ABSTRACT
In the last recent years wireless broadband internet has become more and more popular. Especially WiMAX, which offers wireless broadband internet in urban areas and provides a good quality of service (QoS) for real-time applications due to the available QoS mechanisms at the Medium Access Control (MAC) layer. But WiMAX does not specify, e.g., standard scheduling mechanisms used to allocate available resources among users. This paper discusses requirements that are imposed by real time applications on the WiMAX technology and it describes different solutions that can be used by the WiMAX technology in order to satisfy these requirements.

Keywords
WiMax, QoS, real time applications, wireless, broadband internet.

1. INTRODUCTION
In these days in The Netherlands, wireless internet becomes more and more popular. Many citizens of the Netherlands use IEEE 802.11 (such as 802.11b, 802.11g and 802.11n) access points in their home for wireless internet through wired broadband internet. However, at the moment, internet service providers are testing a new technology for wireless broadband internet in urban areas, named WiMAX [1] (802.16). WiMAX has a higher broadband capacity than other wireless internet connections and a wider network range than 802.11 access points. In the future, with WiMAX, internet providers will have the opportunity to provide wireless broadband internet for customers and organizations [2] [3].

Due to the increase of the broadband internet usage, there are many services available such as Voice over IP (VOIP), video on demand (VoD) and gaming [4]. A good way to handle these services through an internet connection is by using Quality of Service (QoS). QoS is a feature that has the ability to, among others, provide different priorities to different services through an internet connection. However wireless connections are very vulnerable, because the signal can be blocked by all kinds of objects between the sender and the recipient. This might decrease the available bandwidth and could cause packet loss [5]. This makes it hard to implement a good working QoS in wireless networks.

WiMAX is like a master-slave oriented technique, because the base station has full control over the transmission. Due to this it can guarantee QoS much better than 802.11 networks. The main research question that is answered by this paper is:

How real time applications are supported by WiMAX technologies?

In order to explain how real time applications can be supported by WiMAX technologies, the following questions are answered:

1. Which WiMAX IEEE standards are currently available and which are upcoming?
2. Which network scenarios are supported by the WiMAX technologies?
3. What are the QoS requirements imposed by real time applications on WiMAX technologies?
4. Which QoS solutions are supported by the WiMAX technologies?
5. Can the QoS solutions supported by the WiMAX technologies satisfy the QoS requirements?

Since much research has already been done on WiMAX, QoS and QoS in WiMAX, it was sufficient to do this research by means of a literature study and making qualitative comparisons. The first step of this research was to look at the available literature about WiMAX, QoS and QoS in WiMAX. The next step was to answer the sub questions 1, 2, 3 and 4 to define the QoS requirements and the supported solutions by WiMAX technologies. After these four questions were answered, sub question five was answered by the defined QoS requirements, the supported solutions and the test results from the available literature. After this, conclusions were derived, and based on these conclusions the main research question was answered.

This paper is organized as follows: an overview of the WiMAX IEEE standards with possible network scenarios is described in section 2. The QoS requirements of real time applications in WiMAX networks are given in section 3. Section 4 describes possible QoS solutions for real time applications in WiMAX networks. Section 5 describes the test results and gives a comparison of the solutions. Finally, conclusions and future work activities are given in section 6.
2. WIMAX STANDARDS AND NETWORK SCENARIOS

This section presents the WiMAX standards and the network scenarios that are supported by this technology.

2.1 WiMAX Standards

In 1999 the IEEE Standards Board established a new working group called IEEE 802.16 [6]. The purpose of this working group was to develop and publish air interface standards for wireless metropolitan area networks. In 2003 the WiMAX Forum was established in order to promote and enable the deployment of a new broadband access technology based on the 802.16 standards, called WiMAX [7]. The first WiMAX system was based on IEEE 802.16-2004 standard [6]. This standard offers fixed broadband wireless communications using rooftop mounted Customer Premises Equipment. In December, 2005 the IEEE completed the 802.16e-2005 amendment, which added new features to support mobile applications. This resulting standard is now commonly known by the WiMAX Forum as mobile WiMAX, release 1.0 [8]. Based on the mandatory features and some of the optional features needed for enhanced mobility and QoS support of the 802.16e-2005 standard, the WiMAX Forum released the first Mobile WiMAX, release 1.0, in late 2006. This release contains all of the basic features needed to enable WiMAX deployments. [7] describes the following features:

- Access Service Network (ASN) and Connectivity Service Network (CSN) mobility. Which enables WiMAX to use different providers, namely network access providers (NAP) that takes care of radio access infrastructure and network service providers (NSP) that provides IP connectivity through the WiMAX network [7].
- Paging and location management. This is a feature supported by the ASN gateway. The ASN gateway is a complete set of network functions that provides radio access to a subscriber [9].
- IPv4 and IPv6 connectivity. This functionality provides network connectivity.
- Preprovisioned/static QoS. This feature is used to pre-provision QoS capabilities.
- Optional radio resource management (RRM). This is an optional feature in order to use limited radio resources and radio network infrastructure as efficiently as possible.
- Network discovery/selection. This function is required for nomadic, portable and mobile services in the same geographical area where a subscriber may have radio coverage access to an ASN managed network by a single NAP and shared by several NSPs, see [10].
- IP/Ethernet Convergence Sublayer support. Is a functionality that is used for the transport of all packet-based protocols such as Internet Protocol (IP), see [11].
- Flexible credentials, pre- and postpaid accounting. This is a functionality that provides a communication access during a time period.
- Roaming (RADIUS only). Roaming provides the ability to access wireless services by using the network of an operator that is not the standard network operator.
- 3GPP I-WLAN compatible interworking. This is a functionality that controls the services of the subscribers when switching to another network, like WiFi.

In 2007 the WiMAX forum started with the development of network release 1.5. The development of release 1.5 aimed primarily at enabling QoS and provisioning of open remote device and support for advanced network services. This release combines the IEEE 802.16-2004 base standard and IEEE 802.16e, 802.16f, 802.16g amendments [12] [13] [14]. [7] describes that it enables the following key features:

- Over-the-air (OTA) activation and provisioning. This functionality is used to activate and program subscriber stations over the air.
- Location-based services (LBS). LBS make it possible to get the geographical positions of mobile devices.
- Multicast broadcast service (MBS). This functionality enables the support of multimedia streaming service in the MAC layer of WiMAX.
- Internet protocol Multimedia Subsystem (IMS) integration. IMS is a framework for delivering Internet Protocol multimedia services.
- Dynamic QoS and policy and charging (PCC) compatible with 3GPP Release 7. This is a mechanism that can automatically configure QoS settings based upon requests from subscribers [15].
- Telephony VoIP with emergency call services and lawful interception.
- Full NAP sharing support. NAP sharing support provides NSPs over multiple NAPs.
- Handover data integrity. This mechanism minimizes the packet loss during a handover.
- Multihost support, allows subscribers to use multiple base stations.
- Additional Ethernet services: VLAN (Virtual Local Area Network), DSL (Digital Subscriber Line) and IWK (Interworking between WiMAX and Code Division Multiple Access (CDMA) 2000).
- Enhanced open Internet services. An improvement of open internet services for users.
- Diameter-based Authentication, Authorization, and Accounting (AAA). Is the functionality that provide centralized AAA management for subscribers to connect to the network.

At the moment the WiMAX Forum is developing a major release 2.0. This release is based on the IEEE 802.16m standard, see [16], and will improve spectrum efficiency, latency and scalability to achieve wider bandwidths in challenging environments. The WiMAX forum expects to
complete release 2.0 in late 2010 / early 2011. According to [7], release 2.0 will support the following features:

- Multimedia session continuity. This provides the ability to maintain continuity of multimedia sessions during handovers.
- 3GPP/2 interworking (optimized handover). This is a handover improvement in order to switch to 3GPP/2 networks and vice versa.
- Network management, including self-organized/optimized networks (SONs). This functionality provides the ability to (automatic) increase the overall network performance, quality and reliability.
- Seamless WiFi-WiMAX handover. WiFi-WiMAX handover is a mechanism used to support a seamless handover from WiFi to WiMAX and vice versa.
- Roaming enhancements. This is an improvement of roaming, within the mobile WiMAX release 1.0.
- Support for multihop relay stations. This is a mechanism that enables the use of multihop relay stations, which extend or enhance the coverage of a base station.
- Support for femto-cells. This is an ability to use cell phone connectivity around house by using WiMAX.
- Device reported metrics. These are mechanisms that monitor devices.

2.2 WiMAX network scenarios
The IEEE 802.16 supports two types of network deployments scenarios, namely fixed and mobile access. The fixed scenario, is also called nomadic because of its semi-mobile characteristics, is more like a replacement for existing wired broadband internet networks. It offers high-quality broadband wireless services and high-speed data rates. A benefit of this in comparison with wired broadband internet services is that it is nomadic. It can, for instance, be used for mobile offices spreading through a certain area. This scenario consists of a subnet of multiple subscriber stations and multiple interfaces of multiple base stations.

In mobile WiMAX, the mobile access deployment scenario is used. This scenario provides a high-speed data throughput like in wired networks but it also has the same mobility functions as cellular systems. Instead of 802.11 standard, the base stations of 802.16 provide both mobility functions and fixed communication features [17].

