

Server Selection Mechanism for VoIP System in Wireless Mesh Networks

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Abstract— With the rapid development of wireless communication technology and mobile devices, network traffic on smart mobile devices has increased gradually. VoIP is a well-known and mature technology and communication anywhere is emerging as an important application that requires an entirely different way of thinking about communications. In practice VoIP system one kind of issue that vitally impacts the communication is known as service unavailable when loss of Internet connection. In this study we propose a VoIP system architecture that involves the implementation of SIP server and client on a smart mobile device. Every user can play the roles of both SIP server and client, and users can form a self-organized group in Wireless Mesh Networks (WMN) to solve the problem of the unavailability of VoIP in specific area with limited Internet connection. Moreover, this study also proposes a Server Selection Mechanism (SSM) that evaluates the capacity of each smart mobile device to select the most appropriate one to act as the SIP server. The Server Selection Mechanism enhanced the period of whole group connectivity and provided better service for group users. A comparison reveals that the Server Selection Mechanism increase the period of group connectivity to 63% and 28% over those achieved using the non-regular select and random select mechanisms respectively.

Keywords— VoIP, Wireless Mesh Network, Power-Saving, Server Selection Mechanism

I. INTRODUCTION

With the rapid development of wireless communication technology, there are significant amount of users surf the Internet by portable smart devices.. With Internet, smart devices offer communication easiness and convenience for the users through applications. Nowadays, IP telephony, VoIP, has become one of most killer application which could be applied to several purposes such as Internet telephone calls, multimedia conferences, despite of those functions, and one of the most attractive features for us is its scalability. For instance, when traditional telecommunication infrastructure is not available in specific area such as some rural places, mountain areas and post-disaster areas, IP telephony can be implement easily by deploying wireless network. Therefore, a numbers of smart devices such as smart phone or tablet form a self-organized WMN (Wireless Mesh Network) provides IP telephony and multimedia applications for emergency communication is emerging as an important application.

SIP (Session Initiation Protocol) has become a well-known and widely adopted protocol and major protocol for VoIP application because of its compatibility and scalability. SIP, an application-layer control (signalling) protocol for creating, modifying, and terminating sessions with one or more participants, is a protocol which has been developed and designed within the IETF (<http://www.ietf.org>). The protocol has been designed with easy implementation, good scalability, and flexibility [1].

The brief architecture of VoIP system is shown as Figure 1. In a conventional VoIP system, the SIP server is usually implemented on a remote server and connected to an IP network. In this figure, there are some critical requirements so that VoIP application could work appropriately. It must have SIP server for guiding other SIP clients, and SIP server is usually implemented at remote side server. In the other hand, if there has no network infrastructure in specific area, then users cannot connect to internet so that VoIP application cannot work anymore. Therefore, this study provides a solution that SIP server and client are implemented in same device for solving this problem and providing a communicate system by VoIP application for some groups such as rescue team in some post-disaster areas. Owing to every user can play the roles of both SIP server and client, this study also proposes a Server Selection Mechanism (SSM) for choosing the most appropriate user to be the SIP server in every time slot to serve others. The Server Selection Mechanism enhanced the period of whole group connectivity and provided better service for group users. A comparison reveals that the Server Selection Mechanism increase the period of group connectivity to 63% and 28% over those achieved using the non-regular select and random select mechanisms respectively.

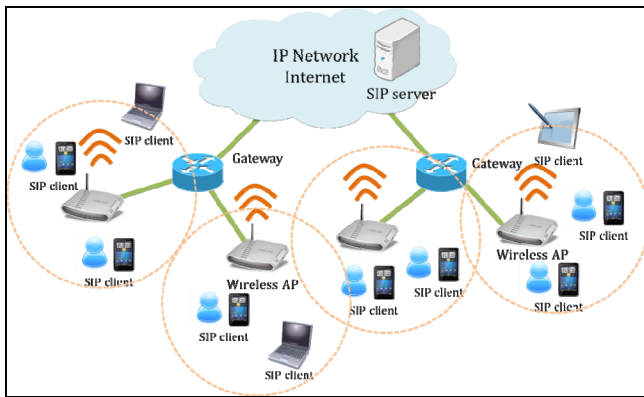


Figure 1. Architecture of Conventional VoIP system

The remainder of the study is structured as follows. In section II shows some related work. In section III presents the system architecture of this study. This study proposes a mechanism for extending the period of whole group connectivity and providing a mobile SIP server by implementing SIP server and SIP client on Android platform. In section IV is the implementation of this study. In section V presents simulation result and performance analysis. Finally, the conclusion and future work of this study are drawn in section VI.

