Mobility and QoS Support for Multi-user Sessions over Heterogeneous Networks

Luis Veloso, Eduardo Cerqueira, Edmundo Monteiro, and Paulo Mendes

***Abstract*—Seamless handover over heterogeneous mobile environments is a major requirement to the success of the next generation of networks. However, seamless movement requires the control of the quality level and connectivity of communication sessions with no perceived service degradation to the users. This seamless characteristic is equally important for communication sessions encompassing either one or multiple receivers, being the latter called multi-user sessions. This paper presents a solution to allow seamless mobility for multi-user sessions over heterogeneous networks with mobile receivers and static senders. The proposed solution integrates end-to-end Quality of Service (QoS) mapping, QoS adaptation and connectivity control with seamless mobility support. The latter is achieved through the use of buffers in mobile nodes and caches in access-routers together with mobility prediction and context transfer schemes. Simulation experiments present the efficiency of this proposal to setup ongoing sessions and its impact in reducing packet losses and improving the quality of the received video sequence during handover.**

***Index Terms*—Heterogeneous networks, multi-user sessions, Quality of Service, seamless mobility.**

I. INTRODUCTION

Nowadays, several wireless technologies exist and are being developed to satisfy the different requirements and expectations of users. Simultaneously, the size reduction and the increase in the autonomy of electronic devices boosted the development and affordability of portable devices. These progresses together with the offer of new multi-user services such as IPTV and video-streaming, create a demand for a communication system able to simultaneously distribute content to multiple users with no perceived service degradation and over networks with different connectivity schemes and capability levels.

On the one hand, the maturity and low cost affordability of the portable devices is responsible for their wide spread utilization, and consequently, motivates the necessity to provide seamless mobility. The seamless attribute refers to the reduction of packet losses and latency of an ongoing multi-user session. In other words, seamless handovers must be assured to allow the uninterrupted reception of a multi-user session during the movement of a mobile device. This will increase the user satisfaction, and consequently, the operator revenues.

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On the other hand, multi-user sessions must be distributed independently of the underlying QoS model, link capacity, access and transport technologies, which may be different in each network along the communication path [1]. For instance, whilst the *Differentiated Services* (DiffServ) model [2] and the *Protocol Independent Multicast for the Source–Specific Multicast* (PIM-SSM) [3, 4] can be implemented inside a network to provide QoS assurance and packet distribution to multiple receivers, between networks the packet distribution can be based on unicast communications since the number of links between adjacent networks is not expected to be very high.

A solution to allow seamless mobility across heterogeneous networks with QoS and connectivity control is presented in this article. This proposal consists in the integration of two mechanisms which are being investigated within the *QoS Architecture for Multi-user Mobile Multimedia* (Q3M) architecture [5, 6] described in Figure 1: the *Seamless Mobility of Users for Media Distribution Services* (SEMUD) [7] and the *Multi-user Session Control* (MUSC) [8].

The SEMUD mechanism aims to assure seamless mobility based on the cooperation between caches in access-routers and buffers in mobile devices. A higher seamless assurance is achieved through the use of session context transfer and mobility prediction. The MUSC mechanism controls the setup of ongoing multi-user sessions in new end-to-end paths based on the multicast remote-subscription method [9]. This control is performed by providing QoS mapping, QoS adaptation and the coordination of connectivity translations on the predicted paths. MUSC carries out its control towards the receivers, which may be different from the reverse-path usually used by multicast routing protocols [10].

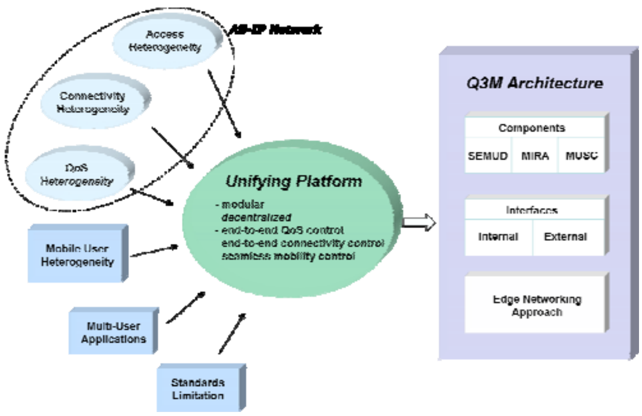

Fig. 1. Q3M architecture.

As a brief overview, the Q3M architecture controls the flow of multi-user sessions across heterogeneous networks through the use of an edge networking approach, in which the functionality of each network is controlled by a group of organized edge devices, as well as by assuring a modular integration of three components: SEMUD, MUSC and *Multi-service Resource Allocation* (MIRA) [11]. The cooperation between these components provides ubiquitous access, resource reservation, QoS mapping, QoS adaptation, connectivity and seamless mobility control for multi-user sessions in an All-IP network system.

The remainder of this paper is organized as follows. Section 2 presents the related work. A brief overview of SEMUD and MUSC is described in Section 3. Following, an illustration of their functionalities is shown in Section 4. The efficiency of the proposal to setup ongoing multi-user sessions and the benefits in reducing packet losses are analysed in Section 5. Finally, conclusions and future work are summarized in Section 6.

II. Related Work

In what concerns mobility control [12], IETF protocols such as *Mobile IPv6* (MIPv6) [13], *Hierarchical MIPv6* (HMIPv6) [14] and *Fast Handovers for IPv6* (FMIPv6) [15], were developed for controlling unicast sessions. These proposals were not aim to support multicast sessions and present several drawbacks regarding the mobility support.

For example, despite the Mobile IP capacity to support mobility in homogeneous and heterogeneous systems, this proposal presents severe latency and significative packet losses during the handover when the home agent and the mobile node are distanced. Similarly, although the HMIPv6 protocol can reduce the delay related with updating the location of mobile nodes after handover inside the same domain, it has no provision to prevent packet losses. FMIPv6 allows an anticipated reaction to movement followed by the transferring and buffering of packets from the old to the new access-router. This scheme minimizes the packet losses but involves the exchange of a large amount of signalling and the construction of tunnels. This way, despite the capacity of HMIPv6 to reduce handover delays and FMIPv6 to reduce packet losses during handovers, individually they cannot achieve seamless handovers. The use of buffers to store packets during handover is also supported by the *Low Latency Handoffs in Mobile IPv4* (LLH) [16] and *Dynamic Buffering Control Scheme for Mobile Handover* [17] proposals. However, none of these approaches are suitable for multicast sessions. The *Session Initiation Protocol* (SIP) [18] can also be used to control handover at the application layer. SIP has the advantage of keeping mobility support independent of the wireless and network layer elements [19]. However, this solution is being investigated only to control handover in unicast environments.

Regarding multicast, two mobility control methods can be emphasized: the bi-directional tunnelling based on MIP, and the remote-subscription technique [20]. With the bidirectional tunnelling, multicast must be supported only in the home network. Also, this approach allows the mobile node to subscribe to local home network multicast groups. However, this scheme leads to the increase of the traffic in the home network, and the routing of the multicast packets will not be optimal. Besides, if several mobile nodes have subscribed to the same multicast group in a foreign network, packet duplication will occur. In the Remote Subscription approach, the mobile node follows the usual behaviour of a stationary node when subscribing or leaving a multicast group. This proposal leads to an optimal routing and avoids the packet duplication existent in the Mobile IP based approaches. However, every time the mobile node moves to a new network (and no subscriber of the same group exists in this network), the multicast tree has to be readjusted causing long interruptions, and consequently, packet losses. Moreover, both proposals lack in terms of QoS support and are dependent of specific connectivity technologies, such as multicast from source to home agent in the former, and the same multicast address realm end-to-end in the latter.

A scheme that allows handover with faster re-establishment of unicast or multicast sessions is the IETF *Context Transfer Protocol* (CXTP) [21]. CXTP can achieve seamless mobility by transferring the session context to new access-routers before the handover [22]. However, CXTP alone is not sufficient to provide seamless mobility of multi-user sessions, since it does not setup ongoing sessions.

In addition to the mobility control of multi-user sessions, the heterogeneity of networks poses requirements for the mapping and adaptation of QoS, as well as for the session translation among different connectivity schemes. A research challenge is the deployment of a dynamic QoS mapping mechanism to map the session requirements into a suitable service class inside or between networks, independently of the underlying QoS models used to assure QoS guarantees for sessions on the communication path. Static QoS mapping schemes or even guidelines for IP QoS mapping alone are not sufficient to assure the quality level of sessions, because they do not shield end-user and network internals from the details of the underlying QoS infrastructures and do not take the current conditions of the network into consideration. In addition, due to the existence of links with different capacities and oscillation of network capabilities, sessions must be adapted to the current network conditions taking into consideration the priority of each flow. An adaptation scheme must be independent from CODECs and must avoid the interaction of users during the adaptation process. The combination of the above approaches avoids session blocking and reduces the impact on the quality perceived by mobile users in congestion periods.

Regarding mapping solutions, IETF proposed a static QoS mapping scheme which provides a guideline for IP QoS mapping in DiffServ networks [23]. However, this mapping approach assumes that all networks are configured with the same QoS model and service classes. Other approaches are used to provide QoS mapping for sessions, but require proprietary modules in the end-systems and “expert” users to decide the most suitable class [24], reducing the system flexibility. Another solution [25] proposes a centralized agent to classify the session requirements into classes of service between networks with different QoS models. However, it is focused on the QoS metrics used by its agents and does not present the cooperation between agents to control QoS mapping along the end-to-end session path. Moreover, Mammeri [26] introduces a mapping solution that provides QoS guarantees for unicast sessions across *Integrated Service* (IntServ) and DiffServ models. However, this approach is dependent of the underlying QoS model. Furthermore, several guideline-based QoS mapping solutions only control the mapping of sessions from the DiffServ or IntServ models to the IEEE 802.16 QoS model [27].

