Experimental Evaluation of Multimedia Services in WiMAX

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ABSTRACT

WiMAX is a new technology that can provide long distance broadband wireless access based on IEEE 802.16 standards. After a short overview of IEEE 802.16 and the WiMAX network architecture, we center on the Quality of Service (QoS) support for multimedia applications in WiMAX. Inherent QoS support is an important factor differentiating WiMAX from other wireless access technologies, but so far has not been studied using off-the-shelf equipment. We fill this gap by empirically quantifying the performance of different multimedia applications and scheduling services in a WiMAX testbed. Specifically, the performance attained by multimedia applications over fixed WiMAX links, configured with Best Effort and Real Time Polling scheduling service classes is presented. We measure performance under both over-provisioned and under-provisioned configurations. When resources are substantially over-provisioned, the WiMAX equipment performs well and application requirements are fulfilled as expected. On the other hand, in under-provisioned conditions, applications using the Real Time Polling scheduling service enjoy better performance than those using the Best Effort scheduling service, but we note that proper configuration of the service flows may not be straightforward in practice.

Categories and Subject Descriptors

C.2.5 [**Computer-Communication Networks**]: Local and Wide-Area Networks; C.4 [**Performance of Systems**]: Measurement Techniques

General Terms

Experimentation, Measurement, Performance, Verification

Keywords

Network Measurement, WiMAX, IEEE 802.16, WiMAX testbeds, VoIP, Video Streaming

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1. INTRODUCTION

New and emerging services such as Video on Demand (VoD), IPTV, and triple play [1] bring multimedia content to end users at an ever-increasing rate. In this diversity of multimedia applications, voice and video applications are the most popular and are expected to dominate the traffic mix in next generation wireless access networks. To address the requirements of multimedia applications in the current specification of the Worldwide Interoperability Microwave Access (WiMAX) technology, the WiMAX Forum [2] defines different traffic models according to application requirements [3] in terms of bandwidth, delay and jitter.

The IEEE 802.16 standard is a wireless broadband access standard that includes two main specifications: IEEE 802.16-2004 [4] for fixed wireless access scenarios and IEEE 802.16e [5], which includes mobility support. One of the novelties introduced by the standard is the native support for Quality of Service (QoS). To enable such support, the standard specifies different scheduling services that are optimized for different kinds of applications. The IEEE 802.16 QoS model includes service flows that characterize the traffic transported in different connections. In fact a significant difference of IEEE 802.16 from other IEEE 802 standards is that connections between the Subscriber Station (SS), or Mobile Station (MS) in mobile environments, and the Base Station (BS) are identified by connection identifiers and not by MAC addresses.

WiMAX is based on the IEEE 802.16 standards and on the ETSI HiperMAN [6] standards. WiMAX complements the specification of IEEE 802.16 standards by defining a complete network architecture that includes the access and connectivity segments. The access service network includes the MS, the BS and the gateway responsible for the network access. The connectivity service network includes functionalities related with IP services, such as Authentication, Accounting, and Authorization (AAA) and IP Multimedia Services (IMS), among others. The WiMAX network reference model [7, 8] also takes into account support for mobility.

The main goal of this work is to evaluate the performance of multimedia applications over WiMAX. Our evaluation is based not on simulation, which is quite common in the literature, but on real WiMAX equipment with QoS support. We emulate voice and video traffic, transmitted over the WiMAX links that are configured with Best Effort (BE) or with Real Time Polling Service (rtPS) scheduling. Although WiMAX specifications cover a larger range of scheduling services, not all equipment support them. As such our evaluation is constrained by the testbed equipment. Nevertheless, we note that to the best of our knowledge this is the first such empirical evaluation published in the peer-reviewed literature. The BE and rtPS scheduling services are configured with different parameters including the maximum sustained rate and the maximum allowed delay, when applicable. Our bandwidth configuration considers an under-provisioned case, corresponding to the configuration with low values of available bandwidth, and an over-provisioned case where an excess of capacity is reserved. Our experiments verify that there is good support for multimedia applications by our WiMAX equipment in over-provisioned conditions. In the under-provisioned test cases there is a differentiation between the rtPS and the BE scheduling service classes, with the former providing better quality of service, in terms of delay and packet loss, when carefully configured.

