Bandwidth Sensitive Adaptation of Applications during MRM Controlled Multi-Radio Handover

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Abstract. In heterogeneous radio access networks mobile terminals can dynamically change their radio access not only between access points but also between different radio access technologies (RATs). We present a network based Multi-Radio Management (MRM) for seamless control of inter-RAT mobility. Access Selection can be triggered by a change of radio channel quality, by the start of an application or by system load and handover is executed by enhanced MobileIPv6. Dynamic adaptation of the IMS services, e.g. adaptation of IPTV to the available bandwidth, is realized by triggering additional mid-call SIP signaling between communicating parties so that Quality of Experience (QoE) can be significantly improved. The described MRM and dynamic adaptation of IMS services are validated by a prototypical implementation in an integrated demonstrator.

Keywords: Multi-radio handover, vertical handover, MRM, MRRM, radio access selection, ASF, IMS, codec adaptation, SIP re-invite, SIP codec adaptation.

1 Introduction

Today, many different radio access technologies (RATs) such as GSM [1], UMTS [2], CDMA2000 [3], WLAN [4] or WiMAX [5] coexist in parallel—quite often in an overlaying deployment by the same operator. Such operators need network solutions that provide their subscribers with ubiquitous mobile services in overlay cells, hotspots, indoor-coverage, and added cells with future wireless technologies. Inter-RAT mobility between 3GPP technologies (GSM, UMTS, HSPA, LTE) and non-3GPP technologies (CDMA2000, WiMAX, WLAN) is currently being standardized in the 3GPP evolved packet system (EPS) [6]. But the specification is still missing a network based decision functions for inter-RAT access selection although some related work has been initiated in 3GPP in the scope of the Access Network Discovery and Selection Function (ANDSF)[1] [6]. Furthermore, the change of

[1] ANDSF supports terminal based multi-radio access selection by network based policies.

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available QoS (e.g. the supported data rate) during handover is not communicated to the application layer. In this paper we describe a multi-radio management (MRM) function that monitors the radio link quality of the serving cell and of the candidate cells, anticipates the change of available bandwidth caused by a planned handover and triggers the IMS application function to adapt the running SIP services, e.g. to change the codec of a video stream.

A major issue for such inter-RAT access selection and for cross-layer optimization of the QoS is that different RATs use different measurement values and metrics to quantify the radio channel quality. The envisioned interworking thus requires abstraction and adaptation functionality between radio layers and MRM to map the specific link quality values to a generic scale of values that can be compared to application QoS values. Some standardization bodies are working on general interworking solutions, such as the IEEE Media independent Handover (MIH) framework [7]), but they do not provide solutions to metrics for inter-RAT access selection. This paper presents an integrated solution comprising the MRM concept [8][9] developed by Alcatel-Lucent Bell Labs and the IMS enhancements [10][11] developed by Deutsche Telekom / T-Systems. This work has been undertaken in the framework of cooperative projects, such the German Federal Ministry of Research and Education's (BMBF) project "Scalable, Efficient and Flexible Networks" (ScaleNet) [12] and the European Union's (EU) project "Ambient Networks" [13].

2 Multi-Radio Management

In a heterogeneous wireless environment mobile terminals can dynamically change their radio access not only between access points but also between different radio access technologies (RATs). For a change of the radio access, i.e. for an inter-RAT handover, the mobile terminal (1) has to measure the radio link quality of the serving radio link and of candidate radio links in other technologies and (2) to take access selection decision to figure out if and when one of the candidate cells offers a better service than the currently serving cell. Due to battery constraints of the terminals it is not possible to continuously scan over all radio channels on all radio interfaces. Instead, a multi-radio capable functionality with knowledge of the terminal's position and of the network topology and status shall advertise nearby candidate cells.

Such a coordinated operation of several RATs requires the introduction of an overarching and generic resource and mobility management residing above the legacy Radio Resource Management (RRM) and Mobility Management (MM) of the individual technologies. This new entity is called Multi-Radio Management[2] (MRM) [8]. MRM functions are typically distributed over the core network, the access networks and the mobile terminals. To take full benefit of MRM and to allow operator control over the load distribution, it is preferable to deploy the access selection function in the network [8].

