SIP-Based Context-Aware Mobility for IPTV IMS Services

Victor Sandonis
Departamento de Ingeniería Telemática
Universidad Carlos III de Madrid
Avda. de la Universidad, 30
28911 Leganés – Madrid (Spain)
vsandoni@it.uc3m.es

Ismael Fernandez
Departamento de Ingeniería Telemática
Universidad Carlos III de Madrid
Avda. de la Universidad, 30
28911 Leganés – Madrid (Spain)
ismafc06@gmail.com

Ignacio Soto
Departamento de Ingeniería de Sistemas Telemáticos
Universidad Politécnica de Madrid
Avda. Complutense, 30
28040 Madrid (Spain)
isoto@dit.upm.es

ABSTRACT

In this paper we propose a solution to support mobility in terminals using IMS services. The solution is based on SIP signalling and makes mobility transparent for service providers. Two types of mobility are supported: a terminal changing of access network and a communication being moved from one terminal to another. The system uses context information to enhance the mobility experience, adapting it to user preferences, automating procedures in mobility situations, and simplifying decisions made by users in relation with mobility options. The paper also describes the implementation and deployment platform for the solution.

Categories and Subject Descriptors

C.2.1 [**Computer-Communication Networks**]: Network Architecture and Design – network communications, packet-switching networks, wireless communications

General Terms

Design, Experimentation, Standardization.

Keywords

Mobility, IMS, SIP, IPTV, Context-awareness

1. INTRODUCTION

Access to any kind of communication services anytime, anywhere and even while on the move is what users have become to expect nowadays. This poses incredible high requirements to telecom operators' networks to be able to serve the traffic of those services. Mobility requires wireless communications, but bandwidth is a limited resource in the air interface. To be able to overcome this limitation the operators are pushing several solutions such as the evolution of the air interface technologies, the deployment of increasingly smaller cells, and the combination of several heterogeneous access networks.

Offering connectivity through a combination of several access networks requires the ability to manage the mobility of terminals across them, while ensuring transparency to service providers. Changing access network implies a change in the IP address of the terminal. To prevent this change from having an effect on the terminal open communications, we need to use a mobility support solution to take care of it.

In telecom networks, operators are pushing the adoption of the IP multimedia subsystem (IMS) to provide, through their data IP-based networks, multimedia services, such as VoIP or IPTV. In this paper we propose a mobility support solution based on SIP, the signalling protocol used in IMS. The solution is completely integrated in IMS, does not require modifications to the IMS specifications, and supports mobility transparently to service providers.

The proposed solution supports terminal mobility, the movement of a terminal between access networks without breaking its communications. Additionally, it also supports session mobility, allowing a user to move active communications among its terminals. The last key point of our proposal is the use of a context information server to enhance the users' mobility experience, for example automating some configuration tasks.

The mobility solution proposed in this paper is valid for any service based on IMS, both for unicast and multicast communications. Nevertheless in the context of the Celtic UP-TO-US project we are focusing on the IPTV service: multicast-based LiveTV and unicast-based Video on Demand (VoD), which will be our use cases through the paper.

The rest of the paper is organized as follows: section 2 reviews the terminology and analyses works in the literature related to the paper; section 3 describes the proposed solution for transparent SIP-based context-aware mobility; section 4 covers the implementation of the solution, with a description of the used tools and platforms, and their behaviour; finally section 5 highlights the conclusions that result from our work.

2. BACKGROUND

The first part of this section presents an overview of the various types of mobility; then the related work relevant to the solution proposed here is discussed.

For the case of IPTV Services two types of mobility have been considered: session mobility and terminal mobility.

Session mobility is defined as the seamless transfer of an ongoing IPTV session from the original device (where the content is currently being streamed) to a different target device. This transfer is done without terminating and re-establishing the call to the corresponding party (i.e., the media server). This mobility service makes possible, for example, that the user can transfer the current IPTV session displayed in her mobile phone to a stationary TV (equipped with its Set-Top-Box).

Session mobility has two modes of operation, namely, push mode and pull mode:

- In the *push mode*, the original device (where the content is currently being streamed) discovers the devices to which a session mobility procedure potentially could be initiated, selects one device and starts the mechanism.

- In the *pull mode*, the user selects in the target device, the on-going session he wants to move. Thus, the target device needs previously to learn the active sessions for this user that are currently being streamed to other devices. After selecting a session, the target device initiates the session mobility procedure.

Terminal mobility allows a terminal to move between access networks (the terminal changes its IP address) while keeping ongoing communications alive despite the movement. This type of mobility is aligned with current network environments that integrate different wireless and fixed access network technologies, using terminals with multiple network interfaces and enabling users to access IPTV service via different access networks.

The mobility solution presented in this paper considers three different ways to initiate the mobility:

- *Manual*: The user decides that he wants to initiate a mobility procedure.

- *Semi-Automatic*: The UP-TO-US system proposes initiating a mobility procedure. To perform this proposal the system takes into consideration the different context information related to the user, the terminals and the networks. The user always has the opportunity to accept or reject this suggestion.

- *Automatic*: The UP-TO-US system taking into consideration all the context information initiates a mobility procedure.

The user has the opportunity to configure as part of his preferences the mobility mode he prefers (manual/semi-automatic/automatic).

Terminal mobility support in IP networks has been studied for some time. In particular, the IETF has standardized the Mobile IP (MIP) protocols both for IPv4 [1] and IPv6 [2] to provide mobility support in IP networks. Mobile IP makes terminal mobility transparent to any communication layer above IP, including the applications and, therefore a node is able to change the IP subnetwork is being used to access the Internet without losing its active communications. Mobile IP is a good solution to provide terminal mobility support but its integration with IMS is far from trivial [3][4][5]. This is essentially because MIP hides from the application layer, including the IMS control, the IP address used by a mobile device in an access network, but IMS requires this address to reserve resources in the access network for the traffic of the services used in the mobile device.

