An Adaptive Gateway Discovery Algorithm to support QoS when providing Internet Access to Mobile Ad Hoc Networks

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***Abstract*—When a node in an ad hoc network wants Internet access, it needs to obtain information about the available gateways and it should select the most appropriate of them. In this work we propose a new gateway discovery scheme suitable for real-time applications that adjusts the frequency of gateway advertisements dynamically. This adjustment is related to the percentage of real-time sources that have quality of service problems because of excessive end-to-end delays. The optimal values for the configuration parameters (time interval and threshold) of the proposed adaptive gateway discovery mechanism for the selected network conditions have been studied with the aid of simulations. The scalability of the proposed scheme with respect to mobility as well as the impact of best-effort traffic load have been analyzed. Simulation results indicate that the proposed scheme significantly improves the average end-to-end delay, jitter and packet delivery ratio of real-time flows; the routing overhead is also reduced and there is no starvation of best-effort traffic.**

***Index Terms*— ad hoc network, Internet gateway discovery, quality of service, performance analysis**

I. INTRODUCTION

Ad hoc networks [1] consist of wireless mobile devices that are able to communicate even if they are outside of their radio ranges because the intermediate nodes will route the packets from the source node to the destination node.

Originally, the investigation was centered in developing isolated and independent ad hoc networks useful to collaborate in certain restricted environments like in the case of natural catastrophes. However, more recently, the attention has been focused in studying heterogeneous networks. The interaction between ad hoc networks and other types of networks, like cellular networks, infrastructure-based WLANs (Wireless Local Area Networks) [2] and, especially, wired networks raises great interest.

The communication between wireless ad hoc networks and infrastructure-based networks is essential to extend Internet beyond its traditional scope, to remote inaccessible areas, making Web services in ad-hoc networks available anytime, anywhere.

The mobile nodes in a wireless ad hoc network must be able to detect available gateways and select one of them if they want to have Internet access.

On the other hand, real-time applications have special Quality of Service (QoS) requirements that must be satisfied to function properly and they are expected to maintain their quality level in heterogeneous networks.

New gateway discovery mechanisms should be designed thinking over the requirements of real-time flows because since Internet access to mobile nodes is provided through gateways, the quality of such service depends on the selection procedure used by ad-hoc mobile nodes to choose the most convenient gateway and register with. Besides, the intrinsic functioning characteristics of the selected gateway mechanism will influence the service level degree that a particular flow is able to obtain and maintain.

Our objective will be to design a new gateway discovery protocol that helps real-time flows in a wired-cum-wireless scenario to maintain their quality of service parameters. Some different approaches have been developed in literature, which propose different gateway discovery schemes, but none of them is related to service differentiation. We have designed a new gateway adaptive discovery mechanism that is able cooperate with real-time flows to improve and maintain their desired quality of service. This is the main contribution of this paper.

The paper is organized as follows: Section II describes related work about Internet gateway discovery methods. Section III remarks the importance of quality of service provision in wireless ad hoc networks. The proposed adaptive gateway discovery scheme is presented in Section IV. Section V shows our simulation results and finally Section VI concludes this paper.

II. Related Work

When an ad hoc node needs to send packets towards Internet, it should discover a gateway (see Fig. 1). This device implements the protocol stack of the ad hoc as well as the fixed network, routing the packets from one network to the other. The protocol stack used by mobile nodes, gateways and Internet nodes is shown in Fig. 2.

The Internet Draft "Global Connectivity for IPv6 Mobile Ad Hoc Networks" [3] describes how to modify ad hoc routing protocols to discover gateways and thus providing Internet connectivity to mobile ad hoc networks.

The ad hoc routing protocols may be extended using three different approaches to detect gateways:

- Proactive gateway discovery [4]:
  The gateways periodically broadcast advertisement messages (GWADVs, Gateway Advertisements) that contain information about the global prefix length and the IPv6 address from the gateway. These messages are flooded throughout the entire network. The routing protocols of the mobile nodes use this information to autoconfigure a new routable IPv6 address and select the address of one of the gateways as default route. The mobile nodes select the best Internet-gateway examining the distance towards it in number of hops or considering other parameters.
- Reactive gateway discovery [5]:
  A mobile node that wants to send packets towards Internet broadcasts a message to the group of gateways within the ad hoc network. The gateways receive this message and reply to it accordingly. The routing protocol of the mobile node selects the gateway which offers the best route towards Internet in terms of number of hops or other parameters.
- Hybrid gateway discovery [6] [7]:
  This method combines the reactive and proactive approaches; it defines a transmission range where the gateways periodically send advertisement messages (GWADVs) and they are propagated around a limited zone (a certain number of hops away from the gateway), as it is shown in Fig. 3. A mobile node receiving these messages can obtain information about the global prefix length and the IPv6 address from the gateways carried in this message to discover the global prefix. Afterwards, the routing protocol of this mobile node autoconfigures a new routable IPv6 address and selects the address of one of the gateways as default route. The mobile nodes select the gateway that is either closer in terms of number of hops or that is more appropriate because of other parameters. If a mobile node wants Internet connectivity and it is outside the gateways transmission range and the propagation zone of the gateways advertisements, it broadcasts a message to the group of gateways in the ad hoc network. If another mobile node receives this message, it rebroadcasts it until it arrives to a gateway that responds sending back a reply. The mobile node selects the reply of the gateway which offers the best route towards Internet in terms of number of hops or due to other parameters.

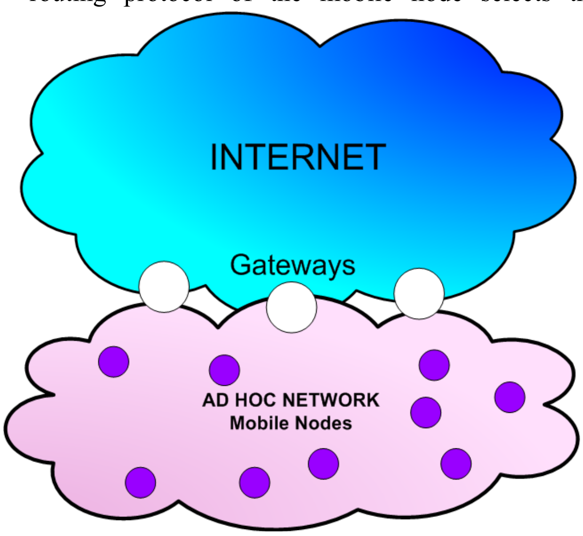

Figure 1. Interworking scenario.

