Evaluating WiMAX QoS performance in a real testbed

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Abstract — Broadband Wireless Access technologies are expected to play a central role in next generation networks. WiMAX, based on the IEEE 802.16 standard, has the potential to form the foundation upon which operators will deliver ubiquitous Internet access in the near future. It is also widely anticipated that the next generation wireless networks will handle an exponentially growth of audio/visual content. In order to evaluate the QoS performance over WiMAX, it is important to test the WiMAX system with real time services, such as VoIP and video streaming. This paper presents an evaluation of the WiMAX QoS performance using these services.

I. Introduction & Related Work

Ubiquitous Internet access is one of the biggest challenges for the telecommunications industry in the near future. User's access to the Internet is significantly growing in the current days and will be a requirement in next generation networks. This is very challenging for the operators that will have to find a way to provide broadband connectivity to the users, independently of their location. Additionally, the demand for high bandwidth services and applications will also be required.

IEEE 802.16 [1] [2], also known as WiMAX, is an attractive solution for this type of next generation environments. It is a Point-to-MultiPoint (PMP) technology, providing high throughputs in Wireless Metropolitan Area Networks (WMANs). The IEEE 802.16 standard reference model is composed by the data (Physical layer and by the Medium Access Control layer of the protocol stack), control and management planes. Multiple physical layers are supported, operating in the 2 – 66 GHz frequency spectrum and supporting single and multi-carrier air interfaces, each suited to a particular environment. This wireless technology supports intrinsically Quality of Service (QoS) functionalities in the MAC layer, through the usage of connections and unidirectional service flows (SFs) between the Base Stations (BS) and the Subscriber Stations (SS). Five scheduling services are defined to meet different QoS needs: Unsolicited Grant Service (UGS – supports real-time SFs that generate fixed size data packets on a periodic basis, such as VoIP); Extended Real-Time Polling Service (ertPS – supports real-time SFs that generate fixed size data packets on a periodic basis, but the allocations are dynamic); Real-Time Polling Service (rtPS – supports real-time SFs with variable sized data packets on a periodic basis, such as video); Non-Real-Time Polling Service (nrtPS – supports non-real-time SFs that require variable size data grants on a regular basis, such as high bandwidth FTP); and Best Effort (BE).

Pioneering and closely related work to this was published by Scalabrino et al. [3], [4]. Using a fixed WiMAX testbed deployed in Turin, Italy, they focus on VoIP performance over WiMAX in particular when service differentiation is employed in the presence of significant amounts of elastic background traffic. Unfortunately, although their testbed included three SSs, the authors do not report any results from their simultaneous use. That is, their evaluation considers only point-to-point links. The same applies to the results reported by Grondalen et al. [5] from a fixed WiMAX field trial in Oslo, Norway. Their main means of evaluation are bulk TCP and UDP transfers. They measure throughput in both Line-Of-Sight (LOS) and Non-Line-Of-Sight (NLOS) conditions and correlate it with the received signal strength indicator (RSSI) values. Grondalen et al. reported that their WiMAX system (employing the same modulation and FEC as the one used in this paper) can deliver 9.6 Mb/s to a single flow in the downlink even at a distance of 5 km from the BS. Mignanti et al. [6] also report on FTP and VoIP performance over WiMAX in the Wind testbed in Ivrea, Italy. Their results indicate acceptable mean opinion scores for VoIP in a cell with a 2 km radius, but do not comment on overall (cumulative) throughput. Unfortunately, the results in [6] are not directly comparable to ours due to differences at the physical layer (WiMAX equipment and modulation schemes are different from ours). More recently, Pentikousis et al. [7] [8] published a set of tests considering video streaming and VoIP in a WiMAX PMP testbed. This was the first publicly available evaluation of VoIP and video streaming over a WiMAX testbed considering the simultaneous use of two SSs in the same cell.

None of these results considers evaluations of the WiMAX QoS performance with real time services in a PMP testbed. In this paper, we will use the same methodology as in [7] [8]. However, we do not only measure the capacity of the WiMAX link using Best Effort as a scheduling service, but we also evaluate the performance of WiMAX when different types of applications, with different requirements, are used.

