Institute of Communication Networks and Computer Engineering
University of Stuttgart
Pfaffenwaldring 47, D-70569 Stuttgart, Germany
Phone: ++49-711-685-68026, Fax: ++49-711-685-67983
Email: mail@ikr.uni-stuttgart.de, http://www.ikr.uni-stuttgart.de

Multicast Mobility in Heterogeneous Technologies: Experimental Validation

[1]Susana Sargento, [2]David Wagner, [1]José Rocha, [3]Fabio Mitrano, [4]Janusz Gozdecki, [2]Jens Mödeker
[1]Instituto de Telecomunicações, Universidade de Aveiro, Aveiro, Portugal
[2]Fraunhofer FOKUS Institute, Germany
[3]Università di Roma "La Sapienza", Dip. di Informatica e Sistemistica, Rome, Italy
[4]AGH University of Science and Technology, Dep. of Telecommunications, Krakow, Poland
susana@ua.pt, david.wagner@fokus.fraunhofer.de, jrocha@av.it.pt, fabio.mitrano@dis.uniroma1.it, gozdecki@agh.edu.pl, jens.moedeker@fokus.fraunhofer.de

***Abstract* – The convergence of services and technologies has been driven by service and network providers with the aim to develop a unified infrastructure. From the services side, we have multimedia through unicast, multicast and broadcast services. From the technologies side, we have wired and wireless technologies, including unidirectional technologies such as Digital Video Broadcasting (DVB). These trends allied to the increasing mobility and QoS demands, introduce strong requirements to future telecommunication networks.**

This paper presents an innovative approach to handle multicast services in a heterogeneous networks environment, including broadcast technologies. The presented architecture aims at guaranteeing end-to-end QoS in mobile scenarios, efficiently handling the underlying network resources and integrating the emerging broadcast technologies. This architecture was developed in a real environment with mobility of multicast sessions through heterogeneous technologies, Wi-Fi and DVB, using also Wi-Fi as a return channel. The results show that the architecture is able to support the seamless mobility of users receiving multicast sessions, with low degradation on the running communications.

Index Terms — **QoS, mobility, multicast, broadcast technologies, media independent handovers.**

I. Introduction

In the last years there has been an increasing trend to unify all telecommunication services, such as telephony, TV and Internet under an unique infrastructure able to seamless integrate both wired and wireless networks. Moreover, several innovative network architectures have been proposed for this purpose (most of the effort has been focused on the emerging IPv6 network protocol and its limits in terms of Quality of Service (QoS) management). Although on one hand several solutions have been proposed to handle services like VoIP based on unicast flows, on the other hand, to handle services like video streaming, multicast forwarding techniques and their efficiency in terms of network resources usage have been neglected due to difficulties to face the additional complexity and scalability concerns.

In this paper we describe a QoS and mobility architecture able to seamlessly support multicast services in heterogeneous technologies, including unidirectional Digital Video Broadcasting (DVB) networks. This architecture was developed under the framework of IST FP6 Daidalos II project [1], is characterized by a hierarchical structure to support QoS, and extends the current Media Independent Handover standard IEEE 802.21 [2] to seamlessly integrate mobility and QoS. To enable the support of multicast services in unidirectional technologies, the Multicast Subscription System (MSS) protocol is introduced: this mechanism needs a return channel only for subscription. This architecture was implemented in a real environment with mobility of listeners through different technologies, Wi-Fi and DVB, using also Wi-Fi as the return channel. The results show that the architecture supports the seamless mobility of users receiving multicast sessions, with low degradation on the running communications. To the best of our knowledge, this is the first real demonstrator containing QoS and mobility of multicast sessions through heterogeneous technologies.

This paper is organized as follows. Section II presents related work in the area and section III describes the general architecture. Then, section IV depicts the multicast and broadcast requirements that guided the design of new protocols and mechanisms, and sections V and VI describe, respectively, the process of both source and listener session setup with QoS, and mobility of multicast sessions with QoS support. Section VII presents the real testbed and the obtained results. Finally, section VIII concludes the paper and introduces future work.

II. Related Work

The efficient integration of mobility and IP multicast is still a challenge mainly due to IP address changes: after handover, sessions must be re-established to receive multicast data on the new position. Mobility of multicast receivers is currently possible in existing multicast routing protocols, such as Protocol Independent Multicast (PIM) [10], by means of Mobile IP (MIP) for IPv4 [3] or IPv6 [4] networks. Unfortunately, IP multicast and MIP can place service degradations during session re-establishment in foreign networks, which is not acceptable for real-time multimedia applications.

In MIPv6, Remote Subscription (MIP-RS) and Bi-directional Tunnelling (MIP-BT) strategies were introduced as attempts to overcome the problems during handover. The main idea behind MIP-RS consists in using the Context Transfer Protocol (CXTP) [5] to allow the transfer of context between multicast-enabled Access Routers (AR). Each mobile node must re-subscribe to the desired multicast group upon entering

978-1-4244-4148-8/09/$25.00 ©2009
This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE "GLOBECOM" 2009 proceedings.

a foreign network. Besides providing optimal routing, remote subscription can place excessive processing and signalling overhead to reconstruct the multicast tree, depending on the frequency of handoffs. Moreover, mobile nodes are forced to re-initiate multicast distribution after handover, and rely on multicast dynamics to adapt to network changes. Multicast Scheme for Wireless Networks (MobiCast) [6] and Mobile Multicast with Routing Optimization (MMROP) [7] are examples of current proposals using MIP-RS. MMROP introduces the Mobility Agent (MA) entity to ensure routing efficiency and no packet losses from roaming. MAs are Foreign Agents (FAs) that route missing packets (via tunneling) to neighboring subnets. MobiCast's key extension is the introduction of the Domain Foreign Agent (DFA) which serves many small adjacent wireless cells, and then a hierarchy is introduced, as in Hierarchical Mobile IP approaches.