This paper is organized as follows. Section 2 introduces the IEEE 802.16 standard and the WiMAX technology. Section 3 describes the multimedia applications and the evaluation process to assess their performance over the WiMAX testbed. Finally, Section 4 concludes the paper.

2. BACKGROUND

This section introduces the basics of the IEEE 802.16 standard and the network architecture specified by the WiMAX Forum.

2.1 An Overview of IEEE 802.16

The IEEE 802.16 standard, often cited as the last-mile wireless broadband access standard, includes a set of features such as native QoS and mobility support. The IEEE 802.16-2004 [4] (also known as IEEE 802.16d) and IEEE 802.16e [5] are the major versions of the standard. They define different functionalities, such as, operation in line of sight (LOS) and in non line of sight (NLOS) conditions, inherent support for different scheduling services, mobility, and extended coverage. The scheduling services supported include the Unsolicited Grant Service (UGS) for VoIP applications with constant bit rates; the Real Time Polling Service (rtPS) for video applications with variable bit rates; Extended rtPS (ertPS), defined in [5], for VoIP applications with silence suppression features; the Non Real Time Polling Service (nrtPS) for file transfer applications; and Best Effort (BE) service for web browsing applications.

In the IEEE 802.16 QoS model, each service flow is a unidirectional flow of packets with a particular set of QoS parameters, such as, traffic priority, maximum sustained traffic rate, maximum traffic burst, minimum reserved traffic rate, minimum tolerable traffic rate, tolerated jitter range, maximum delay, vendor-specific QoS parameters, and request/transmission policy. The standard specifies different types of service flows: Provisioned, Admitted and Active. Only active service flows are allowed to forward packets. The functional entities introduced in the standard are the Subscriber Station (SS), or Mobile Station (MS) in IEEE 802.16e, and the Base Station (BS). The BS is responsible for the centralized QoS scheduling inside its cell based on QoS parameters configured by the management system and the active bandwidth requests received from the SS. The SS or MS must identify a BS, acquire physical synchronization, obtain MAC parameters, and attach to the network.

The IEEE 802.16 reference model distinguishes between the data/control plane and the management plane, which

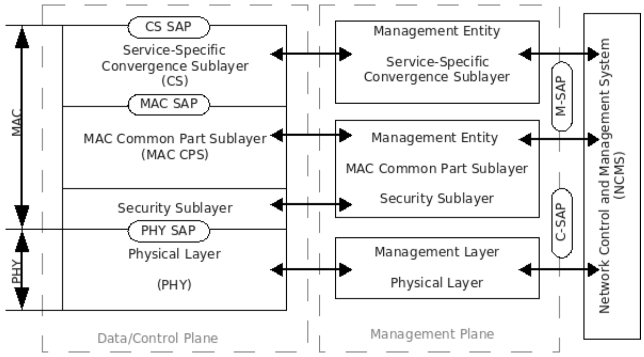

Figure 1: IEEE 802.16 data/control and management planes

is specified by the IEEE 802.16g [9] working group. The diverse functionalities span across the medium access control (MAC) and physical (PHY) layers, as depicted in Fig. 1. The MAC layer is divided into three sub layers: The service-specific Convergence Sub layer (CS), the MAC Common Part Sub layer (CPS) and the security sublayer. The interaction between the different sublayers is done through well defined Service Access Points (SAPs). The CS sub layer interfaces with higher-layer protocols and different CS sub layers are specified to handle different protocols such as Asynchronous Transfer Mode (ATM) and Internet Protocol (IP). The Packet CS is able to transport all packet-based protocols, such as IP, and is preferred in mobile environments.