The paper is organized as follows. This section presented an overview of the WiMAX technology and described the related work in the evaluation of WiMAX performance in real testbeds. Section II defines the scenario and methodology of our experiment, including the measurement of the WiMAX link capacity, traffic emulation and host clock synchronization. Section III presents the performance results. Finally, Section IV describes the conclusions of this work.

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II. SCENARIO AND METHODOLOGY

The demonstrator implemented to validate and evaluate the WiMAX QoS performance is illustrated in Figure 1. We employ multiple competing traffic sources over a PMP WiMAX topology and measure the capacity of the WiMAX link to handle a multitude of VoIP flows between the SSs, while simultaneously delivering a variable number of IPTV streams. We emulated an IPTV service running between the Correspondent Node (CN – connected to the WiMAX BS) and the WiMAX Terminal 1 (WT1 – connected to SS1), in parallel with QoS and Best Effort VoIP conversations, both running between WT1 and WT2. By gradually increasing N, the number of IPTV A/V streams, we determined the "breakpoint" of the WiMAX downlink channel. For each N, we repeated 10 times the run, which lasted 60 seconds. Regarding the performance metrics, we have measured jitter, throughput, packet loss, and one way delay for LOS conditions.

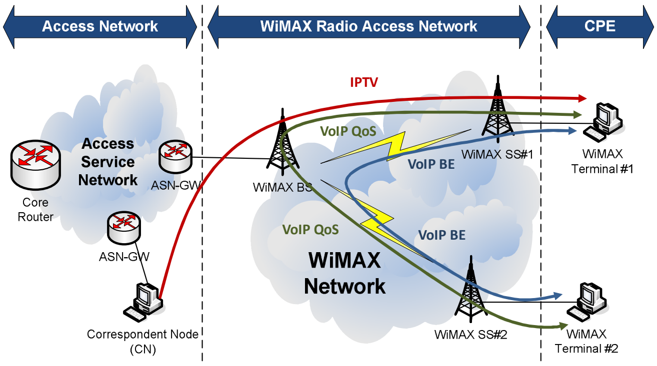

Fig. 1: WiMAX testbed to evaluate QoS performance

Before proceeding with the evaluation, we have measured the maximum throughput that can be obtained in the WiMAX system. We saturated the WiMAX link and measured the maximum application-level throughput, also called goodput, on the downlink and uplink directions. We made the test with different maximum transmission units (MTU), and obtained the best results for application payloads of 1472 bytes (MTU = 1500 bytes, the recommended MTU size for IEEE 802.16 standard-compliant equipment). For the uplink, the average maximum measured goodput was 4.75 Mb/s and for the downlink it was 5.75 Mb/s, with negligible (<0.1%) packet loss.

In order to emulate a set of IPTV streams, we used twenty minutes of live IPTV unicast transmission and created a packet trace. The captured video stream was in H.264/AVC format (also known as MPEG-4 Part 10) [9] and the accompanying audio stream was encoded in MPEG-1 Audio Layer II (also known as MP2) [10]. The content of the transmission was a music video TV channel configured with video stream at 512 kb/s (360×288, 25 f/s), and the audio at 192 kb/s, emphasizing audio over video quality. The captured video stream has a variable bit rate (VBR). The total packet sizes of the video varied greatly, with the biggest value being at 1492 bytes. It was also emulated the corresponding IPTV audio stream using constant bit rate (CBR) traffic with the total packet size fixed at 634 bytes (including codec payload and RTP/UDP/IP/MAC headers). The video and audio parts of the IPTV traffic are separated and streamed to different ports. Using the obtained packet trace, it was possible to create trace files with all packet sizes and inter-arrival times for video and audio. Based on these trace files we "playback" N IPTV A/V streams starting from a random point in the twenty minute long IPTV packet trace. We use JTG [11] to generate the trace-driven IPTV streaming traffic. The source of the N A/V streams is located at the CN, connected to the BS, while the sink is the WT1 connected to SS1.

