QoS Aware Multi-homing in Integrated 3GPP and non-3GPP Future Networks

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***Abstract*—In future networks, collaboration of heterogeneous wireless access technologies is inevitable. 3GPP has already standardized the interconnection of non-3GPP access technologies with the exiting 3GPP access networks. This enables users to make seamless vertical handovers between available access networks. However, the question still remains whether the non-3GPP access technologies can deliver the Quality of Service (QoS) demands of user applications? Within the context of the Open Connectivity Services (OConS) of the SAIL European project, this work investigates the effects of the integration of two network types on user Quality of Experience (QoE). In order to accomplish QoS guaranteed service from non-3GPP access technologies, this paper proposes a novel resource management algorithm. With the help of simulation results it is proved that network operators can get significant performance boost for their networks when the proposed scheme is deployed in an environment with heterogeneous access networks.**

Keywords: ***LTE and WLAN interworking, Efficient resource allocation, User QoE optimization, Multihoming in wireless heterogeneous networks***

I. Introduction

The EU-funded research project SAIL (Scalable & Adaptive Internet Solutions) research the design of the Networks of the Future, investigating new architecture for the future internet. The SAIL project is part of the European Commission's 7th Framework Program [1]. 24 operators, vendors and research institution are working together since 2010 on the research and development of novel networking technologies using proof-of-concept prototypes to lead the way from current networks to the Network of the Future [2]. The work of this paper falls within the solutions that the OConS work package is providing. The work ranges from concepts of OconS architecutre framework and Multi-P transmissions in LTE systems. The OConS approach is working on the proposal of an open and flexible architectural framework to handle the connectivity of networks. To fulfill the requirements and to keep the flexibility and openness of OConS, a component-based architecture frame-work is proposed, in 3 steps: (1) information collection; (2) decision making; (3) decision enforcement. These steps are handled by three functional entities:

- Information Management Entity (IE) is in charge of gathering the information required by decision making, e.g., link quality, power limitation, load and congestion of the networks. IE also pre-processes and filters the gathered information before it is delivered to the other entities.
- Decision Making Entity (DE) uses the information from IE to make the decision to fulfill some pre-defined metrics. For example, the final enforcements can be handover, load balancing and flow splitting. The goals of improving system performance can be maximizing the network throughput, balancing the load etc.
- Execution and Enforcement Entity (EE), finally, executes or performs the decision made by DE.

A. Motivation and Contributions

Nowadays, the end-user equipments are more and more powerful, and normally, have more than one interfaces which can connect to different wireless networks, e.g., 3G/4G mobile systems or wireless LAN (WLAN). On the other hand, the network resources are always scarce for supporting a huge number of users running different applications. The Multi-P algorithm, enables the interconnection of LTE and WLAN, so that end users can exploit all of the available wireless resources, meanwhile operators can also flexibly balance the traffic load among multiple access networks.

In this presented work, besides the general introduction to SAIL, OConS and Multi-P we put our focus on a simulation model realizing the Multi-P transmission of 3GPP LTE and WLAN. This involves development of simulation model according to 3GPP specifications, implementation of MIPv6 extensions to realize multi-homing and flow management techniques as well as the integration of user QoE evaluation tools in the simulator. With the help of intelligent resource management schemes we contribute towards network capacity improvements and user QoE enahancements.

The rest of this paper is organized as follows: Section II introduces the details of the OPNET[1] [20] simulation model for Multi-P. The traffic flow management strategies and the detailed mechanisms are described in Section III. For proof of concept, Section IV shows simulation results of different scenarios to reveal the advantages of our approach. At the end, Section V concludes this work and points out some possible points for our future works.

[1]OPNET Modeler®is a commercial network simulator which accelerates the R&D process for analyzing and designing communication networks, devices, protocols, and applications.