The main functions of MRM are described in more details in the following sections.

[2] MRM is sometimes also called Multi-RRM (MRRM), Joint RRM (JRRM), Heterogeneous RRM (HRRM) or Common RRM (CRRM).

2.1 Abstraction and Adaptation Layer

The mechanisms and procedures for link control strongly differ between radio access technologies, i.e. MRM would have to operate different access selection algorithms and different handover mechanisms for each pair of RATs. This requires specification and interfaces involving multiple standardization bodies and large implementation effort.

Therefore, an abstraction and adaptation layer (AAL) is defined between MRM and the radio interfaces, hiding the RAT-specific properties from the generic MRM (Fig. 1). The main functions of this layer are the translation of generic MRM procedures to RAT-specific procedures and the mapping of RAT-specific values to generic values and vice versa. For example, a generic link setup request of MRM may be mapped for IEEE 802.11 WLAN into a sequence of Mlme.Join, Mlme.Authenticate and Mlme.Associate primitives as specified by the WLAN protocol [4]. More examples for the translation process are given in [8].

The advantage of these abstraction and adaptation function becomes particularly visible when new radio technologies are integrated into existing systems. In this case, the MRM as well as the AAL for existing RATs remain unchanged, while only the adaptation and abstraction functions needs to be implemented for the new RAT.

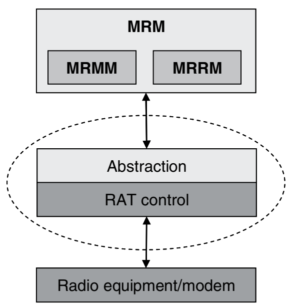

Fig. 1. Abstraction from RAT specific to generic values and commands

2.2 Multi-Radio Measurements

The radio link quality is expressed in different RATs as a radio signal strength (in dBm or as relative RSSI values, e.g. in 802.11 WLAN), as a signal-to-noise value (SNR, SNIR, given in dB), as bit error rate (e.g. in 10^{-4}) or as relative channel quality indicator CQI (e.g. in 3GPP HSDPA). These link quality values are not directly comparable between each other, nor to the QoS requirements of the running service. The abstraction layer thus recalculates these values according to each RATs specifications into an available peek data rate at a given residual bit error rate. Furthermore, if the RAT supports QoS guarantees, it also calculates the granted minimum data rate and the transmission delay [8].

MRM configures generic link quality measurement requests at the interface to AAL to receive periodic and event-based link quality measurements. Candidate measurement requests are based on the advertisement of surrounding cells. For moving users, detection of entering or leaving hotspots with broadband coverage has to be performed fast and efficiently to enable well-informed MRM decisions.

In the same abstracted way, MRM on network side collects load information to assess the availability status of the cell resources.

2.3 Access Selection

Radio access selection denotes the process of choosing for a terminal (at a given location and running a given service) the most appropriate access technology and point of attachment (cell or access point) within a heterogeneous network. However, inter-technology handover is not always reasonable. E.g., real time services may not be well supported by a best effort WLAN hotspot or fast moving mobile terminals may reside within a hotspot for too short an amount of time. Therefore, access selection algorithms need to be configured in terms of operator and user criteria [13].

Access (re)selection decisions are taken proactively based on the abstracted link quality measurements, in order to seamlessly finalize the handover execution before a fading link breaks. This enables always-best connectivity for mobile users.

Next to the provisioning of best connectivity to the user, access selection also offers operators the potential of optimizing network usage and revenue. Obviously, inefficient service distribution between the different access systems may lead to overload on some access systems, while resources in alternative access systems are still available. Moreover, depending on the link performance and the service type, the resource costs of different access technologies differ, leading to a potential for overall capacity increase.

