Improving End-to-End Delay for Real-time Multimedia Services in Mobile Ad-hoc Networks

J. N. Boshoff and A. S. J. Helberg, *Faculty of Engineering, NWU, South Africa*

***Abstract*—In a Mobile Ad-hoc Network (MANET), no fixed access points are present and thus there exists no physical or fixed routing infrastructure. This forces each node in a MANET to act not only as a host, but also as a router. Routing is a critical factor for the effective operation of a MANET. Many proposed MANET routing protocols are best effort protocols and do not consider the quality of service that can be provided by the routes that are used. Modern real-time multimedia applications often require strict bandwidth and delay guarantees.**

We extend the AODV routing protocol for MANETs to consider end-to-end delay requests from applications. Voice and video services are now able to request the maximum delay that the paths it will use should provide. Stationary and Mobile Ad-hoc Networks are simulated in OPNET and results prove that the end-to-end delay is greatly improved for these real-time services. The packet delivery fraction is also improved in most scenarios, thus, selected routes are more reliable than before.

***Index Terms*—Mobile Ad Hoc Networks, Quality of Service, AODV, Real-time Services, End-to-end Delay.**

I. Introduction

MOBILE Ad hoc Networks (MANET) are a subset of wireless networks. In these type of networks, no physical or control infrastructure is present. With no fixed infrastructure, nodes communicate by transmitting data directly to other nodes. However, communicating nodes may not always be in each other's transmission range. In such cases, nodes communicate with each other by using multi-hop routes, thus, any node in a MANET should be able to perform the task of a router and forward packets from a source in the network, to a destination. Furthermore, such networks should be self organized and accommodative to node mobility. [1]

The three main characteristics of MANETs that contribute to their dynamic and uncertain nature are node mobility, limited transmission bandwidth and limited energy resources of mobile nodes [2]. Mobile Ad hoc Networks are very useful in military deployments, disaster rescue operations and electronic classrooms. All of these are scenarios where no network infrastructure is available, but communication is crucial and has to be real-time and reliable. [1, 3]

Routing is a critical factor for the effective operation of a MANET. The dynamic topology of MANETs, limited bandwidth, power limitation of mobile devices, the limited physical security and lack of centralized control make routing a challenging task in these networks [3, 4, 5]. Multiple Quality of Service (QoS), security or other requirements of modern network applications make the routing problem even more involved than before [6].

The growing demand for such infrastructure-less networks brings the question of QoS guarantees to the horizon [5]. Real-time service provisioning in wireless networks is a challenging task. This is due to the variation of QoS requests, the dynamic and distributed nature of mobile networks, high bit error rates and frequent link and node failures in mobile networks. Furthermore, applications often demand high bandwidth, flow synchronization and are delay sensitive [5, 7]. To ensure that the performance requirements of different network applications will be met, a MANET should be able to service multiple QoS requests of applications.

Reactive routing protocols discover, store and maintain a route, or routes, to a destination only when communication with the destination node is required. The route discovery process is thus real-time and reduces control traffic overhead; however, route discovery latency together with frequent route discovery attempts can greatly degrade the real-time performance of a MANET [1, 2]. The most popular and studied on-demand routing protocols for MANETs include Dynamic Source Routing (DSR) [8], Temporally Ordered Routing Algorithm (TORA) [9] and the Ad hoc On-demand Distance Vector (AODV) routing protocol [10].

We propose to extend the AODV protocol, allowing applications to request a maximum end-to-end transmission delay. When new routes are discovered, the node-to-node and end-to-end delay are monitored and recorded in the routing tables. Application packets are transmitted only if a route's recorded delay is less than the requested delay.

The rest of the paper is organized as follows: Section II gives an overview on related work, which is also used as the foundation of this study. Section III summarizes the proposed extension of the AODV protocol. In section IV, the simulation setup is discussed and the results are

Manuscript submitted on April 14, 2007. This work was supported in part by the Telkom S.A. Limited CoE.

J. N. Boshoff is with the School for Electrical, Electronic and Computer Engineering, North-West University, Private Bag X6001, Potchefstroom, 2520, South Africa, (phone: +2718-299-1961; fax: +2718-299-1977; e-mail: 12523267@nwu.ac.za).

