Study, Evaluation and Measurement of 802.11e and 802.16e Quality of Service Mechanisms in the Context of a Vertical Handover Case of Real Time Applications

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Summary
Wireless networking technologies, do not cease to develop and impose day after day. Unlike wired networks, these technologies offer much more mobility, a wide coverage and speed deployment. However, they represent several limits, especially in terms of quality of service (QoS).
We hear these days by heterogeneous networks, these latter, and thanks to the handover mechanisms, can ensure a routing data without a break even in the customers' mobility case. These mechanisms remains unreliable bringing latency and additional losses. It is for this reason that the QoS mechanisms deployment becomes an inevitable necessity.
Several research works handled the QoS mechanisms in the Wi-Fi or WiMAX homogeneous wireless networks (Horizontal Handover). According to our research, no scientific work has been done evaluating the heterogeneous network performance taking into account the presence of both technologies simultaneously (Vertical Handover) as well as their good configurations.
Through this article, we will enhance and complete the earlier works by initiating the following points: (i) studying the QoS mechanisms in 802.11e and 802.16e networks, (ii) discussing the MIPv6 protocol deployment in a wireless network, (iii) evaluating and measuring the QoS impact on the network performance and real-time applications.
These studies were conducted under OPNET Modeler simulator, by using the Voice over IP (VOIP) and Video Conference (VC) applications. The simulation parameters are: The network delay (Wi-Fi and WiMAX), the throughput, the jitter, the end-to-end delay, the MOS score and the loss rate.

Key words:
802.16e; 802.11e; VOIP; QoS; UGS; RTPS; DCF; PCF; HCF; Video Conferencing; Vertical handover; OPNET Modeler.

1. Introduction

With the rapid evolution of new digital communication technologies and the growing users' needs in terms of mobility, wireless networks represent an emerging solution offering to the user mobility and access to information, regardless of its geographical position.
Although there are several heterogeneous wireless access technologies, such as 802.16e, 802.11e, LTE, UMTS, etc… These latter offer wide coverage, high throughput and optimal cost compared to wired access technologies. In addition, these latter constitute the backbone networks of the new generations.
However, these wireless communication technologies do not implicitly guarantee the quality of services to different users. This is why this research axis is an active field to the present day. The issue related to the quality of service can be approached according to two main levels:

1. Quality of service for the selection from one access technology to another (Example of a device that disconnects from the 4G Network, as soon as a Wi-Fi network is available).
2. Quality of service for user applications.

In this paper, we will perform a study, evaluation and measurement of the 802.11e and 802.16e quality of service mechanisms in the context of a vertical handover, using both VOIP and VC applications.
The rest of the introduction will be organized as follows: In the first section, we will study the vertical handover process as well as the mobility management protocol, Mobile Internet Protocol Version 6 « MIPv6 ». Then, and through the second and the third section, we are going to perform respectively, a study on the 802.11e and 802.16e heterogeneous networks with their quality of service mechanisms. The real-time applications used in the scenario will be detailed through the fourth section and to finish we will perform an empirical study on the related works in the last section.

1.1 The vertical handover process

The vertical handover is a technique that allows a multi-interface station to switch between different network access technologies without service interruption or applications disconnection. Handover may occur for a better QoS satisfaction: if the current cell cannot satisfy the quality of service required by the user, or when there is another cell that provides services with a better quality.
Fig.1 illustrates an example of the vertical handover.

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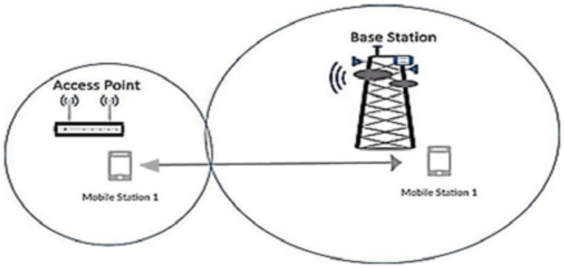

Fig. 1. Vertical Handover

The handover management is a complex process that is based on decision policies to assess where and when the handover should be executed. This process takes place in three stages:

- Collection of handover information;
- Decision;
- Execution.

In general, and in traditional approaches, the information collected to determine the handover execution is related to the received signal strength [1]. On the other hand, in the new approaches, the decision is based on other information in addition to the signal strength. Such decision may be taken either at the network or the terminal level.

The process execution is dictated by a set of criteria that can be grouped mainly in the following way:

- Network Criteria: coverage, latency, BER [2], etc.;
- Services Criteria: QoS and potential services, etc.

The problems to be overcome during the vertical handover are those of connection reservation when it is necessary to switch between different interfaces. More specifically, the problem is to maintain an open connection, then it is necessary to have the same IP address even if there is a change of subnets (due to changing interfaces).

In order to achieve this goal, Mobile IPv6 is used, it can play a leading role in integrating different technologies of the link layer, with the promise to allow transparent mobility through a unified network layer.

Currently, the mobility using the IPv4 protocol with MIPv4 mechanisms [3] and PMIPv4 [4] suffers from a significant problem of triangular exchange during a communication. This method requires packets to go through the parent user agent before arriving to the correspondent, which necessarily increases the delay. MIPv6 [5] has been proposed to solve this problem through an address mapping system which allows the mobile parent user agent to send his new address to his correspondent. And his correspondent can contact him directly through this address via a tunnel he will create for it.