QoS adaptation solutions are used to adjust the overall quality of a session to the capability of different networks. Existing CODEC-aware adaptation proposals control the quality level of sessions in DiffServ networks by discarding packets according to the importance of each frame of a *Moving Pictures Experts Group* (MPEG) video [28]. In congestion periods, less important frames are dropped before important ones. However, this adaptation mechanism is not suitable for future generation networks, because its applicability depends on a specific multimedia CODEC and on a specific QoS model. In addition, other mechanisms control the session rate to the current networks conditions based on receiver or transcoder approaches. Receiver-based approaches reduce the system flexibility by implementing modules in end-systems to join or leave flows of multicast sessions based on notification about the network conditions [29], and transcoder-based solutions require elements in the network to adapt the content coding (re-coding) to the available bandwidth [30], making the network deployment dependent from multimedia CODECs.

The end-to-end connectivity control of multi-user sessions over heterogeneous networks can be accomplished by using tunnel-based [31] or translation-based [32] approaches. While the former requires the same IP multicast address realm in both access-networks, the latter allows the session connectivity mapping between unicast and multicast networks. Nonetheless, existing translation solutions provide only unidirectional conversion between unicast and multicast realms [33, 34]. In general, the translation-based schemes do not control the session connectivity between multicast networks with different address realms, such as networks implementing *Any-Source* and *Source-Specific Multicast*.

Most of the QoS mapping approaches provide a static mapping scheme or operate only in networks with specific QoS model. Additionally, several proposals reduce the system flexibility by requiring the installation of proprietary modules in end-system as well as by using centralized approaches to control the session mapping. Regarding QoS adaptation solutions, receiver-based approaches need the utilization of modules in end-systems to join or leave multicast sessions. The system flexibility is also negatively affected due to the implementation of CODEC-aware agents in the network to control the session quality level.

The analysis of related work has shown that none of the connectivity solutions assures the end-to-end coordination of a chain of connectivity translators to support any layout of heterogeneous networks and end-hosts. In order to overcome the above limitations and to guarantee the distribution of multi-user sessions with seamless mobility, connectivity and QoS support, this paper proposes the integration of SEMUD and MUSC mechanisms.

III. SEAMLESS MOBILITY OVER HETEROGENEOUS ENVIRONMENTS

The cooperation between SEMUD and MUSC mechanisms provides seamless mobility control with end-to-end QoS mapping and adaptation, as well as connectivity support. Due to the heterogeneity of the networks, this proposal is based on the separation of the multi-user session identifier and the network locator as proposed in *Next Step in Signalling* (NSIS) framework [35]. While the session identifier has a global meaning, the network locator is only relevant for the local network. Hence, each multi-user session is described in a *Session Object* (SOBJ) which is identified by a global session identifier. A multi user session can also be scalable and composed by a set of flows whose QoS parameters are described in the QSPEC object [36].

This proposal assumes that receivers obtain from the source, by any off-line/on-line means, information regarding the available sessions encompassing the SOBJ and QSPECs. Each QSPEC includes the flow priority, bit-rate, tolerance to loss, delay and jitter.

The location of the MUSC and SEMUD agents is illustrated in Figure 2, which shows that MUSC agents are implemented at network edges while SEMUD agents are implemented at access-routers and mobile devices. Agents are called access-agents when they encompass the SEMUD and MUSC functionalities. Moreover, agents can have distinct roles in different edges for different sessions: in an edge router, an agent is called an ingress-agent for sessions whose traffic is entering the network in that edge router, or egress-agent if the traffic is leaving the network.

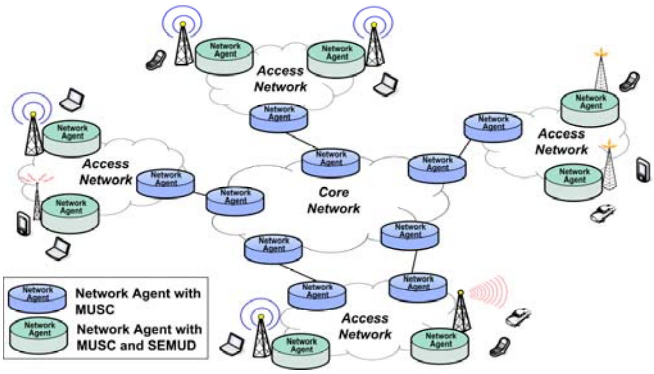

Fig. 2. MUSC and SEMUD location in a generic scenario.

As an overview, the interoperation between MUSC and SEMUD mechanisms can be described as follows. After attaching to an access-agent, receivers send the session description to the MUSC agent located in the access-agent by using SIP and the *Session Description Protocol* (SDP) [37]. After the session configuration (QoS mapping, QoS adaptation and connectivity translation) in all edges of the end-to-end path, MUSC notifies SEMUD in the access-agent regarding the context of the new session. Before handover, the most probable cells to where the mobile device will move are predicted and the session context is sent to the respective access routers, allowing the session to be installed by MUSC and SEMUD in the predicted end-to-end paths. After handover the session will be already present in the new access-agent, which will permit the recovery of missing packets.

A. *SEMUD Overview*

SEMUD aims to provide seamless handover between access-agents belonging to neighbouring networks through the combination of context transfer, mobility prediction and cache and buffer mechanisms. Packet losses and delay are reduced by controlling caches in access-agents and buffers in mobile devices.

The data packets received in the access-agent belonging to the subscribed session are stored in the cache (where the oldest packets are removed) and forwarded to the interested receivers. This storage will permit a subsequent recovery of packets which are eventually lost. Notice that for each multi-user session an independent cache will be created. When the packets forwarded by a cache are received by a mobile device they are stored into the buffer and consumed by the application. In the presence of a handover, the data in the buffer of the mobile device will continue to be read in order to keep the data flow.

The presence of the session in the next access-agent is guarantied through the employing of mobility prediction schemes. This is, the most probable cells to where the mobile device will move are predicted based on parameters such as the moving direction, velocity, current position and historical records. The addresses of the predicted access-agents could be obtained in a static way through agreements, where a machine in the current network exchanges this information with another one located in other network. Alternatively, the addresses discovery could be done in a dynamic way by querying a DNS-alike service. Based on this forecast, the session context is transferred in advance into the predicted cells by the SEMUD-P signalling protocol, as described in Figure 3. This way, the interaction between the SEMUD and the MUSC in the predicted access-agents will permit to reserve the resources and configure translators in advance.

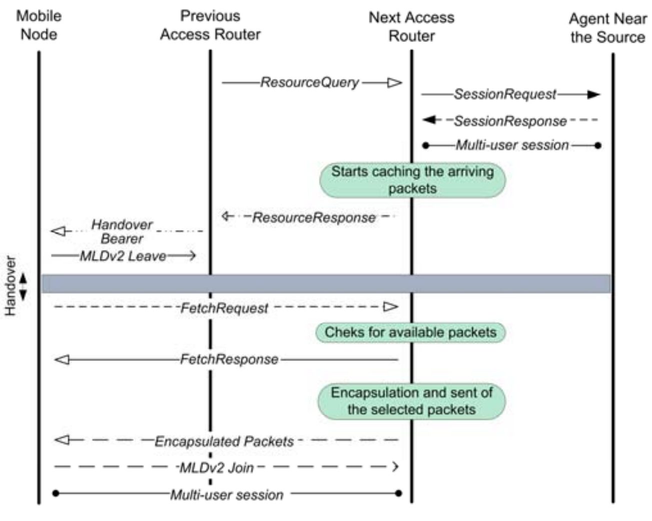

Fig. 3. Message sequence in the handover procedure.

Additionally, information concerning the capabilities provided by the predicted access-agents is collected and conveyed by the SEMUD-P to the current access-agent. At the current access-agent, the probed information regarding the available resources in the predicted access-agents, the signal-to-noise ratio and the knowledge regarding the access technologies, gives support to the handover decision. For example, the selection of the next access router should give preference to access routers that give guaranties of QoS in detriment to those that cannot satisfy the QoS requisites or cannot built a branch to the multicast tree. When the handover decision is taken, a message containing the identification of the next access router and of the multi-user sessions is sent from the current access router towards the mobile node. Simultaneously, the communication between SEMUD and MUSC allows for the release of resources reserved on the old path and on the new paths of the predicted cells (that the mobile device is not going to use).

When the handover procedure is complete, the mobile device updates its buffer by fetching the missing packets from the cache of the new access-agent. The message sent to fetch the packets will carry information regarding the available space in the buffer, the time stamp of the last packet received in the buffer before handover and the intended multicast session represented in the PIM-SSM by <*Source, Group*> or <S, G>. In the present work it is considered that the recovered packets are sent from the cache to the buffer via unicast. The advantage of this approach is to avoid the reception of these packets by other local subscribers of the same multi-user session. This will require encapsulation of the multicast packets into unicast packets, which the mobile node will de-encapsulate before putting them (the multicast ones) in the buffer.