A. S. J. Helberg is with the School for Electrical, Electronic and Computer Engineering, North-West University, Private Bag X6001, Potchefstroom, 2520, South Africa, (phone: +2718-299-1966; fax: +2718-299-1977; e-mail: 12363626@nwu.ac.za).

displayed and discussed in section V. Section VI contains a brief discussion of the study.

II. Related Work

A. The AODV routing protocol [10]

The AODV routing protocol is a layer 3 protocol for ad hoc networks that provides dynamic, self starting, multi-hop routing between nodes in such a network. It can adapt quickly to dynamic link conditions, it has low processing and memory overhead, and low network utilization. Loop freedom is guaranteed, and the Bellman-Ford "counting to infinity" problem is eliminated by using destination sequence numbers. When a specific link is no longer available, AODV notifies all nodes that used this link as part of an active route.

AODV uses three message types in its operation; Route Requests (RREQ), Route Replies (RREP) and Route Errors (RERR). These are UDP packets that use normal IP headers.

When a node wants to send data to a new destination, or a destination to which it does not have a valid route, it creates and broadcasts a RREQ packet. Either the destination node, or an intermediate node with a valid route to the destination, can reply with a RREP packet to this RREQ packet.

A newly discovered route is only accepted as valid if the associated sequence number is as least as high as the sequence number in the RREQ packet. The sequence number in the RREQ is the number provided by the destination for the last known valid route to the specific destination. Each node in a MANET keeps track of a monotonically increasing sequence number for itself. It also records the highest known sequence number of each destination node that it wishes to communicate with. These are known as destination sequence numbers. This destination sequence number is used in all packets that are sent to the specific destination. [2]

On the reception of a RREQ packet, an intermediate node first stores a route back to the originator of the RREQ packet before forwarding the RREQ packet. This is used in the case where a RREP packet needs to be sent back to the originator of the RREQ. Duplicate copies of a specific RREQ packet are discarded by any node. [2]

Each node is responsible to monitor the link status between itself and every next hop in all active routes of which the node is part of. If any link is no longer available, the node should notify all other nodes in its precursor list that used the link in one or more active routes, by sending a RERR message. A precursor list is a list in which a node keeps the IP addresses of all the nodes that is likely to use the node as a next hop toward a specific destination. [2]

Any node can update its own routing information via a RREQ or a RREP packet. Upon receiving updated routing information in such a way, the following update rule is evaluated [2]:

if $(seq\_nr^d_i < seq\_nr^d_j)$ *or*
$((seq\_nr^d_i = seq\_nr^d_j)$ *and*
$(hop\_count^d_i > hop\_count^d_j))$
then
$seq\_nr^d_i := seq\_nr^d_j;$
$hop\_count^d_i := hop\_count^d_j + 1;$
$next\_hop^d_i := j;$
endif

The notation applies for a node i which receives routing information to destination d from a neighbor j. The destination sequence number, hop count and next hop for a destination d at node i is represented by $seq\_nr^d_i$, $hop\_count^d_i$ and $next\_hop^d_i$ respectively.

B. QoS extensions

Some QoS routing protocols for MANET have been proposed [5, 11, 12], as well as extensions to existing protocols, enabling them to consider one or more QoS requests by applications [13, 14, 15, 16]. Only a few extensions have been proposed for AODV and of them, the focus area is mainly the provision of bandwidth guarantees [14, 15, 16].

The overall operation of AODV is relatively lightweight and efficient. Its main drawback when it comes to real-time multimedia services is the fact that it uses the first and freshest discovered route which is not necessarily the best. Our initial simulations, which confirms the results in [17] and [18], showed that AODV in most cases outperforms other well known non-QoS protocols such as DSR and TORA in terms of throughput, delay and jitter. This makes AODV a good candidate for the implementation of new QoS extensions.

III. Delay Aware AODV (DA-AODV)

We used the proposed *QoS for AODV* in [13], with variations on some concepts, as basis for our delay aware extension of AODV.