In MIP-BT approaches, when a mobile multicast source aims to redirect its multicast flow through the home network, it must tunnel the data to its Home Agent (HA). The HA receives the multicast packets from the tunnel and sends out the packets using IP Multicast Routing on behalf of the mobile multicast source. This fundamental multicast solution hides all movement since HAs remain fixed and results in static multicast trees. It may be employed transparently by mobile multicast listeners and sources, at the cost of significant performance degradations due to the overhead on the network and also the delay on the data delivery. The Mobile Multicast (MoM) Protocol [8] is an example of proposals using MIP-BT. MoM's key extension is the use of a Designated Multicast Service Provider (DMSP) that solves a tunnel convergence problem. A DMSP for a given multicast group is a HA chosen by the visited subnet's FA out of the many HAs that forward packets for the specific group to the visited subnet.

Neither MIP-RS nor MIP-BT related proposals have efficient and seamless support to multicast mobility, because they cannot deploy mobility in a transparent manner. The limitations identified in the related work analysis motivated the design of a QoS-aware and media independent architecture to optimize multicast mobility support. Moreover, the use of context transfer in inter-domain scenarios is very complex; the use of our Multicast Subscription System (MSS) protocol supports inter-domain while providing efficient routing through PIM-SM (more information in sections V and VI).

III. QoS AND MOBILITY ARCHITECTURE FOR MULTICAST

The proposed QoS and mobility architecture introduces a hierarchical approach to fully integrate several heterogeneous technologies and to efficiently support seamless mobility among them, also when considering multicast services (Fig. 1). The architecture also considers the support of multicast transmission over broadcast networks. There are three possible scenarios for this purpose: multicast transmission with no return channel, i.e. on pure unidirectional links; multicast transmission in the presence of a permanent return channel; and multicast transmission with a temporary return channel. We will need the support of a return channel to provide the multicast subscription; however, it can be temporary.

In the architecture, access routers (ARs) connect mobile terminals (MTs), listeners or sources, to the Access Network (AN). In addition, all access networks are connected through Core Networks (CNs). The architecture contains an entity that manages the layer 3 QoS inside a Local Mobility Domain (LMD), denoted as Zone QoS-Broker (ZQoSB). The ZQoSB can be centralized or distributed, and is enhanced with the Multicast Manager (MM) module to provide authorization, access control and QoS management as well as handover support for multicast services. The ZQoSB is also a policy enforcement point of the Authentication, Authorization, Accounting, Auditing and Charging (A4C) subsystem, controlling access to the network by QoS mechanisms deployed in ARs.

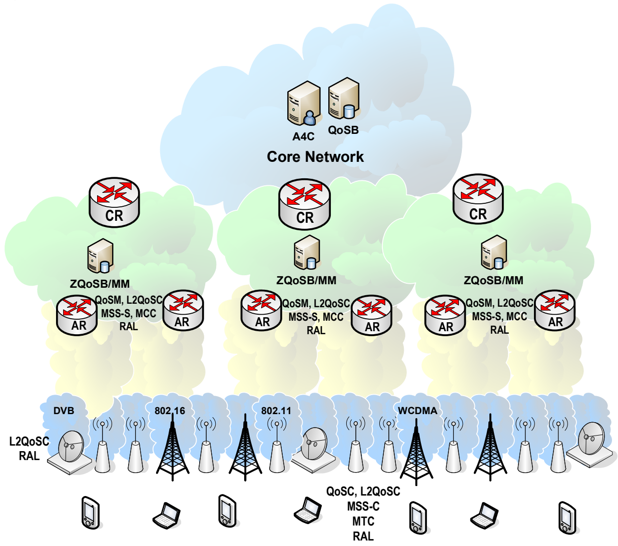

Fig. 1. Multicast QoS and Mobility Architecture

In the access network, both MTs and ARs contain elements that perform the enforcement of the QoS in the network and trigger the QoS process for admission control and resource reservation: QoS Client (QoSC) and QoS Manager (QoSM), respectively in the MTs and ARs. The handling of reservations at Layer 2 is performed by a L2 QoS Controller (L2QoSC). The specific characteristics and reservation handling of each technology are executed by a Radio Access Layer (RAL).

For mobility purposes, we consider the upcoming standard IEEE 802.21 Media Independent Handovers (MIH) [2], extended to seamlessly integrate QoS features. The IEEE 802.21 is the common denominator that abstracts the heterogeneity of the access technologies, and will then be extended to join together QoS provisioning and mobility: such a mobility and QoS integration will contribute to a clean network design and simplified network operations. For mobility control, the MT contains a Mobile Terminal Controller (MTC) module: the process of interface selection considering QoS requirements is performed through the interaction between QoS Client (QoSC) and MTC.

