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Dynamic Novel Cross-Layer Performance Enhancement Approach for SIP over OLSR

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ABSTRACT The SIP signaling performance has a vital role for the overall QoS of SIP-based VoIP applications over MANET. The SIP end-to-end performance metrics have been defined in RFC 6076 to provide a standardized method for the performance evaluation of the SIP signaling system over different platforms. However, to our best acknowledge, the benchmarked values for these metrics have not been proposed yet. Therefore, in this paper, a novel Cross-Layer performance enhancement approach is proposed, implemented, and evaluated to improve the performance of the SIP signaling system over OLSR-based MANET by applying significant dynamic modifications for the routing parameters. The SIP performance metrics seek to accurately reflect the SIP signaling state and the required actions for the routing parameters. The implementation of the Cross-Layer OLSR approach has been successful effectively in reducing the total delays in the SIP processes, enhancing the signaling performance, and increasing the utilization level in the system bandwidth and routing processes.

INDEX TERMS SIP, performance metrics, B2BUA, OLSR, network.

I. INTRODUCTION

The Session Initiation Protocol (SIP) signaling system is used to control and manage the implementation of the VoIP/multimedia applications. Implementing the SIP signaling over wireless Mobile Adhoc NETwork (MANET) has many challenges over all SIP call stages. Several different factors affect the Quality of Service (QoS) of SIP-based VoIP over MANET's routing protocols [1]–[3], such as the mobility model, voice codec, physical distance between calling parties, number of hops, node capacity, Wireless Local Area Network (WLAN) technology, the behavior of the transportation protocol, and call duration. When the route signaling is lost, expired, or delayed, the SIP-based applications will be affected, and consequently the VoIP performance will be degraded. On the other hand, the implementation of SIP-based VoIP depends on three main processes: caller's registration, call initiation, and call termination. These processes depend on the SIP server to relay the connectivity between different callers. The delays in SIP signaling of all processes affect the performance of the VoIP/multimedia calls [4]. In addition, the SIP signaling delays are influenced by the connectivity status between the call parties, which are the caller, the callee, and the SIP server. For the implementation of a SIP server with a Back-to-Back User Agent (B2BUA), the main challenging issue is the fact that SIP generally relies on a centralized architecture between the callers and the SIP server. Therefore, the SIP signaling performance has an important role in the overall QoS for different network systems.

In order to implement SIP-based VoIP applications over MANET efficiently, researchers need to exert more efforts for the sake of improving performance. The importance of this implementation relates to its role as a substitute communication scheme in disasters, emergency recovery system, military operations, and collaborative applications. However, the mobility of MANET nodes and related routing issues increase the required time for the registration, call setup and call termination processes that is reflected on the general performance of the SIP-based VoIP/multimedia applications. Moreover, the centralized nature of the SIP signaling system is not suitable for the MANET nature because of the nodes' mobility, routing mechanism, and signaling issues such as noise, interference, and fading. Therefore, when the route signal is lost or delayed, the SIP signaling system is effected directly and the performance of VoIP applications will be lowered. Therefore, to enhance the SIP signaling performance in MANET, the routing parameters are need to be adjusted dynamically during the SIP processes based on a determined level for the performance enhancement metrics to support the SIP signaling system. This dynamic adoption for the routing parameter during the SIP processes is being reflected positively on the SIP signaling efficiency, and consequently on the performance of the VoIP applications. As a result, the SIP protocol can be a key component for MANET. This component will further support the usability and capabilities of MANET regarding its implementation. As for **motivations behind doing this work.** We are overwhelmingly motivated by the dynamic nature of MANET nodes and related routing issues which predominantly increase the delays in the SIP processes (SIP signaling performance), thus affect the VoIP QoS. Hence, there are no available benchmarked values for the performance metrics of the SIP signaling system. This motivates us to further investigate and develop the performance of SIP Signaling System over MANET.

In brief, the Contributions of this paper are as follows; (1) presenting an intensively done research and conducting a comprehensively made evaluation of the SIP signaling system for SIP-based VoIP applications over MANET, (2) benchmarking the related SIP performance metrics of the RFC 6076 for VoIP applications over OLSR protocol for the SIP processes over MANET in terms of the SIP signaling performance, and system efficiency, (3) proposing a cross layer approach for OLSR routing algorithm to enhance the performance for multimedia applications over MANET, and (4) providing an evaluation study for the current state of art for OLSR routing protocol usage for multimedia applications over MANET. Most importantly, the paper is an implementation of the VoIP and used to work with finding the ordinary level of the active timers that are the requirements to achieve the best implementations.

The rest of this paper is organized as follows; section II discusses the closely related works concisely. In section III, research scenarios and implementation are elegantly drawn. Research methodology is meticulously presented in section IV. Section V presents both the simulation parameters and the basic assumptions of this work. Section VI illustrates the design and implementation of proposed approach. Sections VII and VIII provides the analysis methods for the proposed work of tis paper. Results and discussion are extensively given in sections IX. Finally, section X draws conclusions and future work directions.

II. RELATED WORK

A significant amount of researches investigated the factors that lead to improve the performance of interior gateway protocol (IGP) in the Internet. [5] Open Shortest Path First (OSPF), is one of those popular protocols, but it is a link state routing protocol that has not yet considered to address MANET challenges. However, a few amount of researches studied and evaluated the performance of real-time applications over MANET. Most of the evaluation efforts considered Constant Bit Rate (CBR) or File Transfer Protocol (FTP) traffic with a different number of MANET nodes [6]. Reference [7] proposed the obstacles that emerged from using standard SIP across MANET. This is because MANET is a decentralized group of autonomous self-configured nodes but a SIP infrastructure requires centralized registrar servers and proxies. On the other hands, it lacks the provision of real time solutions to improve the performance of SIP centralized proxies/registrars located in unstable MANET nodes. OLSR is a proactive routing protocol that uses the basic functionality of link state routing and distance vector. Being a proactive routing protocol, OLSR provides the immediate routes available to the destination. In this protocol, each node maintains topology information and updates it periodically using link state messages [8]. Previous studies [9]–[11] investigated different types of reactive and proactive MANET routing protocols. They provided a comprehensive analysis of their performance as well. OLSR provides certain optimization mechanisms to improve its functionality, which are Multipoint Relay Selectors (MRS), Multipoint Relay (MPR) and Control Messages. The Multipoint Relay was used to minimize the control overhead and the number of rebroadcasting nodes. Each node's one-hop and two-hop neighbors are found using periodic broadcasting of HELLO messages. Then each node selects its one hop neighbor to be an MPR, so that all two-hop neighbors are reachable from at least one member of an MPR set. Only for nodes that are MPRs, at least, one node rebroadcast the packet for the selection of an MPR in the entire network, whereas the nodes that are not part of an MPR set receive and process each control message but not to rebroadcast it. On the basis of one-hop and two-hop information, each node calculates its route to destination and saves it into its own routing table. Thus the OLSR protocol is designed for a complete distributive nature platform, and does not require any kind of reliable transmission for its control messages, like HELLO messages and similar. The protocol used a sequence number in its control message in order to deliver the messages. The protocol is designed to perform a hop-by-hop routing, which guarantees the frequent delivery for packet to the destination node [12], [13].

On the other extreme, the signaling performance of SIP has a great influence over the overall Quality of Service (QoS) and Quality of Experience (QoE) in next generation networks. The main performance matrices of SIP based VOIP applications was performed in [14]. Although it concluded that IPv4 MANET acts better than IPv6 due to TCP signaling and header overhead, an enhancement to the SIP performance for different routing algorithms had not been proposed. The Optimized Link State Routing (OLSR) was determined by the IETF MANET working group as the RFC 3626 [15]. SIP signaling delays were relating to the connectivity status between the SIP call parties, which are the Caller, Callee, and SIP proxies. These delays mainly happen during the Caller's registration process, call initiation, call termination, and/or call management [16]. In addition, SIP signaling is affected by the behavior of the Transportation Protocol (TCP or UDP) [17], which SIP relies on during the different connectivity processes of SIP calls. Many standards have been proposed for the performance evaluation of telephony signaling protocols, however, none of these metrics were used to address the SIP signaling performance until the IETF proposed the RFC 6076, the SIP end-to-end Performance Metrics [18]. However, to the best of our acknowledge, there have been no numerical values or benchmark objectives for the RFC 6076 SIP performance metrics.

A. RFC 6076: SIP END-TO-END PERFORMANCE METRICS

The Internet Engineering Task Force (IETF) adopted standardized end-to-end Performance Metrics for a basic SIP-based signaling system as defined in RFC 6076 [18]. These metrics provide Key Performance Indicators (KPIs) and Service Level Agreement (SLA) indicators to support the SIP-based telephony systems and enhance the network utilization. The RFC 6076 defined the following SIP end-to-end Performance Metrics:

1) REGISTRATION REQUEST DELAY (RRD)

The RRD is used to determine the response delay time for the User Agent REGISTER request. The RDD helps to measure and analyze the successful Registration requests. While at the originating User Agent, the output values for RRD should be in milliseconds (ms). This metric is calculated using equation (1):

$$\begin{aligned} RRD = {} & Time\ of\ Final\ Response \\ & - Time\ of\ REGISTER\ Request \end{aligned} \tag{1}$$

The RRD is calculated only for successful registrations. In addition, when the load of SIP calls increases in the network systems, the value of RRD also increases. When there is a low load in the network system the value of RRD will be in the range of the lowest values.

2) SESSION REQUEST DELAY (SRD)

The SRD is a metric designed to detect the faults or defects that cause delays in responding to INVITE requests. SRD considers both successful and unsuccessful session setup requests where the duration for success and failure responses is varied. The SRD is calculated using the following equation (3):

$$\begin{aligned} SRD = {} & Time\ of\ Status\ Indicative\ Response \\ & - Time\ of\ INVITE \end{aligned} \tag{2}$$

Session Disconnect Delay (SDD) The SDD is designed to calculate the time interval between the time that the session completion message (BYE) is sent and the last subsequent acknowledgement of the session completion response received (2xx). The SDD is used to detect the failures or impairments that cause the delays for a session to end. The SDD measures both successful and failed session disconnections where the output values are in milliseconds (ms). The SDD is calculated using the following equation (4):

$$\begin{aligned} SDD = {} & Time\ of\ (2xx)\ or\ Timeout \\ & -Time\ of\ Completion\ Message\ (BYE) \end{aligned} \tag{3}$$

In this paper, hence, the three RFC 6076 metrics of performance are considered during the evaluation phase for the proposed Cross-Layer approaches. These metrics are the Registration Request Delay (RRD), the Session Request Delay (SRD), and the Session Disconnect Delay (SDD).

III. RESEARCH SCENARIOS AND IMPLEMENTATIONS

The key aim of this research is to study a simple, closed, OLSR-based MANET scenario with a high density of mobile nodes using SIP-based VoIP to communicate together. The study assumed that a single SIP server is based on a B2BUA-based SIP signaling system for a MANET network system, which provides the SIP registration, initiation, and termination mechanisms for SIP calls [19]. It also assumed that node A is the caller and node B is the callee. Node A and node B both need to register with the SIP server to identify their existence and IP addresses. When node A wants to call node B, the SIP initiation messages for the call setup process start flooding between both caller nodes through the B2BUA-based SIP server. When the call ends, the SIP termination message will be sent through the B2BUA-based SIP server to terminate the call.