Fig.2 illustrates an example of the MIPv6 protocol operation.

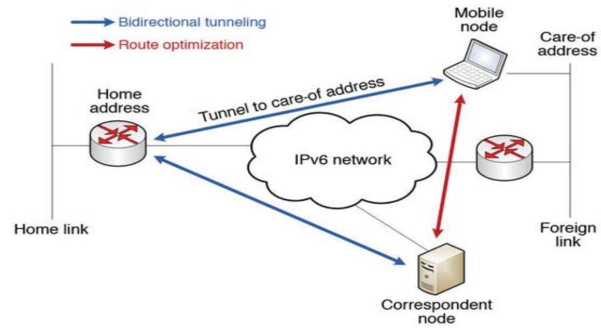

Fig. 2. MIPv6

The rapid handover detection constitutes one of the major concerns in the heterogeneous networks. The shorter the detection time, the better the network performance and the applications transported. For this reason, the IEEE 802.21 standard was created.

The IEEE 802.21 standard, also called MIH (Media Independent Handover) is a handover protocol standardized in 2009 by the IEEE 802.21 working group [6]. This standard has been developed to support interconnection and mobility between heterogeneous networks. The MIH standard includes a set of algorithms to maintain uninterrupted communications between networks based on IEEE (802.11e, 802.16e) or cellular technologies (UMTS [7], LTE [8]). The IEEE Standard 802.21 specifies procedures and primitives that facilitate the handover initiation and the network selection.

Fig.3 shows the handover management by the IEEE 802.21 standard.

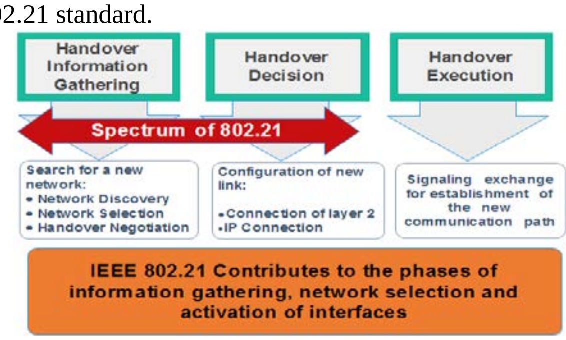

Fig. 3. The IEEE 802.21 operation

1.2 The 802.11e network

The IEEE 802.11 [9] is an international standard describing the characteristics of a Wireless Local Area Network (WLAN), this latter is used to replace the Local Area Network (LAN) or as an extension of the LAN infrastructure.

Wireless networks must make a compromise between the scope and the throughput available. Various developments

are in progress as well to allow extensions for security [10-12] and the quality of service.
The IEEE 802.11e [13] is an improved version of the IEEE 802.11 introducing the QoS at the MAC layer for the voice, data and video transport traffic through the WLAN.
With the IEEE 802.11e, the Distributed Coordination Function (DCF) [14-15] which is an improved variant of the CSMA/CA [16], who avoids collisions during the transmission by the random slowdown after each frame (backoff). The DCF mode has some problems: it only supports the Best-Effort service, it does not guarantee the delay and the jitter, and it degrades the throughput when the load is high.
With the IEEE 802.11e, the PCF (Point Coordination Function) [17] allowing access to the medium wireless without constraint, also presents some problems: the central polling scheme is ineffective, an unpredictable beacon frame delay due to the incompatible cooperation between modes CP (Contention Period) [18] and CFP (Contention Free Period) [19], and finally an unknown transmission time of the polled stations.
The media sharing with DCF and PCF modes does not allow to predict the allocation of the speech duration, which is nevertheless necessary for QoS requirements.
Two new modes were added 802.11g [20] or 802.11n [21]. They are based on a coordination function called HCF (Hybrid Coordination Function) [22-23].
After having obtained access to the media, the station can keep it. Thus, it can emit a certain frames number, during a given period of time (TXOP Transmission Opportunity) [24]. This information is received by the station in a beacon frame.
HCF brings two new access mechanisms: EDCA (Enhanced Distributed Channel Access) [25-26] which provides a restraint access service (based on CSMA/CA) with traffic differentiation, HCCA (HCF Controlled Channel Access) [27] which provides unconstrained access (by polling) for a service with parameterized QoS.
Fig.4 illustrates a synthesis of the 802.11e standard architecture.

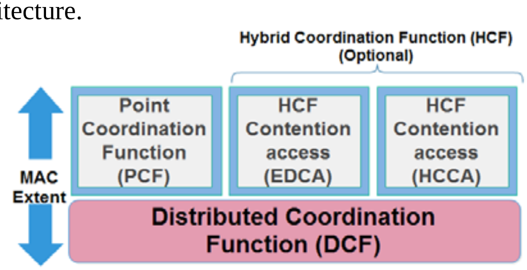

Fig. 4. The 802.11e standard architecture

1.3 The 802.16e network

The IEEE 802.16 standard [28] better known as WiMAX. It is a technology used mainly for metropolitan wireless networks (WMAN), which aims to provide a broadband internet connection on a coverage area of several kilometers' radius.
The theoretical throughput of WiMAX is in the order of 70 Mbps with a range of 50 kilometers. WiMAX technology introduces mobility features into its network: a WiMAX terminal can move while maintaining reliable network access. This feature is introduced by the IEEE 802.16e standard [29-30] which can be classified in wireless wide area network (WWAN).
In addition, the IEEE 802.16 standard has defined four services classes namely: Unsolicited Grant Service (UGS) [31], real-time Polling Services (rtPS) [32], non-real-time Polling Service (nrtPS) [33] and interactive traffic (BE). And a last class recently integrated the standard constituting an extension of the real-time service (ertPS) [34]. The characteristics of each class are defined in Table I below.

