# INNOV 2012

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**INNOV 2012 Editors**

Pascal Lorenz, University of Haute Alsace, France

# INNOV 2012

# Foreword

The First International Conference on Communications, Computation, Networks and Technologies [INNOV 2012], held between October 21-26, 2012 in Venice, Italy, aimed at addressing resent research results and forecasting challenges on selected topics related to communications, computation, networks and technologies.

Considering the importance of innovative topics in today's technology-driven society, there is a paradigm shift in classical-by-now approaches, such as networking, communications, resource sharing, collaboration and telecommunications. Recent achievements demand rethinking available technologies and considering the emerging ones.

We take here the opportunity to warmly thank all the members of the INNOV 2012 Technical Program Committee, as well as the numerous reviewers. The creation of such a high quality conference program would not have been possible without their involvement. We also kindly thank all the authors who dedicated much of their time and efforts to contribute to INNOV 2012. We truly believe that, thanks to all these efforts, the final conference program consisted of top quality contributions.

Also, this event could not have been a reality without the support of many individuals, organizations, and sponsors. We are grateful to the members of the INNOV 2012 organizing committee for their help in handling the logistics and for their work to make this professional meeting a success.

We hope that INNOV 2012 was a successful international forum for the exchange of ideas and results between academia and industry and for the promotion of progress in communications, computation, networks and technologies.

We are convinced that the participants found the event useful and communications very open. We also hope the attendees enjoyed the charm of Venice, Italy.

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# INNOV 2012

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# Automated Generation of Movement Trace for Aircraft Ad Hoc Networks

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***Abstract* — Following our previous our work [1] to get realistic movement trace for aircraft ad hoc networks, in this paper, we propose how to generate the automated scenarios that are supposed to be used in simulation. Even though an approach for realistic mobility model was proposed as well as implemented in our previous work, it lacks of repeatability and takes long time to get the trace file due to manual configuration for parameters of all aircrafts. Thus, automated scenario generation is strongly required for diverse different scenarios in a way of convenient and easy configuration. To achieve this research goal, some additional functions are proposed and implemented under the same software architecture as the previous work.**

***Keywords-movement trace; aircraft ad hoc networks.***

## I. INTRODUCTION

As many researches for AANET (Aircraft Ad Hoc Networks) [2] have been recently conducted, performance evaluation through network simulator is regarded as one of essential tasks while developing new enhanced protocols or network architecture. Nevertheless, since current popular network simulators are not sufficiently equipped for adequately simulating the mobility of moving aircraft, corresponding components for each simulator is strongly required. Among required components, movement trace is one of big research challenges because there are little available sources. Thus, a new source of realistic mobility traces for simulation should be concerned significantly. This requirement was also observed in research experience from pure mobile ad hoc networks to vehicular ad hoc networks.

Related to movement trace, it can be obtained by either mathematical model or actual observation. For former example, a new model was introduced in [3]. In this work, we propose how to create a position and velocity function for an aircraft, given its starting location and speed, altitude, and heading on each leg of its height. In addition, an optimization procedure is used to fit the position and velocity functions to the supplied data. Even though the proposed scheme in [3] enables the researcher to facilitate the movement information; however, the proposed method is not suitable to generate realistic mobility model. The major reason for this drawback is caused by the following fact that there is a large number of aircrafts with different properties and great changes in flying surrounding situations such as wind. Therefore, despite of complex dynamics of each aircraft and external environments, it should be considered together with a mathematical model; for the realistic movement model, mathematical model presented in [3] cannot accommodate all possible situations. This fact leads to unrealistic movement trace in many scenarios. In the latter case, as a current observation, real-world aircraft will be tracked using on-board or subsidiary devices and vehicle positions recorded at regular intervals. However, while such a mobility model will arguably result in the most realistic aircraft movement, its use is limited by this approaches' inherent limitation to a small set of mobility parameters. Changing only one parameter, e.g., the density of vehicles, and keeping all other parameters unchanged is simply infeasible in reasonably large scenarios. The restriction of trace data on what could be recorded from real-world vehicle movements can be easily overcome by generating such movement traces artificially.

To solve problem mentioned above, we proposed how to use commercial flight simulator for movement trace. As far as we know, this is the first approach to use commercial flight simulator for the research in AANET.

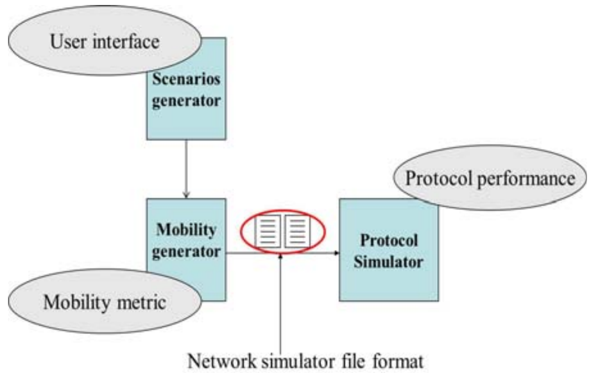


Figure 1. Software architecture

By using flight simulator, actual movement of each aircraft is controlled by each user's manual aviation or automatic flight in the commercial flight simulator depending on dynamic of each aircraft. The software architecture for the movement trace is shown in Figure 1. Through user interface, a user can create several scenarios. And then, mobility generator that is implemented by help of flight simulator is executed to generate raw movement data. Finally, raw data is converted to the adequate format in each network simulator. In Figure 2, we can see the real implementation and experiment results. A user can create each node in developed software and this action brings about

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the creation of aircraft in flight simulator. Similar to this procedure, positional information of waypoints are transferred to corresponding object in flight simulator. After executing simulation, an aircraft starts its aviation and generates the trace for movement. This information is displayed in developed software to prove its operation.

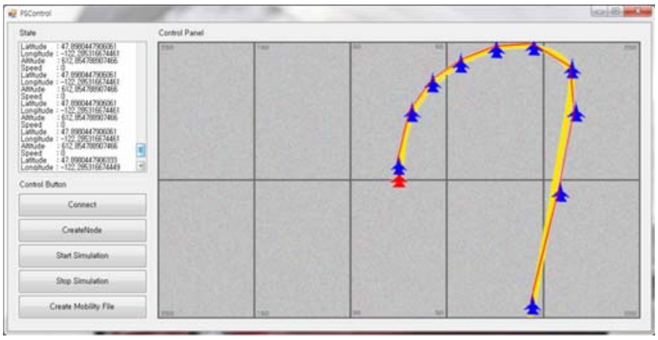


Figure 2. Movement toward waypoint and display of position information at the left side

## II. AUTOMATED TRACE FOR MOVEMENT

By using the works described in the previous section, it is possible for researchers to gather realistic movement trace. But, it is recommended to employ automated scenarios configuration rather than manual one for the diverse scenarios since multiple runs are naturally required in the procedure for performance evaluation. With the same demand, "setdest" [4] external command for mobility generation for random waypoint model is provided in NS-2. In addition, a mobility scenario generation and analysis tool was already released as shown in [5]. In this section, we explain how to extend the current software to generate automated movement trace.

### A. *Analysis of Automated Movement Trace*

The procedure toward automated scenarios requires following information to gather movement trace, number of aircrafts, type of aircrafts, mobility pattern, and simulation time. Figure 3 illustrates the example of user interface for generating scenarios. In this interface, if a user wants to create eight aircrafts totally, four aircrafts are made after creating four ones, respectively. The type of aircrafts can be determined in a group.

### B. *Implementation Issues*

- Initial position of aircraft: When many aircrafts are created, the initial position of them can affect the movement traces. To prevent dense deployment, the position of aircraft is determined in a uniform way where the aircraft were arranged in any position within the square in uniform topology.
- Mobility type: For the mobility pattern, we design two different types. One is random walk model [6] and the other is mission based model. The latter is assumed that a special mission is performed by the corresponding aircraft. Thus, an aircraft starts aviation toward the destination in this mode. After an aircraft arrives the destination and then it change the direction to the original point. On the other hand, an aircraft moves with arbitrary speed and direction during the predetermined time in the random walk model. After the duration is over, new direction is selected and an aircraft change its moving direction according to new one. The major different feature with typical random walk is that moving speed is not changed but fixed in this model. The model is very suitable dynamic situation where there is fighting.

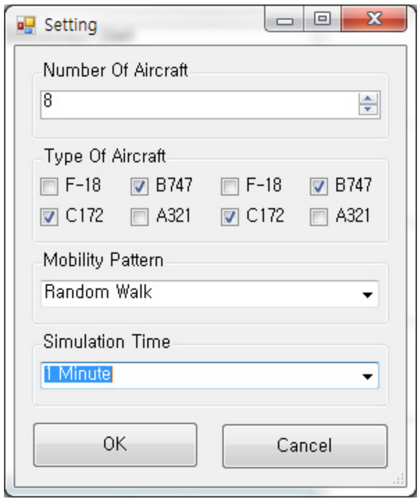


Figure 3. User interface for scenario generator

### C. *Algorithm for Movement Trace*

The algorithm for automated movement is shown below. The main part of algorithm is defined in line 4 as FUNC. This function is provided by commercial flight simulator to model realistic movement of each aircraft.

---

$A_i$ : $i^{th}$ aircraft ( 0 <= $i$ <= number of aircrafts)
$A_i(m)$ : mobility model of $i^{th}$ aircraft
$A_i(k)$ : kind of $i^{th}$ aircraft
$P_i(t)$ : Position of $A_i$ at time $t$
$V(i)$ : Velocity of aircraft $i$
$Distance_i$ ($j$-1,$j$) : Moving distance from time $j$-1 to $j$
$T$ : simulation time

1. for all aircrafts {
2. $P_i(0)$ = initial position of $A_i$
3. while (current_time < $T$) {
4. $Distance_i(j\text{-}1,j)$ = $\mathbf{FUNC}_i(V(i), A_i(k), A_i(m))$
   where $\mathbf{FUNC}_i$ is defined by dynamic of each aircraft in Flight Simulator
5. $P_i(j) = P_i(j\text{-}1) + Distance_i(j\text{-}1,j)$
6. converts $P_i(j)$ to ns-2 format
7. j++
8. }
9. }

---

- STEP 1: for all aircrafts, set initial position.
- STEP 2: Compute the next position with realistic dynamics of aircrafts until whole simulation time is over.
- STEP 3: Generated position information is converted to NS-2 format.

## III. EXPERIMENTS

We implemented the core technologies for mobility and scenario generator. In Figure 4, the initial position of 10 aircraft is shown when each aircraft locates in uniform way. In addition, Figure 5 shows the movement of aircrafts when movement pattern is modeled as random walk. The colored link represented the consecutive positions that aircraft sequentially visited. Moreover, experiment for mission based model is illustrated in Figure 6. Different from case in Figure 5, we can see the respective destination that an aircraft is supposed to visit the place. Each aircraft move toward the destination based on its dynamic model.

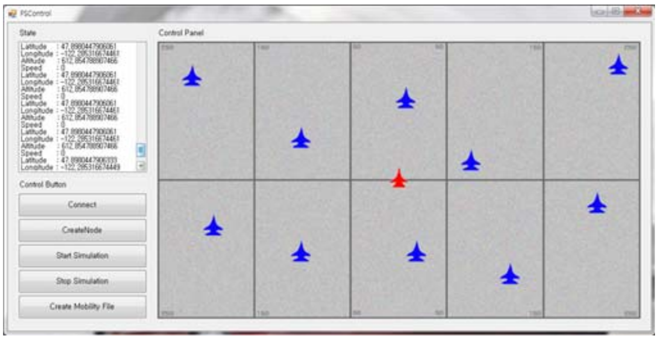

Figure 4. Locates aircraft at the initial point

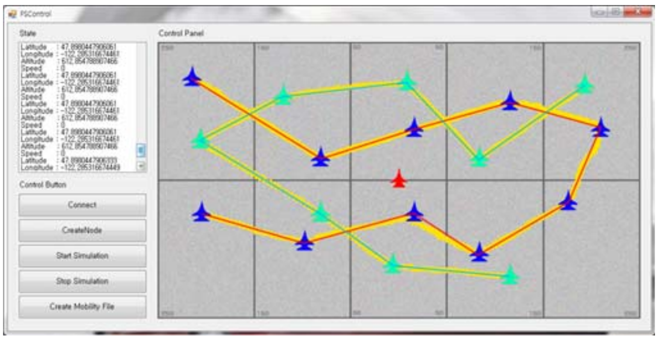

Figure 5. Movement display in random walk model

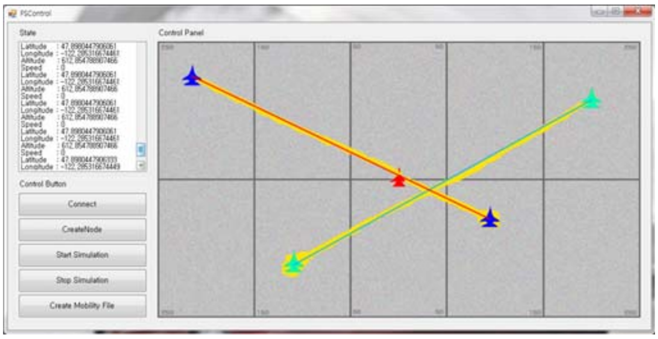

Figure 6. Movement display in mission based model

## IV. COMPARISON WITH PREVIOS WORK

In this section, the proposed scheme is compared with previous work [3], based on mathematical equation. The position is marked in the realistic map as you can see in Figure 7 and Figure 8. First, aircraft in Figure 7 shows the mathematical movement. Since the mobility model is not affected by external environment as well as dynamic of aircraft, movement is illustrated as straight line. Also, direction is changed rapidly at the waypoint. On the other hand, movement for Boeing 747 aircraft shows different pattern with the former one. When corresponding aircraft requires wide range of turning direction where external environment is not always the same, different movement trace around the waypoints are observed. Another point worthwhile mentioning is that the actual movement cannot guarantee to pass waypoints exactly even though the aircraft approaches them nearly. Meanwhile, movement that is managed by the equation, all points are shown to covered exactly by the trace.

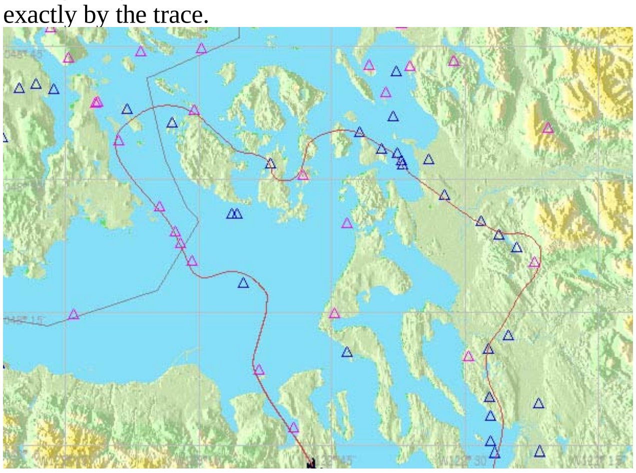

Figure 7. Movement trace by the proposed scheme

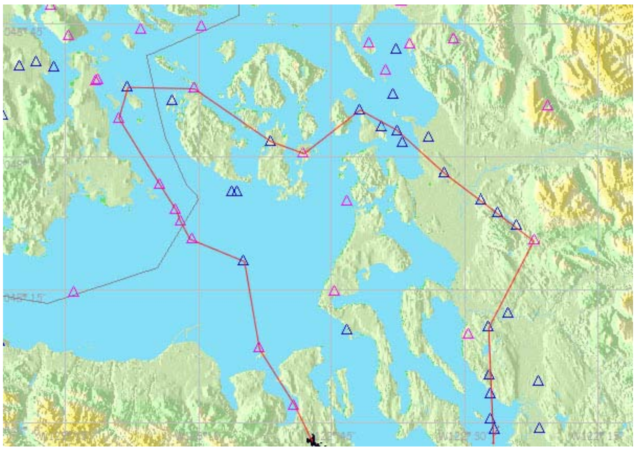

Figure 8. Movement trace by [3]

## V. CONCLUSION

In this paper, we proposed how to implement automated scenarios and mobility generator for aircraft ad hoc networks. Based on software architecture in our previous works, some functions are additionally implemented while considering implementation issues such as mobility pattern. Finally, results for experiment of implementation are shown to prove the correctness of the implementation.

For the further works, we will provide more functions for initial positioning such as random distribution. Also, other mobility patterns such as group mobility will be concerned for implementation. Furthermore, performance evaluation of the routing protocol with generated mobility trace file will be accomplished and analyzed.

## ACKNOWLEDGMENT

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# Transition From IPv4 To IPv6: A Method for Large Enterprise Networks

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***Abstract* - This study analyzes experiences of several large enterprises that had deployed IPv6 addresses. Key factors on the success and failure of IPv6 addresses deployment were synthesized from findings from those enterprises. This research utilizes qualitative method, both inductive and deductive reasoning along with design science approach. Seven guidelines of design science method are followed strictly for better end results. Open-ended interviews will be data collection methods of the study. Documents, such as articles, books, and websites also provide lots of information. Content analysis helps the authors to look directly into context of documents to find the core meaning. The content of this study was examined on two scales: technical side and managerial side. Findings upon data collected reveal several significant factors, which affect IPv6 addresses implementation project. Hence, a solution, the most applicable transition method was concluded. This method was then tested on a virtual environment simulating a large network model. It was proven to be working.**

***Keywords-Internet protocol; IP, IPv4; transition; transition method; IPv6; large enterprise; network; IPv6 readiness.***

## I. INTRODUCTION

On 8 June 2011, over 1000 top websites in the world took part in an event called “World IPv6 Day”. As IPv4 addresses are running out, the need for changing to IP next generation, IPv6, is obvious. This study aims at finding the best method of transition from IPv4 addresses to IPv6 addresses for large enterprise networks.

Since the birth of Internet in 1960s [5], it has completely changed the way of communications forever. With its capabilities, the Internet has already become a world-wide broadcasting capability, a mechanism for information dissemination, and a medium for collaboration and interaction between individuals and their computers regardless of geographic location [20]. However, despite the uncountable Internet's phenomenal impact on business and its reach across all sectors, there is still no model which valuing companies' Internet efforts correctly [1]. Besides, according to its nature in the structure of Internet, the TCP/IP has also played an important role in the global expansion of communications. As a result, the more users join the Internet, the better it would be to spread knowledge in every field around the world. However, this is also the problem as the IP address is not unlimited and the Internet community is witnessing the exhaust of IPv4 addresses not year by year but day by day, which calls for a proper solution. The first group of Internet users that would be affected is Internet Service Providers (ISPs), large enterprises, companies, etc. The reason is that they hold the most number of IPv4 for operation and management and before the IPv4 runs out, they will need an appropriate act to handle the exhaustion, and otherwise, the collapse of the worldwide Internet is foreseeable [17].