When the amount of packets to recover from the cache is larger than the available space in the buffer it is necessary to discard packets. In this situation, the method used in the present proposal is to recover packets sequentially from the cache and discard the ones that found no place in the buffer. However, since packets have different influence in the quality of a session a better approach (used in [38]) will be to recover the cached packets taking in account their role in the referred quality.

B. *MUSC Overview*

MUSC aims to support the mobility of multi-user sessions over heterogeneous wireless environments. A receiver-driven and source-initiated protocol, called MUSC-P, is used to coordinate QoS mapping, QoS adaptation and Connectivity Control mechanisms with other edge-agents along the end-to-end session path based on a soft-state approach. MUSC-P is being specified in the context of the NSIS framework, in which it can be included as an extra *NSIS Signalling Layer Protocol* (NSLP). MUSC-P operates in a receiver-driven approach, since it is triggered at access-agent (agent located in wired or wireless access-router). It is source-initiated since MUSC starts the QoS and connectivity configuration of its agents at the agent nearest to the source, or at the first agent in the path towards the source that contains the requested multi-user session. This functionality aims to build QoS-aware distribution trees, in environments with asymmetric routing, taking into account the QoS characteristics of the path from source to receivers, which may be different from the reverse-path used by multicast protocols.

In ingress and egress-agents, the QoS mapping mechanism accomplishes a dynamic mapping of the session requirements into the available service classes controlled by a network resource allocation controller (e.g., *Service Level Agreement* (SLA) controller between networks) or a QoS bandwidth broker. An interface with resource allocation controllers allows MUSC to collect, from a resource controller, information regarding wired or wireless network classes (e.g., available bandwidth, maximum loss and delay along the session path described in a quantitative or qualitative manner) and to select a suitable service class for a session among networks with different service classes, network resources and/or QoS models. The use of this interface allows operators to use resource allocation controllers of their choice.

Additionally, by exporting the mapping negotiation for the application level, MUSC does not require changes on mobile devices, shields end-user and network internals from the details of the underlying QoS infrastructures and assures an acceptable quality level to sessions, even during handover. In essence, the mapping algorithm compares, one by one, the QoS parameters desired for each flow of the session (collected from the per-flow QSPEC) and the list of available service classes offered inside or between networks (collected from the resource allocation controller). After that, the mapping algorithm chooses the network class that is more appropriated for the requested session based on the perfect match, sub-perfect match and hybrid match methods as follows:

- *Perfect Match Method*: This method aims to support the full QoS requirements and bandwidth committed for all flows of a multi-user session, by selecting the preferred class to be used for each flow. Therefore, it assures that each flow of a session is mapped to a class of service that supports the same QoS requirements as desired in the QSPEC. When the preferred service class does not have enough available bandwidth to assure the minimum packet loss rate for the session, the QoS adaptation mechanism is triggered, which then may decide to try a sub-perfect or a hybrid mapping;
- *Sub-perfect Match Method*: This method maps all flows of a multi-user session to a service class that supports QoS parameters different from the ones described in the QSPEC. The mapping of all flows into another network service aims to avoid the session blocking and the re-ordering of packets. It can be used in a congestion period of a preferred class, while assuring the session full rate;
- *Hybrid Match Method*: This method assures the allocation of, at least, the high priority flows of a session into the preferred class. The remainder flows are mapped to a less significant service class. It can be used when the packet re-ordering is not crucial. It can be suitable for scheduled video and audio, where it is more important to ensure an intelligible audio flow than a perfect video.

When the mapping process is not optimal, for instance due to the selection of an overloaded service class, the adaptation mechanism is triggered. The session is blocked only when a misplaced service class cannot guarantee the minimal QoS requirements desired for each flow of the session. Adaptation operates based on the QSPEC and the current network conditions. Following, a description of the three adaptation methods is presented. When the maximum bandwidth of the preferred class cannot assure the QoS committed for a low priority flow, this flow is removed from the outgoing interface and classified into the sleeping state by MUSC. Sleeping flows are awaked when the network capability becomes available again and the session full rate is supported. On the other hand, the *Re-Mapping Adaptation* method requests the mapping of the session to another class (using the sub-perfect or hybrid mapping). The *Service Class Re-adjustment* method can be used to try the accommodation of the session into the preferred class, by requesting the re-adjustment of the maximum (extra) bandwidth assigned for the service classes. The adaptation process allows the mobile user to keep acceptable quality level of ongoing sessions, independently of its movement. A deeper analysis of both QoS mapping and adaptation mechanisms can be found in [39].

In addition to the QoS control mechanisms, MUSC supports a translation-based *Connectivity Control* mechanism to provide session connectivity between networks that implement the same and/or different address realms. For instance, it assures the continuity of multi-user sessions from unicast to unicast, multicast to unicast, unicast to multicast and multicast to multicast networks. Therefore, it allows senders to offer their content and receivers to access them independently of the devices or networks connectivity technologies (e.g., IP unicast or IP multicast). This mechanism also controls the shaping of distribution trees aiming to build distribution trees with the branch point as close as possible to the receivers. Hence, it avoids the reconstruction of the entire session path due to the mobility of the receivers, senders or networks as well as network failure. Moreover, it controls the packet distribution in a multicast manner to avoid the packet duplication and to save network resources.

C. *MUSC and SEMUD Interfaces*

As depicted in Figure 4, SEMUD, MUSC and MIRA have interfaces to exchange information between themselves and existing solutions and standards. Following, the most significative of these interfaces are described in this section. The MIRA interfaces are presented in [11].

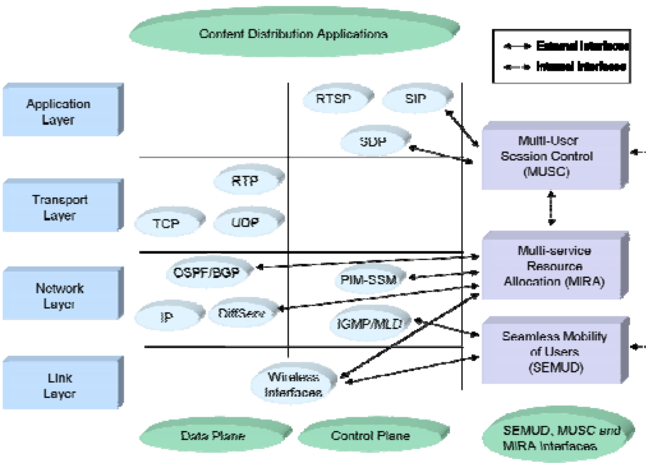

Fig. 4. External and internal Interfaces.

The address allocation controller interface is used by MUSC to control the connectivity of multi-user sessions. In all downstream agents, MUSC interacts with an address allocation controller (MIRA in Figure 4) to request the allocation of channel identifiers for flows of a session. If PIM-SSM is supported, it is triggered by MUSC in edge-agents to create multicast branches associated with each flow. This interface is also used to inform an address allocation controller and PIM-SSM about flows that were removed.

The resource allocation controller interface is used by MUSC to query information about network classes and their available bandwidth towards the access-agent in which the receiver is attached or is moving to. After the mapping process, MUSC informs which network class was selected and which bandwidth is required for each flow of multi-user sessions. In congestion situations, the QoS adaptation mechanism uses this interface to adapt flows of sessions to the current network conditions. The resource controller is also notified about flows that were released by MUSC.

The access controller interface allows SEMUD to keep the session context and to create a cache for the session (if the session still does not exist) when a session is accepted by MUSC. The opposite operation is triggered by MUSC when a session ends. Before handover, MUSC in predicted access-agent(s) will setup the session and collect information concerning the capability and connectivity provided by the latter. At the previous access-agent, the interaction between SEMUD and MUSC allows the delete of resources associated with the session on the old path.

The SEMUD mobility prediction interface for the interaction with a movement prediction module. The last one is used to predict the next most probable access-agents based on the location of the base stations and on the properties of the mobile device such as location, moving direction and velocity. For instance, this interface allows the interaction with a mobility prediction scheme as the one proposed in [40].

The multicast activate interface allows multicast-aware receivers connected to multicast-aware access-networks to leave/join multicast channels associated with each flow of a session notified by MUSC. This is done through the interaction of SEMUD and IGMPv3/MLDv2 [41, 42], where a leave message is triggered during the disconnection from the old access-agent and a join message is sent after attaching to the new access-agent.

IV. Illustration of the Overall Functionality

Figure 5 presents an example of MUSC and SEMUD operations in an inter-network handover scenario. It is assumed the existence of an anticipated handover scheme which obtains the location of the candidate access-agents and notifies SEMUD regarding this information. Moreover, in each candidate agent MUSC triggers a resource and an address allocation controller during the QoS (mapping and adaptation) and connectivity control operations, respectively.

Based on the interaction with the mobility prediction mechanism, SEMUD verifies that R1 (step i) is moving away from the access-agent-A and that access-agent-K is the candidate access-agent (step 1 in Figure 5). Upon receiving the IP address of the predicted access-agent, SEMUD-P sends a *ResourceQuery* message to the SEMUD agent in the access-agent-K (step 2 in Figure 5). This procedure allows the session setup on the new path (by triggering MUSC and notifying it about the SOBJ) and the creation of a cache for the session in the new access-agent.

