OMNeT++ Framework for Efficient Simulation of Vertical Handover Decision Algorithms

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***Abstract*—It is generally considered that the wireless networks of the future will consist of several radio access technologies. In this context, the vertical handover (VHO) decision algorithms will increase in importance and in complexity, and the study of these algorithms becomes a hot research topics.**

In this work we propose a high-level OMNeT++ framework that permits an efficient evaluation of various vertical handover algorithms. Our framework permits relatively short simulations, making it suitable for testing and evaluating different VHO algorithms.

The framework can be easily extended by including new network selection algorithms and new radio access technologies.

I. Introduction

In cellular networks a handover (or handoff) occurs when a mobile user disconnects from the base station that serves its current cell and connects to the base station that serves another cell. When the two cells belong to different network operators, or if they use different radio access technologies, the process is called vertical handover (VHO). The VHO process has three parts: handover information gathering (or network discovery), handover decision, or network selection decision, and handover execution [1]. In this work we concentrate only on network selection decision.

It is largely accepted that the wireless networks of the future (called also Next Generation Networks, or NGN) will consist of different radio access technologies (e.g. GSM/EGPRS, UMTS, LTE, WLAN, etc) and a common, IP based, core network. This model of the NGN will allow the mobile users to select a network based not only on the signal strength, but also on criteria like cost, transfer rate, error rate, battery consumption, geographic coverage, user preferences, etc. In this way the algorithms used for VHO, mainly for network selection decision, become very complex and their study is a hot research area. In this work we present the framework that we created in order to compare by simulation the efficiency of different VHO algorithms.

Because of their complexity, the algorithms used for network selection decision can be based on fuzzy logic (FL), neural networks (NN), multicriteria decision making (MDM) methods, analytical hierarchy process (AHP), etc.

There are also VHO algorithms based on combinations of the above mentioned methods, for example fuzzy TOPSIS in [2], fuzzy analytic hierarchy process [3], fuzzy and neural networks in [4].

It is difficult to compare the performance of such a variety of VHO decision algorithms, and we believe that an efficient and fair comparison can be realized only if the algorithms that we want to compare are simulated in the same framework and in the same conditions. In this respect, we have developed a framework for simulation of network selection algorithms.

Our first decision related to the framework was if to use a detailed simulation model, close to an emulation, based on the implementation of all protocols involved (TCP, IP, different radio protocols, etc), or a to create a model with a higher abstraction level. In order to avoid extremely long simulations, we have chosen to make a high level model of the problem, instead of using a very detailed simulation.

Another argument in the support of our decision to use a high-level approach is the very high complexity of the existing cellular networks (EGPRS, UMTS, LTE). All cellular networks include many protocols and nodes both on radio and on core network ([5], [6]). Also, the complexity of radio algorithms involved for data transfer, link adaptation, cell selection and reselection, is very high ([7]), and hence the task of realizing a very detailed simulation model of many radio access technologies becomes a very difficult task.

The rest of the paper is organized as follows: next section describes our framework, section III contains the simulation results that validate our framework, then we present some related work. The paper ends with a section of conclusions and future work.

II. The simulation framework

A. General considerations about our framework

We aimed to design a framework that has to fulfill the following requirements:

1) to be flexible and modular, in order to allow the inclusion of different radio access technologies (RAT) and VHO selection algorithms. The means that the framework should permit to configure a simulation model consisting of different radio access networks (RANs) that correspond to different RATs (e.g. EGPRS, UMTS, LTE, WLAN, etc) and to run different selection algorithms on that configuration in order to compare their results.
2) to be realized at a high level of abstraction, in order to avoid extremely long simulations.

3) to be able to include the significant details of a radio access technology and to increase the level of details, if necessary.

The most difficult decision in our model was the time level at which to work. This is because the events in the real system occur with very different time granularities: the scheduling in the radio networks happens at intervals of milliseconds (e.g. from 1ms in LTE to 20ms in EGPRS), the radio conditions change at hundreds or milliseconds or seconds, new data is generated at intervals of seconds, users move into or from the current cell at intervals of tens of seconds or even minutes.

We solved this problem by modeling the data transfer at the IP level: we consider that, during the transfer of an IP packet, the radio conditions remain unchanged.

We organize our framework on three levels, corresponding to the above mentioned requirements that our framework should meet.

