Supporting mobility in an IMS-based P2P IPTV service: a Proactive Context Transfer mechanism

Ivan Vidal[a,*], Jaime Garcia-Reinoso[a], Antonio de la Oliva[a], Alex Bikfalvi[b], Ignacio Soto[a]

[a] *Universidad Carlos III de Madrid. Avda. de la Universidad 30 28911 Leganés - Madrid (Spain)*

[b] *IMDEA Networks, Avda. del Mar Mediterraneo 22, 28918, Leganés - Madrid (Spain)*

Abstract

In recent years, IPTV has received an increasing amount of interest from the industry, commercial providers and the research community, alike. In this context, standardization bodies, such as ETSI and ITU-T, are specifying the architecture of IPTV systems based on IP multicast. An interesting alternative to support the IPTV service delivery relies on the peer-to-peer (P2P) paradigm to distribute and push the streaming effort towards the network edge. However, while P2P IPTV was studied in fixed access technologies, there has been little attention paid to the implications arising in mobile environments. One of these involves the service handover when the user moves to a different network. By analyzing previous work from the perspective of an IPTV service, we concluded that a proactive approach is necessary for the handling of inter-network handovers. In this paper, we propose a new general handover mechanism for the IP Multimedia Subsystem (IMS), while studying its applicability to a P2P IPTV service. Our solution, called Proactive Context Transfer Service, incorporates the existing IEEE 802.21 technology in order to minimize the handover delay. The proposal is validated by comparing it against solutions derived from previous work.

Keywords: IP Multimedia Subsystem (IMS), Peer-to-Peer (P2P), Mobile IP (MIP), Television over IP (IPTV), IEEE 802.21

*Corresponding author

Email addresses: ividal@it.uc3m.es (Ivan Vidal), jgr@it.uc3m.es (Jaime Garcia-Reinoso), aoliva@it.uc3m.es (Antonio de la Oliva), alex.bikfalvi@imdea.org (Alex Bikfalvi), isoto@it.uc3m.es (Ignacio Soto)

1. Introduction

Nowadays there are several initiatives and research studies proposing a general architecture to stream TV using the TCP/IP protocols (IPTV) ([1, 2]). As the applicability of IP multicast in the current Internet presents several challenges [3], such as address management, security, support of heterogeneous receivers and charging, other proposals are gaining support for content distribution to multiple users. One of these alternatives is the use of Peer-to-Peer (P2P) techniques to distribute content, by means of an overlay network formed by end user equipment [4]. One of the challenges with this technique in the current Internet is how to distribute a real-time flow over a best-effort network: although there are very efficient ways to construct the overlay and to exchange packets between end nodes, packets can be delayed or lost.

In order to guarantee an acceptable Quality of Service (QoS), some improvements are necessary to the Internet. In this context, standardization bodies like 3GPP and ETSI (TISPAN working group) are working to propose a complete architectural framework for Next Generation Networks (NGNs), capable of providing QoS to end-users. NGNs are centered around the IP Multimedia Subsystem (IMS), which provides access and session control for all-IP services. Initially developed by 3GPP for UMTS cellular networks, currently 3GPP is working together with ETSI-TISPAN to extend the specification for any type of access technology.

In [5], the authors propose a way to deploy P2P IPTV streaming using IMS as the core of the system, providing mechanisms to join, switch between channels and leave the system with similar delays as in the IP multicast counterpart. A brief introduction to the IMS and this P2P IPTV system is included in Sect. 2. In this proposal all peers are connected to a fixed access network, therefore we decided *to explore the impact of user mobility in the previously proposed system.* There are two problems when using mobile devices as peers in a P2P IPTV system: (1) limited uplink bandwidth and (2) packets losses while switching between different networks (changing the IP address).

Section 3 analyses how to enable mobility for the P2P IPTV service, using current proposals for handover across IMS networks, and proposes a buffering mechanism to avoid packet losses. In Sect. 4 we propose a mechanism to minimize the handover delay by transferring in advance, before the User Equipment (UE) moves, the IMS context between the serving and target networks, hence reducing the number of operations performed during the handover. To this end, we use the IMS infrastructure and we introduce IEEE 802.21 signalling, minimizing the changes to current specifications. Although this is a general proposal that can be used for any multimedia service, in order to estimate the desired delays, this section particularizes how this architecture can be used in a P2P IPTV scenario. Because the delay imposed by a mobility solution is critical for an IPTV service, Sect. 5 is focused on its analytical measurement for the solutions presented in Sect. 3 and Sect. 4. Section 6 concludes with the most remarkable points discussed in the paper, and presents the future work.

2. Background on the IMS-based P2P IPTV service

This section presents a brief description of the architecture, technologies and protocols used in the IP Multimedia Subsystem (IMS). In addition, we provide an overview of the P2P IPTV service [5] built on top of the IMS.

2.1. The IP Multimedia Subsystem

Nowadays, the Internet and the cellular networks evolve towards convergence, integrating a broad set of services that are delivered to the end user by means of the IP protocol. In this context, the IP Multimedia Subsystem (IMS) is currently being developed by the 3GPP as a key element to facilitate the convergence. The IMS is a control architecture, based on the IP protocol, that enables the provision of value-added multimedia services by supporting a set of facilities related with session control, QoS, charging and integration of services. Figure 1 shows a simplified overview of the IMS architecture (further details can be found in [6]), where a User Equipment (UE) gets network connectivity by means of a UMTS access[1], consisting of a UMTS terrestrial radio access network and the UMTS packet domain.

As it can be observed from the figure, the architecture follows a layered approach, where three planes have been defined: *the control plane*, *the user plane* and the *application plane*. This organization allows to separate the transport technologies and bearer services, utilized at the transport plane,

[1]Note that IMS can also be used with other access network technologies.

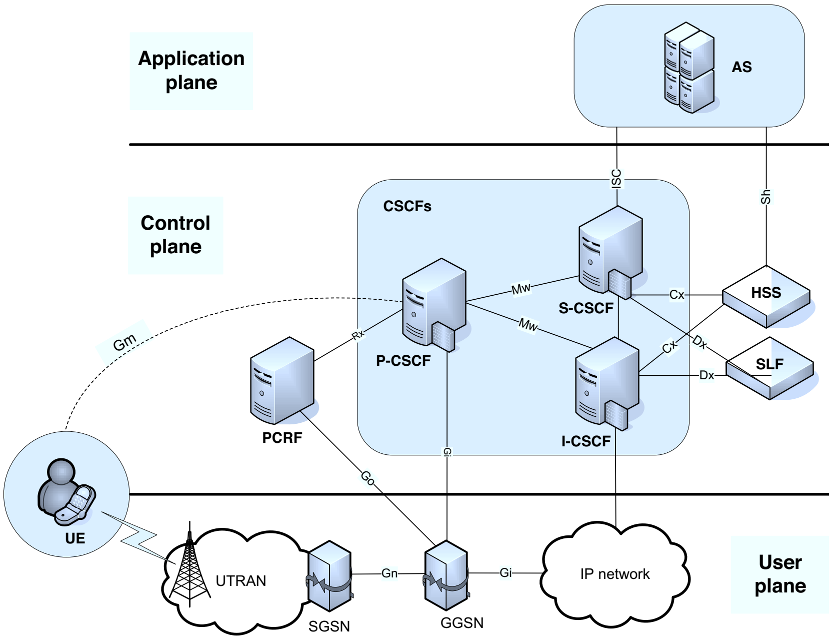

Figure 1: IMS architecture

from the session management functionalities that correspond to the control plane. On top of the control plane, the application plane implements the services that are accessed by the end user. These services are provided with a set of common functionalities from the control plane, and can be delivered to the end user independently from the network access technology utilized in the user plane. In this layered architecture, the Session Initiation Protocol (SIP) [7] plays a crucial role, being the protocol chosen in the IMS for session control functionalities. In addition to SIP, other protocols are specially relevant in the IMS such as Diameter [8], that is utilized to provide AAA (Authentication, Authorization and Accounting) functionalities.

On the other hand, the different functions implemented at each plane are organized in a set of functional entities, which are interconnected by standardized reference points. Focusing on the control plane, the following entities are specially relevant: the Call Session Control Functions (CSCFs), which are in charge of processing all the SIP signaling messages originating or terminating at the UE; the user databases, namely the Home Subscriber Server (HSS) and the Subscriber Location functions (SLF); and the Policy Control and Charging Rules Function (PCRF), which provides policy control decision and flow-based charging control functionalities. The IMS architecture defines three types of CSCFs: the Proxy-CSCF (P-CSCF), the Interrogating-CSCF (I-CSCF) and the Serving-CSCF (S-CSCF). The P-CSCF is the entry/exit point into the IMS control plane for every SIP signalling message originating/terminating at the UE. The I-CSCF is the entry point to the network of an operator for the incoming sessions destined to the operator subscribers. The Serving-CSCF (S-CSCF) performs session control and registration functionalities. This functional entity checks the service profile of the user and verifies whether a given SIP signalling message should be routed to one or more Application Servers (ASs), which provide services to the end user (e.g. IPTV) from the application plane.

In the user plane, Fig. 1 shows the UMTS Terrestrial Radio Access Network (UTRAN) and the UMTS packet domain. The latter includes the Serving GPRS Support Node (SGSN), that links the radio access network with the packet core network, and the Gateway GPRS Support Node (GGSN), that internetworks with external networks and provides the UE with IP-level connectivity by means of PDP contexts. A Packet Data Protocol (PDP) context is a QoS enabled logical connection that supports the exchange of IP packets between the UE and the GGSN.

2.2. P2P streaming in IMS

This section presents the architecture proposed in [5]: a peer-to-peer IPTV service in a fixed IMS scenario, where the video streaming is done using Application Level Multicast (ALM) trees (one tree corresponds to one TV channel). A tree is built between the video server (or IPTV head-end) and one or more UEs, which, in turn, can be used as a normal server to distribute the video stream to other UEs. The overall architecture is presented in Fig. 2.

The weakness point in ALM is the disruption of the service when a parent node (a node serving video to other peers) leaves the tree: all branches below the leaving node will be affected until the tree is completely reconstructed. A node leaves a tree when tuning to a different channel or when leaving the system. The former is a controlled action and the system can react trying to minimize packet losses. When switching between channels there are two parameters that have to be taken into account: packet losses for the orphan children and the channel switching time for the leaving node. In case of an ordered or a non-ordered leaving the only thing that has to be considered is the number of lost packets.