In order to classify the type of data contained in an application packet, the Type of Service (ToS) field in the Internet Protocol (IP) header is used. The default values of 160 for Interactive Multimedia and 192 for Interactive Voice are used.

Whenever an application packet is ready for transmission and no valid route is available, the route discovery process is initiated as in AODV. The RREQ and RREP packets are now modified to contain a delay field in which the accumulated delay up to node i is recorded. Whenever a node receives one of these packets and needs to update its route table, this route delay value is stored with the specific route entry. This happens for the source, destination and intermediate nodes. Now, if an application requests a route to a specific destination node, the requested delay is compared with the route delay stored in the route table if a valid route is available. A route is only selected if the route delay is the same or less than the requested delay.

Once a first route to a destination is discovered, it is stored in the route table, no matter the delay of the route. From thereon, if new routes to the destination are discovered, a modified version of the update rule described in section II [10] determines whether or not the new route is preferred above the existing route. The metrics considered in the AODV update rule, which is primarily route freshness and secondly minimum hop count (for equal route freshness), are now replaced with minimum delay and route freshness on equal basis. Thus, a new route will not be used if it has a higher delay than the existing route.

The modified update rule is as follows:

if $((seq\_nr^d_i <= seq\_nr^d_j)$ *and*
$(route\_delay^d_i > route\_delay^d_j))$
then
  $seq\_nr^d_i := seq\_nr^d_j;$
  $hop\_count^d_i := hop\_count^d_j + 1;$
  $next\_hop^d_i := j;$
endif

As in AODV, each node broadcasts a *Hello* message locally every HELLO_INTERVAL milliseconds to offer connectivity information [10]. The format of a RREP packet is used for *Hello* messages. The delay field in the extended RREP message is also used in *Hello* messages to record link delay. Upon receiving a *Hello* message, a node compares the link delay with a delay threshold which is permitted per link. If the delay exceeds the threshold, the QoS provided by the link is no longer acceptable and the route table entry to the originator of the Hello message is deleted. The Route Error notification mechanism of AODV then detects the “broken link” and notifies all sources that was using this link in one or more active routes.

A more elegant solution would be to inform all nodes using the specific link in an active route of the precise change in delay, but this would require more extensive coding. The proposed solution is very efficient for its simplicity. The fact that the simulated services have the same delay requests contributes to this. Once services with different requests are running on the same network, assigning a threshold to the delay permitted per link would be less efficient. The second proposed solution (delay change notification) would be preferred in such networks.

A final extension is made to AODV to ensure that packets that are queued for route discovery are dropped if they are queued for longer that a maximum accepted delay period, typically 400 milliseconds for real-time traffic [19].

IV. Simulation Setup

OPNET Modeler was used to perform network simulations. The basic scenario setup is as follows: An ad-hoc network containing 81 nodes is set up over a physical area of 100x100 meter. A total of six nodes are set up to perform GSM quality Voice-over-IP (VoIP) communication and six other nodes to perform low quality (128x120 pixels, 10 fps) video conferencing. Simulation runs are divided into four main scenarios. In two of the four scenarios, the transmission power is set to $5x10^{-5}$ W, which enables every node in the network to communicate directly with every other node, thus, creating a single-hop network. In these two scenarios, the one is configured to be a static network and the other to be mobile. In scenarios three and four, the transmission power is set to $2x10^{-5}$ W, forcing nodes in most cases to use multiple hops to communicate with other nodes. Again, one static and one mobile scenario is configured.