In both scenarios it is possible to use point-to-point, point-to-multipoint and mesh architectures, see Figure 1. A point-to-point architecture consists of a permanent link between two endpoints. This is often being used in WiMAX for backhaul connections.

A point-to-multipoint architecture consists of multiple subscribers and one base station. The base station acts as a coordinator and relays all connections. If a subscriber wants to send data to his neighbor, it first has to communicate with the base station. In mobile WiMAX, the point-to-multipoint architecture has also the possibility to use mobility support. Due to this, subscribers can experience a seamless handover between different base stations.

In mesh mode it is not clear what the base station or the subscriber is. Every subscriber can directly communicate with his neighbor without any interaction with a base station. In mesh mode several subscribers can act as a base station that is directly connected to a backhaul like an Internet connection or a public network [18].

3. QoS REQUIREMENTS ON WIMAX
As mentioned before, in the introduction, QoS is useful to handle real-time applications such as VOIP, VOD and gaming. Such applications need reliable internet connections. Due to the fact that wireless connections are very vulnerable it is important to know what the minimum QoS requirements are. Therefore, applications are classified in categories, so different priorities can be allocated to different applications. For example web browsing has a lower priority than VOIP. These priorities are based on the requirements of the applications. The requirements of the applications are based on the following criteria; latency, packet loss, packet delay variation and minimum data rate. In Table 1 an overview is given of the classification of the applications and of their requirements. To meet these applications requirements, the QoS solutions should able to meet the following four requirements, see Table 1:

- Requirement 1: Minimum bandwidth data-rate
- Requirement 2: Maximum jitter tolerance
- Requirement 3: Maximum latency tolerance
- Requirement 4: Maximum packet loss

WiMAX should support mechanisms to provision the needed QoS. There are several features used for QoS provisioning. In this paper we will focus on two main QoS provisioning functions, i.e., packet queuing and packet scheduling.

- Requirement 5: Packet scheduling
  
  A mechanism that is used to control the manner in which queued packets are selected, i.e., scheduled, for transmission, see [21].

- Requirement 6: Packet Queuing
  
  Packets arriving into a node have to be buffered and queued. The QoS provisioning function that provides this feature is denoted as packet queuing, see [21].

In case of an architecture where multiple relaying of signals is allowed, like in a mesh scenario, there should be a mechanism to coordinate the communication. To offer a good QoS in this scenario, a centralized coordinator could be appointed, like some sort of base station, to guarantee the required and thus given QoS level. Each client should also have an access point kind of authority where the access only is granted when the base station authorizes such access. The base station should have an inventory of the capacity of all nodes in the network and based on that capacity, a way of calculating the most effective link via hops to the end-user is available [22]. This leads to the following QoS requirement:

- Requirement 7: Support of end-to-end QoS.

Packets that arrive at the base station from the network are placed in the traffic packet queues. From that, the scheduler decides which traffic to map into a frame from the queues and generate an appropriate MAP information element. These MAP elements contain information on transmission for each frame [23], so packets are scheduled according to their service classes. In this way, scheduling on a frame-by-frame basis, gives a lot of flexibility. Due to this, a scheduler can execute at a frequency of 400 frames per second. In order to guarantee this frequency no complex QoS solutions are desired [24]. Which leads to the following QoS requirement:

- Requirement 8: No complex QoS solutions.

4. QoS SOLUTIONS

The IEEE 802.16 standards provides fast centralized scheduling algorithms that precisely control the uplink and downlink at the base station which helps to increase system efficiency [25]. These scheduling algorithms schedule user packets into predefined service classes. The service classes of the IEEE 802.16 are built into the MAC layer and consist in mobile WiMAX of five different service classes and in fixed WiMAX of four service classes. All the service classes in mobile WiMAX are the same as in fixed WiMAX, but mobile WiMAX has an additional class called extended real-time Polling Service. All the service classes are defined below and summarized in Table 2:

Unsolicited Grant Service (UGS).

This mechanism provides a fixed periodic bandwidth allocation [20]. It is designed to support real-time data streams at periodic intervals with fixed-size data packets like T1 connections do. Examples of such applications that requires such a connection are Multiplayer Interactive Gaming, VOIP without silence suppression and Video conference. Parameters used in this service class mechanism are [26]:

- maximum sustained traffic rate
- maximum latency
- tolerated jitter.

Extended real-time Polling Service (ertPS).

ErtPS is designed to support real-time data streams at periodic intervals that generate variable-size data packets like VOIP with silence suppression. It provides the same feature as UGS, but it also takes care of silent periods, so no traffic is sent during such periods. For that the Base station poll needs to have the Mobile Subscriber during the silent period to determine of the silent period has ended [20]. Parameters used in this service class mechanism are [12]:

- maximum latency
- minimum reserved traffic rate
- maximum sustained traffic rate
- traffic priority.