Another alternative is to use the Session Initiation Protocol (SIP) to support mobility [6][7][8] in IP networks. In this respect, 3GPP has proposed a set of mechanisms to maintain service continuity [9] in the event of terminal mobility or session mobility. Using SIP to handle mobility presents the advantage of not requiring additional mechanisms outside the signalling used in IMS. But the traditional SIP mobility support is not transparent to the IPTV service provider.

Making mobility transparent to the IPTV service provider is a desirable feature that is feasible to provide. TRIM architecture [10] provides terminal mobility support in IMS-based networks without requiring any changes to the IMS infrastructure and without entailing any upgrades to the service provider's application. The main component of the TRIM architecture is a SIP Application Server (AS) located in user's home network. This AS stays on the signalling path between the two communicating end-points. It receives all the SIP signalling messages corresponding to the multimedia sessions and controls a set of Address Translators located also in the user' home network. A particular mobile user is served by one of these Address Translators. The Address Translator is configured by the SIP Application Server to address the media received from the multimedia service provider (i.e., IPTV service provider) to the appropriate location where the mobile terminal is at that moment. Analogously, it forwards the traffic in the opposite direction, i.e. from the mobile terminal to the IPTV service provider. In particular, the address translator simply changes the IP addresses and ports contained in the packets according to the configuration provided by the SIP Application Server. Therefore, the service provider always observes the same remote addressing information for the Mobile Terminal, no matter where the latter is located.

The UP-TO-US mobility solution uses the functionality explained above from the TRIM architecture[1] to provide not only terminal mobility but also session mobility (not considered in TRIM). Additionally, the UP-TO-US mobility solution integrates the concept of context-awareness mobility to enhance the user's mobility experience, for example proposing to the user a session transfer according to his preferences and the available nearby devices.

3. TRANSPARENT SIP-BASED CONTEXT-AWARE MOBILITY

In this section we describe the UP-TO-US IPTV Service Continuity solution that provides transparent SIP-based context-aware mobility.

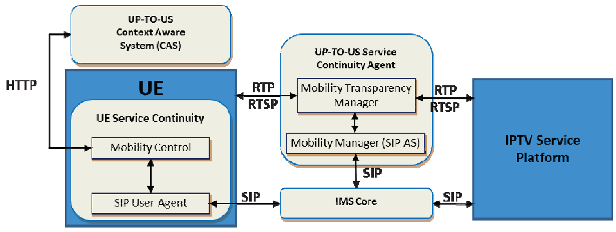

Figure 1. Architecture

3.1 Architecture

The architecture of the UP-TO-US IPTV Service Continuity solution (see figure 1) is composed by the modules explained in next subsections.

3.1.1 UP-TO-US Context-Awareness System

The UP-TO-US Context-Awareness System (CAS) is the module in charge of storing the context information of the user environment, the network domain and the service domain. It provides context information about mobility user preferences, available devices, available access networks, user's session parameters, etc. This information is useful to manage the different mobility procedures.

[1] The TRIM architecture also adds functionality (modifications) to user terminals that we have chosen to avoid in the UP-TO-US mobility solution.

3.1.2 UP-TO-US Service Continuity Agent

The UP-TO-US Service Continuity Agent is located in the home network. In order to provide mobility support functionalities, it is inserted both in the signalling path and in the data plane media flows of user's IPTV sessions. The UP-TO-US Service Continuity Agent is divided in two sub-modules: the Mobility Manager and the Mobility Transparency Manager.

3.1.2.1 Mobility Manager

The Mobility Manager is a SIP Application Server (AS) [6] located in the signalling path between the User Equipment (UE) and the IPTV Service Platform, such that receives all SIP signalling messages of users' multimedia sessions (the IMS initial filter criteria is configured to indicate that SIP session establishment messages have to be sent to the Mobility Manager). The Mobility Manager makes mobility transparent to the control plane of the service provider side. Additionally, it modifies SIP messages and Session Description Protocol (SDP) payloads to make media packets of multimedia sessions travel through the Mobility Transparency Manager. The Mobility Manager controls the Mobility Transparency Manager according to the information about IPTV sessions it extracts from SIP signalling messages. This information concerns the addressing parameters of the participants in the session and the session description included in the SDP payload of SIP messages.

3.1.2.2 Mobility Transparency Manager

The Mobility Transparency Manager is a media forwarder whose functionality is to make the mobility transparent to the user plane of the service provider side. This way, the content provider is unaware of UE movements between different access networks (terminal mobility) or movements of sessions between devices (session mobility) in the client side. The Mobility Transparency Manager is controlled by the Mobility Manager according to the information extracted from the SIP signalling messages of user's multimedia sessions. It is configured to properly handle the media flows of user's IPTV sessions[2]. This way, when the UE moves between different access networks (terminal mobility) or when the session is transferred between devices (session mobility), the Mobility Transparency Manager is configured to forward the media received from the content provider to the current location (access network) of the UE in case of terminal mobility or to the new UE in case of session mobility. In the opposite direction, user plane data received from the UE is forwarded to the content provider. This way, data packets are forwarded to the appropriate destination IP address and port according to the information provided by the Mobility Manager.

The result is that the content provider side always perceives and maintains the same remote addressing information regardless of the UE handoffs between different access networks (terminal mobility) or when the session is transferred between devices (session mobility). Figure 2 describes how the Mobility Transparency Manager forwards data packets in the user plane to the appropriate destination IP address and port.

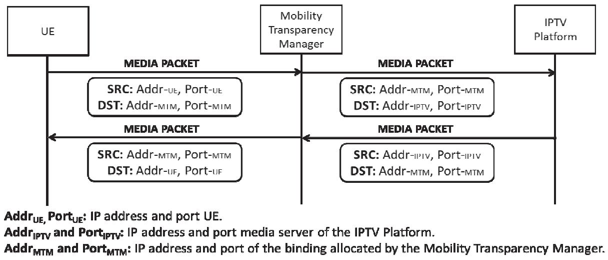

Figure 2. Mobility Transparency Manager forwarding

On the other hand, the Mobility Transparency Manager behaves as a RTSP proxy inserted in the RTSP signalling path between the UE and the IPTV Platform for Video on Demand service[2].

