1]. It is also expected to be flexible in terms of deployment. The Local Area deployment scenarios such as in home or office environment provide services to users in a limited geographical area [2]. This deployment scenario has been considered as an important research area in the latest International Mobile Telecommunications-Advanced (IMT-Advanced) workshop [3]. Keeping this in view it becomes important to investigate techniques to be suitable for such deployment scenarios in order to improve the throughput performance.

Fixed frequency reuse schemes have been considered as effective means to improve throughput performance for many systems by reducing inter-cell interference [4, 5, 6]. In traditional Wide Area (WA) deployment scenarios, this is achieved by a proper network planning, including frequency reuse plans, base station location, controlling transmit power levels and antenna radiation characteristics. In contrast to such scenario the deployment of HeNBs for LA of LTE-A cannot be planned before hand by the operators, because of the random and uncoordinated deployment, since the owner of the each HeNB device will be responsible for the deployment. Hence, such network planning is absolutely not feasible. Therefore the network has to operate in random deployment scenario. It is also natural to envision the lack of coordination between operators providing services over the same geographical area in distributed manner. Moreover, it is also expected that such deployment scenario will lead to a possible environment where all HeNBs share the available radio spectrum based on certain physical layer mechanisms and higher layer policies. Keeping these aspects in view, a mechanism is required to assign spectrum in self organized manner to the HeNBs deployed in the local area. The algorithm should help HeNBs to adapt to the suitable set of the spectrum chunks in order to minimize the mutual interference and improve the system throughput performance.

In the previous study [11] it has been outlined that the fixed frequency reuse scheme with reuse factor two helps to achieve the highest throughput performance compared to all other fixed frequency reuse schemes in the LA deployment scenario. However, the frequency reuse two, like other schemes requires beforehand network planning for spectrum assignment in order to minimize the mutual interference. As an example, fig. 1 shows an optimal frequency plan with reuse two in an indoor office environment with 4 cells, where different colors are used to represent different spectrum chunks. Here a chunk is defined as the part of the available spectrum band for allocation to a HeNB, following a frequency reuse scheme. In our example reuse two has been considered, hence only two spectrum chunks are shown here. In this figure the x and y dimensions have been indicated in meters.

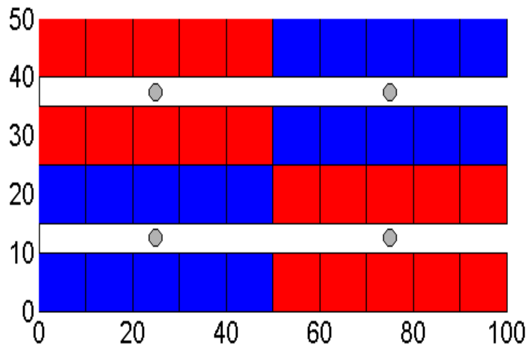

Fig. 1. Spectrum allocation for frequency reuse two with optimal plan.

When we can plan a network beforehand, this configuration can be easily achieved. However in the assumed LA scenarios of LTE-A such planning is not feasible, because of random and uncoordinated deployment. Therefore a mechanism is required to be developed for such scenario in order to assign spectrum in self organized manner and achieve nearly the same performance as is achievable with the planned network deployment. Keeping this in view, a Self-organized spectrum chunk selection has been proposed in this paper. The algorithm starts with random chunk selection for each cell, which may lead to one of the following configurations, considering four cells. In this figure the x and y dimensions have been indicated in meters.

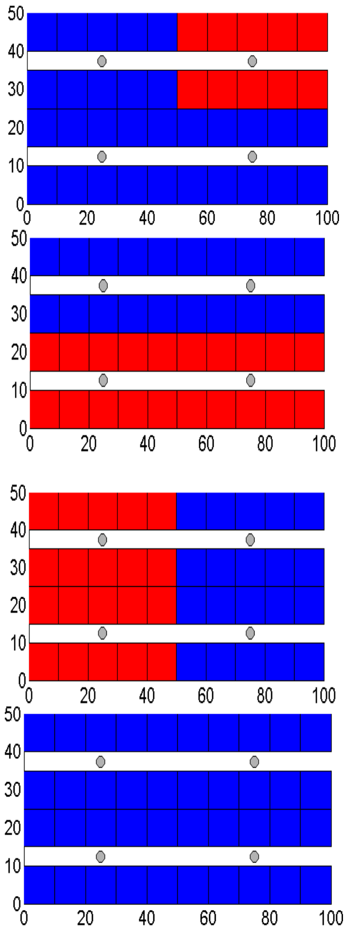

Fig. 2. Possible spectrum allocation for frequency reuse two with random chunk selection

The aim of the proposed algorithm is to tend to reach to the optimal spectrum assignment as shown in the figure 1 from any of the possible configuration obtained in figure 2, due to selection of chunks in random manner. The details description of the algorithm is given in section II.
The rest of the paper is organized as follows: Section II describes in detail the proposed spectrum selection algorithm; Section III describes the considered scenario, assumptions and used methodology. Section IV presents the result obtained by the proposed algorithm under different traffic conditions, compared with different frequency reuse schemes. The conclusions are provided in section V.

II. ALGORITHM DESCRIPTION

The proposed self organized spectrum chunk selection algorithm aims to achieve the performance close to the planned frequency reuse two scheme. Therefore, we consider the whole spectral bandwidth to be divided in two spectrum chunks of equal size. However, this algorithm can be suitably used to give performance close to any other frequency reuse schemes. The flow chart for the algorithm is shown in fig. 3. The algorithm works in the following steps.

Step 1: Initialization phase
As a HeNB is powered on, at first, it randomly selects a spectrum chunk to establish the communication with its users. In this step a HeNB is also initialized with a sequence number and a switch over number. The sequence number determines its position in the queue of HeNBs, and thereby controls the turns of the HeNBs to update their chunk selection. We considered that each HeNB has to wait for its turn to update the spectrum assignment and while waiting for its turn it keeps the previously selected chunk for communication. The switch over number is used to enforce a HeNB to switch over to another spectrum chunk selection in order to avoid the dead lock situation, in which all the HeNBs may become stable over the same non-optimal spectrum assignment. In fact the switch over number indicates the maximum number of times a HeNB is allowed to retain the same spectrum chunk selection. Whenever this number reaches a pre-defined value, the HeNB switches over to the other chunk selection. After this the switch over number is again initialized
.

Step 2: Condition to update the chunk selection
After the initialization phase is over, the HeNB reduces its sequence number by one in each update interval, until it reaches 0, which indicates the turn for the HeNB to update its spectrum chunk selection. After this the sequence number is reset to the pre-specified value.

Step 3: Interference consideration for chunk selection
At each turn, the spectrum chunk selection is based on the information available to compare the level of interference over the two chunks. The uplink (UL) Receive Interference Power (RIP) over a chunk is considered as the interference measurement yard stick. The UL RIP is a well established LTE feature, which is defined as the total power received over the spectrum chunk by a base station.

The following expression is used to determine the spectrum chunk selection

$$\text{Abs}(RIP1 - RIP2) > P\_Threshold.$$

Where, RIP1 an RIP2 indicate the interference levels on the first and the second chunk in dB. Abs indicates the absolute values. The absolute values of the difference of two is compared against the specified threshold (P_Threshold), which ensures a sufficient amount of difference in the interference level of the two chunks before making a decision for chunk selection. This interference threshold may be a tradeoff with regards to how much interference can be tolerated against the increased signaling requirement in case of switching over to the another chunk. Therefore there is a consideration that a HeNB will not switch over the other chunk, unless there is a significant gain for doing so.

If the difference is very small between the two chunks (i. e. < P_Threshold), then the HeNB will not switch to the other chunk for a specified number of times, specified by the switch over number. But when the maximum possible value for the switch over number is reached then the HeNB is forced to switch over to the other chunk. In this step even if the spectrum chunk currently in use may be more suitable, even then a changeover is forced by the switch over number, this helps to avoid the dead lock situation. This step increases the possibility for more appropriate spectrum chunk selection in the subsequent steps.

Step 4: Switching over to other Chunk

If the difference in the RIP of the two chunks is very significant (>P_threshold), the eNB will choose always the chunk with minimum RIP.

The proposed self organized spectrum assignment algorithm has the following features:

1. It operates based on the UL RIP measurement, which a currently used feature in LTE system
2. It works in a random and uncoordinated deployment scenario with very little requirement for the signaling exchange among HeNBs.
3. It allows the HeNBs in sequencer, one after the other , for spectrum chunk selection thereby it avoids the complexity in interference assessment at the time of spectrum allocation.
4. It provides a scalable solution. However, when the number of HeNBs is very high, a long convergence time is required, as only one HeNB is allowed to change its allocation at one time. However, during this convergence time, the transmission within each HeNB still continue.
5. The performance of this algorithm is upper-bounded by the performance of fixed frequency reuse two scheme, since only two spectrum chunks have been considered for this algorithm.
6. The assigned spectrum is used for both Uplink (UL) and Downlink (DL) transmissions.

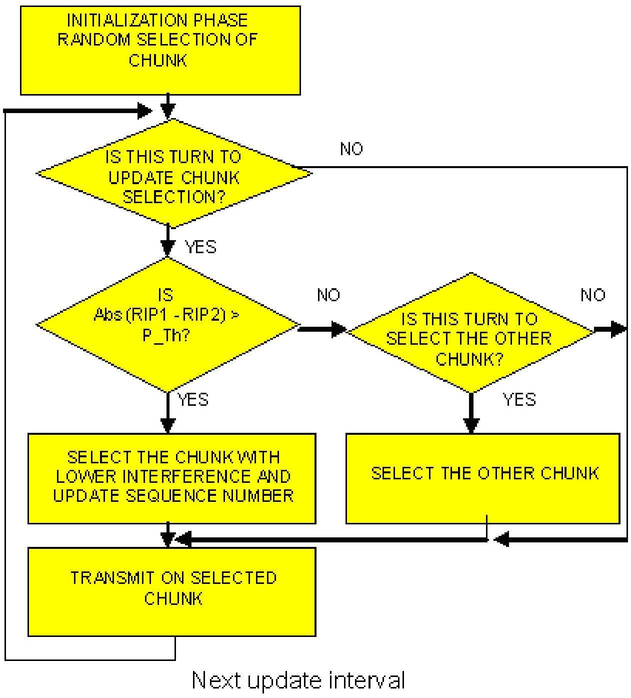

Fig. 3. Flowchart of the Self-organized Spectrum Assignment Algorithm

III. SCENARIO DESCRIPTION, SIMULATION METHODOLOGY AND ASSUMPTIONS

The indoor LA office scenario described in [7] has been used to model the scenario. It contains 4 HeNBs, located at the center of each cell. Each HeNB provides services for the coverage area of 50 meter X 25 meter, consisting of 10 rooms, as shown in fig. 1.The performance has also been examined, when the location of HeNBs are randomized, i. e. the locations of the HeNBs are not assumed at the centers of the cells.

In order to investigate the performance, the snap-shot based simulation method has been used where,

1. within each snap-shot, the cell layout is generated according to the scenario,
2. users are generated with uniformly distributed locations,
3. Signal to Interference and Noise Ratio (SINR) is calculated according to the received signal power and interference level,
4. Throughput is obtained by mapping SINR according to the LTE link-level capacity [8].
5. A few thousands of snap-shots are simulated to get the averaged performance.

As discussed in Step 4 above, for a SINR value within $[SINR_{\min}, SINR_{\max}]$, the capacity has been estimated using [8], given as

$$S = BW_{eff} * \log_2(1 + SINR / SINR_{eff})$$

(1)

Where, S is the estimated spectral efficiency in bps/Hz. For a Single Input Single Output (SISO) system, if SINR is less than $SINR_{\min}$, then $S=0$ is considered and if SINR is larger than $SINR_{\max}$ then S=*5.4* is considered. The BW_{eff} and $SINR_{eff}$ indicate the effective bandwidth and the effective SINR respectively. The assumed values for the above said parameters are given in following table 1.

Table 1. The Assumed Tthrough Mapping Parameters for LTE Link Level Capacity [8].

	BW_{eff}	$SINR_{eff}$	$SINR_{\min}$	$SINR_{\max}$
DL	0.56	2.0	-10 dB	32 dB
UL	0.52	2.34	-10 dB	35 dB

The following metrics are used for the evaluation of the system performance:

1. *Average cell throughput*: This is the cell throughput averaged among all the simulated cells.
2. *Cell edge user throughput*: This is the 5% user outage throughput, which can be obtained by sorting the throughput for all users and taking the one corresponds to the 5% Cumulative Distribution Function (CDF) value.
3. *Chunk selection interval:* This is the time period for one chunk selection operation. It contains an integer number of transmission frames.

To simplify the computations and to mainly focus on the performance evaluation of the proposed algorithm, some simplifications are assumed, such as no power control has been exercised, the Round Robin mechanism has been used for frequency domain scheduling, and fast fading has not been considered.

IV. PERFORMANCE RESULTS

The performance evaluation of the proposed algorithm is carried out based on the LTE specifications [7, 9, 10], with parameters summarized in Table 2.

Table 2. Parameters and Assumptions for System Level Evaluation [7,9, 10].

PARAMETER	**SETTING/DESCRIPTION**
Spectrum allocation	100 MHz at 3.5 GHz
Access scheme	DL: OFDMA UL: SC-OFDMA
Duplexing scheme	TDD
UEs per cell	5 ~ 10 UEs
HeNB characteristics	
Total TX power	24 dBm
Antenna system	“Omni-directional”, 3 dBi gain
Receiver noise figure	7-9 dB
Minimum Coupling Loss	45 dB
UE characteristics	
TX power	24 dBm
Antenna system	“Omni-directional”, 0 dBi gain
Receiver noise figure	9 dB
Propagation model	
Room size	10x10 m
Corridor width	5 m
Internal walls	light attenuation, 5dB
Path loss model	Line of Sight (LOS): 18.7 $\log_{10}$ (*d[m]*) + 46.8 + 20$\log_{10}$ (f[GHz]/5.0) None Line of Sight (NLOS): 20 log10 (d[m]) + 46.4 + nw · Lw + 20log10 (f [GHz]/5.0) where d = direct-line distance [m], f = carrier frequency [GHz], n_w = number of walls between transmitter and receiver, L_w = wall attenuation [dB]
Standard deviation of Shadow fading	LOS: 3 [dB] NLOS: 6 [dB]

A. Downlink average cell throughput

Fig. 4 shows the performance of Downlink (DL) average cell throughput of the proposed algorithm compared with the fixed frequency reuse 1, 2 and 4 schemes. It is observed that the proposed algorithm presents improved performance compared to fixed frequency reuse 1 and 4 schemes. However, its performance is slightly lower compared to reuse 2 schemes. It is important to note that the fixed frequency reuse 2 scheme undergoes beforehand network planning for optimal performance. It is encouraging to have a very close performance in self organized manner in random and uncoordinated deployment scenario.

B. Downlink cell edge user throughput

Fig. 5 shows the DL cell edge user throughput performance. It can be seen that the performance of the proposed algorithm is significantly higher compared to frequency reuse 1 scheme (in the order of 200 % ~ 300 %). We observe nearly the same performance as reuse 4 scheme, however, a lower than reuse 2 scheme is realized.

C. Convergence time

As mentioned before, this algorithm requires some time to converge to the nearly optimal frequency plan. Fig. 6 shows the performance of the algorithm at different times. It can be seen that the performance is stabilized after 10 selection intervals, with significant improvement.

D. Performance in Uplink

With the same chunk allocation as DL transmission, similar performance has also been realized in UL.

E. Performance with randomized HeNB location

The results so far shown are with respect to the fixed HeNB location. However, in reality, it may be difficult to control this location, especially in home scenarios where the owner is assumed to have the main responsibility for HeNB deployment. Keeping this scenario in view, this becomes important to evaluate the performance with random locations of the HeNBs. The result of DL average cell throughput is shown in Fig. 7, where it can be seen that nearly the similar performance is realizable.

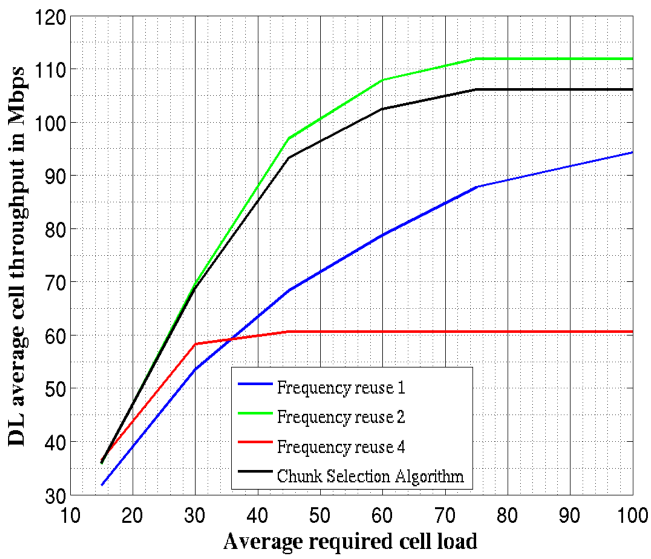

Fig. 4. Comparison of DL Average Cell Throughput

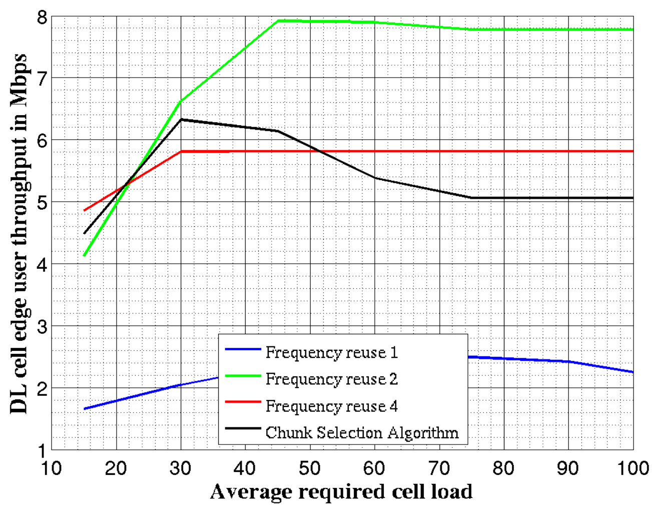

Fig. 5. Comparison of DL Cell Edge User Throughput

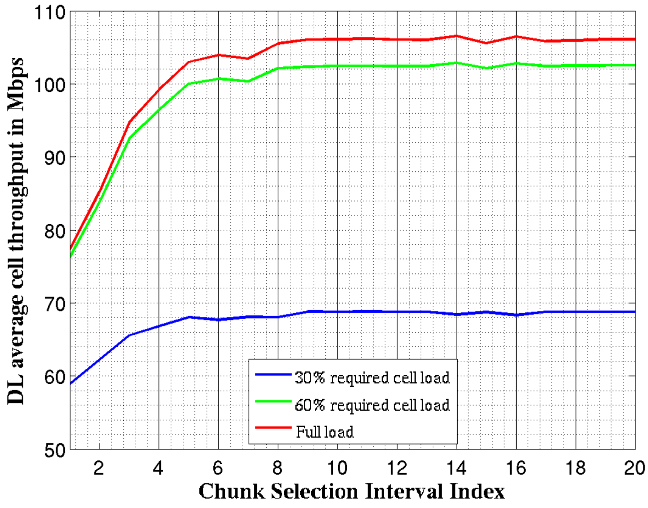

Fig. 6. Convergence Time of Algorithm

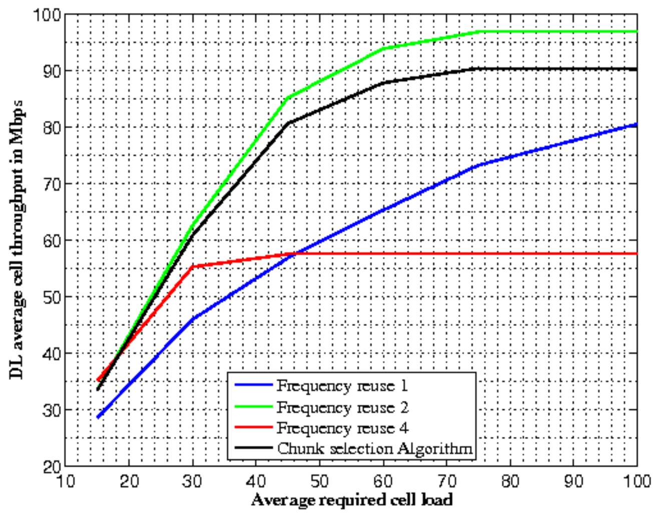

Fig. 7. DL Average Cell Throughput with Randomized HeNB locations

V. CONCLUSION

In this paper, we proposed a mechanism for self organized spectrum chunk selection algorithm. The proposed algorithm aims minimize the mutual inter-cell interference in order to improve the system throughput performance for LA deployment of HeNBs assuming LTE-advances systems deployment scenario. It works with very limited signaling exchange among the HeNBs and requires no additional measurements than what is already established for the LTE. These features make it suitable for LA deployment scenarios, where the optimal frequency plan is infeasible to realize because of the expected random and uncoordinated large scale deployment of HeNBs. Along with the features of scalability and operation simplicity, the algorithm approaches the performance of fixed frequency reuse 2 scheme and provide much better performance than other frequency reuse schemes, such as reuse 1 and 4.

REFERENCES

[1] H. Murai, M. Edvardsson and E. Dahlman, "LTE-Advanced – The Solution for IMT-Advanced", ERICSSON, 2008.

[2] IEEE 802.11 B. P. Crow, I. Widjaja, J.G. Kim, P. T. Sakai, "Wireless Local Area Networks", IEEE Communications Magazine, September 1997, pp. 116-126.

[3] ETSI MCC, "Report of 3GPP TSG RAN IMT-Advanced Workshop", April 7-8, 2008.

[4] Arne Simonsson, "Frequency Reuse and Intercell Interference Co-ordination in E-UTRA", IEEE VTC2007-Spring, pp. 3091-3095

[5] R. Giuliano, C. Monti and P. Loreti, "Wireless Technologies Advances for Emergency and Rural Communications - WiMAX Fractional Frequency Reuse for Rural Environments", IEEE Wireless Communications, June 2008, pp. 60-65

[6] T. S. Rappaport and R. A. Brickhouse, "A Simulation Study of Urban In-building Cellular Frequency Reuse", IEEE Personal Communications, February 1997, pp. 19-23

[7] IST-4-027756 WINNER II, D1.1.2 "WINNER II Channel Models part I- Channel Models", Sept 2007.

[8] P. Mogensen, W. Na, I. Kovács, F. Frederiksen, A. Pokhariyal, K. Pedersen, T. Kolding, K. Hugl and M. Kuusela, "LTE Capacity compared to the Shannon Bound", IEEE VTC2007-Spring, pp. 1234-1238.

[9] TR 101 112 V3.2.0 (1998-04), *Technical Report,* Universal Mobile Telecommunications System (UMTS); Selection Procedures for the Choice of Radio Transmission Technologies of the UMTS (UMTS 30.03 version 3.2.0)

[10] 3GPP TR 25.814 Technical Specification Group Radio Access Network; Physical Layer Aspects for Evolved UTRA, V7.0.0 (2006-6).

[11] Sanjay Kumar, "Techniques for Efficient Spectrum Usage for Next Generation Mobile Communication Networks: An LTE and LTE-A case Study" a PhD thesis at Aalborg University, Denmark, June, 2009 (ISBN 978-87-92328-29-8).

ON THE DESIGN OF ULTRA WIDE BAND ANTENNA BASED ON FRACTAL GEOMETRY

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ABSTRACT

This paper presents ultra wide band circular fractal antenna. The antenna has been fed with coplanar waveguide (CPW) feed. This fractal antenna has been designed and fabricated on FR4 substrate ε_r = 4.3 and thickness h = 1.53 mm with initial diameter of solid circular disc 15 mm. The experimental result of circular fractal antenna exhibits the ultra wide band (UWB) characteristic from 3.295 GHz to 13.365 GHz corresponds 120.88 % impedance bandwidth. The first resonant frequency of fractal antenna shifted to 3.75 GHz in comparison to first resonant frequency 4.31 GHz of conventional simple circular disc monopole antenna. This indicates the size reduction of antenna. The measured radiation pattern of this fractal antenna is nearly omni-directional in azimuth plane throughout the band. This type of antenna can be useful for UWB system and sensing applications.

Keywords: Circular microstrip antenna, Monopole antenna, Multi-band antenna, UWB fractal antenna, multiband, Resonant frequency and CPW-feed.

1.INTRODUCTION

The ultra wideband (UWB) system have become emerging research topic in the field of modern wireless communications since Federal Communication Commission (FCC) band 3-1 to 10.6 GHz declared in Feb. 2006. The UWB system required the UWB antenna of unique features such as transmitting and/or receiving electromagnetic energy in shorter durations and avoiding frequency dispersive and space dispersive [1]. Several schemes have been suggested in recent years for designing the ultra wide band antennas. Some UWB antennas are much more complex than other existing single band, dual band and multi-band antennas . Planar microstrip and fractal antennas have been rapidly developed for multiband and broad band in high data rate systems known as wide band communication systems. A fractal antenna can be designed to receive and transmit over a wide range of frequencies. The applications of fractal shapes are reduced size, multiband and wideband antennas. In open literature several multiband and wide band antennas has been developed [2-5]. Sierpinski gasket and Sierpinski carpet have been reported [2-3]. For wide band applications, several UWB fractal antennas have been reported [4-5]. These antennas have been fed directly with the coaxial probe which required very large ground plane [2-3]. These antennas are also fragile and not suitable to integrate with the Microwave integrated circuits (MIC) and monolithic microwave integrated circuits (MMIC). The J. Liang et. al. [6] has reported the conventional circular disc monopole antenna with CPW-feed. The coplanar waveguide has been reported that offer attractive advantages over conventional Microstrip feed lines as it has lower dispersion characteristics at higher frequencies, broader impedance bandwidth, unipolar configuration and ease of integration with active devices. Recently various UWB fractal antennas have been reported for UWB applications [7-11]. In [7], Crown square microstrip antenna is proposed to reduce the size. The frequency notched ultra-wideband microstrip slot antenna with a fractal tuning stub is proposed to achieve frequency notched function [8-9]. Ding et. al. [10] have proposed a new UWB fractal antenna by adopting the fractal concept on the CPW-fed circular UWB antenna. Ji- Chyun et.al. [11] has reported UWB fractal antenna of bigger in size which is directly fed with coaxial probe and not suitable for integration with MMIC/MICs. In this paper, circular fractal antenna with CPW-Fed is presented for UWB characteristics. This antenna exhibits the properties like miniature size, low resonance, wideband phenomenon and omni-directional radiation pattern. This antenna has advantages such as light weight, low profile, low cost, ease of fabrication, easy to integrate with RF devices and MIC/MMICs. A detail parametric study of antenna has been done with respect to design parameters. The antenna has been characterized experimentally in term of impedance bandwidth, radiation pattern and gain.

2.ANTENNA DESIGN

In this paper, ultra wide band circular fractal antenna has been proposed. The UWB printed circular fractal antenna is useful for wireless communication system for high data rate transmission at low power level. The fractal antenna has been constructed based on Descartes circle theorem [12]. First, monopole antenna with CPW fed of 15 mm diameter has been constructed as shown in the Figure 1a. This is called as 'initiator'or 'zeroth iteration'. The fractal antenna resulted from each iteration have been constructed with the help of Descartes circle theorem [12]. The iterative stage of initial dimension has been given in Table 1. In the first iteration, two inner circles have been taken each of radius 3.673 mm. The radii of these inner circles were determined by dividing the radius of original circle by 2. Now these two circles are subtracted from the original circle of radius 7.5 mm. This is called 1^{st} iteration as shown in Figure 1b. In the 2^{nd} iteration, two circles have been taken each of radius 2.45 mm. The radii of these circles were determined by dividing the radius of original circle by 3. Now these two circles are subtracted from the original circle of radius 7.5 mm. This is called 2^{nd} iteration as shown in Figure 1c. In the third iteration, four inner circles have been taken each of radii 1.16 mm. The radius of these circles is determined by dividing the radius of original circle by 6. Now theses four circles are subtracted from the circle of radius 7.5 mm. This is called third iteration as shown in Figure 1d. In the fourth iteration, again four inner circles have been taken each of 0.62 mm radii. The radius of these inner circles is determined by dividing the radius of original circle by 11. Now these four circles were subtracted from the original circle of radius 7.5 mm. This is called fourth iteration as shown in Figure 1e. In the fifth iteration, again four inner circles have been taken each of radius 0.47 mm. The radius of these circles is determined by dividing the radius of original circle by 14. Now these four circles are subtracted from the circle of radius 7.5 mm. This is called fifth iteration as in Figure 1f. This process can be repeated up to infinite iteration. Practically infinite iterative structure is not possible because of fabrication constraints. The fifth iterative fractal antenna has been finalized to design on the same substrate dielectric constant and thickness as conventional microstrip monopole antenna. This antenna has been fed with the coplanar feed. The CPW-fed and radiating elements both are printed on the top side of a low cost FR-4 substrate with dielectric constant $\varepsilon_r = 4.3$, h = 1.53 mm and tan $\delta = 0.02$.

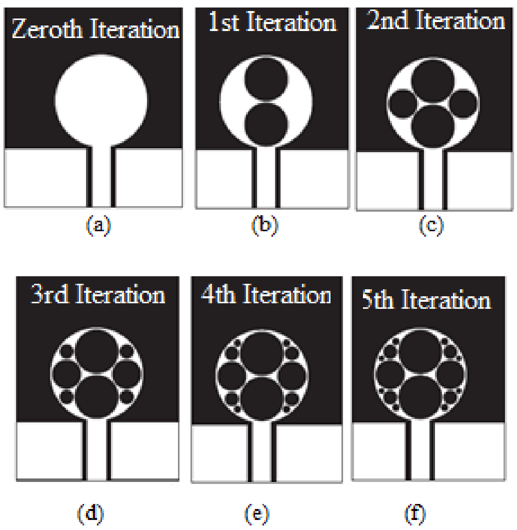

Figure 1 Circular fractal antenna with each iteration

Table 1 Dimension of Each iteration based on Descartes Theorem [12]

Initiator	0^{th} Iteration	1^{st} Iteration	2^{nd} Iteration	3^{rd} Iteration	4^{th} Iteration
no. of circles	1	2	2	4	4
Radii	1	1/2	1/3	1/6	1/11

3.DESIGN OF CIRCULAR MICROSTRIP ANTENNA

The design expression of simple circular microstrip antenna [13] for calculating the resonant frequency is given as

$$f_r = \frac{1.841 v_o}{2\pi r_{eff}\sqrt{\varepsilon_{eff}}} \tag{1}$$

Where v_o is the velocity of light. The effective radius r_{eff} can be calculated by following expression

$$r_{eff} = r_o\left[1 + \frac{2h}{\pi r_o \varepsilon_{eff}}\left\{\ln\left(\frac{r_o}{2h}\right) + (1.41\varepsilon_r + 1.77) + \frac{h}{r_o}(0.268\varepsilon_{eff} + 1.65)\right\}\right]^{1/2} \tag{2}$$

Where r_o is radius of the circular patch. The dimension of the simple solid circular patch is taken as radius 7.5 mm. This patch has been designed on FR4 substrate dielectric constant of $\varepsilon_r = 4.3$ and thickness h = 1.53 mm. The monopole and fractal monopole antenna has been fed with optimized dimension of CPW - feed. The advantage of coplanar feed is that the feed of the antenna, ground and radiating elements all are printed on the same side of the substrate.

4.CPW-FED AND UWB CHARACTERISTIC

The antennas have been fed directly with the coaxial probe need a perpendicular ground plane increases the size [1-2]. They are not convenient for integrating with monolithic microwave integrated circuits (MMIC) and MIC circuits. In this paper, printed circular fractal monopole antennas have been fed by a coplanar waveguide (CPW). It has been found out that CPW - fed offers less dispersion at higher frequency and broader matching, easy fabrication and integration with MIC/MMIC. The CPW-fed antenna not only performs better in respect of bandwidth and but radiation pattern is also good [9]. A simple circular disc monopole antenna with CPW-fed is shown in Figure 1. The current distribution of the proposed antenna is mainly along the circumference of the circular disc. The current density is low in the middle area of the solid circular disc monopole antenna as shown in Figure 2. Therefore, the current will not be affected if the middle area metallization of the solid circular disc monopole antenna is removed by circular or other geometrical pattern. Removing some portion of metallization from solid circular disc increases the effective path of the surface current. In this antenna, the effective length of current path is increased by inscribing circle patterns inside solid circular disc using Descartes circle theorem. This resulted, the first resonance frequency will be decreased and the size of the antenna will be reduced. To achieve the UWB characteristic, the fractal structure can be added to increase the resonance frequency in high frequencies by adding resonance elements in solid circular disc antenna through various circles patterns based on Descartes circle theorem. In this paper, resonance elements have been added by inscribing circles patterns iterative wise using Descartes circle theorem up to fifth iteration as shown in Figure 1. The proposed fractal antenna structure has been shown in Figure 3 with optimized dimension. The impedance matching has been achieved by adjusting the width W = 3.2 mm of the inner conductor and optimizing the gap between the ground plane and feed width to g = 0.7 mm i.e. 70 Ω for wide bandwidth. To achieve the UWB characteristic, the gap between patch and ground has been optimized to h = 0.5 mm. The length of ground plane GL =14.17 mm and width of the ground plane GW=19.7 mm have been taken and optimized.

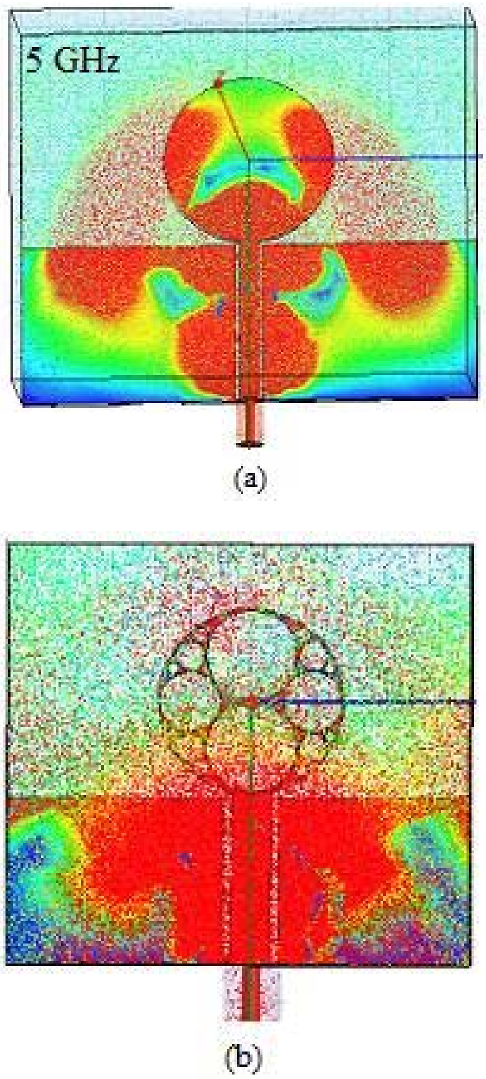

Figure 2 Current Distribution on circular monopole and circular fractal antenna

5.EXPERIMENTAL RESULTS AND DISCUSSION

The proposed circular fractal antenna has been shown in Figure 2. The circular fractal antenna and monopole have been fabricated on substrate ε_r = 4.3 , thickness 1.53 mm and with optimized dimension. The substrate dimension of these antennas has been taken 44 mm x 40 mm. The circular disc monopole and circular fractal antennas have been tested using R&S vector analyzer ZVA40. The experimental result of monopole antenna has been shown in Figure 4. The first resonant frequency of this antenna is 4.31 GHz. The experimental result acquired from vector network analyzer for circular fractal antenna has been shown in Figure 5. This antenna exhibits the excellent ultra wide bandwidth (UWB) characteristics. The impedance bandwidth of this antenna is 10.07 GHz which corresponds to 120.88 %. It has been observed that first resonant frequency of circular fractal antenna has shifted to 3.75 GHz by the application of fractal geometry in comparison to monopole antenna first resonant frequency 4.31 GHz. This indicates the size reduction of the antenna. The proposed circular fractal antenna with CPW- feed is compact one in comparison to circular fractal antenna reported in [10]. Because antenna reported in [10] was fed directly feed by 50 Ω coaxial probe to the driven element with an SMA connector. This need a perpendicular ground plane which increases the size of antenna and not suitable for integrating with monolithic microwave integrated circuits (MMIC). The type of fractal antenna configuration is useful for modern UWB wireless communication system.

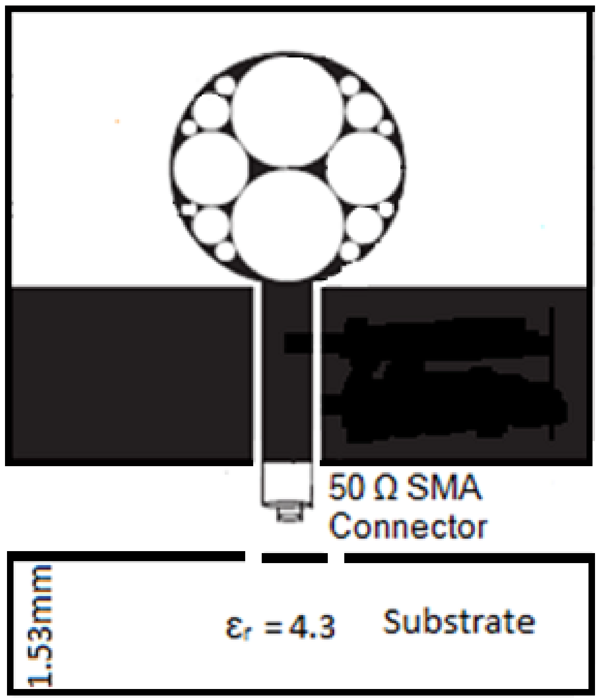

Figure 3 Circular fractal antenna based on Descartes circle theorem [12]

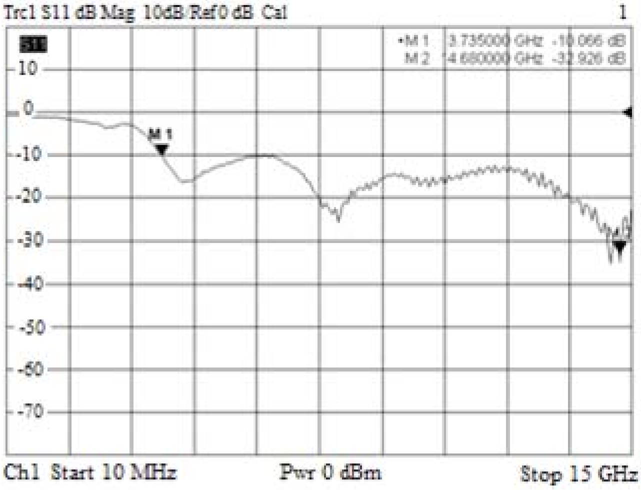

Figure 4 Experimental results of circular disc monopole antenna

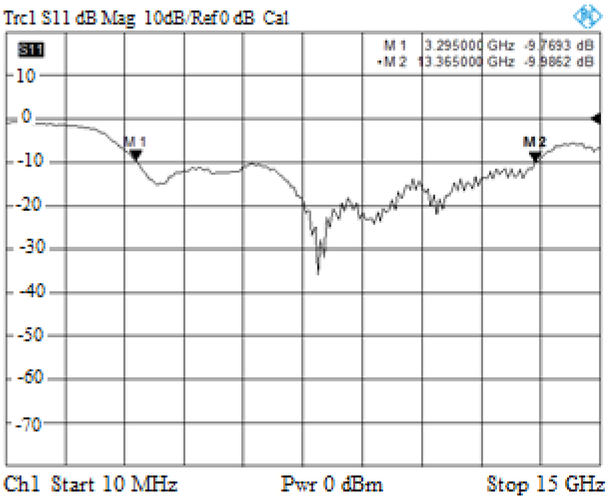

Figure 5 Experimental results of circular fractal antenna

5.RADIATION PATTERN

The radiation patterns of circular fractal antenna have been measured in in-house anechoic chamber. The radiation in azimuth as well as elevation plane has been measured at selective frequencies. The radiation patterns in H-plane have been measured at frequencies 4.2 GHz, 7.0 GHz, 8.775 GHz and 10.1 GHz as shown in Figure 6. The radiation patterns in E-plane have also been measured at frequencies 6.6 GHz, 8.325 GHz and 9.525 GHz as shown in Figure 7. The nature of H-plane radiation pattern is nearly omni- directional. The radiation pattern in H- plane and E – plane is more stable throughout the band in comparison to reported [10]. The radiation pattern nature in E – plane is like monopole radiation pattern. The radiation pattern of circular fractal antenna in [10] is having increasing number of lobes as frequency increases; while side lobe in the proposed fractal is less than [10]. The radiation pattern in H-plane is more stable. The variation is around 3 dBi upto frequency 10.6 GHz; while the variation in [14] is around -18 dBi up to 7.0 GHz. In E-plane side lobe is less in proposed case. The measured gain of this antenna is less than 5dBi up to the frequency 13.36 GHz.

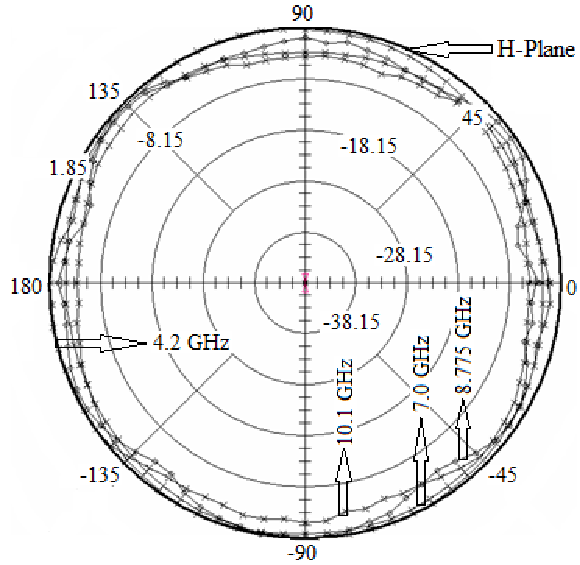

Figure 6 H – Plane radiation patterns of circular fractal antenna

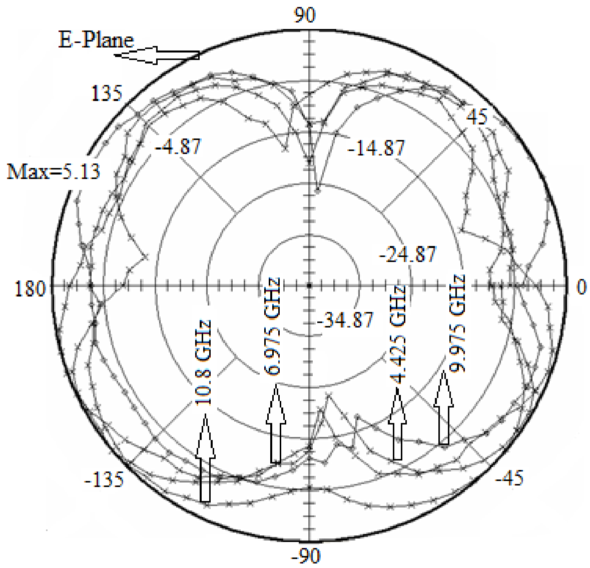

Figure 7 E – Plane radiation patterns of circular fractal antenna

6.CONCLUSION

The circular fractal antenna with ultra-wideband characteristics has been successfully implemented and demonstrated. The proposed fractal antenna has been designed with fifth iteration. It is observed as iterations increases, first resonant frequency shifts to the lower frequency side, thus yielding size reduction of the antenna. The measurement results exhibits the excellent UWB characteristics in the wide range from 3.295 GHz to 13.365 GHz corresponds 120.88 % impedance bandwidth and good radiation pattern. The measured radiation of proposed antenna in azimuth plane is omni-directional and stable throughout the band. The gain of this antenna is less than 5 dBi. The proposed fractal antenna is simple to design and easy to fabricate and integrate with MMIC devices. The antennas of this type can be useful for 4^{th} generation wireless applications.

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REFERENCES

[1] L. Yang and G. B. Giamalkis, “Ultra wide band Communications,” “IEEE signal processing magazine, Nov. pp. 26-28, 2004.

[2] C. P. Baliarda, J. Romeu, R. Pous, and A. Cardama,”On the behavior of the Sierpinski multiband fractal antenna,” IEEE Trans Antennas Propagat. Vol. 46, pp. 517-524, (1998).

[3] R.V. Haraprasad, Y. Purushottam, V.C. Misra, and N.Ashok,,” Microstrip fractal patch antenna for multiband communication,” Electron Lett. Vol. 36, pp. 1179-1180, 2000.

[4] N. P. Agrawall, G. Kumar, K. P. Ray,” Wide-Band Planar Monopole Antennas,” IEEE Transactions on Antennas and Propagation, Vol. 46, no. 2, pp.294-5, February 1998.

[5] M. Hammoud, P. Poey, F. Colombel, ” Matching the Input Impedance of a Broadband Disc Monopole,” Electronics Letters, Vol. 29, no. 4, pp. 406-7, 18th February 1993.

[6] J. Liang, C.C. Chiau, X.D. Chen, and C.G. Parini, “Study of a printed circular disc monopole antenna for UWB systems,” IEEE Trans Antennas Propag. Vol. 53, pp. 3500-3504, 2005.

[7] P.Dehkhod, A. Tavakoli,” A crown square microstrip fractal antenna,” IEEE Antenna Propag. Soc Symp Dig 3, pp. 2396-9, 2004.

[8] VJ Lui, CH. Cheng, Y. Cheng, H. Zhu,” Frequency notched ultra-wideband microstrip slot antenna with fractal tuning stub,” Electron Lett. 41, pp. 294-6, 2005.

[9] W J Lui, CH. Cheng, HB. Zhu,” Compact frequency notched ultra-wideband fractal printed slot antenna,” IEEE Microwave Wireless Compon. Lett. 16, pp. 224-6, 2006.

[10] M. Ding, R. Jin, J. Geng, Q. Wu,” Design of a CPW-fed ultrawideband fractal antenna,” Microwave Opt. Technol Lett. 49, pp. 173-6, 2007.

[11] Ji-Chyun et.al., “Circular fractal antenna approaches with discrete circule theorem for multiband/wideband application,” Microwave and Optical Tech. Lett. Vol. 44, No.5, pp. 404-408, March 2005.

[12] J.C. Lagarias, C.L. Mallows, and A. Wilks,” Beyond the Descartes circle theorem, Amer Math 109, pp. 338-361, (2002).

[13] A. K. Verma and Nasimuddin,”Simple Accurate Expression for Directivity of Circular Microstrip Antenna”, Journal of Microwaves and Optoelectronics Vol. 2, No. 6, Dec. 2002.

DESIGN OF INSCRIBED SQUARE CIRCULAR FRACTAL ANTENNA WITH ADJUSTABLE NOTCH-BAND CHARACTERISTICS

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ABSTRACT

This paper presents the design of an inscribed square circular fractal antenna with notch having adjustable frequency characteristics. The position and width of the notch band can be adjusted in the entire operating band. A prototype of the antenna has been designed on FR4 substrate dielectric constant 4.3 and thickness h=1.53 mm with a U-shape slot in coplanar waveguide feed of length L = 11 mm and slot width W = 0.4 mm. The experimental result of this antenna exhibits ultra wide band characteristics from frequency 3.1 GHz to 15.0 GHz. The notch in operating band helps to reduce the interference with the frequency bands of Worldwide Interoperability for microwave access (WiMAX). The simulated and experimental return loss are found in good agreement. The experimental radiation of this antenna in azimuth plane is nearly omni-directional. This proposed inscribed square circular fractal antenna with notch can thus be used for Ultra wide band (UWB) system, microwave imaging and precision position system.

Keywords -Monopole antenna, Fractal Antenna, Multiband, Resonant frequency and CPW-fed.

1.INTRODUCTION

UWB technology has attracted much attention for use in shortrange high-speed wireless communication applications. UWB system has allocated 7.5 GHz of spectrum for unlicensed use by the FCC in February 2002 for communication applications. With the release of the 3.1 - 10.6 GHz band for ultra wideband (UWB) operation, a variety of typical UWB applications have come up for instance, indoor/ outdoor communication systems, ground penetrating and vehicular radars, wall and through-wall imaging, medical imaging and surveillance [1]. It is certain that many future systems will utilize hand held devices for such short-range and high bandwidth applications. Therefore, the realization of Ultra Wide band (UWB) antennas in printed-circuit technologies within relatively small substrate areas is of primary importance. A number of such antennas with either microstrip monopole, fractal monopole with coplanar waveguide feeds or others in combined technologies have been presented in open literature [2 -8]. However, a part of the bandwidth of ultra wideband (3.1-10.6GHz) released by Federal Communications Commission (FCC). The frequency bands of Wide Local Area network (WLAN) (5.15–5.35/5.725 – 5.825 GHz), Worldwide Interoperability for microwave access (WiMAX) (3.3-3.7GHz) and the C Band used for satellite communication are overlap. Therefore, to reduce the electromagnetic interference interference between the UWB and nearby existing communication systems, notch filters in UWB systems are necessary [2-6]. The notch filter separately connected with UWB antenna increase the size of antenna system. So, some technique to create notch in the antenna operating frequency has be to embedded in the antenna circuit. In this contest, various research work monopole antennas and fractal antennas with notch for UWB characteristic have been reported [2-6]. In [7], a crown square fractal antenna with reduce size for UWB characteristic have been reported. M. Ding [8] has reported the ultra wide band fractal antenna with coplanarwaveguide (CPW) feed. very less research work has been reported of UWB fractal antenna with notch characteristics.

In this paper, a novel CPW-fed UWB fractal antenna with notch characteristic is presented. A prototype of the antenna has been designed and fabricated to reject the frequency band from 3.7 to 4 GHz. This antenna not only satisfies all UWB bands but also rejects the limited band in order to avoid possible interference with the existing WIMAX band. Details of the parametric study of this proposed antenna has been done. Experimental results of the proposed antenna are in good agreement with simulated results.

2. ANTENNA CONFIGURATION

The UWB fractal antenna on printed circuit board has been proposed this paper. This fractal antenna has advantages of compact size, less weight, conformal, easily fabricated and integrated with Microwave monolithic integrated / Microwave integrated circuits (MMIC/MICs) circuits. This antenna has been constructed from simple conventional monopole antenna iteration wise as shown in Figure 1. The solid circular monopole antenna has been designed on FR4 substrate ε_r =4.3, h = 1.53 mm, with radius 9.1 mm. This is called the initiator or zeroth iteration. The first iteration of fractal antenna has been constructed by inscribing the square patch of dimension 12.8 x 12.8 mm inside the circle and subtracted it from circle. This is called 1st iterative inscribed square circular fractal antenna as shown in Figure 1b. The 2nd iteration has been achieved by making the circle of diameter

12.8 mm and an inscribed square of dimension 9.05 x 9.05 mm has been subtracted from this inner one circle as shown in Figure 1c. The 3rd iteration is constructed by making the metallic circle of 9.05 mm diameter inside the square touching the metallic part of its and subtracting an inscribed square of dimension 6.4 x 6.4 mm as shown in Figure 1d. In the fourth iteration, a circle of diameter 6.4 mm is made and an inscribed square of dimension 4.525 mm x 4.525 mm is subtracted as shown in Figure 1e. This process can be repeated upto infinite iteration. Practically infinite iterative structure is not possible because of fabrication constraints. The fourth iterative fractal antenna has been finalized to design on the same substrate dielectric constant and thickness as conventional microstrip monopole antenna as shown in Figure 2. This antenna has been fed with the coplanar feed. The CPW-Fed and radiating elements both are printed on the top side of a low cost FR4 substrate with dielectric constant $\varepsilon_r = 4.3$, h =1.53 mm and loss tangent tan $\delta = 0.02$.

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

Figure 1 Proposed fractal antenna with various iteration

3. COPLANAR FEED WITH SLOT

Figure 2 shows the geometry of the proposed fractal antenna. It is composed of inscribed square circular fractal radiating elements, fed by a U-shape slotted CPW-feed with a very small ground plane. There is no ground plane at the bottom of the substrate. It is known that CPW - feed is advantageous for less dispersion at higher frequency, broader matching, easy fabrication and integration with MIC/MMIC. The CPW-fed antenna not only performs better in respect of bandwidth and but radiation pattern is also good [8]. In coplanar feed, the feed of antenna and radiating elements are printed on the same side of the substrate. A slot is inserted of U-shape in 50 Ω feed line of coplanar waveguide to create the notched frequency band as shown in Figure 2. The resonant frequency of the notched band is defined by the effective length of the slot and band-notched frequencies by width of the slot. It can be seen by changing the L and W, notched band frequencies can be controlled. The notched frequency bandwidth is controlled by varying the width of slot. While notched-band resonant frequency can be controlled by changing the length of slot. To validate the design the band-notched at frequency from 3.635 to 3.935 GHz, the slot length of 10 mm and width 0.4 mm have been taken which can be redesign according to the requirement. A parameter studies with respect to slot width and length have been done. Figure 2 shows the U-shaped slots with dimension having a frequency band-notched function. The inscribed square circular fractal antenna with notch has been designed on an FR4 substrate, having permittivity $\varepsilon_r = 4.3$ and height h =1.53 mm.

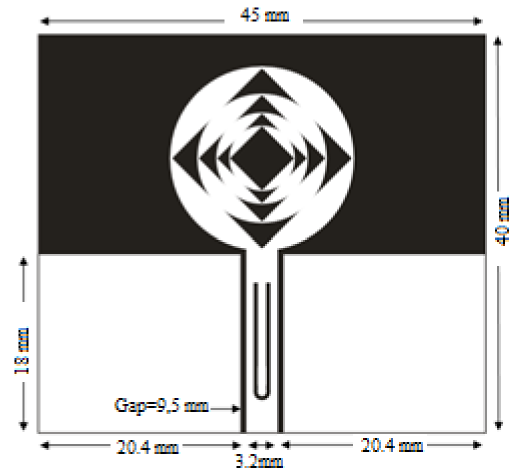


Figure 2 Geometry of proposed UWB fractal antenna

The total size of the antenna including the ground plane is only 40 x 45mm^2. A 50Ω CPW transmission line, which consists of strip width of 3.2 mm and a gap distance of 0.5 mm between the single strip and the coplanar ground plane is used for feeding the antenna. The geometry of the U shaped slot is shown in Figure 3. All dimensions are in mm. The prototype of the antenna with slot has been fabricated and tested as shown in Figure 4.

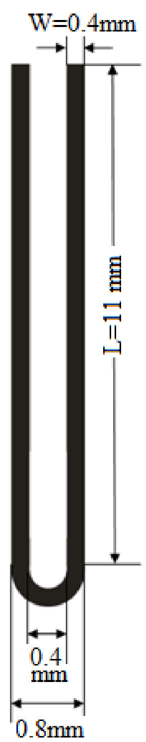


Figure 3 U-shaped slot

4. RESULTS AND DISCUSSIONS

A prototype of the Inscribed Square Circular Fractal antenna with adjustable Notch-band characteristics has been designed on FR4 substrate with ε_r = 4.3 and thickness h = 1.53 mm with the length of U-shaped slot L = 11 mm and slot width W = 0.4 mm. The antenna design has been simulated using commercially available electromagnetic simulators based on FDTD technique. Photograph of the fabricated antenna has been shown in Figure 4.

Figure 4 Photograph of the fractal antenna prototype

Figure 5 shows the simulated result return loss versus frequency with and without U-shaped slot. Without U shaped slot on the feed line, the return loss simulated result of the proposed antenna shows that the return loss is greater than -10 dB from 3.1 to 12 GHz and beyond which satisfies the UWB bandwidth. By introducing U-shaped slot on the feed line, the sharp frequency band-notch characteristic is obtained very close to the desired frequency from 3.7 to 4GHz.

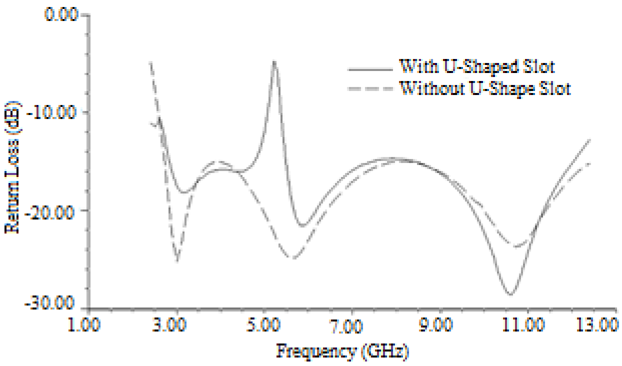


Figure 5 Simulated return loss with and without U-shaped slot

Figure 6 shows the experimental results acquired from the R & S ZVA40 vector network analyzer of the proposed antenna. This fractal antenna exhibits the excellent ultra wide bandwidth from frequency 3.1 GHz to 15 GHz except band-notched frequencies from 3.635 GHz to 3.935 GHz. This corresponds to the 11.9 GHz impedance bandwidth or 132.49 % for VSWR 2:1. It has been observed that first resonant frequency of fractal antenna without slot shifted at 3.2 GHz in comparison to the first resonant frequency of solid circular disc monopole antenna 3.775 GHz. This shift indicates the size reduction of the antenna. The measured and simulated return loss graphs are found to be in good agreement with each other as can be clearly seen from Figures 6 and notch characteristic at L=12 mm, 11 mm and 10 mm in Figure 7.

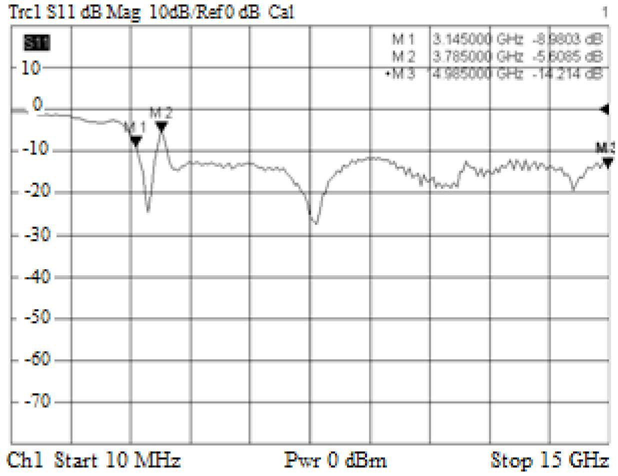


Figure 6 Experimental result of proposed fractal antenna

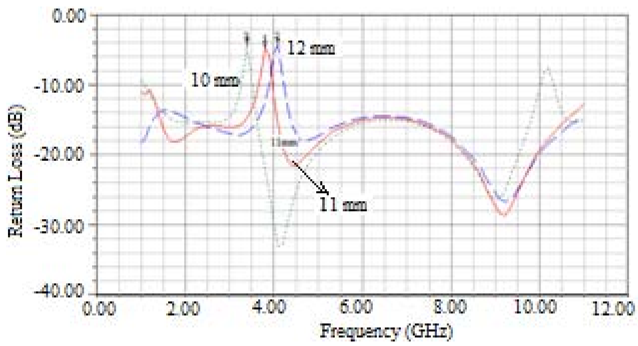


Figure 7 Simulated result of the fractal antenna at 12 mm,11 mm and 10 mm slot length.

Figure 8 illustrates the effect of variation in the U-shaped slot length (L). It can be clearly seen that as the length of the U-shaped slot (L) is increased, the notch band shifts to the lower frequency side. Hence the notch band can be adjusted anywhere on the UWB bandwidth of 3.1 to 10.6 GHz by varying slot length L. In the present design, the slot length has been taken 11 mm and width of the slot 0.4 mm. It has also been observed that variation in the slot width (W) widens the notch band. This is illustrated in Figure 9. As the slot width (W) is increased, the notch band occupies a larger frequency range. So, by adjusting the slot width the notch frequency band can be achieved as shown in the simulated result shown in Figure 9.

The simulated radiation patterns of this fractal antenna have been calculated using electromagnetic simulator at frequency 2 GHz and 7 GHz as shown in Figure 10a and10b.The radiation pattern in azimuth plane is nearly omni directional. The nature of radiation pattern in elevation plane or in E – plane is dumble shape. The red solid line present the h-plane pattern and blue solid graph present the E-plane pattern.

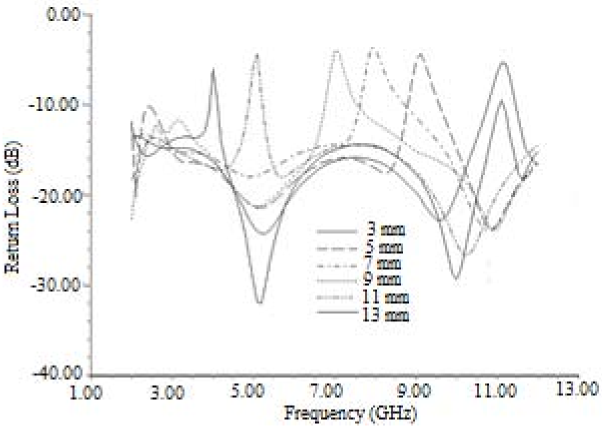


Figure 8 Simulated notch characteristic with respect to the variation of slot length L

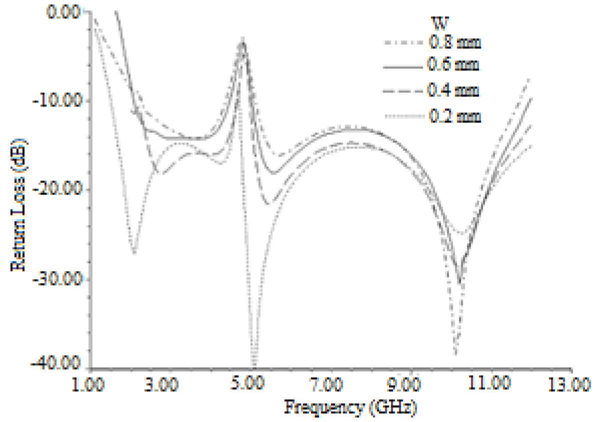
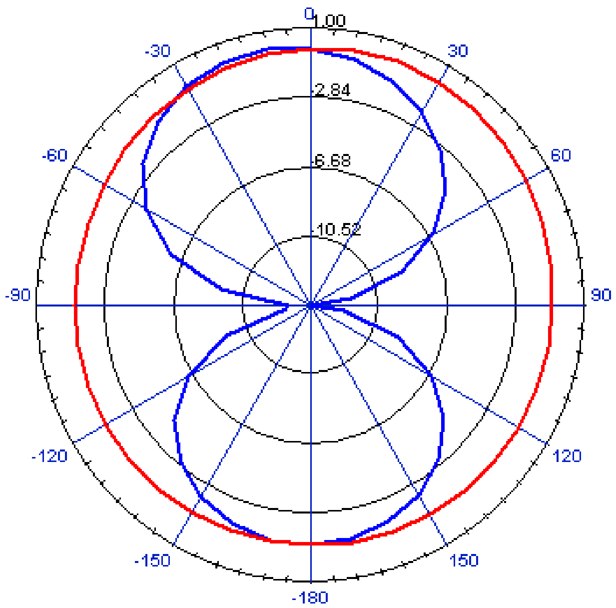
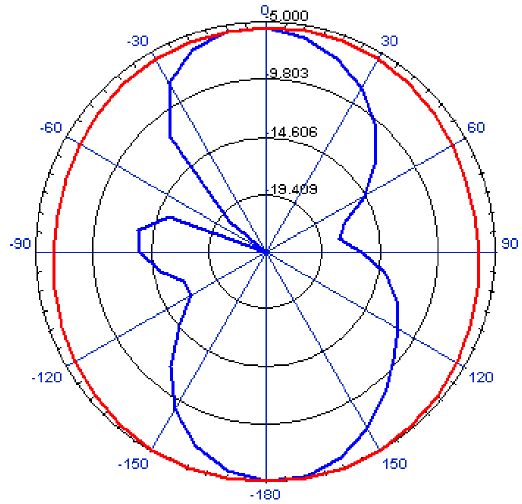


Figure 9 Simulated return loss for variation in slot width W



(a)



(b)

Figure 10 Simulated radiation patterns for the proposed antenna at (a) 2 GHz and (b) 7 GHz

The experimental radiation pattern of this antenna has been measured in the in-house anechoic chamber. The h-plane radiation pattern of this antenna has been measured at frequencies 3.0 GHz and 5.85 GHz. The nature of radiation pattern is nearly onmi-directional as shown in Figure 11. The E-plane radiation pattern has also been measured. The nature of radiation pattern in E-plane is bumble shape. It has been measured at frequencies 6.375GHz and 8.4 GHz in Figure 12 but at higher frequencies 11.25 GHz and 13.75 GHz becoming like monopole radiation pattern as shown in Figure 13. The measured gain of this antenna has been shown in Figure 14. At the notched frequency gain of the antenna dropped drastically as clearly visible in gain graph.

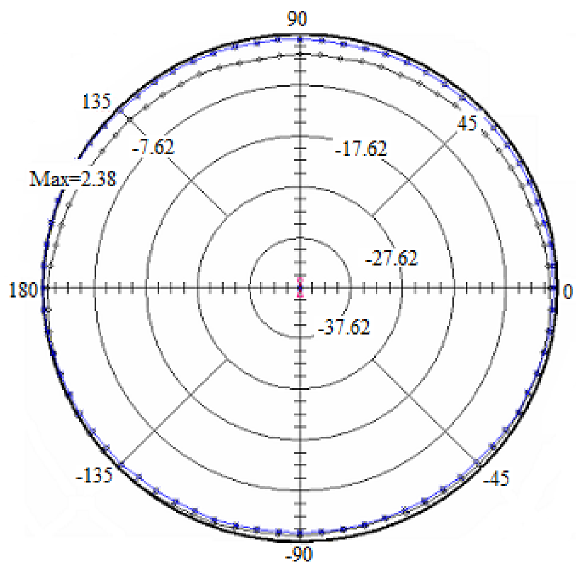


Figure 11 Experimental Radiation Pattern of proposed antenna at frequencies 3.0 GHz (blue line) and 5.85 GHz (black line)

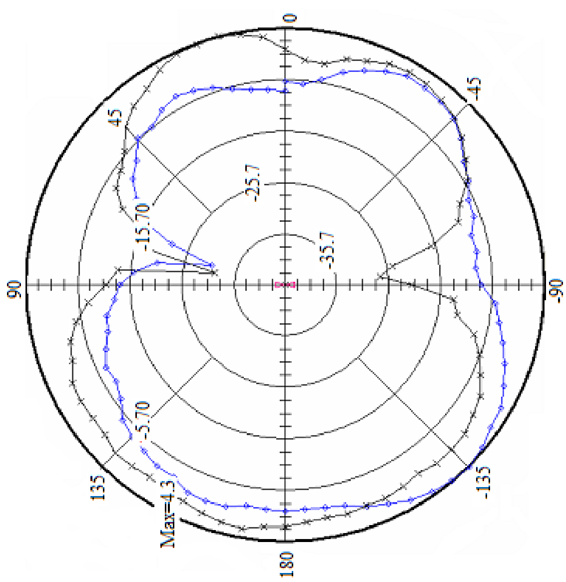


Figure 12 Experimental Radiation pattern in E plane at frequencies 6.375GHz (blue) and 8.4 GHz (black)

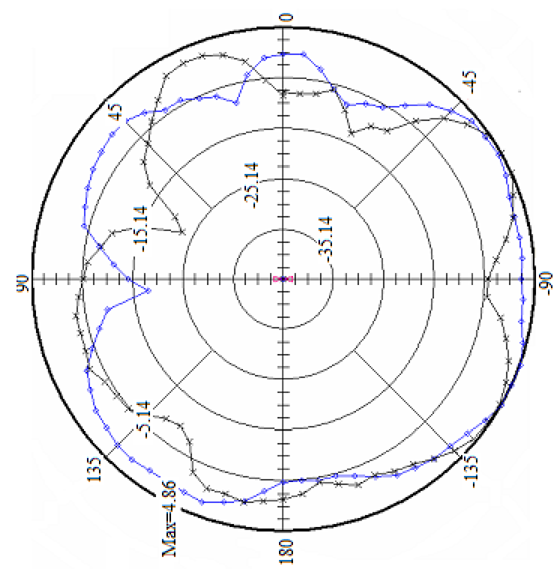


Figure 13 Experimental Radiation Pattern in E-plane at frequencies at 11.25 GHz (black) and 13.75 GHz (blue)

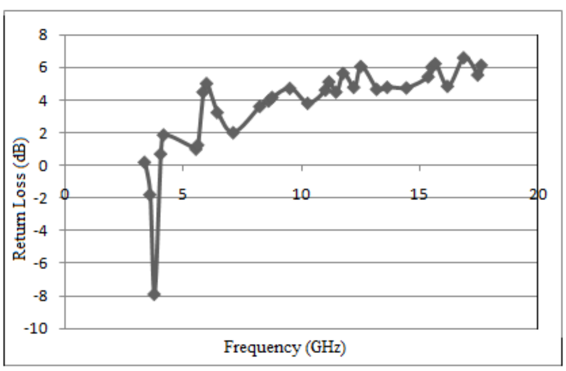


Figure 14 Measured Gain of proposed fractal antenna

5. CONCLUSION

A novel UWB inscribed Square Circular Fractal antenna with adjustable Notch-band characteristics has been designed and implemented. A band-notched characteristic with position and width adjustable over the entire UWB bandwidth from 3.1 to 10.6 GHz is achieved by incorporating a U-shaped slot on the feed line. The prototype of proposed antenna shows measured return loss greater than -10dB for the frequency band from 3.1 to 15 GHz, with a rejection band between 3.635 GHz to 3.935GHz. The proposed fractal antenna is very simple, compact in size, easy to fabricated and can be integrated with MMIC/MICs. The proposed antenna having a frequency band notched characteristic is promising for fourth generation UWB wireless communication system, and vehicular radars and microwave imaging applications.

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REFERENCES

[1] G. R. Aiello and G. D. Rogerson, "Ultra-wideband Wireless Systems," IEEE Microwave Magazine, pp. 36-47, June 2003.

[2] H. J. Zhou, B. H. Sun, Q.Z. Liu, and J.Y. Deng," Implementation and investigation of U-shaped aperture UWB antenna with dual band notched characteristics," Electron Lett. Vol. 44, pp. 1387-1388, 2008.

[3] T. P. Vuong, A. Ghiotto, and Y. Duroc,"Design and characteristics of a small U-slotted planar antenna for IR - UWB," MOTL. Vol. 49 pp. 1727-1731, 2007.

[4] R. Zaker, C. Ghobadi, and J. Nourinia, "Novel modified UWB planar monopole antenna with variable frequency band-notch function," IEEE Antennas Wireless Propag. Lett. Vol. 7, pp. 112-114, 2008.

[5] V. J. Lui, C.H. Cheng, Y. Cheng, and H. Zhu, "Frequency notched ultra-wideband microstrip slot antenna with fractal tuning stub," Electron Lett. Vol. 41, pp. 294-296, 2005

[6] W.J. Lui, C.H. Cheng, and H.B. Zhu," Compact frequency notched ultra-wideband fractal printed slot antenna," IEEE Microwave Wireless Compon Lett. vol. 16, pp. 224-226, 2006.

[7] P. Dehkhod and A. Tavakoli, "A crown square microstrip fractal antenna," IEEE AP-S, Dig 3, pp.2396-2399, 2004.

[8] M. Ding, R. Jin, J. Geng, and Q. Wu, "Design of a CPW-fed ultrawideband fractal antenna," Microwave Opt. Technol. Lett. 49, pp.173-176, 2007.

RESONANT FREQUENCIES OF A CIRCULARLY POLARIZED NEARLY CIRCULAR ANNULAR RING MICROSTRIP ANTENNA WITH SUPERSTRATE LOADING AND AIRGAPS

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ABSTRACT

This paper presents an analysis for the resonant frequencies and its various harmonics of a nearly Circular Annular Ring Microstrip Antenna (ARMSA) with and without air gaps and superstrate loadings. This ARMSA is studied for various radii of the inner and outer radiating circular edges of disc. Three such nearly circular ARMSA are analyzed with an Aspect Ratio of 0.98. By diagonal feeding at the center of ARMSA, circular polarizations are observed with generation of fundamental resonant frequency and higher order modes. Multilayer dielectric ARMSA with and without air gaps are analyzed using effective quasi-static capacitance approach and compared with experimental results using Vector Network Analyzer to provide less than 1% deviation in the resonant frequency. Also the full wave simulated and experimental readings go in good agreement for all the three nearly circular ARMSA for with and without air gaps along with superstrate loadings of various height and dielectric constant material as cover. This closed form model of nearly circular ARMSA is suitable for covered antenna devices CAD and is directly applicable for integration of microstrip antennas beneath protective dielectric superstrates in portable wireless equipments.

Keywords— Nearly Circular Annular Ring, Aspect Ratio, air gap, superstrates, Circular Polarization

1. INTRODUCTION

Recently the use of microstrip antennas (MSA) has acquired a fast ascent in various applications of portable wireless equipment due to its versatile characteristics like compactness, conformal nature, cost effective and ease of design. The designer's basic requirement includes the proper choice of structure geometry, material selection, thickness, feeding techniques, polarization and far field radiation pattern which in turn demand an appropriate analysis technique which predicts accurately the behavior of the antenna under consideration viz. the resonant frequency, and impedance bandwidth. The various applications require that the antenna to be placed out of the sight of the consumer beneath plastic covers or protective dielectric superstrates. Also to protect the MSA from environmental damage like accumulation of snow, oxidation or corrosion, a dielectric cover called superstrate is usually added on the top of the patch. Covering the MSA with such superstrate, shifts the antenna resonant frequency due to change in the effective permeability of the [1-2] microstrip structure. The shifts in the resonating frequency of the antenna are required to be considered while designing of MSA before installation in a portable unit.

Numerous analyses have been carried out to determine the fundamental resonant frequency of the MSA [3-4], but determining the resonant frequency of the higher order modes is lacking, moreover application of higher order modes have also emerged such as GPS receivers and similar devices. In many practical applications circular polarization is required which can be usually obtained by using two feeds located geometrically 90^0 apart and with relative phase shift of 90^0. Circular polarization also can be expected from a slightly elliptical radiator, fed along a line 45° from its major axis. Such an antenna requires only one feed, fed along the edge at Φ^0 from the semi major axis by a coaxial line through a dielectric substrate or by a microstrip line [5-6].

In this paper a detailed study is done on circularly polarized Nearly Circular ARMSA. By using the mapping techniques this circular disc MSA is mapped in nearly circular annular ring geometry. The modified capacitance takes care of the fringe field variations due to different modes which are formulated as in [7]. Then the Nearly Circular annular ring loaded with superstrates with different dielectric constants and thickness along with air gaps of various spacer heights are analyzed both theoretically and practically.

2. THEORETICAL BACKGROUND

The structure under investigation is shown in figure1. The nearly circular metallic disc has a 'b/a' aspect ratio of 0.98. To investigate the properties of the Nearly Circular ARMSA, the inner disc radius was kept half to that of the outer disc's radius and the inner metallic disc was cut from the outer metallic disc to create an annular ring.

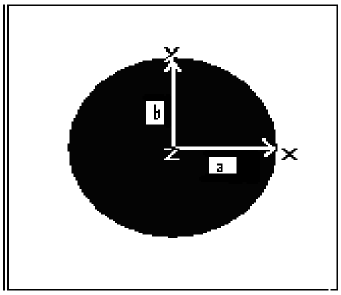

Figure1. Nearly circular disc with 'a' being radius along Major X-axis & 'b' being radius along Minor Y-axis.

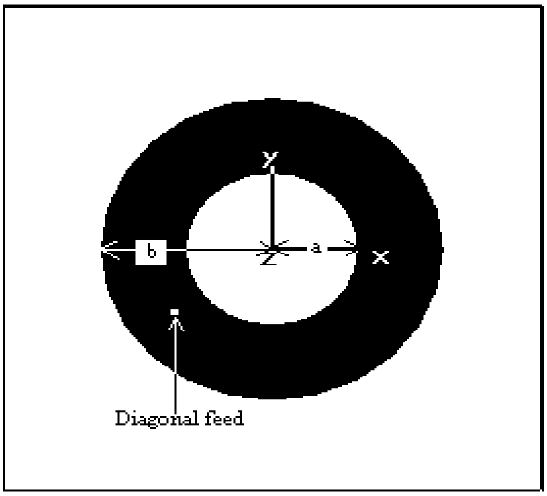

Figure2. Top view of nearly circular ARMSA with 'a' radius of inner disc and 'b' radius of outer disc

Keeping the aspect ratio constant, the major and minor axis was progressively reduced from 70 mm to 8.75 mm.Three such Nearly Circular ARMSA viz. 1st, 2nd and 3rd ARMSA were printed on a dielectric substrate of FR4 with dielectric constant ε_r = 4.3 and material height of h = 1.53 mm. A set of ellipses with varying eccentricities were etched on the PCB having same thickness 'h' and dielectric constant of ε_r as shown in top view of figure 2.

In this paper, the effective capacitance which takes care of the fringe field variations due to the different modes is formulated as in [8-10]. Then the layered ARMSA with and without air gaps are analyzed by calculating the effective capacitance by 'quasi-static capacitance' approach [13].

