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Abstract. Many types of Voice over Internet Protocol (VoIP) are used to provide audio or multimedia sessions over Internet Protocol (IP) networks, multimedia communications became very easy by using applications that provide Internet Protocol VoIP like Skype, Viber, WhatsApp, etc. And network simulation has frequently been used to determine the performance feasibility of any network protocols and the model reliability before practical implementation. Several kinds of studies were carried out on voice efficiency over different types of networks environment, among these experiments and researches, finding a better signaling protocol for communication by evaluating and analyzing the performance of the protocols, such as the most commonly used Session Initiation Protocol (SIP) and (H.323) signaling protocols. This paper provides a comparative study of the performance of the two aforementioned protocols that are most widely used over the IEEE 802.11a wireless network by using the QualNet simulator. The study was performed by analyzing the data packets for both initiator and receiver for both protocols, to evaluate the performance based on five metrics: Session total Bytes, Establishment time, Sent, total Bytes received of Real-time transport protocol(RTP), Average End to End Delay and Overall Throughput. The outcomes demonstrated SIP protocol ascendingly rise and superior over H.323 protocol as a permanently started from the Zero-point around 5.4 (Mbps) of the time until the end of the experiment close to 6.2 with a 0.05-second delay while the rival was always below these values of throughput. Hence its superior provides preferability in multimedia sessions. studying and evaluate the performance of two IPs with a high capability simulator QualNet, SIP outright outperforms H.323 provide preferability recommendations in voice calls.

Keywords: QualNet, Simulation, SIP, H.323, VoIP, Wireless Network, Signaling Protocols.

1 Introduction

Millions of customers communicate every day across the internet and the exponential growth and success of various media communications lead to a produce vast amount of data like audio, video, text, etc, and for the last decade, many communications based on VoIP protocols have been widely used for multimedia large data communications[1]. VoIP services are growing increasingly due to their versatile and straightforward implementation and the low cost of traditional public telephone networks [2]. VoIP is a standard instant message chat technology and commonly used in social networking, has the potential to deliver instant message services with less capital, and is exceptionally scalable [3]. This fact underlines the significance of voice-over IP signaling protocols. For many years, accordingly, two protocols have competed in this field, namely H.323 and SIP[4]. They came from various organizations. H.323 is the product of the ITU Telecommunication Standardization Sector (ITU-T), which has been the most relevant agency in telecommunications and telephony for a long time. In contrast, SIP comes from the Internet Engineering Task Force (IETF) this organization oversees the growth of the Internet [5]. We will provide a study and evaluation of these two IPs in terms of properties and performance and communications behavior even in Quality of Service (QoS) [6], and give details on why one was more effective in this mission, the evaluation characteristics are functions of network components, applied applications in the protocol architecture, and supported applications for networking in addition to peer-to-peer telephony [7].
However, the definition of the user agent back-to-back is significant in the creation of complex SIP network applications [4]. Back-to-back users include two user agents, who are defined somewhat as the most complicated form of a SIP proxy, one uses the incoming session, and the other user agent sets another session. Technology at the framework level monitors two sessions signaling and transmits signals from one to the other (after making necessary modifications to messages). Enables specific programs to be introduced, which need some session control and media stream. In a collection of Request for Comments RFC documents [5]. Protocol specification of the SIP provides. Key RFC Paper defines only the portion and features of the protocol needed for first-party call power. Separate records include case notifications mechanisms, third-party call management elements, the presence, instant message services elements, meeting features, and various other elements of the protocol. Any of them are going to be listed here: The main documents, the generic event of notification machine, notification of presence, Notice of Registration, Presence Information Data Format (PIDF), model for the development of presence and instant messaging applications and Call Processing Language (CPL), message request types.
On the other side, our second scope protocol is H.323 protocol is ITU Telecommunications (ITU-T) standard recommendations are that specified for any packet network to provide audio-visual communication sessions. The H.323 protocol describes multiple connectivity components such as Gateways, Gatekeepers,

