# Experimental Evaluation of Multimedia Services in WiMAX


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## ABSTRACT

WiMAX is a new framework to provide long distance broadband wireless access using the IEEE 802.16 standard. One of the most important characteristics of WiMAX is the support of applications with different requirements in terms of parameters such as delay, jitter and bandwidth, as for instance real-time applications. The WiMAX capability to support applications with special needs is analysed in this paper through extensive experimentation on a WiMAX test-bed. Specifically, the performance obtained for multimedia applications over WiMAX links, configured with Best Effort and Real Time Polling scheduling service classes is presented. The experimental results show that, on one hand, when resources are over-provisioned, the performance of the WiMAX equipment is very good and the requirements of applications are fulfilled, as expected. On the other hand, in under-provisioned conditions, applications using the Real Time Polling scheduling service have better performance than those using the Best Effort scheduling service, given that the configuration parameter that defines the maximum allowed delay for a flow is well configured.



## Categories and Subject Descriptors

C.2.5 [**Local and Wide-Area Networks**]

## General Terms

Experimentation, Measurement, Performance, Verification.

## Keywords

Network Measurement, WiMAX, IEEE 802.16, WiMAX test-beds, VoIP, Video Streaming


## 1. INTRODUCTION

New emerging services such as, Video on Demand (VoD), triple play and IPTV [1] bring multimedia content to the end user. In this diversity of multimedia applications, voice and video applications are the most popular. To address the requirements of these applications in the current specification of the Worldwide Interoperability Microwave Access (WiMAX) technology, the WiMAX Forum [2] defines different traffic models according to the requirements of the applications [3] in terms of bandwidth, latency and jitter requirements.

The IEEE 802.16 standard is a wireless broadband access standard that includes two main specifications, the IEEE 802.16-2004 [4] for fixed scenarios and the IEEE 802.16e [5] to support mobility. 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 QoS model defined by the IEEE 802.16 standard includes service flows to characterize the traffic that can be transported in the different connections. Also the 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 the MAC addresses as in other IEEE 802 standards.

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

The main goal of this work is to evaluate the performance of multimedia applications over WiMAX networks. The evaluation is based on applications emulating the voice and video traffic, transmitted over the WiMAX links that are configured with the Best Effort (BE) or with the Real Time Polling Service (rtPS) scheduling service. The choice of these scheduling classes was constrained by the classes available on the WiMAX equipment used on the test-bed. These scheduling services are configured with different parameters including the maximum sustained rate and the maximum allowed delay, when applicable. The configuration for the bandwidth is performed in two modes. Under-provisioned, corresponding to the configuration with low values of bandwidth and over-provisioned that corresponds to the configuration with an excess of bandwidth.

The experimentation reveals a good support of the WiMAX equipment for multimedia applications 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 first providing a better quality of service, in terms of delay and packet loss, when adequately configured.



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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 the performance of the WiMAX equipment. Finally, Section 4 presents the main conclusions and highlights issues to be addressed in future work.

## 2. BACKGROUND

This section introduces the functionalities of the IEEE 802.16 standard and the network architecture of the WiMAX technology.

### 2.1 An Overview of IEEE 802.16

The IEEE 802.16 standard, known 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. The IEEE 802.16 standard supports different functionalities such as, the operation in Line of Sight (LOS) and in Non Line of Sight (NLOS), the support for different scheduling services, mobility and the extended coverage. The different scheduling services supported include the Unsolicited Grant Service (UGS) for VoIP applications with constant bit rates, the Real Time Polling Service (rtPS) for MPEG video applications with variable bit rates, the Extended rtPS (ertPS), which is only available in the IEEE 802.16e, targeting VoIP applications with silence suppression features, the Non Real Time Polling Service (nrtPS) for file transfer applications, and the Best Effort (BE) for web browsing applications.

In the IEEE 802.16 QoS model, the service flow is a unidirectional flow of packets with a particular set of QoS parameters. The different QoS parameters include traffic priority, maximum sustained traffic rate, maximum traffic burst, minimum reserved traffic rate, minimum tolerable traffic rate, tolerated jitter, maximum latency, 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 or Mobile Station in the IEEE 802.16e standard, and the Base Station. The BS performs centralized QoS scheduling 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 is being addressed in IEEE 802.16g [9]. The diverse functionalities span across the MAC and PHY layers, as Figure 1 depicts. 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 sub layer. The interaction between the different sub layers is done through well defined Service Access Points (SAPs). The CS sub layer performs the interface with higher protocols and different CS sub layers are specified to support 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 mobility environments.

