2013 8th EUROSIM Congress on Modelling and Simulation

SIP Signaling and QoS for ROHC Based Next Generation MANETs Reactive Routing Protocols

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***Abstract*— In this paper, an evaluation of SIP signaling and voice QoS for SIP based VoIP using GSM voice codec system over IPv6 MANETs with *Static*, and *Random* mobility models. This evaluation study considered two types of reactive routing protocols, AODV and DSR. The study examined IPv4, IPv6, and Robust Header Compression (ROHC) as a compression/decompression system for IPv6 headers. The evaluation results show that SIP signaling and VoIP traffic are acting poorly over IPv6 even when applying ROHC. In general, AODV has low performance over different types of *Random* mobility models for MANET nodes, while DSR shows better performance with *Static* mobility models and bad performance with *Random* mobility models. When using ROHC for TCP traffic, a level of enhanced performance had shown for SIP based VoIP calls over IPv6 MANET. However ROHC still has longer delays and poor performance compared with SIP based VoIP over IPv4 MANET. Therefore, the SIP signaling for IPv6 MANET reactive protocols require further enhancements for SIP/TCP registration and retransmission timers to be able to employ the capabilities of IPv6 and ROHC system for SIP based VoIP and real-time applications over MANET.**

Keywords- MANET; AOVD; DSR; SIP; ROHC; IPv6; QoS; VoIP; GSM Voice Codec; OPNET

I. INTRODUCTION

MANET is one of the most common wireless networks with dynamic distributions of mobile devices that move in different types of mobility systems [1]. MANET has different types of routing protocols, and each routing protocol has its own characteristics over different types of applications and mobility models. This study considered the Reactive routing protocols for MANET. It considered the Dynamic Source Routing (DSR), Ad hoc On Demand Distance Vector (AODV), and Temporally Ordered Routing Algorithm (TORA) routing protocols. On the other hand, voice over IP (VoIP) is one of the most common applications over different types of network systems. Different factors affect the VoIP QoS over MANET reactive routing protocols such as the mobility model, voice codec, physical distance between the call parties, hops number, node capacity, WLAN technology system, and calls durations. This study considered VoIP with GSM voice codecs. GSM is a common voice codecs which has a wide range of usages over different types of voice applications. It has a good voice quality and high performance over different types of network systems and with different types of voice applications. The main aim of the study is to examine the QoS and SIP signaling for VoIP applications over MANET reactive routing protocols with moderate node capacity, different types of mobility models, and different voice codecs. In this evaluation study, the IEEE 802.11n considered as the WLAN technology which represents the physical layer technology for the implemented MANET. In addition, we considered IPv4 as the addressing model in all MANET nodes. IPv4 is still widely used for different network systems. The simulation works are implemented using OPNET modeler. The SIP based VoIP applications consist of two types of signaling: SIP signaling (TCP based) and voice Signaling (UDP based). Both signaling systems have an impact on the VoIP QoS depends on the type of the applied network system, applications, user numbers, etc.

A. SIP Signalling:

SIP is a signaling protocol defined by SIP Working Group, within Internet Engineering Task Force (IETF). The protocol was published as IETF (RFC 2543) and currently has the status of a proposed standard [2]. SIP is commonly used for controlling the multimedia communication sessions such as voice and video calls over Internet Protocol (IP). The SIP session can include one or more Participants/Applications and can be used for creating, modifying, and terminating two or more participant sessions by consisting of one or more media streams. SIP is a text encoded protocol with a built in code which allows a different type of modifications and extensions. The modifications could be applied in addresses, ports, participant invitations, and adding/deleting media streams [2]. SIP is an application layer protocol designed to be independent of the existing transport layer, and it depends on the supported Internet protocols. Moreover, SIP can run on the top of the TCP/IP, UDP/IP, or Stream Control Transmission Protocol (SCTP) [3].

SIP based VoIP depends on three main stages: the registration stage, the Call initiation stage, and the call termination stage. These stages are depending on the SIP Proxy Server to relay the connectivity between different callers. The delays of the SIP signals in all stages affect the performance of VoIP calls. As SIP is a TCP based application layer signaling system, all the TCP timers

978-0-7695-5073-2/13 $26.00 © 2013 IEEE
DOI 10.1109/EUROSIM.2013.103

(retransmission, and Round Trip Timer (RTT) are important factors for overall structure of the SIP connectivity system.

B. *Voice Signalling:*

Voice traffic transferred over a special protocol system known as the Real Time Protocol (RTP) which is based on UDP. For SIP based VoIP applications, the call initiation stage which activates the media data transfer process directly from the Caller to the Callee using one of the supported voice codecs. This research study is based on GSM FR Quality Speech voice codec. GSM has its own characteristics and voice quality which affected by the number of hops, routing protocol, voice load, background noise, node movement, mobility models, bandwidth, traffic congestion and etc. The QoS parameters Voice applications are Jitter, End-To-End Delay, Amount of Traffic Received, and the amount of Traffic Sent/Received.

C. *MANET Overview:*

In telecommunication systems, MANET is as self-configured unlike nodes, creating infrastructure-less network, connected with different Wireless Networks as nodes which exchange data packets without a central control system [1]. There are three types of routing protocols in MANET [6]. The *Reactive routing protocols* which are On-demand protocols that discover the routes when needed. They considered as source-initiated route discovery protocols (e.g. DSR, AODV, and TORA). The *Proactive routing protocols* which are traditional distributed protocols that use the shortest paths based on periodic updates, however, they have high routing overhead (e.g. OLSR, and Destination Sequenced Distance Vector (DSDV)). Finally, *Hybrid routing protocols* which have combined functionality from both, reactive and proactive routing protocols with hybrid routing capabilities (e.g. Zone Routing Protocol (ZRP)).