This study is conducted to answer the question, which is the best method for large enterprise networks to transit from IPv4 addresses to IPv6 addresses? Currently, there have been many papers, documents, or reports about IPv4 addresses exhaustion; the invention of IPv6 addresses and the way administrators can apply IPv6 addresses to existing networks, known as transition. However, there are still very few documents for applying the transition from IPv4 addresses to IPv6 addresses in large enterprise networks with many different geographical branches around the world. Therefore, with this paper, we would like to give our suggestion on a solution for a complete implementation of IPv6 addresses into large enterprise networks with no influence on its current operation. Furthermore, this paper does not only focus on the technical aspects, but also the management side. It would provide an insight into the importance of IPv6 address transition, as well as a careful analysis on its influence to the enterprise network and its operation. For all the information above, this research could be used as a source of reference for network administrators, board of directors, information executives, or students and network researchers who have an interest in the network communication and would like to join the community of IPv6.

Therefore, we will have two main parts: the theoretical and the practical. For the first part, we would propose the research question, our approaches with the qualitative method, and also the data collection. In addition, we would give an introduction about computer network, Internet protocol, especially all the main features of IPv4 addresses and IPv6 addresses to indicate the differences between them. For the second part, we would like to apply the Design-Science method to analyze the current network infrastructure, IPv6 readiness in large enterprises to acknowledge the reasons and willingness for changing to IPv6 addresses. Moreover, this method is also deployed in the IPv6 address implementation for its effectiveness and risks.

## II. RESEARCH METHOD

This section provides the research question and research methodology of our study. Research approach will be presented in detail so that readers will comprehend our research model.

*A. Question and Objectives*

The most important and also initial step of a research is to define the research question. Based on the nature of that question, proper methods will be applied to find the expected answers. The research question of this paper is: "*Transition from IPv4 to IPv6: What is the best method for large enterprise networks?*"

These following actions are taken to find the answers:

- Conducting a thorough literature review
- Interviewing some specific large companies which have deployed IPv6 addresses
- Analyzing their experiences and attitudes towards IPv6 address deployment
- Analyzing and comparing some transition methods to find the best one (inductive)
- Building a network model and testing the method
- Concluding the result from data and theory (deductive)

The results from above actions are main objectives of this paper, which include:

- Acquiring thorough understanding about IP as well as definitions, ideas, and arguments IPv6 addresses transition methods.
- Getting better understanding of IPv6 addresses deployment in real life project and experiences from companies who had deployed IPv6 addresses.
- Proposing the best method for transitions from IPv6 addresses to IPv4 addresses for large enterprise networks.

The type of this study's research question is "solution" which means to find a way to solve a problem. Therefore, the purpose of this paper is to define and test the most applicable method for large enterprise networks to transit their current IPv4 network to IPv6.

*B. Research approach and Strategy: Design Science*

Design science, as the other side of the IS research cycle, creates and evaluates IT artifacts intended to solve identified organizational problems. Such artifacts are represented in a structured form that may vary from software, formal logic and rigorous mathematics to informal natural language descriptions [10]. Those artifacts are broadly defined as constructs, models, methods, and instantiations to meet with the business strategy, information technology strategy, organizational infrastructure and information system infrastructure, which is presented in Figure 1.

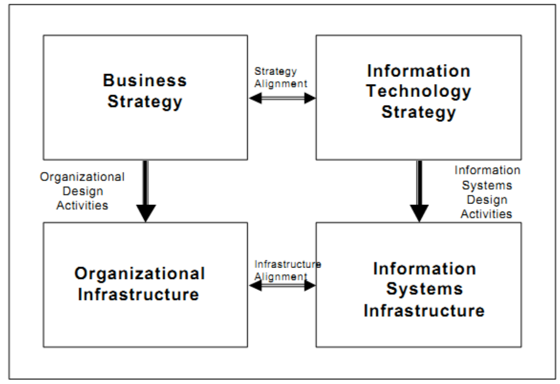


Figure 1. Organizational design and information systems design activities [14]

Therefore, the reason for using design science method is that it is a problem solving process. The fundamental principle of design-science research combines seven guidelines whose knowledge and understanding of a design problem and its solution are acquired in the building and application of an artifact. That is, design-science research requires the creation of an innovative, purposeful artifact (Guideline 1) for a specified problem domain (Guideline 2). Because the artifact is "purposeful," it must yield utility for the specified problem. Hence, thorough evaluation of the artifact is crucial (Guideline 3). Novelty is similarly crucial since the artifact must be "innovative," solving a heretofore-unsolved problem or solving a known problem in a more effective or efficient manner (Guideline 4). In this way, design-science research is differentiated from the practice of design. The artifact itself must be rigorously defined, formally represented, coherent, and internally consistent (Guideline 5). The process by which it is created, and often the artifact itself, incorporates or enables a search process whereby a problem space is constructed and a mechanism posed or enacted to find an effective solution (Guideline 6). Finally, the results of the design-science research must be communicated effectively (Guideline 7) both to a technical audience (researchers who will extend them and practitioners who will implement them) and to a managerial audience (researchers who will study them in context and practitioners who will decide if they should be implemented within their organizations) [10].

*C. Research Method*

Quantitative methods are often used to process random sampling data into numbers and statistics [21]. Quantitative research concerns with testing hypotheses, considers cause and effect, and calculates the size of a phenomenon of interest [19]. The end-results are usually statistical report including both descriptive and inferential statistics. Descriptive method summarizes and presents data in an informative way while inferential method generalizes about a population based on a sample. As such nature of quantitative method, data collection often includes closed-ended questionnaire, surveys that classify various experiences into categories, recording numerical data through observing events etc… [42].

On the other hand, the purpose of qualitative method is to understand and interpret processes underneath an observed event and evaluate people's perception involved in the event [18]. It concerns people, objects, words, images not numbers and statistics. In qualitative research, personal feelings and experiences are analyzed. Qualitative research is often used to construct a new theory from the data collected. For that reason, qualitative data collection methods are interviews with open-ended questions, observation, and document review [42]. This paper aims at studying current network conditions of some large enterprises as well as their attitudes toward the transition from IPv4 addresses to IPv6 addresses. Thus, qualitative research method is applied to this paper. As observation was unable to be carried out, interviews and document reviews were done as data collection method in this paper.

### D. Scope and Limitation

The scope of this research mainly discusses the most applicable transition method from IPv4 addresses to IPv6 addresses for enterprises with large network. The presentation of the method includes literature review, advantages, and configurations as well as simplified models of the method. As this research aims at large networks, large enterprises with big network traffic may find it more useful than small and medium sized network. There are some transition procedures that may not be suitable for small and medium sized networks due to their complexity. Therefore, this paper is most applicable and limited to large networks.

### E. Validity and Reliability

Presently, there are various definitions of validity and reliability in qualitative research method from perspectives of many different researchers. In this paper, the understanding of validity and reliability will be considered and measured by the idea of trustworthiness to establish confidence in the findings. Moreover, Johnson [19] stated that reliability and validity can also be understood as "defensible" [11]. Multiple perspectives from various sources should be compared and tested before the conclusion to strengthen the results and enhance "trustworthiness" [11].

This study relies on a variety of sources, which are from technical papers of leading telecommunication companies. Conclusion is drawn in reference to those data. All the chosen enterprises had carried out IPv6 address deployment with large networks which fall into class B to class A based on IP classes. All the interviewees are people who were in charge of or involved in IPv6 addresses deployment in their companies. All data sources are listed in reference and can be verified. Data collected will be analyzed by proper methods in the right procedures so that the study remains stability, reproducibility and accuracy. It means that data can be analyzed and classified in the same way over a period of time [29].

## III. EVALUATION OF CURRENT TRANSITION METHODS

The transition from IPv4 addresses to IPv6 addresses is not a one-day step and involves a lot of changes in network structures with the use of IP addresses. For the future success of IPv6 addresses, the next step in deploying IPv6 addresses is to vote for the most suitable transition methods and their management. Although many kinds of transition mechanisms have been invented to help with the process such as NAT64, Stateless IP/ICMP translation, NAT-PT, 6over4, and Teredo, the implementation of IPv6 addresses is never said to be easy and simple, even for experienced administrators. As a result, the most difficult problem to make decisions for is which method will be chosen for the implementation process to achieve a smooth and seamless [31].

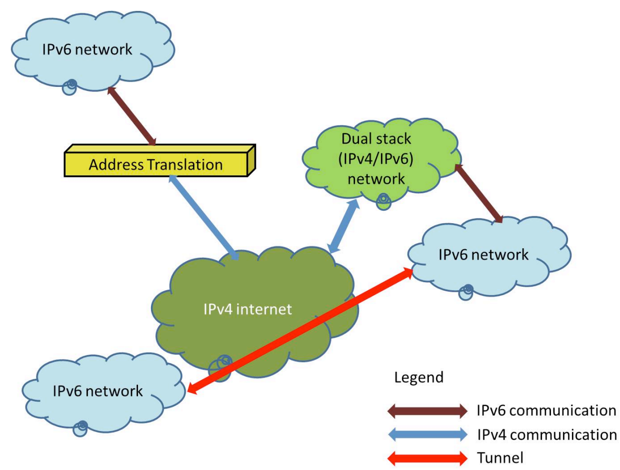


Figure 2. Different transition technologies [38]

According to Figure 2, there are different kinds of technologies which can be applied such as dual stack, tunneling mechanisms, and translation techniques. Firstly, dual stack means to support IPv4 addresses and IPv6 addresses at the same time on all network nodes. Secondly, translation means to convert directly IPv4 addresses to IPv6 addresses or vice versa. Finally, tunneling means to create a tunnel for IPv6 addresses native networks across IPv4 network. Over sixteen transition techniques have been used and tested for the communications between different networks to ensure IPv4 addresses and IPv6 addresses interoperability. Therefore, to make decision on the best suited transition methods, it is really important to have an overview of the current IPv4 networks. In addition, enterprises must analyze needed functionalities, scalability, and securities in the corporation. Besides, "one size does not fit all" and a network can be applied different transition mechanisms together to support a complete distributed system.

In this section, based on the information from the research and literature review, we would present an overview of some major transition methods as well as relevant matter to opt out the best methods for large enterprise networks. Each technique possesses individual attributes and plays an important part in the transition process. In general, we can classify various transition techniques into three categories with respect to connectivity and necessary elements for the implementation.

The first category is Dual Stack which uses two IPv4 and IPv6 stacks for operating simultaneously, which enables devices to run on either protocol, according to available services, network availability, and administrative policies. This can be achieved in both end systems and network devices. As a result, IPv4 enabled programs use IPv4 stack and this goes the same for IPv6. The IP header version field would play an important role in receiving and sending packets. In other words, this kind of IPv6 transition is the encapsulation of IPv6 within IPv4.

Next, is the translation category, which means to convert directly protocols from IPv4 to IPv6 or vice versa, which might result in transforming those two protocol headers and payload. This mechanism can be established at layers in protocol stack, consisting of network, transport, and application layers. The fundamental part of translation mechanism in transition process is the conversion of IP and ICMP packets.

Finally, the last category is tunneling which means to transfer data between compatible networking nodes over incompatible networks. It utilizes a protocol whose function is to encapsulate the payload between two nodes or end systems. This encapsulation is carried out at the tunnel entrance and the payload will be de-capsulated at the tunnel exit.

### A. *Summary of three methods*

Table I below is the summary table containing the advantages and disadvantages for three main methods

TABLE I. SUMMARY OF THREES METHODS

| | Advantages | Disadvantages |
|---|---|---|
| **Tunneling** | - Configure tunnel endpoints only<br>- Simple deployment<br>- No additional management | - Face another problem of NATs<br>- Take more time and CPU power<br>- Harder to troubleshooting and network management<br>- Have single points of failure<br>- Vulnerable to security attacks |
| **Translation** | - The router is used as a translation communicator<br>- Solve network interoperability problems | - Limitations similar to IPv4 NAT<br>- Reduction in the overall value and utility of the network.<br>- Harder to control on a larger scale<br>- Complexity increases in IP addresses |
| **Dual Stack** | - Easy to implement<br>- Low cost<br>- Greatest flexibility<br>- Already supported in all OSs and devices | - Two routing tables<br>- Additional memory and CPU power<br>- Two firewall sets of policies |

### B. *Conclusion*

Based on the above overview of all mechanisms and current practices in researched enterprise networks, nearly all deployments of IPv6 addresses in enterprise networks apply dual stack mechanism as it gives us a way to know more about IPv6 addresses as well as to improve practical experience with a new address family, which plays an important role in the success of transition implementation. Therefore, in this paper, we choose the dual stack mechanism to build a simulated model for large enterprise networks.

### C. *Simulated Enterprise Network Design*

To support the transition process, we will use the Cisco Packet Tracer program [48], which is a powerful network simulation software from Cisco, to create a visual model of an enterprise network. Our model has three main areas. At first, the headquarter network area consists of four groups which are the Demilitarized Zone, the Intrusion Prevention, the Service Provision, and the Client Group. Secondly, a branch network area is a division of the business that can be located in various geographic areas. Therefore, the network model of each branch is similar to the headquarter model only without the DMZ. Thirdly, the network area of ISP routers with VPN users who perform the work outside the enterprise network still needs to get access to data from protected servers inside the network.

After successfully building the simulated network model, we implemented the chosen Dual Stack method [47] with specific technologies such as Dynamic Host Configuration Protocol (DHCP) [49] to assign IP addresses automatically for end devices, Open Shortest Path First (OSPF) [50] as an IP routing protocol, Border Gateway Protocol (BGP) [52] for the simulation of Internet among ISPs, Virtual Private Network (VPN) [51] for simulating the group of mobile users, and finally the security establishment, as illustrated in Figure 3.

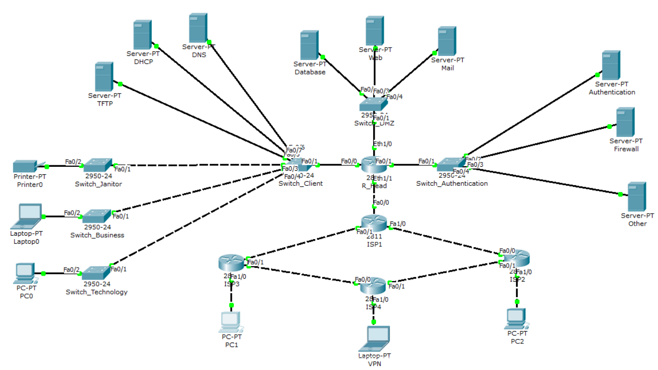


Figure 3. Simulated Enterprise Network Design

Based on practical results from the simulated network model, we have an opportunity to put our chosen method to test and prove that it is the best choice for large enterprise networks to switch from IPv4 to IPv6.

## IV. RESEARCH DATA

This research collected data by interviews with several specific organizations and documents review.

Standardized, opened-ended interviews were conducted with people in charge of IPv6 addresses deployment or network maintenance to find out their experiences on this topic. The questions are made so that answers are open-ended; which means participants can fully express their points of views [41]. The same questionnaire was provided to the interviewees to make the process of analyzing and comparing more easily. Before the interviews were conducted, emails concerning the issues were sent to all appropriate organizations asking for permissions. Those organizations were several large Vietnamese enterprises and Finnish organizations listed in the list of IPv6 addresses deployment status of Finland (derived from www.vyncke.org/ipv6status/detailed.php?country=fi). The authors contacted sixty companies and universities; however, only five of them answered back or gave permission for the interview. It means that interview answer percentage was only 8.3 %.

Overall, the most common reasons for deploying IPv6 addresses were testing, preparing for the future, and gathering user experiences. The organizations that the authors had the permission to conduct the interview are university, Internetworking consulting company, training center, and Telecom Corporation. At the moment, those organizations are running well with IPv4 addresses, they have not hit the wall of IPv4 addresses shortage yet. Since all the organizations mentioned above are related to networking sector, training, and service providers, they were aware of the need for IPv6 addresses preparation. IPv4 addresses exhaustion is obvious and it causes future addressing problems. Our interviews also revealed IPv4 addresses exhaustion as the most common reason stated. As we know that IPv4 addresses free pool is out, new Internet connections are rising fast in developing countries. Once those remaining IPv4 addresses are out, new connections will be provided with IPv6 addresses. If enterprises do not prepare for such parallel existence of IPv4 addresses and IPv6 addresses, they may lose a number of customers who cannot access them on Internet.

As inferred from the interview with the Internetworking Consulting Company, the most important aspect to an enterprise is cost saving. Technical features above may not sound very interesting to them. On cost-saving scale, the transition to IPv6 addresses may offer many benefits for large enterprises. It promises to bring better routing performance, improve security and auto-configuration, which generate lower implementation cost, and daily maintenance basic cost. Therefore, it may save long-term IT cost for enterprises.

The interviews revealed that the most commonly used transition method is Dual-Stack. It is because networking devices nowadays already support IPv4 addresses and IPv6 addresses at the same time. Transition technologies are also important factors of IPv6 addresses transition process. There are several common techniques that will be discussed further in this paper.

Moreover, IPv6 is a certain future that it may bring opportunities to early service provider. Since many enterprises are new to IPv6 addresses, service provider can be a consultant in decision making while making some profit. Chip Popoviciu [12] emphasized the value that early adoption can get. Service providers can sell various services if they can play a role in helping enterprises deploy IPv6 addresses.

On the other hand, the research has found out that the most common issue in an IPv6 addresses deployment project was user training. Firstly, training cost was not considered properly in the project planning phase. Then, IP maintenance staffs were lack of IPv6 addresses experiences and knowledge. It not only raised the cost but also lowered the efficiency of the project.

Lastly, there is no backwards compatibility in transition to IPv6 addresses. Microsoft has no intention to implement IPv6 addresses for older Window OS such as Windows 98 or Windows Millennium Edition or Windows 2000 [23]. Therefore, there are some concerns with the cost of replacing a variety of application, which needs to be IPv6 addresses compatible. However, most of the networking software support IPv6 addresses comes as an upgrades based on old software version, thus there is not much extra cost, even no. Nevertheless, specific software tailored for company is much more different. If an enterprise wishes to transit all software to IPv6 addresses compatible and eliminate IPv4 addresses, it may come up with huge costs. It also costs a lot of time and efforts of applications developer to change the applications according to information from the interview with employee from Telecom Corporation.

A request for Comments (RFCs) is a document system invented by Steve Crocken in 1969 to keep the record as well as improve technology being used on the ARPAnet. An RFC describes a research or applications on networking technology or define a new one. TCP/IP was developed by the RFC method of development. In this research, we consulted RFCs as our data on technical issues. RFCs are the most updated and trustable papers on networking technology. A set of RFC, which will be used in this paper, consists of: RFC 2373, RFC2647, RFC2765, RFC2766, RFC2767, RFC3022, RFC4057, RFC4861, RFC5735, RFC6343, and RFC791. Besides RFCs, a set of journal papers on networking technology is also used as technical data in this study.

## V. RECOMMENDATION OF IPV6 TRANSITTION FOR ENTERPRISES

"*Readiness is a state of preparedness of persons, systems, or organizations to meet a situation and carry out a planned sequence of actions. Readiness is based on thoroughness of the planning, adequacy and training of the personnel, and supply and reserve of support services or systems*" (BusinessDictionary).

In expression of IPv6 addresses, this means being ready for the implementation of IPv6 addresses into a network when business requirements arise.