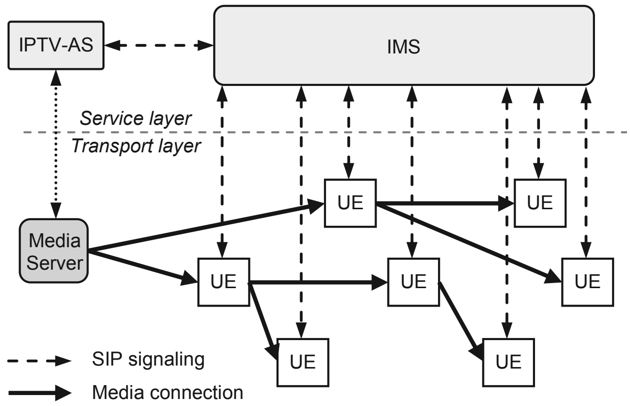

Figure 2: P2P IPTV service architecture

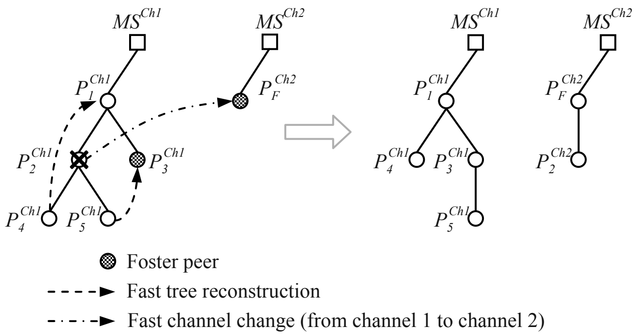

Figure 3: Fast channel change

As reconnecting to the tree through the IMS implies a long delay, [5] introduces the concept of *foster peers*, which are peers with pre-reserved resources that can quickly accept orphan children or peers switching to their channels. In case of a channel change, orphan peers will be reconnected to foster peers in the same tree and then the leaving peer will be associated with a foster peer in the desired channel (see Fig. 3). In case of an ordered leaving, orphan peers will be rehoused in the same way as in the switching channel process and then the leaving node will be authorized to disconnect. For a non-ordered leaving, the process is the same but it will be triggered by a timer or a counter for packet losses at the orphan nodes instead of by the leaving node.

Another key entity introduced in [5] is the *IPTV Application Server* or *IPTV AS*. The IPTV AS is a SIP Application Server, used as a Back-to-Back

User Agent[2] (B2BUA) between two UEs or between a Media Server and a UE. The SIP signalling involved in the ALM construction and maintenance passes through the IPTV AS. The IPTV AS receives SIP messages in order to tune to a channel, switch between channels and leave the system. With all this information, the IPTV AS decides how to construct or modify all trees in the system, adding and removing foster peers when necessary.

Although the main ideas described in [5] can be used for fixed or mobile UEs, there are two problems that must be addressed when the UE is a mobile device:

- In a wireless environment, the access bandwidth is a scarce resource. In general, the uplink should not be used by the UE in order to minimize costs and improve the resource usage.
- If a mobile UE changes its IP address while receiving a stream, packets will be lost. The number of lost packets depends on the delay introduced during the handover.

The first issue can be easily addressed at the IPTV AS. As this is the functional entity in charge of setting up the distribution tree, it can guarantee that a mobile UE never assumes the role of a parent peer in the tree. Regarding the second issue, it is possible to classify handovers in two groups: *soft handovers* (make-before-break) and *hard handovers* (break-before-make). While in *soft handover* the same data is delivered to the mobile device through two access networks simultaneously, in *hard handover* data is always received in one and only in one interface at any time. Although there are some studies using *soft handover* mechanisms like in [9], our goal is to propose a more general mechanism valid for any kind of mobile devices. Therefore, next sections will focus on minimizing the number of lost packets using a *hard handover* approach.

3. Enabling seamless mobility in the P2P IPTV service

As it has been indicated in the previous section, enabling mobility in the P2P IPTV service entails packet losses in the media plane. As long

[2]A Back-to-Back User Agent (B2BUA) is a concatenation of two SIP User Agents connected by some application-specific logic (see [7]). A B2BUA receives SIP requests, and to determine how each request should be responded it can generate further requests.

bursts of packet losses can negatively influence the end user experience, some mechanisms are needed to address this issue. In this respect, we propose to combine the following two approaches:

- Introducing buffering techniques at the parent peer (if delay is still significant).
- Minimizing the handover delay of the mobile UE.

In the following we cover each of these two approaches, buffering techniques are analyzed in section 3.1 while section 3.2 analyses the handover alternatives that have been proposed in the literature for IMS enabled networks.

3.1. Buffering packets when roaming in IMS

Video streaming is a real-time application, sensitive to jitter and long bursts of lost packets. The jitter problem can be mitigated using buffering at the client side, introducing a delay in the play-back process, which can be tolerated by the user if it is not too high. Low packet losses can also be tolerated by the user. The problem arises when a high number of packets are lost due to mobility. In order to minimize packet losses, we propose to send a *pause* to the parent of the moving UE before it starts the handover. After the UE is stable in the new access network, it sends a *resume* to its parent which, in turn, resumes sending the buffered packets at the maximum peak rate. Appropriate buffering at the mobile node side makes the procedure transparent to the users.

Let us denote the delay introduced during the handover process by d, and the bit rate of the video (assuming a Constant Bit Rate or CBR) by R. Then, it is possible to express the total number of bits buffered at the parent while its child is in the handover process as $b = d \times R$. When the mobile node finishes the handover by sending a *resume* message, the parent will restart sending all buffered packets and all incoming packets during the *recovery* phase. In our approach, the parent will send packets at the video bit rate (R) and an additional bandwidth (R_{add}) to recover the original buffer size at the mobile node (the total bandwidth during the recovery phase is then $R + R_{add}$). The delay d_r necessary to recover the stored buffer at the mobile node is $d_r = b/R_{add}$. With the former and the latter equalities, we can obtain Eq. 1 that will be used in Sect. 5 in order to obtain representative recovery delays.

$$d_r = \frac{d \times R}{R_{add}} \tag{1}$$

The handover delay depends on the specific scenario used for mobility in IMS. These scenarios will be described in the next subsection and the delay introduced for each scenario will be analyzed in Sect. 5.

3.2. Alternatives for UE mobility in an IMS-based IPTV service

There has been a huge effort to improve mobility management in IP networks. Standardization bodies such as the Internet Engineering Task Force (IETF) have developed solutions for mobility support in IPv4 [10] and IPv6 [11], for improved IP handover performance [12, 13], and for supporting the movement of networks as a whole [14]. Researchers have also addressed these topics with different points of view ([15, 16, 17, 18, 19, 20]). However, macro-mobility (changing network and IP address) in IMS-based networks is very difficult to achieve with existing standards [21], although this type of mobility is expected to become very common. The challenges of integrating mobility, IMS, and control of the access network have been analyzed in [22] (WLAN and cdma2000 access) and in [23] (GPRS access).

This section presents three existing handover mechanisms for IMS-based services, particularizing to the P2P IPTV service and to the use of the buffering mechanism defined in Sect. 3.1. From now on we assume that the UE is connected through a GPRS access network[3], but the studied mechanisms are applicable to any IMS based network and access technology, except that the procedure for reserving resources in the access network would be different.

Next we introduce several general notions belonging to the IMS and GPRS technologies. In a general mobility scenario, the UE can be located either in the home or in the visited network. The home network maintains the user subscription data and provides services. In general, we assume that the UE moves from an *old* (or *current*) to a *new* (or *target*) network, where both the old and the new network can be the home or a visited network. In addition, the entry-point for the IMS, the Proxy Call Session Control Function (P-CSCF), can be located either in the home or in the visited network,

[3]We use the term GPRS access network to refer to the packet domain of UMTS networks plus the radio access network

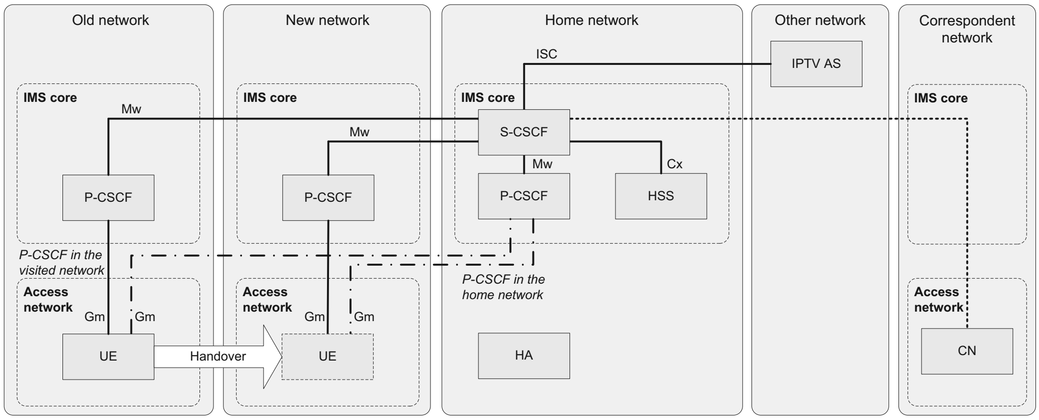

Figure 4: Mobility scenario: UE moves from an old to a new visited network, with the P-CSCF located in the visited network

depending on how its discovery is configured in the UE and in the visited network (Fig. 4).

3.2.1. SIP mobility

This proposal uses the existing IMS infrastructure and specification to restore the multimedia session after the user moves to a new network. For the sake of simplicity, for this scenario, we study only the case where the P-CSCF is located in the visited network and a handover implies a change of the P-CSCF assigned to the UE. The situation where the P-CSCF is located in the home network can be derived as a particular case.

Figure 5 illustrates the IMS signalling flow for the network handover. When the UE detects that it is about to lose connectivity to the current network (through link layer triggers, IEEE 802.21 [24] or technology specific mechanisms), it sends a pause notification (a SIP *NOTIFY* request) that will eventually arrive at the Correspondent Node[4] (CN). At this point, the CN will begin buffering a limited duration of the video stream.

For a GPRS access technology, the UE begins the handover process by deleting all bearer PDP contexts detaching from the old network, attaching to

[4]Correspondent Node is a Mobile IP ([11, 10]) term that refers to a peer communicating with the mobile node. We adopt this term in the rest of the paper, even when the mobile node is not using Mobile IP. In our case, the correspondent node can be a parent peer or a media server.