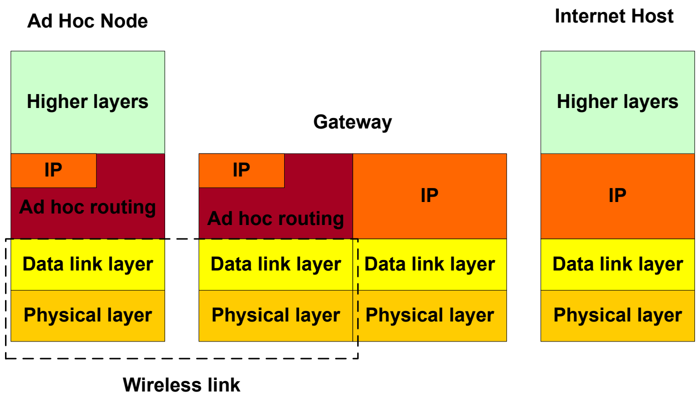

Figure 2. Protocols architecture.

In [8] the authors compare the proactive and reactive approaches by the aid of simulations. The simulation results show that in the proactive approach the overhead load caused by Internet connection is increased, but on the other hand, if a mobile node looses Internet connectivity, it can detect a new gateway quicker; as a result, the packet delivery ratio is higher and the average end-to-end delay is lower, specially if the average link durations are longer.

In [9] the scalability of both approaches (proactive and reactive) is compared with respect to the number of Internet gateways. The simulation results show that the proactive approach is more advantageous because the packet delivery ratio is higher and, although the signaling overhead is larger too, it is reduced for a higher number of Internet gateways, because the amount of periodical

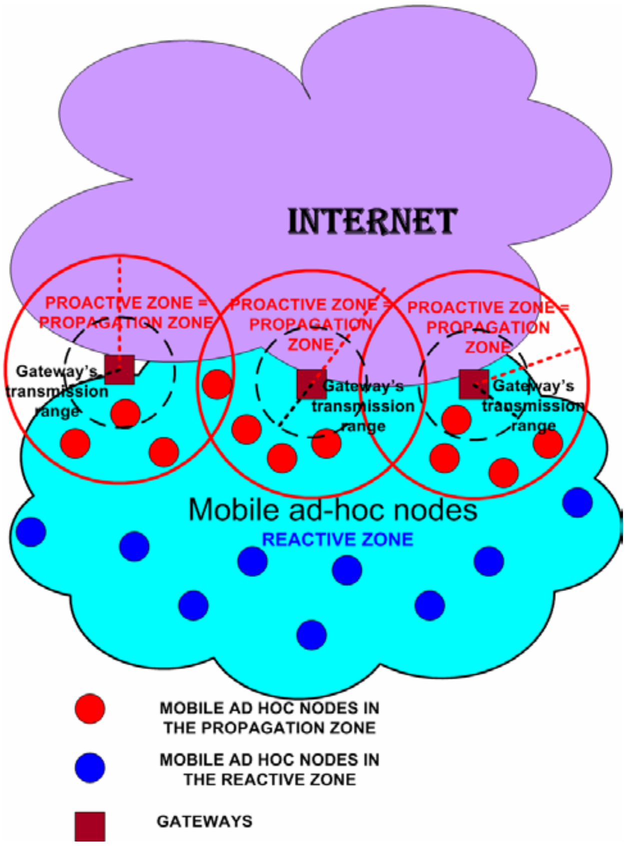

Figure 3. Hybrid gateway discovery.

gateway advertisements is increased but more data packets are transmitted successfully.

The hybrid gateway discovery approach is also compared with the other ones with the aid of simulations that show the average packet delay (defined as the delay for sending packets from the source towards the gateway) and the packet delivery ratio. The hybrid gateway discovery represents a balance between the reactive and the proactive approaches; hence the curves obtained for this approach are always located between the other two schemes.

From here on the different approaches that have been proposed in the literature are modifications of the already mentioned gateway discovery strategies.

In [10] the authors propose an adaptive gateway discovery mechanism based on the hybrid discovery approach that modifies the scope of the GWADV (Gateway Advertisement) messages sent by the gateways to obtain the maximal benefit in terms of overhead savings by avoiding sources to flood the network asking for gateways. The same authors propose in [11] an adaptive gateway discovery mechanism based on the hybrid discovery approach that modifies the scope of the GWADV messages sent by the gateways to reach the maximal number of active sources. A comparison between these gateway discovery schemes with already existing ones is done in [12], where both adaptive approaches have been evaluated with similar results and it has been demonstrated that they outperform existing schemes.

In [7] the authors suggest an adaptive gateway discovery scheme based on the hybrid discovery approach where the gateways send GWADV messages only if node mobility is detected using a source-routing protocol and if the overhead of sending GWADV messages is lower in comparison with the benefit of informing the mobile nodes connected to the Internet about gateways when mobility is detected.

The authors in [13] propose to improve the performance of the hybrid Internet gateway discovery using a feedback controller that adapts the frequency of the GWADVs and the coverage range of the gateways dynamically according to the number of solicitation messages received by the gateways, their hop count, the number of data packets received by the gateways and their hop count.

However, the already mentioned existing approaches are methods to discover gateways that treat all the traffic in the same way and do not consider differences between real-time and best-effort applications. Next sections remark the importance of providing quality of service to real-time applications in wireless ad hoc networks and introduce gateway discovery mechanisms that help to differentiate service levels between best-effort and real-time traffic.

III. Quality of Service Provision in Wireless Ad Hoc Networks

Quality of Service can be defined as the ability of the network to offer a required service demanded by a particular application, establishing some type of control over certain parameters: end-to-end delay, jitter, traffic loss or bandwidth.

It is a very challenging topic to provide QoS in wireless ad hoc networks [14] due to the intrinsic properties of this kind of networks: variable capacity of the links, topologies that change dynamically, etc; furthermore, in wireless networks the packet loss rate and the jitter of the applications are higher in comparison with wired networks due to the existence of fading, interference between neighbouring nodes, etc.

In this paper the performance of multimedia applications in wireless ad hoc networks connected to wired networks has been studied. A specific type of real-time application implying burstiness and containing end-to-end delay information has been selected: VBR Voice-over-IP (VoIP) [15].