In the presented implementation, MRM applies a decision algorithm with a weighted metric regarding abstracted link quality, application QoS requirements, and system load. The Access Selection Function (ASF) can either be implemented distributed over the access networks or as a centralized service on a Heterogeneous Access Management (HAM) server in the operator’s core network [8].

2.4 Mobility Management

Furthermore, MRM comprises a mobility management, which coordinates the time sequence of build-up and release of the radio connections during handover execution. Depending on the capabilities of the mobile terminal different handover protocols and handover sequences can be used [9]. In this article, we assume use of Mobile IP protocol IPv6 [14] and that the terminal is capable of connecting the target radio link before releasing the source link, so that a seamless make-before-break handover can be performed.

3 IP Multimedia Subsystem

The IP Multimedia Subsystem (IMS) [15] is an architectural framework in 3GPP EPS for delivering Internet Protocol (IP) multimedia services independently of the type of wireless access or wired access (fixed mobile convergence, FMC).

The IP Multimedia Subsystem (IMS) is used to forward the complete Session Initiation Protocol (SIP) [16] signaling used in the IMS for session management. A number of Call Session Control Functions (CSCF) are introduced to establish a multimedia session between subscribers and to prepare delivery of the demanded services according to the session characteristics required by users. Some of CSCFs have interfaces to the Home Subscriber Server (HSS) where the complete information about particular subscribers is stored, like their profiles, policies, subscriptions, preferences, etc.

Three types of CSCFs are defined:

- Proxy-CSCF (P-CSCF) is the first point of contact for the IMS users. The main goals of the P-CSCF are the guarantee of signaling messages between the network and subscriber and resource allocation for media flows by the interaction with Resource and Admission Control.
- Serving-CSCF (S-CSCF) is the main control entity within the IMS. It processes registrations from subscribers and stores their current location, is responsible for subscriber authentication and call management. Subscriber policies stored in the HSS control the operations performed by the S-CSCF for a particular subscriber.
- Interrogating-CSCF (I-CSCF) queries the HSS to find out the appropriate S-CSCF for the subscriber. It can also be used to hide operator's network topology from other networks.

4 Bandwidth Adaptation of Applications

MRM monitors serving and candidate radio link qualities with respect to the requested QoS of the running application. When MRM decides on a handover to provide best possible service the provided data rate may change. MRM as described in [8] does not interact with the running applications. Further mechanisms are required to adapt the service to the available bandwidth. For example, when the mobile terminal enters a hot spot with high data rate the resolution of a video stream could be improved from VGA to HDTV. In case the bandwidth reduces, e.g. when leaving a hotspot, the high data rate will rapidly fill the transmission queue of the target RAT and packets will be dropped, either by policing or simply by buffer overflow. Both will cause artifacts or freezing of a video stream as shown in Fig. 6b. The disruption of the video stream reduces the QoE for users dramatically as it is not possible to continuously follow the video content.

Different mechanisms on application layer will be briefly discussed for adaptation of the data rate.

4.1 UDP Applications

The User Data Protocol (UDP) [17] is a best-effort protocol without flow control, accepting incoming data as-is. Often the Real Time Transport Protocol (RTP) [18] is run on top of UDP. Video streaming with a data rate determined by the video source is a typical example for a service running over RTP/UDP/IP. A higher available data rate on the link will not be used, a lower data rate on the link will lead to vey low quality[3].

[3] If the terminal runs some kind of application manager (e.g. the user interface of a PDA) data rate monitoring may be used to stop the application if the data rate drops below a minimum.

4.2 TCP Applications

TCP [19] is a transport protocol providing reliable transmissions and which adapts the data rate according to the round trip time between communicating peers. TCP quickly reduces the data rate in case of congestion and slowly ramps-up to a higher data rate if served well. This efficiently adapts the data rate of applications that can send data as fast as the queue is emptied, e.g. non-realtime services like file downloads using File Transfer Protocol (FTP) [20]. However, the basic TCP does not take benefit from anticipated data rate changes during MRM handover [4] and the ramping is not suitable for applications with a fixed packet rate, like real-time video streaming.