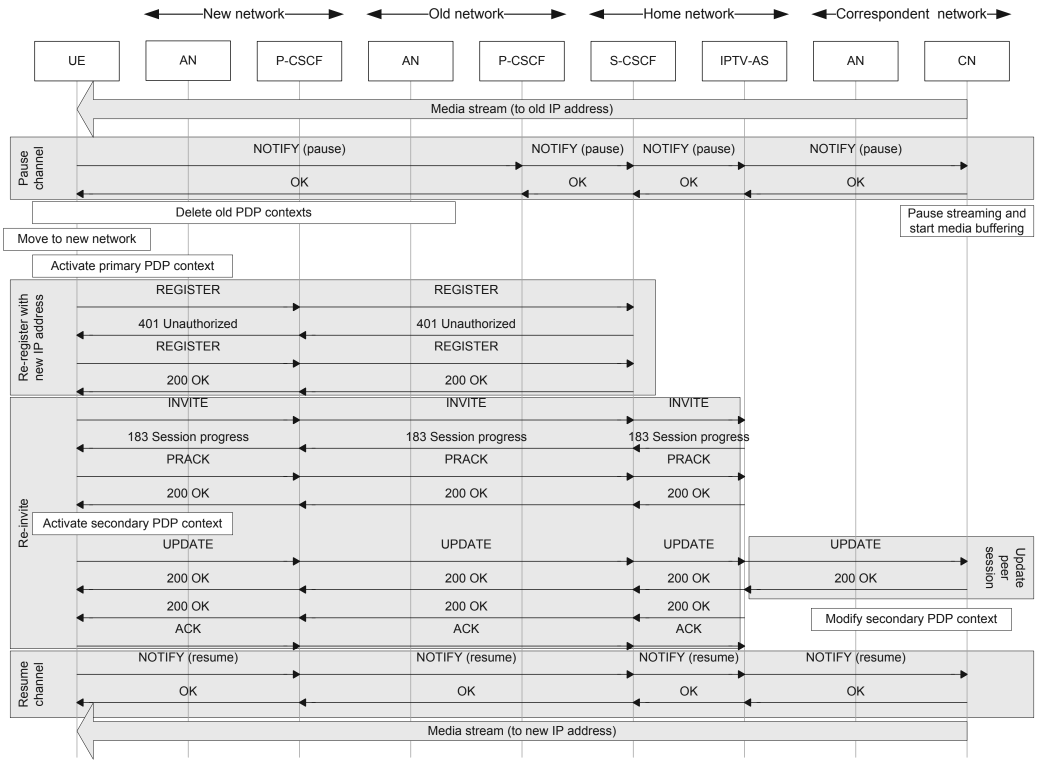

Figure 5: SIP mobility with the P-CSCF located in the visited network

the new network and establishing a primary PDP context for IMS signalling. With the activation of a new PDP context, the UE obtains a new IP address and discovers a P-CSCF in the new network. Afterwards, the UE initiates a regular IMS re-registration procedure, to inform the S-CSCF of the new contact address and P-CSCF address. This procedure is completed in the usual way after two round trips, each of them initiated by a REGISTER request. Further details on the IMS registration procedure can be found in [25].

Following a successful re-registration, the UE modifies the existing IPTV session by issuing a *re-INVITE* via the new P-CSCF. During the *re-INVITE*, the IPTV AS updates the contact URI of the UE, the URI of the new *P-CSCF* and the IP address and port where the media stream will be delivered. In addition, the UE performs a resource reservation by establishing a secondary PDP context. After the secondary PDP context is activated, the UE sends a SIP *UPDATE request* towards the IPTV AS that, in turn, arrives at the CN updating the destination IP address and port. Finally, the UE sends a *NOTIFY* that eventually arrives to the CN to resume the video streaming.

For the mobility scenario to work, the IPTV application at the UE must execute the signalling procedures described here and preserve the state of the existing SIP sessions during the handover (while network connectivity is lost).

3.2.2. Optimised SIP mobility

As an improvement over the previous mobility scenarios, several papers like [23, 26, 27] have proposed an optimization technique that transfers the context information between the old and new P-CSCF. The purpose of the P-CSCF context transfer is to reduce the handover latency by having all the parameters necessary to establish the signalling security associations and the bearer PDP context readily available at the new P-CSCF.

To make the context transfer possible, the IMS architecture has to be modified by changing the P-CSCF and adding a new reference point or interface between two P-CSCFs. Figure 6 shows the re-registration procedure that is executed after the activation of the primary PDP context in the new network.

The process of context transfer starts with the UE sending a re-registration message (1) to the new P-CSCF (containing the old P-CSCF information). This message must be integrity-protected because it involves a new P-CSCF. Because the integrity key is not known to the P-CSCF at this time, the P-

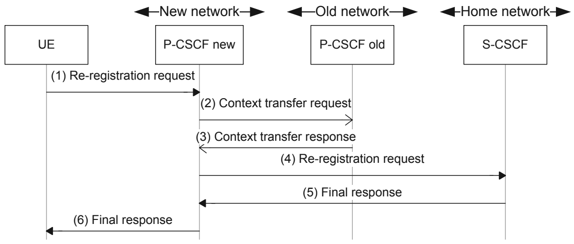

Figure 6: Re-registration with P-CSCF context transfer

CSCF must defer the verification of the message and UE authenticity until after the context transfer. After receiving the context transfer request, the new P-CSCF contacts the old P-CSCF in order to retrieve the UE context parameters including the encryption keys for the security associations, and the media parameters and filters of the previous sessions (2-3). If the context transfer is successful and the integrity of the re-registration request is verified, the S-CSCF is informed of the UE location change (4-5).

Due to the P-CSCF context transfer, the modification of the IPTV session at the UE requires only three message exchanges, an initial *re-INVITE* sent to the IPTV AS containing an SDP with the new media parameters at the UE-side, a final OK response and an ACK. Following the reception of the *re-INVITE* request, the new P-CSCF will create a new traffic filter with the media parameters from the transferred context and new UE. At the same time, the UE establishes a secondary PDP context within the new GPRS access network. In parallel, upon receiving the *INVITE* the IPTV AS will trigger a session update for the media parameters at the CN side. Finally, a notification to resume the streaming informs the CN that the handover process is completed. Figure 7 summarizes the signalling flow.

3.2.3. Mobile IP and IMS

Mobile IP (MIP) [10, 11] solves the mobility problem by allowing the UE to maintain network layer connectivity while moving to a visited network. With MIP, the mobile UE has two IP addresses, the Home Address (HoA) and the Care-of Address (CoA). In order to maintain communications, the mobile UE uses the HoA as a permanent address, making the mobility transparent for applications and the correspondent node. In order to deliver the IP packets to the UE, the UE also receives a temporary address, the CoA, which is topologically correct in the visited network. A Home Agent (HA) in the home network, the network where the HoA is topologically correct,

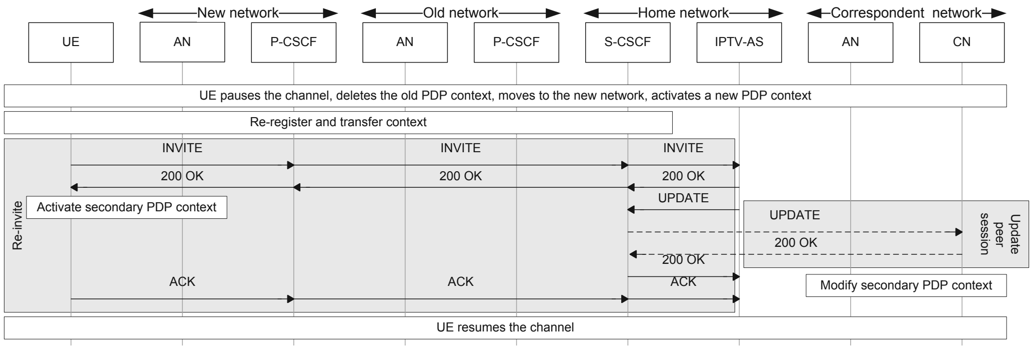

Figure 7: Optimised SIP mobility with P-CSCF context transfer

is responsible for sending and receiving the IP datagrams on behalf of the mobile node, by using a bidirectional IP in IP tunnel.

Figure 8 summarizes the operation of MIP version 6 in tunnel mode[5]. The first step is an authenticated binding procedure, through which the mobile UE informs the HA of its CoA. At the end of the binding procedure a MIP tunnel through the HA is established and a communication between the UE and a CN takes place as follows. The outbound IP datagrams (from the UE to the CN) are tunneled to the HA with the HA Address (HAA) as destination, which in turn will use the topologically correct HoA to forward the packets to the CN. The inbound IP datagrams, having the HoA as destination and arriving in the home network, are intercepted by the HA and sent through an IP tunnel to the UE using the CoA.

While MIP offers an attractive solution for seamless transition to mobile scenarios, its integration with the GPRS access technology and with IMS presents several challenges [23]. The allocation of network resources, including the provision of QoS, involves the creation of appropriate filters in the network gateways (in the GGSN for GPRS). These filters include the IP ad-

[5]A major difference between MIPv6 and MIPv4 is that the protocol used to set up the MIP tunnel is different. In addition, in MIPv4, due to the limited number of addresses and for mobility detection purposes, an entity called Foreign Agent (FA) is present in the visited network. The FA is usually the end of the MIP tunnel and its address can be used as the CoA for every mobile node within the visited network. Finally, MIPv6 includes an optional route optimization procedure that allows the sending of data traffic directly between the CN and the mobile node without going through the HA (not shown in Fig. 8).

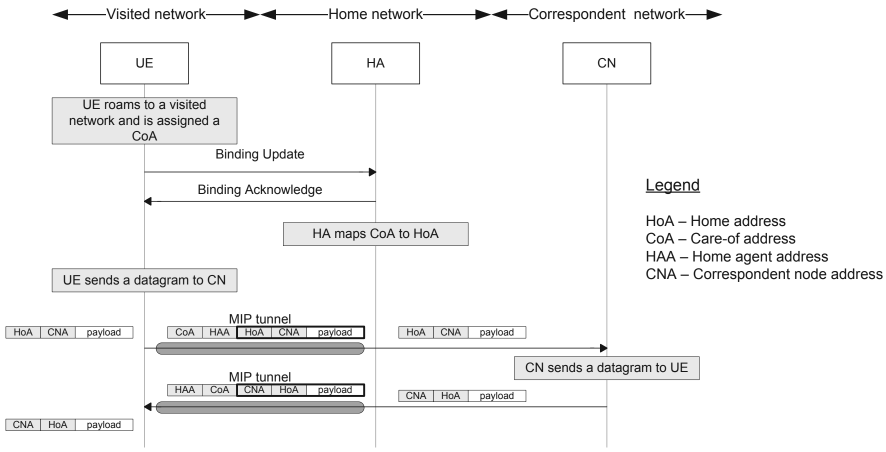

Figure 8: Mobile IP version 6

dresses of the nodes for which the communication is allowed and they are installed during the session establishment according to the Session Description Protocol (SDP) [28] body in the SIP messages.