In addition to the N A/V streams, we injected C bidirectional VoIP flows using JTG with source/sink pairs in the domains of SS1 and SS2. We have chosen Speex [12], an open source audio codec specially designed for VoIP applications over packet switching networks. Speex is designed to be robust against packet loss and has been incorporated in several applications. We emulated C Speex VoIP flows each with a wideband codec bitrate of 12.8 kb/s using JTG. For each VoIP flow, JTG generates 50 packets/s with 32 bytes of codec payload, thus leading to an effective application bit-rate of 17.6 kb/s (including RTP headers). After adding a total of 28 bytes of UDP and IP headers, each JTG instance injects 28.8 kb/s of total emulated Speex CBR traffic into the network. In order to test VoIP backhauling inside the same WiMAX cell, we introduced C = 50 simultaneous, bidirectional flows, yielding an application goodput (Speex payload plus RTP header) of 880 kb/s. This is only 18.5% of the maximum uplink goodput of 4.75 Mb/s, measured with MTU sized UDP packets.

For high-precision one way delay measurements, accurate clock synchronization is necessary, taking care of both absolute time and clock drift at different hosts in the network. For the one way delay measurements, both absolute time and clock drift are important. We used the IEEE 1588 Precision Time Protocol (PTP) open source server (PTPd) [13] to synchronize the clocks of all hosts. Although PTP injects a very small amount of traffic when compared with the rest of the sources in our tests, it is preferable that PTP signaling does not interfere with the measured traffic, and therefore the testbed synchronization was made using a different network. After initializing the PTPd in each machine and waiting the necessary time for achieving synchronization, the offset between the different host clocks was lower than 100 µs.

III. RESULTS

At this section our measurements are presented in boxplots. The box in each figure contains the middle 50% of the measured values. The line in the middle represents the median, whereas the top and the bottom of the box correspond to Q3 (median for the second half of the data) and Q1 (median for the first half of the data). Values outside the whisker lines, shown as crosses, are considered outliers.

A. Tests without QoS

Firstly we have performed some tests without QoS. Four Service Flows (SFs) were created, two per SS (one for uplink and one for downlink), to permit that all the traffic generated by the different sources will pass through WiMAX links and hit the defined sinks. Taking into account that these tests were performed with Best Effort, both IPTV (N streams) and VoIP (50 simultaneous, bidirectional flows) traffic for SS1 can pass through the same SF, as illustrated in Figure 1.

Comparing the results at SS1 for audio, video and VoIP we conclude that audio is the real time traffic that can achieve transmission rates closer to the theoretical rate because of its higher value of jitter. It means that audio is the traffic that is more strongly adjusted at physical level, in order to achieve a transmission rate more close to reality. It is followed by VoIP and finally by video which is the traffic that has the lower value for jitter (see Figure 2).

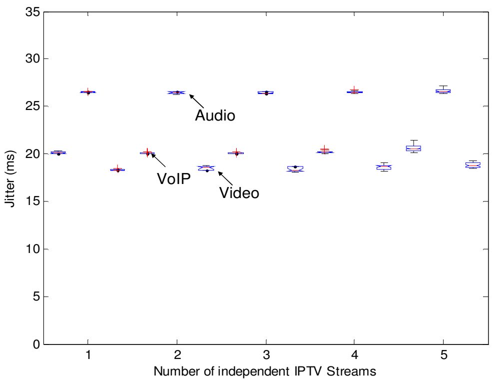

Fig. 2: Measured Jitter

The one way packet delays (see Figure 3), in the BS-SS1 link, as measured by the packet inter-arrival times at SS1, are quite similar across all traffic types. For audio and video the one way packet delays is approximately 20/25 ms – this is the involved delay in the WiMAX. For VoIP it is roughly twice 45/50 ms, because this traffic passes two WiMAX links. Overall, mean delay < 60 ms for N <= 3. This range of one way delays can be tolerated by all applications involved in the examined scenario. When N = 4, the median value of the inter-arrival times jumps to 60/80 ms for audio and video traffics and 80/100 ms for VoIP traffic, as the majority of the received packets are queued in the network buffers. This range of one way delays can be handled with adequate buffering for the IPTV streams, but not for the VoIP calls.