TABLE I
General Network Parameters

Parameter	Value	Units
Network Area	100x100	meter (m)
Wireless Nodes	81	none
Transmission Power	$5x10^{-5}$ or $2x10^{-5}$	watt (W)
MAC Layer Protocol	*IEEE* 802.11b	
Data Rate	5.5	Megabits/second (Mbps)
Buffer Size	256000	bits
Node Speed (mobile scenarios)	[0, 5]	meter/second (m/s)
Link Delay Threshold	0.1	seconds
Active Route Timeout	1	seconds
Simulation Time	1200	seconds

TABLE II
Voice-over-IP Parameters

Parameter	Value	Units
Voice Nodes	6	nodes
Encoder Scheme	GSM	
Voice Frames per Packet	5	frames
Max. Delay Request	0.15	seconds

TABLE III
Video Conferencing Parameters

Parameter	Value	Units
Video Nodes	6	nodes
Frame Inter-arrival Time	10	frames/second (fps)
Frame Size	128x120	pixels
Max. Delay Request	0.15	seconds

These four scenarios simulate large office network configurations where on average six persons are busy with VoIP and six other with video conferencing. In such an office scenario, employees would mainly sit at their desks (static scenarios), but can also move around with mobile devices such as laptops, mobile phones or PDA’s (mobile scenarios).

Tables I, II and III summarize the most important simulation parameters. Other protocol specific parameters are kept default as given in the AODV specification [10].

For each scenario, five tests of 20 minutes each are performed, each with a different seed value, used for random number generation. Random numbers are used for certain stochastic events during the simulation such as node movement and data destination selection.

V. Simulation Results

Figures 1 to 8 illustrate the average packet end-to-end delay obtained from the simulations, for both AODV and DA-AODV. End-to-end delay for video and voice packets is illustrated separately. The graphs show the average delay taken over all five simulation runs for each scenario. In all scenarios and tests, the packet end-to-end delay is dramatically improved, and higher packet delivery fractions compared to AODV, occur in most cases.

Table IV summarizes the average delay of all five tests per scenario, as well as the delay experienced in worst cases based on the percentage improvement on the delay experienced with AODV. It also contains the average and worst case improvements of packet delivery fractions (P.D.F.), delay variance, as well as the increase in routing load.

The packet delivery fraction for video packets is improved in all tested scenarios with a maximum

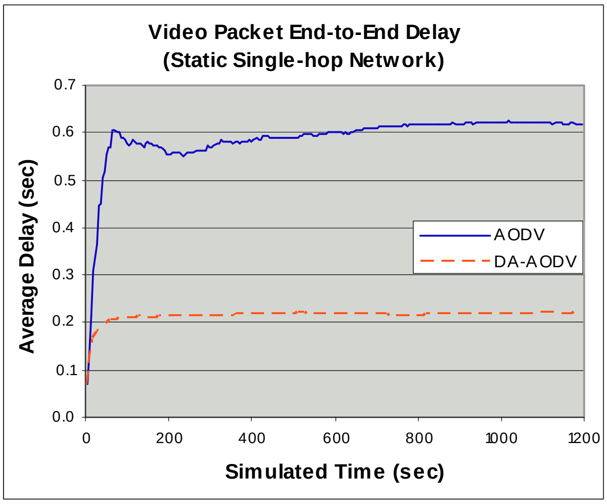

Fig. 1. Static Single-hop Network – Video Packet End-to-End Delay

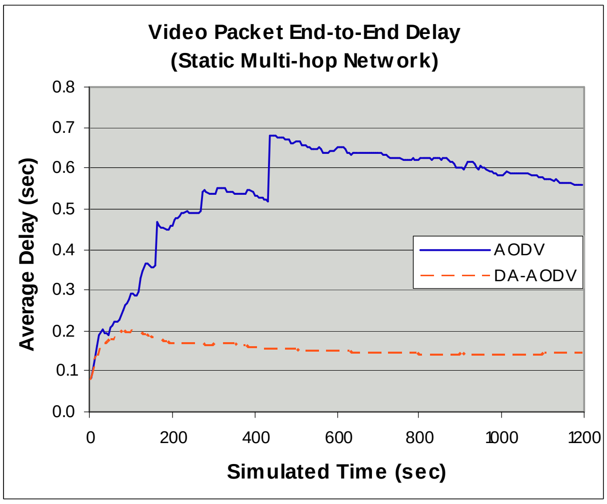

Fig. 3. Static Multi-hop Network – Video Packet End-to-End Delay

improvement of more than 137 times (from 0.66% for AODV to 91.31% for DA-AODV) and a minimum improvement of 26.07%. In 95% of the test runs the improvement was above 100% and in 60% of the tests it was more that 10 times. For voice packets, 70% of the test runs showed a higher packet delivery fraction than AODV with a maximum improvement of 96.13%. The worst delivery fraction of voice packets occurred in a mobile multi-hop network scenario. The delivery fraction was 24.93% worse than that of AODV, but it had a delay improvement of 60.71%. The increase in average routing load varies between 5.20% and 152.15%, with an average increase of 50%.