The University of Helsinki, Axu TM Oy, and FPT Telecom Corporation have already taken initial steps to implement IPv6 addresses into the network system based answers gathered from the research. Although these efforts are now just meant for testing, the message from the international community is clear. The transition from IPv4 addresses to IPv6 addresses will become a must for enterprises, especially ones that currently provide online services based on IP addresses. They must be able to handle large requests from internal and external customers who are applying their emails or web or other services to everyday working lives over the Internet. It is impossible to predict exactly when IPv6 addresses will become mandatory for most companies. However, in accordance with Mr. SeppoSyrjanen, Data Network Specialist from IT Center of University of Helsinki, although IPv6 addresses adoption is a need for current companies to prepare for the future, aside from the exhaustion of IPv4 addresses for stakeholders to take into consideration, it is not an easy and single step that can be achieved in a short time but this requires a great amount of thorough planning and preparation to develop and adjust an IPv6 business case. Therefore, it is not when IPv4 addresses come to the point of complete exhaustion, IPv6 addresses have already been considered as purely strategic because this is not only to establish global connectivity of enterprise networks for the future, but also to guarantee growth as well.

In this part, based on information from the interviews with large companies and enterprises, we would like to categorize different preparation activities that can be applied as a plan in this IPv6 addresses implementation. This part would become a great asset to assure that a common method is ready to make plans and to check IPv6 readiness when it falls into place for each enterprise network. It is an outline of phases which have both technical infrastructure and business readiness taken into account for an enterprise to initiate the transition to IPv6 addresses.

### *A. Business Side*

**Phase 1**: The determination of business grounds and demands to implement IPv6 addresses

An enterprise must have strong and reasonable desires to initiate the IPv6 addresses transition project. They need to realize business requirements, motives and mark the features of IPv6 addresses to those particular objectives. Five main conditions that need to be acutely considered are business operations after the depletion of IPv4 addresses, support for a great amount of network devices, enterprise policies for IPv6 addresses transition, requests from customer, partners, suppliers, and the global-scaled trade, said by Nguyen Dac Thuan, from FPT Telecom.

**Phase 2**: The analysis of profits, expenses, and risks

Enterprises need to assess the impacts of IPv6 addresses transition and which kinds of benefits it brings to the business. Specifically, they need to perform thorough analysis to decide which certain line of business or programs can be benefited from IPv6 addresses transition. In addition, Aleksi Suhonen, from Axu TM Oy, has said that there were other relevant subjects that also need to be taken into serious consideration such as enhancement of new services as well as the maintenance of existing services, development of network efficiency and cost savings (the elimination of NAT or other work-around methods), the high performance of large enterprise network, simple configuration of online operations, and the supply of tactical advantages.

Phi Long, from Nhatnghe Network Traning Center, has said that costs estimation is the most important part and it can decide the progress of the implementation based on his own experiences from many concerned network projects. Therefore, once enterprises would like to initiate the IPv6 addresses transition, they also need to be prepared for the budget that can be used for planning, design (infrastructure upgrades if needed), implementation testing, deployment, personnel training as well as operational costs.

In IPv6 addresses transition, risk includes business, legal, privacy, security, reliability, interoperability and technical risks. Only when we can identify risks and its impact, will we be able to apply action plans to prevent or reduce the influence on the whole project. These plans should put emphasis on major program activities, specific solutions, and impacts.

**Phase 3**: The settlement of a supervised group (SG) for administration of IPv6 addresses transition project. Nguyen Dac Thuan, from FPT Telecom, indicated that the supervised group would temporarily act as a centralized management office (CMO) to make plans, administer, and control the progress of IPv6 addresses transition throughout the entire enterprise. Furthermore, the SG will arrange sufficient resources such as staffing, training, and budget to support the IPv6 project successfully. Specifically, the SG will be responsible for recruiting suitable members to the group for different roles and responsibilities; gaining authority rights within the enterprise to support financial matters for the transition project and set policies to become the priority in case of shortage of resources; organizing an administration structure to guarantee the success implementation of IPv6 addresses transition. The SG will be the leader to set the milestones and targets for the working team and control the progress through successful results.

### *B. Technical Side*

***Phase 4*****:** The assessment of all assets of current network infrastructure

As shown by by Head of IT center, from the University of Helsinki, we suggest that before starting to implement the IPv6 transition project, the enterprises need to carry out a complete analysis of current networks to get an overview of components that may need to be changed or upgraded to be suitable for transition to IPv6 such as address allocation, networks services (IP, wireless, VoIP, DNS, DHCP, NTP…), network management, applications, operational systems and support.

***Phase 5*****:** The establishment of architecture for IPv6 project

When implementing the transition from IPv4 addresses to IPv6 addresses, there must be an overall IPv6 architecture for various impacted areas. It should be standard based and support IPv4 to perform a smooth transition. Moreover, this architecture should also expect new networks and services as well as foreseeable traffic growth after the implementation. There are some concerned major areas such as IPv6 addressing plan, IPv6 routing, IPv6 interconnection, IPv6 foreseeable traffic, IPv6 enabled systems, IPv6 deployment plan, transition mechanism (dual stack, tunneling, and translation), network services, security, management, scalability & reliability, and service level agreements [32].

***Phase 6*****:** The outline of a specific structure on the influence of IPv6 project

The IPv6 project, once established, will place influence on every platform and service in the network. As a result, IPv6 capability and its influence will be decided according to enterprises' standard for each platform and service, which consists of commercial and industry standards. This includes the required resources (devices, personnel, budget, etc.) and the communication between system integrator and vendor, said by Nguyen Dac Thuan, from FPT Telecom.

***Phase 7*****:** The development of an IPv6 project plan

In this phase, the SG is required to gather all information and resources to design a final plan for IPv6 addresses transition in the enterprise network. Because of its importance, this plan is required to contain a schedule of small projects to be implemented along with dependencies and priority. Furthermore, in accordance with Phi Long, from Nhatnghe Network Training Center, there should be a testing environment for members to gain experience with new IPv6 addresses features and also to demonstrate the architecture, plans, and policies... One more thing is that the SG should perform trials on the real enterprise network as well as operational processes to ensure that all devices and services acquired or developed are IPv6 capable.

***Phase 8*****:** The provision of a personnel-training program

This IPv6 addresses transition project involves either business or technical aspects and this also means the attendance of many users from the board of directors to ordinary staff to maintain IPv6 readiness. As a result, training is required to update knowledge and skills for users to familiarize with the new system [36]. However, based on the position of users in the enterprise, there will be many types of training programs to be suitable for all. Based on the information of Nguyen Dac Thuan, from FPT Telecom, we divide the training into four categories:

*General training program* aims to give normal users primary information about IPv6 addresses and its related issues.

*Engineer training program* is to give detailed information about IPv6 addresses technologies and this is suitable for staff members who are responsible for analyzing, planning, designing, testing and deploying IPv6 addresses.

*Operational training program* presents specific IPv6 education to employees who take care of the support for an IPv6 network.

*Special training program* includes advanced information in certain technology are, which is suitable for technical specialists or experts in a certain technology area such as security, mobile, etc.

### *C. Stages of Readiness*

In this part, based on the information from literature review and interviews, we have combined that information to create a checking tool for enterprises to assess IPv6 readiness level in the network. Based on the result from this tool, the board of directors can have an overview of the current network and make decisions or plans according to the result. The stages of IPv6 readiness can be arranged into six ranks, which represent the work to be achieved before implementing IPv6:

***Rank 1*****:** The enterprise has no intention to implement IPv6.

At this stage, enterprises have no business requirements and decide not to integrate IPv6 into the system as they analyze that the expenditure for IPv4 addresses shortage is lower than the effort and budget spent for transition to IPv6 addresses while IPv4 addresses exhaustion will not place influence on their business.

***Rank 2*****:** The enterprise has taken IPv6 into consideration but is still unprepared to initiate it.

At this stage, enterprises may hire IT experts to advice on the IPv6 project or methods to prevent IPv4 addresses exhaustion. Moreover, there may be discussions within the executive group (CIO, CEO, CTO…) to collect information in relation with IPv6 project such as business and technical requirements as well as cost and risk for transiting to IPv6.

***Rank 3*****:** The enterprise has an IPv6 program in place and is determining important issues.

At this stage, enterprises may establish a business case and a budget for the IPv6 migration. A supervised group is also formed to control and manage the progress of IPv6 implementation. The members and roles of the IPv6 Transition Group should be identified. Furthermore, a thorough analysis of current network infrastructure should be done to check the IPv6 capabilities.

***Rank 4:*** The enterprise possesses an IPv6 project associated by a final plan.

At this stage, enterprises may already have a sponsored IPv6 project that includes a detailed report of current infrastructure and a tested architecture design of IPv6 implementation.

***Rank 5*****:** The enterprise is in possession of an IPv6 project without any unresolved crucial issues.

At this stage, enterprises, supported by all detailed documents such as an IPv6 deployment plan, training plan, architecture design, may actively put into practice those plans and design to perform the first testing on the real networks.

***Rank 6:*** The enterprise has successfully accomplished the IPv6 addresses transition project.

At this stage, enterprises have deployed IPv6 addresses into the system and finished the testing part. Furthermore, the training programs are also provided for every user. The system is ready to communicate with other IPv6 networks from customers, partners, and suppliers.

Table II below describes the phases that are suitable for each rank.

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TABLE II. RANK DESCRIPTION

| Phase | Description | Rank | | | | | |
|---|---|---|---|---|---|---|---|
| | | 1 | 2 | 3 | 4 | 5 | 6 |
| 1 | The determination of business grounds and demands to implement IPv6 addresses | | X | X | X | X | X |
| 2 | The analysis of profits, expenses, and risks | | X | X | X | X | X |
| 3 | The settlement of a supervised group (SG) for administration of IPv6 addresses transition project | | | X | X | X | X |
| 4 | The assessment of all assets of current network infrastructure | | | | X | X | X |
| 5 | The establishment of an architecture for IPv6 project | | | | X | X | X |
| 6 | The outline of a specific structure on the influence of IPv6 project | | | | | X | X |
| 7 | The development of an IPv6 project plan | | | | | | X |
| 8 | The provision of a personnel training program | | | | | | X |

In general, it is necessary for enterprises to thoroughly analyze and implement an IPv6 addresses transition with clear instructions to serve expectations. However, because of the specific expectations may change from time to time, and they can be different by various enterprises, a complete approach with careful planning and preparation as listed in this part, accompanied by the details for each phase will allow the IPv6 addresses implementation project to be achieved successfully, which will open a new path for each enterprise to be ready for the next generation of communication networks.

## VI. CONCLUSION AND FUTURE WORK

### A. Research result

There are three transition methods that were most applied i.e. dual stack, translation and tunneling. Each of them has its own advantages and disadvantages.

The second objective was to analyze real life experiences of enterprises that had deployed IPv6 addresses. We learned that the reasons for starting IPv6 could be:

- Preparing for IPv4 addresses shortage coming in near future
- Testing the transition process
- Better features
- Getting support from top executives on the project

On the other hand, some enterprises were not interested in IPv6 addresses transition for the following reasons:

- Business is still going on well
- Training costs
- No instant advantages
- No solution from service providers
- No backward compatibilities

For those above reasons, dual stack seems to be the best method

### B. Recommendations

*Understand the situation.* Most companies are running well with IPv4 addresses. They won't have troubles until several years. What will come then? Developing countries with increasing numbers of new computers and devices connecting to the Internet will need their IPs, which will be IPv6 addresses. Being slow to adopt new technology will lead to losing access to these potential customers. Besides, IPv6 addresses have new features that promise to bring better management and administration as well as improve security. Moreover, service providers who can offer enterprises services in transition obviously can make profit out of it.

*Be prepared.* Proper budget must be considered in advance including planning, design, testing, deployment, personnel training and operational costs. IPv6 support products should be integrated into product lifecycle replacement. The reason is that the network will still be able to communicate with IPv6 addresses from the outside world and it makes the transition process much more fluent in case the organization needs to deploy IPv6 addresses later on. Software or a system tailored for specific organization should be considered an IPv6 matter, as it is very difficult to change in the future.

*Pay attention to human factor.* The human factor includes staffs of project teams and the operational administrator. Project team members must be people who really understand internal network structure because they will decide which method of transition to apply. Choosing the right method will avoid many troubles for administration. The operational administrator must be the one who has knowledge of IPv6.

### C. Methodology

The paper followed seven guidelines of design science method. It was an inductive study looking for a solution to a problem. The interviews in this study were conducted with people who are involved in IPv6 addresses deployment projects in large network enterprises in Vietnam and Finland. Additionally, document review was essential for this study since this paper concerned a lot of technical issues and evaluated situations based on existing techniques, which were documented in various published sources. It was important to study documents on technical experiences of previous projects as well as new technology coming.

Content analysis was the right choice for analyzing those documents and interviews' transcripts.

### D. Limitation and Further Study

The authors of this study are Vietnamese students who are studying in Finland. Finland is a developed country in technology. For that reason the authors intended to focus on Finnish enterprises. However, very few companies had answered back. Then, Vietnam was chosen as the author had more relations to companies in Vietnam and hoped that there would be some more companies participate. About 60 organizations were contacted and there were only five enterprises answered. Four of that were large network enterprises. Half of enterprises that answered back had successfully carried out IPv6 project while the other half had failed. That number might not be representative. The author had reference other sources to synthesize a model of enterprise network. Because there was no specific answer of people who administrate the enterprise network, this study could not give more details. It is the biggest the limitation of this study.

This study was limited to large enterprises that had a network size of over 1000 computers. Consequently, the research results may not be true for smaller network.

Finally, there are various areas for further study based on this research. Firstly, a study with a larger sample or more cases could be done for better results. Another topic could be transition method for small and medium network enterprises. Or it is possible to find the critical factors for the failure of IPv6 deployment in general or large network size enterprises in particular (or small and medium sized ones). Additionally, further study on IPv6 for mobile devices can be considered.

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# A Scheme Improving Performance of IEEE 802.11 Multicast Protocol

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***Abstract*—The IEEE 802.11 multicast protocol is proper to send data to multiple recipients. Nowadays, Wi-Fi equipped smart phones are in the access of almost every person which increases amount of data transmissions. In this paper, we discuss the flaws in existing IEEE 802.11 multicast protocol and propose an efficient way to improve its performance. The proposed system has an assisting station (AS) and the stations that want to join a multicast group have radio frequency monitoring (RFMON) modules which help themselves overhear the data. Thus, for multicasting, the access point (AP) transmits unicast data to AS by the fixed Internet protocol (IP) address while stations in multicast group can also receive the data. The proposed system improves efficiency and reliability, while reducing delay for multicast transmissions.**

***Keywords-IEEE 802.11; Multicast; Wireless Network; RFMON mode.***

## I. INTRODUCTION

Unicast transmission is to send a message to a single destination identified by a unique address in the network. Multicast transmission is the delivery of a message or information to a group of destinations simultaneously in a single transmission from a source. In unicast transmission, a sender transmits data frame to a receiver which in turn sends back acknowledgment (ACK) to notify the sender of its successful reception. If the sender does not receive ACK during a fixed time interval, it automatically transmits data again. Thus, the unicast can guarantee reliable data transmission by feedback whereas multicast does not assure reliability as it lacks such ACK process. For this reason, multicast transmission has the lowest data transfer rate in IEEE 802.11[1]. Using multiple unicast transmissions as an alternative for multicasting incurs an overhead problem because of multiple ACKs. In this paper, we propose an efficient method to improve the performance of multicasting by introducing the concept of an assisting station (AS) and radio frequency monitoring (RFMON) module.

The rest of the paper is organized as follows. Section 2 describes background and related work. Section 3 presents the proposed scheme in detail. Section 4 describes performance evaluation and the achieved results. Section 5 concludes the paper.

## II. BACKGROUND AND RELATED WORK

### *A. IEEE 802.11*

IEEE 802.11 is a set of standards for implementing wireless local area network (WLAN). Wi-Fi is a wireless communication protocol specification used to exchange data. The 802.11 medium access control (MAC) layer has two major modes of operation, namely, the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF). The DCF is the most popular mode of MAC operation in 802.11. The DCF uses carrier sense multiple access with collision avoidance (CSMA/CA) based scheme for its operation.

In unicast transmission, after channel assignment, a sender transmits a request to send (RTS) frame to reserve access of channel. The receiver replies with a clear to send (CTS) frame. Other stations avoid collision by listening the RTS/CTS and refrain from sending data for a given time. After receiving data successfully, the receiver sends an ACK to the sender. If the data transmission does not take place successfully, the sender retransmits that data.

In multicast, a single sender transmits data packet to multiple recipients at the same time. The multicast is an efficient way to transmit data packets to multiple stations that need them. However, unlike unicast, there is no feedback process, i.e., sending ACK, and therefore it does not guarantee reliability. On the other hand, assuring maximum reliability to multiple recipients causes multicast data to be transmitted at the lowest data transmission speed.

In [2], BMW (Broadcast Medium Window) method is proposed to solve the problems of multicasting by performing multiple time unicast for multicast transmission. In BMW, each station has three lists, i.e., NEIGHBOR, SEND BUFFER and RECEIVER BUFFER lists. The NEIGHBOR is a list of current neighboring stations, SEND BUFFER stores multicast frames to be forwarded and RECEIVER BUFFER maintains the sequence numbers of the data frames received from neighboring stations. After a sender that has multicast data to send takes channel, it sends RTS and receives CTS as a response. After sending the data frame and receiving ACK, it moves on to the next recipient in the NEIGHBOR list. This process is repeated until entire data are transmitted successfully. However, the main drawback of BMW is that if the sender has $n$ neighbors to send, there are $n$ time contention phases per multicast operation, which increases delay.

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In [3], the Batch Mode Multicast MAC (BMMM) method is proposed to solve the problem of [2]. In BMMM, in order to send a multicast frame, the sender sends RTS to each station individually and waits for CTS from each of them. Upon reception of CTSs from *all* the intended recipients, the sender starts to send the multicast data. Following this, it sends a special frame called Request for ACK (RAK) to each station, and each station responds to the RAK with an ACK. Upon receiving ACKs from all intended recipients, the transmission is said to be done successfully. If one of ACKs is missed, then the sender again contends for the medium and repeats the above procedure. The RAK frame format is shown in Fig. 1.

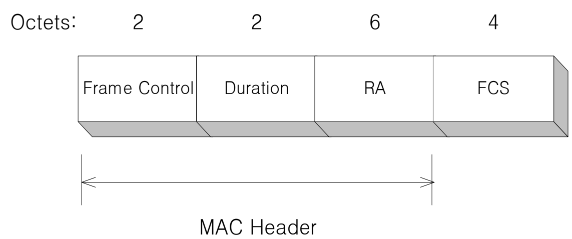


**Figure 1. RAK frame format.**

In BMW, nodes contend for channel assignment to send a control frame to each station. The sender transmits RTS and receives CTS, after which it transmits the data. Upon receiving the data, the receiver sends back ACK. Above procedure is repeated for all intended receivers, which causes overhead. In BMMM, after taking a channel, the sender sends RTS and the receiver which receives the RTS sends back CTS. This RTS/CTS procedure is repeated for all intended receivers. After finishing the RTS/CTS procedure, the sender transmits the data, by multicast, to those stations from which it has received CTS. In this way, BMMM tries to solve the delay and overhead problem of BMW and also improves multicast reliability by sending RAK. However, BMW and BMMM use multiple unicast transmissions and multiple RTS/CTS procedures respectively for multicasting, which increases reliability but incurs delay and overhead.

