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Research Article

A New QoS Management Scheme for VoIP Application over Wireless Ad Hoc Networks

Said El brak,[1] Mohamed El brak,[1] and Driss Benhaddou[2]

[1] *Department of Computer Science, Abdelmalek Essaadi University, 90100 Tangier, Morocco*
[2] *Department of Engineering Technology, Houston University, Houston, TX 77004, USA*

Correspondence should be addressed to Said El brak; elbraks@gmail.com

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Nowadays, mobile ad hoc networks (MANETs) have to support new applications including VoIP (voice over IP) that impose stringent QoS (quality of service) constraints and requirements. However, VoIP applications make a very inefficient use of the MANET resources. Our work represents a first step toward improving aspects at the network layer by addressing issues from the standpoint of adaptation, claiming that effective adaptation of routing parameters can enhance VoIP quality. The most important contribution is the adaptive OLSR-VA algorithm which provides an integrated environment where VoIP activity is constantly detected and routing parameters are adapted in order to meet the application requirements. To investigate the performance advantage achieved by such algorithm, a number of realistic simulations (MANET scenarios) are performed under different conditions. The most important observation is that performance is satisfactory in terms of the perceived voice quality.

1. Introduction

Mobile ad hoc networks (MANETs) are emerging wireless technologies, which can be flexibly and conveniently deployed in almost any environment. MANETs are self-forming and self-healing, enabling peer-level communications between mobile nodes without relying on centralized resources or fixed infrastructure. Voice over IP (VoIP) is one of the fastest growing applications in networking. As wireless components spread, VoIP over wireless networks is becoming increasingly important and supporting voice over ad hoc networks is part of realizing an all-IP goal. Additionally, MANET's attributes enable providing VoIP services in virtually any scenario key applications including disaster recovery, heavy construction, mining, transportation, defence, and special event management.

Naturally, users demand to be able to use the same VoIP services independently of the access network. However, many QoS issues remain unsolved in infrastructure-based networks. So how it will be for the MANETs which combine many challenges? At first, the wireless channel introduces constraints including its inherent broadcast nature and temporal response variability due to fading, absorption, and noise and interference sensitivity. In addition, ad hoc networks suffer from scarcity of resources, lack of a central entity, and volatility of connections. These challenges create new performance limitations for VoMAN (voice over MANET) and make it a new exciting task.

In this paper, we propose a self-adapting routing protocol that adapts its parameters to the VoIP load in the network. It also describes how the rate of HELLO packets is adapted according to the parameters of the voice codec. To the best of our knowledge, adapting the routing parameters to media transmission has never been addressed before.

This paper is structured as follows. Section 2 briefly presents the state of the art. In Section 3, the OLSR routing protocol is presented, and the impact of tuning its parameters on VoIP is discussed. Section 4 illustrates how to make OLSR protocol aware of VoIP by introducing the new adaptive algorithm OLSR-VA. Section 5 presents the performance evaluation method and metrics. Section 6 summarizes the most important simulation results and their interpretation. Finally, Section 7 concludes this paper and also presents some future works.

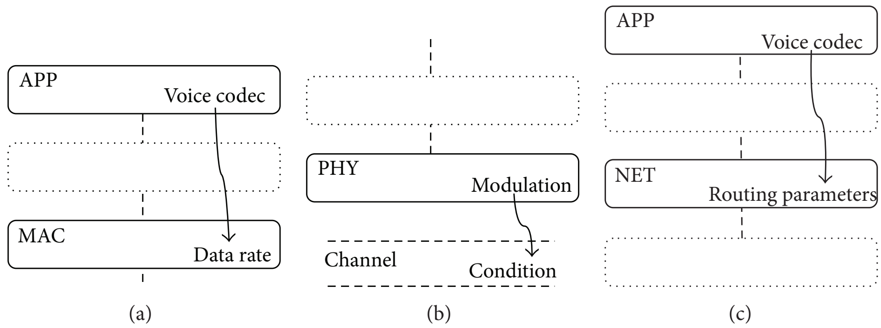

Figure 1: Adaptation techniques.

2. State of the Art

The first QoS solutions for VoIP over MANETs have addressed traditional QoS guaranteeing mechanisms already designed for WLAN, such as resource reservation [1], admission control techniques [2], and differentiated services (802.11e standard) [3], which can guarantee some minimum QoS for multimedia traffic. All these solutions have shown inability to improve the quality of VoIP calls carried by a MANET.