D. *ROHC Overview:*

ROHC is a standardized compression scheme proposed by [4] to support both wired and wireless links with high error rate and long RTT. It used to provide an adopted suitable form of compressed data for IP-based communication systems such as 3GPP and 3GPP2 which act as the core technology for 3G cellular networks [5]. ROHC used to compresses the IP, UDP, RTP, and TCP packet headers to provide the best case with lower size of packet headers for IP-based communication. ROHC is first devised to improve voice and data transmission quality in one hop cellular network, it could have different characteristics when it is adopted into multi-hop wireless mesh backhaul network. ROHC designed to overcome the overhead for streaming application such as VoIP, where IPv4 normally require 40 Bytes compared with 60 Bytes for IPv6. The mechanism of ROHC is depend in placing the compression system before the link port of the source side with limited capacity and the decompression system after the link port of the destination side. The compression and the decompression systems of ROHC used to convert large amount of overhead traffic headers to fewer amount of traffic headers overheads for IPv6 network systems. ROHC performs better than other types of compression schemes for IP-based links with high rate of packet loss such as wireless network systems.

II. RELATED WORKS

Number of evaluation studies used to evaluate MANET reactive routing protocols from a different perspective. Most of the evaluation works considered Constant Bit Rate (CBR) or File Transfer Protocol (FTP) traffic with a different number of MANET nodes. A performance evaluation with OPNET Modeler 14.5 for AODV and DSR using GSM voice traffic, concluded that AODV has the lowest End-to-End delay and lower network load compared with DSR [6]. Furthermore, ADOV has higher average throughput and received traffic while DSR not scale well with large sized networks. The simulation results also showed that ADOV reactive routing protocol is the best suited for MANET, while DSR has very poor QoS in MANET with high node capacity for GSM voice applications. However, there wasn't any consideration for real time applications such as VoIP using different types of mobility model [7]. In [8], an evaluation of AODV (reactive) and OLSR (proactive) in OPNET Modeler with a variable number of MANET nodes concluded that the performance of the routing protocols are vary depending on the network type and the selection of the accurate routing protocols, that affects the applications efficiency. In [9], three random based mobility models implemented for MANET reactive routing protocols over different performance parameters. The results shows that AODV in *Random* Waypoint mobility model performs better than TORA and DSR in *Random* walk and *Random* Direction mobility model, and the study conclude that AODV can be used with intensive mobility models. In [10], a performance comparison of selected MANET routing protocols in a varying network sizes with increasing area and nodes size to investigate *Random* mobility and scalability of the routing process with high CBR traffic flow. AODV performed very consistently and established quick connection between nodes without delays while TORA had high End-to-End delay due to the formation of temporary loops within the network. AODV has the best efficiency with high traffic applications compared to OLSR and TORA.

For ROHC based MANET, very few testbed or field trial measurement efforts reported in the reviewed literature. An evaluation of the transmission of GSM encoded voice using ROHC over wireless link is proposed in [11]. The research used to introduce an evaluation methodology that combines an elementary objective of voice quality metrics using a novel frame with synchronization mechanism of vice transmission to provide an effective and accurate quality evaluation of voice packets. The research considered the impact of ROHC on the consumed bandwidth and the delay jitter in the voice signal regardless to the impact of ROHC on the voice quality. The research findings show that ROHC mostly reduced the bandwidth required for the transmission of GSM encoded voice to the half for a wide range of error probabilities on wireless links. Furthermore, ROHC improved the voice quality compared to the voice transmissions without ROHC. The method of this research focused on general wireless networks without referring to the mobility issues that affect real time applications performance.

A method system design for providing header compression guidance to mobile devices initiating certain applications and protocols [12]. It used a selective header compression (SHC) to utilize the SIP messages to provide ROHC guidance for mobile devices to initiate SIP based applications, protocols and service options. In addition, the ROHC guidance designed to allow mobile devices to provide an appropriate selections for the applications source and/or destination ports for data flows that requiring ROHC. The ROHC guidance used to recommend the mobile device request for ROHC based VoIP flows but not for video flows as the guidance help to specify the suitable range of RTP ports for the applicable ROHC based flows. This method only provide a selective way for ROHC over RTP/UDP flows depends on the performance features of the SIP based flows without any farther testbed or simulations efforts with comparable results.

In [13], an implementation of the header compression algorithm and protocol for TCP/IPv6 transport over wireless links with slow/medium speed. This implementation applied on the underway standardization work in the IETF ROHC working group. The simulation results showed that ROHC over TCP reduced the size of TCP/IPv6 headers with acceptable efficiency, while it is robust for the packet loss of wireless link. In addition, the research efforts indicate that the performance and the type of TCP affect the ROHC based wireless link.

An evaluation of the behavioral effect of ROHC and packet aggregation over multi-hop wireless mesh networks represented in [5]. The study mentioned that ROHC improved the IP based data flow with around 20% of the total wireless link transmissions. However, even if the number of mesh router increased the total number of the achieved rate will gradually decrease. On the other hand, the ROHC cooperation with packet aggregation provides 4 to 10 times of the achieved rate of improvements and results up to 6 times of reductions for the end-to-end delays. However, this cooperation has an effect on the processing time over the behavior of the wireless mesh networks. The research study used to evaluate and improve the ROHC processing time using NS-2 simulation models. The simulation results show an improvement in the general performance for ROHC and packet aggregation where the hardware design is required to speed up the processing units for ROHC based wireless mesh network systems. Accordingly, the study proposed and evaluated a hardware system model for ROHC and packet aggregation by using SystemC Hardware Description Language (HDL). Further simulation results show that the proposed built-in processor has improved the ROHC performance by enhancing the low speed of the processing power.

An analysis presentation of the primary functional blocks of ROHC with an extract for the architectural implications on the next generation network processor for wireless access represented in [14]. The study focused on the memory space, bandwidth, and processing resource budgets for the wireless access hardware. The study examined the resource consumption and the gains of the achievable potential performance using the offloading computationally intensive ROHC functions for the applications with specific hardware assists. In addition, this paper discussed the design trade-offs for hardware which assist in the form of a reconfigurable hardware for the ROHC design and functionality for the network processors and access infrastructure without investigating any real time applications.