The first level is the system level. It contains the main modules that compose our model of a VHO system: a data generator, a storage element, decision modules that contain the network selection algorithms, radio access networks and a sink. This level ensures a flexible and modular structure, where new RANs and new VHO algorithms can be added and different configurations of the system can be set. Subsection II-B contains a detailed description of this level.

The second level, described in subsection II-C, is a technology independent high-level model of a RAN, that can be used for any type of radio technology. It avoids the details of data transmission in radio access networks (e.g. channel allocation, scheduling, retransmissions, link adaptation, etc), and, in consequence, in this way we avoid extremely long simulations. We model the data transfer at the IP level: the length of a data unit (e.g., of a file) is decreased by the length of an IP packet. A delay element models the duration of the transmission of an IP packet.

Third level is a radio model of the changes of the network load and of the radio conditions, for each RAN. This model depends on the radio technology and in this work we present the models for EGPRS and for UMTS RANs (see subsections II-E and II-D). This part of the model can provide a more or a less realistic model of a certain radio access technology, and hence, it can influence the level of accuracy of the entire simulation model.

We believe that our model is a reasonable compromise between the accuracy of the technology description and the high level of abstraction that enables short simulations, in order to be able to test and compare the efficiency of different network selection algorithms.

For implementing our framework we use OMNeT++, a flexible and efficient discrete event network simulator [8]. OMNeT++ has a structural aspect, represented by composed modules that are hierarchical structures consisting on any number of levels, and a behavioural aspect, represented by simple modules, described in C++. The communication between modules is realized using *messages*. OMNeT++ contains tools for visualization and debugging, random number generators, statistics collection, etc.

B. The model of a VHO system

The simulation model of a VHO system is shown in figure 1, where we can identify the following Omnet modules:

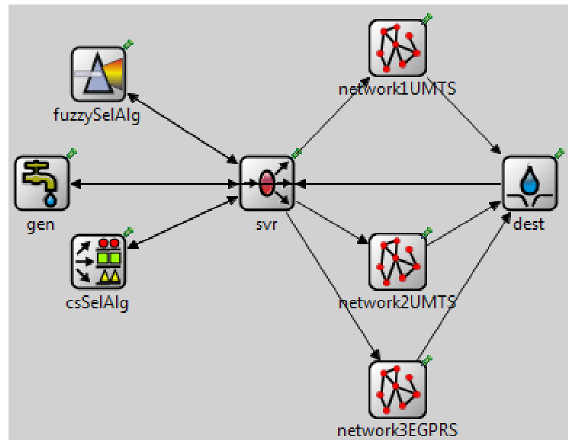

Fig. 1. The OMNET++ model of a VHO system

1) *gen:* the generator module. It creates OMNeT++ messages at certain time intervals, or in certain conditions. The messages can be considered to be ‘files’, where a ‘file’ can correspond to a real IP packet, or to a real file, of a certain length, or to a page of a WWW session, depending on the type of traffic that is modeled.
   If we model the background (e-mail, FTP), or interactive (WWW) traffic, then the generator is informed by the destination module (the sink) when the file has been completely sent, so that the generator can send the next file. If we want to model a streaming traffic, then the generator module will generate data packets at certain time intervals, and the server will have to store the packets waiting to be sent.
2) *svr:* the server. It is actually only a storage element. It stores the messages (i.e. the files) created by the generator module.
3) The selection algorithm is represented in the current implementation by two OMNeT++ modules, the *fuzzySelAlg*, that implements a fuzzy logic based VHO algorithm, and the *csSelAlg*, that contains the consumer surplus (CS) algorithm ([9]). The selection algorithm decides on which of the available networks to send the next ‘file’.
4) *dest:* the destination module. It is a sink type module, that collects statistics and deletes the OMNeT++ messages that model the files that arrive at destination.
5) the radio access networks. In the configuration shown in figure 1 we have two UMTS and one EGPRS networks: *network1UMTS, network2UMTS, network3EGPRS*. The user can chose one of them in order to send the next data packet.

C. The model of a radio access network

The general model of a radio access network is shown in figure 2.

The module called *dataBuffer* stores the file (the OMNeT++ message) received from the server module.

The effective transfer of a file is modeled using the delay element (*dlay* in the figure 2) and the *loop* module. If a part of the file (an IP packet) is send, the length of the file will be decreased by the length of the IP packet. When the file length is zero, it means that the file has been completely sent, and the corresponding OMNeT++ message will be sent to the sink (the destination module). The delay element models the time necessary to send an IP packet. This time depends on the transfer rate provided by the radio access network.