The IEEE 802.16g standard introduces the Generic Packet Convergence Sub layer (GPCS), which is independent of higher-layer protocols, thus supporting multiple packet-based protocols. An important role of the CS is the classification of higher layer Protocol Data Units (PDUs) and their mapping onto the appropriate MAC service flow. The classification process is based on sets of matching criteria, such as IP address, ports and Type of Service (ToS) fields. The MAC CPS includes the necessary functionalities to control the medium access and manages the necessary operations to establish a connection between the SS and the BS.

As mentioned earlier, in IEEE 802.16 connections are identified by a Connection Identifier (CID) and not by the MAC address of the host as in other IEEE 802 standards, for instance IEEE 802.11. The SS MAC address is only used in initial ranging and authentication. The security sub layer provides privacy by encrypting the connections between the SS and the BS.

The mobility support introduced in the IEEE 802.16e standard includes power-saving specifications and handover procedures. With respect to power-saving, two modes of operation are specified: Sleep and Idle. The Idle mode is more power conserving than the Sleep mode, as the MS can turn off completely and become periodically available for downlink broadcast messages without being registered with any BS. With respect to mobility, although different handover types are supported in the standard, such as Hard Handover (HHO), Fast Base Station Switching (FBSS) and Macro Diversity Handover (MDHO), only HHO is mandated to be supported by all equipment. With HHOs, transfer interruptions are possible when a mobile node switches from one BS to another. The handover decision can be taken

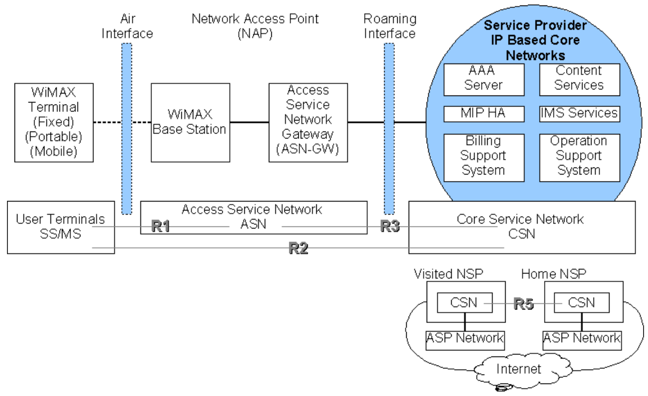

Figure 2: WiMAX Network Architecture

by the BS, MS or by another network entity. The MS gets knowledge of existing neighbors via management messages transmitted periodically by the BSs. Using this information the MS can perform scan and association procedures. Once the handover decision has been made, the MS begins the synchronization process with the target BS.

2.2 WiMAX Description

The WiMAX Forum [2] aims at defining an architecture within which different vendor equipment can interoperate flawlessly while conforming to IEEE 802.16 and ETSI HiperMAN standards. Fixed WiMAX is based on [4, 6] and supports fixed and nomadic access in LOS and NLOS conditions. Mobile WiMAX is based on [5] and, among others, adds mobility support in WiMAX networks. In short, the WiMAX Forum specifies an end-to-end architecture, instead of only focusing on the radio access segment of the network.

Fig. 2 depicts the WiMAX Forum network reference model [7, 8], which includes several different entities. The Network Access Provider (NAP), is a business entity providing WiMAX radio resources to one or more WiMAX Network Service Providers (NSPs) and controls the Access Service Network (ASN). NSPs are business entities that provide IP connectivity and WiMAX services to WiMAX subscribers. An NSP manages its own Connectivity Service Network (CSN). The Access Service Network (ASN) includes network elements such as the BS and the ASN Gateway (ASN-GW), providing network access to Mobile Stations. The ASN contains the network functions necessary to provide radio access to a WiMAX subscriber.