Multipoint Control Units, and Internet-based and Boundary Elements that provide aggregation and review of multimedia connectivity capabilities. H.323 has seven steps of a call which will facilitate the exchanging of messages or supplementary resources between components. The following are six essential phases of the H.323 functionality: setup call, communication initialization, and exchange capability, visual audio communication establishment, call stability with RTP, media service exchange on call, and termination the call. Although the protocol H.323 has more architectural complexity, it offers continuity of control by schedule calling and has an efficient transmission that allows the system more robust and secure [8]. H.323 is a group of protocols coupled with each other to ensure three primary tasks: signaling, negotiation of codecs, and transport of VOIP data. the protocol guarantees control over the quality and usage of services constraints. Its protocol stack consists primarily of the first H.245 protocol that negotiates the opening and usage of channels and the configurations for VOIP contact. The second Q.931 protocol enables the establishment of signals and calls. Third, the protocol of enrollment, admission, status 'RAS' is the protocol used for the terminal to interact with the gatekeeper. Last Usage of the Real-Time Streaming Protocol (RTCP)/RTP protocol to route and manage sources of audio and video[9].

Besides the general specifications for H.323, there are additional specifications[10]. H.225 and H.24 are the two most important, but the list does not end here, since there are H.235 specifications (which specify H.323 security functions) as well as other specifications. Because the communication network simulators are limited abilities, using the QualNet network simulator is considered to be more effective in terms of network scalability and accuracy compared to (OPNET, NS-3, and NS-2 or even MATLAB) simulators [11] [12] [13]. QualNet simulator is a high-fidelity network analysis program that predicts the efficiency of wireless, wired, and mixed-platform networks and networking systems. QualNet supports thousands of network nodes to be simulated. QualNet provides unrivaled portability of the software and versatility of interfaces. And also QualNet operates on UNIX, Windows, Mac OS X, and Linux parallel and sequential operating systems [14] [15].

Throughout the calling growth, VoIP technology will occupy a huge internet communications bandwidth, resulting in a large increase in demand for the network's transmission capacity. In current mobile network pricing schemes, where billing relies on usage frequency, a considerable rise in network resources, would quickly lead to a dramatic increase in cell network rates. A major VoIP technology challenge is the trade-offs between the volume of data flow and the level of touch in voice communications [3]. Our main problem how to pick the best signaling protocol that achieves high efficiency over wireless networks to accommodate the applicable call density, by taking into account the different metrics to be measured which is Session total Bytes, Establishment time, Sent, total Bytes received RTP average end to end delay and Overall Throughput. The paper achieves the flowing points.

- Factors that determine voice quality include the choice of codec, echo control, packet loss, delay, overall throughput, and the network's design of the network Packet loss causes voice clipping and skips.
- The Quality of Service (QoS) can be achieved by managing router queues and by routing traffic around congested parts of the network.

- Improve efficiency by identifying the best VoIP signaling protocol that achieves high performance over wireless networks.

The rest of the paper is organized as following the second section, focusing on related works. The third section discusses the methodology. The fourth section simulation scenario and the fifth section presents the result and analysis, and the last section sixth concludes the paper.