Although Multicast Listener Discovery (MLD) [9] is the protocol used in IPv6 environment to enable the end terminals to join and leave the multicast sessions of interest, regarding QoS handling and heterogeneous access networks including unidirectional broadcast accesses such as DVB, the Multicast Subscription System (MSS) protocol was introduced. It provides a new protocol that only needs the return channel for subscription and, if the receiver loses uplink connectivity, the desired packets will still be forwarded until the subscription timeout is reached. In this sense, the architecture also contains multicast modules in the MTs and ARs. On the MT side, the MSS-Client (MSS-C) is used to subscribe an existing multicast service, by interacting with the QoSC, and to create a new one from a multicast source. On the AR side, the multicast support is provided with the MSS-Server (MSS-S) which interacts directly with the MSS-C, and a Multicast Controller (MCC) in the AR, which is the entity that deals with subscription requests, authorization, and handles the multicast routing.

In summary, the architecture contains the following new contributions: the multicast subscription system that enables flexible subscription lifetime and supports unavailability of return channel; the QoS reservation inherent in the subscription and in the handover; the integration of IEEE 802.21 media independent handover with multicast mechanisms and QoS; the support of handover both of the direct and return channel.

IV. Multicast and Broadcast Requirements

To enhance the Daidalos II QoS architecture with multicast and broadcast support, the following main assumptions and requirements have been taken into account:

- The Source Specific Multicast (SSM) model is considered to add more granularity to the management of network resources; in particular PIM-SM [10] with SSM extensions is adopted. PIM-SM implies a shared multicast tree for two reasons: tree optimization and new source discovery (by means of a Rendezvous Point – RP - acting as root of the shared tree). However, PIM-SM allows the switching from the shared tree to the shortest path tree (from receiver to sources), when the listener access router receives the first multicast packets indicating the source address (thus using the RP only for source discovery). All ARs are configured for this immediate switching in order to both simplify the reservation of resources, by just handling the shortest path trees, and to increase the scalability, by avoiding bottlenecks located at the RPs.
- QoS requests are source driven. This means that the QoS requirements necessary to join a multicast source are associated to the source itself. QoS requirements should be available to the ZQoSB in order to allocate resources for multicast receivers.
- The user should know about the qualitative mapping of service quality (high, low, medium) and prices to source and multicast addresses: it can be informed by a service discovery mechanism. This information will allow the user to choose the preferred source. The users with different requirements and receiving the same service, but with different qualities, belong to different multicast groups, with different QoS reservations (in this case, driven by the service itself).
- Broadcast networks, like DVB ones, have just the unidirectional downlink channel, while the uplink (interaction) channel is provided by other technologies. L2 tunneling mechanisms over uplink paths will hide this asymmetry; therefore, unidirectional ARs are treated by ZQoSB like bidirectional ones.
- Legacy terminals can join a multicast group using network level protocols such as MLDv2 [11]. Instead, Daidalos-compliant terminals will adopt a MSS in order to overtake the MLD limitations and to fully support QoS and broadcast with return link realized by tunneling mechanisms. Both the legacy and the MSS multicast subscription procedures are based on direct interaction between terminals and ARs. The reasons to define a new subscription protocol are the following:
  - The MLD messages do not include any QoS information about the requested multicast sessions.
  - The MLD protocol does not enable the AR to send responses towards the end terminals requesting to join a given multicast session informing them whether their requests can or not be handled with respect to the multicast admission control.
  - The MLD protocol is defined to be used only at the multicast receiver side. The multicast source is not able to express its desire to start sending multicast traffic and its need in terms of QoS and resources.
  - The MLD join messages have pre-defined lifetimes and, therefore they need to be refreshed continuously and almost periodically by the multicast receivers located in the same local sub-network. These fixed lifetimes are not flexible enough when dealing with unidirectional links such as DVB with non-guaranteed uplink availability.
  - MLD routers might request for listener confirmation when a node leaves the group, so a permanent ability to respond to MLD requests is needed at any time.

The MSS is introduced to be used at the edge links between the end terminals and the ARs, as an improvement of the MLD protocol functionalities. The MSS provides enhanced edge multicast management compared with MLD and it also targets the multicast sources. For this purpose, a multicast source should use the MSS protocol to inform its corresponding AR about its will to set a multicast session and its associated QoS parameters. The multicast receivers can specify within their multicast subscription requests the desired subscription lifetime and resource. This flexibility regarding the subscription lifetime is of particular interest in case of unidirectional links where the associated uplink channel cannot be permanently guaranteed, as highlighted above. At the AR, a multicast subscription server is defined in order to handle the terminal requests and inform the internal multicast routing framework about them. Finally, the multicast subscription server sends responses to the multicast subscription request according to the decisions taken by the ZQoSBr.

V. QoS and Multicast Session Setup

Taking into account the previously mentioned requirements, with source driven QoS requests, it is required to perform the session setup mechanism for the source in order to manage access control and resource allocation. The next sections describe this process for both sources and receivers.

A. Source Session Setup

The message flow of a source session setup with QoS support consists of two phases as shown in Fig. 2.