This scenario takes advantage of applying SIP-based VoIP applications as an alternative or backup communication system over mobile nodes that support the MANET network system. This system could be used for disaster and emergency recovery schemes when other communication systems are lost or broken down. The infrastructure-less, multi-hop communication, dynamic topology features of a Mobile ad hoc network differentiate it from other conventional wireless networks [2]. Self-configuring natures, infrastructure-less features are the best suitable form for emergency applications, military operations, and collaborative applications. In emergency applications where pre-available infrastructure is destroyed due to sudden catastrophes (e.g. Tsunami, earthquake, etc.), a Mobile ad hoc network can be used to establish communication with the emergency units, and similarly in collaborative applications like video conferencing, multimedia chat, etc. These implementations can be one-to-many and many-to-many. Therefore a Mobile ad hoc network needs to be deployed in various environments like broadcast and multicast natures. The major requirements of these applications are delays and packet sensitivity [20], where the retransmission mechanism cannot be applied for real-time applications. Thus, in this study, a MANET with a moderate node capacity and different types of mobility models is considered. Finally, it is worth indicating that this work is significantly driven by our previous work [44]. In fact, the work of this paper can be considered as a major extension with a significant optimization for [44], as supported by the drawn motivation along with contributions in section I.

IV. RESEARCH METHODOLOGY

In this research study, the methodology depends on the literature investigations of the current state of the art for SIP signaling over MANET. This work is then going to implement a simple closed MANET system with different mobility scenarios and apply the SIP-based VoIP application between two ends. After that, the study will apply a sequence of implementations to evaluate and optimize the SIP signaling system over MANET [10]. The SIP signaling and QoS parameters for VoIP over MANET will be assessed on the simulation tools to implement the research scenarios. In general, the evaluation of the SIP signaling performance for the simulated scenarios depends on the end-to-end performance metrics of RFC 6076 and the call setup time. In addition, the evaluation of SIP signaling considered the B2BUA SIP server performance regarding SIP messages during the registration, initiation, and termination processes of SIP calls. This study considers the B2BUA-based SIP server because of its features among the proxy-based SIP server.

The first part of the study will evaluate and analyze the SIP signaling system and VoIP performance over MANET. This evaluation will include part of the determination efforts to benchmark the SIP end-to-end performance metrics of RFC 6076 for WLAN in general, and MANET in particular for a B2BUA-based SIP signaling system [21]. In addition, this part determines the main performance issues for the implemented systems that support the studies and implementations for new performance enhancement approaches. The second part of this study is going to employ the ROHC system over an IPv6 SIP-based VoIP application [22] for the same MANET system that was used in the first part of the evaluation study. This implementation will be used to investigate the SIP signaling performance, in particular the enhancement level which ROHC could basically provide for the investigated performance metrics [23]. The third part of this study considers the development and enhancement methods to improve the performance of the SIP signaling system over MANET. This part will depend on the evaluation and analysis findings in the first and second parts. The dynamic adjustments for the parameters of MANET routing protocols is applied for the performance enhancement approaches. These approaches will provide a flexible dynamic accommodation for the SIP signaling system for VoIP applications to meet with the variable connectivity nature of MANET. Most importantly, the third part is considered as the main contribution of this research study. The results of the performance enhancement contribution will be analyzed and compared with the current state of the art to show the enhancement level of the SIP signaling performance over MANET.

V. SIMULATION PARAMETERS AND ASSUMPTIONS

In this work, the IEEE 802.11n is applied as the wireless network standard for MANET, due to its enhanced features, wide usage over WLAN devices, and good mobility support for MANET comparing to 801.11a, b and g, according to [24] and [23]. OPNET® Modeler provides the best implementation capabilities over other simulation tools and test-beds for the SIP signaling system over OLSR routing protocols. OPNET®

Modeler is a Discrete Event Simulation (DES) tool that provides realistic and accurate implementations for SIP-based VoIP applications over MANET, and for a large number of nodes with reliable implementations. As a result, the simulation efforts will be carried out in OPNET®Modeler version 17.1.

The simulation is implemented over four types of mobility models: Static, Uniform, Random Waypoint (RWP), and RWP-All. These models represent the most common real-life mobility scenarios for MANET nodes [25]. The B2BUA-based SIP server is fully controlling and managing the SIP sessions over all the call stages. In addition, the SIP server is assumed to have a high performance for data processing over all received and sent SIP traffic during all the calling stages. In the Static model, the MANET's nodes are stable and not moving. In the Uniform model, all nodes move in the same direction, with different speeds within the identified speed limitation range, except the SIP server which is in static state [10]. In the RWP model, the nodes move in different directions, but the SIP server is stable in the center of the simulation area. In the RWP mobility model, every node in the simulation, except the SIP server, has its own mobility direction and speed, depending on the identified random functionality of the node parameters. In the RWP-All model, all nodes move in varied directions, including the SIP server. The mobile nodes randomly move, and at the same time act as routers that discover and maintain the route statistics for multi-hop communication [26]. The main characteristics that affect the implementations of these mobile nodes are the unpredictable topology changes, low bandwidth, high level of mobility, and variable connections. The reason for examining random mobility using two different models is to study and evaluate the effect of SIP server mobility for VoIP applications and the signaling QoS.

In general, the topology modelling system in the designed scenarios depends on the OLSR mechanisms for route selections in both OLSR which are perfectly implemented in OPNET® Modeler. The topology selection depends on the algorithm of the routing protocol that considers the location, number of hops, and power issues. For both RWP and RWP-All scenarios, MANET has dynamic topologies among its nodes. Thus, the nodes are partially connected during the simulation time. The Queuing theory for the B2BUA-based SIP server system that is applied in the simulation system is for a single server node with an M/M/1/K queue. The single server queue has a limited queue size (K) (i.e. buffer) as represented. In addition, the M/M/1/K queue system is applied for the node's routing system for each node in the MANET, as shown in the equations of (1), (3), and (4) [27], [28]. Furthermore, this study considers both IPv4 and IPv6 MANETs in order to identify the difference in route overhead between the two IP systems. However, IPv6 will not apply the QoS features over MANET applications in this research study and simulation efforts, as it is not supported over MANET in OPNET® Modeler. The GSM voice codec will be used for VoIP applications because of its simplicity, wide usage, efficiency, and compatibility with MANET natures [29], [30]. In most of the simulation, voice data will travel through the B2BUA-based SIP server between the caller and the callee to provide a secure communication system [31]. Table 1 presents the simulation parameters that were identified depending on the features and capabilities of the MANET and SIP-based VoIP applications. Both design and implementation will be used to investigate and evaluate the QoS for SIP-based VoIP over MANET using the previously identified mobility systems.

TABLE 1. System configuration in OPNET® modeler [1].

A. MANET				
Number of Simulations:	56	**Simulation Seed Number:**	128	
Simulation Duration:	30 Minutes = 1800 Seconds			
Mobility Models:	Static, Uniform, RWP, and RWP-All			
MANET Routing Protocols:	OLSR	**Background Traffic:**	30% to 40%	
Number of nodes:	25 nodes	**Area Dimension:**	1 km x 1 km	
Node Speed Range:	uniform speed between 1.4 m/s (5 km/hr) and 9.8 m/s (35 km/hr)			
WLAN Physical Characteristic:	802.11 n	**Data Rate:**	13 Mbps	
Maximum Transmission Range between Nodes:		100 -250 meters		
Frequency Band:	2.4 GHz	**Transmissi on Power:**	0.001 W	
Minimum MTU:	IPv4: 586 Bytes IPv6: 1280 Bytes	**Buffer Size:**	32 Kbytes	
B. Applications: SIP Based VoIP				
SIP Server Connect Timeout:	TCP Based	**Voice Codec:**	GSM (13 Kbps)	
VoIP Calls Unlimited	**Call Duration:**	**Caller:**	**Callee:**	**Total VoIP Calls in 1800 Seconds:**
	10 Seconds	Node 1	Node 24	175 Calls
Maximum Simultaneous Calls:	**B2BUA-based SIP Server:**	**User Agent**	**Total VoIP Calls in 1800 Seconds in the SIP server:**	
	Unlimited Calls	1 Call at time	Unlimited Calls	

All the assumptions and simulation setup parameters of this study are based on other similar studies [25], [27], [32]–[35]. The MANET scenarios and implementation methods that are followed in this research study are driven from [30] and [36]. Furthermore, the SIP server in this design is used as a single SIP server. The SIP server will act as a B2BUA for SIP-based applications. Therefore, the SIP server is considering multiple SIP/TCP connections at the time of multiple instantaneous calls. The calls are in its best effort conditions where background traffic is generated at the same time. The total estimated number of VoIP calls initiated in the best effort conditions is 175 calls in 30 minutes between node 1 and node 24. The simulations generates an unlimited number of sequenced voice calls between different nodes in MANET to provide the background traffic using the VoIP applications. This background traffic has a medium level of saturation for the provided wireless bandwidth in MANET, with 40% as represented in [30], [33], [34], [37], and [38].

In the simulation efforts, using static and simple SIP-based VoIP applications, it is assumed that the Session Description Protocol (SDP) signaling system is not applied in the simulation works of this research study [39]. Most of the research studies in the Literature considered SIP signaling without the SDP signaling system in their research efforts, such as in [33], [34], [37], [40], and [41]. This is because of the limitations and constraints on the analysis and implementation works when applying the SDP signaling system [2], [42]. Moreover, the implementation of SDP signaling in the simulation tools that support SIP systems over OPNET® Modeler is not possible, especially for MANET. This is because these simulation tools do not provide SDP, and the mobility issues have considerable problems over SIP/SDP signaling synchronization during the simulation. Therefore, this research study focused on the static simple mechanism of SIP signaling and its timers to evaluate and enhance the SIP applications.

A. THE EVALUATION METHODS FOR THE SIP-BASED VoIP APPLICATIONS OVER MANET

The assumptions in this work are designed to generate many VoIP calls within a short period of time to provide a comprehensive investigation for all sessions over different MANET models. OPNET® Modeler supports the implementations for SIP-based VoIP applications over OLSR routing protocols with the ability to analyze the simulation results. The B2BUA-based SIP server is used to provide the SIP-based VoIP implementations. The B2BUA-based SIP server provides a secure and controlled communication system that provides a wide range of privacy with dynamic voice connectivity system over MANET. The B2BUA-based SIP server will be implemented using a single server node that uses the M/M/1/K queuing system with a single queue with limited buffer size (K). The SIP signaling and QoS parameters for VoIP are assessed in OPNET® Modeler. The simulations considers simultaneous VoIP applications as background traffic that influences the performance of SIP applications. For Static and Uniform Scenarios, the SIP signaling performance is in its best effort state as the weakest link at the network performs well for routing and data transmission processes. In addition, it is assumed that the connectivity is available for all MANET nodes in the simulation area and no hardware problems are existed. On the other hand, the worst case scenario is represented with the mobility of the SIP server, as shown in the Random Way Point (RWP) scenario. By comparing the efforts over different mobility models, this study highlights the differences between the SIP signaling performance metrics over these examined scenarios. During the results analysis, the average data representation is used as it provides simple and comparable readings that help to understand the overall performance for SIP signaling. To benchmark the SIP end-to-end performance metrics for a B2BUA-based SIP signaling system from the evaluation efforts, the study will depend on the results of the Static and Uniform mobility models. Moreover, the results of the evaluation studies are used with the proposed Cross-Layer Approaches over MANET. These benchmarked values will be utilized to identify the SIP signaling issues that impact the general performance of the SIP-based applications based on the investigated performance metrics.