Due to the tunneled communication between the UE and the HA, the use of MIP cannot be made transparent to the UE and to the IMS network. On the one hand, in MIP the HoA is the permanent address of the mobile UE, and therefore it should be used by the CN and the IMS core. On the other hand, the access network serving the mobile UE requires that resource reservation and filtering are done using the IP address assigned to the mobile in that access network (i.e. the CoA). As a consequence, using MIP in IMS requires changes in IMS functional entities to make them MIP aware as proposed in [23].

Nevertheless, the use of MIP could bring major benefits, since with MIP the mobility of the UE is transparent for CNs and applications as the UE is always addressed by the HoA. MIP could also bring additional advantages, depending on the network where the P-CSCF is located (home or visited), simplifying the IMS procedures after mobility. The different scenarios will be studied next, highlighting the mentioned benefits.

In Sect. 3.1 we proposed a buffering mechanism to be implemented at the parent node of the UE. Using MIP opens the possibility of implementing the buffering at the HA using the technique described in [29]. This has the advantage of not imposing CNs with the burden of buffering packets. In this way, buffering can be offered as a service to mobile users by operators.

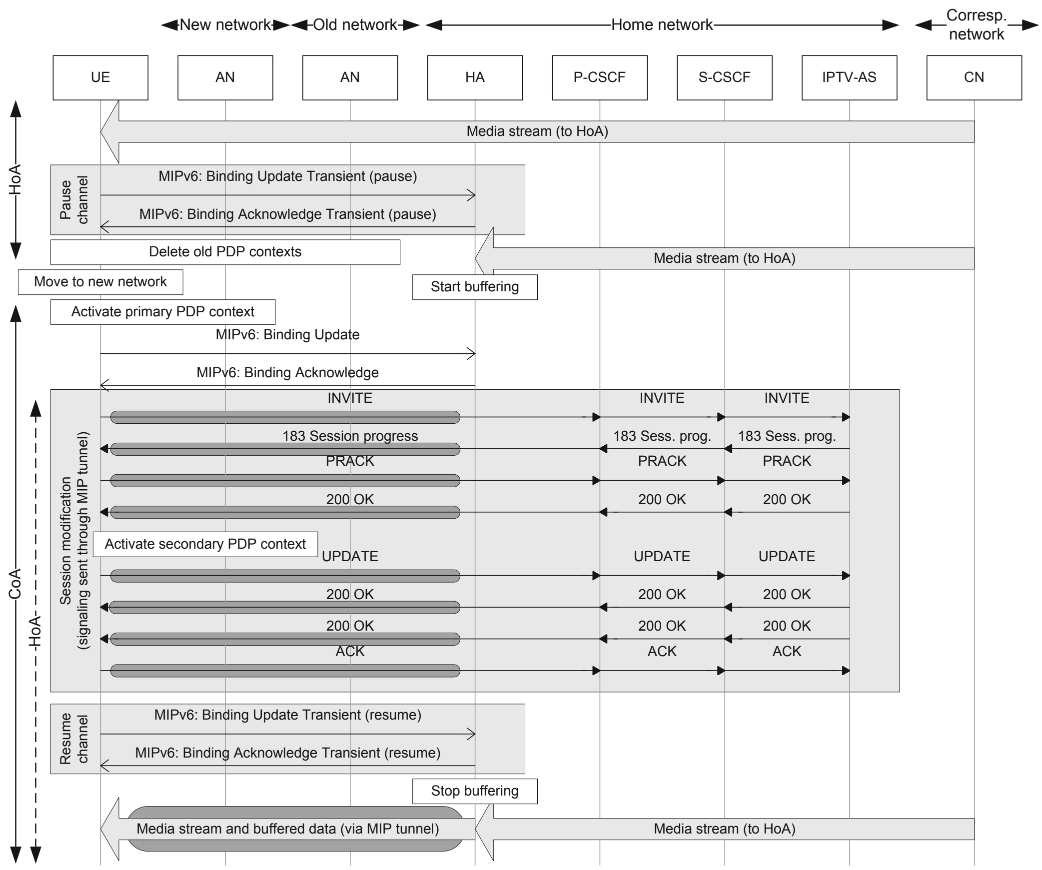

Figure 9: Handover procedure with MIP and P-CSCF in the home network

Buffering in the HA, as described, can be offered only if the MN does not use the Route Optimization feature of MIPv6, if the MN is using route optimization, then buffering can be done in the CN as described in previous section. In the rest of the section, we assume that route optimization is not activated.

Figure 9 illustrates a mobility scenario involving MIP and where the P-CSCF is located in the home network. The IMS signalling is tunneled with MIP via the HA and the UE does not have to re-register to IMS when moving to a visited network. A number of modifications are necessary in the UE, the IMS core and session description information in order to support this MIP scenario. When the UE is in the visited network, all IP datagrams are sent and received using the HoA as if the UE was located in the home network.

The MIP tunnel between the UE and HA must use the CoA and the HAA in order to honor the IP routing. For this reason, the QoS control in the visited access must consider that the CoA and the HAA are the endpoints of the media traffic with respect to the outer IP header.

During the session establishment or modification, the UE must inform the IMS of the use of MIP, and the appropriate IP addresses: HoA, CoA and HAA. Filters must be created and installed in the new access network with end-point addresses the CoA and the HAA (the outer header of the IP tunnel).

If the P-CSCF is located in the visited network, there are two possible approaches: either using MIP for both signalling and media (as if the P-CSCF was located in the home network, shown in Figure 9), or only for media. As in both cases a new P-CSCF is assigned to the UE, an IMS re-registration is needed in order to inform the S-CSCF about the URI of the new P-CSCF and to configure a security mechanism between the UE and that P-CSCF. However, using MIP for signalling (sending SIP messages using the HoA) has the disadvantage that the signalling goes through the MIP tunnel from the visited network to the HA in the home network, and from there it is normally routed back to the P-CSCF in the visited network adding delay to the total signalling round-trip-time.

The alternative, illustrated in Fig. 10, shows the handover procedure when the signalling follows the normal routing and only the media flow is using MIP. The UE uses the CoA for signalling, and therefore, after moving to the new network, it must re-register to IMS. In terms of session establishment and modification, the procedure remains the same. The UE must inform the IMS of the use of MIP and the following addresses: HoA, CoA and HAA. Filters must be created and installed in the new access network with end-point addresses the CoA and the HAA (the outer header of the IP tunnel).

4. Proactive context transfer service

This section describes an optimized solution for mobility, based on the transfer of context information from the current P-CSCF associated with the UE, to the P-CSCF assigned to the UE in the target network. Unlike previous proposals for context transfer (see Sect. 3.2.2), the mechanism detailed in this section performs the context transfer functionalities in advance, before the UE moves to the target network. This approach guarantees a minimum handover latency for the services that are being delivered to the end user.

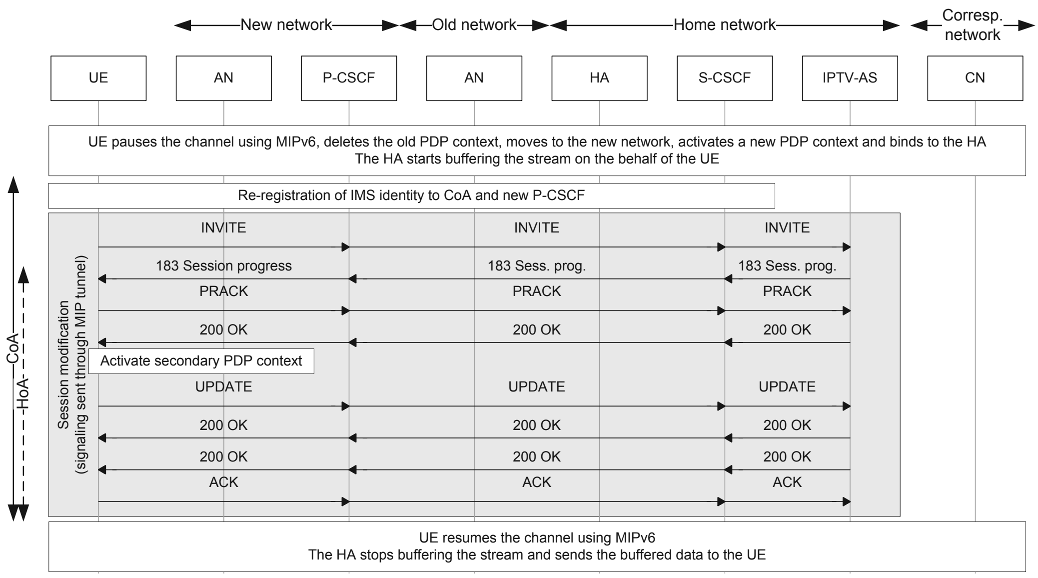

Figure 10: Handover procedure with MIP and P-CSCF in a visited network

The mechanism utilizes the IMS infrastructure, as defined by 3GPP, and IEEE 802.21 methods in order to configure the security mechanism that will be used for the communication between the UE and the new P-CSCF, and to transfer the QoS related information to the new P-CSCF for those services that are being accessed by the end user. Figure 11 depicts the architecture of the proposed solution.

The proposed procedure uses the handover enabling mechanisms of IEEE 802.21 [24]. IEEE 802.21 is a recent standard that aims at enhancing the user experience, by including mechanisms to improve the performance of handovers between heterogeneous technologies. This is done by introducing a set of primitives, which enable the communication between different network entities and the UE. We have chosen IEEE 802.21 to provide the communication mechanisms that enables the exchange of information required for the context transfer.

