Multi-homing to Provide Reliable Connectivity

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Abstract *— There have been tremendous advances over the past decades when it comes to wireless access technologies. Nowadays, several wireless access technologies are available everywhere. Even mobile devices have evolved to support different access technologies in providing the best possible access to the internet. Smart phones and tablet PCs (e.g. Apple Ipad) for example have the capabilities to support different networks access technologies, like 3G, 4G or WiFi networks. Currently these mobile devices can communicate using one access technology at a time. However there is a big potential for improving network capacity and enhancing user Quality of Experience (QoE) if these access technologies are integrated together. Such integration would make access technologies to cooperate and work simultaneously in a heterogeneous environment from which both the end users, as well as, the mobile operators can benefit. With the help of these integrated networks, users can be provided with more reliable connectivity in disaster management scenarios. In this work, it is investigated how to tackle the simultaneous usage of wireless access technologies. For this purpose a practical example of 3GPP (3rd generation partnership project) LTE (Long Term Evolution) and non-3GPP WLAN integrated heterogeneous network is considered.*

Keywords: LTE, WLAN and QoE

I. INTRODUCTION

There are various prevailing standards of wireless access technologies in the current communication market, such as 3GPP, non-3GPP, and 3GPP2 etc. Admittedly, each type of these access technologies has certain advantages which justify its existence in this age of evolution of technology. For example, 3GPP networks are more efficient in terms of handling high traffic demands, providing QoS (Quality of Service) guarantees and the extended coverage. Whereas, the non-3GPP technologies like IEEE 802.11 are simple to operate and therefore need less investment and operation & maintenance cost. On the other hand, the wireless portable devices are becoming increasingly popular and it is widely expected that such devices will outnumber any other forms of smart computing and communication in near future. With the capability of connecting through several types of 3GPP and non-3GPP access technologies these devices run a wide variety of bandwidth demanding services including high speed data delivery and multimedia communication. However due to the limitations of today's network architecture these devices can connect to one access technology a time.

An integration of widespread non-3GPP technologies with existing 3GPP networks can provide several advantages in daily life as well as in disaster situations. Integrated 3GPP and non-3GPP networks, on one hand, can improve the overall network capacity and, on the other, can empower users with multi-homing i.e., the capability to connect to the network through more than one connection simultaneously. In a disaster situation the highest priority is to maintain user connectivity during the communication. Multi-homing servers this purpose very well as associating a user with a network through multiple access technologies makes his connectivity more reliable. This means if one of the connections is broken due to disaster situation, the user is still reachable through his other connections to the network. Moreover, the mobility management solutions (e.g., Mobile IP) ensure a seamless handover from one access technology to the other without interrupting the ongoing user communication. Multi-homing can also be beneficial in relieving user anxiety related to communication issues by enhancing the user QoE (Quality of Experience) in disaster situations. This is because multi-homing can take the advantage of network diversity and aggregated bandwidth resources from multiple access technologies to enhance QoS (Quality of Service) of user communication. Similarly multi-homing, with the help of clever algorithms improves the network capacity which allows serving more users in a disaster situation where available network resources are scares.

3GPP standardization has already envisioned the possible benefits from the cooperation of 3GPP and non-3GPP networks and has come up with such integration standards 0] (see Figure). Non-3GPP technologies can be integrated with 3GPP technologies through one of the three interfaces (S2a, S2b, S2c) provided by EPC / SAE. The description of the each interface is as follows:

S2a – provides the integration path between the trusted non-3GPP IP networks and 3GPP networks (The decision whether a network is trusted is not just based on the access network technology type but subject to the network operator’s policy). In this case the mobility is handled by the network based mobility solution, i.e. Proxy MIPv6.

S2b – provides the integration path between the un-trusted non-3GPP IP networks and 3GPP networks. In this case the mobility is also handled by the network based mobility solution.