VI. THE DESIGN AND IMPLEMENTATIONS FOR THE PROPOSED CROSS-LAYER APPROACHES FOR OLSR-BASED MANET

The RFC 6076 proposed end-to-end performance metrics for SIP signaling to provide a standardized method for evaluating SIP performance over different platforms. In this section, a design for a Cross-Layer approach is proposed to enhance the SIP sessions' performance of SIP-based VoIP over OLSR-based MANETs. This approach employs the SIP performance metrics to maintain the SIP registration, call setup, and termination processes of the SIP calls using a dynamic adjustment system for the routing protocol parameters, as shown in **FIGURE 1**. For the registration process, the Cross-Layer algorithm is implemented on the caller agents: User Agent Client (A) and User Agent Client (B). On the other hand, for the call setup process, the Cross-Layer algorithm is applied over the UAC (A), and the UAS (SIP Server) where each of them has a different algorithm depending upon the related identified signaling system.

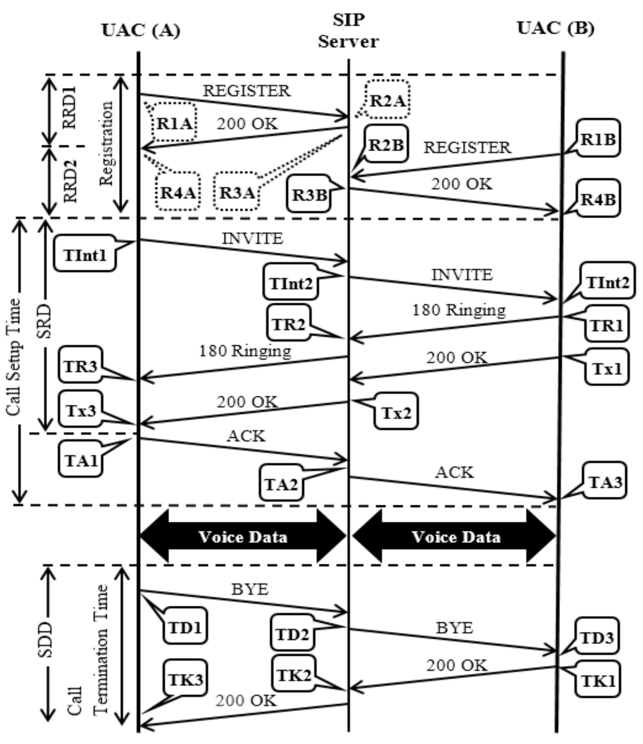

FIGURE 1. The identified parameters of the proposed Cross-Layer Algorithms for the SIP signalling implementations over OLSR based MANET.

Furthermore, the termination process will be implemented on the User Agent Client (A), as shown in **FIGURE 1**. The proposed Cross-Layer approaches are applicable to be implemented over both IPv4 and IPv6 traffic systems as well.

The proactive nature for the OLSR routing protocol is adding extra overhead routing traffic over the OLSR-based MANET, especially with real-time applications. This problem makes the design and implementations of the Cross-Layer approaches for OLSR-based MANET (CLOLSR) more challenging with low levels of performance enhancement. The routing parameters, which are considered for the adjustment and modifications within the proposed CLOLSR, are based on the performance enhancement review for OLSR. The CLOLSR exploits the values of the HELLO messages and the Topology Control (TC) intervals to modify the values of the Multiple Point Relay (MPR) of the OLSR routing protocol. The MPR values increased by the increments of the HELLO messages and the TC intervals' values with the considerations of the OLSR MPR selection mechanism in Algorithm 1. **FIGURE 2** represents the CLOLSR over MANET OSI for all performance enhancement approaches of the investigated SIP processes. The proposed CLOLSR algorithms are not applicable to be implemented over MANET using other routing protocols unless they share the routing concepts and parameters. In this section, the design of the CLOLSR algorithms is professionally introduced to enhance the SIP signaling processes over OLSR-based MANET. Furthermore, the CLOLSR is modelled, implemented, and evaluated by the OPNET® Modeler.

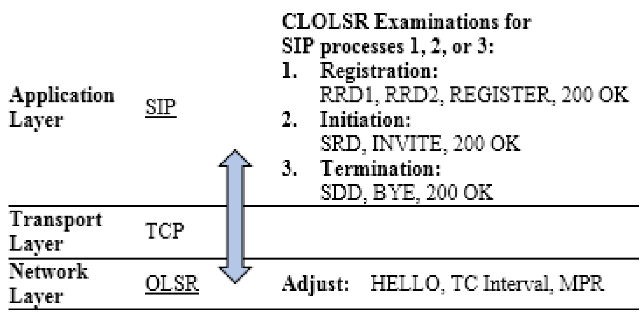

FIGURE 2. CLOLSR representation over MANET OSI for the proposed performance enhancement approaches of the SIP processes.

A. CLOLSR FOR THE CALL REGISTRATION PROCESS

The REGISTER messages of the SIP signaling flow are considered to enhance the performance of the SIP registration processes. The REGISTER messages are used to register the UACs with the SIP server. The SIP performance metrics are employed to improve performance of the registration processes by adjusting the routing parameters of the OLSR to the required level. On the other hand, The RRD values of the SIP performance metrics are employed within the CLOLSR approach to enhance the performance of the registration process by adjusting the route discovery values to the required level depending on the benchmarked values of the

PRD values. Furthermore, the registration intervals of each SIP call are evaluated as well for OLSR-based MANET. The RRD values in the CLOLSR are calculated for each SIP call using the identified timers of the SIP call on the UAC (A) and UAC (B) sides, as demonstrated in **FIGURE 1**. The total registration time is the time interval between the initiated REGISTER messages of UAC (A) (R1A) or UAC (B) (R1B) and the 200 OK response message of the REGISTER message for the UAC (A) (R4A) or UAC (B) (R4B). Therefore, the CLOLSR utilized RRD values of the registration process for the related SIP call and calculated its values using equation (1). The CLOLSR approach for the registration process is applied on the callers' side: the UAC (A) and UAC (B). Once a delay is recognized by the UAC (A) or UAC (B), the CLOLSR approach is triggered to adjust the OLSR routing parameters using the Cross-Layer messages. The adjustments process for the OLSR routing values are based on the investigated SIP performance metrics that passed from the application layer, as represented in **FIGURE 2**.

The CLOLSR algorithm looked at RRD_BenchM parameter as the RRD benchmarked values that will be used to determine the registration processes of the SIP calls over MANET. The values of the RRD_BenchM have been determined from the evaluated RRD values of the benchmark efforts by the evaluation efforts of the Static and Uniform mobility models of the best case scenarios. The RRD values that will be used for the RRD_BenchM are dedicated for both IPv4 and IPv6 traffic by the proposed benchmark sets for the SIP performance metrics. During the registration session, the sequence number for the registration process has the same sequence number as the SIP call number n where "X" refers to the current UAC node. The RRD values for both the caller UAC (A) and UAC (B) will be compared with the considered RRD_BenchM from the benchmarked set during the registration sessions. When the current RRD value becomes greater than the RRD_BenchM value, then the CLOLSR will update the routing table parameters by resending the HELLO messages using longer time periods for the Topology Control (TC) values, as represented in Algorithm 1. The CLOLSR process is activated temporarily for a short period of time until the MPR values of the second hop neighbors (MPR_2) for the current node (X) become more than the values of the first hop neighbors (MPR_1) for the same node. This adoption mechanism is based on the OLSR MPR selection mechanism which is represented in Algorithm 1. Then when the MPR_2 received for node (X) and for the current registration process and its value is greater than the current MPR_1 value, the REGISTER request will be sent again and the TC value will be degraded to its original value to save the CPU cycles and reduce the bandwidth consumption. The proposed algorithm have used the simulation time t which is represented in seconds, T_Registration_Wait which is the maximum specified time in seconds before sending the re-REGISTER message, and T_End is the simulation end time. The total registration time for a registration process is the total of RRD1 and RRD2, or it is the time difference between the first registered UAC of the call parties with the SIP Server and the last received registration response from the call parties.

Algorithm 1 The Call Registration Process for the Cross-Layer OLSR. The Implementation of This Algorithm Is on the $UAC(R(n))$ for SIP Call Number n, Where R Is the User Agent Client A or the User Agent Client B

```
REGISTER(n)
Inputs: RRD_BM,t_rw,t_end
    for t=0, until t ≤t_end, with step t = t+1
      if REGISTER(n)is sent by theUAC(n)
        if OK_200 (n) is received and
          ...REGISTER(n) is received
          RRD(n) = R4(n) – R1(n)
        else if R3(n) > RRD_BM
            ... or RRD(n) > RRD_BM
            ... or TC == 0
            ... or MPR1(X,n) ≥ MPR2(X,n)
          TC_Current = TC
          TC = TC+1
          resend HELLO(n)
          if MPR2(X,n) is received
            ...and MPR2(X,n) > MPR1(X,n)
            ...and R(n) != R(n+1)
            resend REGISTER(n)
            TC = TC_Current
            go to REGISTER(n+1)
          else return
        else return
      else if t > t_rw
        resend REGISTER(n)
        go to REGISTER(n+1)
      else go to REGISTER(n+1)
```

B. CLOLSR FOR THE CALL SETUP PROCESS

The performance of the SIP call initiation processes depends on the messages of the call setup that are being interchanged between the call entities. For the CLOLSR approach over MANET, the SRD values are calculated for all calls which are initiated by the caller node UAC (A) as represented in **FIGURE 1**. The call initiation messages go through the SIP server and all the INVITE messages and parameters are recognized by the SIP server and the caller for the evaluations of the SIP signaling performance. The time difference between TInt1 and TA3 is considered as the call setup time where the SRD value is the time difference between the INVITE message, sent at time TInt1, and Tx3, in equation (3). The SIP call setup intervals will be evaluated for OLSR-based MANET. The CLOLSR approaches have employed the three-way handshake system for the call setup process between the SIP call entities. The SRD SIP performance metrics are also applied in the CLOLSR. This proposed approach is applied over the UAC (A) and the UAS

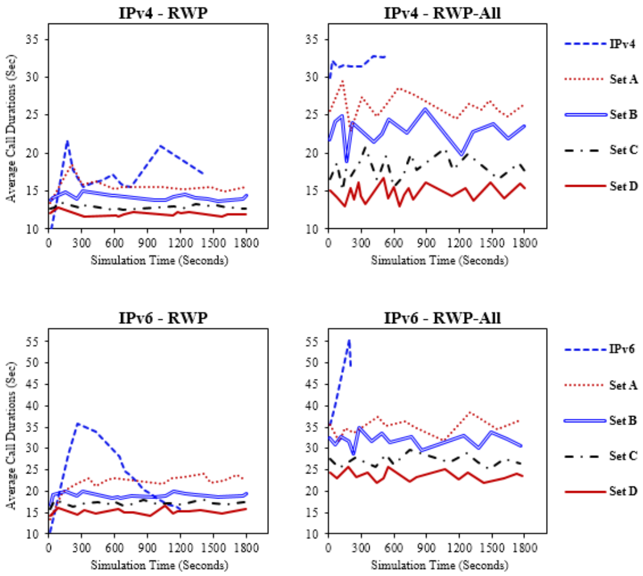

FIGURE 3. Average call durations in seconds for OLSR-based MANET using CLOLSR approach.

Algorithm 2 The Call Setup for the Cross-Layer OLSR. The Implementation of This Algorithm Is on the $UAC(R(n))$ for SIP Call Number n Where R Is the User Agent Client A

```
CALL_SETUP_UAC(n)
Inputs: SRD_BM,t_rw,t_end
    for t=0, untilt ≤t_end, with step t = t+1
        if INVITE(n) is sent by the UAC(n)
            if RINGING_180 (n) is received
                SRD(n) = TX3(n) - T_INT 1(n)
            else if TR3(n) > SRD_BM
                ... or SRD(n)>SRD_BM or TC == 0
                ... or MPR1(X,n)≥ MPR2(X,n)
                TC_Current = TC
                TC = TC+1
                resend HELLO(n)
                if MPR2(X,n) is received
                    ...and MPR2(X,n) > MPR1(X,n)
                    ...and R(n) != R(n+1)
                    resend INVITE(n)
                    TC=TC_Current
                    go to CALL_SETUP_UAC(n+1)
                else return
            else return
        else if t > t_rw
            resend INVITE(n)
            go to CALL_SETUP_UAC(n+1)
        else go to CALL_SETUP_UAC(n+1)
```

(SIP Server). Whenever a delay is recognized by any of the SIP call parties, the CLOLSR approach will be activated to optimize the routing performance using the Cross-layer messages to modify the OLSR routing parameters based on the analyzed performance metrics of the call setup process in the application layer, as illustrated in **FIGURE 2**.