MUSC agent verifies that session S1 is neither locally present nor in the access-network N3. Consequently, a *SessRequest* message is sent towards the source of the session (step 3 in Figure 5). This message is stopped in agent-F, since it has another branch with the same requested flows of S1. In this agent, MUSC interacts with the resource allocation controller to query information about inter-network service classes. Based on the response and QoS parameters described in the QSPEC, MUSC selects the network class and requests to the address allocation controller, the allocation of a pair of IP unicast addresses and transport ports to identify each flow of S1 between N2-N3 (in this case, the source is the agent-F and agent-J is the destination) (step 4 in Figure 5). Afterwards, the resource allocation controller is triggered to configure the required bandwidth for each flow in the selected class.

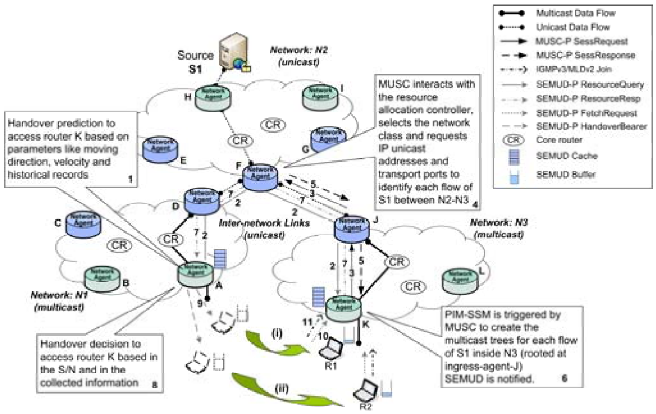

Fig. 5. Example of an inter-network handover.

After resource and address allocation operations, MUSC in agent-F starts the address translation and the packet replication of each flow of S1 in the inter-network link (different flows have the same source and destination, agents-H and F, but are identified by different ports). In the control plane, the agent-F sends a MUSC-P *SessResponse* message to agent-J (the IP address was obtained by the resource controller) (step 5 in Figure 5). The reception of this message allows MUSC in the agent-J to update its state with the channel identifier allocated to the flows of S1. The MUSC interaction with the resource and address allocation controllers in agent-J occurs as described before.

Since the agent-K could be the new access-agent of R1, PIM-SSM is triggered by MUSC to create the multicast trees for each flow of S1 inside N3 (rooted at ingress-agent-J). The resource controller is activated by MUSC to provide service differentiation for each flow on the wireless link. Finally, SEMUD is triggered and receives information about the request, including the new SSM channels used for each flow in N3 (step 6 in Figure 5). After activating a cache for S1, a SEMUD-P *ResourceResp* message is sent to the access-agent-A (step 7 in Figure 5).

Upon the reception of the *ResourceResp* message, SEMUD analyses the information concerning the available resources (in the predicted access-agents) and the signal-to-noise ratio, and decides to handover to the predicated access-agent-K (step 8 in Figure 5). In access-agent-A, SEMUD sends a *HandoverBearer* message to notify R1 about the IP address of the future access-agent and the SSM channel allocated to each flow (step 9 in Figure 5). After the handover, SEMUD informs MUSC to adjust the number of receivers associated with S1 in the previous access-agent. However, the state on the old path is not released by MUSC, because R2 is still receiving data from S1.

During handover, packets are stored in the cache of the new access-agent and after the attachment of R1 to this agent, SEMUD-P sends a *FetchRequest* message to recover the missing packets and to sync the packet reception with the cache in agent-K (step 10 in Figure 5). The recovered packets are sent from the cache to the buffer via unicast connections. This requires encapsulation of the multicast packets into unicast packets, which SEMUD in R1 de-encapsulates before putting them in the session buffer. This functionality avoids packet replication to other receivers subscribing the same multicast group and attached to agent-K. After receiving all the fetched packets, SEMUD triggers the IGMPv3/MLDv2 to join the multicast channel allocated for each flow of the session (step 11 in Figure 5).

After the handover of R1, SEMUD in agent-A identifies the next access-agent to where R2 will move (step ii) and triggers MUSC to pre-set the session on the new path. Upon receiving the SOBJ transferred by the SEMUD-P *ResourceQuery* message, MUSC verifies that S1 is already activated and increments the number of receivers receiving the requested session. After the MUSC reply, SEMUD associates R2 with the existing cache and sends a *ResourceResp* message to the previous agent to complete the handover. The seamless handover process is accomplished as explained for R1. However, MUSC in agent-A releases the state associated with S1, triggers SEMUD to remove the cache, the resource controller to erase network resources and PIM-SSM to delete the multicast trees of each flow in N1.

In agent-D and agent-F, MUSC removes the S1 state because no MUSC-P *SessRefresh* message arrives to these agents before the expiration of the MUSC clean-up interval. This requires the interaction with resource and address allocation controllers to release their state associated with the removed flows.

V. Performance Evaluation

Several simulation experiments were accomplished using the *Network Simulator-2* (NS2) [43-45] to verify the performance of MUSC and SEMUD to control the setup of ongoing sessions with seamless experience to the receivers. The convergence time of the mechanisms, the amount of packet losses and the quality of the received video sequence measured in *Peak Signal to Noise Ratio* (PSNR) are analysed. Simulation results regarding the efficiency of QoS adaptation operations in QoS-aware mobile networks (which are not the focus of this paper) are presented in [46]. There, the benefits of combining QoS mapping, QoS adaptation and mobility are demonstrated through the analysis of the blocking probability, the one-way delay and the throughput of sessions when the QoS mapping adaptation mechanism is enabled and disabled.

Three topologies (A, B and C) were randomly generated by *Boston University Representative Internet Topology Generator* (BRITE) following the same inter-network scenario as illustrated in Figure 5. Each of the three networks in either topology has twenty routers (four edges and sixteen cores). The intra and inter-network links have a bandwidth of 100 Mb/s and the wireless link capacity is 11 Mb/s. The propagation delay inside and between networks is attributed according to the distance between the edges. A PIM-SSM agent for NS2 [47] is implemented to distribute the session packets.

Two mobile nodes are placed in the same access-agent and receive one *Variable Bit Rate* (VBR) flow with an average rate of 86 KB/s. This flow consists in a video sequence denominated News [48] with format YUV, sampling 4:2:0, size 352x288 (CIF) and containing 300 frames sent with a 30 frames/s rate. Each frame was fragmented in blocks with 1024 bytes length which were transported in *Real-Time Transport Protocol* (RTP) packets. Figure 6 depicts some frames extracted from the "News" video sequence.

Fig. 6. Some frames extracted from the "News" video sequence.

Since the mobility prediction is under investigation, it is assumed that SEMUD is notified in advance about the movement of receivers to a predicted access-agent. This notification occurs in a period of time sufficient to allow the session setup on the new path before the disconnection from the old access-agent.

A. Convergence Time

Table 1 describes the SEMUD and MUSC convergence time to setup the session before and after the attachment of each receiver to the new access-agent. The convergence period is the time used by each mechanism to coordinate its operations in order to install the ongoing session with QoS, connectivity and seamless mobility support. Before handover, the SEMUD convergence time includes signalling and the cache configuration procedures. The MUSC convergence time encompasses signalling and the configuration of the session mapping and connectivity for the first receiver. All posterior requests for the same session in the same access-agent are processed locally by MUSC as happens with R2. After the handover, the SEMUD convergence time includes the fetching of packets stored in the cache.

TABLE 1
MUSC AND SEMUD CONVERGENCE TIME (MS) BEFORE AND AFTER THE ATTACHMENT OF THE RECEIVERS TO THE NEW ACCESS-AGENT

Topology	Receiver	Before		After	Total
		SEMUD	MUSC	SEMUD	(ms)
A	R1	26.15	16.23	1.07	43.45
	R2	26.15	-	1.07	27.22
B	R1	28.08	18.23	1.12	47.43
	R2	28.08	-	1.12	29.20
C	R1	27.54	16.82	1.03	43.39
	R2	27.54	-	1.03	28.57

The overall convergence time to setup the session for R2 is reduced in 36 % because the requested session is already available in the new access router. Only MUSC local procedures are done (negligible time) to configure the number of receivers associated with the session and to reply to SEMUD. This functionality minimizes 48 % in the overall signalling overhead, because only SEMUD-P messages are used. The latency to setup the sessions is increased in approximately 1 % by MUSC and SEMUD operations. Note that the MUSC convergence time would be higher if the requested session for R1 would be activated in an agent near the source.

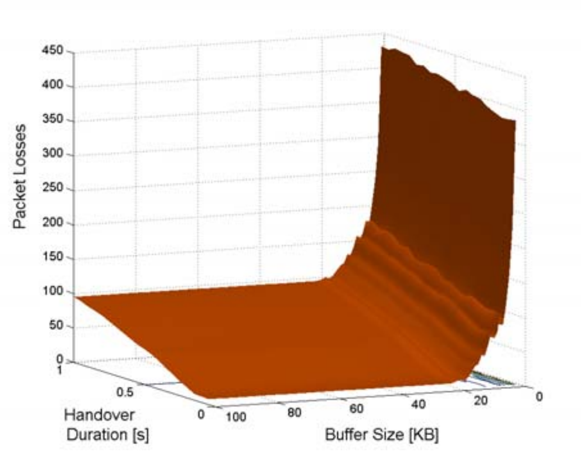

Fig. 7. Packet Losses versus the Buffer Size and the Handover Duration when the SEMUD mechanism is disabled.