3.1.3 UE Service Continuity

The UE Service Continuity module is an element of the User Equipment. The UE Service Continuity interacts with the UP-TO-US Context Awareness System (CAS) and with the UP-TO-US Service Continuity Agent to manage mobility support functionalities. The UE Service Continuity module is divided in two sub-modules: the Mobility Control and the SIP User Agent.

3.1.3.1 Mobility Control

The Mobility Control sub-module has all the logic of mobility and makes mobility decisions taking into account the available context information and user mobility preferences obtained from the interaction with the CAS. This way, the Mobility Control sub-module subscribes to the mobility triggers in the CAS which monitors user context and notifies to the Mobility Control sub-module when a mobility event occurs. Possible mobility events are (1) trigger informing that the user is close to other devices different from the current one to initiate push session mobility, (2) trigger informing about the user proximity to the device to initiate pull session mobility and (3) trigger informing that the terminal has to perform a handover to another access network. On the other hand, the Mobility control sub-module retrieves from the CAS context information about mobility user preferences, available devices, available access networks, user's session parameters, etc. that is useful for handling mobility procedures.

Based on the combination of the existing context information that is available locally in the UE (for instance IP address and access network type of the interfaces of the UE, Wi-Fi signal strength level, parameters of the session that is active on the UE, etc.) and the information retrieved from the CAS, the Mobility Control sub-module makes mobility decisions on which mobility procedure is appropriate under a specific context (push session mobility, pull session mobility or terminal mobility procedures) and requests the SIP User Agent to exchange the appropriate SIP signalling to handle the different mobility procedures with the UP-TO-US Service Continuity Agent. This way, the UP-TO-US Service Continuity Agent is maintained updated with the current addressing information of the UE when it moves between access networks (terminal mobility) or the session is transferred between devices (session mobility).

3.1.3.2 SIP User Agent

The SIP User Agent is in charge of the SIP signalling exchange between the UE and the UP-TO-US Service Continuity Agent to handle the different SIP procedures for registration, session establishment, session modification, session release, etc. SIP

[2] According with the ETSI TISPAN standard [11], the SDP offer of SIP signalling messages contains a media description for the RTSP content control channel. This way, RTSP packets exchanged between UE and the service platform are considered as belonging to a media flow in the data plane. The Mobility Transparency Manager handles RTSP packets as a RTSP proxy.

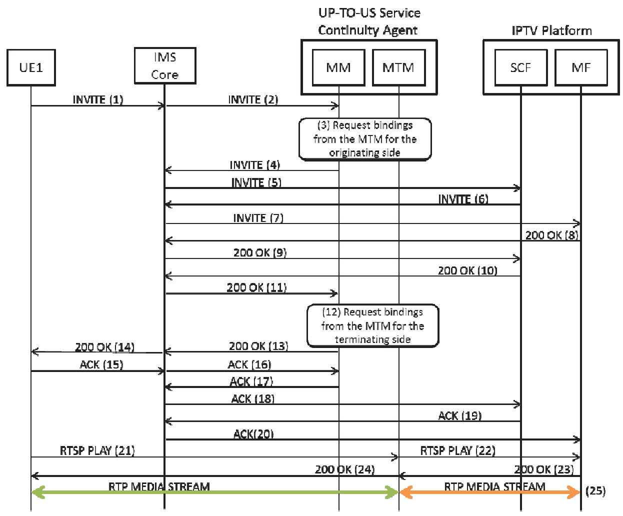

Figure 3. VoD session establishment

signalling messages travel through the IMS infrastructure to arrive to the UP-TO-US Service Continuity Agent.

3.2 Procedures of the proposed solution

The IPTV service continuity solution considers the Video on Demand (VoD) service and the LiveTV service. Next subsections explain the different procedures of the IPTV service continuity solution for VoD and LiveTV services. These procedures are based on TRIM [10] and IETF, ETSI TISPAN and 3GPP specifications [7][8][9] and [11]. The figures presented in the next sections show the exchange of the main messages between modules assuming that the user is already registered in the IMS on UE1 and UE2. The modules described in section 3.1 are shown in the figures (the MM sub-module is the Mobility Manager and the MTM sub-module is the Mobility Transparency Manager), but also the IPTV Service Control Function (SCF), the IPTV Media Function (MF) [12] of the IPTV Platform and the Elementary Control Function and the Elementary Forwarding Function (ECF/EFF) [13] appear in the figures. The SCF is the module of the IPTV platform responsible of controlling the sessions of all IPTV Services (service authorization, validation of user requests based on user profile, MF selection Billing & Accounting, etc.). The MF is in charge of the media delivery to the users. The ECF/EFF can be described as the access router that processes Internet Group Management Protocol (IGMP) [14] signalling and forwards multicast traffic.

3.2.1 Video on Demand Service

This section describes the procedures of the solution for the VoD Service.

3.2.1.1 Session Establishment

The establishment of a VoD session is illustrated in figure 3. After the user selects the content he wants to watch, UE1 begins the session establishment issuing an INVITE request (1) to the IMS Core that according with the IMS initial filter criteria is routed (2) towards the Mobility Manager. The SDP offer of the INVITE message contains the media descriptions of the RTSP content control flow and the content delivery flows, and it is processed by the Mobility Manager (3). For each media component included in the SDP offer:

a) The Mobility Manager obtains the addressing information where the UE1 will receive the data traffic of the media component. This is the IP address and port included in the media component description. The Mobility Manager requests the Mobility Transparency Manager to create a binding for this addressing information. As result, the Mobility Transparency Manager allocates a new pair of IP address and port, and returns it to the Mobility Manager.

b) The Mobility Manager modifies the media description of the SDP payload replacing the IP address and port by the binding obtained from the Mobility Transparency Manager. This way, the data traffic of the media component will be sent by the MF to the Mobility Transparency Manager.