The link and codec adaptation mechanisms [4, 5] were one step ahead to construct QoS solution for VoIP over multirate wireless link. These mechanisms are providing resilience and robustness in cellular networks [6]. Hence, it is required to provide MANETs with the same capabilities. Adaptation mechanisms can be done at several layers of the network protocol stack. Well-known examples of adaptation mechanism for VoIP over wireless media are *PHY/Channel adaptation* which allows each mobile node to adapt its transmission rate dynamically to channel condition [7] (Figure 1(a)) and *APP/MAC adaptation* which adapts the codecs of the active voice flows to the new network conditions (data rate) (Figure 1(b)). The last one is considered one of the most effective solutions of the multirate problem on VoIP calls [4, 5].

In [8], the authors propose a solution that allows the use of VoIP services in distributed wireless networks using a metric based on an objective measure of the QoS of VoIP. In this work authors do not address the routing issue which is the major concern in ad hoc wireless networks. Although the performance of the solution has been evaluated in a low dynamic environment (less than 10 m/s), that might not perfectly reflect a real MANET.

Routing layer is one of the major key components for the MANET connectivity and performance. Nevertheless, few efforts have focused on the impact of routing protocols on voice traffic over MANETs. Furthermore, adaptation mechanisms have to deal with routing plan and integrate routing protocol in their architecture. Based on these considerations, and to further improve the performance of voice over MANETs, we propose an adaptation mechanism that integrates routing in the solution. As mentioned in Figure 1(c), our proposal for quality of service in VoIP over MANET has focused on adaptation between network layer and application layer (*APP/NET adaptation*) by integrating self-adaptation mechanism in the routing protocol.

In this study, we do not implement a full cross-layer design, but we integrate a layer triggers mechanism [9] that predefined signals to notify events such as VoIP activity and codec settings. The implemented triggering mechanism modifies smoothly the interfaces between the layers to accommodate the messages from the application layer (VOIP) passed to the network layer (OLSR-VA routing).

3. Optimized Link State Routing Protocol

3.1. Overview. OLSR, presented in [10], concentrates on routing in ad hoc networks. It is a table-driven proactive routing protocol for MANETs based on the link state approach and uses the shortest path first (SPF) algorithm. Due to its proactive nature, it has the advantage of making the routes ready before they are needed by exchanging topology information with other nodes of the network periodically. OLSR provides the following optimizations to the classic link state algorithm.

(i) It reduces the size of control packets by implementing multipoint relays (MPRs).

(ii) It minimizes flooding of control traffic by only permitting selected nodes, MPRs, to send control traffic through the network and does not generate extra control traffic in response to link failures or arriving nodes.

Each node has to maintain a list of nodes qualified as MPRs. MPRs are used in the flooding mechanism in order to reduce the flooding process and to periodically maintain the topology. Figure 2 compares MPR flooding to full flooding as used by classic flooding mechanisms.

OLSR simply relies on the soft-state approach to maintain the consistency of topology information among nodes in the network. Apart from the periodic control messages, OLSR does not generate extra control traffic in response to link failure and nodes joining/leaving events. OLSR deploys two essential signalling messages: HELLO messages and TC messages.

HELLO Messages. These messages are sent only for one hop and serve to discover neighbours, which are localized in

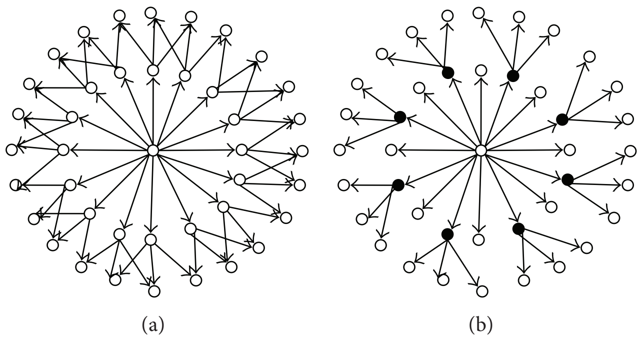

Figure 2: Comparison of full flooding (a) and MPR flooding (b).

the range of the local node. Links can be symmetric or asymmetric; OLSR considers that two nodes which are two-hop neighbours (via another node) cannot behave as neighbours even if links are symmetric. The HELLO message contains a list of neighbours and the status of each corresponding link. For this reason, HELLO messages are considered as an immediate informative packet which updates the quality of links of the neighbourhood of a given node. In addition, when nodes leave the network, or the links between them get broken, the corresponding link state and the neighbour state are removed after the state timeout timers expire without receiving any HELLO messages.

TC Messages (Topology Control). Each node periodically broadcasts TC messages to declare its MPR selector set and populate its topology table. Node records information about the topology of the network as obtained from TC messages. Based on such information, the routing table is calculated based on the SPF algorithm. In addition, periodic TC messages help the remote nodes recover from loss of topology information caused by state inconsistencies or node restarts.