II. BACKGROUND KNOWLEDGE

According to [2], Session control servers are multiplied in server sets. Performance of such replicated session control servers is quantified by transaction control time. Thus, reducing transaction control time enhances performance. Server selection policies (SSP) are crucial in achieving this Goal. The maximum availability (MA) SSP is proposed to improve session control performance in scenarios with server and communication failures.

According to [3], it presents the performance evaluation of Voice over Internet Protocol (VoIP) in Multi-hop Wireless Network (MWN) developed using Multi-radio Access Relay (MAR). The performances of VoIP are investigated using Real Time Protocol (RTP) and Compress Real Time Protocol (CRTP) header techniques. RTP and CRTP are used to transport voice packets using G711.1, G723.1 and G729.2 codec. The performance of VoIP is analysed based on three important elements which are delay, jitter and packet loss. [4] shows performance evaluation and analysis of different VoIP over WLAN mobility schemes is essential in order to meet adequate QoS levels in VoWLAN (Voice over Wireless LAN) deployments. This paper presents a study of real time voice communication QoS in terms of delay, jitter, throughput, and packet loss between three different Wireless Mesh Network mobility schemes in a crowded 802.11 environment. [5-7] present the design and implementation of Asterisk server.

A. SIP (Session Initiation Protocol)

According to [8-11], the Session Initiation Protocol (SIP) is a multi-purpose and flexible signaling protocol for session communications in IP-based networks. SIP mainly handles the

session management domain, and once a session has been established, this session can be used by different communication applications (e.g. VoIP, video conferencing, and instant messaging).

In SIP, there are two types of SIP message. A SIP message is either a Request from a client to a server, or a Response from a server to a client. Both types of messages consist of a start-line, one or more header fields, an empty line indicating the end of the header fields, and an optional message-body [12-14]. The SIP network architecture composed of four types of logical SIP entities and each entity has specific function shown and the SIP network architecture and SIP entities are shown as Figure 2. They setup SIP session to participate in SIP communication and play the roles of client or server or both of them [15-16].

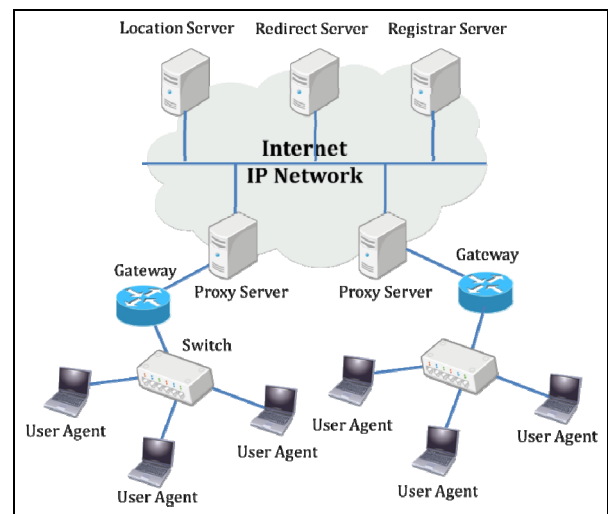


Figure 2. VoIP network architecture and SIP entities

B. Wireless Mesh Network (WMN)

In the last few years, the Wireless Mesh Network (WMN) has gotten lots attention in the research community. WMN has been an emerging technology in recent years due to its attributes such as self-configuration, self-organization, flexibility, high robustness and bandwidth efficiency. Owing to these attributes, WMN are widely regarded as an attractive solution for wireless community networks [17].

