QoS Challenges of Real Time Traffic during UMTS/WiMAX/WLAN Vertical Handovers

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Abstract — **Multi-access mobile devices and overlapping wireless network deployments have emerged as a next generation network fixture. Mobile devices should be capable of handing over between heterogeneous networks seamlessly and automatically. On the other side, the Voice-over-IP protocol (VoIP) and video has attracted a great deal of attention in the field of wireless mobile networks. In this paper we study the vertical handover latency of voice packet transmissions and video traffic during the vertical handovers between UMTS (HSPA), WiMAX (IEEE 802.16) and WLAN (IEEE 802.11) networks. Furthermore, we analyze throughput and packet loss results during VHOs between UMTS and WiMAX considering the effects of DCD/UCD interval, and detect the best values for this parameter.**

Keywords — **Handover, Latency, Packet Loss, Throughput, UMTS, Video, VoIP, WiMAX, WLAN.**

I. INTRODUCTION

PROVIDING users of multi-interface devices the ability to roam between different access networks is becoming a key requirement for service providers. The availability of multiple mobile broadband access technologies, together with the increasing use of real-time multimedia applications, is creating strong demand for handover solutions that can seamlessly and securely transition user sessions across different access technologies. A key challenge to meeting this growing demand is to ensure handover performance, measured in terms of latency and loss.

The main purpose of IEEE 802.21 standard is to enable handovers between heterogeneous technologies (including IEEE 802 and cellular technologies) without service interruption, hence improving the user experience of mobile terminals. The aim of the handover procedure is to maximize the service continuity by providing seamless maintenance of active communications when the user changes its point of attachment to the network, either wired or wireless. To avoid unacceptable disruptions to ongoing communications, the network should establish the link to the new point of attachment prior to releasing the previous link. Such soft handover would prevent any perceptible interruption, for example during a voice call. IEEE 802.21 provides a framework that allows higher levels to interact with lower layers to provide session continuity without dealing with the specifics of each technology.

Packet losses and delay due to vertical handover between heterogeneous mobile and wireless networks can significantly degrade the end-to-end VoIP call quality. Thus, optimization of the parameter values for link triggers and router configuration in IEEE 802.21 is crucial to have the best vertical handover performance metrics results in order to have non interruptible VoIP traffic between the users during vertical handovers.

In this paper we optimize the vertical performance metrics results (vertical handover latency, packet loss and throughput) during vertical handover process between UMTS, WiMAX and WLAN for VoIP (codec G.723.1) and video traffic, using ns-2 simulator and package from NIST [1]. The delay contributed by the synchronization component in the WiMAX standard is the most significant during the vertical handover process. Because of this, the effect of the interval of the MAC management messages (downlink and uplink channel descriptor – DCD/UCD) on delay, throughput and packet loss results of the VoIP traffic is researched and optimized. Throughput performances are also analyzed through 15 random simulations with different random trajectories of the mobile terminal for different DCD/UCD intervals. Results are presented which illustrate that significant improvements in vertical handover latency and packet loss for VoIP G.723.1 codec between UMTS and WiMAX are achievable with DCD/UCD optimization. Throughput performances of different random simulations during vertical handover between UMTS and WiMAX for two DCD/UCD intervals show that throughput results during vertical handover are improved if we use lower DCD/UCD interval.

This paper is organized as follows: Section II is the introduction to the VoIP (G.723.1 codec) and video traffic used in the simulations. Section III presents the simulation model used for the researching. Section IV gives the results and analysis from the simulations and finally Section V concludes this paper.

II. VOIP AND VIDEO TRAFFIC

To maintain a conversation at good quality levels, a VoIP flow requires low packet loss rates. Loss rates up to 10% may be tolerated depending on the type of packet concealment technique employed by the decoder on the side of the receiver. To sustain intelligibility of VoIP communications the total end-to-end delay should remain below 150 ms or lower, for highly interactive conversations. Delays in the range of 150-400 ms are

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considered acceptable, although the annoyance becomes perceptible; delays greater than 400 ms are considered intolerable and thus unacceptable for effective communication.

G.723.1 codec belongs to the Algebraic Code Excited Linear Prediction (ACELP) family of codec and has two bit rates associated with it: 5.3 kbps and 6.3 kbps. The encoder functionality includes Voice Activity Detection and Comfort Noise Generation (VAD/CNG) and decoder is capable of accepting silence frames. The coder operates on speech frames of 30 ms corresponding to 240 samples at a sampling rate of 8000 samples/s and the total algorithmic delay is 37.5 ms. The codec offers good speech quality in network impairments such as frame loss and bit errors and is suitable for applications such as VoIP.