<table>
<thead>
<tr>
<th>Requirement</th>
<th>Amount</th>
<th>Data</th>
<th>Media Content Downloads</th>
</tr>
</thead>
<tbody>
<tr>
<td>Requirement 2: Maximum jitter tolerance</td>
<td>Low</td>
<td>Low to high</td>
<td>High</td>
</tr>
<tr>
<td>Requirement 3: Maximum latency tolerance</td>
<td>0.01Mbp</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>Requirement 4: Maximum packet loss</td>
<td>&gt; 1Mbps</td>
<td>-</td>
<td>-</td>
</tr>
</tbody>
</table>

Table 1: Overview of QoS requirements, from [19] [20].
Real-time Polling Service (rtPS).

This mechanism is designed to support real-time streams consisting of variable-sized data packets of data streams that are issued on periodic intervals. Due to the fact that the requirement of bandwidth vary in this service class, the base stations polls regularly to determine if the silent period has ended [20]. Examples of applications that require such a connection are Multiplayer interactive gaming and streaming media. Parameters used in this service class mechanism are [26]:
- minimum reserved traffic rate
- maximum sustained traffic rate
- maximum latency
- traffic priority.

Non-real-time Polling Service (nrtPS).

This mechanism guarantee a minimum data throughput. It is designed to support applications that require delay-tolerant data streams consisting of variable packets for which a minimum data throughput is required. Application that use nrtPS are FTP, Web Browsing and Instant Messaging. Parameters used in this service class mechanism are [26]:
- minimum reserved traffic rate
- maximum sustained traffic rate
- traffic priority.

Best Effort (BE).

This mechanism provides data traffic that requires no guarantees. Most of the data traffic fall into this service class. Bandwidth for traffic of this class will only be granted if there is remaining bandwidth space available. Parameters used in this service class mechanism are [26]:
- maximum sustained traffic rate
- traffic priority.

Although the standard specifies a centralized scheduling scheme with five or four service classes, it does not specify algorithms for routing and scheduling algorithms. The use of such algorithms have a significant impact on the performance and the provided QoS for the different users, see [22]. Many studies have been done recently in developing such algorithms and improving the scheduling scheme. The following sub sections describe some recent proposed solutions.

### Table 2: Service classes in WiMAX.

<table>
<thead>
<tr>
<th>Service designations</th>
<th>Flow Parameters</th>
<th>Defining QoS Parameters</th>
<th>Application Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unsolicited grant services (UGS)</td>
<td>- minimum reserved traffic rate</td>
<td>maximum sustained traffic rate</td>
<td>VOIP without silence suppression</td>
</tr>
<tr>
<td></td>
<td>- maximum latency</td>
<td>tolerated jitter</td>
<td></td>
</tr>
<tr>
<td>Extended real-time polling service (ertPS)</td>
<td>- minimum reserved traffic rate</td>
<td>maximum sustained traffic rate</td>
<td>VOIP with silence suppression</td>
</tr>
<tr>
<td></td>
<td>- maximum latency</td>
<td>traffic priority</td>
<td></td>
</tr>
<tr>
<td>Real-time polling service (rtPS)</td>
<td>- minimum reserved traffic rate</td>
<td>maximum sustained traffic rate</td>
<td>Streaming audio and video, Motion Picture Experts Group (MPEG) encoded</td>
</tr>
<tr>
<td></td>
<td>- maximum latency</td>
<td>traffic priority</td>
<td></td>
</tr>
<tr>
<td>Non-real-time polling service (nrtPS)</td>
<td>- minimum reserved traffic rate</td>
<td>maximum sustained traffic rate</td>
<td>File Transfer Protocol (FTP)</td>
</tr>
<tr>
<td></td>
<td>traffic priority</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Best-effort service (BE)</td>
<td>maximum sustained traffic rate</td>
<td>traffic priority</td>
<td>Web browsing, data transfer</td>
</tr>
</tbody>
</table>

### 4.1 Call Admission Control in Mobile WiMAX networks

Most of the current researchers focus on packet scheduling. Instead of packet scheduling, [2] proposes a Call Admission Control scheme that extends the solution of [27] for mobile WiMAX networks to complements the capabilities of QoS. The QoS architecture and the role of the base station and subscriber of this solution is presented in Figure 2. In this figure the base station provides connectivity, management and control to the subscriber. And on the other hand the subscriber provides connectivity between subscriber and base station. Therefore, both base station and subscriber are involved in the QoS control of WiMAX.