There are some works related to voice transmission in IEEE 802.11 [16][17]; other authors [18] [19] [20] have addressed the topic of VoIP traffic support in the case of multihop ad hoc networking, from the perspective of doing simulations and implementing working testbeds. We will analyze the transmission of real-time traffic that shares resources with background traffic between a mobile ad hoc network and a fixed IP network.

The ITU-T recommends in its standard G.114 that the end-to-end delay of VoIP traffic should be kept below 150 *ms* to maintain an acceptable conversation quality [21] . Delays from 150 to 400 *ms* are acceptable provided that administrators are aware of the impact on quality, and latency larger than 400 *ms* is unacceptable.

Our goal is to provide QoS to real-time applications in wireless ad hoc networks connected to wired networks, differentiating services between real-time and best-effort traffic.

There exists a relation between the QoS provisioning and the gateway discovery method. The hybrid and specially the proactive gateway discovery mechanisms show a better performance with respect to end-to-end delay of real-time flows, because GWADV messages are sent periodically and not only when it is needed, as in the reactive approach. Thus, real-time applications are able to find a route towards Internet for their traffic sooner. But, on the other hand, if a real-time application has delay problems due to congestion and more GWADV messages are sent, the congestion will be increased and the performance of the delay sensitive applications will be seriously damaged.

Next section presents an adaptive gateway discovery approach that has been mainly designed to reduce congestion problems in an ad hoc network and that helps real-time applications to maintain their QoS parameters even in the presence of excessive traffic.

IV. Proposed Adaptive Gateway Discovery Mechanism

A. Protocol Architecture

A scenario where an ad hoc network is connected to a fixed one via several gateways has been considered (see Fig. 4). Best-effort and real-time sources wish to start sending traffic from the ad hoc towards the fixed network through a gateway.

The routing protocol in the ad hoc network (Ad Hoc On Demand Distance Vector (AODV)) [22] has been modified as described in the Internet draft "Global Connectivity for IPv6 Mobile Ad Hoc Networks" [3] to discover gateways. In this work a new gateway discovery method to find a gateway has been proposed, which is based on the hybrid mechanism (see Fig. 3). Therefore, our proposed approach defines a transmission range where the gateways periodically send advertisement messages and they are propagated around a limited zone (a certain number of hops away from the gateway). If a mobile node wants Internet connectivity and it is outside the gateways transmission range and the propagation zone of the gateways advertisements, it should broadcast a message to the group of gateways in the ad hoc network. The gateways should respond sending back a reply and the routing protocol of the mobile node selects the reply of the gateway which offers the best route towards Internet in terms of number of hops accordingly to the normal functioning of the AODV routing protocol.

The hybrid gateway discovery mechanism has been selected as reference model because it shows a better behavior with respect to latency for delay sensitive applications in comparison with the reactive approach. What is more, the GWADVs of the hybrid approach are propagated only a limited number of hops away from the gateway (advertisement zone) and not through the entire network as the proactive approach. Therefore, less overhead is introduced and congestion is reduced.

Once the routes have been established, traffic is sent towards the fixed network. Quality of service should be provided, differentiating services between real-time and best-effort applications. Consequently, the destination nodes of the real-time traffic in the fixed network periodically monitor the end-to-end delays of these flows. To achieve it, a 'timestamp' or generation time of the packet is introduced in the header of the real-time application protocol (the RTP protocol (Real-time Transport)) and the average end-to-end delay is calculated at the destination nodes as a time difference. If the end-to-end delay of one or more real-time sources becomes greater than a threshold (140 *ms*, because the ITU-T recommends to keep these delays under 150 *ms* [21] and the system needs some reaction time), QoS_LOST messages will be sent to the real-time traffic sources that have latency problems to warn them about the situation (see Fig. 5). When a node in the ad hoc network receives a QoS_LOST message, it will react executing a QoS mechanism to improve the QoS of the real-time flow that has latency problems; for example, the authors in [23] propose to send a QoS_LOST message to the real-time sources generating flows that have problems to keep their end-to-end delays under 150 *ms* and to the intermediate nodes along the routes in the ad hoc network. Then these nodes forward the QoS_LOST message as a broadcast packet to all their neighbours because they may be contending with them for medium access. When a node receives a QoS_LOST message as a broadcast packet, it throttles its best-effort traffic. This QoS model named DS-SWAN provides an improvement of the quality for real-time flows.

But now we are interested in the arrival of the QoS_LOST messages to the gateway that is crossed when these messages travel towards the real-time sources in the ad hoc network (see Fig. 5).

A new mechanism has been proposed where each gateway periodically (each τ seconds) (gateway advertisement interval) checks if it must send a GWADV

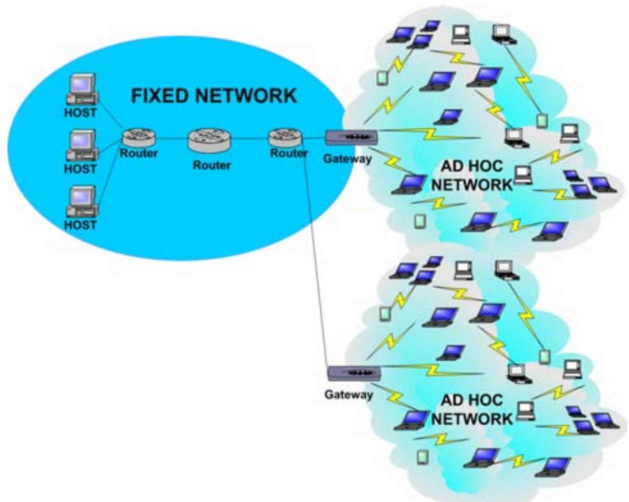

Figure 4. Proposed scenario.

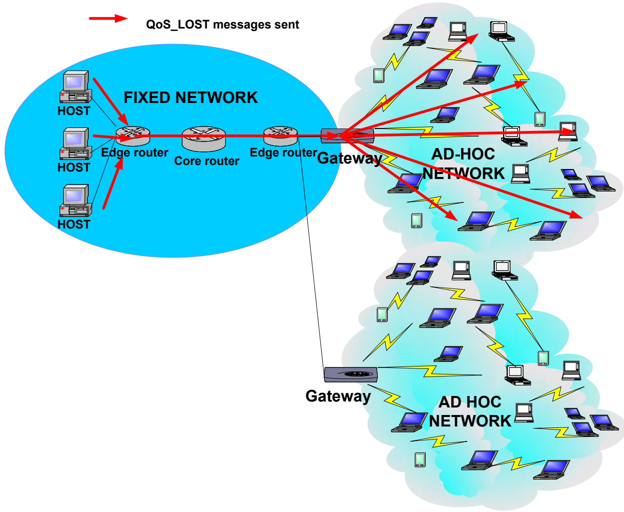

Figure 5. QoS_LOST messages sent from the fixed towards the ad hoc network.

message. Therefore, each gateway should periodically check if it has received QoS_LOST messages during the last τ seconds from real-time flows having problems to keep their end-to-end delays below 150 *ms*. If it is not the case, the gateway sends a GWADV message.