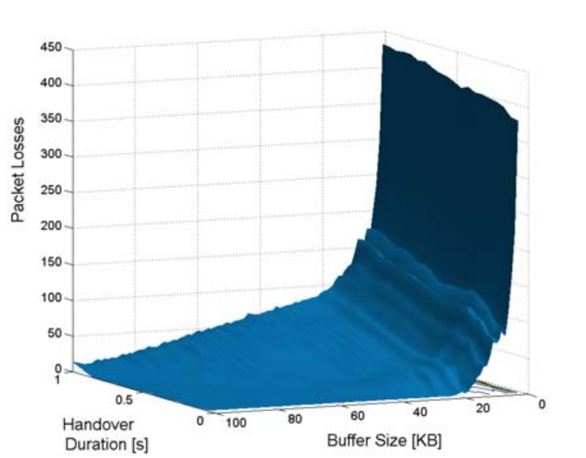

Fig. 8. Packet Losses versus the Buffer Size and the Handover Duration when the SEMUD mechanism is enabled.

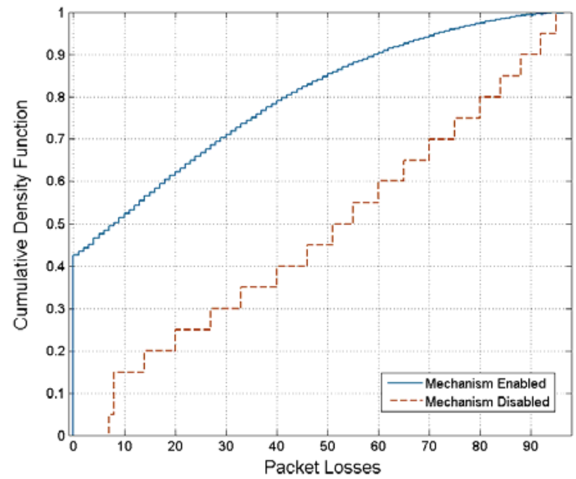

Fig. 9. Cumulative Density Function of the Packet Losses.

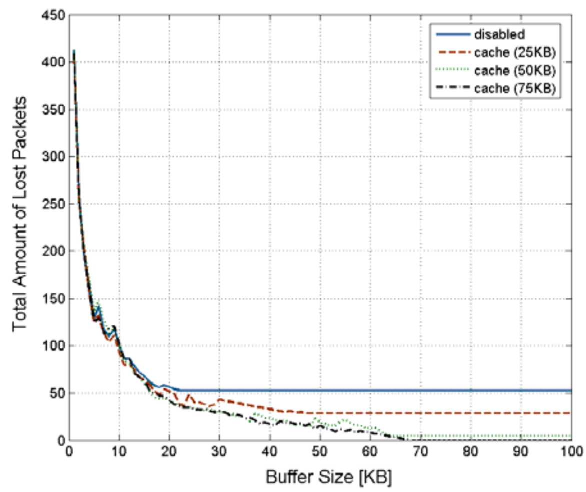

Fig. 10. Total amount of lost packets versus the buffer size with different cache sizes.

B. Packet Losses

The total amount of packet losses for each simulation was obtained by varying the Buffer size and the handover duration and considering a constant Cache size. It was considered a size of 100 KB for the Cache, a variation between 1 and 100 KB for the Buffer size and a variation between 50 ms and 1 s (with a 50 ms step) to the handover duration. To evaluate the improvements introduced by the proposed mechanism, the simulations were effectuated with and without its activation. MUSC mapping and connectivity operations were performed to allow the continuity of the session over heterogeneous networks and to map the session into the available service class based on the *Perfect Match* approach.

As seen Figure 7, when the SEMUD mechanism is disabled the high amount of packets lost during handover is not recovered.

Notice the exponential packet losses occuring when the Buffer size is inferior to a threshold (21.04 KB). Below this value the Buffer will rapidly get full since the packets pertaining to each frame arrive in bursts. Consequently, the packets that can not be placed into the Buffer will be discarded. The threshold value from which the exponential losses will occur depends on the source debit. In other words, the obtained value of 21.04 KB refers to the specific debit used in our simulation.

To Buffer size values above this threshold, the total amount of packet losses remains constant for a specific handover duration. This occurs because the threshold value represents the Buffer size from which the rate fluctuations are absorbed. In this situation, the total number of lost packets corresponds solely to the ones that are lost during handover.

The total amount of packet losses (per simulation) when the SEMUD mechanism is enabled is depicted in Figure 8.

Following the same reasoning presented previously, the total number of lost packets grows exponentially when the Buffer size becomes very small. Nevertheless, as the size increases the improvement obtained in reducing the quantity of lost packets is significative.

The *Cumulative Density Function* (CDF) for the total amount of packet losses with and without the activation of the SEMUD mechanism is shown in Figure 9.

The amount of packet losses was also evaluated versus a buffer size varying between 1 and 100 KB and for a handover duration with 500 ms. The results depicted in Figure 10 were obtained for several values of the cache size when the mechanism is enabled, and compared with the situation when it is disabled.

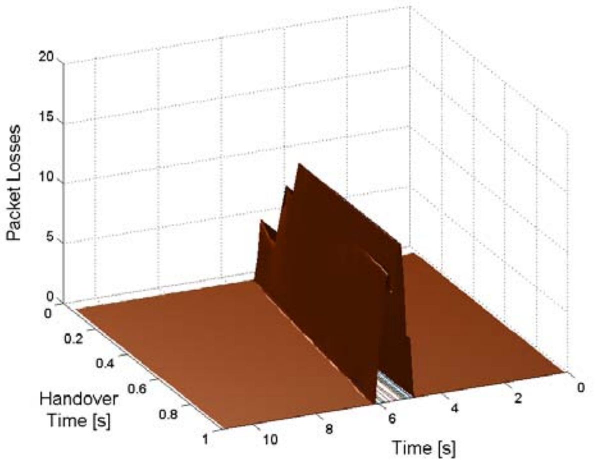

Fig. 11. Temporal variation of Packet Losses versus the Handover Time when the SEMUD mechanism is disabled.

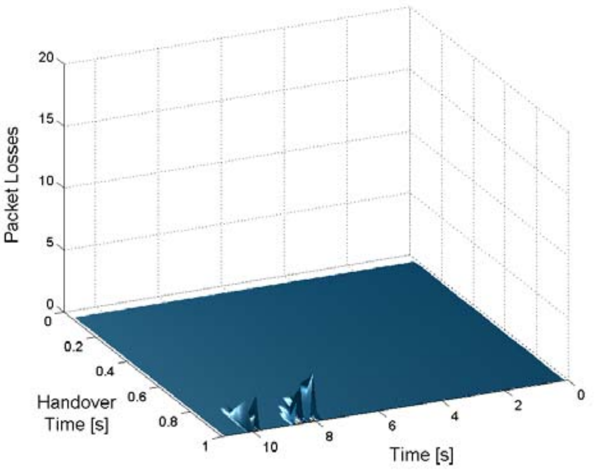

Fig. 12. Temporal variation of Packet Losses versus the Handover Time when the SEMUD mechanism is enabled.

The amount of packets that is possible to recover augments with the cache size, and consequently, the total number of lost packets is reduced. Similarly, as the buffer size increases it is possible to accommodate a larger number of packets coming from the cache in the recovering process. The improvement obtained in reducing packet losses when the SEMUD mechanism is enabled is noticeable. Namely, the packet losses are totally reduced for a combination of a cache size with 54 KB and a buffer size with 68 KB.

Additionally, the temporal variation of the packet losses versus the handover duration when considering a fixed Buffer size (100 KB) and a fixed Cache size (100 KB) was obtained. Note that the temporal variation refers to the duration of the simulation and the sampling period used to obtain the results was 100 ms. The results obtained when the mechanism is disabled are presented in Figure 11.

The results in Figure 12 were obtained after the activation of the SEMUD mechanism. In this situation the improvements in reducing the number of lost packets are evident. The only losses observed are for handover values closer to the maximum used in the simulation. In this case, the number of packets missing during handover is high and its recovery leads to the full occupation of the Buffer.

Afterwards, the occupation of the Buffer will suffer a fluctuation around this value. Obviously, when the fluctuation is above the Buffer size the arriving packets will be lost.

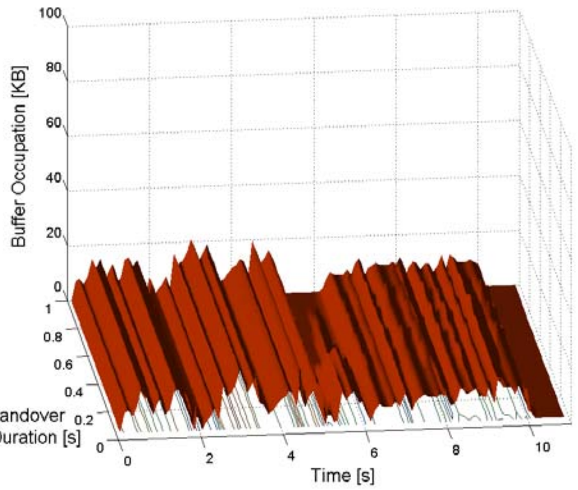

Fig. 13. Buffer Occupation versus the Handover Time when the SEMUD mechanism is disabled.