Communication between the different elements of the network architecture is based on reference points, which form the foundation for seamless interoperability (see Fig. 2). For instance, reference point R1 describes the protocols and procedures between MS and ASN, as detailed in [4, 5, 9]. Since the ASN concentrates on network access functionality, different implementation profiles for the ASN are defined. Profile A includes an ASN-GW and one or more BSs, while Profile B centralizes the implementation of the ASN functions into a single device. Profile C includes a distribution of the ASN functions between the ASN-GW and the BS. With Profile A, the ASN-GW manages handover and controls radio resources. Besides the IP connectivity assured by the CSN, IP address allocation, Internet access, billing operations and IP Multimedia Services (IMS) are also managed by the CSN. This last profile is preferred by the WiMAX Forum.

The mobility management considered by the WiMAX Forum for Mobile WiMAX supports IPv4 [10] and IPv6 [11] mobility management protocols as well as mechanisms for reducing packet loss and handover delay. Two types of mobility are considered: ASN-anchored mobility and the CSN-anchored mobility. ASN-anchored mobility, or micro-mobility, is devoted to the mobility procedures that occur without the need for a MS Care-of-Address update, since the MS moves its point of attachment between BSs of the same ASN. CSN-anchored mobility, or macro-mobility, considers IP mobility between ASN and CSN. Different types of Mobile IP (MIP) implementations are considered to support macro-mobility. The first one is aimed for MIP-enabled clients and the second for those nodes that do not support MIP, and therefore need assistance from the network to perform handovers. This last approach is based on Proxy Mobile IP (PMIP) [12]. With the MIP-aware approach, the MS is compliant with MIPv4 if deployed in IPv4 networks, or MIPv6 if deployed in IPv6 networks.

The WiMAX QoS framework extends the IEEE 802.16 QoS model by defining various QoS-related entities in the WiMAX network and the corresponding mechanisms for provisioning and managing various services flows. The WiMAX QoS framework supports static and dynamic service flow creation, although Release 1.0 [7, 8] only describes static provisioning. Also, QoS mechanisms only focus on the WiMAX radio link connections and no end-to-end QoS mechanisms are specified, therefore there is no provision of QoS in the core network. The WiMAX QoS framework includes the definition of abstract messages to convey triggers, initiate service flows actions, request policy decisions, download policy rules and update the MS location.

3. EVALUATION

This section describes the multimedia application characteristics we are interested in, in general, and presents our experimental results for voice and video applications in a WiMAX testbed.

3.1 Multimedia Applications

VoIP applications require assured bandwidth and specific bounds on delay and delay variation (or jitter) that depend on the employed codec, which encodes human voice into samples that can be transported over the network encapsulated in IP packets. Different codecs are specified in the ITU-T [13] recommendations, including G.726 [14], G.722 [15] and G.711 [16]. Although most VoIP applications support G.711 [17], it is not the most efficient codec in terms of bandwidth conservation or handling packet losses, and it does not perform data compression. On the other hand, G.711 does conserve bandwidth with features such as voice activity detection, which avoids sending full packets in periods of silence. Although it is quite demanding in terms of bandwidth, G.711 is thoroughly studied and used in the literature, and for this reason it is the codec we choose to use in this first study of QoS support in our WiMAX testbed.

Video applications have different QoS requirements than VoIP, determined largely by the data representation format (e.g. MPEG-4), resolution, frame rate, compression rate, color spaces and stream type. Streaming applications are often more tolerant to delay and jitter degradation than voice applications, since buffering can be employed efficiently al-

Table 1: PSNR and Mean Opinion Score for Video

PSNR	MOS	Description
>37	5	Perceptible (Excellent)
31-37	4	Perceptible but not annoying (Good)
25-31	3	Perceptible and slightly annoying (Fair)
20-25	2	Annoying but not objectionable (Poor)
<20	1	Very annoying and objectionable (Bad)

lowing the receiving client to absorb delay variations. The Common Intermediate Format (CIF) and the Quarter Common Intermediate Format (QCIF) are the most representative picture scanning formats of H.261 and H.263 video codecs. For instance, CIF defines a resolution of 352×288 pixels and approximately 30 frames per second. The YUV model is the preferred video color space since it models the human perception of color more accurately than other color spaces such as RGB, which is widely used in computer graphics hardware. The YUV model defines the color space in terms of one lumma component (brightness) and two chrominance (color) components.

