Simulation model for performance evaluation of advanced SIP based mobility management techniques

Rok Libnik, Gorazd Kandus, and Ales Svigelj

***Abstract*—**Wireless technologies have evolved very rapidly in recent years. In the future, operators will need to enable users to use communication' services independently of access technologies, so they will have to support seamless handovers in heterogeneous networks. In this paper, we focus on building the simulation model for testing advanced handover procedures. Simulation model based on real operator network measurements was presented, verified and validated. In addition, advanced procedure for SIP based seamless handover in heterogeneous network using congestion detection was described. Its performance was evaluated using the described simulation model. The results have shown that, by using the CAHP C procedure, we can significantly improve the user experience for VoIP application when performing seamless handover in heterogeneous networks.

***Keywords*—**Congestion detection, Handover, SIP, Simulation.

I. Introduction

IN the last decade we have been witnessing rapid development of new wireless network technologies. The most popular today are wireless LAN, which is usually provided in limited coverage areas, and GSM/UMTS, which is provided over wider coverage areas. Other technologies that have evolved in today's market include WiMAX, the HSPA (High Speed Packet Access) family and pre 4G. To enable users of mobile terminals communication in a variety of networks, equipment manufacturers have started to offer dual mode handsets, while operators are starting to offer fixed mobile converged services. Some of them (BT as the first in 2005) already offer services which enable users to handover seamlessly from GSM to WLAN network, using the UMA (Unlicensed Mobile Access) approach. However, user demands are increasing very rapidly and operators will also need to offer seamless handover between heterogeneous wireless technologies.

The biggest motivation among all applications for using seamless handover is for real-time applications (e.g. Voice over IP (VoIP)). The main challenge in providing seamless handover between heterogeneous networks is the assurance of appropriate quality of experience (QoE) for the user during handover. This is particularly important when the target network is not under the control of the user's operator/service provider, which will usually be the case in future, since users will use different access technologies/networks, run by different operators. In order to be able to develop new services that will maintain quality of experience for the user during movements between different networks, new mechanisms needs to be tested prior starting industrial development.

The availability of special-purpose simulation tools and massive computing capabilities at decreasing cost per operation, have made simulation one of the most widely used and accepted tools in research and system analysis. Simulation can significantly decrease the development of new services as the behavior of newly proposed services can be tested in advance in the simulation environment, exposing also to the worst case scenarios, which can arise in the real network environment. From the telecom operators' point of view the simulation can be used to experiment with new designs or policies as to prepare what might happen. This is even more important in today's highly competitive environment, where the delivery of new services which improves users QoE is of paramount importance. Today it is very hard to find simulation models for handover in heterogeneous networks, thus simulation models with handover functionalities should be developed using appropriate simulation tools.

The main aim of this paper is to build the simulation model of the wireless telecommunication network for testing the advanced newly developed mobility management techniques using Session Initiation Protocol (SIP). In order to make the simulation environment as real as possible, the simulation model should incorporate real signal measurements that are mimicking the user movements in real environment. It enables the simulation any access technology, although in this paper we are focusing on WLAN and HSPA.

This paper is organized as follows. In Section 2 related work on mobility management techniques are presented, from which SIP mobility is described in more details. In Section 3 the analysis of real operator environment is shown. The section describes also the simulation model development and simulation scenario. The validation and verification of simulation model and simulation results for CAHP C procedure are presented in Section 4, followed with conclusions in Section 5.

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II. RELATED WORK AND SIP MOBILITY

In today's networks the majority of handovers are performed in homogeneous networks, in which the handover takes place between the same access technologies, i.e. horizontal handover. Examples of horizontal handover can be found in today's mobile networks (e.g. GSM, UMTS). In the future, there will be many different wireless networks that will support a variety of applications, allowing users seamless connectivity between different access technologies. Our focus is therefore on seamless handover in heterogeneous networks (vertical handover). From the protocol implementation point of view handover in heterogeneous network can be performed at different OSI layers.

At the network layer, authors have mainly selected MIP as the protocol [1-4], as defined in [5]. When using MIP for handover, new elements such as home and foreign agents need to be installed in the network. Because MIP is a network layer protocol which is used just for transportation, the application which is in use is not aware of the handover process. However network involvement in handover execution is large, as handover is executed in the network itself. The usefulness of MIP for handover is limited by its less wide use, and few operators have implemented it to their network.

The most widely used protocols at the transport layer are TCP and UDP. Both have some limitations for mobility support. To overcome these, a new protocol called SCTP (Stream Control Transmission Protocol) was introduced [6]. However SCTP is not capable of changing the IP address once the session has started. A new solution called mSCTP (mobile SCTP) was then developed, which enables IP addresses to be added, deleted and changed during active SCTP association [7,8]. Performing handover on the transport layer has little impact on the network [9]. If MN supports mSCTP, then operators do not need to change their network. As handover is executed on the transport layer, the application, like on the network layer, is not aware of the handover execution and can be used as it is. The biggest drawback of the mSCTP protocol is that is rarely used in a real operator environment.