A routing table is kept at each node and contains routes to all other destinations in the network. This table is built by tracking connected pairs (i.e., pairs which link status are bidirectional) in the topology table. In order to obtain optimal paths, only connected pairs are selected on the minimal path. There is no entry for destinations whose routes are broken or are not fully known. Route table entries contain the destination address, next-hop address, and estimated distance to destination (in number of hops).

3.2. Impact of Tuning OLSR Routing Parameters on VoIP. In our previous study [11], we aimed to gain a better understanding of tuning OLSR routing parameters impact on VoIP codecs. Through simulations, we have manually tuned two OLSR routing parameters and tested their impact on VoIP quality. These parameters are as follows.

Refresh Interval Timers (HELLO Interval). This parameter is used for periodic updates of link information and for subsequent topology maintenance. As results when decreasing the refresh interval this leads to an increase in routing control packet and thus the network bandwidth will be reduced. As known, VoIP codecs have different bandwidth requirements according to their voice payload and sample interval. So, configuring HELLO interval to VoIP codec requirements may optimize the overall network bandwidth and increases VoMAN performance.

Willingness. The willingness is a parameter value announced by nodes participating in OLSR routed network to act as relays for OLSR control traffic for their neighbours. The MPR selection algorithm selects first the nodes with the highest willingness. OLSR has eight values available for the willingness (from 0 “WILL NEVER” to 7 “WILL ALWAYS”). Simulation shows the dependencies between this parameter and VoIP activities in the network. More specifically, for VoIP nodes low value of willingness allows them to preserve resources for VoIP task. It may be possible to tune the willingness dynamically, during operation, by detecting node VoIP activity and configuring their willingness to WILL_LOW.

These results provide useful insights into how an adaptive routing protocol can be designed in the context of VoIP over mobile ad hoc networks. Such adaptive routing protocol requires further investigation.

4. OLSR-VA: Making OLSR Aware of VoIP

4.1. Proposal Approach. Our proposal for quality of service in VoIP over MANET has focused on adaptation between network layer and application layer by integrating self-adaptation mechanism in OLSR routing protocol. We propose a novel model for the OLSR protocol which exploits the signalling traffic to disseminate VoIP activity information in the network. Accordingly, each network element (node) proactively has knowledge about VoIP activities which happened in the MANET. The adaptation process is based on two actions as follows.

Monitoring. Feedback from application layer (VoIP activity and codec used) and network layer (routing);

Adaptation. Self-tuning OLSR routing parameters (HELLO interval and willingness).

Consider a VoMAN scenario (Figure 3), with a number of VoIP calls active (calls 1 and 2). Consider now a new VoIP call (call 3) which will be initiated between two users (nodes A and F). The audio codec is selected according to available resources (e.g., bandwidth). The OLSR routing protocol will find routes for VoIP traffic based on the shortest hop count paths first algorithm (e.g., path A-B-D-E-F) and will not care about VoIP load on this path. However, this leads to network parts with heavy VoIP load more than others, presenting a high level of radio interference or a high level of congestion. The purpose of the extended OLSR protocol is to balance VoIP load in the network by routing VoIP packet over the less loaded paths even if requiring more hop count (e.g., choose path A-G-H-I-J-F). This process will reduce channel interference and congestion, and consequently the QoS will be improved.

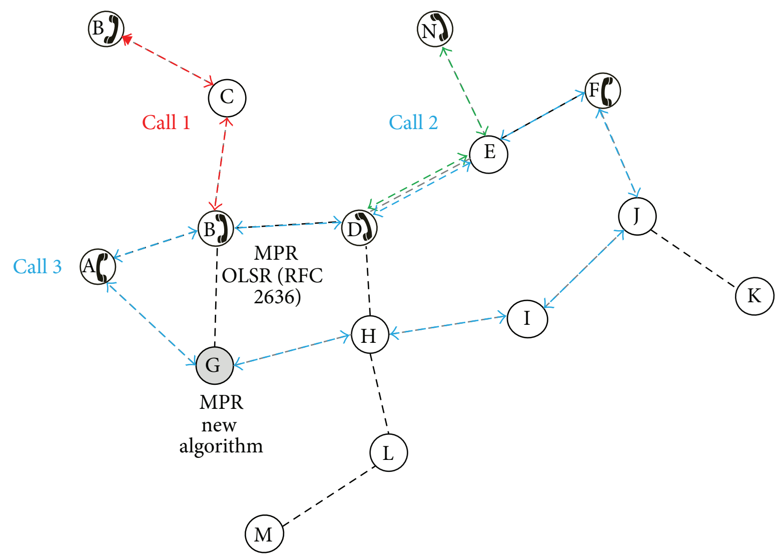

Figure 3: VoMAN scenario.

This solution will make better bandwidth utilization: on the one hand, by adapting signalling overhead associated with OLSR protocol and, on the other hand, by transmitting voice packets via part of the network which has less VoIP activities. A key idea behind this solution is that the proposed solution can provide a part from a general QoS management architecture for VoIP over MANETs.