Figure 3 shows the side view of Nearly Circular ARMSA etched on a substrate of FR4 material with height 'h' and dielectric constant of ε_r = 4.3 having a ground plane beneath it. The annular ring has an inner radius of 'a' and an outer radius of 'b'. The feed position is optimized to be placed diagonally at coordinates of d. For analyzing the effect of superstrates and air gap over the ARMSA, the air gap spacers of various heights are introduced to create an air gap cavity between the actual radiating patch and the superstrates. The superstrates are of materials of different dielectric constants and varying heights placed above each other denoted as ε_{r1}, ε_{r2}, h_1, h_2 respectively.

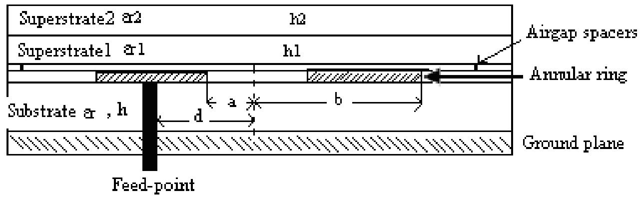

Figure3. Side View of Nearly Circular ARMSA with Air gap and Superstrates

3. ANALYSIS OF NEARLY CIRCULAR ARMSA WITH SUPERSTRATES

For an Annular ring MSA with an inner radius 'a' and outer radius 'b' the effective dielectric constant is determined and the modified eigen values are obtained by solving the following characteristic equation:

$$J_n^{'}\left(k_{nm}^{'}a\right)Y_n^{'}\left(k_{nm}^{'}b\right)-J_n^{'}k_{nm}^{'}\left(k_{nm}^{'}b\right)Y_n^{'}\left(k_{nm}^{'}a\right)=0 \qquad (1)$$

Where 'n' is order of the Bessel function and 'm' is m^{th} zero of the function $J'_n(ka)$ and J_n and Y_n are the Bessel function of order n of first and second kind respectively. The resonant frequency of Microstrip Antenna is governed by the accuracy of the capacitance of the patch resonator as is given in [11-12].

$$f^1 = \frac{1}{2\pi\sqrt{L^1C^1(1+\Delta)}} \qquad (2)$$

where for small (h/a)

$$\Delta = \frac{2h}{\pi a\varepsilon_r}\left[\ln\left(\frac{\pi a}{2h}\right)+1.7726\right] \qquad (3)$$

Where; 'h' is height of substrate

'ε' is dielectric constant of substrate

'a' is radius of the discs

L^1 and C^1 are first order inductance and capacitance respectively. By using the eigen values of (1) the above equations (2) and (3) provide comparable results only for the dominant modes but gives considerable drift between theoretical and experimental values of the resonant frequency for higher order modes. Hence, the capacitance also needs to incorporate the fringe field variations and hence the mode variables 'n' and 'm' to yield a better accuracy. Thus the term 'Δ' of (3) is modified for a better approximation of capacitance for the higher order modes as:

$$\Delta = \frac{2h}{\pi a\varepsilon_r}\left[\ln\frac{\pi a}{2h}F(n,m)+1.7726\right] \qquad (4)$$

Where; $F(n,m) = f(n,m) * g(\beta)^q$

$$f(n,m) = \left(\frac{\beta}{\alpha}\right)^k \ ; \quad \alpha = \begin{cases} n, & n \geq 1 \\ 1, & n = 0 \end{cases}$$

$$\beta = m \quad \forall m$$

The value of k is obtained in the least square error sense and is nearly equal to 2. Then the function $g(\beta)=\beta$ is optimized for minimum error for which we obtain $q\approx1$.

Thus the general form of capacitance for any mode can be given by:

$$C^1 = C^0\left[1+\frac{2h}{\pi a\varepsilon_r}\left(\ln\frac{\pi a}{2h\alpha^2}\beta^3+1.7726\right)\right] \quad (5)$$

The modified first order capacitance is applied to the Annular ring MSA by considering the inner and outer radiating circular edges as edges of disc of smaller and larger radii respectively , giving modified 'a' and 'b' as:

$$a_{eff} = a\left[1-\frac{2h}{\pi a\varepsilon_{eff}}\left(\ln\frac{\pi a}{2h\alpha^2}\beta^3+1.7726\right)\right]^{1/2} \quad (6)$$

$$b_{eff} = b\left[1+\frac{2h}{\pi b\varepsilon_{eff}}\left(\ln\frac{\pi b}{2h\alpha^2}\beta^3+1.7726\right)\right]^{1/2} \quad (7)$$

4. NEARLY CIRCULAR ARMSA WITH AIR GAPS

The capacitance is a function of only the dielectric of the material and the geometry of patch. If we assume a potential difference of 'E_o' between the plates and uniform electric field intensities in the two regions with 'Q' being the magnitude of the surface charge, the effective capacitance is found as:

$$C=\frac{Q}{E_0}=\frac{1}{\frac{h_1}{\varepsilon_1 A}+\frac{h_2}{\varepsilon_2 A}}=\frac{1}{\frac{1}{C_1}+\frac{1}{C_2}} \quad (8)$$

So, the individual capacitances are in series.
The effective dielectric constant is calculated as:

$$\varepsilon_{eff}=\frac{C(h_1+h_2)}{A} \quad (9)$$

Where 'C' is the series capacitance of the layers. To calculate the resonant frequency of ARMSA with an air gap effective inner and outer radii are calculated using (9). Then the patch area responsible for the capacitance of each layer is calculated by the formula:

$$A=\pi\left(b_{eff}^2-a_{eff}^2\right) \quad (10)$$

The series capacitance is obtained and then the resonant frequency for the dominant mode as well as for the higher order modes is calculated.

5. RESULTS AND DISCUSSION

Several set of measurement of the Nearly Circular ARMSA with two different superstrate materials as cover along with and without air gap between them are presented to test the accuracy of the resonant frequency of the antenna discussed in section 2. For all ARMSA the resonant frequency measurements use the minimum return loss specification of resonance. This data as well as the calculated resonant frequencies using the formulations given in above section are comparable and are provided in following sub-sections. Percentage errors use experimental resonant frequency as a reference for the dominant mode as well as for the higher harmonics.

5.1. Simple nearly circular ARMSA

In the Simple nearly circular ARMSA the first ARMSA has FR4 substrate material of 180 x 180 mm^2 dimensions with h = 1.53 mm and ε_r = 4.3. The radiating patch dimensions along X-axis is outer radius=70 mm and inner radius=35 mm while along Y-axis is outer radius=68.6 mm and inner radius = 34.3 mm. The feed position is kept diagonal for circular polarizations with coordinates of (-35,-35) mm. The second ARMSA has the same substrate material of 110 x 110 mm^2. The patch dimensions along X-axis is outer radius=35 mm and inner radius=17.5 mm while along Y-axis the outer radius=34.3 and inner radius=17.15 mm. The feed position is optimized along diagonal line for circular polarizations with coordinates of (-18.2,-18.2) mm. The third ARMSA is of 75 x 75 mm^2. The patch along X-axis is outer radius=17.5 mm and inner radius=8.75 mm while along Y-axis is outer radius=17.15 mm and inner radius=8.575 mm. The feed position is kept diagonal with coordinates of (-8.75,-8.75) mm.
Table1 shows the percentage errors between the experimental and calculated resonant frequencies of all modes without air gap and superstrates above the 1st, 2nd and 3rd ARMSA. The %Error is calculated using the expression as: % error = $(f_{exp} - f_{calc})/ f_{exp}*100$

Table1. Simple 1st, 2nd & 3rd ARMSA without air gap and superstrate.

Patch type	modes	f_{o1}	f_{o2}	f_{o3}	f_{o4}	f_{o5}
1st ARMSA b = 70mm a = 35mm	f_{exp} GHz	0.905	1.330	1.735	2.120	2.500
	f_{cal} GHz	0.907	1.326	1.722	2.100	2.470
	% Err	-0.221	0.315	0.714	0.943	1.200
2nd ARMSA b = 35mm a=17.5mm	f_{exp} GHz	0.940	1.820	2.645	3.440	4.188
	f_{cal} GHz	0.958	1.856	2.694	3.480	4.242
	% Err	-1.900	-1.98	-1.875	-1.16	-1.29
3rd ARMSA b=17.5mm a=8.75mm	f_{exp} GHz	1.960	3.770	5.450	7.055	8.625
	f_{cal} GHz	1.954	3.815	5.431	6.966	8.581
	% Err	0.290	-1.21	0.229	1.261	0.505

The average percent error in the resonant frequencies of the dominant as well as the harmonics between the experimental and calculated values for 1st ARMSA is found to be 0.59%. For the 2nd and 3rd ARMSA the percent error is found as -1.64% and 0.216% respectively. The negative sign in %error indicates that the shift in resonant frequency is towards the upper side with respect to the experimental resonant frequency. As the size of the ARMSA is reducing it is observed that the resonant frequency of dominant as well as the harmonics shifts towards the higher side in frequency.

5.2. Nearly circular ARMSA with one superstrate

Table2 includes the resonant frequencies of all modes without air gaps and one superstrate of FR4 material with 'h'=1.64 mm, ε_r= 4.3 above the ARMSA. The average percent error in the resonant frequencies of the dominant as well as the harmonics between the experimental and calculated values for the 1st, 2nd and 3rd ARMSA is found to be 1%, -0.214% and 2.317% respectively.

Table 2. 1st, 2nd & 3rd ARMSA loaded with one superstrate of FR4

Patch Type	modes	f_{o1}	f_{o2}	f_{o3}	f_{o4}	f_{o5}
1st ARMSA b = 70mm a = 35mm	f_{exp} GHz	0.897	1.317	1.715	2.105	2.490
	f_{cal} GHz	0.894	1.307	1.694	2.065	2.470
	% Err	0.334	0.767	1.236	1.900	0.803
2nd ARMSA b = 35mm a=17.5mm	f_{exp} GHz	0.927	1.804	2.614	3.380	4.102
	f_{cal} GHz	0.931	1.810	2.618	3.377	4.110
	% Err	-0.431	-0.332	-0.175	0.080	-0.21
3rd ARMSA b=17.5mm a=8.75mm	f_{exp} GHz	1.925	3.680	5.295	6.865	8.290
	f_{cal} GHz	1.866	3.629	5.151	6.707	8.001
	% Err	3.064	1.385	1.350	2.290	3.498

Table 3. 2nd & 3rd ARMSA with one superstrate of RT Duroid

Patch type	modes	f_{o1}	f_{o2}	f_{o3}	f_{o4}	f_{o5}
2nd ARMSA b=35mm a=17.5mm	f_{exp} GHz	0.929	1.827	2.659	3.425	4.189
	f_{cal} GHz	0.929	1.827	2.659	3.425	4.189
	% Err	-2.152	-1.018	-0.630	-0.875	-0.400
3rd ARMSA b=17.5mm a=8.75mm	f_{exp} GHz	1.960	3.805	5.405	7.005	8.595
	f_{cal} GHz	1.933	3.769	5.356	6.861	8.420
	% Err	1.367	0.932	0.890	2.050	2.031

Table 3 above shows the resonant frequencies of all modes without air gaps and one superstrate of RT Duroid material having 'h'=0.787 mm, ε_r= 2.2 above the ARMSA. The average percent error in the resonant frequencies of the dominant and the harmonics between the experimental and calculated values for the 2nd and 3rd ARMSA is found as -1.015% and 1.454% respectively.

5.3. Nearly circular ARMSA with air gaps

Table 4 shows the resonant frequencies of all modes with one air spacer of height 0.26 mm between substrate and one superstrate of FR4 with 'h'=1.64 mm, ε_r= 4.3 above the ARMSA. The average percent error in the resonant frequencies of the dominant and the harmonics for the 1st, 2nd and 3rd ARMSA is found to be 0.538%, 1.816% and 1.188% respectively.

Table 4. 1st, 2nd & 3rd ARMSA with one superstrate of FR4 having 'h'=1.64 mm, ε_r= 4.3 & air gap = 0.26 mm

Patch type	modes	f_{o1}	f_{o2}	f_{o3}	f_{o4}	f_{o5}
1st ARMSA b = 70mm a = 35mm	f_{exp} GHz	0.897	1.317	1.760	2.101	2.470
	f_{cal} GHz	0.895	1.300	1.752	2.100	2.450
	% Err	0.220	1.200	0.454	0.047	0.769
2nd ARMSA b = 35mm a =17.5mm	f_{exp} GHz	0.927	1.805	2.615	3.381	4.123
	f_{cal} GHz	0.943	1.893	2.667	3.445	4.194
	% Err	-1.70	-1.88	-1.88	-1.89	-1.72
3rd ARMSA b=17.5mm a =8.75mm	f_{exp} GHz	1.970	3.780	5.440	7.020	8.575
	f_{cal} GHz	1.931	3.788	5.385	6.901	8.482
	% Err	1.954	0.200	1.007	1.695	1.083

Table 5 shows the resonant frequencies of all modes with two air gap spacers of height 0.26 mm each between substrate and one superstrate of FR4 with 'h'=1.64 mm, ε_r= 4.3 above the ARMSA. The average percent error in the resonant frequencies for the 1st, 2nd and 3rd ARMSA is found to be 0.15%, -1.71% and 0.545% respectively.

Table 5. 1st, 2nd & 3rd ARMSA with one superstrate of FR4 having 'h'=1.64 mm, ε_r= 4.3 & air gap = 0.26 mm + 0.26 mm

Patch type	modes	f_{o1}	f_{o2}	f_{o3}	f_{o4}	f_{o5}
1st ARMSA b = 70mm a = 35mm	f_{exp} GHz	0.897	1.318	1.715	2.101	2.475
	f_{cal} GHz	0.899	1.316	1.707	2.082	2.500
	% Err	0.222	0.174	0.461	0.904	-1.01
2nd ARMSA b = 35mm a=17.5mm	f_{exp} GHz	0.928	1.827	2.659	3.425	4.189
	f_{cal} GHz	0.955	1.857	2.692	3.478	4.242
	% Err	-2.90	-1.600	-1.241	-1.547	-1.26
3rd ARMSA b=17.5mm a=8.75mm	f_{exp} GHz	1.970	3.780	5.440	7.050	8.590
	f_{cal} GHz	1.954	3.804	5.414	6.947	8.534
	% Err	0.791	-0.642	0.469	1.460	0.646

Table 6 shows the resonant frequencies of all modes with three air gap spacers of height 0.26, 0.26 and 0.787 mm between substrate and one superstrate of RT Duroid material of 'h'= 0.787 mm, ε_r= 2.2 above the ARMSA. The average error in the resonant frequencies for the 2nd and 3rd ARMSA is found as -1.614% and 0.543% respectively.

5.4. Nearly circular ARMSA with two superstrates

Table 7 includes the resonant frequencies of all modes done with two superstrates, one of RT Duroid material having 'h'=0.787 mm, ε_r=2.2 and another of FR4 with 'h'=1.64

mm, ε_r = 4.3 above the ARMSA. The ARMSA is also analyzed by interchanging the superstrates as well as for two superstrates of same material above it. The average percent error in the resonant frequencies between the experimental and calculated values for the 2nd and 3rd ARMSA is found to be -0.377%, 1.794%, 0.122%, 0.56%, 0.357% and 0.535% sequentially.

Table 6. 2nd & 3rd ARMSA loaded with one superstrate of RT Duroid with 'h'=0.787 mm, ε_r= 2.2 & air gap = 0.26+0.787+0.787 mm

Patch type	modes	f_{o1}	f_{o2}	f_{o3}	f_{o4}	f_{o5}
2nd ARMSA b = 35mm a=17.5mm	f_{exp} GHz	0.929	1.828	2.660	3.426	4.190
	f_{cal} GHz	0.955	1.854	2.692	3.480	4.240
	% Err	-2.700	-1.400	-1.200	-1.576	-1.19
3rd ARMSA b=17.5mm a=8.75mm	f_{exp} GHz	1.970	3.780	5.440	7.055	8.620
	f_{cal} GHz	1.954	3.803	5.426	6.953	8.546
	% Err	0.792	-0.616	0.250	1.430	0.857

Table 7. 2nd & 3rd ARMSA with two superstrates of RT Duroid & FR4.

Patch type	modes	f_{o1}	f_{o2}	f_{o3}	f_{o4}	f_{o5}
2nd ARMSA two covers h1=0.787, ε_{r1}=2.2& h_2=1.64 ε_{r2}=4.3	f_{exp} GHz	0.927	1.804	2.638	3.402	4.144
	f_{cal} GHz	0.935	1.819	2.639	3.406	4.146
	% Err	-0.862	-0.853	0.003	-0.126	-0.048
3rd ARMSA two covers h1=0.787, ε_{r1}=2.2 & h_2=1.64 ε_{r2}=4.3	f_{exp} GHz	1.935	3.685	5.345	6.91	8.425
	f_{cal} GHz	1.895	3.684	5.229	6.819	8.139
	% Err	2.067	0.027	2.170	1.316	3.395
2nd ARMSA two covers h1=1.64, ε_{r1}= 4.3& h_2=0.787 ε_{r2}=2.2	f_{exp} GHz	0.927	1.805	2.614	3.380	4.100
	f_{cal} GHz	0.929	1.805	2.612	3.369	4.098
	% Err	0.215	0	0.049	0.32	0.026
3rd ARMSA two covers h1=1.64, ε_{r1}=4.3& h_2=0.787 ε_{r2}=2.2	f_{exp} GHz	1.925	3.635	5.25	6.82	8.245
	f_{cal} GHz	1.919	3.627	5.15	6.791	8.25
	% Err	0.311	0.220	1.900	0.425	-0.06
2nd ARMSA Two FR4 covers of h_1=h_2=1.64 ε_{r1}=ε_{r2}=4.3	f_{exp} GHz	0.927	1.782	2.590	3.358	4.078
	f_{cal} GHz	0.924	1.791	2.592	3.341	4.663
	% Err	0.323	0.516	0.092	0.488	0.367
3rd ARMSA Two FR4 covers of h_1=h_2=1.64 ε_{r1}=ε_{r2}=4.3	f_{exp} GHz	1.922	3.622	5.201	6.721	8.101
	f_{cal} GHz	1.907	3.613	5.190	6.695	8.016
	% Err	0.780	0.248	0.211	0.387	1.049

6. GRAPHICAL REPRESENTATION OF NEARLY CIRCULAR ARMSA

According to the specifications given in previous section all the three types of nearly circular ARMSA were simulated first using the 3D EM simulator to determine the different modes of resonance, to find out the dependence of resonant frequencies on the size of the patch and to find the effect of shift in the resonant frequencies due to loading of superstrates of various materials and air gaps. A comparison is made between the various sizes of the ARMSA with a superstrate over it and introducing air gaps between the actual radiating patch and superstrate of different spacing height. Figure 4 shows the comparison of simulated graphs of return loss with respect to frequency for the simple type of 1st, 2nd and 3rd ARMSA for the results as tabulated in Table 1. Lower first order resonant frequency is indicated as f_{o1}, whereas $f_{o2,}$ $f_{o3,}$ f_{o4} and f_{o5} indicate the higher order harmonics. From figure 4 it is observed that there is considerable shift in resonant frequency of all modes when the size of ARMSA varies from radius of 17.5 mm for the 3rd annular ring to 70 mm for the 1st annular ring. The first resonant frequency f_{o1} of 3rd ring shifts from 1.96GHz to 0.940GHz for 2nd ring and to 0.905GHz for 1st annular ring. This shift of the resonant frequency is expected towards the lower side, since as the inner dimensions will be larger the effective current path will also be larger. The same frequency shift towards the lower side is observed for all the modes and for all the cases of loading of superstrates and air gaps.

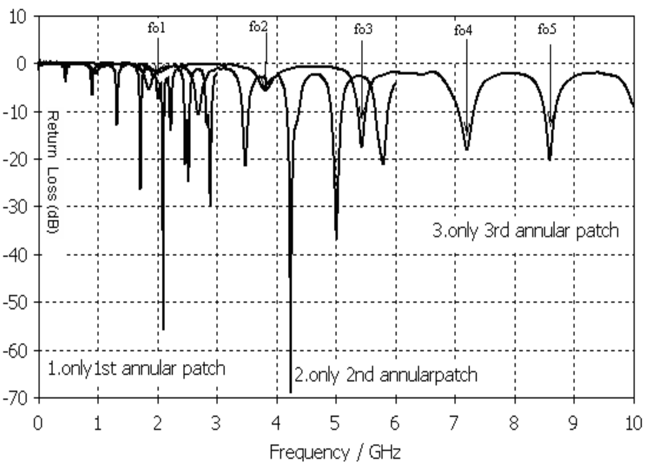

Figure 4. Comparison of resonant frequencies for ARMSA's of Table1.

Figure 5 shows the comparison of simulated graphs of return loss with respect to frequency for the FR4 superstrate loading above each of 1st, 2nd and 3rd ARMSA for the results as tabulated in Table 2. Here the comparison is made in the shift in resonant frequency due to FR4 loading in each case. For the 1st ARMSA f_{o1} shifts from 0.940GHz to 0.927GHz and for 3rd ARMSA f_{o1} shifts from 1.960GHz to 1.925GHz. This shift in the resonant frequency towards the lower side is the effect of superstrate loading on the radiating patch. Figure 6 shows the comparison of simulated

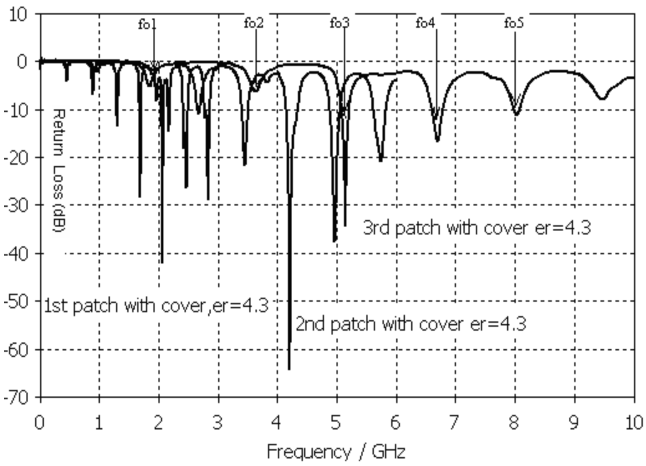

Figure 5. Comparison of resonant frequencies for ARMSA‘s of Table2.

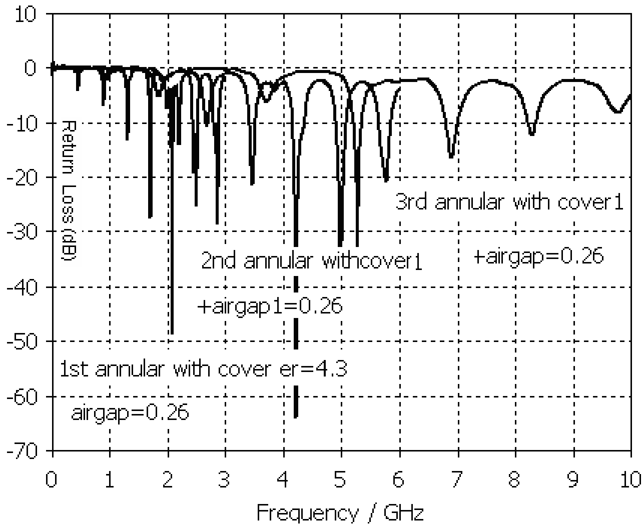

Figure 6. Comparison of resonant frequencies for ARMSA‘s of Table 4.

graphs of return loss with respect to frequency for the FR4 superstrate loading above each of 1st, 2nd and 3rd ARMSA and introduction of an air gap spacing of height 0.26 mm between them for the results as tabulated in Table 4. In this case the shift in f_{01} and all other modes are slightly towards the upper side and are considerably observed in the 3rd ring indicating a shift of 1.925GHz to 1.970GHz. Figure 7 shows the comparison of simulated graphs of return loss with respect to frequency for the FR4 superstrate loading above each of 1st, 2nd and 3rd ARMSA and introduction of two air gap spacing of height 0.26 mm each between them for the results as tabulated in Table 5 and Table 6. In both cases there is no effective shift in frequency observed as the air gap spacing goes on increasing. By comparing the results for double superstrate loading of different materials above the radiating annular ring as tabulated in Table 7, there is again a considerable shift in resonating frequency towards the lower side in each case.

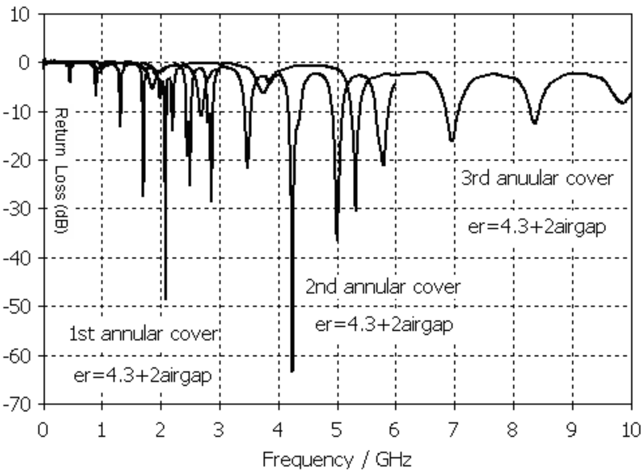

Figure 7. Comparison of resonant frequencies for ARMSA‘s of Table 5.

7. CONCLUSION

The resonant frequencies of a nearly circular annular ring microstrip antenna employing circular polarization with superstrate loading and air gaps between them were analyzed for various radii of the inner and outer nearly circular discs. The full wave analysis of the modified expression for the calculation of the resonant frequency of ARMSA were presented which incorporates the fringing field variations due to different modes excited, which gives an improved accuracy. The quasi-static capacitance shows a simple and efficient analysis to calculate the resonant frequency of multilayer dielectric microstrip antennas. All the three types of ARMSA were fabricated and a comparison is made between the experimental and calculated values of the resonant frequencies for various harmonics. The model demonstrates less than 1% errors on average for each case varying from simple patch, with and without air gaps of various heights to different types of superstrates of various dielectric constants. For the simple ARMSA the average percent error is -0.278%, for ARMSA with one superstrate it is 1.03%, for ARMSA with air gap spacers it is -0.535% and for the ARMSA with two superstrates the overall average percent error is 0.5%. Hence the experimental and full wave analysis methods for determining the resonant frequency of the antenna go in well agreement with each other. The ARMSA shows a shift of the resonant frequency towards the lower side and the same frequency shift towards the lower side is observed for all the modes and for all the cases of loading with superstrates and air gaps. Such type of study is useful for calculating the effect on resonant frequency, improvement of gain and bandwidth of portable antennas. Additionally, the ARMSA model is suitable for CAD and is directly applicable for the integration of microstrip antennas beneath plastic covers or protective dielectric superstrates in portable wireless equipments.

REFERENCES

[1] Debatosh Guha, “Resonant frequency of circular Microstrip Antenna with and without Airgaps”, *IEEE transactions on Antennas and Propagation*, vol.49. No.1, January 2001.
[2] K.M.Luk, W.Y.Tam, and C.L.Yip, “Analysis of circular Microstrip antenna with superstrate”, *IEE proceedings* vol. 136, Pt.H.No. 3, June 1989.
[3] Jennifer T. Bernhard and C. J. Tousignant, “Resonant Frequencies of Rectangular Microstrip Antennas With Flush & Spaced Dielectric Superstrates”, *IEEE transactions on Antennas and Propagation*, vol.47, No.2, February 1999.
[4] Vicente G.P., Daniel S.V., Eva R.L, Jose Luis V.R. and Carlos M.P., “Analysis of short circuited ring patch operated at TM_{01} mode”, in Rev. Fac. Ing. –Univ. Tarapaca, vol.13, No.2, pp. 21-30.
[5] Liang C. Shen, “The Elliptical Microstrip Antenna with Circular polarization” *IEEE transactions on Antennas and Propagation*, vol. 29. No.1, January 1981.
[6] Stuart A. Long, L.C.Shen, D.H.Schaubert and F.G.Farrar, “An experimental study of the circular polarized elliptical printed circuit antenna”, *IEEE transactions on Antennas and Propagation*, vol.29, No.1, January 1981.
[7] Rajanish & T.S.Vedavathy, “Resonant frequency of Higher order modes for circular microstrip antennas”, 1999, *Asia Pacific Microwave Conference*, pp. 936-940.
[8] S.Sriram & T.S.Vedavathy, “Novel analysis Scheme to analyse multilayer dielectric microstrip antennas”, *IEEE 1999, 0-7803-5761-2/99.*
[9] Shun-Shi Zhong, Gang Liu and Ghulam Qasim, “Closed Form expressions for resonant frequency of rectangular patch antennas with multi dielectric layers”, *IEEE transactions on Antennas and Propagation*, vol.42, No.9, Sept. 1994.
[10] K. F. Lee and J. S. Dahele, “The two layered annular ring microstrip antenna”, *Int. J. Electronics* 1986, 61(2), pp. 207-217.
[11] Yi-Fang Lin, Hua-Ming Chen and Shih-Chieh Lin, “A new coupling mechanism for circularly polarized Annular-Ring Patch Antenna”, *IEEE transactions on Antennas and Propagation*, vol.56, No.1, January 2008.
[12] H. M. Chen and K. L. Wong, “On the circular polarization operation of annular ring microstrip antennas”, *IEEE transactions on Antennas and Propagation*, vol.47, pp 289-292, August 1999.
[13] W. C. Chew and J. A. Kong, “Effects of fringing field on the capacitance of circular microstrip disc”, *IEEE transactions on microwave theory techniques* vol.28, pp. 98-104, February 1980.

SESSION 8

FUTURE INTERNET AND THE ENVIRONMENT

A SCHEME FOR DISASTER RECOVERY IN WIRELESS NETWORKS WITH DYNAMIC AD-HOC ROUTING

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ABSTRACT

This paper proposes a hybrid network scheme combining ad-hoc networks into cellular networks. The scheme is aimed to help the networks to recover to service as much as possible after a disaster strike, by maintaining the connection between Base Stations (BSs) and nodes via multi-hopping, where if a node cannot connect to a BS directly, it switches its working mode from cellular mode to ad-hoc mode. A location-based routing protocol has been proposed for building a route from the node to the BS. Simulation results shows that even if only a small part of the nodes can directly connect to a BS, most of the nodes can find a route to a BS via multi-hopping. And it is found that it outperforms a previously proposed solution which is via beaconing in terms of resistance to mobility.

Keywords— Routing, ad-hoc, location-based, multi-hop, disaster-recovery

1. INTRODUCTION

Natural or man-made disasters have posed a great challenge to human society of the world. In Japan, frequently-seen disasters are earthquake, volcano eruption, tsunami, typhoon, rain flooding, and avalanche. Robustness against the above disasters should be considered in the design of information network systems for business, research, and education.

But actually, after such a disaster happens, network congestion is usually found to be serious. Examples in history are like Kobe earthquake in 1995 and the "911" attack in New York, the heavy use of phone call by general population caused sudden and serious congestion in phone system.

In normal circumstances, the current communication infrastructures are sufficient for the resource demand of daily traffic of calls. But after a disaster happens, two things will change: first is that most of people within the affected area will try to make calls to family or friends immediately; and second is that part of the infrastructure may suffer from damage.

To overcome these two issues, this paper discusses about a hybrid network system combining a cellular network with ad-hoc networks, and proposes an ad-hoc routing protocol for the network. The focus is on maintaining the connectivity. The rest of the paper is organized as follows: section 2 talks about the related works on ad-hoc routing, and multi-hopping communication systems. Section 3 describes the network model and section 4 explains the procedures and the protocol. Section 5 shows the simulation results and section 6 draws a conclusion.

2. RELATED WORKS

Routing protocols for wireless ad-hoc networks can be categorized into non-location-based and location-based. Examples of the non-location-based are AODV [8], DSR [9], TORA [10], DSDV [11], and ZRP [12]. These protocols mainly adopt some mechanisms of flooding, either to detect routes on-demand, or to maintain routing information proactively on each node. Differently, location-based routing protocols perform forwarding decisions based on local knowledge of location information. Protocols like BLR (Beaconless Location-based Routing), IGF, and CBF [3] have emerged as new solutions of this category featuring with being beaconless, which is significant to overhead reducing, and is suitable for large-scale network.

Regarding to the application case of disaster recovery, a number of hundreds or above of nodes, after a disaster, may not receive service from centralized network. In order to build communication links, they have to organize an ad-hoc network. So the BLR-like protocols are suitable.

Papers have discussed about employing the technique of ad-hoc networking as a solution to disaster relief and recovery. Article [2] has sketched a framework of an integrated disaster management communication and information system. But implementation of such a framework, e.g. practical networking protocols is not provided.

Article [1] has proposed routing protocols for ad-hoc network for disaster cases. But its focus is on power consumption, since the application case is for electronic staff badges, which have tiny battery storage.

The technique of hybrid networking combining ad-hoc networking and cellular working has drawn attention, and correspondent schemes have been proposed. Multi-hop Radio Access Cellular System (MRAC) [5] focuses on achieving both high speed/high-capacity and good area coverage. However MRAC only allows single or double hops via a hop-station and [5] has not shown the routing mechanism in detail. In article [6] ODMA changes the mode to multi-hopping according to data transmission rate. Mobile stations located in a low bit-rate area transmit data to a high bit-rate-area node via multi-hopping.

Integrated Cellular and Ad-hoc Relaying System, iCAR [7] is a load balancing scheme in wireless network, which introduced ad-hoc relay stations, placed at strategic locations within a cell to divert traffic from a congested cell to a non-congested cell. In a case of large disaster like earthquake, most cells might be congested, so the system will not work well.

Article [4] proposes an ad-hoc routing protocol using unicast-based route discovery in a hybrid wireless network. The protocol requires all the nodes that can access BS to periodically transmit control information on CCCH (Common Communication Channel) [4], which is monitored by the other nodes. Such periodical transmission is like beaconing, which has two defects:

1) traffic overhead
2) obsoleteness in routing table due to mobility

The information carried in the periodical beaconing is for the updating of a node's routing table. If the moving speeds of the nodes are high, the routing tables are unlikely to be updated quickly enough. And packets re-send situation will have to happen. This issue will be discussed again in section 5.3.

Furthermore the protocol requires a channel of TDMA, which restricts its application on systems with other multiple access schemes.

At last, the network model in [4] only consists of one cell, and it is presumed that the BS of the cell is still functional after the earthquake. But in actual disaster cases, situations might be more complicated, where some BSs are down while some others are still functional.

According to the above, the study of this article is to propose a routing protocol that is beaconless, on-demand, and performs better against mobility. Meanwhile a model with more cells and BSs involved is to be discussed.

3. NETWORK MODEL

Here in this paper we consider a model which is composed of a certain group of adjacent cells, each of which is supported by a BS. However, due to a disaster, some of the BSs are down and then the cells on them are dead; further more, even in a cell where the BS is still up, some deterioration of connectivity happens, which means some mobile nodes within the cell area do not have direct connection with the BS, possibly due to damage in wireless unit, or interruption in channels, or blocking by obstacles.

To recover from such deterioration of connectivity, a hybrid network scheme is proposed here, which combines centralized cellular networks and ad-hoc networks. The centralized cellular networks connect the functional BSs and the nodes directly, while the ad-hoc networks connect the nodes each other. The nodes that connect directly to BSs operate in cellular mode; while nodes that do not receive signals from any BS communicate with their neighbors and operate in ad-hoc mode. A node operating in cellular mode is designed to work as gateways, which is able to receive packets from neighbors too and helps to relay packets from a node operating in ad-hoc mode.

In normal circumstances, nodes work in cellular mode. But in the cases of disaster, some nodes that cannot receive signals from BS directly switch to ad-hoc mode. A route connecting such a node to a BS is attempted to be built, and the node requires to its neighbors to relay the packets to a BS. Fig 1 depicts the network model.

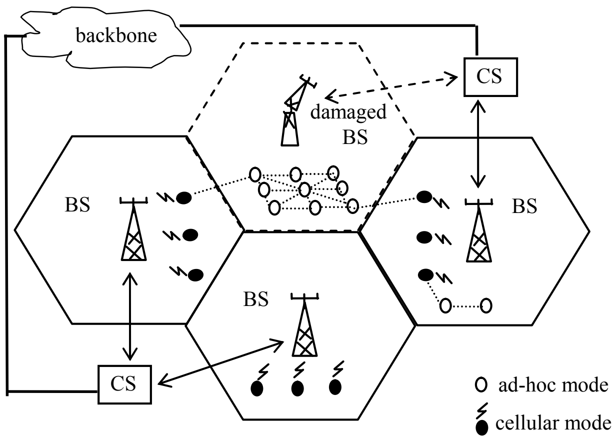

Fig 1. Model of Hybrid Networks

4. PROCEDURES AND PROTOCOL

A node will perform the following procedures to establish calls.

1) If the node detects signal from a BS and is able to build connection in between, the node tries to establish the call directly via the BS with the usual cellular call routines.
2) If the node cannot detect signal from any BS or fails to make connection with it, it switches to work in ad-hoc mode, and tries to find a route to a BS via multi-hopping. First, with the help of GPS, it will determine its own location; and with pre-stored data, it will find out a certain numbers of near-around BSs and their locations; then it will attempt to access the nearest BS with the RSSI-Enhanced BLR routing protocols described below. If the attempt fails (e.g. the BS is destroyed after the disaster), the node will try a secondary nearest one. Such attempts will repeat until at least one operational BS is found or all BSs have been gone thru.
3) If the node has access to at least one BS, it records the BS in its entries, and tries to establish the calls by sending the data to the BS.
4) If the BS is suffering from traffic congestion and the above call is rejected (which is typical after a disaster), the node marks the BS as busy in its entry, and then performs the BS searching attempts described in 2), and tries to make the call via a newly found BS.

A marked "busy" BS will be unmarked after a certain interval. And calls will be tried on non-busy BSs in the ascending order of distances.

To try to access a BS with the knowledge of the BS's location, or to send a packet to the BS, a source node performs according to the following RSSI-enhanced BLR protocol [3]:

1) The source node put the location coordinates of both the BS and of itself into the packet's header. Then the source node broadcast the packet.
2) If there are nodes within the transmission range, they will receive it. Now each of these neighboring nodes has the location information of the source and the destination, and in associate with its own location, they can via geometrical calculation determine whether they are in a forward area and evaluate a progress. The nodes within the forward area become candidates of next-hop relay and apply a Dynamic Forwarding Delay mechanism. The candidate with the shortest delay time will make a broadcast as a relay of the packet and other candidates will receive it and cancel the scheduled relay.
3) Each candidate node will replace in the packet header the previous node's location coordinates with its own location coordinates. Thus, during the whole routing, each intermediate node repeats a same behavior and eventually the packet will reach a node that has direct connection with the BTS.
4) This node will send the packet content to the BS via the uplink. And the forwarding routing is finished.

Fig 2 explains the protocol with an example: node S is to send a packet to BS. First S retrieves the location information of itself and BS, then the direction S→BS is known. Then S broadcasts the packet, and A, B, C1, and C2, which are within the broadcast range, hear the broadcast. But only C1 and C2 are the candidates of next hop, since A and B know they are not in the forward 60 ° sector. Then after some delay, either C1 or C2 will re-broadcast the packet first. The delay time relies on the delay function in [3]. Here let us assume C1, which becomes the next hop and broadcasts the packet with a same range. Since C2 is within the 60 ° sector, C2 should be able to hear C1's broadcast, and thus will not do the relay. In a similar manner, the packet is relayed via C1, D, E, and G. G is a gateway, so the packet is successfully delivered to BS. The sequence of the hops is recorded in the packet, so backward delivery is possible.

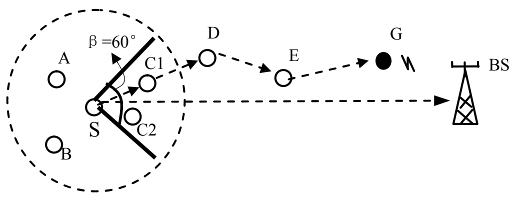

Fig 2. Routing with Enhanced-BLR

5. SIMULATION

5.1. Simulation Conditions

Simulation has been done with OmnNet++. Below are the simulation model and the process.

1) The network is modeled with a certain number of BSs and mobile nodes in an area of a square. Each a BS is serving in the middle of its cell.
2) As if after a disaster, a part of BSs are destroyed. Such BSs are selected by random, and the ratio is according to *FBR* (defined below).
3) According to *IDCNR* (defined below), some of the mobile nodes are randomly selected to have direct access to their own nearer cell. Such nodes may work as gateways as long as they keep the direct access to a BS. Meanwhile, the other nodes do not have the direct access and operate in ad-hoc mode.
4) The nodes are moving at an average speed *v*, changing directions randomly every a certain interval.
5) A node that operates in ad-hoc mode can communicate with another node with a largest distance *d*, noted as communication range.
6) During the test period, nodes are randomly selected to send messages, which initiates route finding.

Such a process is repeated 300 times and the result data is averaged.

Fig 3 depicts the network model of simulation.

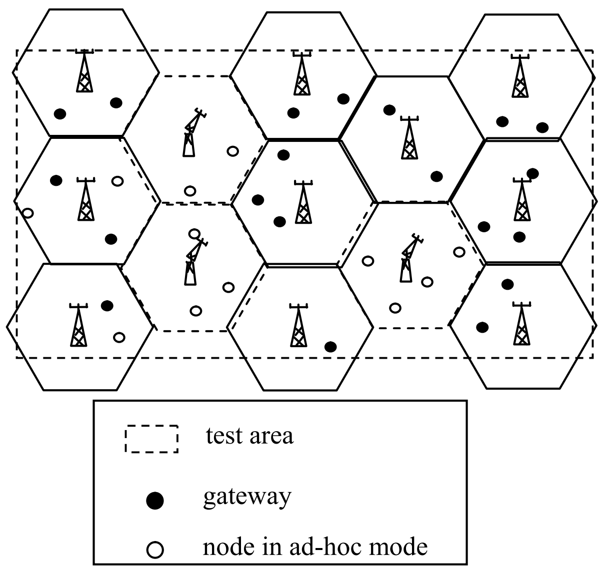

Fig 3. Simulation Model

5.2. Metrics

5.2.1. IDCNR

At the initial state, a number of all nodes are selected to have direction connection with a BS. In article [4], DCNR is defined as the ratio of nodes in a cell to be able to access

BS directly. Here in our simulation, the nodes are moving and the status of direct connection may change. So here we redefine it as Initially Directly Connected Node Ratio (IDCNR) as a pre-set value for the simulation to determine the ratio of such nodes.

$IDCNR = m_1 / N$ (1)

where m_1 is the number of nodes initially have direct connection to a BS, and N is the total number of the nodes in the test area.

5.2.2. FBR

As after a disaster, a certain BSs are destroyed and the rests remain functional. Functional BS Ratio (*FBR*) is defined as

$FBR = m_2 / M$ (2)

Where m_2 is the number of functional BSs after a disaster and M is the total number of BSs before the disaster.

5.2.3. Reachability

A node is called reachable if it can either directly connect to a BS or find at least one route to a BS via multi-hopping within a certain maximum hop-count. Such a maximum hop-count is noted as *MR*. *MR* = 1 if a node connects to a BS directly. As in [4], reachability is defined as the ratio of the nodes that are able to reach BS directly or by multi-hopping.

5.2.4. Average Re-send Times per Packet

With the RSSI-enhanced BLR routing protocol described in this article, when an intermediate node fails to find a next hop to forward the packet, the fact of re-sending the packet by the source node will happen as a retry. To successfully send a packet to the destination finally, several times of retry may be needed.

With the protocol proposed in [4], which requires periodical beaconing, a same retry mechanism also exists.

As a measurement of the performance, average re-send times for sending a packet, noted as *AR*, is defined as a metric.

$AR = RS / TP$ (3)

where *RS* is the total times of the happenings re-sending packets, and *TP* is the number of the total packets to be sent during the test time. *RS* and *TP* are recorded during experiments.

5.2.5. Relative Node Density (RND)

Assume that each node has the same communication range. Let *K* be the smallest number of nodes required to cover a whole test area, i.e., every spot of the test area is within communication range of at least one node. *RND* is defined as

$RND = N / K$ (4)

In the following, if not specified, the values of simulation parameters are as in table I.

TABLE I

Default Values of Main Parameters

Parameters	meaning	Value
C	Cell number	10x10
r	Cell radius	340m
N	The total number of the nodes in the test area	90000
v	Average moving speed of nodes	3m/s
d	A node's communication range when in ad-hoc mode	20m
MR	Maximum number of hopping steps to send packets	10
FBR	Functional BS ratio	0.7
IDCNR	Initially directly connected node ratio	0.5
RND	Relative node density	1

5.3. Results

5.3.1. Reachability

Fig 4 shows the results of reachability versus *IDCNR*, with the curves of different *FBR*. It can be seen that either functional BS ratio is high or low, the reachability is larger than the correspondent *IDCNR*, which means that the service coverage is extended for non-direct-connection users. It also shows that when *FBR* is large (0.7 or 0.9), the left part of the curves climbs up fast, which means that if there are many enough functional BSs, the proposed routing algorithm helps the initially unconnected nodes with recovery of connection in a quantity that is times of the initially connected nodes.

Fig 5 shows the results of reachability versus *INCNR*, with the curves of different *MR* values. As expected, it is seen that the larger the maximum allowed hop-count is, the larger the reachability is. When initially 20% of the nodes have direct connection, *MR = 4* provides ~60% of service coverage, *MR = 6* provides ~85%, and *MR = 10* reaches ~90%. It indicates that if the damage and influence of the disaster is serious, allowing more hops of relay can significantly relief the problem of out-of-service of many nodes, although large number of hopping may bring some side effects.

Fig 6 shows the impact of node density. It is seen that the higher the density is, the higher reachability is, which is easy to explain – because higher density means more nodes to join in relaying. It is seen that the reachability can be maintained fairly high if node density is fairly high, e.g., when *RND* = 0.75 and *FBR* = 0.7, the reachability is over 0.7.

5.3.2. Average Resend Times (AR)

Experiments on comparison of the average re-send times of the Enhanced BLR protocol and the Beaconing protocol in [4] are done. Fig 7 shows the impact of the average speed of the nodes.

It is obvious to see that the impact of the beaconing protocol is much higher than then Enhanced-BLR protocol. This is because when the mobile nodes' speed is fairly high, the stored routing table might be outdated as the topology has changed due to mobility. It is imaginable. E.g. when a relaying node is moving at speed 10 m/s, it takes only 2s to travel the distance of communication range *d*, so the topology is significantly changed.

6. CONCLUSION

This paper has proposed a location-based ad-hoc routing protocol used in a hybrid wireless system, which combines ad-hoc networking scheme with a centralized cellular network. The focus is to maintain connectivity of nodes to BSs in the aftermath of a disaster. Simulation results shows that even only a small number of the nodes can directly connect to a BS, times of that number of the nodes can find a route to a BS via multi-hopping, which means a great deal of users can recover with service. Numerically, 20% ~ 30% of post-disaster service coverage is possible to be increased to 80% ~ 90% with the help of the proposed recovery mechanism. And it is found that it outperforms the solution in [4] in terms of resistance to mobility.

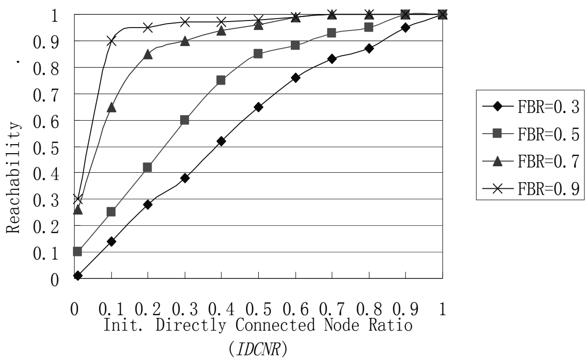

Fig 4. Impact of FBR on Reachability

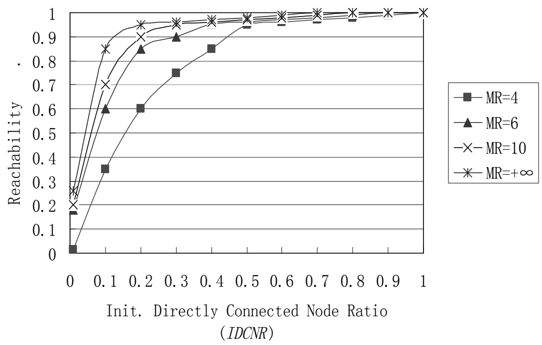

Fig 5. Impact of *MR* on Reachability

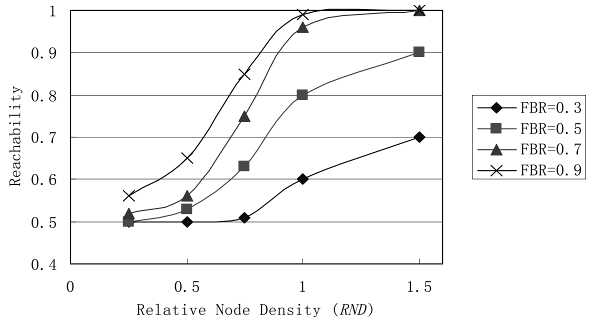

Fig 6. Impact of Node Density

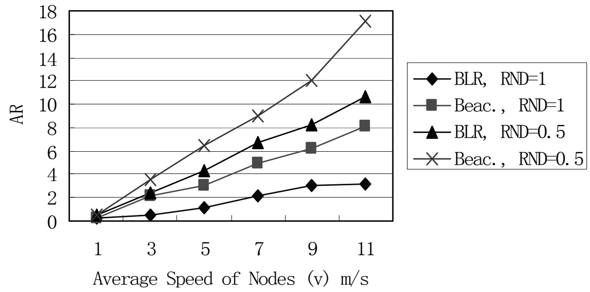

Fig 7. Impact of Speed on Average Re-send Times

REFERENCES

[1] G. Zussman, A. Segall, "Energy Efficient Routing in Ad Hoc Disaster Recovery Networks", Proc. of IEEE INFOCOM 2003, San Francisco, April 2003.

[2] A. Meisser, T. Luckenbach, T. Risse, T. Kirste, H. Kirchner, "Design Challenges for an Integrated Disaster Management Communication and InformationSystem", Proc. of the 1st IEEE Workshop on Disaster Recovery Networks, New York, June 2002.

[3] G. Chen, K. Itoh, T. Sato, "Enhancement of Beaconless Location-Based Routing with Signal Strength Assistance for Ad-Hoc Networks", Trans. of IEICE 91-B(7): 2265-2271, 2008

[4] T. Fujiwara, N. Iida, T. Watanabe, "An Ad-hoc Routing Protocol in Hybrid Wireless Networks for Emergency Communications", icdcsw, vol. 6, pp.748-754, 24th International Conference on Distributed Computing Systems Workshops - W6: WWAN (ICDCSW'04), 2004

[5] Y. Yamao, T. Otsu, A. Fujiwara, H. Murata, S. Yoshida, "Multi-hop Radio Access Cellular Concept for Fourth-Generation Mobile Communications System", Proc. of IEEE PIMRC2002, Lisbon, Sep. 2002.

[6] 3GPP TSG-RAN, "Opportunity Driven Multiple Access", 3GPP Technical Reort, 3G TR 25.924, ver. 1.0.0, Dec. 1999.

[7] S. De, O. Tonguz, H. Wu, C. Qiao, "Integrated Cellular and Ad Hoc Relay (iCAR) Systems: Pushing the Performance Limits of Conventional Wireless Networks", hicss, vol. 9, pp.300, 35th Annual Hawaii International Conference on System Sciences (HICSS'02)-Volume 9, 2002

[8] C. E. Perkins and E. Rayer, "Ad-hoc on-demand distance

vector routing", Proc. WMCSA '99, New Orleans. USA, Feb. 1999, pp. 90-100.

[9] D. B. Johnson, D. A. Maltz. and J. Broch, "DSR: The dynamic source routing protocol for multi-hop wireless ad hoc networks", Ad Hoc Networking. Addison-Wesley. 2001, ch. 5. pp. 139-172.

[10] V. D. Park and M. S. Conon, "A highly adaptive distributed routing algorithm for mobile wireless networks", Proc. INFQCOM '97. Kobe. Japan, Apr. 1997, pp. 1405-1413.

[11] C. Perkins and P. Bhagwat, "Highly dynamic destination-sequenced distance-vector routing (DSDV) for mobile computers", Proc. ACM/SIGCOMM '94 Conference on Communications Architecture, Protocols and Applications, London, UK, Aug. 1994. pp. 234-243.

[12] Z. J. Haas and M. R. Pearlman, "A hybrid framework for routing in ad hoc networks", Ad Hoc Networking. Addison-Wesley, 2002, pp. 221 -253.

A NEW STUDY ON NETWORK PERFORMANCE UNDER LINK FAILURE IN OPS/OBS HIGH-CAPACITY OPTICAL NETWORKS

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ABSTRACT

In this work we analyze the performance and sensitivity to link failure of metropolitan networks based on the technology of optical packet/burst switching (OPS/OBS). We use ring and mesh topologies to evaluate through analytical modeling and computer simulations the impact of link failure on each topology. We adopt the parameters average number of hops and packet loss fraction to evaluate network performance. It is observed that mesh topologies with triple-connection node configuration (3x3) are more robust; consequently in case of link failure the impact of lost data is minimum compared with the other topologies and configurations considered.

Keywords- Network Protection, Photonic Switching, Optical Packets, Optical Fiber Communications.

1. INTRODUCTION

With the popularity of applications such as video streaming and conferencing, IP-TV, VoIP and new interactive services, the amount of traffic through communications networks keeps increasing, generating new challenges to network latency, traffic transport and protection of network and services. This increase in traffic volume due the development of new broadband services and applications is dominant in metro-access networks [1][2][3], requiring speed and granularity, because of its variability (core networks, with multiplexed traffic have other problems). Nowadays, optical network technologies are well defined for core networks, with WDM solutions firmly established [4]. For the metropolitan access level, however, where networks require granularity and flexibility to deal with highly variable traffic, the scene is still open.

With this scenario is important adopt technological solutions to increase granularity without loss of speed and agility, as well as reliability offering greater survivability in case of link failures. There is a tendency to use optical packets and bursts (OPS/OBS) for transport and switching following the same philosophy as conventional optical networks using electronic (IP) packet switching; such solutions are to carry high bandwidths running over single wavelengths; WDM solutions can be used to multiply the capacities discussed here [2]. These are the basic conditions adopted in this work.

This work focuses on metro-access networks with ring and with mesh topologies, the latter being Manhattan St type, and the rings are both conventional (as in SDH/SONET) and reformed connections, as will be seen. We analyze their performances according to established parameters, such as average number of hops and packet loss fraction. Other topologies are not considered here.

Our main purpose in this work is to extend and innovate on previous results [5][6], and evaluate and compare optical network architectures using the conventional models and their variations with modified models, implementing configurations with unidirectional and bidirectional links, and considering link failures for all these topologies. We investigate and find through computer simulations the better architectures, in terms of (higher) capacity and (lower) latency. Simulations save time and energy, and are a valuable tool in network planning; of course, simulations always require validation in test-beds. For the present work, we have chosen basic parameters such as average number of hops and packet loss fraction (PLF) as metrics for network performance. The present proposal of networks in OPS/OBS architectures is original to the best of our knowledge, with switching and routing being performed directly in the optical layer level [3], not requiring OE conversion of optical packet payload at optical nodes, keeping very low network latency.

In Section II, we present the basic theory used in this work and some analytic results; in Section III we present the various configurations of mesh and ring topologies we are using; in Section IV the methodology and dynamics for simulation scenarios are presented. Other sections show the Results, Discussion and Conclusion.

2. BASIC THEORY AND PARAMETERS

This work is based in exploring and analysing the following network parameters: capacity, average number of hops, performance factor, optical packet loss fraction.

The total capacity of the networks considered here is given by the sum of the separate capacity of the network links, valid for mesh and ring (other topologies will behave differently). If the node has R inputs and R outputs, the node configuration is (RxR) [4],[5],

$$C_t = \frac{R.N.S}{\overline{H}} \tag{1}$$

where $\overline{H}$ is the total average number of hops for the optical packets (all packets from all origins to all destinations); N is the number of nodes, and S is the link capacity; the parameter R can be value 2, 3, or 4. The case of nodes in configuration (4x4) is the same as network with bidirectional links.

The traffic was modeled assuming uniform distribution, which means every node generating uniform traffic (number of applications) to every other node except to itself. The effective number of user nodes (simply "users") in the network for this condition is $N_u = N(N-1)$. We define the user-share capacity as C_t/N_u, , which using (1) becomes,

$$C_u = \frac{R.S}{\overline{H}.(N-1)} \quad (3)$$

A figure of merit for network performance is defined as a performance factor $F_p = C_t/\overline{H}$, which can be applied to any multihop environment that follows (1-3). The reasoning is simple: higher capacity with smaller average hop number leads to more efficient networks that will also have lower latency.

The calculation of average number of hops depends on the routing protocol adopted. When we adopt the Store-and-Forward (SF) protocol that is based in the minimum paths matrix gerated by Dijkstra algorithm, the packets/bursts always are transmitted by smallest path to the destination address. When we use Deflection Routing protocol (DR, no optical buffering), in which optical packets that arrive "late" -- when optical switch is blocked by another optical packet of early arrival -- are deflected to the available output port, the calculation is again based in the minimum paths matrix generated by Dijkstra algorithm, but it is not always that the optical packet is sent by shortest path to destination, because deflection may take it through a longer path.

The results of Table 1 use the SF as routing protocol. for average number of hops, capacity and performance factor, and equations (1)-(3). These results will be later used to compare with the simulations results (section VI) using the NS simulator and DR protocol. In the simulation results another useful parameter was considered: the packet loss fraction (PLF); it is defined as the ratio consisting of the difference between number of optical packets that were sent and those that arrived at their destination, divided (normalized) by the total number of generated (sent) packets in the network during the total simulation time.

To analyze impact of link failure, equation (1) for the capacity is modified accordingly:

$$C_t = \frac{(R.N-k).S}{\overline{H}} \quad (2)$$

with k the number of link failures involved. The cases of unidirectional and bi-directional links are separately considered.

3. MESH AND RING TOPOLOGIES

The proposed OPS/OBS optical networks architectures used in this work are based on Manhattan Street (MS, Fig. 1) and Ring (Fig. 2) topologies. The MS is a full-connected mesh topology that has been object of many studies, applicable better to metropolitan optical networks [7][8][9]. In the present work we consider the conventional $N=m^2$ model, where N total number of nodes, and m integer (even or odd), which is an extension of the classical MS model, for which m is always even. Thus, we name node count regular (MS, even) or quasi-regular (MSq, odd). It should be noted that the main feature of the mesh topologies is the option of various paths that the packets can use to arrive at destination address; in case of failure or congestion in a given path, the packet can easily change its route and follow another path.

The optical nodes are interconnected using unidirectional links and the same connections of conventional model of MS plus connections in the diagonal using the same logic connectivity of MS network, leaving the network with more options of paths than MS network with node configuration (2x2). Thus, in the mesh configuration (3x3) we implement a new input and output in the (2x2) optical node. The intelligence of the optical node is the same of our previous model [3] with header recognition and optical switch control electronics, a short fiber delay line (just to allow header processing time), and add/drop functionalities. This new (3x3) model will be also compared with MS topology using bidirectional links, with node configuration being a double (2x2), that is (4x4). In the unidirectional links: a packet from *neighbor* nodes i and j follows from i to j without the possibility of leaving j and return to i; in bi-directional links this possibility may easily occur.

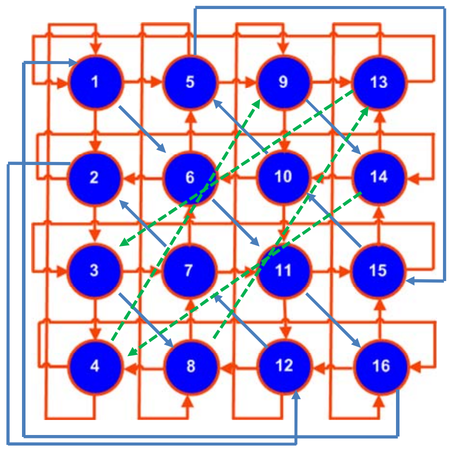

Fig. 1: Optical mesh topology networks: a modified MS-16 with node configuration (3x3)

In the new ring model (Fig. 2) the nodes are connected with next neighbor node in the clockwise and in counter-clockwise directions; we connect with next node after neighbor node being to used only unidirectional links, different of the convention architecture, where the links connect the nodes neighbor in both directions and use bidirectional links.

The optical switching for all MS models (2x2), (3x3), (4x4) and ring (unidirectional and bidirectional) is controlled by fast electronic logic circuits (40ns rise-fall times) operating on packet-by-packet basis, determined only by header processing; thus the overall switching time for all nodes is <2µs for every packet or burst, independent of payload size.

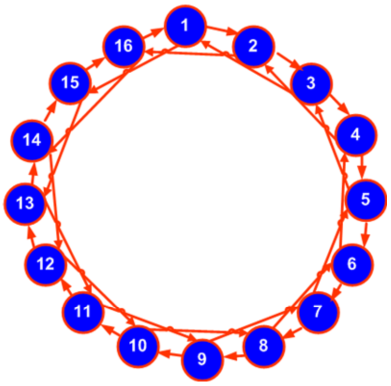

Fig. 2: Optical Ring topology: unidirectional Ring-16 with node configuration (2x2)

These architectures allow the use of deflection routing (DR) as the protocol to resolve contentions and avoid collisions, [8] with the advantage of no need of buffer in the optical layer, which contributes to cost reduction. Electronic buffering at the (external) client interface of the optical node is considered elsewhere [7], and does not affect the results of the present work. Links having typically 10km length, and networks spanning less than 36 optical nodes.

The first evaluation is summarized in Table 1, where we considered the analytic results of parameters such as average number of hops, network capacity and performance factor. In these first evaluations we do not consider link failure and the results are obtained through the use of store-and-forward (SF) protocol, because it is useful in establishing the bases for comparison of the various topologies considered. It is seen in Table I that the new (3x3, unidirectional) and the (4x4, bidirectional) mesh architectures are by far the ones with better performance in the chosen parameters; we will come back to this point later (Section V). It should be further noted that the SF protocol does not consider contention between packets (ideal scenarios for transmission), always takes the best route, but at the cost of significant impact in network latency.

4. Link Failure Protection

In high capacity optical networks it is important to have network protection mechanisms to prevent failures of the system from disrupting services being carried by the network [4]. Here we consider single link failures, that is failures that occur due cuts of fibers or cables, or failure of components in Tx/Rx (transmitter/receiver) pairs [10]. Thus, it is our goal here to study the impact of failure on the OPS/OBS networks, but not to go in detail and evaluate the different mechanisms of protection.

Table II shows the impact of single link failure in parameters as number of hops, capacity and performance factor. Double link failure is three orders of magnitude smaller [10] and needs not be considered.

Topologies		# of nodes N	Average # of hops $\overline{H}$	Network Capacity C_t	Perfor-mance F_p
Unid(2x2)	MSq-9	9	2,0	22,3	11,1
	MS-16	16	2,9	27,3	9,3
	MSq-25	25	3,3	38,1	11,6
	MS-36	36	3,7	48,5	13
Unid(3x3)	MSq-9	9	1,6	40,8	24,7
	MS-16	16	2,1	56,0	26,2
	MSq-25	25	2,5	73,2	28,6
	MS-36	36	2,9	92,1	31,4
Bid(4x4)	MSq-9	9	1,5	60	40
	MS-16	16	2,1	75	35,1
	MSq-25	25	2,5	100	40
	MS-36	36	3,1	116	37,8
Ring (2x2)	Ring-9(bid)	9	2,5	18	7,2
	Ring-9 (unid)	9	2,2	20	9,1
	Ring-16(bid)	16	4,2	18,7	4,4
	Ring-16(unid)	16	3,5	23,0	6,6
	Ring-25(bid)	25	6,5	19,2	2,9
	Ring25 (unid)	25	4.8	25,8	5,3

Table I Network parameters for mesh/ring topologies with unidirectional and bidirectional links.

5. Traffic Simulation Methods and Network Configurations

In this section we summarize the procedures and conditions for simulations in this work. A diagram of the simulation dynamics for SF and DR protocols is shown in Fig. 4. The network architecture models use unidirectional and bidirectional links (treated separately). The traffic distribution is uniform with every node generating the same amount of traffic to every other node during each simulation round time. Every connection was set up using UDP protocol, i.e., no retransmission of the lost packets. In SF mode packet loss is observed to be below 10^{-9}; in the DR mode it is above 10^{-6}, as will be explained later. In Fig. 4 the traffic sources in each node generate optical packets (from client input electronic packets) of same size (500 bytes) and send these optical packets in variable times; the total number of packets generated for simulation rounds is $2x10^5$. We have chosen the MS-type mesh topologies as defined above, MS-16, MSq-25 and equivalent ring topologies with 16, 25 nodes. The optical packets are fixed with a duration equivalent (at link transmission rate of 2.5 Gb/s) to 500 bytes, which we assume to be a representative average in IP-packet networks (variable size packets and bursts will be object of future implementation); link length is 10km for all links.

The results for OPS/OBS network traffic simulations were obtained using the software Network Simulator (NS-

2); for analytic results and data processing and plotting, a standard commercial software was used.

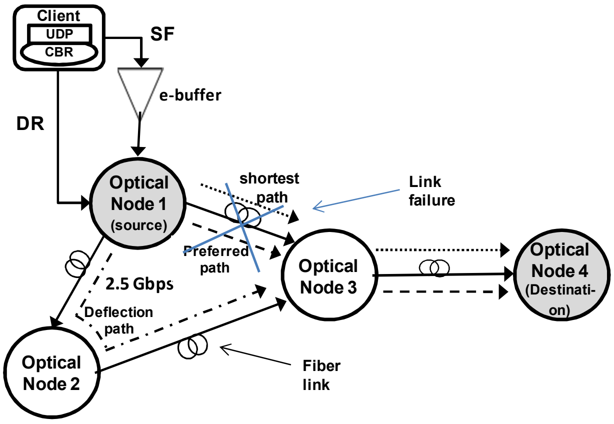

Fig. 4: Simulation Dynamics.

	Topologies	Number of nodes N	Average Number Hops $\overline{H}$	Network Capacity C_t	Performance F_p
Unid(2x2)	MSq-9	9	2,125	21,18	9,97
	MS-16	16	3,017	26,52	8,79
	MSq-25	25	3,347	37,35	11,16
	MS-36	36	3,781	47,61	12,59
Unid(3x3)	MSq-9	9	1,69	38,46	22,76
	MS-16	16	2,16	54,40	25,18
	MSq-25	25	2,56	72,27	28,23
	MS-36	36	2,95	90,68	30,74
Bid(4x4)	MSq-9	9	1,514	59,45	39,26
	MS-16	16	2,142	74,70	34,87
	MSq-25	25	2,507	99,72	39,78
	MS-36	36	3,091	116,47	37,68
Ring (2x2)	Ring-9(bid)	9	2,9	14,66	5,05
	Ring-9 (unid)	9	2,375	17,89	7,53
	Ring-16(bid)	16	4,96	15,63	3,15
	Ring-16(unid)	16	3,479	22,28	6,40
	Ring-25(bid)	25	7,58	16,16	2,13
	Ring25 (unid)	25	4,986	24,57	4,93

Table II. Network parameters for mesh/ring topologies with unidirectional and bidirectional links considering link failure.

6. NETWORK RESULTS AND DISCUSSION

The results presented in figures and graphs in this section compare the average number of hops and packet lost fraction (PLF), and show the impact of link failure on these parameters when failure occur in the more used links [5] , for the various architectures and node configurations considered in this work. The parameter *link load* represents the rate of occupation of the link capacity {0,1}. Results presented here were obtained through computer simulations of optical packet traffic flow on the NS-2 open platform using models developed in our labs.

Figure 5 compares the average of number of hops between the model MS(2x2) with modified model MS(3x3) for networks with 16 and 25 nodes and unidirectional links; Figure 6 between MS(2x2) with MS(4x4) networks; and Figure 7 between models MS(3x3) and MS(4x4).

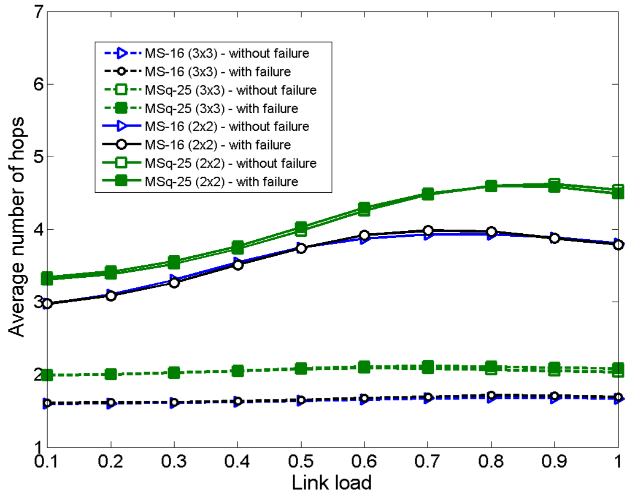

Fig.5 : Average number of hops for MS and MSq (2x2) and (3x3) considering link failure

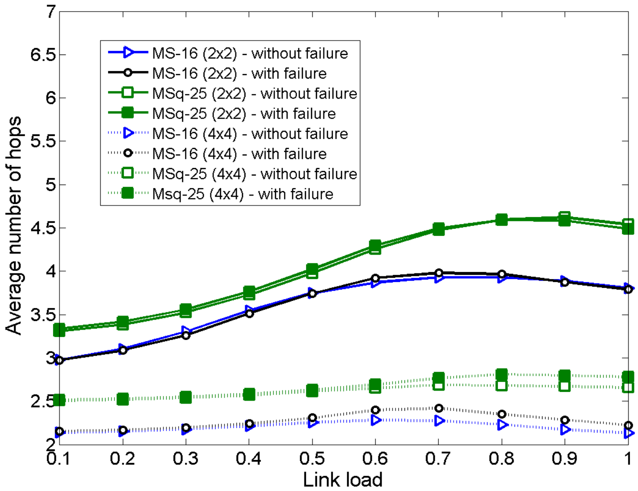

Fig.6 : Average number of hops for MS and MSq (2x2) and (4x4) considering link failure.

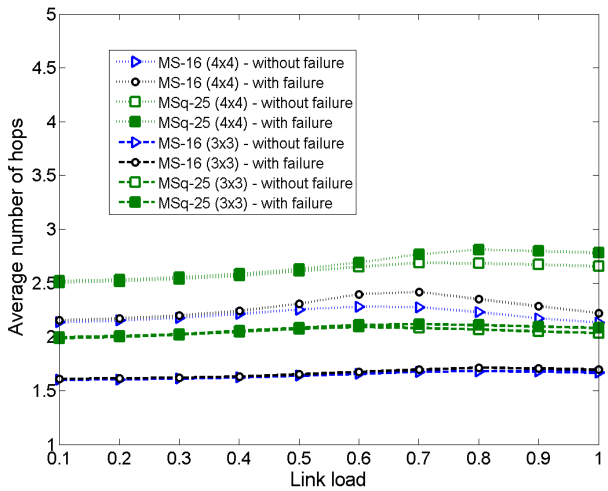

Fig.7 : Average number of hops for MS and MSq, (3x3) and (4x4), considering link failure.

The network model in mesh that presented the best results of average number of hops with and without link failure, as shown by the figures 5, 6, and 7, were MS and MSq with node configuration (3x3). Fig.5 shows that under single link failure the change in average number of hops was not significant for both models (3x3) and (2x2). For model MS and MSq (4x4) we can see in Fig.6 that this model has better results of average number of hops than model (2x2) with and without link failure, but we observe that in load lager than 50% it shows a small but noticeable change.

From Fig.7 we verify that the model (3x3) has better results than (4x4). This result is clearly an innovation, because one would expect that the bi-directional model MS(4x4) had better results due to its larger number of possible paths as compared to MS-(2x2) and MS-(3x3), see Table 1and 2**;** however, the observed result of better performance for MS-(3x3), is interpreted as this topology offers the largest number of different paths between any node pairs for all optical packets, when compared with other models. Of course, the MS(4x4) has the *largest number of paths*, but through DR protocol many packets may thread the same paths several times, with an increasing impact on latency and extra "artificial" traffic. Furthermore, the probability of an optical packet returning back directly to its node of origin is 25% for bi-directional MS(4x4); whereas for unidirectional MS(3x3) this possibility does not occur (zero probability) because the optical packets always move to a different node.

For the Ring models in Figure 8, we obtained a lower figure for average number of hops with the new unidirectional 2x2 Ring model, than that for the conventional bidirectional ring, with and without link failure. Another significant result is that the impact of link failure on the new model is clearly smaller than on the conventional ring, especially for link loads up to 50%,, which is explained by the existence of new alternate connections (as in Fig.2).

Figures 9 & 10 present results of the traffic analysis for PLF in the networks studied here. Before analyzing these results, it should be noted that the absolute values of PLF seems somewhat high because we are actually interested in the relative/comparative behaviour of the networks with different topologies and node interconnections. That has been carefully kept under the same conditions. The packet losses however are higher than expected because intentionally we did not introduce adequate electronic buffering and collision detection protocols – optical packet losses occur mostly at node input, due to lack of synchronism when the optical packets generated at a given node are inserted in optical transport layer. In practical networks, of course, this is avoided with electronic buffering (e-buffers) at the electronic layer of the optical access node [7] and simple collision detection mechanisms at the optical switch input.

On the other hand, we also wanted to investigate quantitatively this controllable loss mechanism. Another loss factor which could impact (but does not) in the different situations analyzed is the time-to-live (TTL) of the optical packets in the network; which can also be controlled by electronic buffering.

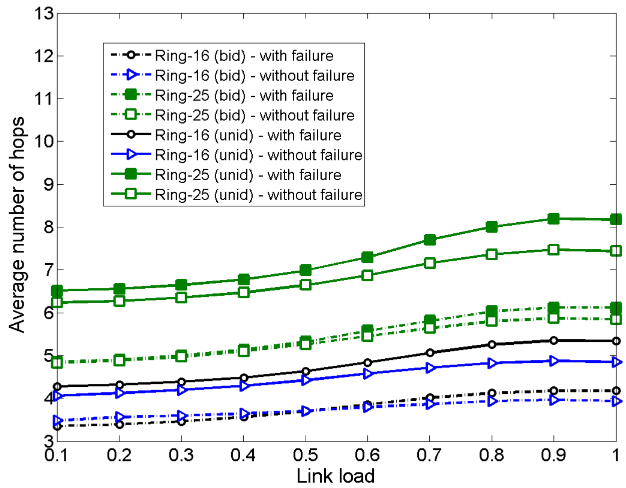

Fig.8 : Average number of hops for MS and MSq, (2x2) and (3x3), considering link failure.

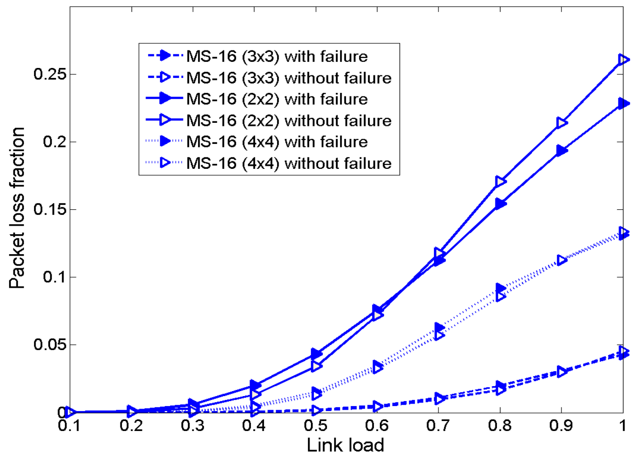

Fig.9 : PLF for mesh networks MS and MSq, with (2x2), (3x3) and (4x4), with and without link failure.

That would keep deflections in DR protocol to a minimum; but then again, we wanted to investigate how much TTL influences. Results in Figures 9 e 10 demonstrate quantitatively that both loss mechanisms (collisions and TTL) add to a rather small PLF, indicating that e-buffers are not necessarily required, contributing to equipment reduction (a techno-economical aspect of interest for network equipment cost). As seen in Figs. 9 and 10, the MS-16 network, with a configuration (3x3) is the one with smallest PLF, with or without failure, meaning also that the impact of link failure is negligible in this case.

Notice that for configurations (2x2) and (4x4) losses are larger both for network under normal operation or with fault. In the case of MS-16 (2x2), PLF is larger simply because it has smaller number of links and paths; for the configuration (4x4), we interpret that although it has more links (25% more than (3x3)), it actually has smaller number of different optical paths for traffic between nodes, which is what really matters in DR .

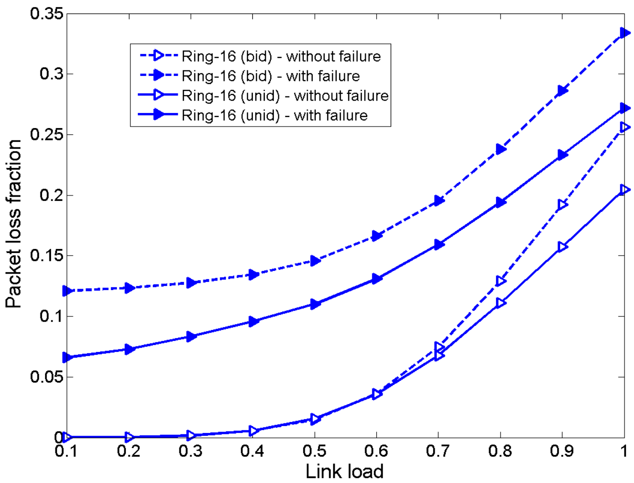

Fig. 10: PLF for Ring-16, uni- and bi-directional.