For mobility management on the application layer, SIP [10] is usually selected as the most favoured protocol [3,11,12]. SIP runs on top of several different transport protocols and is today's most widely used protocol for IP telephony. With minor modifications, SIP can support four types of mobility: Terminal mobility enables devices to move between subnets and be accessible to other hosts and to continue any ongoing session when they move. Session mobility enables users to maintain a session while moving from one terminal to another. Personal mobility enables users to use the same set of services, even when changing devices or network attachment points and service mobility, that enables users to be identified by the same logical address, even if the user is at different terminals. The advantage of using SIP protocol for handover execution is that SIP is an application layer protocol and thus agnostic to lower layers. On the other hand an application usually needs to be improved to support handover. The transport independence of SIP means that it does not require large network involvement, but its biggest advantage is the wide adoption in real operator environments, since almost all operators that are offering VoIP services use SIP for signalization.

Our main aim is to focus on solutions that can be easily deployed in a real operator environment with minor modification to existing operator's network architecture and no modifications in guest networks. SIP is used in many operator environments and has been selected as the primary signaling protocol in IMS (IP Multimedia Subsystem) networks. As it runs on the application layer, SIP is independent of access technologies. Thus, we decided to focus on the use of SIP in mobility management in particular on terminal mobility for which two types of mobility management approaches have been defined – pre-call mobility and mid-call mobility. Although SIP supports several applications in this paper we selected IP telephony.

In pre-call mobility scenario mobile node (MN) moves to another network and gets the new IP address. In the session description of SIP REGISTER message, the MN informs the SIP server about the new IP address. Usually the MN gets the new IP address from the DHCP server, which is located in the network. Because the new IP address is acquired prior to the call this operation does not affect the quality of experience, and thus will not be discussed further. In the mid-call scenario first a call is established between the corresponded node (CN) and the MN that is in the home network When MN moves to another network, it sends the SIP re-INVITE message to CN and informs it about the location change. The new RTP session is then established. Similar to the pre-call scenario, the MN gets the new IP address in the new network. In the SIP re-INVITE message the session description also includes the new IP address. The limitation of this approach is that the SIP server is not informed about the location change. In the literature, some solutions have been presented in which MN informs the SIP server about the location change after sending the SIP re INVITE message [3]. However, in a real operator environment, information about location change needs to be sent to the SIP server prior to starting a new SIP session between MN and CN. This should be done to support proper charging, since prices can differ between networks.

To overcome that limitation in [13] we proposed the enhanced mid-call mobility scenario presented in Fig. 1.

First the initial call is established (steps 1-5). After moving to another network (step 6), the MN sends the SIP re INVITE message to the SIP server (step 7) to inform it about location change. SIP server then forwards the SIP re-INVITE message to the CN (step 8). After the acknowledgement (steps 9, 10), the new RTP session is established (step 11). As in both types of mobility, the MN gets the new IP address when it moves to the target network.

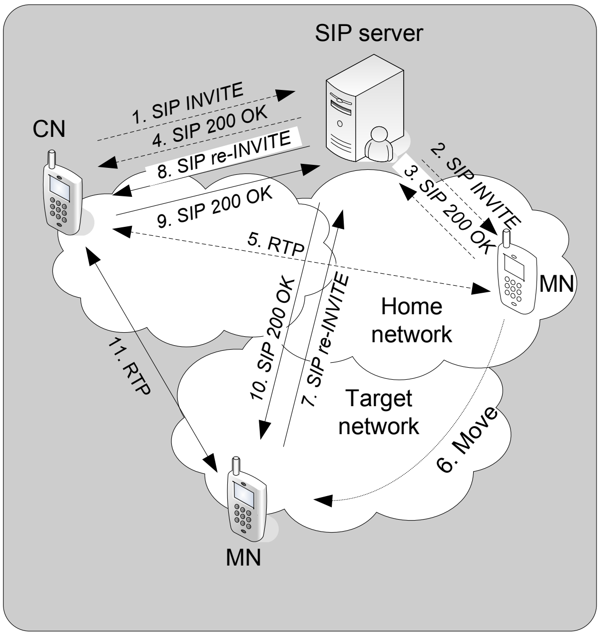

Fig. 1 Enhanced mid-call mobility scenario

Most commonly the trigger used for handover execution is based on received power. This trigger is usually the prerequisite for handover – i.e. handover cannot take place to a network lacking, or with limited, signal coverage. In some cases relaying only on SNR can lead to service degradation, as SNR ratio of the target network can be above the predefined threshold, but at the same time the target network's QoS parameters may not be adequate, due to network congestion (e.g. in the case of handover to a free public WLAN network in a congress centre or hotel). In literature [14-16] it is possible to find proposals of other triggers; such are bandwidth of the target network, economic price, power consumption, user preferences, priority given to interfaces, location, and velocity. For real time applications in particular, the ability to provide appropriate QoS in the target network is crucial. Thus we decided to focus only on QoS parameter, end to end delay in particular. To detect congestion in the network when SNR ratio exceeds the predefined threshold in [17] we defined the Congestion Aware Handover Procedure (CAHP C) procedure, which is used for congestion detection. The congestion detection algorithm is based on sending newly defined SIP messages. The end to end delay of the unreliable (congested likely) network is then calculated based on responses to those SIP messages.