The call initiation processes of the SIP-based VoIP implementation over OLSR have very long delays over RWP and RWP-All mobility models. Therefore, the proposed CLOLSR algorithm is designed to determine and optimize the call setup performance over OLSR based on the evaluation values of SRD parameters. The design of the CLOLSR algorithm considered the SRD_BenchM values to determine the proposed benchmarked sets of the SIP performance metrics. In the proposed algorithm, the actual SRD value for a call setup process of the SIP call number n is calculated using equation (4), and Algorithm 5. The SRD values of the caller agent UAC (A) is compared with the SRD_BenchM values within the call setup process for the evaluation purposes. Once the SRD value of the current call is recognized by the CLOLSR that it is greater than the SRD_BenchM values, the routing update is activated by sending the HELLO messages with longer intervals to support the Topology Control (TC) mechanisms of the MANET with the active routes, as depicted in Algorithm 2 for the UACs. The MPR values of the second hop neighbors (MPR_2) for the current node (X) will be updated with the received routing replies messages. As a result, the MPR_2 values become higher than the values of the first hop neighbors (MPR_1) for the same node. The performance of OLSR and the effectiveness of modifying the MPR selection mechanism have been represented in Algorithm 3. Once the updated MPR_2 is received for node (X) and for the current call setup process with values greater than the current MPR_1 value, the INVITE request will be resent again as a re-INVITE message and the TC values will be degraded to their previous values to save CPU cycles and reduce the bandwidth consumption. On the other hand, the UAS depends on the signaling timers for the INVITE messages and responses that are exchanged between both callers during the call setup processes. The UAS is also able to control the call setup performance for the INVITE messages between both UAC (A) and UAC (B) by examining the TInt2, TR2, Tx2, and TA2 parameters. Therefore, the UAS can determine the SRD values for the UAC (A) call setup process from the sequence numbers and time stamps of the INVITE messages and their acknowledgments. Once a delay is detected, the UAS activates the performance enhancement approach to adjust the routing parameters with the required level to update the routing table with the required active routes. In this proposed algorithms for the call setup processes, the X refers to the current UAC node. The SIP call number is n which also represents the sequence number of the call setup process. The simulation time is t in seconds, the T_Call_Setup_Wait is the maximum specified time in seconds before sending the re-INVITE message, and T_End is the simulation termination time.

The proposed CLOLSR algorithm for the call setup process designed to provide a reliable detection system for any delays and undeliverable SIP messages that will take place

Algorithm 3 Call Termination for Cross-Layer OLSR. The Implementation of This Algorithm Is on the *UAC*(*R*(*n*)) for SIP Call Number *n*, Where *R* Is the User Agent Client A or the User Agent Client B

CALL_TERMINATION(n)
Inputs: *SDD_BM,t_rw,t_end*
for $t = 0$, until $t \le t\_end$, with step $t = t+1$
  if *BYE(n)* is sent by the *UAC(n)*
    if *TK3(n)* is received for *BYE(n)*
      $SDD(n) = TK3(n) - TD1(n)$
    else if $TK3(n) > SDD\_BM$
      … or $SDD(n) > SDD\_BM$
      … or $TC == 0$
      … or $MPR1(X,n) \ge MPR2(X,n)$
      $TC\_Current = TC$
      $TC = TC+1$
      resend *HELLO(n)*
      if *MPR2(X,n)* is received
        … and $MPR2(X,n) > MPR1(X,n)$
        … and $R(n)! = R(n+1)$
        resend *BYE(n)*
        $TC = TC\_Current$
        goto *CALL_TERMINATION(n+1)*
      else return
    else return
  else if $t > t\_rw$
    resend *BYE(n)*
    go to *CALL_TERMINATION(n+1)*
  else go to *CALL_TERMINATION(n+1)*

during this process. The CLOLSR approach considers the SRD performance metric to adjust the level of the routing update values. This approach provides a self-adjustment mechanism for the SIP signaling instead, to enhance the call setup performance over MANET. In addition, the SRD values are employed to be evaluated by the caller nodes and the UAS with active monitoring during the call setup processes to enhance the performance. Whenever the MPR values are not updated during this approach, the call setup performance will not be enhanced where additional delays occur, because the algorithm values need to be recovered with their previous values to complete the re-invitation process. Moreover, the CPU cycles increase with the growing in the number of HELLO messages to update the routing table entries. Consequently, the bandwidth consumption increases with inactive updates for the routing tables.

C. CLOLSR FOR THE CALL TERMINATION PROCESS

The termination performance of a SIP-based VoIP call depends on the transportations of the BYE messages that sent from one of the callers during the call, and gone through the B2BUA SIP Server. The CLOLSR approach considers the SDD values during the performance enhancement for the call termination process to provide dynamic adjustments for the route discovery values. The original existing SDD values have been evaluated and considered for the suggested benchmark values in this section. The SDD values are determined by the caller nodes using the identified timers of the SIP call number n on the UAC that requested the session termination as illustrated in **FIGURE 2**. The SDD values for a SIP call number n is calculated using equation (4), and Algorithm 1. For the call termination process, the SDD values is the time difference between TD1 which is the time of sending the BYE message by the UAC, and TK3 which is the time of receiving the 200 OK response message of the BYE message. During the call session, the CLOLSR system observers the call termination processes between the caller agents. Once a delay being recognized by the UAC (A), the performance enhancement is activated to adjust the routing parameters using the Cross-Layer messages based on the identified SIP performance metrics at the application layer. For the performance enhancement process of the call termination, the CLOLSR algorithm considered the SDD_BenchM parameter as the benchmarked values of the SDDs which are used to determine any existing delays as represented in Algorithm 3. The SDD_BenchM values taken from the values of the investigated scenarios during the benchmarking efforts. In addition, the SDD_BenchM values determined for both IPv4 and IPv6 implementations using the proposed sets of the benchmarked SIP performance metrics. During the termination process, the sequence number for the registration process has the same sequence number the SIP call number n.

The CLOLSR will be activated to update the routing table parameters by regenerating the HELLO messages when the SDD values are greater than the SDD_BenchM. An update for the TC values using longer intervals is applied as shown in Algorithm 3. This activation for the CLOLSR process works temporarily until the MPR values for the second hop neighbors (MPR_2) of the current node (X) become higher than the values of the first hop neighbors (MPR_1) of the same node (X). This modification mechanism of the MPR selection considers the basic MPR routing system in Algorithm 1. Once the MPR_2 values updated for node (X) with the current termination process with the required value, the BYE request will be re-sent , and the current TC value for node (X) will be degraded to its original value to save the CPU cycles and reduce the bandwidth consumptions. The termination process sequence number is the same sequence number for the SIP call. The simulation time t is represented in seconds, T_Termination_Wait is the maximum specified time in seconds before resending the BYE messages, and T_End represents the simulation end time.

VII. ANALYSIS METHODS FOR THE RESEARCH STUDY

The evaluation studies for SIP-based VoIP applications is utilized to determine the SIP signaling performance for IPv4 and IPv6 for OLSR over four different mobility models. Both Static and Uniform mobility models will be used to represent the best effort scenarios which are used to compare the performance of the SIP signaling over MANET with the performance of other platforms, such as LAN, WLAN, Satellite, and WiMAX. This step will indicate the accuracy level of the evaluation works of this research study as the comparison efforts should meet the expected level of similarity with other network systems, as provided in [21], [30], [33], and [37].

The RWP and RWP-All mobility models are expected to have higher levels of difference in their results when compared with the Static and Uniform mobility models. Thus, it will be hard to link these differences with other platforms of network systems as the communication natures are different and simulation parameters are not fully matched with the identified parameters in this research study. Therefore, this research study will focus on the evaluation works on OPNET® Modeler for SIP-based VoIP over OLSR to benchmark the end-to-end SIP performance metrics and the SIP signaling performance for the current state of the art. Then, these values are used as a reference (baselines) to compare them with the performance values of the proposed algorithms to specify the enhancement level that could be provided. The comparison depends on different parameters for both SIP signaling and the MANET routing protocol. The evaluation works will analyze the results and show the differences over different mobility models and routing protocols based on the graphical representations and the benchmark values of the simulation results. In addition, the results analysis will include the performance of both the B2BUA-based SIP server and the SIP callers to provide comprehensive investigations for the SIP signaling system.

VIII. RESULTS AND DISCUSSION FOR THE CLOLSR-BASED IMPLEMENTATIONS

In this section, the evaluation results for the proposed Cross-Layer OLSR approach of the SIP-based VoIP implementations will be discussed with the considerations of the CLOLSR design. The evaluations considered four identified sets of the benchmarked values of the RFC 6076 for the end-to-end SIP signaling performance metrics as shown in Table 2. To investigate the CLOLSR approach over SIP-based MANET, the proposed sets of the benchmarked values are designed to support the SIP signaling system over OLSR-based MANET. The Benchmark sets for the SIP performance evaluation metrics. These sets are representing the best efforts for the Cross-Layer implementations over the investigated scenarios of the SIP-based VoIP over MANET. The RRD, SRD, and SDD values are identified gradually from the longest to the shortest values of the benchmarked sets as given in Table 2. Set A provides the longest possible values for the investigated RRD, SRD, and SDD parameters that can support the performance enhancement for the SIP signaling system over OLSR-based MANET. These values have decreased consequently in sets B, C, and D. Set D represents the best values with low parameters of the Benchmarked end-to-end SIP performance metrics. Set D represents the values of the best case, and set A represents the values of the worst case for benchmarked performance metrics. The SIP signaling performance for the CLOLSR implementations is affected by the identified benchmarked sets that are reflected on the general performance for the registration, initiation, and termination sessions. Using the study assumptions that were proposed in section V, a number of SIP-based VoIP calls were generated with short periods to fulfil the investigation efforts about all sessions of the SIP signaling system during the simulation time using different mobility models. The SIP signaling performance evaluates the registration, initiation, and termination sessions for the SIP-based VoIP calls. The statistical results in OPNET® Modeler uses the successful ratio, number of connected calls, and the call durations for this evaluation [43]. The total number of the initiated calls between the caller and the callee is 175 calls over all the scenarios with duration of 10 seconds for each call.

TABLE 2. Number of initiated VoIP calls for IPv4/IPv6 OLSR-based MANET.