The Proactive Context Transfer Service (PCTS) described in this section will be supported by means of a SIP Application Server (AS). Any user subscribed to this service will be served by a PCTS AS located in its home network, which will be in charge of managing the mobility of the user and initiating the context transfer to a new P-CSCF located in the target network. The user subscription to the PCTS presents a per-service granularity,

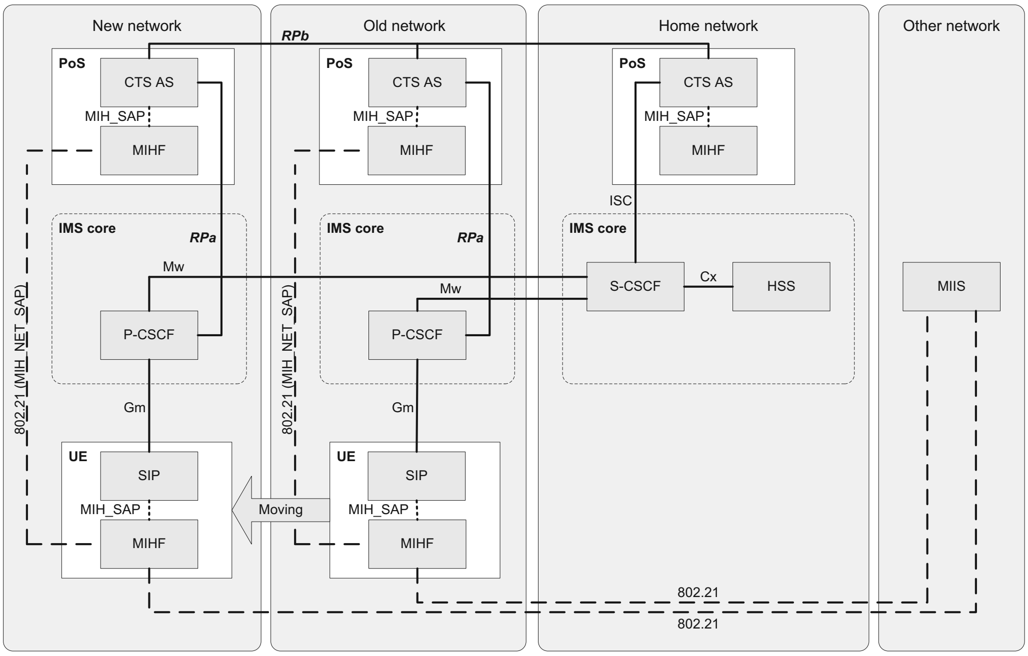

Figure 11: Architecture of the Context Transfer Service

i.e. the end user can subscribe to the PCTS for the subset of services that require reduced handover delays (e.g. an IPTV service). Context transfer functionalities may involve other PCTS ASs located in the current and target networks of the UE.

For context transfer purposes, the PCTS AS supports an intra-domain SIP interface towards the S-CSCF (the standard ISC reference point), a new intra-domain interface towards the P-CSCF (the reference point generically called RP_a in Fig. 11) and a new inter-domain interface enabling the communication with other PCTS ASs[6] (the reference point generically called RP_b in Fig. 11). The purpose of these interfaces will be further clarified in next subsections, which describe the procedures to initialize the PCTS, to perform a context transfer to a new P-CSCF and to route the SIP signalling messages to the new UE location after the context transfer.

[6]The definition of the interfaces towards the P-CSCF and PCTS ASs is out of the scope of this paper

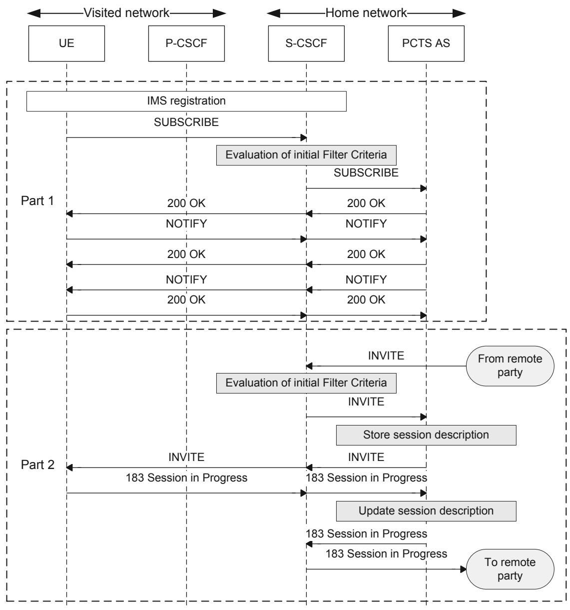

Figure 12: Initializing the Context Transfer Service

4.1. Initializing the context transfer service

Once that the user switches on its UE, and successfully finalizes the IMS registration, the UE starts the initialization process of the proactive Context Transfer Service. This procedure is illustrated in part 1 of Fig. 12 and starts with a SIP SUBSCRIBE request, sent from the UE and addressed to the Public Service Identity (PSI) of the PCTS. This request arrives to the S-CSCF that serves the end user in its home network, where initial filter criteria are evaluated. If the user has a subscription to the PCTS, then one of those criteria will indicate that this request ought to be forwarded to a SIP AS specific for the Context Transfer Service, i.e. the PCTS AS.

This SUBSCRIBE request subscribes the user to the state information related with the context transfer procedures. The PCTS AS answers back to the request with a SIP OK response, that eventually reaches the UE. The PCTS AS sends a NOTIFY request in response to the subscription, which in this case does not contain any context transfer information at this stage. On the other hand, the OK response establishes a SIP dialogue between the UE and the PCTS AS that will be used to route the subsequent SIP signalling messages between this two entities. The response implies an implicit subscription of the PCTS AS to the state information related with the mobility of the user. Therefore, a new NOTIFY request is sent from the UE to the PCTS AS, indicating that the user is in a stable state within the current network.

At this point, if the user is involved in the execution of any service, an INVITE request will eventually be received at its S-CSCF (either originating or terminating at the UE), where the set of filter criteria will be evaluated. If the user has contracted a Context Transfer Service for this specific service, then a criterion will indicate that the request should be routed to the PCTS AS. The PCTS AS will take the role of a SIP Back-to-Back User Agent, as defined in [7]. Assuming this role, the PCTS AS remains in the path of future SIP requests and responses exchanged in the dialogue corresponding to this INVITE request (see part 2 of Fig. 12). This way, the PCTS AS is always provided with updated information about the multimedia session associated with the SIP dialogue (this information is carried in the SDP payloads that are encapsulated in the SIP messages). Therefore, the PCTS AS always keeps the QoS related information for all those services the user has contracted the Context Transfer Service.

4.2. Transferring the context

Assuming that an UE needs to move from its current network (i.e. the serving network) to a new network (i.e. the target network), the UE can request from the PCTS the context transfer for all the subscribed services to a new P-CSCF in the target network. It is important to emphasize that a context transfer implies the following procedures at the PCTS:

- Obtaining the configuration parameters of the UE in the target network, e.g. the IP address and the URI of the P-CSCF that will be assigned to the UE. These steps are shown in Fig. 13.

- Transferring the QoS related information for all the subscribed services (i.e. the service information) to the P-CSCF in the target network. This procedure is shown in Fig. 14.

- Reconfiguration of the security mechanism used for the communication between the UE and the old P-CSCF, in order to be utilized in the target network. This procedure and the termination steps of the Context Transfer procedure are shown in Fig. 15.

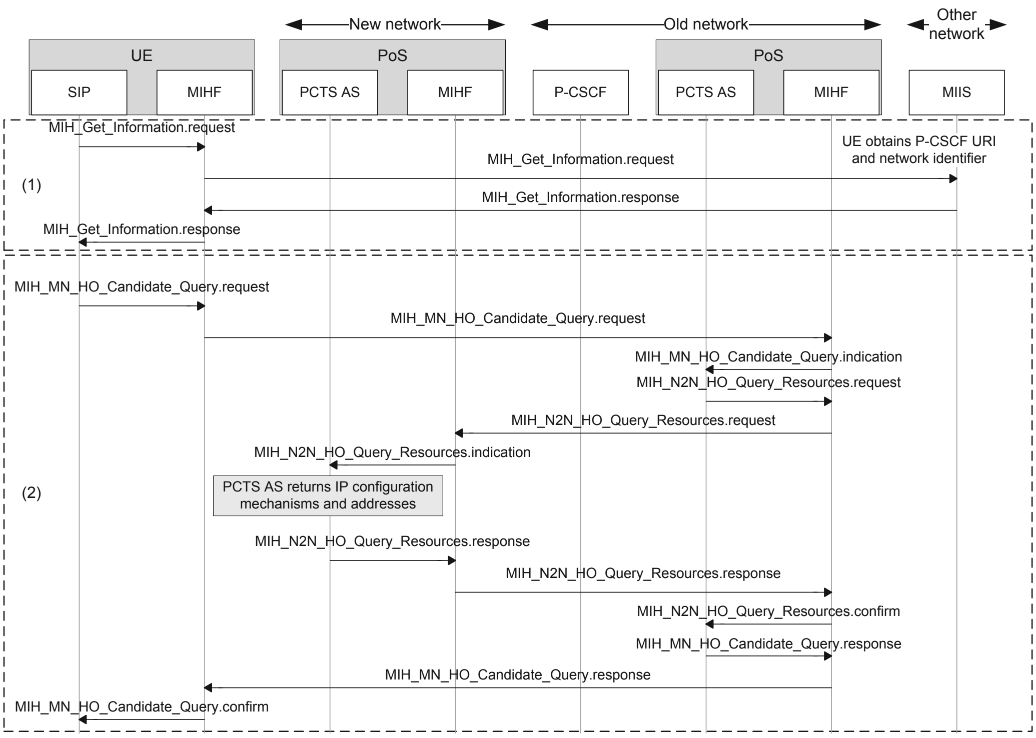

Figure 13: PCTS Phase 1 - Gathering information about the candidate networks

Next, we present the IMS and IEEE 802.21 procedures proposed to perform a context transfer to the new P-CSCF serving the UE in the target network. These procedures could be applied for both the SIP mobility and the MIP scenarios (a delay analysis for both use cases will be presented in Sects. 5.4.1 and 5.4.2). From the point of view of IEEE 802.21, the PCTS AS in each network plays the role of Media Independent Handover User (MIH User), managing the UE mobility. The MIH Users are the entities that use the services provided by the Media Independent Handover Function (MIHF). The PCTS AS controls the Point of Service (PoS) behavior using an MIHF to communicate with other IEEE 802.21 entities. The steps of the proactive context transfer procedure are shown in Figs. 13 to 15, and described in detail below. In these figures, it is assumed that an IPTV service is being delivered to a UE that eventually moves to a new network.

(1) The UE is connected to the serving network through its serving PoS and it has access to the MIH Information Service (MIIS) [30]. The UE queries information about its neighboring networks through the MIH_Get_Information primitive. Through the use of this primitive, the UE is able to contact the MIIS, which contains information regarding existing networks within a geographical area. The response of the MIIS contains among others, the P-CSCF URI and the Network Identifier of the different surrounding networks.

(2) Based on the information received, the UE triggers a mobile initiated handover by sending an MIH_MN_HO_Candidate_Query request message to the Serving PoS. This request contains the information of potential candidate networks, acquired in the previous step. The Serving PoS queries the availability of resources at the candidate networks by sending an MIH_N2N_HO_Query_Resources request message to one or multiple Candidate PoSs.