In contrast, it is interesting to note in Figure 10 that Rings are much more sensitive to link failure than meshes (again due to number of different optical paths between nodes), and the bidirectional ring is worse than the unidirectional ring proposed here, because of the more limited number of optical paths between nodes. Once again, we note superior performance, robustness and efficiency of mesh compared to other topologies.

7. CONCLUSION

We have extended and innovated on previous results [5][6], evaluating and comparing mesh and ring optical network architectures using the conventional models and their variations with modified models, implementing unidirectional and bidirectional links, and considering single link failures for all topologies. We performed comparative analysis of network capacity, average number of hops and packet loss fraction for the various network configurations considered.

We observe for the modeled networks that those with largest number of paths between different node pairs tend to have smallest average number of hops; which is the case for MS(3x3, unidirectional), that has the largest number of different link connections, which in turn leads to least number of hops in end-to-end connections.

We believe that results obtained here can be useful as an input to network planning, in that similar traffic is used in all cases. Equipment cost and availability were not considered. We regard one attractive advantage in our proposal that these innovations should not have significant impact on network installation. But then again, it should be clear that computer simulations are good approximations and allow easy variation of parameters, at no cost, but attention must be kept to real world applications, and their practical limitations. Future work will include cost of link usage and node failure, to further investigate how the present results would alter.

REFERENCES

[1] I.Tomkos, S. Azodolmolky, D. Klonidis, M. Aggelou, and K, Margariti "Dynamic Impairment Aware Networking for Transparent Mesh Optical Networks: Activities of EUProject DICONET," *Proc. ICTON'2008,* June 2008- Athens, Greece.

[2] C. Develder, A. Stavdas, Fabio Neri, J. Solé-Pareta, N. Le Sauze, and P. Demeester, "Benchmarking and Viability Assessment of Optical Packet Switching for Metro Networks," *J. Lightwave Tech.*, Vol. 22, no. 11, Nov. 2004.

[3] F.R. Barbosa, D. Maia, L. Pezzolo, A. C. Sachs, M.R. Salvador, "Optical Packet Switching Node for Metro-Access Networks", paper We4.P-160, *Proceed. 29th. European Conf. on Optical Comm.– **ECOC'2003***, Rimini, Itália, Sept. 2003.

[4] R. Ramaswami, K. N. Sivarajan, "*Optical Networks: a practical perspective*", Morgan Kaufmann, Academic Press, Boston, USA 2nd Edition, 2002.

[5] I. B. Martins ; F. R. Barbosa ; L. H. Bonani ; E. Moschim . "Evaluation of Throughput and Protection in Optically Switched Metropolitan Networks Architectures", *Photonic Network Communications PNET*, number 2, April 2010. DOI: 10.1007/s11107-010-0249-z

[6] I. B. Martins, F. R. Barbosa, L. H. Bonani, and E. Moschim "Improved Method for Evaluation of Network Throughput and Protection in Future Optically Switched Metropolitan Networks," *Proceed. Advanced Intl Conf. on Telecomm. - AICT'2009,* May 2009, Venice/Mestre, Italy.

[7] L.H. Bonani, F. Rudge Barbosa, E. Moschim, and R. Arthur, "Analysis of Electronic Buffers in Optical Packet/Burst Switched Mesh Networks," *Intl. Conference on Transparent Optical Networks-ICTON-2008*, June 2008 – Athens, Greece.

[8] S. Yao, B. Mukherjee, S. J. Yoo, and S. Dixit, "Unified Study of Contention Resolution in Optical Packet Switching Networks," *IEEE J. Lightwave Tech.,* vol.21, no.3, p.672, March 2003.

[9] I. Chlamtac and A. Fumagalli, "An Optical Switch Architecture for Manhattan Networks", IEEE J.Select. Areas Communic. Vol.11, no. 4, .550, May 1993.

[10] D. A. Schupke and R. Prinz. "Capacity Efficiency and Restorability of Path Protection and Rerouting in WDM Networks Subject to Dual Failures", *Photonic Network Comm PNET*, Vol 8, no. 2, p.191, Springer, Netherlands Sept. 2004

BUSINESS SCHEME FOR SHIFTING FROM EXISTING NETWORKS TO TRUSTED GREEN NETWORKS

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ABSTRACT

The future networks have yet to be defined. These are not represented by the next generation of the Internet and they need to satisfy requirements for the sustainability of mankind. These are called Trusted Green Networks (TGNs) in this paper. Although TGNs offer marvelous concepts and excellent functions, they will not always be widely deployed. There have been several initiatives to develop future networks. Their purpose is developing innovative technologies, but not including deployment schemes. We selected "sustainability", "trust and security", and "solving the digital divide by location" as concepts underlying TGNs and clarified their requirements. A business scheme is also proposed that boosts the shift from existing networks to TGNs. And the network layer model of TGNs is introduced.

Keywords—Internet, Future network, Green network, Cap and Trade, Business scheme

1. INTRODUCTION

There are many kinds of communication networks such as PSTN, the IPv4 Internet, and 3G-mobiles throughout the world that enables communication between humans and humans, humans and computers, and computers and computers. These communication networks have made the main contribution to mankind until now. However, various large-scale problems have come to the fore. Threats such as spam mail or computer viruses have increased very rapidly on the Internet and user's intellectual property rights have been infringed [1]. The amount of communication traffic is also expected to increase very rapidly. The calorific value of all our communication devices will represent 10% of the total calorific value in Japan by 2025 [2]. By then, the digital divide between developed countries and developing countries will also be a huge problem [3].

The future networks are needed to solve above three problems for ensuring the future of mankind in adding to improve existing networks. We call these trusted green networks (TGNs) in this paper.

TGNs may not be able to solve these problems, if they cannot be smoothly or widely deployed. For example, IPv6 was standardized to solve the shortage of IPv4 addresses in 1998 [4]. Unfortunately, its penetration was less than 1.5% in 2009 [5, 6]. The probability of penetration will be exceedingly low in the future. However, 3G-mobile services started in 2001 and 2G-mobile systems have efficiently shifted to 3G-mobile systems throughout Japan [7]. The main reasons may be that shifts are necessary for carriers to operate and users demand 3G-mobile systems so that they can use the mobile-Web and Internet mail.

There have been several initiatives to develop future networks. For example, the National Science Foundation (NSF) in the USA has initiated the Future Internet Design (FIND) [8], and promoted the Global Environmental for Networking Innovations (GENI) [9]. In Europe, the Future Internet of the cross-European Technology Platforms (X-ETPs) Group has tried to define research challenges and identify important trends in the future [10]. The European Future Internet Initiative has aimed to establish a Future Internet pan-European coordinated partnership that will bring about clear benefit for Europe [10]. In Japan, the National Institute of Information and Technology (NiCT) has initiated the AKARI project, which is a core research group to study the New Generation Network (NGNW) [11].

The purpose of these initiatives is developing innovative technologies, but not including deployment schemes. I believe there will be no sense in developing technologies, if no Future Internet will be deployed. Hence, demands for TGNs will be clarified based on such admirable ideas in this paper and a scheme to shift the existing network to TGNs will be proposed.

The remainder of this paper is divided into seven parts. First, serious problems related to the future of mankind and existing networks are introduced. After that, I consider the necessity that next generation networks will be widely and smoothly deployed based on the relationship between design concepts and the introduction rate at which existing networks are being deployed in Section 3. I introduce several projects and initiatives to study the future networks in Section 4 and clarify the demands for TGNs and define TGNs in Section 5. A scheme that boosts shifting from existing networks to TGNs is proposed in Section 6. Because of introducing the TGN-trading market to boost shifting, network business model will be changed. I introduce a new business layer model based on OHMN (the open heterogeneous mobile networks) [12] for TGNs in Section 7. Finally, conclusions are drawn in Section 8.

2. SOCIAL PROBLEMS WITH COMMUNICATION NETWORKS

Three social problems are considered in this chapter, but not technical problems with communication networks.

2.1. Energy saving by communication networks

Global warming is a critical problem affecting sustainability. Information and Communication Technology (ICT) will become one of the most efficient tools for solving this problem. However, because of the rapid increase in communication traffic, the calorific value of communication devices is expected to increase rapidly. Japan's Ministry of Economy, Trade and Industry has estimated that the total amount of traffic will be 121 Tbpas by 2025, and the total amount of electricity consumed by communication devices will be 1033 x 10^{12} Wh by then [2]. Green Touch™ is a global consortium organized by Bell Laboratories whose goal is to create the technologies that are needed to make communication networks 1000 times more energy efficient than they are today [13]. A thousand-fold reduction is roughly equivalent to being able to power the world's communication networks, including the Internet, for three years using the same amount of energy that it currently takes to run them for a single day.

TGNs must introduce such energy efficient technologies. They must also be deployed throughout the world as quickly as possible.

2.2. Lack of trust and security

Threats to computers such as spam, computer viruses, hacking, and phishing are increasing very rapidly. The security of computers is in crisis. Users have to pay a great deal of attention to such malware and attacks, while accessing the Internet. The lack of trust is a serious problem for the sustainability of mankind.

TrendLabs has reported more than a 2000 percent increase in Web threats from the beginning of 2005 and the end of 2008 [1]. As updates grow more numerous, network administrators are spending greater amounts of time in managing them. Remote or mobile workers are particularly vulnerable, as they may not receive pattern file updates for hours or days, depending on how long they are off fast networks. Clearly, continual pattern file updates of this magnitude are not sustainable over time. Greater numbers of pattern files generate larger amounts of heat, proving a greater menace in terms of global warming.

Similarly businesses and consumers will continue to suffer from data leaks, financial losses, identity theft, and damaged reputations in the future.

Mobile phones play a critical role in our everyday lives, making them an attractive target for malware authors as they contain confidential personal information. Some mobile phones have IC-chips available enabling them to be used as credit cards or in lieu of cash. Hence, it is not surprising to see mobile malware enter the threat landscape. PC-based threats are already appearing on smartphones and cybercriminals are typically attracted to large target populations, such as increasing iPhone user bases. Cracked iPhones are already experiencing problems in their inability to receive updates from Apple Inc., making them more susceptible to malware attacks.

Because most malware comes from compromised computers, removing compromised computers that are compromised will efficiently decrease those that need to be treated on TGNs.

2.3. Digital divide by location

The lack of communication networks is almost equal to a lack of knowledge. Communication networks are necessary for the sustainability of mankind. However, there is a huge difference in penetration between locations. The International Telecommunication Union (ITU) has collected the most comprehensive range of statistics on the penetration of ICT, its accessibility, and use. Fig. 1 is a bar chart of the penetration per 100 inhabitants of principal communication systems in 2007. Fig. 2 plots the penetration per 100 inhabitants of the Internet from 1997 to 2007. The ITU has pointed out three main issues with the penetration of ICT in Fig. 1 [3].

- Developments in the mobile sector have been able to particularly change the ICT landscape. By the end of 2007, almost one out of two people had a mobile phone. In Europe, penetration has surpassed the 100% mark. More than one out of four Africans and one in three Asians have a mobile phone. A high level of competition and a decrease in prices have substantially been able to reduce the digital divide in the area of mobile telephones.
- Fixed telephone lines remain the exception, especially in Africa, where mobile phones are clearly dominating, and penetration is three per 100 inhabitants, by far the lowest in the world. The limited availability of fixed lines has also been a barrier to the uptake of fixed broadband and it is most likely that Africa's broadband market will be dominated by mobile broadband. ITU started collecting data on mobile broadband subscribers in 2005 and their data indicate that while uptake is on the rise, the rollout of mobile broadband services is concentrated in the developed world. Falling prices and the increasing licensing and availability of 3G is expected to change this over the coming years.
- The digital divide remains a major problem in terms of the Internet and especially broadband uptake. While fixed broadband penetration is growing rapidly and has reached around 15% in Europe and 10% in the Americas, it stands at less than .5% in Africa. Internet use generally remains low in Africa, especially where only 5% of the population is online, compared to over 40% in Europe, the Americas, and Oceania.

The amount of traffic is expected to increase very rapidly in the near future, as explained in the previous chapter. TGNs have to be easy to introduce in not only developed countries but also developing countries, and they may have to have wireless Internet capabilities.

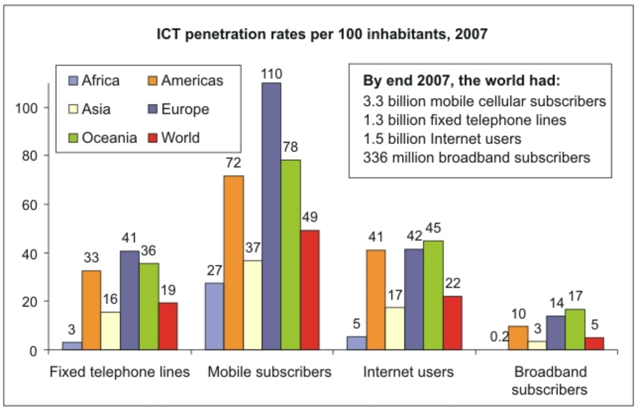

Fig. 1 ICT penetration levels around the world, by region, 2007

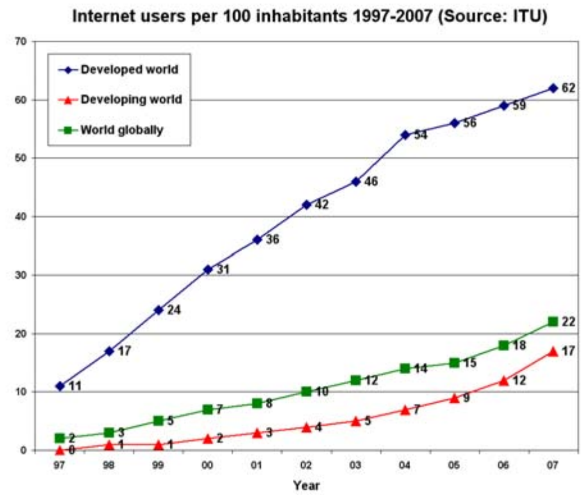

Fig.2 Internet penetration level from 1997 to 2007

3. GENERATION SHIFT IN EXISTING COMMUNICATION NETWORKS

I will try to clarify why the generation shift in communication networks has made a steady progress by analyzing existing networks.

3.1. IPv6

Internet Protocol version 6 (IPv6) is the next-generation Internet protocol. The main driving force in the redesign of the Internet Protocol is the predictable depletion of IPv4 addresses. IPv6 was defined by the Internet Engineering Task Force (IETF) as RFC 2460 in December 1998 [4]. The changes from IPv4 to IPv6 fall primarily into five categories:

(1) Expanded Addressing Capabilities;

IPv6 has increased IP addresses from 32 bits to 128 bits, to support greater levels of addressing hierarchies, greater number of addressable nodes, and simpler automatic configuration of addresses. The scalability of multicast routing will improve by adding a "scope" field to multicast addresses. A new type of address called an "anycast address" is defined, which is used to send a packet to any one of a group of nodes.

(2) Header Format Simplification;

Some IPv4 header fields have been dropped or made optional to reduce the common-case processing cost of packet handling and to limit the bandwidth cost of the IPv6 header.

(3) Improved Support for Extensions and Options;

Changes in the way IP header options are encoded allows for more efficient forwarding, less stringent limits on the length of options, and greater flexibility for introducing new options in the future.

(4) Flow Labeling Capability;

A new capability has been added to enable the labeling of packets belonging to particular traffic "flows" for which the sender requests special handling, such as non-default quality of services or "real-time" services.

(5) Authentication and Privacy Capabilities;

Extensions to support authentication, data integrity, and (optional) data confidentiality have been specified for IPv6.

The working IPv6 ratio for the top-10 countries was less than 1.5% in 2009 [5, 6]. IPv6 has not been adopted as quickly as its designers had expected. Since IPv6 is not backward-compatible with IPv4, both clients and servers have to deploy IPv6 to make full use of it. As IPv6 provides few immediate benefits apart from a larger address space, the operational community has had little motivation to deploy it.

A backward-compatibility is indispensable for a successor to be adopted by a subscriber and on operator.

3.2. IMT-2000[1]

International Mobile Telecommunications-2000 (IMT-2000) was standardized in 1999. The four main reasons it was standardized were to:

(1) Achieve mobile multimedia.
- Faster transmission rate with a maximum of 2Mbps
- Mobile-Web

(2) Achieve global mobility.
(3) Achieve high-quality voice communication.
(4) Achieve spectral efficiency.

NTT DoCoMo started its 3G mobile service in 2001, which was the first in the world. NTT DoCoMo also started its i-mode in 1999, which was the first successful mobile-Web service in the world. Other carriers started 3G-mobiles and similar mobile-Web services within a short time in Japan. Fig. 3 plots the penetration rate for 3G-mobile and mobile-Web from 2000 to 2010 throughout Japan [7]. Both of them increased very rapidly. Carriers in Japan had to shift rapidly from 2G to 3G to solve the shortage of frequency around the Tokyo metropolitan area.

[1] IMT-2000 is better known as 3G mobile communication.

However, users demanded a mobile-Web service, which itself required fast a wireless communication network.

Both carriers and users who were major stakeholders in 3G-mobiles were compellingly motivated to deploy 3G-mobiles.

Table 1 lists the percentage of carriers, which have introduced 2.5G and 3G-mobiles in Japan, Europe, USA, and China. Russia is included in Europe. These data came from Wapedia Wiki [14]. Most major carriers in the world have already provided 2.5G or 3G-mobile services. This high rate of introduction has depended on demands for mobile-Web and multimedia-SMS.

Carriers and users clearly have intense motivation to use 3G-mobiles.

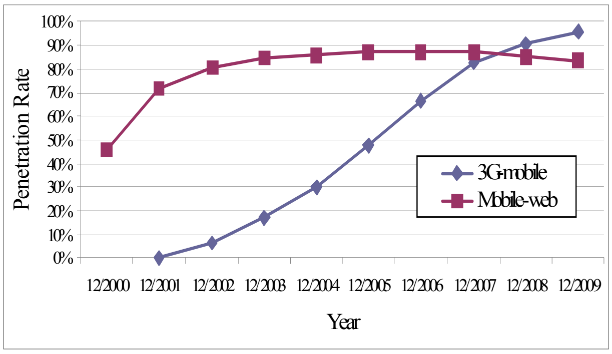

Fig. 3 Penetration rate of 3G-mobile and i-mode in Japan

Table 1 Introduction rate of 2.5G and 3G-mobile in major region or countries

Country or Region	2.5G-mobile (%)	3G -mobile (%)
Japan	0	100
Europe (no MVNO)	44	61
USA (major carriers)	50	100
China	0	100

4. R&D Initiatives for the Future Networks

Several initiatives in some counties or areas have already started to develop the future networks.

The National Science Foundation (NSF) in the USA has initiated the Future Internet Design (FIND) research program [8]. FIND invites the research community to consider what the requirements should be for a global network 15 years from now, and how we could build such a network if we were not constrained by the current Internet, i.e., *if we could design it from scratch.*

At the same time, NFS is promoting the Global Environmental for Networking Innovations (GENI) [9]. GENI is a suite of research infrastructures rapidly taking shape in prototype form across the United States.

Three main issues are being studied by academic and industrial researchers:

- Science issues, where we cannot understand or predict the behavior of complex, large-scale networks.
- Innovation issues, where we currently face substantial barriers to innovation with novel architectures, services, and technologies.
- Social issues, where we increasingly rely on the Internet but are unsure that we can trust its security, privacy, or resilience.

The ten major research areas are:

- Content-distribution services
- Disruption-tolerant networks
- Novel mobility architectures
- Novel routing architectures
- Reliable global networks
- Experimental methodologies
- Experimental exploration of theoretical models
- Network science and design
- Social and behavior aspects of global networks
- Virtualization architecture

The Strategic Research Agenda (SRA) on the Future Internet of the cross-European Technology Platforms (X-ETPs FI) Group in Europe has aimed to define short, medium, and long-term research challenges and identify important trends in the future [10]. Fourteen areas are being researched by the SRA:

- Routing and addressing scalability and dynamics
- The manageability and diagnosability of resources (forwarding, processing, and storage) and data/traffic
- Security, privacy, trust, and accountability
- Availability, ubiquity, and simplicity
- Adaptability and evolvability to heterogeneous environments, content, context/situation, and application needs: The semantic Web and seamless localization
- Operating systems, applications and host mobility/nomadicity: Cloud OSs, embedded OSs
- Sustainability of energy
- Conflicting interests and dissimilar utilities: Stakeholder positioning
- Searchability/localization, selection, composition, and adaption: Search engines
- Beyond digital communication: 3D communication, Behavior communication
- Internet by and for people: Contribution to revolutionary societal challenges to ensure social, economical, legal and cultural viability
- Internet of content and knowledge: Virtual environment
- Internet of things: Intelligence/smart etc.
- Internet of services: Open-service platform

The Ministry of Internal Affairs and Communication (MIC) in Japan has supported a committee to discuss R&D issues with the future network. The New Generation

Network (NGNW) was defined in the committee as a network beyond both the Internet and NGNs. The National Institute of Information and Technology (NiCT) has initiated the AKARI project [11], which is a core research group that is studying the NGNW architecture and protocols. Three important requirements for AKARI are:

- How to cope with complexity (versatile appliances and heterogeneous networking).
- How to attain low-energy consumption to prevent further global warming.
- How to achieve a compromise between openness and transparency vs. high levels of security.

GENI has focused on studying and developing network technologies. Apart from GENI, AKARI has researched not only network technologies but also low-energy-consumption technologies, and security technologies. In X-ETPs FI, as detailed in its vision document, the future Internet is articulated around the Internet by and for people, the Internet of content, the Internet of services and the Internet of things.

As can be seen from the research areas or requirements of these three projects, they have focused on developing technologies and applications and have expressed less interest on how to shift existing networks to the future networks.

5. CONCEPTS OF TGNs

TGNs have to have not only functions and performance to solve three serious problems discussed in Section2 but also charm for network and service providers, and they have to be deployed as quickly as possible for the sustainability of mankind. Therefore, TGNs have to satisfy five key issues:

(1) Business continuity;
- Network and service providers can continuously make profits.
- Low operating and management costs.
- Backward-compatibility with existing services and applications working on existing networks.
- Communication market grows continuously.

(2) Solving problems of existing networks;
- Depletion of IPv4 addressed on Internet.
- Rapid increase in amount of malware.

(3) Energy saving to decease global warming;
- Reduction in electricity consumed by communication devices, servers, and terminals.

(4) Scalability and flexibility;
- Decrease digital divide by location.
- Reduce costs effectively by deploying TGNs from metropolitan to rural areas.
- Flexible for a progress of technologies and user demands.

(5) New functions or service to create new markets.

In addition to these five issues, some ideas or schemes are needed to boost shifting from existing networks to TGNs.

We proposed the open heterogeneous mobile network (OHMN) [12], which will help to open up the mobile communication market. Because the OHMN business model is divided to the terminal layer, the network layer, the connection service layer, the platform layer, the contents and applications layer and the supervising layer, business players within a same layer will become competitive harder and business players will actively collaborate with business players of other layer; and the OHMN business model will make it easier for newcomers to start new business in the market. The OHMN will be expected to generate a positive spiral of activity in the mobile market and continuously enhance the development of this market. Hence, the OHMN architecture is one of strong candidates for architecture of TGNs.

When the TGN-architecture will be based on the OHMN architecture model, TGNs will consist of terminals, access networks, transport networks, connection service systems, platform servers, contents and application servers and third party organization servers. Here, components of TGNs are called TGN-x such as TGN-terminals. Research results of R&D initiatives for the future network introduced in Section 4 will be applied to TGNs to realize above five necessities.

Usually, all their components need to be standardized. However, I propose that the performance criteria of each component, and interfaces and protocols between different components will be standardized, and architectures or protocols of each component will not be standardized. For example, we may just determine the levels of energy consumption, security, and cost performance, and the interface (includes protocol) between components. This scheme should allow manufacturers to freely design network components and achieve cost effectiveness.

6. SCHEME FOR SHIFTING TO TGNs

This section discusses a scheme that can boost the shift from existing networks to TGNs.

6.1. Motivation for TGNs

As I described in Section 3, it is necessary for stakeholders in TGNs to be strongly motivated to shift smoothly from existing networks to TGNs. I recommend the OHMN business layer model [12], which is horizontally divided and not vertically integrated, to enable TGNs to continuously expand. There are seven stakeholders in the OHMN business layer model:

- End users
- Access-network providers
- Transport-network providers
- Connection-service providers
- Content and application providers

- Charging providers
- Supervisors

I believe that all stakeholders feel some degree of motivation as to whether they will satisfy the demands of TGNs or not.

- End users:
  They feel slightly motivated to conserve energy and provide security against malware. Whether end users will positively shift to TGNs or not depends on the tradeoff between decreased electricity bills and the cost of security software. Perhaps more positive motivation is required to enable a smooth shift.
- Access-network and transport-network providers:
  The first demand discussed in Section 5 is crucial to both of them. If IPv4 addresses or telephone numbers will be depleted within the next few years and it will be surely impossible to solve such problems technically, they will positively shift their networks to TGNs the same as they did when shifting to 3G-mobile networks. They will feel strongly motivated, if they can benefit from a thousand-fold reduction in electricity consumption, which is the goal of Green Touch™.
- Connection-service, content-and-application, and charging providers:
  They will feel strongly motivated to save energy the same as access-network and transport-network providers.
- Supervisors:
  They will also feel motivated to conserve energy and also gain a good impression of security against malware.

A dramatic reduction in electricity usage strongly motivates stakeholders. Unfortunately, this means that the shift to TGNs will not eventuate, if such dramatic energy-saving technologies cannot be attained. Also, there is little motivation for solving the digital divide by location. I believe that a shifting scheme is needed, where stakeholders feel motivated by mid-scale technologies to reduce electricity consumption.

6.2. Cap and Trade Scheme

Economic incentives become strong motivators, the same as those that produce dramatic reductions in electricity consumption. I propose the cap and trade scheme that has been applied to reduce the amount of CO_2 [15, 16]. There are two schemes for adopting TGNs, i.e., cap and trade itself and modifications to this scheme. The first one is where the maximum emissions allowed by each provider (not users) transferred from maximum electricity consumption are set as a cap and trading on the market is the same as that in other categories, such as manufacturing by factories. Because existing cap and trade schemes are focused on decreasing CO_2 emissions, it is difficult for them to boost the deployment of TGNs. The second scheme involves modifying existing schemes and establishing an independent market. The latter one must be modified to boost the deployment of TGNs, especially in developing countries. I call this modified one the TGN-Cap & Trade scheme and propose the following:

The cap: A cap is used to set a minimum rate of introduction that is a ratio between TGN units (or traffic) and total units (or traffic) and it penalizes network providers who are less than their allotment.

- The minimum rate of introduction must be equal for each provider.
- **TGN units deployed in developing countries are counted N times**. This becomes the motivation for deploying TGNs in developing countries. Magnified coefficient N will be decided at a standardizing meeting of ITU-T.
- TGN Introduction Rate (TIR):

$$TIR = \frac{TA + N \cdot TB}{TE + TA + TB}$$

TA: The number of TGN units deployed in developed countries

TB: The number of TGN units deployed in developing countries

TE: The number of existing units

N: Magnified coefficient

- Each year, the cap is ratcheted up on a gradual and predictable schedule.
- Providers can plan well in advance to be allowed more and more permits.
- Providers who introduce TGNs that exceed the cap can sell their extra portions.

The trade: Some network providers will find it easy to introduce TGNs to match their number of permits; others may find it more difficult. Trading lets network providers buy and sell their extra portions.

- Trading lets providers introduce more cost-effective shifts to TGNs.
- **Users who use green terminals can sell quantities of traffic that have passed through TGNs.**
- Trading gives stakeholders incentives to invest in deployment and energy-saving technologies.

An image of TGN-Cap & Trade scheme is shown in Fig.4.

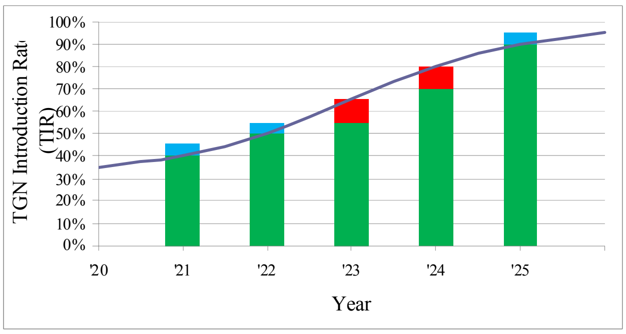

Fig. 4 An image of TGN-Cap & Trade scheme

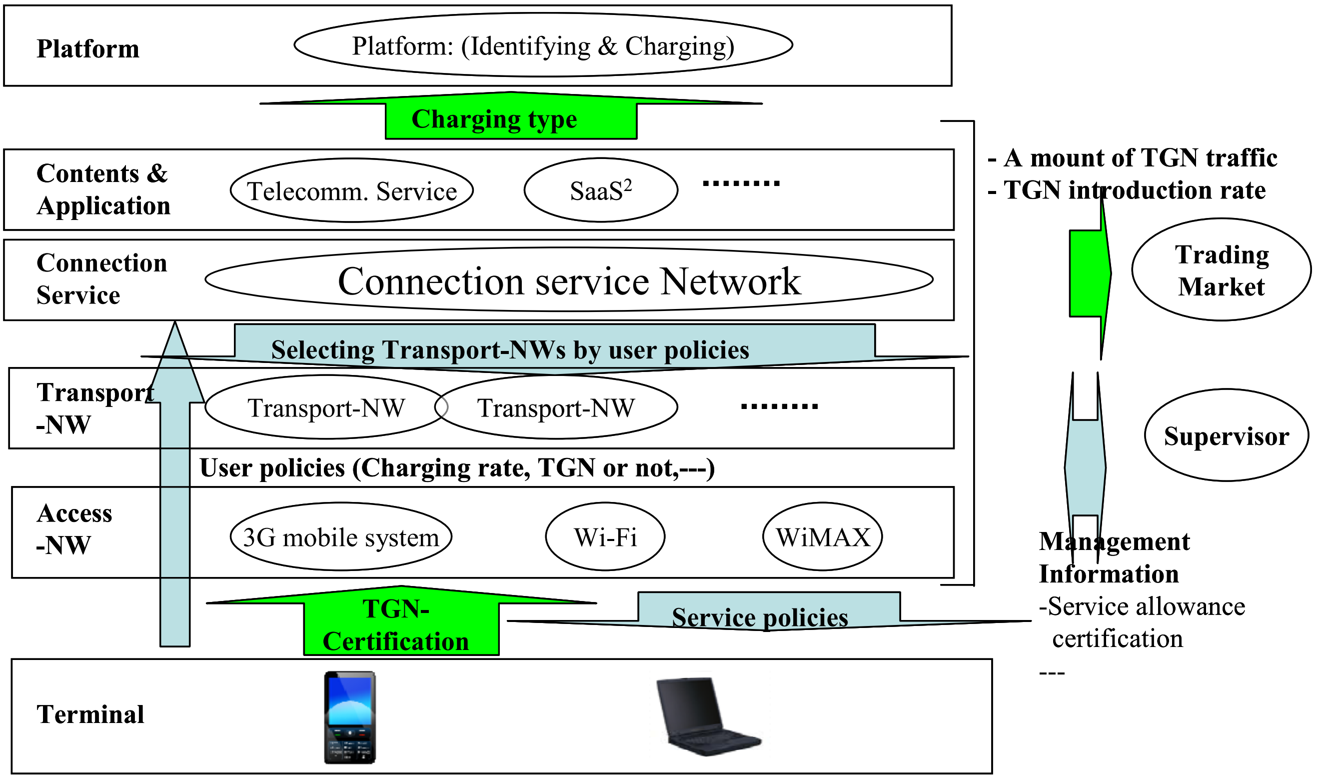

Fig.5 TGN business layer model

7. TGN BUSINESS LAYER MODEL

Because a trading market will be added to TGNs as described in Section 6, the OHMN business layer model will be modified, as shown in Fig. 5. As shown in Section 2, most of malware comes from comprised computers. I propose introduction of the service allowance certification to removing comprised computers. The major additional jobs by each layer of the modified OHMN will be:

Terminal layer:

- Terminal manufacturers take the **TGN certification** for their new terminals and install it in their terminals. All terminals hand such certification information to communication application in a terminal, when they are installed.
- Communication applications add certification information to all packets.
- Users select an access network based on service policies sent from each access network.
- Users send their policies for selecting transport networks.

Access-network layer:

- An access network exchanges the TGN certification information used for establishing a virtual transport network with a terminal. One item of information for establishing a virtual transport network is "selecting TGN".
- An access network exchanges such establishment information with a connection-service system.

[2] SaaS: Software as a service

Transport-network layer:

- A transport network exchanges TGN certification with a connection-service network.

Connection-service layer:

- A connection-service network determines and establish a user path between an end access network and another end one based on information that is exchanged with access and transport networks.
- A connection service system send charging information to a charging system on the platform layer instead of the transport network.
- A connection-service system sends the TGN trading system information that is needed for the modified cap and trade scheme.

Platform layer:

- Charging is done the same way as in the original OHMN.

Content and application layer: The jobs in this layer changes depending on security technologies and their operation. Three main examples are given below;

- The content or application provider obtains the allowance certification from the supervisor.
- When they take the allowance certification, they present it on their content or applications.
- Content or applications servers exchange TGN certification with an application on a terminal and that will be presented to users.

Supervision:

- A supervisor investigates the rate at which TGNs have been introduced by each network provider every year.

- A supervisor investigates whether content and application providers keep rules or not. When they keep rules, a supervisor gives them the allowance certification. This procedure has to be done every year. A supervisor also gathers research information from users.
- A supervisor will present the latest investigation results to users.

Trading Market:

- Network providers will sell and buy extra portions of the cap.
- Users will sell their quantities of green traffic.

8. CONCLUSION

I proposed a definition of the future networks that are expected to solve problems such as global warming, lack of security and trust on the Internet, and the digital divide by location for the sustainability of mankind as TGNs.

I clarified the requirements for TGNs and proposed a modified cap and trade scheme to boost the shift from existing networks to TGNs. Providers can sell extra portions of the rate at which they introduce TGNs and buy fewer portions in this scheme. TGN units deployed in developing countries are only counted as a few times. Users can also sell the quantities of traffic that pass through their TGNs. As a result of this scheme, major stakeholders in TGNs will gain economic incentives and replace smoothly from existing networks to TGNs.

The details on the cap and trade scheme modified to TGNs have to be better considered as to when it can be introduced in practice as I just presented its basic structure here.

REFERENCES

[1] Trend Micro, "Trend Micro 2008 Annual Threat Roundup and 2009 Forecast"; http://us.trendmicro.com/imperia/md/content/us/pdf/threats/securitylibrary/trend_micro_2009_annual_threat_roundup.pdf

[2] Japan Ministry of Economy, Trade and Industry, Green IT Initiative in Japan, 2009; http://www.meti.go.jp/english/policy/**GreenIT**InitiativeInJapan.pdf

[3] ITU, "Global ICT developments"; http://www.itu.int/ITU-D/ict/statistics/ict/index.html

[4] Internet Protocol, Version 6 (IPv6) Specification (RFC 2460); http://tools.ietf.org/html/rfc2460

[5] M. Leber, "Global IPv6 Deployment Progress Report"; http://bgp.he.net/ipv6-progress-report.cgi

[6] Lorenzo Colitti, S. H. Gunderson, and E. Kline, Tiziana Refice, "Evaluating IPv6 Adoption in the Internet", PAM 2010, Zurich, 2010.

[7] Telecommunications Carriers Association; http://www.tca.or.jp/database/index.html

[8] NSF NeTS FIND Initiative; http://www.nets-find.net/

[9] GENI: Opening up new classes of experiments in global networking; http://www.geni.net

[10] European Future Internet Portal; http://www.future-internet.eu/

[11] T. Aoyama, "A New Generation Network: Beyond the Internet and NGN", IEEE Communication Magazine, Vo. 47, No. 5, pp. 82–87, May 2009.

[12] Y. Murata, et al., "Architecture and Business Model of Open Heterogeneous Mobile Network", Proceeding of the 1st ITU-T Kaleidoscope Academic Conference, pp.143–150, 2008.

[13] Green Touch™; http://www.greentouch.org/

[14] Wapedia Wiki, "List of mobile network operators" : http://wapedia.mobi/en/List_of_mobile_network_operators#By_region

[15] Center for American Progress, Cap and Trade 101; http://www.americanprogress.org/issues/2008/01/capandtrade101.html

[16] EDF, Global Warming, "How Cap and Trade Works"; http://www.edf.org/article.cfm?contentID=9112

INNOVATIVE AD-HOC WIRELESS SENSOR NETWORKS TO SIGNIFICANTLY REDUCE LEAKAGES IN UNDERGROUND WATER INFRASTRUCTURES

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ABSTRACT

This paper presents an ICT solution to overcome the problem of water dispersion in water distribution networks. Leakage prevention and breaks identification in water distribution networks are fundamental for an adequate use of natural resources. Nowadays, all over the world, water wasting along the distribution path reaches untenable percentages (up to 80 % in some regions). Since the pipes are buried within the terrain, typically only relevant breaks are considered for restorations: excavations are very expensive and consequently the costs to identify the position of the leakage or just the position of the pipe itself are too high. To address this problem, and simplify the leakage identification process, the authors have designed a wireless network system making use of mobile wireless sensors able to detect breaks and reveal unknown tracks and monitor the pressure spectrum of the fluid flowing in the pipe. The sensors transmit the acquired data from the terrain to the surface by use of a wireless connection. On the surface ground there are stations that receive the signal, process it, and communicate with a central unit where necessary intelligent signal processing techniques are used to detect leakage sources. Compared to other leakage detection solutions already available in the market (such as: Ground penetrating radar (GPR), pure acoustic techniques and tracer gases), the proposed technique appears very efficient and much more inexpensive.

Keywords— Wireless sensor networks, Ad-hoc networks, RFIDs, Green technology, radio-acoustic sensors

1. INTRODUCTION

Breakdowns and damages in fluid distribution systems represent a problem of growing importance, due to their fundamental role as a primary good that water and gas symbolize all over the world [1]. This problem is emphasized by the progressive decrease of water resources not only in Equatorial Countries, but also in Western ones. All over the world, water distribution networks are typically old and suffer from leakages and dispersions. . On the other hand, gas infrastructures are less affected by damages and falls, but the cost and the strategic value of the resource makes relevant the effect of even small, rare or distributed damages.

The restoration of damaged pipelines, especially when pipes are deployed under the ground surface, requires high complexity, first of all because the exact paths are generally unknown, and secondly because, even when the path is known, it is difficult to identify the exact location of the damage along the conduit. Therefore, despite these breaks represent for companies that manage fluid transportation infrastructures, and consequently for the social community, a huge dispersion of primary resources, the renovations are complex and require long times to reach an acceptable solution. From the point of view of the costs, the first factor is represented by the technological process needed to identify the leakage. Hence, it is strongly related to time drawbacks. As an example, the detection of failures using advanced technologies may cost 3200 USD per kilometer when it is easy to identify the break, up to 65000 USD per kilometer if the damage occurs in a complex metropolitan environment. In general, expenses increase with the complexity of the urban scenario, because of indirect costs generated primarily by the excavation process. Especially when works concern relatively huge areas, there can be an unbearable effect on traffic, public services and business activities. All these costs can be lowered by refining the identification techniques with more accurate approaches.

Several monitoring techniques are available in the literature [2], [3], [4], [5]. Among all these, tracer gases [6] and ground penetrating radars (GPR) [7], [8] appear to be very promising since they do not require any direct connection between the pipe and the outside, but they are not able to identify small leakages or to survey the pipe before strong damages occur. Gas tracing makes use of special gas mixture, a mix of hydrogen (5%) and nitrogen (95%), used to identify the conduit leakages within the ground. The gas is inserted in the pipeline and subsequently is investigated from the exterior using special instruments able to detect the concentration of gas in the environment. The system requires service interruption (tracer gases must replace the fluid normally conducted in the pipe) and is very expensive, due to the high cost of the gas itself, together with the gas sensors on the ground surface. GPR, on the other hand, may

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allow an easy estimation of unknown tube paths, but cannot provide a comprehensive monitoring of small pipe damages.

An automatic system able to measure electromagnetic parameters of oil field pipes automatically has been developed [9], so that the correct interpretation of steel pipe defects can be provided. The fault of steel products can be detected based on the eddy current technique, and an automatic measuring technique is used to correlate results. This system is applicable only to steel pipes, since it is based on the measurement of steel electromagnetic properties.

Accurate techniques make use of acoustic sensors [10], able to detect the acoustic noise typically produced by the presence of water losses and generated by the pressure gradient between the inner side and the outer side of the pipeline. The noise can be monitored on the pipe, in the ground or within the pipe itself. I.e., it is possible to apply a correlation technique to two measured acoustic/vibration signals on the pipe, on either side of a leak [11]. The technique that makes use of geophones [12] requires an operator with high professional background and good expertise, in order to identify the acoustic noise produced by losses in the framework of an external acoustic background. For this reason, the technique is critical in urban environments with high background noise. Furthermore, it becomes even more critical in case of large losses and fluids with low hydraulic pressure. An alternative acoustic solution is based on a time-domain technique that makes an analysis of the noise propagation delay. For this purpose, two different sensors are applied on the pipe surface in separate positions. Synchronizing the sensors and calculating the time taken by the noise to reach the two probes, it is possible to identify the position of the damage. Depending on the mechanical properties of the material used to construct the tube, the technique works on distances ranging from 50 to 200 meters.

More recently, it has been possible to deploy microphones in the pipe, processing the data on-site, without the need of a further elaboration. The main drawback, in this case, is represented by the need to keep the flow of the sensor under control, to reconstruct its position and make correlations between sensor position and the monitored acoustic spectrum [13].

Apart from the last one, all the known acoustic techniques normally require a direct wired connection with the sensor on or inside the tube. Therefore they are inappropriate to investigate pipes networks over long distances. Furthermore, they do not provide useful solutions for the detection of underground paths. To overcome these drawbacks, our group has recently proposed a system able to detect breaks and reveal unknown tracks by monitoring the acoustic spectrum of the noise produced by the fluid flowing in the pipe [14]. It transmits the detected information on a wireless channel, hence it does not require a physical connection to the surface, it gives an accurate detection of the leakage location; it allows an easy and repeatable identification of the track.

2. AN APPROACH BASED ON WIRELESS SENSOR NETWORKS

To simplify the construction and management of the system, the architecture is based on the use of underground mobile wireless sensor networks [14]. As shown in Fig.1, the suggested network is made up of ground stations, collocated in fixed or even movable locations, in proximity of known pipe crossing positions. The ground stations are equipped with directive antennas, pointed towards the terrain and communicating with mobile sensors that flow through the pipe network, transported by the liquid. These sensors represent the core of the monitoring system. They are made up of a hydrophone as a sensing unit, and a radiofrequency or microwave radio as a transmitting unit. The wireless component is able to connect to the ground stations, even if it is collocated within the liquid, inside a pipe interred in the terrain. Since the sensor is transported by the liquid under normal working conditions, the proposed solution preserves water provisioning, without the need to take the liquid out of the conduit. The flow of the sensor is controlled by means of hydraulic tricks and kept constant along the path. In this way it is possible to make real time and continuous pressure acquisitions without the use of wires or cables. Furthermore, when the sensor intercepts a ground station, its position is identified and the acquired spectra are correlated to leakage positions.

The proposed network scheme allows detecting breaks by monitoring the spectrum of the fluid pressure, and revealing unknown paths, by tracking the sensor movements. The detected information is transmitted through wireless channels; hence, a physical connection to the surface is not required. An accurate detection of the leakage position is provided. An easy and repeatable identification of the track is possible.

3. SENSOR CONFIGURATION

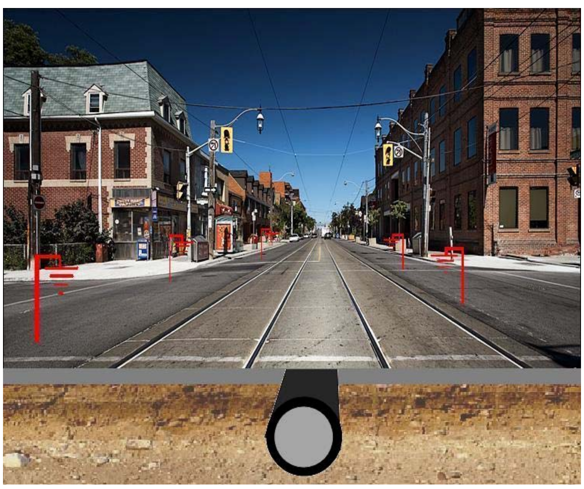

Fig. 1. Mobile Wireless Sensor Network applied to water pipes survey. The ground stations are collocated in fixed or movable positions, where the pipe crosses the normal direction to the ground (typically, where manholes are located).

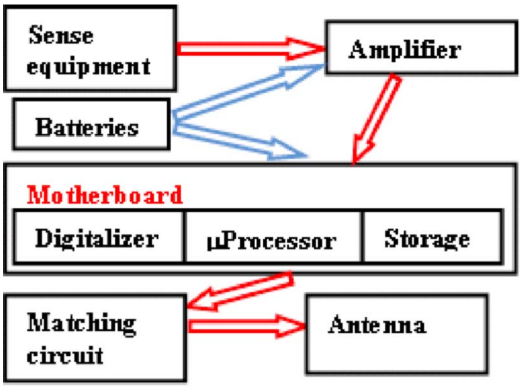

Fig. 2. Block-diagram of the mobile wireless sensor unit.

The sensor must make a measurement of liquid pressure in time domain, which is further processed in the frequency domain, to recognize pipe damages or fluid leakages. Hence, the general scheme of the wireless sensor unit is shown in Fig.2: the pressure is measured by means of a standard hydrophone, interfaced with a digital acoustic card. The card is controlled by a Single Board Computer (SBC), where the detected information is processed electronically, digitalized, stored on a flash memory and transmitted to the surface. Power supply is obtained from standard rechargeable batteries.

4. SENSOR REALIZATION

Several prototypes were designed, working at different frequency ranges, from 100 MHz to 2.4 GHz. Among all, three have been designed, at 180 MHz, 433 MHz and 700 MHz.

As a sensor, different acoustic transducers able to work within water (hydrophones) were considered. Finally, a miniaturized hydrophone with high sensitivity, -198 dB re 1V/µPa, was selected. Figure 3 shows the hydrophone, which does not need a power supply in order to generate acceptable signals based on the acoustic noise. Preliminary experiments showed that the output signal amplitude ranges between 10 mV and 100 mV based on the level of the generated noise.

The microprocessor board hosts and manages the whole mobile system. The main managing tasks are the control of the interface with the hydrophone and the sampling operation, the conversion of the acquired data from analogue to digital, the storing, pre-processing and routing of the data, the control of the RF board through the serial peripheral interface (SPI) interface, and the power control of RF board through a specific and ad-hoc protocol.

Fig. 3. The hydrophone implemented in the wireless mobile unit

Fig. 4. The microcontroller board

Among the numerous of-the-shelf controllers capable to perform most of these tasks, the PIC32 from Microchip Technology Inc® was chosen (see Fig.4 for reference). The main features of the board are: 32-bit 80 MHz core microprocessor, 16 channels 10-bit analog to digital converter (ADC), 16-bit timer, several types of interfacing protocols, 512 KB flash memory, 128 KB SRAM, etc.

The wireless transmission is realized by means of an RF board operating on the industrial, scientific, medical (ISM) band. The operating frequency was chosen, based on previous results [6]. The highest transmission power is 27 dBm. The module has input sensitivity level of -117 dBm with high data rate up to 115.2 kbps. Also, it exhibits analog and digital received signal strength indicator (RSSI). The wireless protocol used between the transmitter, the detection module, and the receiver, the gateway, is being developed in such a way to reduce the overhead exists in other wireless chipsets available in the market. This is required to minimize the time needed to send and receive data. Consumed energy is a critical factor in the proposed setup and reducing transmission time significantly improves the system performance.

As far as the design activity concerns the realization of the antenna, one must take into account that, in order to favor the flow of the sensor within the pipelines, the sensor must be as small as possible. Hence, the antenna must be miniaturized, even if the selected frequency ranges would require antennas with relevant dimensions (especially at 180 Mhz and 433 MHz) to optimize the transmission to the surface. As a matter of fact, the antennas are much smaller than the wavelength.

Being small, electrical antennas are not suitable for the desired application, as they are inserted in conductive media and consequently completely mismatched from the surrounding environment. Hence, the use of magnetic antennas (dipoles) is mandatory. Electrically small magnetic dipoles are characterized by very short dimensions: their circumference C has to be less than one

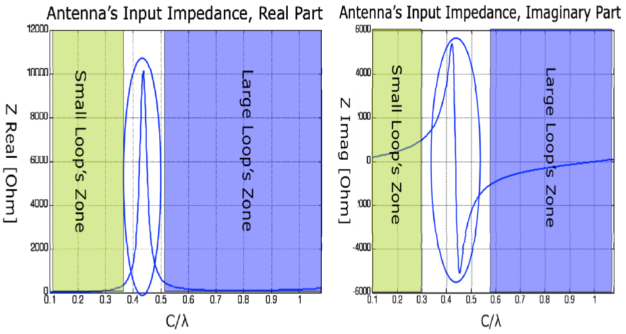

Fig. 5. . *Real and Imaginary Parts of the input loop's impedance versus its circumference normalized to the wavelenght (C/λ)*

fifth of the wavelength and then the input impedance tends to be a short circuit. For this reason they are completely mismatched from the transmitter because of a very high reactive part and a very low (almost zero) real part of the input impedance.

Starting from results shown in Fig.5, the design of the antenna is carried out by fixing the real part of the antenna impedance at 50 Ohm, and consequently manipulating the imaginary part by inserting in series to stubs with input reactance equal to half the reactance of the antenna. The geometrical result is shown in Fig.6 The insertion of the stubs slightly modifies the impedance of the antenna, and consequently their length must be re-adapted, following an iterative algorithm.

6. CONCLUSIONS

The paper presents an innovative concept that uses mobile wireless network technology to monitor pipes for water

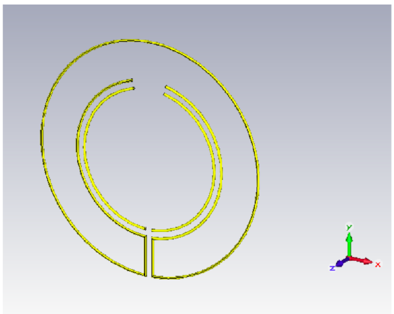

Fig. 6. A magnetic antenna with the matching circuit, specifically designed for the described application, at 433 MHz

provisioning. The numerical electromagnetic predictions, as well as the experimental data, validate the proposed approach and demonstrate its applicability on larger scales. At the moment three different solutions have been implemented. The one working at 750 MHz has been already measured in a real scenario [14], it works well and allows the use of more directive antenna over the ground surface, minimizing the generation of noise. The second one works at 433 MHz, and it has been presented here. The third one works at 180 MHz, it is under construction now and will be presented in future publications.

REFERENCES

[1] Substantive Issues Arising in the Implementation of the International Covenant on Economic, Social and Cultural Rights, November 2002. General Comment No. 15, www.ohchr.org.

[2] H.V. Fuchs and R. Riehle. Ten years of experience with leak detection by acoustic signal analysis. Applied Acoustics, 3:1-19, 1991.

[3] D.A. Liston and J.D. Liston. Leak Detection Techniques. Journal of the New England Water Works Association, 1992

[4] O. Hunaidi, W.T. Chu, A. Wang, and W. Guan. Leak Detection Methods for Plastic Water Distribution Pipes. In Seminars on Water & Sewer Infrastructure Systems: Challenges and Solutions, pages 249-270, Ottawa, Canada, 2000.

[5] X.J. Wang, F.M. Lambert, R.A. Simpson, and J.P. Vitkovsky. Leak Detection in Pipelines and Pipe Networks: A Review. 6th Conference on Hydraulics in Civil Engineering: The State of Hydraulics,pages: 391-400. Barton, A.C.T.: Institution of Engineers, Australia, 2001.

[6] Sensistor AB. The H2 Method for Locating Leaks in Buried Water Pipes. Application Note, Sensistor AB, 1997.

[7] O. Hunaidi. Ground Penetrating Radar for Detection of Leaks in Buried Plastic Water Distribution Pipes. International Conference on Ground Penetrating Radar, 1998.

[8] B.J. Allred and N.R. Fausey. G.P.R. Detection of Drainage Pipes in Farmlands. In International Conference on Ground Penetrating Radar, Delft, Netherlands, 2004.

[9] J. Yin, J. Pineda de Gyves, M. Lu. An Automatic System Measuring Electromagnetic Parameters for Oil Field Pipes. IEEE International Conference on Industrial Technology, 1994.

[10] Hunaidi O., Wing C., Acoustical Characteristics of Leak Signals in Plastic Water Distribution Pipes, Journal of Applied Acoustic, 1998.

[11] Y. Gao, M.J. Brennan, P.F. Joseph, J.M. Muggleton, and O. Hunaidi. On the selection of acoustic/vibration sensors for leak detection in plastic water pipes. Journal of Sound and Vibration, Volume 283, Issues 3-5, pages 927-941, 2005.

[12] http://www.water-technology.net/contractors/pipe_clean/pure_tech/.

[13] M. Thompson, M. L. Harper, The body is the sensor, Insight - Non-Destructive Testing and Condition Monitoring, Volume: 50, Issue: 2, pages: 98-99, 2008.

[14] D. Trinchero, R. Stefanelli, "Microwave Architectures for Wireless Mobile Monitoring Networks Inside Water Distribution Conduits," Micr. Theory and Tech., IEEE Trans. on, vol.57, no.12, pp.3298-3306, Dec. 2009

POSTER SESSION

SHOWCASING INNOVATIONS FOR FUTURE NETWORKS AND SERVICES

BEYOND THE WIFI: INTRODUCING RFID SYSTEM USING IPV6

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ABSTRACT

RFID System suffers from limited address space and local mobility. Moreover, it is a monopoly business with few vendors, which are trying to dominate the market with proprietary standard of RFID reader. The proposed system will replace the expensive RFID reader with cheap Wireless Network Interface Card (WNIC). For the purpose, an innovative scheme for RFID tagging system is introduced which will be benefited by well-defined WiFi protocol. IPv6 (Internet Protocol version 6) address will be used as product identifier. This will provide a universal identity of the objects with seamless global mobility. The EPC (Electronic Product Code) will directly map to IPv6 address by using an auto configuration method. So that 64 bit EPC will take the place of the EUI-64 portion of IPv6 address. The proposed system suggests the mechanism of reducing significant cost, physical location detection and usage of global unique address, which will also be compatible with existing EPC addressing scheme.

Keywords— RFID Tag, EPC, IPv6, WiFi

1. INTRODUCTION

Future network and services for recognition scheme along with wireless technologies are building up some innovative ideas. These ideas possibly will improve environmental, institutional, industrial, healthcare and transport provision. At present, wired services manage to supply infrastructure whereas a wireless service creates available mobility, miniaturization, inexpensive electronic devices which will endow access points everywhere. Currently, wireless technology or wireless devices are an essential part of our everyday lives. In the near future, wireless network technology will be used to link consumers with businesses that cater to their needs.

An RFID is a recognition system, which remotely stores and recovers data from any objects. To store and recover the data, it uses a tag or transponder. It is an object applied to or attached into a product, animal, or even a person for identification and tracking. To track any object it uses an Electronic Product Code (EPC), which is a unique number attached inside the RFID tag. EPC can be 64 bits or 96 bits. The two key parts of a RFID system consists of two fundamental components – a transponder (i.e., the tag itself) and a transceiver (i.e., the reader). An RFID tag is an integrated circuit with an antenna and act as a storage medium. A reader reads tag data by using wireless communication. A RFID tag can be passive (draw power from the reader), active (battery powered) or semi- passive (require battery, but the tag lies inactive until a signal is received from the reader) [1].

Today RFID is used in various areas of public and private sectors. It is used in the hospital and pharmacy; a druggist can fill a prescription from a bottle labeled with a RFID chip to confirm the authenticity of its contents. In addition, some organization uses RFID to track mail and security, to manage hazardous materials, to collect toll, to make people aware about tsunami, bird-flu etc. Moreover RFID has been used in warehouse management, supply chain management, reverse logistics, shipment tracking and asset tracking etc. In addition, RFID has been used at home where pets are set in with chips so that lost animals could easily be identified and returned to their owners.

To identify any object or instance an identification code is required. Unique identification mechanism has the capability of secure and efficient communication. Due to the requirement of global unique address structure a new protocol known as IPv6(IP version 6) will be used for objects-to-objects communications [2]. EPC, EUI-64 (Extended Unique Identifier), MAC (Medium Access Control) addresses, URI/URL, etc are some examples of identification codes. To manage a large number of different identification codes requires a large address space, so IPv6 (128 bits) network infrastructure will become vital for future network. IPv6 will provide universal any-to-any connectivity using unique local address with enough address space. It will enable self-organization and service discovery by using auto-configuration. In addition, it will provide enhanced service capabilities using IPv6 header information. Moreover, it will facilitate multi-homing using IPv6 addressing.

A combination of IPv6 and the wireless systems like IEEE 802.11 protocol, perhaps better known as WiFi will reduce the size of the current problems of the dual scarcity in the IP address, quality of service and security from the IP side and lack of spectrum and bandwidth from the wireless side. By combining the two technologies, the best value possible for the wireless end-user will be ensured [3]. WiFi uses the unlicensed ISM (Industrial, Scientific, and Medical) band centered at 2.4 GHz. Data rates upwards of 11 Mbps will be achieved; however, it is more typical to see data flow at 5 Mbps or less [4]. The 2.4 GHz frequency band exists throughout the world for low cost short-range wireless communications. It has increased the popularity with the production of networking and battery-operated technologies based on WiFi. An increasing number of RFID systems are also designed for operation within the 2.4 GHz ISM frequency bands [5].

RFID systems use many different frequencies for different purpose, which requires different types of receivers. Based on the research survey by NECTEC in 2005, the delay of RFID deployment due to four factors: RFID cost, standard, technology suitability, and lack of knowledge [6]. In data transmission policy an RFID network is less secure and user privacy issue is another major problem [7]. The most important thing is that, though RFID is an emerging technology, it is lagging behind for some limitations such as vendor specific solution and expensive implementation cost. The most expensive part of using RFID is the reader. For example, reader of a typical EM or RF theft detection system cost approximately $1,500 to $7,500 [8]. Also by using the reader, we place a limitation on RFID tag implementation. These drawbacks can be easily overcome by using WiFi network and IPv6 address as tag ID [9].

In this paper, an innovative RFID tagging system is introduced using WiFi system. The tag will use IPv6 address as a unique identifier of the objects. Instead of using RFID reader, it will simply use wireless network card to receive information from the IPv6 transponder. The proposed system will eliminate the expensive vendor specific RFID reader and provide a global identification number to every object.

2. RELATED WORK

It has already been discussed that there are many kinds of identification codes. Using these identification codes, all the devices should be reachable to other devices. To manage a large number of different identification codes it is essential to have a large network infrastructure. Therefore, network infrastructure like IPv6 will become vital. By using IPv6, both location information of IPv6 address and uniqueness information of identification codes will be defined.

Accordingly, by combining IPv6 address with identification codes an end-to-end connectivity will be established. In particular, it is assumed that the EPC of RFID is the most popular applications among all the identification codes.

In this section, a study approach related to the method proposed in this paper is discussed. It consists of the tag in which EPC code and IPv6 is embedded. By using radio frequency identification the object in which the tag including the EPC-IPv6 code is remain, a WNIC can bring about the IP address of the object. In addition, the WNIC has the function of transmitting the IPv6 address. Because it is connected via Internet, the EPC code can be accessible using its code through the WNIC. Therefore, a bidirectional communication using EPC-IPv6 mapping mechanism can be possible between object and Internet.

In a mapped EPC-IPv6 code mapping mechanism, the EUI-64 portion of the network prefix will be replaced by the EPC (64 bit) code value [10].

Usually, an IPv6 address contains 128 bits. It has two parts, network prefix (64bits) and EUI-64 part or Interface ID. There is a simple algorithm of converting the MAC (Medium Access Control) address into a modified EUI-64: the global flag (7th bit) of the MAC address is inverted and the value "fffe" is inserted between the 3rd and 4th byte of the MAC address. This EUI 64 is used as interface ID of an IPv6 address. Figure 1 shows the steps of creating EUI-64 from a MAC [11].

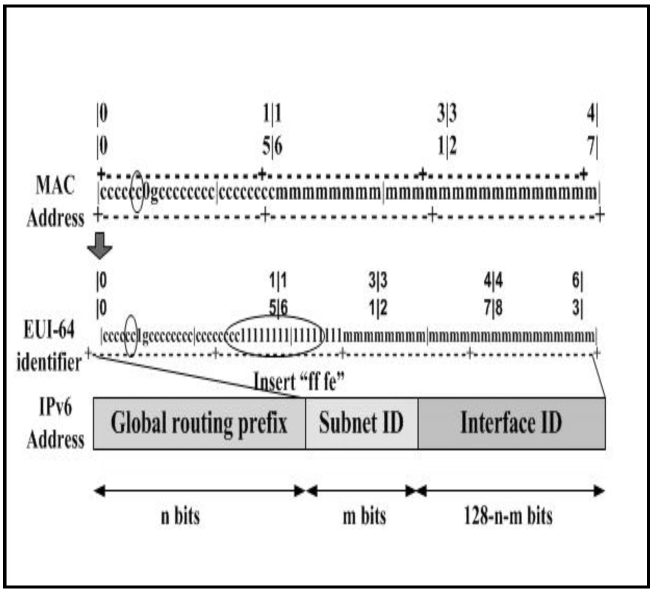

Figure 1. IPv6 address structure for mapping with EUI-64

Similarly, we can map total 64 bits of EPC into IPv6 address structure and used as IPv6 interface ID. Thus, the IPv6 address will contain the whole information of EPC and find out the way to reach any object to the destination. IPv6 address as shown in Figure 2 can be one of IPv6-based unified address structures. EPC information will directly map into the Interface ID portion of the IPv6 address using direct mapping.

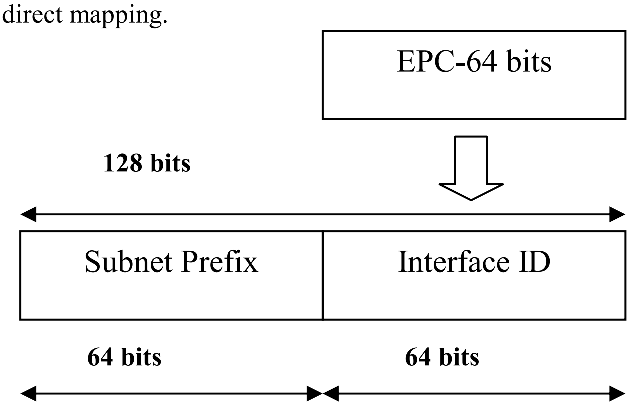

Figure 2. IPv6 Address format after mapping with EPC

IPv6-based unified address field includes several identification codes. Mapping EPC with IPv6 address structure will provide reach ability to physical location detection of any objects. Each identification code becomes addressable in the IPv6 network. The reach ability scope will be defined by IPv6 subnet prefix portion. Location computation software could directly communicate with tagged devices from anywhere within the IPv6 network.

3. PROPOSED MECHANISM

3.1. EPC-IPv6 Tag

In the proposed system, a RFID tag will consists of an antenna and an integrated circuit (IC). A memory block, rectifier, demodulator, modulator, low power digital logic and RF front end will make the complete tag IC. The antenna connected with RF Front End will control the system's data acquisition and communication. The tag contains an electronic microchip, as shown in Figure 3, which will be fabricated as a low power IC.

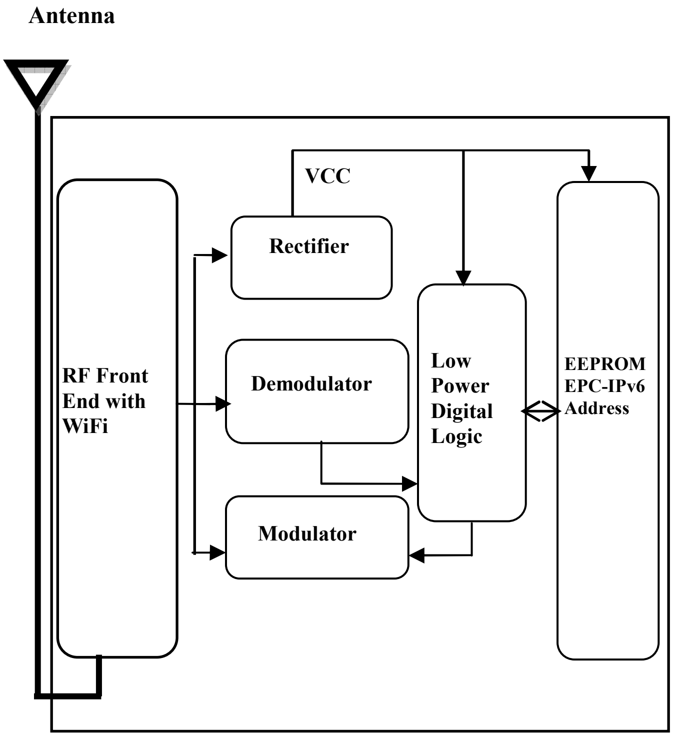

Figure 3. Block diagram of the proposed EPC-IPv6 Tag

The tag memory block can be Electrically Erasable Programmable Read Only Memory (EEPROM), Static Random Access Memory (SRAM) or Ferroelectric Random Access memory (FRAM). In the proposed system EEPROM will be used as a storage medium and data buffer of EPC mapped IPv6 address. It will be used in a wide range of applications due to its low manufacturing cost and high number of possible reprogramming cycle.

The modulator block will be used to power the modulation of the carrier signal received from the WNIC in order to send valid signal. The demodulation block will work as a decoder towards the tag.

The rectifier circuit will be used as there will be a limiter and voltage pump circuit for either limiting or increasing the dc power.

The RF front-end existing herein will be designed in a way that it will be intended to operate in the 2.4-GHz ISM band. It will be conceived to support high data rates. The RF front end will work as everything between the antenna and the digital base band system. It will work as a receiver. For a receiver, this area will include all the filters, Low Noise Amplifiers (LNAs), and down-conversion mixer(s). All these circuits will need to process the modulated signals received at the antenna into signals suitable for input into the base band analogue-to-digital converter (ADC). Therefore, it will work as an analogue-to digital or RF-to-base band portion of a receiver. In the proposed system, this RF Front end with the IEEE 802.11 protocol will use to communicate with the WNIC through the antenna.

3.2. EPC-IPv6 Tag Communication

Figure 4 shows the procedure of communication through IEEE 802.11 protocol between a RFID tag with EPC-IPv6 address structure and a computer, which uses a wireless network interface card.

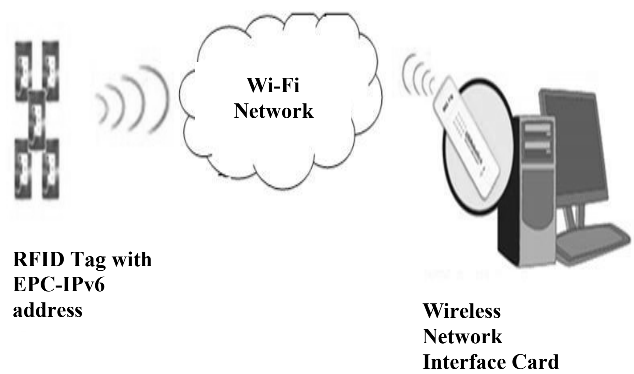

Figure 4. EPC-IPv6 Tag communicate with WNIC through WiFi

By using the mapping mechanism stated in the related work section of this paper, RFID tag will store the whole EPC-IPv6 address. The RFID tag can be consigned to a range of between 3 and 300 feet at 2.4 GHz ISM frequency band. When an IPv6 RFID tag enters into a WiFi network, it will be recognized by the system containing WNIC. At first, the WNIC will broadcast a message. All RFID tag in that WiFi network will receive it. After receiving this message, RFID tag will send the acknowledgement packet. From the acknowledgement packet the WNIC get the RFID tag IP address.

3.3. WiFi Receiver

In the proposed mechanism of this paper, the IP address could produce through the IP address generation process. The generated IP address will map with the RFID EPC and is stored in the RFID tag or transponder.

Figure 5 shows the procedure of extracting an EPC id from the mapped EPC-IPv6 address structure.

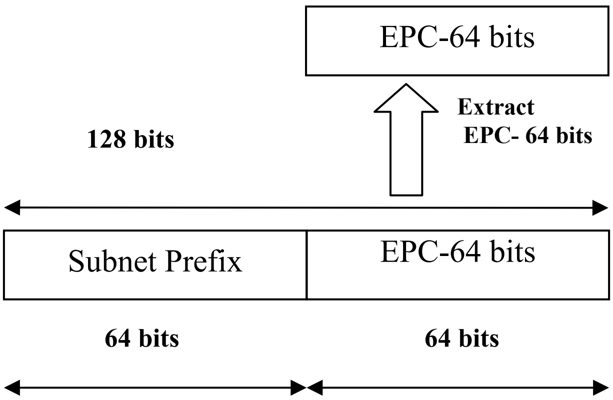

Figure 5. IPv6 address after EPC extraction

RFID tag data received by WNIC will process by an application to decrypt EPC id from IPv6 frame. However, EPC and IPv6 designed for use by very different applications. Thus, the partition identifiers in the two identification schemes have different interpretations. Due to their interpretations, the EPC namespace is effectively managed by the manufacturers of products and the IPv6 namespace is effectively managed by network administrators [12].

CONCLUSION

The RFID technology faces some problems with huge accomplishment expenditure and vendor precise solution. Also, tag ID is having the problem with physical location identification. In the proposed system, an RFID tagging scheme consists of EPC mapped IPv6 address. The RFID tag can communicate with WNIC in WiFi networks without using the expensive RFID tag reader. As the RFID tag contains IPv6 address, it will be accessed from any location. Additionally, the EPC will provide the general information about the objects, which is presently used. As a result, it is possible to find out the physical location of any objects. Thus, the proposed system will reduce the cost of RFID tagging system for future network of things.

REFERENCES

[1] N. C. K. Sreven Preradovic, and Isaac Balbin, "RFID Transponders," *IEEE, Microwave Magazine* vol. 9, pp. 90-103, 2008.

[2] J. K. C. Gyu Myoung Lee, and Taesoo Chung, "Address Structure for supporting Ubiquitous Networking using IPv6," in *ICACT*, 2008, pp. 1088 - 1090.

[3] T. T. Nielsen, "IPv6 for Future Wireless Networks," *Wireless Personal Communications,* vol. 17, pp. 237- 247, 2001. [4] B. Senese. (2003), "Implementing Wireless Communication in Hospital Environments with Bluetooth, 802.11b, and Other Technologies ". *Medical device And Diagnostic*.

[5] R. Bridgelall, "Bluetooth/802.11 Protocol Adaptation for RFID Tags," in *Tags, Proceedings of the 4th European Wireless Conference*, 2002.

[6] SRII, "Report on Development plan for RFID in Industry and Service (RFID)," NECTEC,2006.

[7] S. E. S. Stephen A. Weis , Ronald L. Rivest , and Daniel W. Engels, *Security and Privacy Aspects of Low-Cost Radio Frequency Identification System* , vol. 802/2004: SpringerLink, 2004.

[8] R. W. Boss. (2003), *Library Technology Reports on ALA [Online]*. Available: http://www.ala.org

[9] Dinesh Vadhia and R. Gupta. (2004), *homepage on World Internet Center. [Online]*. Available: http:// www.worldinternetcenter.com/pubs

[10] D. G. Yoon, Lee , D.H. , Seo , C.H. , and Choi , S.G., "RFID Networking Mechanism Using Address anagement Agent," in *4th International Conference n Networked Computing and Advanced Information anagement*, 2008, pp. 617 - 622.

[11] M. Dunmore, *An IPv6 Deployment Guide*: The 6net onsortium, 2005.

[12] E. D. W. (2002), *homepage on Auto-ID Center [Online].* Available: http://www.autoidlabs.org

COMPARATIVE ANALYSIS OF EXTENDED GEOGRAPHICAL WIRELESS NETWORKS BASED ON DIVERSITY TRANSMISSION SYSTEMS

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ABSTRACT

The paper analyses the performance of wireless networks working over unlicensed frequency ranges, making use of diversity transmission systems, as an effective means to increase outdoor coverage capabilities in rural scenarios. To this purpose, a network has been designed and realized, providing coverage to a wide rural area in the hills of Piedmont, a region located in North-Western Italy. Once constructed, the network performance has been monitored and characterized, in terms of coverage capabilities, signal quality, and noise immunity.

Keywords— Wireless networks, spatial diversity, polarization diversity, WLAN, Wi-Fi, WiMax

1. INTRODUCTION

The fight against the digital gap in rural regions is addressed as a strategic need in Developed [1] and Developing [2] Countries, as one of the most efficient means to increase living conditions, avoiding emigration, maintaining popular heritage and traditions. IP based intra-connectivity represents an efficient way to establish basic vital services like telemedicine and distance learning, together with the possibility to deliver low cost IP extranet services (VOIP, internet provisioning, access to centralized administrative databases).

Out of the city it is hard to favor the realization of IP communication infrastructures on the basis of commercial interests: not only in developing regions, but also in several remote areas located in developed countries. For this reason, it is fundamental to identify adequate inexpensive solutions to trigger the digital inclusion that can be followed by systems with increasing complexity and performance, as the connectivity demand increases. Compared to wired-based connections, wireless technology guarantees about cost reduction, since less infrastructures are required. Moreover, many wireless architecture can be implemented in the unlicensed frequency bandwidths, m inimizing the maintenance costs. Finally, several solutions are available off-the-shelf, in a wide range of prices and performance.

As a natural consequence, wireless networks have been largely used to overcome digital gaps all over the World [3]. Satellite connections, Third Generation (3G) mobile networks, (Wi-Max) licensed systems and Wi-Fi unlicensed networks lead to different implementations with the same aim: providing adequate connectivity in remote and isolated places.

Among all, satellites have been considered an effective solution to provide wideband connectivity. Satellite coverage reaches any remote place in the world and does not require intermediate repeaters (but the satellite itself). Unfortunately, bandwidth on satellite is still really expensive, latencies affect the quality of many real time services, and X-band propagation suffers environmental diseases and negative weather conditions. For all these reasons, satellites represent an excellent means to provide Internet connection to very isolated premises [4].

3G is regarded by many observers as a better solution to provide wideband connectivity to large rural areas, as mobiles are spread all over the World, with a surprisingly but realistic distribution not only in Developed, but also in Developing Countries. Unfortunately, 3G access is much more expensive than other forms of bandwidth provisioning; moreover, in rural regions cellular base stations (BSs) are spread over large areas, cells have huge dimensions and the quality of the connection suffers of luck of signals and interference [5].

Licensed Wireless Metropolitan Access Networks (WMANs) represent a better and more reliable way to provide wideband connectivity to rural areas, with lower costs for end-users and better insensitivity to the interference. Wi-Max is being implemented in Developed Countries, but it is still too expensive for applications id Developing areas [6].

Unlicensed WMANs derived by the Wireless Local Access Network (WLAN) standards, IEEE 802.11x, represent an adequate alternative for rural areas. Obviously, these systems suffer more than any other the interference that limits dramatically coverage capabilities and bandwidth provisioning [7].

An incoming standard is being considered now for the construction of Wireless Regional Access Networks (WRAN). Once approved, it will make use of licensed and unlicensed frequencies in the part of UHF bandwidth that is not going to be used after the switch-off from analogue to digital television broadcasting. The use of lower frequencies will allow the coverage of huge areas, but more

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than for WMANs, a strong insensitivity to interferences will be required to the system [8].

As a matter of fact, coverage opportunities that can be achieved with the listed systems (but the satellite one) are affected by interferences, especially when applied in the unlicensed spectrum. Hence, any Physical Layer implementation that allows a statistical improvement of the signal-to-noise ratio, is regarded positively.

2. NETWORKS BASED ON DIVERSITY TRANSMISSION TECHNIQUES

Last generation transmitting systems, including Multiple Input Multiple Output (MIMO) ones, implement the concept of diversity for the realization of the physical channel between the transmitter and the receiver, thanks to the use of more than one antenna at both sides of the link [9]. The last mobile and networking standards take into account the use of diversity, at least at one side the channel.

Spatial diversity is not favored by the necessity to separate antennas, since space is needed on the side of the client, and it is not easy to accomplish this requirement within client premise equipments (CPEs). On the contrary, polarization diversity can be implemented without increasing the space, since antennas with double polarizations can be mounted within the same radome, without increasing the overall dimensions of the system.

Time diversity may increase significantly the performance of the link, but it requires a huge bandwidth. For this reason, its application is limited to Wireless Personal Area Networks (WPAN), mainly for Ultrawideband connections [10].

A detailed analysis of the performance obtainable by the application of the MIMO concept as well as the spatial, polarization and time diversity can be found in the literature [11] and it is not reported here, as this is not the main subject of the presented analysis. The scope of the paper is to describe an experiment where the diversity concept is extensively applied over a pre-existing non-diversity network, in order to collect data about network stability, interference sensitivity, coverage improvements.