III. Simulation Model

The main aim of our research was to develop a simulation model of a telecommunication system resembling the experimental environment, where we can evaluate different handover procedures based on SIP protocol. Our main focus was on analyzing the VoIP service. The steps of building simulation model are described in the following subsections. First the analysis of VoIP service in a real operator environment will be presented. Next modeling of the telecommunication network will be described, which will be further divided to description of modeling of the architecture and description of implementation of CAHP C procedure in the simulation model.

A. Analysis of VoIP service in a real operator environment

All real-time applications, such as video and voice, are very sensitive to QoS parameters, usually measured by jitter, end-to-end delay, and packet loss. In [18] ITU recommends the upper limit values for each parameter. If these are exceeded, the user will notice degradation of the service. For our work we selected end to end delay, which should be by ITU recommendation less than 150 ms. However, the generally accepted limit for good quality voice connection is 200 ms delay one way [19].

In general, there are two main factors in the wireless environment which have the biggest impact on the user's QoE for voice calls: the quality of the signal, measured in terms of SNR ratio, and utilization of the network or utilization of the most congested link. In order to get as realistic result as possible we have decided to analyze the influence of the above parameters on the quality of the VoIP call. Obtained results will be used as the input parameters in our simulation model for testing advanced seamless handover procedures.

When setting up the testing scenario some characteristics of a real operator environment need to be taken into account. Operators are obliged to provide some additional functionality that can change the architecture of the IP telephony solution. Usually they add a Session Border Controller (SBC) to their network that provides several functionalities including [20]:

- perimeter defence (access control, topology hiding, and denial of service (DoS) detection and prevention);
- functionality not available in the endpoints (NAT traversal, protocol interworking or repair);
- traffic management (media monitoring and QoS).

By adding SBC to the architecture, some practices can be in conflict with SIP architectural principles. SIP based SBCs typically handle both signalling and media. The SBC also provides controlling and protecting functions and is configured and managed by the operator.

Taking account of the above characteristics and requirements, we have set up an experiment in a real operator environment. The architecture is presented in Fig. 2.

The SIP client on laptop NB was connected via a WLAN interface to access point AP in our room and with HSPA USB modem to mobile network. The AP was connected to a hub to which computers CPU1, IP phone 2 and CPU3 were also connected. The hub was further connected to a switch, which was connected to the LAN network and to computer CPU2. The purpose behind using the hub was to monitor the status of the access connection, which is usually more problematic for assuring QoS. In the core networks of a real operator environment there is sufficient bandwidth and enough mechanisms for QoS assurance. Another IP phone (i.e. IP phone 1), which was connected to internet, was also used. For the analysis we established a call between SIP client on laptop computer NB and IP phone 1 or IP phone 2. SIP client and both IP phones were registered to Telekom Slovenije IP telephony service.

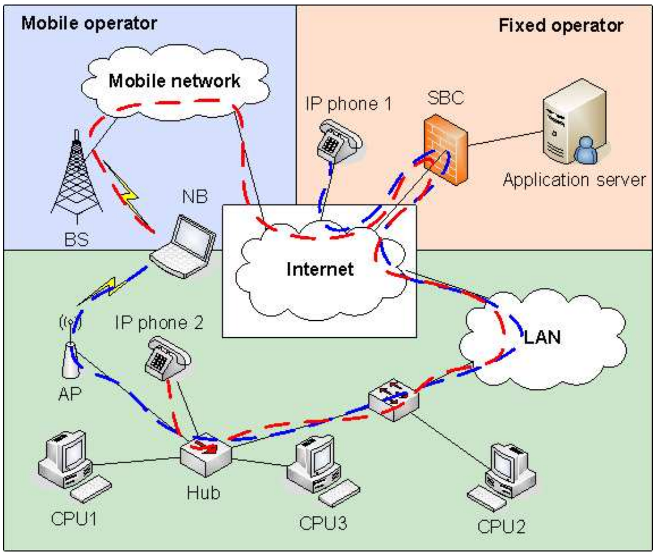

Fig. 2 Network architecture of a demonstration setup

We tested four experimental scenarios:

1. ES_BASIC_HSPA: This scenario was used for enough the reference status of the mobile operator network. We established a call between SIP client on NB and the IP phone 2. The call was established via HSPA USB modem connected to NB. The RTP stream of scenario is presented with red dashed line.
2. ES_BASIC_WLAN: This scenario was also used for enough the reference status of the fixed operator network. We established a call between SIP client on NB and the IP phone 1. The call was established via WLAN interface on NB. There was no other traffic generated in the access network and we were close to AP (i.e. very good signal). The RTP stream of scenario is presented with blue dashed line.
3. ES_TRAFFIC: In this scenario, first the call between SIP client on NB and IP phone 1 was established. In the middle of the call we started to produce heavy traffic between computers CPU1 and CPU2, causing congestion in the network. The RTP stream of scenario is presented with blue dashed line.
4. ES_WALK: First of all, the call between SIP client on NB and IP phone 1 was established. In the middle of the call we started to move with the laptop away from the access point and back. In this scenario no other traffic was generated in the network. The RTP stream of scenario is presented with blue dashed line.

In all four experimental scenarios no other devices were connected to the access point. To obtain a better understanding of QoS of the VoIP service we ran a Wireshark [21] analyzer on computer CPU3. At the same time we used Network Stumbler [22] on NB to measure signal and noise strength of the WLAN access point. We performed measurements several times; typical results are presented as follows. To compare the different experimental scenarios we used pre-recorded speech, which was played on NB during a call.