	IPv4 OLSR				
	IPv4	Set A	Set B	Set C	Set D
RWP	49	68	79	91	104
RWP All	34	42	49	56	61
	IPv6 OLSR				
	IPv6	Set A	Set B	Set C	Set D
RWP	40	61	69	76	88
RWP All	3	17	21	24	32

The total number of initiated VoIP calls over the IPv4 OLSR-based MANET for both RWP and RWP-All mobility models are represented in Table 2. Set D has a larger number of initiated calls when compared the actual normal IPv4 traffic and set A has a lesser number of successful calls. The total number of the implemented calls is increased over sets B, C and D. Set D represents the best level of successful VoIP calls with 104 calls for RWP, and 61 calls for RWP-All. The enhancement percentage of successful calls for set D is 59.43% of the total implemented calls, which is 31.43% more than the IPv4 traffic implementations over RWP without CLOLSR. On the other hand, for the RWP-All mobility model, set D has 34.9% of successful calls which is more than the actual IPv4 traffic for RWP-All with 15.47%. For IPv6 implementations of OLSR-based MANET, the implementations of the CLOLSR showed variable levels of enhancement upon the implemented VoIP calls with the proposed benchmarked sets, as shown in Table 2. Set A has a lower number of implemented VoIP calls when compared with the IPv6 based VoIP calls for both RWP and RWP-All mobility models. While the number of successful calls for RWP had increased to 88 calls with set D when compared with 40 calls for normal IPv6 traffic with a percentage of increase of 27.43%. Furthermore, set D have increased the number of successful calls for the RWP-All mobility model to 32 compared with only 3 for normal implementations of IPv6 traffic with a percentage increase that reached 16.58%.

The increase by the CLOLSR is still limited and does not provide a higher level of enhancements over the considered implementations. However, this level of enhancement over OLSR-based MANET with the considered mobility scenarios is still good when compared with the actual low level of performance for SIP calls.

TABLE 3. Number of rejected VoIP calls for IPv4/IPv6 OLSR-based MANET.

	IPv4 OLSR				
	IPv4	Set A	Set B	Set C	Set D
RWP	133	116	107	98	85
RWP All	140	138	135	132	128
	IPv6 OLSR				
	IPv6	Set A	Set B	Set C	Set D
RWP	144	137	131	126	118
RWP All	173	168	159	151	140

The total number of rejected VoIP calls over IPv4 and IPv6 OLSR-based MANET for both RWP and RWP-All mobility models are given in Table 3. The number of rejected calls is decreased with sets B and C, where set D has the lowest number of rejected calls for IPv4 traffic with 85 for RWP, and 128 for RWP-All. The total percentage of the rejected calls has been reduced to 48.57%, which is 27.43% less than the actual normal IPv4 traffic for RWP, and 73.14% which is 6.86% less than the actual normal IPv4 traffic for RWP-All. For the IPv6 traffic, the number of rejected calls decreased with sets B, C and D. Set D has the lowest number of rejected calls for IPv6 traffic with 118 for RWP, and 140 for RWP-All. The percentage of the rejected calls has reduced to 67.43%, which is 14.86% less than the actual normal IPv6 traffic over RWP, and 80% which is 18.87% less than the actual normal IPv6 traffic for RWP-All.

The total call duration is considering the SIP-based call processes and data transmission intervals. The call duration for a single call is the total required time for the registration, initiation, voice data transmission, and termination processes. The average durations of the VoIP calls within the implementations of the CLOLSR approaches over OLSR-based MANET using the RWP and RWP-All mobility models are shown in VIII-A2. The considered VoIP calls are for duration of 10 seconds for SIP-VoIP calls as identified and implemented in OPNET® Modeler scenarios.

For IPv4 traffic with the RWP mobility model, the average call duration is between 15 and 21 seconds for IPv4. It is enhanced to be between 13.45 and 17.82 seconds for set A, between 13.13 and 15.18 seconds for set B, between 12.72 and 17.33 seconds for set C, and between 11.86 and 13.77 seconds for set D. With the RWP-All mobility model, the IPv4 traffic has an average between 28 and 37 seconds. This average is enhanced with the CLOLSR implementations to be between 22.78 and 29.65 seconds with set A, between 17.42 and 25.43 seconds with set B, between 14.93 and 21.87 seconds with set C, and between 12.76 and 17.13 seconds with set D. On the other hand, for the IPv6 traffic, the average call duration is between 14 and 36 seconds and is enhanced to be between 13.75 and 24.37 seconds for set A, between 16.64 to 20.28 seconds for set B, between 16.14 to 17.73 seconds for set C, and between 14.26 to 16.92 seconds for set D. With the RWP-All mobility model, the IPv6 traffic has very long call durations with an average of 35 to 55 seconds. These average values have been enhanced with the implementations of the CLOLSR to be between 31.46 and 38.85 seconds with set A, between 27.94 and 30.14 seconds with set B, between 25.1 and 30.12 seconds with set C, and between 21.86 and 25.74 seconds with set D. From the results shown in VIII-A, set D has the best enhancement level with implementations of CLOLSR over the total SIP-based call durations over both RWP and RWP-All mobility models and for both IPv4 and IPv6 traffic. The call duration intervals reduced for IPv4 traffic with about 5 seconds for RWP and 14.65 seconds for RWP-All. For IPv6 traffic, the call duration intervals reduced on average to 8.67 seconds for RWP and 21.82 seconds for RWP-All.

A. SIP REGISTRATION INTERVAL WITH CLOLSR

The earliest stage of the SIP-based VoIP calls is the registration process. The implementations of the CLOLSR approach considered the registration processes for the user agents with the system SIP server. The implementations of the CLOLSR approach have different levels of performance enhancements over the intervals of the registration processes for both IPv4 and IPv6 SIP-based, with the considerations of the identified sets for the SIP signaling metrics in Table 1.

The average registration time for both the caller and the callee to register with the SIP server for the SIP sessions for IPv4 traffic over OLSR-based MANET with the RWP mobility model is represented in Table 4. The average registration time for IPv4 traffic is 2.49 seconds and is reduced by the implementations of the CLOLSR to 1.92 seconds with set A, 1.76 seconds with set B, 1.42 seconds with set C, and 973.83 ms with set D. The average time of the registration intervals have reduced for IPv4 traffic with an average of 1.2 seconds with the RWP mobility model. The enhancement percentage with set D is 39.12% when compared with 2.49 seconds of the average registration intervals for the original IPv4 traffic. For the RWP-All mobility model, the average registration time for IPv4 traffic is reduced by the implementations of the CLOLSR from 6.02 seconds to 3.03 seconds with set A, 2.54 seconds with set B, 1.96 seconds with set C, and 1.44 seconds with set D. The average time of the registration intervals have reduced for IPv4 traffic with an average of 4.46 seconds with the RWP-All mobility model. The enhancement percentage with set D is 49.53% when compared with 6.02 seconds of the average registration intervals for the original IPv4 traffic.

On the other hand, the average registration for IPv6 traffic over OLSR-based MANET with the RWP mobility model

TABLE 4. Average SIP registration time for SIP clients over IPv4/IPv6 OLSR-based MANET with CLOLSR implementations.

	IPv4 OLSR in milliseconds (msec)				
	IPv4	Set A	Set B	Set C	Set D
RWP	2489.19	1918.31	1757.82	1420.92	973.83
RWP All	6017.38	3029.72	2541.66	1963.59	1443.56
	IPv6 OLSR in milliseconds (msec)				
	IPv6	Set A	Set B	Set C	Set D
RWP	5326.59	3779.22	2875.68	1972.04	1243.58
RWP All	13819.4	5530.92	4601.45	3799.73	2803.15

is illustrated in Table 4. The average registration time for IPv6 traffic is 5.33 seconds and is reduced with the OLSR implementations to 3.78 seconds with set A, 2.88 seconds with set B, 1.97 seconds with set C, and 1.24 seconds with set D. Therefore, the average time of the registration intervals have reduced for IPv6 traffic with an average of 3.34 seconds for the RWP mobility model. The enhancement percentage with set D is 76.73% when compared with 5.33 seconds of the average registration intervals for the original IPv6 traffic. For the RWP-All mobility model, the average registration time for IPv6 traffic reduced with the CLOLSR implementation from 13.82 seconds to 5.53 seconds with set A, 4.6 seconds with set B, 3.80 seconds with set C, and 2.80 seconds with set D. The average time of the registration intervals have reduced for IPv6 traffic with an average of 10.26 seconds over the RWP-All mobility model. The enhancement percentage with set D is 20.29% when compared with 13.82 seconds of the average registration intervals for the original IPv6 traffic.

B. SIP CALL SETUP INTERVAL WITH CLOLSR

The main process for the SIP-based VoIP calls is the call initiation process. The general performance of this process is related to the amount of signaling flow between the callers that go through the SIP server. The average time spent for the call setup processes over the OLSR-based MANET with the implementations of the CLOLSR approach is represented in **FIGURE 4**. The values of the average call setup time are always more than the values of the SRD performance metrics during all the initiated SIP calls.

The average call setup time for the IPv4 traffic with the RWP mobility model is between 2.93 to 5.18 seconds for IPv4 traffic, between 2.1 to 4.83 seconds for set A, between 1.84 to 3.95 seconds for set B, between 1.35 to 3.78 seconds for set C, and between 0.94 to 2.26 seconds for set D. In the RWP-All mobility model, the average call setup time for IPv4 traffic over OLSR-based MANET is in the range of 5.14 to 7.23 seconds for IPv4, between 4.13 to 6.32 seconds for set A, between 3.65 to 5.27 seconds for set B, between 2.18 to 4.98 seconds for set C, and between 1.87 to 3.96 seconds for set D. The results of the IPv4 implementations showed that the implementations of set D of the CLOLSR enhanced the call setup process and reduced the average call setup time to about 2.52 second for RWP and 3.1 to 3.93 seconds for RWP-All.

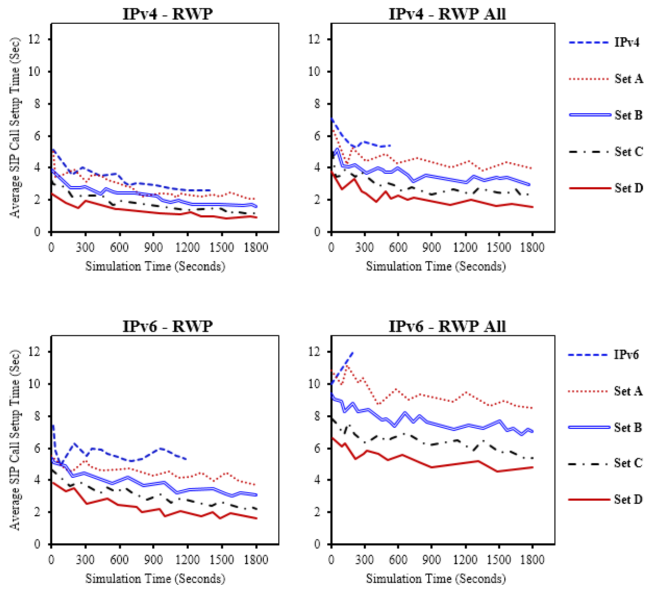

FIGURE 4. Average SIP call setup time in seconds for OLSR-based MANET using CLOLSR approach.

On the other hand, the implementations of the IPv6 traffic with the RWP mobility model showed an average call setup time between 4.91 to 7.18 seconds. The implementations also showed a level of enhancement on the average call setup time which is between 4.27 to 5.67 seconds for set A, between 3.41 to 5.32 seconds for set B, between 2.21 to 4.43 seconds for set C, and between 1.83 to 3.97 seconds for set D. In the RWP-All mobility model, the average call setup time for IPv6 OLSR-based MANET is in the range of 9.88 to 12.19 seconds for IPv6 traffic. The average call setup time over IPv6 traffic is enhanced to be between 8.69 to 11.14 seconds for set A, between 7.38 to 9.43 seconds for set B, between 5.95 to 7.92 seconds for set C, and between 4.89 to 6.73 seconds for set D. The results conclude that set D of the CLOLSR has the best level of enhancement for the average call setup time for IPv6 traffic as it reduced the average call setup time with 2.93 to 3.26 seconds for RWP and 5.27 to 6.15 seconds for RWP-All.