4.2. OLSR-VA Protocol Design. Successful voice transmission over MANET needs the design of a routing protocol which takes in mind VoIP activity (VA) and the audio codec requirements. This codec may be changed according to network condition. We propose an adaptive proactive routing algorithm that applies adaptation functions so as to achieve optimum voice quality with less control overhead. Essentially, the protocol's behaviour (i.e., parameters) is tuned according to VoIP activity and audio codec configurations. The basic structure of this proposal can be seen in a detailed algorithm flow chart in Figure 4.

In OLSR-VA, the monitoring phase is a constant procedure for triggering, focusing particularly on three types of events: local VoIP activity detection (local VoIP monitor), neighbour VoIP activity detection (neighbour VoIP monitor), and routing performance (routing monitor). In the initial start-up phase, a node may have event triggers from the upper layer, informing a VoIP session establishment or a new received HELLO message which contains new information about VoIP session established in neighbourhood. Hence, the basic algorithm is based on three major "*monitoring-adaptation*" processes.

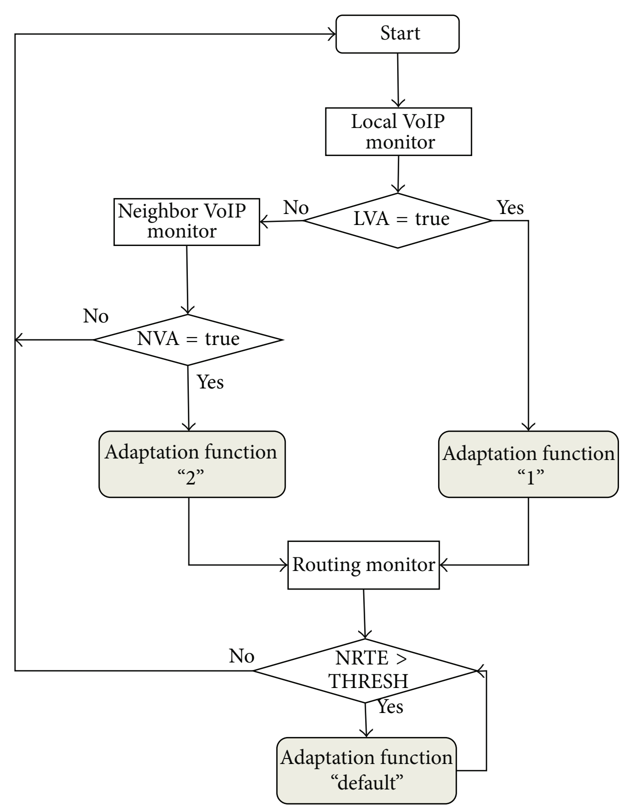

Figure 4: OLSR-VA algorithm flow chart.

(i) *Monitoring Local VoIP Activity- (LVA-) Adaptation Function "1."* In this process, the node monitors VoIP activity and codec changes from the upper layers and then, if appropriate, applies adaptation function "1" or goes to the next monitoring process.

(ii) *Monitoring Neighbour VoIP Activity- (NVA-) Adaptation Function "2."* In this process, the node monitors VoIP activity and codec changes from its neighbourhood and then, if appropriate, applies adaptation function "2."

```
    Input: node i, Local VoIP Activity LVA_i.
    Output: HELLO interval of i H_i
(1) Begin
(2)   if LVA_i = true then
(3)     read codec parameters: sample_size, sample_interval
(4)     if sample_size is high then
(5)       Increase H_i
(6)       else if sample_interval is high then
(7)         decrease H_i
(8)       end if
(9)     end if
(10)  end if
(11)  W_i ← WILL_LOW
(12)  return H_i
(13) End
```

Algorithm 1: OLSR-VA_LVA_adaptation_function(i, LVA_i).

(iii) *Monitoring Routing Performance-Adaptation Function "Default."* This process occurs when adaptation function "1" or "2" is applied. The node monitors the routing performance and then, if appropriate, applies the default adaptation function.

Each of these processes is analyzed next in detail.

(i) LVA Monitoring-Adaptation Function "1." Normally, OLSR sets up and maintains routes regardless of application layer communication demands. OLSR-VA monitors VoIP activity from upper layers using layer triggers mechanism. OLSR-VA is notified about a local VoIP session establishment and the audio codec used from a VoIP signalling protocol such as SIP [12]. If a local VoIP activity is detected, OLSR-VA has to adapt its routing parameters (HELLO interval and willingness) and disseminate VoIP information to neighbourhood. The resulting solution is presented in Algorithm 1.