The average improvement in delay variation is rather consistent (between 80% and 100%) for video traffic, as well as voice traffic in static scenarios. The average improvements for voice delay variation in the mobile scenarios are 73.71% and 65.98%. Only one test run showed worse delay variation than AODV. This occurred in the mobile multi-hop scenario of which the average delay was also the worst.

The fact that MANETs have a very random nature contributes to the variation in the performance improvements. The availability of routes that meet the specified criteria are the most critical, yet uncertain factor

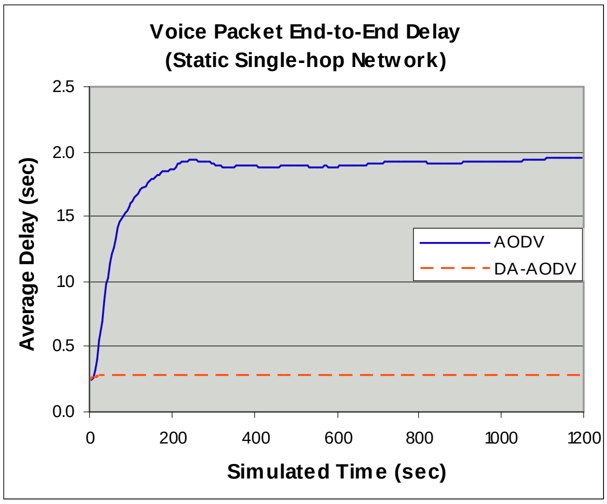

Fig. 2. Static Single-hop Network – Voice Packet End-to-End Delay

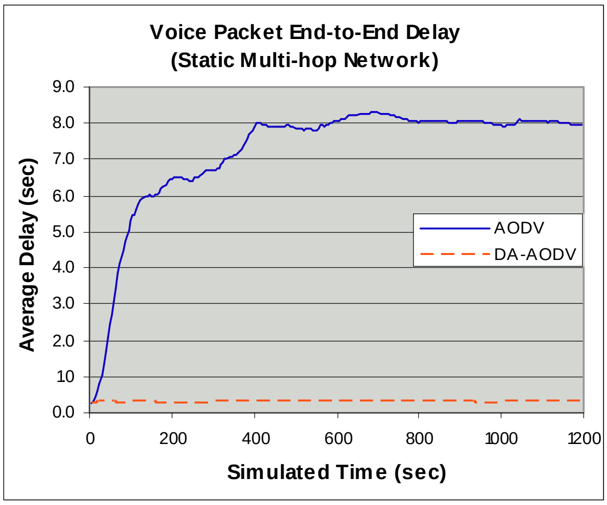

Fig. 4. Static Multi-hop Network – Voice Packet End-to-End Delay

that influences the performance of the protocol. Although the improvements in the mentioned parameters vary within limits, the end-to-end delay improvement, which is the focus of this study, is rather consistent (Table IV).

In the single-hop scenarios, it might seem as if the protocol can't really improve delay since only one single-hop route exists between every source-destination pair. DA-AODV does not produce smaller link delays in these scenarios (or multi-hop scenarios), but does improve packet