A study for VoIP traffic over fixed WiMAX had introduced in [15] to evaluate ROHC and the application layer aggregation of voice for VoIP performance in a fixed WiMAX testbed with one base station and two subscriber stations. The result shows that ROHC increases the number of simultaneous bidirectional emulated VoIP flows by 6% compared to plain VoIP, while the aggregation and ROHC allows 86% more flows than standard VoIP to be sustained. The evaluation steps of this research effort could be applied for ROHC over MANET to improve the general performance of SIP-based VoIP applications. In [16], a performance evaluation for VoIP over Wireless MAN-OFDMA air interface of a state of the art mobile WiMAX testbed operating at the 3.5 GHz frequency band and quantify the benefits of employing VoIP aggregation and ROHC. In addition, the study proposed VoIP aggregation with ROHC and evaluated using simulation and modelling. The study concluded that the combined use of VoIP aggregation and ROHC over WiMAX can increase the number of effective data flows without loss by approximately three times the regular VoIP transmission. Furthermore, another study used to investigate ROHC over the Universal Mobile Telecommunications System (UMTS) to provide an optimal combination of Radio Access Bearers (RAB) for VoIMS (VoIP with IP Multimedia Subsystem in the Core Network) to provide an efficient use of ROHC to improve the QoS of the physical layer [17]. The main improvement of this RAB combination is with the adaptions of the throughput for SIP signaling to improve the call setup delays for the transmission of the packet switch by using a Transport Format selection algorithm to combine flexible rate of data flows to match the physical layer. The researches efforts in [15], [16], and [17] could inspire the research efforts of the evaluation and the improvements of VoIP over IPv6 MANET using ROHC.

III. Simulation Parameters and Assumptions

In this study, we considered IEEE 802.11n as the wireless network standard for MANET models because of its new features and good mobility support for MANET compared with 801.11a, b and g [18]. The evaluation efforts implemented in OPNET Modeler 17.1. The simulation works applied over four types of mobility models to study the performance with different mobility aspect and related positions. The mobility models are: *Static*, *Random Waypoint*, *Random Waypoint All* and *Random Waypoint All with Trajectory SIP Server*. Table I represents a brief summary of the simulation parameters. These parameters identified depending on the features and the capabilities of MANET and VoIP applications compared with other evaluation studies as in [6], [7], [9], and [18].

TABLE I. SIMULATION PARAMETERS IN OPNET

<table>
<tr><th colspan="5">A. MANET</th></tr>
<tr><td>Number of Simulations</td><td>20</td><td colspan="2">Simulation Seed Number</td><td>128</td></tr>
<tr><td>Simulation Duration:</td><td colspan="4">10 Minutes = 600 Seconds</td></tr>
<tr><td>Mobility Models:</td><td colspan="4">Static, Random Waypoint, Random Waypoint All, and Random Waypoint All with Trajectory SIP Server</td></tr>
<tr><td colspan="3">MANET Reactive Routing Protocols</td><td colspan="2">AODV, DSR</td></tr>
<tr><td>Number of nodes:</td><td>25 nodes</td><td colspan="2">Area Dimension:</td><td>1 km x 1 km</td></tr>
<tr><td>Node Speed Range:</td><td colspan="4">Uniform Speed between 1 m/s (3.6 km/hr) and 10 m/s (18 km/hr)</td></tr>
<tr><td>WLAN Physical Characteristic:</td><td>802.11 n</td><td colspan="2">Data Rate:</td><td>13 Mbps</td></tr>
<tr><td colspan="2">Transmission Range between Nodes:</td><td colspan="3">up to 250 meters</td></tr>
<tr><td>Frequency Band:</td><td>2.4 GHz</td><td colspan="2">Transmission Power:</td><td>0.001 W</td></tr>
<tr><td>Packet Size:</td><td>512 Bytes</td><td colspan="2">Buffer Size:</td><td>32 Kbytes</td></tr>
<tr><th colspan="5">B. Applications: SIP Based VoIP</th></tr>
<tr><td colspan="2">SIP Server Connect Timeout:</td><td>TCP Based</td><td>Voice Codec:</td><td>GSM 13 Kbps</td></tr>
<tr><td rowspan="5">VoIP Calls (Unlimited)</td><td>Durations</td><td>Caller</td><td>Callee</td><td>Number of Intended Calls in 600 Seconds</td></tr>
<tr><td>5 Sec</td><td>Node 1</td><td>Node 24</td><td>115</td></tr>
<tr><td>10 Sec</td><td>Node 22</td><td>Node 3</td><td>60</td></tr>
<tr><td>20 Sec</td><td>Node 5</td><td>Node 20</td><td>30</td></tr>
<tr><td>40 Sec</td><td>Node 15</td><td>Node19</td><td>15</td></tr>
<tr><td rowspan="2">Maximum Simultaneous Calls</td><td>SIP Server</td><td colspan="2">User Agent (Caller/Callee)</td><td>Total VoIP Calls in 600 Seconds</td></tr>
<tr><td>Unlimited Call/ Second</td><td colspan="2">1 Call/ Second</td><td>220 Calls</td></tr>
</table>

In the *Static* model, MANET's nodes are stable and not moving. In *Random Waypoint* model, nodes are moving in different directions, but the SIP server is stable with no movement in the center of the simulation area. Fig. 1 shows the simulation implementation in OPNET for a MANET with *Random Waypoint* mobility model after 20 seconds from the beginning of the simulation where each node, except the SIP Server which is not moving. In *Random Waypoint All*, nodes are moving in different directions including SIP server depending on the identified *Random* functionality of the node parameters. Finally, In *Random Waypoint All with Trajectory SIP Server model*, nodes are moving in different *Random* directions where the SIP server is in a special trajectory identified for a specific times and locations that SIP Server move through during the simulation progress. The reason for examining the *Random* mobility using three different models is to study and evaluate the effect of the SIP server mobility over VoIP applications and the signaling QoS. All the assumptions and simulation works of this evaluation study are based on IPv4/IPv6 MANET for VoIP applications using GSM voice codec.