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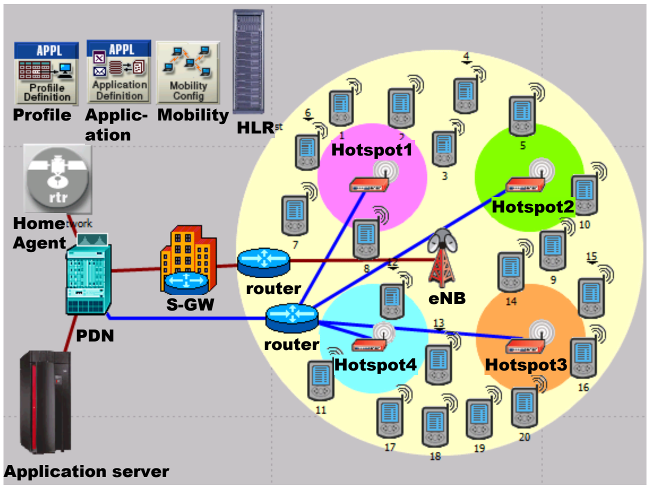

Fig. 1. Overview of the considered scenarios in OPNET. The large circular area represents the LTE access coverage. The small circular areas represent the coverage of WLAN access technology which overlaps LTE access coverage.

II. System Models

A. Applying Multi-P of OConS into interconnection of 3GPP LTE and WLAN

The goal of the presented work is to investigate the Interconnection (and cooperation) of 3GPP (e.g., LTE) and non-3GPP systems (e.g., Wireless Local Area Network (WLAN)) by means of the Multi-P transmission. The key part of this approach is the decision process: selection of interfaces of the UE and access alternatives (at the network side) that should be used for the transmission and how to manage the traffic flows.

From an architectural point of view as depicted in Fig. 1, the interconnection between 3GPP and non-3GPP (in our case we have chosen LTE and WLAN as case study) is possible at the Packet Data Network Gateway (PDN GW). Hence, it becomes a reasonable location for the DE of the network controlled Multi-P decision. For the LTE side, the IEs are located at the eNodeB (eNB) and the access gateway (aGW). For the WLAN side, the IE is located at the Wireless Access Point (WAP). UE also has IE and DE to gather the downlink information for the Multi-P decisions and to execute the Multi-P transmission.

B. OPNET Simulation Model

3GPP specified SAE [19] architecture allows a mobile user to roam between 3GPP and non-3GPP access technologies. In order to provide users with seamless mobility Proxy Mobile IPv6 (network based mobility) and Dual stack Mobile IPv6 (host based mobility) have been proposed [19]. We follow this proposal in the integration of 3GPP access technology (namely LTE) and trusted non-3GPP access technology (namely, legacy WLAN 802.11g), where host based mobility solution i.e., Dual stack Mobile IPv6 is considered. According to current 3GPP specification multi-homing is not supported. This implies that a user can either be associated to LTE network or WLAN network but cannot connect to both networks simultaneously. This work extends the 3GPP specified architecture to give users multi-homing capabilities. This is achieved by extending the implementation of MIPv6 to support multiple care-of address [14] and flow management functionality [12][13].

Fig. 1 shows an overview of the simulation network implemented in OPNET. All entities of SAE architecture which are necessary to carry out multi-homing scenario have been implemented. As per 3GPP proposal home agent (HA) function is located at PDN gateway. All users are considered to be out of home network during the complete simulation time. Remote server acts a correspondent node (CN) from where mobile users access application services like VoIP, Video and FTP. Users receive router advertisements from eNB and possibly from WLAN access point to configure care-of addresses. These care-of addresses are then registered with their HA through standard MIPv6 signaling. In this way all user traffic is tunneled from HA to the user.

It should be noted that our focus is only on the downlink access for LTE and WLAN. This implies that no uplink transmissions are performed for WLAN during the whole simulation time. Instead all uplink traffic (e.g., TCP ACK packets etc.) is transmitted by the user through LTE access link.

The original OPNET simulator does not adjust PHY data rate of WLAN users dynamically based on the received signal strength. This behavior has been modified based on literature survey [15] to bring it closer to the reality. As a result of this extension the user PHY data rate changes dynamically based on the received signal strength. However, the users are served by the access point only if their channel conditions are good enough allowing them to transmit at PHY data rate of 6Mbps or better. This is to avoid WLAN netwrok performance degradations due to users with low PHY data rate e.g., 1Mbps, 2Mbps etc.

OPNET provides no mechanism to evaluate voice call quality for wideband G.722.2 [17] codec. A procedure according to ITU-T recommendation has been introduced to evaluate voice call quality for simulation results as detailed in [18]. Furthermore, OPNET has also no support for realistic video traffic generation using any standard codec. Another extension has been made to generate realistic video call traffic according to MPEG-4 codec and evaluate the video call quality at the receiving end as proposed in [16].