![Figure 2: QoS architecture with proposed solution [2].](image)

The proposed Call Admission Control blocks unwanted calls and reduces the buffer needed for packet scheduling, in order to guarantee the QoS. The scheme is based on the service classes
(BE, nrtPS, rtPS, ertPS and UGS) in order to guarantee the forced termination probability of handovers. It builds a complete sharing approach in which the channels in each subscriber are shared amongst the different service classes. The scheme uses different traffic types and multiple thresholds in order to fulfill the specific QoS requirements. The scheme is dedicated to a voice, data and multimedia integrated system in which four thresholds are used in order to provide different traffic types with different QoS classes. The main advantage is that it enables the overall bandwidth to be completely shared, thus leading to efficient usage of the scarce wireless resources [2].

4.2 Bandwidth allocation to subscribers.
In [24] it has been emphasized that prior work focused on one or two service classes that could be realized by uplink schedulers. The uplink scheduler that [24] proposes supports all the service classes. It uses three queues of a different priority level, called: low, intermediate and high. The low priority queue stores BE bandwidth request. The intermediate queue stores bandwidth requests sent by rtPS and nrtPS connections. These requests can be migrated to the high priority queue to guarantee that their QoS requirements are met. The high priority queue stores bandwidth requests sent by ertPS and UGS connections. The base station executes the uplink scheduler at every frame and it broadcasts the scheduling agenda to the subscribers.

In order to guarantee UGS and ertPS the scheduler in [24] generates periodic grants and inserts them into the high priority queue at predefined intervals. These intervals are specified by the subscriber at the connection times, and the base station defines the interval between the bandwidth request opportunities. To guarantee the delay bound requirement used for rtPS connections, the base station assigns a deadline for each bandwidth request in the intermediate priority queue. In order to guarantee the minimum bandwidth requirement of rtPS and nrtPS traffic the scheduler uses a window duration. Each time the scheduler is executed it calculates a priority value for each request. Requests of connections that already received the minimum requirement are assigned to low priority values. And other requests are assigned based on the rule: the lower the bandwidth received by the connection, the higher the priority of its requests.

4.3 QoS scheduling based on grants per subscriber
[28] proposes a complete QoS framework based on Grants per Subscriber Station. This framework contains a QoS model that has to send an access request to the base station before the subscriber establishes a connection. After that the base station decides whether the request is confirmed or not, this is based on the available bandwidth and the QoS requirements. If the request is permitted, the base station will submit the details of the QoS to the subscriber, see Figure 3. Also presented in Figure 3, the framework of [28] consists of two scheduling algorithms. One where the base station adapts a simple fair Max-Min scheduling method [29]. This method guarantees fairness between subscribers and simplifies the scheduling module on the base station by transferring some parts of the scheduling functions to the subscribers. The other scheduler is a subscriber upstream scheduler. This scheduler adopts priority scheduling structure to satisfy the QoS requirements of traffic with high priority. Besides the scheduling algorithms, [28] also uses an admission control mechanism to control the data with the highest priority traffic and ensures that traffic with low priority won't be starved.

4.4 QoS in WiMAX mesh networks
In WiMAX mesh networks subscribers can communicate with each other without a base station. Due to this fact the base station can't control the QoS of the subscribers. [22] present algorithms for routing and scheduling in order to meet the QoS requirements based on the service classes. The first presented algorithm provides routing in scheduling for the aggregate traffic for each source-destination pair of subscribers. This algorithm determines the fraction of the average traffic requirement of the link between the source and destination. Based on this average throughput requirement, the algorithm will decide which connections will pass through which route (constant bit rate, variable bit rate or other). The second scheduling algorithm is designed to guarantee constant bit rate (UGS and ertPS) traffic. It ensures that the total (constant) amount of traffic generated during a frame will hold. The last scheduling algorithm is designed to guarantee variable bit rate (nrtPS and rtPS) traffic. This algorithm ensures that the variable bit rate will follow the same route, and calculates delay in order the meet the delay requirement. Other traffic doesn't have QoS requirements and will only be let through if there is still some available bandwidth left [22].

4.5 QoS scheduling for group mobility in WiMAX
[5] focuses on QoS in a group mobility scenario, with attention to vehicles, mobile subscribers, that move away from the base station. This movement decreases the signal to noise ratio and reduce the available bandwidth to the mobile subscriber that is moving away. For that reason, [5] provides a MAC scheduling algorithm that will adaptively allocate the available bandwidth between real-time and non-real-time flows. When the available bandwidth decreases the scheduling algorithm increase the available bandwidth in the real-time service flow and decrease the available bandwidth in the non-real-time service flow. But that should be done in such a way that the non-real-time service flow should not starve. This is accomplished by using a polling scheme. The base station polls every mobile subscriber which can send bandwidth requests for real time traffic during the slot allocated to it. The mobile subscriber implements the scheduling algorithm. Each time the base station polls, the scheduling algorithm generates an aggregate bandwidth request equals to the sum of average data rates of each flow within the
mobile subscriber. After that the base station provides bandwidth based on the aggregate bandwidth request of the mobile subscriber.

