Effective Multimedia and Multi-party Communications on Multicast MANET Extensions to IP Access Networks

Pedro M. Ruiz[1], Antonio F. Gomez-Skarmeta[2], Pedro Martinez[2], Juan A. Sanchez[1], and Emilio Garcia[1]

[1] Agora Systems S.A, Aravaca, 12 3-B, 28040 Madrid, Spain
{pedro.ruiz,jantonio.laguna,emilio.garcia}@agoratechnologies.com

[2] University of Murcia, Dept. of Information and Communication Engineering, Campus de Espinardo, 30100 Espinardo (Murcia), Spain
{skarmeta,pma}@dif.um.es

Abstract. We investigate the problem of offering a good user-perceived QoS for real-time multimedia traffic multicast over self-organizing ad hoc networks extensions connected to IP access networks. At the network layer, we propose a new protocol called MMARP (Multicast MANET Routing Protocol) which is able to provide efficient multicast communications within the ad hoc network extension, and preserves backward compatibility with traditional IP multicast equipment. However, MMARP is not enough because in wireless ad hoc networks the radio link and the mobility of the nodes prevent the network resources to be stable and guaranteed. To overcome that limitation we propose an adaptive application architecture which allows multimedia applications to adapt in real-time their internal settings to the specific network conditions in order to preserve a good user-perceived QoS.

1 Introduction

IP multicast is one of the most reasearched areas since the 80's. Its main benefit is that the bandwidth consumption for group communications is dramatically reduced, something of particular interest for 'all-IP' and 'beyond 3G' mobile networks where the number of user terminals is high and there are many services (e.g. paging) which could benefit from multipoint communications. These networks are expected to be formed by a core IP network which is extended by self-organising multihop ad hoc networks, in which a user terminal employs those of other users as relay points to provide multihop paths between the distant nodes and the fixed network architecture.

The provision of efficient IP multicast communications in such extensions is not straightforward. The usual intra-domain IP multicast protocol suite for fixed network consists of the IGMPv2 [1] protocol for multicast group membership in combination with PIM-SM[2], which is in charge of the IP multicast routing. These protocols are not able to deal with the quick and unpredictable link changes which characterise ad hoc networks, because they would consume too much overhead. Other multicast routing protocols (e.g. CAMP[3], ODMRP[4], ...) have been proposed particularly for ad hoc networks. These protocols incorporate specific functionality which enables them to cope with the particular characteristics of ad hoc networks; however, they are only suitable for isolated ad hoc networks and can neither interoperate with a fixed IP network nor support standard-IP multicast sources or receivers.

In this paper we propose the Multicast MAnet Routing Protocol (MMARP), a new multicast ad hoc routing protocol which is able to deal with the complexity of supporting traditional IP nodes whilst interoperating smoothly with fixed IP networks. MMARP nodes are able to intercept and process IGMP messages. They further permit standard-IP nodes to participate in IP multicast communications as they do when attached to a fixed IP network.

However, the nature of ad hoc networks in which even the routers can move, makes the delivery of a good quality a real challenge for real-time multimedia applications. As network-layer reservations cannot offer strict QoS guarantees in these scenarios, we propose the use of 'adaptive applications', which adapt their behaviour to the network conditions. They are a complement to the network layer QoS mechanisms, which allows applications to preserve a good user-perceived QoS even when the available network resources vary during the session.

This idea of adaptive applications was already known for fixed networks[5]. However, these results are not directly applicable to wireless and mobile scenarios in which in addition to congestion there are many other factors which affect the user-perceived QoS (e.g. fading, mobility, multipath, etc.). There are also works focused on wireless networks ([6–8]) which offer some interesting ideas but their requirements are very difficult to be met in our ad hoc network extensions. So, we present our adaptive application architecture and use it to demonstrate its benefits when used in multihop ad hoc network extensions.