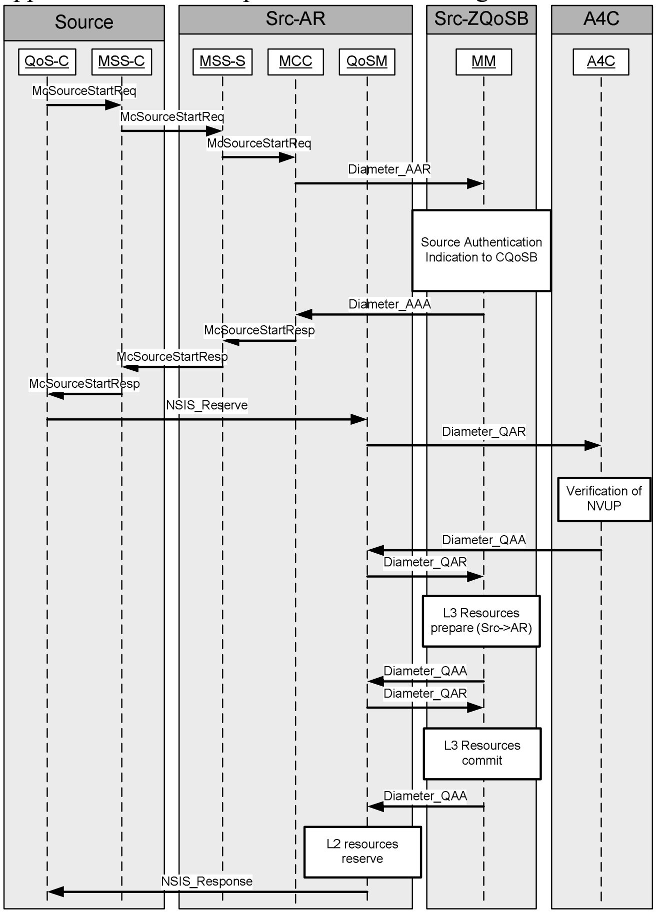

Fig. 2. Source subscription and QoS resource reservations

The first phase covers the check for available resources and provides the QoS requirements of the service to be started to the ZQosB in the source domain. This information is needed to handle future intra- and inter-access network join requests. The process is based on MSS for the communication between AR and MT, and Diameter [12] for the communication between AR and the respective ZQosB. The second phase covers authentication and authorization combined with resource reservation in case of successful completion of the first phase. It uses Next Steps In Signaling (NSIS) [13] in the access network (between the MT and AR) and Diameter for communication with A4C for source authentication according to its profile (Network View of User Profile - NVUP), and with the ZQosB for resources reservation.

Upon this process, the AR also performs L2 reservation in the corresponding technology through the RAL, and a positive notification is sent to the MT. After completion of this process, the source may start sending data which will be forwarded to the Rendezvous Point according to PIM-SM. QoS is only guaranteed for the access network of the source; additional resources will be reserved on demand for the path to subscribers (customers subscribing multicast services) including their access network.

B. Listener Session Setup

Fig. 3 shows the process of listener session setup, considering that this listener is connected to a DVB AR and to a Wi-Fi AR return channel through a link-layer tunnel [14].

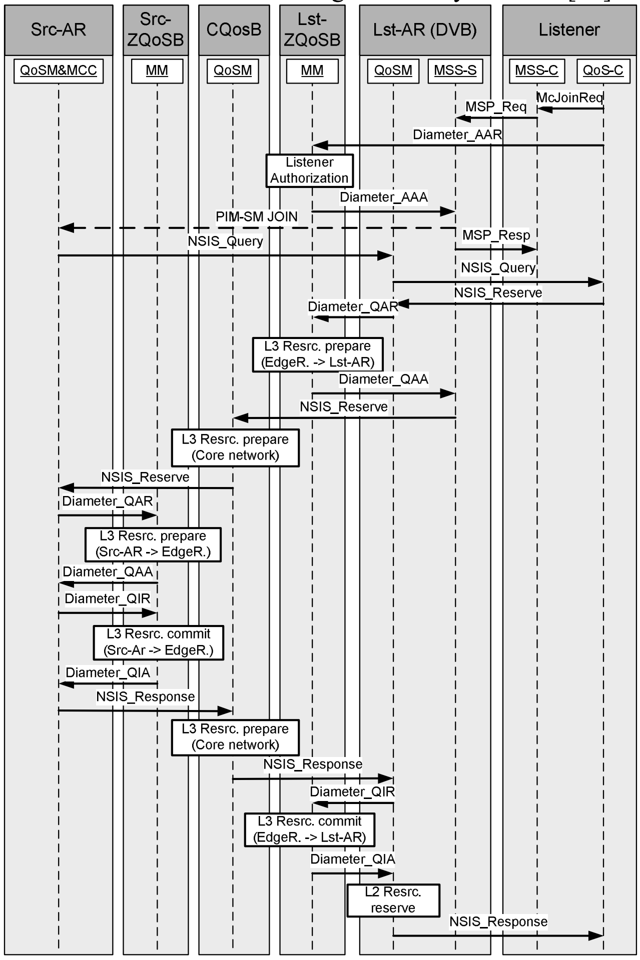

Fig. 3. Listener subscription and QoS resources reservation

When the first user in an access network subscribes to a multicast service, the process is based on the same protocols as the source session setup, but it may cross several administrative domains. First, the MSS request is sent by the MT to the connected AR. The AR checks for authorization using Diameter in the ZQoSB. If this request succeeds, the Multicast Routing Daemon (MRD6) [15] joins the PIM-SM tree of the specified group and source. This ensures that the optimal routing path is used. When this request reaches the AR of the source, it provides an answer with the requirements of the requested service using NSIS to the MT. The MT triggers the resource reservation from the source AR to the access network it is connected to (in the example scenario the DVB access network). Since the source may be located in another domain, the listener ZQoSB may need to contact a core QoSB that in turn contacts the source ZQoSB. The reservation is