IV. SIMULATION RESULTS AND DISCUSSION

The simulation results show two main aspects for VoIP applications; the SIP signaling system between the call's parties, and the voice QoS. Each evaluation aspect is considered over the defined mobility models as explained and justified in the previous section. The X axis in all results represents the simulation time in Seconds (S) which shows a comparable graphical statistics between different mobility models and the voice codecs over the simulation time. The Y axis represents the call's duration in Seconds (S), the Number of Calls (N), or the Number of Bytes (B). The performance of most of the VoIP applications is low at the beginning of the simulation which make the results difficult to be discussed and analyzed. However, after 150 to 200 seconds from the beginning of the simulation, the results show comparable statistics for VoIP related statistics as VoIP statistics become stable. These delays happen in the early stages of the simulation which affect VoIP applications because of the initiations of the routing tables for MANETs depending on the protocol type. In addition, the delays of the nodes registration process in the SIP server adds more delays to VoIP applications at the beginning of the simulation.

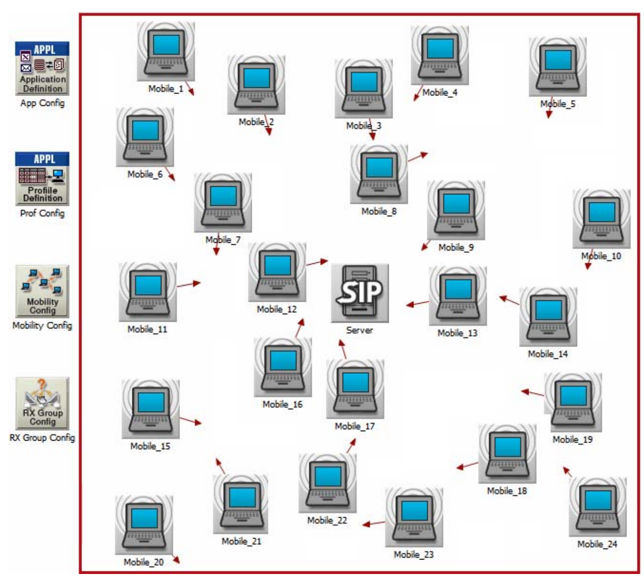

Figure 1. MANET *Random* Mobility Model at the simulation time of 20 Seconds. Mobility directions represented in red arrows.

The simulation works in this part had run for both AODV and DSR MANET over five types of IP Headers for SIP based VoIP applications which are: IPv4, IPv6, IPv6 using ROHC for All headers, IPv6 using ROHC for RTP UDP/IP headers only, and IPv6 using ROHC for TCP/IP headers only. The simulation results used to compare each type of the SIP based VoIP application over different mobility models to study the general signaling performance and VoIP QoS.

A. *Results Evaluation for AODV Routing Protocol:*

VoIP calls over AODV show variable percentage of successful and rejected calls as in Table II. All mobility models with AODV MANET have number of delayed and rejected calls. The best performance is with *Static* mobility model where no mobility related issues occurred in this model. IPv4 has the highest percentage of successful VoIP calls where VoIP calls with ROHC TCP/IP Compression have good results over IPv6 while the ROHC RTP headers has the worst performance. In general, with different mobility models for MANET, the performance of SIP signaling and VoIP QoS affected with more than half of the performance of *Static* models. This is because of the nature of route calculations in AODV regarding to the mobility parameters. The mobility of the SIP server makes the performance VoIP applications worst in *Random* mobility models with considerable low efficiency and bad QoS for IPv6. Furthermore, the location of the SIP server affects the performance of the real-time applications where its central position provides good connectivity for all nodes as the hop numbers between different nodes remain in the best effort conditions. However, the *Random* mobility of the SIP server improved the general performance of VoIP applications.

TABLE II. VOIP SIMULATIONS STATISTICS FOR AODV MANET

Mobility Model	IP Headers Type	Total of successful VoIP Calls out of 220 calls	% of Successful VoIP calls	% of Rejected VoIP calls
Static	IPv4	193	87.7 %	12.3 %
	IPv6	176	81.4 %	18.6 %
	ROHC All Headers	67	30.5 %	69.5 %
	ROHC RTP UDP/IP	32	14.5 %	85.5 %
	ROHC TCP/IP	179	86.4 %	13.6 %
Random	IPv4	162	73.6 %	26.4 %
	IPv6	95	43.2 %	56.8 %
	ROHC All Headers	61	27.7 %	72.3 %
	ROHC RTP UDP/IP	21	9.9 %	90.1 %
	ROHC TCP/IP	121	55.4 %	44.6 %
Random All	IPv4	167	75.9 %	24.1 %
	IPv6	107	51.4 %	48.6 %
	ROHC All Headers	75	34.1 %	65.9 %
	ROHC RTP UDP/IP	29	13.2 %	86.1 %
	ROHC TCP/IP	125	45.8 %	43.2 %
Random All with SIP Server Trajectory	IPv4	141	64.1 %	35.9 %
	IPv6	70	31.8 %	68.2 %
	ROHC All Headers	59	26.8 %	73.2 %
	ROHC RTP UDP/IP	54	24.5 %	75.5 %
	ROHC TCP/IP	86	39.1 %	60.9 %

A.1 *SIP Signalling Evaluation:*

Fig. 2 shows the average calls setup time for VoIP applications over different mobility models using AODV MANET routing protocol. The setup time for *Static* is ideal with 0.2 second for IPv4 where it is too long for IPv6 with 0.7 second. For different mobility models ROHC has no enhancement for the calls setup times while all IPv6 applications shown a considerable call setup time. ROHC for TCP/IP headers has the best performance over other types of IPv6 ROHC. In *Random All* mobility models, ROHC over RTP and *All Headers* act poorly as SIP signaling for VoIP application interrupted until the end of the simulation time. The performance of IPv6 is very low compared with IPv4 and the ROHC header compressions couldn't provide the effective performance enhancements in IPv6 call's setup time for AODV MANET. The uncompleted representation for VoIP calls mean that the VoIP call sequence has stopped and the SIP connectivity system in the network had been crashed due to undelivered SIP registration or termination messages.