The BS-SS1 WiMAX downlink can handle N <= 3 simultaneous A/V streams in parallel with the VoIP traffic with negligible packet loss. When N = 4, packet loss increases rapidly, and even the packet loss average does not exceed 5%, there are occasional situations in which it exceeds, for VoIP, which is unacceptable (see Figure 4). Even for the Speex codec, which is the most robust and tolerant to packet loss of the three codecs emulated, this situation would degrade the performance considerably. The IPTV video streams suffer packet losses that does not exceed 5% for video and audio, with N<=4, which could be handled satisfactorily by a real-world IPTV client. When N > 4, packet losses exceed 15%, which is also unacceptable.

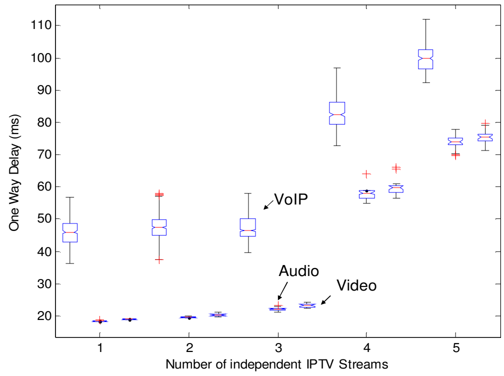

Fig. 3: Measured One Way Delay

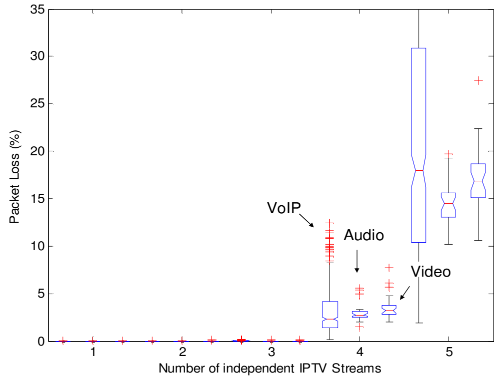

Fig. 4: Measured Packet Loss

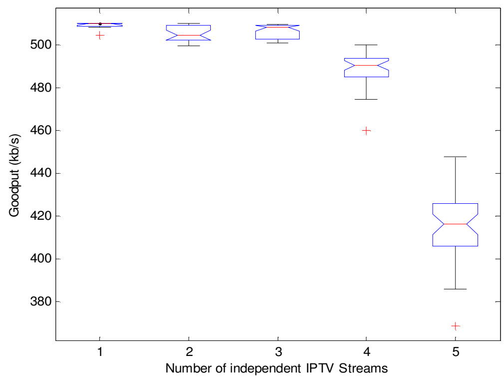

Fig. 5: Measured Goodput for Video

The difference between the VBR H.264/AVC encoded video streams and CBR audio and VoIP is visible. Goodput of video ranges between 500 kb/s and 512 kb/s, when N <= 3, as one would expect (see Figure 5). Meanwhile, the throughput of the audio and VoIP streams remains very close to 178 kb/s (see Figure 6) and 17.6 kb/s (see Figure 7), respectively. When N > 3, the capacity of the WiMAX downlink becomes a restrictive factor and the median goodput of all traffic types starts to fall. The spread of the average goodput in different runs starts to increase, as packets are dropped due to backlogs at the WiMAX interface. When the normalized values of goodput are examined, we note that the behavior of the three different traffic types is practically the same when N > 3, which is the “breakpoint” of the WiMAX downlink.