The transfer rate of the radio access network depends on the cell load, on the radio conditions and on the type of the network (i.e. EGPRS, UMTS). This is modeled by the module *extCondGen*. It is an OMNeT++ composed module, consisting on three sub-modules, as shown in figure 3 [10]:

- the *loadCondGen* module, for modeling the increase or decrease of the cell load.
- Radio condition generator (the *radioCondGen* module) models the quality of the radio link for the user.
- external condition analyzer: *extCondAnlyzr*. It receives the information from the previous two modules and, based on the specificity of the radio access technology, computes the transfer rate of the user. This information (the transfer rate) is sent to the delay element.

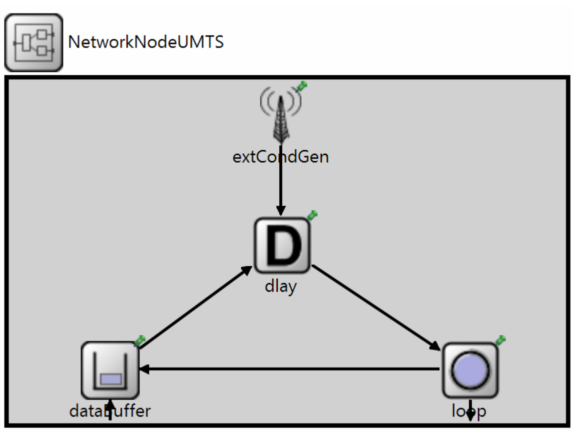

Fig. 2. The OMNET++ model of a radio access network

D. The model for an UMTS cell

The model of an UMTS cell considers that the cell has a certain capacity (of 1024 kps in this work), and that the cell has an initial load (for example 512 kbps, or 256 kbps). In our model, a user (mobile station, or MS) can have the following set of transfer rates, specific to UMTS cells without High Speed Uplink or Downlink Packet Access: {32, 48, 64, 80, 96, 112, 128, 192, 256, 320, 384} kbps. The module *loadCondGen* changes the network load (i.e. cell load) at certain time intervals, by increasing or decreasing the current cell load with one of these values. If a new user comes into the cell, then the cell load will be increased with the capacity used by this user, while if the user leaves the cell, then the cell load will be decreased with that value.

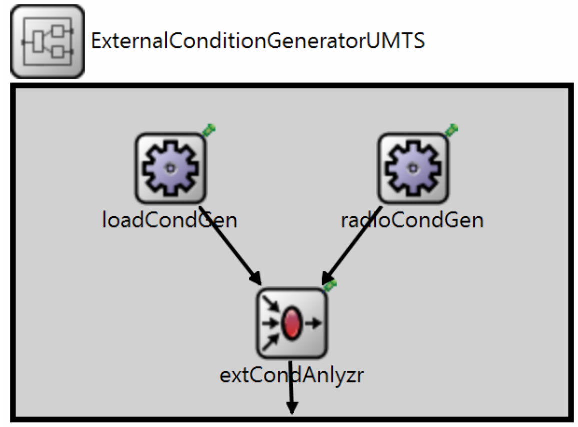

Fig. 3. The OMNET++ model of a the external condition generator module

The *radioCondGen* module models the quality of the radio link of our user, by randomly generating one of the above values (from 32 kbps to 384 kbps), which is denoted *the possible transfer rate* of the modeled user.

The *extCondAnalyzer* module combines the information received from *loadCondGen* and *radioCondGen* and computes the transfer rate of the modeled user as the minimum between the available capacity in the cell and the possible transfer rate of the modeled user.

For example, if the available cell capacity is 256 kbps and the *radioCondGen* generates a possible transfer rate of 32 kbps, then the transfer rate received by the modeled user will be 32 kbps, but if the possible transfer rate is 320 kbps, then the transfer rate received by the modeled user will be only 256 kbps. In our model this value remains unchanged during the transfer of an IP packet.

E. The model for an EGPRS cell

The model of an EGPRS cell considers that the capacity of the cell is given by one carrier (i.e. frequency) entirely dedicated to EGPRS, which means that the 8 time slots (TS) available on one carrier are allocated only for data. A user (mobile station or MS) can receive time slots only on one carrier at a time.