In experimental scenarios ES_BASIC_HSPA and ES_BASIC_WLAN the end-to-end delay and jitter of the VoIP call were measured (Table I). It can be seen that both networks performed well and there was no degradation of voice during the call.

Table I Results for basic experimental scenarios

Access	End-to-end delay		Jitter	
	Mean	Standard deviation (σ)	Mean	Standard deviation (σ)
HSPA	29.9 ms	15.7 ms	22.4 ms	182.2 ms
WLAN	29.9 ms	11.6 ms	17.6 ms	183.4 ms

Additional traffic was generated in experimental scenario ES_TRAFFIC, using UDP generator D ITG [23] on CPU1. The generated traffic was sent to CPU2. In Fig. 3 a traffic intensity model for UDP generator is presented. The packets sent had a constant payload size of 500 bytes. Including headers the total size of a packet was 542 bytes.

The results from experimental scenario ES_TRAFFIC are reported in Fig. 4. The red line represents link utilization. As can be seen, the link was almost congested between 120 and 190 s of the demo. The blue line represents utilization caused by RTP traffic (i.e. voice call) and represents around 2% of all link utilization. The green line represents end-to-end delay of the IP telephony service. All three parameters were measured with Wireshark on CPU3. It can be seen that end-to-end delay increases when link utilization approaches 100%. As the laptop computer remained stationary, the WiFi SNR ratio was constant at around 50 dB. During the congestion the quality of voice call was affected significantly (i.e. whole words were missing), thus conversation was almost impossible.

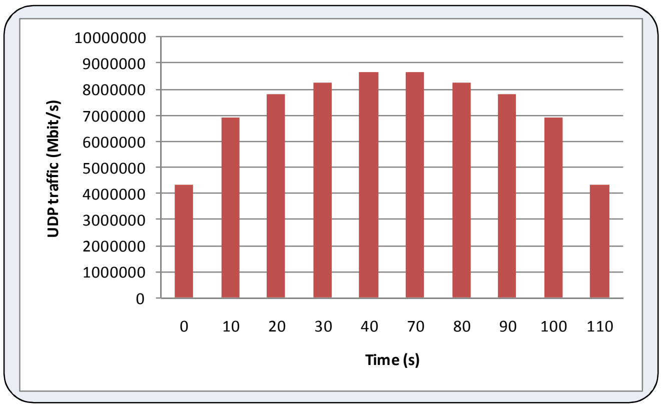

Fig. 3 Traffic intensity model for UDP generator

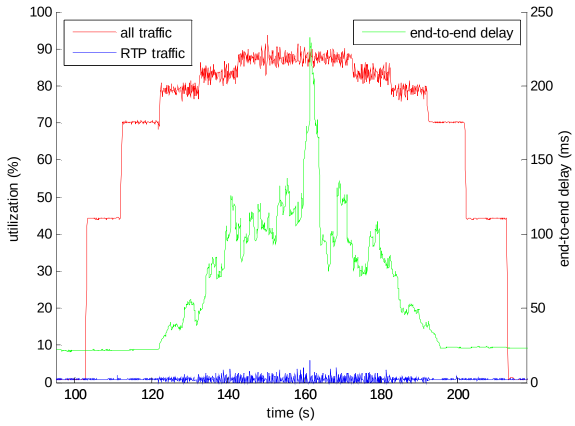

Fig. 4 Results of experimental scenario ES_TRAFFIC

The results of experimental scenario ES_WALK are presented in Fig. 5. The red line represents SNR ratio measured with NetStumbler on NB. The green line represents end to end delay of the IP telephony service measured with Wireshark on CPU1. Degradation of service becomes apparent when SNR fell below 10 dB. The low SNR resulted in high packet loss, thus the measurement of delay in Wireshark are not calculated properly, so the results after 590 s of experimental scenario are omitted.

The experimental results of the ES_BASIC_HSPA scenario lead to the conclusion that the new generation of today's mobile packet networks (i.e. HSPA) enables use of the IP telephony service in those networks. The results of experimental scenarios ES_TRAFFIC and ES_WALK show that the service was degraded significantly (i) when the link to the network was almost congested and (ii) when the SNR ratio fell below 10 dB.

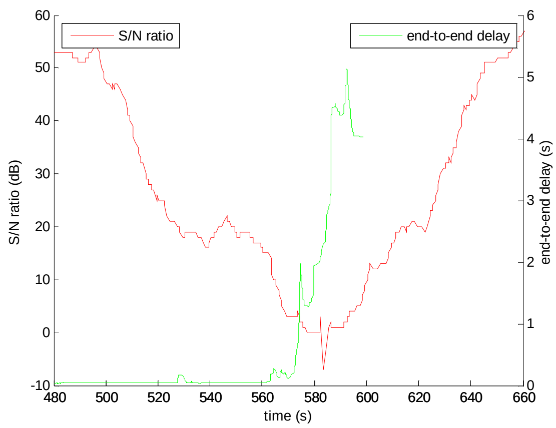

Fig. 5 Results of experimental scenario ES_WALK

We can summarize that QoS of VoIP service is affected not only by the SNR ratio, but also the by conditions in the WLAN access network. This is especially problematic since conditions in the network are not predictable and applications for VoIP do not support testing of the network. All these factors have to be considered when performing seamless handover between different networks. These measurements were carried out in order to be able to prepare a simulation environment that would be as close as possible to real environment characteristics and for testing advanced handover mechanisms.