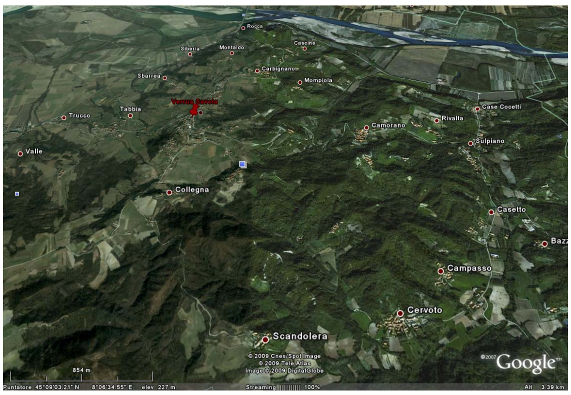

Fig. 1. Digital map of the Municipality of Verrua Savoia, near Torino, Italy. The center is located where the red cross is printed. Image taken from Google Earth. Digital maps origin to Spring 2010.

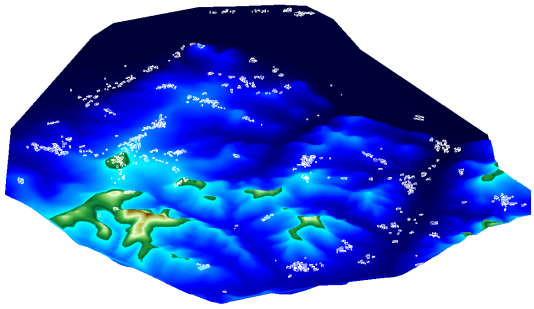

Fig. 2. Colormap orographic representation of the Municipality of Verrua Savoia, near Torino, Italy. The image has been obtained by use of the radiofrequency coverage software developed by our Labs. Geographical Database courtesy of Piedmont Region. The white boxes represent the houses.

3. THE EXPERIMENT

The experiment has been assembled in the Monferrato region, a huge hilly area in North-Western Italy, not far from the city of Torino. The area is mainly agricultural, with vineyards, apple and nut trees fields mixed to small forests. Only small villages are present, housing density is low, and houses are scattered almost uniformly over the territory. As a real case, the Municipality of Verrua Savoia has been selected.

Verrua Savoia is a Municipality that covers a territory of about 16 square kilometers, with approximately 1400 inhabitants, and half of the population aging more than sixty-five years old. The economical background is formed by small factories, few family-run shops, a couple of restaurants and bars. Most of the working people are commuters. For the rest, the economy is mainly agricultural, mainly devoted to family needs.

Topographically, the Municipality area is very well representing the whole Monferrato region. Fig.1 shows a three-dimensional digital map, illustrating the kind of environment and vegetation than can be found there, while Fig. 2 shows the orographic profile, together with the urbanization density and the almost uniform scattering of the houses. Fig. 3 and Fig. 4 reports a picture of the landscape, clearly showing the anarchical distribution of trees and vegetation.

As a matter of fact, providing ICT services on the territory of the Municipality is a losing affair. Hence, for long times, no wideband connectivity was available for the inhabitants.

Fig. 3. Picture showing the landscape in Verrua Savoia

Fig. 4. Picture showing the landscape in Verrua Savoia

4. A NON-DIVERSITY NETWORK WORKING IN THE UNLICESNED BANDWIDTH

In 2005 our Lab, in accordance with the Municipality administration, started the implementation of wireless networks to fill the digital gap of the village.

Apart from the social impact of the study, technically speaking, the territory represented a significant case of study, thanks to the variety of the landscape, the hilly nature of its orographic profile and the presence of larger, more inhabited and better connected neighboring villages. Verrua Savoia is located on the terminal front of the hills, facing the flat. As a consequence, transmitters located over the hills of Verrua Savoia are exposed to a significant number of radio-electric interference sources.

The first realization was carried out by means of Radiolan transmitters, working in the unlicensed 2.4 GHz WLAN bandwidth (IEEE 802.11g). Four BSs were realized, in strategic positions, providing coverage to approximately the 65% of the territory. Fig. 5 shows their positions. Either sector or omnidirectional radiators were used, depending on coverage requirements. After a short time, that solution evidenced the tremendous limits deriving from the implementation of an unlicensed system in the 2.4 Ghz bandwidth, with BSs and CPEs strongly limited in terms of effective isotropic radiated power (EIRP): 20 dBm.

As a matter of fact, in such a situation, the network performance is significantly affected by the presence of the

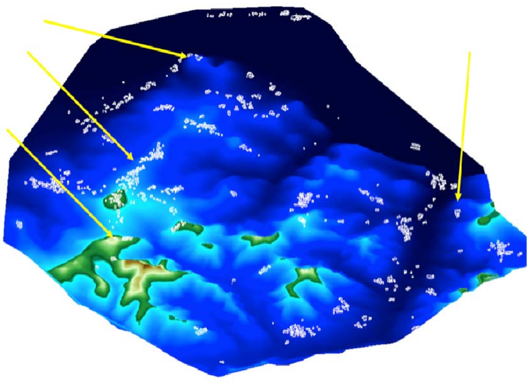

Fig. 5. Implementation of the 2005 network, in the Radiolan standard (unlicensed 2.4 GHz WLAN)

interference. For this reason, we decided to document the historical behavior of the network by registering the signal-to-noise ratio. In less then eight months, a constant decrement of the signal-to-noise ratio reduced significantly the coverage range (down to the 90% of the original value), thanks to an exponential increase of indoor and outdoor WLAN systems in the neighboring municipalities. Data have been measured by the transmitters themselves and registered hourly on board. Interferences do not affect symmetrically BSs and CPEs, since BS are deployed in strategic places being exposed to stronger interference. Moreover, the BS does not use directive antennas, resulting in a lower level of the received signal, if compared to the CPE.

After the first year, the network became unusable, mainly because of the interferences, and signal data were not collected anymore. It has been discontinued in September 2010.

In late 2005 the Radiolan network was overlapped by an unlicensed Hiperman network, implemented in the standard IEEE 802.11a + IEEE 802.11h. In terms of interferences, the allocated bandwidth is more robust than the one at 2.4 GHz, as the use of such a frequency is allowed only to wireless internet service providers (WISPs). Hence, the number of spurious transmissions was much lower, if compared to the 2.4 GHz case.

Fig. 6 shows the network working in the Hiperlan standard. The design was slightly modified, as BSs could be assembled in more strategic sites, with better line-of-sight coverage even if less protected radio-electrically.

Again, a decrement of the signal-to-noise ratio was measured in the four years for all the four BSs (especially the ones facing the flat), thanks to an exponential increase of the use of Hiperlan systems in the neighboring municipalities. Data have been measured by the transmitters themselves and registered hourly on board. Consequently, also for the Hiperlan case there has been constant decrement of the signal-to-noise ratio, even if less significant than the Radiolan case. The current state of the network shows that the coverage capabilities are 50% reduced, if compared to the ones measured in the first two years.

More detailed analyses showed the presence of lightly stronger signals originated far from the Municipality, not consistent with an adequate EIRP limitation. As a matter of fact, this represents a non-standard (illegal) situation. But still, we kept it into account, as a demonstration of the realness of the experiment

5. A POLARIZATION DIVERSITY NETWORK WORKING IN THE UNLICESNED BANDWIDTH

Observing the negative trend of the signal-to-noise ratio of the RadioLan and HiperMan experiments, in early 2010 our group decided to test the performance capabilities of diversity systems. For this reasons, a new network was designed and constructed in the HiperMan bandwidth (5.5

to 5.7 GHz), by implementation of BSs and CPEs working with double channels and dual linearly polarized antennas. The 2010 network is shown in Fig.7: as one can see, four BSs were deployed, making use of more strategic positions, even if potentially affected by stronger interferences. The use of diversity, in a scenario characterized by strong interferences, significantly favored the improvement of the average signal-to-noise ratio. On the other hand, when only the BS is using diversity, only slight improvements (if compared to the ones characterizing bidirectional diversity) are obtained.

The design of the networks involved the use of radio prediction tools, developed in our Labs. Also, designing the three different networks, involved the definition of complex antenna systems, by implementation of sector and/or directive antennas.

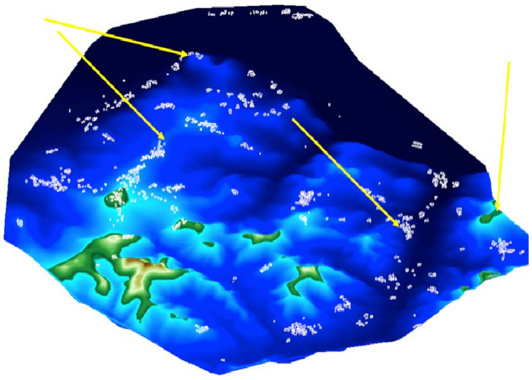

Fig. 6. Implementation of the 2006 network, in the Hiperlan standard (unlicensed 5.4 GHz WMAN)

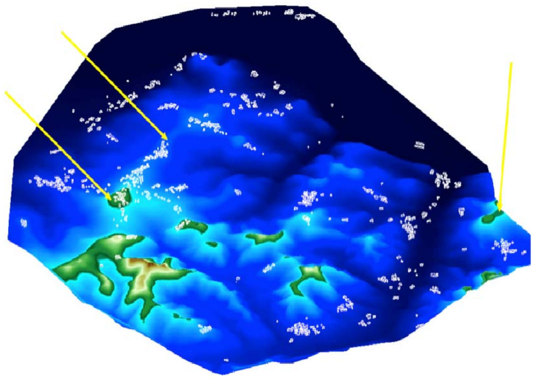

Fig. 7. Implementation of the 2010 network, in the Hiperlan standard (unlicensed 5.4 GHz WMAN) + double channel diversity (polarization diversity)

6. CONCLUSIONS

This paper presents the experimental construction of wireless networks making use (or not) of diversity transmission schemes. Experimental data show the significant increase of the performance provided by WMAN networks making use of bidirectional polarization diversity systems.

REFERENCES

[1] M. D. Chinn and R. W. Fairlie, “The determinants of the global digital divide: a cross-country analysis of computer and internet penetration”, *Oxf. Econ. Pap*. (2006)

[2] W. Chen B. Wellman, “The Global Digital Divide – Within And Between Countries”, *It&Society*, Volume 1, Issue 7, Spring/Summer 2004, Pp. 39-45

[3] S. M. Mishra, J. Hwang, D. Filippini, T. Du, R. Moazzami, and L. Subramanian, “Economic Analysis of Networking Technologies for Rural Developing Regions”, *1st Workshop on Internet and Network Economics*, Dec 2005.

[4] Spectar, J.M., “Bridging the Global Digital Divide: Frameworks for Access and the World Wireless Web”, 26 N.C.J. Int'l L. & Com. Reg. 57 (2000-2001)

[5] E. Pimenidis, A. B. Sideridis, E. Antonopoulou, “Mobile devices and services: bridging the digital divide in rural areas”, International Journal of Electronic Security and Digital Forensics, Volume 2, Number 4 / 2009, Pages: 424 - 434

[6] IEEE 802.16.X standards, 2004-2010, IEEE

[7] L. Subramanian, S. Surana, R. Patra, S. Nedevschi, M. Ho, E. Brewer, A. Sheth, “Rethinking Wireless in the Developing World”, *Hot Topics in Networks (HotNets-V)*, November 2006.

[8] C. Cordeiro, K. Challapali, D. Birru, Sai Shankar N, “IEEE 802.22: An Introduction to the First Wireless Standard based on Cognitive Radios”, JOURNAL OF COMMUNICATIONS, VOL. 1, NO. 1, APRIL 2006

[9] JH Winters, J Salz, RD Gitlin, “The capacity of wireless communication systems can be substantially increased by the use of antenna diversity”, Conf. Inform. Sci. Syst, 1992

[10] Standard ECMA-368, “High Rate Ultra Wideband PHY and MAC Standard”, 3rd edition (December 2008)

[11] J. Moon and Y. Kim. “Antenna Diversity Strengthens Wireless LANs.” Communication Systems Design, pages 15-22, Jan 2003

SIP TRUNKING THE ROUTE TO THE NEW VOIP SERVICES

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ABSTRACT

This work gives an overview of SIP-Trunking solution and explains the existing difficulties in implementing VoIP services, when this architecture is deployed in multivendor environment. The causes of these problems are explained with two existing approaches used by carrier to solve interoperability: they are Full Jacket SIP-Trunk and Customized SIP-Trunk. A solution to cover the lackings of SIP standards is the introduction of a SIP adaptation device called "Inter-Domain Adaptation Device" which will increase the potentiality of SIP-Trunking based solutions. It is also proposed a method for assessing the complexity of different approaches to SIP-Trunking applications.

Keywords — SIP-Trunking, T.38, MTP, SBC, PBX.

1. INTRODUCTION

There is an evident evolution of technology used by Telecom Providers tending to migrate voice services from traditional TDM networks to new VoIP networks, especially involving SIP-Trunking solutions. The number of this type of migrations is expected to grow dramatically at 106% Compound Annual Growth Rate (CAGR) in the next few years, to reach nearly 170000 in 2012 in West Europe [1]. The main driver of this migration is reduction in capital expenditures (CAPEX) and decrease of the operational expenditures (OPEX). But this migration determines also a slight quality of voice service reduction; depending on speech coder of VoIP the Mean Opinion Score (MOS) [2] may vary and usually is near to 4 [3] a bit less than the traditional TDM fixed networks but similar to mobile users experience.

The technology change determines also migration problems for already deployed services. If it's problematic to implement consolidated services like analogical fax and modem (see e.g. Sip-Forum that realizes a FoIP Task Group [4],[5]), it's also easy to understand how it's difficult to deploy new services like IP Automated Trading Desk or complex mandatory services like support for handicapped (bars brail, hearing, etc.).

The introduction of new VoIP solutions at Enterprise level is a great opportunity to redesign and standardize services. SIP-Trunking makes possible to implement new services (like Presence, videoconference, telepresence, virtual-fax) accessible as "*advanced communications as a service*" and that can change customer perception and improve productivity, collaboration and travels costs reduction. Service Providers (SPs) can obtain savings in terms of CAPEX and OPEX, but this new solution introduces interoperability problems between different technologies and vendors.

This situation is similar to the ISUP solution in the beginning of 90' [6]; but today there are a wide range of players that develop VoIP products for business market like IP-PBXs and Corporate Switching Nodes (CSN). There are big vendors (like Alcatel-Lucent, Ericsson,…) as well as small software houses that develops applications [7] on commercial hardware. This implementation simplicity determines solutions based on different interpretation of SIP protocol, permitted by laxity of SIP standards, which leaves room for proprietary adjustments in features such as advanced calling features, security or QoS. Hence compliance with SIP standards doesn't guarantee seamless communication between end-users that leverage on different IP-PBX systems. The SIP-Forum [8] tries to solve this problem with an architecture framework proposal (SIPconnect) that define a minimal set of IETF and ITU-T standards that must be supported and provides precise guidance in the areas where the standards leave multiple implementation options and specifies a minimal set of capabilities that should be supported by the SPs and enterprise's networks [9].

Some vendors try to promote the sharing of knowledge and experience through on-line community [10]. All this approaches demonstrate the existing difficulty in SIP-Trunking Architecture deployment, specifically in very large and multi country scenarios; but new services and voice traffic cost reduction push Large Enterprises to issue Request For Quotation for SIP-Trunking solutions in order to put in competition SPs and gain the best ROI (Return Of Investment). So customers ask for quick deployment and SPs must be able to respect customers requirements.

This paper addresses some topics encountered during the implementation of complex systems integration projects implemented by Italtel for Large Enterprises and Service Providers.

2. SIP STANDARDIZATION MODEL

The SIP protocol is standardized by IETF Requests For Comments (RFCs) developed in an open and communal environment. To have the greatest possible consensus in committees and satisfy the largest number of participants, usually the RFC bloated in both size and flexibility. The specifications are full of weak terms like "May" and "Should" that allows the developers of SIP-based systems to make plenty of "free decisions". So two VoIP systems may be incompatible with each other while being both compliant with specs; e.g. there are many standard ways to transport DTMF (Dual-tone multi-frequency) tones:

- RFC2833 [11]: specialized payload packets in the RTP;
- RFC4730 [12]: SIP-NOTIFY with XML docs;
- SIP-INFO [13]: a SIP message with specific payload.

This approach is entirely different from ITU-T specifications that the Telecommunications Industry was regulated or standardized for the last four decades.

Hence the problem does not start with the technology, but with the approach in creating standards.

3. SIP BUSINESS TRUNKING ARCHITECTURAL MODEL

The SIP Business-Trunking reference architecture is a five level hierarchical architecture divided into two domains: Public and Private.

The *Public Domain* has two levels represented by:

- Class 5 softswitches (SSW-C5), that are the first level of the network model;
- Network Border-Elements (second level) that is the frontier of the public domain towards customers.

The *Private Domain* has three hierarchical levels:

- Corporate Border-Element, the frontier towards the public network;
- Corporate Switching Node (CSN);
- IP-PBXs.

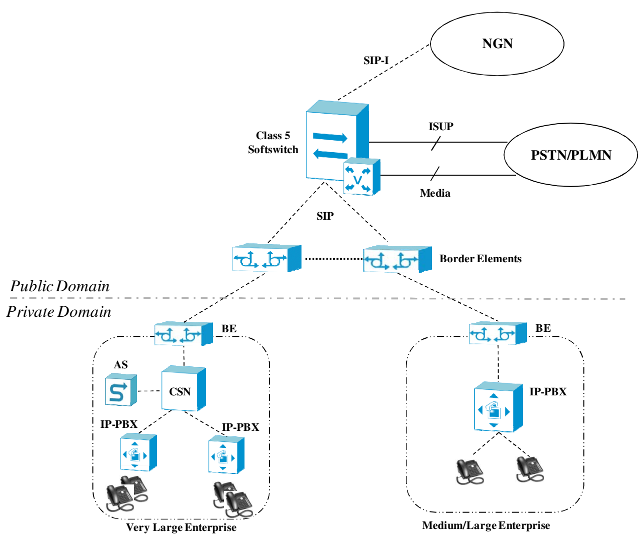

Figure 1: Network Architectural Model

The SSW-C5 are generally connected northbound to the SP's transit network (Class-4 Exchange) via ISUP or via SIP-I/SIP-T whereas southbound SSW-C5 provides a SIP User to Network Interface (UNI) that allows the end user access to the PSTN/PLMN; the Business-Trunking UNI is defined combining a set of RFCs with appropriate policy defined by the carrier.

Between the SSW-C5 and Private Domains there is a layer of Border-Elements (BE) that provide the following main features:

- network topology hiding;
- application layer firewalling;
- NAT-SIP aware.

These devices provides termination and reorigination of both signaling and media between Public and Private domain.

The CSN can be inserted in the architecture when the private VoIP network is very large and there are benefits having a local session routing capabilities.

The CSN provides the following main functionalities:

- Corporate's IP-PBX interconnection point;
- Centralized session routing;
- On-Net/Forced On-Net calls management;
- Corporate's private numbering plan management;
- signaling mediation between Corporate VoIP network and SP's Business Trunking UNI;
- private/public phone identity translation.

The CSN represents also the Network Element (NE) that interconnects Application Server like: Fax-Server, Microsoft Office Communicator , Presence-Server, etc.. Lower the CSN there are multiple IP-PBX with attested IP-Phones.

The described architecture can be declined in two different ways depending on the treatment modality of the user-plane (RTP flow) that can be:

- Fixed-Media modality;
- Variable-Media modality.

3.2 Fixed vs Variable media

In the VoIP world the RTP stream (user-plane) typically is peer-to-peer, this means that end-points negotiate end-to-end media informations and they are always involved in each media stream variation (e.g.: when an hold with music is invoked). The signaling passes end-to-end through al network elements involved in the path. This type of user-plane treatment is called *Variable-Media*. In the traditional ISDN-PBX the media is anchored by the PBX itself that terminates BRA/PRA links and, when needed, perform signaling and media loop splitting the call in two different legs.

IP-PBX's vendors have realized some dedicated elements that perform media anchoring and emulates traditional PBX behavior called *Fixed-Media*; for example Cisco has MTP (Media Termination Point) function [14] and Avaya has Prowler Cards [15].

The *Fixed-Media* architecture in IP-PBX dramatically simplifies the interconnection with carriers via SIP-Trunking because all the calling scenarios are reduced to a "basic-call"; so the *Fixed-Media Architecture* is characterize by the presence of anchoring equipments.

Unfortunately elements like MTP etc., increase costs without added value to the customer and in some cases creating limitations to the evolution of the services (for example they are not able to treat video); hence recommended choice is to adopt *Variable-Media* model.

3.4 SIP Business Trunking Benefits

The SIP Business-Trunking allows to realize a centralized dedicated telephony infrastructure at the enterprise's premises that permits to conform services delivered to all offices with the added value of customizing them to the needs of the end users.

A centralized platform management permits the control of the delivered service's quality (think of companies with so many small/medium sites all around the world, where telecommunication is a commodity).

SIP also revolutionizes the way through which Enterprises can realize interconnection with SPs; so it is possible to concentrate the interconnection with the public network in a centralized point with a substantial saving of cost of interconnection.

Moreover the Large Enterprises can choose more than one SPs and can choose the trunk to be used depending on type of traffic (towards PLMN, PSTN, international…) and rate applied by the SP's.

4. SIP BUSINESS TRUNKING SIGNALING MODEL

The setup of a voice call with SIP protocol is based on signaling flow that allows the network to perform the session routing and the bearer capabilities are negotiated by the endpoints through the offer/answer procedure as specified in the RFC3264 [16].

Referring to the SIP business trunking architectural model, in the following figure an example of a VoIP signaling call flow is depicted. The call is initiated by an IP-Phone directing towards PSTN, through a SIP-Trunk interconnection. The call is terminated by a SSW-C5 with trunking gateway functionalities and redirected toward PSTN. In the example is supposed that CSN use an Inter-Cluster-Trunk protocol to dialog with corporate's IP-PBXs and the IP-PBXs use custom protocols versus controlled IP Phones (e.g. Cisco Skinny Protocol).

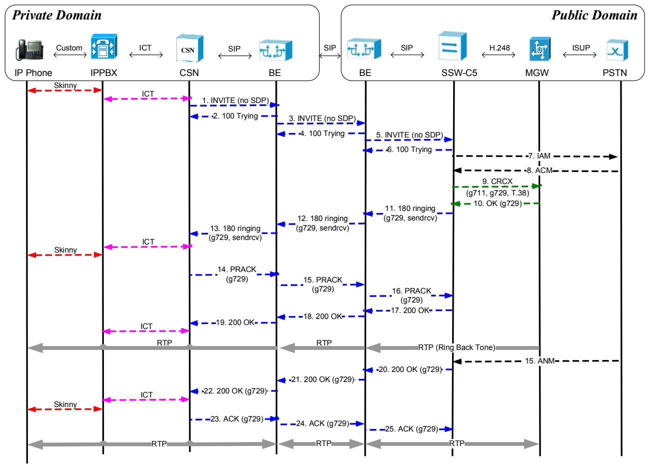

Figure 2: Call Setup in the reference architecture

The signalling model varies depending on whether you choose the *Fixed-Media* or the *Variable-Media* modality; depending on the choice made for media we will speak respectively about:

- *ISDN-like SIP-trunk* signalling model;
- *Pure SIP-trunk* signalling model.

ISDN-like SIP-trunk model is not detailed in this paper because it does not give any added advantage as described in 3.2; so we will focus our attention on *Pure SIP-trunk model* and we will introduce a newly proposed *Hierarchical SIP-model* that maintains the flexibility of pure model and simplifies the interworking between different technologies.

4.1 Pure SIP trunk signaling model

In a pure SIP signaling model with variable media, it is expected that messages originated in a corporate premise will be propagated to other corporate premises involved in the scenario. The following figure is an example of the simplified signaling flow after a successful call setup, the called user performs a blind call transfer towards a PSTN number:

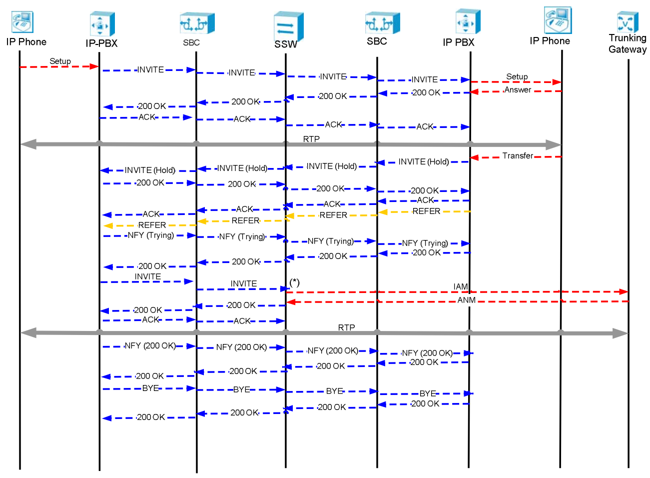

Figure 3: Call flow example (simplified without CSN)

The initial INVITE-200OK-ACK [23] exchange determines the establishment of the call; after that the end-user performs a call transfer. The call transfer invoked by the user of Enterprise-B, triggers a sequence of signaling that cross all the network elements and impacts also the IP-PBX of Enterprise-A (see REFER [17] in figure 3 that crosses all the network).

The IP-PBX of Enterprise-A must be able to understand signaling and perform requested service; in this example it must manage the REFER method and implement the call transfer service. In the pure SIP signaling model the service logic is distributed only between the two IP-PBX, whereas the SSW-C5 performs the routing functions.

This is a simplified approach but implies that all the IP-PBX are able to perform requested service logics and, when a new behavior is introduced in the network by a new element, all the interoperability tests must be performed to be sure that no regression happened. In the figure's example there is also another point of attention (marked with *): in the signaling flow proposed by RFC5589 [18] a new INVITE is issued by transferee. The INVITE could not be correctly associated to the call transfer service, so the transferee can be in the condition to be charged for the entire leg of the call (from transferee to target). The transferor will not be charged for the second leg of the call (and this can be not compliant with country charging requirements). By the way "plain" SIP networks are simpler than hierarchical one but the latter better fulfill SP's requirements like Call Detail Records generation (CDR), (crucial for billing purposes) and SP's UNI technical specifications.

4.2 Hierarchical SIP signaling model

To establish successfully sessions between two users and to manage the various scenarios that can happen during the call, it is important to consider and manage different aspects such as:

- Identity management & certification;
- Voice-codec compatibility;
- DTMF interoperability;
- Encryption;
- Fax support;
- CDR generation.

The call-flow in figure 3 highlights how each new NE or new service introduced in the network may have impacts towards all other NEs present in the Public or Private domains. The pure SIP signaling model has a big constraint so in this period of standardization laxity it is better to implement a hierarchical architecture where it is foreseen some *harmonization functions* that split various domains and simplify the interworking procedures.

The harmonization function can be performed either in a dedicated network element or embedded in the SSW-C5 or BE. The right choice depends on the specific characteristics of the network; a hierarchical architecture allows adopting some specific rules on the SSW-C5 that force a well defined behavior on the UNI and guarantee SPs to maintain the control on user's operations.

5. GO TO MARKET APPROACH

Waiting for an efficient standardization of SIP, the SPs have two possibilities:

- Full Jacket (FJ)
  If the volume (number of enterprise customers) is high, the SP can define some pre-packaged solutions. This approach is usually utilized by the incumbent Carrier

- Customer Tailored (CT)
  The SP tries to satisfy all the customer's requests without imposing technical limitations or pre-packaged service bundles

5.1 Full Jacket

The Full Jacket approach defines all the various equipments and services that can be supported by the solution and combine them into bundles; each bundle is pre-certified and can be sold as an out-of-the-shelf product. The SP sells to the Enterprise all the necessary platform and terminals to provide services. It's more easy to propose a FJ bundle when the customer is a green field.
Each time it is required to introduce a new service and/or a new device, it is necessary to update the bundle with dedicated study and test sessions.
The expected effort for this kind of activity can be evaluated as a proportion of the possible combinations between all the devices as show in the following formula:

$$[1] \quad \sum_{f=1}^{TF} \sum_{i=1}^{NS-1} \sum_{j=i+1}^{NS} n_{f,i} * k_{f,j}$$

Where TF is the total number of functionalities that must be supported, NS is the number of Enterprises, $n_{f,i}$ is the number of typology devices that are involved for the f function on the enterprise i. The $k_{f,j}$ is the number of typology devices that are involved for the f function on the enterprise j.
The effort to validate the solution is made at the first implementation, with this approach we operate every time in a Homogeneous Domain (HoDo).

5.2 Customer Tailored

The Customer Tailored approach complies with all the requests of the customer case-by-case. This means that all the devices, terminals and services are selected by the Enterprise based on its commercial/technological requirements.
All the devices and terminals are customer ownership and usually are managed by the SP with a specific contract of maintenance. The delivery effort depends on the typology

and the number of devices involved, not only in current implementation, but also in the previous one. So from the SP's point of view the complexity increases with the increase of acquired customers.
The expected effort can be evaluated as indicate in the formula [1].
Each time a new Enterprise with different devices is added to SP's network, the effort put in solving the problems can be evaluate as show in the following formula:

$$[2]\ \sum_{f=1}^{TF} \sum_{j=1}^{NS} n_f * k_{f,j}$$

Where n_f is the number of typology devices involved in the f function by the new enterprise. NS is the number of already existing Enterprise. The $k_{f,j}$ and TF are the same of formula [1]. With this approach we operate every time in a Inhomogeneous Domain (InDo).
Usually an Enterprise which selects InDo has the trend to realize multi vendor domains in their premise, so the interoperability problems start already at customer's home.

6. RECONCILATION BETWEEN THE APPROACHES

Both approaches (FJ and CT) can have serious compatibility problems with the volumes expected in the next three years in the SIP business trunk market.
Each SP will have a complex InDo, composed by islands of HoDo; there is a concrete risk to have an unmanageable network with exponential growth of OPEX. We propose to adopt the SIP Business Trunking and signaling architectural model (cfr. 3, 4.2) with a new network function (or element) denominated *Inter-Domain Adaptation Device* (IDAD) which understands simultaneously all the "standard" SIP-messages and call flows and can harmonize them.

The IDAD perform the following main functionalities:

- Media Anchoring;
- Dynamic Audio/Video transcoding;
- Signaling decoupling & normalization;
- Interdomain features harmonization;
- Session Admission Control;
- Fax adaptation;
- DTMF interworking;
- Encryption termination.

Italtel has developed a feature called Media Termination Function (MTF) on board of its own Softswitch (i-SSW [19]) that performs some of IDAD's functionalities.

6.1 Media Anchoring

In a standard SIP session the RTP flow goes peer-to-peer; Media Anchoring feature allows IDAD to anchor the media stream splitting the RTP flow in two segments and to monitor both session's user and control plane and perform transcoding if necessary. The usage of this feature makes also possible to perform signaling normalization as described in 6.3. The anchoring point represents a unique interworking point for all other domains towards which the originating domain will be connected with.

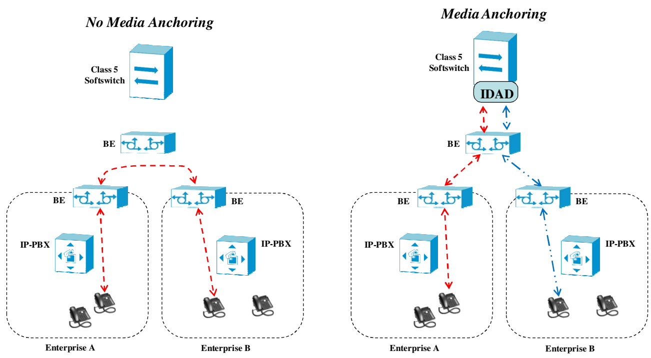

Figure 4: Media path with/without media anchoring

The singularities of the originating domain will be managed by IDAD preserving other domains thus simplifying interoperability. The Media Anchoring is also an opportunity to be able to complete successfully a session between two domains with no common shared codec (IDAD intercepts the RTP and performs the required media transformation).

6.2 Dynamic Audio/Video transcoding

During the SIP session setup it is performed the offer/answer procedure [16] that is used by user agents (UA) to agree codecs that must be used for communication. If the two parties don't have a common audio/video codec, the negotiation will fail unless IDAD intercept the codec mismatch and engage transcoding resources. Depending on the session setup scenario the IDAD can decide to book transcoding resources based on one of the following events:

- The originating domain presents a set of codecs insufficient for the destination domain (based on provisioning information);
- Catching an error response received by destination UA.

In the first case the IDAD enriches the codec bouquet with all codecs defined by provisioning information and, if the answer will select a codec that the origin domain doesn't support, IDAD will perform requested transcoding.
In the second case when IDAD reveals an error message for codec mismatch, it will issue a new INVITE with an SDP with all codecs supported by IDAD and performs requested transcoding.
An alternative to perform a provisioning of codecs supported by various domains can be represented by the auto-learning feature of IDAD that will learn dynamically (based on error messages revealed during session instauration) the set of supported codecs by various domains and it applies resource reservation and transcoding following self-learned rules.

6.3 Signaling decoupling & normalization

The IDAD, in combination with SSW-C5, performs a decoupling of the signaling among domains terminating and re-originating SIP session in a Back-to-Back logic. This behavior allows IDAD to perform a normalization of headers used/expected from/by various domains. A typical example: *SIP Diversion Header* [20] and *History Info Header* [21], both permitted by standards to indicate in which point a session has suffered a transfer. IDAD performs requested adaptation to make different "SIP-dialects" compatible by different IP-PBX vendors.

6.4 Interdomain features harmonization

Some features or methods not declared mandatory by standards, can create interworking problems between IP-PBX of different vendors; for example some domains are using UPDATE method [22] while others do not support it or direct fax session setup with Fax Relay (T.38). The IDAD performs the interdomain harmonization by interpreting and filtering the particular features used by a specific domain and spreading only basic and more common procedures towards others domains.

For example a fax call that starts directly in T.38 (without a previous establishment of a session that uses a voice codec) the IDAD can emulate towards called domain a more widespread behavior that consists in setting up as first step a G.711 session and try later to negotiate a T.38 fax relay codec; in this way the percentage of a successful session setup is higher and the fallback to fax pass-through mode is guaranteed.

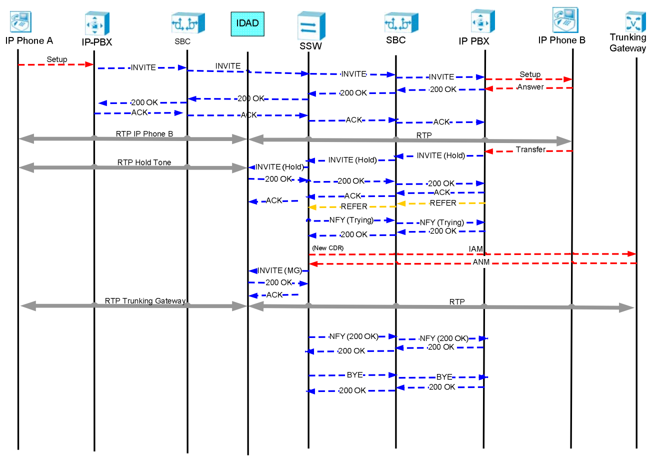

Figure 5: Signalling harmonization

6.5 Session Admission Control

The access bandwidth (data link between enterprise and SP) is a parameter subject to the contract's Service Level Agreement negotiated between SP and Enterprise; so the amount of access bandwidth should be determined as the trade-off between traffic expected and bandwidth's cost. But a well designed access can also go in defect for some rare events when the profile of traffic heavily violates the traffic model considered in design phase.

The Session Admission Control performed by IDAD allows to monitor the saturation of available bandwidth; IDAD per each access considers the used codec and the bandwidth occupation. When the amount of occupied bandwidth surpasses a specific percentage threshold new sessions request will no more be authorized for the specific access.

The IDAD can also differentiate the admission policy depending on the codec used, so for example video can be allowed only when e specific percentage of bandwidth is available or for a limited number of sessions.

6.6 Fax Adaptation

In the IP world there are two common ways to transmit fax:

- Fax Passthrough (G.711 clear channel);
- Fax Relay (T.38).

This two methods are incompatible with each other; the best practice is to use T.38 which is more reliable, but sometimes T.38 cannot be used. So in the real world a SP can have domains that use T.38 (with no fallback to G.711 capability) and others that support only Fax Passthrough. In this case IDAD is able to detect the incompatibility of fax codecs during the session setup phase and perform the necessary transcoding operations and signaling adaptation to have a successful fax transmission.

6.7 DTMF in-band/out-of-band transformation

Once the session is established it's possible for user agents to exchange DTMF end-to-end. DTMF can be transmitted in two different modalities:

- In Band
  The digit is transmitted in the media flow marking in a particular way an RTP packet following RFC2833[11] ;
- Out-of-band
  The digit is transmitted following the signaling path using:
  - SIP-INFO Method
  - KPML (RFC4730 [12])

  SIP-INFO is more widely supported although less well standardized.

In this heterogeneous DTMF transmission modes present today, the IDAD presence in the network resolves DTMF exchanging problems performing transformation among various methods.

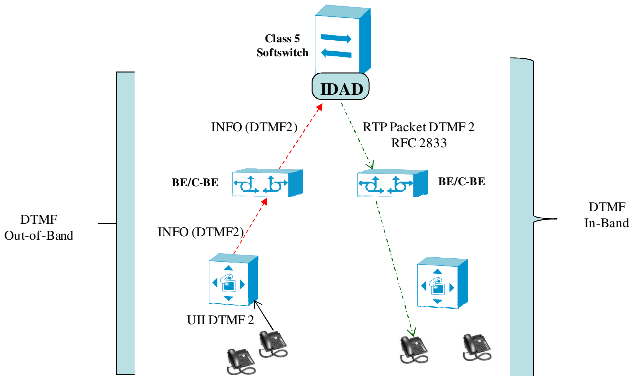

Figure 6: DTMF transformation

Based on the domains' characteristics IDAD decides when it is necessary anchor the media flow and to perform DTMF transformation ensuring interoperability.

6.8 Encryption termination

There are some enterprises which request to have the voice encryption. Typically this kind of requirement can be satisfied inside the customer premises but sometimes this requirement exists also between different enterprises that belongs to the same group.

It is difficult for SP's to provide encryption in this enlarged scenario with different methods of encryption used in different domains, non sharing of key/certificate between different organizations, obligation to be able to apply lawful interception.

The IDAD allows SPs to offer encryption of voice flow between different enterprises maintaining the capability to intercept the voice stream if requested by the authorities; indeed IDAD can anchor the user-plane and perform the required decryption/encryption process.

6.9 Simplification of the complexity

The deployment of a new technology domain implies a campaign of certification tests that covers all possible combinations between different technologies for their interworking because of weak standardization of SIP-Trunk interface.

With the introduction of IDAD platform, the expected effort to insert a new HoDo, will be proportional to the possible combinations between all the devices and the IDAD. See the following formula:

$$[3]\quad \sum_{f=1}^{TF}\sum_{i=1}^{NHoDo} n_{f,i}$$

Where NHoDo is the number of Enterprise in the new HoDo. The TF and $n_{f,i}$ are the same of formula [1].

For example when a new island HoDo, realized by a single enterprise, is inserted, the effort can be evaluate as show in the following formula:

$$[4]\quad \sum_{f=1}^{TF} n_f$$

Where n_f is the number of typology devices that are involved for the f function on the added enterprise.

The effort without IDAD is significantly major than with IDAD as showed by the following formula:

$$[5]\quad \sum_{f=1}^{TF}\sum_{j=1}^{NS} n_f * k_{f,j} > \sum_{f=1}^{TF} n_f$$

7. DISCUSSIONS AND CONCLUSIONS

The objective of this paper was to provide an overview of SIP-Trunking solution, their state of art and show the tendencies of their development. Some problems that impact on the services were analyzed. We strongly believe the most likely evolution of VoIP will be SIP-Trunking, that will permit the creation of new services for end users. But both models used today by Carrier (FJ and CT) may have serious compatibility problems with the market volumes expected in the next years.

The strengthening of the standards is the main way to go forward for multi-vendor interoperable solutions, but we must also consider the market presence of many small-medium players coming onto the world of telecommunications with the advent of VoIP.

These players prefer the development of products/services that are partially compliant with product of other vendors, but meet the needs and timing of the market. It is difficult that these players will waiting for standard's maturity.

For these reasons in this study we have proposed a solution (IDAD) to cover the lapses/lackings of SIP standard, without limiting the network flexibility. We hope this study will serve as a stimulus for improve the SIP standardization and for further research in above-mentioned subject area.

REFERENCES

[1] Salmeron, "Western European SIP-Trunking Market, 2007-2012", Decembre 2008, IDC #WS06Q, volume:1;
[2] ITU-T Recommendation P.800 Methods for subjective determination of transmission quality;
[3] D.Collins, "Carrier Grade Voice Over IP",McGraw-Hill;
[4] SIP Forum – Fax_Over_IP_Task_Group – Problem Statement – V1.0;
[5] SIP Forum – Fax_Over_IP_Task_Group – Addressing the Identified Problems;
[6] Cazzaniga,Garavelli, "Implementation of SS7: Italtel's experience",ieee, jul.1990,vol.28;
[7] http://www.asterisk.org/;
[8] http://www.sipforum.org/;
[9] SIPconnect Whitepaper, "The SIPconnect Technical Recommendation ",rev1;
[10] http://www.siptrunk.org/;
[11] H.Schulzrinne,S.Petrack, RFC2833, "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals";
[12] E.Burger,M.Dolly, RFC4730, "A Session Initiation Protocol (SIP) Event Package for Key Press Stimulus (KPML)";
[13] S.Donovan, RFC2976, "The SIP INFO method";

[14] http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/3_1_1/ccmsys/a05mtp.html#wp1030050;
[15] http://www.avaya.com/rt/master-usa/en-us/resource/assets/applicationnotes/gateway.pdf;
[16] J.Rosenberg,H.Schulzrinne, RFC3264 “An Offer/Answer Model with the Session Description Protocol (SDP)”;
[17] R.Sparks, RFC3515, “The Session Initiation Protocol (SIP) Refer Method”;
[18] R.Sparks,A.Johnston,D.Petrie, RFC5589 ” Session Initiation Protocol (SIP) Call Control–Transfer”;
[19] http://www.italtel.com/allegati/2Solutions-products/products/
[20] S.Levy, Internet Draft, “Diversion Indication in SIP draft-levy-SIP-diversion-11”;
[21] M.Barnes, RFC4244, “An extension to SIP for request history information”;
[22] J.Rosenberg, RFC3311, “The Session Initiation Protocol (SIP) UPDATE Method”;
[23] J.Rosenberg,H.Schulzrinne,G.Camarillo, RFC3261 “SIP: Session Initiation Protocol”;

GLOBAL E-PUBLIC SERVICE

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ABSTRACT

This paper proposes a Global e-Public Service (GePS) to electronically deliver public services in a more efficient way by maximizing the utilization of resources and providing all countries a common platform to help its citizens and reduce the inequity in the world.

Most of primary services which all governments need to provide to its citizens could be delivered electronically through e-government initiatives. These e-services were identified through research of leading e-government portals and a reference model for an e-government portal was developed.

Resources required by each government to provide services common to all are a wasteful exercise. The Global e-Public Service defines a mechanism to develop these electronic public services centrally by a global agency and to implement them by various governments as integrated and decentralized systems around the world.

Keywords— Global, electronic, public, service, government

1. INTRODUCTION

The Global e-Public Service (GePS) is a mechanism for governments to efficiently deliver public services electronically for effective economic and social development.

Today many e-government initiatives are investing heavily to re-invent the wheel, which is a wasteful exercise unless the wheel itself could be improved. These investments sometimes provided by international development agencies for the delivery of public services are held selfishly by governments once systems are developed.

This paper proposes a mechanism to overcome the problem of delivering e-services equitably to public by sharing a superior system and maximizing the return on investment.

Section two on Global e-Public Service identifies critical components such as architecture, implementing organization, team, infrastructure, application development, integration support, and delivery of global services.

Section three on provision of e-public services elaborates on e-government, e-governance, e-health, e-education, e-services to citizens by life-cycle needs, e-services for businesses and entrepreneurs, e-services for visitors to the country, common e-services and help.

Reference model for e-government portal is also proposed.

Section four discusses challenges such as global Internet penetration, literacy rate of public, availability of e-services, willingness of governments, and global organization for implementation.

Section five looks at opportunities such as reducing inequity, inter governmental collaboration, sharing of resources, business opportunities, provision of novel services.

Section six makes recommendations such as independent organization for implementation, composition of implementation team, provision of infrastructure facilities, strategy for implementation, changes to government policy, changes to future Internet standards.

Finally section seven makes conclusions on the proposed GePS.

2. GLOBAL E-PUBLIC SERVICE

E-public services are too sacred to be held by commercial interests and provided by businesses. Even when they aren't provided commercially, it would deprive the public of most countries who cannot afford them. Usually the technological expertise is with commercial enterprises and they have the right to provide services at a profit.

Therefore, on humanitarian grounds alone, it is essential for governments to have the capability to provide at least the minimum services to public electronically immaterial of their economic capabilities.

Practical way of achieving this is for an international organization to setup the necessary framework to make it a reality. This could be their greatest service to human kind as Information and Communication Technology (ICT) can reduce the inequity among human beings.

Solutions for Internet infrastructure, accessibility, affordability, technological and language literacy need to be addressed separately and GePS looks at the shared delivery mechanism.

2.1. Architecture of GePS

The primary architecture of Global e-Public Service as depicted in Figure 1 consists of a central suite of web applications known as GePS and a corresponding database to hold the common data integrated with other e-government systems.

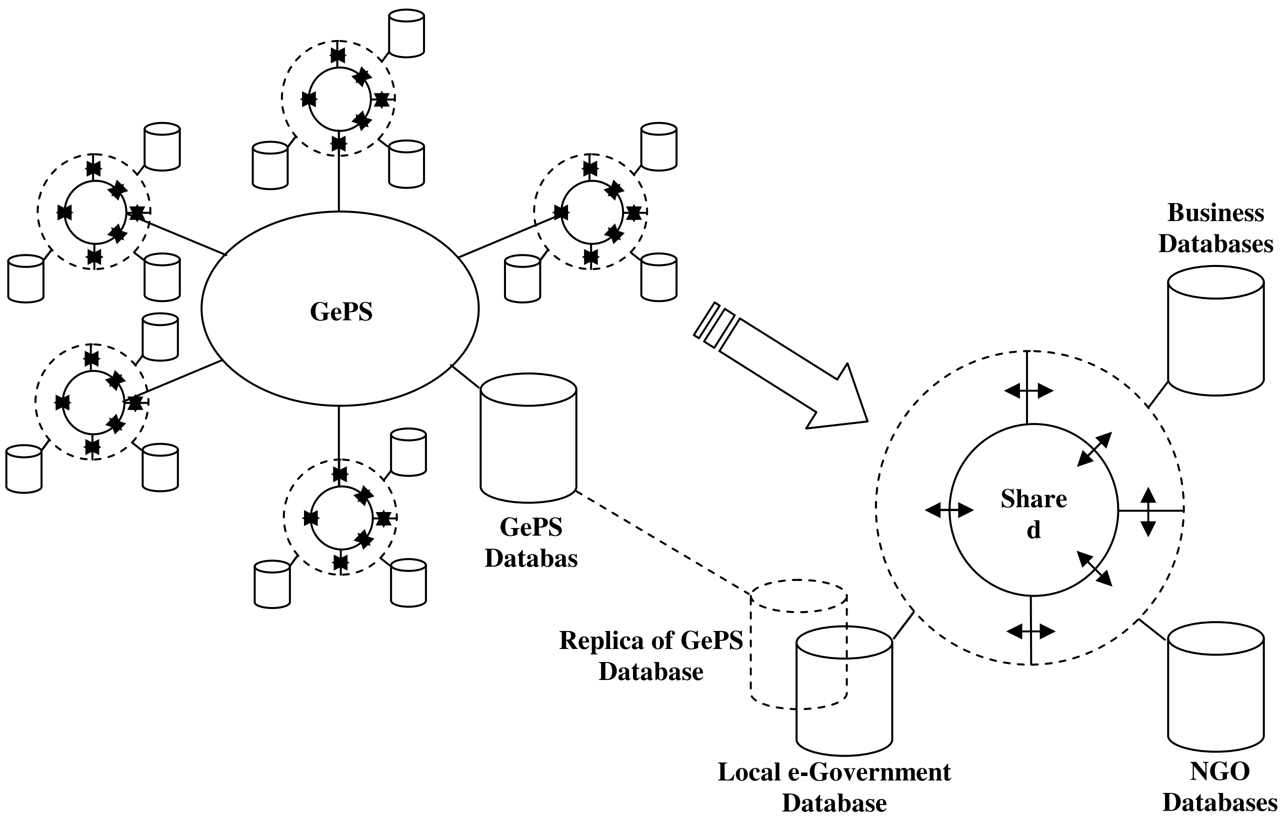

Figure 1: Architecture of Global e-Public Service (GePS)

Each country could integrate GePS applications into their e-government systems without compromising them and provide interfaces to existing and new technology.

At country level, the GePS system could be further integrated with registered businesses and nongovernmental organizations link with their international partners too.

E-government systems of each country would have their local databases, while the businesses and NGOs would have their local/international databases.

Replicas of the GePS database partitions holding data of each country could be held locally for availability and better performance and provide mirroring for further safety.

The GePS database should contain only non sensitive data which countries are willing to share, while sensitive data is stored in their local databases.

The updating of shared information in the GePS database should be done using secure login as depicted in Figure 2.

2.2. Implementing organization

All governments do not have the same resources and expertise to quickly and effectively provide e-services to their citizens. Therefore, an organization such as United Nations or World Bank would be ideal to setup a separate agency for implementation.

2.3. Implementation team

Setting up an effective implementation team would be the primary task of the implementing agency to make GePS a reality. The team at minimum must consist of a project manager, database expert, security expert, web application developers and web site/database administrator.

2.4. Infrastructure needs

The underlying theme of GePS must be "simple and efficient". It should be implementable from day one with basic modules after little or no configuration. It must be designed in such a way that each country could easily adapt it to their requirements and platforms available.

Therefore, the GePS should be able to run on any open or proprietary operating system and use whatever the available database management system rather than forcing one on any.

This would enable them to be easily implemented and seamlessly integrated with other existing, legacy and future applications either in open or proprietary platforms.

Implementing organization need to provide the reliable web hosting and database facilities for the GePS.

2.5. Developing global applications

The success of the GePS primarily depends on the availability of applications to provide public services electronically. As these public services are needed by all countries and some countries may not have the resources to develop them on their own, they need to be developed centrally and made part of the GePS.

The countries which do not have e-government systems could use them as it is; the ones who already have some of them could integrate the ones they don't have; and those who already have them could share their information with the GePS to take them to the next level.

2.6. Integration support

Most countries, whether they already have e-government systems or not, would be having some form of data stores in multitude of systems. Therefore, it is essential to provide necessary expertise to develop interfaces to transfer existing information to GePS or to integrate with it.

For example, sample data connections and scripts for various types of databases and sources should be provided.

Provision of expertise to setup country level data centres and basic data centre facilities for countries who cannot afford such facilities during startup stage would be helpful.

The initial training and support to create country level administrator accounts, update information, and use/integrate with existing systems too should be provided.

2.7. Delivery of global e-public services

Country specific e-Public services could be provided through standard URLs for each country with the following format.

http://countrycode.agency.domain

3. E-PUBLIC SERVICES

There are common public services which most governments provide or would like to do so in the future. Depending on the e-government maturity, some of these services would be delivered electronically. This section identifies some of the most common public services provided by governments in the top twenty five rankings of the United Nations global e-government survey [1].

The success of GePS depends on the initial selection of services which could be provided with the minimum amount of customization. For example, providing a directory of public institutions would need very little customization, whereas a tax system would need high levels of customization and regular modifications.

3.1. E-government

"The use of ICT and its application by the government for the provision of information and public services to the people." - United Nations

The provision of information and public services using ICT is done by various governments in multitude of ways.

Some governments will be delivering them entirely manually, while some using ICT for backend operations and the more developed using electronic means both for frontend and backend operations. However, there are basic services which every government either provides or would like to provide electronically. Some of the critical services which the public would like to obtain are as follows.

- Directory of government institutions such as ministries, departments, boards, commissions, authorities, embassies etc., with at least the following.
    - Contact information of organizations.
    - Contact information of key employees.
    - Services provided with rules, regulations, procedures and required documentation.
    - Frequently asked questions.
- Country specific information, demographics and useful statistics.
- Judicial, legislative, and executive systems.
- Government policies and standards.
- Geographic regions, states/provinces, cities/towns, area codes, and maps.
- Daily weather forecasts, holidays, news and events.

United Nations Knowledge Base (UNKB) of global e-government information and data is a valuable source for research, education and planning purposes [2].

3.2. E-governance

"E-governance is the public sector's use of information and communication technologies with the aim of improving information and service delivery, encouraging citizen participation in the decision-making process and making government more accountable, transparent and effective…" - UNESCO

E-participation is an integral part of e-governance and e-democracy. All citizens would like to interact with their governments and it's prudent for any government to obtain feedback from its citizens. Therefore, governments should provide a basic framework for the public to interact with the government along with information such as electoral lists to be available online for the public use.

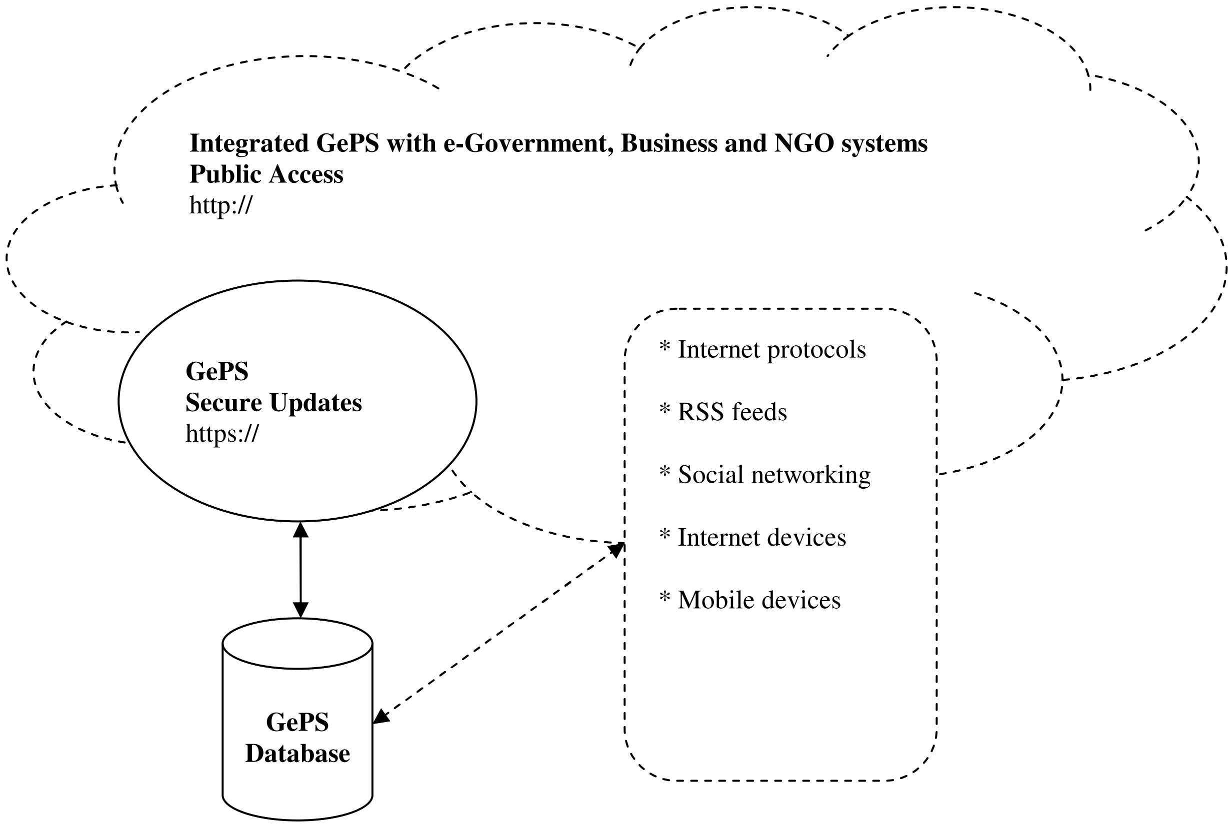

Figure 2: Usage of Global e-Public Service (GePS)

3.3. E-health and e-education

Health is a primary need of any human being and it is also a primary responsibility of any government. Therefore, e-health should be an essential component of e-public services. Governments should provide the information on hospitals and medical facilities, immunization plans, and online health related services and advice.

Similarly future of any nation depends on the education of its citizens. Each country has its own education systems, curricula and examinations. Governments should provide information on schools, tertiary institutions, universities and mechanism to provide education and training electronically.

3.4. E-services to citizens by life-cycle needs

E-services would be found easily by citizens when they are provided as per their life-cycle needs such as follows.

Children: Birth certificates, immunization, schooling, identification cards, national examinations.

Adults: Tertiary and university education, adult and e-learning opportunities, education loans, employment, social security/employee benefits, marriage certificates, housing and property, driving licenses, passports.

Seniors: Pension/retirement benefits, retirement homes, social support.

Common: Medical facilities, death certificates.

3.5. E-services for businesses and entrepreneurs

There are certain services the business organizations and entrepreneurs would like to obtain such as business registration, business guides and directories, laws and regulations, banking, insurance and financial institutions, accounting, auditing and taxing authorities, land and labour, trade related organizations, foreign currency and monetary policies, import/export services, customs and duties, logistical companies, business consultants, investment opportunities etc.

3.6. E-services for visitors to the country

All countries have visitors in the form of tourists, expatriate workers, business personnel, official visitors and immigrants and would need information on embassies, visa services, accommodation, travel services, places to visit, cultural events, recreational activities, local customs etc.

3.7. Common e-services and help

Information on common public services such as health, security, transport, communication, utilities, weather, etc. is needed.

There are also essential help services such as emergency contact numbers, emergency warnings, disaster management, government help desk, frequently asked questions, search, etc.

3.8. Reference model for e-government portal

E-services to be implemented must be what are most valued by the public and easily implementable to provide.

Table 1 illustrates a reference model for an e-government portal based on those services. These are common services any citizen would like to obtain from their governments and GePS could provide the underlying mechanism to provide them at whatever the e-government maturity level each government is.

This is only a reference model and each government could use it to selectively modify and integrate its services for the benefit of the public.

3.9. Implementation of e-services and advantages

Even a country which does not currently have any e-government portal would be still having traditional government institutions providing various services. Therefore, they could use this to provide at least the contact information and services provided by them.

Government institutions which already have a basic web presence could use this to provide links to their own web sites in addition to contact information and services provided.

Those who already have online services can go further and integrate them to provide a globally compatible system.

Therefore, GePS would be invaluable to whatever the stage of e-government the country is.

Basic services such as contact information of government agencies, the services they provide, and procedures followed could be provided by the implementation agency without any investment on infrastructure or technology by governments.

The only requirement would be the provision of country specific information online by responsible officials through secure login. This would not compromise the confidentiality and privacy of any government or institution as what would be updated is only what is to be shared by the public.

The unprecedented advantage is that all countries in the world would be sharing a common searchable service immaterial of their own systems.

Even the poorest of countries could be part of it and will encourage them to move upwards in their e-government maturity level.

4. CHALLENGES

There are multiple challenges to be faced and addressed in order to maximize the benefits of GePS.

4.1. Global Internet penetration

In order to provide e-public services to the global population, first of all they need to have access to the Internet. As per the “Internet World Stats” it is evident that over 80% of the world population is having an Internet penetration of less than 20% [3]. Therefore, all governments need to improve the accessibility to its citizens by improving infrastructure as well as making it affordable. In fact to reduce inequity the free Internet access must be available for the poor, which may be done through Universal Service Fund or similar mechanisms.

4.2. Literacy rate of public

Even when access to the Internet is available, the public must have the necessary literacy and skills to make use of e-public services. Therefore, the people must have the basic ICT skills, ability to read the content in the language provided. To overcome these challenges, all governments need to improve the basic ICT skills of the citizens, content to be localized whenever possible and improve the language skills as well as supporting the differently abled people.

4.3. Availability of e-services

While the challenges of infrastructure for accessing Internet and skills to use the content are been resolved, we need to address the availability of e-services. This is the primary objective of the GePS and it needs to provide much as possible applications for the delivery of public services electronically.

4.4. Willingness of governments

The success of GePS depends on the willingness of governments to share their information and resources to the greater good of humans. Due to security or other considerations, governments can selectively share only the information it usually provides to general public without any restrictions depending on the country environment and also taking legislative conformance into account.

4.5. Global organization for implementation

The real value of GePS is the ability to provide common public services electronically to citizens of the world. Therefore, it must be done through a global organization acceptable to all governments with necessary commitment and required funding support to make it a reality.

Table 1: Reference Model for an E-Government Portal

Government	Citizens	Businesses	Visitors	Services	Help
Executive Head of government **Legislature** Head of parliament Parliament **Government** Ministries Departments Authorities **Judiciary** Courts Laws & Regulations Policies and standards **Governance** Feedback & Interaction Electoral lists **Country** Country statistics National symbols	**Children** Registration of birth Immunization Schooling National identification National examinations **Adults** Tertiary education University education Employment Social security Marriage registration Housing & Property Driving license Passport **Senior citizens** Retirement benefits Retirement homes Social support **Other support** Medical facilities Registration of death	**Registration** Business registration Guides & Directories **Financial** Banks Insurance Accounting Auditing Taxing Stock market **Regulations** Monetary policies Foreign currency Import/export Customs & Duties **Business support** Trade related organizations Logistical companies Business consultants Investment opportunities	**Travel permits** Embassies Visa services **Travel services** Airlines Hotels & Accommodation Taxi services Travel agents **Tourism** Places to visit Cultural events Recreational activities Local customs Maps **Other travelers** Business travelers Expatriate workers Immigrants	**Public health** Hospitals Medical services **Public transport** Railway service Bus service **Public security** Police Civil defense Military **Utilities** Electricity Water Gas **Communication** Telecom Postal services **General** Weather Holidays News & Events	**Emergency** Emergency numbers Emergency warnings Disaster management **Help desk** Government help desk Frequently asked questions Internet security **Utilities** Virus guards Archival software Currency converter **Login** Citizen login Administrator login **Search** Search site Search web **Policies** Privacy policy Usage policy

5. OPPORTUNITIES

GePS would bring in multitude of opportunities for governments and the industries.

5.1. Reducing inequity

This would be a great opportunity to reduce the inequity of people for countries which are lagging behind or do not have the resources to embark on e-government initiatives by having a minimum set of e-public services to provide.

5.2. Inter governmental collaboration

This would pave the way for future government to government seamless collaboration. Some governments may wish for open interaction with each other while the others may prefer to do so selectively as needed basis.

5.3. Sharing of resources

Even though this is not a must, a novel way to fund and share resources would be, if each and every country contributes 1/5th of what they spend on e-government services to a common fund and used to develop the GePS.

For an example, if each country is spending USD 1 million and 1/5th of that would be USD 200,000 and this into 200 countries would be USD 40 million. That is by losing 20% of country budget you get a return of 4,000% worth system.

The return would be skewed as the richer countries would be contributing more and the poorer contributing less. But this is the ideal opportunity to reduce the inequity by giving more to the poor. The value is even more when you compare the expertise which could be harnessed and the ability to bring in best practices into the common GePS. The further advantage is each country still has 4/5th of their e-government budget for customization and create their unique services.

5.4. Business opportunities

When more and more citizens are exposed to e-services the demand for faster, better and more value added services would be generated. Hence, there will be many opportunities for businesses to provide these value added services to the public who could afford a premium price.

5.5. Provision of novel services

When public services are delivered through a unified mechanism, new service models could be developed to cater to inter governmental and global citizens. Even global early warning and disaster management systems, e-learning, and e-business opportunities across countries could be created.

6. RECOMMENDATIONS

Turning ideas into reality is the most difficult task. The most common reason for that is making them overly complex at the beginning itself. Therefore, the strategy should be to start with smaller projects which can deliver benefits in shortest possible time frame. The GePS is within our reach and every day delayed is a day deprived to the most needed.

6.1. Independent organization for implementation

All governments do not have the same resources and expertise to quickly and effectively provide e-services to their public. Even those who have would be wasting their resources to re-invent the wheel to provide the basic e-services. Therefore, an organization such as United Nations would be the ideal implementing organization or to setup a separate agency with funding from World Bank or other regional development bank.

6.2. Composition of implementation team

Success of e-government is dependent on the implementation team who needs to transform the vision to reality. Therefore, the team needs to be IT savvy, but with common sense to understand technology per se would not make GePS successful. They must work in small teams based on expertise and must have the urgency and sense of accountability for rapid implementation of e-public services.

6.3. Provision of infrastructure facilities

The countries with the most advanced ICT sectors present the highest levels of competitiveness, suggesting that having a country enabled by ICT improves the overall performance of its economy in the long run [4].

The GePS applications and common interfaces should be developed centrally and back-end support provided to maintain a consistent central repository of information, while allowing decentralized update of information by various government institutions on their own. This would enable the governments to have the most up to date information from the direct owners of such information and also the ability to present them in a uniform manner to the general public. Therefore, initially a central database could be provided as a data repository and a secure web site as the common interface for information update.

Mobile cellular has been the most rapidly adopted technology in history and today the most popular and widespread personal technology on the planet, with an estimated 4.6 billion subscriptions globally by the end of 2009 [5]. This too needs to be taken account for future delivery of GePS.

6.4. Strategy for implementation

The implementing organization that would setup a project management team needs to set timelines for implementation. The ideal scenario would be to initially select a country with high level of e-government maturity along with one which is moderate and another which is very low for a pilot implementation that are passionate to get involved and showcase for the citizens of the world.

The first stage is to provide each government with a super user account with the ability to create administrator accounts for each of its government and other institutions. These administrator accounts in turn should be able to create organization unit level administrator and user accounts so that all pertinent information could be updated by the owners of such information itself.

The second stage should be to help the governments to use their existing information through bi-directional data interfaces to protect the investments already made on such initiatives.

Third stage should be to allow governments to customize the usage of information in GePS to suit their individual requirements without compromising the GePS.

Therefore, while encouraging government institutions to have their own web sites, the GePS should provide the common applications and interfaces to update the information, which is to be provided to the general public in a uniform manner. But this is in no way to curtail the creativity and innovation of individual institutions as this same information available centrally can be used dynamically by their own web sites using different layouts and formats creatively and innovatively.

Most of the countries would need to provide the e-government portal in multiple languages. Hence, menu options should be database driven with translated text.

6.5. Changes to government policy

The success of the GePS should be ensured by governments by making changes to its e-government policies. It should make it mandatory for all government institutions to have a GePS account. This could be extended to all government registered businesses and NGOs too.

Each government needs to take legal issues pertaining to each country and use features conforming to its legislature.

There is no limit to the information you could provide, but there should be a minimum amount of up to date information available for each government entity made mandatory by government policy.

This would be a win-win situation to both the government and each institution as any changes to the information would be immediately reflected at every access point dynamically instead of using static and not so up to date information on various web pages. The greatest advantage is that the governments can improve applications and interfaces centrally and provide them on the fly based suggestions provided by institutions and the general public.

6.6. Changes to future Internet standards

The future Internet standards should incorporate mechanisms to support the delivery of GePS through better protocols and delivery mechanisms. Initially they could be XML based, but with dynamic ability to share e-public services within and among e-government systems. Location based automatic delivery of disaster warnings and advertisement of news and events embedded into to future HTML would be an example.

7. CONCLUSIONS

The proposed Global e-Public Service (GePS) is to unify the disconnected systems of various governments, businesses and nongovernmental organizations with the primary aim of providing better services to public and reducing inequity among the citizens of this earth.

Therefore, it is prudent for governments to openly welcome the GePS with the intention of incorporating the positive features into their local e-government initiatives.

International development agencies that fund and support e-government initiatives could use their resources much more efficiently across countries and provide more effective services to the public.

The future Internet standards could incorporate mechanisms to support the delivery of such services through better protocols and delivery mechanisms.

An organization such as International Telecommunication Union (ITU) would be ideal to initiate a project such as GePS with the expertise its members have and garner the support of international development agencies and banks for funding and implementation, which would be the most economical, efficient and effective way to help the public.

REFERENCES

[1] United Nations Public Administration Network (UNPAN), “Global Survey”, http://www2.unpan.org/egovkb/global_reports/index.htm

[2] Division for Public Administration and Development Management (DPADM), “UN E-Government Development Knowledge Base”, United Nations Department of Economic and Social Affairs (UNDESA), http://www.unpan.org/egovkb/

[3] Internet World Stats, “World Internet Usage Statistics” http://www.internetworldstats.com/stats.htm

[4] World Economic Forum, “Global Information Technology Report”, http://www.networkedreadiness.com/gitr/main/fullreport/index.html

[5] Telecommunication Development Sector (ITU-D), “Information Society Statistical Profiles 2009”, Statistics of Telecommunication Development Bureau, International Telecommunication Union (ITU), http://www.itu.int/ITU-D/ict/

INTEGRATING WIRELESS SENSOR NETWORKS AND MOBILE AD HOC NETWORKS FOR AN ENHANCED END-USER EXPERIENCE

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ABSTRACT

Wireless sensor networks (WSNs) sense and aggregate ambient information (e.g. space, environment or physiological data). Ambient information can enhance end-user experience and is made available to end-user applications (which may reside in another network) via gateways. Gateways are usually centralized and fixed. Mobile ad hoc networks (MANETs) are networks that can be deployed "on the fly". They are useful in situations such as emergency response operations. When the ambient information collected by WSNs is intended for applications residing in a MANET, centralized and fixed gateways are not practicably feasible. This paper proposes an overall two-level overlay architecture to integrate WSNs (with mobile and distributed gateways) and MANETs, for an enhanced end-user experience. It also proposes a new architecture to interconnect the two overlays of the overall architecture. Motivating scenarios are presented, requirements are derived, and the two-level overlay is discussed along with the proposed interconnection architecture. The prototype is also presented.

Keywords— Wireless sensor networks, mobile ad hoc networks, peer to peer, overlay interconnection, session initiation protocol

1. INTRODUCTION

Wireless sensor networks (WSNs) are sets of distributed nodes that collaborate to monitor physical, environmental and physiological conditions [1]. They are made up of sensors that do the actual sensing; aggregators that aggregate information; and sinks and gateways that enable communication with the end-user applications. The information they collect is known as ambient information and can significantly enhance end-user experience.

Reference [2] illustrates how end-user experience can be enhanced in the specific case where WSNs are integrated with 3G networks, and the gateway is centralized and fixed. In health monitoring, for instance, the status of patients can be tracked using ambient information collected by WSNs. The information is relayed to a fixed and centralized gateway which will then transmit it to end-user applications residing in 3G networks. If a patient's physical condition, as measured by the WSNs, so indicates, these applications could automatically establish a 911 call between that patient and the PSAP (Public Safety Answering Point), thus greatly enhancing the patient's end-user experience (and possibly saving their life).

MANETs are transient networks formed dynamically by a collection of arbitrarily located wireless mobile nodes without relying on any existing infrastructure or centralized administration [3]. They are particularly useful in emergency situations such as wars and natural disasters. When the ambient information collected by WSNs is intended for end-user applications residing in MANETs, centralized and fixed gateways are usually not practicably feasible. This is due mainly to the infrastructure-less nature of MANETs and the constraints on the end-user devices that are most often used in MANETs (e.g. PDAs and smart phones with limited processing power and memory capacity that cannot host centralized gateways).

This paper proposes an overall two-level peer to peer (P2P) overlay architecture to integrate WSNs (with distributed and mobile gateways) and MANETs, to enhance the end-user experience of the MANET participants. It also tackles the challenging problem of interconnecting the two overlays of the overall architecture. P2P overlays are logical application layer networks built on top of the real networks. In these networks, nodes, known as peers, communicate and collaborate with each other in a distributed and ad hoc manner without using a centralized control point [4].

The next section presents a motivating scenario and derives the requirements. A discussion of the overall two-level overlay architecture follows. The fourth section is devoted to the interconnection architecture. The fifth section deals with the details of the interconnection protocols and procedures. Implementation of our prototype is then described in the sixth section. We then present some related work and end with our conclusion.

2. MOTIVATING SCENARIOS AND REQUIREMENTS

2.1. A motivating scenario

Let us imagine a large-scale disaster, such as an earthquake occurring in a city. In this situation, the existing infrastructure may be damaged or destroyed. Victims need to be evacuated and require intervention from paramedics. Because a hospital is either too far from the area or it is already overwhelmed from the disaster, the paramedics decide to provide on-site examinations of the patients ("stay and play attempts"). They install physiological sensors to monitor the patients' health situation (e.g. body temperature, heart rate) continuously. Some similar scenarios can be found in [5].

In order to quickly and efficiently establish emergency medical services, a MANET is formed between the paramedics and this MANET is connected to the network(s) of hospitals with specialized doctors. The end-user devices carried by paramedics (e.g. PDAs, mobile devices) collectively act as distributed and mobile gateways towards the applications residing in the MANET and that are distributed over the very same end-user devices. This is due to the non-feasibility of fixed and centralized gateways. Gateway devices can gather, aggregate, and transmit ambient information collected by the WSNs when the paramedics are walking in the disaster area.

From an end-user application point of view, each paramedic may subscribe to specific emergency events related to each patient in order to receive alerts (e.g. instant messages) on their health status. In addition, they may need some interaction between each other, as well as cooperation and instructions from the specialized doctors in the hospital. Therefore, a multimedia conferencing session may be established as the result of a trigger being hit (an urgent situation) based on WSN sensed data or on demand. Patients' information may be captured, displayed and shared in real-time for remote interactive consultations. Remote specialized doctors can give some audio or video guidance and feedback to the paramedics, based on the ambient information provided by the WSNs, to assist them with the situation.

2.2. Requirements

In this section, we derive general requirements for the overall system as well as the specific requirements of each part of the system (i.e. gateway and end-user application functional entities).

2.2.1. General requirements

Since participants in a MANET may leave and join anytime, and the whole system will be built on top of MANET end-user devices, the first requirement is that none of the entities in the proposed approach, including gateway and application, should be permanently centralized. In addition, as a second requirement, the system should be self-organized and support self-recovery.

As the third requirement, the approach should scale both in terms of the number of WSN nodes and the number of MANET participants; meaning that it should function in the same manner even in large-scale deployments. It should also be simple to set up. The fifth requirement stipulates that it should use resources such as memory and processing power in an efficient manner, since mobile devices deployed in a MANET environment are heterogeneous and resource-constrained. This implies the design and deployment of light-weight protocols.

The last requirement is that the solution should be completely based on the application layer protocols and independent of lower layer protocols. The protocols used should of course be standard whenever possible to ease deployment and inter-operability.

2.2.2. Application and gateway related requirements

As a consequence of the first general requirement, only distributed gateways and applications can be used. Furthermore, they must be mobile in the scenarios we have envisioned and have previously illustrated with an example.

The support of different types of applications, including both session-based and non session-based, is required to cater to the various needs of the MANET participants. In session-based applications, the application entities need to establish a session and store information about that session (e.g. any multi-party application such as conferencing). In non session-based applications, nodes do not need to establish a session and they do not have previous knowledge about any messages (e.g. event notification applications, stateless request response systems).

The gateway should aggregate, filter and store the ambient information it receives from the WSN. It should also translate the information model used internally in the WSN to the one required by the end-user applications and vice a versa. In addition, it should provide both synchronous and asynchronous communication modes.

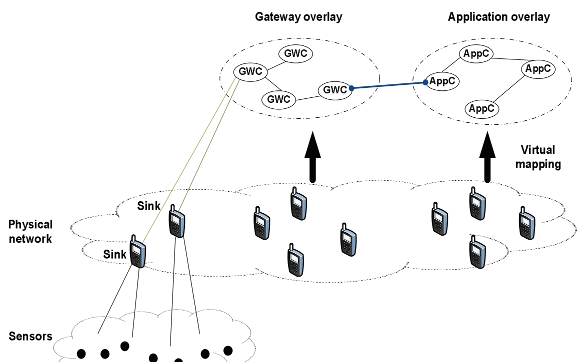

Figure 1: WSN and MANET integration architecture

3. THE TWO-LEVEL OVERLAY ARCHITECTURE

Figure 1 depicts our overall architecture. It comprises two overlays built on top of the MANET; the first is the gateway and the second is the application. GateWay Component (GWC) nodes are components which collectively act as a gateway. AppC nodes are Application Components that collaborate to fulfill application tasks.

Although the figure shows only one application overlay, there may be several applications using the same gateway. Using overlays fulfils our general requirements. They enable self-organization and self-recovery. They also decouple applications and gateways from the lower layer protocols. Previously proposed overlay-based gateway architecture in [6] to interconnect WSNs to 3G networks meet many of our gateway-specific requirements and could be used as a starting point for the design of the gateway overlay level of our architecture. The same observation applies to overlay-based architectures proposed in [7, 8] to provide enhanced end-user multimedia services. They can be used as the starting point for the design of the application overlay of our two-level architecture.

The key remaining issue to be solved is how gateway and application overlays exchange information. A specific architecture is therefore needed to provide interconnection between these two overlays.

4. THE OVERLAYS' INTERCONNECTION ARCHITECTURE

In this section, we derive the requirements for overlays' interconnection. This is followed by our architectural assumptions and principles.

4.1. Requirements on interconnection of gateway and application overlays

The first requirement is that the proposed interconnection architecture should be independent of both P2P overlay architecture and middleware since there exists a wide range of architectures (e.g. structured vs. unstructured) and middleware (e.g. JXTA, Chord). The second requirement is that the system should provide synchronous and asynchronous communication modes. Our third requirement is that the architecture should not rely on the lower layer protocols used in the overlays that are being interconnected, since these protocols may change in different environments. The fourth requirement is scalability, in terms of the number of overlays as well as the number of nodes in each overlay. This is necessary because there will be several end-user applications and there may also be an important number of participants in the MANET.