Otherwise, each gateway should calculate:

$$\alpha(t) = \frac{P}{F}, \tag{1}$$

where P = number of real-time sources having end-to-end latency problems and F = total number of real-time sources using that gateway.

A threshold γ has been set, where $0 \leq \gamma \leq 1$. It is fulfilled:

If $\alpha(t) > \gamma$, no GWADV messages should be sent by the gateway to the ad hoc network, because if real-time flows have QoS problems due to excessive congestion, it is not recommended to introduce more traffic overload in the network with these messages.

The GWADV messages have a higher priority than real-time packets because they are control packets and they are normally served in the queues of the mobile ad hoc nodes at the MAC layer before real-time or best-effort traffic when they try to access the wireless medium. Therefore, if GWADV messages are not temporarily sent, the congestion in the ad hoc network is noticeably reduced.

On the contrary, if $\alpha(t) \leq \gamma$, GWADV messages should normally be sent towards the ad hoc network.

The value of γ broadly represents an upper bound for the probability that a new real-time flow will fail the quality of service requirements. Therefore, gateways with lower values of γ have a higher probability of satisfying a flow's quality of service requirements.

This gateway discovery method serves the purpose that real-time sources do not increase their end-to-end latency problems if congestion is excessive. The proposed mechanism adapts the rate at which the gateway advertisement messages are sent considering the threshold γ. Thus, the gateway examines the real-time flow conditions before taking the decision to send GWADVs. If the real-time flows, which are connected to a gateway, experience congestion problems, no GWADV messages are sent by this gateway; as a result, if certain routes towards Internet fail or new sources wish to start sending traffic towards the wired network, they will use other gateways that are sending GWADVs to establish a connection towards Internet or they will have to initiate a new route discovery process searching for a default route; consequently, the traffic load is distributed.

B. Considerations

We have made the assumption aiming to design our adaptive gateway discovery approach that congestion appears in the ad hoc and not in the fixed network. A DiffServ domain in the fixed network should prevent that congestion is introduced in the core routers because it has been considered that the wired network is overprovisioned.

It is important to notice that the designed mechanism is able to act in the presence of congestion due to excessive traffic in the ad-hoc network. The end-to-end delays of the real-time flows will probably not be higher than expected due to the propagation delays because the distances between nodes in the ad hoc network are not as long. However, if there is an increase in the end-to-end delay of real-time flows due to the quality of the wireless links (fading), our proposed approach will not solve the problem. Nevertheless, if the link quality is very bad or the intermediate nodes in the ad hoc network move and as a consequence there is a link failure, with the aid of the AODV routing protocol and our proposed adaptive gateway discovery approach a better route can be found by the source node to route its packets towards Internet and the QoS of the real-time flow could be improved. Besides, congestion is reduced because best-effort traffic is throttled in the ad hoc network and less GWADV messages are sent.

On the other hand, it is possible to monitor the end-to-end delay of real-time traffic with the aid of timestamps because the nodes are GPS (Global Positioning System)-enabled and they are able to synchronize the global time.

In addition, the case where traffic is sent towards Internet has been studied in order to design an efficient way to allow nodes in the ad hoc network to detect gateways. However, the proposed scheme can function well when traffic is sent in both directions.

What is more, it is important to think about best-effort background traffic. It does not have an active role in the decisions taken by the gateway to send GWADV packets, because these decisions are related only to delay-sensitive real-time applications. If best-effort traffic suffers higher delays due to congestion problems it is not as crucial; if it would have been decided not to send GWADV packets when best-effort traffic flows suffer congestion problems, this means that perhaps more real-time traffic sources would have had to start a route discovery process to find a route towards Internet (the gateways would have not advertised that they are available) and therefore the signaling overhead of the network would have been increased. Best-effort traffic flows would have profited at the expense of the QoS parameters of real-time traffic flows, which is not logical. Therefore, it has been decided that the decision to send or not GWADV messages should only be related to the QoS parameters of real-time flows. Best-effort traffic flows sometimes will take advantage of this gateway discovery mechanism because congestion is reduced; sometimes they will be at disadvantage because they will not receive GWADV messages and they will have to start a new discovery process to find a gateway.

C. Functioning Example

The functioning of this adaptive scheme is illustrated in Fig. 6. It shows an example of an ad hoc network where three VoIP real-time and two CBR best-effort flows have been established to send packets towards Internet through the gateway. If VoIP flows VoIP1 and VoIP3 have problems to keep their end-to-end delays under 150 *ms*, QoS_LOST messages will be sent to these VoIP sources in the ad hoc network through the gateway to warn them about the situation. The gateway takes advantage of this information and it periodically calculates the percentage of VoIP sources that route their packets towards Internet through it and that have end-to-end delay problems. In our example this percentage is $\alpha(t) = 2/3$. If the threshold for latency problems is set to be $\gamma = 0.4$, then it follows that $\alpha(t) > \gamma$, which means that any GWADV messages should be sent by the gateway to the ad hoc network, because the percentage of VoIP sources having delay problems due to excessive congestion is larger than the threshold and this means that the network should not be overloaded with more traffic if it is not strictly necessary. If afterwards one of the VoIP

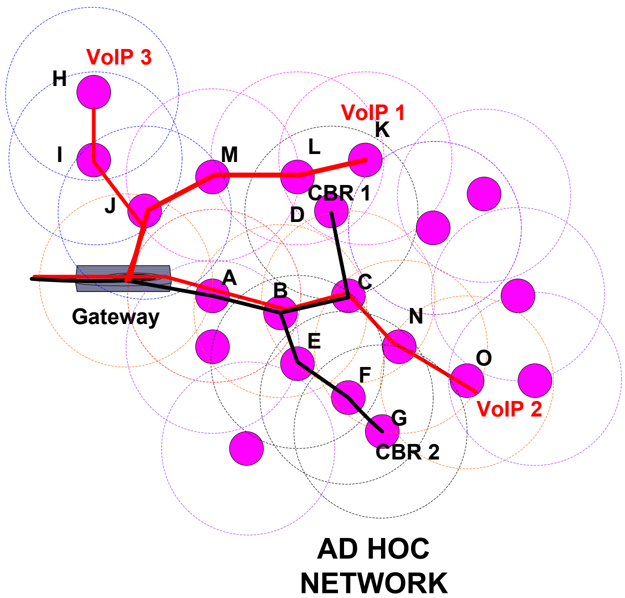

Figure 6. Example network.