B. Development of simulation model

There are several simulation tools available today on the market e.g. ns-2, pdns, SSFNet, J Sim, GloMoSim, OPNET. For our work we selected OPNET Modeler [24] simulation tool. It has an open source code of commonly used protocols, which is very convenient for performance evaluation of user developed / enhanced mobility management mechanisms [25]. It enables network modeling and simulation for designing new protocols and technologies, together with performance evaluation of existing and newly developed optimized protocols and applications. It has hierarchical modeling environment consisting of three levels: (i) project level, (ii) node level and (iii) process level. The project level graphically represents the topology of a communications network. It allows users to create node and link objects to represent network topology elements and configure them quickly. The node level captures the architecture of a network device or system by depicting the flow of data between functional elements, which typically represent network protocols or algorithms and are assigned process models to achieve any required behavior. The process level uses a powerful finite state machine (FSM) approach to support detailed specification of protocols, resources, applications, algorithms, and queuing policies. FSMs are dynamic and can be spawned during simulation in response to specific events. The C/C++ code that governs each state of a process model can be customized rapidly.

As not all requirements for our simulation model were met by OPNET, we developed some modifications to existing processes which can be divided in two main groups: (i) modification of links and terminals and (ii) modification of VoIP models. Both are shortly presented below.

C. Modification of links and terminals

Our main aim was to simulate handover in heterogeneous networks. In OPNET several wireless access technologies are already supported (e.g. UMTS, WLAN, Wimax). However we found some limitations in modelling real environment, as OPNET by default support ideal conditions. We intended to build such a simulation model where we can import signal measurements. In such a way, we can model user movements and user behaviour (i.e. user movements within the building, user entering elevator etc.). Some modifications were needed, which enabled us to import SNR measurements from real environments. In order to achieve that we defined generic terminal that connects to the network using IP links. By default OPNET does not support multimode terminals. But it offers tool by which it is possible to configure such a terminal. Terminal with two IP connections was defined. We defined access technology by choosing appropriate modified link. On links we applied link models such as error correction and error model that defined characteristics of particular network. This enabled us to simulate any access technology and also to import experimental SNR measurements of moving user in real environment. The SNR was measured with Netstumbler (Fig. 5 – red line) which enables export of data in CSV files. Measured values were parsed to fit in to pre-defined tables, which were then imported into OPNET.

D. Modification of VoIP processes

As we decided for handover on the application layer, which is not supported by OPNET, we customized some pre defined process models that incorporate SIP procedures to be able to support enhanced mid-call scenario with CAHP C procedure.

The OPNET's main process which is responsible for VoIP calls simulation was thus enhanced with handover described functionalities. New SIP messages were defined for testing the unreliable network prior the handover and also during the call. We also defined new interrupts for triggering the handover.

OPNET uses the FSM to implement the behaviour of a particular protocol/module. FSMs use states and transitions to determine what actions the module can take in response to an event. It consists of two types of states: A red state is one that returns control of the simulation to the Simulation Kernel after executing its executives. A green state is one that does not return control, but instead immediately executes the executives and transitions to another state. The Fig. 6 shows schematic FSM of the CAHP C procedure for congestion avoidance.

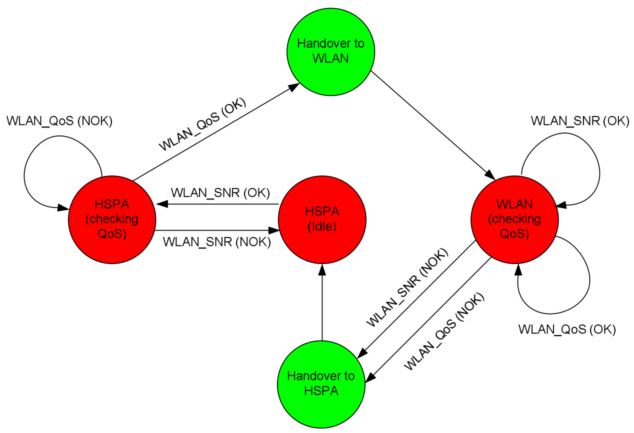

Fig. 6 FSM of CAHP C procedure

In our simulation scenario we preformed handover between WLAN and HSPA. We assumed that the HSPA network is widely available and has good QoS characteristics, while the WLAN network covers only limited area, is unreliable and can get congested likely. Based on user settings the WLAN network is prioritized. The initial call is established via HSPA network. When the call is established the process is in the state HSPA (idle). The SNR of WLAN network is constantly monitored and when it rises above predefined threshold, interrupt WLAN_SNR (OK) triggers the process to move to the state Checking QoS in which the WLAN network is tested prior handover as it is unreliable in terms of QoS assurance. We defined SIP pre_PROBE message that is used for WLAN network testing. After the SIP pre_PROBE messages are send the process waits for responses. When responses are received, QoS parameters are calculated. If QoS parameters are sufficient, interrupt WLAN_QoS (OK) triggers the process to the state Handover to WLAN where the handover is executed. After the new session is established process continues to the state WLAN (checking QoS). If calculated QoS parameters in the state Checking QoS are not sufficient, based on interrupt WLAN_QoS (NOK) the SIP pre_PROBE messages are sent again. If the SNR of the WLAN network falls below threshold while sending the messages or waiting for responses the process is returned back to HSPA (Idle) state with interrupt WLAN_SNR (NOK).