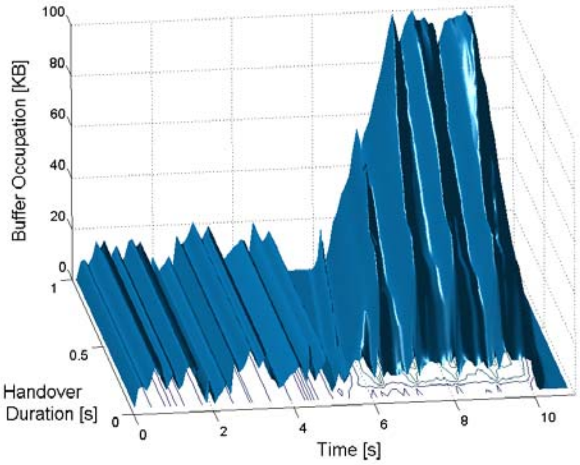

Fig. 14. Buffer Occupation versus the Handover Time when the SEMUD mechanism is enabled.

C. *Buffer Occupation*

Following, the temporal variation of the Buffer occupation versus the handover duration is analyzed. The handover time includes MUSC and SEMUD signalling and operations to configure the connectivity, QoS, Cache and Buffer mechanisms. The variation when the SEMUD mechanism is disabled is depicted in Figure 13, in which it is considered a Cache size of 100 KB.

The oscillations present in Figure 13 result from the fragmentation and transport of each frame in several IP packets. In other words, to send a frame it is necessary to send several IP packets. The number of IP packets resulting from a frame depends on the maximum IP packet size (1052 bytes in this scenario) and on the frame size (e.g., different sizes for different frame types (I, P or B)). This way, the removal or insertion of a frame into the Buffer corresponds to the remotion or insertion of several packets that convey it.

This fact justifies the oscillations in Figure 13 occurring in the Buffer occupation during the simulation time when the mechanism is disabled. During the handover period (initiated at the 4.984 s instant) it is observed a decrease in the Buffer occupation. After the handover completion, the Buffer reinitiates the reception of packets with the same oscillatory behaviour as before.

When the SEMUD mechanism is enabled, the variation of the Buffer occupation will be significantly different as observed in Figure 14. In this situation, after the handover completion the mobile node requests from the Cache the packets that were not received during handover. The signalling used to accomplish this request contains the identification of the last packet received in the Buffer and the available space in it. With this information the Cache can select and send the available packets to the Buffer. This process of recovery leads to an increase in the Buffer occupation after the handover completion.

An increase in the handover duration leads to a higher number of packets which are not received during this period, and consequently, to a higher number of packets which are necessary to recover. In this manner, the slope in the Buffer occupation occurring after the handover increases with the number of packets which are necessary to recover, and consequently, with the handover duration.

D. PSNR (Peak Signal to Noise Ratio)

The advantage of the proposal in reducing the impact of handovers in the user perceived quality was also analyzed. To accomplish that, it is imperative to compare the received frames with the original ones, and consequently, a metric for the evaluation of the reconstructed frames was necessary. With this objective in mind, several metrics *Signal to Noise* (SNR) have been developed to estimate the quality of reconstructed frames comparatively to the original ones. The most widespread metric to evaluate the quality of the video transmission is the PSNR [49].

This metric is easily defined through the *Mean Square Error* (MSE). Considering the luminance (Y) of the reconstructed and original frames and assuming frames with MxN pixels, the MSE is defined through:

$$MSE = \frac{1}{MxN}\sum_{i=0}^{M-1}\sum_{j=0}^{N-1}\left\|Ys(i,j)-Yd(i,j)\right\|^2$$

In this equation, while Ys(i,j) designates the pixel in the position (i, j) of the original frames, the Yd(i,j) represents the pixel located in the position (i, j) of the reconstructed frame. Since a video signal possesses a wide dynamic variation the PSNR is express in a logarithmic scale. Based in the MSE definition it is relatively easy to express PSNR through the expression:

$$PSNR = 10\log_{10}\left(\frac{MAX^2}{MSE}\right) = 20\log_{10}\left(\frac{MAX}{\sqrt{MSE}}\right)$$

Here $MAX = 2^K - 1$ and k represents the number of bits per sample. Considering 8bits/sample the final expression becomes:

$$PSNR = 20\log_{10}\left(\frac{255}{\sqrt{\frac{1}{MxN}\sum_{i=0}^{M-1}\sum_{j=0}^{N-1}\left\|Ys(i,j)-Yd(i,j)\right\|^2}}\right)$$

Through this metric each received frame will be compared with the original one, and the obtained value will give a measure for the degradation of the original frame. The quality of the received sequence will be obtained by using this technique with all the frames that constitute the sequence.

Nevertheless, this is an objective quality evaluation metric and it is necessary to estimate the opinion of the users relatively to the received sequence. The metric of the human impression regarding the quality uses a scale between 1 (bad) and 5 (excellent) and is designated by *Mean Opinion Score* (MOS). For video quality the equivalence between the MOS and PSNR can be approximate through the Table 2:

TABLE 2
CORRESPONDENCE BETWEEN THE *MEAN OPINION SCORE* (MOS) AND THE *PEAK TO SIGNAL NOISE RATIO* (PSNR)

Mean Opinion Score	**PSNR [dB]**
5 (Excellent)	>37
4 (Good)	31 – 37
3 (Fair)	25 – 31
2 (Poor)	20-25
1 (Bad)	<20

At this point, it is necessary to remember that the exploitation of temporal redundancy to obtain compression in video sequences leaded to the appearance of three different frame types [49]: I (Intra) frames, P (Predicted) frames and B (Bidirectional) frames. The I frames only resort in spatial redundancy. These frames avoid the error propagation but require higher transmission resources. On the other hand, the P frames are “predicted” from a former frame (I or P). These frames are composed by vectors that describe the localization of similar areas in the former frame. This way, they require approximately half of the resources comparatively to I frames. Finally, the B frames are encoded using backward and forward temporal redundancy, relatively to I and P frames, and require approximately a quarter of resources comparatively to I frames. The group of frames formed by an I frame and all the subsequent frames until the next I frame is denominated *Group of Pictures* (GOP). Notice that the lost of an I frame makes impossible to correctly decode the remaining frames pertaining to the respective GOP.

The composition of a video sequence and the interdependency between frames is described in Figure 15. The basic element of the sequence is a “block” formed by 8x8 pixels. A group of four adjacent blocks forms a macro-block, which together with other macro-blocks form a slice. The ensemble of several slices originates a frame.

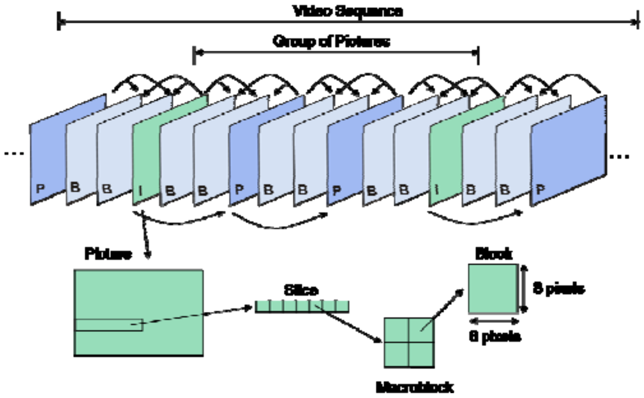

Fig. 15. Video Sequence composition.

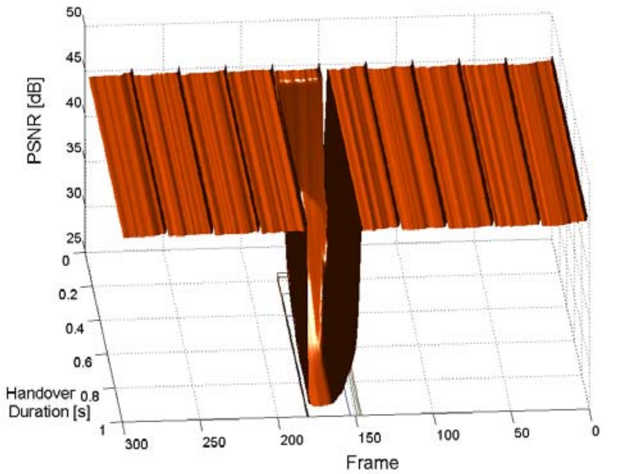

Fig. 16. PSNR versus the Handover Duration when the SEMUD mechanism is disabled.

Following the quality of the received frames during the simulation versus the handover duration is presented. It was considered a constant Buffer and Cache size (100 KB) and the frames quality (PSNR) was evaluated versus the handover duration (between 50 ms and 1 s with a 50 ms step).

The influence of the packet losses occurring during the handover period in the quality of the frames which are received during this period is noticeable when the SEMUD mechanism is disabled, as shown in Figure 16. Notice that the amount of frames with low quality is surprisingly high for almost all the values of the handover duration. This occurs because an I frame is lost among the packet losses that occur during handover. Loosing an I frame leads to the impossibility to decode the remaining frames pertaining to the GOP.

As shown in Figure 17, when the SEMUD mechanism is enabled, a substantial improvement in the quality of the received sequence is observed. The packets which are not received during handover are now recovered when it finishes. Consequently, the frames transmitted during the handover period do not suffer degradation.