In Figure 3, WMN consists of two types of nodes: (1) mesh routers and (2) mesh clients. The mesh routers form the infrastructure of a mesh backbone for mesh clients. In general, mesh routers have minimal mobility and operate just like a network of fixed routers, except that they are connected by wireless links through wireless technologies such as IEEE 802.11. The connections between mesh nodes can be set up and maintained dynamically making a WMN a dynamic self-organized and self-healing network [18]. In WMN, mesh router is equipped with a traffic aggregation device (similar to an 802.11 access point) that interacts with individual mesh clients, so the routing protocols play an important role in WMN to improve the performance of wireless clients and increase network capacity [19-22].

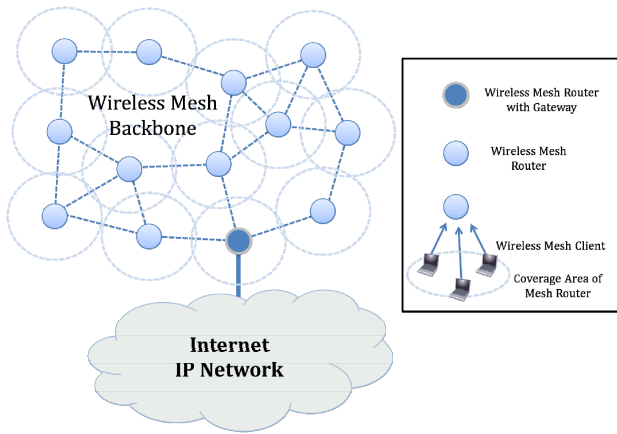


Figure 3. Scenario of wireless mesh network

The primary advantages of WMN are its inherent fault tolerance against network failures, simplicity of setting up a network, and its broadband capability. WMN is dynamically self-organized and self-configured that can be deployed incrementally as needed. While more nodes are installed, the reliability and connectivity of WMN will increase for mesh clients.

III. PROPOSED ARCHITECTURE OF INTEGRATING SIP SERVER AND CLIENT WITH SSM

This section introduces the overview of the proposed VoIP network architecture, subsequently, the section introduces the server selection mechanism for choosing one of the most appropriate user to be the SIP server to provide a qualify communication.

A. System Overview

The study proposes a solution that implements the SIP server and SIP client on Android platform to form a self-organized temporary WMN network provides IP telephony and multimedia applications for emergency communication. The proposed architecture is shown as Figure 4.

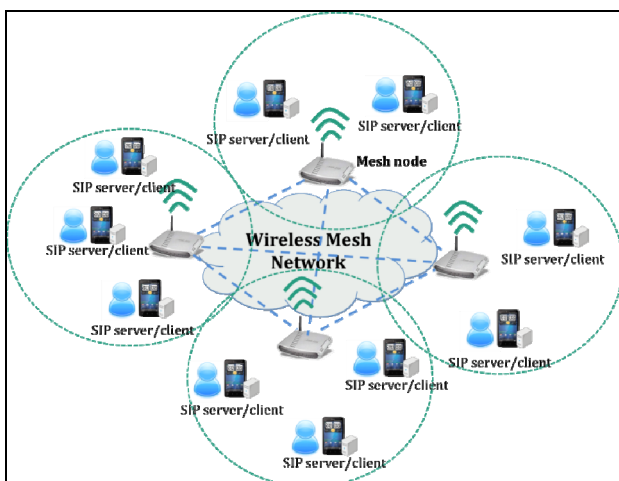


Figure 4. Proposed architecture of communication system

Each user agent can play the roles of both SIP server and client. Figure 5 illustrates the defined modules which are integrated in each user agent. In SIP client, Status Detection is the module to detect every user's status such as power capacity, Wi-Fi signal strength, and CPU utilization of the device. Level Evaluation is the module that evaluates user's each status then acquire correspond level value. Finally, Data Transmission module transmits user's level value, acquired from Status Evaluation module, to the SIP server of the group. In SIP server, Status Request is the module that broadcast to every SIP client for requesting their status due to select the next user to be the SIP server. Server Selection is a mechanism for deciding which user is the most appropriate to be the SIP server in next timeslot. Message Notification is the module to notify the most appropriate user to be new SIP server in next timeslot. The Warning Trigger is the module defines the case of the end of each timeslot and the case of SIP server is getting unavailable. These cases will trigger the module for warning every client and trigger the Status Request module that selecting the next SIP server.