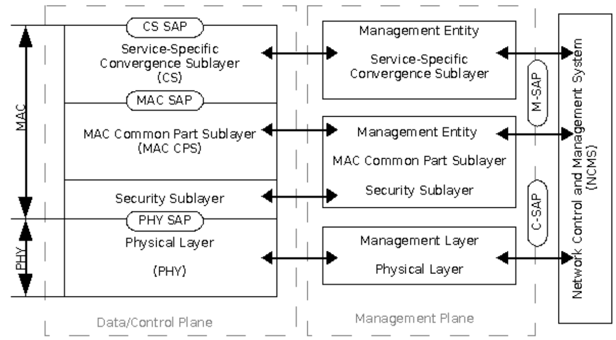


**Figure 1 - IEEE 802.16 data/control and management planes**

The IEEE 802.16g standard introduces the Generic Packet Convergence Sub layer (GPCS) that is independent of upper layer protocols, thus supporting multiple packet-based protocols. An important role of the CS is the classification of higher layer Protocol Data Unit (PDU) to map it to 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. The necessary operations to establish a connection between the SS and the BS are managed by the MAC CPS. 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 MAC address of the SS is only used in the 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. The Sleep and Idle modes are two power-saving modes specified. The Idle mode is more power conservative, when compared to the Sleep mode, since the MS can completely turn off and become periodically available for downlink broadcast messages without being registered with any BS. Although different handover modes are supported in the standard, such as Hard Handover (HHO) mode, Fast Base Station Switching (FBSS) and Macro Diversity Handover (MDHO), all the handover procedures are specified for the HHO mode. The HHO mode has the disadvantage of implying an abrupt transfer of connection from one BS to another when compared to the other optional modes. The handover decision can be made by the BS, MS or by a network entity. The MS gets knowledge of existing neighbours in management messages sent periodically by the BSs, with this information the MS can perform scan and association. Once the handover decision has been made, the MS begins the synchronization process with the target BS.

### 2.2 An Overview of WiMAX

The WiMAX Forum [2] is specifying the WiMAX technology, aiming the interoperability of equipments that conform to the IEEE 802.16 and ETSI HiperMAN standards. The different versions of WiMAX are: Fixed WiMAX and Mobile WiMAX. The Fixed is based on the IEEE 802.16-2004 and ETSI HiperMAN [6] standards and supports fixed and nomadic access in LOS and NLOS conditions. The Mobile WiMAX is based on the IEEE 802.16e standard and adds support for mobility.

The novelty of WiMAX is the specification of an End-to-End architecture, instead of only focusing the radio access segment of the network. Figure 2 depicts the network reference model [7, 8] specified by the WiMAX Forum.

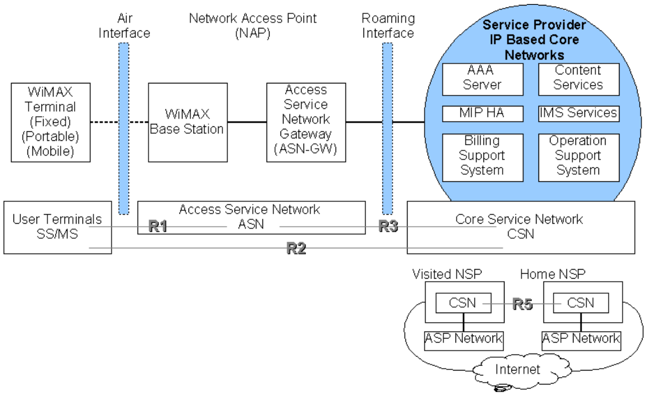


**Figure 2 - WiMAX network architecture**

The WiMAX network architecture comprises different entities. The Network Access Provider, a business entity providing WiMAX radio resources to one or more WiMAX Network Service Providers and controls the Access Service Network (ASN). The Network Service Provider, which is a business entity that provides IP connectivity and WiMAX services to the WiMAX subscribers and manages the Connectivity Service Network (CSN). The Access Service Network includes network elements such as the BS and the ASN Gateway (ASN-GW), providing network access to the Mobile Stations. The ASN contains the network functions needed to provide radio access to a WiMAX subscriber.