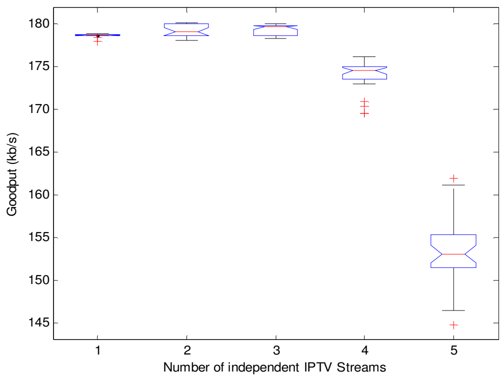

Fig. 6: Measured Goodput for Audio

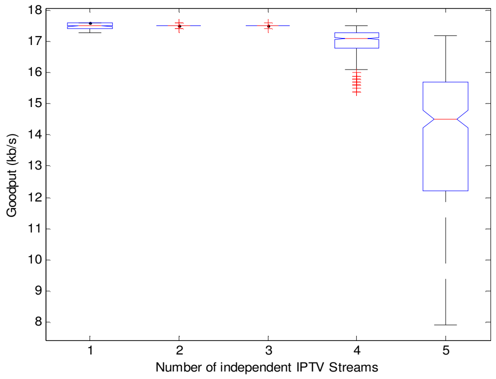

Fig. 7: Measured Goodput for VoIP

B. Tests with QoS (and background traffic as Best Effort)

To study the behavior of the WiMAX system using different service classes, we have used the rtPS service class for both VoIP QoS traffic and IPTV traffic, but giving lower priority to the IPTV traffic. The rtPS service class had a minimum bandwidth allocated of 1440 Kb/s (50 flows of VoIP x 28.8 Kb/s). For the IPTV traffic we have decided to assign the rtPS service class without associating a minimum bandwidth – IPTV traffic may have associated some delay. For the VoIP QoS traffic, four SFs were created, two per SS (one for uplink and one for downlink). For IPTV traffic we have created a downlink SF on SS1 domain. The VoIP BE traffic between SS1 and SS2 is emulated in a similar way to the VoIP QoS traffic, that is, 50 simultaneous VoIP flows are sent between SS1 and SS2, through the BE service class, in order to differentiate it from the VoIP QoS traffic. The SFs created in the WiMAX system are illustrated in Figure 1.

When N >= 4, the one way delay for IPTV increases faster compared with the results with Best Effort. In this case the priority for IPTV traffic is lower than for VoIP (with QoS) traffic. Therefore, the IPTV traffic has to wait for the WiMAX channel to be free (see Figure 8). For VoIP (with QoS) the results were expected because this is the most priority traffic. The delay is associated with the WiMAX equipment used because it is usually the delay involved in a WiMAX link. The Best Effort (VoIP without QoS) traffic has to wait more time to be serviced when the WiMAX link starts to saturate because it has a lower priority (see Figure 9).

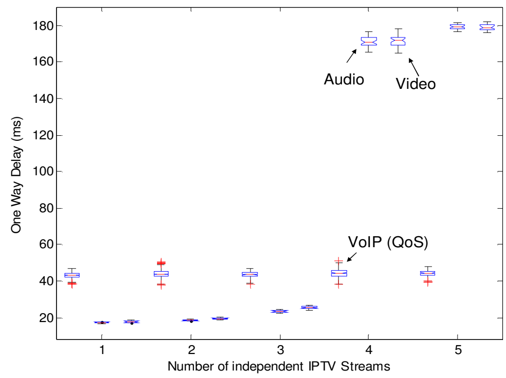

Fig. 8: Measured One Way Delay for Traffic with QoS

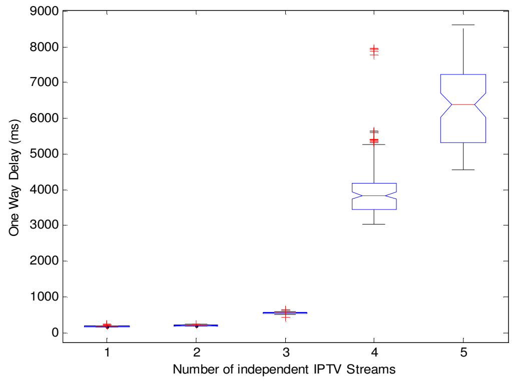

Fig. 9: Measured One Way Delay for BE traffic

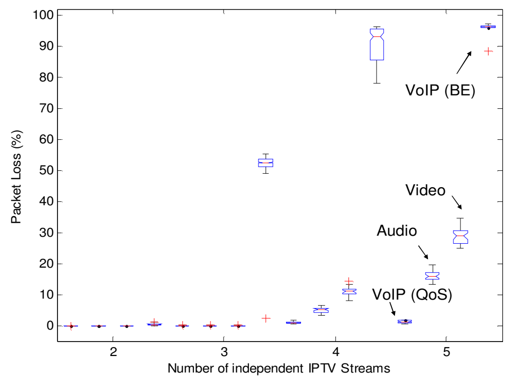

Fig. 10: Measured Packet Loss

Accordingly with the results at SS1 for audio and video, the BS-SS1 WiMAX downlink can handle N <= 3 simultaneous A/V streams in parallel with the VoIP traffic with negligible packet loss (see Figure 10), as for tests with Best Effort. When N >= 4, packet loss for IPTV increases rapidly, which is unacceptable. The packet loss values for video and audio are higher than in the tests performed only with Best Effort. In this case, the priority for IPTV traffic (both audio and video) is lower than for VoIP (with QoS) traffic, and then