C. SIP CALL TERMINATION INTERVAL WITH CLOLSR

The call termination process is the last part of the SIP-based VoIP call where one of the call's parties finishes the call by sending a call termination request through the B2BUA-based SIP server. The effect of the call termination process over the performance of the SIP-based VoIP is the lowest when compared with the registration and call setup processes. The call termination intervals are equal to the SDD values of the end-to-end SIP performance metrics. The CLOLSR implementations with identified sets for the call termination in Table 2 showed variable level of enhancements over both IPv4 and IPv6 traffic.

TABLE 5. Average SIP termination time for SIP callers over IPv4/IPv6 OLSR-based MANETfor CLOLSR implementations.

	IPv4 OLSR in milliseconds (msec)				
	IPv4	Set A	Set B	Set C	Set D
RWP	1088.14	902.71	712.21	563.72	298.39
RWP All	1818.38	1420.66	1127.38	917.59	723.87
	IPv6 OLSR in milliseconds (msec)				
	IPv6	Set A	Set B	Set C	Set D
RWP	1886.68	1175.54	912.76	753.44	427.63
RWP All	5412.85	3897.27	2952.36	2009.54	1109.35

In Table 5, however, the average call termination time for the initiated SIP calls between the caller and the callee for IPv4 traffic over the RWP mobility model is 1.1 seconds. The implementations of the CLOLSR reduced the average call termination time for IPv4 traffic to 9082.71 ms with set A, 712.21 ms with set B, 563.72 ms with set C, and 298.39 ms with set D. The total average time of the call termination intervals have been reduced for IPv4 traffic with an average of 524.37 ms for RWP. The enhancement percentage is 27.42% for set D when compared with the original average call termination intervals of IPv4 traffic. For the RWP-All mobility model, the average call termination time for IPv4 traffic is 1.82 seconds. The implementations of the CLOLSR reduced the average call termination time for IPv4 traffic to 1.42 seconds with set A, 1.13 seconds with set B, 917.59 ms with set C, and 723.87 ms with set D. Therefore, the average call termination time have reduced for IPv4 traffic with an average of 0.9 second for the scenarios of the RWP mobility models. The percentage of the enhancement level is 39.81%, compared with 1.82 seconds for the original average call termination time for IPv4 traffic.

On the other hand, Table 5 shows the average call termination time for the initiated SIP calls between the caller and the callee for IPv6 traffic over the RWP mobility model. The average call termination time for IPv6 traffic is 1.89 seconds and is reduced by the implementations of the CLOLSR to 1.18 seconds with set A, 912.76 ms with set B, 753.44 ms with set C, and 427.63 ms with set D. From the results, the average of the call termination time for IPv6 traffic reduced with an average of 743.33 ms to 1313.74 ms for RWP, where the enhancement percentage is from 39.41% to 69.62% of the original average of the call termination intervals. For the RWP-All mobility model, the average call termination time for IPv6 traffic is reduced by the implementations of the CLAODV from 5.41 seconds to 3.91 seconds with set A, 2.95 seconds with set B, 2.01 seconds with set C, and 1.11 seconds with set D. The average time of the termination intervals reduced for IPv6 traffic with an average of 3.45 to 4.37 seconds for the RWP-All mobility model. The enhancement percentage is from 37.13% to 20.52%, compared with the original average of the call termination processes for IPv6 traffic.

D. ROUTING PERFORMANCE WITH CLOLSR

The performance of the SIP signaling on MANET is affected by the routing performance. For OLSR-based MANET, the routing parameters have an important role over the performance enhancement for the implementations of the SIP-based applications. In this section, the related OLSR routing parameters will be investigated regarding the enhancement level of the CLOLSR implementations over the SIP signaling performance. In **FIGURE 5**, the average number of HELLO messages for OLSR routing processes that were sent during the simulation time is represented. The route discovery status between the nodes within the dynamic mobility nature of MANET depends on the number of activated HELLO messages. As the number of HELLO messages increases, the connectivity between the nodes becomes better over the dynamic nature. However, the routing overhead increases with the increase of HELLO traffic in the MANET. The increasing number of HELLO messages means that only a few of the known routes are known between the communicating nodes because of the nodes' dynamic mobility that affects the reachability, and availability, and causes rapid disconnections. For IPv4 traffic over the RWP mobility model, the average HELLO traffic sent in the MANET is between 4.4 Kbits/s and 5.9 Kbits/s. The average number has increased with the implementations of the proposed CLOLSR approaches to between 5.5 Kbits/s and 9.2 Kbits/s with set A, between 6.7 Kbits/s and 10.3 Kbits/s with set B, between 7.6 Kbits/s and 11.7 Kbits/s with set C, and between 7.3 Kbits/s and 12.2 Kbits/s with set D. For the RWP-All mobility model, the average number of HELLO traffic that were sent for IPv6 traffic implementations increased to be between 6.5 Kbits/s and 7.8 Kbits/s. With the implementations of the CLOLSR, the average number of HELLO traffic

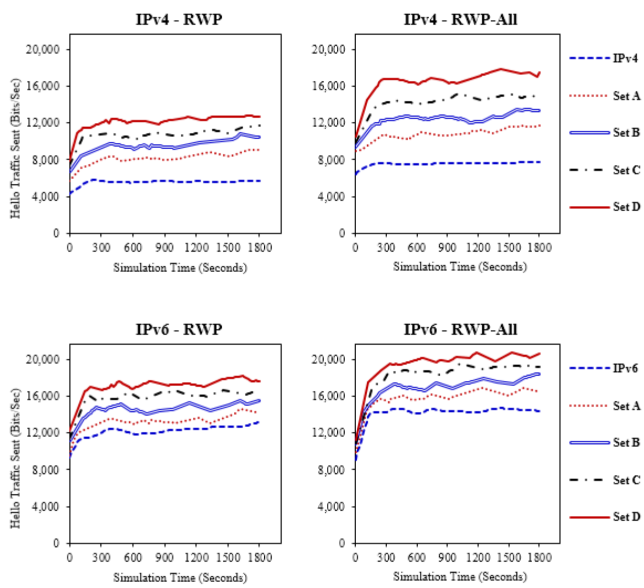

FIGURE 5. Average HELLO traffic sent in MANET for OLSR-based MANET using CLOLSR approach.

sent for IPv6 traffic increased to between 8.8 Kbits/s and 11.7 Kbits/s with set A, 9.1 Kbits/s and 12.5 Kbits/s with set B, 9.6 Kbits/s and 15.1 Kbits/s with set C, and 10.1 Kbits/s and 17.8 Kbits/s with set D.

On the other hand, for IPv6 traffic over the RWP mobility model, the average HELLO traffic sent in the MANET is between 10.1 Kbits/s and 12.9 Kbits/s. This average number increased with the CLOLSR implementations to between 10.3 Kbits/s and 14.4 Kbits/s with set A, between 10.9 Kbits/s and 15.4 Kbits/s with set B, between 11.7 Kbits/s and 16.8 Kbits/s with set C, and between 12.1 Kbits/s and 18.3 Kbits/s with set D. The average number of HELLO traffic sent for IPv6 traffic over the RWP-All mobility model increased to be between 9.5 Kbits/s and 14.3 Kbits/s. With the implementations of the CLOLSR, the average number of HELLO traffic sent for IPv6 traffic increased to between 10.6 Kbits/s and 16.4 Kbits/s with set A, 11.2 Kbits/s and 18.2 Kbits/s with set B, 10.8 Kbits/s and 18.5 Kbits/s with set C, and 11.1 Kbits/s and 20.3 Kbits/s with set D. From the previous results, the implementations of the CLOLSR increased the number of HELLO messages over both RWP and RWP-All mobility models which indicated the active number of requested route updates during the nodes' mobility for SIP applications.

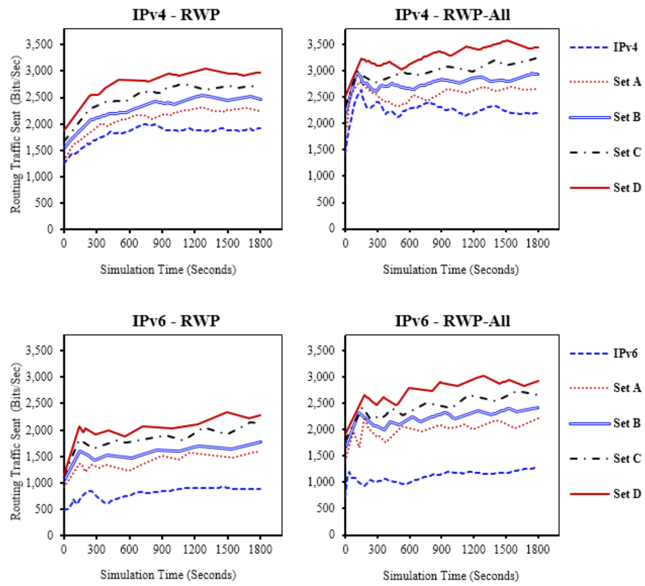

FIGURE 6. **Average routing traffic sent by the caller node in (Bits/Sec) for OLSR-based MANET using CLOLSR approach.**

FIGURE6, on the other extreme, shows the average routing traffic that is sent by the caller node for the implementations of the SIP-based VoIP applications with the investigations of the CLOLSR approaches. For IPv4 traffic, the average number of sent routing traffic from the caller node is in the range of 1.23 to 1.93 Kbits/s over the RWP mobility model. The average number of the sent routing traffic increased with the CLOLSR implementations to between 1.39 to 2.17 Kbits/s with set A, between 1.52 to 2.52 Kbits/s

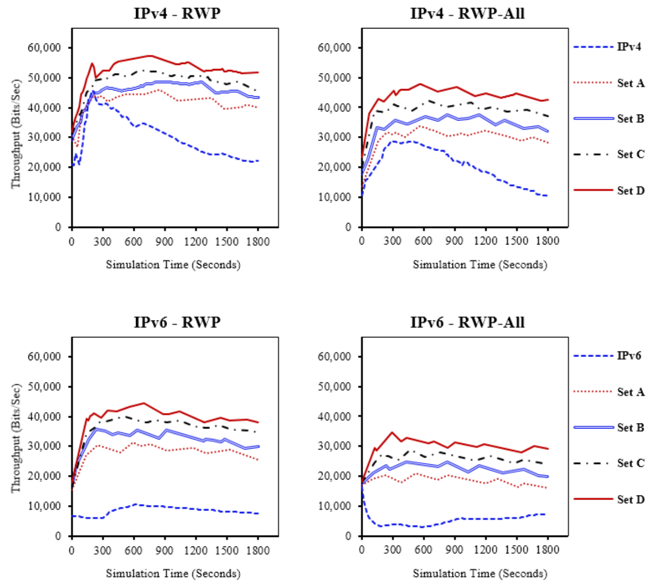

FIGURE 7. **Average routing traffic sent by the caller node in (Bits/Sec) for OLSR-based MANET using CLOLSR approach.**