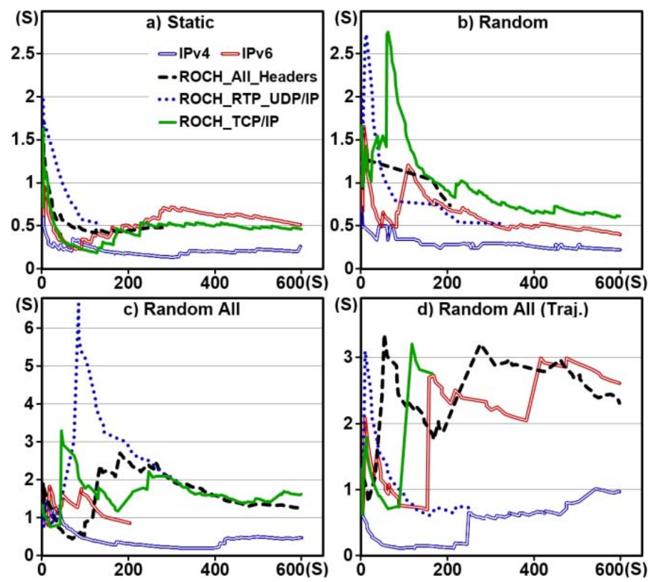

Figure 2. Average SIP Call Setup Time in Seconds for AODV MANET.

Fig. 3 shows the average number of the rejected SIP based VoIP calls over different types of IP traffic headers. The number of the rejected calls is too high for ROHC over RTP and All headers compared with IPv6 and ROHC TCP in *Static* model, while in *Random* models there is a higher number of rejected calls affect the general performance of the VoIP call sessions. On the other hand, IPv4 has the lowest number of rejected SIP calls in all mobility models while in *Random* mobility models, IPv6 has double number of rejected calls because of the unreachability and the connectionless for IPv6 data flows. The mobility of SIP server has a direct influence on the SIP signaling system which affects the SIP timers for the retransmission messages for each rejected call. In general, ROHC TCP has a considerable improvements over IPv6 traffic as the overhead of SIP/TCP traffic is lightly reduced over the *Random* mobility models.

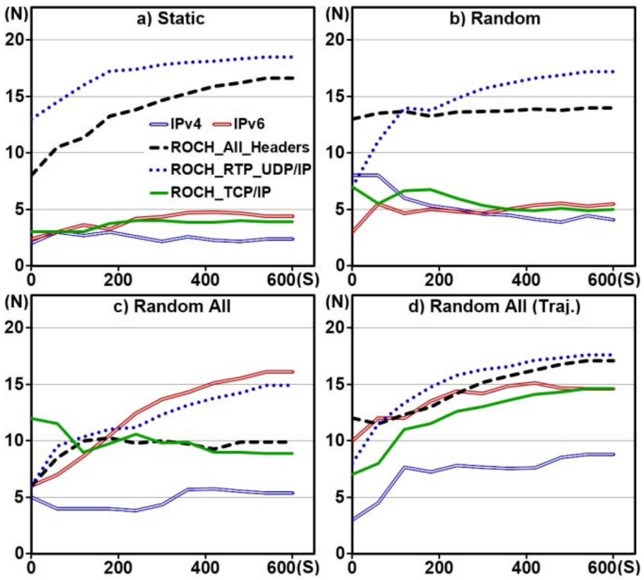

Figure 3. Average Number of Rejected SIP Calls for AODV MANET.

A.2 SIP Server Efficiency:

The SIP server used to initiate, terminate and modify the SIP calls. The more number of VoIP calls that SIP server can support, the better performance it can provides. Fig. 4 shows the average number of active VoIP calls per second which managed by the SIP server. The AODV MANET has a converged number of calls/second over different mobility models. For IPv4, IPv6, and ROHC TCP, the SIP server has a similar number of SIP requests for *Static* and *Random* mobility models while it has lower number of SIP calls request in *Random* mobility models. The ROHC TCP signaling is more efficient with AODV as it is able to interact with the SIP server over different mobility models even with the mobility of the SIP server.

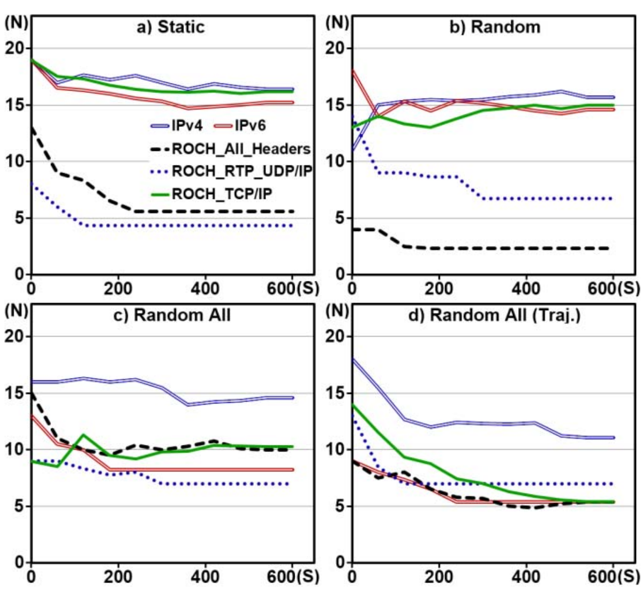

Figure 4. Average Number of Active Calls managed by the SIP Server.