978-1-4244-4148-8/09/$25.00 ©2009
This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE "GLOBECOM" 2009 proceedings.

prepared through NSIS messages from the listener to the source access network and the resources are reserved on the way back.

Differently from the unicast case, local or remote resources can be already allocated due to other users listening to the same service. For this reason, the admission control algorithm will perform a resource check only when it is necessary to build a new branch of the multicast tree. There is caching and intelligent reservation at several levels: if the listener ZQoSB already knows the requirements, and forwarding and resource reservation are provided up to its domain, it will not contact the source ZQoSB.

When the PIM join reaches the source AR, it will start forwarding traffic on best effort basis. Two cases are possible: either the Join message is sent via the shared tree, then the listener AR will switch immediately to the shortest path tree since QoS is only guaranteed along it; or the Join message is directly sent via the shortest path tree so the traffic will profit from resource allocation from the beginning.

VI. QOS AND MULTICAST MOBILITY

The handover management on the MT is based on IEEE 802.21 MIH and includes management of multicast handovers with QoS support. The MIH based handover consists of three phases as depicted in Fig. 4:

- Initiate phase: the MT finds out if a handover to a specified network is possible, taking A4C and QoS into account.
- Commit phase: MT decides to execute the handover, the resources in the new access network are allocated, and the MT triggers the L2 handover.
- Complete phase: MT is attached to the new network and the resources in the old access network are released.

All three phases use a MIH signaling between MIH functions in different entities in the network. The process starts when the Mobile Terminal Controller (MTC) decides to move to another point of attachment, most probably based on MIH events e.g. informing about a low Signal to Noise Ratio (SNR) for the old access network. The MTC is a MIH user and triggers the MIH Function (MIHF) to send a Handover Initiate Request; this message contains the information of the network the MT intends to move. This information is forwarded to the ZQoSB which checks for resource availability and prepares the resources in the new network. If the ZQosB of the two networks differ but are in the same domain or in sufficiently federated domains, NSIS signaling is used between the brokers (not covered in the figure). Notice that NSIS is only processed in access networks. When a positive answer reaches the MTC, it will switch to commit phase and send a MIH handover commit message. When it reaches the ZQoSB, it reserves L3 resources to the new access network and triggers L2 reservations in the specific technology through the RAL in the ARs. If the MT listens to a multicast session, it will also trigger the Multicast Controller (MCC) of the AR of the new access network to join the respective multicast streams (Source, Group) in advance. When this process is completed, the MT is informed by a MIH response message. At this stage all resources in the network are ready for the MT to handoff.

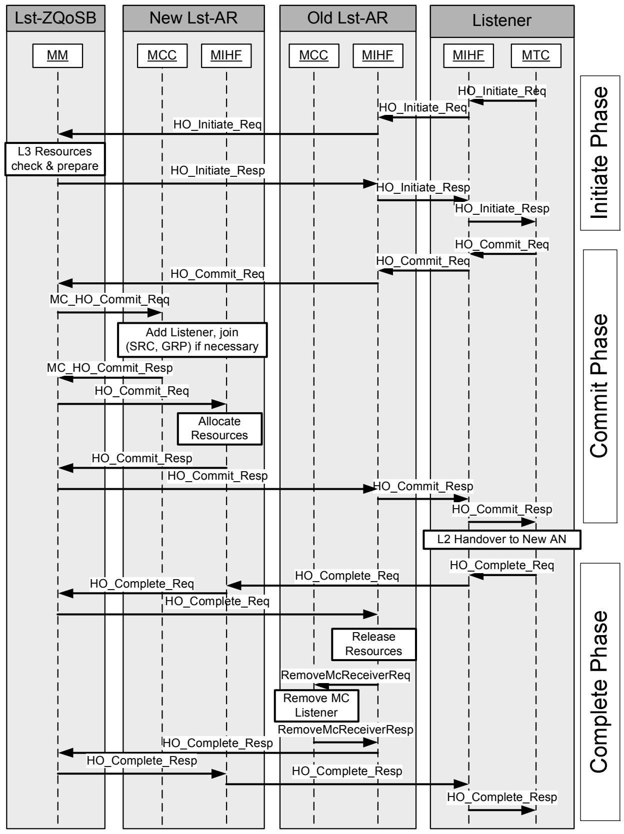

Fig. 4. Multicast Handover with QoS Support

After the link layer handover, the MT will be able to continue receiving the multicast streams it previously joined. It immediately informs the new AR about the successful handover by a MIH Handover Complete message. This message is forwarded to the ZQoSB which triggers the release of resources in the old AR. This applies to network resources managed by the QoSM as well as for the Multicast Group Membership: the MCC is informed that this listener has left the network; since MCC manages group membership explicitly, it knows if there are still listeners for these groups and can leave all multicast groups which are not needed for this access network without the need for active querying. Then, the MT is informed about the successful completion of the handover.