The user-perceived video quality can be measured by calculating the Peak Signal Noise to Ratio (PSNR). PSNR is determined by comparing each pixel in the original frame with the (possibly) distorted received frame, thus allowing the evaluation of the distortion introduced by network transmission. Table 1 presents the relation between PSNR, measured in dB and the Mean Opinion Score (MOS) evaluation as given in [18]. MOS is defined in a five-point scale, quantifying the user-perceived video quality.

3.2 Testbed and Tests Description

This subsection describes the test conditions and the tools used to evaluate the performance of voice and video applications in our WiMAX testbed. The common measurement values for video and voice evaluation were packet loss ratio, one way delay, and jitter. The video applications were also evaluated according to the user perceived video quality measured in the MOS scale. To determine the WiMAX equipment performance, in different setups, and to assess the effectiveness of WiMAX QoS mechanisms, two distinct scenarios were evaluated:

- Under-provisioned: The bandwidth reserved is less than the application requirements.
- Over-provisioned: The bandwidth reserved is in excess of the application requirements.

3.2.1 Voice Tests

The evaluation of voice applications was performed with voice sessions with the duration of sixty seconds. Voice traffic was generated with the Distributed Internet Traffic Generator (D-ITG) [19]. The voice codec used was the G.711 with no voice activity detection and with one sample encapsulated per IP packet. The VoIP traffic was transmitted from the SS towards the ASN, i.e. the WiMAX uplink. The codec payload of 80 bytes corresponded to 10 ms of emulated conversation. The voice tests were performed in two configurations:

- Single client: A single flow was instantiated to determine the QoS differentiation of the different scheduling services.

Table 2: Voice Applications: Single-flow Tests

Test Case	B (kb/s)	d (ms)	s
160kb_2_rtPS	160	2	rtPS
160kb_100_rtPS	160	100	rtPS
160kb_150_rtPS	160	150	rtPS
160kb_300_rtPS	160	300	rtPS
160kb_na_BE	160	na	BE
80kb_2_rtPS	80	2	rtPS
80kb_100_rtPS	80	100	rtPS
80kb_150_rtPS	80	150	rtPS
80kb_300_rtPS	80	300	rtPS
80kb_na_BE	80	na	BE

- Multiple Clients: Different voice sessions were emulated, each one representing a VoIP client, to determine the level of support for simultaneous users in aggregated service flows.

Table 2 summarizes our test configurations for the single client case. Each test was identified by the reserved bandwidth (B) (160 kb/s or 80 kb/s), by the maximum allowed delay (d) (when applicable) and by the scheduling service (s) configured. For instance, the "160kb_2_rtPS" test was conducted with a reservation of 160 kb/s of bandwidth, a configured delay of 2 ms and employed rtPS scheduling.

In the single-flow cases, the tests with 80 kb/s correspond to under-provisioned configurations since the minimum required bandwidth for the generated traffic was 100 kb/s, whilst the 160 kb/s test cases correspond to over-provisioned configurations. The multiple client tests used the same parameters as the single flow cases (delay and scheduling service), but introduced also the simultaneous number of users (n). In all tests, a service flow was pre-configured with 1 Mb/s of reserved bandwidth. The tests with 25, 50 and 75 simultaneous clients represented the over-provisioned test cases, since only 0.3, 0.6 and 0.9 Mb/s were required, respectively. The test cases with 100 and 180 simultaneous clients correspond to under-provisioned cases since 1.2 and 2.15 Mb/s of bandwidth is required, respectively.

The different values for the delay parameter were based on the ITU G.114 recommendation [20], which specifies 150 ms for one-way delay between the sender and the receiver of voice applications and defines a maximum bound of 400 ms for an acceptable one-way delay.