2 Related Works

Many studies were performed to evaluate the performance of VoIP by changing the codecs other studies discussed the signaling protocols of the VoIP.
Like in [16] study conduct on the performance of VoIP service with the G.729 and G.711 codecs based on the same network topology focus only on the QoS of the Voice over IP platform and the study conducted without taking into account signaling protocols and limitations of the simulation software plus the outcome.
The researchers in [9] evaluate (VoIP) performance and two signaling protocols SIP and H.323, the study performed in a homogeneous 802.11e, and the findings showed the SIP protocol's efficacy in terms of call setup time, but the study focused on MIPv6 and its limitations, especially on mobility speed, and no overall throughput.
The results in [3] concluded that: SIP voice quality is higher than Red5 through conducted comparison of SIP and Red5 flow, and on voice quality acceptance level of the general public In addition to SIP, the focus was put on a different protocol were not H.323.
As discussed in [17] paper draws a comparison using an (OPNET Modeler) between four combinations of H.323 and SIP Protocols and with G .711 and G.729 codec, The combination H.323 and G.729 proved to be the most appropriate one in the connection process, the study limitation was the protocols were not individually performed.
A study performed in [18] to compare the performance in the network of two types of codecs G.711 and G.729 on IEEE 802.11ah standard for voice over IP services, the evaluation concentrated on the following scales packet, delay ratio, average delay, and throughput, the study conducted using network simulators (NS-3), hence the research is a little out of scope.

3 Methodology

To create and manage multimedia sessions, signaling protocols are implemented. There are currently two standard protocols on the market that are commonly used, SIP and H.323 [19]. Our research went through the behavior of two VoIP protocols modulated, and the performance will be evaluated through building scenarios on a QualNet simulator, using the Random Waypoint Mobility Model(RWP) the random nodes will choose the direction and its constant speed in each movement randomly and then after shortstop period it will choose another destination and another constant speed, therefore, it considers close to reality and better than choose various steady scenarios[20]. To investigate the performance of this protocol mechanism, the statistical results will be analyzed to come up with a comparison between the protocols, as in Fig.1.

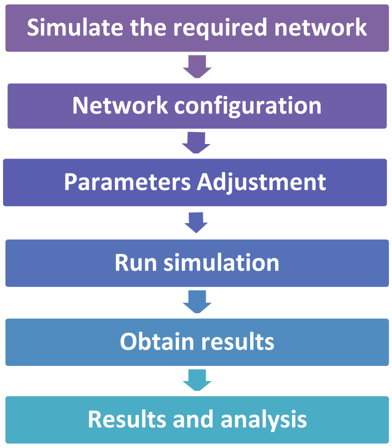

Fig. 1. Methodology steps

4 Simulation Scenario

In the simulation process, the experiment scenarios implemented using the simulator QualNet by following steps and as illustrated in Fig. 2.

- Firstly configure the general properties which apply to the whole Scenario.
- Specify the network topology by creating subnets, placing nodes, and define g nodes mobility
- Configure the protocol stack for individual nodes or groups of nodes as neces sary.
- Configure parameters for collecting simulation results and controlling runtim e performance.
- Analyzing the results.

The last step is to use the file editor to compare the performance of routing protocols based on the performance metrics selected.

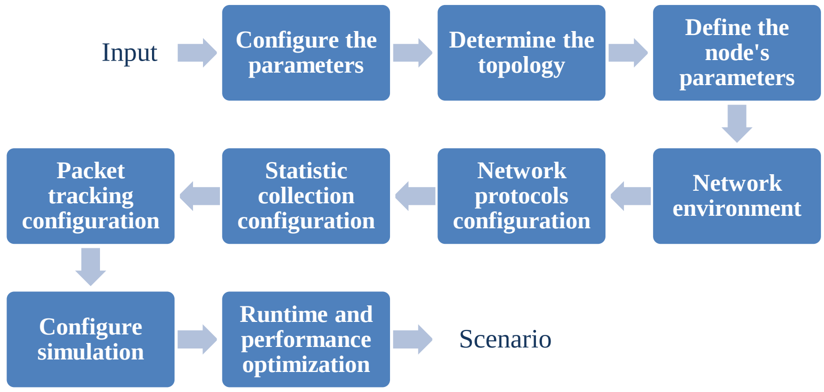

Fig. 2. Creating a scenario

Our Scenario in Fig. 3 and general parameters, were considered a WLAN network that has 10 nodes. Three of them are RWP mobility nodes. Node 10 is used to perform a generalized function for both of the VOIP networks (H.323 and SIP). Node 1, 2, 3, 4, 5, and node 6, 7, 8, 9 are connected to wireless network1 and wireless network2, respectively. Node 10 is locally connected to nodes 5 and 6.