5. SATISFACTION OF QoS SOLUTIONS

In section 4 several types of QoS solutions were described. In order to give an explanation of how real time applications are supported by WiMAX technologies, it is important to know how these solutions are tested and if these solutions really meet the QoS requirements of real-time applications. This section describes the way how the previous mentioned QoS solutions are tested and which QoS requirements are met. An overview of all the solutions, described in section 4, and on which requirements these solutions can satisfy, are presented in Table 3. In Table 3 the grades, *Bad*, *Fair* and *Good* are used to emphasize if and how each QoS solution satisfies each of the 8 QoS requirements.

The grades *Bad*, *Normal* and *Good* are interpreted by each requirement in the following way:

- **Requirement 1 (Minimum bandwidth data rate):**
  - *Bad*: Several bandwidth data rate requirements given in Table 1 are NOT satisfied.
  - *Normal*: All the bandwidth data rate requirements given in Table 1 are satisfied, but some values are critical.
  - *Good*: All the bandwidth data rate requirements given in Table 1 are easily satisfied.

- **Requirement 2 (Maximum jitter tolerance):**
  - *Bad*: Several jitter requirements given in Table 1 are NOT satisfied.
  - *Normal*: All the jitter requirements given in Table 1 are satisfied, but some values are critical.
  - *Good*: All the jitter requirements given in Table 1 are easily satisfied.

- **Requirement 3 (Maximum latency tolerance):**
  - *Bad*: Several latency requirements given in Table 1 are NOT satisfied.
  - *Normal*: All the latency requirements given in Table 1 are satisfied, but some values are critical.
  - *Good*: All the latency requirements given in Table 1 are easily satisfied.

- **Requirement 4 (Maximum packet loss):**
  - *Bad*: Several packet loss requirements given in Table 1 are NOT satisfied.
  - *Normal*: All the packet loss requirements given in Table 1 are satisfied, but some values are critical.
  - *Good*: All the packet loss requirements given in Table 1 are easily satisfied.

- **Requirement 5 (Packet scheduling):**
  - *Bad*: The solution does not support packet scheduling
  - *Normal*: The solution supports only a limited number of packet scheduling mechanisms.
  - *Good*: The solution supports several types of packet scheduling.

- **Requirement 6 (Packet queuing):**
  - *Bad*: The solution supports only drop tail queuing, i.e., when a packet arrives and the queue is full then this packet is dropped.
  - *Normal*: The solution supports only a limited number of packet queuing mechanisms.
  - *Good*: The solution supports several types of packet queuing.

- **Requirement 7 (Support of end-to-end QoS):**
  - *Bad*: It does not support end-to-end QoS.
  - *Normal*: It supports end-to-end QoS only in combination with other solutions/technologies.
  - *Good*: It supports end-to-end QoS.

- **Requirement 8 (No complex QoS solutions):**
  - *Bad*: Solutions are much more complex than the standardized ones.
  - *Normal*: Solutions are slightly more complex than the standardized solutions
  - *Good*: Solutions are equally complex than the standardized solutions.

5.1 Call Admission Control in Mobile WiMAX networks

The proposed Call Admission Control scheme in [2] was tested in a scenario with one base station and three subscribers. Based on a certain interval the traffic through the network was increased in order to test the proposed Call Admission control scheme. The test results prove that the resource utilization in the subscriber goes up to 100%. It also proves that the dropping and blocking probability of UGS connections are kept below 1% and the blocking probability of rtPS and nrtPS were reduced. Due to this fact the Call Admission Control handle traffic more efficiently, which means that the Call Admission Control scheme makes it possible to accept more connections while it handles UGS traffic.

However the test results that the Call Admission Control scheme handle traffic more efficiently, it only proves that there is more bandwidth available for other traffic flows (rtPS, nrtPS and BE), [2] did not mention anything about delay and jitter which is essential for UGS connections. [2] also did not mentioned anything about support of end-to-end QoS. Due to this it is not sure if this proposed solution really satisfies all QoS requirements of real time applications. This solution satisfies the requirements in the following way, see Table 3:

- **Requirement 1: Good**, the test results prove that there is more bandwidth available.
the increase in offered load, but still guarantee the minimum stays in the required value. The goodput of nrtPS decreases with intervals. However the delay of rtPS connections increases but means that scheme is able to provide data grants at fixed rtPS connections was not affected by the load increase, which service flows. The test results show that the delay of UGS and consists of a base station with the subscribers uniformly distributed around it from 5 to 50 subscribers in steps of 5 units. [24] tested the proposed scheduler in a simulated network that consists of random generated data packets. and one mobile base station. The data sent through the network consisted of a fixed number of subscribers and connections per subscriber. The second scenario consists of a variable number of subscribers, where the workload and the type of the traffic can be varied. Both simulation scenarios were evaluated based on throughput, delay and packet loss.