4.3 IMS/SIP Applications

IMS/SIP applications also use UDP or TCP transport. But on top of the transport layer, the data rate of the applications is negotiated between the communicating peers (e.g. between a terminal and a video server) during session establishment using SIP INVITE messages. Within these messages a list of supported codecs is exchanged. SIP RE-INVITE messages can be used to adapt the codec at any time during the running service. In IMS, the IMS Application Function (AF) is introduced to be located on a CSCF and that acts as a SIP proxy mapping a codec to the required QoS parameters (e.g. bit rate, BER) on the IP layer. Doing so, AF is able to change the codec depending on the IP resources available for user on the transport plane.

Interworking of MRM and IMS AF opens the possibility to use the knowledge of abstracted link performance not only for MRM access selection but also for adaptation of the codec and of codec parameters of running multimedia sessions. During start of a SIP session the IMS AF registers at MRM for QoS status information, supplied by AccessFlow.Indications with the actual user plane data rate. When the data rate changes, e.g. at an MRM induced handover event, the IMS AF receives the newly available data rate from MRM. AF then enforces codec adaptation by sending SIP RE-INVITE messages with the new codec to the terminal and to the media server (Fig. 2).

For the pro-active handover execution two cases have to be distinguished:

- When the AccessFlow.Status.Indication of the target RAT shows a higher data rate the current codec shall be used until the handover is executed. After the data plane has been switched (i.e. after MIPv6 Binding Update) the AF enforces the use of a larger codec to provide better QoE to the user utilizing the increased data rate.
- If the candidate radio link offers lower bandwidth than the current radio link (e.g. for a fading link when leaving coverage of a hotspot) the change to a codec requiring a lower bit rate must happen before the switching of the data plane to the new radio link. Otherwise the data rate will rapidly fill the transmission queue and buffer overflow or policing of RRM will lead to dropping of packets and bad user experience. Instead, MRM must send an AccessFlowStatus.Indication to the AF before executing the handover. Only after the codec has been re-negotiated MRM can trigger the MIPv6 binding update.

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[4] Like for UDP transport, an application manager can be used to terminate the application if the data rate drops below a minimum, or the remaining download time increases unacceptably.

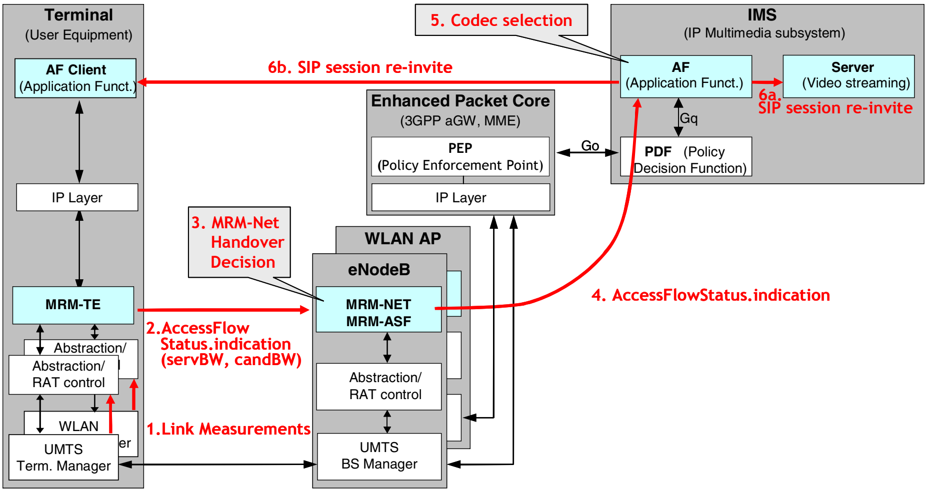

Fig. 2. Signaling of MRM and IMS for bandwidth adaptation to radio link bandwidth

5 Demonstrator

In order to validate the MRM and IMS enhancement concept, a demonstrator [22] has been set up comprising two different radio access networks and a dual-radio mobile terminal.