The communication between the different elements of the network architecture is based on reference points, which set the foundation for seamless interoperability (see Figure 2). For instance, reference point R1 describes the protocols and procedures between MS and ASN, these MAC and physical specifications are detailed in the IEEE 802.16-2004, 802.16e and 802.16g standards. Since the ASN concentrates on the network access functionality, different implementation profiles for 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 control and radio resources. Besides the IP connectivity assured by the CSN, the IP address allocation, Internet access, billing operations and IP Multimedia Services (IMS) are also managed in the CSN. This last profile is preferred by the WiMAX Forum.

The mobility management defined by the WiMAX Forum for Mobile WiMAX supports IPv4 and IPv6 mobility management protocols as well as, the reduction of packet loss and handover latency. Two types of mobility are considered in the architecture: ASN-anchored mobility and the CSN-anchored mobility. ASN-anchored mobility, or micro-mobility, is devoted to the mobility procedures that occur without a Care-of-Address update of the MS, since the MS moves its point of attachment between BSs within the same ASN network. CSN-anchored mobility or macro-mobility considers the IP mobility between ASN and CSN. Different types of Mobile IP implementations are considered to support macro-mobility. The first one is based on MIP-aware clients and the other one is based on clients that do not support MIP functionalities, therefore needing some kind of assistance from the network to perform handover. This last approach is based on the Proxy Mobile IP (PMIP) implementation. With the MIP-aware approach, the MS is compliant with MIPv4 [10] if deployed in IPv4 networks, or MIPv6 [11] 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 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 envisions the static provisioning. Also, the 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 networks. 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 MS location.

# 3. EVALUATION

This section, on one hand describes the multimedia application characteristics. On the other hand presents the experimentation of voice and video applications, a subset of multimedia applications, in a WiMAX test-bed.

## 3.1 Multimedia Applications

VoIP applications require assured bandwidth and specific bounds on delay and jitter that depend on the configured codec, which codes human voice into samples that can be transported in IP packets. Different codecs are specified in the ITU-T [12] recommendations, such as G.726 [13], G.722 [14] and G.711 [15]. Although most VoIP applications support G.711 [16], G.711 does not achieve the best performance in terms of packet loss and bandwidth conservation, since it does not perform data compression. Despite this, G.711 provides features for bandwidth conservation such as voice activity detection, which avoids sending full packets in periods of silence.

Video applications have different QoS requirements, which are determined by the data representation format (e.g. MPEG-4), resolution, frame rate, compression rate, colour spaces and stream type. Streaming applications can be more tolerant to the delay and jitter effects than voice applications, since buffer mechanisms allow to absorb the delay variation issues. 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 pixels per line and 288 lines per pixel and approximately 30 frames per second. The YUV model is the preferred video colour space since it models the human perception of colour more closely than other colour spaces like RGB, widely used in computer graphics hardware. The YUV model defines the colour space in terms of one lumma component (brightness) and two chrominance (colour) components.

The user perceived video quality can be measured by calculating the Peak Signal Noise to Ratio (PSNR). The PSNR is determined by comparing each pixel in the original frame with the distorted frame, thus allowing the evaluation of the distortion introduced by the propagation in the network. Table 1 presents the relation between PSNR and the Mean Opinion Score (MOS) evaluation. MOS is defined in a 5-point scale, quantifying the user-perceived video quality.

**Table 1. PSNR and Mean Opinion Score for Video**

| PSNR (dB) | MOS | Description |
|---|---|---|
| > 37 | 5 | Perceptible (Excellent) |
| 31-37 | 4 | Just 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) |

## 3.2 Tests Description

This subsection describes the test conditions and the tools used to evaluate the performance of voice and video applications in WiMAX links.

The common measurements considered for video and voice evaluations 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 requirements.
- **Over-provisioned**. The bandwidth reserved is in excess of the 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) [17]. The voice codec used was the G.711 with no voice activity detection and with one sample per packet. The VoIP traffic was generated from the SS towards the ASN with a payload of 80bytes which include 10ms of conversation. The voice tests were performed in two variants:

- **One Client**. A single flow was created to determine the QoS differentiation of the different scheduling services.
- **Multiple Clients**. Different voice sessions, each one representing a client, was created to determine the support of simultaneous users in aggregated service flows.