sources solves its QoS problems, the gateway will calculate a new percentage $\alpha(t) = 1/3$. Now GWADV messages should be sent towards the ad hoc network because $\alpha(t) \leq \gamma$. The advertisement messages will be propagated around a limited zone (a certain number of hops away from the gateway); in this case it has been defined an advertisement zone of TTL = 4 hops. This means that the gateway advertisement messages will be received by the sources VoIP1 (route gateway-J-M-L-K), VoIP3 (route gateway-J-I-H) and CBR1 (route gateway-A-B-C-D). The other sources will not receive the GWADV messages because they are more than 4 hops away from the gateway and they would have to do a route discovery in the case that the route towards the gateway fails.

In [24] the authors present a preliminary version of the proposed adaptive gateway discovery mechanism. In the present paper the main algorithm has been simplified. We discuss in more detail the decisions on the design of this approach and the pros/cons of the proposed solution. Besides, we have run new simulations to evaluate the performance of the proposed scheme with respect to mobility considering a more overloaded network (with 10 best-effort traffic sources instead of 6). What is more, we have carried out other simulations with respect to the best-effort traffic load and we have studied which are the optimal values for the parameters of the proposed adaptive gateway discovery mechanism (time interval and threshold) for the selected network conditions.

V. Simulations

A. Simulation Environment

We have run simulations with the NS-2 tool [25] to investigate the performance of our proposed approach. The system framework is shown in Fig. 4. A scenario where an ad-hoc network is connected via two gateways to a fixed IP network has been selected. The chosen scenario consists of 20 mobile nodes, 2 gateways, 3 fixed routers and 3 corresponding hosts.

The mobile nodes are uniformly distributed in a rectangular region of 1000 *m* by 500 *m*. The gateways are placed with x, y coordinates (150,250) and (850,250). Each mobile node selects a random destination within the area and moves toward it at a velocity uniformly distributed between 0 and 3 m/s. Upon reaching the destination, the node pauses for a pause time, selects another destination and repeats the process. Five different pause times have been chosen: 0, 20, 50, 125 and 200 seconds.

The dynamic routing algorithm is AODV [22] and the wireless links are IEEE 802.11b.

Background traffic is generated by 10 of the mobile hosts, while VBR VoIP traffic is generated by 15 of the mobile hosts. The destinations of each of the background and VoIP flows are chosen randomly among the three hosts in the wired network.

Best-effort CBR background traffic and real-time VBR VoIP traffic are transmitted. CBR has been proposed as background traffic instead of TCP. The reason is that TCP performs poorly in an ad-hoc network because packets that are lost due to link failure and route changes trigger TCP's congestion avoidance mechanisms [26]. On the contrary, many authors [27] use CBR as background traffic successfully.

The VBR mode is used for VoIP traffic. A silence suppression technique in voice codecs is employed so that no packets are generated in the silence period. For voice calls, the ITU G. 726 or "adaptive differential pulse code modulation (ADPCM)" codec has been used. The VoIP traffic is modeled as a source with exponentially distributed on and off periods with 1.004 s and 3.587 *s* average each and two frames (20 *m*s audio sample each frame) are carried in each packet (80 + 80 bytes payload). Frames are generated during the on period every 20 *ms* with size 80 bytes and at a constant bit rate of 32 *Kbps* without any compression. VoIP is established over real-time transport protocol (RTP), which uses UDP/IP between RTP and link layer protocols. Packets have a constant size and are generated at a constant inter-arrival time during the on period. The VoIP connections are activated at a starting time chosen from a uniform distribution in [10 *s*, 15 *s*].

Background traffic is Constant Bit Rate (CBR) with a rate of 48 Kbit/s and a packet size of 120 bytes. To avoid synchronization, the CBR flows have starting times chosen randomly from the interval [15 *s*, 20 *s*] for the first source, [20 *s*, 25 *s*] for the second source and so on.

B. Methodology

The hybrid approach "Hybrid scheme" has been evaluated and compared with our proposed adaptive scheme discussed in Section IV "Proposed adaptive scheme". It is important to know which gateway discovery mechanism outperforms the other one in terms of QoS parameters (average end-to-end delays, jitter and packet loss) for real-time traffic flows because this means that it is more suitable for this type of applications. Furthermore, it is essential to study the impact of this scheme on the throughput of best-effort traffic. What is more, the comparison of the signaling introduced in terms of routing overhead is interesting, too, because it shows how much network resources does each protocol need to do its work [11].

In both approaches a TTL = 5 hops is used as proactive zone (or propagation zone) for the GWADV messages. In the proposed new adaptive mechanism the gateway every τ = 5 seconds checks if it must send a GWADV packet, whereas in the hybrid approach the gateway does always send a GWADV message every 5 seconds.

C. Evaluation

We have run simulations with the aim of analyzing the impact of mobility and scalability of the proposed mechanism with respect to mobility. Fig. 7 shows the average end-to-end delay for VoIP traffic. This parameter is defined as the time it takes for data packets to arrive from the source to the destination node.

In both schemes the end-to-end delays for VoIP traffic are increased with smaller pause times, because when the pause time is very low the routes of the existing flows break frequently and the routing protocol continuously does new route discovery processes that increase latency. On the contrary, when the pause time is larger, the average link duration is increased as well as the duration of the routes. The average end-to-end delay for VoIP sources is lower with our proposed adaptive scheme, because less GWADV messages are sent in congestion conditions. Each gateway periodically checks if it has received QoS_LOST messages associated with VoIP sources having end-to-end delay problems. If the percentage of VoIP traffic sources having latency problems exceeds a predefined threshold (in this case this threshold is set to $\gamma = 0.15$), no GWADV messages are sent by the gateway. Therefore, no more traffic overload is introduced in the congested network and as a consequence the latency of the VoIP flows is diminished; hence with the adaptive scheme the reduction of congestion is more effective in comparison with the hybrid scheme. With the hybrid scheme the average end-to-end delay for VoIP traffic is higher than 150 *ms* for all pause times except for a pause time of 200 *s*, which means that the VoIP quality is severely degraded. On the contrary, with the proposed adaptive scheme, the average end-to-end delay is always maintained around or below 150 *ms* except for a pause time of 0 *s*, which means that the VoIP quality is adequate.