The remainder of the paper is organised as follows: Section 2 discusses the problem of multicast routing in ad hoc network extensions. Section 3 describes the adaptive application architecture. Section 4 shows the empirical results from the MMARP and adaptive application architecture implementations, and finally section 5 gives some conclusions and future work.

2 MMARP for Ad hoc Network Extensions

In ad hoc network extensions, standard multicast ad hoc routing protocols proposed to date are not backwards compatible. In addition, traditional IP multicast routing solutions for fixed networks are backwards compatible but do not offer a good performance. Therefore, we propose the MMARP[10] protocol as a trade-off in which we provide the efficiency of standard ad hoc routing and the smooth interoperability of standard-IP multicast protocols. The key point is that all the new functionality is confined within the ad hoc fringe, and standard protocols like IGMP are used in the boundary with the access network and standard-IP nodes. As we demonstrated in [9], this approach outperforms the other alternatives including the tunnel-based approach which UMTS follows.

For the remaining text we use the terms standard-IP source or standard-IP receiver to refer to a traditional IP Multicast source or receiver and we use the term ad hoc sender or ad hoc receiver to refer to pure ad hoc nodes.

MMARP uses the IGMP protocol as a means to interoperate both with the access network and standard-IP nodes. The interoperation with the access routers is performed by the Multicast Internet Gateways (MIGs), which are the ad hoc nodes situated just one hop away from the fixed network. They are responsible for sending IGMP Reports to the access routers to notify group memberships within the ad hoc network extension. This approach shields the MMARP operation from the protocols performing the intra-domain or inter-domain multicast routing in the IP network.

2.1 Creation and Maintenance of the Distribution Mesh

MMARP uses a hybrid approach to construct a multicast distribution mesh. Multicast routes among nodes within the ad hoc extension are established on-demand whereas multicast routes to the multicast sources in the fixed network are established proactively. This mesh-based distribution structure, offers a good protection against the mobility of the nodes.

The reactive part of the protocol consists of a request and reply phase. When an ad hoc node has new data to send, it broadcasts a MMARP_SOURCE message which is flooded within the entire ad hoc network and it serves as a backward learning mechanism. The proactive part of the protocol is based on the periodic advertisement of the MIGs as default multicast gateways to the fixed network. A node realises it is a MIG when it receives IGMP Queries from the access routers. MIGs periodically broadcast a MMARP_DFL_ROUTE message which is flooded to the whole ad hoc network. Ad hoc nodes receiving these messages are able to select its preferred MIG based on any metric (e.g. less hops).

When one of these messages arrives at a receiver, or at a neighbour of a standard-IP receiver, it broadcasts a MMARP_JOIN message including the IP address of the previous hop that it learnt in the request phase to reach the sources within the ad hoc extension and its preferred MIG, to reach those in the Internet. When an ad hoc node detects its IP address in MMARP_JOIN message, it recognises that it is in the path between a source and a destination. It then activates its MF_FLAG (Multicast Forwarder Flag) and rebroadcasts a MMARP_JOIN message containing its previously learnt previous hop towards the source. In addition, when a MIG receives a MMARP_JOIN message addressed to it, it then sends an IGMP Report towards the multicast router in the access network, to make IP multicast data from sources in the fixed network reach the destinations within the ad hoc network extension. This process ends up creating a shortest multicast tree between the source and the destinations. When there are different sources for the same group, the combination of the different trees form a multicast distribution mesh.

In the case of a standard-IP node becoming an active source, the process for creating the distribution mesh is similar except that the MMARP_SOURCE message is actually generated by the neighbouring ad hoc nodes which receive the data packets from the source. The behaviour of MMARP nodes is presented in Fig. 1.