5.2 Bandwidth allocation to subscribers.
[24] tested the proposed scheduler in a simulated network that consists of a base station with the subscribers uniformly distributed around it from 5 to 50 subscribers in steps of 5 units. [24] consider five types of traffic: voice, voice with silence suppression, video, FTP and WEB which are associated to five service flows. The test results show that the delay of UGS and rtPS connections was not affected by the load increase, which means that scheme is able to provide data grants at fixed intervals. However the delay of rtPS connections increases but stays in the required value. The goodput of nrtPS decreases with the increase in offered load, but still guarantee the minimum bandwidth requirement. But the throughput of BE connections decreased sharply. However [24] did only consider a point-to-multipoint scenario without handovers. Due to the fact that handovers could cause packet-loss it is not certain if the proposed scheduler in [24] is able to meet the end-to-end QoS requirement. This solution satisfies the requirements in the following way, see Table 3:

- Requirement 4: Good, due to the fact that the blocking probability is kept below 1%
- Requirement 5: Good, the solution is able to use several types package scheduling.
- Requirement 6: Bad, the proposed solution did not mention anything about packet queuing. For that it is assumed that the proposed solution only supports drop tail queuing.
- Requirement 7: Not applicable, [24] did not mention anything about end-to-end QoS.
- Requirement 8: Normal, it proposes a additional CAC and it is slightly more complex than the standardized solution.

5.3 QoS scheduling based on grants per subscriber
In [28] a QoS scheduling based on grants per subscriber has been presented. The used solution was tested using simulations. In this simulation two scenarios were tested. The first scenario consists of a fixed number of subscribers and connections per subscriber. The second scenario consists of a variable number of subscribers, where the workload and the type of the traffic can be varied. Both simulation scenarios were evaluated based on throughput, delay and packet loss.

The results of the first scenario are able to meet the QoS requirements 1,2,3 and 4 better than the original framework. In the second scenario the delay of rtPS increases when the network is heavy loaded, but it is still superior than other algorithms and it stays within the QoS delay requirements (< 50 ms). However in comparison with normal scheduler algorithms the throughput of BE traffic is lower since the proposed framework first meets the demand of high priority services. But since there are no requirements for BE connections the second scenario also meets the QoS requirement 1,2,3 and 4 better than the original framework. It improves the throughput and delay of UGS and rtPS traffic due to the simple schedule algorithms. This solution satisfies the requirements in the following way, see Table 3:

- Requirement 1: Good, the test results prove that there is more bandwidth available.
- Requirement 2: Not applicable, [28] did not monitor everything about jitter.
- Requirement 3: Good, due to the fact that the latency was not affected and easily stays within the required range of values.
- Requirement 4: Good, [28] shows that the packet loss is lower than other algorithms and easily meet the packet loss requirement of table 1.
- Requirement 5: Normal, the solution uses limited types of packet scheduling.
- Requirement 6: Bad, [28] did not mentioned anything about queuing, for that it is assumed that the proposed solution only supports drop tail queuing.
- Requirement 7: Not applicable, [28] did not mentioned anything about end-to-end QoS.
- Requirement 8: Normal, [28] propose a new scheduling architecture that consists of two additional schedules. This architecture is slightly more complex than the standardized solution.

5.4 QoS in WiMAX mesh networks
The proposed algorithms in [22] were tested in a network simulation environment that consists of 20 mobile subscribers and one mobile base station. The data sent through the network consisted of random generated data packets.
In [22] the following were observed. No packet loss for constant bit rate (UGS and erTPS) and variable bit rate (nrtPS and rtPS) traffic in the simulations. The average delay of the constant bit rate packets was less than 28.01 milliseconds and of the variable bit rate packets was less than 28.09 milliseconds in a heavily loaded simulation scenario. In the same situation the average throughput of TCP connections stays above the minimum required throughput for different TCP connections. Based on these results, it could be concluded that the proposed algorithms could comply to the delay, packet loss and throughput requirements of voice and video streaming connections. This solution satisfies the requirements in the following way, see Table 3:

- Requirement 1: Good, the test results prove that there is more bandwidth available, also in the more heavily loaded scenarios.
- Requirement 2: Not applicable, [22] did not monitor anything about jitter.
- Requirement 3: Good, the latency stays below 30ms. Based on the above it can be deduced that the latency requirement could be satisfied.
- Requirement 4: Good, during the simulation runs presented in [22], no packet loss was observed. This means that the packet loss requirement could also be satisfied.
- Requirement 5: Normal, the solution uses limited types of packet scheduling.
- Requirement 6: Normal, the solution uses limited queues
- Requirement 7: Normal, the solution supports QoS in combination with other networks.
- Requirement 8: Bad, [22] uses several routing and scheduling algorithms, which means that the proposed solution is much more complex than the standardized solution.