IPTV traffic has to wait for the WiMAX channel to be free. It causes the increase of packet loss because the queue in the WiMAX segment, for IPTV traffic, will saturate earlier than in the previous situation. For VoIP (with QoS) the results are as expected because this is the most priority traffic with lower packet loss. However, the Best Effort traffic (VoIP without QoS) presents a high level of packet loss, as we expected, since it has the lower priority value amongst all others. Note that with 5 IPTV streams, it keeps almost 100% of packet loss. As expected, this demonstrates that good quality in Best Effort is only possible when the link is not saturated.

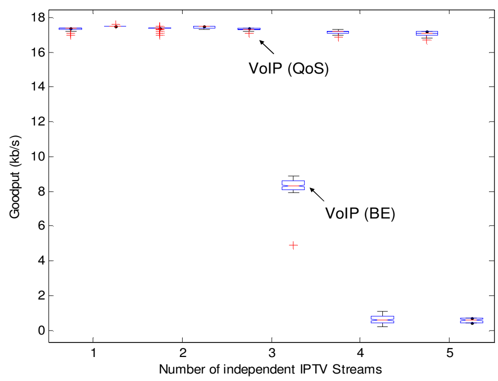

Fig. 11: Measured Goodput for VoIP

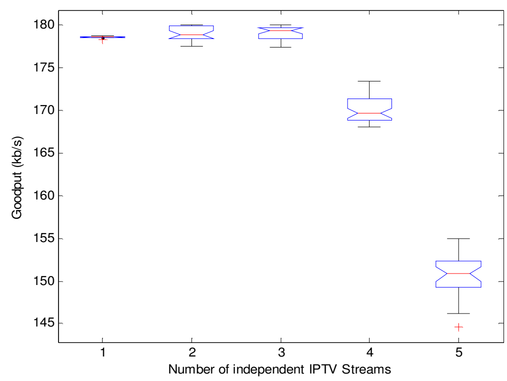

Fig. 12: Measured Goodput for Audio

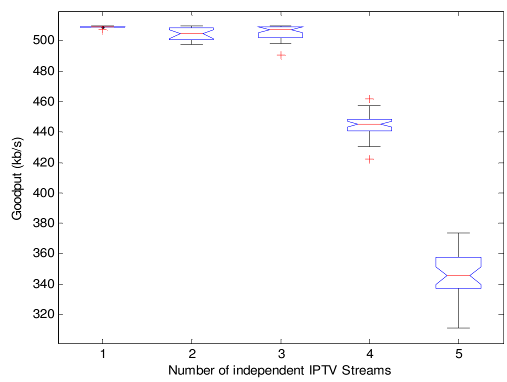

Fig. 13: Measured Goodput for Video

The application goodput results provide the same conclusions than the packet loss and one way delay results. In result of the higher service class for VoIP (with QoS), this traffic always has the bandwidth of WiMAX segment that it needs, whatever the number of IPTV streams (see Figure 11). When N >= 4, goodput for IPTV decreases rapidly, which is unacceptable (see Figure 12 and Figure 13). This is again reflecting the lower priority of IPTV traffic compared to the VoIP (with QoS) one. The Best Effort traffic has no bandwidth available when the WiMAX link starts to saturate because of its lower priority (see Figure 11).

IV. CONCLUSION

We have demonstrated that the WiMAX technology proves to be efficient and compliant with real time services and next generation environments. In order to evaluate the QoS performance over WiMAX, we performed several tests with the WiMAX system using real time services, such as VoIP and IPTV, in a PMP real testbed. In terms of the level of QoS achieved, WiMAX is able to support the different service requirements. Different service classes can be created, each one applied for a specific service, allowing an efficient scheduling mechanism. In our specific case, the requirements of both VoIP and IPTV traffic were different, with more strict requirements to VoIP, achieving then the best service from the network.

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