HELLO interval is adapted according to the codec configuration (sample size and sample interval). When using codec with high sample size, MANET must guarantee bandwidth. However, routing control packets consume a good part of this one. Therefore, increasing HELLO interval leads to some bandwidth preservation, obviously without affecting routing protocol performance. Codecs with small sampling interval generate voice frames in shorter time interval. So, high overhead is required to identify an appropriate path from the sender to the receiver.

Willingness is a willing or interest of the node in the ad hoc network to give a contribution or commitment to the other nodes in order to send a data in the network. In OLSR-VA, each node can declare an appropriate willingness. We decided to base the willingness selection on VA metrics. *Willingness* is set to *WILL_LOW* if the node has LVA = true. In this case, it will announce its inability to carry VoIP traffic on behalf of other nodes which makes provisions of their resource.

(ii) NVA Monitoring-Adaptation Function "2." The neighbour VoIP monitor is activated if there is no local VoIP session. The monitor simply processes HELLO message received from neighbours, which contains information about VoIP activity in the neighbourhood. If the NVA monitor detects a VA, OLSR-VA applies the same adaptation functions as described in previous section; only the willingness is set to *WILL_DEFAULT*.

This function seems to implement the same policy as the adaptation function "1." However, the modification of willingness leads to heuristic routing mechanism using a new algorithm for MPR computation. In OLSR (RFC 2636), each node selects their MPR permitting to compute the routes based on the knowledge of state of the network. The original heuristic for MPRs selection constructs the *MPR-set* that enables a node to reach any node in the symmetrical strict 2-hop neighbourhood through relaying by one MPR node with willingness different from *WILL_NEVER*. However, in OLSR-VA, heuristic routing allows a measure of route optimization based on recent knowledge of the state of the network and also VoIP activities occurring on it (VA metric). Therefore, the MPR must be selected in such a way that they will ensure voice transmission over links presenting the less VoIP activity.

(iii) Routing Monitoring-Adaptation Function "Default." Routing monitor is activated after one of the adaptation processes discussed above is applied. As previously described, OLSR-VA is an adaptive routing protocol which adapts their parameters with respect to events (LVA and NVA). If this adaptation leads to routing performance degradation, OLSR-VA must tune their parameters to default ones (that of RFC 2636 standard [10]).

NRTE metric is used to evaluate routing performance. This metric represents the fraction of dropped packets by no available route per the total number of sent packets. According to [13], the routing protocol is unable to forward packets to their destinations if the NRTE is greater than

Input: node i, number of dropping packet cause of no available route reason NRTE_i.
Output: default parameters values

Begin
  if LVA_i = true or NVA_i = true **then**
    for $t = \mathit{Current\_time}$ **to** (5)
      Calculate AVG(NRTE_i)
    End for
    While AVG(NRTE_i) > 10% **do**
      $H_i \leftarrow$ defout_values
      $W_i \leftarrow$ defout_values
    End while
  End if
End

Algorithm 2: OLSR-VA_Default_adaptation_function(i, NRTE_i).

a defined threshold. We calculate the NRTE ratio each 5 seconds, and if the average is greater than 10%, the default adaptation function (OLSR RFC-2636 default parameters) will be applied. The LVA and NVA monitoring cannot be activated until achieving an average less than 10%. The resulting solution is presented in Algorithm 2.

4.3. Dissemination of VoIP Information. As previously explained, to establish and maintain the OLSR repositories information, a number of different OLSR messages are defined and exchanged periodically by the nodes participating in the network. Together they form the OLSR control traffic. OLSR exchanges periodic HELLO messages and collects 2-hop neighbourhood and MPR information to be able to construct the routes. This mechanism can be easily extended to carry the VoIP information as well.

To implement OLSR-VA core functionality new HELLO message type has to be supported. Thanks to reserved fields, original HELLO message structure can be easily extended to supply the necessary information of VoIP activities for the originating node itself and its listed 1-hop neighbours. The proposed modified message structure is shown in Figure 5.

We propose using the first half of the reserved field within the local information section for signalling local VoIP activity (LVA) (1 byte) and the second half for voice codec used (1 byte). Hence, in addition to the original local information (such as willingness and Htime), node also includes its own VA information in the HELLO message. The reserved field in the link information section is used for signalling 1-hop neighbours VoIP activity (NVA) (1 byte); that is, upon receiving a HELLO message from its neighbour, a node reads the neighbour LVA value and includes its neighbours VA information in the HELLO messages.

As a result, through the exchange of this new HELLO message structure, there is no extra overhead introduced as the unused parts of HELLO messages are utilized for the dissemination of the VoIP activity information.