To avoid QoE degradation when the call is handed over to WLAN network the network is constantly checked with newly defined SIP mid_PROBE messages for congestion avoidance. Based on the responses the QoS parameters are calculated. If QoS parameters meet the recommended values the process stays in the state WLAN (checking QoS) and SIP mid_PROBE messages are sent again. If QoS parameters are not sufficient the handover back to HSPA network is triggered with interrupt WLAN_QoS (NOK) and it is executed in the state Handover to HSPA. After new session establishment the process returns to state HSPA (Idle). Besides end to end delay in the state WLAN (checking QoS) SNR is also monitored. If the SNR falls below threshold, handover execution back to HSPA is triggered by interrupt WLAN_SNR (NOK).

In states HSPA (idle), Checking QoS and WLAN (checking QoS) RTP packets are received and sent (states for receiving/sending packets are omitted in figure for simplicity).

E. Simulation scenario

The simulation model comprises two networks, WLAN and HSPA. Two hosts, MN and CN, that are using IP telephony as an application, were also defined. The G.711 codec was used for VoIP. Silence suppression was not used, resulting in a constant RTP traffic stream of 100 kbit/s.

The network architecture of the simulation model is presented in Fig. 7.

The following assumptions were made in the simulation scenario:

- HSPA network is always available.
- WLAN network has limited coverage (for example congress centres), but could probably become congested.
- WLAN network is prioritized, which means that MN will always try to perform a handover when WLAN network is available.
- MN is a dual mode handset capable of sending RTP packets and SIP INVITE messages at the same time via different interfaces. By this capability we lowered the overall handover execution time, since SIP re INVITE message was sent via the second interface while the first interface was still used for RTP traffic.

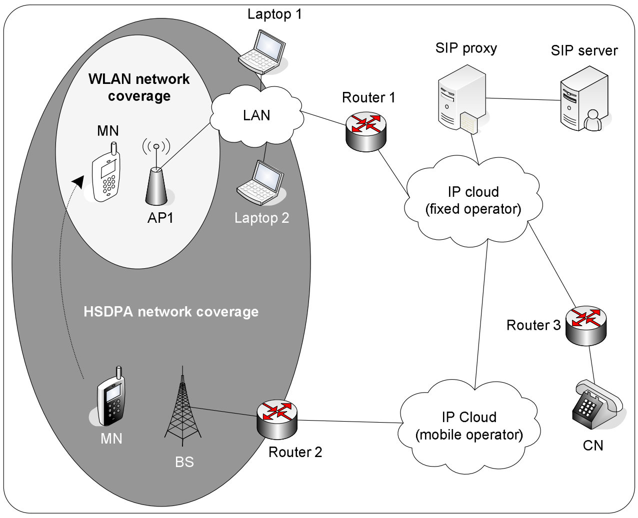

Fig. 7: Network architecture of the simulation environment

The MN is a dual mode terminal, capable of connecting to WLAN (representing a fixed operator) and to HSPA network (representing a mobile operator). Both networks are connected via IP clouds to the SBC. The CN is an IP phone connected to SBC.

To ensure that the simulation model is as similar as possible to a real operator environment, the parameters from the experimental scenarios presented in section 3.1 have been taken into account. The latencies in the IP clouds were set according to experimental measurements presented in Table 1. To increase traffic in the WLAN network, other clients were added (represented by laptops in Fig. 7), which generated additional UDP traffic in the LAN network.

Next we defined threshold that will be used as triggers for newly defined interrupts. End to end delays smaller than 200 ms do not affect QoS of real time applications, thus this value was used as the threshold. As presented in the experimental scenario in section 3.1, the QoS starts to decrease when SNR ratio falls below 10 dB; the threshold for SNR was thus set to 10 dB.

IV. Simulation Results

The focus of this paper was validation and verification of simulation model in comparison with real environment, thus the results in this chapter will be focused on validation and verification. In addition the results of using the CAHP C procedure for congestion avoidance using the developed simulation model will be presented briefly. More results are presented in [17].

A. Comparison of simulation model and experimental scenario results

In the first series of simulation results the simulation model was verified and validated. To be able to compare real measurements with simulation model results, first the experimental scenario ES_TRAFFIC was repeated in the simulation model. We generated same load of traffic that was used in experimental scenario. As the user did not move we set SNR to constant of 50 dB which does not have effect on QoS parameters. The access link utilization comparison between ES_TRAFFIC and simulation model is presented in Fig. 8.

It can be seen that simulated utilization resembles the experimental utilization sufficiently. Next we compared end to end delay of IP telephony. In the first simulation run we noticed that simulation model needs to be calibrated, as effect on QoS of VoIP was not the same as in real measurement. This happened due to hardware limitation in the hub that we used. Thus the capacity of the link was lowered for 15%. The results are shown in Fig. 9. It can be seen that the behaviour of delay in simulation model is similar to real measurement, thus approving our simulation model.