Table 2 summarizes the different tests for the single flow cases. Each test was identified by the reserved bandwidth (*B*) (160Kb/s or 80Kb/s), by the maximum allowed delay (*d*) (when applicable) and by the scheduling service (*s*) configured. For instance, 160kb_2_rtPS test had 160Kbytes of bandwidth, a configured delay of 2ms and used the rtPS scheduling service class.

**Table 2. Voice applications: Test with one flow**

| Test Case | Bandwidth - *B* (kb/s) | Delay - *d* (ms) | Scheduler –*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 | N/A | BE |

In the single flow cases, the tests with 80Kb/s represented the under-provisioned cases, since the minimum required bandwidth, for the generated traffic was 100 Kb/s, whilst the 160Kb/s test cases represented the over-provisioned situations.

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 the tests, a service flow was pre-configured with 1Mb/s of bandwidth. The test with 25, 50 and 75 simultaneous clients represented the over-provisioned test cases, since only 0,30, 0,60 and 0,90Mb/s were required, respectively. The test cases with 100 and 180 simultaneous clients represented the under-provisioned cases with 1,20 and 2,15Mb/s of required bandwidth.

The different values for the delay parameter were based on the ITU G.114 recommendation [19], which specifies 150ms for one way delay between the sender and the receiver of voice applications and defines a maximum bound of 400ms for an acceptable one way delay.

### *3.2.2 Video Tests*

Evalvid is a framework that provides a set of tools to convert raw video files into a MPEG format in order to be transmitted over the network [18]. Evalvid is a complete framework that evaluates network performance for video transmission, not only based on the common network parameters, such as delay, jitter and packet loss but also on more objective measures like PSNR and the MOS scale, allowing with this to measure the user perceived video quality. Evalvid determines the PSNR of the transmitted videos by comparing the original video files with the transmitted videos. And from the PSNR value it is possible 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. The video traffic was based on the Foreman video file, which was prepared with the Evalvid tools to be transmitted in the WiMAX links. The different video tests performed are depicted in Table 3.

**Table 3. Video tests for each video file**

| Scheduler - *s* | Delay - *d* | Bandwidth - *B* (Mb/s) |
|---|---|---|
| BE | N/A | 2 |
| rtPS | 2 | |
| | 100 | |
| | 150 | |
| BE | N/A | 1 |
| rtPS | 2 | |
| | 100 | |
| | 150 | |

The minimum reserved bandwidth in all the rtPS test cases was 500Kb/s. The video file had a bit rate of 1Mb/s (as was measured by the Evalvid tools), therefore 2Mb/s of bandwidth represented the over-provisioned cases while 1Mb/s of bandwidth represented the under-provisioned cases.

The video evaluation process using the Evalvid framework consisted on different steps, which went from the conversion of the original raw video file format to the MPEG video file format, passing by the creation of the reference video files until the comparison of the transmitted and reference video files to determine the PSNR and, consequently, the respective MOS.

## 3.3 Test-bed

This subsection describes the test-bed used for the evaluation of voice and video applications.

Figure 3 depicts the layout of the test-bed, which was based on an Ethernet loopback, deployed in a GNU/Linux 2.6.22 Kernel with the self-to-self patch [20]. The kernel was patched, since the standard kernel does not allow the functionality of a loopback between two network interface cards, on the same machine. This functionality was required to avoid synchronization issues between different machines in order to measure one-way delay.

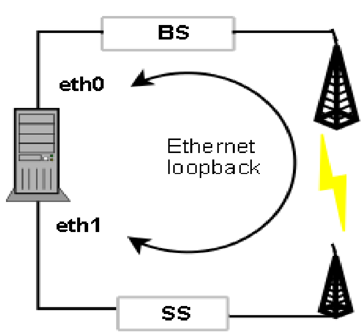


**Figure 3 – Test-bed for voice and video evaluation**

The WiMAX equipment used in the test-bed was based on the Redline RedMAX AN-100U Base Station and on the Redline RedMAX Subscriber Unit outdoors Station and was configured according to the parameters summarized in Table 4.

**Table 4. Test-bed configured parameters**

| Description | Value |
|---|---|
| RF Downlink Channel | 3488000 KHz |
| Tx Output Power | 0 |
| Channel Size | 7 MHz |
| Cyclic Prefix | 1/16 |
| DL Ratio | 56 % |
| Cell Range | 5 Km |

## 3.4 Results

This subsection presents the results of the experimentation of voice and video applications in the WiMAX test-bed.