In other case of studies, they proposed some methods to improve multicast efficiency based on ad-hoc networks [4]-[6].

## III. PROPOSED SCHEME

In this paper, we propose an efficient method to overcome the aforementioned problems of multicasting. The proposed method has an assisting station (AS) and the stations that intend to receive the multicast data and have the RFMON mode.

For multicasting, AP transmits unicast data to AS through the fixed IP address and stations in the multicast group can also receive the data with the help of RFMON mode. Stations with RFMON mode as a way of monitoring WLAN can collect the entire WLAN packets and monitor the existence of network effectively [7]-[9]. It is receive-only mode for data.

### *A. Assisting station (AS)*

The AS receives multicast frames from AP and sends back ACK which ensures reliability. It has the fixed known IP destination address. Usually AP transmits multicast data to multiple recipients. However, in the proposed method, AP sends the multicast data to AS by using the fixed IP address. The stations that want multicast data can also receive that data simultaneously by using the RFMON mode. Because of that, the proposed method can solve the overhead problem. The AS can be located at the edge of AP radio coverage, which can guarantee that every station can receive the data if AS responds to data by ACK.

In the proposed method, even though AS is a special station, AP recognizes it as an ordinary station. Therefore, the proposed method can be used while compatible with IEEE 802.11 without any change in its infrastructure. The AP sends the multicast data to the AS by using a fixed IP address. If the AP assigns IP address to the other stations except the AS, the AP assigns one of the remaining IP addresses except the IP address assigned to the AS.

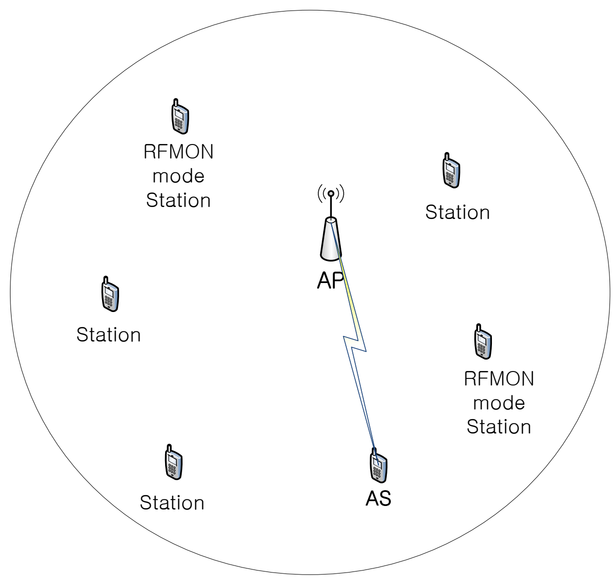


Figure 2. The AS receives multicast data from the AP by using the proposed method.

### *B. Use of RFMON mode*

In the proposed scheme, in order to become a member of multicast group, a station should be equipped with a module or dongle that supports the RFMON mode. The module or dongle is programmed to monitor the packets in the network which AS joins. In this way, the stations equipped with the RFMON module can also receive and process the multicast data. If a station does not want to receive the multicast data, the RFMON mode can be disabled. Thus, owners of the stations can make a decision about data reception by either installing or removing the RFMON mode.

Fig. 2 illustrates multicast transmission using the proposed method. The AP sends the multicast data to one destination, i.e., AS using the fixed IP address, while the multicast group equipped with RFMON module receives the data simultaneously together with AS. Other stations that are not in the multicast group cannot receive the data because they are not equipped with the RFMON module.

## IV. PERFORMANCE EVALUATION

To evaluate the proposed scheme, we conducted simulations in QualNet 5.0, which provides a platform to test protocol design, analysis and verification prior to implementing the system in practice. The complete set of simulation parameters is shown in Table 1.

TABLE I. PARAMETERS FOR SIMULATION

| PARAMETERS FOR SIMULATION | |
|---|---|
| ***Parameters*** | ***Values*** |
| PHY/MAC | IEEE 802.11g |
| Packet Size | 512 bytes |
| Antenna model | Omnidirectional |
| Total number of station | 50 |
| RFMON mode station | 5, 10, 15, 20, 25, 30 |
| Assisting Station | 1 |
| Application data type | CBR |
| Simulation time | 500sec |
| Number of total packets | 100 |

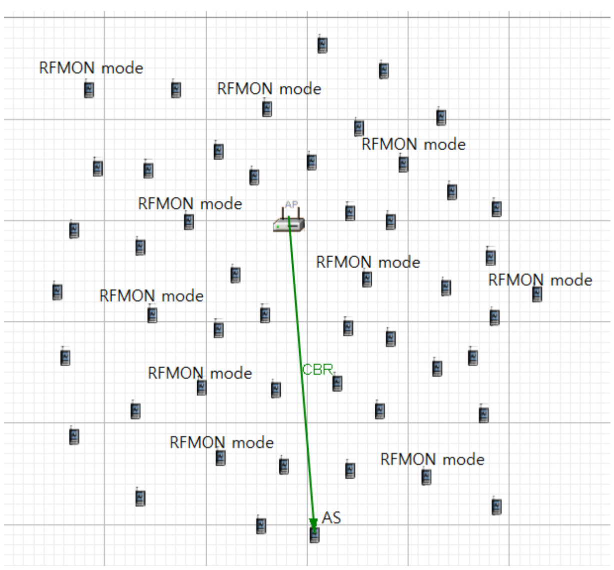


Figure 3. Network model for simulation.

The network model for simulation is shown in Fig. 3. Simulation shows that there are 10 RFMON-mode stations. RFMON-mode stations are randomly chosen and the number of them is increased from 5 to 30. The AS is located around the edge of AP radio coverage.

Fig. 4 shows the average end-to-end delay for the legacy multicast of IEEE 802.11 protocol and the proposed scheme. Evidently, IEEE 802.11 multicast performs poorly in comparison with the proposed scheme.

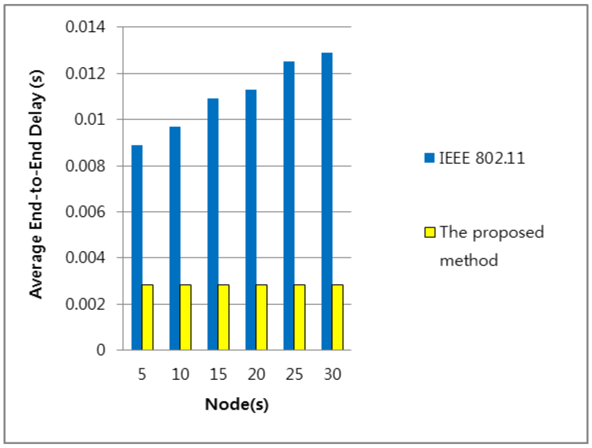


Figure 4. Average end-to-end delay.

## V. CONCLUSION

Delay and overhead are major issues in multicast transmission in IEEE 802.11. This paper proposes an efficient method to overcome these issues. The simulation results show that the proposed scheme has 70~80% less delay than IEEE 802.11 multicast and there is no packet loss in the proposed scheme. Thus, it ensures reliability in multicast transmission without incurring overhead. Furthermore, it does not need to set up or replace the infrastructure but simply needs to change the IP address assignment. The stations can decide to receive multicast data by either enabling or disabling the RFMON mode.

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# Performance Evaluation of Load-Balancing Gateway Selection Method in Multi-hop Wireless Networks

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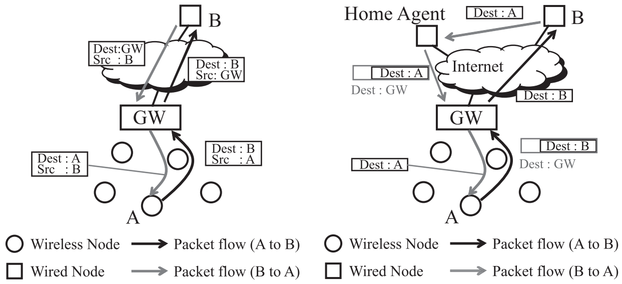
***Abstract*—In wireless multi-hop networks having multiple gateways to an external wired network, a host should select its connected gateway when it would like to communicate with an outside host. One possible and simple way is minimum hop gateway selection policy. In our previous study, we have already revealed that when the minimum hop gateway is selected, imbalance of traffic distribution in a network causes significant performance degradation. Wireless communication channel around the gateway is very important resource because all the external traffic goes through the gateway. So, in our previous work, we proposed load-balancing gateway selection, which takes account of not only traffic intensity at a gateway but also interference of wireless channel around a gateway. With a simple scenario, we showed that our proposed method improves gateway throughput performance. In this paper, we evaluate our proposed method with more sophisticated model and show that it improves throughput performance of load-concentrated gateway. We also show that it surprisingly improves throughput performance of not-load-concentrated gateway, i.e., this gateway obtains more throughput even though the number of its connected hosts is increased when our proposed method is applied. Our performance evaluation in this paper newly reveals that our proposed gateway selection method improves total network throughput by significantly reducing generated control packets for route error recovery.**

***Keywords-****Wireless Multi-hop Network; Gateway Selection; Load Balancing.*

## I. INTRODUCTION

Wireless multi-hop networks enable wireless hosts to communicate with each other by relaying packets at intermediate nodes, even when a destination host is located beyond single hop wireless channel area. When a wireless host in a wireless multi-hop network would like to communicate with a wired host located outside a wireless multi-hop network, packets should be transmitted through a wireless multi-hop network to a gateway node, which is connected to a wired network. For this gateway approach for expanding wireless multi-hop networks, several schemes have been proposed.

For communication between a wireless host and the gateway node, i.e., in a wireless multi-hop network part, two approaches, a proactive approach and a reactive approach, have been proposed. In a proactive approach [1]-[3], a gateway node periodically broadcast a control packet and a wireless host can identify its connected gateway. In a reactive approach [4][5], a wireless host transmits a control packet in a flooding fashion, e.g., a RREQ (Route Request) packet in AODV (Ad Hoc On Demand Distance Vector Routing) [6], and a gateway node responses it. When a wireless host receives this response packet, it can identify a gateway host and can transmit data packets.




(a) NAT approach (b) Mobile IP approach

Figure 1. Communication through gateway node

For communication between the gateway node and a wired host, i.e., in a wired network part, some mechanisms, which enable a corresponding gateway to receive response packets from a wired host, should be implemented. There have been proposed several approaches, which make use of existing networking technologies, Mobile IP [7] and NAT (Network Address Translation) [4]. In NAT approach (Figure 1 (a)), a gateway node behaves as a NAT router and exchanges source or destination address in arrived packets. In this approach, a node outside ad hoc network sends a packet to the designated gateway, which means even when an ad hoc host moves and the best gateway to be connected is changed, connection between this host should be through this designated gateway. In Mobile IP approach (Figure 1 (b)) [1]-[3], a gateway node behaves as a foreign agent. Mobile IP approach can be combined with the proactive approach in wireless network part. This is because a wireless host should identify its connected gateway in order to notify foreign agent change in case of change in a connected gateway and only the proactive approach enables a wireless host to identify its connected

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gateway. With this mechanism of notification of connected gateway, Mobile IP approach can be applied to the case of mobility of hosts and is preferable for ad hoc networks.

MIPMANET (Mobile IP for Mobile Ad Hoc Networks) [1] has been proposed as this Mobile IP approach. In MIPMANET, when there are multiple candidate gateways to be connected, a wireless host selects the shortest hop gateway. In the proactive approach, all the gateway nodes broadcast its advertized packet (control packet) and some useful information for gateway selection can be conveyed with this control packet.

We have already proposed load-balancing gateway selection [8], which takes account of not only traffic intensity at a gateway but also interference of wireless channel around a gateway. Simulation results in previous paper showed that our proposed method brings total throughput performance improvement when it was applied only to the wireless hosts, which have several candidate gateways of similar hop distance. In our previous work [8], we used simple simulation model of a square shape. In this model, there are two gateways located at each of two opposed corners. Hosts located close to the diagonal line have a tendency of having similar hop distances to these two gateways. These hosts with similar hop distance has great improvement of throughput with gateway selection because of no (or small) hop distance increase caused by a change of default gateway. This means that it is not surprising that our proposed method in this simple model improves gateway throughput performance. In this paper, we would like to use more general and sophisticated simulation model and show that our proposed method generally improves throughput of load-concentrated gateway. As our newly obtained performance evaluation results in Section 4 shows, our proposed method improves not only throughput of load-concentrated gateway but also of other gateways. Even though the number of hosts selecting these (other) gateways as their default gateway increases with our proposed method, throughput of these gateways are surprisingly improved. We newly reveal that our proposed method can reduces generated control flooding packets, which greatly improved total network performance.

The paper is structured as follows. First, in Section 2, we review the MIPMANET. In Section 3, we explain our proposed method in detail. We evaluate performance using our proposed method in Section 4. Finally, we conclude a paper in Section 5.

## II. MIPMANET

MIPMANET is one of examples, which combine proactive approach in wireless part and Mobile IP approach in wired part. Figure 2 shows overview of MIPMANET behavior. In MIPMANET, a gateway node behaves as a foreign agent of Mobile IP. Each wireless host has global IP address, which is assigned at its home location. A gateway node periodically broadcast an advertizing packet, which

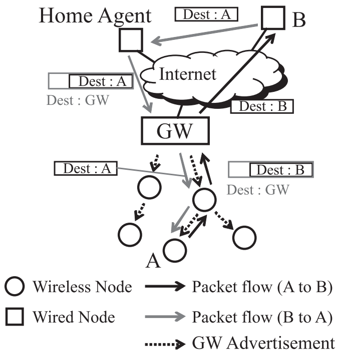



Figure 2. Packet flow in MIPMANET

includes its IP address and some information useful for gateway selection. When each wireless host receives this advertizing packet, it selects one adequate gateway, called default gateway. And a wireless host notifies to its home agent and registers this default gateway as a foreign agent.

Wireless multi-hop networks are generally constructed with hosts with global IP address without shared subnet part. So, when wireless host A would like to communicate with host B, it should first identify whether host A is inside its belonging wireless multi-hop network. Wireless host A can identify by sending RREQ packet of AODV in flooding fashion. When host B exists inside the same wireless multi-hop network, RREP (Route Reply) will be replied by host B. When host A cannot receive any RREP message even with sufficient latency, it can identify that host B is outside its wireless network. In this case, host A encapsulate data packet (to B) and transmits it to its default gateway. When default gateway receives this encapsulated packet, it decapsulates it and transmit this data packet (to B) simply to host B. Host B simply replies to host A of its global IP address, which means this replied packet is forwarded to home agent. Home agent forwards this replied packet to default gateway (foreign agent) of host A by encapsulation. When default gateway receives this forwarded reply, it decapsulates it and forwards it to host A (by using AODV RREQ flooding).

In MIPMANET, when there are multiple gateways in a wireless multi-hop network, all wireless hosts receive an advertizing packet from each gateway. Advertizing packets can convey hop count, which is incremented at each relay node, so a wireless host can identify its hop count to each gateway. MIPMANET [1] has proposed minimum-hop gateway selection policy. However, with minimum hop gateway selection policy, geographical imbalance of generated traffic may cause traffic load imbalance at gateway nodes. A gateway node has generally a tendency of traffic concentration because all external traffic goes through it, so traffic load imbalance will couse serious performance degradation.

## III. Load Balancing Gateway Selection

In this section, we explain about our previously proposed gateway selection method [8], cost function, which takes account of traffic load and wireless channel contention and our gateway selection method.

### A. Cost Function

First, we define cost function of gateway node $i$ affected by host $k$, $C_{ki}$, as follows.

$$C_{ki} = \sum_{h=1}^{h_{ki}} h^{-1} \quad (1)$$

where $h_{ki}$ is the number of hops between host $k$ and gateway node $i$.

When there exist $n_i$ wireless hosts connected to gateway node $i$, we define total cost of gateway node $i$, $C_i$, as follows.

$$C_i = \sum_{k=1}^{n_i} C_{ki} \quad (2)$$

Each gateway node can obtain the number of hops, $h_{ki}$, by receiving RREQ packet from active hosts. It calculates total cost of gateway $i$, $C_i$, and distributes this calculated total cost to all hosts by broadcasting periodically an advertizing message. In equation (1), we considered wireless channel affection around the gateway node is approximately inversely proportional to distance to the gateway node. For details of our cost function, please refer to our previous paper [8].

### B. Gateway Selection Method

When each host receives advertizing message from multiple gateways, it calculates new gateway costs. New gateway cost consists of gateway costs brought by currently connected hosts (gateway costs explained above), and costs brought by the corresponding host.

Figure 3 depicts one example of cost calculation and gateway selection. In Figure 3, two gateways, GW1 and GW2, have 4 and 2 connected hosts, respectively. When a focused host (gray colored one) would like to send data to an external host, it should select an adequate gateway.

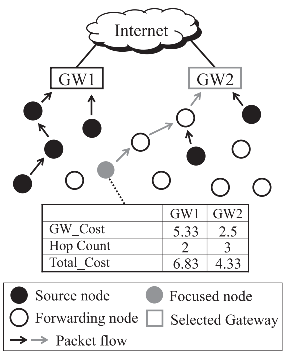



Figure 3. Load-balancing gateway selection

In this example, total gateway cost of GW1 and GW2 is 5.33 (=1/3+2*(1/2)+3*(1/1)+1/1) and 2.5 (=1/2+1/1+1/1), respectively. Focused host's hop count to GW1 and GW2 is 2 and 3, respectively. So, when this host is connected to GW1, total cost of GW1 is 6.83 (=5.33+1/2+1/1). When this host is connected to GW2, total cost of GW2 is 4.33 (=2.5+1/3+1/2+1/1). So, this focused host selects GW2 because GW2 gives smaller cost as its own default gateway.

As shown in this example, each host selects gateway node giving minimum total cost. This cost function takes account of not only traffic load but wireless channel contention around the gateway node. So, with this minimum cost gateway selection, throughput performance of communications through gateway node can be improved.

### C. Threshold for Hop Count Difference

Li et al. [9] shows that throughput performance of wireless multi-hop session is seriously degraded with increase of the number of hops. When a gateway selection policy gives selection of a distant gateway, throughput performance might not be improved. In extreme case of selecting extremely distant gateway, throughput performance through gateway node may be degraded seriously, which means a gateway selection might bring worse throughput.

To prevent this situation, we restrictively apply our gateway selection policy to subset of gateways. We restrict candidate gateways as the following way. When there are multiple gateways (candidate gateways), hop count of the shortest gateway is set as baseline of gateway selection. Gateway nodes whose hop count is within threshold when compared with this baseline, are candidate of gateway selection. Other gateways are not candidate to be selected. For example, when hop count to GW1, GW2 and GW3 is 2, 3 and 4, respectively, and threshold is 1, only GW1 and GW2 are candidates. GW1 is the shortest gateway and its hop count is 2, so 2 is baseline. GW2 is within threshold when compared with this baseline (3=2+1), so GW2 is also a candidate. However, GW3 is outside of threshold from the baseline, so GW3 cannot be a candidate. When there is only one candidate, i.e. only the shortest gateway nodes in a candidate and other nodes are not inside the threshold, a host simply selects the shortest gateway node.