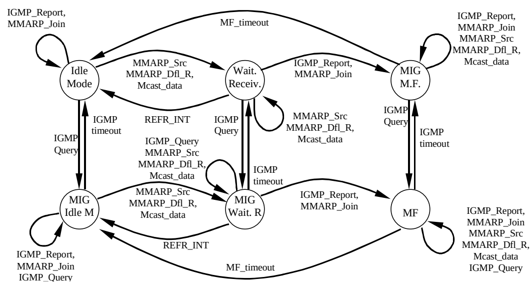


Fig. 1. MMARP state diagram

2.2 Data Forwarding, Reliability, and Loop Avoidance

Data forwarding is very straightforward: when a data packet addressed to group 'G' arrives at a node whose MF_FLAG for group 'G' is active, it checks that the message is not a duplicate and in that case retransmits the packet. In any other case the packet is silently dropped.

In addition, to overcome link breakages during the creation of the distribution mesh, a local repairing mechanism is used. Whenever a node is unable to deliver a MMARP_JOIN message to its next hop after four retries, it broadcasts a MMARP_NACK message to its one-hop neighbours. Upon the reception of this message, the neighbours use their own route to reach that next hop. Should any of them not know an alternate path, they repeat the process until a path is found. Although this recovery process does not offer optimal routes, it offers a quick recovery before the next topology refresh.

Loop avoidance is guaranteed by means of the inclusion of a unique identifier (sequence number) in each packet, which allows intermediate nodes to discard duplicate packets and create only loop-free routes.

2.3 Improving Backward Compatibility Support

In order to be compatible with the standard-IP multicast model, MMARP nodes in the neighbourhood of a standard-IP node have to send MMARP_SOURCE or MMARP_JOIN messages on behalf of the standard-IP node. This means that messages generated by standard-IP nodes may be received by all neighbours and processed independently, creating unnecessary paths. The MMARP protocol has been designed to avoid unnecessary generation of these messages. It includes a field which allows identification of the node which actually triggered the sending of the control message; this allows ad hoc nodes to identify all the MMARP packets which are triggered by a specific standard-IP node, independently of the ad hoc neighbour which actually generated it.

3 Architecture for Real-Time Adaptive Applications

Quality of Service (QoS) is defined in ITU-T recommendation E.800[12] as *the collective effect of service performance, which determines the degree of satisfaction of a user of a service. It is characterized by a combination of service performance factors such as operability, accessibility, retainability and integrity.* Placing some additional features in the application layer would allow presenting a better QoS to the user in environments in which traditional solutions would perform badly. The main items in this architecture are shown in Fig. 2.

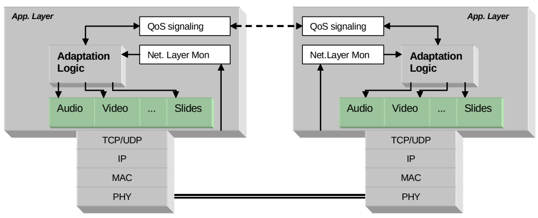


Fig. 2. Framework for Adaptive Applications

3.1 Adaptation Logic

The main problem at the application layer with respect to mobile and wireless networks is that of adapting data transmitted to or received from the application to the characteristics of the different networks, including throughput and delay concerns.

Therefore, the Adaptation Logic can be seen as a somewhat complex function, which receives some inputs (e.g. jitter, bandwidth, loss-rate, user-preferences, etc.) and generates an output (i.e. set of application settings like audio codec, video codec, frame rate, video size, mute of some components, etc.). The complexity lies in making the output take into consideration the subjective aspects of user preferences.

Most of the bad effects perceived by the user are due to packet losses, and therefore the most important input for the Adaptation Logic will be the end-to-end loss rate per reporting period. End-to-end delay problems may also make the user feel a bad quality but this problem can usually be avoided with a proper buffer management without needing to reduce the used bandwidth.