III. Traffic Flow Management

A network operator can manage the traffic flow of a multi-homed user either by switching the complete traffic flow from one path to another or by splitting it into multiple sub-flows in a way that each path carries one sub-flow. 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 is why in this work, flow management with flow splitting option is implemented and analyzed with the help of simulations.

A very basic question related to flow splitting option is: What should be an appropriate size of sub-flows (in bits/sec) transported to a multi-homed user over each path? In other words, how much traffic should be sent to user on each of the available access links? And a straightforward answer would be: each path or link should be loaded according to its bandwidth capacity. Moreover, when considering the flow splitting option one should also think how these sub-flows will be aggregated at the receiving end. In the following subsections, answers to these two questions are addressed.

A. Estimation of Link Capacity

1) WLAN Access Technology: Legacy WLAN (802.11 a/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 the users in a “First Come First Serve” manner. That’s why overall throughput of a hotspot and that of the 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. One way to estimate the throughput of a user is by knowing how much data have been sent to the user in a certain time window. However, this option has a serious drawback; it can only be used if there is already some data flowing to the user over WLAN link. Therefore, during time when a user has just attached to an access point and has not yet received any data over WLAN link, its potential bandwidth capacity over WLAN link cannot be estimated. Sending an arbitrary amount of data traffic on WLAN link without the knowledge of its capacity may lead to excessive queuing delays and buffer overflows at the WLAN access point.

This work proposes a novel way of scheduling available WLAN bandwidth resources in an efficient way which also provides an accurate estimation of user bandwidth capacity over its WLAN link. This approach needs following pieces of information to work i.e., number of active users attached to WLAN access point and their PHY data rate at a particular time instance. Once the number of active users associated to an access point N is known and they are assumed to be served in round robin manner, throughput of a user i denoted by $\lambda^{\rm rr}_{{\rm user}i}$ can be easily computed. Let’s take t_i as the time required to transmit one complete IP packet of size p_i bits. The value of t_i can be computed based on user’s current PHY data rate, packet size and MAC/PHY protocol overhead bits. Now that

$$\lambda^{\rm rr}_{{\rm user}i} = \frac{p_i}{\sum_{i=1}^{N} t_i}. \tag{1}$$

Similarly the access point throughput $\lambda^{\rm rr}_{\rm AP}$ is given by

$$\lambda^{\rm rr}_{\rm AP} = \frac{\sum_{i=1}^{N} p_i}{\sum_{i=1}^{N} t_i}, \tag{2}$$

The assumption that access point serves users in round robin manner can be realized by performing an intelligent flow splitting at HA. Actually HA is the entity where flow management function decides to which user’s care-of address a packet will be forwarded. For all users who are being served by the same access point, their data traffic packets are sent towards the serving access point in the round robin manner i.e., one packet from first user, next packet from second user, and so on. Owning to the fact that no packet re-ordering takes place on the transport link from PDN-GW to the access point, all of these packets will be buffered in the FIFO queue of the access point in the sent order. In this way, when these packets are transmitted to the users it can be claimed that users are being served by the access point in round robin manner.

The round robin way of scheduling WLAN resources, however, does not make optimum use of the resources. This point can be elaborated with following example. Consider a single active user attached to a WLAN access point who is receiving a UDP flow comprised of fixed IP packet size of p bits. Assuming 54Mbps PHY data rate, the user experiences a throughput of $p/t_{\rm 54Mbps}$ where $t_{\rm 54Mbps}$ is the time to transmit one packet. As soon as another user with 6Mbps PHY data rate (who is also receiving a similar UDP flow) associates to the same access point, the overall throughput now amounts to $2p/(t_{\rm 54Mbps} + t_{\rm 6Mbps})$. Considering basic channel access mechanism of 802.11g $t_{\rm 6Mbps} \simeq 5.6 \cdot t_{\rm 54Mbps}$ which implies that joining of second user reduces the overall access point throughput by 70%. This is because round robin is a fair scheme which gives equal chance of medium access to all active users irrespective of their channel conditions.