## IV. Performance Evaluation

In this section, performance of our proposed method is comparatively evaluated with the minimum-hop gateway selection. We would like to use more general and sophisticated simulation model and show that our proposed method generally improves throughput of load-concentrated gateway.

### A. Simulation Model

For simulation tool, we use Qualnet 4.5 [10]. In order to evaluate basic performance of our proposed method, we use a static model where no mobility of hosts are considered. Semiautomated node placement model (called in Qualnet) where square field is divided into cells and one host is located randomly in each cell, in used for node placement. When we use pure random model, there may be some heterogeneity of connectivity among wireless host. We would like to avoid evaluating our proposed method in this extremely heterogeneous situation, so semi-automated node placement, which avoids extremely heterogeneous host location is used.

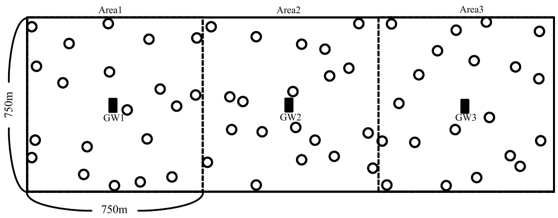



Figure 4. Simulation model (3 gateway)

Table I
SIMULATION PARAMETER

| | | | |
|---|---|---|---|
| number of gateway | 3 | packet size | 512 [byte] |
| number of node | 60 | packet interval | 100 [msec] |
| wireless band | 2 [Mbps] | advertisement interval | 5 [sec] |
| radio range | 250 [m] | simulation time | 1000 [sec] |

In this paper, we use a new network model as shown in Figure 4. One gateway node is located at the center of each square area, thus totally there are 3 gateway nodes. In each square area, 20 wireless hosts are located randomly with semi-automated node placement model. So, there are totally 60 wireless hosts in a whole network. GW advertisement is broadcast every 5 second. Each host is assumed to have exponentially distributed active time and holds communication to outside area through a gateway during this active time period. Inactive time period is also assumed to have exponential distribution and each host is alternately in each of active or inactive time.

We assume imbalanced traffic model where there are 4 active hosts in Area 1 and Area 3, and 16 active hosts in Area 2 on average. So, in Area 1 and Area 3, average active time and inactive time is 2.5[sec] and 10.0[sec], respectively. In Area 2, average active time and inactive time is 10.0[sec] and 2.5[sec], respectively. So, in this traffic model, Area 2 has more generated traffic than Area 1 and 3.

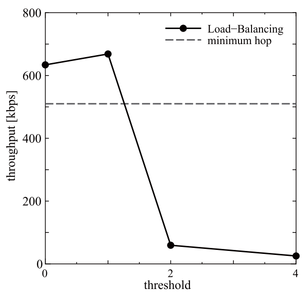
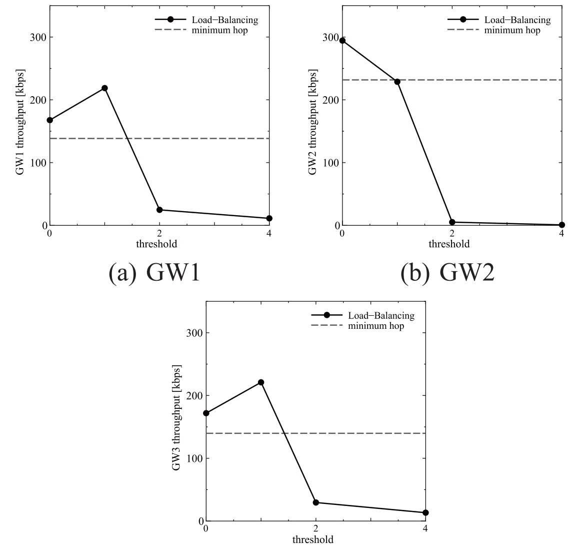



Figure 5. Gateway throughput characteristics




(a) GW1

(b) GW2

(c) GW3

Figure 6. Throughput characteristics of each GW

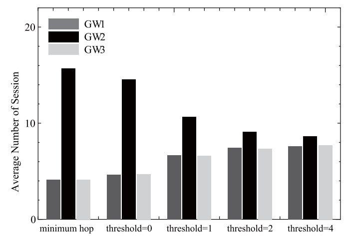



Figure 7. Average Number of Session

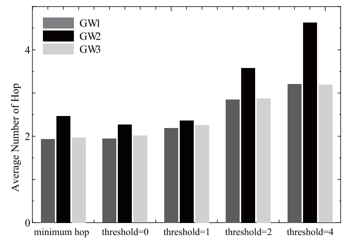



Figure 8. Average Number of Hop

MAC and routing protocol is IEEE 802.11DCF and AODV, respectively. Other simulation parameters are shown in Table 1.

### B. *Gateway Throughput*

In this section, performance of our proposed method is comparatively evaluated with the minimum-hop gateway selection. Figure 5 shows total throughput performance of our proposed method. vertical axis shows total throughput of 3 gateway nodes. Horizontal axis shows threshold defined in Section III.C. The dotted line in this figure is total throughput of the minimum-hop gateway selection. As shown in this figure, total throughput of our method of threshold=1

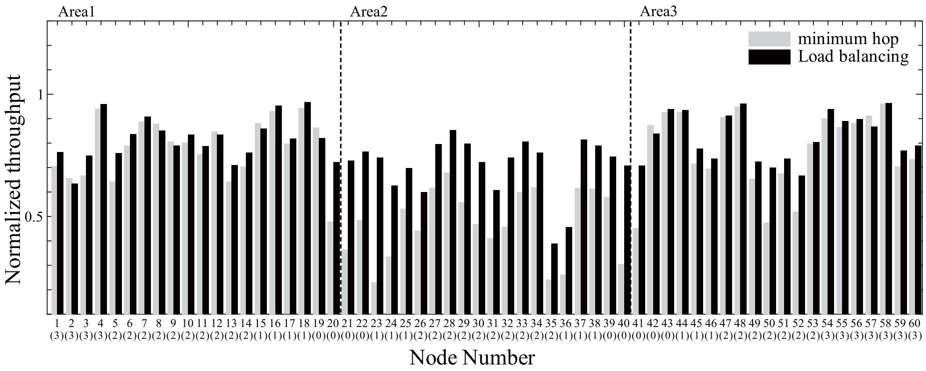



Figure 9. Normalized throughput of each node

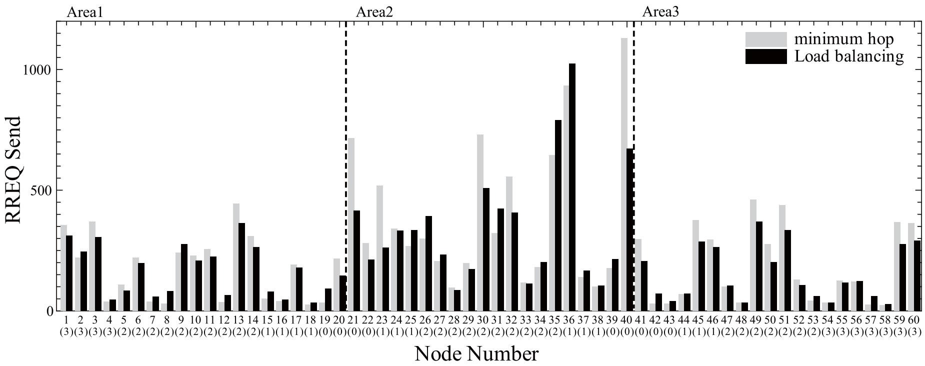



Figure 10. The number of RREQ sent by each node

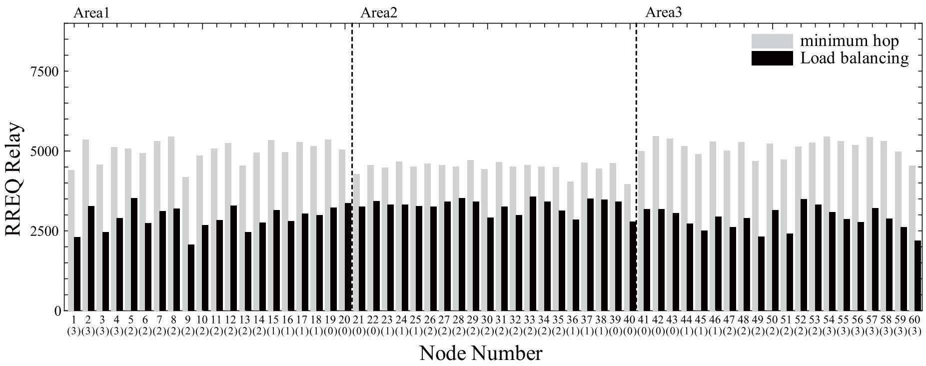



Figure 11. The number of RREQ relayed by each node

is improved approximately 31% when compared with the minimum-hop gateway selection.

In Figure 6 (a), (b) and (c), we show respective throughput of GW1, GW2 and GW3. Figures 7 and 8 show average number of session and hop in our proposed method, respectively. In a simulation model in this paper, more nodes generate traffic in Area 2 and more traffic is generated here. As threshold increases, the number of sessions arranged to each gateway is more balanced, as shown in Figure 7. This means from the viewpoint of load balancing, large threshold is preferable. However, as shown in Figure 8, the number of hops for each session increases with increase of the threshold. In ad hoc networks, increase of hops leads to significant throughput degradation. So, as shown in Figure 6, total throughput of GW1 and GW2 with larger threshold than 1 is degraded even though the number of its connected sessions is increased.

In the evaluation of our proposed method, hereafter, we use the best parameter value of threshold=1.

### *C. Detailed discussion for throughput performance*

In this section, we evaluate throughput characteristics for each host. Figure 9 shows normalized throughput characteristics of each host. Vertical axis shows throughput

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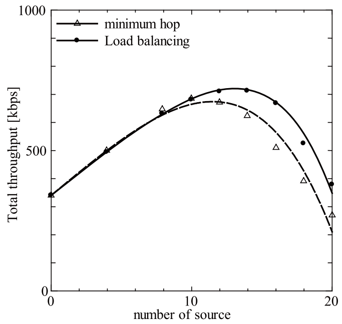


Figure 12. The number of souces v.s. gateway throughput characteristics

normalized with generated traffic volume. Horizontal axis shows node number. The number in parenthesis is hop count difference between the closest gateway and the second closest one. The node ID is allocated in the order of this number. In our simulation model, more active source hosts are located in Area 2 than other two areas, Area 1 and Area 3. In our proposed method, Area 2 traffic around the edge will be induced to other areas with load balancing effect. So, reduction of traffic intensity to GW2 is expected to improve throughput performance of all nodes in Area 2. Simulation results shown in Figure 9 confirm this load balancing effect in Area 2.

Figures 10 and 11 show RREQ generation characteristics and RREQ relay characteristics, respectively. As shown in Figure 10, the number of generated RREQ packets is generally decreased with our proposed method. This leads to great improvement of RREQ relay performance. With load balancing effect, some traffic concentrated to GW2 in minimum hop selection is shifted to other gateways. This leads increase of traffic intensity in Area 1 and Area 3. In ad hoc networks, increase of traffic generally causes decrease of normalized throughput even though absolute throughput increases. However, as shown in Figure 9, our proposed method can slightly increase normalized throughput in Area 1 and Area 3 even though traffic intensity to these two areas is increased. This surprising results are brought by decrease of control packets, i.e., generated and relay RREQ shown in Figures 10 and 11.

Figure 12 shows total throughput characteristics. In this evaluation, number of active sources in Area 2 is changed as simulation parameter and is horizontal axis of Figure 12. Vertical axis shows total throughput of the whole network, i.e., summation of GW1, GW2 and GW3 throughput. As shown in this figure, with our proposed method, concentrated traffic to Area 2 is adequately guided to other areas by load balancing effect, so the number of sources giving maximum throughput is shifted from 12 to 14. This means Area 2 can include more sources in our method.

Newly revealed features of our proposed method are as follows:

- Improvement in normalized throughput is obtained not only in heavy-loaded area but also in light-loaded area because of reduction of control overhead.
- This surprising win-win relationship for heavy-loaded and light-loaded area improves also total throughput of the whole network.

## V. CONCLUSION

In this paper, our proposed method is comparatively evaluated with the minimum-hop gateway selection policy in a new topology. Our simulation results show that our proposed method takes into account gateway load balancing and improves total throughput with threshold 1. We revealed that our proposed method improves not only load-concentrated gateway throughput but also other gateway's throughput. We carefully investigated a reason for its throughput improvement and reveal that our proposed method can reduce the number of generated RREQ floodings and thus improves throughput performance of all hosts in a whole multi-hop wireless network.

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# Multipath Channel Model for MIMO-based Broadband Power Line Communications

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***Abstract*—Broadband power line communication (BPLC) is a promising medium for network access technology by which broadband services can be offered, such as smart grid, broadband Internet access, digital entertainment, and home networking services. This paper presents a modified multipath channel model for multiple-input multiple-output (MIMO) systems and analyzes the channel capacity and bit-error-rate performance of MIMO orthogonal frequency division multiplexing (MIMO-OFDM) based BPLC over that channel model. The widely-used single-input single-output (SISO) channel model (also called Zimmermann's model) is extended to the MIMO channel model considering coupling effects among conductors.**

***Keywords-MIMO; OFDM; crosstalk; channel capacity; broadband power line communication.***

## I. INTRODUCTION

The greatest advantage of power line communication (PLC) is that there is no need for new infrastructure, which is cost efficient in the implementation. The other merits of PLC are as follows: it contributes to low power consumption; it can be applied for the use of energy control and monitoring; it reduces the wiring and lightens the transportation system (vehicles, aircrafts, etc.). In other words, PLC is more "green". Definitely, PLC provides a green ubiquitous concept, economical realization and installation. Furthermore, Broadband PLC (BPLC) can provide high-rate or high performance data services through power line channels. Hence, it can be sufficiently capable of serving as a means of communication able to support future green-energy based smart grid applications. Since the BPLC standard (IEEE 1901) is recently adopted [1], there has been growing interest to BPLC.

Several techniques to model channel transfer characteristic of PLC networks have been presented in various literatures [2] – [4]. Zimmermann and Dostert in [3] state the variety of loads connected to the network terminals and the presence of several cable branches which cause impedance mismatches. Due to these impedance mismatches, many reflections of signal appear during its transfer along the cable. This is called multipath fading or frequency selective fading. The signal distortion experience as a result of frequency-selective fading can be significantly reduced, even if not completely eliminated, by the introduction of efficient multicarrier modulation techniques. One of these efficient multicarrier modulation techniques is orthogonal frequency division multiplexing (OFDM) [5]. OFDM is a smart technique for reliable broadband communications over a wireless channel, which reduces inter-symbol interference and can be applied in PLC [6]. Multiple-input multiple-output (MIMO) communication is a well-known technique to improve data capacity in radio transmission system and can be similarly applied to PLC by substituting transmit and receive antennas with signal feed and receive ports as well as the wireless channel with the existing electrical wiring [7], [8]. In order to investigate the performance of the MIMO power line network, a reliable MIMO multipath channel model needs to be exploited. However, until now, there is no generally established PLC channel model for MIMO based BPLC systems.

The objective of this paper is to design a multipath channel model that could be reliable in analyzing MIMO BPLC systems. We modified the widely used single-input single-output (SISO) channel model [3] into MIMO channel model which defines multi-antenna channel characteristics including coupling effects among conductors. The contributions of this paper include: (a) MIMO multipath channel characterization based from multi-conductor transmission line (MTL) theory [9]; (b) analytical model which considers crosstalk caused by coupling effects between conductors; (c) bit error rate (BER) and channel capacity analysis for a suggested MIMO-OFDM BPLC system.

For MIMO BPLC, a pair of conductors forms a single antenna path (even there has no real antenna). Thus the coupling effect among conductors should not be avoidable [9]. However, existing MIMO systems [7], [8], which have been introduced until now, do not consider crosstalk between conductors forming MIMO channels such that their simulation results are too optimistic. In this paper, the widely used multipath channel model proposed in [3], which is intended for SISO systems, is modified into a multipath channel model for MIMO systems with coupling effects.

The remainder of this paper is organized as follows: transmission line analysis and MIMO channel modeling are presented in Section II. Section III shows the coding scheme and MIMO-OFDM implementation. Section IV shows the simulation results. Concluding remarks are drawn in Section V.

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## II. Transmission Line Analysis and MIMO Channel Modeling

In this section, we will discuss how the channel model of power cable is derived. MIMO is well-suited for BPLC because of its great advantages. One of its advantages is very high capacity and spectral efficiency achieved by simultaneously employing space, time, and frequency domains. A key component of MIMO BPLC is the improved communications reliability, i.e., reduced bit error rate (BER) and improved data capacity, which can be achieved with reasonable computational complexity. However, for the implementation of MIMO BPLC, coupling effects between conductors should be considered.