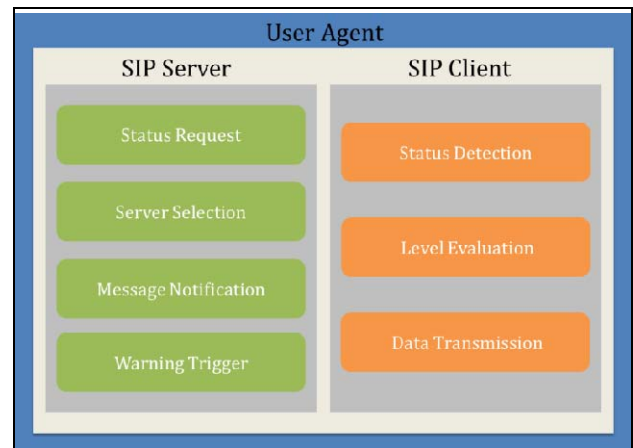


Figure 5. Defined modules in user agent

B. Server Selection Mechanism (SSM)

The proposed SSM mechanism in this study assumes that the communication costs of SIP server and SIP client for server selection is zero and security issue is not considerate. Figure 6 illustrates the scenario of SIP server selection. Initially, this study defaults one of the users, i.e. client A, as the SIP server.

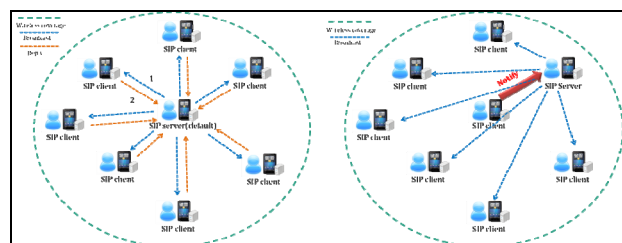


Figure 6. Scenario of SIP server selection

At first, SIP server will trigger the Status Request module and broadcast to every client in the group for requesting each client's status. After clients receive the broadcast they will detect their power capacity status (C_{i_power}), Wi-Fi signal strength status (S_{i_wifi}) and CPU utilization status (U_{i_cpu}), then evaluate correspond level value of each status (L_{i_power} , L_{i_wifi} and L_{i_cpu}). $W_{i(L_power)}$ means the weight value of L_{i_power} , $W_{i(L_wifi)}$ is the weight value of L_{i_wifi} and $W_{i(L_cpu)}$ is the weight value of L_{i_cpu} . At initial stage, all weight values are set to 1, but if 80% of user's level value is less than 3, then corresponding weight value will set to 2. Table 2 shows the parameters of each status. Level evaluation policy is shown as Figure 7.

TABLE 1. PARAMETERS OF EACH STATUS

Symbol	Unit	Definition
C_{i_power}	%	Power Capacity of user i
S_{i_wifi}	dBm	Wi-Fi Strength of user i
U_{i_cpu}	%	CPU Utilization of user i
L_{i_power}	Level	Level of user i's Power Capacity
L_{i_wifi}	Level	Level of user i's Wi-Fi Strength
L_{i_cpu}	Level	Level of user i's CPU Utilization
$W_{i(L_power)}$	N/A	The Weight Value of user i's L_{i_power}
$W_{i(L_wifi)}$	N/A	The Weight Value of user i's L_{i_wifi}
$W_{i(L_cpu)}$	N/A	The Weight Value of user i's L_{i_cpu}
V_i	N/A	User i's Final Value