TABLE I. Synthesis on 802.16e QoS mechanisms

Class of service	Description	Application
Background services BE (Best Effort)	The BE class is dedicated to traffic requiring no particular level of performance.	Internet navigation
Data transfer service nrtPS (non-real-time Polling Service)	The nrtPS class must in turn ensure the good management of traffic insensitive to the delay but require a minimum throughput. So, the packets size can be variable, as well as the delay between two packet transmissions.	File transfer (FTP)
Real-time services rtPS (real-time Polling Services)	The rtPS class is designed to handle real-time traffic for which the data stream packets size is variable and at regular intervals. This class therefore respects the delay-sensitive traffic avoiding to update queries, the collisions involved by the contention interval.	Video stream such as the MPEG, H.263
Extended real-time service ertPS (extended real time Polling Service)	The ertPS class is intended to support real-time data streams characterized by a variable packets size received periodically.	Voice over IP (VoIP) with silence removal
Unsolicited Grant Services (UGS)	The UGS class is intended to support the real-time data streams characterized by a fixed packets size received periodically.	Voice over IP (VoIP) without silence removal

1.4 Real-time Applications

Among the applications that we used for our study, we cite: The VOIP and VC. We will present briefly the interest of each of these applications as well as their constraints in an IP network.

The VOIP is a technology for digitizing telephone calls in order to transmit them over the IP network with the aim of minimizing costs. The VOIP technology inherits the same IP network limits such as:

- The jitter: If two consecutive packets leave the source node with time stamps t1 & t2 and are played back at the destination node at time t3 & t4, then: jitter = (t4 - t3) - (t2 - t1). Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node.
- The latency: The total voice packet delay, called "analog-to-analog" or "mouth-to-ear" delay = network_delay + encoding_delay + decoding_delay + compression_delay + decompression_delay Network delay is the time at which the sender node gave the packet to RTP to the time the receiver got it from RTP. 300 msec is the maximum threshold to be tolerated according to ITU Telecommunication Standardization Sector [35].
- The packet losses: Packet loss may be measured as frame loss rate defined as the percentage of frames that should have been forwarded by a network but were not. ((Number of Sent packet - Number of receveid packets) * 100 / Number of sent packets)
- MOS score (Table II): Stands for Mean Opinion Score, gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using various codec. [36]

TABLE II. MOS Score

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

The study of the VOIP performance is an active research field, several studies have been conducted for its evaluation. Various studies have been conducted for evaluating the VOIP performance by varying codecs [37-40], these studies have shown the effectiveness of the G.729 codec over other codecs.

Other works were interested in studying the impact of routing protocols on the VOIP quality [41-46]. These articles show by simulation that the EIGRP protocol is the most recommended thanks to its efficient bandwidth management and quick failure detection. Other works have studied the impact of the IPV6 [47].

As for the VC, it allows through the video and audio to convey to interlocutors the information regardless of their geographical position. VC technology shares the same constraints as VOIP, particularly in terms of latency, jitter and especially loss rate.

Real-time applications essentially go through two phases to create a session: the call signaling and the traffic transport.

The part of signaling is a critical phase for admission control, the codecs negotiation and the channel establishment. Among the standardized protocols and widely used, we mention the Session Initiation Protocol acronym of SIP [48] and the H.323 protocol [49]. According to the literature [50-52] it can be noted that the SIP protocol is the most deployed and efficient.

After the negotiation and communication establishment phase, The Real Time Protocol « RTP » [53], based on UDP, transports VOIP packets with a very high reliability, but offers no mechanism of communication reporting. For this reason, the Real Time Control Protocol "RTCP" generates from time to time a report of the communication quality.

The rest of the paper is organized as follow: In second part, we will detail related works. The third part will present evaluation scenarios and used parameters of simulation. The interpretation and analysis of the obtained results will discussed in fourth part. Finally, we will conclude.

2. Related works

The articles [54-57] compare between PCF and DCF, these works present the smooth configuration. The first article deployed a web traffic for evaluation and the second used the VOIP and simulated the rising load. Both works led to the efficiency of the PCF protocol compared to DCF. The third article has conducted an evaluation of the network parameters between the PCF and DCF modes under different variants 802.11a, b and g. While the fourth paper concluded on the efficiency of DCF versus PCF in terms of throughput, this seems a bit far from the technology fundamentals.

The whole works did not deal with HCF mode. The articles [58-62] integrate the HCF mode, but does not take into account the nodes mobility.

Through the articles [63-64], the authors conducted a comparative study between the different 802.16e QoS mechanisms: UGS, RTPS, eRTPS using VOIP as application. They have led to that ertps is best suited for VOIP. While recent works [65-67] have evaluated the VOIP performance under WiMAX and have demonstrated the UGS mechanism effectiveness for VOIP. However, none of these works has dealt with the vertical handover issue.