VII. TESTBED AND EXPERIMENTAL RESULTS

This section contains the description of the testbed used in the real experiments, and both control (signalling) and data performance results.

The testbed scenario is depicted in Fig. 5. It fulfils the hierarchical concept proposed in the overall architecture. The lowest layer includes the MTs, source and listener. The source is a desktop PC that is connected to the source AR via Ethernet (fixed node); the listener is a laptop PC connected to the Wi-Fi network. Both terminals run a Daidalos II modified kernel with all the changes required for the QoS and mobility support, including the Mobile IPv6 [4] support. The second layer is composed by the access networks, the ARs, and the ZQoSB

This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE "GLOBECOM" 2009 proceedings.

which contains a Multicast Manager. The top layer is the core network that contains a Home Agent that maintains a map of Care-of-Addresses (CoAs) and their corresponding Home Addresses (HoAs), allowing the routing of data to the right destination, including the handling of handover situations. The A4C is also used to perform user authentication and authorization. The core router connects to the core network; it also works as local mobility anchor, which will be used, together with the Home Agent, to control local mobility. The core and access networks are built in Ethernet connections with a maximum bitrate of 100 Mbps.

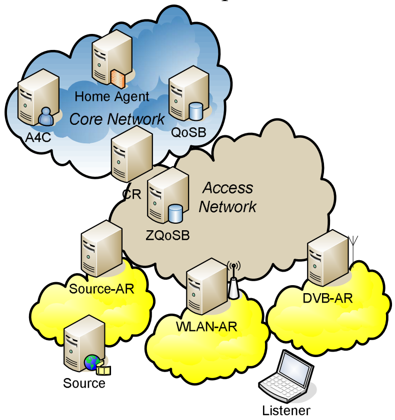

Fig. 5. Testbed Scenario

The DVB transmission parameters used in the testbed are the following: bandwidth is 8 MHz, FEC is 1/2, the modulation is 64-QAM, the transmission mode is 8k,the guard interval is 1/8 and there is no hierarchy. The testbed used a DVB-H bursting scheme on this physical basis which provided 4.5 MBit/s for the logical DVB-H network used. Please note that the queuing strategy of the used equipment heavily influences the delay of this link: although pure DVB-T setups can achieve a delay below 15msec, this setup causes a varying delay of about 550msec.

The following results contain the mean of 10 different experiments and, whenever relevant, we present *boxplots* that represent the mean values and their deviation.

A. *Signaling processes*

This sub-section contains the time required for the different signalling processes.

1) *MSS Source session setup*

Based on the performed measurements, the average time needed for the MSS source session setup is only 47 msec. For the source registration, the average time is 43 msec. These values are very small as expected, since the source is in a fixed network.

2) *Listener interface scanning and bootstrap*

The listener interface bootstrap is divided in 3 stages: Wi-Fi AP scanning, attachment to Wi-Fi AP and Wi-Fi interface bootstrap.

The participants in this process are the listener, the Wi-Fi AR and the ZQoSB. The Wi-Fi AP scanning process is measured in the MTC, starting when it sends a MIH_*ScanRequest* to the MIHF at the listener, and ends when the MTC receives the corresponding MIH_*ScanResponse*. This process takes a mean of 1.37 sec. The attachment to the Wi-Fi AP starts when the MTC sends a MIH_*LinkSwitchRequest* to the MIHF in the listener and ends when it receives the corresponding answer. This process takes 2.04 sec. The Wi-Fi interface bootstrap requires the authentication process and routing table updating, including the binding to the Home Agent. The average measured value is 0.62 sec.

3) *Listener session setup and registration*

The listener setup time starts when the application sends a multicast subscription to the QoS Client and finishes when the application receives the first packet of multicast data. For a clearer visualization of the timings, 3 stages were distinguished: MSS session setup, listener registration and multicast data reception.

In the middle of the MSS session setup process, there is the listener registration stage, consisting of listener authorization and registration at the ZQoSB. The listener registration stage begins when the MCC at the Wi-Fi AR sends a *UserJoinReq* (through Diameter) to the Multicast Manager at the ZQoSB, and finishes when it receives the corresponding answer. As in the source setup case, the main influence for the total time is the registration time. In this case, this dependence is stronger due to the time that the ZQoSB needs to access its database checking if the listener bootstrap is successful and if the user is registered there. The MSS listener session setup takes an average time of 136 msec and the listener registration takes 96 msec. The time required for the listener to receive the first multicast data packet after the Wi-Fi AR sends the PIM Join upwards towards the source AR is 400 msec.