One way to overcome this drawback is performing flow splitting in such a manner which gives users with medium access in proportion to their PHY data rate values. In other words, the users are given equal share of time slice and in this way the users with the higher PHY data rate can transmit more packet compared to the users with lower PHY data rate. This scheduling effect can be achieved in the above described example if 56 packets from first user and 10 packets from second user are sent to WLAN access point by flow management function residing at HA. This will enhance the overall system throughput by 196% compared to simple round robin scheme. However, this overall system performance gain comes at the cost of reduction in throughput of second user. Nevertheless, the proposed scheme is fair enough to give a user system throughput share in proportion to his channel condition while considerably improving the overall system throughput.

In order to compute the throughput of a system which follows above mentioned WLAN resource scheduling scheme, let’s define r_i as the achievable data rate for a user who is the only active user in the system. This amounts to $r_i = p_i/t_i$, where t_i is the time taken by user i to transmit a packet of size p_i bits when transmitting with a certain PHY data rate. Now, the user i’s fair share from throughput resources in proportion to his achievable data rate r_i is given by w_i such that $w_i = r_i/(\sum_{i=1}^{N} r_i)$. This way, the overall system throughput $\lambda^{\rm ch}_{\rm AP}$ will be computed as following

$$\lambda^{\rm ch}_{\rm AP} = \frac{\sum_{i=1}^{N} w_i \cdot p_i}{\sum_{i=1}^{N} w_i \cdot t_i} \tag{3}$$

and the throughput of user i is given by

$$\lambda^{\text{ch}}_{\text{user}_i} = \frac{w_i \cdot p_i}{\sum_{i=1}^{N} w_i \cdot t_i} \quad (4)$$

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 HA.

2) LTE Access Technology: In case of LTE, it is not simple to compute the bandwidth capacity available to a user. This is because its value 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. This problem can, however, be solved by introducing a throughput metering function between PDCP and RLC layers at eNodeB. This metering function reports the average throughput of the user data flowing from PDCP to RLC layer in downlink direction. The reported throughput value is then taken as LTE link capacity of that particular user. In addition to frequent user throughput reports, the metering function also provides occupancy level of user PDCP buffer at eNodeB. PDCP buffer occupancy actually reflects the tendency of increase or decrease in user throughput. For example, when user 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 which in turn makes PDCP buffer occupancy to increase. The opposite is true, when user throughput increases. Owing to this fact, flow management function at HA tries to keep PDCP buffer occupancy at a certain target level. During the events of decrease in PDCP buffer occupancy more user traffic is sent to LTE link till the target buffer occupancy level is achieved and vice versa. The amount of target buffer occupancy is decided dynamically as explained later in this section.

B. Sub-flow Aggregation Function

In multi-path communication, packet may arrive out of order at the destination [7]. 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.

On the other hand, TCP based application are very sensitive to packet re-ordering. This is because an out-of-sequence packet leads TCP overestimate the congestion of the network, which results in a substantial degradation in application throughput and network performance [6]. A literature survey shows that there are several proposals to make TCP robust against packet re-ordering [8]-[11]. The analysis and implementation of these schemes in our simulator is currently not within the focus of this research work. Instead, we implement a simple TCP re-ordering buffer at user side, which is very similar in functionality to a play-out buffer. Simulation analysis shows that re-ordering buffer length must be set as less than TCP protocol time out value. In this work, re-ordering buffer length has been kept between 100ms to 500ms. TCP reordering buffer length τ_{tcp} is a key factor in deciding target PDCP buffer occupancy level μ_i of a user i.e.,

$$\mu_i = \lambda^{\text{LTE}}_i \cdot \tau_{\text{tcp}}, \quad (5)$$

where λ^{LTE}_i is the estimation of LTE link throughput of the user.

With the help of this strategy, LTE link delay is controlled not to exceed the TCP re-ordering buffer length, and hence, avoid unnecessary TCP time-outs.