The next requirements is that the overlay interconnection approach should be self-organized and have no permanently centralized node, as it is built on top of an existing MANET. The sixth requirement is that our interconnection mechanism should be efficient in its use of the resources of the mobile devices involved. The seventh requirement is that the proposed solution should be based on lightweight protocols to fulfill the optimal usage of resource-constrained mobile devices. The last requirement is that the protocols used in our architecture should be standardized and commonly used protocols so that they can be easily deployable.

4.2. Architectural assumptions and principles

4.2.1. Assumptions

Our fundamental assumption is that the overlays are designed with interconnection in mind. This means, for instance, that the nodes in the overlays to be interconnected are aware of the roles they play in the overlays and are able to eventually play additional roles related to interconnection. It also means that some nodes belong to both gateway and application overlays (e.g. they may be nodes that already play roles in both overlays or they may be introduced, for the purpose of interconnection, into the real network on which the overlays are built).

4.2.2. Principles

For interconnection purposes, we define two new roles: *interconnector* and *super interconnector*. An *interconnector* is a node which belongs to both overlays. It can be seen as a node that plays a role in each overlay (i.e. APPlication InterConnector (APPIC) and GateWay InterConnector roles (GWIC)). One or several *interconnectors* may exist at the same time. A *super interconnector* (Super APPlication InterConnector (SAPPIC) or Super GateWay InterConnector (SGWIC)) belongs to a single overlay and has an address known by all the other nodes in the same overlay, and we assume there is only one per overlay. It is actually an interconnection bootstrapping node. It keeps track of all the *interconnectors*.

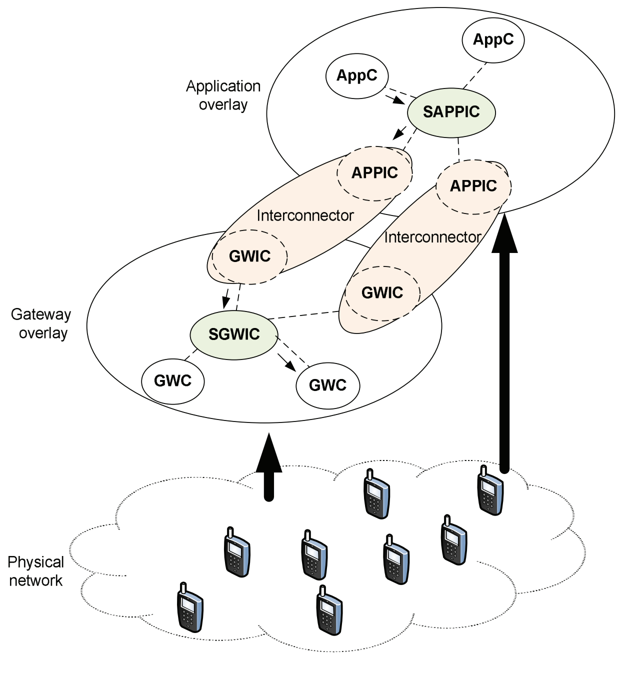

Figure 2: Overlays' interconnection architecture

Whenever a source node (e.g. an Application Component (AppC)) in an overlay wants to communicate with a destination node (e.g. a GateWay Component (GWC)) in another overlay, it sends the message to the *super interconnector* of its overlay. The *super interconnector* will select an *interconnector* based on defined criteria such as node power or load level, and forward the message to that *interconnector*. The *Interconnector* will then relay the message to the destination overlay and transfer it to the *super interconnector* in the destination overlay, which will finally send it to the corresponding node (destination node) in that overlay.

Figure 2 depicts an example of the information flow between two overlays. This exchange relies on a protocol we call the interconnection protocol and is described in the next section. This protocol must be supported by all the nodes in the overlays to be interconnected, which ensures that the proposed architecture is independent of the protocols used in the overlays, and consequently of the architectures of the overlays to be interconnected.

5. THE OVERLAYS' INTERCONNECTION PROTOCOLS AND OPERATIONAL PROCEDURES

5.1. Overlays' interconnection protocol

We have selected the Session Initiation Protocol (SIP) [9]. SIP meets all the previously presented requirements. It is a light-weight standard protocol which is easily extensible and widely deployed. It is interoperable with a variety of mobile devices. It is independent of P2P middleware and can carry the required information with acceptable overhead.

Furthermore, SIP has also been used successfully in overlay self-organization procedures. The only entity of SIP that we use is the SIP User Agent (SIP UA). Two extensions of SIP methods are deployed: INFO method [10], and SUBSCRIBE/NOTIFY methods [11].

5.2. Overlays' interconnection operational procedures

In this section, we describe the operational procedures induced by the newly introduced interconnection roles which are related to self-organization in each overlay (also known as overlay churn). We also detail the information exchange flow between the gateway and the application overlays.

5.2.1. Self-organization procedures

After joining the overlay, an *interconnector* declares its role to the *super interconnector* by sending a SIP INFO message. *Super interconnector* will then respond with a 200 OK message. This is feasible because the *super interconnector* is a bootstrapping node and its address is known by all of the *interconnectors*.

The *interconnector* will then send a SIP SUBSCRIBE message to the *super interconnector* so that the former can dynamically be aware of the set of *interconnectors* in its overlay.

For the sake of simplicity, we assume that all departures from the MANET are voluntary. When an *interconnector* is leaving the MANET, it informs the *super interconnector* by sending an unsubscription message to it. This can be done by deploying the SIP SUBSCRIBE message and setting the *Expires* field to zero.

5.2.2. Information exchange procedures

Our procedures rely on SIP subscription and notification methods. The SIP SUBSCRIBE message for required information is sent by the application overlay (AppC) to the application *super interconnector* (SAPPIC). The SAPPIC relays the request by sending a SIP SUBSCRIBE message to a selected application *interconnector* (APPIC). The request will then be forwarded to the gateway *interconnector* (GWIC). In the gateway overlay, the GWIC sends a SIP SUBSCRIBE message to the gateway *super interconnector* (SGWIC). The SGWIC will then relay the request to one of its subscribed gateway components nodes (GWC). When the subscribed event is triggered, the required information will be retrieved by the gateway and will be sent back to the application overlay as a SIP NOTIFY message, passing through the gateway *super interconnector* (SGWIC), the *interconnector*, application *super interconnector* (SAPPIC) and the destination application node.

6. IMPLEMENTATION

In this section we describe an operational scenario, the software modules implemented to realize the interconnection mechanism and our implemented prototype.

6.1. Operational scenario

This section consists of a basic scenario to illustrate how the proposed two-level overlay architecture is used along with the overlay interconnection mechanism to provide WSN ambient information for enhancing end-user experience of MANET participants.

Our scenario is illustrated in Figure 3. This procedure is triggered when an application end-user requests information that should be provided by the gateway overlay.

After all the involved nodes have joined their corresponding overlays, application and gateway *interconnector* nodes (APPIC and GWIC) declare their roles to their corresponding *super interconnector*s (SAPPIC and SGWIC). They will be now ready to perform information exchange operations. For the specific application of the emergency scenario that we have described in section 2, this can be a simple request, such as subscription for an urgent situation related to the patients, followed by notifications sent whenever the subscribed event is triggered.

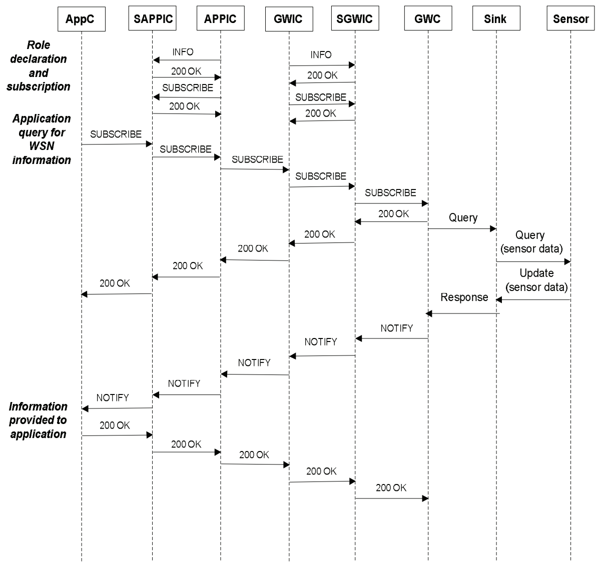

Figure 3: Operational scenario

When a SIP SUBSCRIBE message arrives at the gateway from the application overlay via the defined interconnection path, the GWC (gateway component) will first send a 200 OK response and then perform the appropriate operations on the subscription message, such as translating this message to an understandable message for a sink. The translated subscription message is then transmitted to a sink which then processes the message and sends the appropriate query to the sensors. When the requested data is received from the sensors, the sink node sends it back to the gateway in an appropriate format. The received data will be stored, translated, mapped to a SIP NOTIFY message and relayed to the application via the defined interconnection path. The related 200 OK response will then be sent back by the application overlay to the gateway.

6.2. Software architecture

The software architecture of the *interconnector* is depicted in Figure 4. It consists of five modules and three protocol stacks. The self-organization module is responsible for role declaration of the *interconnector* to the *super interconnector*s and for its departure procedure. The APPIC and GWIC message relay modules deal with transferring messages from one overlay to another. Self-organization and message relay modules provide all the interconnection functionalities using the SIP stack. Application and gateway overlay-specific modules, which use corresponding overlay protocol stacks, provide overlay-specific functionalities.

Figure 5 illustrates the software architecture of an application *super interconnector*, which is very similar to the gateway *super interconnector*. Self-organization, APPIC message relay and application overlay-specific modules are as they were defined earlier. The APPIC selection module is in charge of selecting the appropriate application *interconnector*, based on some defined criteria.

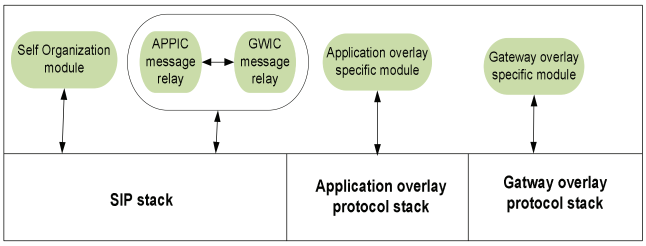

Figure 4: Interconnector software architecture

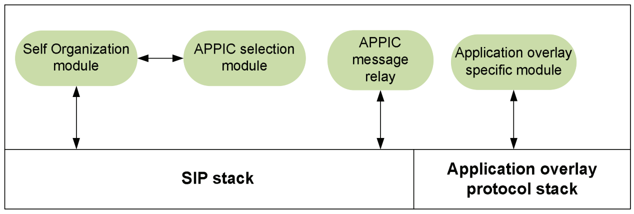

Figure 5: Application super interconnector software architecture

6.3. Prototype

To provide a proof of concept, a subset of the scenario depicted in Figure 3 was implemented. A MANET environment was created, consisting of five nodes (2 laptops and 3 PCs) equipped by 802.11 adaptive cards. Two nodes (AppC and SAPPIC) are joined the application overlay which is a JXTA P2P overlay [12]. In a similar way, GWC and SGWIC nodes join the gateway overlay, which is an Open Chord P2P overlay [13]. The fifth node is the *interconnector* between the application and gateway overlays. The interconnection is then established between the application and gateway overlays to provide the end-user service, which is a text notification sent to the application end-user whenever a subscribed event is triggered.

We implemented a subset of *interconnector* and *super interconnector* nodes as well as two nodes in different overlays that do not play any interconnection role but that wish to intercommunicate. We deployed and tested our implementation on aforementioned nodes and observed that our architecture is working properly. JXTA and Open Chord are two existing P2P middleware that provide a generic framework and a set of building blocks to develop P2P applications, so that these applications do not have to be built from scratch. We used JAIN SIP [14], a standardized Java based interface to SIP protocol stack, to provide SIP UA required functionalities that should be supported by all the defined nodes in each overlay.

7. RELATED WORK

In [15], the authors present an architecture for the deployment of small-scale sensor networks, consisting of P2P ad hoc mobile sensors and gateways in the 3G mobile environment. Sensors are carried by the operators and gateway functionality is developed on mobile phones or PDA devices. This type of gateway supports almost all our defined gateway-specific requirements, such as mobility, processing, synchronous and asynchronous communication. However, the gateway in this system is centralized and there is no support for self-organization, scalability and fault tolerance. Furthermore, it does not address the application overlay.

The gateway architecture proposed in [6] and which we discussed earlier in this paper, is a P2P overlay for the integration of mobile sink-based WSNs with IMS. IMS is a global and standardized architecture which enables connectivity and service control functionalities for the 3G networks based on the Internet protocols [16]. The gateway overlay is built on top of the same mobile phones that also act as mobile sinks, and end-user devices. This gateway meets all our gateway-specific and many of our general requirements but it assumes that the fixed application users reside at 3G network.

DUMBONET [17] is a hybrid combination of MANETs, satellite IP network, and the Internet which aims at providing a collaborative emergency response system used for mass causality events. This work describes the design and proof of concept for the proposed MANET and application platform. The application motivation in DUMBONET is similar to one of our motivating scenarios in which medical paramedics are provided with ad-hoc mobile devices in order to interact among themselves, along with remote communication with the specialists at hospital. This system can be combined with physiological and environmental sensor networks that provide useful information as required by the emergency staff. However, this work does not provide any explanation on how the sensors' data is transferred to the application end-user and it is also based on the lower layer protocols.

8. CONCLUSION

In this paper we proposed an overall two-level overlay architecture to integrate WSNs (with mobile and distributed gateways) and MANETs, thus enhancing the end-user experience of MANET participants. A scenario based on emergency conditions was presented to motivate such situations. Mobile gateway and application nodes can run on end-user ad-hoc devices and form respective overlays on top of a MANET. We derived specific requirements for the interconnection of gateway and application overlays and proposed an architecture to interconnect these two overlays of the overall architecture. Protocols and operational procedures for self-organization and information exchange between these two overlays were presented.

As a proof of concept, we implemented a prototype for event notification using JXTA as an application overlay and Open Chord as a gateway overlay. We used JAIN SIP to provide intercommunication between overlays, based on SIP protocol.

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REFERENCES

[1] I. F. Akyildiz,W. Su, Y. Sankarasubramaniam and E. Cayirci, "Wireless Sensor Networks: A Survey," *IEEE Comm. Mag*, August 2002.

[2] M. El Barachi, A. Kadiwal, R. Glitho, F. Khendek, and R. Dssouli, "The Design and Implementation of a Gateway for IP Multimedia Subsystem/Wireless Sensors Networks Interworking," *IEEE Vehicular Technology Conference*, Barcelona, Spain, April 2009.

[3] J. Liu and I. Chlamtac, "Mobile Ad-Hoc Networking with a View of 4G Wireless: Imperatives and Challenges," *Mobile Ad Hoc Networking*, chapter 1, Wiley-IEEE Press, July 2004.

[4] K. Lua, J. Crowcroft, M. Pias, R. Sharma, and S. Lim, "A survey and comparison of peer-to-peer overlay network schemes," *Communications Surveys & Tutorials, IEEE*, pp. 72-93, 2005.

[5] B. H. Thomas, G. Quirchmayr, and W. Piekarski, "Through walls communication for medical emergency services", *International Journal of Human-Computer Interaction,* 16:3, pp. 477 – 496, July 2010

[6] M. Velez Pulgarin, R. Glitho, and A. Quintero, "An Overlay Gateway for the Integration of IP Multimedia Subsystem and Mobile Sink Based Wireless Sensor Networks," *IEEE Vehicular Technology Conference (VTC))*, Ottawa, Canada, Fall 2010.

[7] C. Fu, R. Glitho, and F. Khendek, "Signaling for Conferencing in Integrated 3G/Mobile Ad Hoc Networks," *iscc, 11th IEEE Symposium on Computers and Communications*, pp. 838-843, 2006.

[8] D. Ben Kheder, R. Glitho, and R. Dssouli, "Media Handling Aspects of Conferencing in Broadband Wireless Ad Hoc Networks," *IEEE Network*, pp. 42-49, March/April 2006.

[9] RFC 3261, "Session Initiation Protocol (SIP)," http://www.ietf.org/rfc/rfc3261.txt, June 2002.

[10] RFC 2976, "The SIP INFO Method," http://www.ietf.org/rfc/rfc2976.txt, October 2000.

[11] RFC 3265, "Session Initiation Protocol (SIP)-Specific Event Notification," http://www.ietf.org/rfc/rfc3265.txt, June 2002.

[12] JXTA, http:// www.jxta.org

[13] Open chord, http:// open-chord.sourceforge.net

[14] JAIN SIP, https://jain-sip.dev.java.net/va.net

[15] S. Krco, D. Cleary, and D. Parker, "P2P Mobile Sensor Networks," In *Proceedings of the Proceedings of the 38th Annual Hawaii international Conference on System Sciences,* vol.09, 2005.

[16] G. Camarillo and M. Garcia-Martin, "The 3G IP Multimedia Subsystem (IMS): Merging the Internet and the Cellular Worlds," Chapter 3, 2004.

[17] K. Kanchanasut, T. Wongsaardsakul, M. Chansutthirangkool, A. Laouiti, H. Tazaki, and K. R. Arefin. "DUMBO II: a V-2-I emergency network," *In Proceedings of the 4th Asian Conference on internet Engineering*, Pratunam, Bangkok, Thailand, 2008.

TELECOMMUNICATIONS BUSINESS MODEL FOR CONVERGED NETWORKS FOCUSING FINAL USERS

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ABSTRACT

The telecommunications converged networks, specially inside of new Internet environment, makes possible supply a plenty of services to users, as voice and multimedia services with assured perceived quality of services (QoS) through different access technologies, however this brings a complex scenario in terms of technology in order to fit the user perceived QoS needs to appropriate technical QoS requirements, and also delivery the right service to a particular user. Therefore, to enable the appropriate service delivery for each user, this article proposes a business model for the telecommunications segment, aiming delivery services according to particular User Service Level Agreements (USLAs), prepared transparently of the technologies involved, and using QoS parameters according to the users literacy. The proposed business model considers four providers: Services, Infrastructure, Content and Access aiming to facilitate the relationship between users and providers, and to clarify the roles and responsibilities of each actor, as well. This paper presents the proposed business model, and discuss the needs for an regulatory frameworks necessary to meet the requirements of proposed business model with focus on users.

Keywords— Business Model, Converged Networks, Regulatory, QoS, Final User.

1. OBJECTIVE

A converged network allows to offer interactive mobile services with multimedia content to users independently of specific technologies or providers. But the most common business model adopted today considers that the user needs to choose appropriate service provider for each part of service, i.e. for a complete service the user usually needs to contract an access provider, a mobile service provider and a content provider at least, so user must manage more than one contract and more than one bill, and manages specific QoS issues with each provider, having a lack of general responsibility of service delivery as a whole. More than that particular perceived QoS for each user is impossible to be stated on the contracts, causing misunderstandings and difficulties on users providers relationship. One of main issues is how to face the user lack of technical literacy for preparing a contract (USLA) understandable for both sides that reflects user's wishes and the technical requirements visible for access, content infrastructure or service providers. The main purpose of this paper is to propose a business model focusing users of the services, thus enabling him to understand the service purchased without worrying with involved technology, specially the expected QoS. The paper presents the adequacy of business processes of each kind of provider involved in service providing that makes up the proposed business model, and changes on regulatory frameworks. The paper also presents the adherence of proposed business model to eTOM (Enhanced Telecom Operations Map) and integration of data architecture aiming the implementation of business model in a telecommunication provider.

2. INTRODUCTION

The main focus of our society is information, available under format of audio, data, images and video; and increasingly, the users want to obtain, to supply, to share and to interact with information using interactive mobile services with multimedia content in any place, any time, using any device, and on move. This scenario has created new opportunities to offer new services by providers that compose the segment of telecommunications businesses.

Next generation networks (NGN) provides several services, and enables different access technologies can be chosen by the users, and according their necessities the features of interactivity, mobility and quality of service (QoS) must be considered. The evolution in wireless network devices, allowing the transmission of data, audio, image and video, including broadcasting over several wireless access technologies, in order to support interactive mobile applications with multimedia content. However, choice of technology and QoS (in a technical point of view) provided by each services offered by traditional telecommunications service provider are beyond of understanding of user, considering the average users' technical literacy. [1] [2].

Traditional telecommunications service providers usually do not offer all possible access technologies and QoS that fits with the desired QoS as perceived by the particular user. The proposal is to create a User Service Level Agreement (USLA) to regulate the relationship between provider and user, where the service is well described and the QoS as well, the point is: the descriptions of the service and QoS are made according user's literacy. So QoS requirements are ones related to the user's perceived QoS for particular service being contracted. USLA would not include technical requirements that cannot be easily measured by user, for example, bandwidth, or access technology to be used [3].

By other hand, current telecommunications business model does not allow users to receive services from several providers at same terminal, when it is possible users must manage the services in order to determine which one is most adequate to him at each time.

To face this issue and considering the current environment of telecommunications business segment, the new demand requested by users, and new scenario that has been built by Internet of the Future concepts, some features must be handled carefully by the current telecommunications services providers:

- Provision of services: services must be offered to users, independently of access technology and providers;
- Assured quality: services should assure the contracted QoS, independently of the technology, QoS requirements must be agreed and understood by users, and services must fit the cost/benefit relation defined by users. This means services must be managed by providers considering each user individually;
- Optimizing the use of the network: The services must be provided to users with the technology that meets the telecommunications services providers' cost-benefit relation;
- Focus on business: Telecommunications services providers must be specialized and make investments in their core business.

Therefore, current environment is propitious to discuss and to deploy new business models for telecommunications segment, aiming to meet the new characteristics of the market, and enable the provision of services focusing the user and not in the technology, hurrying up the adaptation for new scenario provided by the Internet of the Future [1] [2].

Today, in general way, mobile interactive services are provided by telecommunications services providers, and they cannot guarantee the service as a whole, even if the service level agreement (SLA), because, as the service is composed by multimedia content, telecommunication services (access, backbone), and user's devices, at least, the telecommunications services providers cannot interfere in content or in the user's device, and content providers are not liable to providing access. The result is the user needs to track, analyze and verify the performance of each provider, and always it is very difficult to identify the problem and correctly penalize the service provider according SLA. The proposed solution is to create a unique interface between user and providers managed by specifics USLAs.

3. PROPOSED BUSINESS MODEL

A business model based on convergent networks, which organize the relationship between service providers and users, and among providers responsible to implement the whole service, and establish a clear responsibility over the whole service under user's view with a single interface and agreement is wished. This business model must be adherent to NGN and Internet of the Future concepts. [6][7].

Proposed business model, divides the telecommunications segment in four main providers, and each provider represents a portion of the value chain, as depicted in figure 1. The hypothesis is that each enterprise focuses and specializes on a single activity that compounds the whole service to be provided in accordance with USLA, but nothing prevents an enterprise provides services in more than one area.

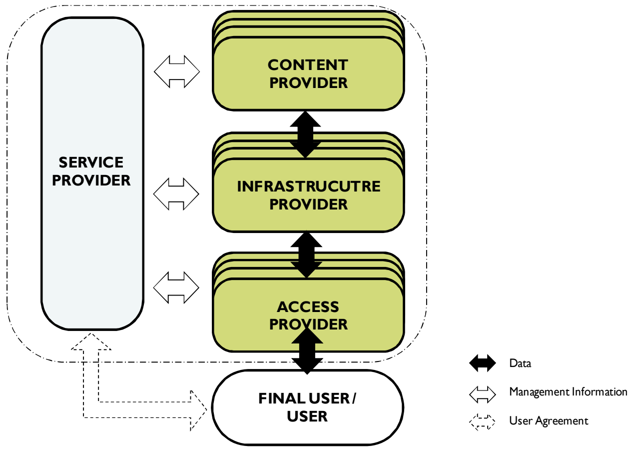

Fig. 1 Proposed business model

3.1. Service Provider

Service Provider (SP) is responsible for commercialization of services to user, whose goal is to offer services within the necessity, quality and cost-benefit expected by user. SP should be carry out services level agreements with other providers – Provider Service Agreements (PSLA) – involved in service providing, aiming assure the QoS. And especially monitors and manages information related to the actual QoS parameters values to decide on "real time" the best route for each contracted service by the individual user.

The service can be delivered to user by the SP through several providers, but the SP is responsible to assure the perceived QoS as stated on USLA.

Is important notice that SP is the unique contractual interface between user and the services providing mechanism and the hole of SP is manage the contracts, using the USLAs and PSLAs.

3.2. Access Provider

Access Provider (AP) is responsible for providing the last mile to user. The last mile should be implemented using wireless or wired communications, but today, as mobility is mandatory, wireless technologies are focused in this paper.

For user point of view, the services can be received from several APs, depending on SP choice, that is based on the actual QoS parameters values. The handover must be transparent to user.

3.3. Infrastructure Provider

Infrastructure Provider (INP) is responsible for providing all necessary communications infrastructure between Content Providers and APs. The SP can use several INPs to implement the communication needed for delivering services, and can promote handover procedures, in order to guarantee QoS requirements stated on USLAs, and cost/benefit relation. Again the handover must be transparent to users.

3.4. Content Provider

Content Provider (CP) is responsible for providing the content that compose the service contracted by users, and contrasting with Access and Infrastructure Providers that use technologies related directly to the telecommunications segment, the CP uses a large variety of technologies in their business not directly inside of telecommunication field. Currently CPs are represented by banks, schools, video producers, games organizations among others. Notice that as content is an important part of users' perceived QoS, so SP must monitor the content quality in order to assure the QoS requirements stated on USLA, and SP can also promote a handover of CPs in case of risk of QoS lost.

3.5. Service Level Agreements

SP has the whole responsibility for delivering the services, and has a formal contract with users represented by an USLA (dashed arrow in figure 1), that regulates the service and user's perceived QoS whished.

By the other side, as the SP needs content, telecommunication infrastructure and last mile to compound and provide the services, he must contract Access, Infrastructure and Content Providers to proceed, these contracts are represented by PSLAs that states QoS requirements, services descriptions and prices. Empty arrows in figure 1 represents the PSLAs and QoS data used on the monitoring task performed by SP. [2][3].

3.6. Horizontal and Vertical Handover

The SPs need to consider redundancies in content, infrastructure and access services, in order to guarantee USLAs in any case, lowering the risks in their businesses, the redundancies are managed monitoring the QoS parameters data from each active or inactive partner provider and triggering a vertical or horizontal handover when one of the QoS parameters values reach to a critical point. Handover refers to the process of transferring a service session from one channel to another without loss of QoS or interruption of service, transparently to user. Proposed business model considers handover inside of providers and between providers of same layer [8] [9].

3.7. Data Integration

In order to realize handover it is necessary to receive QoS parameters values and status information from each equipment of each provider in use or not, that can compose the services being delivered to users. The architecture proposed in this paper to implement the data exchange uses a middleware and connectors between providers as depicted in figure 2.

Beyond status information, the providers exchange service orders to request network configuration needed to guarantee each specific USLA. After receive service orders, the providers analyzes the status of their resources and verify possibility to accept the request, it is necessary to consider PSLA agreed with Service Provider as well.

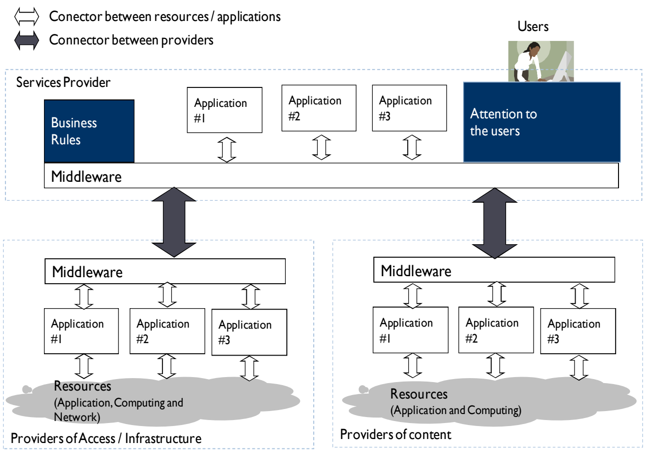

Fig. 2 Providers integration of data

Referring to figure 2, providers' data integration architecture considers five following main components:

- Middleware: It is responsible for integration of internal applications running in each provider. Middleware allows different applications to exchange information between them and with workflow engine ensuring that right information will be available to the application on appropriated time;
- Connector: is responsible for standardizing the information exchange between middleware and applications, and also between middlewares. The connectors allow the middleware of each provider to receive information of each resources / applications in a standardized way. Connectors between middlewares aim standardization of

communication between middlewares. With the connectors standardization a new provider can be easily incorporated to supply chain of services, this feature is desirable considering that proposed business model requires a seamless integration between providers;

- Applications: are software components that perform specific functions in architecture, such as fault, performance, service order, network & services inventory management, among others. Following, the Attention to the User and Business Rule applications are highlighted due to their importance to proposed business model:
- Attention to the User Application: it is the primary responsible for providing information to users regarding to the services supplied by SP. As unique responsible for managing USLAs, the SP has in this application all the information related with USLA monitored in each provider aiming to control the QoS of each particular service;
- Business Rule Application: it implements the SPs business rules, for example: rule based decision machine, fuzzy logic decision machine, among others, it is used for ensuring the profit of services, because through this application will be decided which provider should be changed in handover procedures, and it depends of PSLAS and particular USLA.

The architecture can be implemented using open system middleware, like J2EE platform, which allows for greater interoperability between providers allowing SP to get the best decision to meet USLA.

3.8. Business Processes for Each Provider

The implementation of proposed business model needs a small adequacy of eTOM model proposed by TMF (TeleManagement Forum). The eTOM [4][5], shown in figure 3, offers a general business processes model that describes the best practices for providing telecommunications services applied to providers, and adopted by ITU-T (International Telecommunication Union) in recommendation M.3050.

The eTOM, as stated, is composed by three processes groups:

- Enterprise management group: It consists of activities relating to the strategic and enterprise planning, enterprise risk management, human resources management, financial and asset management, enterprise effectiveness management, knowledge and research management and stakeholder and external relations management;
- Strategy, Infrastructure & Product group: It aims to achieve the management of the life cycle of products and infrastructure, aligned with the strategic needs of the enterprise;
- Operations group: It consists of processes relating to the operation and maintenance services offered by the provider, treating the needs for support, configuration, service assurance and billing.

Besides these three groups, eTOM also has horizontal and vertical levels.

The horizontal levels are composed by groups of processes that represent the functionality vision of business processes, and define area responsible for creating the capacity to implement, support and automate these processes. The vertical levels represent end to end processes required for delivering services applied to business of telecommunications services providers.

This paper proposes the creation of four providers to comply the proposed business model to eTOM, aiming the information exchange regarding to status of equipment and services in a transparent manner.

The figure 4 shows a general vision of eTOM applied to business model proposed in this paper.

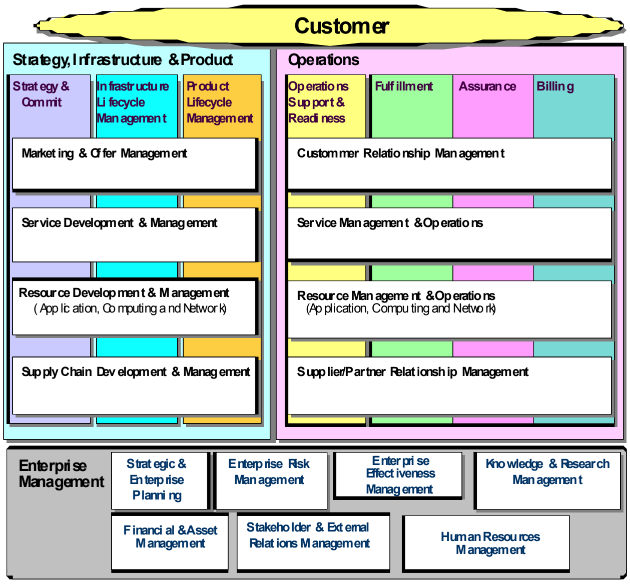

Fig. 3 eTOM [4]

In figure 4 can be noticed that Access, Infrastructure and Content providers have contract with SP, so the SP behaves as user of the services provided by them, and they do not need to have direct access to users, by the other side, the SPs have direct interface with the user.

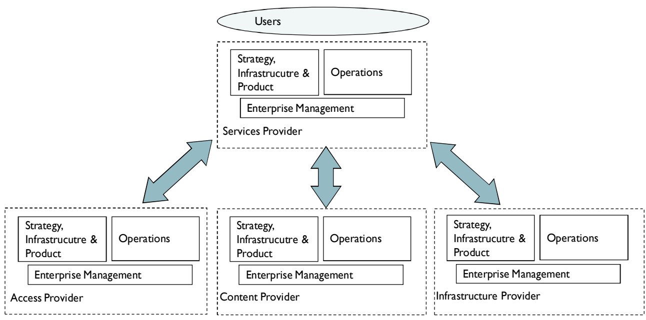

Fig. 4 General vision of new model of eTOM

These means that providers have different operations in terms of eTOM application, as follows:

- SP does not have any kind of telecommunication resources (applications, computing and network), and its operation is focused on activities of management services, and agreements (USLA and PSLA), services order management, CRM (Customer Relationship Management) and billing. So, the eTOM application for SPs is focused on the customer, services, and partners (providers), see figure 5. The SPs does not perform any activity relating to resource management & operations and resource development & management. But at the processes management of brand & advertising become an item of extreme importance, due to direct contact with user. In simplified way, SPs will have their professionals focused on sales & marketing areas, and in activities to create and manage new services.

- Access, Infrastructure and Content providers, by their side, have all telecommunications resources (applications, computing and network) to be managed and kept in perfect working status in order to the agreed upon PSLAs. The activities of these providers are directly related to telecommunication resources, such as fault and performance management, billing data collection, network availability and resource configuration. So, considering that in the proposed business model these providers do not have relationship with users, they cannot worry about marketing & offer management, customer relationship management, service development & management and service management & operations, focusing user, see figure 6.

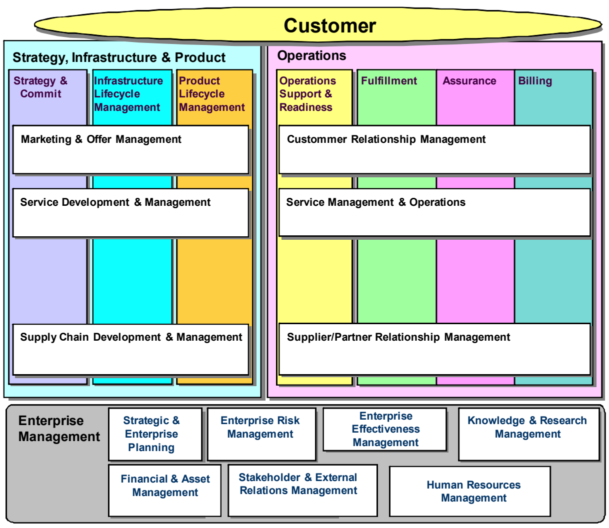

Fig. 5 eTOM for Service Providers

The Access, Infrastructure and Content providers have their professionals focused on technical areas and maintenance of resources.

Considering the specialization of each provider, operational activities undertaken by each provider becomes more focused, unlike what is currently used, in the approach proposed by this paper, each provider should dedicate at main activities and, consequently, can reduce operational costs and improve performance. Moreover, the simplification of enterprises of telecommunications segment will allow better management of their business processes and vision of real problems of enterprise.

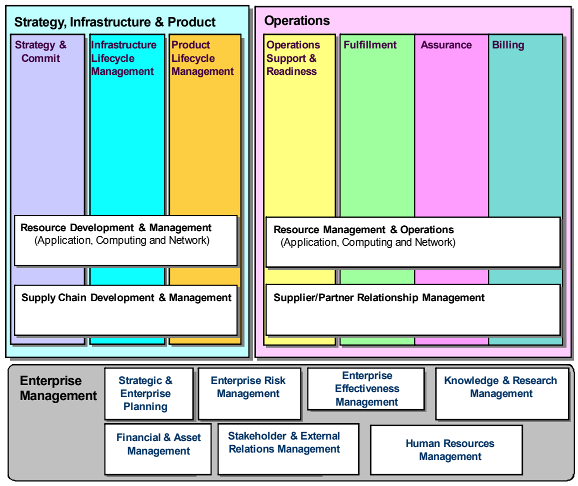

Fig. 6 eTOM for Access, Infrastructure and Content providers

4. EXAMPLE OF WORKFLOW ON PROPOSED BUSINESS MODEL

This section presents an example aiming to make clear the understanding of workflow between the providers proposed on business model. According to Figure 7, the information is transmitted from CP to user device (terminal) through Infrastructure and Access providers, without passing through SP. To ensure the USLA, the SP collects information about equipment status (QoS parameters values) to be aware of the current situation of services in each provider and at user terminal.

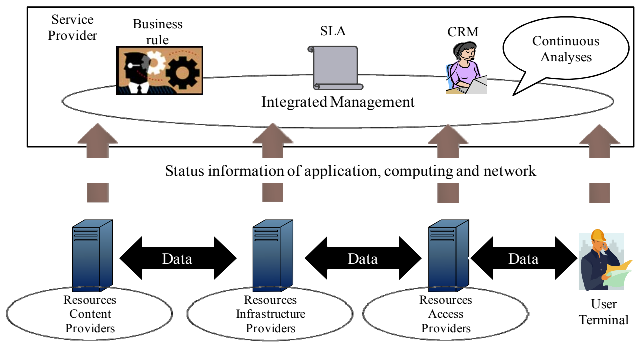

Fig. 7 Systemic view of proposed business model

The status information of each equipment and service allows SP to know the current situation and predict future situation of services delivered to user, to select providers that better comply the cost-benefit relation for providing service according specific USLA agreed with specific user. SP has an integrated management system that receives the status of equipment and services of all providers that have contracts (PLSAs) with, to allow the selection of set of providers that better meet the needs of USLA.

The SP should integrate all providers' necessities for provision of services contracted by user, then it is necessary the integration of the Operation Support Systems (OSS) and the Business Support Systems (BSS) of each provider to be integrated, and exchanging at least the following information [10][12]:

- Fault events of equipment, servers and applications;
- Performance data of equipment, servers and applications;
- Data regarding to availability of equipment, servers and applications;
- Data regarding to billing of the product set;
- Transmission and reception of service orders for equipments' configuration, servers and applications.

The information exchange mentioned above will permit SP to realize the following activities [11][13][14]:

- QoS evaluation and prediction for services delivered or to be delivered to final, using data related to fault and performance of each provider, determining whether the services is meeting or not each particular USLA;
- If necessary to change the configuration of services, SP will evaluate availability of providers, that it has PSLAs, in accordance with available equipments, servers and/or applications, and send services orders to perform an appropriated configuration, triggering the beginning of service delivery or handover process;
- Through billing information will be selected the better cost-effective solution, if the values achieved are not agreed on respective PSLA.

To illustrate the above, consider that SP is supplying a service to the user, according to the USLA signed, and the service was configured as shown in figure 8. The user does need to know the configuration of contracted services, this must be transparent to him.

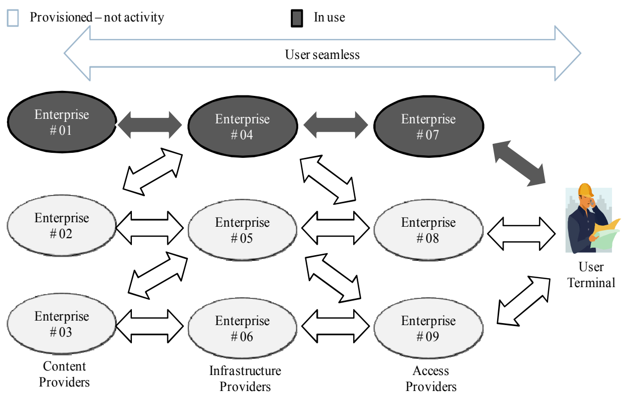

Fig. 8 Example of configured service

The service is being provided through the enterprise # 01 (CP), #04 (INP) and #07 (AP). Now consider a fault in the content server of the enterprise #01, event 1 shown in figure 9.

The SP receives the fault information of enterprise #01 and through Management System performs the analysis of impact (event 2 of figure 4) based on QoS parameters values, business rules, network situation in each providers, PSLA and USLA, and may require a reconfiguration of providers through service orders (event 3 of figure 4), to remain supplying the service on a continuous basis for user according to USLA. As mentioned, the new provider, chosen by the Management System, can refuse the service order, in this case, new analyses will be performed and another provider will receive the service order.

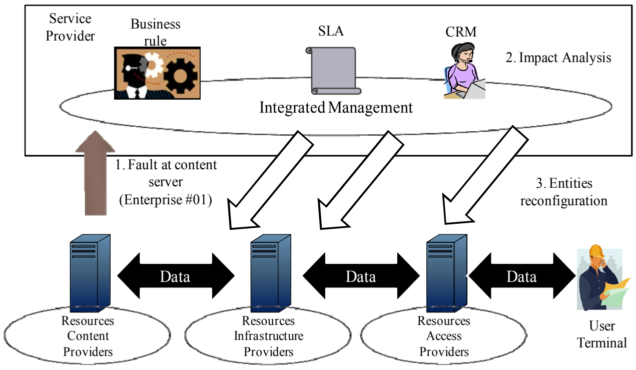

Fig. 9 Analyze at Service Provider

After the reconfiguration of providers (handover), the final configuration is shown in figure 10, however for the user this reconfiguration task must be transparent and cause no impact on the services that has been supplied.

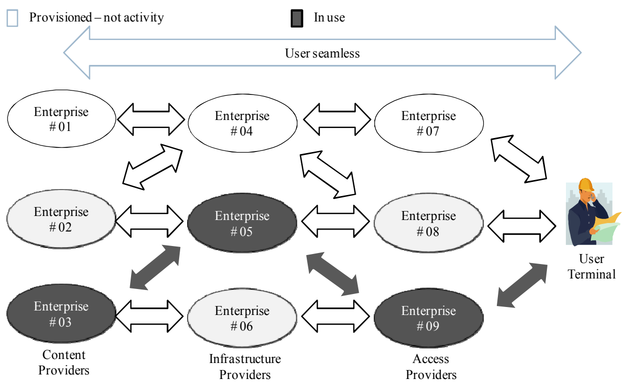

Fig. 10 Example of final configuration

At final configuration the service is supplied by enterprises #03, #05 and #09, in this example the SP changed all providers.

5. LEGISLATIVE AND REGULATORY ISSUES

This paper proposes a business model for telecommunications segment where the focus is the user and the SP is responsible to choose the adequate services to compound the service to be delivered to user.

The proposed business model allow the user to have a desired service independent of Access, Infrastructure or Content providers, and mainly to have the adequate services configuration for each particular situation. The business model permits the user to have a unique interface with providers, because all negotiation will be realized through SP.

Actually, in majority of countries, the telecommunications segment does not have a business model like the proposed one, the most used one is to have in same enterprise all technologies necessaries to provide the service to user, but the services do not support handover of providers.

In order to support proposed business model will be necessary to revise legislative and regulatory framework, not only related to technical issues, but also to political issues.

At technical point of view is necessary to revise at least:

- Revision on business process map (eTOM) of telecommunications segments;
- Definition of interface contract between the providers for data integration;
- Definition of requisites and parameter for USLA and PSLA;
- Definition of mechanisms to realize handover between enterprises.

At political point of view is necessary to revise at least:

- Operations license emitted by regulatory agencies to providers;
- Regulate the operations realized by Content Providers;
- Create the figure of Service Provider.

6. CONCLUSIONS

This article proposed a business model for telecommunications segment, and proposed a change in current business processes model proposed by TMF in eTOM, in order to allow the implementation of proposed business model, also present an open system architecture that permit providers integration. Proposed business model allows greater freedom for user of telecommunications services, thus enabling it to decide the service without having to evaluate the technology that provides the service. This specialization allows that each provider focus on your core business and has a better cost-benefit.

Open system architecture allows cheaper development and greater interoperability between providers, and standardize the communication between them, allowing providers based anywhere in the world to participate as a provider.

The benefits of proposed user oriented business model instead of technology oriented business are:

- User does not need to know about technologies that meet his necessities, respecting him literacy;
- User needs only to know his necessities in terms of services;
- User can manage the SLA;
- Provide more effective services to user;
- SP can use better resource in each situation;
- SP could supply new services with minimum investment;
- All providers can focus in their specialty;
- It is fully compatible with Internet of Future concepts.

6.1. Contribution

The article presents a proposed business model, which enables the converging heterogeneous networks, facilitating absorption of new technologies by user in general. It simplifies the choice of user, but without losing the characteristic of flexibility and convergence of new networks, as stated in NGN. The proposed discussion starts the design of a new business model through adaptation of business processes proposed by the TMF.

6.2. Implementation

The LSA (Open System Laboratory) of Computing and Digital Systems Department of Escola Politecnica of Universidade de São Paulo is implementing the proposal presented in this article. At current phase of project, the system for managing the workforce for the contracting of services has been developed, in order to manage order of service to be sent to Access, Infrastructure and Content providers. The architecture is being developed on J2EE platform and is focused to meet effectively the user requirements. The first phase has been funded by Projects and Studies Brazilian Funding Agency FINEP (contract 0108051900) and by Thales Information Systems a Brazilian subsidiary of Thales, that is a global technology leader for aerospace, space, defense, security markets.

6. REFERENCES

[1] D. Karam, “Modelo de negócio para mobilidade e interatividade em ambientes convergentes heterogeneos”, Thesis, Escola Politécnica da Universidade de São Paulo, 80p, 2006.

[2] A.P.G. Serra, “Método para identificação de parâmetros de qualidade de serviços aplicados a serviços móveis e interativos”, Thesis, Escola Politécnica da Universidade de São Paulo, 120p, 2007.

[3] Telemanagement Forum, “GB917 SLA Management Handbook Release 2.5”, Telemanagemen Forum, 2005.

[4] Telemanagement Forum, “GB921 Business Process Framework Release 8.0”, Telemanagement Forum, 2008.

[5] Telemanagement Forum, “TMF519 Service Delivery Framework Business Agreement Release 1.1”, Telemanagement Forum, 2009.

[6] International Telecommunication Union, “Y.2001 General Overview of NGN”, International Telecommunication Union, 18p, 2004.

[7] International Telecommunication Union, “Y.2173 Management of Performance Measurement for NGN”, International Telecommunication Union, 2008.

[8] H. Attaullah, F. Iqbal, Y. Muhammad, "Intelligent vertical handover decision model to improve QoS", IEEE

Third International conference on Digital Information Management, pp. 119-124, 2008.

[9] Y. Hwang, A. Park, "Vertical handover platform over applying the open API for WLAN and 3G LTE systems", IEEE 68th Vehicular Technology Conference, pp. 1-5, 2008.

[10] P. Bellavista, "Middleware for next-generation converged networks and services: myths or reality?", IEEE 31st annual international conference on computer software and applications, vol. 1, pp. 24-27, 2007.

[11] Y. Gorur, "Converged Network Management: challenges and solutions", IEEE Optical fiber communication conference, 10p, 2006.

[12] M. Ajansa, M. Boulmalf, H. Harroud, H. Hamam, “A Policy Based Event Management Middleware for Implementing RFID Applications”, IEEE International Conference on Wireless and Mobile Computing WIMOB 2009, Networking and Communications, p. 406-410, 2009.

[13] L. Seung, G. Jee, K. Minho, “Performance Enhancement in Future PON and Mobile Convergence Networks”, ICACT 2009 11th International Conferece on Advanced Communication Technology, p. 233-236, 2009.

[14] Z. Zhenyu, X. Xiaoyaho, “Research the Effectiveness of Neural Network for Telecom Planning Prediction”, ICIEA 2008 3rd IEEE Conference on Industrial Electronics and Applications, p. 56-60, 2008.

ON DEMAND FINE GRAIN RESOURCE MONITORING SYSTEM FOR SERVER CONSOLIDATION

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ABSTRACT

Server consolidation and virtualization technology are used in modern data center as a method to improve the server utilization rate by encouraging the sharing of physical resources between virtual machines (VM). Although, this can be realized by plenty of virtualization softwares available in markets, creating a good management policy for the server consolidation environment is still a challenging research problem. The policy should be able to maintain the high resource sharing rate, while keeping the performance drop rate as low as possible, which are totally contradict requirements. In order to progress the research in this area, a monitoring system, which can effectively monitor current system state and collect essential information, is indispensable. However, due to many differences in use case scenarios, monitoring systems designed for conventional management domain cannot be applied directly. This needs us to reconsider the design of the monitoring system. This paper proposes some key information that is still missing from existing monitoring systems as well as the design of Pantau, a monitoring system designed for capturing information necessary for server consolidation.

Index Terms— Server Consolidation, Measurement, Monitoring System, Distributed System

1. INTRODUCTION

In order to maintain good quality of service, most servers in today's data center are being over provisioned to accommodate peak workload which results in low server utilization rate for the rest of the time. Many analyst firms estimate that resource utilization of 15 to 20 percent is common[1]. Server consolidation is an effective way to increase server utilization rate by encouraging the sharing of physical resources between VMs.This can be realized using various virtualization software, including VMWare[2], Xen[3], OpenVZ[4], and KVM[5]. Figure 1 and figure 2 demonstrate how server consolidation technique can increase the server utilization rate. The number in the star represents CPU usage on each machine. The icon at the bottom of the server's icon represents an application running on that server.

Some researchers[6] take this idea to even further, by trying to create standard interfaces between these data centers, so that all data centers can be connected to create a virtually unlimited computing resource pool where resources for a VM can be provisioned from any physical machines. We argue that just connection interfaces are not enough to connect those data centers together. In order to maintain the service quality while migrating VMs to run on different data centers or even on different physical machines in the same data center, we need to understand thoroughly the performance behaviour of the VMs in these systems. This is a notoriously difficult task because the performance of the VM depends on a lot of factors such as the overhead caused by virtualization software and the resources contention between other VMs residing on the same physical machine. In order to overcome this challenge, a monitoring system that can capture us the necessary data is indispensable.

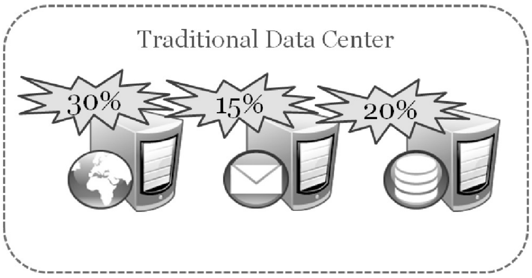

Fig. 1. Traditional Data Center

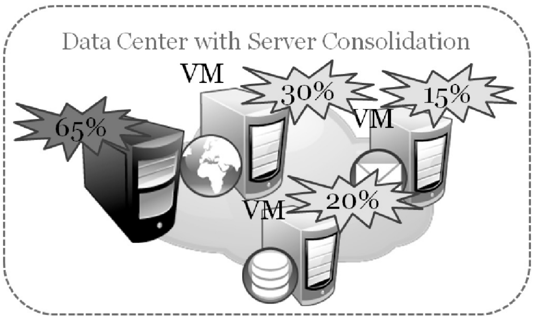

Fig. 2. Data Center with Server Consolidation

Several monitoring systems[7, 8, 9, 10, 11, 12] have been proposed so far. However, those systems could not be applied directly to monitor the server consolidation environment due to the fact that virtualization technology allows multiple virtual machines to run on the same physical machine. Therefore, it is inevitable that the machines affect one another in some ways. Thus, monitoring on resource usage of each machine independently is not sufficient to understand the characteristic of the system. Moreover, most of conventional monitoring systems do not capture resource usage data at a fine grain time interval which makes the captured data insufficient for making extensive analysis. This encourage us to design yet another monitoring system. For the rest of our paper, we will refer to conventional monitoring systems as monitoring systems designed for management domain and refer to our proposed system as a monitoring system designed for research domain. The main differences between these domains are the amount of data and the accuracy of data which users expect from the monitoring systems. Specifically speaking, in the management domain users usually need the systems to capture as many information as possible, so that if any problems occur they can investigate from several dimensions. On the other hand in research domain, users usually know which resource usage data they are currently interested in, and they expect that data to be captured as accurate as possible.

This paper proposes some key information that is still missing from existing monitoring systems as well as the design of Pantau the monitoring system designed to serve the purposes in server consolidation research. The rest of the paper is organized as follows. Next section, we discuss some of existing resource monitoring systems and point out why we think they do not suit for monitoring the server consolidation environment. Then we discuss common management tasks and the challenges in designing our system in section 3. In section 4, we describe the design decision and the implementation of Pantau, the monitoring system that realizes the requirements proposed in this paper. And then we conclude this paper in section 5

2. RELATED WORK

The Simple Network Management Protocol (SNMP)[13] is widely used and supported by many monitoring softwares. It provides an open protocol format that can be used to monitor a variety of different types of equipments, using standard management information base (MIB). This MIB provides monitoring systems with information specific to each hardware device. However, it does incur some amount of overhead due to being designed as a general purpose protocol. Some monitoring systems thus choose to rely on their proprietary protocol that incurs less overhead, but yet still meeting their needs.

There are a number of research efforts focusing on proposing resource monitoring systems. One of the most widely used in cluster monitoring is Ganglia[7]. It is scalable distributed monitoring system and is in use on over 500 clusters around the world. Ganglia makes use of hybrid approach monitoring which inherits the desirable properties of listen/announce protocols including automatic discovery of cluster membership, while at the same time still permitting federation in a hierarchical manner.

In cluster environment where one physical machine can host applications from different users, it might be possible that a well-behaved application suffers from the poorly-behaved one. To allow the entire community to be able to spot the problems on nodes while trying to preserve user's privacy, CoMon[11] introduces the concept of slice centric information. The idea of this concept is to provide a user with the aggregated resource usage information of other users without providing access to the information on a per-process basis. It relies on two types of daemon, slice centric and node centric daemon, to gather the resource usage data every 5 minutes. The slice centric daemon reports the aggregated resources used by all processes within each slice. The node centric daemon on the other hand reports over 57 values of node specific resource usage information.

Several network monitoring tools such as Zenoss[8], Nagios[9] and Munin[10] are based on manager-client architecture. Compared with hierarchical architecture, this architecture has lower scalability; however the advantage of this architecture is its simplicity. The manager process is deployed on a monitoring server, while a client process acts as a data gathering agent is deployed to every node. The manager process polls each agent for data at specific interval. These monitoring tools can be extended easily using plug-ins which can be any kinds of executable scripts that can output data in a predefined format.

Despite being able to provide us with useful information, these systems usually work at coarse time interval such as 5 minutes interval. This is because they were initially designed to support the work in management domain which requires them to keep track of a large set of resource usage data over a long period. We argue that this coarse grain monitoring is not sufficient for research purpose. Kamoshida[12] proposed a method to achieve realtime monitoring by trying to reduce the communication cost. The main idea is to send only significant updates to the manager. Although this method can reduce the workload dramatically, tick by tick update is lost. Our work achieves fine grain monitoring without sacrificing tick by tick update using a much simpler method that makes use of the fact that most research works focus on a relatively small set of data at any specific time. In addition to the coarse time interval, none of above monitoring tools provides us with convenient methods to visualize the correlation between the resource usage of related machines nor does it provides any methods to visualize the correlation between application performance parameters and corresponding resource usage data.

3. MONITORING SYSTEM FOR SERVER CONSOLIDATION ENVIRONMENT

In this section, we discuss the requirements and challenges for our monitoring system. Section 3.1 shows examples of fundamental management tasks that arise in this environment. These management tasks are still unsolved research questions that need additional information to guide the research direction. We are trying to provide an easy to use tool to help researchers in this area. Section 3.2 explains why these requirements are difficult to achieve using traditional monitoring systems.

3.1. Management Task

3.1.1. Virtual Machine Scheduling

Virtual machine scheduling is the process of determining to which physical machine a VM should be allocated. In its simplest form, this problem can be formulated as a vector packing problem. Each resource represents one dimension in the problem[14, 15, 16]. However, recent works[17, 18] have shown that resource usage information cannot be linearly added to get the overall resource consumption due to the fact that current virtualization technology cannot provide a complete isolate executing environment for VMs residing on the same physical machine. Therefore, only resource usage information of each VM and the capacity of physical machine are not sufficient to create a realistic policy.

3.1.2. Resource Provisioning

Resource provisioning focuses on ensuring that an application has sufficient resources to provide its service. This is a common problem that has been studied in other architectures like cluster and grid computing. However, server consolidation complicates the problem by the overhead introduced by virtualization software. One solution is to estimate the required resources from historical resource usage data of the same application when running on native platform. Wood *et al.*[19] proposed a regression-based model that maps the native system usage profile into a virtualized one. In this work, they focused on estimating CPU usage of an application based on the resource usage of several representative applications both on virtual and native platform. Another interesting method is to estimate necessary resources from existing applications with similar architecture or have similar characteristic such as being CPU intensive applications. From this reason it might be useful if the monitoring system can provide the correlation between application performance parameters and the required resources.

3.2. Challenges

Traditional monitoring systems were designed to work in management domain. They focus on providing a complete set of resource usage information, in order that if any problems occur, the administrator can investigate the problems through several dimensions. These requirements are different for monitoring in research domain. Researchers usually prefer that the most accurate data be captured at the finest time interval, so that extensive analysis can be made. Also, in research work what is more important than the data themselves is the correlation between them and other related values. Traditional tools usually do not provide convenient methods to visualize those correlations. Another challenge is we would like researchers to be able to conduct their experiments independently at the same time. Therefore, our system should be able to give each researcher different resource usage values according to their interests. Our challenges thus are:-

- *Correlation between data* Several management tasks need to know the correlation between resource usage data of related machines. For instance, it's interesting to know what will happen to the CPU usage of other VMs on the same physical machine, if we allocate a large amount of CPU to a particular VM on that machine. It's important to note that each researcher might be interested in different correlation, therefore our system should provide flexible methods for users to define how they want to visualize those correlations.

- *Monitoring time interval* In research domain, usually we are required to capture data at fine grain interval. Traditional monitoring systems cannot satisfy this requirement due to the fact that at very fine grain interval the resources consumed by the monitoring systems themselves put too much burden on the systems they are trying to monitor. Some works, such as proposed in [12], suggest methods to achieve realtime monitoring by giving up on tick by tick data and focus on only update that have significant different from latest update, which is not acceptable for research work.

- *Target machine and interested resource values addressing method* We assume that there are many researchers working on the same computer cluster. At any specific time period, each researcher run his experiment using different machines and is interested in different sets of resource usage values. From above scenarios, we can see that, unlike conventional monitoring systems that normally keep track on a large resource usage data sets, if our system can provide the users with a method to specify their interested resource values and target machines, we can greatly reduce the data that needed to be captured.

4. IMPLEMENTATION

In this section, we describe the design decision and architecture of "Pantau" our monitoring system that realized the features proposed in previous section. Pantau was implemented as an extension of "munin" an opensource monitoring system. We summarize the design decision of Pantau in section 4.1 before describing the detailed architecture in section 4.2. The data gathering process is described in section 4.3. Section 4.4 shows the demonstration of Pantau system as well as some screenshots.

4.1. Design Decision

This section explains how challenges in section 3.2 are addressed in our design.

4.1.1. Correlation between data

Considering several essential resource information proposed in section 3.1, we found that existing monitoring systems need modifications for the purpose of capturing the relations between a physical machine and its VMs and provide a better visibility on their resource contention. The key solutions are twofold enhancements on existing monitoring system; (i) a new structure that relies on comprehensive monitoring process, and (ii) two level mapping processes introduced as new management functions.

- *Comprehensive monitoring* is a complete monitoring process to obtain the following required data:

  1. Physical machine's resource utilization (e.g. CPU, memory, disk I/O, etc.)
  2. Physical machine's VM's process's resource utilization (e.g. CPU and memory utilization for a VM process)
  3. VM's resource utilization (same parameters as for physical host's resource utilization)
  4. Application parameters on the VM (e.g. throughput of HTTP requests per second, database query response time, etc.)

  This comprehensive monitoring process requires a management server to monitor a VM as exactly as an ordinary physical machine. Since some management tasks may require resource utilization data to be updated at a high rate, while some only keep track of historical data and do not need frequent updates as this will consume too much storage space. Thus it is necessary that the monitoring system can offer different update rate for different kind of resource data.

- *Two level mapping process.* The 1st level mapping process is resource to resource mapping. This step maps resource's utilization of the physical machine to each VM running on the host. It also maps resource's utilization of one VM to another VM running on the same host. 2nd level mapping is a resource to application parameter mapping. We map each resource utilization parameter to each application parameter. This mapping process leads to the visibility of resource utilization of a VM on its physical host. These two level mapping process , as shown in Figure 3 shall allow us to perform a more extensive analysis on the collected data such as estimating virtualization overhead, capacity planning and optimum scheduling algorithm.

These two enhancements are realized through the use of *plug-in* and *flexible graph display* concept. The *plug-ins* are simply any executable scripts which can output in a predefined format. These *plug-ins* are what we use to do the real monitoring tasks. They defines how each monitoring task should be performed. The advantage of using *plug-ins* is our system is easy to extend. Our system provides users with default *plug-ins* to capture the data proposed in comprehensive monitoring section. However, it's impossible to predict ahead of time all the user's interested resource values especially for application performance parameters. *Plug-ins* allow users to extend our system to capture their interested

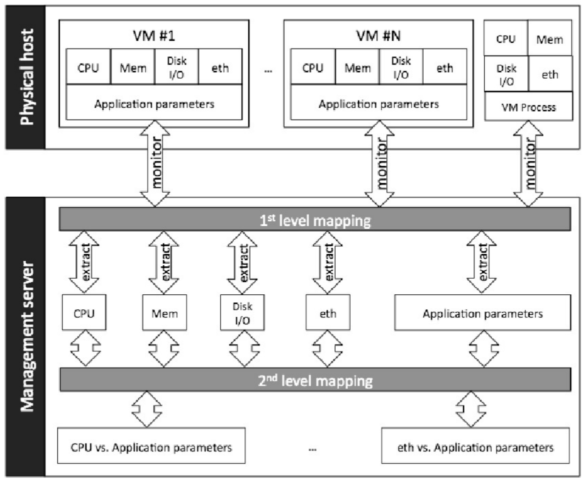

Fig. 3. Proposed Enhancement

data. The *flexible graph display* allows users to create graphs of their interested resource values from their target machines. Moreover, users can overlay two or more graphs to visualize the correlation between the resource usage data.

4.1.2. Monitoring time interval

Most monitoring systems support monitoring at much coarser time granularity. This is due to the fact that monitoring at fine time granularity consumes too many system resources and most of management scenarios do not need detailed resource usage information. However, this is not the case for research works. Pantau is designed so that it can capture information at very fine time granularity to support extensive analysis. Instead of giving up on tick by tick update to reduce the amount of the monitored data, we leverage the fact that normally not all the information is of interest all the time. Therefore, unlike other monitoring systems those capture data from all machines all the time, our system start capturing data only when it receives a monitoring request from user. Moreover, it only monitors on requested resource data from requested machines. This can greatly reduces the data needed to be captured. Fig 4 shows the structure of monitoring request. The start time, duration, and interval are self explaining. They tell us when to start at which frequency and for how long this monitoring task takes. The *plug-in* is explained in previous section. Users can specify many *plug-ins* to be used in a single monitoring request. The target host allows users to specify from which machines the resource usage data should be captured. Next section will discuss about this in more detail.

4.1.3. Target machine and interested resource usage values addressing method

We assume that researchers may conduct several experiments concurrently. Each of this experiment is conducted on different machines to avoid the side effect from other experi-

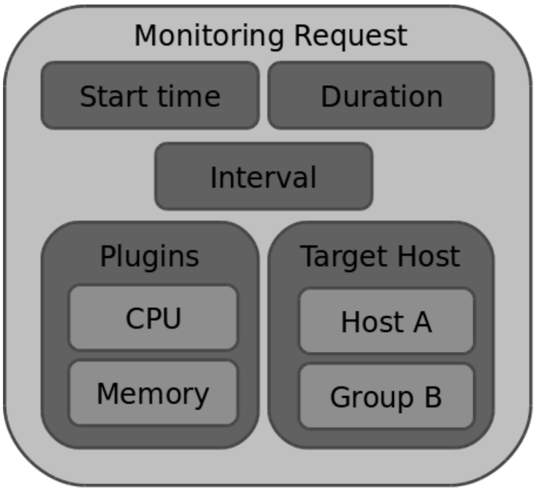

Fig. 4. Monitoring Request Structure

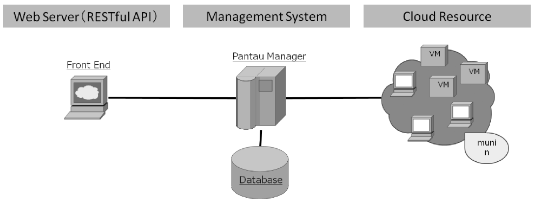

Fig. 5. Pantau Architecture

ments. Besides the *host name* concept, our system introduces the *group name* concept to allows users to specify their target machines in more systematic way. This *group name* is a collection of machines in the system. It can consist of both physical machines and VMs. A machine can belong to many groups at the same time. This allows users to specify monitoring tasks for a semantic group of machines. For instance, they can create a group consist of a physical machine and all VMs residing on that machine to study the effect a VM may have on other VMs residing on the same physical machine.

4.2. Architecture

This section explains the architecture of the Pantau system. For the sake of simplicity, we implement it using manager-client model with no federated capability. Pantau comes with user friendly web interfaces for managing monitoring tasks. One good point of Pantau is users can choose to capture only their interested resource usage data from their interested machines. This is suitable for research work because normally only a small set of resource usage is needed at the same time. For instance, users can choose to capture only CPU usage from all VMs running on physical machine S. Pantau consists of 4 components as depict in Figure 5.

4.2.1. Front end

Front end provides users with web interfaces to interact with the system. It consists of 2 subcomponents, the management screen and the graph drawing component. The management screen is implemented with simple HTML and java-script. From the management screen, users can schedule monitoring tasks, register new physical machines and query resource usage information. The graph drawing component is implemented using action script's FlexChart component. This component implements the functions referred as two level mapping in section 4.1.1. It can display graph from any captured data. It's also able to overlay graph of two or more captured data to show the correlation between any two resource usage data.

4.2.2. Pantau manager

Pantau manager is implemented using java. It behaves like a coordinator in the system. For instance, upon receiving a monitoring request from front end, it'll query database for all IP addresses of physical machines and plug-in's locations and communicate with munin system to carry out a specified monitoring task.

4.2.3. Database

The database is where our data is stored. This includes all captured resource usage data and also the IP addresses of all physical machines in the system. The IP addresses of VMs are not store in the database, instead we rely on each physical machine to keep track of all the hosted VMs. We use MySQL with MyISAM as the database engine. Due to the space and performance limitation, the database is archived after one month.

4.2.4. Munin Agent

We install unmodified munin agents on both physical machines and VMs. This component allows us to monitor a VM as if it is a physical machine. It communicates with Pantau Manager through munin protocol to get details on the monitoring task. Each munin agent launches munin plug-in periodically according to specified interval to capture resource usage data. To prevent network from flooding, all the usage data is stored locally. Upon receiving stop monitoring command, the munin agent sends back the captured data to Pantau manager.

4.3. Data Gathering

While most monitoring systems gather resource usage data and display them immediately at every specific interval, this approach consumes a lot of network bandwidth in case we need to monitor at a very fine granularity. Pantau tries to avoid this problem by storing captured data locally. Moreover, to reduce the amount of captured data, Pantau allows users to monitor only their interested value. Figure 6 shows the interaction between each component of the system.

The monitoring process starts by front end sends a monitoring request to Pantau manager. This monitoring request con-

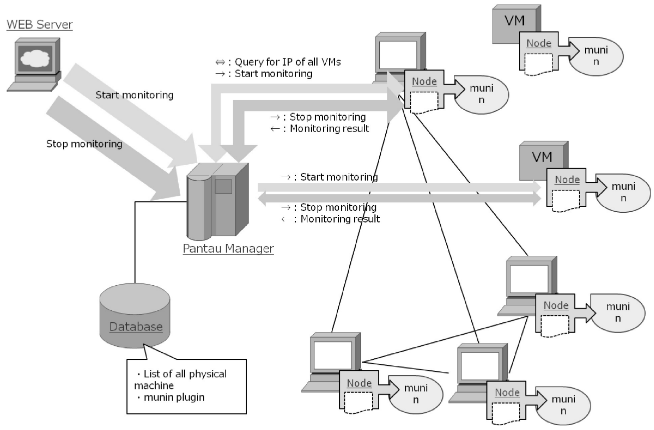

Fig. 6. Resource Usage Information Query

tains 4 information; monitoring start time, monitoring duration, interested machines and interested resource. Upon receiving the request from the front end, the request is put in a queue waiting for the start monitoring time to come. Pantau manager then queries the IP addresses of the target machines and the plug-ins's locations from the database. If it could not find from the database it will send queries to all physical machines assuming that the target machine is a virtual machine residing on one of those physical machines. The target machines can be specified either by host name or by group name which is a collection of predefined target hosts. Users can create new groups using the web interfaces provided by the front end. Pantau manager sends a start monitoring request to the munin agents on the target machines. This request contains information about which plug-ins are supposed to be launched at which time interval. The munin agents execute the requested plug-ins and stores the captured data locally. When the monitoring duration is over, the Pantau manager again sends a stop monitoring request to the munin agents on the target machines. The munin agents reply with the captured resource usage data. Pantau manager stores the data into database.

Users can query the resource usage data using web interfaces provided by the front end. The FlexChart on the front end visualizes the resource usage data into a graph. This graph can be overlaid on other graphs to show the correlation between two or more resource usage values.

4.4. Demonstration

This section shows a demonstration and screenshots from Pantau. The demonstration does not cover all possible usages of Pantau, but should give a better understanding on how our system works. We deployed Pantau on our testbed consists of 20 physical machines. Although the operating system used by all machines in this demonstration is debian lenny, it should be able to work on any unix-like platform.

In the demonstration, we assume that a user has already registered all required plug-ins and information of all cluster nodes using web interfaces. Figure 7 shows the scheduling

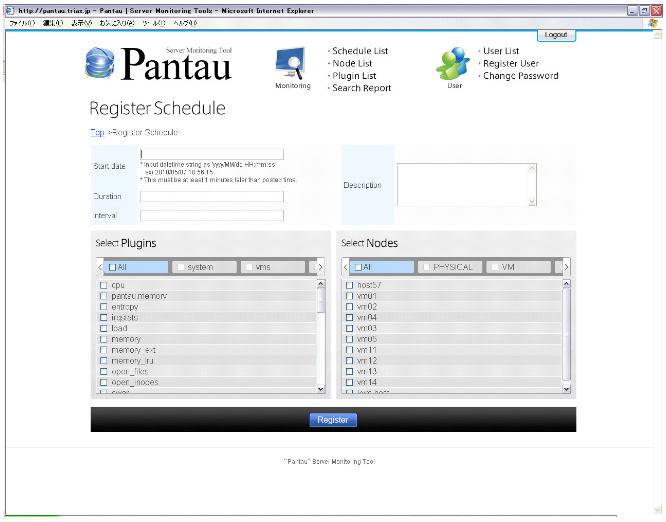

Fig. 7. Scheduling a new monitoring task

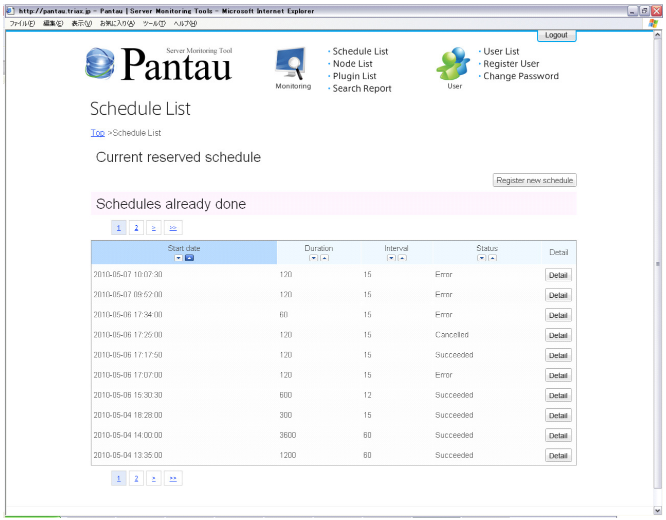

Fig. 8. List of all scheduled tasks

new monitoring task screen. A user can specify the start time, the duration, and the interval of the monitoring task. A user can choose resource usage data to capture and the target machine from the lists at the bottom of the screen.

After the monitoring task finished, it will be shown in schedules already done list as shown in figure 8

A user can choose how to visualize the captured data from create graph screen as shown in figure 9. The data captured from selected plug-ins on selected cluster nodes will be overlaid on the same graph. Figure 10 shows the graph result.

5. CONCLUSION

In order for research to progress smoothly, the first step is to have a monitoring system that can capture accurately the state of the system. However, existing monitoring systems were designed to work in management domain which has different requirements from research domain. More specifi-

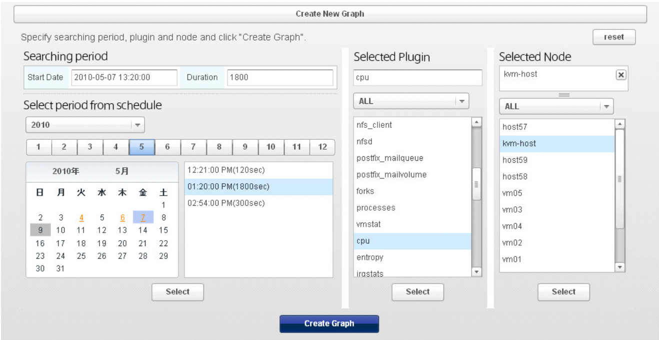

Fig. 9. Create Graph Screen

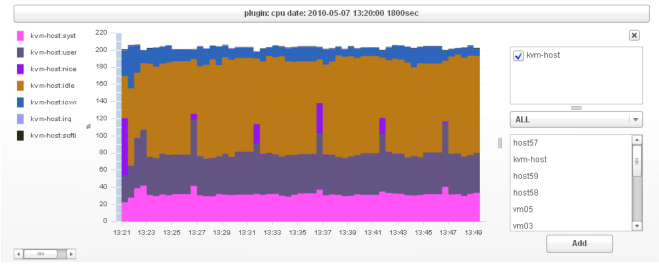

Fig. 10. Graph Result

cally, they usually do not provide enough information on the correlation between each monitored data. The monitoring time interval is also somewhat too coarse to draw any accurate conclusions. In this paper, we discussed design the criteria and challenges associated with research domain monitoring system. We also showed "Pantau", the monitoring system that implements the requirements proposed in this paper. We are planning to use the data captured from "Pantau" to study the characteristic of server consolidation system in more detail .

6. REFERENCES

[1] W. Vogels, "Beyond server consolidation," *ACM Queue*, vol. 6, no. 1, pp. 20–26, 2008.

[2] VMWare, "http://www.vmware.com," .

[3] Xen, "http://www.xen.org," .

[4] OpenVZ, "http://openvz.org," .

[5] KVM, "http://linux-kvm.org," .

[6] E. Levy A. Galis B. Rochwerger, D. Breitgand and K. Nagin et al, "The reservoir model and architecture for open federated cloud computing," *IBM Systems Journal of Research & Development*, vol. 53, no. 4, 2009.

[7] B. N. Chun M.L. Massie and D.E. Culler, "The ganglia distributed monitoring system: Design, implementation, and experience," *Parallel Computing*, vol. 30, no. 7, July 2004.

[8] Zenoss, "http://www.zenoss," .

[9] Nagios, "http://www.nagios.org," .

[10] Munin, "http://munin-monitoring.org," .

[11] K. Park and V. Pai, "Comon: A mostly-scalable monitoring system for planetlab," *Operating Systems Review*, vol. 40, no. 1, pp. 65–74, January 2006.

[12] Y. Kamoshida and K. Taura, "Scalable data gathering for real-time monitoring systems on distributed computing," in *Proceedings of the 8th IEEE International Symposium on Cluster Computing and the Grid (CCGrid2008)*, Lyon, France, 2008, IEEE, pp. 425–432.

[13] M. Schoffstall J. Case, M. Fedor and J. Davin, *A simple network management protocol (SNMP), RFC-1157*, May 1990.

[14] Y. Ajiro and A. Tanaka, "Improving packing algorithms for server consolidation," in *Proceedings of the International Conference for the Computer Measurement Group (CMG)*, 2007.

[15] T. Setzer M. Bichler and B. Speitkamp, "Capacity planning for virtualized servers," in *Proceedings of the Workshop on Information Technologies and Systems*, Milwaukee, Wisconsin, USA, December 2006.

[16] O. Loques V. Petrucci and D. Mossé, "A dynamic configuration model for power-efficient virtualized server clusters," in *Proceedings of the 11th Brazillian Workshop on Real-Time and Embedded Systems (WTR)*, 2009.

[17] X. Zhang D. Newell P. Apparao, R. Iyer and T. Adelmeyer, "Characterization and analysis of a server consolidation benchmark," in *Proceedings of the International Conference on Virtual Execution Environments*, 2008.

[18] Z. Wanf S. Singhal P. Padala, X. Zhu and K. Shin, "Performance evaluation of virtualization technologies for server consolidation," Tech. rep. hpl-2007-59, HP Labs, 2007.