TABLE IV
ACHIEVED END-TO-END DELAY AND OTHER QOS IMPROVEMENTS

Static Single-hop Network	Avg. Video Delay	Video Delay Improvement	Avg. Voice Delay	Voice Delay Improvement	P.D.F. (Video)	P.D.F. (Voice)	Delay Var. (Video)	Delay Var. (Voice)	Routing Load Inc.
Average	0.220	64.42%	0.279	85.47%	1624%	5.42%	95.87%	99.67%	58.35%
Worst Case	0.246	59.90%	0.286	82.07%	1009%	1.84%	93.82%	99.50%	75.65%
Static Multi-hop Network									
Average	0.136	64.81%	0.292	93.27%	6108%	24.86%	92.85%	99.73%	9.03%
Worst Case	0.194	34.40 %	0.308	86.60%	26.07%	-15.42%	75.88%	99.26%	20.53%
Mobile Single-hop Network									
Average	0.274	54.14%	0.342	82.95%	1122%	1.41%	85.79%	73.71%	100.05%
Worst Case	0.304	46.78%	0.433	70.60%	484%	-9.38%	80.98%	28.96%	152.15%
Mobile Multi-hop Network									
Average	0.169	55.40%	0.829	78.71%	737%	-5.48%	96.41%	65.98%	33.48%
Worst Case	0.202	45.79%	1.304	60.71%	192%	-24.93%	93.86%	-29.93%	43.60%

end-to-end delay by prohibiting packet transmission if the link delay exceeds the requested delay. The result is better network utilization since resources are not wasted if communication is inefficient or unusable. It is also possible, but rarely occurs, that a multi-hop route is selected over a single-hop route if it offers lower delay (since delay is the main metric and not hop count as in AODV).

VI. CONCLUSIONS

In this study, a solution on how to improve packet end-to-end delay in Mobile Ad-hoc Networks is proposed. Results illustrating the effectiveness of the solution are provided.

The focus was to improve packet end-to-end delay for real-time multimedia applications. Although the delay was

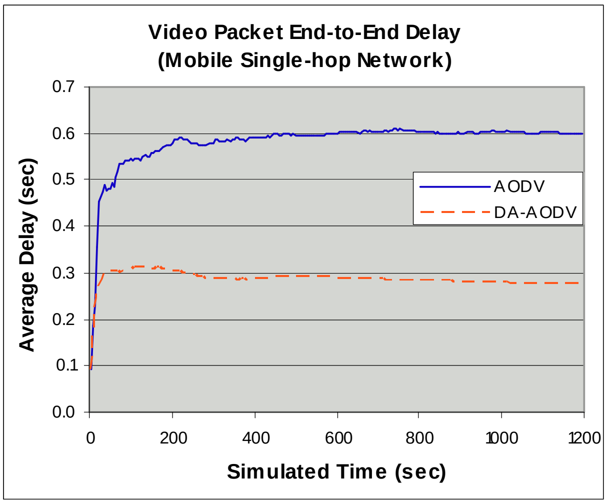

Fig. 5. Mobile Single-hop Network – Video Packet End-to-End Delay

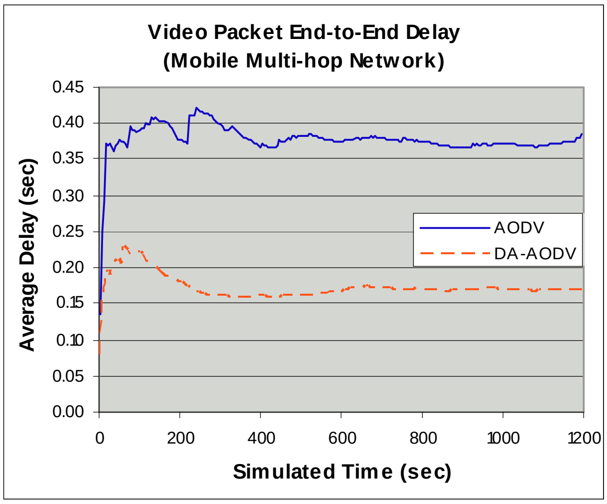

Fig. 7. Mobile Multi-hop Network – Video Packet End-to-End Delay

dramatically lowered in all cases, the delay requests of applications were not fully met (150 milliseconds for voice and video). This is mainly due to factors in other layers of the protocol stack which were not included in the scope of this study. A potential influencing factor is packets being