S2c – provides the integration path between both trusted and un-trusted non-3GPP IP networks and 3GPP networks. In this case the mobility is handled by the host based mobility solution, i.e. Dual Stack MIPv6.

Though 3GPP SAE standard enables mobile users to roam seamlessly between 3GPP and non-3GPP access technologies, it does not support the user multi-homing. In order to investigate the achievable advantages through the support of user multi-homing, the existing 3GPP standards needs extensions. Moreover, the issues related to an efficient management of aggregated bandwidth resource to enhance user QoE and improve network capacity should also be addressed when multi-homing support is realized.

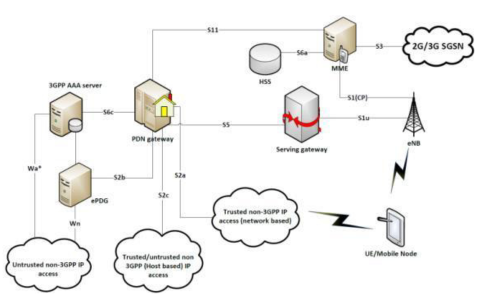


Figure 1: 3GPP standard for integration of 3GPP and non-3GPP technologies

The rest of the paper is organized as follows: Section II provides details the development of a heterogeneous network simulator according to 3GPP specifications, where 4G LTE and WLAN coexist. Additionally it is also explained how multi-homing support for users can be realized according to IETF (Internet Engineering Task Force) standards. Section III highlights the significance of flow management function in assigning network resources to multi-homed users. This section also explains the user bandwidth estimation techniques to assist the flow management function. Finally section IV presents simulation based case study to highlight the advantages of multi-homing support in future heterogeneous networks.

II. NETWORK SIMULATION MODEL

This work follows the proposal of 3GPP specifications in the integration of 3GPP access technology (namely LTE) and trusted non-3GPP access technology (namely legacy WLAN 802.11g) where host based mobility solutions, i.e. Dual Stack Mobile IPv6 is considered. For this purpose a simulation network model has been implemented using the OPNET [1] network simulator [0]. This includes the detailed implementation of LTE network entities following the 3GPP specifications. The details of the LTE network implementation can be found in 0] Simulation models of WLAN access points as well as, the common protocol layers like application, TCP/UDP, IP, MIP, Ethernet etc. comes from the OPNET standard library. As per 3GPP proposal the home agent (HA) function is located at the Packet Data Network (PDN) gateway. The remote server acts as a correspondent node (CN) from where mobile users access application services like VoIP, video, HTTP and FTP (see
Figure).

The OPNET model library implements the basic MIPv6 functionality. This implies that a mobile node may have several care-of addresses but only one, called the primary care-of address, can be registered with its home agent and the correspondent nodes. In order to support multi-homing for the users, an extension has been implemented according to the IETF RFC for multiple care-of address (MCoA) registration [0]. This enables the user to register the care-of addresses from all of its active network interfaces with its home agent. This work assumes that the user never attaches to its home network, and both LTE and WLAN networks are seen as foreign networks by the user as per 3GPP specifications. Therefore a user configures one IPv6 care-of address when it is in the coverage of LTE and another care-of address is obtained when WLAN access is available.

Though MCoA extension enables a user in this simulation environment to register up to two care-of addresses with its home agent, the user cannot communicate over the two network interfaces simultaneously. This is because MCoA recommends using only that single UE care-of address which has been most recently registered / refreshed. This calls for the need of another MIPv6 extension namely Flow Binding Support [0] that permits UE to bind one or more traffic flows to a care-of address. A traffic flow, in this extension, is defined as a set of IP packets matching a traffic selector [0]. Traffic selector helps identify the flow to which a particular packet belongs through the matching of the source and destination IP addresses, transport protocol number, the source and destination port numbers and other fields in IP and higher-layer headers. Traffic selector information is carried as a sub-option inside the new mobility option "Flow Identification Mobility Option" introduced by flow binding support extension.