The article [68] deals with problems related to vertical handover, unfortunately, its study was conducted without taking into account the stations mobility, also the quality of service impact evaluation has been performed on each network without showing where the quality of service implementation is strongly needed. The other works [69-

70] studied the interest of QoS in a vertical handover, but they did not measure its performance by implementing it in a simulation testbed.
Taking into account all our remarks about the related works, we will complement and enhance them by:

- Showing the interest of QoS in a vertical Handover;
- Evaluating the different QoS mechanisms in 802.11e and 802.16e networks;
- Taking into account the node mobility;
- Diversifying applications (VOIP and Video conferencing);
- Showing where the QoS is the most influencing.

3. The Evaluation Environment

To conduct our studies, we used the OPNET Modeler tool [71], several simulators can be used, such as NS2 [72], NS3 [73] and OMNET [74]. OPNET Modeler is currently considered as one of the best simulators in the field of wireless networks compared to other simulators [75].
The book [71] is a good guide to learn OPNET Modeler simulator.

3.1 The evaluation scenarios

The scenario chosen in the evaluations is shown in Fig.5

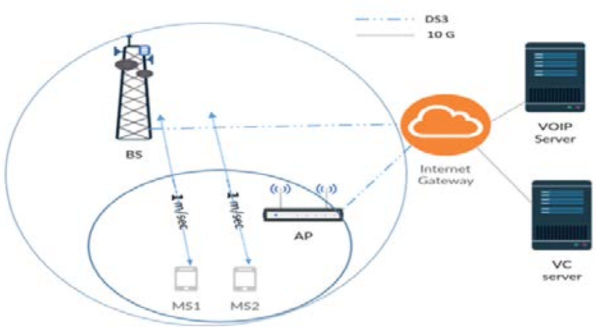

Fig. 5. Evaluation Simulation Model

Based on this model, we have created five scenarios (Table III). Reminded that the purpose of these scenarios is to measure the induced quality using each access method or class of service, and deduct where the quality of service is much more relevant.

TABLE III. Evaluation Scenarios

	DCF	PCF	HCF	WiMAX
Scenario 1	✓			
Scenario 2		✓		
Scenario 3			✓	
Scenario 4				✓
Scenario 5			✓	✓

3.2 The simulation parameters

The mobile terminals MS1 and MS2 are equipped with both Wi-Fi and WiMAX interfaces. The following are the settings used for the WiMAX antenna (Table IV):

TABLE IV. Base station parameters

Parameter	Value
Antenna Gain	15 dBi
Number of transmitters	SISO
Maximal transmission power	500 mW
PHY profile	OFDM
Maximal power density	-60 dBm
Minimal power density	-110 dBm
The resource retention time	200 msec

The simulation parameters used in Wi-Fi scenarios are listed in the below Table V:

TABLE V. Access Point Parameters

Parameter	value
PHY mode	Extended Rate PHY
Throughput	11 Mbps
Transmission power	0.005 W
Beacon interval	0.02 Secs
Buffer size	256 Kilobits

Fig. 6 illustrates the configuration of MIPv6 Client

⊟ Mobile IP Host Parameters	
⊞ Mobile IPv4 Parameters	Disabled
⊟ Mobile IPv6 Parameters	(...)
Node Type	Mobile Node
Route Optimization	Enabled
Home Agent Address	2005:0:0:5:0:0:0:1
⊟ Binding Parameters	(...)
Binding Update Timeout Inter...	10
Binding Update Max Retry Att...	6
Lifetime Requested	100
⊟ Return Routability Parameters	(...)
Routability Test Timeout Inter...	2.0
Routability Test Max Retry Att...	6
Mobility Detection Factor	3

Fig. 6. MIPv6 Client configuration

Fig.7 illustrates the configuration of MIPv6 Home Agent.

⊟ Mobile IP Router Parameters	
⊞ Mobile IPv4 Parameters	Disabled
⊟ Mobile IPv6 Parameters	(...)
Number of Rows	1
⊟ IF1	
Interface Name	IF1
Interface Type	Home Agent
⊟ Home Agent Parameters	(...)
Preference Level	0
Binding Lifetime Granted	100
Maximum Number of Hosts ...	100

Fig. 7. MIPv6 Home Agent configuration

3.3 Application parameters

Parameters of applications used and criteria of evaluation are listed in the tables VI, VII and VIII.

TABLE VI. VOIP parameters

Parameter	Value
Traffic	VOIP
Codec	G729 A [76]

Voice frames per packet	1
Traffic generation	Continuously and infinite (from the start to end of the simulation)
Type of Service	Interactive Voice

TABLE VII. Video Conference parameters

Parameter	Value
Traffic	Video Conferencing
Frame Interarrival Time Information	10 frames per sec
Frame Size Information (Bytes)	128x120 pixels
Traffic generation	Continuously and infinite (from the start to end of the simulation)
Type of Service	Streaming Multimedia

TABLE VIII. Criteria Evaluation

Criteria	Signification
End to End Delay	Section I – D)
MOS Score	Section I – D)
Network Delay	This criteria represents the end to end delay of all the packets received by the wireless LAN MACs of all Wireless nodes in the network and forwarded to the higher layer
Throughput	Represents the total data traffic (in bits/sec) forwarded from Wireless layers to higher layers in all other nodes of the network.
Jitter	Section I – D)
Loss Rate	Section I – D)