5. Performance Evaluation: MANET Scenario Case Study

What advantage might adaptation offer? In order to answer this question, in this section, we study the performance of

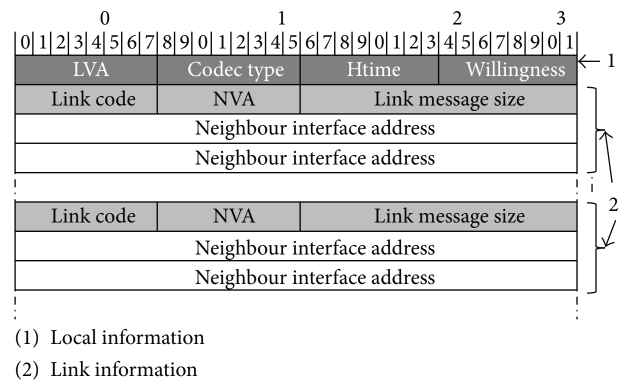

Figure 5: OLSR-VA HELLO message.

OLSR-VA considering a scenario of MANET environment. We chose this scenario because it presents a critical situation where voice transmission is needed and the network infrastructure is unavailable.

The proposed solution is implemented and evaluated using the ns-2 network simulator [14] version 2.35. We compare the performance of our scheme with that of the original protocol to show the improvements in the performance. The performance metrics include two metrics (quantitative and qualitative metrics) which will be defined later.

5.1. Simulation Scenario Description. In this study, a mobility traffic generator was used combined with ns-2, aiming at a significant level of simulation accuracy. The traffic simulator is needed to generate realistic mobility traces, used as an input for the network simulator.

To generate trace files reflecting nodes movements, we consider typical scenarios that focus on the unicast transmission of voice signals between nodes moving at rate of 0–20 km/h, with an average internode distance of 50 to 200 meters in increments of 7 meters. In order to analyze how various conditions affect the quality of the voice, two scenarios are considered (details are summarized in Table 1). Each scenario defines a network area size, which is simulated with varying conditions: size area, VoIP traffic, and densities.

Table 1: MANET simulation scenarios.

	Scenario "1"			Scenario "2"		
Area size		$2500\,m^2$			$5000\,m^2$	
Vehicles	10	→	30	30	→	60
VoIP calls	3	→	10	10	→	20

Table 2: Simulation and protocol parameters.

Simulation parameter	Value or protocol
Simulator	Ns-2.35
Simulation hardware	Intel C2, 2 GHz, 4 GB RAM
Simulation time	500 s
Simulation warm-up time	100 s
Transmission range	100 m
Fading model	Shadowing
Topology model	Random
Call duration	30 s
Protocol parameter	Value
Routing protocol	OLSR (3626)/OLSR-VA
PHY/MAC protocol	802.11b
Transport protocol	RTP/UDP
Buffer size	100 packets, drop-tail queuing
Application layer	Ns2voip++
Voice codecs	ITU-T codecs standard
Talkspurt/silence periods	Weibull distribution
Default OLSR-VA parameter	Value
HELLO interval	3 s
TC interval	5 s
Willingness	3
LVA	False
NVA	False
NRTE_TRESH	100

In application level, *Ns2voip++* modules [15] were used to generate VoIP traffic. The VoIP source is configured to draw the duration of the talkspurt and silence periods from *Weibull* distribution. A number of different codecs are considered. To this end, we consider that end users support multiple VoIP codecs. Table 2 summarizes the important features of the network used in our simulations.

The default values of OLSR-VA are considered when there is no VoIP activity in local node or in the neighbours. Also, these values are applied if the routing performance is decreased in order to stabilize the routing system. However, these values will be revised if a notification comes from the upper layers (layer triggers mechanism). Through this loop control, we can see that management process is similar to the concept of an automatic control system.

5.2. Evaluation Metrics. We aim to evaluate the performance of OLSR-VA gathering both quantitative and qualitative metrics of VoIP quality.

(i) Quantitative Metric. In order to map the QoS requirements at the network level, we use QDVP metric. This metric quantifies the quality degradation of VoIP packet (QDVP) and then gives the percentage of lost and late packets measured at the destination. As a result, $0 \leq \text{QDVP} \leq 1$, while the smaller the QDVP the better. Based on [16] this metric is defined as

$$\text{QDVP} = \frac{P_{\text{lost}} + P_{\text{late}}}{P_{\text{total}}}, \quad (1)$$

where P_{lost} is the number of packets lost, P_{late} is the number of packets arriving to their destination after 150 ms, and P_{total} is the total number of packets sent.

NB. QDVP does not take delay variation into account explicitly; the receiver can employ a play-out buffer to smooth out variations in packet arrival.

(ii) Qualitative Metric. QDVP metric reflects the total performance measurement at network level, while the measurement at user level (perceived voice quality) is not clear. To this end, we use the subjective MOS metric, mean opinion score, which is a numerical indication of the estimated listening quality of the received audio stream. MOS calculation is on E-model [17] method using *Ns2measure* framework [18].