In reality a cell contains several carriers and the 8 time slots of each carrier are shared between data (EGPRS) and voice (GSM). Usually GSM has higher priority than EGPRS, which means that, in case of congestion, GSM can take time slots from EGPRS.

Several mobile stations (MSs) can be multiplexed on the same TS, and we denote by $MSperTS$ the number of mobile stations that share a time slot. It means that the capacity of that time slot is shared by all the MSs that are using it. In order to obtain a good transfer rate, the network operators use to limit this number to 5. Then, we can express the capacity of an EGPRS cell in parts of time slot (denoted $PartsOfTS$), obtaining 40 $PartsOfTS$ (8 time slots per cell multiplied by maximum 5 MS per TS). A MS cannot use more than 2 TS for uplink (UL) and 4 TS in downlink (DL). If we consider that we model a downlink transfer, then the loadCondGen module will increase or decrease the capacity of the cell by a number of $PartsOfTS$ between 1 and 4 at certain time intervals. An increase in the capacity means that an EGPRS user has left the cell or finished its data transfer session, while a decrease of the cell capacity means that a new user has arrived to the cell or it has started a new data transfer session.

The modeled user (mobile station) can receive a number of time slots (denoted $NbOfTS$) which is the minimum value between the number of requested times slots (which will be 4 for DL) and the number of available $PartsOfTS$.

We denote by $Parts\_Of\_TS\_in\_use$ the number of parts of time slots already used in the cell. Then we have to compute the number of mobile stations per time slots, i.e. $MSperTS$, which will be

$$\lceil Parts\_Of\_TS\_in\_use/8 \rceil$$

, where $\lceil x \rceil$ represents the smallest integer bigger than x.

The module *radioCondGen* will randomly generate the transfer rate per time slot (denoted $TrPerTS$), given by the modulation and coding scheme (MCS) that the modeled user can use for transmission, each MCS having a certain transfer rate. In EGPRS there are 9 modulation and coding schemes and the transfer rate per TS ranges from 8.8 kbps for MCS1 to 59.2 kbps for MCS9. The value of the transfer rate will result as the $NbOfTS \cdot TrPerTS/MSperTS$. This value will be sent by the *extCondAnalyzer* module to the delay element and will represent the momentary transfer rate of the user during the transmission or reception of an IP packet.

Although quite realistic, this model of the EGPRS RAN makes several simplifying assumptions, like: retransmissions due to errors are not considered, the radio link quality for a MS is considered to be the same for all its time slots, and the number of MS per TS is rounded in order to be the same for all the time slots received by a MS. We consider that these simplifying assumptions are acceptable and that they do not have a big influence on the final results. Some of the assumptions can be removed from the model, for example we can consider that the *radioCondGen* module provides also a certain block error rate, not only the MCS used by a MS.

III. Results

In this section we describe a consumer surplus (CS) algorithm, from [9], [11], and we use it in order to validate our framework by comparing our simulation results with the results obtained in [9] using a more detailed simulation model.

A. The Consumer surplus algorithm

The consumer surplus (CS) network selection algorithm has been proposed by Ormond et al in [9]. In [9], the algorithm is applied for non-real time traffic, like FTP, and we use the same type of traffic in this work. The main idea of the algorithm is to determine the predicted consumer surplus (i.e. benefit) for the user, for each file and for each of the candidate networks. The consumer surplus is the difference between an utility function U_i (the price that a user would pay for the transfer of a file i) and the cost charged by the network for the transfer of the same file, denoted C_i. A user sets for each file two threshold times, denoted $tc1$ and $tc2$, where $tc1$ is the threshold of the satisfaction zone and $tc2$ is the maximum tolerated delay [9]. This means that, if the delay of a file is smaller than $tc1$ the utility function has the maximum value (30 in our simulations), while if the file delay is bigger than $tc2$, then the utility function has a zero value (the file worth nothing for the user). Between the two thresholds we use a linear shape for the utility function, like in [9]. More elaborated functions can be used, as presented in [11].

The CS algorithm is implemented by the OMNeT++ module csSelAlg from figure 1. In order to predict the transfer rate of each of the candidate networks, the module stores the values of the delays of the last five files transferred on each network and computes a weighted average for the network transfer rate. The weights of the five delays are different: the weight for the first (i.e. the oldest) file delay is 5, then 10 for the second file, 15 for third file, 20 for fourth file, and 30 for the fifth file (the file that has been transferred most recently). At the beginning of the simulation, when there are no delay values stored, the transfer rate advertised by the networks is considered.