After this processing, the Mobility Manager issues a new INVITE message (4) that includes the modified SDP offer that according to the IMS initial filter criteria reaches the Service Control Function of the IPTV platform (5). The SCF performs service authorization and forwards the INVITE message (6) to the appropriate MF. Since the SDP payload received by the MF includes the IP addresses and ports of the bindings allocated by the Mobility Transparency Manager, the data traffic of the different media components will be sent by the MF to the Mobility Transparency Manager. On the other hand, the MF generates a 200 OK response that includes a SDP payload with the media descriptions of the content delivery flows and the RTSP content control flow for the MF side. The MF includes in the SDP payload the RTSP session ID of the content control flow and the RTSP URI to be used in RTSP requests. The 200 OK is routed back and arrives to the Mobility Manager (8 -11). The Mobility Manager processes the 200 OK message from the MF and its SDP offer. For each media component included in the SDP offer:

a) The Mobility Manager obtains the addressing information of the MF for the data traffic of the media component. This is the IP address and port included in the media component description. The Mobility Manager requests the Mobility Transparency Manager to create a binding for this addressing information. As a result, the Mobility Transparency Manager allocates a new pair of IP address and port, and returns it to the Mobility Manager.

b) The Mobility Manager modifies the media description replacing the IP address and port by the binding obtained from the Mobility Transparency Manager. This way, the data traffic of the media component will travel through the Mobility Transparency Manager.

c) The RTSP URI parameter included by the MF in the SDP payload is modified in order to insert the Mobility Transparency Manager in the RTSP signalling path between the EU and the MF.

This way, both the content delivery flow and the RTSP content control flow travel through the Mobility Transparency Manager. The result is that the MF always perceives and maintains the same remote addressing information (the Mobility Transparency Manager binding) regardless the session mobility or terminal mobility. Therefore, mobility is transparent to the MF. Then, the

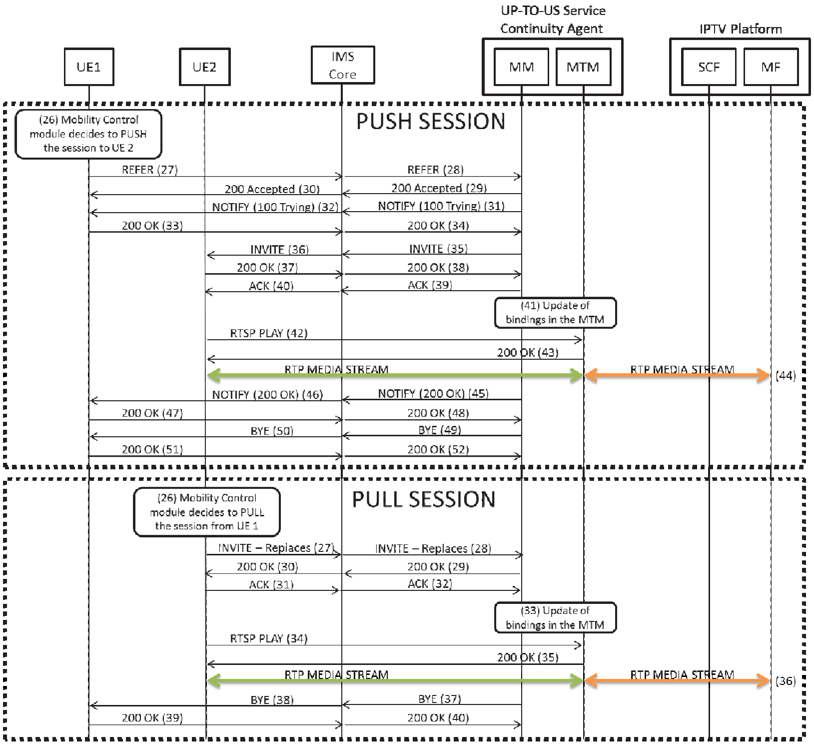

Figure 4. VoD push/pull session mobility

Mobility Manager issues a new 200 OK message with the modified SDP payload (13-14). The UE1 sends an ACK message to complete the SIP session establishment (15-20). Finally, the UE1 sends an RTSP PLAY message (21) to play the content. The RTSP PLAY message is issued to the RTSP URI using the RTSP session ID extracted from the SDP payload of the previous 200 OK message (14). This way, the RTSP PLAY message reaches the Mobility Transparency Manager, which generates a new RTSP PLAY message (22) and sends it to the MF. The RTSP 200 OK response is routed back to the UE 1, travelling through the Mobility Transparency Manager (23-24). At this moment (25), the MF begins streaming the RTP flows that travel through the Mobility Transparency Manager before arriving to the UE1. The Mobility Transparency Manager makes the mobility transparent to the MF by forwarding data packets according to the bindings created (steps (3) and (12)).

3.2.1.2 Push Session Mobility

Push session mobility procedures for VoD service are shown in figure 4. The figure represents a scenario where the UE1 transfers a multimedia session to the UE2. It is assumed that the VoD session is already established on UE1. The Mobility Control module of UE1, after receiving a trigger from the UP-TO-US CAS or under a user request, decides to push the session to UE2. UE2 addressing information to route subsequent messages is obtained by the Mobility Control module from CAS (26). In order to transfer the multimedia session to the UE 2, UE1 sends a REFER message [15] that arrives to the Mobility Manager after being routed by the IMS Core (27-28). The REFER message includes:

- Refer-To header that indicates the destination of the session transfer. In this case the addressing information of the user in UE2 (SIP URI of the user in UE2).
- Target-Dialog header that identifies the SIP session that has to be transferred from UE1 to UE2.
- Referred-By header that indicates who is transferring the session (SIP URI of the user in UE1).