### A. Modified MIMO Power Line Channel Model

In this paper, the widely used channel model [3] for SISO system is modified into MIMO system which considers the coupling effects among conductors. Zimmermann and Dostert in [3] state that PLC channel can be regarded as multipath scenario with frequency selective fading. They also state the variety of loads connected to the network terminals and the presence of several cable branches which cause impedance mismatches. These impedance mismatches produce multipath propagation of the signal in the BPLC environment. Due to impedance mismatches, many reflections of signal appear during its transfer along the cable. Summing up multi-path propagation, signal attenuation and delay leads to the channel transfer function (CTF) [3]:

$$H(f) = \sum_{p=1}^{N_p} \underbrace{g_p}_{\text{weighting factor}} \underbrace{A(f,d_p)}_{\text{attenuation portion}} \underbrace{e^{-j2\pi f d_p/V_d}}_{\text{delay portion}}, \tag{1}$$

where $N_p$ is the total number of fading paths, $g_p$ is the $p$th path weighting factor, $d_p/V_d = \tau_p$ is the $p$th path propagation delay, $d_p$ is the $p$th path length and $V_d$ is the phase velocity. As depicted in (1), the propagation signal is affected by attenuation, $A(f,d_p)$ increasing with the $p$th path distance $d_p$ and frequency $f$. The cable loss derivation further gives the formula of attenuation factor, which is mainly affected by primary cable parameters, in the form of complex propagation constant [3]:

$$\gamma = \sqrt{(R' + j\omega L')(G' + j\omega C')} = \alpha + j\beta, \tag{2}$$

where the real part of the propagation constant $\gamma$ is the attenuation constant $\alpha$, while $\beta$ is the phase constant. Since we are only interested in the attenuation constant, $\beta$ is negligible. The primary cable parameters $L'$ and $C'$ (inductance and capacitance per unit length, respectively) can be estimated by the geometric dimensions and the material properties while $R'$ and $G'$ (resistance and conductance per unit length, respectively) depend on frequency. Tonello, D'Alessandro, and Lampe in [5] also state that the attenuation of a power line cable can be characterized by

$$A(f,d_p) = e^{-\alpha(f)d_p}, \tag{3}$$

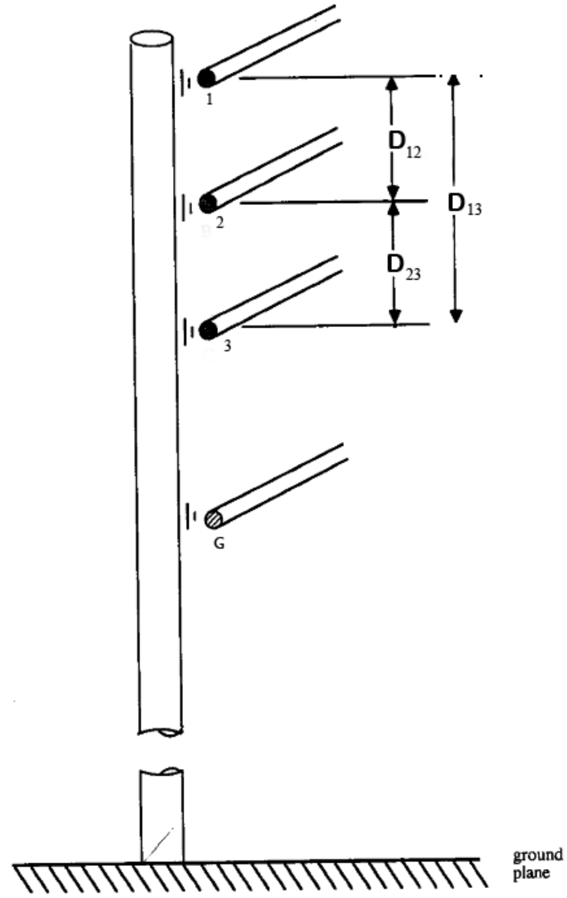


Fig. 1. Typical overhead MV power line

where $\alpha$ can be extracted from (2). Therefore, by substituting (3) to (1), the frequency response can be simply given as

$$H(f) = \sum_{p=1}^{N_p} g_p e^{-\alpha(f)d_p} e^{-j2\pi f d_p/V_d}. \tag{4}$$

We will enhance this SISO channel model into the MIMO channel model. If we are going to transform SISO system into MIMO system for channel capacity improvement, the parallel conductors definitely affects the other conductors due to coupling effects. This coupling effect is commonly present even on different types of multiconductor configuration: low voltage (LV) overhead, LV underground, medium voltage (MV) overhead, and MV underground. Throughout this paper, for simplicity, we mainly focus on MV overhead power lines to construct a MIMO channel, as depicted in Fig. 1.

In MIMO system, the channel transfer function (CTF) for the $i$th transmit antenna to the $j$th receive antenna path (where $i = 1, 2, \cdots, I$ and $j = 1, 2, \cdots, J$) can be written as

$$H_{i,j}(f) = \sum_{p=1}^{N_p} g_p e^{-\alpha_{i,j}(f)d_p} e^{-j2\pi f \tau_p}. \tag{5}$$

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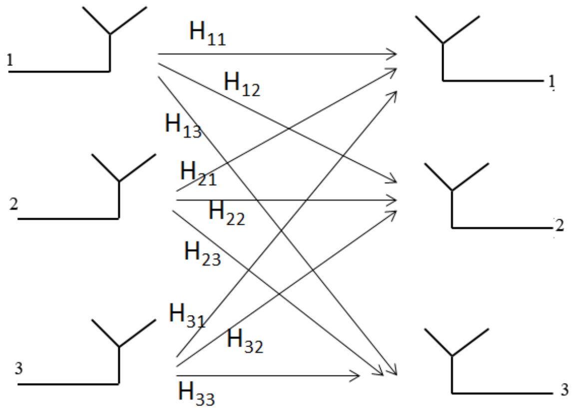


Fig. 2. $3 \times 3$ MIMO System

Equation (5) can then be extended to the overall transfer function matrix:

$$H_{(MIMO)} = \begin{bmatrix} H_{1,1} & \dots & H_{1,J} \\ \vdots & \ddots & \vdots \\ H_{I,1} & \dots & H_{I,J} \end{bmatrix}, \quad (6)$$

where $I$ is the number of transmitting antennas and $J$ is the number of receiving antennas.

For outdoor network with three-phase conductors configuration, as shown in Fig. 1, $3 \times 3$ MIMO system can be obtained by simply pairing Wire 1 to Wire G (ground), Wire 2 to ground and Wire 3 to ground to form antenna 1, 2 and 3 respectively, as shown in Fig. 2.

For this MIMO system consisting of three transmitter and receiver ports, the channel matrix can be written as

$$H_{(MIMO)} = \begin{bmatrix} H_{1,1}(f) & H_{1,2}(f) & H_{1,3}(f) \\ H_{2,1}(f) & H_{2,2}(f) & H_{2,3}(f) \\ H_{3,1}(f) & H_{3,2}(f) & H_{3,3}(f) \end{bmatrix}, \quad (7)$$

where diagonal terms of $H_{(MIMO)}$ indicate co-channels and its anti-diagonal terms indicate cross channels. Since MIMO system is considered, the coupling effect should be considered such that the anti-diagonal terms $H_{i,j}$ (where $i \neq j$) might not be zero.

The attenuation constant $\alpha_{i,j}$ in the attenuation portion of $H_{i,j}$ in (5) can be extracted from

$$\alpha_{i,j} = \text{Real}\left\{ \left( \sqrt{(R'' + j\omega L'').*(G'' + j\omega C'')} \right)_{i,j} \right\}, \quad (8)$$

where the operator $.*$ indicates the element-wise matrix multiplication. This element-wise operation of (8) allows us to derive $\alpha_{i,j}$ for a specific antenna channel, $H_{i,j}$, while assuming all other channel inputs except $i$ are zero (by MTL theory [9], it implies that we can derive $\alpha_{i,j}$ while assuming all other circuit inputs except $i$ are zero , i.e., $V_m = 0$ & $I_m = 0$ for $\forall m, m \neq i$). $R''$, $L''$, $C''$ and $G''$ correspond to transmission line matrices [9], which represent the mutual interactions between conductors. The equivalent per-unit-length (p.u.l) parameter model [9] can be used to characterize the overhead transmission line, as shown in Fig. 3.

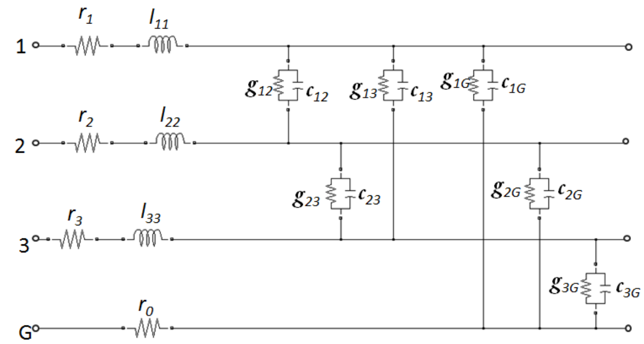

Fig. 3. Equivalent per-unit-length for multiconductor transmission line. (Note that $c_{12} = c_{21} = c_{23} = c_{32} = c_{13} = c_{31}$ and $g_{12} = g_{21} = g_{23} = g_{32} = g_{13} = g_{31}$ are assumed throughout this paper [9].)

The resistance matrix is

$$R'' = \begin{bmatrix} r_1 + r_0 & r_0 & r_0 \\ r_0 & r_2 + r_0 & r_0 \\ r_0 & r_0 & r_3 + r_0 \end{bmatrix}, \quad (9)$$

where $r_0$ is the ground resistance while $r_1$, $r_2$ and $r_3$ are the resistances for line $N$ (which indicates line 1, line 2 or line 3 in Fig. 3) per unit length and computed as

$$r_1 = r_2 = r_3 = \frac{1}{2}\sqrt{\frac{\pi f \mu_c}{\sigma_c}},$$

where $f$, $\mu_c$ and $\sigma_c$ are the wave frequency, permeability and conductivity of conducting material, respectively.

The inductance matrix is

$$L'' = \begin{bmatrix} l_{11} & l_{12} & l_{13} \\ l_{21} & l_{22} & l_{23} \\ l_{31} & l_{32} & l_{33} \end{bmatrix}, \quad (10)$$

where the diagonal terms for $L''$ are the self-inductances for line $N$ per unit length and the anti-diagonal terms for $L''$ are the mutual inductances. The computation for self-inductance per unit length is given as

$$l_{11} = l_{22} = l_{33} = \frac{\mu_o}{2\pi} \ln \frac{GMD}{GMR_L},$$

where

$$GMD = \sqrt[3]{D_{12} D_{13} D_{23}}.$$

Note that geometric mean distance (*GMD*) is a function of equivalent conductor spacings $D_{12}, D_{13}$ and $D_{23}$ between three phase conductors when assuming that the line has one conductor per phase (which is applicable for overhead power line cables). $\mu_o$ is the permeability of dielectric material between conductors and geometric mean radian $(GMR_L)$ is the actual conductor radius $r$.

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The anti-diagonal terms in (10) represent the mutual inductances between conductors and can be computed as

$$l_{12} = l_{21} = k\sqrt{l_{11}l_{22}}$$
$$l_{13} = l_{31} = k\sqrt{l_{11}l_{33}}$$
$$l_{23} = l_{32} = k\sqrt{l_{22}l_{33}}$$

where the constant $k$ is called the coefficient of coupling, and lies in the range $0 \leq k \leq 1$.

The capacitance matrix is

$$C'' = \begin{bmatrix} c_{11} & -c_{12} & -c_{13} \\ -c_{21} & c_{22} & -c_{23} \\ -c_{31} & -c_{32} & c_{33} \end{bmatrix}, \tag{11}$$

where $c_{11}, c_{22}$ and $c_{33}$ are the self-capacitances for line $N$ per unit length and given as

$$c_{11} = c_{1G} + c_{12} + c_{13},$$
$$c_{22} = c_{2G} + c_{21} + c_{23},$$
$$c_{33} = c_{3G} + c_{31} + c_{32},$$

where $c_{NG}$ (i.e., $c_{1G}$, $c_{2G}$ or $c_{3G}$) are the capacitances between line $N$ and ground (G), which can be computed as

$$c_{NG} = \frac{2\pi\varepsilon_o}{\ln\frac{GMD}{GMR_C}}.$$

$\varepsilon_o$ is the permittivity of dielectric material between conductors and $GMR_C$ is the actual conductor radius $r$. The anti-diagonal terms in $C''$ (11) represent the mutual capacitances, denoted by $c_m$ $(= -c_{12} = -c_{13} = ... = -c_{23})$, between conductors and can be calculated as

$$c_m = 4\pi\varepsilon_o.$$

The conductance matrix is

$$G'' = \begin{bmatrix} g_{11} & -g_{12} & -g_{13} \\ -g_{21} & g_{22} & -g_{23} \\ -g_{31} & -g_{32} & g_{33} \end{bmatrix}, \tag{12}$$

where $g_{11}$, $g_{22}$ and $g_{33}$ are the self-conductances for line $N$ per unit length and given as

$$g_{11} = g_{1G} + g_{12} + g_{13},$$
$$g_{22} = g_{2G} + g_{21} + g_{23},$$
$$g_{33} = g_{3G} + g_{31} + g_{32},$$

where $g_{NG}$ (i.e., $g_{1G}$, $g_{2G}$ or $g_{3G}$) are the conductances between line $N$ and ground (G), which can be computed as

$$g_{NG} = 2\pi f c_{NG} \tan\delta.$$

$\delta$ is the skin depth of the conducting material. The anti-diagonal terms in $G''$ (12) represent the mutual conductances, denoted by $g_m$ $( = -g_{12} = -g_{13} = ... = -g_{23})$, between conductors and can be calculated as

$$g_m = 2\pi f c_m \tan\delta.$$

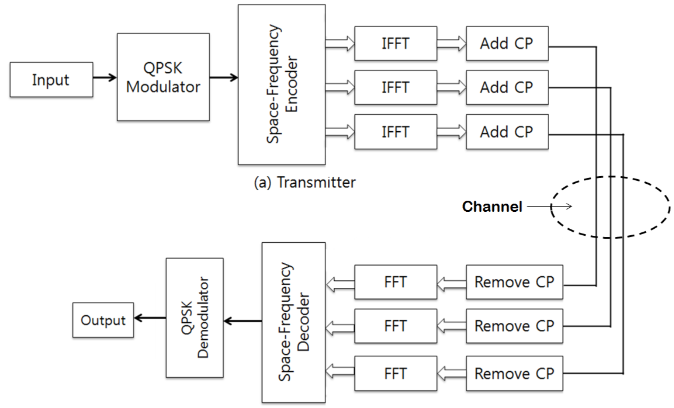


Fig. 4. Block diagram of the proposed $3 \times 3$ MIMO-OFDM System

Performing the required mathematical operation in each element of the transmission parameter matrices to solve for the attenuation factor $\alpha_{i,j}$ in (8), (i.e., $\alpha_{1,1} = \sqrt{(r_1 + r_0 + j\omega l_{11})(g_{11} + j\omega c_{11})}$, $\alpha_{1,2} = \sqrt{(r_0 + j\omega l_{12})(g_{12} - j\omega c_{12})}$, ..., $\alpha_{3,3} = \sqrt{(r_3 + r_0 + j\omega l_{33})(g_{33} + j\omega c_{33})}$), the channel matrix in (7) can be obtained. Hence, this overall transfer function includes crosstalk between conductor. This procedure can be also implemented with the primary line parameters computation for other cases such as indoor with three wires and outdoor with underground power lines as well. For further details of computing those line parameters, please refer to [9] and [10].

## III. Coding Scheme and MIMO-OFDM

Fig. 4 shows the block diagram of a MIMO-OFDM system. In the transmitter side, input signal is fed to the QPSK encoder/modulator block. Accordingly, the modulated symbols are mapped in space frequency (SF) encoder [12]. Then, inverse fast Fourier transform (IFFT) and cyclic prefix (CP) insertion for each antenna path are performed sequentially.

The symbols are then transmitted to the receiver via a MIMO BPLC channel, as shown in Fig. 2. Multiple antennas are applied in order to achieve many desirable objectives for wireless communications, such as capacity increase without bandwidth expansion, transmission reliability enhancement, and co-channel interference suppression for multi-user transmission. The receiver performs the reverse operation of the transmitter. The received signal is carried out by cyclic prefix removal, FFT operation, space frequency decoding, QPSK demodulation and channel decoding. Especially, through the demodulation process, maximum ratio combining (MRC) can be executed to combat the damaging effects of channel fading and effectively combine multiple antenna and multipath fading signals [13]. In this paper, the antenna and fading MRC (AFMRC) scheme [13] is implemented in order to achieve improved bit-error-rate (BER), which is a combining technique of multiple antenna MRC (AMRC) and multipath fading MRC (FMRC).

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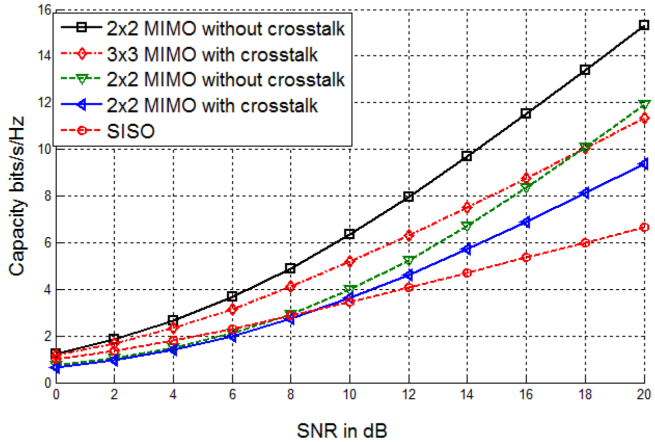


Fig. 5. SISO and $3 \times 3$ MIMO channel capacity

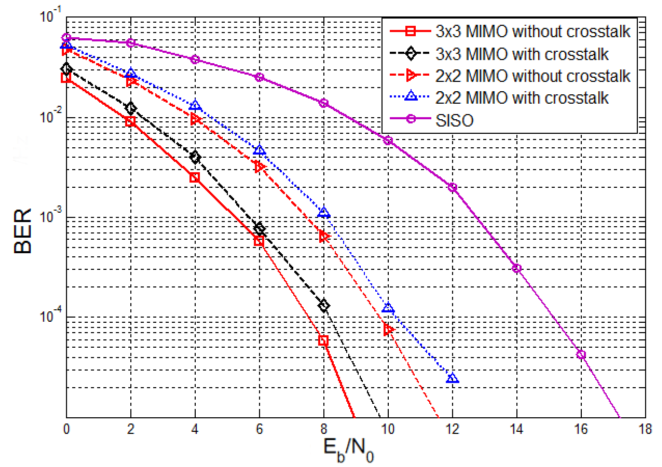


Fig. 6. BER performance comparison of SISO and MIMO with and without crosstalk

TABLE I
SIMULATION PARAMETERS

| Parameters | Value |
|---|---|
| Transfer rate | 6.4 Mbit/s |
| Baseband modulation | QPSK OFDM |
| Baseband Nyquist bandwidth | 2 MHz |
| Information transfer rate | 8 Mbps |
| Spacing between tones | 15.625 KHz |
| Maximum path delay | 0.5 sec |
| OFDM symbol duration | $65\mu$ sec |
| MIMO channel model | $2 \times 2$ and $3 \times 3$ path quasi-static |
| Number of OFDM tones (N) | 128 |
| Noise model | Middleton's impulsive noise [15] |
| Cyclic Prefix Length | 16 |
| Additive White Class A Noise (AWCN) A parameter [15] | 0.1 |

## IV. DATA AND RESULTS

The MIMO channel capacity via power line channels has been simulated using the water filling algorithm [11]. In that algorithm, the optimum power to the parallel subchannels, represented by the diagonal elements of the MIMO channel matrix $H_{(MIMO)}$, is obtained by performing the singular value decomposition on that matrix. In Fig. 5, the channel capacity for a SISO system, a $2 \times 2$ MIMO system (which can be obtained for indoor case with three conductors [14]) and a $3 \times 3$ MIMO system over MV overhead power lines has been plotted. As can be seen from Fig. 5, the channel capacity of $3 \times 3$ MIMO system is 2.15 times greater than SISO. Fig. 5 also depicts the comparison of MIMO without crosstalk and with crosstalk. It is observable that crosstalk damages the data capacity of $3 \times 3$ MIMO system up to 20%.

For the BER performance evaluation, the MIMO-OFDM system presented in Section III is simulated with the parameters listed on Table I. For practical simulation, power line impulse noise [15] is also added to the simulated channel. The performance of MIMO-OFDM is also compared with SISO-OFDM over the power line channel. The BER performance of the proposed model is evaluated both in SISO and MIMO systems and plots are provided in Fig. 6. It can be depicted that the proposed MIMO systems show performance improvement over the conventional SISO system. At the BER of $10^{-4}$, for example, MIMO-OFDM has a performance gain of about 8dB (in energy per bit to noise power spectral density ratio ($E_b/N_0$)) over SISO-OFDM. Fig. 6 also shows the BER results of MIMO with crosstalk consideration, where a performance degradation of 0.7dB can be noticed for both $2 \times 2$ and $3 \times 3$ MIMO at the BER of $10^{-4}$.