TABLE I
SIMULATION CONFIGURATIONS

Parameter	Configurations
Total Number of PRBs	50 PRBs (10 MHz specturm)
Mobility model	Random Direction (RD) with 6 km/h
Number of users	5 VoIP, 3 HD video & 5 Skype video call, 7 FTP downlink users
LTE Channel model	Macroscopic pathloss model , Correlated Slow Fading [3]
LTE MAC Scheduler	TDS: Optimized Service Aware [4], FDS: Iterative RR approach
WLAN technology	802.11g, RTS-CTS enabled, coverage $\approx$ 100 m
VoIP traffic model	G.722.2 wideband codec, 23.05kbps data rate
Skype video model	MPEG-4 codec, 512kbps, 640x480 resolution, 30fps, play-out delay: 250 ms
HD video model	MPEG-4 codec, 1Mbps, 720x480 resolution, 30fps, play-out delay: 250 ms
FTP traffic model	FTP File size: constant 10 MByte continuous file uploads one after the other.
Simulation run time	10^3 seconds,13 seeds, 98% confidence interval

IV. SIMULATION SCENARIOS AND RESULTS

The target of this section is to highlight the gains that can be achieved by extending the 3GPP inter-working architecture to support the simultaneous use of the multi interfaces. That means the aggregation of several wireless interfaces, in order to enhance the system performance. In this section, two main scenarios are compared against each other, that is, the 3GPP default architecture, where multi-homing is not supported however user can perform seamless Handover (HO) to switch between the multiple wireless networks, and this will be referred to as “3GPP HO”. Whereas, the second scenario is the novel proposal of this paper, to extend the 3GPP architecture into supporting the simultaneous use of wireless interface, this will be referred to as “Multi-P”.

For each aforementioned scenario, two simulations setup are investigated. The first setup is composed of 20 users with mixed traffic of: Voice over IP (VoIP), File Transfer Protocol (FTP), video conference (i.e., Skype video call), and High Definition (HD) video streaming. The users move within one LTE eNodeB cell, and within this cell four wireless access points or hot-spots are present as shown in Fig. 1. The second setup is a special case where 5 FTP users are moving within the restricted coverage of a wireless access point where no LTE access is available. The motivation behind the later setup is to show how the resource management function of "Multi-P" scenario outperforms the default "3GPP HO" case. The simulation configuration parameters are shown in Table I.

A. Mixed User Traffic

In this subsection, the mixed traffic setup investigations are discussed. As stated earlier, two different cases are compared against each other, these are: "3GPP HO" scenario and "Multi-P" scenario. The first scenario does not make simultaneous use of LTE and WLAN access technologies. In this case, all traffic takes its path to the user through LTE access which is available everywhere. However, when user enters the WLAN coverage its traffic is completely handed over to WLAN. It is worth mentioning here that in the "3GPP HO" scenario the users make vertical handover of hard nature, i.e., the user are disconnected completely from one network, and establish a new connection to the other one. Though MIPv6 keeps all IP layer connections alive through seamless handover, users might lose some buffered data on the previously connected network. On the other hand, the "Multi-P" scenario enables the users to use WLAN access when they are in its coverage while still keep the LTE connection alive and using it at the same time. As a result, a bandwidth aggregation process of both wireless links is achieved.

Fig. 2 shows the spider web graph of the average user delay values. 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 Fig. 2 has four different axes, each representing one performance metric, mainly VoIP, Skype video, HD video, and FTP file transfer end-to-end delay.

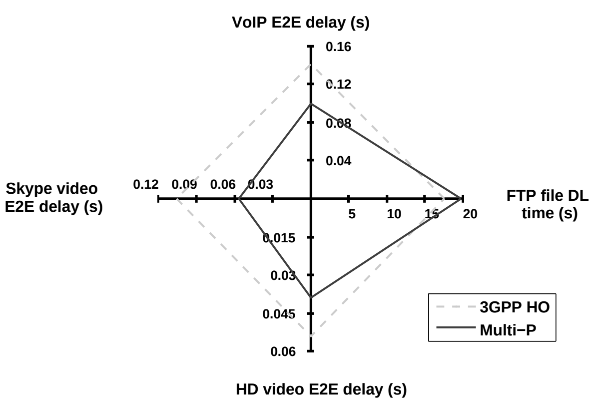

Fig. 2. End-to-end delay comparison spider web graph

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 and videos traffic scenario. As for the FTP performance, it can be seen that the "3GPP HO" scenario has a slightly lower FTP file download time. However, the total number of 10MByte downloaded files is higher in the "Multi-P" scenario compared to the "3GPP HO" one (see Fig. 3). This is because in "3GPP HO" case some of the TCP connections abort due to excessive packet loss and sudden huge changes in TCP round trip time during the handover. In order to have performance evaluation comparison between the two scenarios, the Mean Opinion Score (MOS) values for the VoIP and the video traffics are shown in Fig. 4.