The Mobility Manager processes the REFER method and informs to UE1 that it is trying to transfer the session (29-34). In order to transfer the session to the UE2, the Mobility Manager initiates a new SIP session with the UE2. The SDP payload of the INVITE message (35) includes the bindings generated by the Mobility Transparency Manager, the RTSP session ID of the content control flow and the RTSP URI to be used in RTSP requests obtained when the session was established on the UE1 (see figure 3)[3]. After the session is established with the UE2, the Mobility Manager updates the Mobility Transparency Manager (41) with the addressing information of UE2 extracted from the 200 OK message (38) (media descriptions of the RTSP content control flow and the content delivery flows for UE2). This way, the Mobility Transparency Manager can forward the content delivery flows and the RTSP content control flow to the new device, UE2. Note that the MF always perceives and maintains the same remote addressing information (the Mobility Transparency Manager binding) regardless the session is transferred between devices

[3] Note that if the UE2 does not support the codecs used in the session, the Mobility Manager could re-negotiate the session (by means of a re-INVITE) with the MF to adapt the multimedia session to the codecs supported by the UE2.

(session mobility), and it is also not involved in the mobility SIP signalling. Therefore, the session mobility is transparent to the MF.

Finally, in order to play the content, UE2 sends a RTSP PLAY (42) message to the MF. The RTSP PLAY message is issued to the RTSP URI using the RTSP session ID extracted from the SDP payload of the previous INVITE message (35). This way, the RTSP PLAY message reaches the Mobility Transparency Manager that answers with a RTSP 200 OK message (43) to UE2. Note that the Mobility Transparency Manager does not issue a new RTSP PLAY message to the MF because the session is already played. Since the Mobility Transparency Manager has been updated with the addressing information of the UE2 (step 41), it can forward the content delivery flows and the RTSP content control flow to the new device, UE2 (44). Once the session has been transferred to the UE2, the Mobility Manager informs the UE1 about the success of its requested session transfer, NOTIFY message (45-46), and terminates the session with the UE1 (49-52).

3.2.1.3 Pull Session Mobility

Pull session mobility procedures for VoD service are shown in figure 4. The figure represents a scenario where the UE 2 obtains a multimedia session that is active on the UE 1. It is assumed that the VoD session is already established on UE1. The Mobility Control module of UE2, after receiving a trigger from the CAS or under a user request, decides to pull the session that is established on the UE 1 (26). Information about the session established on UE 1 needed for performing the pull session mobility is obtained by the Mobility Control module from the CAS. In order to transfer the multimedia session from UE 1 to UE 2, UE 2 SIP User Agent triggered by the mobility control module sends an INVITE request (27-28) that includes:

a) Replaces header [16] that identifies the SIP session that has to be replaced (pulled from UE 1).

b) SDP payload with the media descriptions of the RTSP content control flow and the content delivery flows for UE 2.

The Replaces header allows the Mobility Manager to identify the SIP session that the UE 2 wants to pull. This way, the Mobility Manager sends a 200 OK response (29) to UE 2 that includes the bindings generated by the Mobility Transparency Manager, the RTSP session ID of the content control flow and the RTSP URI to be used in RTSP requests obtained when the session was established on the UE1 (see figure 3)[2]. Once the session establishment between UE2 and the Mobility Manager has finished (29-32), the Mobility Manager updates the Mobility Transparency Manager with the addressing information of the UE2 extracted from the INVITE-Replaces message (28) (media descriptions of the RTSP content control flow and the content delivery flows for UE2). This way, the Mobility Transparency Manager can forward the content delivery flows and the RTSP content control flow to the new device, UE2. Note that the MF always perceives and maintains the same remote addressing information (the Mobility Transparency Manager binding) regardless the session is transferred between devices (session mobility), and it is also not involved in the mobility SIP signalling. Therefore, the session mobility is transparent to the MF.

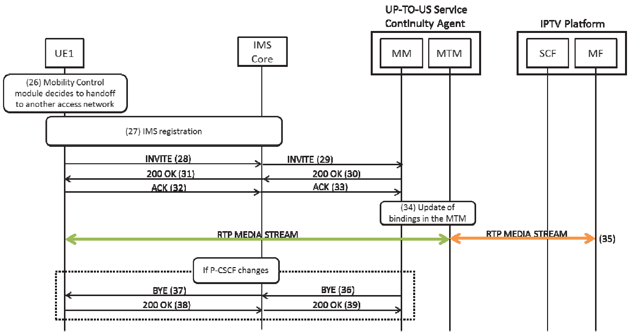

Figure 5. VoD terminal mobility

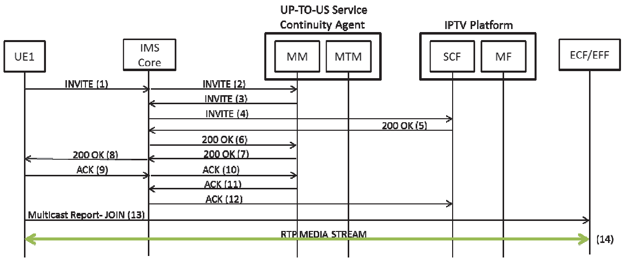

Figure 6. LiveTV session establishment

Finally, in order to play the content, UE2 sends a RTSP PLAY (34) message to the MF. The RTSP PLAY message is issued to the RTSP URI using the RTSP session ID extracted from the SDP payload of the previous 200 OK message (30). This way, the RTSP PLAY message reaches the Mobility Transparency Manager that answers with a RTSP 200 OK message (35) to UE2. Note that the Mobility Transparency Manager does not issue a new RTSP PLAY message to the MF because the session is already played. Since the Mobility Transparency Manager has been updated with the addressing information of the UE2 (step 33), it can forward the content delivery flows and the RTSP content control flow to the new device, UE2 (36). Once the session has been transferred to the UE2, the Mobility Manager terminates the session with the UE1 (37-40).