Surprisingly, there is unexpected quality degradation in frames that do not correspond to the handover period. These frames correspond to results that were obtained for high handover values (close to 1 s). In this case, the number of packets not received during handover is high and its recovery leads to the full occupation of the Buffer. Afterwards, the occupation of the Buffer will suffer a fluctuation around this

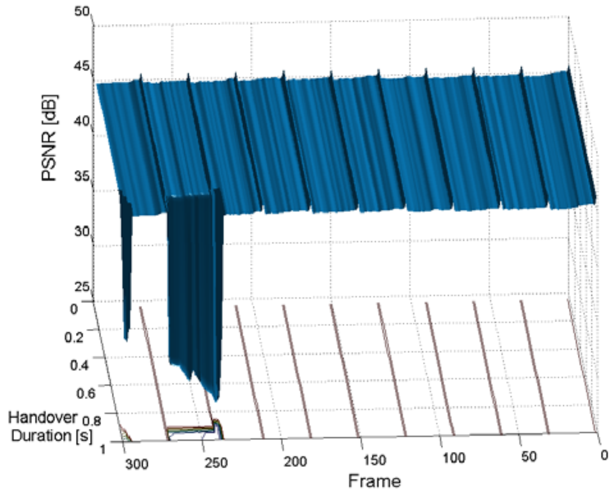

Fig. 17. PSNR versus the Handover Duration when the SEMUD mechanism is enabled.

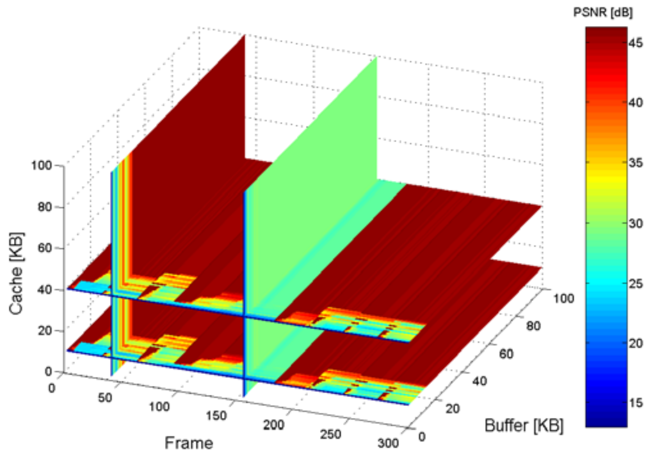

Fig. 18. PSNR versus the Cache and the Buffer Size when SEMUD mechanism is disabled.

value. Obviously, when the fluctuation is above the Buffer size the arriving packets will be lost, and consequently, degradation in the quality of the frames will be noticed. Particularly, when the affected frame is an I frame it will be impossible to decode the frames pertaining to its GOP. This way, the frames affected by the quality degradation in Figure 17 do not result from the losses that occur directly because of the handover but from the ones that occur due to the Buffer overflow.

The consequences of the packet losses in the quality of the received video sequence versus the Cache and Buffer sizes are show in Figure 18 when the mechanism is disabled.

Notice the reduction in quality of the frames pertaining in the [150, 179] interval. This behaviour was expected for the frames in the [150, 164] because they are directly affected by the packet losses that occur during handover (occurring during the [4.984; 5.484]s interval). The same is not true for the [165, 179] frames since they are not directly affected by the handover. However, these frames belong to a GOP whose frame I was lost during handover. For this reason, it is impossible to decode these frames despite their integrity has not been affected. As show in Figure 19, when the proposed mechanism is enabled the results are significantly improved.

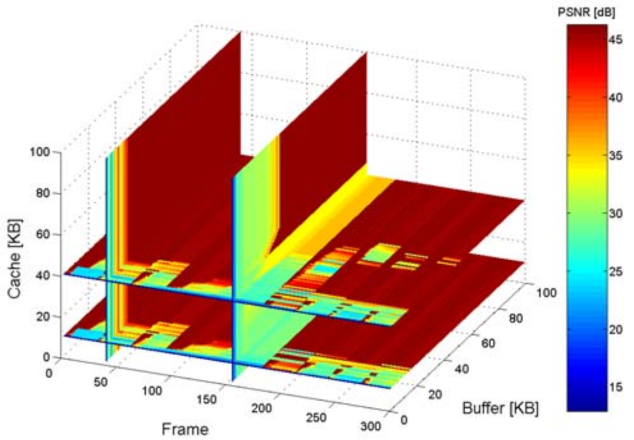

Fig. 19. PSNR versus the Cache and Buffer Sizes when SEMUD mechanism is enabled.

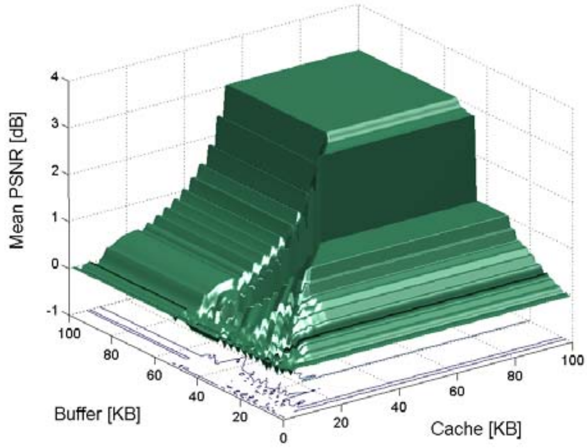

Fig. 20. Mean PSNR improvement versus the Cache and Buffer Sizes.

Figure 20 presents the mean PSNR improvement of the "News" video sequence versus the Buffer and Cache sizes. As before, it was considered a handover duration of 500 ms and a Cache and Buffer sizes varying between 1 and 100 KB.

The enhancement obtained through the use of the proposed mechanism can reach the 3.0191dB.

E. Example

Following, an illustrative example of the improvement achieved when using the proposed mechanism is presented in Figure 21. The depicted frames pertain to the video sequence entitled "News", have the YUV format, 4:2:0 sampling and CIF (352x288) dimension. The YUV video sequence frames were compressed using a MPEG4 encoder into a sequence with a GOP composed of 30 frames and transmitted with a 30 frame/s rate. Each frame was fragmented in blocks with 1024 bytes length which were transported in RTP packets. Those packets were sent using a VBR with an average rate of 86 KB/s. The results were obtained for a constant cache size (100 KB), a constant buffer size (100 KB) and a constant handover duration (500 ms). Moreover, the mobile node was in handover during the [4.934, 5.434]s interval.

Fig. 21. Comparison between received frames when the mechanism is disable (above) and enabled (below).

The quality degradation which occurs when the mechanism is disabled can be observed in the frames presented in Figure 21. The decoder tries to recover the lost/degraded frames by resorting in the last frame that was correctly decoded. When the proposed mechanism is enabled the quality of the frames is greatly improved, as shown in Figure 21.

Additional simulation results concerning the efficiency of the proposed mechanisms in assuring end-to-end QoS guarantees for ongoing multi-user sessions along heterogeneous environments are included in previous work [46]. The results present the impact in terms of throughput, packet delay and percentage of packet losses caused by the mobility of users inside and between networks when the preferred service class is overloaded. For instance, the adaptation of sessions to less important inter-network service classes in a congestion period avoids the session blocking and keeps the session with acceptable quality (the one-way delay is increased only in 5%, which remains acceptable for the session).

VI. Conclusions and Future Work

This paper presents a proposal to allow seamless mobility of multi-user sessions over heterogeneous networks. This goal is achieved through the integration of *Seamless Mobility of Users for Media Distribution Services* (SEMUD) and *Multi-user Session Control* (MUSC) mechanisms. SEMUD supports seamless mobility based on the cooperation between caches placed in access-routers and buffers placed in mobile devices, multi-user session context transfer, and interaction with mobility prediction schemes. MUSC controls mobility by assuring the control of QoS mapping, QoS adaptation and connectivity of sessions among networks.

The performance evaluation shows that MUSC has a convergence time and a signalling overhead independent of the number of receivers of the same session in the same access-network. Moreover, the expected advantages of SEMUD in reducing packet losses are confirmed by the obtained results. For example, for a handover duration of 500 ms and an average rate of 86 KB/s the packet losses are reduced 75% with a 50 KB cache size and a 55 KB buffer size.

As future work, further evaluation will be done to confirm the performance of MUSC and SEMUD in an experimental scenario, and to analyze the scalability of the proposal.

REFERENCES

[1] F. Hartung, N. Niebert, A. Schieder, R. Rembarz, S. Schmid, and L. Eggert, “Advances in Network-Supported Media Delivery in the Next-Generation Mobile Systems”, *IEEE Communications Magazine*, vol. 44, no. 8, August 2006.

[2] S. Blake, D. Black, M. Carlson, E. Davies, and W. Weiss, “An Architecture for Differentiated Services”, *IETF RFC 2475*, December 1998.

[3] H. Holbrook and B. Cain, “Source-Specific Multicast for IP”, *IETF RFC 4607*, August 2006.

[4] S. Bhattacharyya, “An Overview of Source-Specific Multicast (SSM)”, *IETF RFC 3569*, July 2003.

[5] E. Cerqueira, L. Veloso, A. Neto, M. Curado, E. Monteiro, and P. Mendes, “A Unifying Architecture for Publish-Subscribe Services in the Next Generation IP Networks”, in *Proc. IEEE GLOBECOM 2006*, San Francisco, CA, USA, November 2006.

[6] E. Cerqueira, L. Veloso, Augusto Neto, M. Curado, E. Monteiro, and P. Mendes, “Q3M: QoS Architecture for Multi-user Mobile Multimedia Sessions in 4G systems”, in *Proc. 10th IFIP/IEEE International Conference on Management of Multimedia and Mobile Networks and Services (MMNS 2007)*, San José, California, USA, October 2007.