5.1 Demonstrator Set-Up

The MRM demonstrator [8,9,22] consists of a 3GPP UMTS/HSDPA overlay cell, two small IEEE WLAN hotspots and a dual-radio mobile terminal. The integrated MRM & IMS demonstrator extends this with the extended IMS functionality based on the IMS implementation [21]. On the network side an IPv6 application server and an IPv4 IMS server have been integrated (Fig. 3). The demonstrator supports IPv6 mobility for handover during ongoing services of different QoS classes, i.e. FTP based file transfer, Web browsing via Hypertext Transfer Protocol (HTTP) [23], video streaming via UDP and RTP controlled manually or using SIP/IMS.

Generic MRM instances are implemented in the terminal, the core network, and the access networks of each RAT. Between the MRM instances an interface has been specified and implemented based on the Diameter [24] protocol. For the cross-layer interaction between MRM and MIPv6 in the terminal the standard interface to the IP kernel functions is used. The interface between MRM and the abstraction function has been implemented for an UMTS base station (Alcatel-Lucent Evolium NodeB) and for an WLAN 802.11a-based access point (Signalion Sorbas AP). Real radio signal strength measurements are acquired. The movement of the dual-radio terminal is emulated by changing the link performance with a radio frequency (RF) attenuator at the WLAN antenna. The network based MRM Access Selection Function (ASF) thus has access to generic resource status information from MRM modules communicating with UMTS and WLAN interfaces in the terminal and in the access networks. It triggers handovers according to the requested QoS level, the currently available data rate for both access interfaces of the terminal, and the handover policies. MIPv6 handover execution is caused by changing the interface preference in the terminal.

Most parts of the demonstrator are implemented on LINUX platform, but the UMTS control function of the HSDPA test mobile and the IPv4 SIP client run on WINDOWS OS with IPv4 interfaces. Therefore, the User Terminal comprises a network of three computers and compatibility to the IPv6 demonstrator has been achieved by IP6-in-IP4 tunneling mechanisms.

The MRM-demonstrator setup with a distributed MRM deployment is shown in Fig. 3.

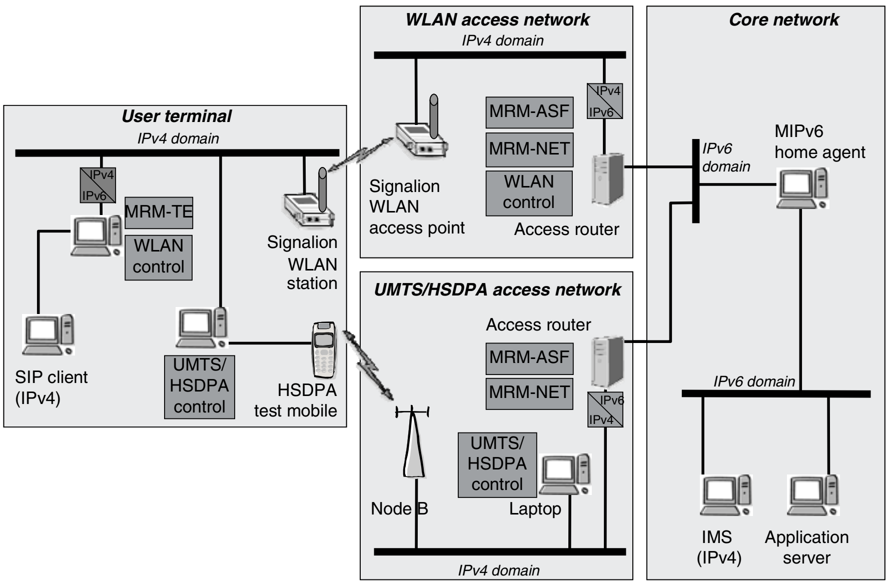

Fig. 3. Demonstrator Setup

The user terminal implements a standard IMS client to be able to initiate and to maintain video streaming services controlled by the IMS. The IMS server [10,11] has been extended with an IMS-MRM Proxy function that communicates with IMS AF deployed on the P-CSCF using standardized Gq' interface. On the other side, it implements the proprietary MRM interface so that signaling messages regarding QoS requests can be translated to MRM signaling for management of resources on the transport plane. The IMS-MRM Proxy function allows the IMS controlled session management to request for an IMS application a minimum and maximum usable QoS level on the access. It receives data rate reports by MRM AccessFlow-Status.Indications and executes dynamic application codec selection when the measurement value exceeds this range.