3.2.2 Video Tests

In this paper we make use of the Evalvid framework, which provides a set of tools to convert raw video files into MPEG format in order to be transmitted over the network [18]. Evalvid can be used to evaluate network performance for video transmission, not only based on common network parameters, such as delay, jitter and packet loss, but also on more objective measures like PSNR and MOS that measure the user-perceived video quality. Evalvid determines the PSNR of the transmitted videos by comparing the original video files with the received videos, as described earlier. Based on PSNR, it is straightforward to determine the user-perceived video quality in the MOS scale.

The video evaluation was performed using a single client (located in the MS side) which received video traffic from the server located on the ASN side. In other words, the video flow was transmitted in the WiMAX downlink. The

Table 3: Video tests for each video file

s	*d* (ms)	*B* (Mb/s)
BE	N/A	2
rtPS	2	
	100	
	150	
BE	N/A	1
rtPS	2	
	100	
	150	

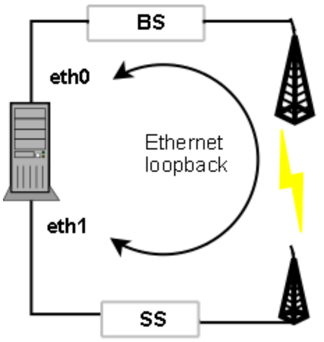

Figure 3: Testbed Schematic

video traffic was based on the well-known Foreman video file, which was prepared with the Evalvid tools to be transmitted in the testbed WiMAX link. The various video tests performed are listed in Table 3.

The minimum reserved bandwidth in all rtPS test cases was 500 kb/s. The video file had a bit rate of 1 Mb/s (as measured by the Evalvid tools), therefore 2 Mb/s of bandwidth represented the over-provisioned case while 1 Mb/s of bandwidth represented the under-provisioned case. The video evaluation process using the Evalvid framework consisted of different steps, which started with the conversion of the original raw video file format to the MPEG video file format, continuing with the creation of the reference video files and ended with the comparison of the transmitted and reference video files to determine the PSNR and, consequently, the corresponding MOS.

3.2.3 *Testbed*

Fig. 3 illustrates the layout of WiMAX testbed, which was based on an Ethernet loopback, deployed in a PC running GNU/Linux kernel ver. 2.6.22 with the "self-to-self" patch [20]. The kernel was patched because the standard kernel does not allow the functionality of a loopback between two different network interface cards on the same machine. This testbed setup was chosen for simplicity and convenience of measurement setup and it allows us to measure one-way delay while avoiding synchronization issues. The WiMAX equipment used in the testbed was the Redline RedMAX AN-100U Base Station and the Redline RedMAX Subscriber Unit outdoors Station and was configured according to the parameters given in Table 4.

3.3 Results

This subsection presents the results of our empirical study of voice and video applications in the WiMAX testbed.

Table 4: Testbed Configuration Parameters

Parameter	Value
RF Downlink Channel	3488000 KHz
Channel Size	7 MHz
Cyclic Prefix	1/16
DL Ratio	56%
Cell Range	5 Km