### *3.4.1 Voice Results*

The evaluation of voice traffic was performed using a single client and multiple clients, resorting to delay and jitter measurements.

In the case of the single client with a configured bandwidth of 80Kb/s, the one way was high, since bandwidth was under-provisioned for the flow. When the bandwidth was configured with 160Kb/s, the delay of the different classes was similar, with an average value of 10ms, as depicted in Figure 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 300ms had the best performance. Such fact was due to a non stringent value of delay (2 or 100ms).

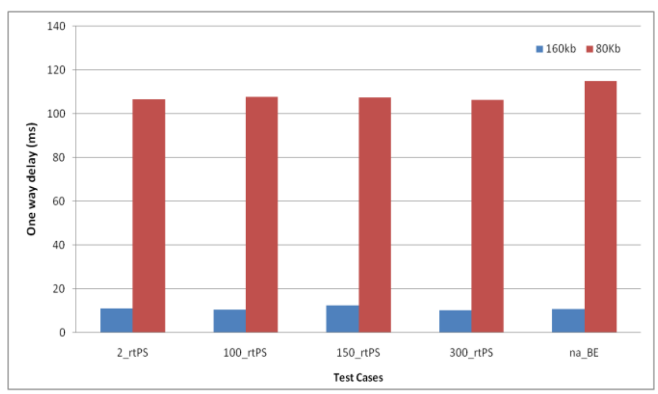


**Figure 4 - One way delay in the case of a single voice client.**

Jitter, in the single voice client test, had an average value of 1,20ms in the over-provisioned test cases (160Kb/s). In the under-provisioned test cases jitter had higher values, lying around 14ms.

The packet loss with the single voice client tests was only visible in the under-provisioned test cases with an average packet loss of 28%, as Figure 5 depicts.

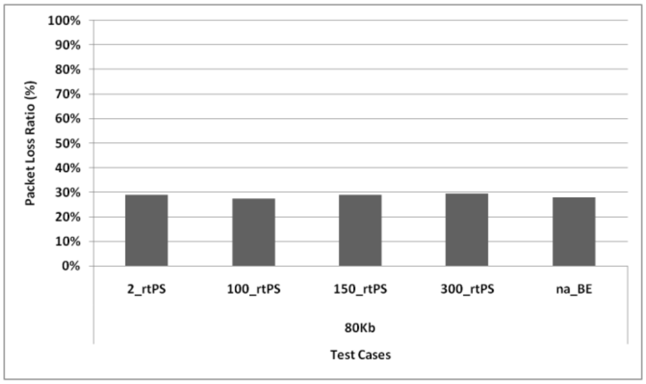


**Figure 5 - Packet loss in the case of a single voice client.**

In these cases, the BE test case had not a high packet loss as some test cases of rtPS, such as the 2ms test case. This behaviour was due to their rigorous criteria of the maximum allowed delay in the rtPS. In the test cases configured with the BE scheduling service such criteria did not exist.

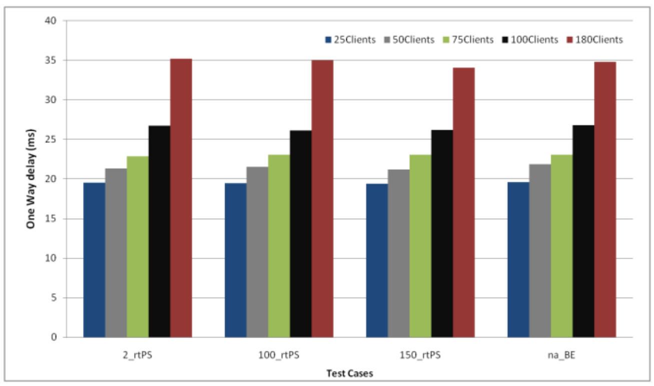


**Figure 6 - One way delay in the case of multiple voice clients.**

Figure 6 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 19ms with the 25 voice clients and an average of 35ms with the 180 voice clients. The rtPS scheduling service had a better performance in terms of delay, when compared to the BE scheduling service. The 150ms test case had an improved performance since it had not stringent criteria, in terms of delay to respect (2 or 100ms).

Jitter, in the multiple voice client tests, increased with the number of simultaneous voice clients. For instance, jitter had a minimum value around 1ms in the over-provisioned cases and a maximum value of 7ms with the 180 voice client tests.