A similar trend is observed regarding the jitter for VoIP traffic, as it is illustrated in Fig. 8.

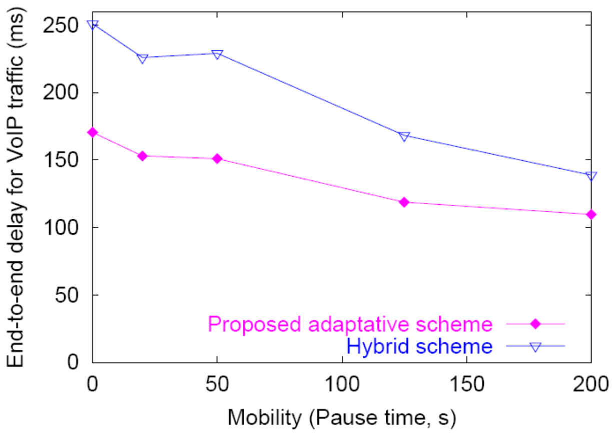

Figure 7. Average end-to-end delay for VoIP traffic.

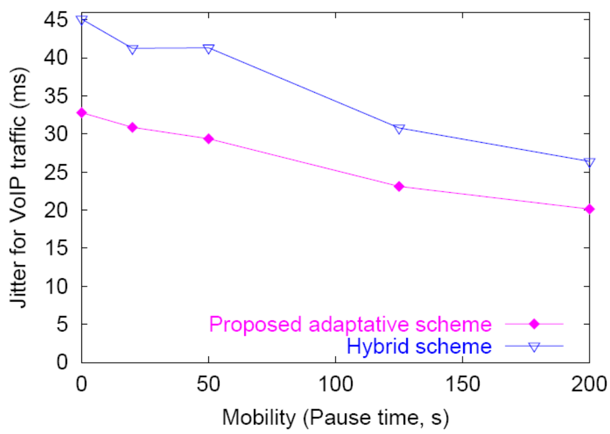

Figure 8. Jitter for VoIP traffic.

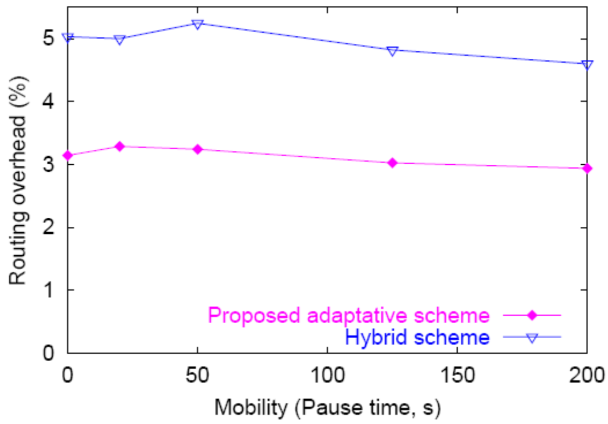

Figure 9. Overhead of control packets.

This QoS parameter is always lower when the proposed adaptive scheme is used (values between 32.8 and 20.1 *ms* instead of values between 45.1 *ms* and 26.4 *ms*). This means that the jitter values are more appropriate if the proposed adaptive gateway discovery mechanism is selected.

Moreover, the packet delivery ratio can be defined as the number of real-time (VoIP) packets successfully delivered over the number of real-time (VoIP) packets generated by the sources and this parameter is very significant to check the quality of service of real-time flows, too. This ratio is decreased when mobility is increased. However, in both mechanisms the values are always maintained over 98.5%, that is, the VoIP packet loss rate is always limited (lower than 1.5 %) and it is very acceptable for real-time traffic.

In addition, the routing overhead (see Fig. 9) has been defined as the amount of control packets (for gateway discovery and routing) divided by the sum of control packets plus data packets; this percentage has been reduced using the proposed adaptive mechanism (to around 3% instead of around 5%).

Besides, with the proposed scheme there is an improvement in the throughput of best-effort traffic due to the decrease of congestion conditions (throughput around 24-25 *Kbps* instead of 21-23 *Kbps* (see Fig. 10)).

Due to the good results our new adaptive scheme has been investigated in depth. The optimum configuration parameters of the proposed adaptive mechanism according to the network conditions have been studied.

In the proposed new mechanism the gateway periodically (each τ seconds) checks if it must send a GWADV message. The performance of the protocol has been evaluated when the value of the time interval τ is changed. The same simulation parameters as previously described have been applied (the pause time for the ad hoc nodes is 20 *s*).

The average end-to-end delay for VoIP traffic and the routing overhead are illustrated in Fig. 11 and Fig. 12 respectively. When the time interval is increased, the routing overhead is reduced and therefore the end-to-end delay for VoIP traffic is improved because less GWADV messages are sent. The end-to-end delay for VoIP traffic is kept below 150 *ms* for a time interval τ larger than 4 *s*, although from the value of τ = 10 *s* on the end-to-end delay for VoIP traffic does not present substantial differences (when the routing overhead is maintained around 1-2 %).

Additionally, the jitter shows a similar behaviour than the average end-to-end delay for VoIP traffic (jitter values are decreased when τ is increased and they are kept around 15 *ms* for a time interval τ over 10 *s*).

Furthermore, the packet delivery ratio for VoIP packets can be observed in Fig. 13. The number of VoIP packets that reach their destinations properly varies between 97.2 and 99.4%.

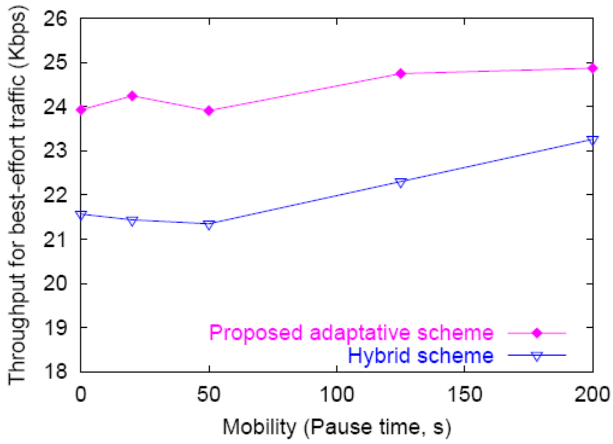

Figure 10. Average throughput for best-effort traffic.