### 5.5 QoS scheduling for group mobility in WiMAX

The WiMAX PMP group mobility scenario of [5] was tested in a simulator called QualNet. Which simulated a scenario with six mobile subscribers and one base station. In order to test the performance of the proposed scheduler, [5] compared the delays incurred by real-time and non-real-time traffic flows in three scenarios. In the first scenario, all the nodes are stationary. In the second scenario the nodes are mobile without the adaptive scheme applied. And in the last scenario the nodes are mobile and the adaptive scheme is employed.

In the results of the evaluated scenarios it seems to be that the first scenario has the lowest real-time and non-real-time delay. In the second scenario, when the nodes are mobile, the real-time and non-real-time traffic delay increases due to the adaptive changes. In the third scenario it is shown that the real-time delay can be reduced at the expense of non-real-time delay, which has increased marginally. This demonstrates the effectiveness of the algorithm for different traffic rates of real-time and non-real-time traffic.

In all the evaluated scenarios, see [5], the delay did not exceed the delay requirement. The maximum delay was 22 milliseconds. But in order to meet the requirements for real-time applications it is also required to observe the jitter, packet loss and end-to-end QoS requirements. This solution satisfies the requirements in the following way, see Table 3:

- Requirement 1: Not applicable, [5] did not mention anything about data rate. Therefore, it is not certain whether this solution satisfies this QoS requirement.
- Requirement 3: Good, the maximum delay of the test results was measured to be 22 milliseconds.
- Requirement 5: Normal, the solution uses limited types of packet scheduling.
- Requirement 6: Normal, due to that the solution uses limited queues (real-time and non-real time).
- Requirement 7: Not applicable, [5] did not mention anything about end-to-end QoS.
- Requirement 8: Normal, [5] proposes an additional scheduling algorithm. This makes the proposed solution slightly more complex.

### 6. CONCLUSION AND FUTURE WORK

This paper focuses on the support of real-time applications in WiMAX networks. Real-time applications require connections with certain quality of services in order to guarantee delay, packet-loss and jitter limitations and minimal bandwidth throughput. In WiMAX there are several systems namely fixed WiMAX, mobile WiMAX 1.0, mobile WiMAX 1.5 and upcoming mobile WiMAX 2.0. These systems are based on multiple IEEE 802.16 standards and supports different networks scenarios, described in section 2.

In order to fulfill the quality of service for real-time applications in the WiMAX systems, there are 8 QoS requirements introduced in section 3, namely; minimum bandwidth data-rate, maximum jitter tolerance, maximum latency tolerance, maximum packet loss, packet scheduling, packet queuing, support of end-to-end QoS, and no complex QoS solutions. These QoS requirements were used to study and compare existing QoS solutions [2] [24] [26] [22] [5] on how they satisfy the derived QoS requirements. As a result of this comparison the following conclusions can be drawn.

Although real-time applications require multiple requirements, most of the solutions only focus on one certain requirement that was tested in a simulation. All the solutions meet the QoS requirement(s) they were tested on. However it is not certain if all these solutions really satisfy all QoS requirements, due the fact that none of the presented solutions were tested on all the requirements. Although [22] meet most of requirements, except the jitter and the complexity requirement. The jitter requirement was not measured and the complexity requirement was not satisfied.

Future research activities in this area could focus on the evaluation of the presented QoS solutions in real life environments, by applying e.g., quantitative analysis and comparison and by using the same simulation environment and workload parameters.

### ACKNOWLEDGMENTS

I would like to thank my supervisor dr. ir. G. Karagiannis for helping me setting up this paper and giving me feedback during the process. Further I would like to thank my peer review group...
for giving me useful comments and helping me to improve the readability of my paper.

REFERENCES
[4] C. Cicconetti, L. Lenzini, E. Mingozzi et al., “Quality of Service Support in IEEE 802.16 Networks - The authors focus on mechanisms that are available in an 802.16 system to QoS and whose effectiveness is evaluated through simulation,” IEEE network., vol. 20, no. 2, pp. 50, 2006.