4. The interpretation and analysis of obtained results

4.1 QoS on IEEE 802.11e

DCF uses the CSMA/CA mechanism, this results in additional latency. An MS waits for a DIFS (DIFS = SIFS+ 2 SlotTime) duration before transmitting, this results in two facts: a queue overflow and an inefficient use of the bandwidth

As we can notice data dropped on DCF scenario exceeds PCF (Fig. 8)

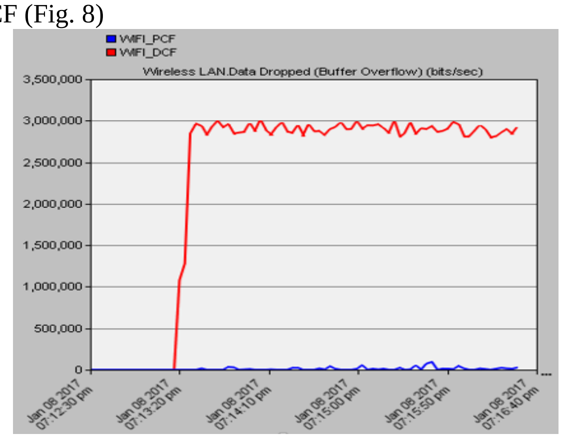

Fig. 8. Wireless Data Dropped DCF vs PCF

Therefore, a very high loss rate (Fig.9), and a high jitter (Fig.10).

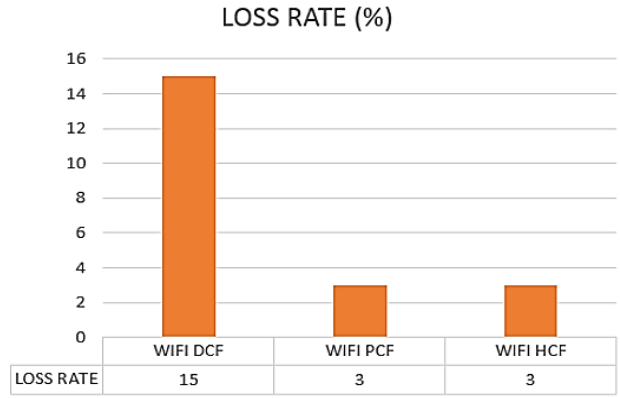

Fig. 9. Loss Rate PCF DCF HCF

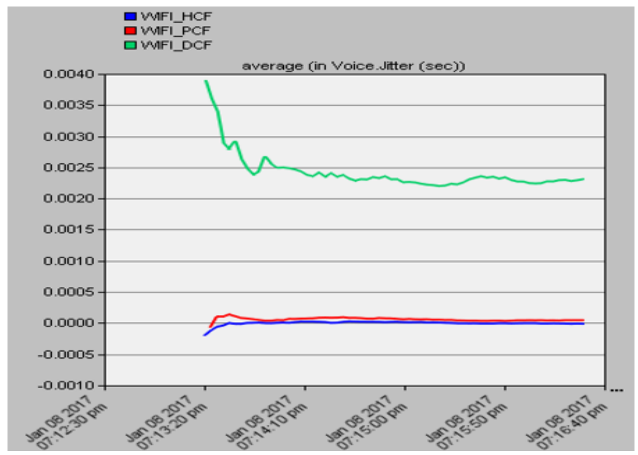

Fig. 10. Voice Jitter HCF PCF DCF

Because of the queue overflow, we can justify the inefficient use of the bandwidth (Fig. 11)

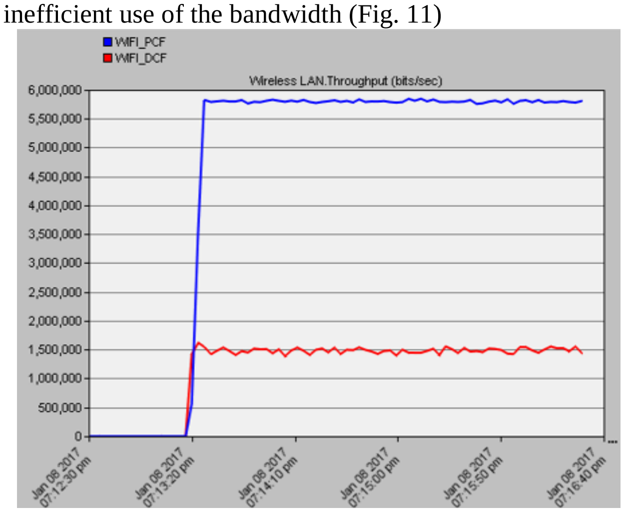

Fig. 11. Throughput of DCF and PCF

Fig.12 shows End to End delay of PCF DCF and HCF modes.