B. Simulation setup

We performed several simulations using the following configurations:

1) a configuration with two UMTS networks (cells), a ‘good’ UMTS network, with better radio condition, and a ‘bad’ UMTS network, where the radio conditions are worse than those of the first network.
2) a configuration with two UMTS cells, where the second network (the ‘bad’ one) has not only worse radio conditions, but it is also congested.
3) a configuration with an UMTS and an EGRS network (the UMTS network is the bad one, but it is not congested).

All simulations were run for 100000 seconds, simulation time, and for each file length there have been 10 different runs, the resulted values being the average between the 10 runs. The IP packets have a length of 1000 bytes in our simulations, in order to be compatible with [9].

The cell load is modified at intervals of 0.6 s for the ‘good’ UMTS network, and at 0.3 s for the ‘bad’ UMTS network and for EGPRS, with a value given by a normal distribution for the two UMTS networks and by a uniform distribution for EGPRS. The parameters of the normal distribution are: a zero mean value and a standard deviation of 150 in the case of the good network, and of 200 for the bad network. The initial load of the good network is 256 kbps, while for the bad network, the initial load is 512 kbps. It means that the average load of each cell will remain close to the initial value.

The difference between the good and the bad cell is given by the radio conditions. For the good cell, the possible transfer rate (see subsection II-D) is given by a normal distribution with an average value of 256 and a deviation of 100, while for the bad network the possible transfer rate is given by a normal distribution with an average of 80 and a deviation of 70. The radio conditions are modified at intervals of 0.4 s for the good UMTS network, and 0.2 s for bad UMTS network.

In the second case, when the bad UMTS cell is congested, the radio conditions are generated in the same way as in the previous case, but the change in the network load is given by a normal distribution with a mean of 80 (and a deviation of 200). The positive value of the mean of the normal distribution means that there will be more positive than negative values for the change of the cell load. This means, of course, that the cell load will have an increasing trend, hence the maximum cell load will be reached many times, which means that the cell is congested.

For the EGPRS network, the radio conditions are given at intervals of 0.2 s by an uniform distribution which gives equal frequencies to all the nine modulation and coding schemes. The cell load variation is given also by an uniform distribution (the intuniform OMNeT function) with values between -4 and 4, which means that the cell load will remain in average close to the initial value of 28 parts of time slot (see II-E).

We consider that in case of congestion, or in case of bad radio conditions, the modeled user still receives radio resources, but with a minimum transfer rate. This is according to a real-life situation, when a user, once admitted into a cell, is not dropped by the network, even in case of high network load. For the results presented here, the minimum transfer rate is considered to be 8.0 kbps for all networks.

C. Results

Figure 4 gives the average file delays for the case of two UMTS cell. The green line is for the case when the second UMTS cell is congested. The average file delays are represented in function of the file lengths, for files from 10 kB to 200 kB. The file lengths are expressed in bits, including the headers, as in the table 1 from [9] The difference between the good and the bad UMTS cells can be better seen for the longer files.

Our results are closed to the results from [9], table I. For example, for the files of 200 kB, we obtain an average delay of 7.57 s, on the good UMTS network, that is close to the delay of 8.48 s, from [9], table 1, corresponding to a transfer rate of 200 kbps. For the bad network, we obtain an average delay of 45.42 s for the 200 kB files, in case of non-congested UMTS networks, and 61.37 s for a congested network. These values are between the values from [9], of 33.93 s for a transfer rate of 50 kbps, and 84.82 s for 20 kbps transfer rate. Of course, the differences between our values and the values from Ormond et al, are given by the differences in the simulation model, but these differences are reasonably small.

For the configuration with one UMTS and one EGPRS cell, the results obtained for the EGPRS cell are very close to those obtained for the UMTS cell and hence we did not represent them.

Unlike [9], we have considered that a file that exceeds the tolerated delay (that we call a late file) is still accepted by the user, and we count these files. The percentage of late files from the total number of files transferred on each network is shown in figure 5 for the good and for the bad UMTS networks. The tolerated delay is 10s in our simulations.

Again, the results obtained for the EGPRS cell are very close to those obtained for the bad UMTS cell and are not shown here.