3.2.1.4 Terminal Mobility

Terminal mobility procedures for VoD service are shown in figure 5. The figure represents a scenario where the UE1 performs a handover between two access networks. It is assumed that the VoD session is already established on UE1. The Mobility Control module of the UE 1, after receiving a trigger from the CAS or under user request, decides to perform a handover to another access network (26). This way, after obtaining IP connectivity in the new access network, the UE 1 registers in the IMS to be reachable in the new IP address (27). In order to inform the Mobility Manager about the new addressing information in the new access network, the UE 1 sends an INVITE message (28) that includes the SDP payload with the media descriptions of the RTSP content control flow and the content delivery flows updated to the new addressing information in the new access network. This INVITE is a re-INVITE if the Proxy-Call Session Control Function (P-CSCF) is the same regardless of the handover between access networks or a new INVITE with a Replaces header in case that the P-CSCF has changed with the handover between access networks (a new dialog has to be established because the route set of proxies traversed by SIP messages changes). When the Mobility Manager processes the INVITE request, it answers with a 200 OK response (30) to UE 1 that includes the bindings generated by the Mobility Transparency Manager, the RTSP session ID of the content control flow and the RTSP URI to be used in RTSP requests obtained when the session was established (see figure 3). Once the Mobility Manager receives the ACK message (33) that completes the session establishment, it updates the Mobility Transparency Manager with the new addressing information of the UE 1. This way, the Mobility Transparency Manager can forward the content delivery flows and the RTSP content control flow (35) to the new location of the UE 1 (new access network). Note that the MF always perceives and maintains the same remote addressing information (the Mobility Transparency Manager binding) regardless the UE handoffs between different access networks. Therefore, the terminal mobility is transparent to the MF. Note that the UE 1 does not issue a new RTSP PLAY message to the MF because the session is already playing on the device. Finally, in the case the P-CSCF changes with the handover between access networks, the Mobility Manager close the session of the old SIP dialog to release the reserved resources in the old access network (36-39).

3.2.2 LiveTV Service

This section describes the procedures of the solution for the LiveTV Service.

3.2.2.1 Session Establishment

The establishment of a LiveTV session is illustrated in figure 6. After the user selects a BroadCast (BC) service, UE1 begins the session establishment issuing an INVITE request (1) to the IMS Core that according with the IMS initial filter criteria is routed (2) towards the Mobility Manager. The SDP offer of the INVITE message contains the media descriptions of the BC session that is processed by the Mobility Manager. Since the request is about a BC service, the Mobility Manager does not modify the SDP offer because the data traffic of the BC session will be sent by multicast. Therefore, it is not needed that the data traffic travels through the Mobility Transparency Manager. Then, the Mobility Manager forwards the unaltered INVITE message (3), that is routed towards the SCF of the IPTV platform (4) according with the IMS initial filter criteria. After performing service authorization, the SCF checks the SDP offer and generates a 200 OK response that includes in the SDP payload the same media descriptions of the received INVITE. The 200 OK is routed back towards the Mobility Manager (5-6). Then, the Mobility Manager sends the 200 OK message without any modification in its SDP payload because it is not needed that the data traffic travels through the Mobility Transparency Manager (it is multicast traffic). When UE1 receives the 200 OK message, it completes the SIP session establishment (9-12). Finally, UE 1 joins the multicast group of the BC session by sending an IGMP JOIN message (13) to the ECF/EFF which starts forwarding the multicast traffic of the BC session to the UE 1 (14).

3.2.2.2 Push Session Mobility

Push session mobility procedures for LiveTV service are shown in figure 7. The figure represents a scenario where the UE1 transfers a multimedia session to the UE2. It is assumed that the LiveTV session is already established on UE1. Regarding the SIP signalling, it is similar to the VoD service case (see previous explanation of SIP signalling for VoD service push session mobility for more details). The main difference between the procedures for VoD service and LiveTV service is that the Mobility Manager does not modify the SDP payload of SIP messages because the data traffic of the BC session is sent by multicast. Therefore, it is not needed that the data traffic travels