## V. CONCLUSION

This paper has developed a modified multipath model for MIMO BPLC systems, where the overall channel transfer function taking into account the crosstalk between conductors forming antenna channels has been derived. This paper has also analyzed channel capacity and BER performance of a MIMO-OFDM BPLC system over MV overhead power line channels.

## ACKNOWLEDGMENT

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# Cloud Applications Versus Web Applications: A Differential Study

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***Abstract*-Cloud computing revolution started reshaping the entire ICT industry and adding new concepts; cloud application is one of these new concepts. This paper aims at differentiating between cloud applications and web applications and to position on the question "what are the differences between cloud applications and web applications?" The paper gives an overview of cloud applications and web applications and discusses differences between both of them. The paper emphasizes that although there are overlapped between cloud applications definitions and web applications definitions; there are also many differences in their characteristics. This work aims to promote the new services that cloud computing is providing nowadays due to its ability to offer many new characteristics through its cloud applications services.**

***Keywords-cloud computing; cloud applications; web applications.***

## I. INTRODUCTION

The terms "Cloud Applications" and "Web Applications" are being used almost interchangeably, but there are real fundamental differences between the cloud applications which are cloud solutions and the web applications which are hosted solutions.

Cloud applications are developed and designed to be hosted by software as a service (SaaS), which is one of a cloud computing service delivery models.

So, what are cloud applications? What are Web Applications? How do they differ from each other? What are the general challenges and issues for both of them?

In answering these questions, this paper will explore the key concepts and characteristics surrounding cloud applications and web applications and will be focusing at differences between them.

The rest of this paper is organized as follows. Section II shows related work. Section III discusses cloud computing definition and its five essential characteristics. Section IV describes cloud computing service models. Section V focuses on the software as a service model (SaaS). Section VI presents cloud applications and its characteristics. Section VII presents web applications and its characteristics. Section VIII analyses the differences between cloud applications and web applications. Finally, Section IX is the conclusion from the finding.

## II. RELATED WORK

There are many technical challenges involved in applications development. One of them is multi-tenancy, which allows a single instance of software to serve multiple organizations by accommodating their unique requirements through configuration at the same time [1].

Xuesong et al. [2] analyzed the similarities and differences between multi-tenancy and isolated tenancy applications and showed that both of them share several features in common. They are both spanning several layers from bottom to top: database, data access, business logic and web user interface. But, Multi-tenancy applications distinguish itself from isolated tenancy applications by their service pattern. The differences of their service patterns result in the differences of their technical implementations:

- The data model in isolated tenancy applications is designed for describing the business requirement, while for multi-tenancy applications besides catering to business requirement, its data model should also be able to accommodate dozens of tenants and ensure data isolation among these tenants.
- Isolated tenancy applications conduct authentication and access control just for users in single tenant, while multi-tenancy applications should also safeguard against intrusion among tenants. So tenant authentication is indispensible in multi-tenancy applications.
- Multi-tenancy applications need a tenant management console for tenant management, which does not exist in isolated tenancy applications.

The above mentioned publication discussed only the multi-tenancy and the isolated tenancy characteristics, and showed how these characteristics affect the applications development and design, and how these characteristics differ from each other, which is not enough to focus only in two characteristics. In this paper, we focus not only on the mentioned characteristics, but also on the other characteristics that make differences between cloud applications and Web applications.

## III. CLOUD COMPUTING

As mentioned in Section I, cloud applications are developed and designed to be hosted on one of the cloud computing service delivery models. This section presents an overview of cloud computing.

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The overarching goal of cloud computing is to provide on-demand computing services with high reliability, scalability and availability in distributed environments [3].

Cisco Systems [3] described cloud computing as: "IT resources and services that are abstracted from the underlying infrastructure and provided "on-demand" and "at scale" in a multitenant environment." Recently, the Information Technology Laboratory at the National Institute of Standards and Technology (NIST) [4] has posted a working definition of cloud computing: "Cloud Computing is a model for enabling ubiquitous, convenient, on-demand network access to a shared pool of configurable computing resources (e.g., networks, servers, storage, applications, and services) that can be rapidly provisioned and released with minimal management effort or service provider interaction."

This definition includes five essential characteristics of cloud computing:

- On-demand self-service. A consumer can unilaterally provision computing capabilities, such as server time and network storage, as needed automatically without requiring human interaction with each service provider.
- Broad network access. Capabilities are available over the network and accessed through standard mechanisms that promote use by heterogeneous thin or thick client platforms (e.g., mobile phones, tablets, laptops, and workstations).
- Resource pooling. The provider's computing resources are pooled to serve multiple consumers using a multi-tenant model, with different physical and virtual resources dynamically assigned and reassigned according to consumer demand. There is a sense of location independence in that the customer generally has no control or knowledge over the exact location of the provided resources but may be able to specify location at a higher level of abstraction (e.g., country, state, or datacenter). Examples of resources include storage, processing, memory, and network bandwidth.
- Rapid elasticity. Capabilities can be elastically provisioned and released, in some cases automatically, to scale rapidly outward and inward commensurate with demand. To the consumer, the capabilities available for provisioning often appear to be unlimited and can be appropriated in any quantity at any time.
- Measured service. Cloud systems automatically control and optimize resource use by leveraging a metering capability at some level of abstraction appropriate to the type of service (e.g., storage, processing, bandwidth, and active user accounts). Resource usage can be monitored, controlled, and reported, providing transparency for both the provider and consumer of the utilized service [5].

## IV. Cloud Computing Service Models

One of the important characteristics of cloud computing is its ability to deliver information technology (IT) capabilities or resources as a services. As shown in Fig. 1, these services are broadly divided into three service models: Infrastructure-as-a-Service (IaaS), Platform-as-a-Service (PaaS) and Software-as-a-Service (SaaS). This models are not new but cloud computing integrate and combine these approaches together.

1. Infrastructure as a Service (IaaS): Products offered via this mode include the remote delivery (through the internet) of a full computer infrastructure (e.g., virtual computers, servers, storage devices, etc.);

2. Platform as a Service (PaaS): To understand this cloud computing layer one needs to remember the traditional computing model where each application managed locally required hardware, an operating system, a database, middleware, web servers, and other software. One also needs to remember the team of network, database, and system management experts that are needed to keep everything up and running. With cloud computing, these services are now provided remotely by cloud providers under this layer;

3. Software as a Service (SaaS): Under this layer, applications are delivered through the medium of the Internet as a service. Instead of installing and maintaining software, simply it can be accessed via the Internet, freeing the customer from complex software and hardware management. This type of cloud service offers a complete application functionality that ranges from productivity (e.g. office-type) applications to programs such as those for Customer Relationship Management (CRM) or enterprise-resource management [6].

## V. Software as a Service (SaaS)

As mentioned before, cloud applications are hosted by SaaS, which is just one of the multiple kinds of services available through the cloud computing. These include Infrastructure as a Service, or IaaS (available through both Public and Hosted Private Communication Service Providers), Platform as a Service, or PaaS, Cloud-based Storage, and even Cloud Based Communications.

The U.S. Information Technology Laboratory at the National Institute of Standards and Technology (NIST) defines Software as a Service (SaaS) as [8]: The capability provided to the consumer is to use the provider's applications running on a cloud infrastructure. The

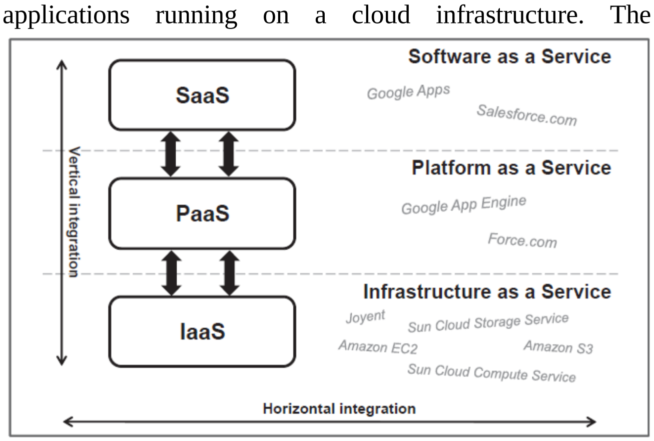


Figure 1. Cloud Computing Service Model [7].

applications are accessible from various client devices through either a thin client interface, such as a web browser (e.g., web-based email), or a program interface. The consumer does not manage or control the underlying cloud infrastructure including network, servers, operating systems, storage, or even individual application capabilities, with the possible exception of limited user-specific application configuration settings.

SaaS is a web-based software deployment model that makes the software available entirely through a web browser as shown in Fig. 2. The users of SaaS software do not care where the software is hosted, what kind of operating system it uses, or whether it is written in PHP, Java, or .NET. And, Also, there is no need to install a single piece of software anywhere [9].

SaaS is basically a term that refers to application in the cloud. Although not all SaaS systems are cloud application, but most of them are.

One of the essential attributes of SaaS model is multi-tenancy, which is defined as an architecture in which a single instance of a software application serves multiple customers; each customer is called a tenant. Tenants may be given the ability to customize some parts of the application, such as color of the user interface (UI) or business rules, but they cannot customize the application's code. Multi-tenancy enables sharing of resources and costs across a large pool of users thus allowing for centralization of infrastructure in locations with lower costs and peak-load capacity increases and utilization and efficiency improvements for systems that are often only 10–20% utilized.

## VI. CLOUD APPLICATIONS

Cloud applications are developed and hosted by the SaaS services providers, accessed by the user customers over the Internet. The SaaS services providers own the software and run it on computers in their data centers. The customers do not own the software, but effectively rent it; usually for a monthly fee. Cloud applications are sometimes also known as hosted software or by its more marketing-friendly cousin, “on-demand”.

The essence of cloud applications is that the customer


Figure 2. SaaS Structure [8].

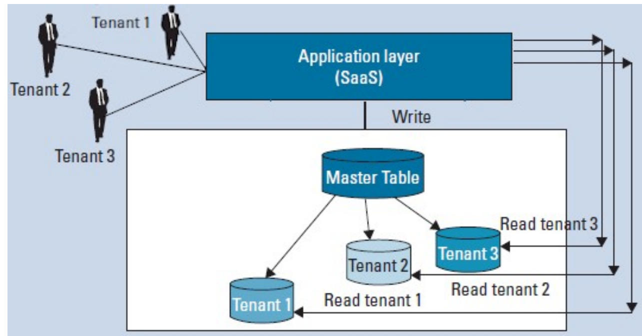


Figure 3. An overview of a general multi-tenancy cloud architecture [10].

does not buy the software, but pays for the service which it provides. The customer does not install software on the servers, but uses capacities of the developer, needing only to pay for the service, cloud server maintenance and consultation services. As a result the customer gets access to the necessary application on a cloud server of the developer which allows saving money and quickly introducing the software to the business. It is needless to say that as a result everyone wins: customers do not need to spend money on system administration, and suppliers, thanks to software installation on their cloud hosting, quickly provide the clients with the necessary service [9].

Mäkilä et al. [10] mentioned five characteristics which are associated with the cloud applications definitions:

1) The application is used through a web browser.

2) The application is not tailored made for each customer.

3) The application does not include software that needs to be installed at the customer’s location.

4) The application does not require special integration and installation work.

5) The pricing of the application is based on actual usage of the software.

Multi-tenancy is becoming a key technology for the success of cloud applications since clients reduce the cost of software use by sharing expenditures, whereas software vendors maximize sales profits. As shown in Fig. 3, multi-tenancy Architectures (MTA) allows multiple customers (tenants) to be aggregated into the same application. Tenants share not only application, but also capital and operational expenses. Moreover, tenants are also able to customize their applications both in endpoint presentation and data structure according to their particular needs but they cannot customize the application's code [11].

## VII. WEB APPLICATIONS

A web application is an application that is invoked with a web browser over the Internet. Ever since 1994, when the internet became available to the public, and especially in 1995, when the world wide web put a usable face on the Internet, the internet has become a platform of choice for a large number of ever-more sophisticated and innovative web applications. In just one decade, the web has evolved from being a repository of pages used primarily for accessing

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static, mostly scientific, information to a powerful platform for application development and deployment; see Fig. 4.

New web technologies, languages, and methodologies make it possible to create dynamic applications that representing a new model of cooperation and collaboration among large numbers of users. Web applications development has been quick to adopt software engineering techniques of component orientation and standard components [12].

Also, web applications can be defined as applications that are accessed over a network such as the internet or an intranet. The term also mean a computer software application that is coded in a browser-supported language (such as JavaScript, combined with a browser-rendered markup language like HTML) and reliant on a common web browser to render the application executable.

It is utilizing web browser technologies to accomplish one or more tasks over a network, typically through a web browser.

Web applications are popular due to the ubiquity of web browsers, and the convenience of using a web browser as a client, sometimes called a thin client. The ability to update and maintain web applications without distributing and installing software on potentially thousands of client computers is a key reason for their popularity, as is the inherent support for cross-platform compatibility. Common web applications include webmail, online retail sales, online auctions, wikis and many other functions [13].

Web applications software and database reside on a central server rather than being installed on the desktop system and is accessed over a network.

The benefits of a web applications start with relieving the developer of the responsibility of building a client for a specific type of computer or a specific operating system. Since the client runs in a web browser, the user could be using an IBM-compatible or a Mac or mobile device. They can be running Windows XP or Windows Vista. They can even be using Internet Explorer or Firefox, though some applications require a specific web browser.

Web applications commonly use a combination of server-side script (ASP, PHP, etc.) and client-side script (HTML, Javascript, etc.) to develop the applications. The client-side script deals with the presentation of the information while the server-side script deals with all the hard stuff like storing and retrieving the information [12].

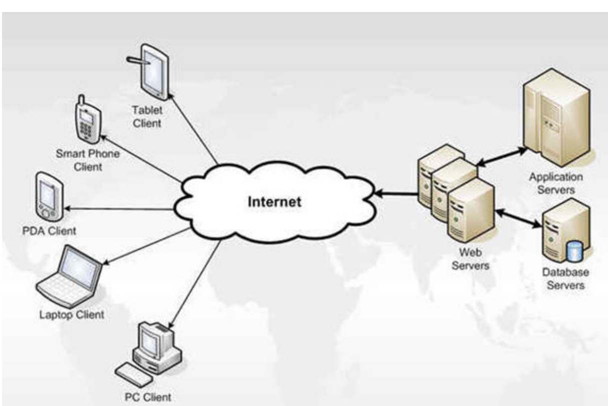


Figure 4. Web application Model.

## VIII. The Differences

A cloud application has an architecture that has the data and the majority of the compute cycles happening in a data center somewhere; all the components of a cloud application are supported by a sophisticated back-end that ensures uptime, security and integration with other systems and supports as many access methods as necessary [14], while a web application is an application program that is stored on a remote server and delivered over the Internet or an intranet through a browser interface.

The following major characteristics play a primary role in the success of the cloud applications, but they are not available in the web applications: inherently scalable and very high uptime.

- Inherently Scalable is an absolute requirement for cloud applications. The cloud applications are written in such a way that it takes full advantage of the underlying platform to be scalable; it must not have limits on number of users or workloads. On the other hand the web applications are generally written for a given platform and are limited by the scalability.
- Very high Uptime is also extremely important for the cloud applications. With a mirrored installations in multiple locations the cloud applications are been deployed, so that the applications are always available (~100% uptime). This high availability architecture calls for hardware redundancy, data mirroring and rapid data synchronization. On the other hand the web applications installed in one location and limited by the availability (uptime).

Most of the web applications - online banking, e-ticketing, flight status checking etc. are really web applications and are limited by scalability and availability.

The cloud applications is essentially a platforms that provide a particular services while the web applications are essentially services that can be accessed over the internet from anywhere on any device.

The cloud applications can be installed on a public cloud or a private cloud and accessed there; conversely, the web applications can be installed on Internet or intranet and accessed there.

Although web applications share some of the same characteristics of cloud applications, they are located elsewhere and are accessible from almost anywhere. Web applications may help in accessing cloud services, but this does not mean they are the same as shown in Table 1. They can be standalone things, too, like applications that allow converting bitmaps to vectors, add drop shadows to images or find out who's stopped following you on Twitter.

Box [15], Dropbox [16], ShareFile [17] and Sugar Sync [18] are almost universally considered cloud apps and, while they have a web interface, they are in no way web applications. They run on back-end systems designed to

TABLE I. COMPARISON BETWEEN CLOUD APPS AND WEB APPS

| | Cloud Application | Web Application |
|---|---|---|
| 1 | The application is accessed through the internet or other computer network | The application is accessed through the internet or other computer network |
| 2 | All cloud applications are web applications | Not all web applications are cloud applications. |
| 3 | Majority of the compute cycles happening in a data center somewhere | Located elsewhere and are accessible from almost anywhere |
| 4 | Inherently Scalable | Limited by scalability |
| 5 | Very high Uptime | Limited by availability |
| 6 | User Data & Business Process store in a multiple replicated data centers | User Data & Business Process store in single data center |
| 7 | Can run on the users' computing systems or the provider's web servers. | Only run on the provider's web servers. |
| 8 | The provided application is standardized for all customers | Each customer uses its own instance of the application. |
| 9 | Multi-tenancy solution. | Isolated -tenancy solution. |

scale and support many users with different requirements. A web interface is just another way to use these services.

Email is also a cloud application. In fact, it was a cloud application back when it was called E-Mail, even though we did not know it. We use Outlook, Mail.app and other mail applications that allow us to access our email stored in Gmail (widely considered a cloud application), Exchange (almost never considered one), Zimbra (if work for VMware) and all the other back ends out there.

Let us consider SalesForce.com; it is accessed primarily over the web, like many other solutions that could be considered web applications, but if the access method is what determines the classification of the application, then you cannot ignore the mobile applications designed to make it easier to use SalesForce or an expense tracking and travel system like Concur [19] (which owns TripIt).

In both cloud applications and web applications, the application is accessed through the Internet or other computer network and the vendor charges service fees [10]. One of the differences is that in cloud applications the provided applications are standardized for all customers, whereas in web applications each customer uses his own instance of the application.

The main difference between cloud applications and web applications is the multi-tenancy. Without the multi-tenancy the solution cannot get the cost efficiencies of true cloud computing, and truly cost effective elasticity and reliability. Web applications could provide elasticity and reliability in a hosted environment, but in practice this is often not truly provided as the costs are too great without multi-tenancy.

## IX. CONCLUSION

Although the cloud applications definition is overlapping with many web applications concepts, this paper discussed the major features of cloud applications that make them different from web applications. The locations where we can store the "user data" and the "business process" is one of the key factors for differentiate between cloud application and web application. Scalability is the key enabler technology of clouds, as it is the basis for cloud application features. Availability is also an important property of cloud applications. Also, the security enhancements for sensitive data which have been added to cloud environments make the cloud applications different from the web applications.

The simple rule is that if a solution is not multi-tenant, it is not a cloud application and it will not give all the benefits that flow from being true cloud application.

Cloud computing has the potential to transform the way information technology (IT) departments relate to, and even think about their role as providers of computing services to the rest of the enterprise. The emergence of cloud computing as an effective software delivery mechanism creates an opportunity for IT departments to change their focus from deploying and supporting applications to managing the services that those applications provide. Successful cloud applications providers produce more value for the business by providing services that draw from both internal and external sources and align closely with business goals.