In this way, a user in the simulation environment is capable of using all of its active network interfaces simultaneously. Depending on the configuration of the scenario, the user traffic flows can either be switched from one network interface to another or one large traffic flow can be split into two sub flows carried to the user over the two network interfaces. This will be discussed in greater detail in the next section.

Generally speaking with the help of above described MIPv6 extensions a user may have an arbitrary number of the active network interface to communicate through the available access technologies in a heterogeneous network environment. This work, however takes an example of the heterogeneous network where only two access technologies, namely the LTE and the WLAN, are available. Therefore a user can have maximum two active network interfaces, one for each access technology.

In order to perform flow management operation, all user traffic must travel through the HA during the whole simulation time. That's why "Route Optimization" option of MIPv6 is not enabled. Flow management function resides at HA and has two logical interfaces; one interface is used to receive necessary information from the base station of the two access technologies and second interface is used to interact with the extended MIPv6 function to manipulate user traffic flows at the two access technologies

III. FLOW MANAGEMENT

In the developed simulation environment a user can communicate simultaneously through 3GPP access technology i.e. LTE as well as non-3GPP access network i.e. WLAN. In general, there are two options of managing traffic flows for a multi-homed user. First option is to change the traffic route of one complete application traffic flow from one path to the other, this is known as "traffic flow switching". For example, a user can decide to keep his TCP based traffic flows on WLAN access network while his VoIP/video traffic follows its way over the LTE network. Second option is to divide a traffic flow into several smaller traffic sub-flows where each sub-flow is carried over one active network interface. This will be called "traffic flow splitting". For example, a user watching HD video streaming of a football match can distribute the video traffic flow over WLAN and LTE network as long as he is in the overlapped coverage of both networks. These sub-flows are then aggregated at the destination to reconstruct the original traffic flow of the application. Though the option of splitting a traffic flow involves more sophisticate techniques, it provides greater flexibility in network load balancing. That's why in this work, flow management with flow splitting option is implemented and analyzed with the help of simulations.

A very basic question that arises when deciding for flow splitting option is: What is an appropriate size of sub-flows transported over each path to a multi-homed user? In other words how much traffic should be sent to user on each available access link? And a straightforward answer would be: each path or link should be loaded according to its bandwidth capacity. Another important question is: how sub-flows will be aggregated at the receiving end. In the following sub-sections answers to these two questions are addressed.

From the architecture viewpoint the flow management functionality is split into two functional parts. One part resides at the network and the other part is implemented at user side. Each functional part receives the necessary information from "Information Management Entities", forwards the information to "Decision Making Entities" and then executes the decisions with the help of "Execution and Enforcement Entities". The flow management functional part on UE side (named as flow management client function) has the additional responsibility to respond to the information queries and decision execution commands sent by the flow management function residing at the network.

A. *Estimation of WLAN link capacity*

In 802.11 network where a number of users are contending for medium access, the network capacity and the individual user throughput is highly variable due to several time varying factors like number of active users, type of traffic flow (TCP or UDP), user channel conditions, as well as, mobility pattern of the users. This makes it extremely difficult if not impossible to mathematically compute individual user throughput in such a network. This work proposes two simple approaches to estimate user throughput along the time. First approach is valid only for downlink communication in which case any uplink traffic is transmitted through the LTE access. Second approach is applicable to a situation where many users are transmitting in uplink direction.

Downlink communication: This approach provides twofold benefits i.e., on one hand, it precisely estimates bandwidth resources of WLAN network, and on the other hand, it manages the network resource assignments to the users in an efficient manner. In this scheme, the flow management function residing in the network does not require the participation of flow management client function of UE.