Case POWER Capacity { if C_{i_power} is greater than 90% L_{i_power} is assigned 10 if C_{i_power} is ranging from 80% to 90% L_{i_power} is assigned 9 if C_{i_power} is ranging from 70% to 80% L_{i_power} is assigned 8 if C_{i_power} is ranging from 60% to 70% L_{i_power} is assigned 7 if C_{i_power} is ranging from 50% to 60% L_{i_power} is assigned 6 if C_{i_power} is ranging from 40% to 50% L_{i_power} is assigned 5 if C_{i_power} is ranging from 30% to 40% L_{i_power} is assigned 4 if C_{i_power} is ranging from 20% to 30% L_{i_power} is assigned 3 if C_{i_power} is ranging from 10% to 20% L_{i_power} is assigned 2 if C_{i_power} is less than 10% L_{i_power} = 1 }	Case Wi-Fi Signal Strength { if S_{i_wifi} is greater than -30dBm L_{i_wifi} is assigned 7 if S_{i_wifi} is ranging from -31dBm to -40dBm L_{i_wifi} is assigned 6 if S_{i_wifi} is ranging from -41dBm to -50dBm L_{i_wifi} is assigned 5 if S_{i_wifi} is ranging from -51dBm to -60dBm L_{i_wifi} is assigned 4 if S_{i_wifi} is ranging from -61dBm to -70dBm L_{i_wifi} is assigned 3 if S_{i_wifi} is ranging from -71dBm to -80dBm L_{i_wifi} is assigned 2 if S_{i_wifi} is ranging from -81dBm to -90dBm L_{i_wifi} is assigned 1 }	Case CPU Utilization { if U_{i_cpu} is greater than 90% L_{i_cpu} is assigned 1 if U_{i_cpu} is ranging from 80% to 90% L_{i_cpu} is assigned 2 if U_{i_cpu} is ranging from 70% to 80% L_{i_cpu} is assigned 3 if U_{i_cpu} is ranging from 60% to 70% L_{i_cpu} is assigned 4 if U_{i_cpu} is ranging from 50% to 60% L_{i_cpu} is assigned 5 if U_{i_cpu} is ranging from 40% to 50% L_{i_cpu} is assigned 6 if U_{i_cpu} is ranging from 30% to 40% L_{i_cpu} is assigned 7 if U_{i_cpu} is ranging from 20% to 30% L_{i_cpu} is assigned 8 if U_{i_cpu} is ranging from 10% to 20% L_{i_cpu} is assigned 9 if U_{i_cpu} is less than 10% L_{i_cpu} = 10 }
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Figure 7. Level evaluation

After the adjustment of weight value according to critical condition of L_{i_power} , L_{i_wifi} and L_{i_cpu} , the SIP server calculates each user's final value V_i , which the maximum V_i will be selected as the new server in the next time slot, 1800 seconds. From above symbols, the SSM formula denotes as Eq.(1) to Eq.(3). In Eq.(2), n means the number of alive nodes in the group. In Eq.(3), n is the number of users, m means the number of factors, W means the weight value of each factor and L means the level value of each factor. If there are some users with same highest final value, the user with highest level

value of priority status will be selected and the priority is sorted as L_{i_power} , L_{i_wifi} then L_{i_cpu} .

$$\{W_{i(L_power)}, W_{i(L_wifi)}, W_{i(L_cpu)}\} = 1, \text{ at initial stage} \quad (1)$$

$$\{W_{i(L_power)}, W_{i(L_wifi)}, W_{i(L_cpu)}\} = 2, \text{ if } \sum_{i=1}^n L_{i_power} \leq 2.4n, \sum_{i=1}^n L_{i_wifi} \leq 2.4n, \sum_{i=1}^n L_{i_cpu} \leq 2.4n \quad (2)$$

$$\text{Max}_{i=0 \rightarrow n} V_i = \sum_{j=1}^m \left(\frac{W_j}{\sum_{k=1}^m W_k} \right) * L_j \quad (3)$$

With the proposed Server Selection mechanism, the user with highest final value, most appropriate user (ex: user B), has been selected to be the new SIP server. Therefore, the SIP server (ex: user A) will notify the user B that he will be the new SIP server in next time slot. Finally, the new SIP server will broadcast to the entire group users for informing them SIP server has been changed in next time slot. Afterwards, users register to the new SIP server for keep communicating to each other.