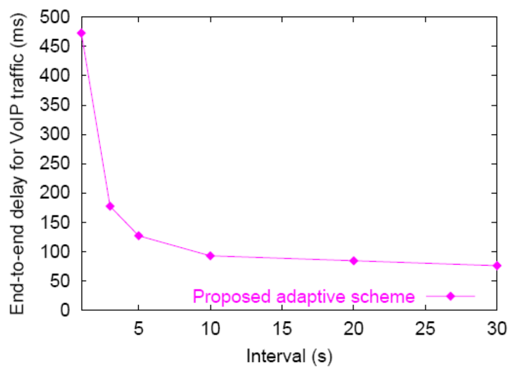

Figure 11. Average end-to-end delay for VoIP traffic.

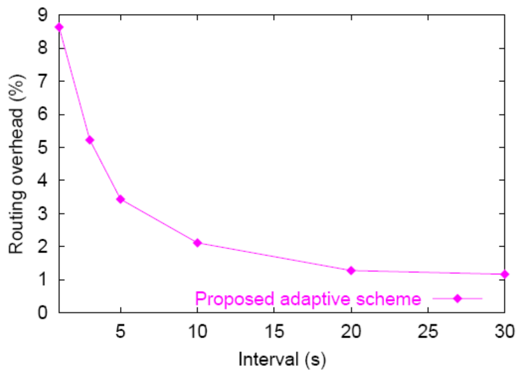

Figure 12. Overhead for control packets.

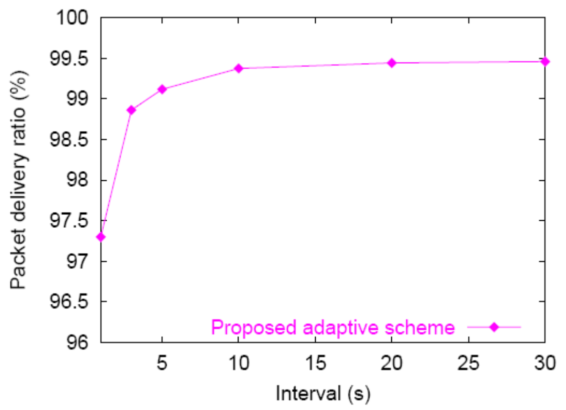

Figure 13. Packet delivery ratio for VoIP packets.

What is more, the throughput for best-effort packets is increased when the time interval between two consecutive GWADV messages is increased (see Fig. 14) because less congestion is introduced in the wireless ad hoc network.

After having processed the variation of the different parameters with respect to the time interval τ, we can conclude that according to the established network conditions a time interval τ higher than 4 *s* should be selected; besides, values of τ higher than 10 *s* do not introduce substantial differences in the variation of the selected parameters.

Now the functioning of the proposed protocol has been evaluated when the value of the threshold γ was modified (time interval value is set to τ = 5 *s*). γ broadly represents an upper bound for the probability that a new real-time flow will fail the quality of service requirements. Therefore, gateways with lower values of γ will theoretically have a higher probability of satisfying a flow's quality of service requirements. We want to prove if our hypothesis is correct.

Fig. 15 represents the average end-to-end delay for VoIP flows with respect to the threshold γ. As we expected, the average end-to-end delay for VoIP flows is

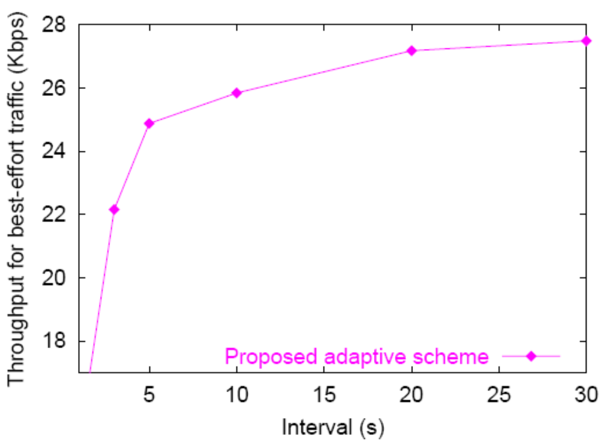

Figure 14. Average throughput for best-effort traffic.

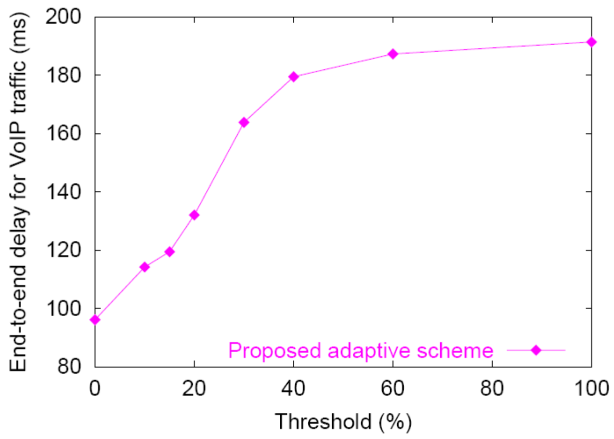

Figure 15. Average end-to-end delay for VoIP traffic.

increased with higher values of γ because, if the value of γ is increased, it is less probable to fulfill that $\alpha(t) > \gamma$, the necessary inequality not to send GWADV messages. Consequently, the routing overhead is increased (see Fig. 16). For threshold values higher than 25% the end-to-end delays are unacceptable for a good conversation quality (larger than 150 *ms*), because the ad-hoc network is congested and the gateway discovery mechanism uses a threshold γ too high to reduce the congestion conditions effectively.

The jitter for VoIP traffic is always low (values between 18.7 and 34.5 *ms*).

The percentage of lost VoIP packets is increased when the threshold value is increased, although it is always kept below 1.2 % with respect to the transmitted VoIP packets.

The throughput for best-effort traffic does not suffer starvation and it is always maintained between 26 *Kbps* (for γ=0) and 22 *Kbps* (for γ=100%).

If the threshold value γ is 100%, the proposed adaptive scheme does behave exactly in the same way as the hybrid scheme. Therefore, we can conclude that the hybrid scheme does behave worse with respect to the parameter values represented in the figures.