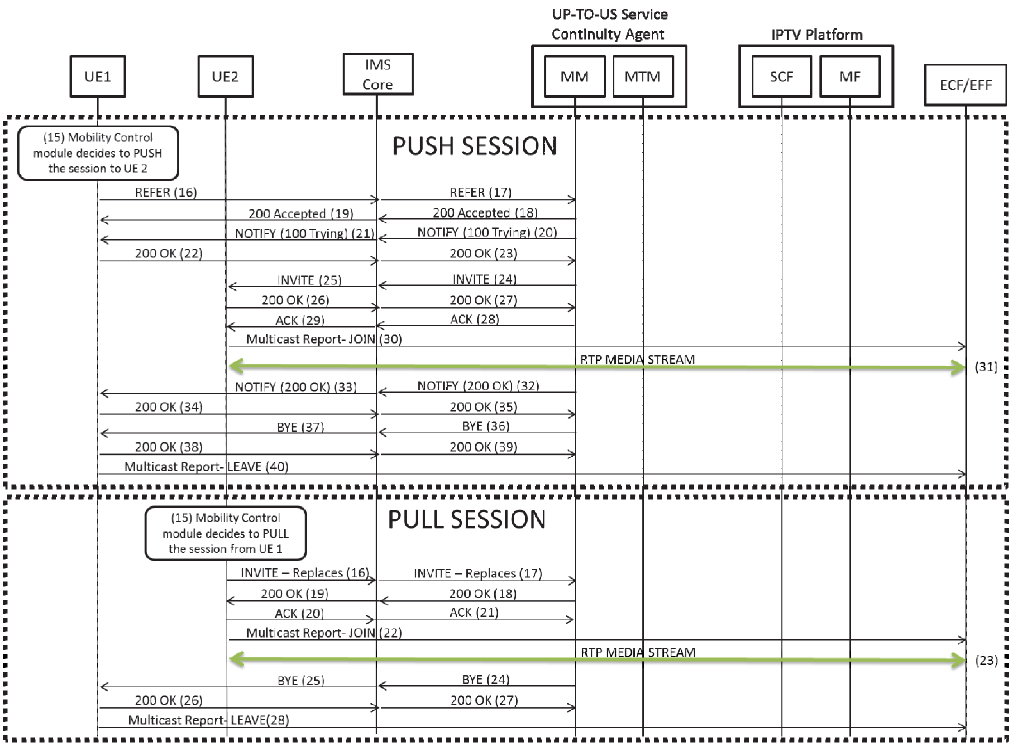

Figure 7. LiveTV push/pull session mobility

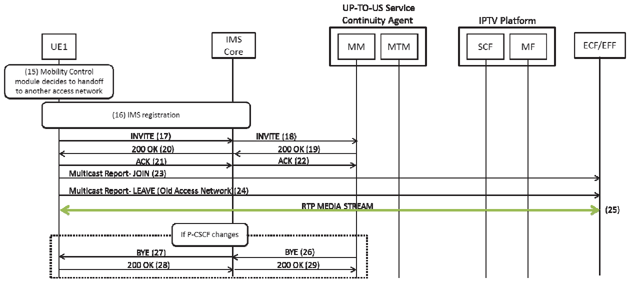

Figure 8. LiveTV terminal mobility

through the Mobility Transparency Manager. Another difference in the procedure is that after the new SIP session establishment, between the Mobility Manager the UE2, is completed (24-29), the UE 2 joins to the multicast group of the BC session by sending an IGMP JOIN message (30) to the ECF/EFF to receive the multicast traffic of the BC session (31). On the other hand, once the session has been transferred to the UE 2, the UE 1 leaves the multicast group of the BC session by sending an IGMP LEAVE message to the ECF/EFF (40).

3.2.2.3 Pull Session Mobility

Pull session mobility procedures for LiveTV service are shown in figure 7. The figure represents a scenario where the UE 2 obtains a multimedia session that is active on the UE 1. It is assumed that the LiveTV session is already established on UE1. As with the push session mobility, the SIP signalling is similar to the VoD service case (see previous explanation of SIP signalling for VoD service pull session mobility for more details). The main difference is the same, it is not needed that the data traffic travels through the Mobility Transparency Manager because it is sent by multicast, thus the Mobility Manager does not modify the SDP payload of SIP messages. Besides, the UE 2 joins the multicast group of the BC session by sending an IGMP JOIN message (22) to the ECF/EFF to receive the multicast traffic of the BC session (23) after completing the new SIP session establishment between the UE2 and the Mobility Manager. Finally, the UE 1 leaves the multicast group of the BC session by sending an IGMP LEAVE message to the ECF/EFF (28) once the session has been transferred to the UE2.

3.2.2.4 Terminal Mobility

Terminal mobility procedures for LiveTV service are shown in figure 8. The figure represents a scenario where the UE1 performs a handover between two access networks. It is assumed that the LiveTV session is already established on UE1. Again, the SIP signalling is similar to the VoD service case (see previous explanation of SIP signalling for VoD service terminal mobility for more details). The difference between the VoD service case and LiveTV service case is that the Mobility Manager does not modify the SDP payload of SIP messages because the data traffic of the BC session is sent by multicast and it is not necessary to make it travel through the Mobility Transparency Manager. Moreover, the UE1 joins to the multicast group of the BC session in the new access network and leaves it in the old access network by sending the appropriate IGMP messages (23-24) to the ECF/EFF to receive the multicast traffic of the BC session.

4. DEPLOYMENT PLATFORM

In this section we present the implementation of the UP-TO-US mobility solution. We have deployed a testbed (see figure 9) with all the components of the architecture already presented: an IMS Core, a Mobility Manager Application Server which controls the Signalling Plane, a Transparency Manager that controls the Media Plane, the CAS with only simplified functions to perform mobility, an IPTV Server, and the user clients.

We have used the Fokus Open IMS Core[4] as our IMS platform; this is a well-known open source implementation. In our testbed, all the components of the OpenIMS core have been installed in a ready-to-use virtual machine image.

We have provisioned two public identities in the HSS, UE1 and UE2, which represent two terminals of a user. We have also configured the Mobility Manager as an Application Server in the Core, and when the user calls the IPTV server from one of his terminals the corresponding INVITE message is routed to our Mobility Manager AS, so mobility support can be provided.

The Mobility Manager has been developed as an application deployed in a Mobicents JAIN SLEE Application Server[5]. Mobicents JAIN SLEE is the first and only open source implementation of the JAIN SLEE 1.1 specification[6]. With Mobicents JAIN SLEE we have a high throughput, low latency event processing application environment, ideal to meet the stringent requirements of communication applications, such as network signalling applications, which is exactly our case.

The Transparency Manager is a standalone Java application, which offers an RMI Interface to the Mobility Manager and uses Java sockets to redirect media flows.

In order to simulate the user terminals and also the IPTV Server, we have used SIPp[7] scripts. SIPp is an open source test tool and traffic generator for the SIP protocol, which can also send media traffic, and is capable of executing shell commands.

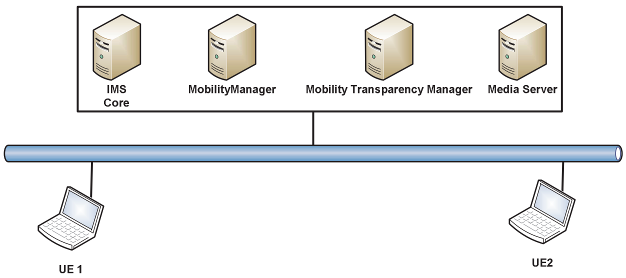

Figure 9. Testbed

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[4] Open IMS Core, Fraunhofer Institute for Open Communication Systems (http://www.openimscore.org/).