Legacy WLAN (802.11a/b/g) provides no QoS when scheduling user traffic. Essentially there is only single queue in a WLAN access router where all incoming traffic is received and then transmitted over the air to users in a "First Come First Serve (FCFS)" manner. That's why throughput of a hotspot and that of users being served is highly variable based on number of active users in the system, their offered traffic load as well as their channel conditions. This scheme modifies the FCFS behavior of WLAN MAC by introducing a quasi-packet-scheduling based on number of active users associated to an access point as well as their PHY data rates. The quasi-packet-scheduling is performed by flow management function at home agent which is the anchor point for all user traffic. In order to perform this function flow management requires the knowledge of user data rates at physical (PHY) layer. This information is readily available at the hotspot through the cross layer communication.

The proposed scheduling scheme resembles the well-known Time Domain Multiple Access (TDMA), where users are given equal share of time slices during to transmit a certain number of packets depending on their PHY data rate. Consider an example of two users associated with a hotspot; first user can transmit with 56Mbps PHY data rate and the second can operate at 6Mbps PHY data rate. Considering basic channel access mechanism of 802.11, time to transmit one IP packet of 1500 bytes with 6Mbps PHY data rate is approximately equal to transmitting six IP packet of the same size with 56Mbps PHY data rate. In this way, if flow management function sends six packets from the first user traffic flow and only one packet from second user traffic flow, the desired TDMA scheduling effect with equal time sharing can be achieved at the hotspot. Following this scheduling scheme WLAN network throughput λ_{AP} and user throughputs λ_{user} can be computed as below

$$\lambda_{AP} = \frac{\sum_{i=1}^{N} w_i \cdot p_i}{\sum_{i=1}^{N} w_i \cdot t_i} \quad \text{and} \quad \lambda_{user\, i} = \lambda_{AP} \cdot w_i$$

$$r_i = \frac{p_i}{t_i} \quad , \quad w_i = \frac{r_i}{\sum_{i=1}^{N} r_i}$$

where t_i is the time duration required to transmit a packet of size p_i bits to user i operating at a certain PHY mode. N is the total number of active users associated to the hotspot.

The above described method of scheduling WLAN bandwidth resources is just one of the possible ways. In general any scheduling scheme can be imposed using flow management function at home agent.

Uplink communication: In this approach a metering function, as a part of information management entity (IE), is introduced at the WLAN MAC layer buffer of the user device. This metering function is responsible for measuring the outlet data rate from the buffer, as well as, the buffer occupancy level. These two values are obtained periodically and sent to flow management client function residing at UE. In the beginning, flow management client function directs a sufficient amount of user traffic to WLAN MAC for transmission. This data stays at the MAC layer buffer before transmission over the radio interface. The flow management client function now continuously receives the buffer occupancy level reports and accordingly adjusts the size of traffic flow to the WLAN network path to keep the buffer occupancy at the desired level. In this way, if buffer occupancy level increases, it hints a reduction in available WLAN path capacity due to some reason e.g. congestion, poor channel conditions etc. Therefore flow management function accordingly reduces the traffic flow amount directed towards the WLAN path. The opposite is true if a reduction in MAC buffer occupancy is observed which suggests an improvement in the path capacity. The flow management client function takes advantage of this by sending more traffic towards the WLAN path. Following this approach time varying WLAN network path capacity can be tracked and used by the flow management function.

The practical implementation of the suggested approach is straightforward as, it does not require any modification in the UE hardware or WLAN MAC protocol. However an appropriate amount of the MAC buffer occupancy μ_i for a smooth operation of this approach still has to be determined. For this purpose following relation is used for every user *i* i.e.,

$$\mu_i = \gamma_i^{WLAN} \cdot \tau \qquad (1)$$

where γ_i^{WLAN} is the throughput estimation of WLAN path and τ is the sub-flow aggregation timer. The concept of sub-flow aggregation will be introduced later in this paper. For a TCP based application flow $\tau = \tau_{tcp}$ which is the TCP re-ordering timer as discussed later. For real time applications like, VoIP or video, $\tau = \tau_{de-jitter}$ (which is the length of de-jitter buffer in units of seconds) is used for that particular application.