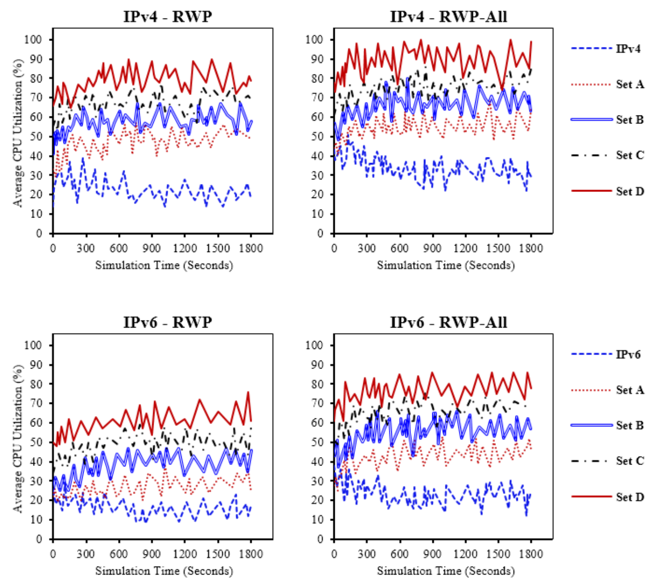

FIGURE 8. **Average routing traffic sent by the caller node in (Bits/Sec) for OLSR-based MANET using CLOLSR approach.**

with set B, between 1.62 to 2.54 Kbits/s with set C, and between 1.91 to 2.92 Kbits/s with Set D. In the RWP-All scenarios, the average of sent routing traffic is in the range of 1.48 to 1.57 Kbits/s, which have been increased with the implementations of the CLOLSR sets. The average number of the sent routing traffic increased to be between 1.75 to 2.67 Kbits/s with set A, between 2.26 to 2.97 Kbits/s with set B, between 2.41 to 3.99 Kbits/s with set C, and between 2.51 to 3.49 Kbits/s with set D. On the other hand, for IPv6 traffic, the average number of sent routing traffic from the caller node is between 500 to 960 bits/s with the RWP mobility model. The average number of the sent routing traffic have increased with the CLOLSR implementations to be between 0.98 bits/s to 1.51 Kbits/s with set A, between 1.05 to 1.67 Kbits/s with set B, between 1.11 to 2.28 Kbits/s with set C, and between 1.1 to 2.39 Kbits/s with set D. In the RWP-All scenarios, the average number of sent routing traffic from the caller node for IPv6 traffic is between 0.62 to 1.21 Kbits/s. The average number of the sent routing traffic have increased to be between 1.41 to 2.46 Kbits/s with set A, between 1.72 to 2.41 Kbits/s with set B, between 1.82 to 2.56 Kbits/s with set C, and between 1.92 to 2.98 Kbits/s with set D. The growth of the total routing traffic sent resulted from the increasing number of the HELLO and TC messages sent for the dynamic route adjustment processes of the OLSR parameters as a result of the implementations for the proposed CLOLSR approaches.

E. BANDWIDTH CONSUMPTIONS AND CPU UTILIZATION FOR CLOLSR IMPLEMENTATIONS

The implementations of the CLOLSR approaches for the three SIP processes have different levels of effect over the network performance in general and the SIP calls' agents in particular. The CLOLSR implementations affect the total consumed bandwidth in the MANET and the CPU utilization level of the B2BUA SIP server for the SIP-based VoIP calls. **FIGURE** represents the average consumed bandwidth for the OLSR-based MANET routing processes during the implementations of the CLOLSR for the investigated scenarios. For IPv4 traffic, the average consumed bandwidth for routing data in the OLSR-based MANET is between 21 to 43 Kbits/s over the RWP mobility model. The implementations of the CLOLSR have increased the average consumed bandwidth for the routing data to between 37 to 38.1 Kbits/s with set A, 29 to 49.3 Kbits/s with set B, 31.2 to 51.5 Kbits/s with set C, and 31.5 to 58.5 Kbits/s with set D. In the RWP-All scenarios, the average consumed bandwidth for the routing data of the IPv4 traffic is in the range of 10.5 to 29.7 Kbits/s, which increases with the CLOLSR implementations. The CLOLSR implementations have increased the average of total consumed bandwidth to be between 12.5 to 32.8 Kbits/s with set A, 17.2 to 38.3 Kbits/s with set B, 20.1 to 41.5 Kbits/s with set C, and 22.7 to 48.6 Kbits/s with set D.

The total consumed bandwidth for routing messages in the IPv6 OLSR scenarios over the RWP mobility model were in the range of 5 to 10 Kbits/s. The implementations of the CLOLSR have increased the total average of the consumed bandwidth to between 15.7 to 30.4 Kbits/s with set A, 15.7 to 46.4 Kbits/s with set B, 15.8 to 38.9 Kbits/s with set C, and 15.8 to 44.7 Kbits/s with set D. In the RWP-All scenarios, the average consumed bandwidth for the routing data of the IPv6 traffic is between 1.8 to 14.7 Kbits/s, which increases with the implementations of CLOLSR. The CLOLSR implementations have increased the average of the total consumed bandwidth to between 18.6 to 21.7 Kbits/s with set A, 18.9 to 24.7 Kbits/s with set B, 18.9 to 28.7 Kbits/s with set C, and 18.9 to 25.3 Kbits/s with set D.

During the implementations of the CLOLSR for SIP-based VoIP calls, the CPU utilization level for the B2BUA SIP server has variable increases depending on the identified values of the proposed sets. The increase in the CPU utilization level results is normal as the number of the running processes of the routing enhancement procedures of the proposed algorithms had also increased. **FIGURE 6** shows the average level of the CPU utilization for the B2BUA SIP server in the OLSR-based MANET during the simulation time for the investigated scenarios. For IPv4 traffic, the average CPU utilization level for the B2BUA SIP server is in the range of 12% to 39% per second for the total capacity of the CPU processes in the RWP mobility model. The CLOLSR implementations have increased the average CPU utilization level in the SIP server to between 29% to 54% per second with set A, 41% to 68% per second with set B, 58% to 78% per second with set C, and 64% to 89% per second with set D. For the IPv4 traffic over the RWP-All scenarios, the average CPU utilization level of the B2BUA SIP server is in the range of 22% to 52% per second from the total capacity of the CPU processes. The CLOLSR implementations have increased the average CPU utilization level in the SIP server to between 41% to 66% per second with set A, 47% to 79% per second with set B, 61% to 86% per second with set C, and 71% to 99% per second with Set D.

On the other hand, for IPv6 traffic, the average CPU utilization level for the B2BUA SIP server is in the range of 8% to 23% per second for the total capacity of the CPU processes in the RWP mobility model. With the implementations of the CLOLSR, the average CPU utilization percentage in the SIP server increased to between 18% to 38% per second with set A, 23% to 46% per second with set B, 33% to 58% per second with set C, and 48% to 74% per second with set D. In the RWP-All scenarios, the average CPU utilization level of the B2BUA SIP server for IPv6 traffic is in the range of 12% to 36% per second for the total capacity of the CPU processes which increases with the implementations of the CLOLSR sets. The CLOLSR implementations have increased the average CPU utilization level in the SIP server to between 25% to 52% per second with a set A, 37% to 66% per second with set B, 49% to 76% per second with set C, and 60% to 86% per second with set D.

IX. DISCUSSION ABOUT THE CLOLSR IMPLEMENTATIONS OVER SIP-BASED VoIP

The proposed CLOLSR approaches for the implementations of SIP-based VoIP applications over OLSR-based MANET have been investigated in the previous subsections. The implementations of the CLOLSR have considered the sets identified in Table 2. With the considerations of related performance metrics of RFC 6076 and the evaluation efforts of the current state, the evaluation of the CLOLSR implementations have been studied and compared regarding the related OLSR routing performance factors. At the beginning of the simulations, all the MANET nodes were used to build its own routing tables depending on the OLSR algorithms. This process consumes a high amount of bandwidth and requires a longer time for MANET with a high density of nodes as in the implemented scenarios. During the simulations, the initial TC routing values of the OLSR-based MANET have longer delays that were reduced until the system reached its initial level of stability with the required initial entries of the routing table values. Mostly, the MANET has reached its initial level of stability at 250 to 350 seconds from the beginning of the simulation. With continuous and dynamic mobility of MANET nodes, the update processes of the routing tables increases the number of sent HELLO and TC messages. However, this growth in the routing messages is not reflecting the actual health and efficiency of the MANET routes. In addition, the high number of route messages in OLSR does not mean that the reachability with the requested node/nodes does not exist. It means that the nodes used to update their routing tables based on the proactive nature of OLSR to fulfil with any requested routes in the MANET.

With the implementations of the CLOLSR algorithm over the three processes of the SIP-based VoIP calls, the OLSR routing parameters are relatively updated depending on the considered reachability delays or destination availability in the routing tables. The modifications on the routing parameters are related to the level of the dynamic adjustments in the routing table. The values of the HELLO and the TC intervals are used to modify the values of the Multiple Point Relay (MPR) of the OLSR routing protocol. Therefore, the MPR values are recorded to increase with the increments of the HELLO and TC values with regard to the OLSR MPR selection mechanism in Algorithm 1. With the implementations of the CLOLSR, the SIP server succeeded in providing better performance by increasing the frequent updates for the routing table entries of the correspondent nodes. The performance levels for the registration and termination processes with the CLOLSR implementations provide good enhancement levels when compared with the actual current signaling system for both IPv4 and IPv6 traffic for the RWP and RWP-All mobility models. The call setup processes observed to have also an acceptable level of enhancement over the RWP mobility model. However, for RWP-All, the call setup processes showed variable levels of enhancement with CLOLSR implementations, especially with IPv6 traffic. The CLOLSR approaches also save the CPU cycles and reduce the consumed bandwidth by returning the MPR values for the correspondent nodes to their original default values. This happens just when the SIP calls' processes reach the required level of the performance enhancement. However, the results show that the CLOLSR implementations have consumed larger amounts of bandwidth with higher levels of CPU utilizations. This growth happened as a result of the frequent adoptions in the routing parameters because of the mobility of the correspondent nodes. The successful adoptions for the OLSR routing parameters depend on the application layer values to enhance the SIP signaling system performance that increased the level of CPU utilization and the consumed bandwidth. The call initiation process consumed the largest amount of traffic and used higher CPU cycles comparing with the registration and termination processes.

The OLSR routing efficiency during the implementations of the SIP-based VoIP calls reflects on the general performance of the SIP signaling system. The CLOLSR used a frequent adoption for the number of the HELLO and Topology Control (TC) messages to reduce the route discovery time between the MANET nodes with a dynamic mobility nature. The dynamic routing enhancements of the CLOLSR approaches improve the efficiency of the Multipoint Relay (MPR) sets for the OLSR routing protocol. Therefore, with the implementations of the CLOLSR, the MPR selection mechanism will be enhanced with a dynamic modification for the values of the OLSR routing table. The CLOLSR provides a reliable detection system for any raised delays or undeliverable SIP messages over both RWP and RWP-All mobility models for the SIP processes. The proactive routing nature of the OLSR is still the main issue within the implementations of the CLOLSR approaches over MANET nodes. This can be enhanced through reducing the number of required second hops during the route discovery processes to provide routes with shorter update periods.

X. CONCLUSION AND FUTHURE WORK

This paper represented an evaluation and comparison study for SIP signaling and VoIP performance metrics over OLSR for both IPv4 and IPv6 MANET. As IP networks still cannot meet the required QoS of VoIP, therefore, the QoS of VoIP improved by controlling the values of these parameters to be within the acceptable range. The differences for the call setup delays for SIP based VoIP applications over IPv4 and IPv6 are in different ranges depend on network system, bandwidth, and the connectivity statue. The VoIP metrics such as end-to-end delays, jitter, throughput, and packet loss are quite comparable for both IPv4 and IPv6 over RWP and RWP-All MANET scenarios. Most of the successful VoIP calls during MANET mobility scenarios occur in the first half of the simulation as the nodes' initial positions provide better connectivity and reachability before they began moving.

The proposed CLOLSR algorithms provide better SIP signaling performance, and flexible adoptions for routing parameters depending on the SIP application status. In addition, the proposed approaches are better than the regular basic SIP signalling system and other related solutions from the literature in terms of the efficiency, the implementations' flexibility, and the QoS criterions. The usages of the determined sets in the proposed Cross-Layer approaches are considered from the benchmarking efforts. These sets were proposed with regard to the required level of enhancements, where sets with too short values could not allow the SIP processes to be generated to provide good services. The CLOLSR provide a dynamic reachability nature for the correspondent nodes to reduce the connectivity delays, save the CPU cycles, and reduce the bandwidth. In addition, these approaches are applicable for both IPv4 and IPv6 implementations over MANET.