IV. SYSTEM IMPLEMENTATION

This study uses an open source software Asterisk as SIP server, and every connection is established by native bridge method that media session is established outside of SIP server. Figure 8 shows the example of native bridge method.

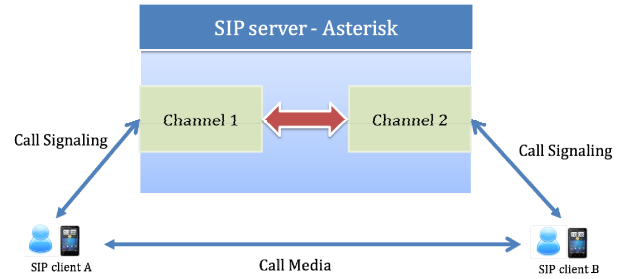


Figure 8. Example of native bridge

Moreover, the number of user and each user account must be set in advance. Four extra libraries, ncurses, SDL, sqlite and termcap to cross-compiler for Asterisk are necessary. After accomplish the compilation and configuration of Asterisk, a partitioned and formatted MMC/SD card with Android system and Asterisk is ready to run on Android based smart device. For SIP client, an open source video SIP phone named Linphone VoIP application is adopted in our test environment.

V. PERFORMANCE ANALYSIS

In this section we estimate the performance of our SSM mechanism with two communication schemes.

A. Single-area Communication System in Wireless Mesh Network

In this subsection we present simulation results to investigate the effectiveness of the study. The communication

scheme could be referred to Figure 6. The results include two cases: (1) Decrease of numbers of nodes with SSM, Random Select Mechanism and Non-Regular Select Mechanism. (2) The variation of the memory usage while changes the server.

In the first case, this study simulates 20 users in the group and sets the time period to 30 minutes. Therefore, when the time is up or SIP sever is no longer available, SSM (Server Selection Mechanism) will be triggered for selecting new SIP server.

Case (1): Figure 9 shows the simulation result of node decrease with SSM, Random Select Mechanism and Non-Regular Selection Mechanism. Random Select Mechanism means it selects the new SIP server randomly and without considering any factors in every trigger. The non-regular select mechanism means the default SIP server doesn't broadcast for changing server in every time slot, it broadcasts only when it's getting unavailable and the new SIP server will be selected randomly.

According to the simulation result, this study summarizes some advantages with SSM as follows:

- Extending the period of whole group connectivity, 20 users still alive, from 5.5 hours to almost 9 hours. Because SSM balances the loading of being SIP server among group users so that the connectivity of whole group can last longer.
- With SSM, the most qualify user is selected to be SIP server in each time slot, and provides better service to other users than other two mechanisms. Because there are some factors has been considered and change server periodically in SSM.

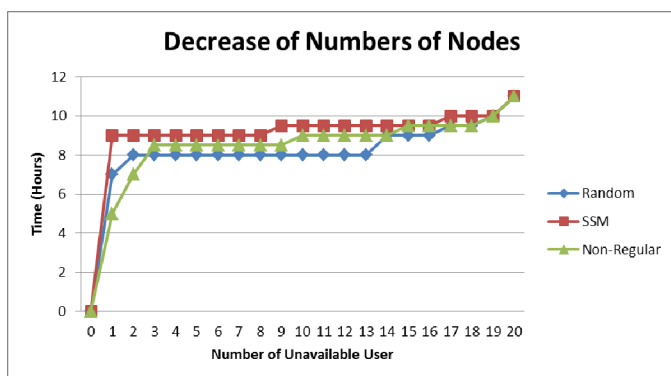


Figure 9. Node decrease with SSM, Random Select and Non-Regular Select

Case (2): Figure 10 shows the variation of current SIP server memory usage and next SIP server memory usage. The wire in green is the SIP server in current time slot and the wire in purple is the SIP server in next time slot. In this case, it simulates there are 20 users in the group. In figure 10, the memory usage of current SIP server is about 62% and the memory usage of next SIP server is about 39%. After the next server has been decided by SSM, the redirects process to new SIP server takes 2.5 to 3.5 second, and the memory usage is decrease to about 41% because of the loading is not heavy as being SIP server. The next SIP server receives the requests

queued in previous SIP server from about 3 to 4 second and it caused the memory usage increase to almost 60%.