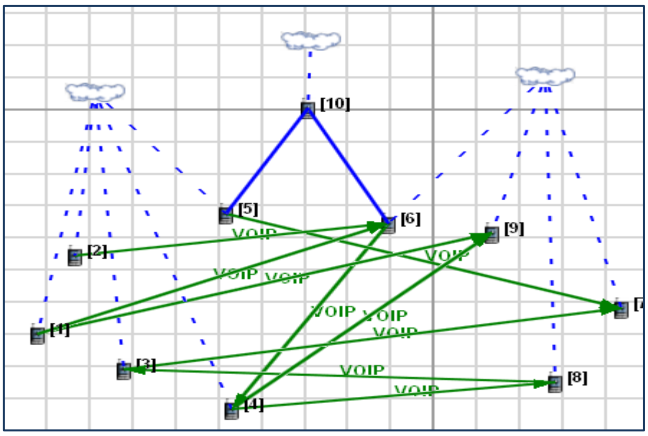

Fig. 3. Wireless ten nodes with VOIP application scenario

We have chosen the mobile nodes as hosts 1, 7, and 8. Host 7 and 8 have a slow movement in comparison to 1. Then we simulated the scenario in the QualNet simulator two times for different protocols (SIP and H.323), the scenario parameters are listed in Table 1.

Table 1. Scenario parameters.

Parameter	Specification
number of nodes	*10*
Simulation Area(Meters)	*1500 x 1500 m*
Mobility	*Nodes(1,7,8) Random, others with no mobility*
Application	*VOIP with SIP and H.323*
Codec	*G.711*
Simulation time	*for both protocols 256 s total 134 s*
Network type	*Two wireless networks*
Mobility type	*RWP*

5 Result and Analysis

Experiments for this study are to analyze SIP and H.323 performance. The purpose is to identify the best transmission protocol. From our experiments, As shown in Fig. 4 and 5. The collected result analyzed and used to compare the performance. The scenario was simulated in the Qualnet environment two times the total simulation time has taken for both protocols 256 seconds with RWP mobility type. The initiator session establishment time for the SIP application is 125 seconds, whereas H.323 has required 131 seconds to establish the session.