[19] K. Ozonat T. Wood, L. Cherkasova and P. Shenoy, "Profiling and modeling resource usage of virtualized applications," in *Proceedings of the 9th ACM/IFIP/USENIX International Conference on Middleware*, 2008, pp. 366–387.

DESCRIBING AND SELECTING COMMUNICATION SERVICES IN A SERVICE ORIENTED NETWORK ARCHITECTURE

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ABSTRACT

Today networks offer communication services ranging from a rather simple and unsecure one to secure and reliable data transmission for communicating on the network. In the future, it is expected that networks will offer a large number of different communication services. With so many services available, determining which service to select and use becomes much more difficult. Here we propose a description schema including an ontology for describing communication services. For service selection a decision making process called Analytic Hierarchy Process (AHP) is utilized which is specially adapted and extended for automatic processing.

Keywords—Ontology, Communication Services, Service Oriented Network Architecture, Future Internet, Next Generation Internet, Analytic Hierarchy Process (AHP)

1. INTRODUCTION

The term ‘Service’ is widely used but the definition varies from domain to domain. The Organization for the Advancement of Structured Information Standards (OASIS) defines a service as “a mechanism to enable access to one or more capabilities, where the access is provided using a prescribed interface and is exercised consistent with constraints and policies as specified by the service description” [1]. Other definitions exist as well [2, 3].

Services can be either atomic or composite [4]. An atomic service is a service of unbreakable functionality. On the other hand, a composite service consists of a number of atomic services to constitute a complex service. In this paper, service means either atomic service or composite service.

Services are the central design elements of a service-oriented architecture (SOA) which can be defined as “[an enabling] framework for integrating business processes and supporting information technology infrastructure as [loosely coupled and] secure, standardized components — services — that can be reused and combined to address changing business priorities [4].” Three fundamental roles of a service-oriented architecture are Service Providers, Service Consumers and Service Brokers [5] as shown in Figure 1.

The Service Provider produces and supplies the service, the Service Consumer utilizes the service and the Service Broker provides the facility to advertise, search and discover the desired services.

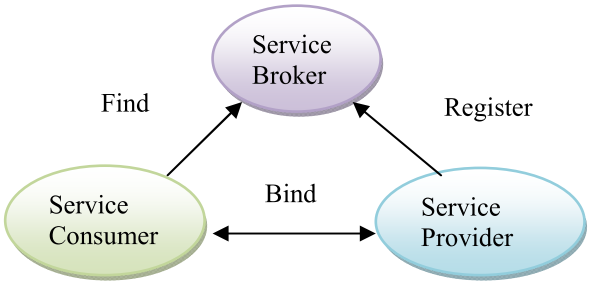

Figure 1. Roles and Operations of SOA

It has been proposed to build networks in the future following the SOA paradigm in order to achieve a flexible and well maintainable network architecture. For example, (SONATE) [6, 8] is a prototype architecture ensuring the SOA paradigm. In this architecture, the authors did not provide a facility for the users/application providers to define their preferences of one effect over another (for instance, success rate over delay, delay over cost) and did not provide a model for selecting the best service from a set of available services based on preferences specified by users/applications.

Searching and selecting a suitable service is not a problem today as there are not many communication services. In other words, the number of communication services in the Internet is limited. For example, an internet-service provider (ISP) offers an infrastructure for using certain services such as connection management, reliability, and routing by utilizing the TCP/IP protocol. Other Service Providers offer their services using various methods such as SOAP, HTTP, SIP, etc.

The future internet will have an even wider selection of communication services to choose from. With such a variety of services, each network user needs to have a way to select the communication service most appropriate for his needs.

Today application requirements are specified by application providers and end-users cannot directly influence those fixed requirements. For example, a user might want to communicate with his friends without worrying about security, but wants to do so affordably. On the other hand, other users might be willing to pay more to have secure communications. Future internet should provide a facility for users to influence their application requirements and thus the requirements will not always be same.

The ability to describe services is crucial. Here it is presented how to describe service from different "services providers", common protocol stacks as well as service of highly dynamic future networks. The central element for service descriptions is a common ontology and the ability to express precedence. It will be shown how the Analytic hierarchy process (AHP) [13] algorithm from decision theory can be adapted to an automatic process of service selection.

The outline of the paper is as follows. Section 2 provides a model for selecting services in the service oriented future network architecture. Use of ontology in this architecture is proposed in section 3. How to describe services in a generic way is shown in section 4. Section 5 illustrates the automatic service selection process by using and extending the Analytic Hierarchy Process (AHP). At last, section 6 concludes the paper.

2. A MODEL FOR SELECTING SERVICES IN THE SERVICE ORIENTED NETWORK ARCHITECTURE

By using the Service-oriented Architecture (SOA) paradigm, a service oriented network architecture (SONATE) has been proposed. As with SOA paradigm, the main three components of SONATE are Service Provider, Service Consumer and Service Broker as shown in Figure 2.

The numbers of providers for communication services are expected to increase in the future. Different providers will offer different types of communication services ranging from services provided by current protocols to fully dynamic composition services.

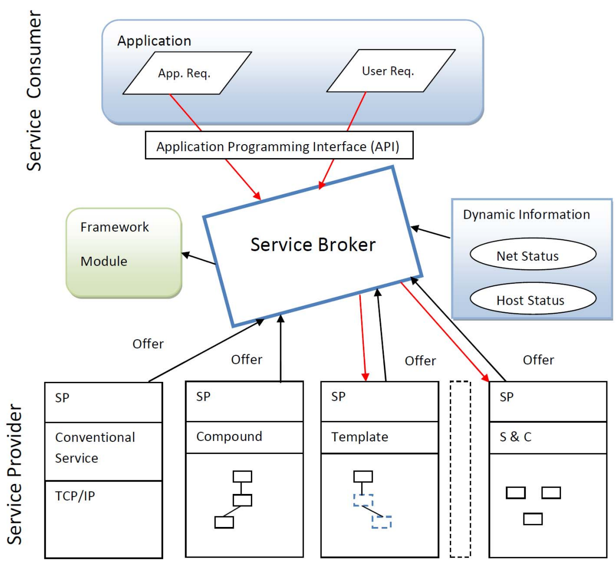

Figure 2. Components of a service-oriented network architecture

2.1. Service Provider

The main functionality of the Service Provider is to offer communication services. Different Service Providers can provide different types of services ranging from a conventional TCP/IP services towards fully dynamic services for the future. For example, one Service Provider might provide already existing communication services, for instance, services provided by UDP/IP which is required for backward compatibility.

Another Service Provider might provide new services which are pre-defined, pre-compiled and pre-composed. Several methods have already been developed for doing the pre-composition. For example, the Netlets approach [14].

A Service Provider might also provide a template for constructing a service which is slightly more flexible compared to the pre-composed approach. Here, certain functionality can be added or deleted based on the needs of the user.

A fully automatic selection and composition (S&C) can be provided by one of the Service Providers which take the requests from the user and compose a service on-the-fly based on the requirements specified by the user.

Taking the number of services and their composition time into account, different types of Service Providers are considered here. The number of services in the conventional Service Provider is limited and can be accessed very quickly whenever necessary. Compound Service Providers which store a lot of pre-composed services might consume time to search and select the appropriate one. The template Service Provider requires computation to compose the appropriate service after getting the requirements. A S&C Service Provider might take very long time to compose a service.

2.2. Service Consumer

Most of the communication requirements are usually specified by application programs. As a result, they are the primary consumer of a communication service. Another important consumer of a service is human. But, human users cannot provide their requirements to the Service Broker directly. They need an application program by which they can specify their demands. Users can specify their requirements by using the Graphical User Interface (GUI) which is usually associated with the application.

The Consumer needs a common Application Programming Interface (API) to make it transparent to him which Service Provider will be selected, and of course a large number of different APIs cannot be supported. It is notable that there may be different APIs for different classes of services, but the API supported by a service could be just one mandatory requirement. Definition of such an API is beyond the scope of this paper.

2.3. Service Broker

Residing in the middle of the Service Provider and the Service Consumer, the Service Broker accepts the user and

Figure 3. An ontology for effects to describe communication services

application requirements from the Service Consumer, receives dynamic information from the host and the network, gets services from the Service Providers and then selects the appropriate service based on the application requirements and dynamic information. For achieving the brokering task, the Service Broker might play a role as a negotiator.

In addition, the broker takes into account that

- There may be a large number of services,
- Some Service Providers need requirements to perform composition and
- Some of them may need a non-neglectable amount of time for composing a service (i.e., add delay to the decision making process)

The different types of Service Providers in the SONATE architecture facilitate consumers (users/applications/nodes) by providing services considering their time and demand. This is handled by using an API between Application and Broker which must provide limits for the selection time. Since the process of "rating" a service, which will be discussed in section 5, uses vector multiplication and can be done quickly, it is possible to promptly handle a large number of services.

After selecting the service, the Service Broker provides the workflow to the Framework Module for execution.

3. ONTOLOGY IN SONATE

In case when the requested service is not available to the Service Broker or template and/or S&C can offer a better service, the service needs to be composed partially or completely on-the-fly. It is impossible to compose a service without its clear and meaningful description. In addition, for matching a requested service with the offered services require explicit description as well.

When more than one Service Provider offers the same service but with different quality attributes (loss ratio, delay), the Broker must decide which provider to select for service provisioning which requires clear description

Mere works [6] have been accomplished concentrating on service description providing semantics for requested and offered services. Comparing the semantics of requests and offerings is performed only by matching strings which reference terms of a common ontology.

We defined an ontology [10] for a list of communication services as shown in Figure 3. Provided that each service in

the ontology has a specific clear meaning, all of the services can be described by the Service Provider and can, in turn, be consumed by the Service Consumer.
Using the ontology, we described the name of communication services and their properties. Communication services can have two types of properties: mandatory properties, and optional properties. Each of those types of properties can contain qualitative properties and/or quantitative properties which can be defined by using an ontology. So, it can be concluded that an ontology is essential for describing services in the service oriented network architecture.

4. SERVICE DESCRIPTION

Having different types of properties and their units make the service complex in structure which should be described in a generic way so that it can be handled easily.
The description must not be limited to a fixed ontology; it must be extendable so that new features can also be described in the future without changing the Service Broker.
A service can be described by a set of visible effects [6]. So, a service, S, is

$$S = \{VE_1, VE_2, \ldots\ldots\ldots VE_n\} \ldots.. \quad eq(1)$$

Services can be either requested or offered. The service that is requested by a user or an application is called requested service, can be denoted by S_R and the service that is offered by the Service Provider is called offered service and can be denoted by S_O.
For getting the optimal service, it is required to compare S_R and S_O and the result can be one of the followings:

Conditions	Action
$S_O < S_R$	User requirements cannot be fulfilled
$S_O \geq S_R$	User requirements can be fulfilled
$S_O = S_R$	Perfect match according to user expectation
$S_O \neq S_R$	No match and user cannot proceed

According to eq (1), comparing of services can be done by comparing visual effects (VEi, where i = 1 to n). The requirements of the Service Consumer can be fulfilled when the following equation is satisfied

$$S_O = \{VE_i\} \supseteq \{VE_j\} = S_R$$

Every individual visual effect VEi can either be a mandatory property, or an optional property. A mandatory property is a property that should be compared at first and must be fulfilled. As the name indicates, an optional property is not required to match exactly but can be negotiated between the application and the provider which is done by the broker. This property is optional and can be omitted as well when fulfilling the property takes an unacceptable amount of time. Both mandatory and optional properties can have qualitative and quantitative properties. Examples of qualitative properties (low/medium/high) are cost and security whereas examples of quantitative properties are success rate and throughput. These properties can be used for the following purposes

- find a service that is appropriate (has all mandatory properties)
- find the best service by optimization of optional properties

When more than one Service Provider offers the same service fulfilling mandatory properties but with different optional property, it is required to calculate or measure the weight of these properties (measured values) to choose the optimal service. The service description helps in the process of service selection by describing those properties in a generic way.

5. SERVICE SELECTION

A client can ask for a service from a set of alternatives specifying the criteria for selection. Analytic Hierarchy Process (AHP) [13] is a mathematical model used to choose one alternative from a given set of alternatives, usually when multiple decision criteria are involved. How this process can be adapted to an automatic process of service selection is shown here.

5.1. Analytic Hierarchy Process (AHP)

AHP, originally proposed by Saaty [13], is a process designed for human decision making. Basically, AHP is used for determining priorities of different alternatives. The flowchart of the process is shown in Figure 4.
The first step of the process is to define a hierarchy. The first and last levels in the hierarchy contain the goal and the alternatives to choose from respectively. One or more hierarchies in the middle contain evaluation criteria.
The second step is to assign pairwise priority to the criteria. The pairwise priority is the preference or satisfaction feelings of one evaluation criteria over another. For defining pairwise priority, a scale between -9 to +9 is used as shown in Figure 5.
The next step is to calculate the overall priority value or priority vector which provides the relative weight among the things or criteria we compare. The calculation detail is beyond the scope of the text.
The subsequent step is to check the consistency of the priority vector by using the method proposed by Saaty. If the vector is not consistent, the next step is to change the pairwise priority of the criteria and repeat the process from the second steps. When the vector is proved to be consistent, the priority vector of the next levels in the hierarchy is calculated.
Except for the first criteria level (i.e., the start level having criteria) in the hierarchy, the priority vector of subsequent hierarchy levels is calculated by multiplying the priority vector calculated from the nearest upper hierarchy which is consistent and the priority vector of the hierarchy.

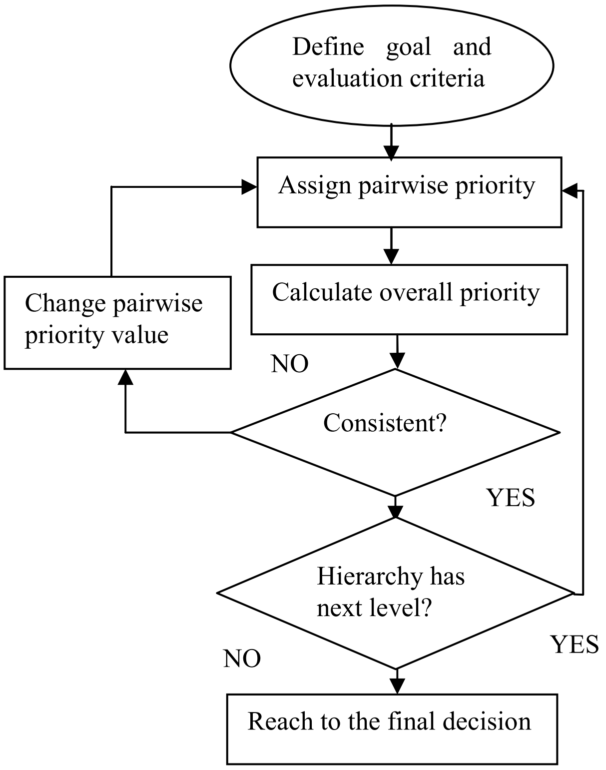

Figure 4. Analytic Hierarchy Process (AHP)

The priority vector of the last hierarchy provides the final decision.

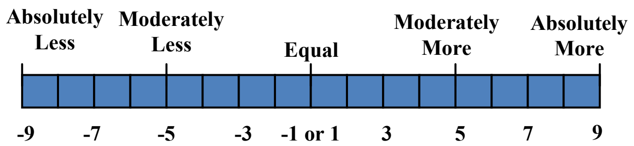

Figure 5. Pairwise comparison scale

The existing AHP process is described in this section. The next section will show how the process can be extended to select a service automatically.

5.2. AHP in service description and selection

AHP can be used for describing services and selecting the best one from available services. The procedures for selecting the best service are

1. Define the effects for selecting a service
2. Assign pairwise priority to the effects
3. Calculate a priority vector of the effects not violating consistency
4. Pass the priority vector of the effects to the broker along with the requirements specified by the user/application.
5. Calculate pairwise priority among the offered services based on the requirements specified by the user and effects provided by the services. This requires a mapping mechanism which cannot be done by AHP. That is why, we propose a mapping mechanism in the section 4.2.1.
6. Calculate priority vector of the offered services preserving consistency
7. Calculate the overall priority vector of the offered services by multiplying the priority vector coming from the application and the priority vector calculated in the offers.
8. Select the service with the highest priority.

The hierarchy for selecting the best service is shown in Figure 7. The top of the hierarchy exposes the goal and the next hierarchy defines the effects that are necessary for achieving the goal. The effects can be either qualitative properties or quantitative properties or both.

Defining a priority is not necessary for mandatory properties because these properties need to match exactly. Instead, the priority needs to be defined for optional properties for achieving the optimal result/service.

The priority is usually assigned by the application (application developer). But, users might influence to define such priorities.

Besides the priority, users/applications need to specify some more values of the effects that are required for the broker to calculate which services in a certain effect should be given more priority than others. These values can be expressed as <effect-name,lb,ub>. Examples of effects are success rate, delay, and cost. The lower bound (lb) and upper bound (ub) are the quantitative limits of an effect where lb is the minimum value of the effect, ub is the maximum value of the effect. <Success rate, 80%, 0> means that the minimum value of the success rate for the application/user is 80%. <Delay, 0, 100ms> means that the maximum required delay is 100ms. <Cost, 5, 5> means that the desired cost is 5.

5.2.1. Pairwise prioritization of services per effect

Defining pairwise prioritization on services for every effect is a critical task, and cannot be provided by the Service Provider which has no idea of the application needs. The question is who then can provide the prioritization of services and how can this be done. In general, this crucial task can be accomplished by using the following process

1. Offerings (given by the Service Provider) describe measured values for every effect.
2. Requirements contain hints for mapping measured values into prioritization as shown in Figure 6.
3. The Service Broker uses the hints of the requirements to parameterize a mapping algorithm.
4. Use the resulting mapping to calculate the prioritization of services with the “measured values” provided by the offerings.

The mapping should have certain properties. Firstly, the mapping must be generic, i.e. not specific to effects or units of measured values. Secondly, the mapping must be monotonic. Thirdly, a liner mapping of measured values to prioritization is not adequate (for instance: delay of 10ms = +9 50 ms = +1 500ms = -9)

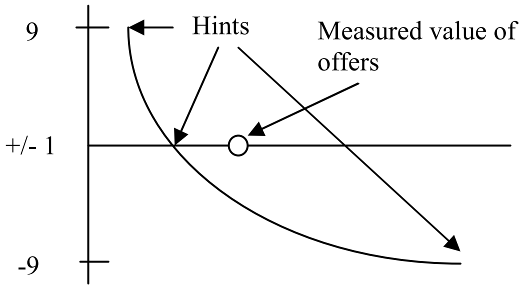

Figure 6. Values measured in terms of hints

An approach for mapping is proposed here to use a monotonic interpolation/extrapolation. In this case, requirements provide value points for interpolation/ extrapolation (must be monotonic). A monotonic interpolation/ extrapolation of these points are used to define mapping. In addition, the specific "measured values" of the offerings are then mapped to its priority.

Let us assume that f () is a function used to define mapping. As an example, considering interpolation, the requirements must contain at least the following two points

- x_0 where f (x_0)= - 9
- x_n where f (x_n)= + 9

If there are measurement values y not within the interval [x_0,x_n], we can extrapolate

- if y < x_0, then f (y) = - 9
- if y > x_n, then f (y) = + 9

Using inter-/extrapolation, a requestor must provide two points but can specify as many parameters as he wants to be more precise.

The aforementioned mapping mechanism facilitates the broker to assign a priority of one service over another for every single effect.

5.2.2. Example Scenario of AHP in service selection

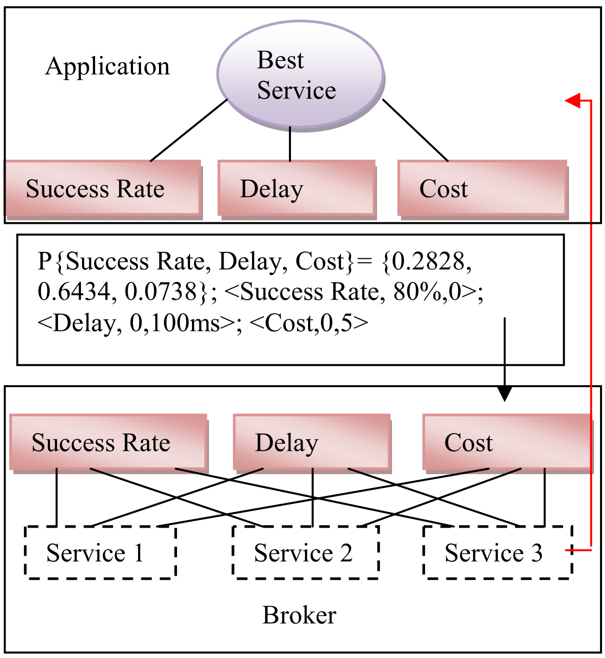

Figure 7. Example scenario for getting the best service

An example scenario for selecting the best service is shown in Figure 7. Here, the application developer defines the pairwise priority as {Success Rate, Delay} = {1,-3}, {Success Rate, Cost} = {5,1} and {Delay, Cost} = {7,1}. Moreover, the expected values of Success Rate, Delay and Cost are specified as shown in the Figure. After calculating the priority matrix, the application sends these values (priority matrix and user/application defined value which is called requirements) to the broker. The broker compares the values of the effects of the offered services with the requirements and then assigns the pairwise priority to the services based on effects. In this case, based on success rate, delay and cost, the assigned pairwise priorities are shown in Table 1 where Service 1, Service 2 and Service 3 are denoted as S1, S2 and S3 respectively.

Table 1. Pairwise priority for different effects

Effects	{S1, S2}	{S1, S3}	{S2,S3}
Success Rate	{-3,1}	{1, -4}	{-2,1}
Delay	{-4,6}	{1, -6}	{-3,1}
Cost	{-3,1}	{1, 4}	{7,1}

Then, the priority vectors are calculated using the values which are shown in Table 2. Finally, the overall priority vector is calculated by multiplying the priority vector received from the application and the priority vector calculated by the broker providing the final result of the priority {Service1, Service 2, Service 3} = {0.11, 0.315, 0.574}. Now, the broker returns the best service (Service 3) to the application.

Table 2. Calculated priority vector for different effects

	{S1, S2, S3}
Success Rate	{0.1226, 0.3202, 0.5571}
Delay	{0.0869, 0.2737, 0.6393}
Cost	{0.2684, 0.6555, 0.0796}

6. CONCLUSION

The number of communication services is expected to increase in the future. Each provider will provide specific types of communication services ranging from services provided by current protocols to fully dynamic composition services. As the number and variety of communication services increase, it is essential to describe them in a generic way so that the Service Consumer can select the most appropriate one. For solving this, the description schema was extended to include an ontology for describing services and their properties meaningfully. Furthermore, the Analytic Hierarchy Process (AHP) was extended to select between similar services according to the client's requirements.

A Service Broker, residing in the middle, takes the requests from the consumer and offered services as inputs and selects the best offered service based on the requests. A schema including an ontology has been described to provide the meanings of the offered services so that client requirements are met.

For selecting the best service by using AHP, it is required to calculate the priority of one service over another. Defining pairwise prioritization on offered services for every effect is a critical task, and cannot be provided by the Service Provider which has no clue of the application needs. Here, we proposed a mapping mechanism for doing this task.

By extending the Service Oriented Network Architecture (SONATE) with an ontology and AHP, the problem of describing services and selecting the best one is solved. To implement this type of solution, a standard description scheme and ontology need to be developed and agreed upon. The properties upon which AHP or other similar methods relies on also need to be standardized. A standard organization might play a role in the standardization process.

Appendix A. Examples of Communication Services

Some examples of communication services are given below:

Security: Security is one of the most popular and necessary communication services which mean that the data is kept safe from intruders/middle-man during communication. This service is necessary for online banking transactions, military communication, medical communication, emergency needs and much more. There are several security services: *integrity*, *data-origin authentication* and *confidentiality*. *Data origin authentication* is a security service that verifies the identity of the claimed source of data. This ensures that the information is sent to or from a trusted partner. *Integrity* is a security service that ensures that modifications to the data are detectable. Even if the intruder obtains the information, *confidentiality* ensures that the man-in-the-middle cannot understand the information by changing the information into an unintelligible form.

Users can ask either request 'Security' in general or one or more of those security services: data-origin authentication, integrity, or confidentiality. Here, it is assumed that, when users request the 'Security' service, all of those security services will be provided.

Routing: This service, in general, routes the packet from source to destination.

RTT_Information: User or application can get Round Trip Time information by using this service

Hop_Information: Using this service, users or applications can get the number of hops between the sender and the receiver

Addressing_Conversion: Using this service, the addresses can be converted from one type to another. For example, from IPv4 to IPv6 address.

Prioritization: When the user or the application needs to give priority of one class of traffic to another, this service can be used.

Signal_Conversion: In case the application needs to convert from one signal type to another, this service can be used, for instance, conversion from analog to digital signal or the other way around.

Size_Reduction: If the application cannot send a file because of its size, a Size Reduction service can be taken to do the task. Compression is one type of size reduction service. The user can request for either of the services to get the desired task done.

Availability: Availability covers different services. The most common one is monitoring which observes whether the peer host is still up and the connection is still alive. Employing the monitoring service as a foundation, a path management service can be created. These can have two different types: the basic one is multihoming. In this case, there are multiple available paths. If for certain circumstances, one path fails, it switches to another path that is not erroneous. A drawback of this service is that always only one path is active. There is a service called *Load_Sharing* which uses different paths simultaneously. Another availability service is Denial-of-Service availability (*DoS_Availability*) which ensures that the authorized users are still able to get served even when the system is under attack. Users can ask explicitly for one or more of the availability services or for the 'Availability' service in general when only the Monitoring service will be provided.

Addressing: This is a common communication service that identifies the source and destination process and its devices. Users or applications can request for one or more of the addressing services explicitly.

Connection_Management: This service provides connection management including connection establishment and connection termination. Users or applications can explicitly request either of the services or in general 'Connection Management' where both of the services will be provided.

Reliability: The Reliability service ensures that the data must reach the destination without any corruption. There are several reliability services:- error detection, data flow limiting, order preservation and error control. As the name indicates, error detection service detects errors that have been happened on the way. Data flow limiting is an important service in a shared network which is used to avoid source, destination and network congestion by limiting data flow. The order preservation service ensures that the data arrives at the destination in the same order as the data has been sent. When 'Reliability' service in general is requested, all of the reliability services will be provided. But, the user can clearly ask for one or more of the reliability services for decreasing the cost of communication.

Packet_Boundaries_Preservation: In case of user or application needs, this service ensures that the packet will not be segmented or fragmented.

Path_MTU: This service provides the size of the maximum transfer unit between the source and the destination.

Loop_Avoidance: This service avoids loop during routing data.

REFERENCES

[1] OASIS Reference Model for Service Oriented Architecture 1.0, Official OASIS Standard (Normative PDF), Oct. 12, 2006, http://docs.oasis-open.org/soa-rm/v1.0/soa-rm.pdf, accessed on 08th April 2010.

[2] A. Kabzeva, M. Hillenbrand, P. Müller, and Ralf Steinmetzy, Towards an Architecture for the Internet of Services, 35th Euromicro SEAA conference, 27-29 AUgust 2009, Patras, Greece.

[3] E. A. Marks & M. Bell, Executive's guide to service-oriented architecture, John Wiley & Sons, 2006

[4] J. Lawler and H. Howell-Barber, Service-oriented architecture : SOA strategy, methodology, and technology, Auerbach Publications, 2008

[5] T. Erl, Service-Oriented Architecture – Concepts, Technology, and Design, Prentice Hall, 2005, ISBN: 0-13-185858-0

[6] B. Reuther & D. Henrici, A model for service-oriented communication systems, Journal of Systems Architecture, 2008

[7] B. Reuther, D. Schwerdel, A. Siddiqui, Z. Dimitrova, P. Müller, A Protocol Framework for a Service Oriented Future Internet Architecture, GI/ITG KuVS Fachgespräch Future Internet, Munich, 2009

[8] B. Reuther and P. Müller, Future Internet Architecture - A Service Oriented Approach. In Oldenbourg Verlag, München, 2008.

[9] D. Schwerdel, Z. Dimitrova, A. Siddiqui, B. Reuther, P. Müller, Paul, Composition of Self Descriptive Protocols for Future Network Architectures, EuroMicro 2009, 27-29 August, Patras, Greece

[10] T. R. Gruber, Toward principles for the design of ontologies used for knowledge sharing. Int. J. Hum. Comput. Stud., 43 (5–6):907–928, 1995.

[11] L. Liu and M. T. Özsu, Encyclopedia of Database Systems, Springer-Verlag, 2009.

[12] Protégé ontology editor, http://protege.stanford.edu/

[13] T. L. Saaty. Relative Measurement and its Generalization in Decision Making: Why Pairwise Comparisons are Central in Mathematics for the Measurement of Intangible Factors - The Analytic Hierarchy/Network Process, RACSAM (Review of the Royal Spanish Academy of Sciences, Series A, Mathematics), , (2008-06), 102 (2): 251–318.

[14] L. Voelker, D. Martin, I. E. Khayat, C. Werle & M. Zitterbart, An Architecture for Concurrent Future Networks, 2nd GI/ITG KuVS Workshop on The Future Internet, GI/ITG Kommunikation und Verteilte Systeme, Karlsruhe, Germany, Nov 2008.

[15] G. Klyne, J.J. Carroll (Eds.), Resource Description Framework: Concept and Abstract Syntax, W3C 2004

VIRTUALIZED PASSIVE OPTICAL METRO AND ACCESS NETWORKS

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ABSTRACT

This paper outlines and proposes the vision of an open architecture framework for a virtualized passive optical metro access network. An open lambda environment based on this architecture framework and its benefits to different stakeholders are described. The paper provides analogous comparisons with existing proposals and discusses two practical examples of applying this architecture framework in a metro-access network and in mobile backhaul. Socio-economic impacts, challenges, as well as possible directions in standardization are analyzed.

***Keywords*—**Network architecture, policy and economic issues

1. INTRODUCTION

The broadband access data rate is expected to grow at a significant rate in the foreseeable future. The quest for more bandwidth at an affordable cost is forcefully driving innovations in metro access technologies. As the vast array of new technologies and new service requirements are being introduced, one of the challenges the industry faces is the metro access architecture design to support and sustain the growing bandwidth demand in the next 5 to 20 years.

This topic has triggered many discussions for the past several years. Public funded projects in Europe, such as PIEMAN, NOBEL, and STRONGEST represent examples. In the United States, the Federal Communications Commission (FCC) unveiled the National Broadband Plan [1] to outline the strategy "*for achieving affordability and maximizing use of broadband to advance consumer welfare, civic participation, public safety and homeland security, ...*" One major challenge is thus the architecture design to achieve the ambitious goals set forth by countries globally.

Although the majority of current commercial deployments in the wireline access market are copper wire based xDSL networks, it is expected that fiber optic network will soon be required to support the bandwidth growth. Industry standards such as Gigabit Passive Optical Network (G-PON) and Gigabit Ethernet PON (GE-PON) have long been completed [2, 3]. Countries in Asia are leading the way for GE-PON deployments, while G-PON deployments are gaining momentum in other parts of the world. Standards for 10GE-PON and 10G-PON have been consented in the IEEE and in ITU-T, respectively.

All of the above mentioned PON systems are based on TDM technology and employ strict and static wavelength assignments for video, upstream and downstream channels, leaving most of the available fiber spectrum un-used. This invariably leads to the questions of how best to utilize the potential bandwidth more efficiently, what such architectures may be like, and what benefits this will bring upon service providers, vendors, and individual subscribers.

In this paper, we propose an open architecture framework to realize a virtualized passive optical metro access network. The key motivations for this proposal are threefold: to understand and leverage implications expected with the introduction of innovative technologies to PON networks; to create new opportunities previously unavailable in PON networks; to maximize benefits for different stakeholders while fostering new innovations. With these motivations in mind, an independent open discussion forum, the Open Lambda Initiative (OLI) [4], is proposed to facilitate the discussions of formulating such an open architecture.

This paper is organized into the following parts. Section 2 provides an overview of several existing proposals for open spectrum sharing architectures and makes an analogous comparison of an OLI enabled network to wireless networks. Section 3 describes the vision of an open lambda environment and the benefits that can be achieved through the deployment of networks in such an environment. Two use case examples of applying the OLI concept in a metro-access network and in mobile backhaul are explained. Section 4 outlines the ecosystem envisioned to foster an open lambda environment. Issues and requirements for component technologies, system design, and management layer are discussed. Section 4 also describes the potential socio-economic impacts and the path forward to standardization, timeline and challenges. Section 5 concludes and summarizes the proposed open architecture framework for a virtualized passive optical metro access network.

2. EXISTING PROPOSALS

The concept of OLI can be thought of as an extension of the spread spectrum and frequency division multiplexing concepts in wireless to the optical domain. In wireless networks, there is an acute need for frequency optimization and protocols have been proposed to ensure optimal use of the spectrum. Often the variability of user requirements implies a disparity in traffic demands. OLI is hence a good technique that caters to the traffic variation – offering passbands that are tuned to user requirements.

2.1. Comparison with existing approaches

The OLI concept for the first time puts forth a conceptual framework for flexible wavelength spacing and lists the potential advantages and challenges of such a framework. However, there have been approaches that exhibit functionalities similar to the OLI concept. All of these approaches have been proposed with specific applications as drivers. On the other hand, OLI has been proposed as a generic framework for next generation communication in metro access networks. It must hence be noted that OLI represents a new approach for spectrum utilization in optical communications and is particularly useful for the last mile as well as metro access especially in the developing world, as can be seen in Section 3.3. in more detail.

Three architectures have been proposed that are similar to the OLI concept, without specifically mentioning the conceptual difference between existing ITU-grid optics and OLI optics. These approaches are: SLICE [5], ROAMTS [6] and Elastic Optical Networks Project (EO-Net) [7]. It must be noted that there is a conceptual relationship between these three proposals and OLI, whereby OLI provides a generic framework whilst the three proposals are specific implementations.

SLICE, spectrum-sliced elastic optical path network, was introduced by researchers at NTT as a spectrum efficient and scalable optical transport architecture to support various data rates with various wavelength granularities. It was proposed to serve as a mid-term solution before optical packet switching becomes mature enough to provide efficient bandwidth allocation for wavelength routed optical path networks.

SLICE is designed to provide proportional optical bandwidth based on bandwidth requirement or traffic volume; but unlike Virtual Concatenation, it cannot aggregate scattered optical resources to optimize the usage of unused optical spectrum although required optical resources are assigned for a service. a

SLICE differs from OLI in that OLI is aimed as a mid-to-long term solution to provide a cooperative management framework to facilitate network operations, management, monitoring, and configurations among all operators. SLICE in its current definition, although also target for optimized bandwidth efficiency, does not support this mode of operation amongst operators. OLI is well suited to passive architectures and provides the optimum efficiency for metro access networks, whilst SLICE is proposed for optical networks in general.

ROAMTS or Reach Optimized Architecture for Multi-Rate Transport System utilizes the OLI concept and proposes a hardware architecture for a node configuration in the metro area. The ROAMTS node architecture is based on a staged-WSS (wavelength selectable switch) concept and partitions the fiber bandwidth into multiple contiguous bands. Each band is composed of channels that have a similar spacing, but the spacing of channels in different bands is different. The bands may not be contiguous in the spectrum and hence ROAMTS is different from conventional waveband switching. ROAMTS fully uses the OLI concept but does not showcase such a conceptual framework.

The third concept proposed in the EO-Net is similar to SLICE and is based on OFDM as a frequency enhancer. It is one of the new research projects under the CELTIC program.

2.2. Comparison with wireless network

Wireless systems have since inception used the concept of flexible frequency bands for better channel utilization. However, these systems work at significantly lower carrier frequencies and hence data rates, implying that the ability to switch or modify channel bandwidths and make the bands flexible is significantly easier than in the optical domain. Customization of wireless access devices using ASICs and DSPs have realized the flexible frequency use in wireless systems is now a standard phenomenon and is hence commercially viable and easily available. Given the benefits of an OLI enhanced environment, it can be expected that such a concept would gain quick commercial acceptance leading to a revolution in the entire product cycle from system vendors, to component vendors, to service providers and aided by the regulator. Aspects of these enhancements of the product cycle are discussed in this paper.

3. DESCRIPTION OF AN OPEN LAMBDA ENVIRONMENT

3.1. Benefits and challenges

First and foremost OLI describes a network configuration which as such does not yet exist with any of the currently available technologies, whether these are TDM or WDM based PON systems. An open lambda enabled network concentrates on implementing a “management” system which needs a solid framework with practical rules to make it function efficiently. Such a framework will have to address the technical challenges that may beset such a system.

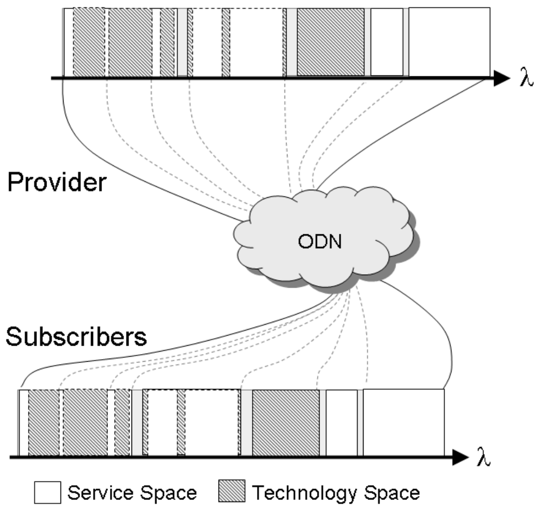

Fig. 1: High level abstract view of spectrum usage in an OLI environment

An OLI environment is in some ways analogous to a wireless radio. In an OLI environment, all channels are transmitted from one or several Optical Line Terminals (OLTs) to all users on a common Optical Distribution Network (ODN). An implementation example is described in [8] where each user is assigned a specific wavelength where the ONU automatically tunes and locks on to. The channel spacing is kept flexible by being adjustable and reconfigurable as and when required. As long as all network elements follow a set of pre-determined rules, a number of network elements may operate on the same ODN, preferably including only passive and maintenance free components as channel combiners. This implementation offers a logical point-to-point link provisioned with flexible wavelength granularity for each user.

Fig. 1 shows an abstract view of spectrum assignment of the OLI model where the partitions of the spectral band are divided into *service space* and *technology space*. A service or technology space refers to a portion of the spectrum allocated to a particular service or technology. Each *space* may encompass additional sub-spaces. In such an environment, several different service providers providing business, residential, and mobile backhaul services may share the same fiber infrastructure using the same technology.

One of the most pressing requirements is to automate subscriber functions without any active involvement from the operator. This becomes even more important if the physical medium, “the fiber”, may not belong to the network or service operator. In this case, services are provided in a virtualized plane, each independent of each other. This is one of the advantages of an OLI system, but at the same time, can be of substantial complexity in detail.

In an OLI environment, the same fiber could be shared by multiple service providers structurally independent of each other, whereas in a typical bitstream sharing environment, the service providers are inter-dependent on the access technology. If such a physical independence is to work, then strict rules are needed to ensure that these different systems do not interfere with each other, which could otherwise lead to service degradation or total loss of service. In order to minimize the risk of potential interference, the OLI management system must maintain a database with information about its physical environment and be aware of selected technical parameters. The balance of controlling such technical parameters while allowing each system maximum freedom within its own spectral region is subject to further discussion [4].

For example, two different operators, a mobile and a fixed networks provider, may require handling of two different types of technologies on the same fiber. Currently, operators are almost always required to build independent fiber networks to support their individual requirements. Whilst this is technically a much simpler solution, it does however require more resources that lead to a major CAPEX and OPEX spending. OLI enabled fiber access aims to ensure that both technologies are working on the same fiber.

In its simplest form, each channel is aware of the other spectral occupancies in its fiber medium. This may be achieved by an OLT sending a metadata stream in addition to the signal payload. This metadata stream may then contain a list of all the other channels or spectral occupancies and their associated operating parameters. The parameters may include, for example, channel assignment, system/service/operator IDs for each spectral band, modulation scheme, and reach information among others. The list of parameters becomes more complex when more than one OLT is considered. All the OLTs need to be aware of each other’s signal channels to avoid interference. However, different OLTs should not have to arrange each others channel plans by a direct link with each other and a fixed wavelength plan dictated by an operator. Rather, each OLT should be able to extract enough information from the metadata to make coexistence possible. This degree of automation would be unique in the data world.

To achieve coexistence, a new concept which we propose to call the *wavelength hotel* is required. This concept describes the management of a pool of available channels which any OLT is capable of working with. Each attached OLT must understand the pool of resources of the other OLTs present and ensure that it itself only operates on channels or spectral areas not yet utilized by other OLTs. Likewise the *wavelength hotel* needs to be aware of foreign channels which do not have any associated metadata and thus likely represent a rogue or foreign technology. This flexibility works best in an ODN design that is without any wavelength selective/limiting components.

There needs to be a spectrum assignment/ management authority, who may be the infrastructure owner or one of the service providers. The spectrum assignment authority facilitates the communication and interaction amongst the network providers, service providers, and OLTs. A set of regulatory rules comparable to those for free space radio transmission need therefore be established in an open lambda environment The key objectives of the OLI rules are to make the fibre an open infrastructure for spectrum sharing; to increase the speed of introducing new services and technologies to market; to create a highly competitive landscape; and to encourage new competing technologies sharing the same infrastructure. Municipalities, whole sale network providers or other investors would be the ones to build out and operate such fiber infrastructure.

Envision for example an OLI-enabled system running on the same ODN with a legacy GPON/EPON system. In this case, an OLI enabled OLT would need to mark the legacy wavelength channels in its *wavelength hotel* as occupied and would operate its own wavelengths well clear of these. Legacy elements themselves must be protected with WDM type filters to prevent overloading its receivers from OLI laser sources.

3.2 Economic benefits in a shared infrastructure

Several benefits can be derived from an OLI enabled network environment. In a shared fiber infrastructure, reduced civil works, simplified equipment inventory, and shared management platform can be expected. These all lead to lower CAPEX and OPEX spending and to the creation of new business opportunities. Community fiber providers or municipalities may share a common business platform. Eventually, in a truly virtualized network environment, facility sharing or collocation may no longer be required at all. In addition, service providers will have the flexibility to negotiate SLAs directly with subscribers and may benefit from completely independent FCAPS management.

An OLI environment profits from its future proof network segmentation of the fiber medium by providing unhindered access for different stakeholders. In such an environment all providers are physically independent of each other and can dictate their own time to market for new and innovative services. Competition will not be governed by who owns the fiber, but rather by what applications one can provide on it.

Imagine the scenario where a consumer contacts his network virtual operator, and upon providing the “PON SIM card” number, the consumer is able to have the data rate dynamically adjusted or switch to a new service provider by dynamic channel assignment. On the other hand, a service provider can set up a different charging structure for different service plans utilizing different technologies. Such advantages and flexibility can only be achieved in an open lambda environment.

4. USE CASE EXAMPLES

4.1. Use case example – metro-access network

A promising use of the OLI concept is in the metro access network, especially in the developing world for broadband access and in the developed world for cable companies. We outline this use case in Fig. 2. Shown in the figure is a metro core network that is characterized by a Reconfigurable Optical Add-Drop Multiplexer (ROADM) and adjoining distribution ring at each network Point of Presence as represented by the ROADM Network Element (NE). The ROADM typically drops wavelengths at the POP, intended for distribution in the distribution ring for metro access or last mile network.

In developing countries, the needs of bandwidth are lesser than those in the developed world [9]. This implies that a classical PON model is not always beneficial. A metro access/last mile local fiber loop results in both fiber savings and equipment savings. Such a fiber loop must have as its inherent characteristic a passive network; however, not all nodes in this kind of a fiber loop require less bandwidth, with some nodes being enterprise type and could require large bandwidth. By using OLI concept in such a fiber access loop, we are able to meet the dual needs of the residential customers and enterprise users while ensuring wavelength segregation and appropriate treatment. The available spectrum can be flexibly shared between multiple users through the OLI enabled network. A user can be given any amount of spectral bandwidth implying support of multi-rate communication in this passive network. Similarly multiple low-intensity users can each be given low granular wavelengths that are all tightly spaced and grouped together. This concept works well when the bandwidth needs of the residential users increase. When such a situation occurs, each user can be given a wider bandwidth to support a higher data rate until the full fiber capacity is used up.

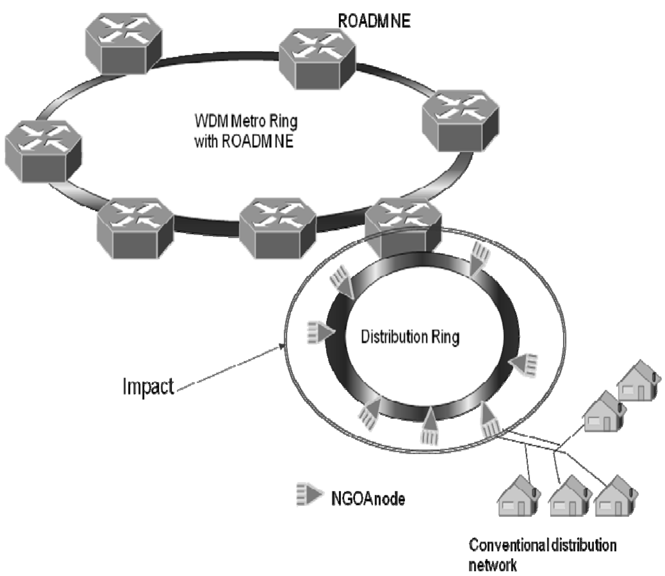

Fig. 2. Enhanced metro access network

4.2. Use case example – mobile backhaul

The growth of mobile services, especially moving into 3G and 4G technologies has created a set of new requirements for metro and access transport. Mobile backhaul is seen as a critical service, especially since it is one of the top-most revenue bearing services in the metro. As we move from 2G to 3G and towards 4G, the requirements juxtaposed by the backhaul are both critical and daunting.

Amongst these requirements is the need for fast-provisioning of bandwidth between base-station systems (BSSs). Bandwidth pipes must be provisioned between BSSs on-demand and further the granularity of these pipes need not necessarily be static. A second problem of such a system is the need for ensuring high QoS to the provisioned connections. A third aspect, especially for 3G and 4G, is the ability to provision connections between BSSs – bypassing mobile switching centers and eNodeB.

Open lambda initiative compliant systems are an excellent technology platform for the mobile backhaul. They provide on-demand connections between users as well as the ability to support multi-granular traffic. The OLI scheme provides multiple channels with flexible spacing – both of which enable scalability of the backhaul while ensuring good support for service provisioning. Existing fixed wavelength spacing solutions are not able to meet the three pronged requirements of dynamism, multi-granular support and heavy peer-to-peer communication.

5. ISSUES AND REQUIREMENTS

5.1. Technology Impacts

Existing standards are based on the assumption that the very large available spectrum of an optical fiber can be sliced every time a new technology arises. With this methodology, there will soon be no spectrum left for reconfiguration to accommodate future technology innovations. If now, with OLI, the spectrum is divided in a flexible, open and fair way amongst technologies and services, possibly considering a multitude of operators and service providers, then this will result in a paradigm shift of the available technologies.

To take full advantage of an open physical media access, new tolerance ranges as well as new technical approaches have to be introduced. The fact of assuming reconfigurability and media sharing, leads to two basic requirements: tunability and wavelength discrimination. Full tunability is required for a fully OLI compliant technology. Tunable optical sources are required and tunable receivers can be either realized by optical filters whose spectral shape is given by the occupied spectral band or by coherent receivers with their intrinsic arbitrary narrow reception bandwidth.

Legacy technologies can be handled by the *wavelength hotel*. Technologies aiming for ultra low cost may sacrifice large part of the spectrum by operating with broader filters and coarsely tunable optical sources. Again, as long as such systems do not disturb the co-existing systems on the same ODN, each equipment/system vendor can and will find the best compromise between component cost and system performance.

On the management layer, two network ownership models can be envisioned. A dedicated network provider may only manage the fiber infrastructure as well as the overall bandwidth assignment; a wholesale provider on the other hand may own the fiber infrastructure and provide services. In addition to the requirements mentioned in Sec. 3.1, providers will also need to address topics related to authentication, security mechanisms and network segmentation, governing of multi-level/multi-service domains, performance monitoring and fault management, regulatory, lawful interception as well as emergency communications. Agreements and designs of an OLI management system are subject to further studies [4].

5.2. Socio-economic impacts

As the OLI proposal represents a completely new landscape for optical metro access networks, many socio-economic impacts can be expected.

Incumbent service providers may expect a smooth migration to higher data rates, an easier introduction and more flexible choice of new technologies whilst encountering direct competition from alternative operators. New or alternative operators will have the unbundled access to network infrastructure and independent technology platforms on a shared fiber infrastructure. Equipment and component vendors may see open doors for new market opportunities and innovations encouraged for while facing the challenges to maintain volume pricing. Investors can rest assured that their investment is best protected by the future proof OLI environment which allows flexible utilization of technologies based on individual subscriber requirements; whereas consumers may enjoy the freedom of selecting services similar to the offerings in the wireless world. As for the regulators, OLI provides an opportunity to ensure fair competition and achieve the goals, such as those outlined in the FCC's National Broadband Plan, to offer "*affordability and maximizing use of broadband… to advance various national purposes*".

5.3. Future directions

In order to assure interoperability and guarantee full advantage of an open lambda environment, the industry will need to come to an agreement of the common set of requirements through standardization.

Standardization of PON systems has been carried out in both ITU-T and IEEE. The Full Service Access Network (FSAN) forum, lead by major global service providers, facilitates discussions amongst service providers and component/system vendors to produce whitepapers prior to consent in the ITU-T. The collaboration between ITU-T and IEEE on this topic is mainly through member contributions. These organizations are currently evaluating technologies towards future PON specifications. However, the proposed open lambda environment is not yet part of the consideration at this time. In addition, although the Broadband Forum is collaborating with FSAN on interoperability of deployed systems, architecture and technology evaluations for PON systems are not their focus.

The Open Lambda Initiative calls upon the industry to jointly discuss the benefits and challenges, and define the initial set of requirements governing such an open spectrum architecture framework. The set of requirements may include means to mitigate the physical limitations of fiber, rules for dynamic spectrum assignment, authentication, security and dynamic channel management, to name but a few. These agreed requirements may then be submitted to a standards organization for official standardization.

The expected time frame for an OLI environment may be sooner than anyone can anticipate. However, it will be up to the joint vision and discussions by the service providers, vendors, and research institutes.

6. CONCLUSION

Several existing proposals for open spectrum sharing architectures – SLICE, ROAMTS and EO-Net which exhibit similarities with the OLI architecture have been highlighted. The ultimate vision of an OLI enabled network concentrates on implementing an overarching management system which needs a solid framework with practical rules to make it function. Two practical examples of applying the OLI concept in a metro-access network and in mobile backhaul describe how support for multi-rate communication in these networks can be achieved.

To foster an OLI environment, several issues and requirements for component technologies, system design, and management layer must be carefully considered. An OLI enabled network provides several potential socio-economic impacts to incumbent and alternative service providers, to investors, to component and equipment vendors, as well as to subscribers and regulatory bodies. The path forward to standardization, timeline and challenges depend upon further discussion and collaboration amongst the different stakeholders within the industry.

In conclusion, this paper proposes an open architecture framework on a fiber medium which introduces heretofore unconsidered wavelength channel flexibility based on research on novel transceiver technologies. Furthermore, it outlines the capabilities of such a system based on co-existence of multiple technologies operating on the same fiber infrastructure.

Whilst such an approach to metro access type infrastructures is as yet not widely known, the technology will ultimately become available and enable the industry to build network infrastructures with an enhanced flexibility in comparison to today's implementations.

ACRONYMS

CAPEX	Capital Expenditure
eNodeB	Evolved NodeB (Base Station)
FCAPS	FCAPS is the ISO Telecommunications Management Network model and stands for Fault, Configuration, Accounting, Performance, Security
NE	Network Element
NGOA	Next Generation Optical Access
OPEX	Operating Expenditure
QoS	Quality of Service
ROADM	Reconfigurable Optical Add-Drop Multiplexer
ROAMTS	Reach Optimized Architecture for Multi-Rate Transport System
SLA	Service Level Agreement
TDM	Time Division Multiplexing
WDM	Wavelength Division Multiplexing
xDSL	Digital Subscriber Line

REFERENCES

[1] National Broadband Plan. The United States Federal Communications Commission. *http://www.broadband.gov/plan/*.

[2] ITU-T G.984 series specifications for Gigabit-capable passive optical networks (GPON). 2009. See more detail in *http://www.itu.int/rec/T-REC-G/en*.

[3] IEEE 802.3ah Gigabit Ethernet PON standards 2004. *http://grouper.ieee.org/groups/802/3/ah/index.html*

[4] Open Lambda Initiative forum. See more detail in *http://www.openlambdainitiative.org*.

[5] M. Jinno, H. Takara, B. Kozicki, Y. Tsukishima, Y. Son, and S. Matsuoka, "Spectrum-efficient and scalable elastic optical path network: Architecture, Benefits, and Enabling Technologies," *IEEE Comm. Mag*., vol.47 no.11, pp.66-73, 2009.

[6] A. Gumaste and N. Ghani, "Reach Optimized Architecture for Multi-Rate Transport System (ROAMTS): One Size Does Not Fit All" 25th, *IEEE/OSA Optic Fiber Communications Conference (OFC)* 2009, San Diego.

[7] Elastic Optical Networks Project. Online Reference: *http://www.celtic-initiative.org/Events/Coordinators-Day/PCO-Workshop2010/Presentations/EO-Net_presentation.pdf,* 2010

[8] J. S. Wey, C. Badstieber, and H. Rohde, "Open lambda initiative for ultra high capacity optical access networks," accepted for presentation at the *OSA ANIC Conference*, June 2010.

[9] A. Gumaste, A. Dhar and P. Gokhale, "On the State and Guiding Principles of Broadband in India," *IEEE Comm. Mag.*, August 2009.

ADAPTIVE RESOURCE ALLOCATION FOR REAL-TIME SERVICES IN OFDMA BASED COGNITIVE RADIO SYSTEMS

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ABSTRACT

In this paper an adaptive resource allocation algorithm in OFDMA based cognitive radio (CR) system is proposed that not only meets the quality of service (QoS) of real-time (RT) services but also increases the dynamic capacity of the system. In contrast to existing algorithms for multi-user OFDM systems which are unable to guarantee maximum sum data rate when applied to CR system in real-time, the proposed joint sub carrier and power allocation algorithm (JSPA) ensures maximum sum data rate under power constraint while improving the system capacity. We model the resource allocation problem with the goal of maximizing the overall system throughput. By judicially assigning resource in primary and secondary sources based on the types of application JSPA demonstrates that the system efficiency and throughput can be dynamically enhanced. JSPA is simulated under two types of services: constant bit rate (CBR) and variable bit rate (VBR).

Keywords — Cognitive Radio, OFDMA, resource allocation, real-time, Iterative water filling.

1. INTRODUCTION

The rapid growth in wireless mobile applications like web browsing, multimedia streaming etc. necessitates the development of next generation broadband wireless networks. The high bit rate in wireless transmission is restricted by inter symbol interference (ISI) because of multipath fading. This problem is basically conquered by orthogonal frequency division multiplexing techniques (OFDM) as modulation technique. Because of its flexibility and improved performance over other access techniques, OFDMA has become the de facto paradigm for the next generation wireless networks [1-3]. The high data rate requirement of many multimedia services motivates to adopt some intelligent technique to further enhance the system capacity and efficiency. Here comes the concept of cognitive radio (CR), which may be used to alleviate the looming spectrum shortage crisis [4].

The CR clubbed with orthogonal frequency division multiplexing becomes an attractive candidate for the future mobile networks. However, it is a challenging problem due to the requirement of peaceful coexistence of both primary/ licensed and secondary/unlicensed users as well as the wide range of available radio spectrum [5-6]. The fundamental tasks of cognitive radio network (CRN) are defined as spectrum sensing, spectrum management, spectrum mobility, and spectrum sharing. Our focus of attention in this paper is on spectrum management and sharing.

The OFDMA allows multiple users to transmit simultaneously on the different sub-carriers per OFDM symbol. Since the broad band channel is represented by number of narrow band sub channels, it can be assured that sub-carriers are assigned to the users based on their quality of service (QoS) requirement, as the capacity of each sub carrier can be predicted with accuracy [7]. The medium access control protocol of an OFDMA system in cognitive environment can not only meet the QoS requirement of real-time services but also increase the dynamic capacity of the system.

In this paper, to support different types of real-time applications, we define a parameter called "adaptability" which signifies the dynamic resource requirement of user defined services. This parameter is embedded in objective function as constraints. The magnitude of adaptability lies between zero and one based on service. For example, adaptability is zero for constant bit rate (CBR) services and highest for the available bit rate (ABR) services. The adaptability for variable bit rate (VBR) can take any value in the range zero to one.

The proposed JSPA algorithm maximises the system capacity by making use of cognitive (primary user's) spectrum which is dynamically shared among different secondary users. JSPA contains two major functions: sub carrier selection and power allocation. The power allocation process is based on iterative water filling algorithm. It assigns a water level for the active sub-carrier and estimates the power to be loaded on each sub-carrier. This helps in maximizing the total bit rate under the constraint of limited power at transmitter.

The remainder of this paper is organized as follows. In Section 2, a concise literature survey is presented. Section 3 presents a system model for objective function with necessary constraints in a typical OFDMA based CR system. This section also presents the derived result of optimization problem. In Section 4, JSPA algorithm is formulated. Section 5 contains simulation results and discussion. Conclusion and scope for future work is presented in Section 6.

2. RELATED WORK

The resource allocation (RA) problem is a challenging issue in OFDMA based cognitive radio network since the primary user's usage behaviour is unpredictable in many cases. The RA algorithm by Rahulamathavan et al, [8] formulates an integer linear programming problem (ILP) which is based on maximization of total data rates under interference power constraint to primary users (PU), individual data rate constraints for secondary users (SU), and total transmission power constraint at the secondary network base station. This radio resource allocation algorithm has been solved using branch and bound method.

The time-varying nature of available spectrum resources in an OFDM-based CR system could be another constraint in resource allocation. Yonghong Zhang and Leung,C [9] proposed RA algorithm for non-real time (NRT) services to ensure that CR rates are satisfied by maintaining proportionally while improving the system throughput. Their aim is to guarantee that CR user rates are maintained in proportion to predefined target rates. Although this paper is in line with our goal, it does not consider the real-time requirements of applications.

The selfishness of users complicates the critical resource allocation problem, as both parties target at maximizing their own utility. Some researchers [5,10,11] have attempted optimization framework based on Nash Bargaining solutions to fairly and efficiently address resource allocation between primary and secondary networks. Dusit Niyato and Ekram Hossain [10] considers the case of bounded rationality in which the secondary users gradually and iteratively adjust their strategies based on the observations on their previous strategies. The proposed algorithm by Hong Xu and Baochun Li [11] allows secondary users to freely optimize the use of channels for transmitting primary data along with their own data, in order to maximize performance in co-operative paradigm.

A cross layer approach by Yonghong Zhang and Cyril Leung [12] is to provide satisfactory quality of service to both real-time and non-real-time applications, despite the rapid variations in available resources caused by the activities of the primary users. Their problem-feasibility issue is addressed using a goal-programming approach.

Considering primary user activity in resource allocation for cognitive radio helps in improving the overall system performance. Ngo D. T et al., [13] has proposed a risk-return model and a general rate-loss function, which gives a reduction in the attainable throughput whenever primary users reoccupy the temporarily accessible sub-channels. They have targeted a multicast OFDMA cognitive network for their simulation.

Diego Piazza et al., [14] have proposed a resource allocation algorithm for real-time streaming in cognitive networks which is very much in line with our work. They have taken an ON-OFF channel model for the primary link, where traffic statistical characteristics are taken into account. A video traffic controller is introduced in their input traffic. Although they have measured video frame loss probability vs. throughput, their work does not consider any resource maximization/optimization techniques.

Our approach not only considers supporting QoS requirement of real-time service but also the dynamic capacity maximization in OFDMA based cognitive radio systems.

3. SYSTEM MODEL

The resource requirement of primary users is protected by enforcing the cognitive users to send their data in unused subcarriers called spectrum hole. Furthermore cognitive users have to leave transmission whenever primary users return. It is assumed that a perfect spectrum sensing mechanism is available and cognitive users do not try to interfere with primary users.

We consider a CR system with a total bandwidth of W Hz and total number of primary and secondary users K. It is assumed that one sub-carrier is used by only one Cognitive Radio User (CRU) and it is not shared.

The optimization problem (OP) can be formulated as

$$\text{OP:} \quad \max \sum_{k=1}^{K}\sum_{m=1}^{M} a_{k,m} r_{k,m} \tag{1}$$

subjected to constraints

$$\text{C}_1\text{:} \sum_{k=1}^{K}\sum_{m=1}^{M} a_{k,m}\left(2^{r_{k,m}}-1\right)\Gamma\sigma_0^{\,2}/g_{k,m} \le TS \tag{2}$$

$$\text{C}_2\text{:} \sum_{k=1}^{K} a_{k,m} = 1, a_{k,m} \in \{(0,1), m \in M; (0), m \in M'\} \tag{3}$$

$$\text{C}_3\text{:} \frac{Nsub_{allot}}{Nsub_{SN} + Nsub_{PN}} \le 1 \tag{4}$$

$$\text{C}_4\text{:} \sum_{k=1}^{K} \alpha_a = 1 \tag{5}$$

In OP $r_{k,m}$ is the number of bits per OFDM symbol which can be supported by m^{th} sub channel for k^{th} user. In (2), S is the average total power per time slot, T is the OFDM symbol duration. $(2^{r_{k,m}} - 1)\ \Gamma\ \sigma^2_0 / g_{k,m}$ is the power needed for OFDM symbol to support data rate $r_{k,m}$ with Γ is the SNR gap and σ^2_o is the noise power. In (3) M(M') Є {1, 2, …M} denotes the set of available (unavailable) sub-carriers. In (4), $Nsub_{allot}$, $Nsub_{SN}$, $Nsub_{PN}$ are the number of sub-carrier allotted, number of sub-carrier in secondary network, and number of sub-carrier in the primary network respectively. In (5) we define a term α_a and it refers to the adaptability of an application which can take any real value in the range [0,1]. We try to classify various RT and NRT services within this range.

For solving OP we relax constraint C_2 to allow $a_{k,m}$ to take on real values in [0,1] and we use the $r_{k,m} = r_{k,m} / a_{k,m}$

transformation and the constraints C_3 and C_4 are expressed as follow:

$$C_3\text{'}: \frac{\sum_{i=0}^{Nsub_{allot}} r_i}{\sum_{i=0}^{Nsub_{SN}} r_i + \sum_{j=0}^{Nsub_{PN}} r_j} \le 1 \quad (6)$$

$$C_4\text{'}: \sum_{k=1}^{K} n_k r_{k,m} = 1 \quad (7)$$

In C_3' r_i is the rate function for i^{th} sub-carrier in secondary network and r_j is the rate function for j^{th} sub-carrier in primary network. In C_4' n_k is a constant factor which indicates number of times the basic rate required by an application. We make use of Karush-Kuhn-Tucker (KKT) conditions [15] to solve this OP.

In general, for m^{th} sub-carrier the optimal sub-carrier allocation strategy is initialized with $r_{k,m} = 0$ when $a_{k,m} = 0$; and when $a_{k,m} = 1$, we derive the optimal bit loading strategy as

$$r_{k,m} = \log_2\left(\frac{\beta_k g_{k,m}}{\Gamma \sigma_o^{\ 2}}\right) \quad (8)$$

where $$\beta_k = \frac{1 + \lambda(\kappa_{i\,j}) + \varphi \sum_{i=0}^{K} n_i}{\gamma \ln 2} \quad (9)$$

where γ, λ, φ are the Langrangian multipliers for C_1, C_3' and C_4' respectively.

we ensure that no power is allocated if the equivalent noise, $\Gamma\ \sigma^2_0\ /\ g_{k,m}$ is higher than the water level, otherwise the power allotted is: $\beta_k - \Gamma\, \sigma^2_0 / g_{k,m}$. Based on the water level{ β_k , k=1, 2, ….K } bit loading on sub-carrier is performed which is governed by (8).

4. FORMULATION OF JSPA

The simulation set up of JSPA consists of CBR and VBR services as input traffic. Because of randomness in user traffic, an iterative approach is implemented while observing any system parameter for performance evaluation. The simulation set up comprises physical and medium access control layer of typical cellular system.

After initialization, the sub-carriers are sorted based on their prevailing SNR. Sum rate is computed to estimate the number of sub-carriers. We proceed by estimating a global water level for all users. During sub-carrier allocation, we perform the calculation of water level for every user by making use of iterative water filling algorithm. This is used to calculate the power loaded on each sub-carrier.

Step1: Initialise K, N, α_a and $g_{user,n}$. Here $g_{user,n}$ is the channel gain for the n^{th} user.

Step2: Sort in descending order $g_{i,j}$ {i=1,2,...K}, {j=1,2....N}.

Step3: Calculate the total sum rate for all users *sumrt*+ = r_k for k = {1,2...K} employing (8).

Step4: Calculate the proportional number of sub-carriers to be allocated to every k^{th} user, number of sub-carriers in secondary network and number of sub-carriers in available primary network. N_s contains the proportional number of sub-carriers that need to be allocated to user from secondary network by considering their requirements: N and *sumrt*.

$Nsub_k = (r_{d,k} / sumrt)N$;

$Np_k = (1 - \alpha_{a,k}) Nsub_k$;

$Ns_k = \alpha_{a,k} Nsub_k$;

$\forall k \subseteq K$

Before proceeding to sub-carrier allocation phase we need to perform the initialisation of global water level where $\beta^{\infty}_{k,m} = \{k = 1,2,...K\}$ is the global water level.

Step5: Initialise water level $\beta_{k,m}$ and power loaded on each sub-carrier $P_{k,m}$ where $k \subseteq K, m \subseteq M$

$\beta_{k,m} = S / N$;

$P_{k,m} = \beta_{k,m} - (\Gamma \sigma_0^2 / g_{k,m})$;

Step6: Here we calculate the difference between the water level and noise factor $\Gamma \sigma_o^{\ 2} / g_{k,m}$ of each sub-carrier.

Let ξ denotes the number of sub-carriers from secondary network that is to be allocated to each user in order to satisfy the rate requirement. This count should be approximately less than or equal to *Ns* for a particular user. If the averaged sum power considering all sub-carrier is less than *S* then water level for every sub-carrier allocated to k^{th} user is increased by a term *S/C* where *C* >1 and r_k is calculated with respect to new water level.

If the average power is greater than *S* then water level for every sub-carrier allocated to k^{th} user is increased by a factor *S/C* and the value of *C* is updated to 1/*C*.

$suballoc(r_{d,k}, Nsub_{SN}, g, \Gamma, \sigma_o^{\ 2}, \xi, C)$

{

// Initialization.

$\forall k \subseteq K$ and $\forall j \subseteq N, unalloc = N$

// Calculate bit rate ($r_{a,k}$) of j^{th} sub-carrier using (8) and (9).

// Compare assigned data rate with requested data rate ($r_{d,k}$).

while $(r_{a,k} < r_{d,k})$

{

$r_{a,k} = r_{a,k} + \log_2\left(\frac{\beta_{j,k} g_{j,k}}{\Gamma \sigma_o^{\ 2}}\right)$;

$if((\xi_j \le Nsub_{SN^k}) \,\&\&(r_{a,k} < r_{d,k}))$

{

// Set 'sub-carrier allocated' bit to 1

$\chi_{j,k} = 1;$

// Incrementing assigned sub-carriers to k^{th} user.

$\xi_j = \xi_j + 1;$

// Decrement number of available sub-carriers.

$unalloc = unalloc - 1;$

$diff = \beta_{j,k} - \left({g_{j,k}}/{\Gamma\sigma_o^2} \right);$

$avg\_pow = unalloc * diff;$

/* Compare average power with total transmit power available */

$if(avg\_pow \leq S)$

{

$\forall j \subseteq J$

/* Increment water level for j^{th} sub-carrier allocated to each of k^{th} user in the secondary network. */

$\beta_{j,k} = \beta_{j,k} + S/C;$

$P_{j,k} = P_{j,k} + S/C;$

// Decrement total transmit power.

$S = S - S/C;$

}

else

{

$\forall g \subseteq K, \forall l \subseteq N$

{

// For all previously allocated sub-carriers for k^{th} users.

$if(\chi_{g,l} = 1)$

// Decrement water level

$\beta_{g,l} = \beta_{g,l} - S/C;$

// Increment available transmit power

$S = S + S/C;$

$P_{g,l} = P_{g,l} - S/C;$

}

$C'' = (1/C);$

$C = C'';$

}

}

}

/* The same procedure is repeated for primary network with suballoc called with same parameters except for ξ . Let ζ denotes the number of sub-carriers from primary network that is allocated to each user in order to satisfy the rate requirement. */

$if((\zeta_i > Np_k) \| (r_k < r_k'))$

$suballoc(r_{d,k}, Nsub_{PN}, g, \Gamma, \sigma_o^2, \zeta)$

5. SIMULATION RESULTS

We consider a multi-user OFDM-based CR system with uniform distribution of users. In simulation the number of users in the system were varied from 2-10. In addition, SNR gap Γ=3.162dB, total transmit power S_t =10W, total bandwidth W = 5 MHz, number of sub-carriers N=128 and 128-QAM are assumed.

First, we simulate the capacity of JSPA under two different input traffic: CBR and VBR. As shown in Fig.1, the capacity remains same for lower number of users (upto 4) and when number of users increase, the capacity corresponding to VBR service increases which is attributed to the dynamic behaviour of JSPA towards VBR services.

We also estimate the throughput (sum data rate) of JSPA (Fig.2). For lower number of users, there is no change in throughput but for higher number of users interestingly CBR shows higher throughput and this is due to the ease in mapping user data rate requirement onto the sub-carriers.

In Fig.3, we observed the CPU time on Intel Pentium4 at 2.2GHz with 4GB RAM. As shown in Fig.3, JSPA handles VBR services more effectively compared to CBR.

6. CONCLUSION

We have simulated JSPA under CBR and VBR services. It is concluded that JSPA handles VBR services more effectively and provides better efficiency. Although JSPA shows slightly lower throughput for VBR services, the higher system capacity overtakes this shortcoming.

Estimating bit error rate of OFDMA CR system employing JSPA in presence of fading channel could be the future work of this research.

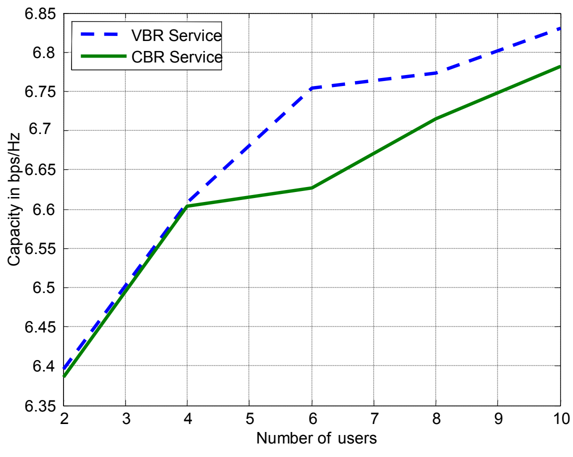

Fig.1: Simulated average capacity of JSPA.

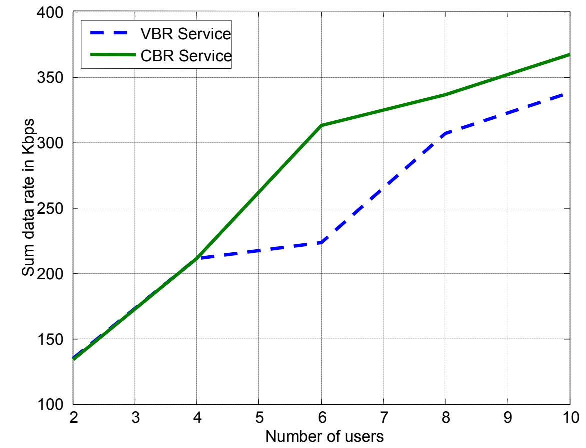

Fig.2: Average sum data rate.

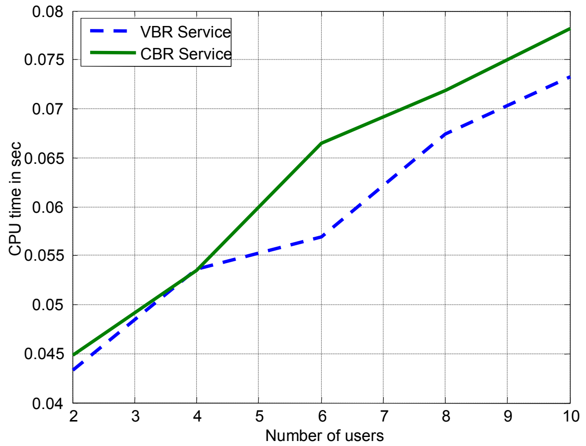

Fig.3: Average CPU time vs. number of users

REFERENCES

[1] Megumi Kaneko, Petar Popovski, and Kazunori Hayashi, "Throughput-Guaranteed Resource-Allocation Algorithms for Relay-Aided Cellular OFDMA System", Vol.58, No.4, pp. 1951-1964, May 2009.

[2] Mylene Pischella and Jean-Claude Belfiore, "Resource Allocation for QoS-Aware OFDMA Using Distributed Network Coordination", *IEEE Transaction on Wireless Communications*, Vol. 58, No. 4, pp. 1766-1775, May 2009.

[3] Jun Xu, Xuemin (Sherman) Shen, Jon W. Mark, and Jun Cai, , " Real-Time Video in OFDM Wireless Systems", *IEEE Transaction on Wireless Communications*, Vol. 7, No. 4, pp. 1417-1427, July 2008.

[4] Rui Wang, Vincent K. N. Lau, Linjun Lv, and Bin Chen, "Joint Cross-Layer Scheduling and Spectrum Sensing for OFDMA Cognitive Radio Systems", ", *IEEE Transaction on Wireless Communications*, Vol. 8, No. 5, pp. 2410-2416, May 2009.

[5] Dusit Niyato, and Ekram Hossain, "Competitive Spectrum Sharing in Cognitive Radio Networks: A Dynamic Game Approach", *IEEE Transaction on Wireless Communications*, Vol. 7, No. 7, pp. 2651-2660, July 2008.

[6] Zhiqiang Li, F. Richard Yu, , and Minyi Huang, "A Distributed Consensus-Based Cooperative Spectrum-Sensing Scheme in Cognitive Radios", *IEEE Transaction on Vehicular Technology*, Vol.59, No.1, pp. 384-392, January 2010.

[7] Kumar, Dhananjay. Srividhya, S. Mariappan, P. Martheeswaran, M. Chellappan, C., "Dynamic resource management for downlink multimedia traffic in OFDMA cellular networks" *ITU-T Kaleidoscope Event*: Innovations for Digital Inclusions, (K-IDI 2009) Aug. 31 - Sept. 1, 2009.

[8] Rahulamathavan, Y, Cumanan, K, Musavian, L, Lambotharan, S. "Optimal Subcarrier and bit allocation techniques for cognitive radio networks using integer linear programming' 15th *IEEE Workshop on Statistical Signal Processing*, pp 293-296, Aug 31 2009-Sep 3 2009.

[9] Yonghong Zhang and Leung,C, "Resource allocation for non-real-time services in OFDM-based cognitive radio systems" *IEEE Commun Letter* vol. 13 no.1 pp.16-18 Jan 2009.

[10] Dusit Niyato and Ekram Hossain "Competitive Spectrum Sharing in Cognitive Radio Networks: A Dynamic Game Approach", *IEEE Transaction on Wireless Communications*, Vol. 7, No. 7, pp. 2651-2060, July 2008.

[11] Hong Xu and Baochun Li, "Efficient Resource Allocation with Flexible Channel Cooperation in OFDMA Cognitive Radio Networks", *IEEE INFOCOM 2010.*

[12] Yonghong Zhang and Cyril Leung, "Cross-Layer Resource Allocation for Mixed Services in Multiuser OFDM-Based Cognitive Radio Systems", *IEEE Transaction on Vehicular Technology*, Vol.58, No.8, pp. 4605-4619, October 2009.

[13] Ngo D. T, Tellambura C, and Nguyen H, "Resource Allocation for OFDMA-based Cognitive Radio Multicast Networks With Primary User Activity Consideration", *IEEE Transactions on Vehicular Technology*, Vol. 1, No. 99, February 2010.

[14] Diego Piazza, Pamela Cosman, Laurence B. Milstein and Guido Tartara, "A Resource Allocation Algorithm for Real-Time Streaming in Cognitive Networks", *IEEE Wireless Communication and Networking Conference (WCNC)* April 2009.

[15] S. Boyd and L. Vandenberghe, *Convex Optimization*. Cambridge University Press, 2004, www.stanford.edu/~boyd/cvxbook/bv_cvxbook.pdf.

ALL PHOTONIC ANALOGUE TO DIGITAL AND DIGITAL TO ANALOGUE CONVERSION TECHNIQUES FOR DIGITAL RADIO OVER FIBRE SYSTEM APPLICATIONS

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ABSTRACT

Wideband electronic analogue to digital conversion (ADC) systems have critical problems encountered in high-frequency broadband communication systems that the recent electronic ADCs (EADC) have experienced those such as uncertainty of sampling time. In this paper, an all photonic sampling and quantization ADC and photonic digital to analogue conversion system with six effective number of bits (ENOB) is designed. By using this photonic ADC (PADC), a novel digital radio over fibre link for wireless radio frequency (RF) signal transportation over 20 Km single mode fibre has been designed whose performance is investigated in this paper. In the digital radio over fibre, the dynamic range is independent of the fibre length.