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# A Study of Information Server over Seamless Service Continuity during Network Handover

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***Abstract*—Mobile devices become more popular nowadays and as they are portable, roaming between heterogeneous networks is essential. With roaming technology, mobile device can perform handover from one network to another network seamlessly. Hence, users may enjoy service on mobile continuously. There are a few approaches discussed among researchers to perform handover for mobile device across heterogeneous network. Most of those existing experiments test bed were built according to the IEEE 802.21 Media Independent Handover standard that is to facilitate a seamless mobility in a heterogeneous network. Information Services (IS) is one of the functionalities in IEEE 802.21 standard for the handover purpose. In this paper, we proposed an experimental test bed setup for seamless video streaming with Information Server in a wireless overlay of 802.11 WLAN. Information Server framework was developed and applied to video streaming application to see the impact of IS over service continuity handover. Results show seamless video streaming service while handoff between two networks and achieved the objective of the study.**

***Keywords-IEEE 802.21; Information Server; network handover; heterogenous network.***

## I. INTRODUCTION

With the proliferation of mobile device, roaming between heterogeneous networks is necessity nowadays. With roaming technology, mobile users can always connected to network services and internet when moving across different access technologies during an ongoing session. Roaming technology can be realized through IEEE 802.21 standard framework that provides Media-Independent-Handover (MIH) [1] to facilitate handover between different networks and different mobile devices.

In IEEE 802.21 standard [2], there is a functionality called media-independent information services which uses Information Database or Information Service (IS) [2] to provide information about neighboring networks surrounding a mobile device. This network information is a combination of several information elements such as Service Set Identifier (SSID), Received Signal Strength Indication (RSSI), Operator Name, Network Type, Encryption Method, Media Access Control (MAC) addresses, etc. Moreover, this network information is structured in such a way that a query/response mechanism can be used to convey the information elements to mobile devices. These information elements are essential for mobile devices to refer to, in the event of performing seamless handover between heterogeneous networks. Fig. 1 shows an example of handover scenario between WIMAX and Wi-Fi that uses IS applied in network handover. It shows the implementation of IS named media-independent handover function (MIHF) that responsible to assist in handover decision making. It provides detail characteristics and services required by Mobility management entity for mobile phone to handoff from WIMAX to Wi-Fi.

Literature studies proven that by applying IEEE 802.21 approach will assist in handover across heterogeneous networks; but, the key issues are service continuity while handing over between heterogeneous access technologies. Most of the existing works were tackling on the network layer to trigger a handover from one network to other network. However, they were not considering the handover process from application layer to support seamless service continuity during network handover. As such, we developed an IS on application layer to see the impact on seamless video streaming during network handover. The network handover decision is done on application layer and controlled by end users. An IS framework created was applied to video streaming in a wireless overlay of 802.11 WLAN [3]. The end target of the experiment is to see the seamless service continuity of video streaming on mobile phone during handover from one network to another network facilitated by IS. A performance study also was conducted to examine that the objective is achieved. This is a preliminary study and application developed as a proof-of-concept for further research on IS in network handover.

The remainder of this paper is organized as follows. Section 2 describes reviews on several existing IS technologies. Section 3 describes on the system scenario and architecture, experiment setup as well as test results. Finally, in Section 4, the paper is concluded.

## II. RELATED RESEARCH WORKS

There are many research works done for network handover that based on IEEE 802.21 standard. Here, we briefly describe some of the existing methods, which are quite relevant to our study. Dutta et al. [4] manipulate service and functionalities in IEEE 802.21 to perform handover between heterogeneous access networks, such as 802.11 [5], 802.16 [6], Code Division Multiple Access (CDMA) [7] and General Packet Radio Service (GPRS) [8]. Experiment result

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shows that disconnection time and packet loss during handover can be reduced by applied 802.21-based approach. Usman and Tinku [9] have proposed an extended model for IEEE 802.21 MIH that offers seamless experience to heterogeneous handover in single and multi-hop network scenarios. Research by San-Jo Yoo et al. [10] have proposed a cross-layer based predictive handover architecture and mechanisms that are implemented in IEEE 802.21 media independent handover architecture. Experiment result demonstrated that the proposed method achieved seamless and proactive mobility for various network environments. Mussabir et al. [11] exploited 802.21 Information Services to optimize the Fast Handovers for Mobile IPv6 (FMIPv6) [12]. The proposed mechanism increase the probability of

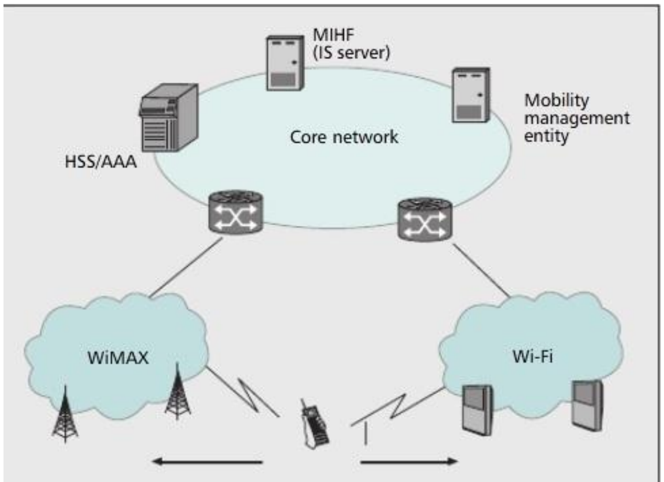


Figure 1. Example of scenario of network handover [3].

predictive mode; so, it reduces the overall expected handover latency in FMIPv6 [12]. Moving-Information-Server (MIS) was proposed by Yaqub [13]. It was implementing in a moving network (e.g. NEMO), where MIS acts both as Reporting Agent (RA) and Information Server (IS). Instead of populating IS repeatedly with redundant network elements, MIS will only store new network elements of available networks. Furthermore, MIS also maintains the updated lists of the known networks mapped with the location information. By using MIS approach, mobile node is capable to know information about the available networks in the geographical domain ahead of time.

## III. FRAMEWORK IMPLEMENTATION

### A. Framework Scenario

In Fig. 2, we assume that the scenario has taken place in an environment where Bob's mobile device will run a video application and will be attached to one access point or in our case WiFi AP 1. A continuous streaming video will stream from Streaming Server through the Internet cloud to WiFi AP 1 before finally reaching Bob's mobile device. As Bob moves to different office departments, WiFi AP 1 strength will become weak; therefore; Bob will experience distortion on his video application. In order to cater this issue, IS is introduced to assist mobile device in performing a seamless handover. This could be achieved by allowing Bob's mobile device to query the IS for other available network, which might currently surrounding it. The query could be performed as soon as Bob's mobile device enters the overlapping zone between WiFi AP 1 and WiFi AP 2. When Bob's mobile device receives a response from the IS, it then perform an auto-connect to WIFI AP 2. As a result, the streaming video will remain in a good quality and Bob will not even notice that his mobile device is already connected to a different AP.

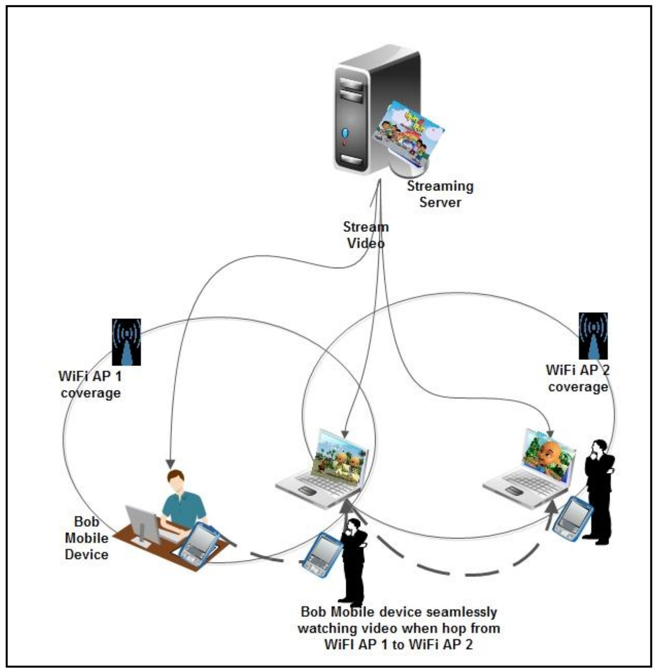


Figure 2. Framework Scenario.

### B. Framework Testbed

The testbed used for carrying out our proposed framework is shown by Fig. 3. Testbed component are: streaming server equipped with VLC Media Player as a content server to continuously stream videos; a desktop as IS, one network switch for Internet connection, two access points, and one laptop, as a mobile device. All devices are connected together through network switch. The two APs are used in order to stimulate seamless handover from one network to another. Mobile device is connected to these APs to perform a handover from AP 1 to AP 2. Streaming server and Bob's mobile device are installed with Ubuntu 9.0 while IS is installed with Windows Vista and equipped with MySQL [14]. IS with response and receive message to

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device is developed in Java program and interact with MySQL server. The experiment was implemented by streaming server streams the video to Bob mobile device.

Fig. 4 shows the flowchart of our experiment. It starts with Bob watched video on his mobile phone. At certain time, Bob pressed a key on his mobile device to perform handover. Upon key pressing, a message was sent to IS contain query such as; the lists of names of the nearest access points to Bob, their signal strength, the security involved in order to attached to the AP, the AP type (i.e., wireless AP, WIMAX), the operators offering the APs, the access availability (with some charging rate), and finally, their distances from Bob's mobile phone.

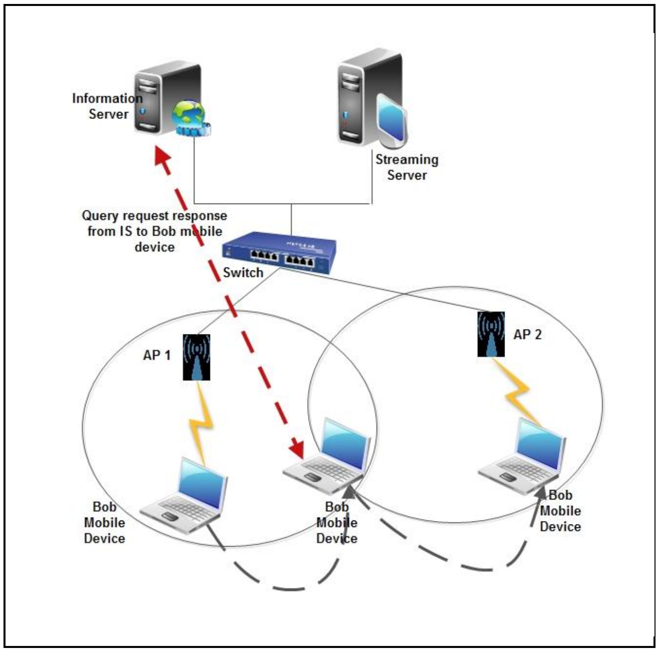


Figure 3. Framework Testbed Setup.

IS will do a searching and checking and provide Bob with a list of information required. For example, the sorting could be based on the APs signal strength and it's distances from Bob's mobile phone. Consequently, Bob will select an AP with the highest signal strength and nearest distance to him.

In this test experiment, AP 2 is the most eligible candidate. Therefore, Bob will switch his mobile device connection from AP 1 to AP 2 and the video service streams continuously to Bob's mobile phone, without Bob realizes the network handoff by his mobile device.

### C. *Testing Methodology and Result*

As mentioned earlier, the objective of this experiment is to see the impact of IS in service continuity during network handover. As such, in this experiment, we measure the performance of network handover process from AP 1 to AP 2. According to [2], characteristics of service continuity and seamless are considered by two performance metrics:

- The handover latency should be no more than a few hundred milliseconds
- The QoS provided by the source and target systems should be nearly identical or should not aware the changes to his communication during and after handover.

The process of handover starts from the key pressed on Bob's mobile phone until the mobile is connected to AP 2 and video packet is sent to Bob's mobile phone by AP 2, as shown by t in Fig. 5. Thus, handover time (t) is measured when a key is pressed on Bob's mobile phone.

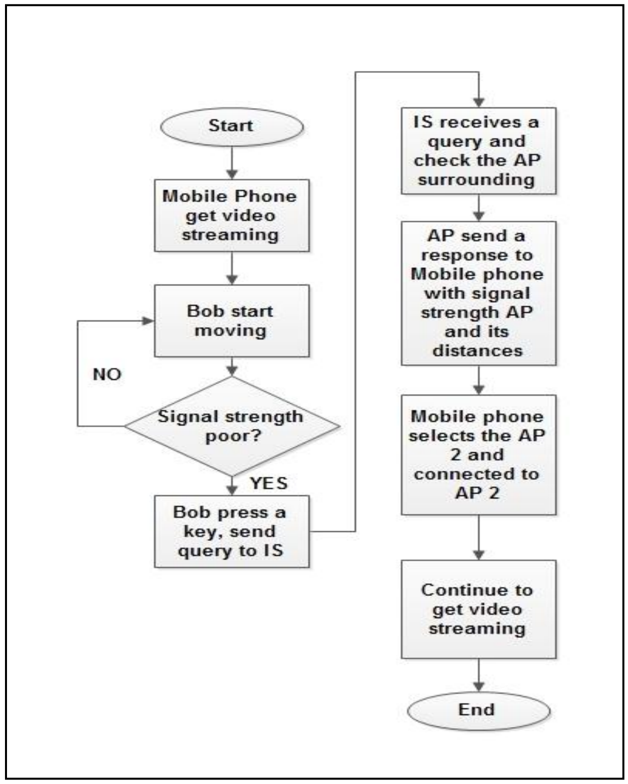


Figure 4. Flowchart of IS Framework.

Wireshark tool was used to gather and evaluate the data. The number of handover attempted was 20 times to tackle deviation issues. Figure 6 depicts the time taken (in seconds) for seamless network handover from AP 1 to WiFi AP 2 when Bob's mobile device pressed the switching key. Seamless handover took time between 0.017 seconds at attempt number 4 and 14, and up to 0.211 seconds at attempts number 7. A little bit delay at attempts number 7; 0.211 seconds. Allover, the average time taken for all 20 network handover is 0.135 seconds, which can be considered as seamless since the time is very minimum. Moreover, while the experiment was conducted, users did not experience any video distortion; video lagging and also did not even notice/aware that the wireless connection had moved from WiFi AP 1 to WiFi AP 2.

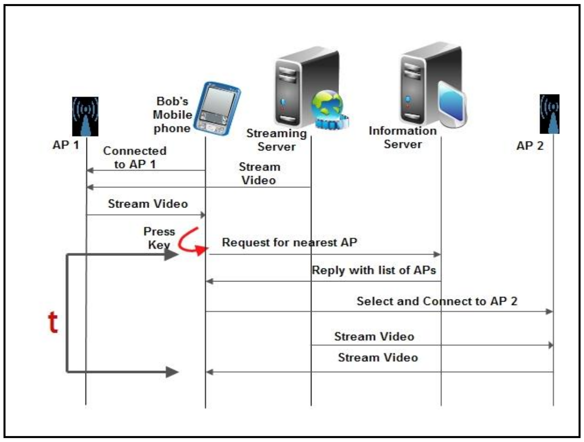


Figure 5. Framework message flow and timing handover.

## I. CONCLUSION AND FUTURE WORK

This paper presented reviews of several different approaches and implementation method of Information Services (IS) based on IEEE 802.21 standard. As explained in the introduction, IS is one of the functionalities in IEEE 802.21 standard for handover purpose. Information provided by IS will facilitates the handover process between different networks. We developed an IS framework as a preliminary study and applied to video streaming in a wireless overlay of 802.11 WLAN to examine the impact of IS over service continuity during handover. The results obtained from performance evaluation show that the streaming video service is seamless while handoff between two networks thus achieved the objective of the experiment. Since it is still in the preliminary study, there is still missing point that should be considered in future research. The point that should be considered in future research are error handling code and also to be implemented across heterogeneous network.

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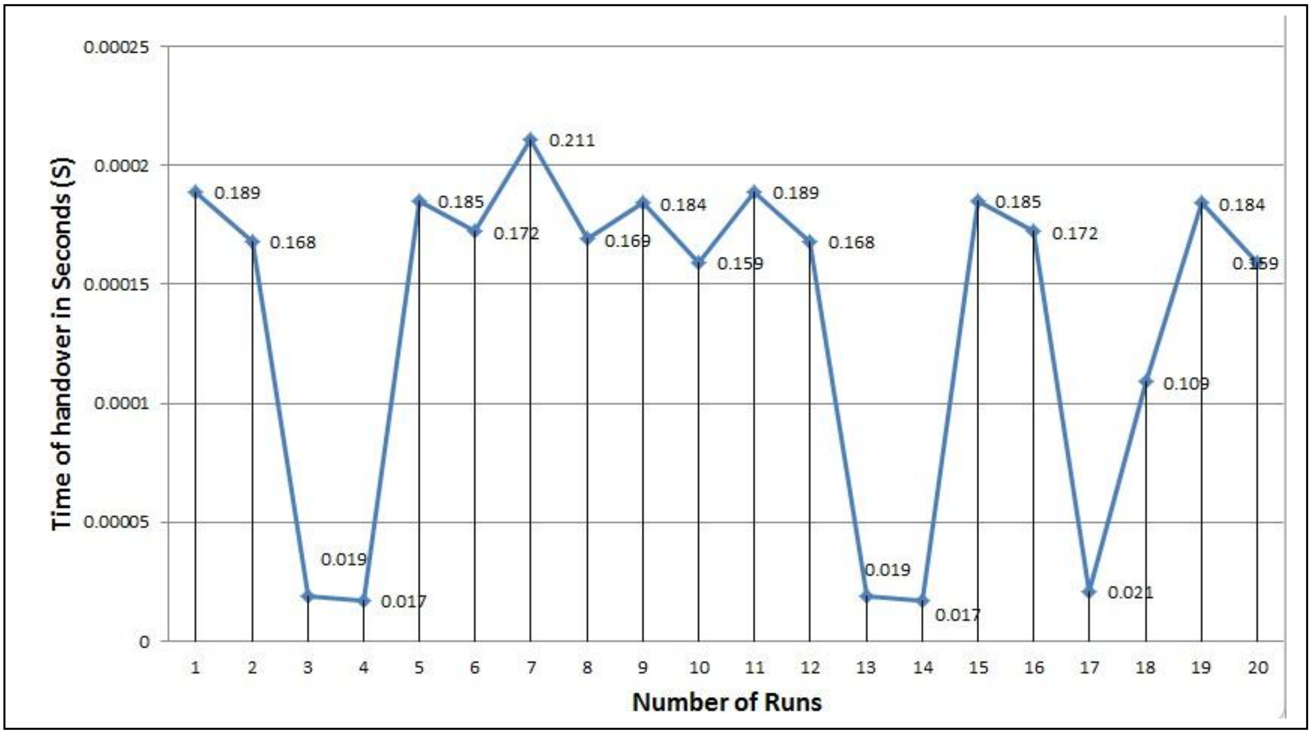


Figure 6. Result of Handover Time