On the other hand, ROHC RTP and All Headers have low number of SIP initiated calls over different mobility models. This mean that the server is not reachable, or not responding for the SIP initiation or termination requests. Furthermore, the SIP server stops receiving any SIP requests after while in *Random* mobility models and this comes worst with the mobility of the SIP server.

A.3 Voice Signalling QoS:

Fig. 5 shows the average voice traffic sent from the Caller to the Callee in Bytes. The amount of received traffic is lower than the amount of sent traffic because of the traffic drops between both sides. On the other hand, Fig. 6 shows the average voice traffic which received from by the Callee in Bytes.

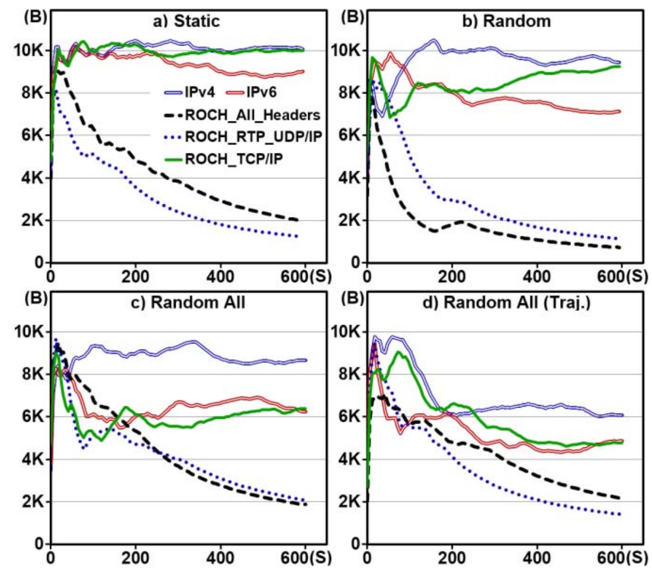

Figure 5. Average Voice Traffic Sent for AODV MANET.

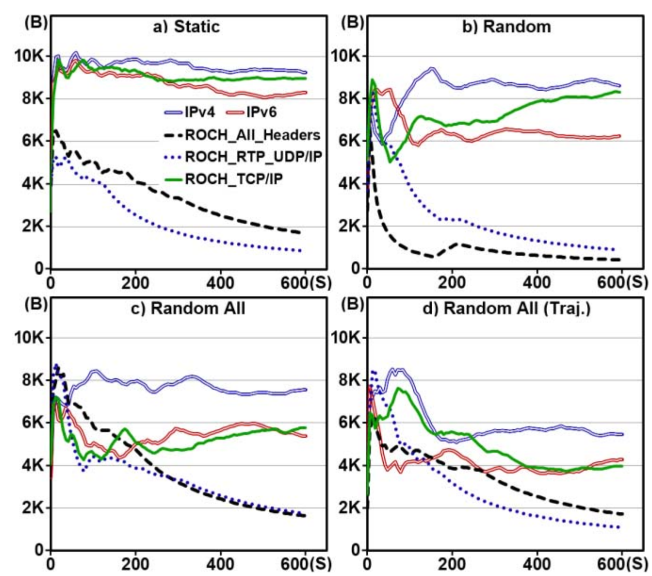

Figure 6. Average Voice Traffic Received for AODV MANET.

In *Random* models the amounts of transferred voice data is high at the beginning of the simulation, but it falls down later. This is because of the initial location and the reachability of the MANET nodes at the beginning of the simulation time. The comparison between both the sent and the received amount of traffic show that ROHC TCP for IPv6 is the best available simple support for IPv6 overheads compared with the performance of IPv4. In addition, the mobility of SIP server with *Random* All model shows better performance than the mobility model with Trajectory. Both the amount of Traffic sent and the traffic received are too low compared with other types of IP Headers. This is happen because of the high number of compression and decompression processes for UDP packets that effect the overall performance of the SIP based applications in AODV MANET.

B. Results Evaluation for DSR Routing Protocol:

For VoIP calls over DSR based MANET, a variable percentage of successful and rejected calls are shown in Table III. The best performance is for *Static* mobility model where no mobility related issues occurred in this model. VoIP calls with IPv4 has the highest percentage of successful VoIP calls where VoIP calls with ROHC TCP/IP compression system enhanced IPv6 SIP system whereas ROHC RTP headers has low performance. DSR has critical performance over IPv6 MANET SIP signaling and VoIP QoS which equal to the third of the total performance for *Static* models as a result of the routes calculations nature. The mobility of the SIP server makes the performance for VoIP applications worst in *Random* mobility models with a considerable low efficiency and bad QoS for IPv6.

TABLE III. VOIP SIMULATIONS STATISTICS FOR DSR MANET

Mobility Model	IP Headers Type	Total of successful VoIP Calls out of 220 calls	% of Successful VoIP calls	% of Rejected VoIP calls
Static	IPv4	206	93.6 %	6.4 %
	IPv6	172	78.2 %	21.8 %
	ROHC All Headers	164	74.5 %	25.5 %
	ROHC RTP UDP/IP	159	72.3 %	27.7 %
	ROHC TCP/IP	180	81.8 %	18.2 %
Random	IPv4	108	49.1 %	50.9 %
	IPv6	61	27.7 %	72.3 %
	ROHC All Headers	57	25.9 %	74.1 %
	ROHC RTP UDP/IP	48	21.8 %	78.2 %
	ROHC TCP/IP	75	34.1 %	65.9 %
Random All	IPv4	121	54.9 %	45.1 %
	IPv6	87	39.5 %	60.5 %
	ROHC All Headers	74	33.6 %	66.4 %
	ROHC RTP UDP/IP	64	29.1 %	70.9 %
	ROHC TCP/IP	93	42.3 %	57.7 %
Random All with SIP Server Trajectory	IPv4	114	51.8 %	48.2 %
	IPv6	53	24.1 %	75.9 %
	ROHC All Headers	51	23.8 %	76.2 %
	ROHC RTP UDP/IP	46	20.9 %	79.1 %
	ROHC TCP/IP	72	32.7 %	67.3 %