[7] L. Veloso, E. Cerqueira, A. Neto, M. Curado, E. Monteiro, and P. Mendes, “Mobility Support of Multi-User Services in Next Generation Wireless Systems”, in *Proc. 26th IEEE International Performance Computing and Communications Conference*, New Orleans, USA, 2007.

[8] E. Cerqueira, L. Veloso, A. Neto, M. Curado, E. Monteiro, and P. Mendes, “Multi-user Session Control in the Next Generation Wireless System”, in *Proc. 4th ACM International Workshop on Mobility Management and Wireless Access (MobiWac 2006)*, Malaga, Spain, October 2006.

[9] Y. Min-hua, Y. Lv-yun, L. Yu, and Z. Hui-min, “The Implementation of Multicast in Mobile IP”, in *Proc. IEEE Wireless Communication and Networking Conference*, New Orleans, USA, March 2003.

[10] Y. He, M. Faloutsos, S. Krishnamurthy, and B. Huffaker, “On routing asymmetry in the Internet”, in Proc. *IEEE Global Telecommunications Conference (GLOBECOM 2005)*, St. Louis, USA, November 2005.

[11] A. Neto, E. Monteiro, E. Cerqueira, P. Mendes, and A. Rissato, “A Resource Reservation Protocol Supporting QoS-aware Multicast Trees for Next Generation Networks”, in *Proc. 12th IEEE Symposium on Computers and Communications (ISCC 2007)*, Aveiro, Portugal, July 2007.

[12] L. Veloso, P. Mendes, and E. Monteiro, “Mobility Management Techniques in Heterogeneous Environments”, DoCoMo EuroLabs EUL-IFN-107, August 2007.

[13] D. Johnson, C. Perkins, and J. Arkko, “Mobility Support for IPv6”, *IETF RFC 3775*, June 2004.

[14] H. Soliman, C. Catellucia, K. Malki, and L. Bellier, “Hierarchical Mobile IPv6 Mobility Management (HMIPv6)”, *IETF RFC 4140*, August 2005.

[15] R. Koodli, “Fast Handovers for Mobile IPv6”, *IETF RFC 4068*, July 2005.

[16] K. Malki, “Low Latency Handoffs in Mobile IPv4”, *IETF Internet Draft*, October 2005.

[17] A. Dutta, E. Berg, D. Famolari, V. Fajardo, Y. Ohba, K. Taniuchi, T. Kodama, and H. Schulzrinne, “Dynamic Buffering Control Scheme for Mobile Handover”, in *Proc. IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Helsinki, Finland, September 2006.

[18] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, “SIP: Session Initiation Protocol”, *IETF RFC 3261*, June 2002.

[19] K. Oberle, S. Wahl, and A. Sitek, “Enhanced Methods for SIP based Session Mobility in a Converged Network”, in *Proc. IST Mobile and Wireless Communications Summit*, Budapest, Hungry, July 2007.

[20] L. Zhou and Y. Sun, “An Analysis of Multicast Support for Mobile Hosts Using Mobile IPv6”, in *Proc. IEEE International Conference on Wireless Communications, Networking and Mobile Computing*, Wuhan, China, September 2006.

[21] J. Loughney, M. Nakhjiri, C. Perkins, and R. Koodli, “Context Transfer Protocol (CXTP)”, *IETF RFC 4067*, July 2005.

[22] I. Miloucheva, "Context Management for Efficient Mobile Multicast Services", in *Proc. International Workshop on Context in Mobile Human Computer Interfaces*, Salzburg, Austria, September 2005.

[23] J. Babiarz, K. Chan, and F. Bake, “Configuration Guidelines for DiffServ Service Classes”, *IETF RFC 4594*, August 2006.

[24] R. Schantz, J. Loyall, C. Rodrigues, and D. Schmidt, “Controlling Quality-of-service in Distributed Real-time and Embedded Systems Via Adaptive Middleware: Experiences with Auto-adaptive and Reconfigurable Systems”, *IEEE Transaction on Software Practice &.Experience*, vol. 36, no. 11, September 2006, pp. 1189–1208.

[25] M. Ryu, H. Park, and S. Shin, “QoS Class Mapping over Heterogeneous Networks using Application Service Map”, in *Proc. IEEE International Conference on Networking, International Conference on Systems and International Conference on Mobile Communications and Learning Technologies (ICNICONSMCL'06)*, Mauritius, April 2006.

[26] Z. Mammeri, “Approach for End-to-End QoS Mapping and Handling”, in *Proc. IFIP International Conference on Wireless and Optical Communications Networks (WOCN 2005)*, Dubai, United Arab Emirates, March 2005.

[27] J. Chen, W. Jiao, and Q. Guo, “An Integrated QoS Control Architecture for IEEE 802.16 Broadband Wireless Access Systems”, in *Proc. IEEE Global Telecommunications Conference*, St. Louis, USA, November 2005.

[28] H. Zhao, N. Ansari, and Q. Shi, “Layered MPEG Video Transmission over IP DiffServ”, in *Proc. IEEE International Conference on Information Technology: Coding and Computing*, Las Vegas, USA, April 2005.

[29] S. Johansen, A. Kim, and A. Perkis, “Quality Incentive Assisted Congestion Control for Receiver-Driven Multicast”, in *Proc. IEEE International Conference on Communications*, Glasgow, UK, June 2007.

[30] J. Kim, G. Kwon, N. Kim, A. Morales, and S. Ko, “Efficient Video Transcoding Technique for QoS-based Home Gateway Service”, *IEEE Transactions on Consumer Electronics*, vol. 52, no. 1, February 2006.

[31] T. Inoue and R. Kurebayash, “Evaluating the Impact of Tunneling on Multicast Efficiency”, in *Proc. IEEE Asia-Pacific Conference on Communications*, Perth, Australia, October 2005.

[32] E. Pearce, “System and Method for Enabling Multicast Telecommunications”, in Application Patent, US7079495 B1, July 2006.

[33] L. Begeja, “System and Method for Delivering Content in a Unicast/Multicast Manner”, in Application Patent, US708142 B1, July 2005.

[34] Z. Jumbiao, “Multicast over Unicast in a Network”, in Application Patent, MX2006PA03857, July 2006.

[35] R. Hancock, G. Karagiannis, J. Loughney, and S. V. Bosch, “Next Steps in Signalling (NSIS): Framework”, *IETF RFC 4080*, June 2005.

[36] G. Ash, A. Bader, C. Kappler, and D. Oran, “QoS NSLP QSPEC Template”, *IETF Internet Draft*, October 2007.

[37] M. Handley, V. Jacobson, and C. Perkins, “SDP: Session Description Protocol”, *IETF RFC 4566*, July 2006.

[38] L. Veloso, E. Cerqueira, P. Mendes, and E. Monteiro, “Selective Mobility support of Multi-user Services in Wireless Environments”, in *Proc. 18th Annual IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC'07)*, Athens, Greece, September, 2007.

[39] E. Cerqueira, T. Sousa, A. Neto, L. Veloso, M. Curado, E. Monteiro, and P. Mendes, “QoS Mapping and Adaptation in Next Generation Networks”, in *Proc. IEEE International Symposium on Applications and the Internet Workshop (SAINTW'07)*, Hiroshima, Japan, January 2007.

[40] M. Sricharan, V. Vaidehi, and P. Arun, “An Activity Based Mobility Prediction Strategy for Next Generation Wireless Networks”, in *Proc. IFIP Conference on Wireless and Optical Communications Networks*, April 2006.

[41] B. Cain, B. Fenner, and A. Thyagarajan, “Internet Group Management Protocol, Version 3”, *IETF RFC 3376*, October 2002.

[42] R. Vida and L. Costa, “Multicast Listener Discovery Version 2 (MLDv2)”, *IETF RFC 3810*, June 2004.

[43] K. Fall and K. Varadhan, “The ns Manual (formerly ns Notes and Documentation)”, The VINT Project, September 2007, Available online at: http://www.isi.edu/nsnam/ns/ns-documentation.html.

[44] J. Klaue, B. Rathke, and A. Wolisz, “EvalVid - A Framework for Video Transmission and Quality Evaluation”, in *Proc. 13th International Conference on Modelling Techniques and Tools for Computer Performance Evaluation (TOOLS 2003)*, Urbana, Illinois, USA, September 2003.

[45] C. Ke, C. Lin, C. Shieh, and W. Hwang, “A Novel Realistic Simulation Tool for Video Transmission over Wireless Network”, in *Proc. IEEE International Conference on Sensor Networks, Ubiquitous, and Trustworthy Computing (SUTC2006)*, Taiwan, June 2006.

[46] E. Cerqueira, L. Veloso, M. Curado, E. Monteiro, and P. Mendes, “QoS Mapping and Adaptation Control for Multi-user Sessions over Heterogeneous Wireless Networks”, in *Proc. 3rd ACM International Mobile Multimedia Communications Conference (Mobimedia 2007)*, Nafpaktos, Greece, August 2007.

[47] T. Camilo, “SSM Extension for NS-2”, December, 2006, Available online at: http://eden.dei.uc.pt/~tandre/ssm_extension.

[48] Video Traces Research Group, "YUV 4:2:0 Video Sequences", Arizona State University, Available online at: http://trace.eas.asu.edu/yuv/index.html.

[49] L. Veloso, P. Mendes, and E. Monteiro, "MPEG over IP Networks", DoCoMo EuroLabs EUL-IFN-96, May 2007.

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