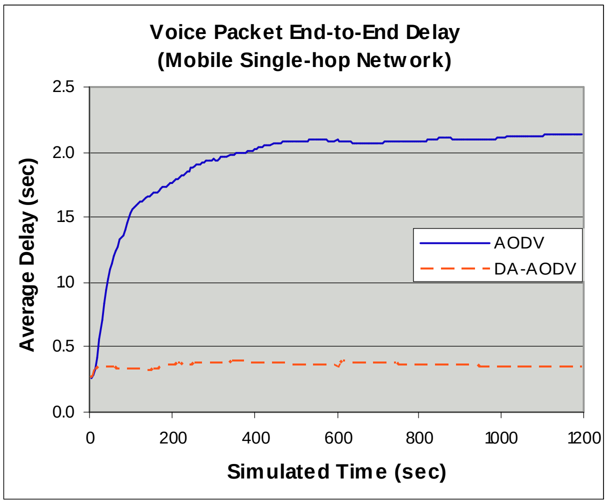

Fig. 6. Mobile Single-hop Network – Voice Packet End-to-End Delay

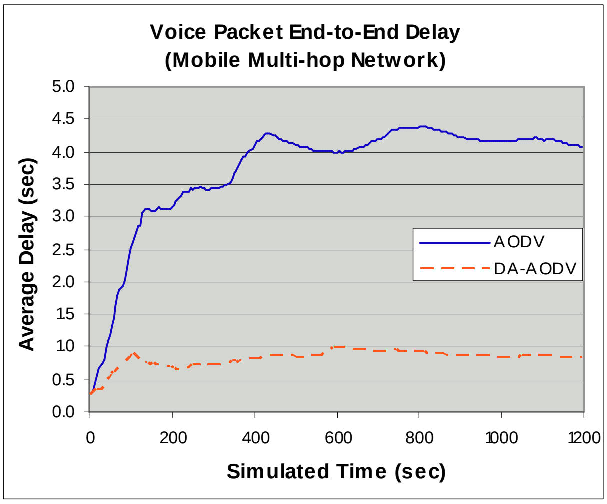

Fig. 8. Mobile Multi-hop Network – Voice Packet End-to-End Delay

queued on the Medium Access Control (MAC) layer, waiting to access the physical medium, as part of the Collision Avoidance (CA) scheme used by the 802.11 MAC layer protocol. This hypothesis is based on the fact that for too strict delay requests (in the order of 50 milliseconds), routing overhead increases significantly since less acceptable routes are available, occupying more bandwidth. This increases the delay of application packets as they are queued for longer periods in order to get access to the shared medium.

Nevertheless, the average packet end-to-end delay was found to be less than 0.4 seconds (for 150 milliseconds requests) in 75% of all tests. The ITU specifies that end-to-end delay of up to 400 milliseconds is acceptable, as long as users are aware of the high delay and the impact on transmission quality due to it [19].

Higher packet delivery fractions in most cases show that routes that are selected, not only provide lower delay, but are also more reliable. Reasons for this could include higher and more reliable signal strengths and the use of less occupied nodes as intermediate hops, avoiding possible bottleneck problems.

References

[1] W. Mateen, S. Raza, Z. A. Uzmi, S. Baqai, "Adaptive Multi-path On-demand Routing in Mobile Ad Hoc Networks", In *Proceedings of ISORC*, 2005, pp 237-244, http://ieeexplore.ieee.org.

[2] M. Marina, S. Das, "On-demand Multipath Distance Vector Routing in Ad Hoc Networks", Ninth Int. Conf. on Network Protocols, November 2001, pp. 13-23, www.computer.org.

[3] Z. Ye, S. V. Krishnamurthy, S. K. Tripathi, "A routing framework for providing robustness to node failures in mobile ad hoc networks", Ad-hoc Networks, Vol. 2, Issue 1, January 2004, pp 97-107, www.sciencedirect.com.

[4] H. Hassanein, A. Zhou, "Load-aware destination-controlled routing for MANETs", Computer Communications, Vol. 26, Issue 14, September 2003, pp. 1551-1559, www.sciencedirect.com.

[5] T. Reddy, S. Sriram, B. Manoj, C. Murthy, "MuSeQoR: Multi-path failure-tolerant security-aware QoS routing in Ad hoc wireless networks", Computer Networks, Vol. 50, Issue 9, June 2006, pp. 1349-1383, www.sciencedirect.com.