Streaming video traffic has a highly variable frame size which is much larger than voice. Packets are fairly large in size ranging from 65 bytes to 1500 bytes. There are two key parameters in which video traffic differ from data traffic: low tolerance for delay and packet loss. Compared to voice traffic, streaming video applications require higher bandwidth (dependent of the encoding protocol, between 10s of Kbps to 10s of Mbps), but are not highly delay or jitter sensitive. They can buffer 4-5 seconds of traffic which can smoothen out jittery traffic considerably. They can also tolerate more losses than voice traffic.

III. SIMULATION SCENARIO AND METHODOLOGY

Simulation scenario presented in Figure 1 consists of a WLAN cell located inside WiMAX cell, both of them located inside an UMTS coverage area, configured on a 2000x2000 m simulated area. Mobile node moves with a pedestrian speed of 10 km/h across all three coverage areas and makes vertical handovers between them. 15 simulation runs were performed with different mobile node trajectories across the three networks. All trajectories are straight lines. In each simulation 4 vertical handovers take place between the three technologies: UMTS/WiMAX, WiMAX/WLAN, WLAN/WiMAX and WiMAX/UMTS. All simulations were run on ns-2 version 2.29.

VoIP is simulated as a two-state voice traffic model. Both states are distributed exponentially with mean for the ON-period _on = 1000 ms and OFF-period _off = 1350 ms. During the ON-period the voice IP flow carried a payload of 24 bytes transmitted at 30 ms intervals. Such data rate specification is compliant with G.723.1 codec at 6.3 Kbps data rate [6].

Video traffic is simulated with constant bit rate so that the packet size is 1500 bytes and a packet is sent with 10 ms interval. The selected rate is nearly sufficient to provide a MPEG-1 video stream, which needs a data rate up to 1.5 Mbit/s [7]. Since the goal was not to test the capacity, the low data rate was reasonable.

Simulation parameters used in the simulations for this paper are recommended and optimized parameter values for link triggers and router configuration based on the previous work in this field [8-11]. Router configuration parameters changed in the default values from the package tool in order to have optimized results are:

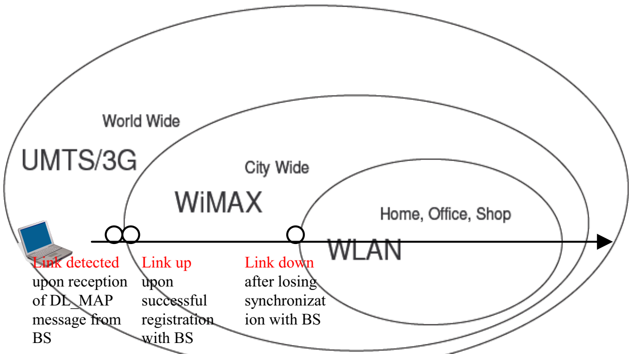


Fig. 1. Simulated scenario (one of the 15 simulated trajectories).

* MIN_RA_DELAY=200 s;
* Router lifetime=1800 s;
* MIN_DELAY_BETWEEN_RA=0,03 s;
* MAX_RA_DELAY= 0 s;

and link down generation parameters that are changed in the module are:

* Missed beacon threshold=2;
* Packet error threshold=4.

DCD and UCD synchronization messages in IEEE 802.16 standard (WiMAX) as we mentioned previously are transmitted by the BS periodically. The maximum interval is 10 s in the standard. In our simulation, with the 802.16 model from NIST, the DCD and UCD interval is set to 5 s. In order to optimize the vertical handover latency, packet loss and throughput performances we compared the results with DCD/UCD set to 1 s and run 15 random simulations. Average results of all simulations for vertical handover latency and packet loss are presented below.

IV. SIMULATION RESULTS AND ANALYSIS

A. *Vertical Handover Latency*

Figure 2 presents the results of average vertical handover latency over 15 random simulations when MN performs 4 vertical handovers while moving through the simulated scenario. Results show that UMTS/WiMAX average vertical handover latency has the worst results. This happens because 802.16 interface before the vertical handover is turned off. If 802.16 is OFF, during the vertical handover process, synchronization with BS (2.5s with DCD/UCD interval of 5s) is needed. During the other 3 vertical handovers 802.16 interface is turned on. Hence the results in Figure 2 show significantly lower vertical handover latency compared to UMTS/WiMAX vertical handover. When we decrease the DCD/UCD interval to 1 second, we decrease the synchronization time during the vertical handover process. That’s why the vertical handover latency has decreased significantly for UMTS/WiMAX vertical handover. Decreasing DCD/UCD interval to 1 s doesn’t impact the vertical handover latency on other 3 vertical handovers, because synchronization is not needed (802.16 interface is still turned on). VHO latency results are similar for simulated VoIP and video traffic. When DCD/UCD interval is 1 s, average VHO latency of the 15 simulations is 641,49 ms for VoIP G.723.1 and 642,94 ms for video MPEG1 traffic.