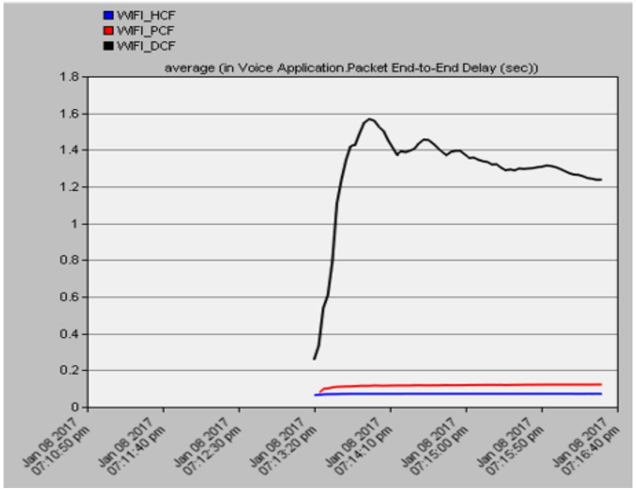

Fig. 12. Voice End to End Delay

Concerning real time applications, we can see that in the scenario with DCF (the default mode) the end-to-end delay parameter, is extremely too high, up to 1600 msec, exceeding by far the tolerable threshold of 150 msec. Unlike the DCF mode, the PCF offers a good threshold. This is justified by the fact that the coordination point uses the pooling principle, which means, authorizes the MS to transmit after interrogating them. This method is much faster than the previous, because the MS will have to wait for a PIFS duration to transmit after the channel release, with PIFS<DIFS. In addition, the CFP interval is configured by 10 msec (Fig.13), which adds much more speed in terms of the channel access.

⊟ PCF Parameters	(...)
PCF Functionality	Enabled
CFP Beacon Multiple	1
CFP Offset	0
CFP Interval (secs)	0.01
Max Failed Polls	2

Fig. 13. PCF parameters

Despite the robustness of PCF. Through the results obtained, we notice that it remains misplaced compared to the HCF mode. Because the PCF mode does not take into account the differentiation of flows Voice, video and data. While the HCF mode allows it.

We notice also that in the HCF mode, the amount of VOIP traffic sent is greater than that of video conferencing traffic (Fig.14), and the reverse in the PC mode (Fig.15). This is justified by the HCF mode nature which gives preference to the flow having the highest DSCP code.

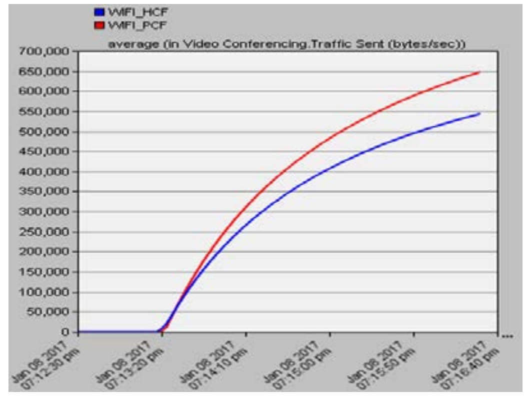

Fig. 14. Video Sent Traffic HCF PCF

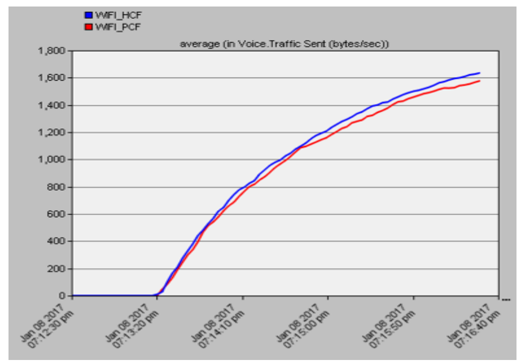

Fig. 15. Voice Sent Traffic HCF PCF

Fig.16 illustrates the configuration used for the HCF mode scenario. The contention window for VOIP traffic is much smaller than that of the video, the data has a very high contention window

⊟ HCF Parameters	(...)
Status	Supported
⊞ EDCA Parameters	Default
⊞ Traffic Category Paramet...	(...)
Block ACK Capability	Supported
⊟ AP Specific Parameters	(...)
⊟ Parameters Advertised ...	(...)
EDCA Parameter Set ...	Enabled
⊟ EDCA Parameter Set	(...)
⊟ Voice	(...)
CWmin	(PHY CWmin + 1) / 4 - 1
CWmax	(PHY CWmin + 1) / 2 - 1
AIFSN	2
⊞ TXOP Limits	Default
⊟ Video	(...)
CWmin	(PHY CWmin + 1) / 2 - 1
CWmax	PHY CWmin
AIFSN	2
⊞ TXOP Limits	Default
⊟ Best Effort	(...)
CWmin	PHY CWmin
CWmax	PHY CWmax
AIFSN	3
⊞ TXOP Limits	Default

Fig. 16. HCF parameters

This justifies why the HCF mode has proposed a smaller access delay compared to the PCF mode (Fig.17).

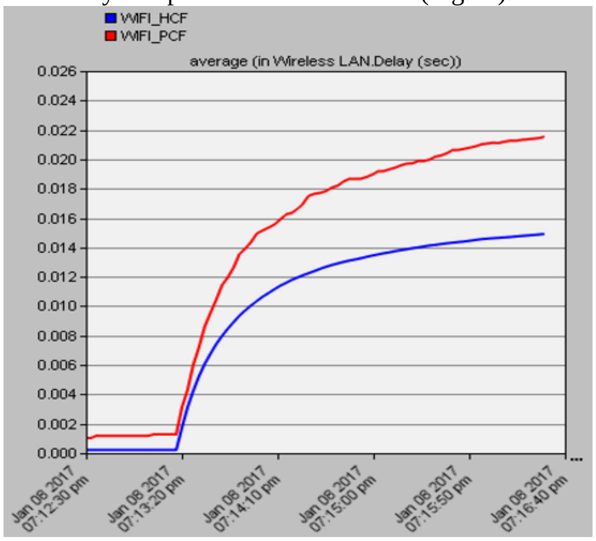

Fig. 17. Wireless LAN Delay HCF PCF

The loss rate affects the quality of the VOIP. According to Fig.9 of losses rates, we found that both HCF and PCF modes proposed a tolerable rate while the DCF has reached 15%. This justifies why the MOS score of the DCF mode offers an unacceptable quality compared to the other two modes (Fig.18).