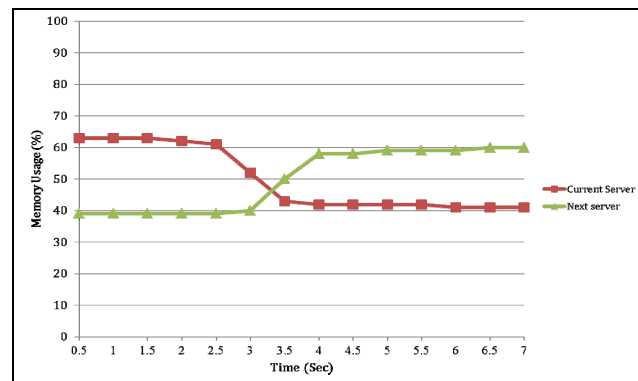


Figure 10. Variation of SIP server memory usage

B. Multi-areas Communication System in Wireless Mesh Network

In this subsection, this study proposes a communication scheme for a wider wireless mesh network, which is composed of several access points. In Figure 11, access points form a wireless mesh network that users can contact other users in different access point wireless coverage via several mesh nodes. In coverage of each access point, there is only one SIP server to serve other clients.

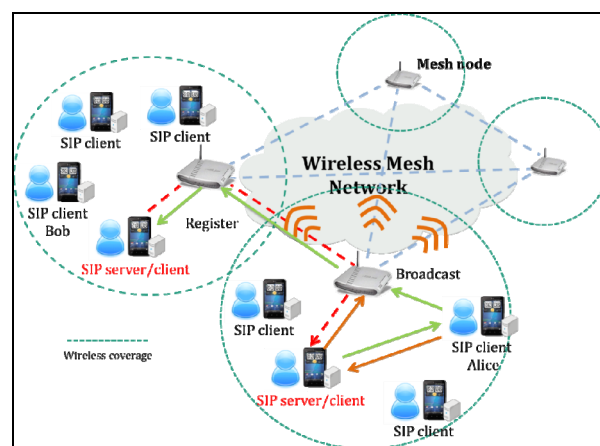


Figure 11. Scenario of cross-area communication

The following describes the simulation results and the results include two cases: (1) Comparison of the delay time (2) Comparison of total packet loss and packet loss rate. The experiment duration in this study is 10 minutes and this study simulates a wireless mesh network environment. The simulation tool is NS2 network simulator, intra-area communication means one hop communication with codec G.711, cross-area communication means three hops communication with codec G.711 and the result of conventional VoIP communication is measured in real environment.

Case (1): Figure 12 shows the comparison the delay time among intra-area communication, cross-area communication, and conventional VoIP communication. Delay time refers to the amount of time it takes to transmit data from source to destination. To compare with the conventional VoIP communication, which the average delay time is about 1.06 second, intra-area communication and cross-area communication is much lower than it because of the number of hops decreased.