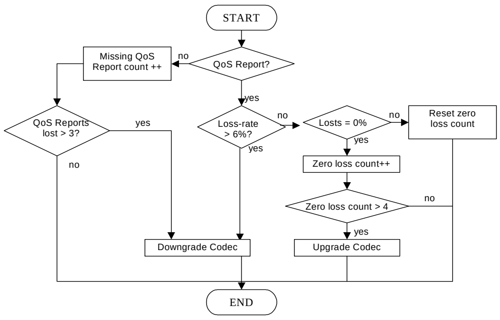


Fig. 3. Schematic diagram of the adaptation logic

The detailed description of the Adaptation Algorithm which is applied to every QoS report received is presented in Fig. 3. As shown, a downgrade in the quality will be only performed when the end-to-end packet loss-rate exceeds 6% or 3 consecutive QoS Reports are lost – possibly due to network congestion or too much interference. Additionally, the quality is upgraded whenever 10 consecutive QoS reports indicating 0% packet loss arrive. These parameters have been set according to our own experience. The complexity of calculating them automatically lies on the existence of subjective components in the user-perceived QoS which may vary from person to person and which are very difficult to model mathematically.

3.2 QoS Signalling Mechanism

The QoS signalling is another key point of the adaptation architecture as it is the only feedback that the source has from the other end. It is basically an end-to-end transport mechanism for signaling data; no special protocol is needed. In fact, it may be enough with a TCP/UDP socket in both sides, and even a standard protocol like Session Initiation Protocol (SIP). In our implementation this quality will be measured in terms of percentage of packets lost and mean delay experienced by the data packets in the network. This information is carried in a special signaling packet called 'QoS Report'. A QoS Report message presents the structure shown in Fig. 4.

Seq	% lost	Delay	User preferences	Estimated BW

Fig. 4. QoS Report Message Format

Some experiments performed in [?] demonstrate that a UDP transport is much more appropriate to carry the feedback than a transport using TCP. TCP retransmissions result in stale QoS information especially on a congested network. An additional issue is that the feedback packet itself has to traverse the network back to the server, and the probability of it actually making it there on time is inversely proportional to its importance. That is, a feedback packet is most important when it carries information about a congested network and it is not important when it is just saying that all is going well. To overcome this problem, we use a periodic reporting approach, by which the clients send periodic and sequenced reports towards the sources. This way, whenever network problems come up, the sender can detect missing reports. The sender uses the heuristic of downgrading the quality when a certain number of QoS Reports are lost. We demonstrated [11] this approach to be very effective in scenarios with abrupt changes in bandwidth.

4 Empirical Results

In order to evaluate the effectiveness of our proposal, we have set up a real MMARP-based ad hoc testbed, on which we will compare the performance of real-time videoconferencing both with traditional applications and with 'adaptive applications'. The testbed has been deployed in the basement of the CS Faculty at the Univ. of Murcia (see Fig. 5). MMARP nodes are numbered from 1 to 4, R is a standard-IP receiver and S is a standard-IP source. The source (S) follows (at walking speed) the path which is shown in Fig. 5 while it runs a videoconference session with node (R).

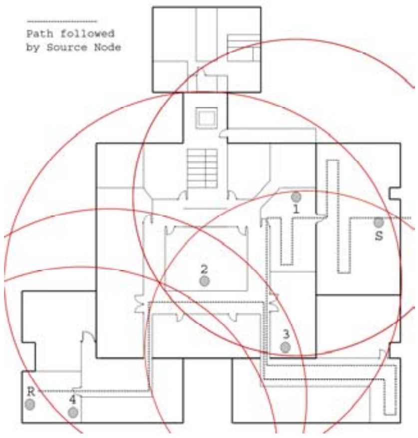


Fig. 5. Map of the testbed scenario

The route has been specifically selected so that link breaks and MMARP route changes take place during the videoconferencing session . Furthermore, the signal strength changes due to the variation of the distance to MMARP nodes and the number of intermediate walls to traverse. This makas the available bandwidth vary during the session.

The trials have been performed using our MMARP implementation for Linux. It is a user-space daemon which handles MMARP packets before they are processed by the TCP/IP stack. In addition, we have also extended the RTP-based ISABEL-lite videoconferencing application to use our adaptive application framework. The settings which are used by the application as adaptation steps are given in Table 1. In order to guarantee a fair comparison of both approaches, the adaptive application starts with the quality step number 3, which is the only one used by the non-adaptive application, and is around the mean bandwidth which we calculated during the whole session. The results which we present are extracted from the RTP traces which are generated by the videoconferencing application. We have used the same route, at the same speed and in the same network conditions for the adaptive and non-adaptive trials.