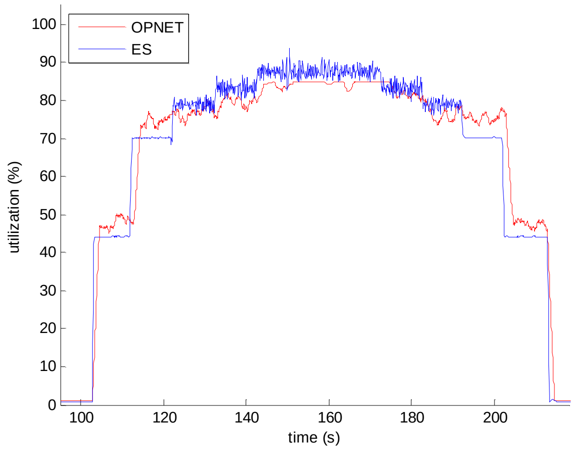

Fig. 8 Comparison of link utilization

Next the comparison between ES_WALK and simulation model was made. In this comparison we focused on packet loss during the conversation which was caused by insufficient signal strength. As in ES_WALK the SNR of WLAN network was measured, we selected cck-11 as a modulation on model of WLAN link. Each modulation has its own curve that defines BER. The BER values at different SNR ratios for cck-11 modulation are presented in Table II. As it can be seen for selected modulation SNR ratio which is lower than 10 dB increases the possibility for packet loss.

Table II cck-11 BER modulation table

SNR	(+0dB)	(+0.25dB)	(+0.50dB)
+ 5.00	0.0081	0.005	0.003
+ 5.75	0.0021	0.0014	0.00078
+ 6.50	0.00044	0.00025	0.00014
+ 7.25	0.000081	0.000044	0.000022
+ 8.00	0.00001	0.0000048	0.0000021
+ 8.75	0.00000081	0.00000037	0.00000016
+ 9.50	0.000000061	0.000000016	0.0000000056

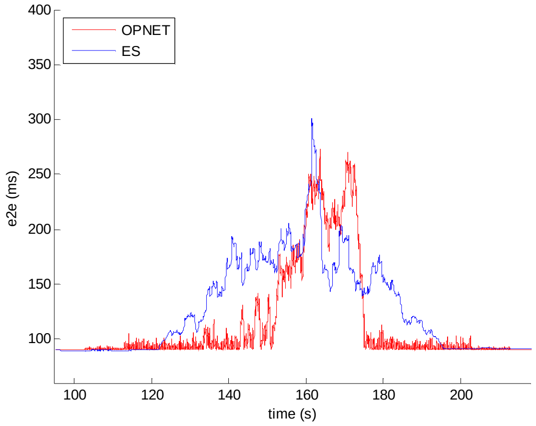

Fig. 9 Comparison of e2e delay

Packet loss measured in simulation model in correlation with SNR is presented in Fig. 10. It can be seen that packet loss increases when SNR falls below 10dB, which is expected result, approving our model.

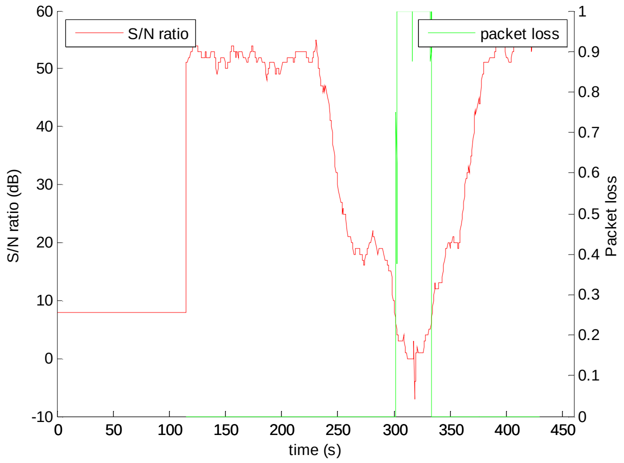

Fig. 10 Signal strength and packet loss in the simulation model

Besides the packet loss there was also end-to-end delay increase in ES_WLAK, as seen in Fig. 5. In simulation model we did not notice the delay increase. This is because OPNET works as an ideal system, where packets are only lost based on modulation curve. The others are not delayed.

From the comparisons it can be seen that there are some differences, but they do not have major impact on results, thus we can conclude that the modified simulation model works accurately enough for using the developed simulation model for evaluating advanced handover mechanisms using SIP protocol.

B. Performance evaluation of CAHP C procedure

The results for performance evaluation of CAHP C procedure will be shortly presented in this subchapter. A simulation scenario consists of a user starting a call in the WLAN network, which is unreliable and congested likely (e.g. a congress centre). In the middle of the call the WLAN network gets congested. After some time the user starts to move outside the area covered by public WLAN and handover is performed from WLAN to HSPA network. Then (still during the same call) the user starts to move back to WLAN coverage. As this network is also in use by other users, the network is temporarily congested when the handover process starts. The input parameters of the simulation scenario were combined from experimental scenarios ES_WALK and ES_TRAFFIC as depicted in Fig. 11. The ES_TRAFFIC scenario was modified with bigger packets (number of packets stayed the same) in order to get the link completely congested. We used 10 Mbit/s link and as seen from Fig. 11 the congestion occurred twice.