Over all the investigated proposed sets of the benchmarked values for both CLOLSR implementations. Therefore, as much as the set values for the CLOLSR approaches reduced the performance level of the SIP signaling system, bandwidth consumptions and CPU utilization of the SIP server increased. In addition, the average of consumed bandwidth for IPv4 scenarios for CLOLSR implementations is slightly lower compared with the consumed bandwidth in IPv6 because of the packet overhead of IPv6 traffic that slightly increased the amount of the consumed bandwidth. In addition, the IPv6 implementations in this research study do not support the mobility features for the mobile nodes.

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Future works could study and improve the SIP registration and retransmission timers over MANET to enhance the SIP signaling and QoS for SIP-based applications over MANET reactive/proactive routing protocols. This would include the topology modelling for the SIP-base VoIP applications over MANET. In addition, study and enhance the SIP signaling system over TCP for both IPv4 and IPv6 as the evaluation results of the SIP/TCP performance showed bad performance over the implemented random mobility models. Furthermore, the retransmission timers for SIP/TCP signaling over MANET are still an open research issue. The retransmission of the SIP/TCP messages causes the duplication, latency, and delays for the INVITE/Re-INVITE messages that impact the SIP call setup processes. Employing the benchmarked values of the RFC 6076 for SIP-based VoIP applications in other platforms such as WLAN and LTE is also a valuable investigation that needs to be added. Lastly, it is worth referring that we are driven to prefer investigating SIP signaling and its performance more than concerning over TCP/UDP due to the importance of SIP applications in terms of Vice over Internet (VoIP) services. In the sense that, the main application for SIP is the ability to provide Vice over Internet (VoIP) services, where it routes telephone calls from a client's to the public switched telephone network (PSTN). SIP-enabled video surveillance cameras can initiate calls to alert the operator of events, such as motion of objects in a protected area. SIP is used for broadcasting to platform independent machines.

REFERENCES

[1] *OPNET Modeler 17.1 Documentation*, OPNET Technol., Bethesdsa, MD, USA, Oct. 2011.

[2] A. M. Abbas and O. Kure, "Quality of service in mobile ad hoc networks: A survey," *Int. J. Ad Hoc Ubiquitous Comput.*, vol. 6, no. 2, pp. 75–98, 2010.

[3] M. K. Gulati and K. Kumar, "QoS routing protocols for mobile ad hoc networks: A survey," *Int. J. Wireless Mobile Comput.*, vol. 5, no. 5, pp. 107–118, 2012.

[4] A. B. Johnston, Ed., *SIP: Understanding the Session Initiation Protocol*. Norwood, MA, USA: Artech House, 2009.

[5] M. Goyal *et al.*, "Improving convergence speed and scalability in OSPF: A survey," *IEEE Commun. Surveys Tuts.*, vol. 14, no. 2, pp. 443–463, 2nd Quart., 2012.

[6] J. Kumar, A. Singh, M. K. Panda, and H. S. Bhadauria, "Study and performance analysis of routing protocol based on CBR," *Procedia Comput. Sci.*, vol. 85, pp. 23–30, Jan. 2016.

[7] E. Gurumoorthi, A. Ayyasamy, M. Archana, and J. V. Barathy, "Performance enhancement for QoS in VoIP applications over MANET," *J. Adv. Comput. Electron. Eng.*, vol. 2, no. 5, pp. 47–54, 2017.

[8] P. Kuppusamy, K. Thirunavukkarasu, and B. Kalaavathi, "A study and comparison of OLSR, AODV and TORA routing protocols in ad hoc networks," in *Proc. 3rd Int. Conf. Electron. Comput. Technol.*, Apr. 2011, pp. 143–147.

[9] A. Shrestha and F. Tekiner, "On MANET routing protocols for mobility and scalability," in *Proc. Int. Conf. Parallel Distrib. Comput., Appl. Technol.*, Dec. 2009, pp. 451–456.

[10] D. E. M. Ahmed and O. O. Khalifa, "Performance evaluation of enhanced MANETs routing protocols under video traffics, for different mobility and scalability models using OPNET," *Amer. J. Eng. Res.*, vol. 6, no. 7, pp. 329–346, 2017.

[11] S. Mohapatra, P. Kanungo, "Performance analysis of AODV, DSR, OLSR and DSDV routing protocols using NS2 simulator," *Procedia Eng.*, vol. 30, pp. 69–76, 2012.

[12] J. Yi, A. Adnane, S. David, B. Parrein, "Multipath optimized link state routing for mobile ad hoc networks," *Ad Hoc Netw.*, vol. 9, no. 1, pp. 28–47, Jan. 2011.

[13] M. Zuhairi, H. Zafar, and D. Harle, "The impact of mobility models on the performance of mobile ad hoc network routing protocol," *IETE Tech. Rev.*, vol. 29, no. 5, pp. 414–420, 2012.

[14] M. Alshamrani, H. Cruickshank, Z. Sun, V. Fami, B. Elmasri, E. Danish, "Signaling performance for SIP over IPv6 mobile ad-hoc network (MANET)," in *Proc. IEEE Int. Symp. Multimedia*, Dec. 2013, pp. 231–236.

[15] T. Clausen and P. Jacquet, *Optimized Link State Routing (OLSR)*, document RFC 3626, IETF Networking Group, 2003.

[16] H. Sinnreich and A. B. Johnston, *Internet Communications Using SIP: Delivering VoIP and Multimedia Services With Session Initiation Protocol*. Hoboken, NJ, USA: Wiley, 2006.

[17] S. K. Singh *et al.*, "Performance impact of TCP and UDP on the mobility models and routing protocols in MANET," in *Intelligent Computing, Networking, and Informatics*. New Delhi, India: Springer, 2014, pp. 895–901.
[18] D. Malas and A. Morton, *Basic Telephony SIP End-to-End Performance Metrics*, document RFC 6076, 2011.
[19] J. Rosenberg, *SIP: Session Initiation Protocol*, document RFC 3261, 2002.
[20] U. Lee, J. Lee, J.-S. Park, and M. Gerla, "FleaNet: A virtual market place on vehicular networks," *IEEE Trans. Veh. Technol.*, vol. 59, no. 1, pp. 344–355, Jan. 2010.
[21] M. Voznak and J. Rozhon, "SIP back to back user benchmarking," in *Proc. 6th Int. Conf. Wireless Mobile Commun.*, Sep. 2010, pp. 92–96.
[22] S. Thomson and T. Narten, *IPv6 Stateless Address Auto Configuration*, Draft Standard 2462, Internet Engineering Task Force, 1989.
[23] S. Sendra, P. Fernandez, C. Turro, and J. Lloret, "IEEE 802.11a/b/g/n indoor coverage and performance comparison," presented at the 6th Int. Conf. Wireless Mobile Commun., Sep. 2010, pp. 185–190.
[24] V. Sharma, H. Singh, M. Kaur, and V. Banga, "Performance evaluation of reactive routing protocols in MANET networks using GSM based voice traffic applications," *Optik*, vol. 124, no. 15, pp. 2013–2016, Aug. 2013.
[25] D. Skordoulis, Q. Ni, H.-H. Chen, A. P. Stephens, C. Liu, and A. Jamalipour, "IEEE 802.11n MAC frame aggregation mechanisms for next-generation high-throughput WLANs," *IEEE Wireless Commun.*, vol. 15, no. 1, pp. 40–47, Feb. 2008.
[26] E. Hyytia, P. Lassila, and J. Virtamo, "Spatial node distribution of the random waypoint mobility model with applications," *IEEE Trans. Mobile Comput.*, vol. 5, no. 6, pp. 680–694, Jun. 2006.
[27] D. Mitsche, G. Resta, and P. Santi, "The random waypoint mobility model with uniform node spatial distribution," *Wireless Netw.*, vol. 20, no. 5, pp. 1053–1066, Jul. 2014.
[28] F. Gao, Q. Liu, and H. Zhan, "Research of SIP DoS defense mechanism based on queue theory," in *Proc. Int. Conf. Comput. Sci., Environ., Ecoinformatics, Educ.*, Berlin, Germany, 2011, pp. 99–104.
[29] *Full Rate Speech; Transcoding (GSM 06.10)*, Standard ETSI EN 300 961 v. 8.1.1, 2000.
[30] K. Pandey and A. Swaroop. (2011). "A comprehensive performance analysis of proactive, reactive and hybrid manets routing protocols." [Online]. Available: https://arxiv.org/abs/1112.5703
[31] M. Happenhofer, C. Egger, and P. Reichl, "Quality of signalling: A new concept for evaluating the performance of non-INVITE SIP transactions," Sep. 2010, pp. 1–8.
[32] J. Singh and R. Mahajan, "Performance analysis of AODV and OLSR using OPNET," *Int. J. Comput. Trends Technol.*, vol. 5, no. 3, pp. 114–117, 2013.
[33] A. Gupta, "Performance analysis of MANET routing protocols in different mobility models," *Int. J. Inf. Technol. Comput. Sci.*, vol. 5, no. 6, pp. 73–82, 2013.
[34] H. Kaur and E. J. Singh, "Performance comparison of OLSR, GRP and TORA using OPNET," *Int. J. Adv. Res. Comput. Sci. Softw. Eng.*, vol. 2, no. 10, pp. 260–267, 2012.
[35] M. Thompson, "A method of proactive MANET routing protocol evaluation applied to the OLSR protocol," in *Proc. 6th ACM Int. Workshop Wireless Netw. Testbeds, Exp. Eval. Characterization*, 2011, pp. 27–34.
[36] D. Tepšić and M. Veinović, "Comparative analysis of unicast routing protocols in MANET networks," in *Proc. Int. Sci. Conf. IT Bus.-Rel. Res.*, 2015, pp. 1–4.
[37] M. Voznak and J. Rozhon, "Approach to stress tests in SIP environment based on marginal analysis," *Telecommun. Syst.*, vol. 52, no. 3, pp. 1583–1593, Mar. 2013.
[38] P. Palta and S. Goyal, "Comparison of OLSR and TORA routing protocols using oPNET modeler," *Int. J. Eng. Res. Technol.*, vol. 1, no. 5, pp. 984–990, 2012.
[39] M. Handley, *SDP: Session Description Protocol*, document RFC 4566, IETF, 2006.
[40] S. Narayan and Y. Shi, "Application layer network performance evaluation of VoIP traffic on a test-bed with IPv4 and IPv6 LAN infrastructure," in *Proc. IEEE Region 8th Int. Conf. Comput. technol. Electr. Electron. Eng. (SIBIRCON)*, Jul. 2010, pp. 215–219.
[41] S. S. Kolahi, N. Chand, B. K. Soorty, and Z. Qu, "Performance comparison of IPv4 and IPv6 in peer-peer and client server local area networks," in *Proc. 10th IEEE Int. Conf. Comput. Inf. Technol.*, Jun./Jul. 2010, pp. 41–45.
[42] S. Massner and C. Richter, "SIP trunking—General requirements for interconnecting enterprise networks," *J. Netw.*, vol. 8, no. 10, pp. 2195–2212, 2013.
[43] T. Hoeher, M. Petraschek, S. Tomic, and M. Hirschbichler, "Evaluating performance characteristics of SIP over IPv6," *J. Netw.*, vol. 2, no. 4, pp. 40–50, 2007.
[44] M. Alshamarani, "Enhancing SIP signaling system over MANET," Ph.D. dissertation, Fac. Eng. Phys. Sci., Univ. Surrey, Guildford, U.K., 2015.

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