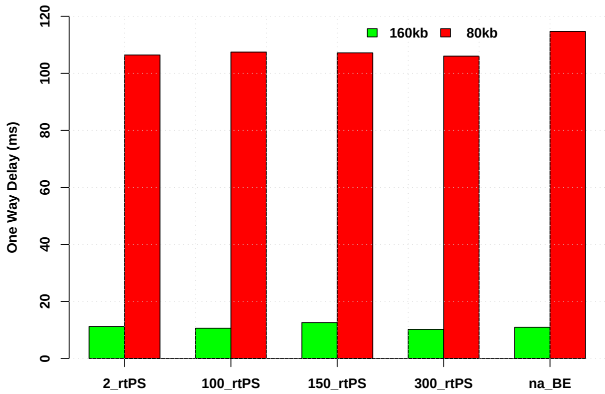

Figure 4: One-way delay for a single voice flow

3.3.1 *Voice Results*

The evaluation of voice traffic was performed using a single client and multiple clients, resorting to delay, jitter and packet loss measurements. In the case of the single client with a configured bandwidth of 80 kb/s, one-way delay was high, since bandwidth was under-provisioned for the flow. When the bandwidth was configured to 160 kb/s, the delay of the different classes was similar, with an average value of 10 ms, as depicted in Fig. 4. In the under-provisioned test cases the delay of the BE scheduling service was higher than the delay with the rtPS scheduling service. The test cases configured with the rtPS and with a delay of 300 ms had the best performance. Such fact was due to a non stringent value of delay (2 or 100 ms). Jitter in the single voice client test, had an average value of 1.2 ms in the over-provisioned test cases (160 kb/s). In the under-provisioned test cases jitter had higher values, around 14 ms.

Packet loss with the single voice client tests was negligible in the overprovisioned tests. In the under-provisioned test cases loss was on average around 28%, as Table 5 details. In these cases, BE scheduling performed actually slightly better than some of the rtPS test cases, in particular the 2 ms test case. This behavior is due to the stringent requirement on the maximum allowed delay. In the test cases configured with the BE scheduling service such criteria does not apply. Fig. 5 depicts the delay for the multiple voice client tests. As expected, delay increased with the number of simultaneous voice clients. For instance, delay had an average of 19 ms with the 25 voice clients and an average of 35 ms with the 180 voice clients. The rtPS scheduling service had a slightly better performance in terms of delay, when compared to the BE scheduling service. The 150 ms test case had an im-

Table 5: Packet Loss for Single Voice Client

Test Case	Packet Loss
2_rtPS	28.98%
100_rtPS	27.52%
150_rtPS	28.98%
300_rtPS	29.79%
na_BE	28.12%

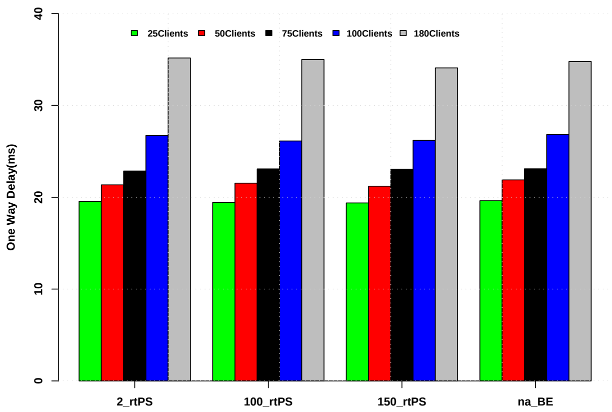

Figure 5: One way delay in the case of multiple voice clients

proved performance since it has less stringent requirements, when compared to the other two rtPS configurations (2 and 100 ms). Jitter, in the multiple voice client tests, increased with the number of simultaneous voice clients. For instance, jitter had a minimum value around 1 ms in the overprovisioned cases and a maximum value of 7 ms with 180 voice clients.

As expected packet loss increased with the number of simultaneous voice clients, in the multiple voice clients case. Although in the over-provisioned test cases packet loss was negligible, in the under-provisioned test case packet loss reached up to 46%.

In the over-provisioned test cases, both video and voice applications conformed to the ITU G.114 and ITU Y.1541 [21] recommendations. The G.114 recommendation specifies a bound of 150 ms for one way delay of voice conversation, and the Y.1541 recommendation presents different QoS classes and defines, for each class, different values for the network performance parameters. The Y.1541 classes 0 and 1 characterize voice traffic and necessitate a packet loss ratio bellow 0.1% for best performance. The under-provisioned test cases had acceptable delay bounds but extremely high packet loss, and thus unacceptable quality, with loss rates around 28%. The multiple voice client tests supported 75 simultaneous clients with a good conversation quality in terms of delay and packet loss metrics. On the other hand, with 180 simultaneous clients in a 1 Mb/s aggregated service flow we measured a packet loss rate of 46%.