    The Candidate PoSs respond with an MIH_N2N_HO_Query_Resources response message and the Serving PoS notifies the Mobile Node of the resulting resource availability at the candidate networks through an MIH_MN_HO_Candidate_Query response message. During the query of resources, the serving PoS also obtains the relevant information for IP configuration in the candidate network; the parameters that can be obtained are, among others: the address of the Foreign Agent (in the case MIPv4 is used), the IP configuration methods used in the candidate network, the candidate network DHCP server address (if required) and the IPv6 address of the Access Router.

(3) At this point, the UE has obtained all the relevant information required to decide a target network for the handover and can proceed to perform a QoS context transfer. To this end, it sends a SIP NOTIFY request to the PCTS AS. This request indicates that the mobility state of the UE has changed and it has to move to any of the candidate networks. The message also includes all the acquired information regarding to these networks. Eventually, the PCTS AS receives the INVITE request, and answers it back with a SIP OK response. This AS will be in charge of managing the mobility of the UE and controlling the context transfer.

(4) Once the PCTS AS has received the information, it is able to take a decision regarding to which network the UE must be handed over. This decision can be taken based on the roaming agreements between the home and target networks and on certain user defined preferences (e.g. the user prefers WLAN to UMTS) that could be stored in a user profile. Once the target network has been selected, the PCTS AS retrieves the service information for all the multimedia sessions established within the scope of the subscribed services. This service information is updated to include the IP address that the UE will be assigned in the target network. Finally, the URI of the new P-CSCF and the updated service information is included in an authorization request that is sent to a PCTS AS in the target network.

    The PCTS AS in the target network examines the request and makes a policy decision, verifying if the QoS context can be installed in the P-CSCF indicated in the request. If so, the authorization request is sent to the new P-CSCF. At this point, the P-CSCF will behave as if the service information contained in the request had been derived from an SDP exchange in the target network, and will contact the policy and charging control system in IMS in order to authorize the service information. The outcome of the authorization request will be received back at the P-CSCF, and will be propagated back following the same path as the request towards the PCTS AS in the home network.

(5) Assuming that the QoS context transfer completes successfully, the PCTS AS sends a SIP NOTIFY request to the UE, informing it about

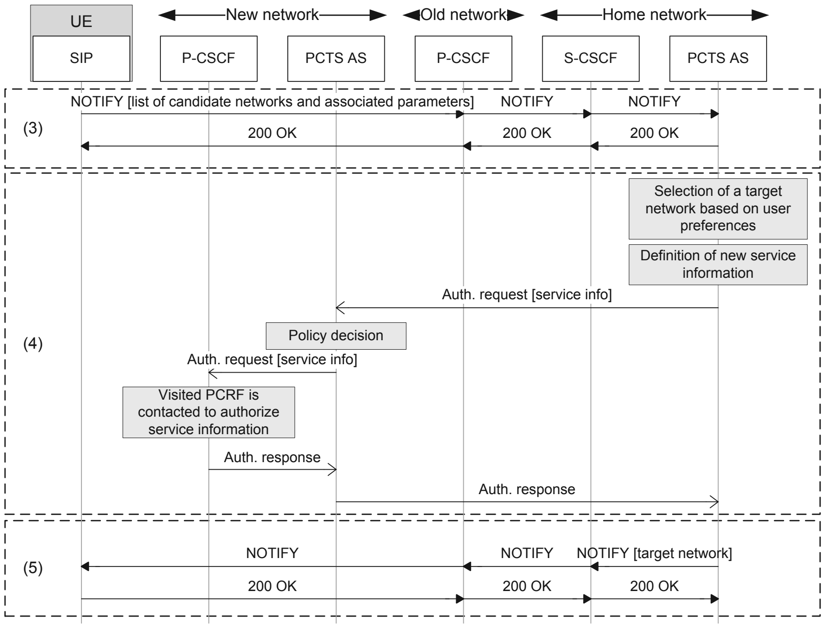

Figure 14: PCTS Phase 2 - Network selection procedure

the changes in its connection status, i.e. the target network that has been selected, the IP address and the URI of the P-CSCF that have been assigned to the UE in this network and an indication that the QoS transfer has been transferred successfully.

In order to improve the performance of the handover, a new parameter has been introduced in the MIH_N2N_HO_Commit primitive. This new parameter is able to carry the IPsec parameters of a connection, being used to transfer the security context from the Serving PoS to the Target PoS in such a way that the UE does not need to re-authenticate itself during the handover. The IPsec parameters transported in the primitive are the ciphering algorithm used by IPsec, the SPIs (Security Parameter Index) defining the security association, the ports used for the secure communication and the ciphering and integrity keys. This extension to the IEEE 802.21 standard is required to improve the performance of the handover, but the rest of the procedure follows the standard mechanisms.

(6) Upon receiving this indication from the PCTS AS, the UE notifies the information about the selected target network to the Serving PoS, by

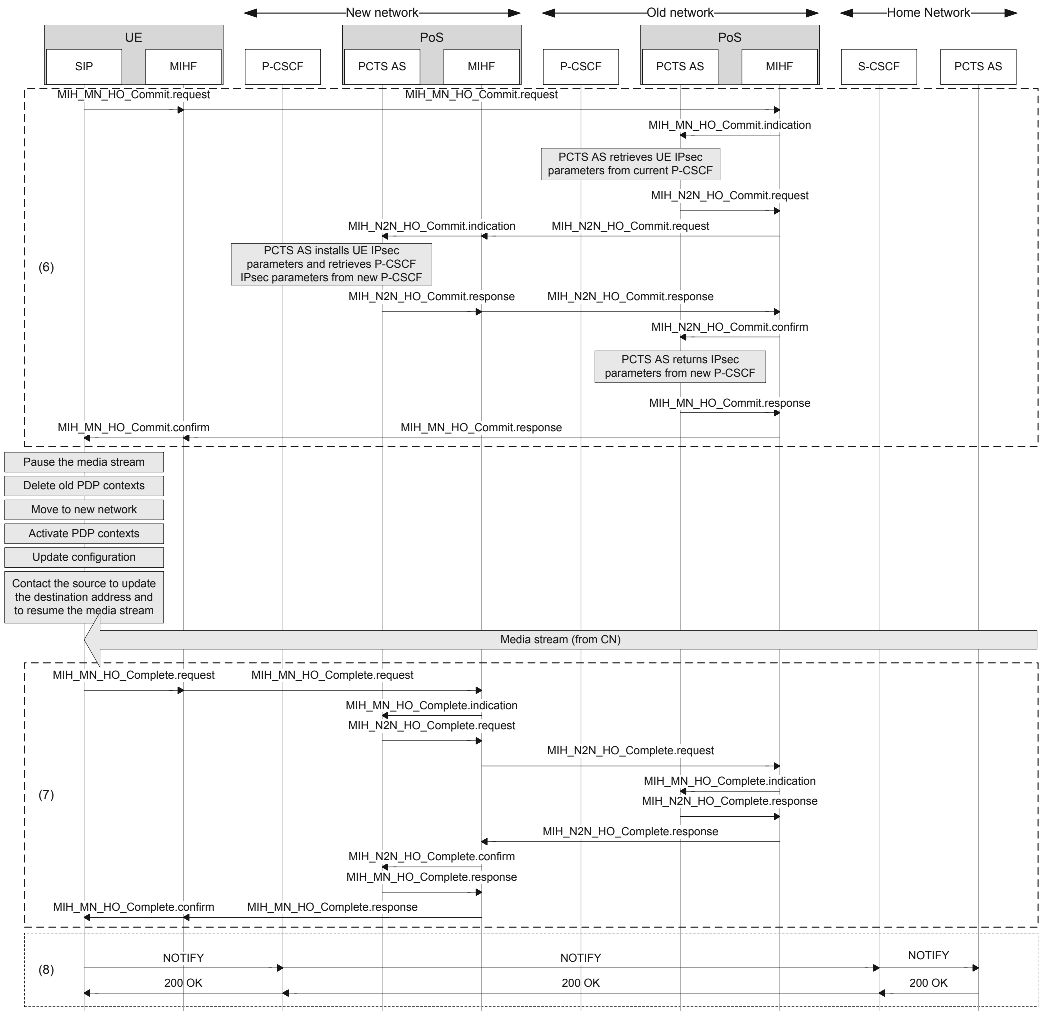

Figure 15: PCTS Phase 3 - Handover completion

sending the MIH_MN_HO_Commit request message. The Serving PoS then starts the signalling of the handover to the target network. This procedure will also result in the proper configuration of the security mechanism to be utilized between the UE and the new P-CSCF. For the purpose of the example illustrated in Fig. 15, an IPsec security mechanism has been considered.

Once the Serving PCTS AS (MIH User) receives the MIH_MN_HO_Commit indication message from the Serving PoS, it retrieves from the current P-CSCF the IPsec parameters that are being utilized by the UE (i.e. cryptographic algorithm, identifiers for the security associations, protected client and server ports, integrity key and encryption key). These parameters are provided to the target PoS through an MIH_N2N_HO_Commit request. This information is given to the target PCTS AS, which is able to install the IPsec parameters in the new P-CSCF, obtaining at the same time, a new set of IPSec parameters which must be set in the UE (i.e. cryptographic algorithm, identifiers for the security associations, client and server protected ports used by the new P-CSCF). Finally, these parameters are returned to the Serving PoS through the response of the MIH_N2N_HO_Commit primitive.

After the new IPSec parameters are returned to the UE through the MIH_MN_HO _Commit response primitive, the UE can start the handover. It is important to note, at this point, that there is the possibility that the P-CSCF does not support the security mechanism or algorithm that are used between the UE and the old P-CSCF. In this case, the UE would not receive any security related parameters in the MIH_MN_HO_Commit response primitive. In this way, although the QoS contexts would have been transferred, the UE would still need to register in the target network after moving, so as to establish a secure communication with the new P-CSCF.

Nevertheless, and before the handover, the UE performs the procedures that are necessary to pause the transmission of the media stream. This way, packet lost is prevented and the UE would be able to resume the media transmission after moving to the target network. At this point, the UE can start the handover process, which comprises the following steps: deleting the old PDP contexts, establishing a new layer 2 connection, activating the necessary PDP contexts for signalling and media and updating the UE configuration. The latter consists of updating the IPsec related information and the URI of the P-CSCF for future SIP requests. Next, the UE informs the source of the media stream about the new destination address, and the media is resumed.

The procedure to pause and resume a video stream is different for SIP mobility and MIP use cases. In the former, as described in Sect. 3.2.1, the procedure is based on SIP signalling and, in the latter, as described in Sect. 3.2.3, the procedure is based on MIP signalling.

(7) In order to finalize the handover procedure, the UE sends an MIH_MN_HO_Complete request message to the target PoS. The target PoS sends an MIH_N2N_ HO_Complete request message to the previous Serving PoS to release resources, which were allocated to the UE. After identifying that the resources are successfully released, the target PoS sends an MIH_MN_HO_Complete response message to the UE.