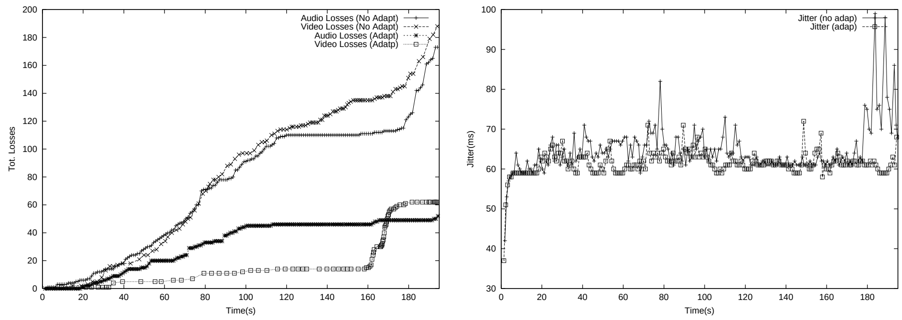
Table 1. Quality steps for the real-time adaptive application

Step #	A. Codec	V. Codec	V. Size	V. Rate	V. Quantity	Estimated BW
0	GSM	-	0 fps	-	-	20 Kbps
1	GSM	MJPEG	4 fps	SQCIF	50	80 Kbps
2	G.722	MJPEG	8 fps	SQCIF	40	140 Kbps
3	G.722	MJPEG	6 fps	QCIF	50	190 Kbps
4	G.722	MJPEG	4 fps	CIF	30	230 Kbps
5	G.722	MJPEG	6 fps	CIF	50	350 Kbps

The results presented in Fig. 6(a) show that the use of adaptive applications is able to reduce the overall packet losses both for audio and video to approximately 1/3. As expected, the differences are higher in the periods in which there is less bandwidth available. This is also noticed in the variation of the delays depicted in Fig. 6(b). In the same critical periods, the non-adaptive approach is not able to control the growing of the end-to-end delay, whereas the adaptive one is able to quickly restore the original state.

The overall packet losses is a good reference to identify the points of the trial in which the network conditions are most critical. This is identified by an increase in the slope of the total packet loss curve. However, what really affects the user perception of QoS is the instantaneous loss-rate, which is what causes the service disruptions. For example, a 20% packet loss-rate can be considered as the point in which an audio flow is perceived with poor quality.

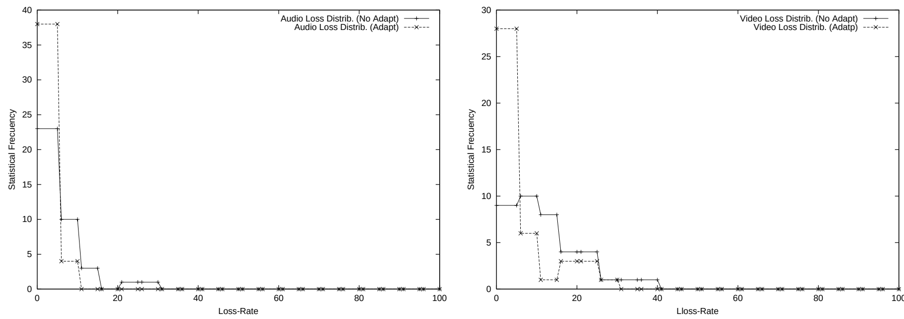
In Fig. 7(a), we compare the statistical histogram for the distribution of the audio loss-rate for both approaches. The same statistical analysis is performed for the video flow in Fig. 7(b). For example, for the audio flow, the adaptive application approach is able to keep the loss-rate below 10% all the time. In fact, it keeps the loss-rate below 5% during the 91% of the time. For the video



(a) (b)

Fig. 6. Total Losses (a) and Audio Jitter (b)

flow, the loss-rate is kept under the 5% the 64% of the time, and its has been under the 10% the 78% of the time.