3.3.2 Video Results

Fig. 6 depicts the MOS classification of the Foreman video, assuming one video client. All over-provisioned test cases

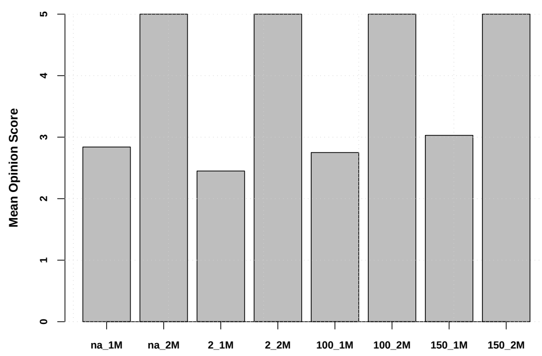

Figure 6: MOS of Foreman Video.

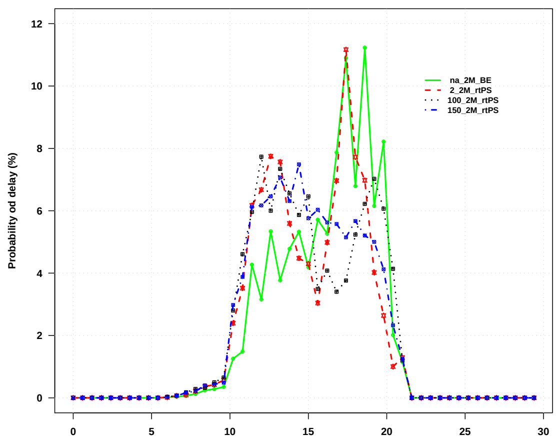

Figure 7: PDF Delay of Foreman Video(2Mbps).

had maximum points in the 5-point scale, as there was no packet loss. In the under-provisioned test cases the video classification depended on the scheduling service and on the delay configured. For instance, the rtPS test cases configured with a maximum delay of 2 and 100 ms had a lower classification when compared to the test cases configured with the BE scheduling service. This is due to the high packet loss in the 2 and 100 ms test cases caused by the rigorous admission criteria (excessively low delay bounds). Packet loss with these low delay bounds occurred as soon as the buffers dedicated to this service flow were full.

Fig. 7 draws the PDF delay for the Foreman video in the overprovisioned test cases. The tests with the BE scheduling service had a higher probability for higher values of delay. For instance, the BE test cases had more probability of having a delay of 20 ms than the rtPS test cases. In the rtPS test cases, the 150 ms configured delay had the lowest probability of high delay when compared to the other test cases (2 and 100 ms). In the underprovisioned test cases (not shown) the delay had probability of values around 30 ms.

Video files transmitted in service flows with different QoS

Figure 8: Foreman Video with 1Mbyte

configuration had different behaviors according to the bandwidth reserved for the service flow. With overprovisioned test cases (2 Mb/s), the video was received with an excellent quality. On the contrary, in the underprovisioned cases, video quality was lower and presented annoying features, as depicted in Fig. 8.

4. CONCLUSIONS

The WiMAX capability to support multimedia applications, and in particular VoIP and video streaming, were empirically evaluated on a testbed. Two overall test conditions were considered, one where resources are over-provisioned and another with considerable under-provisioning. The results show that both VoIP and video streaming behave according to the recommendations of ITU G.114 and ITU Y.1541 in the over-provisioned case. However, in under-provisioned conditions, both voice and video applications do not follow these recommendations. Nonetheless, the rtPS scheduling service offers better QoS support when the maximum allowed delay for the service flow is well configured. Additionally, the results have shown that it is possible to support up to 75 simultaneous G.711 VoIP clients in a service flow configured with 1 Mb/s, which is an good indication about the potential of fixed WiMAX as a backhaul and coverage extension technology.

Clearly, there is still lots of empirical work lying ahead. We barely scratched the surface of what is possible to do with state of the art WiMAX equipment. Moreover, we plan to validate and contrast the results in the lab with those obtained using one of the popular simulators. We expect that our measurements will be of interest to simulationists and practitioners alike.

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