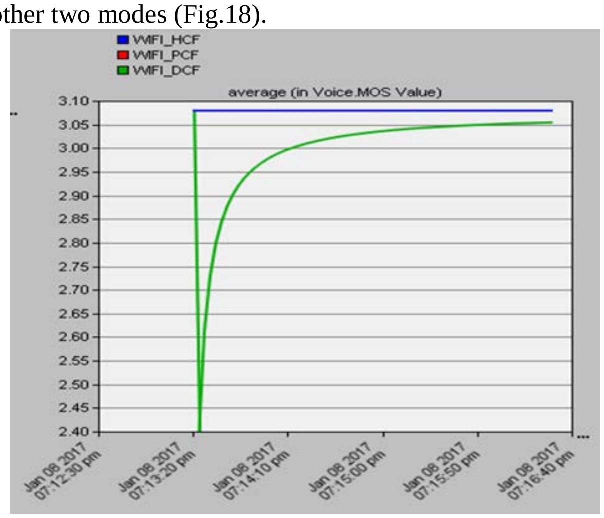

Fig. 18. Wireless VOIP Score MOS

4.2 QoS on IEEE 802.16e

From the results obtained (Fig. 19) it is clear that the quality of service in WIMAX is recommended. This is reflected in the fact that VOIP data, having the most favored class of service, benefit from a much longer processing delay. However, they are not maintained for long in the queue. It is for this reason that jitter is too small.

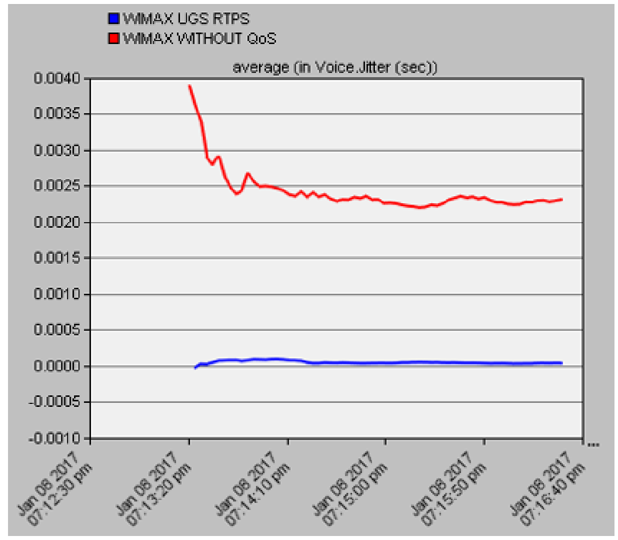

Fig. 19. WiMAX voice jitter

The same effect is observed on the end-to-end delay (Fig.20).

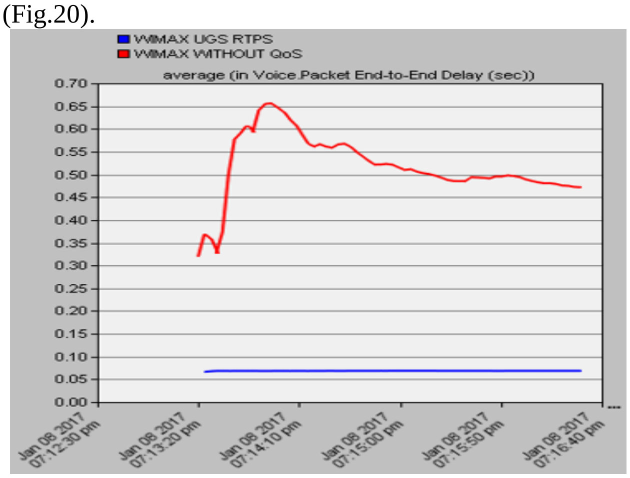

Fig. 20. WiMAX Voice End to End delay

While the quality of service mechanisms in a WIMAX network remain better compared to the WIFI network. This is justified by the fact that the WIMAX network is intended for broadband networks. The obtained result showed that the QoS in WIMAX proposes a lower voice End to End delay than that of the HCF by a factor of 15.3846%, and compared to the PCF mode by a factor of 28.5714%. (Fig. 21)

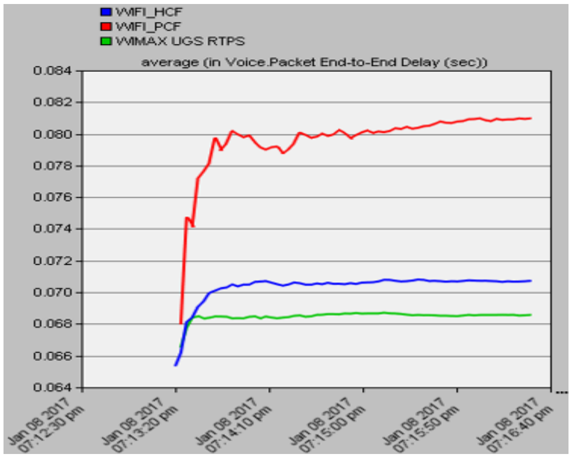

Fig. 21. Voice End-to-End Delay WiFi Vs WiMAX

As expected, the loss rate using the UGS mechanism for VOIP decreased dramatically from 15% to 3% (Fig.22), which is similar to the WIFI scenarios. So we got the same MOS score.