The packet loss, in the multiple voice clients, increased with the number of simultaneous voice clients. Although in the over-provisioned test cases the 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 150ms 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 define the packet loss bellow 0,1% for a best performance. The under-provisioned test cases had acceptable delay bounds but exhibit high packet loss, with values around 28%.

The multiple voice client tests supported 75 simultaneous clients with a good conversation quality in terms of delay and packet loss metrics. For instance, with 180 simultaneous clients in a 1Mb/s aggregated service flow there was 46% of packet loss.

### *3.4.2 Video Results*

The video results are presented in terms of the MOS classification, packet loss and delay which is determined on a Probabilistic Distribution Function (PDF).

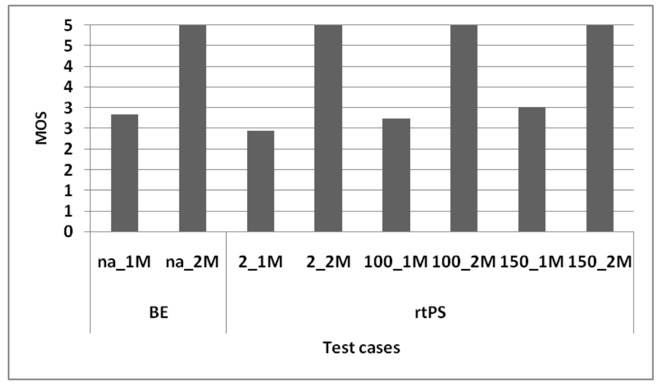


**Figure 7 - MOS of the Foreman video**

Figure 7 depicts the MOS classification of the Foreman video, assuming one video client. All the over-provisioned test cases had the maximum classification in the 5-point scale of MOS. 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 100ms had a lower classification when compared to the test cases configured with the BE scheduling service. Such fact was due to the high packet loss in the 2 and 100ms test cases caused by the rigorous admission criteria (low delay bounds). The packet loss with these low delay bounds occurred as soon as the buffers dedicated to this service flow were full.

With the over-provisioned test cases there was no packet loss, while in the under-provisioned test cases packet loss depended on the scheduling service. The test cases configured with the BE scheduling service had a lower packet loss when compared to the test cases of rtPS configured with a delay of 2 and 100ms.

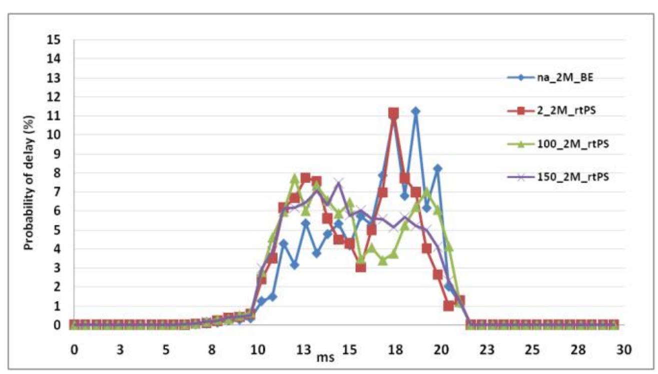


**Figure 8 - PDF delay of Foreman video (2Mbytes)**

Figure 8 depicts the PDF delay for the Foreman video in the over-provisioned 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 20ms than the rtPS test cases. In the rtPS test cases, the 150ms configured delay had the lowest probability of high delay when compared to the other test cases (2 and 100ms). In the under-provisioned test cases (not pictured) the delay had probability of values around 30ms.

Video files transmitted in service flows with different QoS configuration had different behaviours according to the bandwidth configured for the service flows. With the over-provisioned test cases (2Mb/s), video was received with an excellent quality. Nevertheless, in the under-provisioned cases, video quality was lower and presented annoying features, as depicted in Figure 9.


**Figure 9 - Foreman video with 1Mbytes**

## 4. CONCLUSIONS

WiMAX capability to support multimedia applications, namely voice and video applications, were evaluated by experimentation on a test-bed. Two test conditions were created, one where resources are over-provisioned and another with under-provisioning. The result showed that both applications behave according to the recommendations of ITU G.114 and ITU Y.1541 in the over-provisioned case. However, in the under-provision conditions, both voice and video applications do not strictly 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 result have shown that it is possible to support up to 75 simultaneous clients in a service flow configured with 1Mb/s, being an indication for the scalability potential of WiMAX.

## 5. ACKNOWLEDGMENTS

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