Keywords– Photonic ADC, Radio over Fibre, ENOB, Wireless Communication.

1. INTRODUCTION

Radio over fibre technology is currently receiving large attention due to its ability to provide simple antenna front ends, increased capacity, and wireless access coverage. Radio-over-Fibre system (RoF) is the technique of modulating the radio frequency (RF) sub-carrier onto an optical carrier for distribution over a fibre network. RoF technique has been considered a cost-effective and reliable solution for the distribution of the future wireless access networks by using optical fibre with vast transmission bandwidth capacity. RoF link is used in remote antenna applications to distribute signals for Microcell or Picocell base station (BS). The downlink RF signals are distributed from a central station (CS) to many BS known as Radio Access Point (RAP) through the fibres. The uplink signals received at RAP are sent back to the CS for any signal processing. RoF has the following main features: (1) it is transparent to bandwidth or modulation techniques. (2) Needs simple and small BSs. (3) Centralized operation is possible. New wireless subscribers are signing up with an increasing demand of more capacity for ultra-high data rate transfer at speeds of 1 Gbps and up, while the radio spectrum is limited. This requirement of more bandwidth allocation places heavy burden on the current operating radio spectrum and causes spectral congestion at lower microwave frequency. Millimetre Wave (mm-Wave) communication system offers a unique way to resolve these problems [1].

Digital signal processing has revolutionized modern communication systems by offering unprecedented performance and adaptivity. Since digital systems are flexible and more conveniently interface with other systems, and are more reliable and robust against additive noises of devices and channel and achieve better dynamic range than analogue systems. Analogue to digital and digital to analogue converters (ADC and DAC, respectively) are the link between the analogue world and the digital world of signal processing and data handling. In an analogue system the bandwidth is limited by devices performance and parasitic components introduced. Thermal noise generated in active and passive components limits the dynamic range of an analogue system. The ratio between the maximum allowable analogue signal and the noise level determines the dynamic range of the system.

Wideband analogue to digital conversion is a critical problem encountered in broadband communication and radar systems. The recent electronic analogue to digital conversion systems experience problems such as jitter in sampling clock, settling time of the sample and hold circuit, speed of comparator, mismatches in the transistor thresholds and passive component values. These limitations imposed by all of these factors become more severe at higher frequencies. Photonic ADCs by using the Mode-locked laser (MLL) and Mach Zehnder Modulator (MZM) are able to scale the timing jitter of the laser sources to the sub-femtosecond level, which will allow the designers to push the resolution bandwidth by many orders of magnitude beyond what electronic sampling systems can achieve currently [2].

This paper is organized as follows: Section 2 describes analogue RoF (ARoF) and Section 3 introduces the digital RoF (DRoF) system and its architecture. Section 4 provides our system simulation results. Finally, conclusions are presented in Section 5.

2. ANALOGUE ROF SYSTEM

Figure 1, shows an analogue RoF (ARoF) link that includes optical source, modulator, optical amplifier & filters, optical channel and photodiode as a receiver and electronic amplifiers and filters. In some cases the optical source is directly modulated by RF signal, but as the laser is usually a significant source of noise and distortion in a radio over fibre link, so laser diode normally exhibits nonlinear behaviour. When it is well driven above its threshold current, its input/output relationship can be modelled by Volterra series of order 3, [3]. So, in the recent RoF systems, semiconductor laser is used for the optical source and external modulator. Therefore, analogue optical link suffer from nonlinearity of both microwave and optical components that constitute the optical link.

Usually a Mach Zehnder interferometer fabricated from LiNbO3, impress the RF signal on the optical intensity. Also, a wide range of other modulator can be used [4][5].

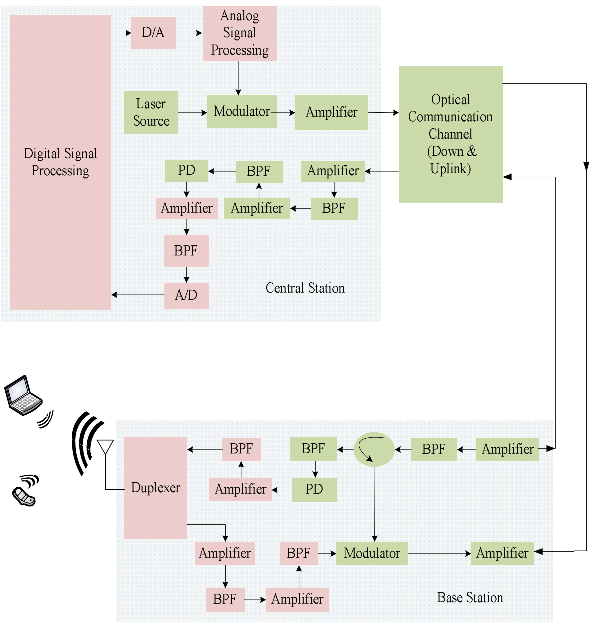


Figure 1: A generic Analogue RoF Link.

3. DIGITAL ROF SYSTEM

The digital RoF (DRoF) link can maintain the dynamic range more independent than optical fibre link distance and can employ the present infrastructure for transporting the digitized radio traffic, [6][7]. Figure 2 shows the proposed converged analogue and digital radio over fibre system. In this system, the analogue RF signal has been digitized by using the PADC. With generation of digital data and proportional multiplexing technique, data stream is transported over optical fibre network.

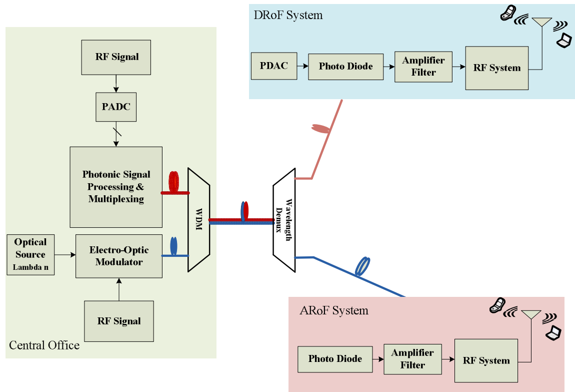


Figure 2: The proposed downlink of converged ARoF and DRoF systems.

In this design, both the baseband and digitized RF data traffic signals are transported through metro and access networks by using wavelength division multiplexing technique. For compensating the chromatic dispersion, the system uses the chromatic dispersion compensation fibre. The photonic DAC of RF system converts digital photonic signal to its analogue version of optical modulated signal. The RF signal is detected by single high speed photo diodes. The RF signal after processing by an electronic RF signal processing system has been fed to the base station transmitter antenna.

3.1. Photonic Analogue-to-Digital Conversion

High-speed sampling and quantization of photonic analogue-to- digital conversion (ADCs) systems have wide variety of applications in todays high-speed signal processing electronic circuits and communication systems, [2][6][8]. Figure 3 shows the proposed photonic sampling and quantization ADC system architecture.

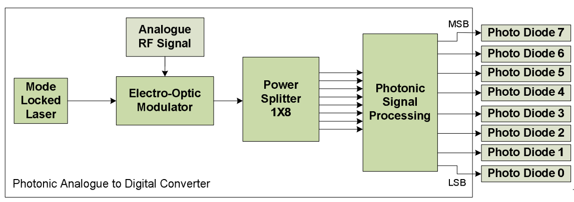


Figure 3: The photonic ADC's architecture.

The digitized optical signal's power have been split into 8 channels and fed to photonic signal sampling and quantization block. In this block, the digitized signals have been processed, multiplexed and combined together for transporting over single mode fibre medium. In this architecture, eight-bit photonic ADC has been designed. For performance investigation of PADC, the photonic quantized signals have been converted to electrical signals by using a high-speed photo diode array.

Despite the variety in ADC's, their performance can be summarized by a relatively small number of parameters, first: stated resolution (number of bits per sample), second: signal to noise ratio (SNR), third: spurious-free dynamic range (SFDR) and forth: power dissipation. SNR and SFDR are the most important parameters of dynamic performance for high-speed applications. The SNR and SFDR provide a more accurate measure of ADC performance than the stated number of bits. The SFDR is the ratio of the single-tone signal amplitude to the largest non-signal component within the spectrum of interest. The noise spectrum contains contributions from all the error mechanisms present. These include the noises of quantization, circuit, aperture, and aperture uncertainty and comparator ambiguity. The only error mechanism present in an ideal ADC is quantization, [9].

The Q denoted as the quantization error is the difference between the analogue signal and digital sampled signal, which is the least significant bit (LSB) of the binary representation of that value given by Equation 1.

$$Q = \frac{V_{FS}}{2^N} \qquad (1)$$

That value of N shows the ADC resolution bits and V_{FS} is the full scale voltage. In this design N=8 bits and V_{FS}=1 Volt. Therefore, Q is 125 mV. To simplify the calculations, the error of quantization can be defined as a simple linear function during a sampling interval. Equation 2 denotes the quantization error function and the noise rms power given by Equation 3.

$$e(t) = Q.\left(\frac{t}{T} - \frac{1}{2}\right) \qquad (2)$$

$$NP_0(rms) = \sqrt{\frac{\int_0^T [e(t)]^2 dt}{T}} = \frac{Q}{12} \qquad (3)$$

and the SNR calculation is given by Equation 4.

$$SNR(dB) = 20log_{10}\left(\frac{V_{FS}(rms)}{NP_0(rms)}\right)$$

$$= 6.02.N + 1.76\ dB \qquad (4)$$

So, the SNR can be improved by increasing N. The effective number of bits is given by Equation 5:

$$ENOB = \frac{SFDR(dBc)}{6.02} \qquad (5)$$

3.2. Base Station

Figure 4, shows the block diagram of the down-link section of photonic and RF system at the base-station for the proposed DRoF architecture. In this system, the photonic DAC converts photonic digital waveform to analogue optical signal. By using the proposed PDAC, the necessity of optical to electrical conversion will be fulfilled by only a high-speed photo diode. For reducing the cost and complexity of the system, the optical sampling pulse has been reused at base-station for sampling the uplink RF signal. Therefore, the need for new optical carrier source and sampling pulse has been satisfied. In the uplink data transportation, the same technique has been implemented.

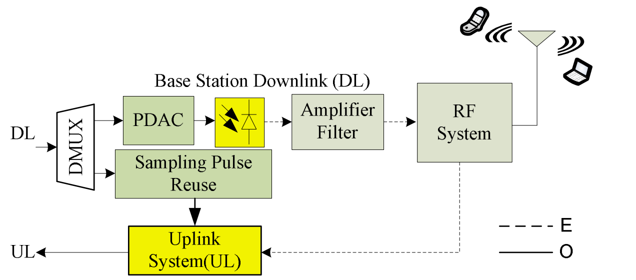


Figure 4: The Base-Station System that uses PDAC.

By realizing this scheme, it is possible to use free spectrum capacity of metro and access networks for transporting the broadband wireless and wireline data traffic. This technique centralises the signal processing, system management, and monitoring processes. Therefore, wireless network could be integrated with existing optical networks that reduces the future super-broadband access-network-system implementation overheads and service costs to the end-users.

4. SIMULATION RESULTS

In this section, the results of simulations that have been performed by Optiwave-Optisystem and Matlab environments are presented. According to the calculations and simulations the SNR of quantized signals are equal to 49.92 dBm. By noticing the simulation results that are shown in Figure 5, the SFDR and ENOB at 160 Gigasample/s sampling rate are equal to 9.82 dB and 1.63, respectively.

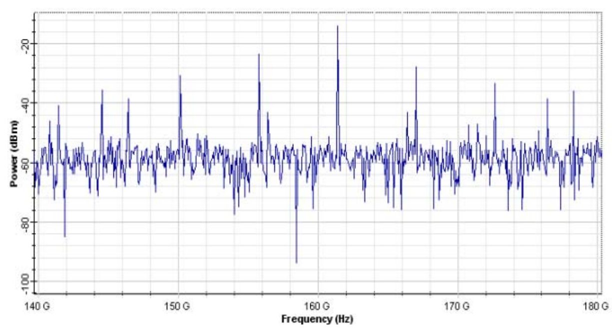


Figure 5: Power spectrum of the digital electrical signal.

For selecting the optimum sampling rate, the PADC performance has been investigated for various sampling rates. Figure 6, shows the ENOB variations with sampling rate. The best ENOB resolution is evaluated at 80

Gigasample/s about 6. As the transfer function of MZM is inherently nonlinear. The dynamic range has been converted into ENOB and plotted against the RF link sampling rates.

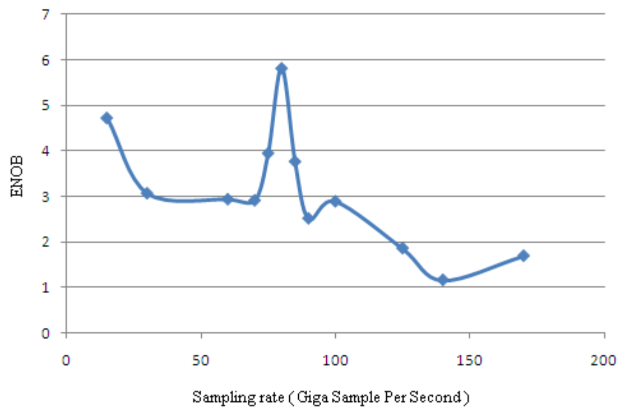
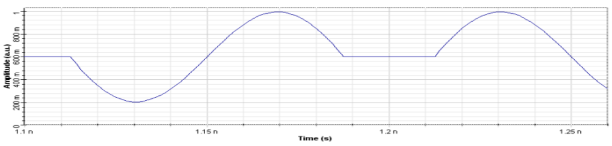
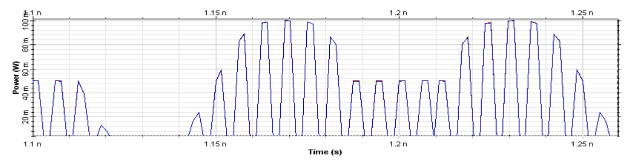


Figure 6: ENOB variation versus sampling rate.

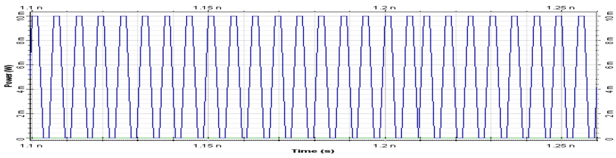
The simulations show that using a nonlinear modulator severely limits the PADC resolution for practical applications. Figure 7, shows a sample RF signal and the sampled signal and two MSB and LSB quantized digital optical signals. In this design, sampling and quantization have been implemented in optical domain. As illustrated in the figure, the 160 GHz optical sampling pulse is generated by a 1550 nm mode-locked laser and a sample 15 GHz, RF signal has been fed to MZM electro-optical modulator as a photonic sampling device. Another limitation factor for PADC's performance is the effect of laser's jitter. If the laser pulses do not occur exactly when they should be, the signal will be sampled at the wrong time, and so the digitized output will differ from the signal at the assumed ideal jitter-free sampling time. The effects of jitter is illustrated in Figure 7.



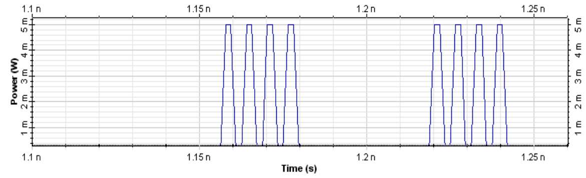
(a)



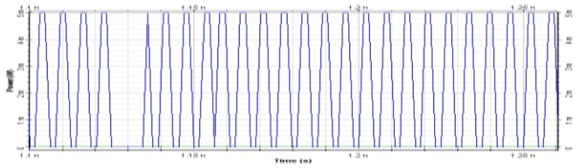
(b)



(c)

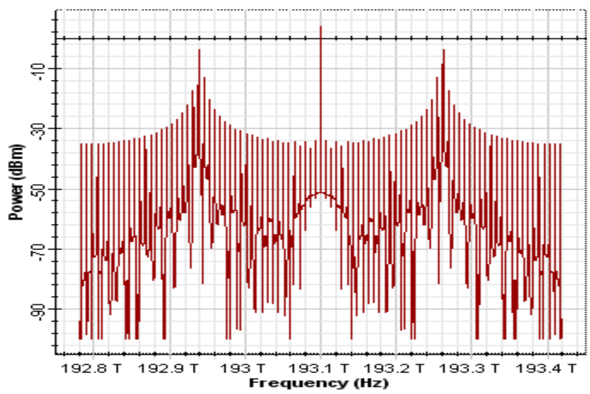


(d)

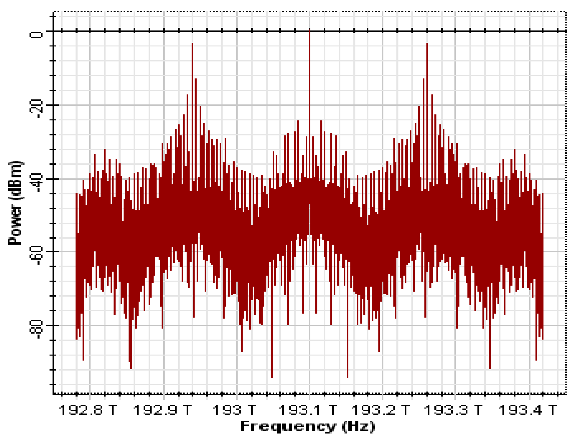


(e)

Figure 7: (a) RF signal, (b) Sampled optical signal, (c) Sampling Clock pulse , (d) Optical signal of MSB, (e) Optical signal of LSB.



(a)



(b)

Figure 8: (a) Optical power spectrum of Mode Locked Laser. (b): Optical power spectrum of quantized signal.

Figure 8, shows the power spectrum of 160 Gigasample/s optical sampling pulse with 1550 nm mode locked laser.

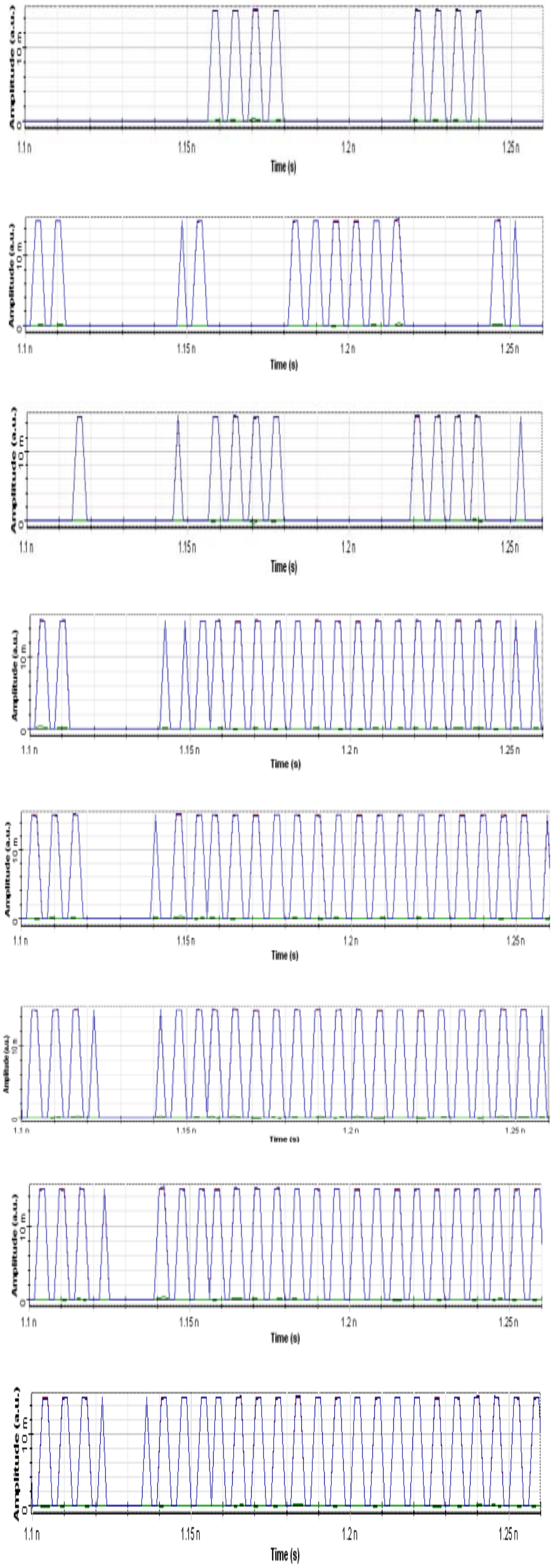
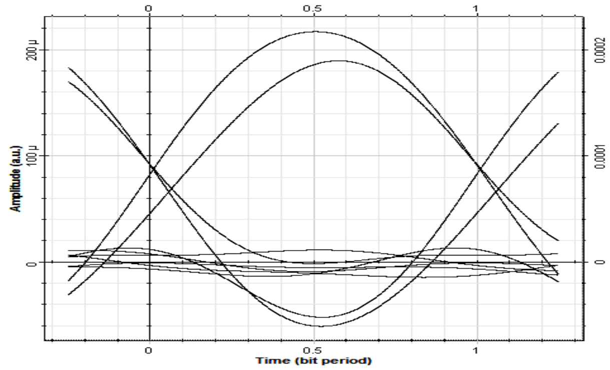


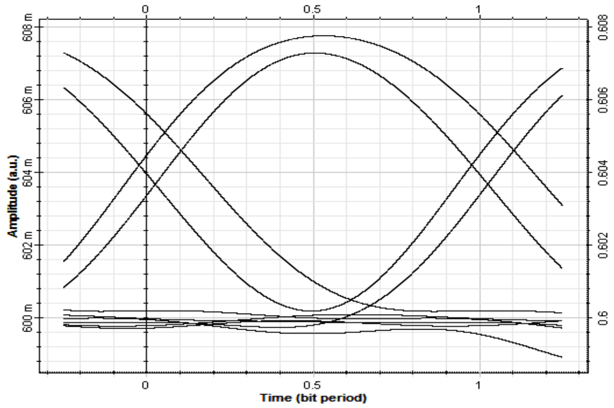
Figure 9: Digital electrical waveform of photonic sampled and quantized optical signal.

Figure 9 shows the digital waveforms of sampled RF signal. The data retiming and regeneration have been performed in optical domain by using the MLL signal. These RF digitized traffic data has been transported by using wavelength division multiplexing technique over fibre network infrastructure.

For investigation of the performance of the proposed digital radio over fibre system (DRoF), a sample 1 Gbps NRZ pseudo-random data is ASK modulated with 10 GHz RF carrier digitized with 30 Gigasample/s PADC and transported by the proposed DRoF system. Figure 10, shows the transmission system performance with the eye diagram of the received digital system.



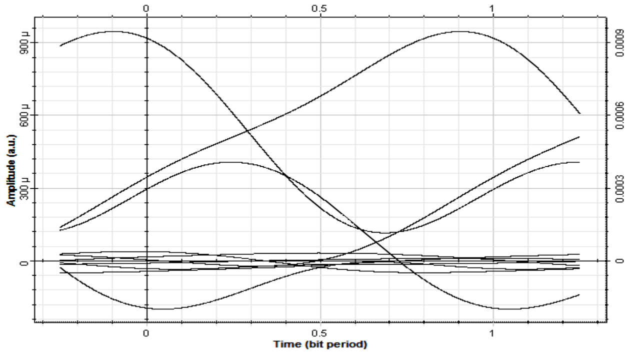
(a)



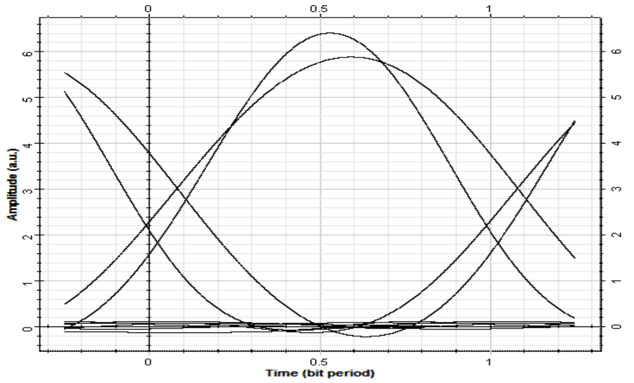
(b)

Figure 10: Eye diagram of 1 Gbps ASK modulated signal with 10 GHz carrier over 15 km length of single-mode fibre: (a) ARoF system, (b) DRoF system.

In this figure, the performance comparison of received signal by using digital and analogue systems has been shown over 15 kilometres dispersion compensated single mode fibre length. In this system, the single mode fibre chromatic dispersion is assumed about 17 ps/(nm.km). Figure 11, shows detected 1 Gbps electrical signal of the (Analogue Radio over Fibre) ARoF and DRoF systems after transporting through 20 km single mode fibre, by using no perfect chromatic dispersion compensation fibre, the total fibre link dispersion has been reduced to 5.75 ps/(nm.km).



(a)



(b)

Figure 11: Eye diagram of 1 Gbps ASK modulated signal with 10 GHz carrier over 20 km length of single mode fibre: (a) ARoF system, (b) DRoF system.

5. CONCLUSIONS

In this paper, the proposed DRoF and conventional ARoF systems are simulated by Optiwave-Optisystem and Matlab simulation tools for super-broadband wireless signal transportation and distribution applications over 20 Km single mode fibre. The performance of the DRoF link is investigated and compared with ARoF. In the DRoF system digitizing the analogue radio frequency (RF) signal and finally returning it back into analogue at destination is done by using our designed all photonic 8-bit ADC with 30 Gigasample/s MLL and DAC convertors. PADC's performance is affected by the laser's jitter, the nonlinearity of MZM, photonic amplifier and other photonic devices performance. In the digital radio over fibre, the dynamic range is independent of the fibre length.

REFERENCES

[1] H-C Ji, H. Kim, and Y. C. Chung, " Full-duplex radio-over-fiber system using phase-modulated downlink and intensity-modulated uplink", *IEEE Photonics Technology Letters*, vol. 21, no. 1, pp. 9-11, Jan. 2009.

[2] G. C. Valley, "Photonic analog-to-digital converters," *Journal of Lightwave Technology*. vol. 15, no. 5, pp. 1955-1982, 2007.

[3] H. Al-Raweshidy and S. Komaki, Radio over Fiber Technology for Mobile Communication Networks, Artech House, 685 Canton Street, MA 02062, 2002. pp 136-138.

[4] C. H. Cox, III, Analog Optical Links, Cambridge University Press, Cambridge UK, 2004.

[5] G. L. Li. and P. K. L. Yu, "Optical intensity modulators for digital and analog applications," *Journal of Light wave Technology*. vol. 21, pp. 2010-2030, 2003.

[6] P. A. Gamage, A. Nirmalathas, C. Lim, D. Novak and R. Waterhouse, "Design and analysis of digitized RF-Over-Fiber Links," *Journal of Lightwave Technology*, vol. 27, no. 12, pp. 2052-2061, 2009.

[7] C. Lim, A. Nirmalathas, M. Bakaul, P. Gamage, K. L. Lee, Y. Yang, D. Novak and R. Waterhouse, " Fiber-Wireless Networks and Subsystem Technologies," *Journal of Lightwave Technology*, vol. 28, no. 4, pp. 390-405, 2010.

[8] J. Kim, M. J. Park, M. H. Perrott and F. Kartner, " Photonic subsampling analog-to-digital conversion of microwave signals at 40-GHz with higher than 7-ENOB resolution," Optics Express, vol. 16, no. 21, pp. 16509-16515, 2008.

[9] R. H. Walden, " Analog-to-digital converter survey and analysis," *IEEE Journal of Selected Areas in Communications*, vol. 17, no. 4, pp. 539-550, 1999.

ENHANCING CYBERSECURITY FOR FUTURE NETWORKS

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ABSTRACT

Next Generation Networks, including specialized implementations such as Cloud Computing and Smart Grids, must be significantly more robust than today's networks – with "baked in" capabilities supporting an array of assurance and cybersecurity capabilities. One of the most significant emerging means of achieving these capabilities is by applying an ensemble of new specifications being developed under the aegis of ITU-T Study Group 17 known as CYBEX – the Cybersecurity Information Exchange Framework. This paper describes the CYBEX Framework, how it came into existence, and its potential application to Future Networks.

Keywords— Cybersecurity, CYBEX, NGN, Cloud Computing, Future networks, SmartGrids

1. INTRODUCTION

One of the most essential as well as challenging set of requirements for all potential species of Future Networks is to achieve significantly greater robustness and security than exists in today's PSTN and IP based networks. These requirements are often simply referred to as cybersecurity and apply to all the many existing and potentially new specialized network implementations such as Cloud Computing and SmartGrids.

Although the exact architectures and platforms of Future Networks remain unclear, and no doubt will be subject to constant evolution, it is entirely feasible to develop sets of extensible specifications to achieve three enduring network requirements: 1) "locking down" the ensemble of network infrastructure, applications, and attached devices to achieve desired assurance levels, 2) instituting "watch and warning" capabilities through that ensemble to know when adverse events or incidents occur which can be used both to defend and improve the defenses, and 3) enable evidence forensics to be handed over to law enforcement or judicial authorities when legal remedies must be pursued.

This paper describes how the emerging Cybersecurity Information Exchange Framework known as CYBEX and being pursued in ITU-T Study Group 17 (Security) provides for the three network requirement, and its potential application to Future Networks.

A visualization of the requirements is depicted in Fig. 1 with needed information exchange capabilities and related actions depicted as dashed flows. This cybersecurity "backplane" of capabilities was initially developed among the senior fellows and staff of the Georgia Tech Center for Information Security as the cyber security ecosystem and introduced into ITU and other for over the past several years. Within ITU-T Study Group 17 (Security) the ecosystem has emerged as the underpinning of CYBEX. [1]

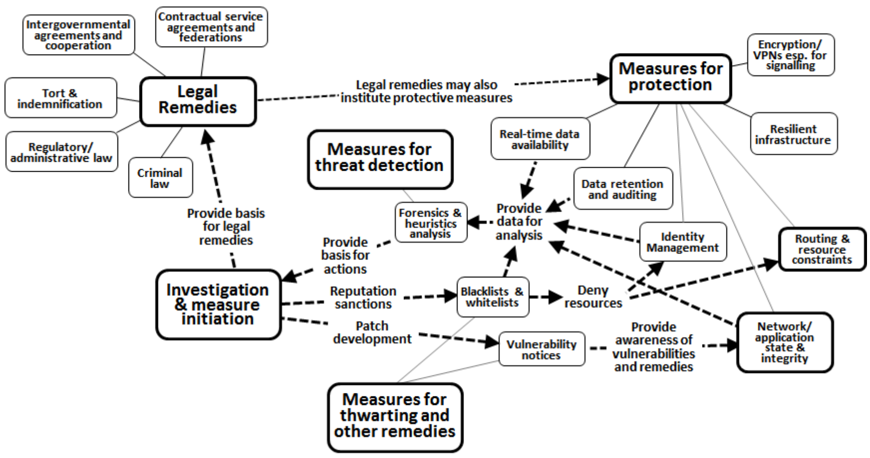

Figure 1. The Cyber Security Ecosystem

Mr. Puri and Mr. Rutkowski are senior management at Yaana Technologies. Mr. Rutkowski is also the ITU-T Cybersecurity Rapporteur (Q4/17) responsible for this subject matter and a Senior Fellow at the Georgia Tech Sam Nunn School. Yaana was a major contributor to the CYBEX material contained in this paper, and Mr. Rutkowski is the principal editor of the emerging standard. This paper describes for the first time, how CYBEX can be applied to Future Networks – linking together work across multiple forums.

2. THE CYBERSECURITY INFORMATION EXCHANGE FRAMEWORK

Although the Cybersecurity Information Exchange Framework emerged as a distinct work item in ITU-T Study Group 17 in 2009, its constituent elements have existed for more than a decade in the diverse communities of information assurance, system vendors, operators, computer incident response teams, and law enforcement authorities. [2] All of the capabilities in Fig. 1 became essential network elements for cybersecurity as IP networks became increasingly used for global network infrastructure and services. [3] The CYBEX initiative sought to identify the "best of breed" protocols and platforms actually being used or developed by user communities for cybersecurity information sharing.

The draft Recommendation ITU-T X.cybex now scheduled for approval balloting in December 2010 contains five extensible functions [4]:

- structuring cyber security information for exchange purposes
- identifying and discovering cyber security information and entities
- requesting and responding with cyber security information
- exchanging cyber security information over networks
- assured cyber security information exchanges

From a cyber security perspective, these functions were organized into seven "clusters" that were mapped to a sets of largely existing standards to meet specific needs:

- Weakness, vulnerability and state exchange
- Event, incident, and heuristics exchange
- Evidence exchange
- CIRT Policy exchange
- Cyber security organization identity and assurance
- Cyber security heuristics and information request

The basic objective is to facilitate coherent, comprehensive, global, timely, and assured exchange of cybersecurity information. This objective also includes structured global discovery and interoperability of that information in a way that allows for continual evolution to accommodate the significant activities and specification evolution occurring in numerous cybersecurity forums, including cloud computing and new applications such as SmartGrid and eHealth cybersecurity. [5]

For these purposes, the CYBEX model is a very simple one as shown in Fig. 2, below, and adapted from the customary OASIS query-response information exchange construct. [6]

Figure 2. CYBEX Exchange of Information

2.1. Cybersecurity Structured Information [7]

For the exchange of cybersecurity information to occur as messages between any two entities as shown in Fig. 2, above, it must be structured and described in some consistent manner that is understood by both of those entities. CYBEX specifications enable this exchange. The goal is to make it easier to share cybersecurity information that often includes "common enumerations," that is, ordered lists of well-established information values for the same data type. Common enumeration allows distributed databases and other capabilities to be linked together, and to facilitate cybersecurity related comparisons.

The application of the model to any arbitrary network infrastructure is depicted in Figure 3 – which was recently added to the draft Recommendation to portray what is encompassed in the specification and the external relationships.

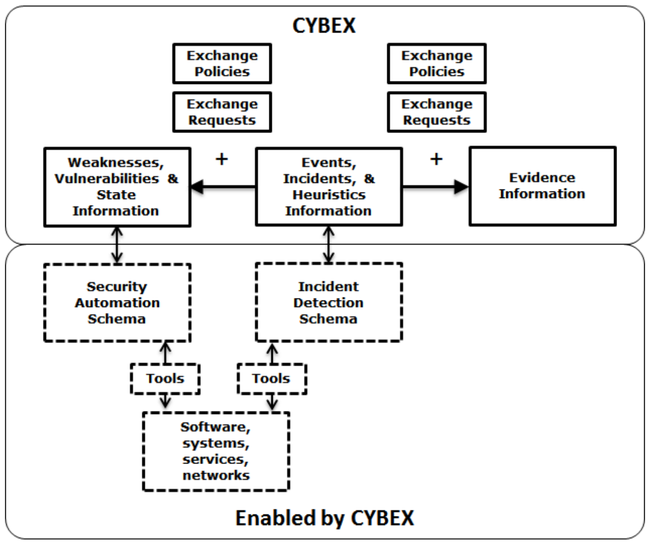

Figure 3. Application of the CYBEX Model

At present, CYBEX identifies nine specifications related to vulnerability/state exchange and five related to event/incident/heuristic exchange, with many dependencies as shown in Fig. 4, below.

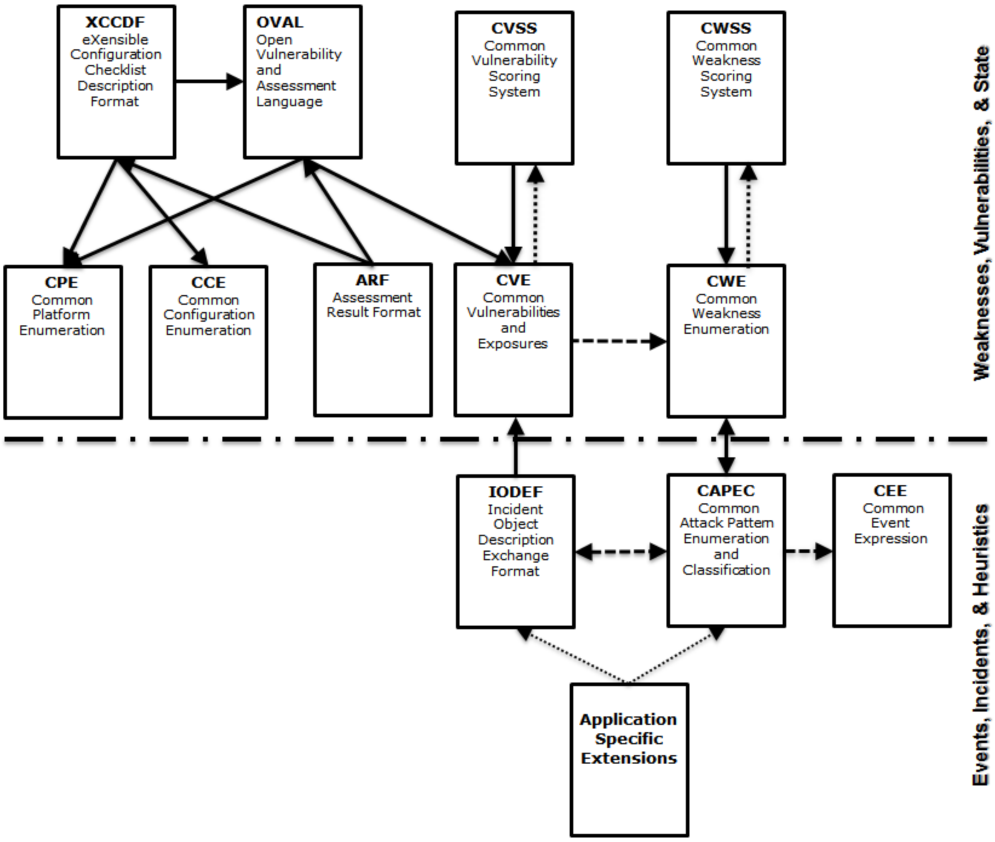

Figure 4. Relationships among assurance and incident cluster platforms

These specifications are all generic in nature and can be applied to any kind of application, system, or device – including any species of Future Network. If specialized implementations are required, extensions to the specification can be readily created and to the extent desired, made discoverable to others. The sections below contain short descriptions of each specification.

2.1.1. Weakness, Vulnerability and State Exchange

Common Weakness Enumeration (CWE). CWE is an XML/XSD based specification in widespread use for exchanging unified, measurable sets of software weaknesses that enable more effective discussion, description, selection, and use of software security tools and services that can find these weaknesses in source code and operational systems as well as better understanding and management of software weaknesses related to architecture and design.

Common Weakness Scoring System (CWSS). CWSS provides for an open framework for communicating the characteristics and impacts of software weakness described by CWE.

Common Vulnerabilities and Exposures (CVE). CVE is an XML based specification in widespread use for exchanging information security vulnerabilities and exposures that aims to provide common names for publicly known problems.

Common Vulnerability Scoring System (CVSS). CVSS provides for an open framework for communicating the characteristics and impacts of ITC vulnerabilities described by CVE.

Open Vulnerability and Assessment Language (OVAL). OVAL is an international, information security, community standard to promote open and publicly available security content, and to standardize the transfer of this information across the entire spectrum of security tools and services. It is used by most of the other specifications.

eXensible Configuration Checklist Description Format (XCCDF). CXCCDF is a specification language for writing security checklists, benchmarks, and related kinds of documents. It also is used by many of the other specifications.

Common Platform Enumeration (CPE). CPE is a structured naming scheme for information technology systems, platforms, and packages. It is used for used for binding other enumerations to specific platforms.

Common Configuration Enumeration (CCE). CCE provides unique identifiers to system configuration issues in order to facilitate fast and accurate correlation of configuration data across multiple information sources and tools. It is used to understand the state of a particular system or device implementation.

Assessment Result Format (ARF). ARF is a standardized ITC facilitates the exchange of assessment results among systems to increase tool interoperability and allow for the aggregation of those results across large enterprises that utilize diverse technologies to detect patch levels, policy compliance, vulnerability, asset inventory, and other tasks.

2.1.2. Event/Incident/Heuristics Exchange

Common Event Expression (CEE). CEE standardizes the way computer events are described, logged, and exchanged. By using CEE's common language and syntax, enterprise-wide log management, correlation, aggregation, auditing, and incident handling can be performed more efficiently and produce better results.
Incident Object Description Exchange Format (IODEF). IODEF defines a data representation that provides a framework for the exchange of information commonly exchanged by Computer Incident Response Teams (CIRTs) about computer security incidents.
Common Attack Pattern Enumeration and Classification (CAPEC). CAPEC is a sophisticated specification for the identification, description, and enumeration of attack patterns. In significantly improves on IODEF.
Phishing, Fraud, and Misuse Format. The Phishing, Fraud, and Misuse Exchange Format extends IODEF to support the reporting of phishing, fraud, other types of electronic crime.
Malware Attribution Enumeration and Characterization Format (MAEC). MAEC builds on CCE, CAPEC and other specifications as part of a formal language for characterizing malware with two core components consisting of enumerated elements (vocabulary) and schema (grammar).

2.1.3. CIRT Policy Exchange

When cybersecurity information is conveyed among parties, it is almost always subject to some understood set of policies with associated conditions. Policy expressions provided structured enumerations for this purpose.

2.1.4. Evidence Exchange

Handover Interface and Service-Specific Details (SSD) for IP delivery. SSD defines a data representation that provides a framework for the exchange of information between a network mediation point and a law enforcement facility to provide an array of different real time network forensics associated with a designated incident or event.
Handover Interface for the Request and Delivery of Retained Data. The Handover Interface for the Request and Delivery of Retained Data specification defines a data representation that provides a framework for the exchange of information between a network mediation point and a law enforcement facility to provide an array of different stored network forensics associated with a designated incident or event.
Architecture for Lawful Intercept in IP Networks. The Architecture for Lawful Intercept in IP Networks specification defines a data representation that provides a framework for the exchange of information between a network access point and a provider mediation facility to provide an array of different real time network forensics associated with a designated incident or event.
Handover for Location Services. The Handover Interface for Location Services specification defines a data representation that provides a framework for the exchange of information between a network mediation point and an external facility to provide a real-time or stored location forensics associated with a network device.
EDRM, Electronic Discovery Reference Model. The Electronic Discovery Reference Model defines a data representation that provides a framework for the exchange of information between a network mediation point and a juridical designated party to request and provide an array of different stored network forensics associated with a designated incident or event.
Digital Evidence Exchange Format. The Digital Evidence Exchange specification defines structures and data elements for structured digital evidence exchange file exchange. Electronic evidence means information and data of investigative value that is stored on or transmitted by electronic device.

2.1.5. Cybersecurity Heuristics and Information Request

Cybersecurity Heuristics and Information Request Protocol (CYIQL). CYIQL defines a flexible data representation that provides a framework for requesting information commonly exchanged by Computer Security Incident Response Teams (CIRTs) about computer security incidents.

2.1.6. Cybersecurity entity identification and discovery

Identifying and discovering parties and information in the cyber security environment remains a challenge because of the substantial diversity and insularity of communities involved. Different cybersecurity organizations are implementing common cybersecurity protocols for the capture and exchange of system state, vulnerability, incident forensics, and incident heuristics information in operational applications. Cybersecurity information exchange protocols can be used by anyone, anywhere, at any time. So there is no way to control their use.

However, common interests may exist among cybersecurity communities regarding cybersecurity identifiers and their creation, administration, discovery, verification, and use. At present, it is not clear how the challenge will be met, although some steps are being taken.

The ITU-T and ISO SC6 who jointly manage the Object Identifier (OID) namespace have allocated a new Arc (2.48) for the purposes of using OIDs to facilitate cybersecurity identification and discovery. A draft recommendation - Guidelines for Administering the OID arc for cybersecurity information exchange – has been prepared and designated X.cybex.1. The namespace would exist for cybersecurity:

- information identifiers
- organization identifiers
- policy identifiers

An additional specification is being developed to provide methods and mechanisms which can be used to identify and locate sources of cybersecurity information, types of cybersecurity information, specific instances of cybersecurity information, methods available for access of cybersecurity information as well as policies which may apply to the access of cybersecurity information.

2.2. Cybersecurity assured exchange [8]

Within the Information Exchange Framework, the achievement of necessary assurance levels and the actual exchange of structured information can occur many different ways. Several specifications have been tentatively identified for these purposes, but additional work and consensus building is necessary.

2.2.1. Assurance of Identities

Entity authentication assurance. The ITU-T and ISO are jointly developing a specification for a life cycle framework for managing the assurance of an entity's identity and its associated identity information in a given context. Four levels of assurance are recognized and the provisions draw on NIST and EU work.

Extended Validation Certificate Framework (EVcert). The EVcert consists of an integrated combination of technologies, protocols, identity proofing, lifecycle management, and auditing practices that describe the minimum requirements that must be met in order to issue and maintain EVcerts concerning a subject organization. The platform is in widespread use and the baseline specification in maintained by the C/A Browser Forum. A version of the specification is also maintained within the EU by ETSI as **Policy requirements for certification authorities issuing public key certificates.**

2.2.2. Information Exchange Protocols

Five transport level protocols are known to be used for the exchange of cybersecurity information "on the wire."

Transport Protocols supporting Cybersecurity Information Exchange. This is a new recommendation that provides an overview of exchange protocols which have been adopted and or adapted for use within CYBEX.

Blocks eXtensible eXchange Protocol Framework for CYBEX. In addition to defining BEEP's channel management profile, this document defines the TLS transport security profile, and the SASL family of profiles.

Simple Object Access Protocol for CYBEX. This specification describes how to use SOAP in combination with HTTP and HTTP Extension Framework.

Transport of Real-time Inter-network Defense (RID) Messages. This specification specifies the transport of RID messages within HTTP Request and Response messages transported over TLS.

Handover Interface and Service-Specific Details (SSD) for IP delivery. The Handover Interface and Service-Specific Details (SSD) for IP delivery specification contains protocols and their implementation for trusted delivery of forensic information to law enforcement and security authorities.

3. APPLYING CYBEX TO FUTURE NETWORKS

The Cybersecurity Information Exchange Framework described in the above section represents a major body of activity, experiences, and tools developed by information assurance, incident response, and forensic analysis. It has, however, remained largely disconnected from the telecommunication standards forums until very recently. It is a major gap in standards collaboration.

There has been little or no focus on any of the capabilities identified in the CYBEX Framework, including many of those in widespread use in other ICT communities. Even the ETSI technical report dealing explicitly with NGN Threat, Vulnerability and Risk Analysis simply provides a general discussion of potential risks and a methodology for risk analysis. [9] The only exception is the treatment of lawful interception and retained data capabilities. [10]

This section examines the emerging NGN Future Network model with the aim to begin bridging that gap. Although it is not possible to know what kind of future networks will emerge, they are likely to be IP-based at least in part. One potential widespread implementation is those modeled on the ITU-T Y-Series Next Generation Networks (NGN) specifications and the ETSI equivalent standards based on IMS implementations. [11]

3.1 Next Generation Networks

Recommendation ITU-T Y.2011 provides the general principles and general reference model for Next Generation Networks. In the NGN work that has occurred over the past decade, conceptual architectures largely based on IMS – divide the infrastructure into basic strata and planes. If this model is considered in the context of the CYBEX framework, as can be seen in Fig. 5 below, the various CYBEX functions straddle all planes and strata of the basic NGN Reference Model.

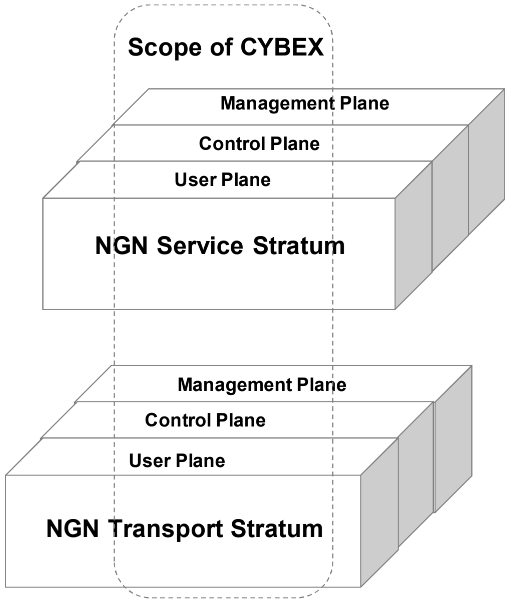

Figure 2/Y.2011

Figure 5 - CYBEX and the NGN Reference Model

In the context of NGN General Functional Model, CYBEX capabilities likely constitute some set of services that are both distinct as well as threaded through the various management and control functions to enhance their integrity as shown in Fig. 6.

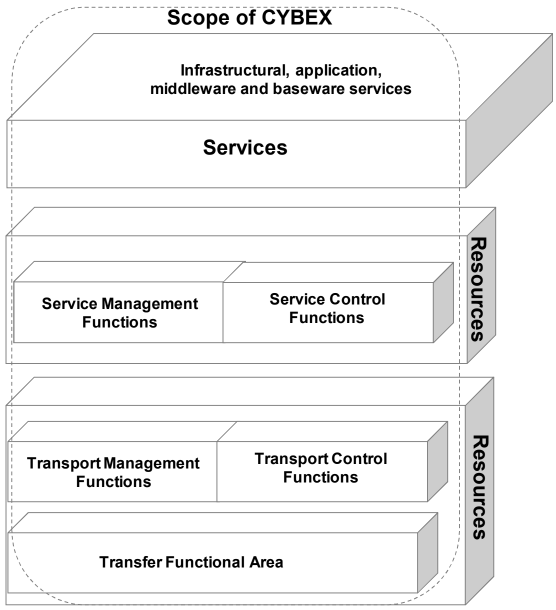

Figure 3/Y.2011

Figure 6 - CYBEX and NGN General Functional Model

Although the ITU-T NGN Recommendations have associated "Release 1" security requirements, those requirements presently are generic in nature and rely largely on trust zones without any specified means of achieving the kind of comprehensive cybersecurity ecosystem provided by the CYBEX framework. [12]

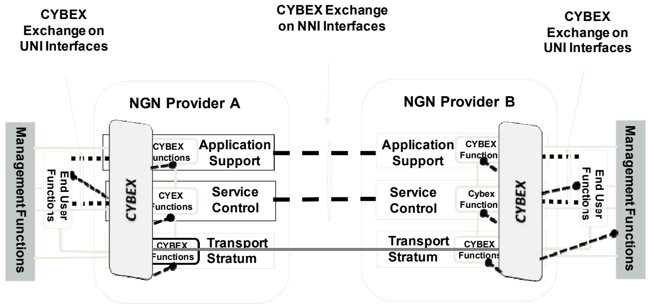

Figure 7 – A CYBEX Reference Model for NGN

A potential implementation of a CYBEX reference model for NGN is depicted in Fig. 7, above. This model is adapted from a similar approach already being taken for NGN Identity Management. [13] In this model, NGN providers would play a substantial CYBEX framework-support function with understood assurance levels among themselves and all network devices and capabilities within their domain. Under this approach, each of the CYBEX framework capabilities would be adapted as necessary through the use of extensions as follows, and reflected in a new extensible Y-series Recommendation that can be adapted to all NGN compliant future networks:

- Weakness/Vulnerability/state exchange for NGNs
- Event/incident/heuristics exchange for NGNs
- Exchange of policies for NGNs
- Evidence exchange for NGNs
- Cybersecurity heuristics and information request for NGNs
- Cybersecurity entity identification and discovery for NGNs

An example of the applying CYBEX weakness, vulnerability, and state capabilities is show in Fig. 8 below.

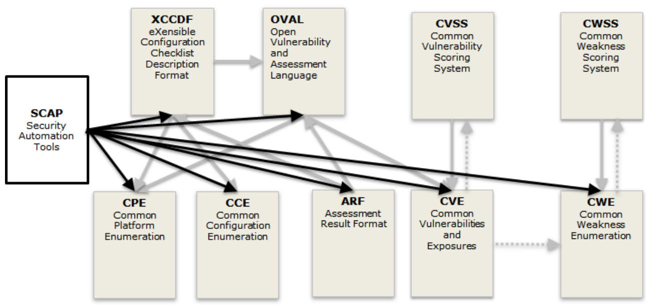

Figure 8 – SCAP for an NGN function

Here a Security Automation Protocol (SCAP) mechanism for a specific NGN function is portrayed. SCAP is simply an automated means for using the other available CYBEX based tools to be used to understand the current security state of the NGN function. [14] In deployed systems, a great many SCAP implementations would exist.

For CYBEX evidence exchange capabilities, ETSI is already in the process of developing a Dynamic Triggering specification for NGN Future Networks. [15] Fig. 9, below depicts the current reference model as applied to multiple operators where the Service and Transport Strata are divided among two operators.

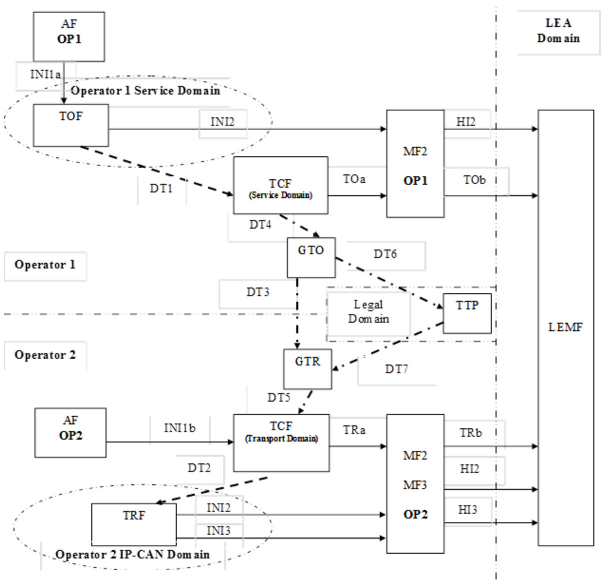

Figure 9 – Dynamic Triggering Multiple Operator Reference Model

3.2. Specialized future network implementations – Cloud Computing and SmartGrids

3.2.1. Cloud computing

Cloud Computing is basically network-based computing, whereby shared resources, software, application and information are provided to computers and other devices on-demand, like a public utility. [16] The concept was pioneered as part of the SS7-based Intelligent Network platforms designed and implemented in the 1990s, and adapted contemporaneously to IP-based networks. The architecture is synergistic with NGN reference models described above.

At present, however, the cloud computing environment has no specific accepted cybersecurity specifications, and is widely viewed as constituting significant network security risks. [17] Here also, the CYBEX framework can be readily adapted to this specialized environment by a combination of adaptations of the existing platforms as necessary for cloud computing implementations:

- Weakness/vulnerability/state exchange for cloud computing
- Event/incident/heuristics exchange for cloud computing
- Exchange of policies for cloud computing
- Evidence exchange for cloud computing
- Cybersecurity heuristics and information request for cloud computing
- Cybersecurity entity identification and discovery for cloud computing

3.2.2. SmartGrids

SmartGrids are constituted by the integrated amalgamation of communications and electrical power distribution network infrastructures and applications in a manner that optimizes the use of electrical power. [18]

At present, however, the SmartGrid environment has no specific accepted cybersecurity specifications. [19] Here also, the CYBEX framework can be readily adapted to this specialized environment by a combination of adaptations of the existing platforms as necessary for SmartGrid implementations:

- Weakness/vulnerability/state exchange for SmartGrids
- Event/incident/heuristics exchange for SmartGrids
- Exchange of policies for SmartGrids
- Evidence exchange for SmartGrids
- Cybersecurity heuristics and information request for SmartGrids
- Cybersecurity entity identification and discovery for SmartGrids

REFERENCES

[1] Draft Rec. ITU-T X.cybex, "Cybersecurity information exchange framework," ITU-T Study Group 17, Q4/17 e-meeting, Doc. 010R2, July 2010.

[2] Q4 Rapporteur, Yaana Technologies, "Initial Draft candidate Rec.X.gcief, Global Cybersecurity Information Exchange Framework," ITU-T Study Group 17, Q4/17 Interim Meeting, Input Doc. 003, June 2009. See also, e.g., "CVE," http://cve.mitre.org; "Common Vulnerability Scoring System," http://www.first.org/cvss; "Incident Object Description and Exchange Format (IODEF)," http://xml.coverpages.org/iodef.html; ETSI, TS102232, "Telecommunications security; Lawful Interception (LI); Handover specification for IP delivery," http://www.etsi.org/WebSite/document/PlugtestsHistory/2006LI/TS102232.pdf.

[3] Draft Rec. ITU-T X.cybex, [1], Fig. 6. This conceptualization of cybersecurity emerged from work initially accomplished at the Georgia Institute of Technology Sam Nunn School and subsequently presented at the ISS World State of the Industry Session in Dec 2007. Rutkowski, "Significant Developments of Note," ISS World, Washington DC, 10-12 Dec. 2007. It was subsequently introduced into the ITU High Level Experts Group on Cybersecurity [HLEG]. Rutkowski, draft Report on Technical and Procedural Measures for Cybersecurity, documents of the ITU HLEG, Jan 2008.

[4] Draft Rec. ITU-T X.cybex, [1], Sec. 6.1

[5] Draft Rec. ITU-T X.cybex, [1], Sec. 1.

[6] Draft Rec. ITU-T X.cybex, [1], Sec. 6.1

[7] The text provided is taken from the CYBEX Framework draft – which is imported from the development communities responsible for their development of the individual specifications – especially the MITRE sites, e.g., http://cve.mitre.org, http://cwe.mitre.org, etc.

[8] Draft Rec. ITU-T X.cybex, [1], Sec. 8.

[9] ETSI Technical Report "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); TISPAN NGN Security (NGN_SEC); Threat, Vulnerability and Risk Analysis," TR187002 V3.0.2 (2010-01); ETSI Technical Standard, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Methods and protocols; Part 1: Method and proforma for Threat, Risk, Vulnerability Analysis," ETSI TS102165-1, TS102165-1 (2006-12).

[10] ETSI Technical Standard, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Release 2 Lawful Interception; Stage 1 and Stage 2 definition," TS187005 V2.1.1 (2009-09); ETSI Technical Standard, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Security; Report and recommendations on compliance to the data retention directive for NGN-R2," TS187012, V2.1.1 (2009-11); ETSI TC LI, "Development in NGN supporting DT," ETSI LI(10)0133 (June 2010).

[11] ITU-T Y-Series Standards, "Global Information Infrastructure, Internet Protocol Aspects and Next Generation Networks Next Generation Networks," especially ITU-T Rec. Y.2011 (10/2004), "General principles and general reference model for Next Generation Networks;" ETSI European Standard, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Functional Architecture," ETSI ES282001 V3.3.0 (2009-02).

[12] Rec. ITU-T Y.2701, "Security requirements for NGN release 1," (04/2007); Rec. ITU-T Y.2704 "Security mechanisms and procedures for NGN," (01/2010).

[13] Rec. ITU-T Y.2720, "NGN identity management framework" (01/2009).

[14] Draft Rec. ITU-T X.cybex, [1] "Appendix A - Security Automation Schema Use Cases."

[15] ETSI Technical Specification "Lawful Interception (LI); Dynamic Triggering of Interception," DTS102677 V0.5.2 (2010-06)

[16] Monique J. Morrow, "Network Virtualization-Cloud, Management and Inter-Cloud," ITU-T Focus Group on Cloud Security, Doc. Cloud-I-10 (Jun 2010).

[17] Koji Nakao, "Cloud Security," ITU-T Focus Group on Cloud Security, Doc. Cloud-I-22 (Jun 2010).

[18] China Academy of Telecommunication Research, "Proposal for Terminology," ITU-T Focus Group on Smart Grid, Doc. Smart-I-9 (Jun 2010)

[19] Chairman, Requirements WG, "Descriptions of requirement extracted from incoming document for WG," ITU-T Focus Group on Smart Grid, Doc. Smart-I-52 (Aug 2010).

TOWARDS A SERVICE-ORIENTED NETWORK VIRTUALIZATION ARCHITECTURE

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ABSTRACT

Network virtualization is an emerging concept that enables the creation of several co-existing logical network instances (or virtual networks) over a shared physical network infrastructure. There are several motivations behind this concept, including: cost-effective sharing of resources; customizable networking solutions; and the convergence of existing network infrastructures. In this paper, we analyze the existing (conventional and virtualized) business models and propose a new business model for virtual networking environments. Our proposed model is a service-oriented hierarchical model, in which different levels of services (i.e. essential services, service enablers, service building blocks, and end-user services) offered by various players, can be dynamically discovered, used, and composed. Using this business model as basis, we also define a layered service-oriented network virtualization architecture and discuss some of the issues related to its operation.

Keywords — Network virtualization, business modeling, service-oriented architecture, context management, resource management.

1. INTRODUCTION

The concept of virtualization [1] has been used for many years. It consists in the addition of a layer of abstraction between users and physical resources, while giving the users the illusion of direct interaction with those resources. Several forms of virtualization already exist, such as: operating systems virtualization (e.g. virtual machines); computational resources virtualization (e.g. cloud computing); and links virtualization (e.g. the MPLS technology). Even router vendors are following this trend by supporting the development of virtual routers.

Network virtualization [2] is an emerging concept that extends the concept of virtualization from individual nodes (or resources) to entire networks. The main idea consists in the creation of several co-existing logical network instances (or virtual networks) over a shared physical network infrastructure. The virtual networks (VNets) can potentially be built according to different design criteria and operated as service tailored networks – thus offeringfull administrative control and full customization capability. It should be noted that Virtual Private Networks (VPNs) are an existing primitive form of virtual networks, which are limited to traffic isolation capabilities and do not allow customization nor administrative control.

In this paper, we propose a new business model for virtual networking environments and use it as basis for the definition of a service-oriented network virtualization architecture. Our business model is inspired by two existing models, namely: the TINA-C model (from the telecommunication field) and the web service composition model (from the data comm. field). It is a service-oriented hierarchical model, in which different levels of services (i.e. essential services, service enablers, service building blocks, and end-user services) offered by various players, can be dynamically discovered, used, and composed. The term service is used here to signify not only value-added services offered to end users but rather any type of functionality that could be offered by a network resource – thus ranging from low level routing/transport functionalities to high level application logics. In the paper, the different roles of the business model are detailed, and a split of functionality between these roles is proposed and used as basis for our architecture. Furthermore, some of the issues related to the architecture's operation are discussed, and hints are given on how to address them.

The rest of the paper is organized as follows: In section 2, we discuss some of the main motivations and challenges related to the network virtualization concept. In section 3, the existing business models are analyzed and our proposed model is presented. This is followed by an elaboration of our initial network virtualization architecture and a discussion of some of its related issues. We end the paper with our conclusions and hints about future work.

2. MOTIVATIONS AND CHALLENGES OF THE NETWORK VIRTUALIZATION CONCEPT

Network virtualization is a technically challenging concept that is driven by several motivations. In this section, we highlight some of these motivations and discuss some of the challenges related to this concept.

2.1 Motivations

Figure 1 illustrates three motivating scenarios that could be enabled by the concept of network virtualization.

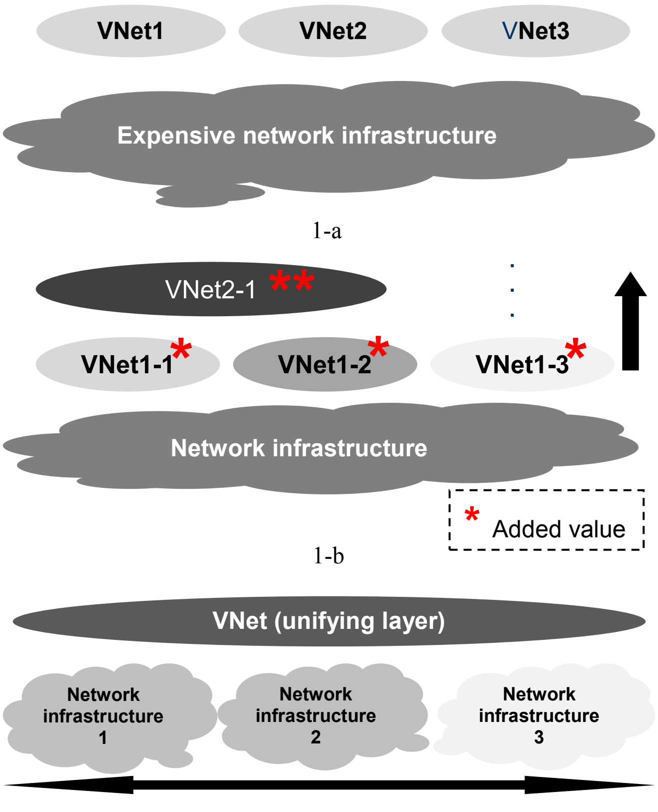

Figure 1. Motivating scenarios for network virtualization: a) Cost-effective sharing of resources; b) Customizable networking solutions; c) Convergence of existing infrastructures

As shown in figure 1a, the first motivation is the cost effective sharing of physical networking resources. In fact, due to the high costs associated with the deployment of physical networking infrastructures, many service providers may be prevented from entering the market due to their limited resources. The network virtualization technology can be used in this case to partition the resources of an existing infrastructure into slices and to associate these slices to different VNets operated by different service providers. This would enable the reduction in the cost of market entry for smaller players who do not have enough resources to build costly infrastructures, but rely on the leased resources of an existing network infrastructure to offer their services.

Unlike the first scenario that focused on the effective partitioning of the physical network resources without enhancing its original functionality, the second scenario (shown in figure 1b) introduces the possibility to add value in the virtualization layer, as means to: 1) introduce new technologies (e.g. new QoS schemes) to existing infrastructures – thus enabling the evolution of existing networks; and 2) customize existing protocols to tailor the network to specific services (i.e. potential for customizable, service-tailored networks). It should be noted that, in this scenario, the instantiated VNets could implement different technologies with respect to each others. Furthermore, as shown in figure 1b, there is a potential of forming a vertical hierarchy in which each VNet builds on the capabilities of the network below it (i.e. a support of the concepts of recursion and inheritance).

Beside resource sharing and customization/evolution of existing infrastructures, another important motivation behind network virtualization is its potential use for the convergence of (possibly heterogeneous) networking infrastructures. In fact, the cooperation between different types of networks is a challenging task that usually requires the reliance on complex gateways and multiple agreements between different operators. The third scenario (shown in figure 1c) addresses this cooperation that could be achieved by deploying a VNet, acting as unifying layer, on top of the different networking infrastructures to enable their interworking. In this case, the value (i.e. the interworking functionality) is added horizontally, via the virtualization layer. This horizontal virtualization dimension could be combined with the vertical dimension presented in the second scenario.

2.2 Challenges

There are many challenges associated with the realization of the network virtualization scenarios presented in the previous section. Examples of these challenges are:

- *Definition of interfaces/interactions between the different levels of the virtual networking hierarchy:* Virtualized networking environments open the door for the existence of various players/roles acting at different levels of the hierarchy. To enable the effective interaction between these roles, suitable interfaces must be defined, at the appropriate levels of abstraction, and must be standardized.
- *Definition of control functions related to the instantiation and configuration of virtual nodes/links:* Virtual networks mainly consist of a set of virtual nodes connected by virtual links. To support their creation, suitable control functions enabling the instantiation and configuration of virtual nodes and virtual links, must be defined.
- *Ensuring scalability at the level of virtual nodes and virtual links:* The main principle behind network virtualization is the coexistence of many VNets operating over a shared physical infrastructure. To achieve this goal, network virtualization environments must scale to support the operation of an increasing number of VNets, without affecting their performance.
- *Suitable framework enabling the dynamic discovery of available physical/virtual resources:* To enable the dynamic creation of (potentially multiple levels of) VNets spanning across different administrative domains, a suitable description and discovery framework is needed. This framework would enable the dynamic discovery of available resources needed for the creation of new VNets or the expansion of existing ones.
- *Definition of global resource management strategies enabling the efficient allocation of resources to VNet:* One of the main challenges in virtual environments is the efficient management of physical resources that are

allocated to VNets. In fact, efficient dynamic resource management strategies, operating at the global network level (i.e. taking into consideration entire VNets rather than individual nodes and spanning the different levels of the hierarchy), must be defined to maximize the utilization of available resources while satisfying the VNets requirements.

- *Definition of suitable business models and charging schemes for virtual environments:* Virtual networks are meant to be open, flexible, and heterogeneous networking environments in which various parties/roles collaborate to offer a diversity of services. To achieve this goal, suitable business and charging models are needed in order to clearly define the different business roles involved in service provisioning and their relationships/interactions. Such models become even more important for enabling the practical deployment of network virtualization solutions in commercial environments (e.g. the future Internet).

3. BUSINESS MODEL FOR VIRTUAL NETWORKING ENVIRONMENT

According to the Telecommunication Information Network Architecture Consortium (TINA-C), a business model describes the different parties (or business roles) involved in service provisioning and their relationships/interactions [3]. Business models are very important because they are usually used as the starting point for standardization and may be used as basis for the definition of architectures. In this section, we start by analyzing the existing business models, proposed for conventional and virtualized networking environments, and then present our proposed model.

3.1 Analysis of Existing Business Models

Existing business models proposed for conventional networks can be divided in two categories, namely: telecommunications business models; and data-communications business models.

There are two main business models in the telecommunication domain: The TINA model [3] and the Parlay model [4]. TINA consists of a set of specifications and a service architecture enabling the provisioning of telecommunication and information services. The TINA business model is a widely reused model that defines five business roles: consumer; retailer; third party service provider; broker; and communication provider. The consumer is the actual user of the service (i.e. the end user) or the entity having the agreement for the service usage (i.e. the subscriber). The retailer is the entity providing (its own or sub-contracted) services and that has business agreements with subscribers for service usage. A third party service provider supports retailers or other third parties with services. It has a business agreement with the retailer and no direct business agreement with subscribers. As for the communication provider, it owns and manages the physical network infrastructure, while the broker provides information to find other parties/services.

Parlay [4] is a set of open, technology-independent, APIs aimed at facilitating the development of telecom-based applications by giving developers controlled access to the capabilities of telecommunications networks. The Parlay business model treats services as service capability features (SCFs) and defines three main roles: The client application that consumes/uses the Parlay services; the enterprise operator that subscribes to the services; and the framework operator handling the subscriptions.

The main business model proposed/used in the data communication domain is the standard Web Service (WS) model [5]. This model constitutes a realization of the Service Oriented Architecture (SOA) in which loosely coupled, platform independent services can be described, published, discovered, used, and composed. Three main roles are defined in the standard WS model: The WS provider that owns (one or more) web service(s) it wants to make available for use; the WS requestor that wishes to make use of a web service; and the WS registry that puts the providers and requestors in contact.

Building on the standard WS model, several other models have been proposed to serve specific purposes. Examples of these models include: The extended service oriented architecture (xSOA) model [6] and the web service composition business model [7] that were proposed to respectively support static and dynamic web service composition – a concept enabling the creation of new web services by reusing and combining existing web services. The xSOA introduces three new entities to the standard WS model, namely: The service aggregator responsible for aggregating services from different service providers into value added composite services; the service operator responsible for service operation/management functionalities; and the market maker responsible for market management functionality. As for the web service composition model, it introduces the following new roles to the standard WS model: WS composer offering WS composition functionality; third party WS provider supporting WS composers with primitive services (or service building blocks) that can be used to compose more complex services; and WS composition registry offering information related to available primitive services.

Beside the conventional business models, two main business models have been proposed for virtualized networks. The first model, which has been adopted in several architectures [8, 9], distinguishes between two roles: the role of infrastructure provider managing the physical infrastructure; and the role of service provider creating VNets by aggregating resources from multiple infrastructure providers and offering end-to-end services. The second model, which has been proposed in the 4Ward project [10], refines the first model by defining four roles: physical infrastructure provider (corresponding to the role of infrastructure provider in the first model); virtual network provider responsible of finding and composing an adequate set of virtual resources from one or more infrastructure providers into an empty virtual topology; virtual network operator that deploys different protocols over the VNet topology and is responsible for the control and management of the VNet; and service provider using the VNet to offer end-to-end services.

By analyzing the presented business models, we found that the models proposed for virtualized environments are still not developed enough and have several limitations. One of these limitations is the *lack of support of the brokerage role*, which is essential in such a dynamic, on-demand networking environment relying on the collaboration between various parties for the provisioning of services. Another limitation is *the lack of support of the concept of (vertical) hierarchy* among various VNets. In fact, the relationship between different virtual infrastructure providers, building their VNets one on the top of the other, is not clearly defined in the existing models. Yet another limitation is that the existing models *do not support the idea of service building blocks*, which is an interesting idea that is widely used in the conventional models.

Among the analyzed conventional business models, we found the *TINA model* and the *WS composition model* to be particularly interesting. In fact, the TINA model is a seminal model with many sound concepts, such as the separation between the physical network operator and the service provider roles and the introduction of the third party service provider role – concepts that are applicable in virtual networking environments. Other interesting ideas to learn from TINA are: the reliance on two levels of services (end user services and service building blocks); and the use of a brokerage function to put different parties in contact. As for the WS composition model, it brought us to make a parallel between the problem of web service composition and the problem of resources discovery and composition in virtual environments. In fact, the functionalities provided by the different types of network resources could be considered as a variety of services that could potentially be discovered, used, and composed. Some interesting ideas to learn from this model include: the support of three levels of services – simple end user services, composite end user services, and non end user services (or service building blocks); and the reliance on two types of registries for the discovery of services and service building blocks.

Inspired by the TINA and the WS composition models, and aiming at addressing the limitations of existing virtualization models, we propose in the coming section a new business model for virtual networking environments.

3.2 Proposed Business Model

Figure 2 depicts our proposed model. Inspired by the TINA and the WS composition models, our model is a service-oriented, hierarchical model, in which different levels of services (offered by various players) can be dynamically discovered, used, and composed. Four levels of service are defined in relation to our model, namely: Essential services; service enablers; service building blocks; and end-user services. *Essential services* are mandatory services needed for the basic operation of the network (i.e. routing and transport services). *Service enablers* consist of the common functions needed to support the operation of end-user services (e.g. session management, subscription management, charging, security, and QoS control)- although they are not necessarily apparent to the end user. As for s*ervice building blocks,* they can be considered as elementary services (or service capabilities) that can be used or combined to form more complex services. Examples of service building blocks could include call control, presence, and messaging. Finally, *end user services* constitute the value added services (VAS) offered to users and which can be simple or composite (i.e. formed by the combination of service building blocks).

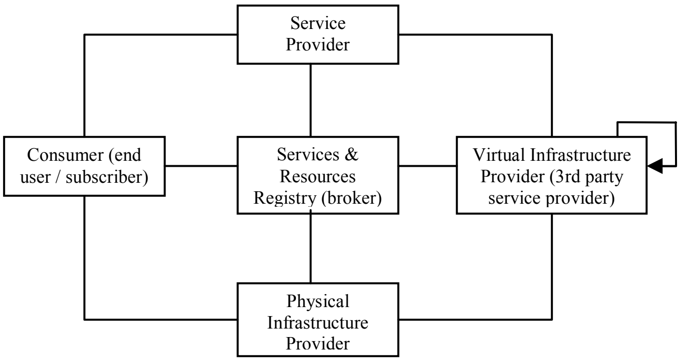

Figure 2. Proposed business model for virtual networking environment

As shown in the figure, our model defines five business roles:

- ***The Physical Infrastructure Provider (PIP):*** Owns and manages a physical network infrastructure and can partition its resources into isolated slices, using some virtualization technology. The services offered by the PIP are essential bearer services (i.e. routing/transport services). This role is similar to the role of communication provider in the TINA model.
- ***Service Provider (SP):*** Entity that has a business agreement with the subscriber and offers value added services. The services offered by a SP could be simple or composite (i.e. formed by combining service building blocks). This role is similar to the role of retailer in the TINA model and the role of WS provider in the WS architecture.
- ***Virtual Infrastructure Provider (VIP):*** Playing the main role in virtual networking environments, the VIP finds and puts together virtual resources (offered by one or more PIPs), deploys any protocols/technologies in the instantiated VNet, and operates its as a native network. Typically, the services offered by VNets are service enablers (i.e. common support functions) or service building blocks, rather than end user services. Like third party service providers in TINA, the VIP supports SPs or other VIPs with services and has no direct business agreement with subscribers. We envision three potential variations of the VIP role: 1) A VIP that adds value in the virtualization layer by introducing a new technology or customizing existing protocols – the resulting VNet can be used by a SP to offer VAS running on it or resold to another VIP that leverages its capabilities to form

another VNet on top of it (i.e. forming a vertical hierarchy); 2) A VIP that uses virtualization to achieve interworking between heterogeneous physical infrastructures – the result being a unified network for others to use; and 3) A VIP that implements more advanced services in the virtual layer to offer application building blocks that can be used by service providers to compose new value added services. It should be noted that we do not envision any conflict of interest between PIPs and VIPs, since they target different categories of customers – PIPs targeting customers seeking basic bearer services, while VIPs targeting customers requiring more sophisticated networking services (e.g. new technologies or customization of existing protocols).

- ***Consumer:*** Acting as end user and service subscriber, the consumer uses the value added services provided by SPs and which rely on physical/virtual infrastructures in their operation.
- ***Services and Resources Registry (SRR):*** Similar to the role of broker in the TINA model, the SRR puts all parties in contact by providing information to find other parties and the services/resources they offer. This may be achieved via some kind of description and discovery framework.

Figure 3 illustrates how we envisage the split of the protocol stack functionality among the five defined roles.