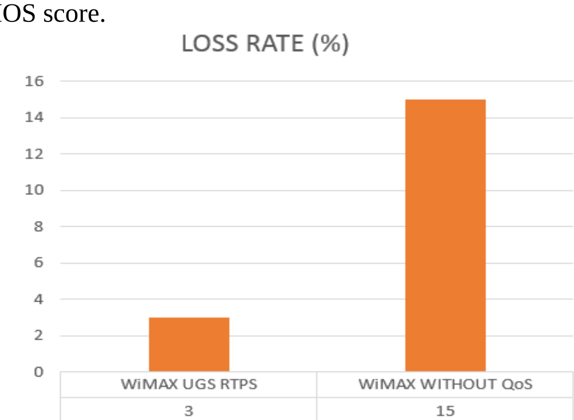

Fig. 22. WiMAX VOIP Loss Rate

To conclude, and thanks to the quality of service, it can be seen that the WIMAX network delay knows improvement up to 700%. (Fig.23)

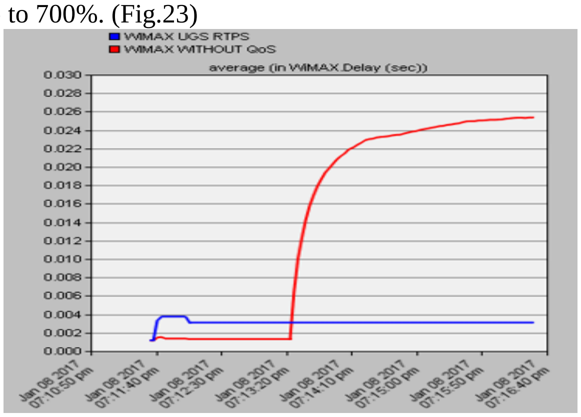

Fig. 23. WiMAX Network delay

4.2 QoS on IEEE 802.16e + IEEE 802.11e

Before running the simulations, we predicted the following preference order: WIMAX quality> WIFI HCF quality> WIMAX + WIFI > WIFI PCF > WIFI DCF. This is justified by the fact that the quality of service involves additional classification and queuing delay, this delay will be multiplied by almost two if we implement the quality of service in both networks.
The following figures illustrates the end-to-end delay for VOIP (Fig.24) and video conferencing (Fig.25) for all scenarios, which supports our expectations:

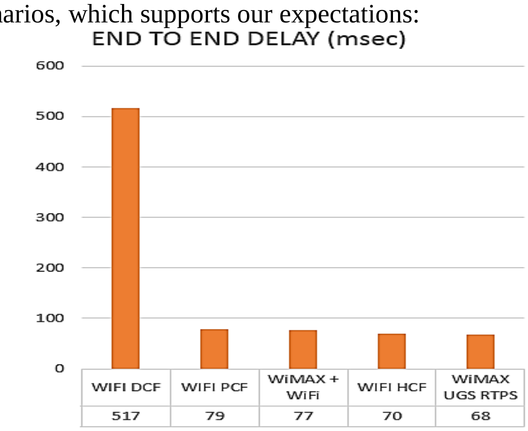

Fig. 24. VOIP End-to-End delay

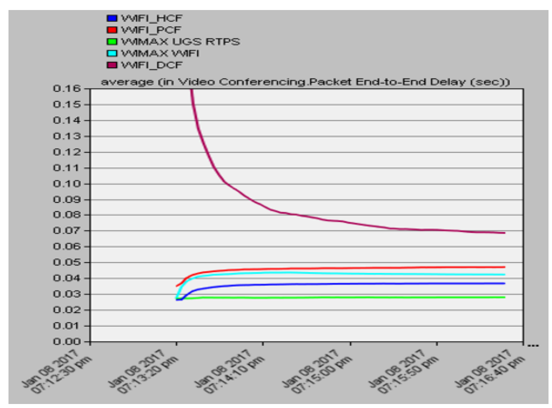

Fig. 25. Video Conference End-to-End delay

5. Conclusion

Through this paper, we discussed a current issue, it is the quality of service in the context of the vertical handover.
We studied the 802.11e and 802.16e standards as well as their QoS mechanisms. And by realized scenarios under OPNET Modeler, we measured and analyzed the impact of QoS mechanisms on heterogeneous network performance and VOIP and Video Conference applications.
The results obtained showed the effectiveness of the HCF method compared to the PCF and DCF. However, we

found a good network performance during the quality of service deployment in the 802.16e network. We have shown also that the QoS implementation in the two networks 802.11e and 802.16e brings an additional delay, a jitter and a high loss rate.

As perspectives, in the next article we will discuss the impact of Dynamic and Multipoint VPN protected by IPsec Protocol in the IEEE 802.16e Network.

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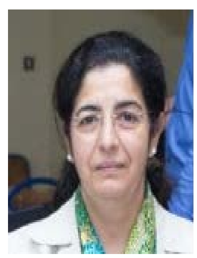

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