(a) (b)

Fig. 7. Audio Loss-rate distribution (a) and video loss-rate distribution (b)

These figures clearly demonstrate that the adaptive application approach on top of our MMARP implementation has been able to offer a very good user-perceived QoS in a scenario in which traditional solutions offer less performance. Furthermore, both for audio and video flows the adaptive applications approach outperforms the traditional real-time multimedia approach, demonstrating that adaptive applications are a good approach for dealing with the complex network conditions by which ad hoc networks are characterised.

5 Conclusions and Future Work

We present our integrated solution (i.e. ad hoc routing and adaptive application framework) to provide efficient real-time multiparty communications in ad hoc network extensions. We demonstrate through experimentation that MMARP offers a good performance supporting IP multicast communications to standard-IP nodes in such scenario. In addition, we demonstrate that the use of our adaptive application framework offers a much better user-perceived QoS than traditional approaches.

As future work, we are working in making the adaptation logic to be completely automatic and intelligent using artificial intelligence techniques.

Acknowledgements

This work has been partially funded by the Spanish Science and Technology Ministry, by means of the project ISAIAS(TIC2000-0198-P4-04).

References

1. Fenner, W.: Internet Group Management Protocol, Version 2. RFC 2236, November, 1997.
2. Estrin, D., Farinacci, D., Helmy, A., Thaler, D., Deering, S., Handley, M., Jacobson, V., Liu, C., Sharma P, Wei, L.: Protocol Independent Multicast Sparse Mode (PIM-SM): Protocol Specification. RFC 2362, June 1998.
3. Garcia-Luna-Aceves, J.J., Madruga, E.L.: The Core Assisted Mesh Protocol. IEEE JSAC, Vol 17, No. 8, August 1999, pp.1380–1394.
4. Lee, S.-J., Su, W., Gerla, M.: On-Demand Multicast Routing Protocol in Multihop Wireless Mobile Networks. ACM/Kluwer Mobile Networks and Applications, 2000.
5. Sisalem, D.: End-to-end Quality of Service Control using Adaptive Applications. IFIP International Workshop on Quality of Service, 1997.
6. A. Alwan, R. Bagrodia, N. Bambos, M. Gerla, L. Kleinrock, J. Short, and J. Villasenor, *"Adaptive Mobile Multimedia Networks"* IEEE Personal Communications, April 1996, pp. 34-51.
7. M. Mirhakkak, N. Schult, and D. Thomsom, *"Dynamic Bandwidth Management and Adaptive Applications for a Variable Bandwidth Wireless Environment"* IEEE JSAC, October 2001, pp. 1985-1997.
8. R. Ramanathan, and R. Hain, *"An Ad Hoc Wireless Testbed for Scalable, Adaptive QoS Support"* IEEE WCNC, November 2000, pp. 998-1002.
9. Ruiz, P.-M., Brown, G., Groves, I.: Scalable Communications for Ad hoc Extensions connected to Mobile IP Networks. Proceedings of the (PIMRC'2002). Lisbon, September, 2002. Vol. 3, pp. 1053–1057.
10. Ruiz, P.-M., Gomez-Skarmeta, A., Groves, I.: Multicast Routing for MANET Extensions to IP Access Networks: The MMARP Protocol. International Workshop on Mobile IP-based Network Developments. London, October, 2002. pp. 75–81.
11. Ruiz, P.-M., Garcia, E.-J.: Improving User-perceived QoS in Mobile and Wireless IP Networks Using Real-Time Adaptive Multimedia Applications. Proceedings of (PIMRC'2002). Lisbon, September, 2002. Vol. 3, pp. 1467–1471.
12. ITU-T Recomendation E.800 (0894): Terms and definitions related to quality of service and network performance including dependability.