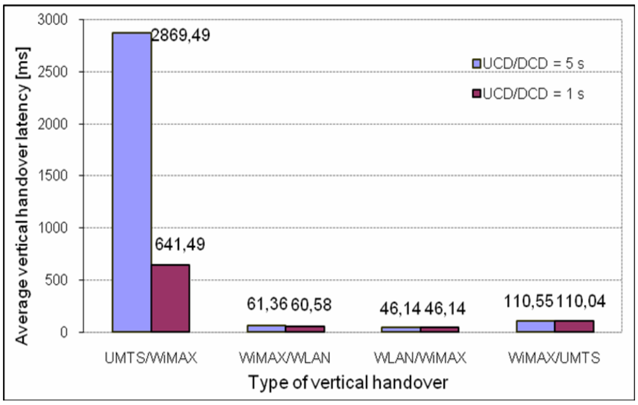


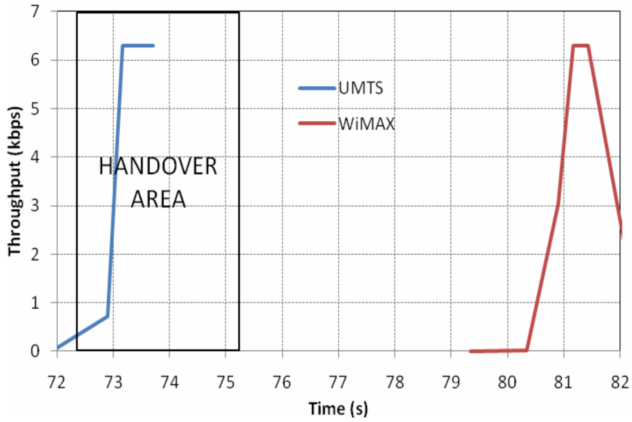
Fig. 2. Average vertical handover latency of 15 random simulations for VoIP (G.723.1) traffic.

B. *Throughput performances for UMTS-WiMAX*

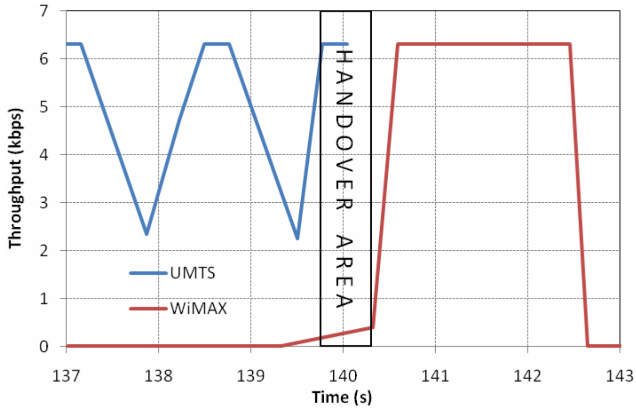
Comparing the results in Figure 3 and 4, we can conclude that throughput degradation during the vertical handover between UMTS and WiMAX networks has great improvement when DCD/UCD interval is decreased to 1 s. When the interval is 1 s, just after the handover, throughput drastically increases. Throughput gap between the end of the vertical handover process and the initiation of the VoIP traffic between the MN and WiMAX BS is much higher when DCD/UCD interval is 5 s. Thus, we can resume that when synchronization time is reduced, the whole VHO process is decreased and packet drop is also decreased, which results into the slight degradation of throughput.

But, reducing the synchronization time by increasing the frequency of the channel descriptor messages comes generally at the cost of a higher bandwidth overhead (less bandwidth available for user traffic). This effect reduces the capacity of the WiMAX network, so the total user traffic will be decreased at the cost of better vertical handover latency results. Thus, we have to decide do we need better VoIP QoS of the users, or better capacity of the cell at the cost of poor VoIP results.

Figure 4 shows the throughput results during vertical handover process between UMTS and WiMAX for video traffic for one of the 15 random simulations that is closest to the average vertical handover latency of all 15 simulations. We can conclude from the results that throughput between the MN and the UMTS base station has high value for most of the vertical handover duration between UMTS and WiMAX and very low results – almost 0 most of the handover duration for the throughput between MN and WiMAX base station. This was not the case with the VoIP traffic if we compare the second part of Figure 3 with Figure 4. VoIP traffic had variable throughput results with UMTS and WiMAX base stations during the vertical handover when we observed results of the 15 simulations. Because of the lack of space we cannot display here all results from the 15 simulations. Reasons for this difference between the VoIP and video results are because of the different type of traffic model. VoIP in the simulations was simulated as two-state model with on-off periods exponentially distributed and video was simulated as constant bit rate traffic.



a) DCD/UCD interval set to 5 seconds



b) DCD/UCD interval set to 1 second

Fig. 3. Throughput performance for UMTS/WiMAX vertical handover for value of average VHO latency of the simulations for VoIP (G.723.1) traffic.