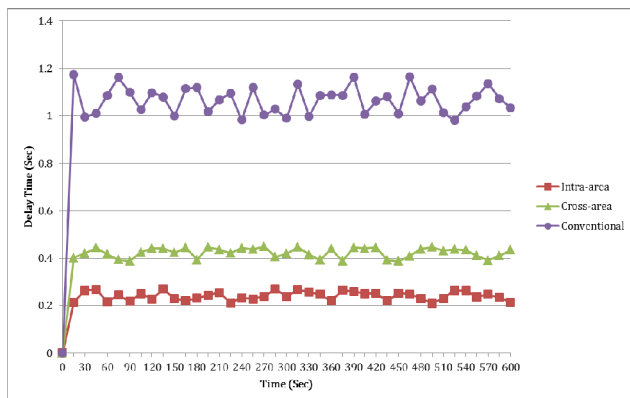


Figure 12. Delay time comparison

Case (2): Figure 13 shows the comparison of accumulative packet loss among intra-area communication, cross-area communication, and conventional VoIP communication. According to the Figure 13 and Figure 14, the accumulative packets loss in intra-area communication is about 87 packets in average and packet loss rate is about 0.29%. The accumulative packets loss in cross-area communication is about 140 packets in average and packet loss rate is about 0.46%. The accumulative packet loss in conventional VoIP communication is about 327 packets in average and packet loss rate is about 1.09%. The comparison of packet loss rate, intra-area communications reduce 0.8% and cross-area communications reduce 0.63%.

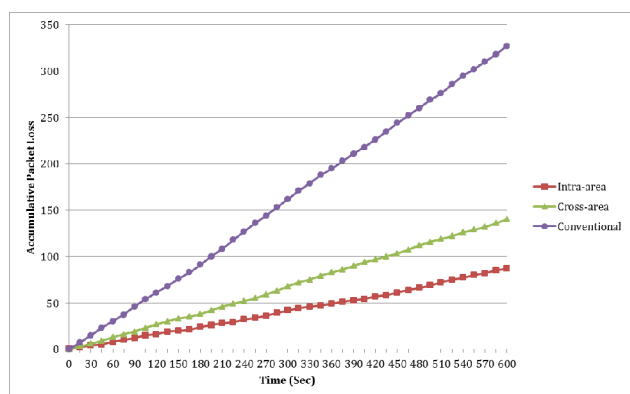


Figure 13. Accumulative packet loss comparison

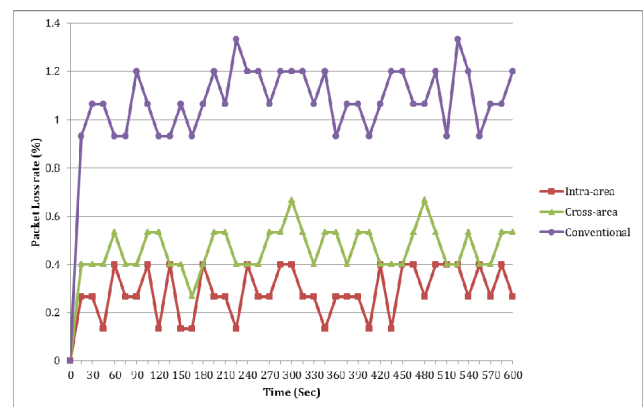


Figure 14. Packet loss rate comparison

With the proposed scheme, the whole communication system coverage area could be scaled up by connecting several access points together. In contrast, the number of hop increases the quality will decrease such as higher packet loss rate and higher delay time.

VI.CONCLUSION

With the rapid development of Internet technology, smart portable devices are getting more popular and widespread. Therefore IP telephony, VoIP, application has become one of most popular and well-known application. This study proposes the architecture of integrating SIP server and client by implementing them on the same device, and proposes a mechanism SSM for choosing the most appropriate user to be SIP server for serve others. With this integrated architecture, it's not only providing the mobility for SIP server but also solves the problem of IP telephony cannot work without connecting to internet in specific area where has no network infrastructure. Moreover, according to the experiment result, with SSM, it can extend the period of whole group connectivity and it can also provide better communication quality for group users.

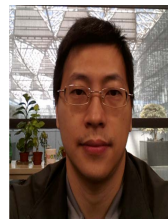
The future works of this study are categorized into two main goals. The first goal is the compatibility. This study integrates the SIP server and client on the Android device. Therefore, this study will research in developing the general solution that can compatible with other operating system such as iOS, Windows 10 in the future. The second goal is enhance the reliability and performance in wireless mesh network. In the section V, this study has introduced the cross-area communication in WMN. To improve the performance of wireless clients and increase network capacity and reliability in WMN, routing protocols is an important factor. Also how to provide better server selection mechanism to improve overall performance is expected. Furthermore, we believe that the study points out a new trend for large-scale emergency communication based on VoIP networks.

ACKNOWLEDGMENTS

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