(8) At this point, the UE notifies to the Home Network PCTS AS the finalization of the handover and its new location. This message ends the Context Transfer procedure. From this moment on, the PCTS AS will route subsequent SIP requests within the scope of the subscribed services to the new UE contact address (the new contact address simply updates the IP address), through the new P-CSCF.

It is important to point out that, at this stage, the S-CSCF in the user home network has not been yet notified about the new contact address of the UE and the URI of the new P-CSCF that it has been assigned. Therefore, new INVITE requests for non-subscribed services, addressed to the public URI of the end user, would not be delivered from the S-CSCF to its UE. To address this issue, the UE needs to re-register with the S-CSCF. However, this registration process is not necessary from the perspective of the services that are already subscribed in the PCTS (i.e. the PCTS AS will be in charge of routing the SIP signalling corresponding to these services towards the UE), and consequently it can be performed after the context transfer procedure.

5. Delay analysis for mobile streaming in IMS

After presenting the previous proposals to perform a handover in IMS in Sect. 3, and our proposal in Sect. 4, this section presents an analytical study to obtain the mobility delay for each of them, in order to compare all

Parameter	Value (ms)
$T_{act,PDPp}$	2340
$T_{act,PDPs}$	1940
$T_{registration}$	1280
$T_{proc,CSCF}$	25

Table 1: Delay values used for the analysis

mechanisms in a scenario where the UE moves to a UMTS network. We are interested in analyzing the duration that the video is paused in the CN (or the HA in the MIP case), during the UE handover procedure, because this is the time during which the CN (HA) has to buffer the video packets on behalf of the UE. We denote this delay by T_{buff}.

The final results are presented in two ways: a general equation and an approximate value, in order to compare all mobility mechanisms. Some numerical input values for our analysis were extracted from [31], and are summarized in Table 1. These values are defined as follows:

- $T_{act,PDPp}$ is the time necessary to activate a primary PDP context.
- $T_{act,PDPs}$ is the time necessary to activate a secondary PDP context.
- $T_{registration}$ covers all signalling to register a terminal in the IMS network.
- $T_{proc,CSCF}$ is the time a SIP message needs to traverse a CSCF device (i.e P-CSCF, S-CSCF or I-CSCF).

The time to attach a mobile node to a UMTS network (T_{attach}) is not included in that work. Therefore, we experimentally evaluated realistic values of T_{attach} via simulation with the OPNET tool[7]. In order to do so, a network scenario for UMTS was used (the large UMTS scenario included in OPNET v15) comprising 58 UE, which arrive to the network at random times. Using this scenario, we performed 100 simulations and, for each of them, we obtained the average value of the time required for a UE to attach to the

[7]OPNET University Program,http://www.opnet.com/services/university/

UMTS network. The worst case of these average values was then used as T_{attach} in the following analysis, resulting in $T_{attach} = 1390\,\text{ms}$.

Another important value is the time necessary to delete a PDP context (both a primary and a secondary). In our proposal, we assume that the terminal just sends the request to delete all its PDP contexts and it leaves immediately the old access network; hence $T_{del,PDPp} = T_{del,PDPs} = 0$ (even if the terminal does not send this request, all PDP context are removed after a time-out). The remaining values used in this section are:

- T_{sip} is the delay for all SIP signalling involved in a given mechanism. This value includes other SIP related delays, as it will be seen in next subsections.
- $T_{sip,pause}$ is the delay for the SIP OK response sent by the CN after receiving the NOTIFY request with the *pause* indication.
- $T_{re-invite}$ gathers all delay involved in a re-invite message sequence.
- $T_{sip,resume}$ is the delay for the SIP NOTIFY request sent by the mobile node to the CN, requesting a *resume* of the paused video.
- T_{mip} is the delay introduced by all the MIP signalling.

The delay corresponding to any SIP message may be decomposed in five components: (1) the delay in the UMTS access network, (2) the processing delays at the traversed *CSCFs*, (3) the processing delay at the IPTV AS, (4) the delays within the core transport network and (5) the delay in the CN access network.

Regarding the SIP delays in the UMTS access network (1), a partial implementation of the IPTV service was developed in Java (version 1.5.0), utilizing the JAIN-SIP API[8]. Using this Java implementation, the real size of every SIP signalling message corresponding to the IPTV service was obtained. With these sizes, an experiment was designed to measure the average delay experienced by each SIP message in a real UMTS access network during the period of high load.

For the experiment a testbed was deployed, consisting of a UE connected to the Internet by means of a real UMTS access. With this infrastructure,

[8]JAIN SIP Developer Tools, https://jain-sip.dev.java.net/

Message	Average delay (ms)
INVITE	134.37
Session in Progress	133.33
PRACK	133.52
OK_{PRACK}	121.26
UPDATE	136.62
OK_{UPDATE}	121.26
OK_{INVITE}	78.04
ACK	80.87
NOTIFY	207.32
OK_{NOTIFY}	78.13

Table 2: UMTS delays for SIP messages

it is possible to measure the Round Trip Time (RTT) of a message of a given size from the UE to its corresponding GGSN. This way, an acquisition process of RTT values was scheduled. In each execution of the process, five values of RTT were obtained for the size of each SIP signalling message. As the result, the execution produces five traces of time delays, each trace containing one RTT value for each of the SIP message sizes. The executions were planned daily with a period of fifteen minutes (from 00:00 to 23:45), and the acquisition process was maintained for over one month. These delays are summarized in Table 2.

With respect to the processing delay at each *CSCF* (2), an average value of 25 ms is used as indicated in [31]. Notice that a SIP message may traverse a set of *CSCFs* in the IMS networks of the UE and the CN. As it has been indicated in [5], the processing delay at the IPTV AS (3) and the delays in the core transport network (4), can be considered negligible for the mathematical study presented in this section. Finally, regarding to the delays in (5), it is assumed that the CN is attached to an ADSL access network with a 3 Mbps/1 Mbps (downlink/uplink) bandwidth.

The following subsections evaluate, from an analytical perspective, the handover delay (T_{buff}) associated with the proposals presented in Sect. 3 and Sect. 4.

5.1. SIP mobility delay

Using Fig. 5 it is easy to obtain the delay:

$$T_{buff} = T_{sip} + T_{attach} + T_{act,PDPp} + T_{act,PDPs} \quad (2)$$

In the previous equation, T_{sip} can be expressed as $T_{sip} = T_{sip,pause} + T_{registration} + T_{re-invite} + T_{sip,resume}$. To evaluate the previous equation, it is important to notice that some messages are processed and/or generated in parallel with other messages. For example, in Fig. 5, after the UE receives the *Session in progress*, it starts activating the secondary PDP context and, at the same time, sends the *PRACK* using the primary PDP context. Other SIP messages that are transmitted in sequence (both *OK* responses corresponding to the *UPDATE* and the *INVITE* requests, and an *ACK* with a *NOTIFY* request) have to be properly considered.

Taking into account the delay in the UMTS access network (as shown in Table 2), the number of traversed *CSCFs* and the delay in the ADSL access network for every SIP message, T_{sip} can be estimated as $T_{sip} = T_{sip,pause} + T_{registration} + T_{re-invite} + T_{sip,resume} = 0.206\,\mathrm{s} + 1.280\,\mathrm{s} + 0.753\,\mathrm{s} + 0.168\,\mathrm{s} = 2.407\,\mathrm{s}$. Finally, Eq. 2 is evaluated as $T_{buff} = 2.407\,\mathrm{s} + 5.67\,\mathrm{s} = 8.077\,\mathrm{s}$.

5.2. SIP context transfer

Let $T_{sip,standard}$ be the delay introduced by the standard SIP messages in Fig. 7 and $T_{sip,new}$ be the delay introduced by the new messages defined in Fig. 6. In this scenario, Eq. 2 is still valid, so the only thing to do is to estimate $T_{sip} = T_{sip,standard} + T_{sip,new} = 0.688\,\mathrm{s} + 0.365\,\mathrm{s} = 1.053\,\mathrm{s}$. In this case, $T_{buff} = 1.053\,\mathrm{s} + 5.67\,\mathrm{s} = 6.723\,\mathrm{s}$.

5.3. Mobile IP

In this case, T_{buff} has to be estimated at the HA and not at the CN, as we are using the ideas proposed in [29] to *pause* and *resume* the media.

5.3.1. P-CSCF in the home network

Using Fig. 9 as a reference, we obtain:

$$T_{buff} = T_{sip} + T_{mip} + T_{attach} + T_{act,PDPp} + T_{act,PDPs} \quad (3)$$

In the previous equation T_{mip} denotes the time necessary to perform a binding update plus the delays of a *pause* response and a *resume* request using MIP. As a round-trip time (RTT) for the typical packet size of a MIP binding update in UMTS is around 50 ms, T_{mip} can be estimated as $T_{mip} = T_{binding} + T_{pause,resp} + T_{resume,req} = 100\,\text{ms}$.

On the other hand, it is important to notice that, in this scenario, it is not necessary to re-register in IMS, because the UE uses the HoA as the registered IP address, which does not change during the handover. Therefore, in this case $T_{sip} = T_{re-invite} = 0.744\,\text{s}$. Finally, T_{buff} is expressed as $T_{buff} = 0.744\,\text{s} + 0.1\,\text{s} + 5.67\,\text{s} = 6.514\,\text{s}$.

5.3.2. P-CSCF in the visited network

Eq. 3 is still valid in this scenario, and again $T_{mip} = T_{binding} + T_{pause,resp} + T_{resume,req}$, but in this case, as a MIP tunnel is not used for the SIP signalling, it is not necessary to wait for the *binding update* acknowledgment so $T_{binding} = 0$ and $T_{mip} = 50\,\text{ms}$. In this scenario it is necessary to re-register (the mobile terminal registers the CoA in IMS), hence $T_{sip} = T_{registration} + T_{re-invite} = 1.280\,\text{s} + 0.744\,\text{s} = 2.024\,\text{s}$ and $T_{buff} = 2.024\,\text{s} + 0.05\,\text{s} + 5.67\,\text{s} = 7.744\,\text{s}$.

5.4. PCTS AS

The mechanism described in Sect. 4 can be used with or without mobile IP: there are two scenarios to analyze the delay.

5.4.1. Delay without MIP

From Fig. 15 it is easy to see that Eq. 2 is still valid. In this case, $T_{sip} = T_{sip,pause} + T_{sip,resume} = 0.374\,\text{s}$ and $T_{buff} = 0.374\,\text{s} + 5.67\,\text{s} = 6.044\,\text{s}$.