B. Estimation of LTE link capacity

LTE radio interface, the interface between UE and eNB consists of four main protocol layers to transfer the data between eNB and UE securely and with certain reliability. They are Packet Data Convergence Protocol (PDCP) layer, Radio Link Control (RLC) layer, Medium Access Control (MAC) layer and Physical (PHY) layer. All these four protocol layers are part of the protocol stack in UE and eNB. At the eNB side, each radio bearer has one PDCP entity which processes the IP packets in the user plane. The PDCP layer mainly performs header compression/decompression, security including data integrity/verification and ciphering/deciphering, and also reordering during handovers. The main data buffers of Uu interface, in downlink direction, are located at the PDCP layer where the data is held before its transmission to the destination UE. Each PDCP entity has its own PDCP buffers implemented on a per radio bearer basis. Similarly at the UE side, the shared PDCP buffer capacity is allocated on a per-bearer basis for the uplink transmission.

LTE link throughput of a user depends on several factors e.g. MAC scheduler type, channel conditions of all users, QoS requirements of traffic from all users, cell load level etc. In order to avoid complex user throughput computations based on aforementioned factors, a simple mechanism is introduced. When the MAC scheduler at eNB selects a user for transmission in a particular time slot, a certain amount of data flows from PDCP buffer to RLC buffer and then to MAC layer from where it is transmitted over the air interface. Following this process, if a metering function is introduced between PDCP and RLC an estimation of the user throughput can be obtained. However the question still remains how much occupancy at PDCP buffer should be achieved; large buffer occupancy would cause high queuing delays while small buffer occupancy could lead to buffer under-runs. We propose to use sub-flow aggregation timer τ to control the total link delay. This scheme is similar in functionality to the technique used for link capacity estimation of WLAN in uplink direction. According to the proposed scheme, initially the data sent over LTE link by flow management function is increased in small steps and the corresponding PDCP buffer occupancy is observed. When PDCP buffer occupancy reach a certain level which causes a queuing delay less than sub-flow aggregation timer, the further increments in sent IP traffic are stopped. Now when user air interface throughput reduces due to some reason (e.g. cell overload or bad channel conditions) the egress data rate from PDCP buffer becomes lower than the ingress data rate. This in turn makes PDCP buffer occupancy to increase. In this case flow management function reduces the IP traffic sent to LTE link until a stable PDCP buffer occupancy level is achieved. The opposite is true when decrease in PDCP buffer occupancy is noticed. The amount of target buffer occupancy ξ_i for a user *i* is decided dynamically based on current estimation of his throughput α_i^{LTE} and sub-flow aggregation timer value τ. i.e.

$$\xi_i = \alpha_i^{LTE} \cdot \tau \tag{2}$$

The purposed scheme is applicable for the LTE link capacity estimation both in uplink and downlink directions.

In multi-path communication packet may arrive out of order at the destination [0]. Real time applications usually deploy a play-out (or de-jitter) buffer which is intended to get rid of jitter associated with packet delays. However it can also perform packet reordering if packets arrive within time window equal to play-out buffer length. In this way, real time applications face no problems when receiving out of order packets in multi-path communication unless delay of all paths is less than play-out buffer length τ_{de_jitter} On the other hand TCP based applications are very sensitive to packet re-ordering. This is because an out-of-sequence packet can lead to TCP overestimate the congestion of the network which results in a substantial degradation in application throughput and network performance [0]. A literature survey shows that there are several proposals to make TCP robust against packet re-ordering [0]-[0]. However, the analysis and implementation of such schemes are currently not within the focus of this research work. Instead we implement a simple TCP re-ordering buffer at the user side which is very similar in functionality to a play-out buffer. Simulation analysis shows that the re-ordering buffer length must be less than the TCP protocol time out value. In this work the re-ordering buffer length has been kept between 100ms to 500ms. TCP re-ordering buffer length τ_{tcp} is a key factor in deciding target PDCP buffer occupancy level ξ_i and WLAN MAC buffer occupancy level μ_i for a user *i* as explained earlier. With the help of this strategy WLAN and LTE link delay is controlled not to exceed the TCP re-ordering buffer length and hence avoid unnecessary TCP timeout.