[5] http://www.mobicents.org/slee/intro.html

[6] http://www.jcp.org/en/jsr/detail?id=240

[7] http://sipp.sourceforge.net/

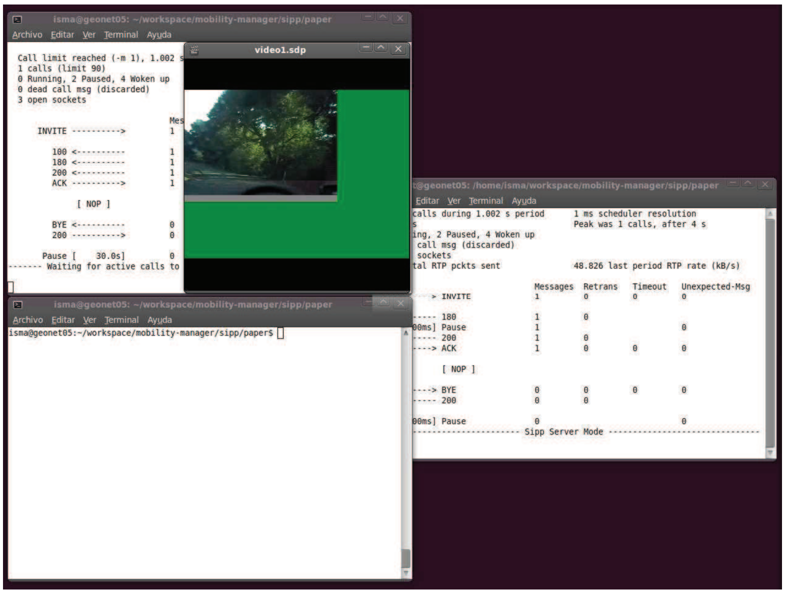

Figure 10. Pull session transfer: Initial snapshot

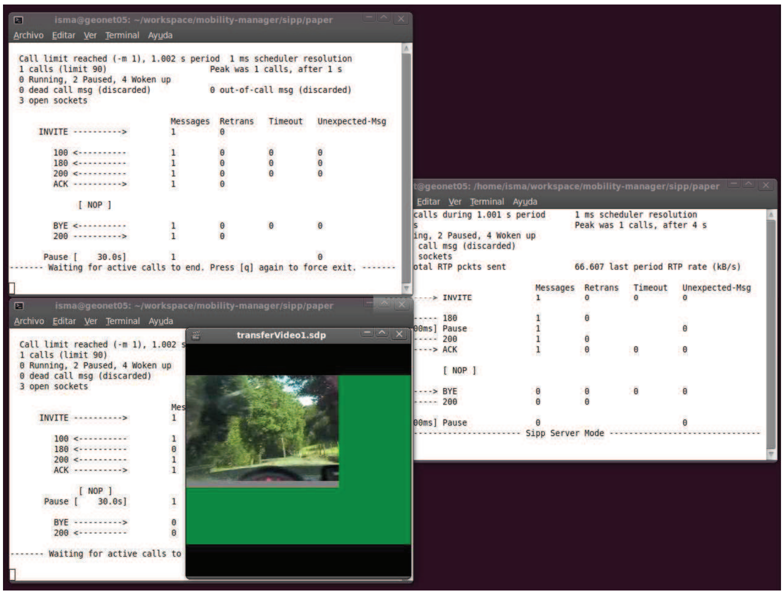

Figure 11. Pull session transfer: Final snapshot

The IPTV Server is a SIPp script which receives and accepts SIP session requests, and uses the ability of sending media traffic to deliver content to the clients.

Regarding the clients, we have developed several SIPp scripts which show the SIP flows, and call shell commands to reproduce the content being received. For that last purpose it uses Totem, the official movie player of the Gnome Desktop Environment.

The client also writes in a file the necessary information to perform a pull session transfer. This is a simple way in which we simulate the CAS.

Our test platform is formed by three Linux Ubuntu computers. One of them runs the Mobicents JAIN SLEE with the Mobility Manager application deployed, the Transparency Manager, the SIPp scripts which emulate the IPTV Server and the IMS Core (installed in a virtual machine). The other two Linux Ubuntu computers are the clients UE1 and UE2.

We have tested pull session mobility and push session mobility in our testbed. In figures 10 and 11, which directly represent two snapshots taken in our test platform, we can see the pull session mobility process. In figure 10 we can see a SIPp script representing the first terminal of the user calling the IPTV server (which is also a SIPp script). The IPTV server is sending the content that is being played using Totem.

In figure 11, we can see another SIPp script representing the second terminal of the user, which initiates the pull session mobility. The session with the first terminal is closed, and the content is now being played in the second terminal.

In order to do preliminary tests about the mobility performance, we have taken two different measurements, one for pull session mobility and the other one for push session mobility. For pull session mobility, we measure the time it takes since the INVITE message with Replaces header is sent till the first content packet is delivered to the new terminal. We repeated the measurement 30 times obtaining an average pull session mobility delay of 58 ms.

For push session mobility, we measure the time it takes since the first terminal sends the REFER message till the first content packet is delivered to the second terminal. We repeated the measure 30 times obtaining an average push session mobility delay of 73 ms, a bigger value than the pull session mobility case. This is logical because the procedure is more complex.

5. CONCLUSION

In this paper we propose a SIP-based mobility support solution for IMS services. The solution makes mobility transparent to service providers, it is integrated with the IMS infrastructure, it does not require modifications to the IMS specifications, and it works with unicast and multicast services. The solution supports both terminal mobility, a terminal that changes access network; and session mobility, a communication moved from one terminal to another. Context-awareness is used in the solution, through a context server, to enhance users' mobility experience: facilitating typical operations when a particular situation is recognized, highlighting to the user mobility-related options that are relevant in those situations, and providing information useful to take decisions regarding mobility.

We have implemented our mobility support solution using standard tools and platforms such as the Fokus Open IMS Core as IMS Platform and the Mobicents application server as the SIP application server. We have made preliminary trials showing that our solution works as intended supporting terminal and session mobility with standard servers that are not affected by the mobility of terminals communicating with them.

As future work we intend to do extensive testing of the implementation, also considering the analysis of performance metrics, for example related with terminal handover delays.

6. ACKNOWLEDGMENTS

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