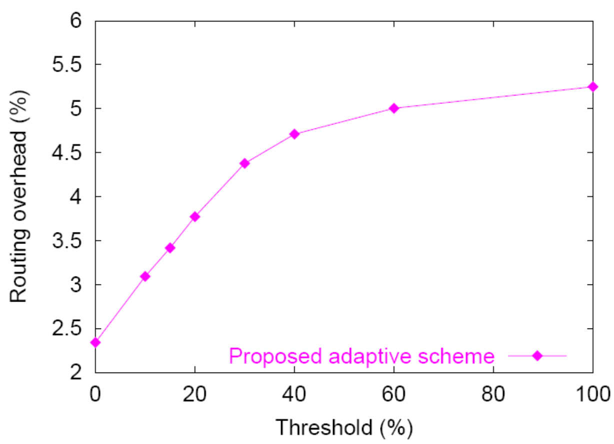

Figure 16. Overhead of control packets.

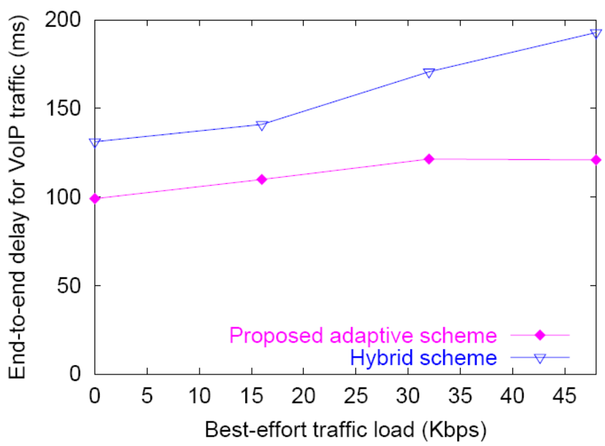

Figure 17. Average end-to-end delay for VoIP traffic.

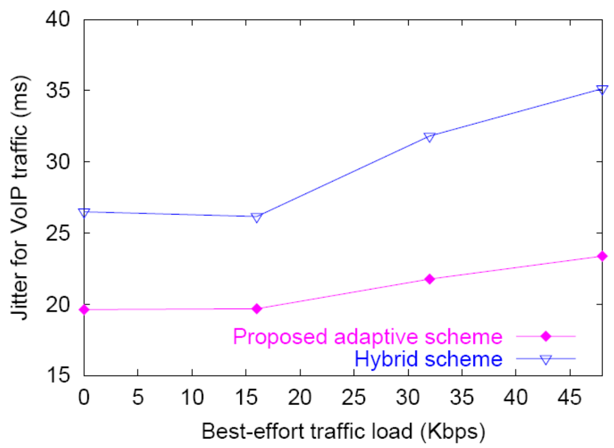

Figure 18. Jitter for VoIP traffic.

After having analyzed the variation of the different parameters with respect to the threshold γ, we can conclude that according to the established network conditions a threshold lower than 25% should be selected if we want to preserve the QoS parameters of the real-time flows, that is, a certain service differentiation to benefit real-time traffic is necessary.

Now we have evaluated the impact of best-effort traffic load on the VoIP flows for both schemes. With the hybrid approach the advertisement interval is 5 *s*, whereas with the proposed adaptive scheme τ = 5 *s* and the threshold γ = 15%. The rest of parameter values is maintained.

Fig. 17 shows the average end-to-end delay for VoIP traffic. The end-to-end delays for VoIP traffic are increased with a higher best-effort traffic rate, because the network is more overloaded. For lower best effort traffic loads both protocols show a good performance because real-time flows don't have QoS problems or only exceptionally. Consequently, both schemes function properly and GWADVs are sent periodically (although the hybrid scheme introduces more congestion and consequently the average end-to-end delays of the VoIP flows are higher). On the other hand, when the best-effort traffic load is increased, the proposed adaptive gateway discovery protocol favors that the end-to-end delays of real-time flows are lower (120.8 *ms* instead of 192.8 *ms* for best-effort CBR traffic load = 48 *Kbps* and latency larger than 150 *ms* does degrade the VoIP conversation quality with the hybrid scheme) because less GWADVS are sent in congestion conditions.

A similar trend is observed regarding the jitter for VoIP traffic (jitter is increased with best-effort traffic load) (see Fig. 18).

The packet delivery ratio for VoIP packets is always kept over 98% (VoIP packet loss rate is less than 2%).

Fig. 19 shows the overhead of control packets. The number of routing packets is increased with best effort traffic load because more congestion is introduced in the network. However, the routing overhead is diminished when the best-effort traffic load is increased. The reason is that the routing overhead has been defined as the amount of control packets (for gateway discovery and routing) divided by the sum of control packets plus data packets and now more data packets (including those of best-effort traffic) are transmitted successfully.

VI. CONCLUSIONS

We have proposed a new gateway discovery scheme that is able to mitigate the congestion conditions in an ad hoc network connected to a fixed network.

We have run simulations to study which are the optimal values for the configuration parameters (time interval and threshold) of the proposed adaptive gateway discovery mechanism for the selected network conditions.

We have shown with the aid of simulations that the proposed adaptive approach outperforms the hybrid scheme in terms of QoS parameters for real-time flows (average end-to-end delay, jitter and packet delivery ratio), without incurring starvation of best-effort traffic. What is more, this adaptive mechanism is scalable with respect to mobility and best-effort traffic load.

Thus, we can conclude that the proposed approach is more adequate than the hybrid scheme in heterogeneous networks because it is able to differentiate services between applications, improving the desired quality of service of real-time flows.

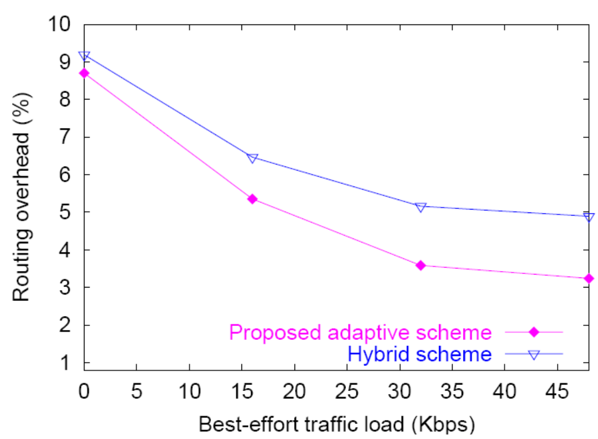

Figure 19. Overhead of control packets.

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