IV. SIMULATION RESULTS

This section presents the findings of a case study performed with the help of developed simulation environment. For this purpose two scenarios are considered. In one scenario users do not make simultaneous use of LTE and WLAN access technologies. Instead user traffic is completely handed over to WLAN on entering in the hotspot coverage which otherwise follows its route over the LTE link. This is default policy for a multi-homed user according to 3GPP specifications and therefore it will be referred as "3GPP HO" case. Whereas, the second scenario extends the 3GPP architecture into supporting the simultaneous use of wireless interfaces, this will be referred to as "Multi-P". In this case user traffic flow is split into two sub flows when user has both WLAN and LTE access network available. The policy here is to push traffic as much as possible over WLAN and any deficits can be fulfilled using LTE access technology.

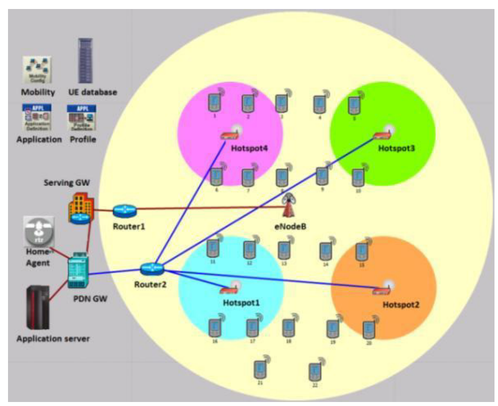


Figure 2: OPNET simulation model example scenario. The large circular area denotes the coverage of LTE network and smaller circular areas represent the WLAN coverage.

In order to distinguish the performance gains provided by multi-homing and flow management in uplink and downlink communications, two simulation analysis are presented. In first simulation analysis only the downlink communication is focused while second analysis is dedicated to uplink communication.

A. Downlink communication analysis:

The system is populated with 20 users generating a rich traffic mixture i.e. 5 Voice over IP (VoIP) users, 5 downlink File Transfer Protocol (FTP) users, 5 Hyper Text Transfer Protocol (HTTP) users, 5 video conference (i.e. Skype video call) users, and 3 High Definition (HD) video streaming users. The users move within one LTE eNB cell, and within this cell four wireless access points or hot-spots are present as shown in Figure 3. The simulation configuration parameters are shown in Table .

It is worth mentioning here that in “3GPP HO” scenario user undergoes through vertical handover which is of hard nature, i.e., the connection is broken from one network, and a new connection is established to the other one. Though MIPv6 keeps all IP layer connections alive through seamless handover, user might lose some buffered data on the previously connected network. On the other hand “Multi-P” scenario makes users use the WLAN access when it's in the coverage, and can still keep the LTE connection and use it at the same time. As a result, a bandwidth aggregation process of both wireless links is done.

Figure (a) shows the spider web graph of the average delay values of user applications. A spider web graph is a visualization technique that can show multiple results in one graph, and is used to compare different scenarios. The graph in Figure (a) has five different axes, each representing one performance metric, mainly VoIP, Skype video, HD video end-to-end packet delay, FTP file transfer time, and HTTP page response time.