In the following section, we show and discuss our simulation results investigating the impacts of the introduced routing mechanisms.

6. Results and Analysis

The overall system performance was tested using two sets of measures. In the first one, the network does not implement any adaptive algorithm and OLSR routing protocol is used. In the other set of simulations, each mobile node provides OLSR-VA functionalities. For each simulation scenario we perform 10 runs with varying random simulation seeds (node density and initial node position in simulation area). We measure QDVP and MOS metrics in function of simulation time (500 s). The VoIP calls are started after the 100 s of network simulation warm-up. In order to show the impact of our adaptive algorithm we switch the VoIP codecs randomly after 250 s of the simulation time.

We first present simulation results for scenario "1." Figure 6 plots the QDVP and MOS as a function of the simulation time for both OLSR and OLSR-VA. Measures are taken every 10 seconds and each value represents the average of 10 simulations repetitive results. As it can be observed, before applying codecs switching (i.e., before 250 seconds of simulation time), both protocols have the same behaviour whether for QDVP or MOS, and the maximum MOS obtained is about 3.4. In this phase, both protocols are in principle the same and have very similar routing paths for the traffic sessions.

The performance difference is revealed when the variation phase of codecs starts (after 250 s). Results clearly show that OLSR-VA deals perfectly with the changing codecs. These can be explained by the adaptive behaviour of OLSR-VA with regard to codec used in each period of the simulation time, while changing codec affects the OLSR performance. As it can be seen, the MOS goes down to 3 which is unacceptable value for our application interest (voice communication in emergency response).

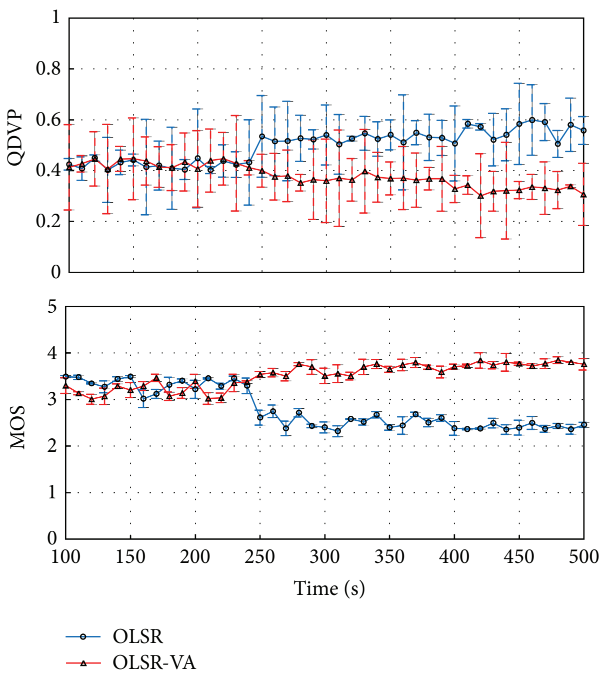

Figure 6: QDVP and MOS as a function of simulation time in scenario "1."

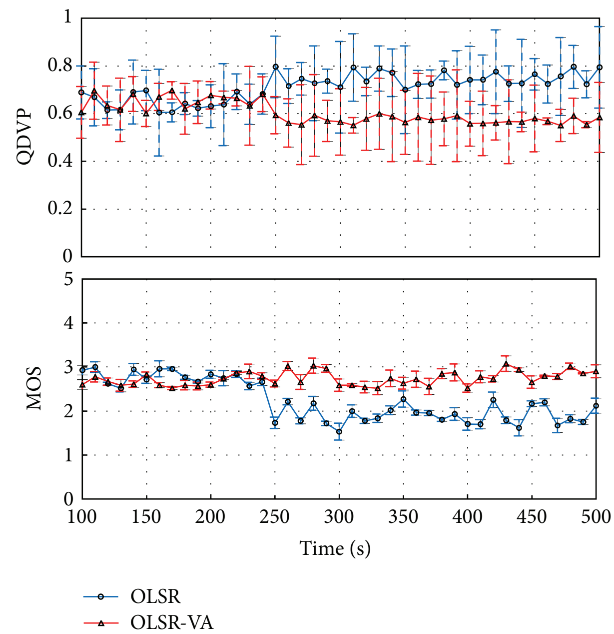

Figure 7: QDVP and MOS as a function of simulation time in scenario "2."