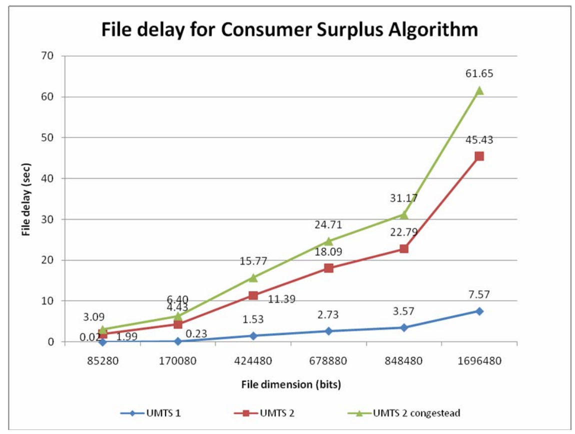

Fig. 4. File mean delay for different file lengths with CS algorithm and two UMTS networks

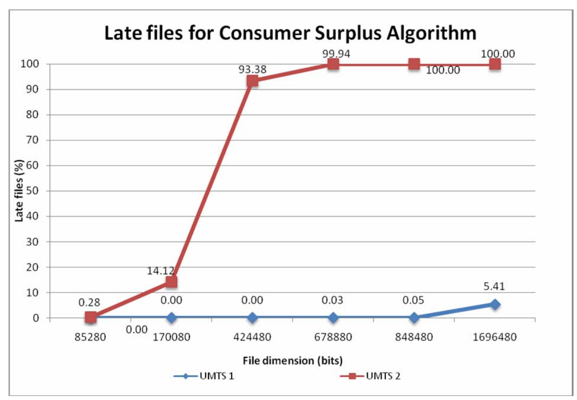

Fig. 5. The percentage of late files with CS algorithm and two UMTS networks

The duration of simulations on a laptop equipped with Intel Core i7-2670QM CPU at 2.20 GHz, Windows 7 Enterprise 64bit is only 266 seconds for the CS algorithm, for all the six file lengths, 10 runs for each file length, two UMTS cells, and 227 seconds for the case with one UMTS and one EGPRS RANs.

IV. Related works

Overviews of VHO techniques and VHO decision strategies can be found in [1], [12] and [13], while a survey and comparison of the weighting algorithms used for MDM techniques is given in [3].

Considering other VHO models presented in literature, they lay between high level models of the networks, like in [14], where the authors do not use simulations, only throughput models for the technologies involved (in their case, WiFi and WiMAX), to hardware platforms for mobile terminals, like in [15], or combination of hardware platforms and simulations, like in [16]. In their study "A User-Centric Analysis of Vertical Handovers" Calvagna and Di Modica ([17]) combine ns2 simulation with hardware implementation. A simulation framework for the evaluation of network selection algorithms, called mCASE, is described in [18]. However, mCASE is a proprietary simulation tool, which limits its availability to researcher community.

Our approach for RAN modeling is somehow similar to that from [19], where the authors use a hybrid simulation framework for modeling radio access technologies, i.e., they combine analytical models with discrete event simulation.

Our model can be compared to that used by Ormond et al in [9] and [11], in the sense that we consider also a user centric approach and, the same as in [9], we consider that an entire file is transferred on the same network. Also, in this work we consider the same type of traffic, i.e., FTP. While Ormond et colleagues use ns2 for their simulations, involving a detailed description of the protocols involved, we consider a higher level of abstraction in our OMNeT++ model. Also, we model here two different RANs, i.e., an EGPRS and an UMTS networks, not only one type of RAN (i.e. WLAN), like in [9], [11].

V. Conclusions and Future Work

This work presents an OMNeT++ framework for efficient simulation of the VHO process. We have implemented two radio access technologies, EGPRS and UMTS. In order to validate the simulation model, we implemented the consumer surplus algorithm from [9]. Our results for the consumer surplus algorithm are close to the results obtained by Ormond and colleagues in [9], using a simulation model more complex than our model.

As future work, we plan to include in our framework other radio access technologies (WLAN, HSDPA and HSUPA, LTE), to implement other network selection algorithms, and toto compare the efficiency of different VHO decision algorithms by performing a sensitivity analysis, in order to see for what values of the parameters is each algorithm better. This is quite a difficult task, due to the potentially very high number of parameters involved.

Also, we hope to obtain new VHO selection algorithms, starting from the algorithms presented in literature.

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