Table 1: Simulation scenario configuration

Parameter	Configuration
LTE spectrum	10 MHz (50 PRBs)
Mobility model	Random Way Point (RWP) with 6 km/h
LTE Channel model	Macroscopic path loss model 0], Correlated Slow Fading, and Jake's Like Model fast fading.
LTE MAC scheduler	TDS: Optimized Service Aware, FDS: Iterative Round Robin approach
WLAN access type	802.11a, RTS-CTS enabled, coverage ~100 m, non-overlapping channel
VoIP traffic model	G.722.2 wideband codec with 23.05kbps data rate and 50 fps, one voice frame per IP packet, play-out delay: 150ms
Skype video model	MPEG-4 codec, 512kbps, 30fps, 640x480 resolution, play-out delay: 250ms
HTTP traffic model	100 bytes html page with 5 objects each of 100Kbytes. Page reading time: 12s
HD video model	MPEG-4 codec, 1Mbps, 30fps, 720x480 resolution, play-out delay: 250ms
FTP traffic model	File size: 10MByte, when one file finishes the next FTP file starts immediately
Simulation run time	1000 seconds, and 14 random seeds, 98% confidence interval

Since all the axes represent delay, the algorithm producing the smaller shape has the best performance. In this case, it is clear that the “Multi-P” algorithm achieves the best results for the VoIP, FTP and videos user traffic. As for the HTTP performance, it can be seen that the “3GPP HO” scenario has a lower HTTP page response time. The reason behind this is the fact that, when the 3GPP algorithm switches from one access technology to another, it immediately starts sending the full amount of traffic. Whereas, “Multi-P” algorithm performs the link capacity estimation by a slow step up increase in the amount of traffic until reaching the maximum capacity of the networks. Since HTTP files are relatively short, the page download complete before full capacity of the link is estimated.

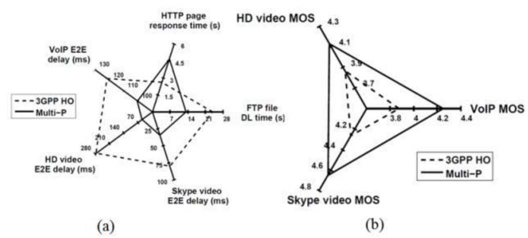
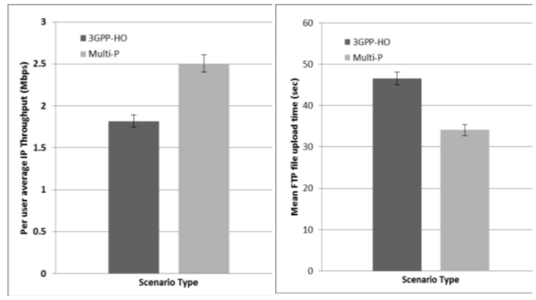


Figure 3: Downlink user application performance

In order to have performance evaluation comparison between the two scenarios, the MOS (Mean Opinion Score) [7] values for the VoIP and the video traffics are shown in Figure (b). The MOS values of VoIP and video applications are computed following the procedures listed in [0] and [0] respectively. The results show that the “Multi-P” algorithm provides very good performance for the VoIP, as well as, for both video traffic types (Skype and HD). The “3GPP HO” scenario achieves lower MOS value because when the users move from one access network to the other (from LTE to WLAN, and vice versa), the buffered data in the previous network is lost, and this affects the quality of the services significantly as reflected in the MOS value. Furthermore when VoIP and video traffic is transmitted over LTE, it is prioritized over FTP to achieve required QoS (i.e. throughput and delay). But when in “3GPP HO” scenario when this traffic type is handed over to WLAN the required QoS cannot always be achieved due to lack of QoS differentiation support by 802.11g. Thanks to algorithms of “Multi-P” scenario which manages 802.11g resources in a way that not only the required QoS for real time traffic is met but also the optimum throughput performance of WLAN access point is accomplished. Moreover in “Multi-P” scenario the loss of buffered data in network is avoided in the following manner. (i) LTE connection is always kept alive hence no buffered data is lost there. (ii) in WLAN, the flow management function sends user traffic over WLAN link only when user PHY data rate is 9Mbps or higher. This is because when a user enters in 6Mbps mode it implies users is almost at edge of coverage which is a strong indication that loss of WLAN link is imminent. Hence no new traffic data is sent on WLAN link for that user which gives him a chance to receive already buffered data from the access point before the loss of link.