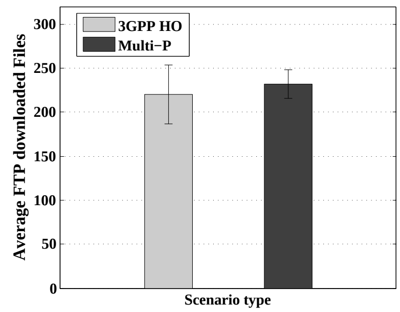

Fig. 3. Number of FTP downloaded files comparison

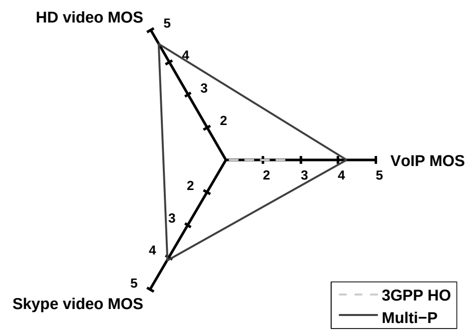

Fig. 4. VoIP and video MOS comparison

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). On the other hand, the "3GPP HO" scenario achieves very low MOS value for the VoIP users, the reason behind is, when VoIP 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 VoIP calls 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 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) As far as WLAN link is concerned, the flow management function at HA sends user traffic on WLAN link only when user PHY data rate is 9Mbps or higher. This is because when a user has PHY data

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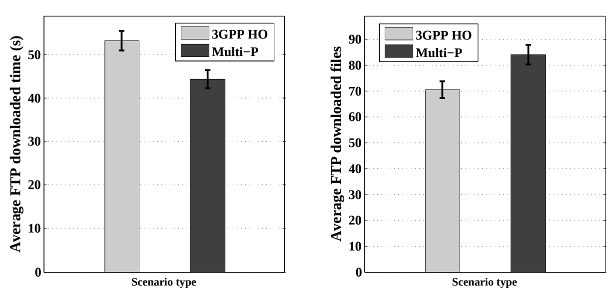
rate as 6Mbps it is a strong indication that loss of WLAN link is imminent. Hence, no new traffic data is sent on WLAN link for that user which gives him a chance to receive already buffered data from the access point before the loss of link.

It can be noticed that video users MOS values in “3GPP HO” scenario are not shown. The reason for that is the large packet loss rate due to two reasons: (i) buffered data loss at previously connected WLAN AP(ii) high end-to-end packet delay which makes play-out buffer discard the packets with delay greater than 250ms. Due this high packet loss rate, EvalVid tool [16] cannot evaluate the exact MOS value of the received video call implying that received video quality to too bad to be watchable. This limitation of EvalVid has already been discussed by the authors of the tool on their website.

B. FTP User Traffic

In this subsection, a special setup is investigated in order to highlight the advantages of using WLAN resource management algorithm discussed in Section III-A1. The scenario comprises of 5 FTP users, all moving within the coverage of the wireless access point without having LTE access. In both scenarios, the FTP users download constant 10MByte file size one after the other. Fig. 5(a) shows the average FTP file download time for both scenarios.



(a) Average file download time (b) Average downloaded files count

Fig. 5. FTP download performance

It can be seen, that the “Multi-P” scenario achieves a lower FTP file download time compared to the other scenario where no management of WLAN resources is performed. Moreover, the users in “Multi-P” scenario manage to download more files than in “3GPP HO” scenario. This shows that “Multi-P” scenario’s algorithm of WLAN resource management does its job in optimum resource utilization to increase overall network throughput as explained in Section III-A1.

V. Conclusion

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 proposed an extension to the specifications to allow a user benefit from all available access technologies by connecting to them simultaneously. The legacy WLAN does not provide QoS guarantee, and therefore not suitable for realtime interactive applications. However, through the use of suggested algorithms for resource management and accurate bandwidth capacity estimation WLAN bandwidth resources can be utilized for multi-homed users running QoS sensitive applications. 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 proposed scheme succeeds not only in providing QoS aware service to multi-homed users, but also optimizing the network bandwidth resource utilization. By outperforming the current 3GPP proposal, the new scheme assures a win-win situation for network operators as well as for users in future wireless networks.

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