4) *Signalling timings for handover*

The multicast handover is divided in the 3 stages: Wi-Fi AP and DVB channel scanning, handover initiate and commit, and interface switching and multicast data reception.

a) *WiFi AP and DVB channel scanning*

The scanning procedure for Wi-Fi takes considerably more time than for DVB, respectively 1366 msec for Wi-Fi and 49 msec for DVB. The time for DVB channel scanning is small because the RAL DVB scans only channels it knows, i.e., the DVB cell IDs and respective transmission parameters are previously registered in a configuration file. The scanning time will therefore depend on the number of channels identified in that file.

b) *Handover initiate and commit*

The handover initiate phase starts when MTC sends a MIH_*HandoverInitiateReq* to the MIHF at the listener, and the handover commit begins when MTC sends the MIH_*HandoverCommitReq* message to the MIHF at the listener and ends when the response is received. After the MIH_*HandoverCommitReq* reaches the ZQoSB, an unsolicited *Join* is generated at the Multicast Manager that is sent to the DVB AR with the required data to join the multicast group. It also informs MRD6 [15] that the listener must receive the packets through the DVB interface. Fig. 6 depicts both times for the handover initiate and commit.

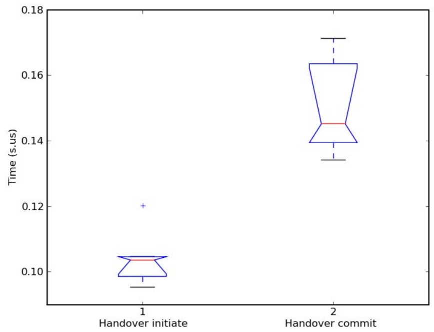

Fig. 6. Handover Duration

The handover initiate phase takes 104 msec and does not contain large variations between experiments due to its small complexity. In the handover commit phase, the ZQoSB will forward the MIH message to the DVB AR and it will need to communicate with the Multicast Manager, which in turn will generate the unsolicited *Join* (in the MIH_ *HandoverCommitReq*). All these steps result in a higher average delay of 151 msec.

c) Interface switching and multicast data reception

The switching of interfaces starts after the commit phase and takes an average time of 24 msec. After the interface switching is performed, the application can now receive the multicast packets coming from the DVB AR. For the multicast data reception time, the interval between the instant when the MCC at the DVB AR sends the PIM Join upwards to create the new branch of the multicast tree and the instant when the first multicast packet arrives at the listener through the DVB interface is 1049 msec. A large amount of this delay is caused by the queuing of the DVB-H multiplexer which could be avoided in a DVB-T setup. Nevertheless, the measured time can be too large for videoconferencing and other similar demanding services since important data is subject to be lost. The effect of handovers in the data performance will be analyzed in the next sub-section.

B. *Data performance*

This sub-section contains the effect that the handovers introduce in the data traffic. The measured data performance parameters are the disruption time, average delay and packet loss. A remote server located in another domain network is added to the testbed to send traffic towards the listener. Both server and listener are synchronized through the Precision Time Protocol daemon (PTPd) [16] allowing the time inference with significant accuracy. The values presented in the following box plots correspond to average values calculated by means of *jtg_calc* [17].

The tests are based on the following description: the server sends a test stream of packets of 1000 bytes at a bitrate of 384 kbps – this is the test stream – and will also send a variable number of other streams in parallel - the background streams (with packets of 44 bytes at a bitrate of 64 kbps). In its turn, the listener simultaneously receives the test stream and all the other streams. Then, a handover is performed and the data is analyzed during this time. By analyzing the data in the received test stream, the performance parameters will be computed through comparison with the transmitted test stream. The tests were executed in the following situations: varying the number of background streams and the number of test streams.

1) Number of background streams

In these tests the test stream is transmitted with an increasing number of background flows: 0, 5, 10, 20, 30, 40 and 50, all in the same QoS class. Fig. 7 shows that the number of background streams does not influence the value of the disruption time. However, there is a clear slight increasing trend, from 1.3 to 1.5 sec. With more traffic in the network, the increase on the network congestion status slightly increases the handover time, as expected.

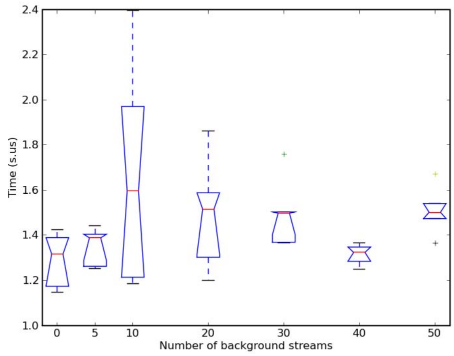

Fig. 7. Disruption Time with Different Background Streams

The delay that a packet experiences in each node results in the transfer delay of that packet across the network (Fig. 8). Packet transfer delay is influenced by the level of network congestion, the number of nodes along the path, and the disruption time due to handovers. As expected, the average delay remains almost stable until the test with 40 background streams, resulting in a total bandwidth of 2.9 Mbps including the test stream. The test with no background traffic also contains a large delay, which may be due to instability on the signal quality in the real testbed.

The packet loss results (Fig. 9) show the expected behaviour: without background streams, the average packet loss is 0.22 % and rises with 5 background streams (704 kbps); beyond this value, the packet loss remains almost constant until 40 background streams (2.9 Mbps). With 50 background streams (3.6 Mbps), the average packet loss reaches 4.99%. We consider that this is the upper limit for the occupied bandwidth without damaging the data performance.