B.1 SIP Signalling Evaluation:

Fig. 7 shows the average calls setup time for VoIP applications over different mobility models using DSR MANET routing protocol. The setup time for *Static* is ideal with 0.1 second for IPv4 where it is too long for IPv6 with about 0.8 second. In mobility models for MANET, ROHC has no enhancement for the calls setup times, although all IPv6 applications shown a significant long call setup time. Using ROHC for TCP/IP headers lightly enhanced the performance for IPv6 applications. In *Random All* mobility models, ROHC over RTP and *All Headers* has bad performance as the SIP signaling for VoIP application interrupted until the end of the simulation time. In general, the performance of IPv6 is very low compared with IPv4 and the ROHC header compressions couldn't enhance the performance of IPv6 call's setup time for DSR MANET. Some of the VoIP calls represented with uncompleted calls sequence which mean that VoIP call sequence had stopped and the SIP connectivity system in the network had crashed because of the undelivered SIP registration/termination messages.

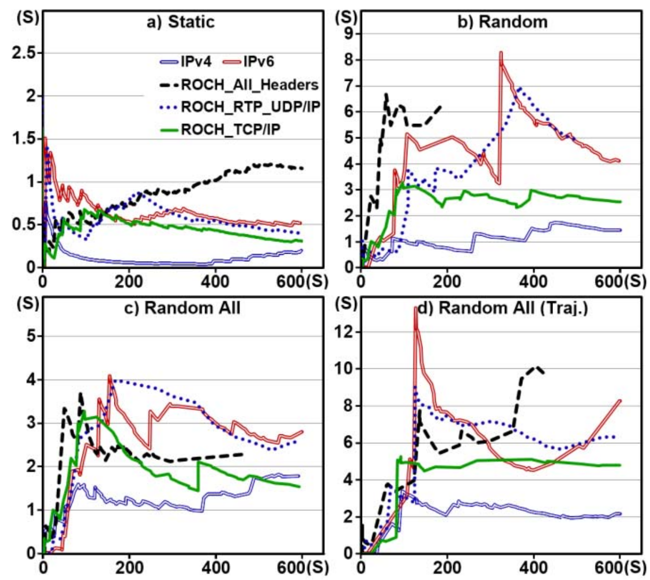

Figure 7. Average SIP Call Setup Time in Seconds for DSR MANET.

Fig. 8 shows the average number of the rejected SIP based VoIP calls over different types of IP traffic headers. The number of the rejected calls is too high for ROHC over RTP and All Headers compared with IPv6 and ROHC TCP in *Static* model, while the *Random* models have higher number of rejected calls affect the general performance of the calls sessions. IPv4 has low number of rejected SIP calls with all mobility models while in *Random* mobility models, IPv6 mostly has the double number of rejected calls. The mobility of SIP server has an impact on the SIP signaling system which affect the retransmission timers for SIP messages during the session initiation and termination processes. Furthermore, ROHC TCP signaling system also has a considerable support for IPv6 traffic as the overhead of SIP/TCP traffic reduced over the *Random* mobility models.

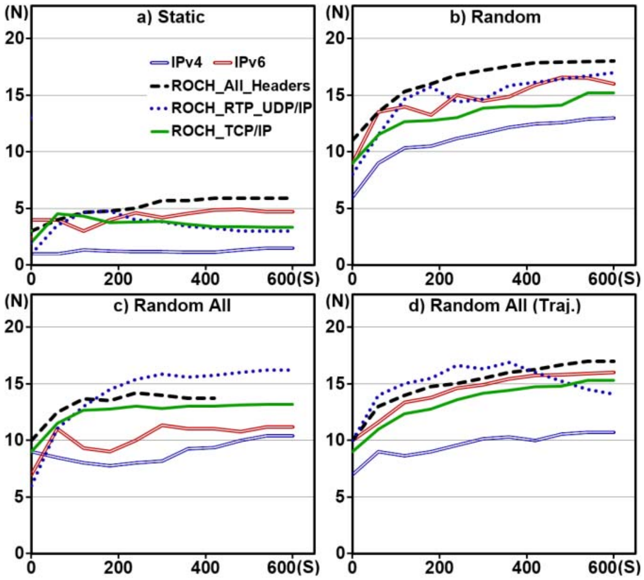

Figure 8. Average Number of Rejected SIP Calls for DSR MANET.

B.2 SIP Server Efficiency:

Fig. 9 shows the average number of active VoIP calls per second that managed by the SIP server. The IPv4 VoIP applications have the highest number of served call by the SIP server while the IPv6 has lower number of served calls. The ROHC TCP and the ROHC UDP have good performance over different types of mobility models.

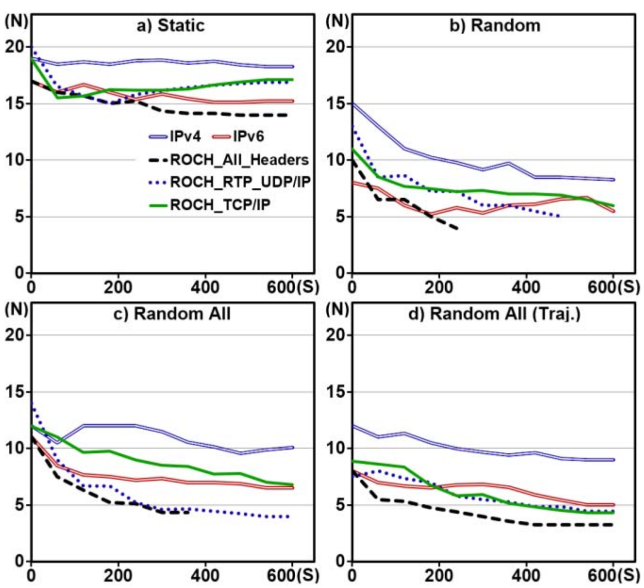

Figure 9. Average Number of Active Calls managed by the SIP Server.