5.4.2. Delay with MIP

Eq. 3 is valid also here, but as it is not necessary to send any SIP message (registration is not necessary in MIP, and *pause* and *resume* are done using MIP packets towards the HA as defined in [29]) then $T_{sip} = 0$. For MIP signalling, and using [29] $T_{mip} = 0.1\,\text{s}$ then $T_{buff} = 0.1\,\text{s} + 5.67\,\text{s} = 5.77\,\text{s}$.

5.5. Summary of the delays

Table 3 summarizes all delays calculated in previous subsections. The column T_{buff} presents the final delay in seconds for each of the considered scenarios. This value represents the time during which the CN (or HA) has to buffer the video packets on behalf of the UE. The column $T_{buff} - T_{umts}^{sig}$ represents the same time as the previous column, but excluding the UMTS signalling delays, i.e. the time to attach a UMTS node to the network and the time necessary to activate a primary and a secondary PDP context. This

Proposal	T_{buff}[s]	$T_{buff} - T_{umts}^{sig}$[s]	**Section**
SIP	8.077	2.407	5.1
SIP CT	6.723	1.053	5.2
MIP (P-CSCF at home)	6.514	0.844	5.3.1
MIP (P-CSCF at visited)	7.744	2.074	5.3.2
PCTS AS	6.044	0.374	5.4.1
PCTS AS (with MIP)	5.770	0.100	5.4.2

Table 3: Handover delay for all proposals

column is included to emphasize the difference on the performance achieved by the different proposals.

For example, assuming that MIP is used, the gain of using the proactive context transfer service defined in Sect. 5.4.2 compared with the scenario where the *P-CSCF* is in the visited network (Sect. 5.3.2), is 1.974 s.

5.6. Recovery phase duration

Using the handover delays from Sect. 5.5, now it is easy to obtain the delay necessary to recover the buffer the mobile node had prior to the handover process. Using Eq. 1, we know that this time depends on the handover delay and on the ratio between the video bit rate R and the additional bit rate used to recover the buffer R_{add}. For example, using our PCTS AS proposal with MIP in a UMTS access network, and a ratio $R/R_{add} = 10$, the delay for the recovery phase is $d_r = 57.7$ s, i.e., less than one minute. In other words, if the mobile node, after performing a handover process, stays in the new network for one or more minutes, we can assure that the buffering will be recovered to its original state.

Even with this buffer recovery mechanism, it is necessary to minimize the handover delay, as longer handover delays will require more buffering at the mobile node, and will mean more added delay to start playing a new TV channel. In other words, buffer recovery and PCTS AS are complementary techniques to increase the quality of experience for the end users.

6. Conclusions

In this paper, we analyzed how to enable mobility for an IMS-based P2P IPTV service. We have studied current proposals for supporting mobility in

IMS-enabled networks in the particular case of a P2P IPTV service. Handovers involving changes of the IP address in IMS-based networks require complex protocol interaction that can lead to long handover delays. To avoid long bursts of packet losses in this paper we propose two novel techniques: (1) a buffering mechanism and (2) a Proactive Context Transfer Service (PCTS).

The purpose of the PCTS is to minimize the service re-establishment delay during inter-network handovers. The main idea is to use the existing Media Independent Handover technology (IEEE 802.21) in order to take a proactive approach in regards to detecting the state information (such as user equipment and proxy addresses) in the new network, applying the necessary QoS policies and updating the location at the IPTV application server and at the correspondent node.

To validate the benefits of our proposal, we compared the obtained service suspension time (or buffering time, i.e. the duration of time the IPTV content has to be buffered at the correspondent node or an intermediate equipment) with three other existing solutions. Our results demonstrate that using the Context Transfer Service results in a significant improvement. This has the benefit of requiring smaller buffer sizes at peer user equipment, and reducing the overall play-back delay.

Acknowledgements

This article has been partially granted by the Spanish MEC through the CONPARTE project (TEC2007-67966-C03-03/TCM) and by the Madrid Community through the MEDIANET project (S-2009/TIC-1468).

References

[1] International Telecommunications Union, IPTV Architecture, FG IPTV-DOC-0181, draft (2007).

[2] ETSI-TISPAN, Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IPTV Architecture; IPTV functions supported by the IMS subsystem IPTV IMS-based, Internet (February 2008).

[3] B. Quinn, K. Almeroth, IP Multicast Applications: Challenges and Solutions, RFC 3170 (Informational) (Sep. 2001).
URL `http://www.ietf.org/rfc/rfc3170.txt`

[4] A. Sentinelli, G. Marfia, M. Gerla, S. Tewari, L. Kleinrock, Will IPTV ride the peer-to-peer stream?, IEEE Communications Magazine 45 (6) (2007) 86–92.

[5] A. Bikfalvi, J. García-Reinoso, I. Vidal, F. Valera, A peer-to-peer IPTV service architecture for the IP multimedia subsystem, International Journal of Communication Systems.
URL http://dx.doi.org/10.1002/dac.1063

[6] 3GPP, IP Multimedia Subsystem (IMS); Stage 2, TS 23.228, 3rd Generation Partnership Project (3GPP) (Sep. 2008).
URL http://www.3gpp.org/ftp/Specs/html-info/23228.htm

[7] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler, SIP: Session Initiation Protocol, RFC 3261 (Proposed Standard), updated by RFCs 3265, 3853, 4320 (Jun. 2002).

[8] P. Calhoun, J. Loughney, E. Guttman, G. Zorn, J. Arkko, Diameter Base Protocol, RFC 3588 (Proposed Standard) (Sep. 2003).
URL http://www.ietf.org/rfc/rfc3588.txt

[9] G. Cunningham, P. Perry, J. Murphy, L. Murphy, Seamless handover of iptv streams in a wireless lan network, Broadcasting, IEEE Transactions on 55 (4) (2009) 796–801. doi:10.1109/TBC.2009.2030466.

[10] C. Perkins, IP Mobility Support for IPv4, RFC 3220 (Proposed Standard), obsoleted by RFC 3344 (Jan. 2002).
URL http://www.ietf.org/rfc/rfc3220.txt

[11] D. Johnson, C. Perkins, J. Arkko, Mobility Support in IPv6, RFC 3775 (Proposed Standard) (Jun. 2004).
URL http://www.ietf.org/rfc/rfc3775.txt

[12] R. K. Ed., Mobile IPv6 Fast Handovers, RFC 5568 (2009).
URL http://www.ietf.org/rfc/rfc5568.txt

[13] H. Soliman, C. Castelluccia, K. Elmalki, L. Bellier, Hierarchical Mobile IPv6 (HMIPv6) Mobility Management, RFC 5380 (2008).
URL http://www.ietf.org/rfc/rfc5380.txt

[14] V. Devarapalli, R. Wakikawa, A. Petrescu, P. Thubert, Network Mobility (NEMO) Basic Support Protocol, RFC 3963 (Proposed Standard) (Jan. 2005).
URL http://www.ietf.org/rfc/rfc3963.txt

[15] C. Vogt, M. Zitterbart, Efficient and scalable, end-to-end mobility support for reactive and proactive handoffs in IPv6, IEEE Communications Magazine 44 (6).

[16] P. Vidales, C. J. Bernardos, I. Soto, D. Cottingham, J. Baliosian, J. Crowcroft, MIPv6 Experimental Evaluation using Overlay Networks, Computer Networks 51 (10).
URL http://dx.doi.org/10.1016/j.comnet.2006.12.004

[17] R. Aguiar, S. Sargento, A. Banchs, C. J. Bernardos, M. Caldern, I. Soto, M. Liebsch, T. Melia, P. Pacyna, Scalable QoS-aware Mobility for Future Mobile Operators, IEEE Communications Magazine 44 (6) (2006) 95 – 102.

[18] C. J. Bernardos, I. Soto, J. I. Moreno, T. Melia, M. Liebsch, R. Schmitz, Experimental evaluation of a handover optimization solution for multimedia applications in a mobile IPv6 network, European Transactions on Telecommunications 16 (4) (2005) 317 – 328.
URL http://dx.doi.org/10.1002/ett.1005

[19] R. Chakravorty, P. Vidales, K. Subramanian, I. Pratt, J. Crowcroft, Performance Issues with Vertical Handovers - Experiences from GPRS Cellular and WLAN hot-spots Integration, in: Proc. Second IEEE International Conference on Pervasive Computing and Communications (PerCom'04), 2004, pp. 155–164.

[20] S. Balasubramaniam, J. Indulska, Vertical handover supporting pervasive computing in future wireless networks, Computer Communications 27 (8) (2004) 708 – 719.

[21] G. Heine, H. Sagkob, GPRS: gateway to third generation mobile networks, Artech House Publishers, 2003.

[22] T. Chiba, H. Yokota, A. Dutta, D. Chee, H. Schulzrinne, Performance Analysis of Next Generation Mobility Protocols for IMS/MMD Net-

works, in: International Wireless Communications and Mobile Computing Conference (IWCMC'08), 2008.

[23] T. Renier, K. Larsen, G. Castro, H. Schwefel, Mid-Session Macro-Mobility in IMS-Based Networks, IEEE Vehicular Technology Magazine 2 (1) (2007) 20–27.

[24] Media Independent Handover Services, IEEE Standard for Local and Metropolitan Area Networks, 2008.

[25] 3GPP, Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3, TS 24.229, 3rd Generation Partnership Project (3GPP) (Sep. 2008).
URL `http://www.3gpp.org/ftp/Specs/html-info/24229.htm`

[26] K. Larsen, E. Matthiesen, H. Schwefel, G. Kuhn, Optimized macro mobility within the 3GPP IP multimedia subsystem, in: Wireless and Mobile Communications, 2006. ICWMC'06. International Conference on, 2006, pp. 82–82.

[27] A. Dutta, K. Manousakis, S. Das, F. Lin, T. Chiba, H. Yokota, A. Idoue, H. Schulzrinne, Mobility Testbed for 3GPP2-Based Multimedia Domain Networks, IEEE Communications Magazine 45 (7) (2007) 118–126.

[28] M. Handley, V. Jacobson, C. Perkins, SDP: Session Description Protocol, RFC 4566 (Proposed Standard) (Jul. 2006).

[29] F. Xia, B. Sarikaya, B. Patil, Mobile IPv6 Handover Using Home Agent Buffering, Internet Draft (November 2008).

[30] A. De La Oliva, A. Banchs, I. Soto, T. Melia, A. Vidal, An overview of IEEE 802.21: media-independent handover services, IEEE Wireless Communications 15 (4) (2008) 96–103.

[31] D. Pesch, M. Pous, G. Foster, Performance evaluation of SIP-based multimedia services in UMTS, Computer Networks 49 (3) (2005) 385–403.