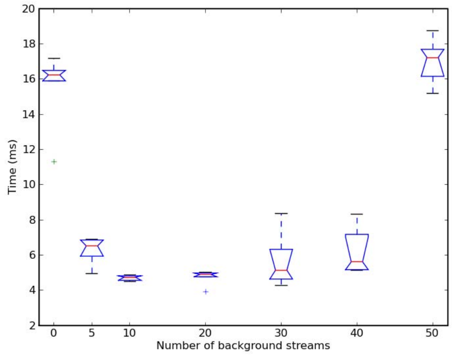

Fig. 8. Delay with Different Background Streams

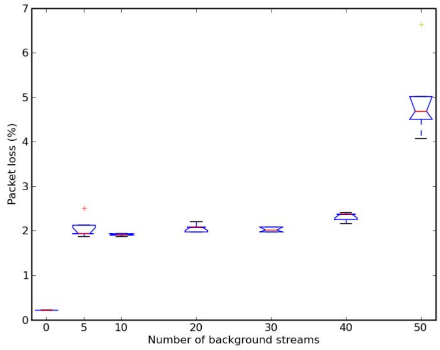

Fig. 9. Packet Loss with Different Background Streams

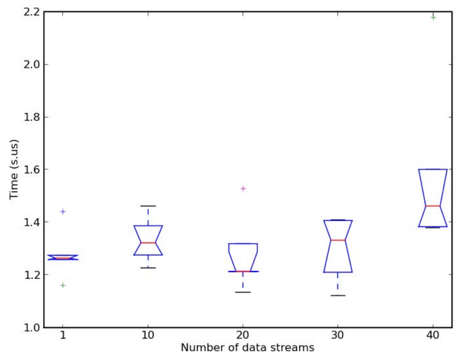

Fig. 10. Disruption Time with Different Test Streams

2) Number of test streams

This test includes a variable number of data streams that have identical characteristics of the test stream, i.e., with packets of 1000 bytes at a bitrate of 384 kbps. The goal of this test is to check the number of data streams the testbed supports with an acceptable performance.

Here we just include the graph for the disruption time (Fig. 10). Again, the number of data streams does not influence considerably the value of the average disruption time, which is similar in both tests. The other performance parameters also have similar values as the ones in the previous test.

VIII. Conclusions and Future Work

This paper showed the final results achieved in the context of the support of both QoS and mobility in multicast services in heterogeneous technologies, both at the design of an architecture able to integrate several technologies and services, and its real deployment in an experimental testbed. The proposed approach has the following characteristics: it is scalable (through a hierarchical architecture), supports legacy and enhanced terminals, decouples the underlying transport plane and the overlying session and application layers, and seamless integrates both unicast and broadcast technologies. The real experimental results show that the handover of multicast sessions between different technologies, Wi-Fi and DVB in our case, is performed without degrading the current communications, even when the network is highly loaded.

Future work includes the support of both sources and listeners mobility in a fully mobile environment.

Acknowledgment

We would like to thank all partners involved in the several parts of the DAIDALOS II project. Their contribution has been fundamental to realize this work.

References

[1] IST FP6 DAIDALOS II project www.ist-daidalos.org.
[2] IEEE 802.21 WG, IEEE Draft Standard for Local and Metropolitan Area Networks: Media Independent Handover Services, IEEE P802.21/D10.0, April 2008.
[3] C. Perkins, "IP Mobility Support for IPv4", IETF RFC 3220, January 2002.
[4] D. Johnson, C. Perkins and J. Arkko, "IP Mobility Support for IPv6", IETF RFC 3775, June 2004.
[5] J. Loughney, M. Nakhjiri, C. Perkins, R. Koodli, "Context Transfer Protocol (CXTP)", RFC 4067, July 2008.
[6] C. Tan and S. Pink, "MobiCast: A Multicast Scheme for Wireless Networks", ACM MONET, 5(4):259-271, 2000.
[7] J. Lai, W. Liao, M. Jiang, and C. Ke, "Mobile Multicast with Routing Optimization for Recipient Mobility", Proceedings IEEE ICC2001, pp. 1340-1344, June 2001.
[8] T. Harrison, C.L. Williamson, W.L. Mackrell, and R.B. Bunt, "Mobile Multicast (MoM) Protocol: Multicast Support for Mobile Hosts", Proceedings ACM MOBICOM, Budapest, Hungary, September 1997.
[9] S. Deering, W. Fenner, B. Haberman "Multicast Listener Discovery (MLD) for IPv6", IETF RFC 2710, October 1999.
[10] B. Fenner et al., "Protocol Independent Multicast - Sparse Mode (PIM-SM) Protocol Specification", RFC4601, August 2006.
[11] R. Vida and L. Costa "Multicast Listener Discovery Version 2 (MLDv2) for IPv6", RFC 3810, June 2004.
[12] P. Calhoun et al., "DIAMETER Base Protocol", IETF RFC 3588, September 2003.
[13] R. Hancock et al, "Next Steps in Signaling (NSIS): Framework", IETF RFC 4080, June 2005.
[14] E. Duros et al., "A Link-Layer Tunneling Mechanism for Unidirectional Links", IETF RFC 3077, March 2001.
[15] H. Santos, "MRD6, an IPv6 Multicast Router" – http://fivebits.net/proj/mrd6
[16] Protocol Precision Time daemon website: http://ptpd.sourceforge.net/.
[17] Jugi's Traffic Generator, website: http://hoslab.cs.helsinki.fi/savane/projects/jtg/.