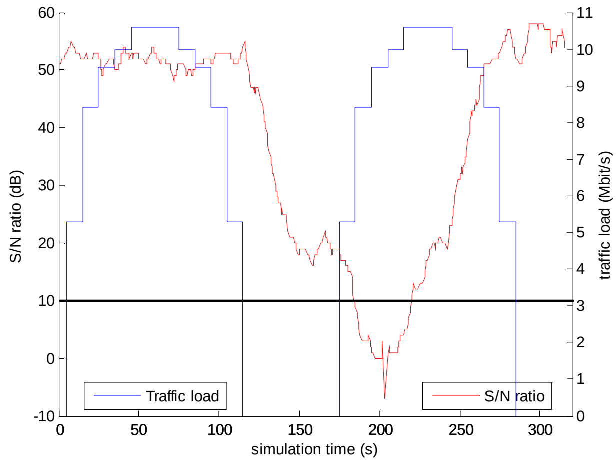

Fig. 11 Input parameters of the simulation scenario for WLAN network

The horizontal line shows a SNR ratio of 10 dB, which represents the initial trigger for starting the handover process.

The simulation scenario was run twice. In the first run the scenario without CAHP C procedure was used (named Basic scenario), while the second run was named Advanced as CAHP C procedure for congestion avoidance was used. Simulation results for both runs are presented in Fig. 12. Red line represents SNR ratio of WLAN network that was imported into OPNET. Blue line presents WLAN access link utilization. The utilization was caused by generated UDP traffic. Black and green lines presents network used in particular scenario.

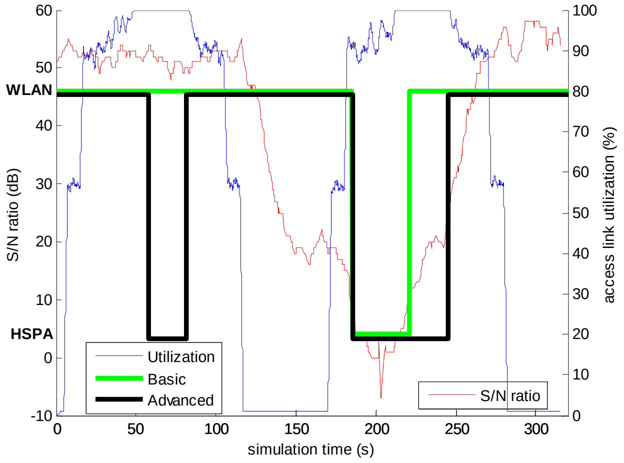

Fig. 12 Simulation results

Based only on SNR the handover (Basic scenario) is seen to occur twice. The first is from WLAN to HSPA network at 185.4 s of simulation time. The second handover is performed from HSPA to WLAN network at 220.6 s of simulation time.

In the Advanced scenario the handover occurred four times. The first and the second handovers occur due to congestion in WLAN network. The third handover was at the same time as in the basic scenario (due to SNR). The forth was delayed due to congestion in WLAN network.

Table III Summarized results of CAHP C procedure

Parameter	*Basic* scenario	*Advanced* scenario
Average end-to-end delay	158 ms	95 ms
Maximum end-to-end delay	624 ms	289 ms
Cumulative conversation time with the delay above 200 ms	20%	2%
No of handovers	2	4
Simulation time, when handover was executed	185.4 s (from WLAN to HSPA) 220.6 (from HSPA to WLAN)	58.0 s (from WLAN to HSPA) 81.4 s (from HSPA to WLAN) 185.4 s (from WLAN to HSPA) 245.4 (from HSPA to WLAN)

From the results it can be seen that with CAHP C procedure the user experience can be significantly improved compared to Basic scenario (handover based only on SNR), as there is lower average and maximum end to end delays and also the cumulative conversation time, with the delay above 200 ms decreases from 20 % to 2 %. The results also show that there are additional two handovers when CAHP C procedure is used (at 58.0 s and 81.4 s). This is because the algorithm detects the congestion in the WLAN network and performs handover back to the HSPA network. When the utilization of the link falls the handover is performed back to WLAN. When the second handover is triggered based on SNR, the algorithm detects that the WLAN network is congested and thus the execution of handover was delayed for 24.8 s.

The quantified results are presented in Table III.

V. Conclusions

In this paper we have presented simulation model for testing advanced mobility management techniques. The simulation model was developed in OPNET Modeler, where some predefined network elements needed to be enhanced. The verification and validation of the simulation model was done based on real operator environment testing. From the comparisons it can be seen that there are some minor differences, but they do not have major impact on results, thus we can conclude that the modified simulation model works accurately enough for using the results from the simulator for simulating handover using SIP protocol. We also presented advanced handover mechanism with congestion detection, based on delay testing. CAHP C procedure was tested in developed simulation model. The results show that user's QoE can be significantly improved when performing handover in unreliable heterogeneous networks. In order to improve handover decision algorithm, we will in our further work evaluate more parameters (triggers) which will give us even better view about target network status, allowing even more efficient handover execution, while the simulation model will be further expanded with abilities to incorporate even more parameters (e.g. battery consumption) from the real networks, which will be included in handover decision phase.

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