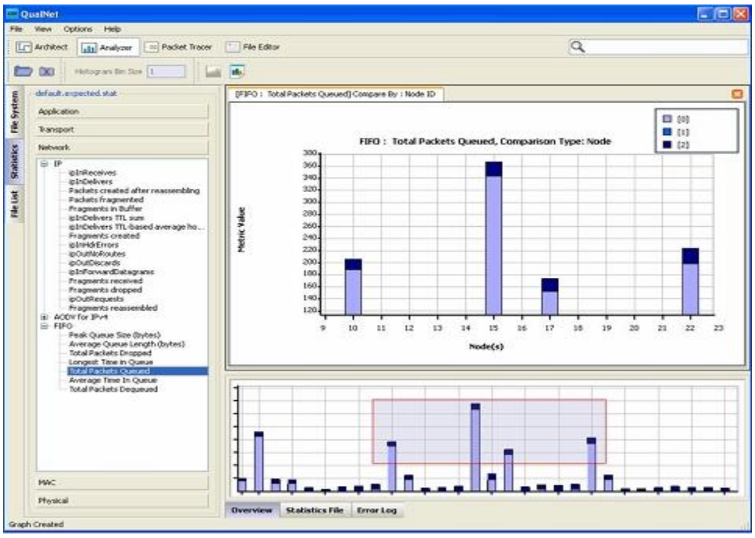

Fig. 4. QualNet analyzer

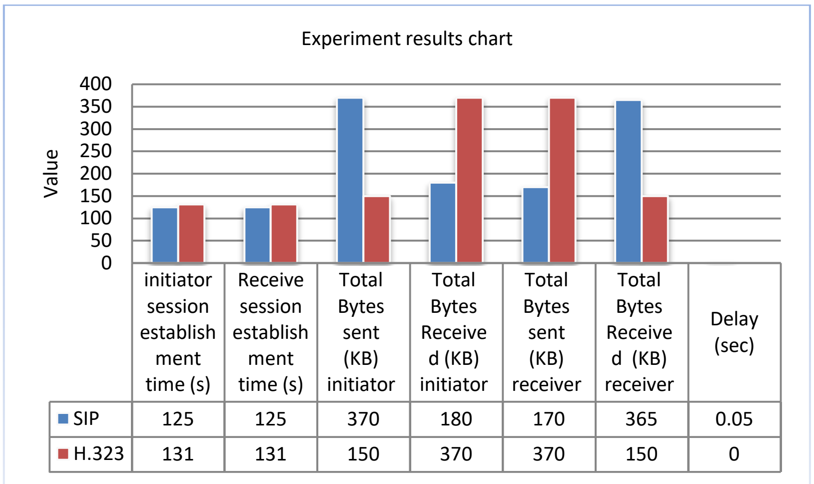

Fig. 5. Experiment results

The receiver session establishment time for SIP and H.323 application is the same as the initiator session establishment time. As we notice in the initiator process of SIP has sent more Bytes 60% compare to H.323 for the same period.
Correspondingly, the receiver packets were the total bytes sent increased by 53% in SIP. while dropdown around 59% in H.323.
This comparison is based on a G.711 codec. depicts the delay amount in SIP protocol was 0.05 sec. the following equation is used to calculate the average end-to-end delay, where

$$T_{E2E} = (T\_R - T\_S) \qquad (1)$$

But the delay period is not fixed In VOIP applications it is different for the different codec.
Finally, the overall throughput in Fig. 6, of the two signaling protocols concerning the duration of audio length. The result gives a good comparison between the protocols for the matter of throughput calculation. In SIP the maximum throughput was 6.182 Mbps at the duration of 123 ms and 6.122 Mbps at the same duration for SIP and H.323.While SIP shows absolute superiority all duration.

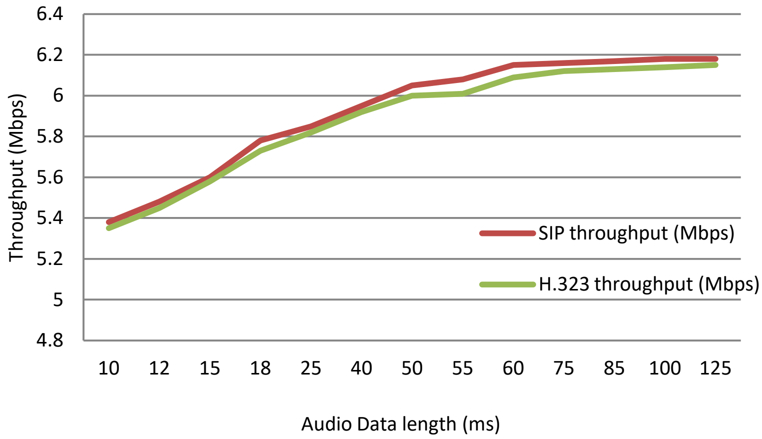

Fig. 6. The overall throughput

6 Conclusion

In this paper, the two common signaling protocols SIP and H.323 simulated by QualNet-based network to evaluated and analyzed both protocols performances in VoIP and 802.11a and wireless network environment, Based on the proposed simulation and overall SIP superiority in throughput especially in long period audio session, moreover the establishment time and session initiation are better in SIP, With an unnoticeable delay in the end to end delay result which would not impact the protocol performance. The future direction includes more different signaling protocols for comparison.

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