Fig. 4 shows the detailed architecture of the integrated MRM & IMS demonstrator.

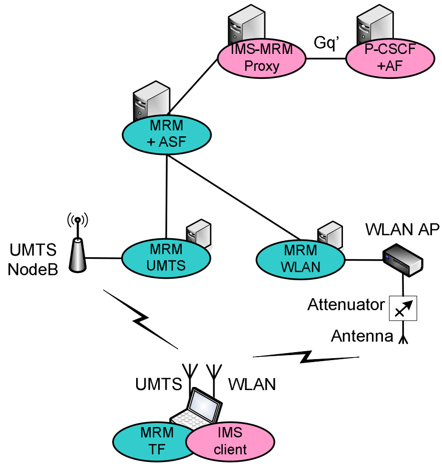

Fig. 4. Architecture of the integrated MRM & IMS demonstrator

5.2 Demonstrator Results

The feasibility of the MRM concept and the scalability by MRM distribution were verified successfully. The distributed MRM instances on network side (MRM-NET) trigger handover decisions in a central or co-located decision function (MRM-ASF) based on abstracted radio link measurements, application type, operator preferences and cell loads. MRM instructs the user terminal to perform a MIPv6 handover between the two heterogeneous RATs. The trigger levels, the decision algorithm, and the simultaneous availability of both radio interfaces during handover (make-before-break versus break-before-make) can be configured. The resulting handover performance is determined by user-perceived service interruption and by measurements of packet loss and transmission time.

The data rate of an FTP file download over UMTS/HSDPA is limited in the demonstrator to 300kBit/sec due to the 80 msec delay on the UMTS uplink. During the handover from HSDPA to WLAN, TCP ramps up the download rate to 3.5MBit/sec and in the other handover direction it successfully prevents congestion by returning to 300kBit/sec. For UDP video streaming with a data rate below 1MBit/sec both RATs support the data rate and MRM performs pro-active handovers seamlessly in both directions, adaptation by IMS AF is not required in this case.

MRM decided handovers have also been successfully performed during video streaming service controlled by IMS/SIP. Fig. 5 shows the reduction of WLAN signal strength caused by a stepwise increase of the attenuator. Handover is performed at 20% relative signal strength where the abstraction layer calculates that the currently used WLAN data rate drops below the data rate estimated from the measurements of the UMTS candidate link (1000 kBit/sec in Fig. 5). MRM sends an AccessFlow-Status.Indication to the MRM-IMS Proxy with information about a newly available data rate for the terminal of 300 kbit/s. MRM-IMS Proxy informs the AF via the Gq' interface which then initiates the adaptation of the codec bit rate parameter for the established video session. After the UMTS target link is established in a make-before-break handover, the data is then transported via the UMTS radio link. Due to limited computing power of the IMS server (implemented on a desk top PC) the re-coding of the video stream leads to a short interruption of the data stream at the source which is sometimes perceivable to the watchful eye. Fig. 6 shows a screenshot of the video quality over WLAN before the handover (Fig. 6a) and the reduced quality after adaptation of the service to the data rate over UMTS/HSPA (Fig. 6c). Please note that without the codec adaptation, the transmission buffer of the NodeB would be quickly filled and about 90% of the packets would be dropped. The video and sound quality would be completely inacceptable without such a codec adaptation (Fig. 6b).