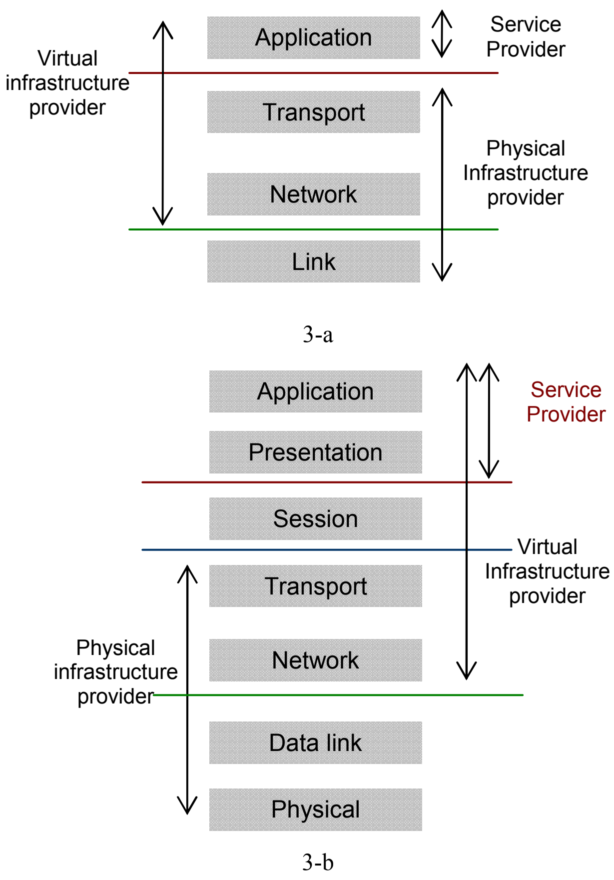

Figure 3. Split of protocol stack functionality among the different roles: a) The TCP/IP model; b) The OSI model

As shown in figure 3a, the TCP/IP protocol stack could be divided as follows: The essential bearer services of the network should be supported by the PIP that should support the link, network, and transport layer functionalities. The VIP could implement application level protocols related to the support of common functions (e.g. session control and media handling) and service building blocks (e.g. presence and conferencing), with the possibility of customizing the transport and network level protocols (originally supported by the PIP) in the virtualization layer. The SP would focus on the implementation of additional application level protocols needed for the operation of its VAS. This split of functionality is further clarified in figure 3b, related to the distribution of the OSI model functionality.

4. NETWORK VIRTUALIZATION ARCHITECTURE: INITIAL PROPOSAL AND RELATED CHALLENGES

In this section, we start by presenting an initial version of our network virtualization architecture, which was designed based on the defined business roles and the proposed functional split between them. We then discuss some of the remaining challenges related to the architecture's operation and how they can be addressed.

4.1 The proposed architecture

As shown in figure 4, the network virtualization architecture we are proposing is a layered architecture that introduces data and control planes at each level of the hierarchy. While the data plane is used for the provision of essential data transportation functionality, the control plane encompasses all the control and management functions needed for the provisioning of different levels of services.

At the lowest level of the hierarchy, we find the physical network (managed by the PIP) that is divided into a physical data plane and a physical control plane. The physical data plane contains regular and virtual routers connected to form the physical network infrastructure. The physical control plane contains network control and management entities that are responsible of achieving the following functionality: Publishing a description of available resources; negotiation of resources with VIPs, allocation of virtual resources, and instantiation of virtual topologies; forming/maintaining a global view of the physical/virtual network contexts; dynamic resource allocation/re-allocation to VNets taking into consideration the resources status and the needs of VNets; as well as authentication, authorization, and charging of VIPs.

At the second level of the hierarchy, we find a VNet (managed by a VIP) consisting of virtual data and control planes. The virtual data plane encompasses a set of virtual nodes connected by virtual links, which is essentially a subset of the underlying physical topology. As for the virtual control plane, it is responsible of the following functionality: Discovery and negotiation of resources, composition/instantiation of the VNet topology, deployment of protocols, and operation of the VNet; publication of description of the VNet available resources; in addition to other support functions (e.g. subscription/session management, QoS control, and AAA) and possibly some service building blocks (e.g. call control, presence).

At the highest level of the hierarchy, we find a service network (VNet 2 managed by a SP) also consisting of a virtual data and control plane. In this case, the virtual control plane contains functions needed for the provisioning of VAS, such as: Discovery of available virtual resources and the negotiation/utilization of those resources for service provisioning; subscription management; authentication/authorization/charging; the service logic; as well as signaling and media handling.

Distributed across all the levels of the hierarchy, we find the services and resources registry that supports the following functionality: A data model enabling the description of various services and physical/virtual resources; in addition to information publication and discovery.

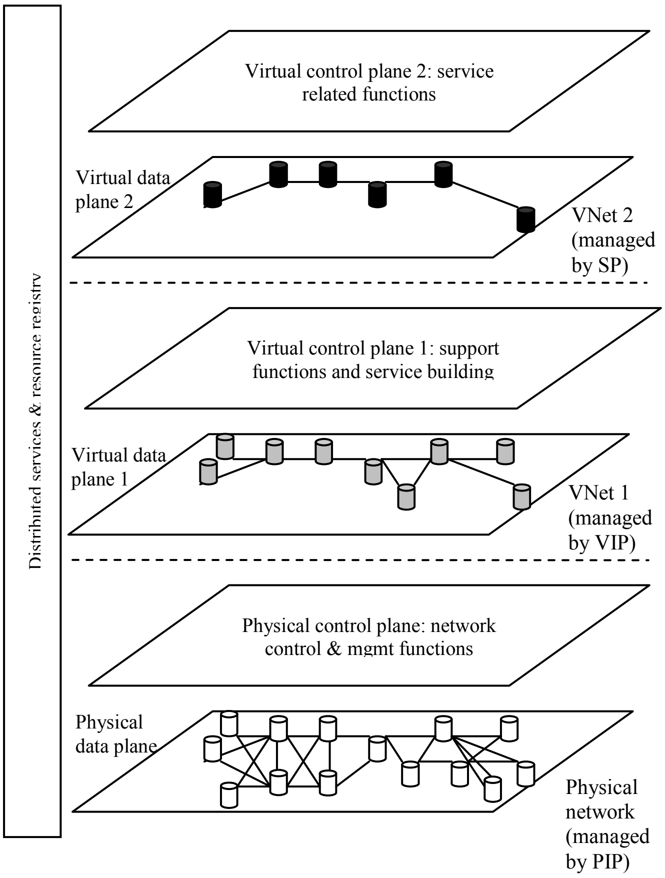

Figure 4. The proposed network virtualization architecture

4.2 Related issues and hints

Context management and dynamic resource allocation are two sub-issues related to the problem of global resource management in our architecture. In this section, we discuss those issues and give hints on how to address them.

4.2.1 Hierarchical clustering for scalable context management

Context management, which implies the acquisition/ modeling/storage/dissemination of situational information, is an important support function needed in virtual environments. In fact, to be able to manage resources efficiently and potentially support advanced context-aware value added services, the system needs to be able to form and maintain a global view of the physical/virtual networks contexts, in a scalable and resource efficient manner. Clustering (a concept which assembles nodes into logical groups or clusters) could be used to ensure the scalability of the context management operations, as follows: Instead of dealing the global context as a whole, it could be divided into smaller units. For instance, two types of context views could be defined: Physical context view (at the physical node level); and logical context view (at the VNet level). Then, each physical node would be responsible of managing its personal physical context view, while each VNet would be represented as a cluster in which a cluster-head is responsible of managing the logical network's context view.

4.2.2 Efficient resource management via dynamic virtual topology adaptation

To enable the efficient management of physical resources (allocated to VNets), the system must be capable of dynamically (re)-allocating resources to VNets, taking into consideration the resources status and the needs of VNets and their users. This may be achieved via the dynamic adaptation of virtual topologies, in response to changes in the physical network resource status or specific requests from VNets. One possible approach to realize dynamic virtual topology adaptation, while minimizing the disruption to ongoing sessions, could be to focus on the replacement of strategic nodes (e.g. junction nodes traversed by several VNets) by less loaded nodes.

5. CONCLUSIONS AND FUTURE WORK

Network virtualization is a promising and technically challenging concept. In this paper, we proposed a new service-oriented, hierarchical, business model for virtual networking environments. We also used this model as basis for the definition of our network virtualization architecture, and discussed some of its related issues. In the future, we plan to refine our architecture and detail the scenarios related to its operation. Moreover, we plan to elaborate the cluster-based context management solution and the dynamic topology adaptation mechanism we briefly discussed, and to validate our solution using simulations.

REFERENCES

[1] P. Barham et al., "Xen and the art of virtualization," in proceedings of the 9th ACM symposium on operating systems principles, New York, 2003, pp. 164-177.

[2] N. Chowdhury and R. Boutaba, “Network Virtualization: State of the Art and Research Challenges,” IEEE Communications Magazine , vol. 47, no. 7, pp. 20-26, July 2009.

[3] TINA Business Model and Reference Points, v.4.0, May 1997, http://www.tinac.com/specifications/documents/bm_rp.pdf, accessed 7 June 2009.

[4] ETSI ES 203 915-3 Version 1.1.1, Parlay 5.0, “Part 3: Framework,” Apr. 2005.

[5] Web Services Architecture Specification, W3C Working Group Note, Feb. 2004, http://www.w3.org/TR/ws-arch/, accessed 28 July 2009.

[6] M. P. Papazoglou, “Extending the Service-Oriented Architecture,” Business Integration Journal, pp. 18-21, February 2005.

[7] R. Karunamurthy, F. Khendek, and R. H. Glitho, “A Business Model for Dynamic Composition of Telecommunication Web Services,” IEEE Communications Magazine, Special issue on ‘Web Services for Telecommunications’, vol. 45, no. 7, pp. 36-43, July 2007.

[8] N. Feamster, L. Gao, and J. Rexford, “How to Lease the Internet in Your Spare Time,” SIGCOMM CCR, vol. 37, no. 1, pp.61–64, 2007.

[9] Y. Zhu, R. Zhang-Shen, S. Rangarajan, and J. Rexford, “Cabernet: Connectivity Architecture for Better Network Services,” In Proceedings of ReArch ’08. ACM, 2008.

[10] G. Schaffrath et al., “Network virtualization architecture: Proposal and initial prototype”, in Workshop on Virtualized Infrastructure Systems and Architectures, August 2009.

THIN APPS STORE FOR SMART PHONES BASED ON PRIVATE CLOUD INFRASTRUCTURE

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ABSTRACT

A novel approach to implement cloud computing for smart phone devices has been presented based on Eucalyptus, an open source cloud-computing framework that provides infrastructure as a service (IaaS). It has full support of Virtualization and is Amazon Web Services interface compatible. A private cloud has been designed using Eucalyptus to develop a smart phone application store. The architecture, physical and network implementation of Eucalyptus Private Cloud on Intel based platform has been discussed in details. We have developed two sample thin applications for mobile based on mobile learning that can be downloaded from our private cloud using Amazon Web Services Platform (PaaS). These thin apps use the private cloud as computing platform, and perform better even on low processing smart phones.

Keywords—Mobile Computing, Private Cloud, Cloud Computing Infrastructure, Augmented Virtually, Thin Apps, Smart phone applications, Virtual Machines, Virtualization.

1. INTRODUCTION

In the recent years, there has been a revolutionary growth in the use of mobile devices for supporting consumer applications like high speed internet access, Mobile/IP TV, augmented virtually based gaming and context aware mobile learning. For example now a person in a museum can learn about the relevant work from his PDA. Smart phones have clearly redefined conventional cell-phone domain by supporting wide range of high profile applications. However energy and bandwidth constraints are the two limiting factors in developing new applications and solutions for mobile environment. The future mobile devices must come up with high capability of supporting sophisticated graphical processing and large memory requirements within its battery constrained environment. Though 3G and 4G networks have significantly reduced the problem of limited bandwidth, the gap between the present processing power and processing requirements of sophisticated mobile applications do not seem to come closer enough unless we integrate the next generation cloud computing technology.

We have developed a private cloud using Eucalyptus, an open source cloud-computing framework. We have also developed a fully integrated context aware mobile learning applications that assert low computational load on mobile devices through the use of cloud computing. The application has been implemented on the unique cloud computing platform using virtualization to enhance scalability. This paper has been divided in 7: Sections I is Introduction and Section II is Technical description of Cloud computing and Virtualization. Section III deals with features of Eucalyptus that is used for Private cloud Infrastructure, its implementation on Intel based Platform and Network Addressing scheme. Section IV deals with overview of integration of Smart phone with cloud. Section V and VI give details application developed followed by conclusion in Section VII.

2. CLOUD COMPUTING WITH VIRTUALIZATION SUPPORT

Cloud computing for mobile devices is widely accepted future technology. It focuses on Internet based development and use of present computer technologies supported through internet. The concept incorporates Infrastructure as a service (IaaS), Platform as a service (PaaS) and Software as a service (SaaS) layers as well as Web 2.0 and other recent technology trends [1] [2] as shown in Fig 1.

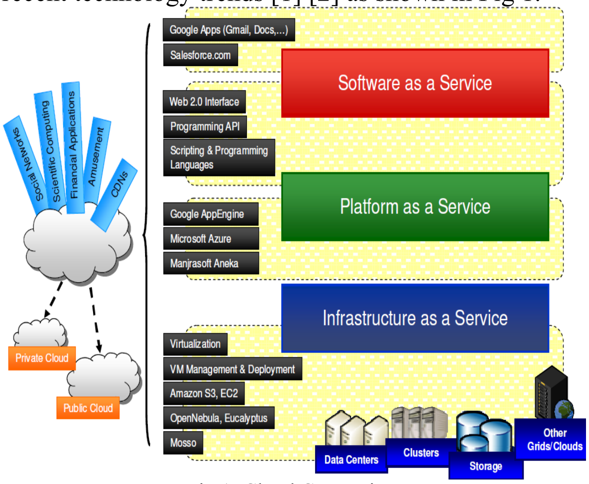

Fig 1: Cloud Computing Layers

One of the most important ideas behind cloud computing is scalability, and the key technology that makes that possible is virtualization. Virtualization allows better use of a server by aggregating multiple operating systems and applications on a single shared computer. Virtualization also permits online migration so that if a server becomes overloaded, an instance of an operating system (and its applications) can be migrated to a new, less cluttered server. Cloud computing for mobile devices provides the key concept of transferring mobile device computation to the cloud.

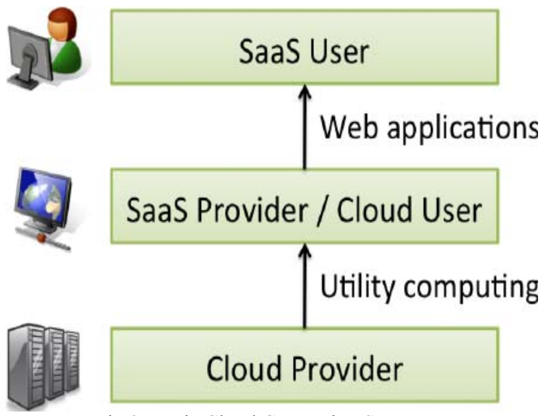

Fig 2: Basic Cloud Computing Structure

Basic cloud computing structure consists of 1) a cloud provider, 2) a SaaS Provider/ Cloud user and 3) SaaS User/ end user (refer Fig 2). The end user just requests for the service from SaaS Provider which is then accomplished by the cloud service provider. The mobile host acts as a client saving its computing time from complex user applications. The computing resources like processor, memory are owned and managed by the cloud service provider and client (Cloud Used/ Saas Provider) access these resources via internet [2]. Cloud based services like Amazon Simple Storage Services provides feature of storing personal data on the cloud and computation infrastructure is supported by Elastic Compute Cloud (EC2) of Amazon. This type of computation infrastructure provides several attractive features like higher utilization of resources through virtualization, lower operational and maintaining cost, ideal launching platform for new services leading to low capital requirements.

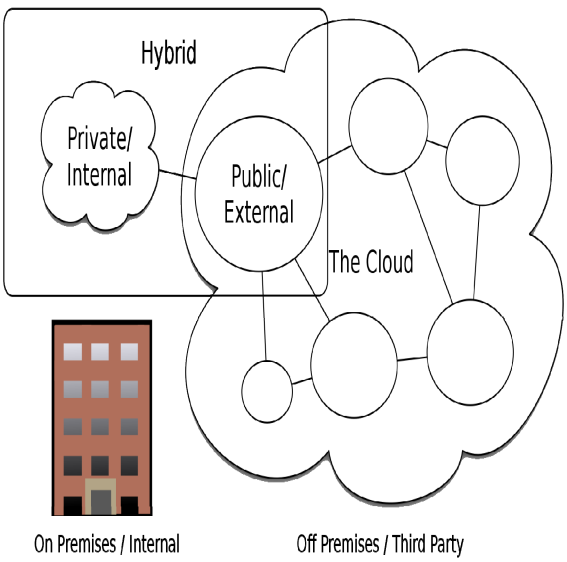

Fig3: Types of Cloud Computing

Technically there are 3 types of clouds: Private clouds, Public clouds and Hybrid Clouds as shown in Fig3. In a public cloud, the provider of the cloud provides the various resources such as the memory and processors to the client through the internet and is implemented using the normal cloud architecture. A private cloud refers to a specific cloud architecture that provides required resources to a certain fixed number of clients on some payment for these services. A hybrid cloud is a combination of cloud computing where the resources offered by the cloud are utilized and in situ computing (device using its own resources).

Mobile devices are operated under a single private cloud and these sets of privately held clouds are connected to the remote cloud. These local cloud units further serve as a successor of remote cloud in the hierarchical structure shown in Fig 4. The private cloud units serve as a computational as well as storage infrastructure for mobile devices.

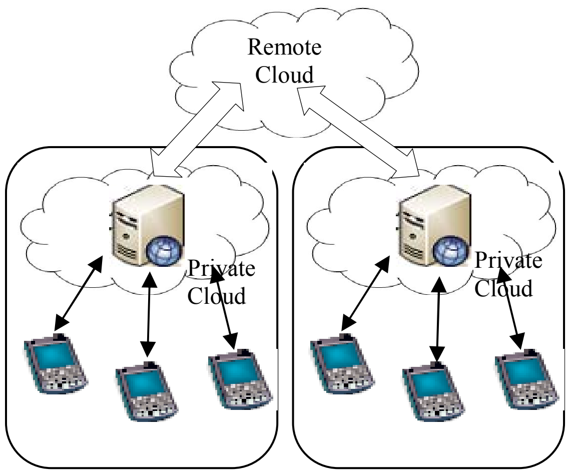

Fig 4: Unification of Private Clouds

3. PRIVATE CLOUD INFRASTRUCTURE

Eucalyptus is an open source cloud-computing framework that we are using to provide Infrastructure as a Service (IaaS). It gives users the ability to run and control entire virtual machine instances deployed across various physical resources. It provides users services, tools and interface similar to its alternate Amazon EC2.

The latest version Eucalyptus v1.6 incorporates following features

- Amazon Web Services interface compatibility.
- Advanced storage integration (iSCSI, SAN, NAS) enables you to easily connect and manage your existing storage systems from within the Eucalyptus cloud. Fig 7.
- Deployment on multiple clusters.
- Deployment of components (Cloud controller, Walrus, Storage Controller, Cluster Controller) on different machines.
- Enhanced concurrency management: cloud requests are serviced asynchronously with minimal locking using eventual consistency for scale.
- Support for open-source monitoring and health/status: Ganglia and Nagios interaction.
- Support multiple hypervisor technologies like Xen, KVM, vSphere, ESX, and ESXi, providing single management console for different virtualized environments as shown in Fig 5.

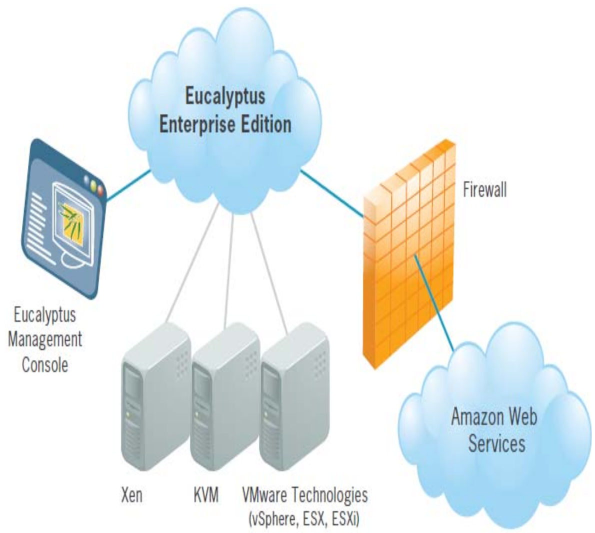

Fig 5: Management Console and Multiple Hypervisors

The architecture of eucalyptus is simple, modular and flexible. It consist of following components (refer Table 1). Fig 6 shown Block diagram of structure of Eucalyptus

Table 1: Components of Eucalyptus

Component	Function
Node Controller	Controls the execution, inspection, and terminating of VM instances on the local host
Cluster Controller	Gathers information about and schedules VM execution on specific node controllers, as well as manages virtual instance network.
Storage Controller (Walrus)	Put/get storage service that implements Amazon's S3 interface, providing a mechanism for storing and accessing virtual machine images and user data.
Cloud Controller	Entry-point into the cloud for users and administrators. It queries node managers for information about resources, makes high-level scheduling decisions, and implements them by making requests to cluster controllers.

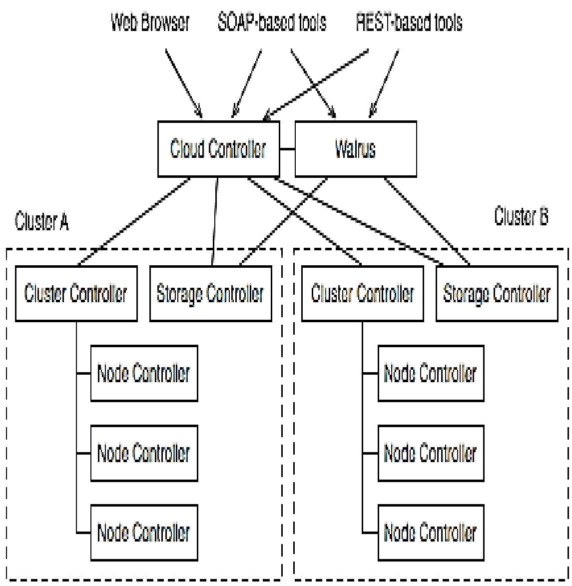

Fig 6: Block Diagram of Structure of Eucalyptus

We are using Euca2ools are command-line tools for interacting with Web services that export a REST/Query-based API compatible with Amazon EC2 and S3 services. The tools can be used with both Amazon's services and with installations of the Eucalyptus open-source cloud-computing infrastructure. The tools were inspired by command-line tools like api-tools and ami-tools developed by Amazon and largely accept the same options and environment variables. However, these tools were implemented from scratch in Python, relying on the Boto library and M2Crypto toolkit. Euca2ools have following features

- Image management (bundle, upload, register, list, deregister)
- IP address management (allocate, associate, list, release)
- Query of availability zones (i.e. clusters in Eucalyptus)
- SSH key management (add, list, delete)
- VM management (start, list, stop, reboot, get console output)
- Security group management
- Volume and snapshot management (attach, list, detach, create, bundle, delete)

Fig 7 shows physical implementation of eucalyptus Intel Based Platform. We are using Intel Xeon processor 5500 series code named Nehalem for Node controllers, Xeon Processor X5570 for Cloud Controller (CLC), Cluster Controller (CCa & CCb), and Walrus Storage Service. Cisco's networking equipment like Routers, Cisco 3750 Switches have been used for configuring and routing the network. The network design has been simulated and tested on Cisco Packet Tracer, a powerful network simulation program.

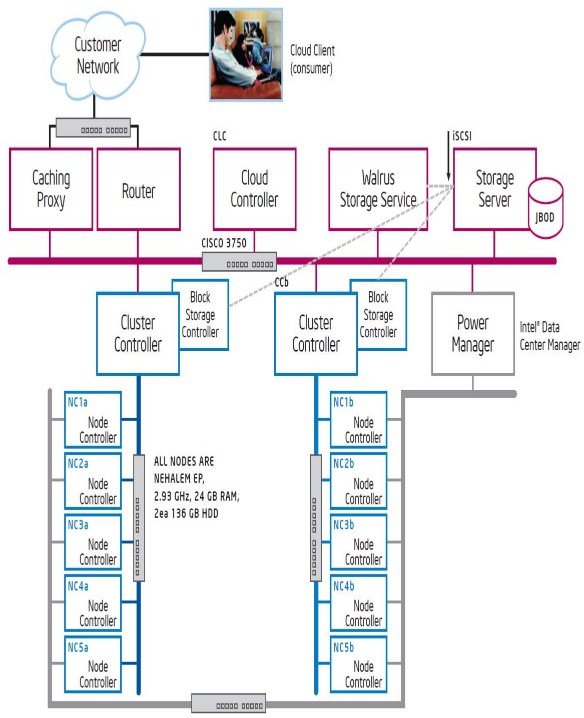

Fig 7: Physical Implementation of EUCALYPTUS

Table 2 shown system configuration details of various components of our private cloud that has been implemented on Intel Xeon based Platform. Similarly, Table 3 shows Network addressing details of the cloud network. All these details are necessary for physical implementation:

Table 2: Private Cloud Systems Configuration

System	Processor	Memory	Form Factor
Cloud / Cluster Controller	Intel Xeon Processor X5570	24 GB	2U Rack Mount
Node Controller	Intel Xeon Processor X5570	24 GB	1U Rack Mount
Walrus Storage Service	Intel Xeon Processor X5570	48 GB	2U Rack Mount
Storage Server	Intel Xeon Processor X5570	24 GB	5U Tower
Proxy Server	Intel Xeon Processor 5140	4 GB	1U Rack Mount

Table 3: Allocated IPv4 Addresses

Network	IP range	Subnet mask	Broadcast
Cloud Network	192.168.16.0/20	255.255.240.0	192.168.31.255
Private VM network	10.0.0.0/8	255.0.0.0	10.255.255.255
Cluster Controller [CCa]	192.168.32.0/24	255.255.255.0	192.168.32.255
Cluster Controller [CCb]	192.168.33.0/24	255.255.255.0	192.168.33.255
CCa Public IP for instance	192.168.17.1/254	255.255.255.0	192.168.17.255
CCb Public IP for instance	192.168.18.1/254	255.255.255.0	192.168.18.255

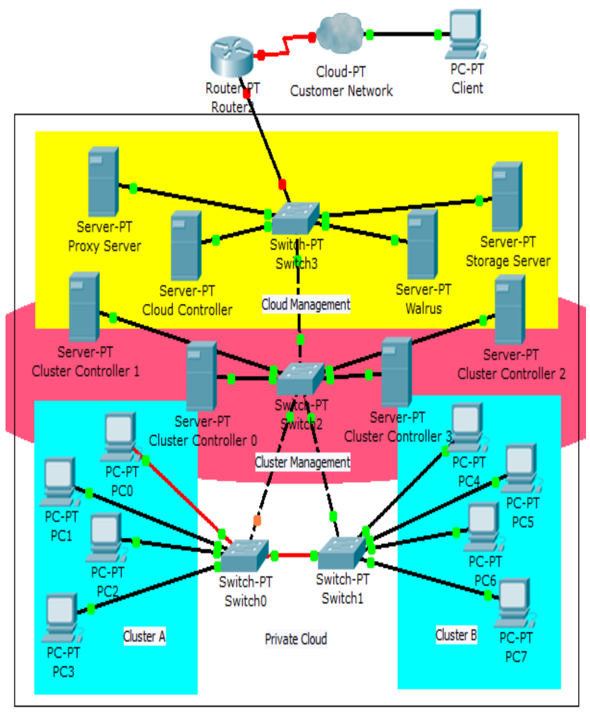

Fig 8: Network Simulation on Cisco Packet Tracer

Fig 8 shows network architecture and simulation of Cloud on Cisco Packet Tracer. Eucalyptus allows users to create one or more VLAN. A VLAN can spread across many physical hosts and virtual machines so that when a virtual machine is migrated from one physical host to another the relationship between VLAN and virtual machine remains same. Eucalyptus takes full advantage of virtualization and provides Virtualized Networks, Virtual Computers and Virtual Storage. It also provides Remote Management Console and Interfacing with external clouds hence contributing to a hybrid cloud as Fig 9.

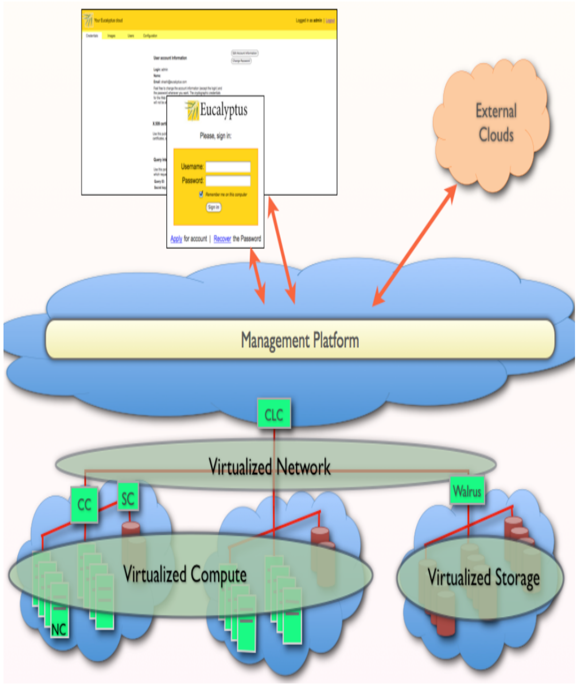

Fig 9: Virtualization In Private cloud

4. SMARTPHONE CLOUD INTEGRATION

Today Smartphones with Internet access are in high demand. Recent mobile phones like Apple iPhone, Blackberry smartphones, and the Google Android phone have processors having clock frequency in GHz and Memory in in GB. Suck mobiles have their own OS onto which different softwares can be installed. But still the performance/ prize ratio for such devices is low and high multimedia applications are not fully supported.

We are addressing these limitations with cloud computing. We are offloading some tasks on a private cloud. We have designed light web-applications that can take input from users, perform computational tasks on cloud, can show results back to users. This process is called augmented execution. The private cloud boosts such applications by providing distributed computing platform.

The Augmented execution is performed in four steps

1) A clone of the smartphone is created within the cloud
2) The state of the primary phone and the clone phone is periodically or on-demand synchronized
3) Applications are executed in the clone, automatically or upon request. This is done by a virtual machine hosted on a node/ cluster or by a physical node.
4) Results from clone execution are re-integrated back into the smartphone state.

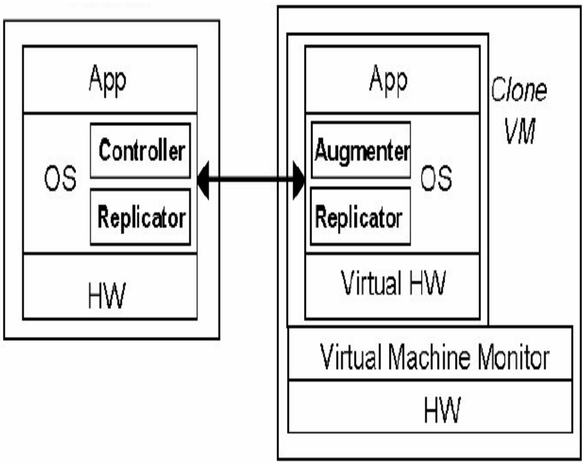

Fig 10: Smartphone and Private cloud integration

To perform augmented execution we need to install Controller and Replicator on mobile OS. The Controller running in the smartphone invokes the augmented execution and merges its results back to the smartphone. On the other hand, The Replicator synchronizes the changes in phone software and state to the clone.

5. APPLICATION 1: AUGMENTED VIRTUALITY BASED LEARNING

In this prototype learning application, context data from the mobile devices can be integrated with the augmented virtual scenario with the real life data which helps in learning. Augmented virtuality based environment can prove worthy in mobile learning. The Fig 11 shows the components of mixed reality in which we are using Augmented Virtuality based application to inter relate real world with virtual environment for mobile learning.

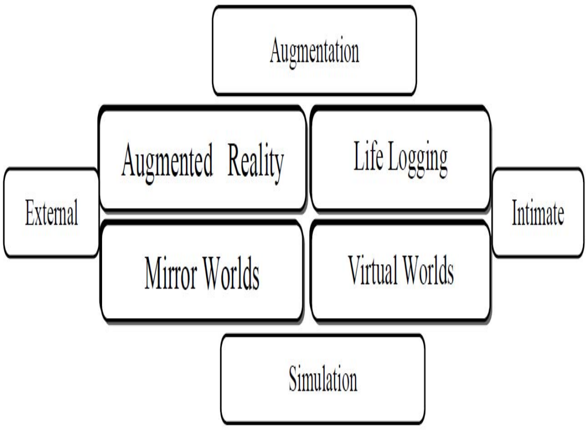

Fig 11: Mixed Reality Composition

We have used smart phones having context gathering modules which includes speed and location data, which then transferred into the virtual world where Avatar is integrated and operated though this real time data. The virtual environment creates an Avatar having interaction with the real data, helps as a feedback in learning process as analyser and assister especially for athletics.

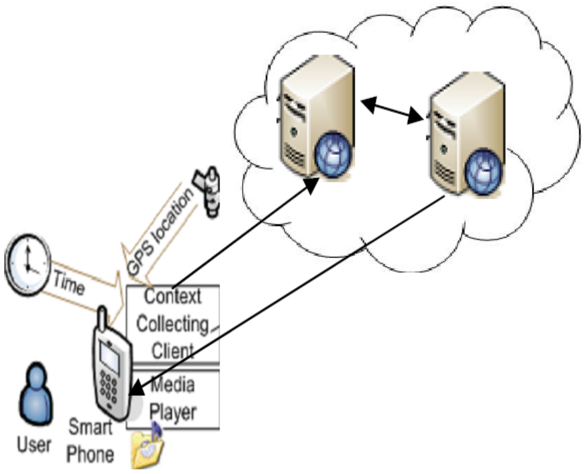

Fig 12: Working of Augmented Virtuality Based Learning Application

The avatar mimics the user location, speed and other body movements in the virtual world .It can be seen later in a synchronized way by the user on his smart phone. Symbian OS based Smart phone is used in which context collection API is installed. It monitors the context aware sensor data and periodically uploads this information on the context database held on the Private Cloud. On the Cloud tiles of user information is integrated from the Google Maps service and provides virtual environment combined with real time data .The latitudes, longitudes and time instances triplets along with speed data serve as inputs for creating user's walk on the virtual environment. The working of the application designed is explained in Fig 12.

Fig 13: Snapshot of Augmented Virtuality Based Learning

Fig 13 shows the snapshot of Augmented Virtuality Based Learning application hosted on private cloud.

6. APPLICATION 2: VIRTUAL GUIDE

Another application that is implemented through the described mobile cloud architecture is the 'virtual guide'. It is an application which implements context learning by receiving inputs – position inputs from sensors such as the GPS and/or GSM receivers, video/ image inputs from camera device in the smart phone or PDA. These inputs are sent to the mobile cloud which processes these inputs to deduce the current location of the device connected to the

cloud by synchronizing the received inputs with the Google maps. Further precision is introduced by matching the image/video inputs with the set of location specific information obtained through the Google map. For Ex if a person goes to a shopping mall, the Google map will only be able to tell the name of the shopping mall by the location data that is obtained through the sensors. Further if the person needs to know the smallest details of a shop that is present in that mall (all the items available or search for a specific item in a specific shop) the application would narrow down to that shop through the context learning of the image/video that the device may record.
Once a specific location is reached (decided mutually by context learning from both the inputs), the application on the cloud would then browse the web to get to the website of that particular place/shop etc. and start running the virtual guide of that place /shop. Thus all the places/shops etc. would need to upload a virtual guide on their websites which would not only give all the details of the place but would also be continuously updated and might even serve as an attendant answering queries of the user. Fig 14 shows the snapshot of virtual guide application hosted on private cloud.

Fig 14: Snapshot of Virtual Guide

7. CONCLUSION

A noble work of implementing Eucalyptus based cloud computing infrastructure for hosting mobile applications store has been presented. Critical details of physical implementation of cloud like servers used for different cloud components like node controller, cluster controller, storage controller and proxy server, their network assembly and IPv4 based addressing scheme have been discussed in details. We have also briefly discussed development of two sample applications for hosting on our Mobile Application Store.

REFERENCES

[1] “Cloud computing - Issues, research and implementations”, ITI 2008 30th International Conference on Information Technology Interfaces, June 23, 2008 - June 26, 2008.

[2] ” Above the Clouds: A Berkeley View of Cloud Computing” Michael Armbrust, Armando Fox, Rean Griffith, Anthony D. Joseph, Randy Katz,Andy Konwinski, Gunho Lee, David Patterson, Ariel Rabkin, Ion Stoica, and Matei Zaharia, February 10, 2009.

[3] Phyoung Jung Kim and Young Ju Noh, "Mobile Agent System Architecture for supporting Mobile Market Application Service in Mobile Computing Environment" in proceedings of IEEE conference on Geometric Modeling and Graphics, July 2003.

[4] George H. Forman and John Zahorjan University of Washington “The Challenges of Mobile Computing “, COMPUTING MILIEUX, April 1994.

[5] Lamia Youseff, Maria Butrico and Dilma. Da Silva, “Toward a Unified Ontology of Cloud Computing”, 2000

ABSTRACTS

Session 1: Keynote speakers	
	Uday B. Desai (Director, Indian Institute of Technology, Hyderabad, India) Tadao Saito (Professor Emeritus the University of Tokyo, Japan) Detlev Otto (CTO, Vodafone Customer Business Team, Nokia Siemens Networks, Germany)

Session 2: Rethinking the network1	
S2.1	Invited paper: Toward a polymorphic future internet: a networking science approach *Kavé Salamatian (Professor, Université de Savoie, France)* In this paper, I will develop two major claims. First the, Future Internet should be polymorphic and conciliate different architectural paradigms networking. The second claim is that the Future Internet should be build on strong theoretical basis from a Networking science that is in course of development. In this paper, I have used the concept of cooperation as an interpretation lens. Specifically, I will describe how virtualisation make possible a polymorphic future Internet and enables the easy deployment of new cooperation schemes. The next aspect that I describe in this paper is relative to security in the future Internet. Particularly the paper advocates the necessity of three major components: a secure execution platform, an authentication mechanism, and a monitoring component. Finally, I will show that it is possible to build scalable addressing and routing scheme but at the condition of following a clean slate approach.
S2.2	Introducing elasticity and adaptation into the optical domain toward more efficient and scalable optical transport networks* *Masahiko Jinno (NTT Network Innovation Laboratories, Japan); Takuya Ohara (NTT, Japan); Yoshiaki Sone (NTT Network Innovation Laboratories, Japan); Akira Hirano (NTT, Japan); Osamu Ishida (NTT Network Innovation Laboratories, Japan); Masahito Tomizawa (NTT, Japan)* There is growing recognition that we are rapidly approaching the physical capacity limit of standard optical fiber. It is important to make better use of optical network resources to accommodate the ever-increasing traffic demand to support the future Internet and services. We first introduce an architecture, enabling technologies, and the benefits of recently proposed spectrum-efficient and scalable elastic optical path networks. In these networks, the required minimum spectral resources are adaptively allocated to an optical path based on traffic demand and network conditions. We then present possible adoption scenarios from current rigid optical networks to elastic optical path networks. We also discuss some possible study items that are relevant to the future activities of ITU-T. These items include optical transport network (OTN) architecture, structure and mapping of the optical transport unit, automatically switched optical network (ASON) control plane issues, and some physical aspects with possible extension of the current frequency grid
S2.3	Introducing Multi-ID and Multi-locator into Network Architecture* *Ved P. Kafle (National Institute of Information and Communications Technology, Japan); Masugi Inoue (National Institute of Information and Communications Technology, Japan)* The present day Internet has no separate namespace for host IDs. It uses IP addresses as host IDs, which are in fact locators. This dual role is problematic for mobility, multihoming, security, and routing on the Internet. To solve these problems, research has recently begun on ID/locator split architectures. Some standardization activities based on this concept are also progressing in ITU-T Study Group 13 and in the IETF. We expect that introduction of the ID/locator split concept into the new generation network or future Internet architecture can bring about additional functions, such as heterogeneous network protocol support, multicast, QoS, resource or service discovery, and flexible human-network interaction. Toward realization of these functions, this paper presents a study on an approach of introducing multi-ID and multi-locator support into the network architecture. The paper also lists items that have the potential to be standardized in ITU-T.

[1] Papers marked with an "*" were nominated for the three best paper award.

S2.4 How can an ISP merge with a CDN?*

Kideok Cho (Seoul National University, Korea); Hakyung Jung (Seoul National University, Korea); Munyoung Lee (Seoul National University, Korea); Diko Ko (Seoul National University, Korea); Taekyoung Kwon (Seoul National University, Korea); Yanghee Choi (Seoul National University, Korea)

As delivering contents has become the dominant usage of Internet, the efficient content distribution is being one of the hottest research areas in network community. In future network, it is anticipated that network entities such as routers will be equipped with in-network storage due to the trend of ever-decreasing storage cost. In this paper, we propose a novel content delivery architecture called Internet Service Provider (ISP) centric Content Delivery (iCODE) by which an ISP can provide content delivery services as well. iCODE can provide efficient content delivery services since an ISP can cache the contents in routers with storage modules considering traffic engineering and the locality of the content requests. Compared with CDN and P2P systems, iCODE can offer reduced delivery latency by placing the contents closer to end hosts, and incentives to ISPs by reducing inter-ISP traffic and allowing traffic engineering. We also discuss the technical and business issues to realize the iCODE architecture.

Session 3: The future internet is for all

S3.1 Invited paper: Can computational thinking reduce marginalization in the future internet?

Peter Wentworth (Professor, Rhodes University, South Africa)

Maths is presently regarded as the key driver that underpins Science, Education and Technology (SET) skills. In spite of significant studies, investment and efforts, math skills and widespread enthusiasm for SET remain elusive. In South Africa's disadvantaged communities, poor quality maths teaching and poor maths performance, both legacies of past political engineering, further fuel marginalization.

Computational thinking is a new characterization of some specific procedural thinking, abstraction, problem solving and organizational skills that are finding their way from computer science programs into other fields.

The paper describes our refocus of content in BingBee, a SET skill-building kiosk project targeting disadvantaged communities. As we shift to emphasize computational thinking more explicitly, we speculate that these skills could complement, and perhaps eventually displace, some elements of maths as the dominant driver of SET.

The confluence of better tools, open service interfaces, and the rapid spread of handsets and devices into marginalized communities is an opportunity to build more widespread computational thinking skills. This could in turn facilitate a future Internet which is more inclusive, and in which users are able to create their own services.

S3.2 Invited paper: Challenges the Internet poses to the policymaker

Arun Mehta (President, Bidirectional Access Promotion Society, BAPSI, India)

This paper addresses policymakers at national and international levels -- regulators, standards bodies, politicians – arguing that there is no “beyond” the Internet. With the Internet so intimately intertwined with the lives of people, being used to build the backbone of large, important communities, an attempt to replace it with a new network would generate immense friction, and cost a lot. The transition would take long, because lots of complex software would need to be written, disrupting critical processes of the economy, indeed of governance. A plethora of regulators with very different manners and degrees of control would have to learn to work together at an international level, otherwise we might revert to the lawlessness of the Internet. The lost opportunity of Minitel, the botched attempt to look beyond the Internet in the 1990s via X.400 and the bankruptcy of large telecommunication companies in the wake of the dotcom boom are useful in appreciating the historical context and learning lessons from. Instead of looking beyond, the ITU should play a constructive role vis-à-vis the Internet. Suggestions presented are elimination of spam, and making the Internet accessible to all. These make commercial sense too.

S3.3 Participatory Approach To The Reduction Of The Digital Gap In Amazon Region of Ecuador In The Framework Of The "Innovation For Development" Program

Alessandro Galardini (Politecnico di Torino, Italy); Benedetta Fiorelli (Politecnico di Torino - iXem Labs, Italy); Salvatore Pappalardo (University of Padova, Italy); Daniele Trinchero (Politecnico di Torino, Italy)

This work illustrates the methodological approach followed in the Province of Orellana, Eastern Ecuador, for the realization of a telecommunication network infrastructure between the capital of the Province, the city of Puerto Francisco de Orellana (also known as El Coca), and some peripheral communities located in the surrounding of the tropical moist forest. The project has been implemented in one of the poorest countries of Latin America, in a remote and disadvantaged area where the lack of communication infrastructures and the absence of almost all public services generates a strong migration towards the capital. In this context, in 2008, it was conceived a project for the development of a communication system that allows the provisioning of basic intranet services for distance learning, telemedicine and internet connectivity. The main scope of the project was the development of an approach focused on the technological transfer to the local population, to start a reduction process of the digital gap in the area. The aim of the project has been achieved thanks to the direct enrolment of local municipalities, small entrepreneurs, communities and local NGO. The technological transfer to local players and the choice of a suitable platform, designed for a simplified, low cost management, guarantee the sustainability and scalability of the project. The declaration of interest in the infrastructure by the Municipality enables the economic sustainability of the project.

Session 4: Protocol evolution and the future internet

S4.1 Invited paper: A vision on the information and communication technologies (ICT) using cloud computing environment

Hiroshi Yasuda, Professor (Tokyo Denki University, Japan)

The government of Japan has announced the new ICT policy in June 2010. One of the points of the new policy is to start the 3D motion image content market in order to create new key industries in the near future as 3D motion image content will become most powerful media for CGM (Consumer Generated Media). In order to activate 3D motion image content industries, the development of an effective and simple tool for making 3D motion image content even by non-experienced people, is required. The Digital Movie Director (DMD) developed by the author, is being evolved as such an effective and simple tool. However, the big computational power requirement in making 3D motion image content has prevented DMD from being widely deployed. The cloud computing technology is supposed to solve this problem, thus, in this paper, the future prospects of the 3D motion image content industries with the cloud computing technology will be explained.

S4.2 Hybrid Circuit/Packet Networks With Dynamic Capacity Partitioning

Chaitanya S. K. Vadrevu (University of California, Davis, USA); Menglin Liu (University of California, Davis, USA); Massimo Tornatore (Politecnico di Milano, Italy); Chin Guok (Energy Sciences Network, USA); Evangelos Chaniotakis (Energy Sciences Network, USA); Inder Monga (Energy Sciences Network, USA); Biswanath Mukherjee (Dept. of Computer Science - University of California Davis, USA)

In this paper, we consider hybrid circuit/packet networks. A hybrid circuit/packet network consists of a circuit network co-existing with a packet network; generally the packet network is embedded on top of the circuit network. However, in certain cases such as the DOE energy sciences network (ESnet) [4], the circuit network and the packet network are deployed side-by-side (e.g. they have common end-node sites and equipment), but they are logically separate and they may have physically disjoint links. Currently, there is no capacity sharing between the packet and the circuit sections of the networks. In this paper, we propose and investigate the characteristics of schemes that enable efficient capacity partitioning between packet and circuit networks while ensuring survivability and robustness of the services. We conduct simulative experiments on ESnet topology with realistic traffic demands. We observe that capacity partitioning between packet and circuit networks enables to support services with enhanced quality of service and robustness along with improved resource utilization.

S4.3 A New Protocol Layer for User Space Functionality

Pankaj Chand (Independent Researcher, India)

Evolution of the Internet user has brought attention to the lack of standards for ideal levels of user interaction. The core Internet architecture has not evolved much since its inception, and its user-driven limitations typically constrain one's personal computing infrastructure so that the goals of pervasive and ubiquitous computing are only incipiently achieved. We propose to consider the user's image, or user space, as a significant entity in the Internet model by introducing a new layer of protocols into the Internet protocol stack to support future usage in the Internet. We also present the Identifier/Interlocutor/Locator split architecture for flexible addressing. Standards for such architectures would provide generic user support across heterogeneous networks.

S4.4 Quality of Service in the Future Internet

Jorge Carapinha (PT Inovação S.A., Portugal Telecom Group, Portugal, Portugal); Christoph Werle (Universität Karlsruhe (TH), Germany); Konstantin Miller (Berlin Institute of Technology, Germany); Roland Bless (Karlsruhe Institute of Technology (KIT), Germany); Andrei Bogdan Rus (Technical University of Cluj-Napoca, Romania); Virgil Dobrota (Technical University of Cluj-Napoca, Romania); Horst Roessler (Alcatel-Lucent, Germany); Heidrun Grob-Lipski (Alcatel-Lucent, Germany)

Whatever the Network of the Future turns out to be, there is little doubt that QoS will constitute a fundamental requirement. However, QoS issues and the respective solutions will not remain unchanged. New challenges will be raised; new ways of dealing with QoS will be enabled by novel networking concepts and techniques. Thus, a fresh approach at the QoS problem will be required. This paper addresses QoS in a Future Internet scenario and is focused on three emerging concepts: Network Virtualization, enabling the coexistence of multiple network architectures over a common infrastructure; In-Network Management, improving scalability of management operations by distributing management logic across all nodes; the Generic Path based on the semantic resource management concept, enabling the design of new data transport mechanisms and supporting different types of communications in highly mobile and dynamic network scenarios.

Session 5: Service innovations in the future internet

S5.1 Cross-Language Identification Using Wavelet Transform and Artificial Neural network*

Shawki A. Al-Dubaee (Aligarh Muslim University, India); Nesar Ahmad (A.M.U. Aligarh, India, India)

With the advent of the Internet, search engines were developed for English language because English language was a lingua franca. Currently, most of popular search engines such as Google and Yahoo! are available in more than 50 languages. However, these search engines have received less attention in South Asian languages especially, Urdu language. In this paper, we propose a novel approach for feature extraction and classification of queries in cross-language search engines. This novel approach presents an automatic method for classification of English and Urdu languages identification. The classifier used is a three-layered feed-forward artificial neural network and the feature vector is formed by calculating the wavelet coefficients. Three wavelet decomposition functions (filters), namely Haar, Bior 2.2 and Bior 3.1 have been used to extract the feature vector set and their performance results have been compared. The performance results of the Haar filter have given superior results than other filters.

S5.2 GeoHybrid: a hierarchical approach for accurate and scalable geographic localization*

Ibrahima Niang (University Cheikh Anta DIOP of Dakar, Senegal); Bamba Gueye (Université Cheikh Anta Diop de Dakar, Senegal); Bassirou Kasse (Université cheikh anta diop, Senegal)

Geographic location and Grid computing are two areas that have taken off in recent years, both receiving a lot of attention from research community. The Grid Resource Brokers, which tries to find the best match between the job requirements and the resources available on the Grid, can take benefits by knowing the geographic location of clients, for a considerable improvement of their decision-taking functions. A measurement-based geolocation service estimates host locations from delay measurements taken from landmarks, which are hosts with a known geographic location, toward the host to be located. Nevertheless, active measurement can burden the network. Relying on database-driven geolocation and

	active measurements, we propose GeoHybrid. GeoHybrid estimates the geographic location of Internet hosts with low overhead as well better accuracy with respect to geolocation databases. Afterwards, we propose a geolocation middleware for grid computing. By defining the architecture and the methods of this service, we show that a promising symbiosis may be envisaged by the use of the proposed middleware service for grid computing.
S5.3	Context-Aware Smart Environments Enabling New Business Models and Services *Christian Mannweiler (University of Kaiserslautern, Germany); Jose Simoes (Fraunhofer FOKUS, Germany); Boris Moltchanov (Telecom Italia, Italy)* This work describes innovative smart environments with embedded context-awareness technologies, enabling new business models and consequently the creation of new services. The context-awareness framework presented in this paper is taken from the results of an EU Framework Programme (FP) 7 Information and Communications Technologies (ICT) project. Major novelties include a business shift from traditional and conventional telecommunication or ICT services towards highly personalized, customized and user targeted services, empowered by a myriad of pervasive and ubiquitous interconnected environments employing various kinds of context information. In this work, we show how these context data can be technically made available as a service and business enabler and be used by any entity or application built within these environments, using context for adapting service logic or for targeted service customization. Moreover, it considers customer's needs and privacy aspects, providing users with a more immersive and less intrusive experience at the same time.
S5.4	Innovative Tangible User Interface as a Mean for Interacting Telecommunications Services *Klemen Peternel (University of Ljubljana, Faculty of Electrical Engineering, Slovenia); Luka Zebec (University of Ljubljana, Faculty of Electrical Engineering, Slovenia); Andrej Kos (University of Ljubljana, Slovenia)* While modern telecommunications are ever more useful and even necessary in everyday life, not all groups of people are equally capable of using them. Due to inevitable demographic changes the elderly are growing in numbers, yet they are not very well served by user interfaces for the various telecommunications tools. The prime target group for our proposed technology is people with cognitive and motor disabilities, whether due to age, illness or traumatic events. They require a user interface which enables them to make or redirect calls, create conferences, set forwarding and/or access different voice XML services - without the complexity of keyboards or menus with tree structures. The motivators behind are: simplicity, accessibility, usability and efficiency - all in the scope of potential user groups and usage scenarios. The key enablers are Next Generation Network (NGN) open interfaces and Near Field Communication (NFC) technology as a part of Radio Frequency Identification (RFID) family.

Session 6: Regulation, standardization and stakeholder participation	
S6.1	How Many Standards in a Laptop? (And Other Empirical Questions) *Brad Biddle (Arizona State University, USA); Andrew White (Arizona State University, USA); Sean Woods (Arizona State University, USA)* An empirical study which identifies 251 technical interoperability standards implemented in a modern laptop computer, and estimates that the total number of standards relevant to such a device is much higher. Of the identified standards, the authors find that 44% were developed by consortia, 36% by formal standards development organizations, and 20% by single companies. The intellectual property rights policies associated with 197 of the standards are assessed: 75% were developed under "RAND" terms, 22% under "royalty free" terms, and 3% utilize a patent pool. The authors make certain observations based on their findings, and identify promising areas for future research.

S6.2 A user-centric approach to QoS regulation in future networks*

Eva Ibarrola (University of the Basque Country, Spain); Jin Xiao (University of Waterloo, Canada); Fidel Liberal (University of the Basque Country, Spain); Armando Ferro (University of the Basque Country, Spain)

The evolution of current networks to Next Generation Networks (NGNs) constitutes arguably the most significant transformation in the Telecommunication sector in recent decades. Quality of Service (QoS) is one of the key aspects in this evolution. In the NGN environment, networks are designed to be multiservice, supporting a wide range of premium services. Each of these services may have different QoS requirements which should be established based on the overall end user's perception. In this emerging context, novel QoS policies are required to adapt the traditional QoS regulatory model to the new scenario. This paper presents an approach to identify key factors that contributes to the development of future Internet quality of service regulation. A case study on the application of our user-centric QoS model to the Internet QoS regulation in Spain is described. The results of the study demonstrate the need for adapting current regulatory frameworks in order to ensure competition, pluralism and diversity in the new network environment.

S6.3 Competition and Cooperation in the formation of Information Technology Interoperability Standards: A Process Model of Web Services Core Standards*

Jai Ganesh (Infosys Technologies, India)

Standards formation is a key dimension in the competitive strategy of ICT firms, as a successful strategy would result in the emergence of favorable IT interoperability standards. This paper examines the standardization efforts of core Web services standards and the results indicate that resource dependencies and strategies adopted by dominant firms to extend their platforms influence the standards formation process. Communities of practice and standard-setting bodies are leveraged by dominant firms in the formation and adoption of standards. We propose a process model of standard setting consisting of five intertwined states: resource pooling, linkages, signaling and implementation, institutionalization, and extension.

Session 7: Radio technologies and the future internet

S7.1 Performance Comparison of Intelligent Jamming In RF (Physical) LAYER with WLAN Ethernet Router and WLAN Ethernet Bridge

Rakesh Jha (SVNIT, India); Upena D. Dalal (Sardar Vallabhbhai National Institue of Technology, Surat, India)

The very nature of Radio Frequency (RF) technology makes Wireless LANs (WLANs) open to a variety of unique attacks. Most of these RF-related attacks begin as exploits of Layer 1 (Physical - PHY) & Layer 2 (Media Access Control - MAC) of the 802.11 specification, and then build into a wide array of more advanced assaults, including Denial of Service (DOS) attacks. In Intelligent Jamming the jammer jammed physical layer of WLAN by generating continuous high power noise in the vicinity of wireless receiver nodes. In this paper, we study the threats in an Intelligent jamming Comparison with WLAN Ethernet Router and WLAN Ethernet Bridge and the security goals to be achieved. We present and examine analytical simulation results for the throughput for different scenario performance, using the well-known network simulator OPNET 10.0 and OPNET Modeler 14.5 for WiMAX Performance. IEEE 802.11b has two different DCF modes: basic CSMA/CA and RTS/CTS. Intelligent jamming, which jams with the knowledge of the protocol, the jamming describe in our paper is based on the basis of Fake AP Jamming. When we have applied same concept in WiMAX system under the influence of jamming we have received same effect of router performance.

S7.2	Self-organized Spectrum Chunk Selection Algorithm for Local Area LTE-Advanced *Sanjay Kumar (Birla Institute of Technology, Mesra, India); Yuanye Wang (Aalborg University, Denmark); Nicola Marchetti (Aalborg University, Denmark)* This paper presents a self organized spectrum chunk selection algorithm in order to minimize the mutual intercell interference among Home Node Bs (HeNBs), aiming to improve the system throughput performance compared to the existing frequency reuse one scheme. The proposed algorithm is useful in Local Area (LA) deployment of the Long Term Evolution-Advanced (LTE-A) systems, where the HeNBs are expected to be deployed randomly and without coordination in distributed manner. The result shows that the proposed algorithm effectively improves the system throughput performance with very limited signaling exchange among the HeNBs,
S7.3	On the Design of Ultra Wide Band Antenna Based on Fractal Geometry *Pranoti Bansode (DIAT (DU), India); Raj KumarS (University of Pune, India)* This paper presents ultra wide band circular fractal antenna. The antenna has been fed with coplanar waveguide (CPW) feed. This fractal antenna has been designed and fabricated on FR4 substrate $\varepsilon r = 4.3$ and thickness $h = 1.53$ mm with initial diameter of solid circular disc 15 mm. The experimental result of circular fractal antenna exhibits the ultra wide band (UWB) characteristic from 3.295 GHz to 13.365 GHz corresponds 120.88 % impedance bandwidth. The first resonant frequency of fractal antenna shifted to 3.75 GHz in comparison to first resonant frequency 4.31 GHz of conventional simple circular disc monopole antenna. This indicates the size reduction of antenna. The measured radiation pattern of this fractal antenna is nearly omni-directional in azimuth plane throughout the band. This type of antenna can be useful for UWB system and sensing applications.
S7.4	Design of Inscribed Square Circular Fractal Antenna with adjustable Notch-Band Characteristics *Raj KumarS (University of Pune, India); Kailas Sawant (DIAT (DU), India); Jatin Pai (DIAT (DU), India)* This paper presents the design of an inscribed square circular fractal antenna with notch having adjustable frequency characteristics. The position and width of the notch band can be adjusted in the entire operating band. A prototype of the antenna has been designed on FR4 substrate with $\varepsilon r = 4.3$ and thickness $h = 1.53$ mm with a U-shape slot in coplanar waveguide feed of length $L = 11$ mm and slot width $W = 0.4$ mm. The experimental result of this antenna exhibits ultra-wide band characteristics from frequency 3.1 GHz to 15.0 GHz. The notch in operating band helps to reduce the interference with the frequency bands of Worldwide Interoperability for microwave access (WiMAX). The simulated and experimental return loss are found in good agreement. The experimental radiation of this antenna in azimuth plane is nearly omni-directional. This proposed inscribed square circular fractal antenna with notch can thus be used for Ultra wide band (UWB) system, microwave imaging and precision position system.
S7.5	Resonant Frequencies Of A Circularly Polarized Nearly Circular Annular Ring Microstrip Antenna With Superstrate Loading And Airgaps *Jayashree Shinde (Sinhgad Academy of Engineering,Kondhwa,Pune, India); PRATAP SHINDE (NMIMS University, India); RAJ KUMAR (DAIT university, India); Mahadeo Uplane (Shivaji University, India); BrajKishor Mishra (NMIMS University, India)* This paper presents an analysis for the resonant frequencies and its various harmonics of a nearly Circular Annular Ring Microstrip Antenna (ARMSA) with and without air gaps and superstrate loadings. This ARMSA is studied for various radii of the inner and outer radiating circular edges of disc. Three such nearly circular ARMSA are analyzed with an Aspect Ratio of 0.98. By diagonal feeding at the center of ARMSA, circular polarizations are observed with generation of fundamental resonant frequency and higher order modes. Multilayer dielectric ARMSA with and without air gaps are analyzed using effective quasi-static capacitance approach and compared with experimental results using Vector Network Analyzer to provide less than 1% deviation in the resonant frequency. Also the full wave simulated and experimental readings go in good agreement for all the three nearly circular ARMSA for with and without air gaps along with superstrate loadings of various height and dielectric constant material as cover. This closed form model of nearly circular ARMSA is suitable for covered antenna devices CAD and is directly applicable for integration of microstrip antennas beneath protective dielectric superstrates in portable wireless equipments.

Session 8: Future internet and the environment

S8.1 A scheme for Disaster Recovery in Wireless Networks with Dynamic Ad-hoc Routing

Guowei Chen (Waseda University, Japan); Aixian Hu (Waseda University, Japan); Takuro Sato (Waseda University, Japan)

This paper proposes a hybrid network scheme combining ad-hoc networks into cellular networks. The scheme is aimed to help the networks to recover to service as much as possible after a disaster strike, by maintaining the connection between Base Stations (BSs) and nodes via multi-hopping, where if a node cannot connect to a BS directly, it switches its working mode from cellular mode to ad-hoc mode. A location-based routing protocol has been proposed for building a route from the node to the BS. Simulation results shows that even only a small part of the nodes can directly connect to a BS, most of the nodes can find a route to a BS via multi-hopping. And it is found that it outperforms a previously proposed solution which is via beaconing in terms of resistance to mobility.

S8.2 A New Study on Network Performance under Link Failure in OPS/OBS High-Capacity Optical Networks

Felipe Rudge Barbosa (University of Campinas - Unicamp, Brazil); Indayara Martins (Unicamp, Brazil); Edson Moschim (State University of Campinas - Unicamp, Brazil)

In this work we analyze the performance and sensitivity to link failure of metropolitan networks based on the technology of optical packet/burst switching (OPS/OBS). We use ring and mesh topologies to evaluate through analytical modeling and computer simulations the impact of link failure on each topology. We adopt the parameters average number of hops and packet loss fraction to evaluate network performance. It is observed that mesh topologies with triple connection node configuration (3x3) are more robust; consequently in case of link failure the impact of lost data is minimum compared with the other topologies and configurations considered.

S8.3 Business Scheme for Shifting from Existing Networks to Trusted Green Networks

Yoshitoshi Murata (Iwate Prefectural University, Japan)

The future networks have yet to be defined. These are not represented by the next generation of the Internet and they need to satisfy requirements for the sustainability of mankind. These are called Trusted Green Networks (TGNs) in this paper. Although TGNs offer marvellous concepts and excellent functions, they will not always be widely deployed. There have been several initiatives to develop future networks. Their purpose is developing innovative technologies, but not including deployment schemes. We selected "sustainability", "trust and security", and "solving the digital divide by location" as concepts underlying TGNs and clarified their requirements. A business scheme is also proposed that boosts the shift from existing networks to TGNs. And the network layer model of TGNs is introduced.

S8.4 Innovative ad-hoc wireless sensor networks to significantly reduce leakages in underground water infrastructures

Daniele Trinchero (Politecnico di Torino, Italy); Riccardo Stefanelli (Politecnico di Torino - iXem Labs, Italy); Luca Cisoni (iXem Labs, Politecnico di Torino, Italy); Abdullah Kadri (QU Wireless Innovations Center, Qatar); Adnan Abu-Dayya (QUWIC, Qatar); Mazen Omar Hasna (Qatar University, Qatar); Tamer Khattab (Qatar University, Qatar)

This paper presents an ICT solution to overcome the problem of water dispersion in water distribution networks. Leakage prevention and breaks identification in water distribution networks are fundamental for an adequate use of natural resources. Nowadays, all over the world, water wasting along the distribution path reaches untenable percentages (up to 80 % in some regions). Since the pipes are buried within the terrain, typically only relevant breaks are considered for restorations: excavations are very expensive and consequently the costs to identify the position of the leakage or just the position of the pipe itself are too high. To address this problem, and simplify the leakage identification process, the authors have designed a wireless network system making use of mobile wireless sensors able to detect breaks and reveal unknown tracks and monitor the pressure spectrum of the fluid flowing in the pipe. The sensors transmit the acquired data from the terrain to the surface by use of a wireless connection. On the surface ground there are stations that receive the signal, process it, and communicate with a central unit where necessary intelligent signal processing techniques are used to detect leakage sources. Compared to other leakage detection solutions already available in the market (such as: Ground penetrating radar (GPR), pure acoustic techniques and tracer gases), the proposed technique appears very efficient and much more inexpensive.

Poster Session: Showcasing innovations for future networks and services	
P.1	Beyond the WiFi: Introducing RFID system using IPv6 *Labonnah F Rahman (Universiti Kebangsaan Malaysia, Malaysia); Mamun B.I Reaz (Universiti Kebangsaan Malaysia, Malaysia); Mohd Alauddin Mohd Ali (Universiti Kebangsaan Malaysia, Malaysia); Mohammad Marufuzzaman (Universiti Kebangsaan Malaysia, Malaysia); Muhammad Raisul Alam (Universiti Kebangsaan Malaysia, Malaysia)* RFID System suffers from limited address space and local mobility. Moreover, it is a monopoly business with few vendors, which are trying to dominate the market with proprietary standard of RFID reader. The proposed system will replace the expensive RFID reader with cheap Wireless Network Interface Card (WNIC). For the purpose, an innovative scheme for RFID tagging system is introduced which will be benefited by well-defined WiFi protocol. IPv6 (Internet Protocol version 6) address will be used as product identifier. This will provide a universal identity of the objects with seamless global mobility. The EPC (Electronic Product Code) will directly map to IPv6 address by using an auto configuration method. So that 64 bit EPC will take the place of the EUI-64 portion of IPv6 address. The proposed system suggests the mechanism of reducing significant cost, physical location detection and usage of global unique address, which will also be compatible with existing EPC addressing scheme.
P.2	Comparative analysis of extended geographical wireless networks based on Diversity transmission systems *Daniele Trinchero (Politecnico di Torino, Italy); Alessandro Galardini (Politecnico di Torino, Italy); Riccardo Stefanelli (Politecnico di Torino - iXem Labs, Italy)* The paper analyses the performance of wireless networks working over unlicensed frequency ranges, making use of diversity transmission systems, as an effective means to increase outdoor coverage capabilities in rural scenarios. To this purpose, a network has been designed and realized, providing coverage to a wide rural area in the hills of Piedmont, a region located in North-Western Italy. Once constructed, the network performance has been monitored and characterized, in terms of coverage capabilities, signal quality, and noise immunity.
P.3	SIP Trunking the route to the new VoIP services *Ivan Gaboli (Italtel, Italy); Virgilio Puglia (Italtel, Italy)* This work gives an overview of SIP Trunking solution and explains the existing difficulties in implementing VoIP services, when this architecture is deployed in multivendor environment. The causes of these problems are explained with two existing approaches used by carrier to solve interoperability: they are Full Jacket SIP-Trunk and Customized SIP Trunk. A solution to cover the lackings of SIP standards is the introduction of a SIP adaptation device called "Inter-Domain Adaptation Device" which will increase the potentiality of SIP-Trunking based solutions. It is also proposed a method for assessing the complexity of different approaches to SIP-Trunking applications.
P.4	Global e-Public Service (GePS) *Priyantha K Weerabahu (DMS Electronics, Sri Lanka)* This paper proposes a Global e-Public Service (GePS) to electronically deliver public services in a more efficient way by maximizing the utilization of resources and providing all countries a common platform to help its citizens and reduce the inequity in the world. Most of primary services which all governments need to provide to its citizens could be delivered electronically through e-government initiatives. These e-services were identified through research of leading e-government portals and a reference model for an e-government portal was developed. Resources required by each government to provide services common to all are a wasteful exercise. The Global e-Public Service defines a mechanism to develop these electronic public services centrally by a global agency and to implement them by various governments as integrated and decentralized systems around the world.

P.5	Integrating Wireless Sensor Networks and Mobile Ad hoc Networks for an Enhanced End-user Experience *Saba Hamedi (Concordia University, Canada); Mohammadmajid Hormati (Concordia University, Canada); Roch Glitho (Concordia University, Canada); Ferhat Khendek (Concordia University, Canada)* Wireless sensor networks (WSNs) sense and aggregate ambient information (e.g. space, environment or physiological data). Ambient information can enhance end-user experience and is made available to end-user applications (which may reside in another network) via gateways. Gateways are usually centralized and fixed. Mobile ad hoc networks (MANETs) are networks that can be deployed "on the fly". They are useful in situations such as emergency response operations. When the ambient information collected by WSNs is intended for applications residing in a MANET, centralized and fixed gateways are not practicably feasible. This paper proposes an overall two-level overlay architecture to integrate WSNs (with mobile and distributed gateways) and MANETs, for an enhanced end-user experience. It also proposes a new architecture to interconnect the two overlays of the overall architecture. Motivating scenarios are presented, requirements are derived, and the two-level overlay is discussed along with the proposed interconnection architecture. The prototype is also presented.
P.6	Telecommunications Business Model For Converged Networks Focusing Final Users *Cledson Sakurai (University of Sao Paulo, Brazil); Moacyr Martucci Jr (University of São Paulo, Brazil); André Hiyuiti Hirakawa (University of São Paulo, Brazil)* The telecommunications converged networks, specially inside of new Internet environment, makes possible supply a plenty of services to users, as voice and multimedia services with assured perceived quality of services (QoS) through different access technologies, however this brings a complex scenario in terms of technology in order to fit the user perceived QoS needs to appropriate technical QoS requirements, and also delivery the right service to a particular user. Therefore, to enable the appropriate service delivery for each user, this article proposes a business model for the telecommunications segment, aiming delivery services according to particular User Service Level Agreements (USLAs), prepared transparently of the technologies involved, and using QoS parameters according to the users literacy. The proposed business model considers four providers: Services, Infrastructure, Content and Access aiming to facilitate the relationship between users and providers, and to clarify the roles and responsibilities of each actor, as well. This paper presents the proposed business model, and discuss the needs for an regulatory frameworks necessary to meet the requirements of proposed business model with focus on users.
P.7	On Demand Fine Grain Resource Monitoring System for Server Consolidation *Arnupharp Viratanapanu (The University of Tokyo, Japan); Ahmad Kamil Abdul Hamid (The University of Tokyo, Japan); Yoshihiro Kawahara (The University of Tokyo, Japan); Tohru Asami (The University of Tokyo, Japan)* Server consolidation and virtualization technology are used in modern data center as a method to improve the server utilization rate by encouraging the sharing of physical re- sources between virtual machines (VM). Although, this can be realized by plenty of virtualization softwares available in markets, creating a good management policy for the server consolidation environment is still a challenging research problem. The policy should be able to maintain the high re- source sharing rate, while keeping the performance drop rate as low as possible, which are totally contradict requirements. In order to progress the research in this area, a monitoring system, which can effectively monitor current system state and collect essential information, is indispensable. However, due to many differences in use case scenarios, monitoring systems designed for conventional management domain can- not be applied directly. This needs us to reconsider the design of the monitoring system. This paper proposes some key information that is still missing from existing monitoring systems as well as the design of Pantau, a monitoring sys- tem designed for capturing information necessary for server consolidation.

P.8	Describing and Selecting Communication Services in a Service Oriented Network Architecture *Rahamatullah Khondoker (University of Kaiserslautern, Germany); Bernd Reuther (University of Kaiserslautern, Germany); Dennis Schwerdel (University of Kaiserslautern, Germany); Abbas Siddiqui (University of Kaiserslautern, Germany); Paul Müller (University of Kaiserslautern, Germany)* Today networks offer communication services ranging from a rather simple and unsecure one to secure and reliable data transmission for communicating on the network. In the future, it is expected that networks will offer a large number of different communication services. With so many services available, determining which service to select and use becomes much more difficult. Here we propose a description schema including an ontology for describing communication services. For service selection a decision making process called Analytic Hierarchy Process (AHP) is utilized which is specially adapted and extended for automatic processing.
P.9	Virtualized passive optical metro and access networks *Jun Shan Wey (Nokia Siemens Networks, USA); Curt Badstieber (Nokia Siemens Networks, Germany); Ashwin A Gumaste (Indian Institute of Technology, Bombay/Massachusetts Institute of Technology, India); Ali Nouroozifar (Nokia Siemens Networks, Canada); Antonio Teixeira (Nokia Siemens Networks, Portugal); Klaus Pulverer (Nokia Siemens Networks, Germany); Harald Rohde (Nokia Siemens Networks, Germany)* This paper outlines and proposes the vision of an open architecture framework for a virtualized passive optical metro access network. An open lambda environment based on this architecture framework and its benefits to different stakeholders are described. The paper provides analogous comparisons with existing proposals and discusses two practical examples of applying this architecture framework in a metro-access network and in mobile backhaul. Socio-economic impacts, challenges, as well as possible directions in standardization are analyzed.
P.10	Adaptive Resource Allocation for Real-Time Services in OFDMA Based Cognitive Radio Systems *Dhananjay Kumar (Anna University, India); Shanmugam Mahalaxmi (Anna University Chennai, India); Jayakumar Sharad Kumar (Anna Univeristy, India); Rangarajan Ramya (Anna University Chennai, India)* In this paper an adaptive resource allocation algorithm in OFDMA based cognitive radio (CR) system is proposed that not only meets the quality of service (QoS) of real-time (RT) services but also increases the dynamic capacity of the system. In contrast to existing algorithms for multi-user OFDM systems which are unable to guarantee maximum sum data rate when applied to CR system in real-time, the proposed joint sub carrier and power allocation algorithm (JSPA) ensures maximum sum data rate under power constraint while improving the system capacity. We model the resource allocation problem with the goal of maximizing the overall system throughput. By judicially assigning resource in primary and secondary sources based on the types of application JSPA demonstrates that the system efficiency and throughput can be dynamically enhanced. JSPA is simulated under two types of services: constant bit rate (CBR) and variable bit rate (VBR).
P.11	All Photonic Analogue to Digital and Digital to analogue conversion techniques for digital radio over fibre system applications *Seyed Reza Abdollahi (Brunel University, United Kingdom); Hamed Saffa Al-Raweshidy (University of Brunel, United Kingdom); Sied Mehdi Fakhraie (University of Tehran, Iran); Rajagopal Nilavalan (Brunel University, United Kingdom)* Wideband electronic analogue to digital conversion (ADC) systems have critical problems encountered in high-frequency broadband communication systems that the recent electronic ADCs (EADC) have experienced those such as uncertainty of sampling time. In this paper, an all photonic sampling and quantization ADC and photonic digital to analogue conversion system with six effective number of bits (ENOB) is designed. By using this photonic ADC (PADC), a novel digital radio over fibre link for wireless radio frequency (RF) signal transportation over 20 Km single mode fibre has been designed whose performance is investigated in this paper. In the digital radio over fibre, the dynamic range is independent of the fibre length.

P.12	Enhancing CyberSecurity for Future Networks *Raj Puri (Yaana Technologies, USA); Anthony Rutkowski (Georgia Institute of Technology, USA)* Next Generation Networks, including specialized implementations such as Cloud Computing and Smart Grids, must be significantly more robust than today's networks - with "baked in" capabilities supporting an array of assurance and cybersecurity capabilities. One of the most significant emerging means of achieving these capabilities is by applying an ensemble of new specifications being developed under the aegis of ITU-T Study Group 17 known as CYBEX - the Cybersecurity Information Exchange Framework. This paper describes the CYBEX Framework, how it came into existence, and its potential application to Future Networks.
P.13	Towards a Service-Oriented Network Virtualization Architecture *May El Barachi (Concordia University, Canada); Nadjia Kara (École de Technologie Supérieure, Canada); Rachida Dssouli (UAEU, UAE)* Network virtualization is an emerging concept that enables the creation of several co-existing logical network instances (or virtual networks) over a shared physical network infrastructure. There are several motivations behind this concept, including: cost-effective sharing of resources; customizable networking solutions; and the convergence of existing network infrastructures. In this paper, we analyze the existing (conventional and virtualized) business models and propose a new business model for virtual networking environments. Our proposed model is a service-oriented hierarchical model, in which different levels of services (i.e. essential services, service enablers, service building blocks, and end-user services) offered by various players, can be dynamically discovered, used, and composed. Using this business model as basis, we also define a layered service-oriented network virtualization architecture and discuss some of the issues related to its operation.
P.14	Thin Apps Store for Smart Phones Based on Private Cloud Infrastructure *Ashish Tanwer (Thapar University, India); Abhishek Tayal (Thapar University, India); Muzahid Hussain (Thapar University, India); Parminder Reel (Thapar University, India)* A novel approach to implement cloud computing for smart phone devices has been presented based on Eucalyptus, an open source cloud-computing framework that provides infrastructure as a service (IaaS). It has full support of Virtualization and is Amazon Web Services interface compatible. A private cloud has been designed using Eucalyptus to develop a smart phone application store. The architecture, physical and network implementation of Eucalyptus Private Cloud on Intel based platform has been discussed in details. We have developed two sample thin applications for mobile based on mobile learning that can be downloaded from our private cloud using Amazon Web Services Platform (PaaS). These thin apps use the private cloud as computing platform, and perform better even on low processing smart phones.

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