VoIP has on and off periods and vertical handover process between UMTS and WiMAX for random simulations has started at different positions of this periods. That is the reason for the variety of throughput behavior for VoIP traffic compared with video traffic, where the simulation runs has similar results for throughput behavior during the vertical handover process between UMTS and WiMAX.

C. *Packet loss for UMTS-WiMAX*

Figure 5 shows the packet loss during the vertical handover process between UMTS and WiMAX of each of the 15 simulation runs. The average packet loss during vertical handover of the MN with the UMTS base station (when DCD/UCD interval = 5 seconds) is 2, and with the WiMAX base station is 5,13.

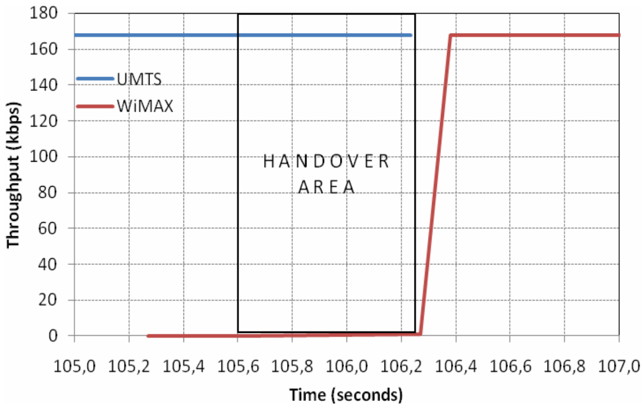


Fig. 4. Throughput performance for UMTS/WiMAX vertical handover for value of average VHO latency of the simulations for video traffic (DCD/UCD set to 1 second).

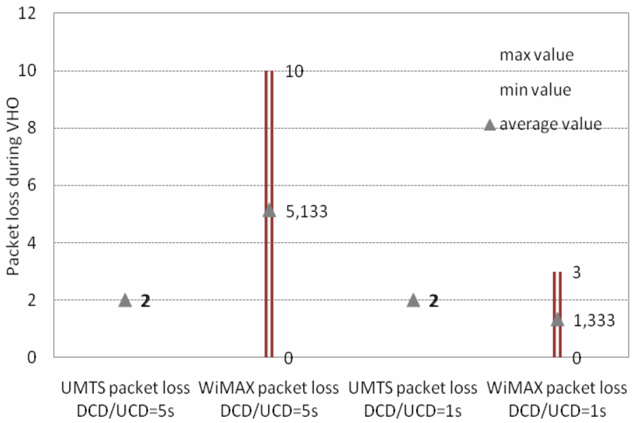


Fig. 5. Packet loss during UMTS/WiMAX vertical handover process for each of the 15 random simulations for VoIP (G.723.1) traffic.

Decreasing the DCD/UCD interval to 1 second improves the results of average number of packet loss between MN and WiMAX BS during vertical handover process. It is significantly decreased to 1.33. Packet loss between MN and UMTS BS during the vertical handover normally has the same value of 2 in both cases, because DCD/UCD interval is WiMAX parameter. Hence, using DCD/UCD interval of 1 second or sleep mode of the IEEE 802.16 interface will give satisfying results of packet loss during vertical handover process between UMTS and WiMAX in our simulations.

V. CONCLUSION

The main task in this paper was to find the best result on the performance metrics (handoff latency, packet loss and throughput) during the vertical handovers for real time traffic (VoIP and video), considering significant impact of the layer 2 vertical handover synchronization parameter (DCD/UCD interval). The other parameters are optimized for getting the best results according referenced papers.

In this paper we have done research regarding the effect of the DCD/UCD interval on the vertical handover latency, throughput and packet loss (also during vertical handover process) using VoIP (codec G.723.1) traffic between UMTS (HSPA), WiMAX (IEEE 802.16) and WLAN (IEEE 802.11) networks. Furthermore we compared the optimized VoIP results for vertical handover with video results using the same simulation parameters.

We have showed that decreasing of the DCD/UCD interval from 5 seconds to 1 second gives great improvement of the vertical handover latency, packet loss and throughput performances during vertical handover process. We can conclude that for better results for the quality of the VoIP and video sessions between the users during vertical handovers between UMTS and WiMAX, we must decrease the DCD/UCD interval, which will also impact the capacity in the networks.

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