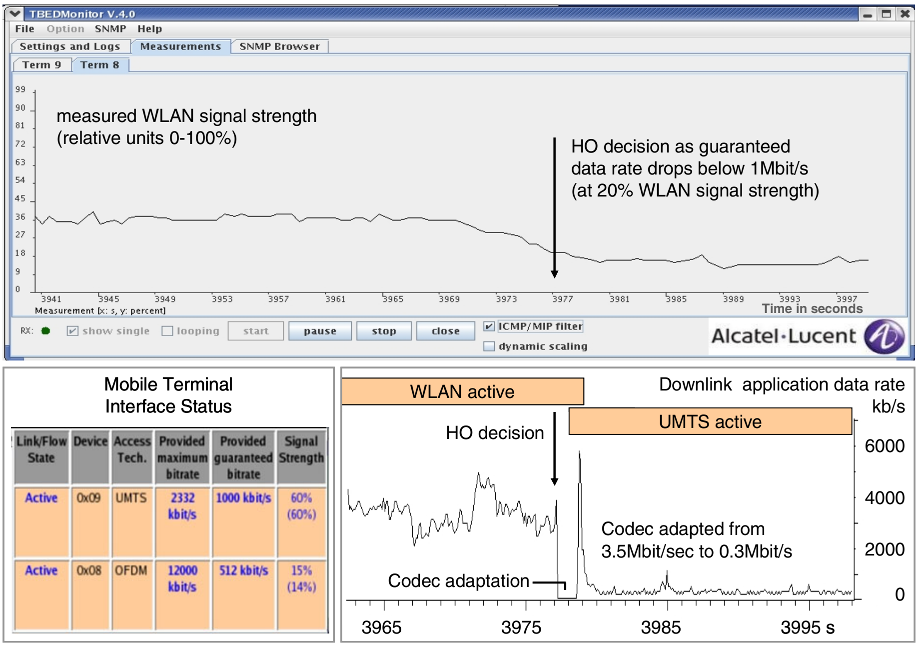

Fig. 5. Signal Strength and Video Streaming Downlink Data Rate during handover to a radio link with lower data rate capability

Fig. 6. By adaptation of IMS codec before Handover streaming can continue with reduced resolution (screen shot of video (a) before handover, (b) after handover without adaptation and (c) after handover with MRM/IMS codec adaptation)

5 Conclusion

Fourth generation wireless networks will probably consist of multiple, heterogeneous radio technologies, requiring intelligent interworking solutions. The multi-radio management (MRM) concept is one necessary step towards a pervasive and effective integration of current and future access technologies. MRM handles the differences between heterogeneous access technologies in a unified way by abstraction from RAT-specific parameters and adaptation to RAT-specific functionality in an adaptation and abstraction layer (AAL). The proposed general approach leads to a significant reduction of standardization and implementation efforts compared to bilateral interworking solutions between each pair of radio access technologies.

The MRM functions and architecture were discussed, the key functionality of MRM is a network-based, pro-active radio access selection mechanism that considers radio link performance, resource usage, and user and operator preferences. For seamless inter-technology handovers, MRM synchronizes IP mobility protocols with link and session layer procedures. This approach enables inter-RAT access selection and common resource and mobility management for all RATs, offering the potential to optimize network usage.

Even though MRM performs handovers seamlessly, the data rate may change in case of inter-RAT handover and this can spoils high user QoE. By interworking of MRM with the IP Multimedia Subsystem (IMS) the MRM information about the actual link performance can be utilized to match the available data rate by adapting the codec with a SIP RE-INVITE, either before or after handover execution.

The feasibility of the MRM concept and its integration with IMS have been realized and validated in an MRM/IMS demonstrator offering seamless IP-mobility between a cellular UMTS/HSDPA network and a WLAN hotspot during ongoing multi-media sessions. MRM measurements of radio link QoS and handover triggers are leveraged in the IMS for pro-active SIP codec adaptations. Nearly seamless adaptation of the quality of a video stream has been demonstrated for handover in both directions.

Based on these results the proposed MRM/IMS integration is a promising solution for future 3GPP enhancements of the EPS architecture, enabling added-value services and providing enhanced QoE in fourth generation wireless networks.

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