[6] M. Ikeda, L. Barolli, A. Koyama, A. Durresi, "Performance evaluation of an intelligent CAC and routing framework for multimedia applications in broadband networks" Journal of Computer and System Sciences, Vol. 72, Issue 7, November 2006, pp. 1183-1200, www.sciencedirect.com.

[7] K. Farkas, O. Wellnitz, M. Dick, X. Gu, "Real-time service provisioning for mobile and wireless networks", Computer Communications, Vol. 29, Issue 5, March 2006, pp. 540-550, www.sciencedirect.com.

[8] D. Johnson, Y. Hu, D. Maltz, "The Dynamic Source Routing Protocol (DSR) for Mobile Ad Hoc Networks for IPv4", IETF RFC 4728, 2007, http://tools.ietf.org/html/rfc4728.

[9] V. Park, S. Corson, "Temporally-Ordered Routing Algorithm (TORA) Version 1 Functional Specification", Internet Draft, 2001, http://www3.ietf.org/proceedings/02mar/I-D/draft-ietf-manet-tora-spec-04.txt.

[10] C. E. Perkins, E. Belding-Royer, S. Das, "Ad hoc On-demand Distance Vector (AODV) Routing", IETF RFC 3561, 2003, http://tools.ietf.org/html/rfc3561.

[11] M. Wang, G. Kuo, "An application-aware QoS routing scheme with improved stability for multimedia applications in mobile ad hoc networks", In *Proceedings of Vehicular Technology Conference*, 2005, Vol. 3, pp. 1901-1905, http://www.ieeexplore.ieee.org.

[12] R. S. Al-Qassas, M. Ould-Khaoua, L. M. Mackenzie, "Performance Evaluation of a New End-to-End Traffic-Aware Routing in MANETs", In *Proceedings of ICPADS*, 2006, pp. 49-54, http://ieeexplore.ieee.org.

[13] C.E. Perkins, E. Belding-Royer, "QoS for AODV", Internet Draft, November 2001, http://tools.ietf.org/html/draft-perkins-manet-aodvqos-00.

[14] I. Gerasimov, R. Simon, "A Bandwidth-Reservation Mechanism for On-Demand Ad hoc Path Finding", In *Proceedings of Simulation Symposium*, April 2002, pp. 27-34, http://ieeexplore.ieee.org.

[15] I. Gerasimov, R. Simon, "Performance Analysis for Ad Hoc QoS Routing Protocols", In *Proceedings of MobiWac*, 2002, pp.87-94, http://ieeexplore.ieee.org.

[16] D. Espes, Z. Mammeri, "Routing Algorithm to Increase Throughput in Ad hoc Networks", In Proceedings of ICN/ICONS/MCL, 2006, p. 66, http://ieeexplore.ieee.org.

[17] S. R. Das, C. E. Perkins, E. M. Royer, "Performance Comparison of Two On-demand Routing Protocols for Ad Hoc Networks", In *Proceedings of Infocom*, 2000, pp. 3-12, http://ieeexplore.ieee.org.

[18] L. Layuan, L. Chunlin, Y. Peiyan, "Performance evaluation and simulations of routing protocols in ad hoc networks", Available online 28 February 2007, www.sciencedirect.com.

[19] International Telecommunication Union (ITU), Recommendation G. 114 – One-way Transmission Time, www.itu.int/itudoc/itu-t/aap/sg12aap/history/g.114.

Jacobus N. Boshoff obtained his B.Eng degree (cum laude) in computer and electronic engineering from the North-West University in 2005. He is currently enrolled at the same university for his M.Eng degree. His current research includes the improvement of Quality of Service in Mobile Ad-hoc Networks.

Albertus S. J. Helberg (Senior Member SAIEE), received his B.Eng (cum laude), M.Eng (cum laude) and D.Eng from the Rand Afrikaans University in 1989, 1991 and 1993 respectively. He is currently professor at the North-West University. His research interests include Telecommunication QoS, Forward Error Correction Coding and Network Coding.