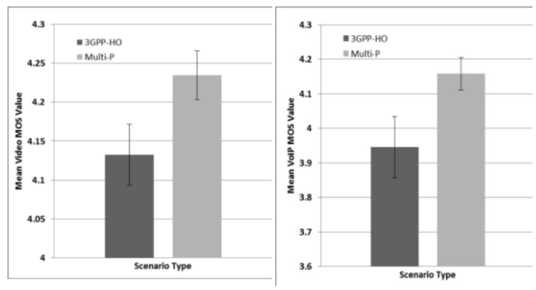
B. Uplink communication analysis:

In this analysis, the simulation setup is composed of 20 users generating mixed traffic i.e. 5 Voice over IP (VoIP) users, 11 uplink File Transfer Protocol (FTP) users and 4 Skype video streaming users. As mentioned earlier in “3GPP HO” scenario the users make handover between two access technologies without following make-before-break approach which leads to loss of data packets buffered at the lower protocol layers of the previously connected network interface. For example, LTE buffers the received IP packets at PDCP, RLC and MAC layers while WLAN keeps all the data buffered at MAC layer before transmission over the radio interface. Therefore when making complete handover from one access technology to another, this buffered data is discarded and has to be recovered by upper layers through retransmissions. This behavior leads to applications performance degradation for both TCP and UDP based applications. “Multi-P” approaches, in contrast to “3GPP HO” scenario, keeps buffered data at the minimal required level through the use of network path capacity estimations to avoid big data losses.



(a) IP throughput of FTP user (b) Mean file upload time

Figure 4: FTP uplink performance



(a) Mean MOS value of wideband VoIP application (b) Mean MOS value of Skype video application

Figure 5: VoIP and video application performance

Figure (a) shows the mean IP uplink throughput as experienced by a FTP user. It can be seen that the “Multi-P” approach manages to provide approximately 38% higher IP throughput compared to ``3GPP HO" scenario which helps him achieve lower file transfer time as seen in Figure 5(b). This indicates that “Multi-P” approaches outperform the “3GPP HO” approach by making better use of aggregated bandwidth resources through network path capacity estimation.

Figure shows the MOS values of VoIP and video applications as experienced by the users. The results show that the “Multi-P” algorithms provide very good performance for the VoIP, as well as, for video conference traffic type. The “3GPP HO” scenario achieves lower MOS value because when the users move from one access network to the other (from LTE to WLAN, and vice versa), there is a chance of some data loss as explained earlier, and this affects the quality of the real time services. Similar to downlink communication, LTE always prioritizes real time traffic over FTP to achieve required QoS (i.e. throughput and delay). But in “3GPP HO” scenario when real time traffic type is handed over to the WLAN the required QoS cannot achieved due to lack of QoS differentiation support by 802.11g. Thanks to algorithms of “Multi-P” scenario which estimates and manages 802.11g resources in a way that not only the required QoS for real time traffic is met but also an enhanced throughput performance of WLAN access point is accomplished.

V. CONCLUSIONS

3GPP SAE architecture specifies how non-3GPP access technologies can be integrated in 3GPP networks and a seamless handover between these access technologies can be performed. This work proposes an extension to the specifications to allow a user benefit from all available access technologies by connecting to them simultaneously. The multiple user connections to the network help achieve reliable connectivity, enhanced QoE as well as improved network capacity in disaster situations. Through the use of suggested algorithms for resource management and accurate network bandwidth capacity estimation the achievement of aforementioned benefits is realized. In order to validate the proposed algorithm and procedures an implementation of integrated network of LTE and legacy WLAN access technologies in OPNET simulator has been carried out. The simulation results provide proof of the concept where the proposed scheme succeeds not only in providing QoS aware service to multi-homed user but also improves the network bandwidth resource utilization.

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