The ROHC TCP and ROHC UDP signaling system are slightly efficient with DSR as the SIP server offered good number of connecting/registering call over different types of mobility models even with the mobility of the SIP server. On the other hand, ROHC over All Headers have lowest number of SIP initiated calls over all mobility models which indicate that the server is not reachable, or not responding for the SIP initiation or termination requests. Moreover, the SIP server used to stop the receiving of any SIP requests after while in *Random* mobility models and this comes complicated with the mobility of the SIP server.

B.3 Voice Signalling QoS:

Fig. 10 shows the average voice traffic sent from the Caller to the Callee in Bytes for DSR MANET. On the other hand, Fig. 11 shows the average voice traffic which received by the Callee in Bytes. The comparison between the two figures show that the amount of the received traffic for several VoIP call sessions is lower than the amount of the sent traffic because of the traffic drops between the Caller and the Callee.

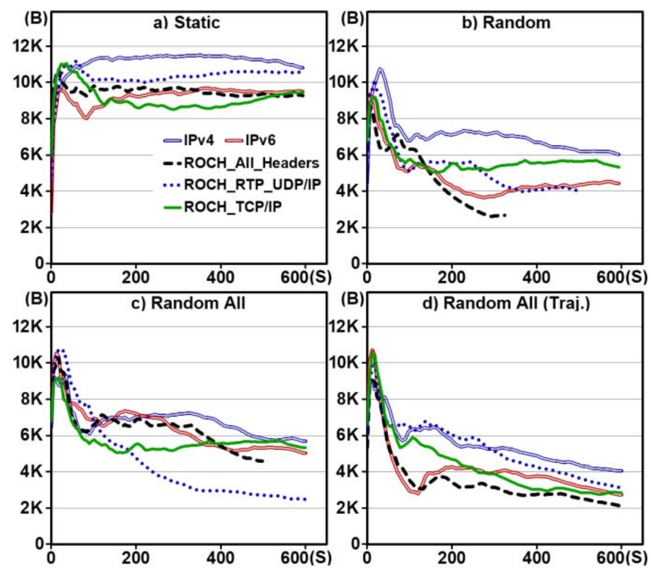

Figure 10. Average Voice Traffic Sent for DSR MANET.

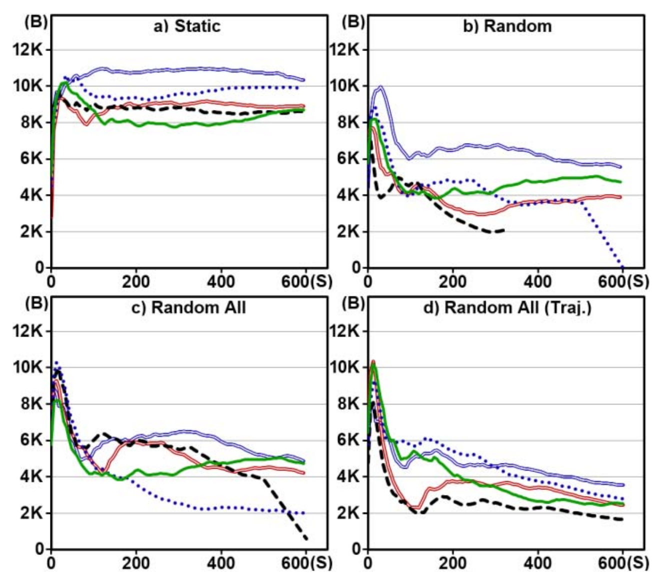

Figure 11. Average Voice Traffic Received for DSR MANET.

For *Random* Mobility models, the amount of transferred VoIP data is high at the beginning of the simulation and it reduced later because of the initial movement of the MANET nodes and the reachability for the SIP reiteration signals at the beginning of the simulation. The ROHC TCP for IPv6 shown acceptable performance for IPv6 overheads compared with other type of ROHC header compression system. The mobility of SIP server with *Random All* mobility model has better performance than the mobility model with SIP server Trajectory movement. DSR in general has a very low VoIP traffic for both, the sent and the received, compared with AODV because of the large amount of routing overhead for DSR.

V. CONCLUSION AND FUTURE WORKS

The evaluation results shown that IPv6 has very long delays with bad performance for SIP based VoIP over both AODV and DSR MANET. In the other hand, as ROHC has recently proposed to reduce the large amount of Internet protocol header overhead when transmitting voice and continuous media over IPv6 wireless networks, however the performance and QoS are still not efficient. The simulation results shown that ROHC for TCP/IP headers has lightly enhanced the signaling performance and VoIP QoS while ROHC over RTP/UDP Headers and ROHC over All Headers for SIP based traffic flows have weak performance for MANET's SIP based applications. All SIP performance factors could affect the SIP based applications over IPv6 MANET. One of these factors is the retransmission timers of SIP signaling that control the calls initiation, termination, and modification processes for VoIP applications. The delays in SIP/TCP messages affect the calls cancelations, terminations, and modifications of the SIP sessions. Furthermore, the nodes capacity, position, and the mobility of the SIP server are all affect the performance of VoIP calls over MANET reactive routing protocols. In addition, the modifications on TCP timers and/or applying another TCP version will enhance the general performance of SIP signaling for IPv6 MANET systems. The retransmission timers for SIP signaling over IPv6 MANET is still an open research issue. The future works will focus on studying and improving the SIP registration and retransmission timers over Next Generation MANET to enhance the SIP signaling and QoS for SIP based applications over MANET reactive routing protocols.

ACKNOWLEDGMENT

The use of OPNET Modeler® version 17.1 in this research was facilitated through OPNET's University Program.

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