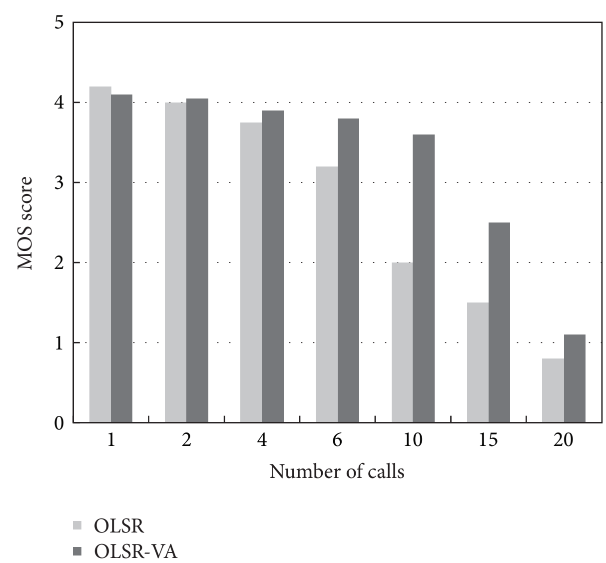

Figure 8: MOS versus number of calls.

QDVP and MOS almost appear as a mirror image, but they are not always presented in a mutually proportional manner. For example, in 340 s of the simulation time, the QDVP value of OLSR-VA is 0.37 and MOS value is 3.5, while in 370 s MOS obtained is 3.8 for the same QDVP value (0.37). This can be explained by the calculation of MOS which is based on metrics used in the calculation of QDVP (delay and packet loss) in addition to other parameters related to codec (such as codec impairment).

In scenario "2," the analysis focus is on the results considering large urban scenarios sizes with a significant increase in density of vehicle and VoIP calls. Figure 7 shows simulation results exhibiting QoS measured in terms of QDVP and MOS metrics. In general, the behaviour of the OLSR and the adaptive scheme OLSR-VA shows degradation in quality. This can be explained by the critical environment of the scenario which could negatively affect the network performance. Since, the probability of link features tends to be higher because of fading radio propagation model.

Moreover, in this scenario, longer paths are expected which result in longer delay due to more queuing at intermediate hop. In turn, OLSR-VA keeps its distinctiveness in relation to OLSR, where it was able to remain the desired minimum value of the MOS. However, in this situation, the major concern is to design and develop an efficient solution for voice communication in such harsh environment.

Alternatively, we have to investigate quality enhancement which adaptation offers under a given load condition. We note that the capacity of voice calls is reduced by DCF mechanism; additionally, in networks with high VoIP calls, a substantial percentage of link capacity is wasted due to control overhead. In this scenario, we investigate whether the adaptive routing protocol solution (OLSR-VA) has any benefit over the nonadaptive one in terms of call capacity.

In Figure 8, we measure the average MOS versus the number of calls which is increased to see the impact on performance. We observe that both protocols give similar voice quality when the number of calls is fewer (less than 6 calls). This is expected as the network is not loaded. In this way, less bandwidth is wasted and less delay is introduced. From six calls a fair listening quality is shown for both protocols. However, for an acceptable voice quality delivered (MOS = 3.6), adaptation accommodates more call capacity when compared to traditional, nonadaptive approaches. We can have call capacity until 10 calls with this score when applying the adaptive mechanism. This is because OLSR-VA uses some of the nonloaded paths to deliver VoIP packets while OLSR uses some of the less stable ones and hence saturating them.

In summary, simulation results have shown that the proposed scheme gives better performance compared to traditional approaches following the layered architecture by selecting paths with high bit-rate links while also avoiding areas of MAC congestion. Additionally, based on investigating various scenarios, our solution OLSR-VA performs better than OLSR to deal with codecs change and provides acceptable voice quality. Additionally, our adaptation mechanism might increase calls capacity with trade-off between the achieved quality and the quantity of accepted calls.

7. Conclusion and Future Works

In this paper, we basically tried to make VOIP smoothly operable in MANET through the adaptive OLSR-VA algorithm which is the most important contribution of this work. Our solution provides an integrated management environment where VoIP activity is constantly detected and the adaptation mechanism addresses it efficiently. In addition, the proposed solution is a lightweight adaptation scheme offering significant advantages for MANETs which are characterized by a scarcity of resources. The core of the contribution is a heuristic to include VoIP traffic activity and network errors in parameter to guide routing decisions.

To investigate the performance advantage achieved by such algorithm, we use simulation scenarios which provide a good cost effective environment. A number of realistic simulations under different conditions are performed (MANET scenarios). The most important observation is that performance is satisfactory, in terms of the perceived voice quality and call capacity. Results have shown ability of the solution to successfully achieve an acceptable voice quality even over long routes and under reasonable load conditions.

As a matter of future works, we intend to develop a prototypical implementation of the whole mechanism which will help validate the simulation results and calculate more precisely the additional cost. Additionally, we wish to test the deployment of our solution in larger-scale environment.

Conflict of Interests

The authors declare that they have no financial or personal conflict of interests which may interfere with the study outcome.

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